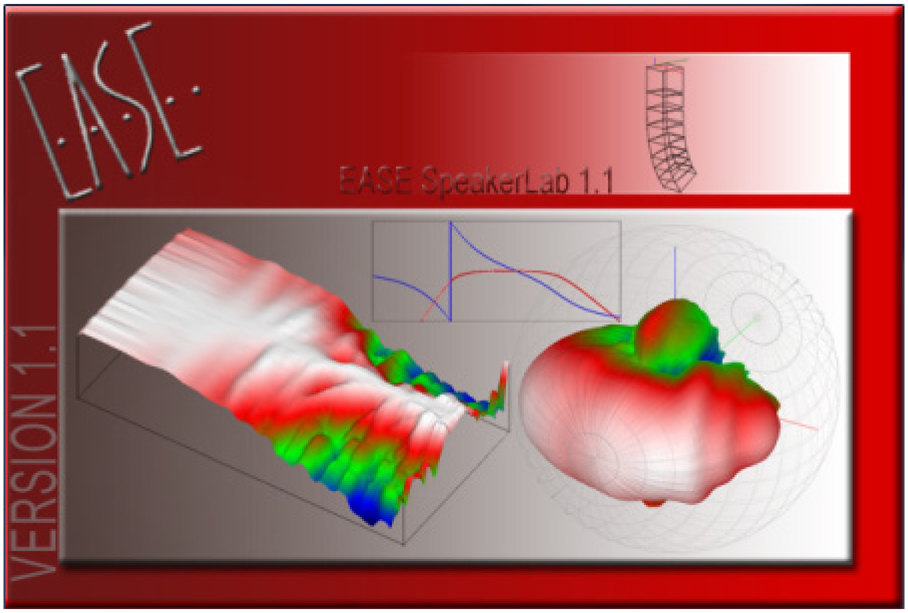
EASE SpeakerLab



Developed by
AFMG (Ahnert Feistel Media Group)

Creators of EASE, EASERA, & SysTune

www.afmg.eu

Software Manual, Rev. 1.0, December 2016

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1 Introduction to EASE SpeakerLab

1.1 Historical Overview

EASE SpeakerLab is a software tool to view and edit loudspeaker models for use with the ***EASE*** family of simulation products: ***EASE***, ***EASE Focus***, ***EASE Address***, and ***EASE Evac***. The Pro version of ***EASE SpeakerLab*** adds various advanced simulation and viewing functions as well as capabilities to export to other loudspeaker modeling data formats.

The accuracy of prediction software is related to the accuracy of its database of electro-acoustic sources. Thanks to the object-oriented approach of the GLL loudspeaker data format, ***EASE SpeakerLab*** provides a means to manage anything from simple point-sources to complex electro-acoustic systems such as modern line arrays.

In order to predict the spatial response of a simple loudspeaker or a complex sound system one needs to know the on-axis and directivity response of every source in the system. There are three main issues to manage: how to measure the spatial response, how to model the source, and store the data. In the past, when computers were not available, engineers used to draw coverage angles on paper, using data based on measured polar diagrams. This approach changed with the adoption of modeling software on PCs, as the first ***EASE*** DOS version in 1990. Until version 4.2 of the ***EASE*** software the loudspeakers were modeled using a point-source with a given on-axis frequency response. The data was only in one-third octave bands and an attenuation balloon with a 5 degree resolution (older versions were using 10 or 15 degree resolution). Everything was limited to a frequency range of 100 Hz to 10 kHz.

With the introduction of more complex sound systems used in arrays - and thus taking advantage of the interaction between sources - accurate prediction using this coarse magnitude-only data was not possible. Another approach had to be developed. A first answer to this problem was the introduction of DLL loudspeaker models, i.e. dedicated software modules to predict the response of line arrays that can be called by the ***EASE*** simulation software. The drawback of this approach is that it is "closed". DLLs also require a lot of effort in programming, skills that are outside the capability of most audio engineers. A second answer was the introduction of phase data for loudspeaker models in ***EASE 4.0***. This allows a reduction in the error of simulations using closely coupled loudspeaker boxes [1], as in the case of clusters and arrays.

Following the experience gained from DLL programming and design, and the advancements in source modeling, a new approach was introduced with ***EASE 4.2***: the GLL, for Generic Loudspeaker Library. Due to its object-oriented nature the GLL overcomes the one box – one point-source approach used previously and also enables the native handling of arrays of boxes.

The basic version of ***EASE SpeakerLab*** is included with any regular ***EASE*** license. An enhanced version, ***EASE SpeakerLab Pro***, includes advanced features and may be purchased separately.

1.2 System Requirements

Minimum Software Requirements

- Microsoft Windows Vista (Service Pack 1) and Windows 7, Windows 8, Windows 10
- Adobe Acrobat Reader 4.0 (or later)
- Microsoft .NET Framework version 3.5, which can be downloaded from the link https://www.microsoft.com/en-us/download/details.aspx?id=21

Minimum Hardware Requirements

- 2 GB RAM
- 1000 x 600 display resolution (1024 x 768 or higher recommended display resolution)
- 1.5 GHz processor speed or higher
- Multiple processors (cores) are supported and recommended

1.3 Installation & Licence Registration

Note: Dynamic Virus Protection (DVP) can cause installation problems. Before installing ***EASE SpeakerLab***, please make sure to temporarily disable any DVP software programs.

Installing and licensing ***EASE SpeakerLab*** requires very little effort. The procedure is identical to other recent AFMG software releases like ***SysTune*** or ***Reflex***. An overview of the process is described below. See Appendix A where the process is covered in detail.

1. Install the ***EASE SpeakerLab*** software package (Also install the ***AFMG Licence Manager*** if not already installed)
2. Install the ***EASE SpeakerLab*** User Certificate
3. Start the ***AFMG Licence Manager***, select ***EASE SpeakerLab***, then download a licence
4. Start ***EASE SpeakerLab***

1.4 Disclaimer

It should be noted that throughout this manual there are screenshots and photographs showing various manufacturers and products. The use of these neither implies nor suggests any preferences for these manufacturers or their products. The use of these images should also not imply any data accuracy judgments over other products or brands. Similarly, no inference should be made about any products or brands not shown in this manual. All displayed labels, products, and/or brands are used at random and without specific intent. Any trademarks that may be shown are the property of their respective owner.

1.5 Data Acquisition Overview

The correct modeling of electro-acoustic systems needs the knowledge of how the system converts the signal fed to its electrical inputs into an acoustic signal radiated into space. The correct measurement of an acoustical source's directivity response for use in GLL models is not trivial. Before starting to collect measurement data please read this manual carefully as well as the suggested literature.

At the time of writing the most recent standard on directivity data measurement is the AES56-2008 standard, published by the AES in late 2008 [2].

The radiation of an acoustical source, like any linear, time invariant (LTI) system is characterized by sampling its impulse response over the space surrounding it. A simple means to do this is to regularly sample over the surface of a sphere of a constant radius; scanning it using a meridians/parallels approach. This is described in detail later in the manual; please refer to the section 2.7 – Create New Source Definition, GSS – The Source Editor as well as section 8 – Background.

The directivity measurements can be carried out in anechoic chambers or also in non-anechoic spaces when applying impulse response windowing measurement techniques [3]. The ideal place to perform polar measurements is the anechoic chamber. However, sometimes practical limitation can lead to the use of large semi-anechoic chambers or simply suitable large non-anechoic rooms. Outdoor measurements are highly discouraged due to the uncontrolled acoustic environment in the presence of temperature change and wind. This can be especially disruptive to the measured phase response, particularly at higher frequencies.

The measurement distance between source and microphone must be sufficiently large to reduce phase errors in complex data [4]. In this instance "sufficiently large" means large compared to the source dimensions. For typical loudspeakers this usually means a minimum distance of 3 – 4 m and ideally a distance of 8 m.

Angular and frequency resolution should be adequate to describe the acoustic source accurately, that is, including all of its radiation properties in the frequency range of interest [1], [5]. Either spatial or spectral under-sampling may lead to severely reduced accuracy in subsequent simulations using such data and should be avoided.

The measurement hardware requires the use of a PC-based electro-acoustic testing system together with one or two turntables, or a robot arm, to rotate the loudspeaker along the directions of the meridians and parallels. ***EASE SpeakerLab*** can import measurement files from various test systems:

- EASERA (www.easera.com)
- MLSSA (www.mlssa.com)
- LMS (www.linearx.com)
- Monkey Forest (www.four-audio.com)
- Clio (www.audiomatica.com)
- TEF (www.gold-line.com)

SpeakerLab can also import generic wav or text impulse response files that can be acquired with any means.

There are also a few turntables and robot arms available on the market. The majority of the above software packages can control them:

- LinearX LT360 computer-controlled turntable (www.linearx.com)
- Outline ET250-3D computer-controlled turntable (www.outline.it)
- Four Audio ELF computer-controlled 3D robot arm (www.four-audio.com)

1.6 Data Formats

As previously introduced there are three ***EASE*** loudspeaker model types:

1.6.1 SPK

The SPK format is the legacy loudspeaker modeling data format used by ***EASE***. The loudspeaker system's response is modeled by a single point-source. The point-source directivity is described by means of tabular data for both magnitude and phase information in 1/3 octave resolution ranging from 100 Hz to 10 kHz. Sensitivity and maximum power allowed is provided in 1/3 octave bands as well.

This format suffers from the usual drawbacks of tabular point-source formats. It cannot implement items such as DSP filter settings or different mechanical configurations. It is also limited in its ability to accurately describe multi-way loudspeaker systems, often requiring workarounds. In addition, the fixed data format may lead to measurement data that is too coarse or it may force manual interpolation and averaging to comply with it. However, its memory requirements are fairly small and it is easily understood and produced.

1.6.2 DLL (Dynamic Linked Library)

The DLL is the most flexible format. A loudspeaker box or system can be modeled with any level of detail and using proprietary algorithms. The DLL is more flexible than the GLL but it requires programming of a win32 library which is not as straightforward as editing a GLL model with ***EASE SpeakerLab***. Also, the DLL is specific to the simulation software, that means a loudspeaker DLL for ***EASE*** can normally only be used in ***EASE***. A separate DLL for the same loudspeaker may be required for other modeling programs.

1.6.3 GLL (Generic Loudspeaker Library)

The GLL is the newest ***EASE*** loudspeaker model data format. A loudspeaker box or system can comprise multiple sound sources connected to multiple inputs as well as crossover networks. Multi-way boxes can be modeled. Line arrays and clusters are supported due to an object-oriented approach that can accurately describe acoustical, electrical, and mechanical data. Acoustical sources can be modeled using high-resolution impulse response data sets instead of fractional-octave tabular data. In contrast to previous approaches, the maximum output capability of the acoustic source is characterized by its maximum input voltage rather than its maximum input power. Entry and editing of GLL files is done using the ***EASE SpeakerLab*** software.

To summarize, the GLL is a major step forward in the features and accuracy of loudspeaker systems modeling compared to the SPK.

1.6.4 Data Conversion

In order to create a loudspeaker model, a set of information about the sources and boxes is needed. This includes the electrical and acoustical behavior of the sound sources with associated crossover filters and the mechanical description of the box and of the rigging system (in case of line arrays and clusters).

For every acoustical source the complex directivity response, the on-axis response, and the maximum voltage rating (we will treat this aspect in detail later) is needed. The measurements must be carried out with a suitable testing system (testing systems supported by ***EASE SpeakerLab*** are listed above) in a high resolution format. With the ***EASE SpeakerLab Pro*** version it is also possible to superimpose a directivity pattern from theoretical acoustical sources other than a point source onto the response of a measured source.

If the system is modeled with crossover filters the response of the filters can be included using standard IIR filters, FIR filters, or the impulse response of the actual filter can be measured and imported.

For a single loudspeaker the required mechanical data are only the dimension of the box and its weight. For line array systems some mechanical data on the suspension system are needed in order to correctly model the entire system.

1.6.5 GLL Data Types

There are three different types of GLL models that can be constructed using ***EASE SpeakerLab***: Loudspeaker, Cluster, and Line Array.

The Loudspeaker type GLL is intended to be used for modeling a single loudspeaker box that is not designed to be arrayed. Of course single loudspeakers can be arrayed by placing them side by side in ***EASE*** using multiple instances of the same GLL.

The Cluster type GLL is for loudspeaker systems that are intended to be clustered or arrayed using one or more different types of loudspeaker boxes within the GLL. It is possible to create many different fixed array configurations. There is also a provision to allow the end user to create, save, and load cluster configurations using the pre-defined loudspeaker box types in the cluster type GLL.

The Line Array type GLL is for loudspeaker systems with vertically arrayed, suspended boxes. This GLL type allows for the managing of all the issues related to these kinds of systems, such as splay angles, suspension bars, and so on. The array configuration can only be modified by the end user as allowed by the manufacturer, that is, the creator of the GLL.

1.6.6 GLL Project & Compiling

Using ***EASE SpeakerLab*** a GLL project can be edited and saved as an .xgll file. This is the file format which contains the source-code for a GLL model. The GLL model can be distributed to end users in a closed, binary .gll file format which is created, or "compiled", by ***SpeakerLab*** based on the .xgll project file.

1.7 Data Creation

GLL loudspeaker models can be created by any ***EASE SpeakerLab*** user. Nevertheless the tool is intended to be used by manufacturers or independent labs since the measurements require a set of tools that are normally only available in R&D environments. In this context ***EASE SpeakerLab*** can be used, not only as a data creation tool for modeling software, but as a loudspeaker design tool. This is possible since it provides for the simulation of the complex spatial response of electro-acoustical systems. As an example it can be used to test different crossover strategies to optimize the directivity response through the crossover region.

The ***EASE SpeakerLab Pro*** version features an export function for different simple balloon formats (the supported formats are ***EASE*** .xhn, ***CADP2*** .gdf, ***Ulysses*** .unf and ***Common Loudspeaker Format*** version 1 and 2 text files). The Pro version also has the possibility to employ directivity models different from the point source which is very useful in R&D. Additional features improve the design process, such as an automatic recalculation function for the simulated results data.

1.7.1 Manufacturers

A GLL model can be authorized by manufacturers using an .author authorization file. The .author file is supplied by AFMG upon request (please send a request email to support@afmg.eu). Additional software licenses may have to be acquired for AFMG products that are free to the end user, e.g. for EASE Focus or EASE Address.

1.7.2 Independent Labs

There are several independent labs and consultants that provide services related to loudspeaker measurements and GLL data file creation. Please consult the AFMG websites for more details at the following links.

- www.afmg.eu/index.php/partners.html
- www.afmg.eu

- speakerlab.afmg.eu
- ease.afmg.eu

1.8 Data Distribution

GLL models can be authorized and directly distributed by the manufacturers. These GLL models can also be sent to AFMG and are included in the distribution of and updates to their simulation products such as ***EASE***, ***EASE Focus***, ***EASE Address***, and ***EASE Evac***. Some of these programs, like ***EASE Focus*** and ***EASE Address***, may require an additional manufacturer license in order for the end user to be able to load the GLL into the software.

1.9 Shortcut List

SpeakerLab - Main Program

Shortcut	Description
F1	Help; Opens this User's Manual
F5	Run GLL
Ctrl+F5	Compile GLL
Ctrl+N	New Project
Crtl+O	Open Project
Ctrl+S	Save Project
Ctrl+Shift+S	Save Project As
Alt+F4	Exit the Program

View GSS - Source Editor

Shortcut	Description
F4	Properties
F5	Balloon Data
Ctrl+N	New Project
Crtl+O	Open Project
Ctrl+S	Save Project
Ctrl+A	Save Project As
Alt+F4	Exit the Program
Ctrl+Shift+B	Attenuation Balloon
Ctrl+Shift+H	Horizontal Map
Ctrl+Shift+V	Vertical Map
Ctrl+Shift+F	Frequency Spectrum

Ctrl+Shift+T	Transfer Function
Ctrl+Shift+N	Normalized Transfer Function
Ctrl+Shift+P	Polars
Ctrl+Shift+D	Directivity Index
Ctrl+Shift+I	Impedance
Ctrl+Shift+L	Maximum Input Voltage
Ctrl+Shift+S	Sensitivity
Ctrl+Shift+M	Maximum SPL
Ctrl+Alt+H	Horizontal Beamwidth
Ctrl+Alt+V	Vertical Beamwidth
Alt+3	3D Perspective
Alt+X	Side View
Alt+Y	Plan View
Alt+Z	End View

View GLL

Shortcut	Description
F5	Calculate
Ctrl+F5	Recalculate
Ctrl+S	Save Configuration
Ctrl+A	Save Configuration As
Ctrl+Shift+C	Case Drawing
Ctrl+Shift+B	Attenuation Balloon
Ctrl+Shift+H	Horizontal Map
Ctrl+Shift+V	Vertical Map
Ctrl+Shift+F	Frequency Spectrum
Ctrl+Shift+T	Transfer Function
Ctrl+Shift+N	Normalized Transfer Function
Ctrl+Shift+P	Polars
Ctrl+Shift+D	Directivity Index
Ctrl+Shift+I	Impedance
Ctrl+Shift+L	Maximum Input Voltage
Ctrl+Shift+S	Sensitivity
Ctrl+Shift+M	Maximum SPL
Ctrl+Alt+H	Horizontal Beamwidth
Ctrl+Alt+V	Vertical Beamwidth
Alt+3	3D Perspective
Alt+X	Side View

Alt+Y	Plan View
Alt+Z	End View

2 Loudspeaker Type GLLs

2.1 Single-Source GLL

The simplest GLL loudspeaker model consists of a single electric signal input connected to a single acoustic source which represents the entire loudspeaker system. A one-way loudspeaker may be modeled in this manner. A two-way (or multi-way) loudspeaker may also be modeled as a single-source GLL. This would be very similar to the older SPK loudspeaker model. For this scenario the loudspeaker system is measured with all of its crossover filters in place. As a result no filters need to be modeled inside the GLL since they are already included in the measured data of the acoustical source.

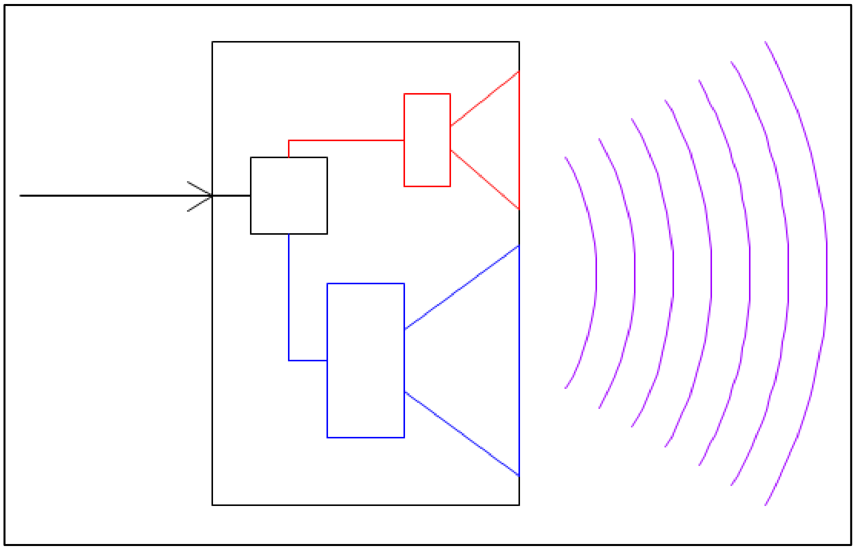


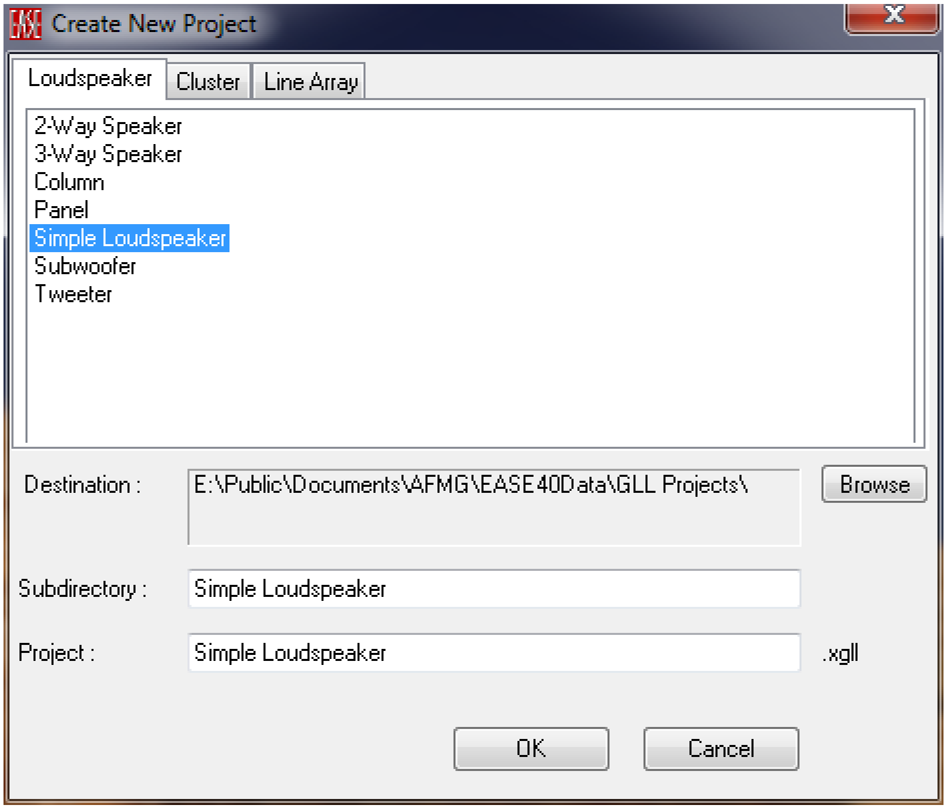
Figure 2-1 – Single-source GLL: one electrical input and one measured acoustical output

2.2 New GLL Project: Simple Loudspeaker

Our first step in creating a new GLL is by selecting the type of project we want to create. Start ***SpeakerLab*** and then click on FILE > NEW PROJECT. This opens the Create New Project wizard. The three tabs at the top select the type of GLL to be created. Select the Loudspeaker type GLL. There are several selections under the LOUDSPEAKER tab. For this example (single-source) we would like to use the SIMPLE LOUDSPEAKER.

Below this selection area we can tell ***SpeakerLab*** the folder location where we want to place a new subfolder that will contain all of our project files. Do this by clicking on the BROWSE button and selecting the desired folder location. Next type the name of the new subfolder in the SUBDIRECTORY field. Finally type the name of the loudspeaker, or other descriptive name, into the PROJECT field. This will be the name of the .xgll project file and the default name of the complied .gll file. Don't worry if you want to change it later. This is easy to do.

Clicking the OK button will create the new project, close the wizard, and open the ***SpeakerLab*** main window.



Destination : E:\Public\Documents\AFMG\EASE40Data\GLL Projects\

Subdirectory : Simple Loudspeaker

Project : Simple Loudspeaker .xgll

Figure 2-2 – Create New Project dialog box

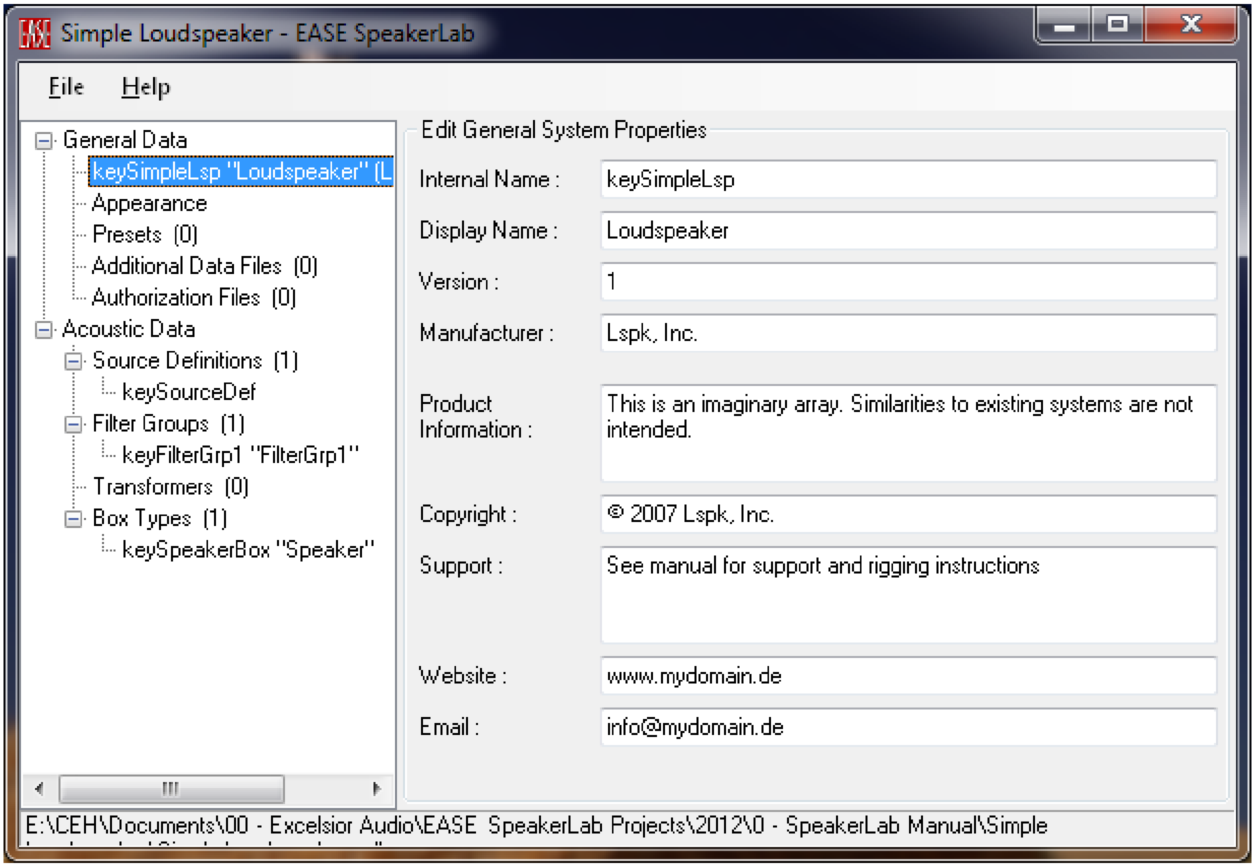
2.3 SpeakerLab Main Window

The ***SpeakerLab*** main window is divided into two main areas. The area on the left shows the property tree of the loudspeaker model. This tree is divided into several categories. For the simple loudspeaker there are only two, General Data and Acoustic Data. Loudspeaker models requiring more detail, such as Clusters and Line Arrays, will also have a Mechanical Data category.

Selecting an item in the tree on the left will change the display on the right to show all of the properties associated with that item on the tree. Any of these properties on the right can be changed as desired.

2.4 General Data

Let's start with the first item under General Data. These are the General System Properties. The first property here is the Internal Name. This is the name the different AFMG programs use internally to reference this GLL. This can be left as is or it can be changed if something else if you desire. The Display Name is the name of the loudspeaker model that will be shown when working with this GLL in different programs (***EASE***, ***Focus***, ***Evac***, etc.) This should be descriptive to indicate something similar to the name of the real loudspeaker being modeled. As we will see later on, there are other items in the GLL that have both internal name property and a display name property.



Edit General System Properties

Internal Name : keySimpleLsp

Display Name : Loudspeaker

Version : 1

Manufacturer : Lspk. Inc.

Product Information : This is an imaginary array. Similarities to existing systems are not intended.

Copyright : © 2007 Lspk. Inc.

Support : See manual for support and rigging instructions

Website : www.mydomain.de

Email : info@mydomain.de

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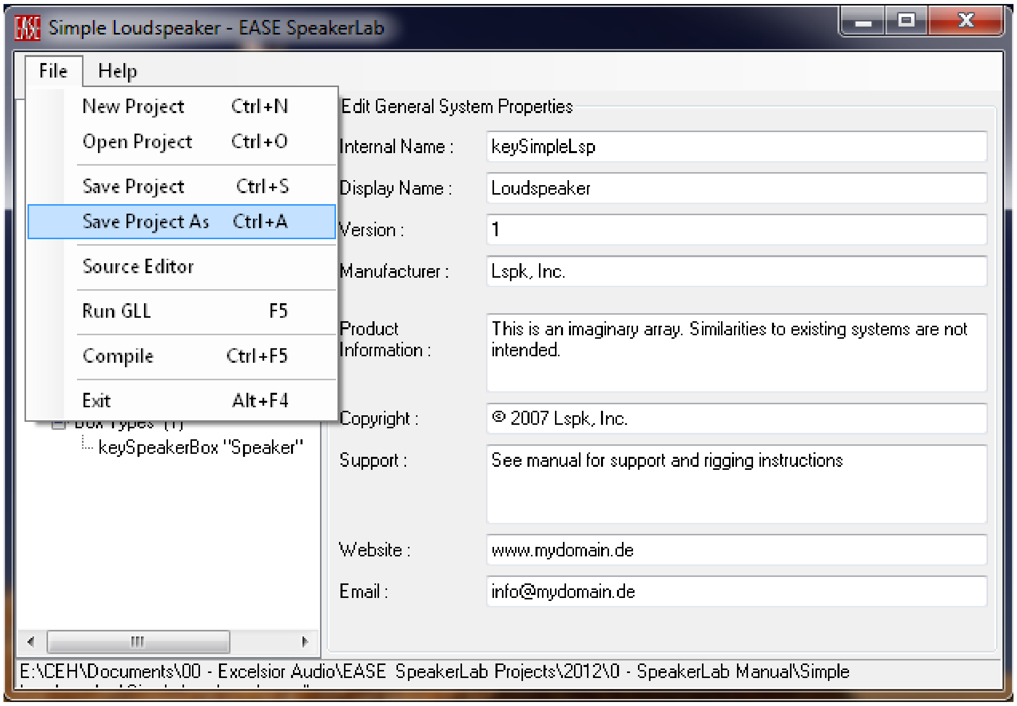
Figure 2-3 – Edit General System Properties page

The Version property is a very useful way to track changes to a project as well as for updates released for publication to end users. By incrementing the Version number it is possible to easily tell if one GLL is more recent or if it is older than another GLL for the same loudspeaker. It is often useful to start a project with the version number at 0.1. As changes are made and tested, the version number can be incremented as needed. When the final GLL is ready for release it can go to version 1.0 or higher for example.

The other properties shown here (Manufacturer, Product Information, Copyright, Support, Website, and Email) are fairly self-explanatory. All of these items are displayed in the GLL when the end user selects to display the Info about the GLL. It can be very useful to put important notes, instructions, or tips in the Product Information property.

2.5 Saving the XGLL Project

Once the GENERAL SYSTEM PROPERTIES have been modified it is a good idea to save the XGLL project using FILE > SAVE PROJECT or FILE > SAVE PROJECT AS. The SAVE PROJECT AS option allows a new file name to be specified prior to the file being saved.



Edit General System Properties

Internal Name : keySimpleLsp

Display Name : Loudspeaker

Version : 1

Manufacturer : Lspk, Inc.

Product Information : This is an imaginary array. Similarities to existing systems are not intended.

Copyright : © 2007 Lspk, Inc.

Support : See manual for support and rigging instructions

Website : www.mydomain.de

Email : info@mydomain.de

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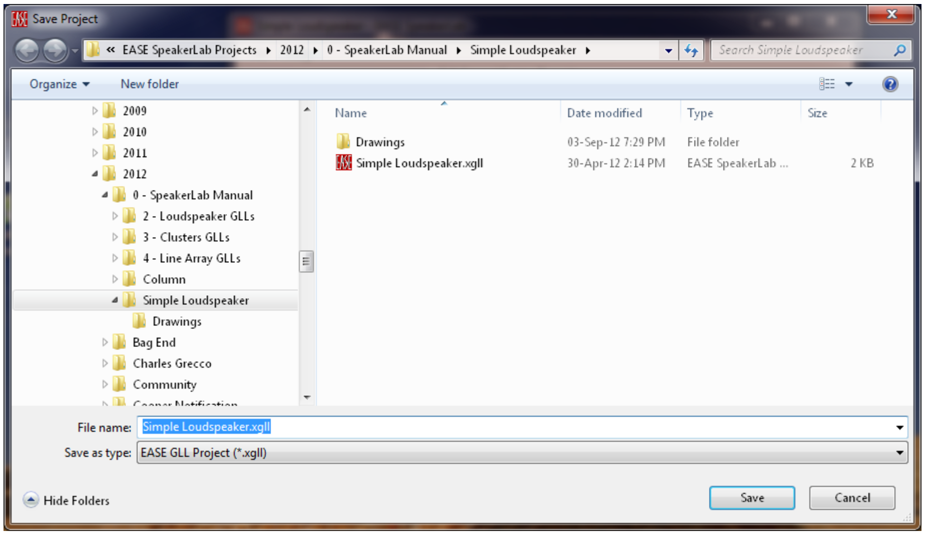
Figure 2-4 – Save the xgll project file



Name	Date modified	Type	Size
Drawings	03-Sep-12 7:29 PM	File folder	
Simple Loudspeaker.xgll	30-Apr-12 2:14 PM	EASE SpeakerLab ...	2 KB

File name: Simple Loudspeaker.xgll

Save as type: EASE GLL Project (*.xgll)

Figure 2-5 – Select the folder and file name

2.5.1 Verify the GLL Before Closing

It is recommended that prior to saving and closing an XGLL file that the GLL be run to test that there are no errors. This can be done by selecting `FILE > RUN GLL` or by pressing F5. If errors are present in the XGLL that prevent it from running as a GLL, SpeakerLab will display this. They can then be corrected before saving the XGLL. Saving an XGLL file with errors and then closing the project file may make it impossible for ***SpeakerLab*** to open the XGLL file later. These errors will have to be corrected manually in order to salvage the XGLL file.

2.5.2 XGLL File Format

The XGLL file is actually a simple text file. It contains all the information that can be edited using ***SpeakerLab***, organized by means of a markup language.

The XGLL file can be edited with any text editor program. The sample XGLL file shown in Figure 2-6 is for the GLL model shown above.

Note that there is a specific hierarchy for the information contained in the XGLL file. If an XGLL file is edited and the something is not where ***SpeakerLab*** expects it to be, ***SpeakerLab*** may not be able to open the file. For full details on the formatting and content of the XGLL file please see Appendix A – XGLL Project File – Text File Organization.

```
"GLL"
"Format", "3D"
"FormatVersion", "1.0"

"System", "Loudspeaker", "keySimpleLsp", "LS"
"SystemVersion", 1.00
"Company", "Lspk, Inc."

"InfoText", "This is an imaginary array. Similarities to existing systems are not intended."
"CopyrightText", "© 2007 Lspk, Inc."
"SupportText", "See manual for support and rigging instructions"
"WebsiteText", "www.mydomain.de"
"EmailText", "info@mydomain.de"

"Logo1", ""
"Logo2", ""
"BackgroundColor", 0, 0, 0

"BoxTypes", 1
   "BoxType", "Speaker", "keySpeakerBox"
       "Sources", 1
          "Source", "Source", "Source", 0.00, 0.00, -0.20, 0.00, 0.00, 0.00, "keySourceDef"
       "Input Configurations", 1
          "Input Configuration", "Default", "InputConfigA"
       "Inputs", 1
          "Input", "Input"
        "Links", 1
            "Link", "Source", ""
       "GeometryFiles", 1
          "GeometryFile", "Drawings\CaseSimpleLsp.xed"
       "ReferencePoint", 0.00, 0.00, -0.15
       "NextPivot", 0.00, 0.00, 0.00
       "CenterOfMass", 0.00, 0.125, -0.15
       "Weight", 5.00

"SourceDefinitions", 1
   "SourceDefinition", "keySourceDef"
       "File", "<SPHERE>"

"FilterGroups", 0
```

Figure 2-6 – Sample .xgll file

2.6 Acoustic Data

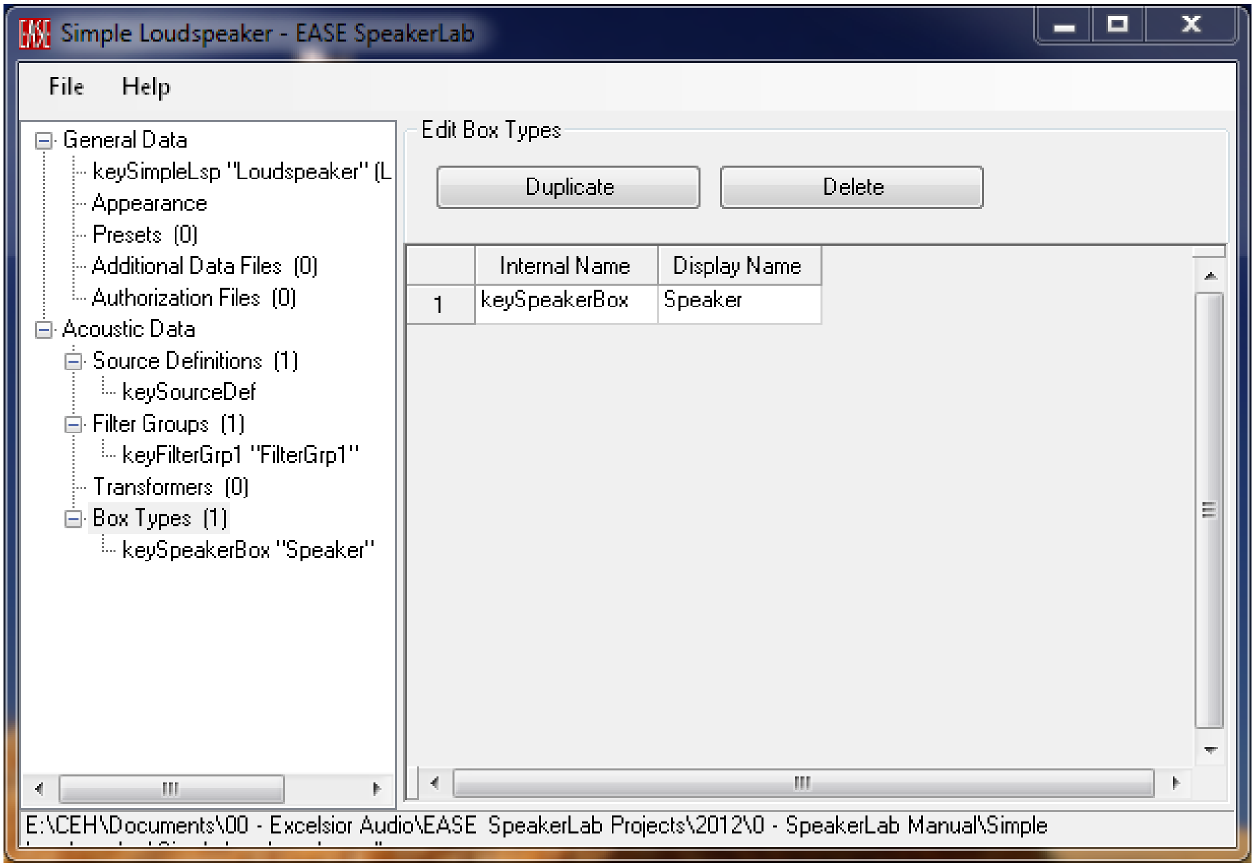
In this first example we will skip the other General Data properties for now (they’re covered in section 2.9 – General Data, Part 2) and move on to the Acoustic Data properties. In this example model we will have one box type, one source, one input, and no filters.

Clicking on the + sign next to Acoustic Data will expand the property tree to show the different groups of acoustical properties for the loudspeaker model.

2.6.1 Box Types

By clicking on Box Types the Edit Box Types page is displayed on the right side of the main window. Since we are working with a simple loudspeaker-type GLL only one box type is allowed so no others can be added. Multiple box types are allowed in Cluster and Line Array models.

Each box type has an Internal Name and a Display Name. These may be changed to more descriptive names if desired. It is recommended that the Display Name be something easily identified by end users of the GLL.



- General Data
  - keySimpleLsp "Loudspeaker" (L
  - Appearance
  - Presets (0)
  - Additional Data Files (0)
  - Authorization Files (0)
- Acoustic Data
  - Source Definitions (1)
    - keySourceDef
  - Filter Groups (1)
    - keyFilterGrp1 "FilterGrp1"
  - Transformers (0)
  - Box Types (1)
    - keySpeakerBox "Speaker"

	Internal Name	Display Name
1	keySpeakerBox	Speaker

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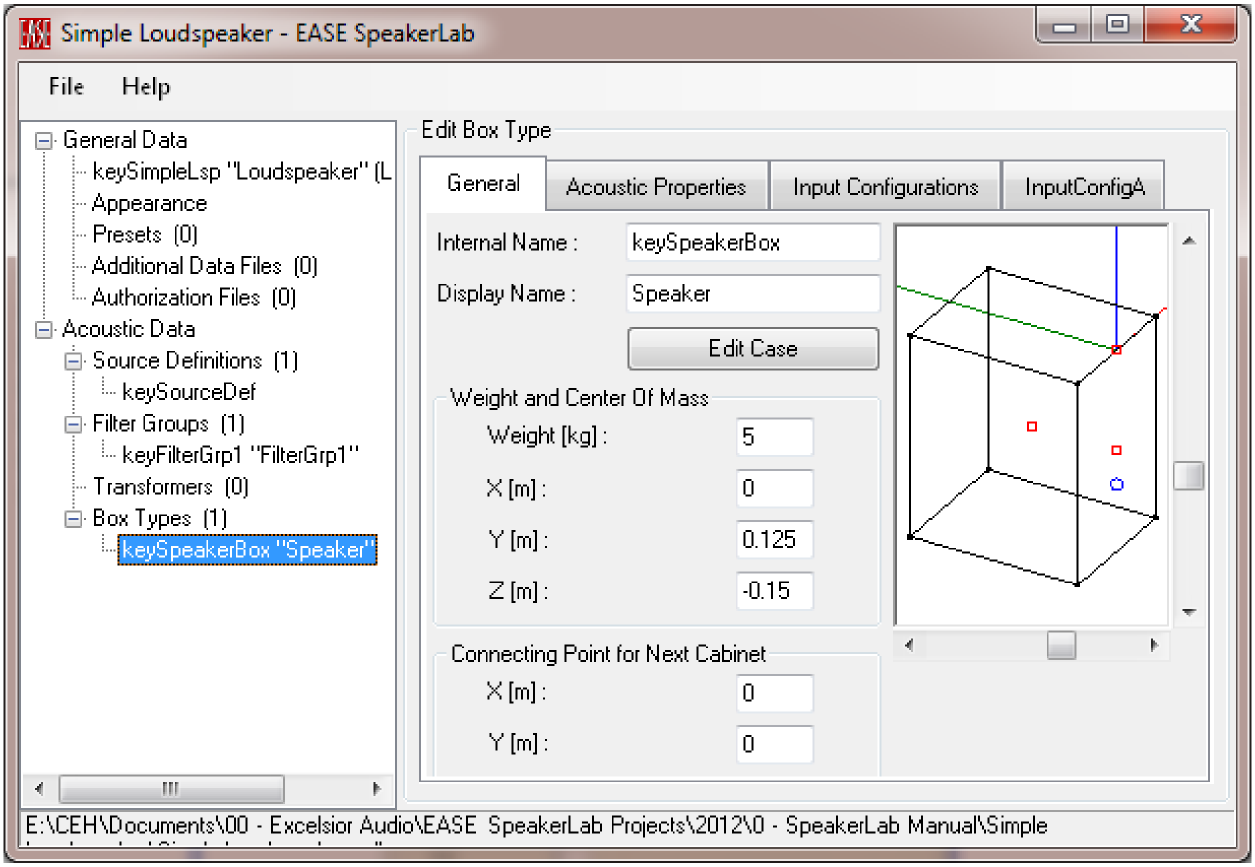
Figure 2-7 – Selecting the Box Type from the property tree

Clicking on the element KeySpeakerBox "Speaker" in the property tree opens the Edit Box Type page on the right. Here we have several different tabs: General, Acoustic Properties, Input Configurations and a dynamic tab labeled InputConfigA. This last tab will change its name based on the Internal Name entered on the Input Configurations tab.

2.6.1.1 General Tab

The General tab allows editing the Internal Name and the Display Name of the box, changing the case drawing that is shown (Edit Case button), specifying the Center Of Mass, Weight, and

Connecting Point for the Next Cabinet. The Weight, Center of Mass, and Connecting Point for Next Cabinet are only used for calculations of Line Array type GLLs. They are informational for any other type of GLL and may be omitted.



Internal Name : keySpeakerBox

Display Name : Speaker

Weight and Center Of Mass

Weight [kg] : 5

X [m] : 0

Y [m] : 0.125

Z [m] : -0.15

Connecting Point for Next Cabinet

X [m] : 0

Y [m] : 0

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Figure 2-8 – General tab on the Edit Box Type page

A drawing of the loudspeaker box used in this example is shown in Figure 2-9. Here we can see the overall dimensions of the box as well as the locations of some of the 3D properties.

The Case Data is stored in a geometry data file. This is a file defining the edges of the box in either a .xed (***EASE*** text edge) or .fed (***EASE*** binary edge) file. Either of these file formats may be selected by clicking the EDIT CASE button. A drawing with the location of each vertex is shown in Figure 2-10.

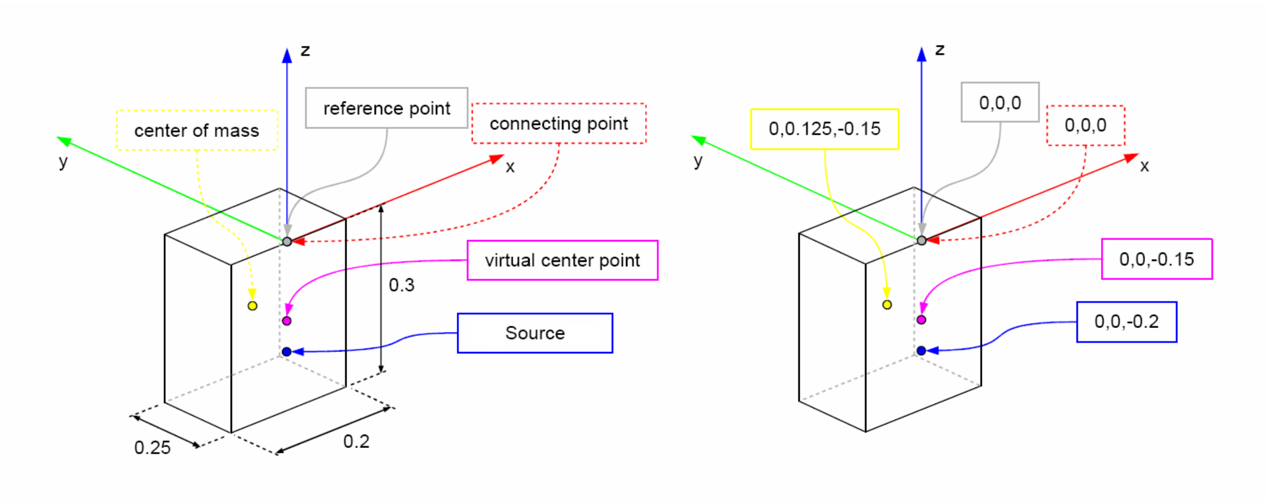


Figure 2-9 – Loudspeaker box drawing

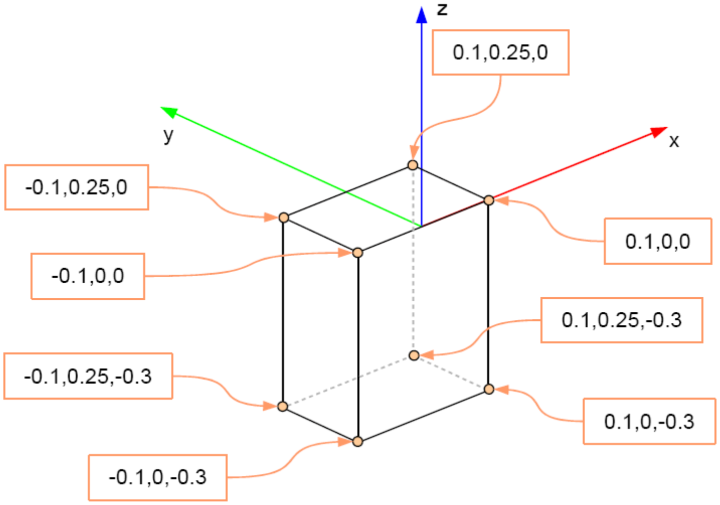
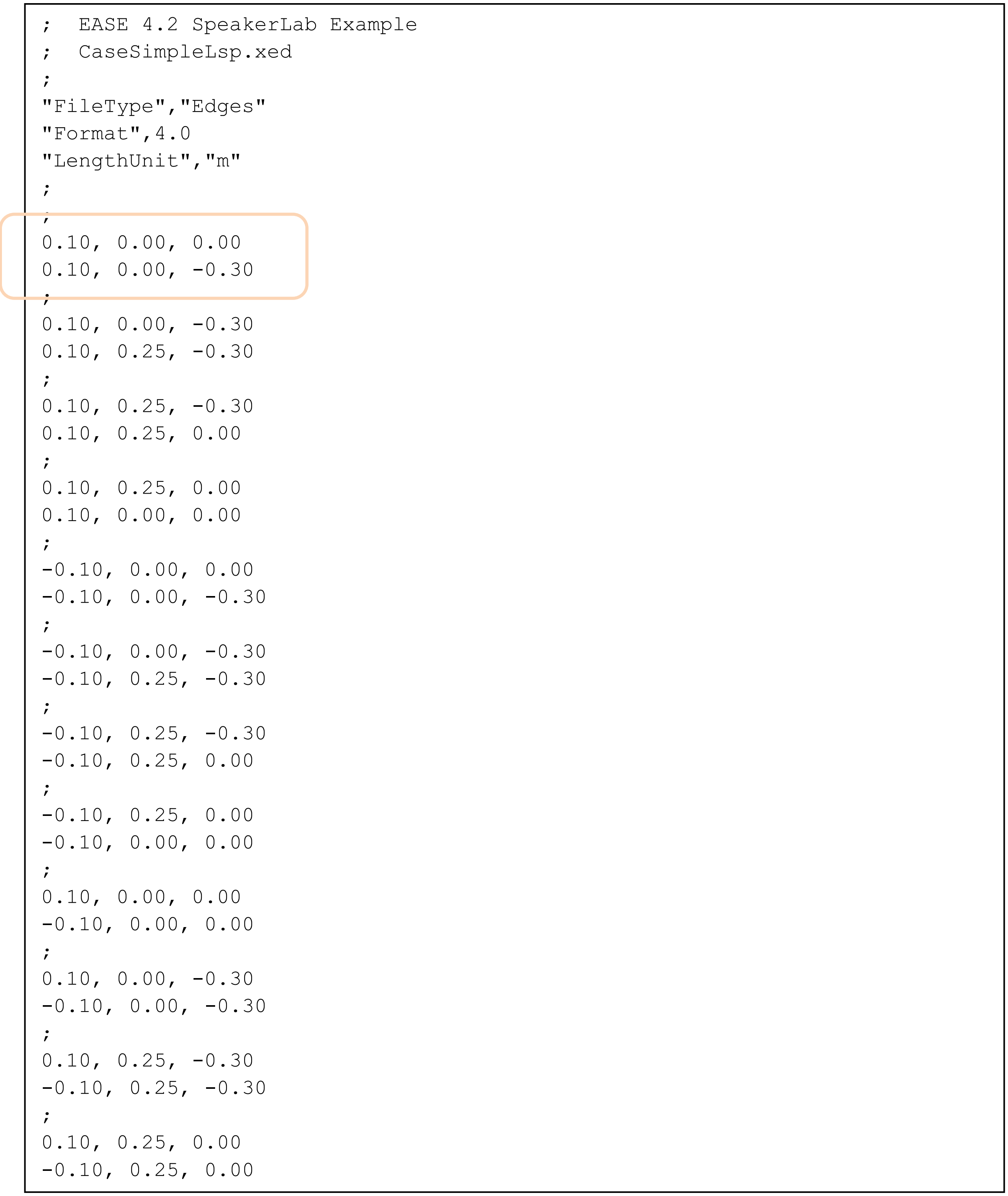


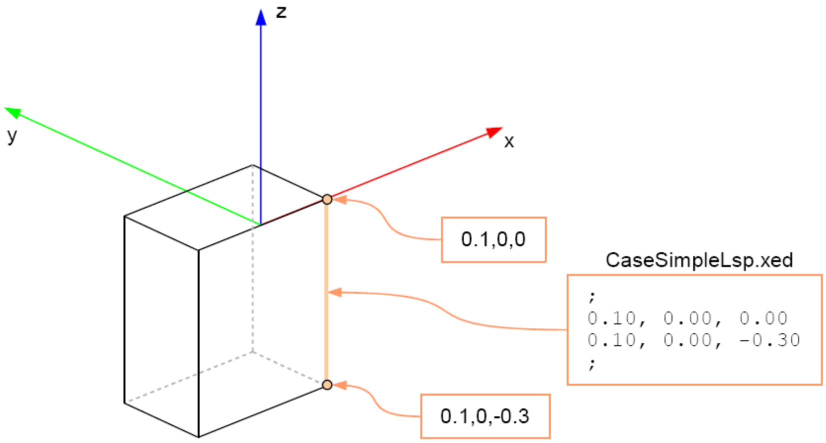
Figure 2-10 – Loudspeaker box drawing with vertex locations

The .xed text file can be edited directly using any text editor. Each edge is defined by the start and end points of each edge. An example is shown in Figure 2-11. The edge defined by the circled lines in the file is illustrated graphically in Figure 2-12.



```
;  EASE 4.2 SpeakerLab Example
;  CaseSimpleLsp.xed
;
"FileType","Edges"
"Format",4.0
"LengthUnit","m"
;
;
0.10, 0.00, 0.00
0.10, 0.00, -0.30
;
0.10, 0.00, -0.30
0.10, 0.25, -0.30
;
0.10, 0.25, -0.30
0.10, 0.25, 0.00
;
0.10, 0.25, 0.00
0.10, 0.00, 0.00
;
-0.10, 0.00, 0.00
-0.10, 0.00, -0.30
;
-0.10, 0.00, -0.30
-0.10, 0.25, -0.30
;
-0.10, 0.25, -0.30
-0.10, 0.25, 0.00
;
-0.10, 0.25, 0.00
-0.10, 0.00, 0.00
;
0.10, 0.00, 0.00
-0.10, 0.00, 0.00
;
0.10, 0.00, -0.30
-0.10, 0.00, -0.30
;
0.10, 0.25, -0.30
-0.10, 0.25, -0.30
;
0.10, 0.25, 0.00
-0.10, 0.25, 0.00
```

Figure 2-11 – Sample .xed file



CaseSimpleLsp.xed

```
;
0.10, 0.00, 0.00
0.10, 0.00, -0.30
;
```

Figure 2-12 – Edge of loudspeaker box drawing

It is also possible to graphically edit the case drawing using the ***EASE Project Data*** module or the ***EASE SpeakerBase*** Edit Case tool and export the drawing as a .xed file. Alternatively, the .fed file can be used directly without exporting the drawing as a .xed file. The .fed file is part of every ***EASE*** project that includes edges and is located in the same folder as the .frd main project file itself. The only difference between using a .fed or a .xed files in the GLL, is that the .fed file will also display the vertices (points) used in the case drawing along with the edges. The .xed file will display only the edges for the case drawing along with the points at each end of the edge. The .xed is made of only edges so there are no points to be seen other than those establishing the edges. Placing extra points in a drawing at the location of some of the properties (sources, center of mass, connecting point for next cabinet, etc.) and having them displayed when working on the .xgll project file can be useful as a check that these property coordinates are set correctly.

2.6.1.2 Acoustic Properties Tab

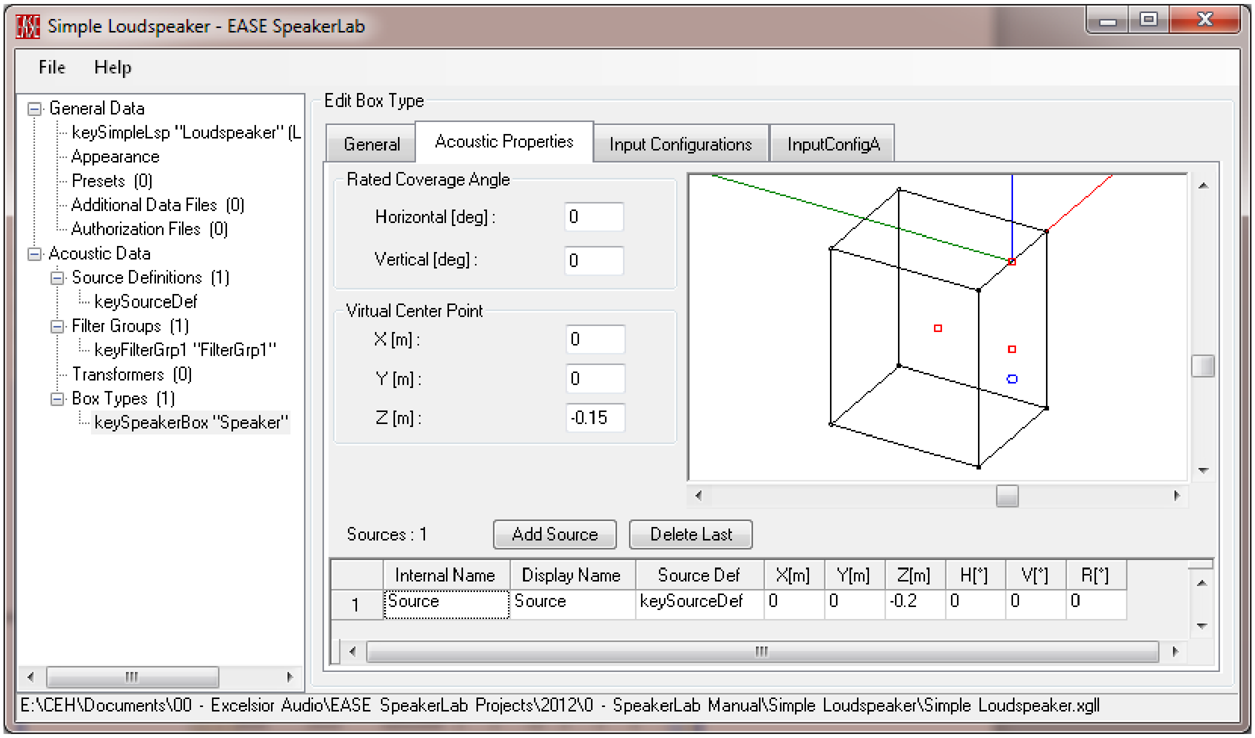
The Acoustic Properties tab is where the properties of the acoustical sources in the loudspeaker model are specified. In this simple case we have only a single source, but we will see later on how to add more sources.

At the bottom of the Acoustical Properties tab there is a section for adding and deleting sources. For every source we have an Internal Name, Display Name, a link to a Source Definition, and the source position and orientation (aiming) relative to the coordinate system of the box. The Internal Name and Display Name can be any name desired. Something short but descriptive is recommended.

Clicking a cell in the Source Def column will open the Select a Source dialog box. This dialog box allows one of the existing Source Definitions to be selected and assigned to this particular source in the loudspeaker box. The selection is made via the Internal Name given to the Source Definition.

The remaining columns (X[m], Y[m], Z[m], H[°], V[°], and R[°]) allow the position of the source and its orientation to be specified. The coordinates are in meters and are relative to the origin (0, 0, 0) of the coordinate system for the loudspeaker case drawing. When using a theoretical point source (IDEAL SPHERE), as an initial design exercise for example, these coordinates should be the actual location of the source in the loudspeaker box.

However, when actual measured data is used (GSS DATA FILE) the location for the Point of Rotation (POR) used during the measurement should be specified for the location. This is an extremely important item. Failure to specify the correct, actual POR as the location for the source will result in errors in the loudspeaker model simulations [4].



Rated Coverage Angle

Horizontal [deg] : 0

Vertical [deg] : 0

Virtual Center Point

X [m] : 0

Y [m] : 0

Z [m] : -0.15

Sources : 1

	Internal Name	Display Name	Source Def	X[m]	Y[m]	Z[m]	H[°]	V[°]	R[°]
1	Source	Source	keySourceDef	0	0	-0.2	0	0	0

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\Simple Loudspeaker\Simple Loudspeaker.xgll

Figure 2-13 – Acoustic Properties tab

Above the list of sources we can specify the RATED COVERAGE ANGLES for the loudspeaker. This defines the broadband beamwidth for this box type. This value is mainly for informational purposes. However, it is used in some applications like ***EASE Address*** and ***EASE Evac*** as a visual orientation when designing loudspeaker layout patterns. Leaving these values at 0 will cause problems when attempting to use the GLL in ***EASE Address*** or ***EASE Evac***.

Since the RATED COVERAGE ANGLES property is critical for the GLLs functionality in certain programs these values should be accurate to aid end users in their design efforts with these programs. While no standard yet exists for determining these values a recommended method is to calculate a weighted average of the loudspeaker's beamwidth for the two octave frequency range of 1 – 4 kHz that is critical for speech intelligibility. The recommended weighted average calculation is detailed in Figure 2-14.

```
A = Average beamwidth from 1-4 kHz
B = Average beamwidth from 2-4 kHz
Weighted Average = (A + B) / 2
```

Figure 2-14 – Recommended calculation of Rated Coverage Angles

This calculation can be easily programmed into a spread sheet. The tabular beamwidth data can then be exported from the GSS or the GLL (section 2.7.11 – Exporting Data) and copied into the spread sheet to automatically calculate the weighted average.

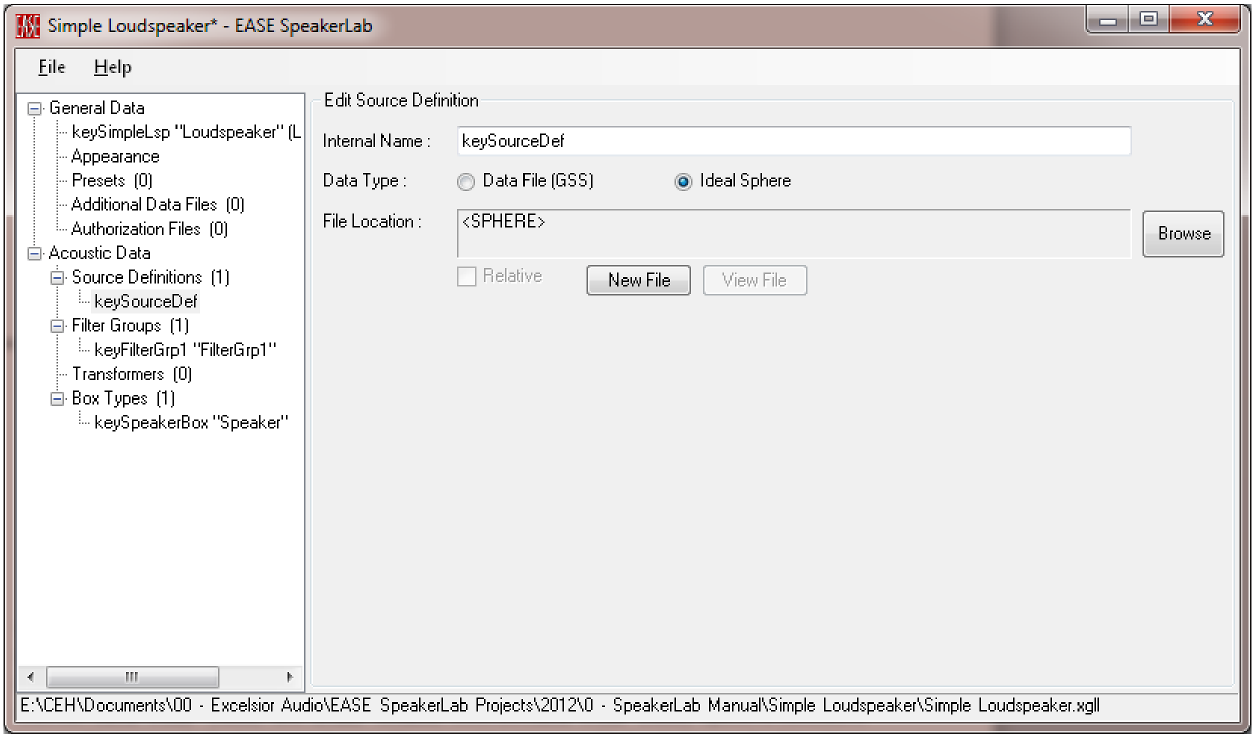
The Virtual Center Point is used for several auxiliary calculations relating to the box as a whole. If there is just a single source, the Virtual Center Point should have the location of this source. Otherwise the geometric center of all sources or of the box should be used.

2.6.1.3 Input Configuration Tabs

The other two tabs (Input Configurations and InputConfigA) are used to specify the signal path inside the box, for example between the electrical input to the box and the acoustical source(s) inside the box. If there are multiple configurations defined (as we will show later), it is possible for the end user to switch between these configurations from the GLL user interface. In the current example we have only one input configuration that connects the single electric input to the single acoustic source.

2.6.2 Source Definitions

At this point we need to specify which acoustic source (the directivity data) to use. Just as with Box Types all of the Source Definitions referenced by the GLL project are available as elements in the property tree on the left. Expanding the Source Definitions and selecting the keySourceDefinition from the property tree shows the Edit Source Definition screen in the page on the right.



Internal Name : keySourceDef

Data Type : ○ Data File (GSS) ● Ideal Sphere

File Location : <SPHERE>

□ Relative

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\Simple Loudspeaker\Simple Loudspeaker.xgll

Figure 2-15 – Edit Source Definition page

Here we can specify the Internal Name of the source and which Data Type to use by choosing between Data File (GSS) and Ideal Sphere. Once the Data File (GSS) option is selected it is possible to specify the location of a .gss file or create a new .gss file by clicking on the New File button. This button opens a new window with the GSS Source Editor and viewer. This editor can be also launched from the file menu under File > Source Editor.

Since our model is "Simple Loudspeaker" with one electrical input and a single acoustic source, we need to create the GSS source on the basis of the balloon measurements of directivity response for the whole loudspeaker box. Please refer to the "Source Editor" section following below for more details on how to create a GSS data file.

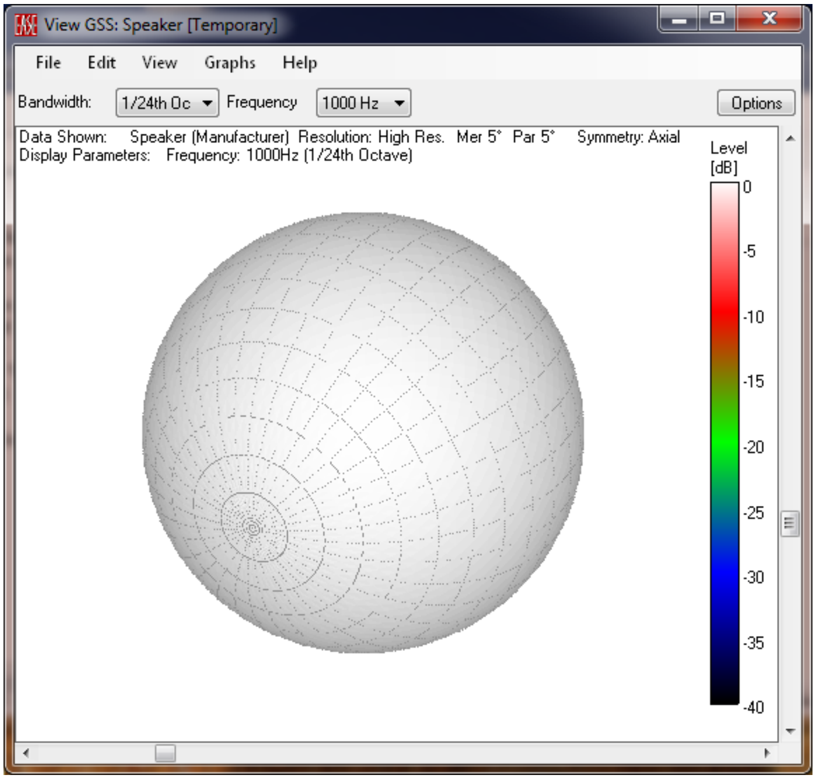
2.6.3 Relative File Paths

It can be noticed that after selecting the .gss data file it is possible to check the Relative checkbox. This checkbox controls if the .xgll project file stores the path to the selected file relative to the location of the .xgll project file. When it is checked the path is relative to the .xgll project file. When it is unchecked the path is absolute. It is strongly recommended to always check the Relative checkbox and use relative paths. This will make it possible to move entire projects to other computers if needed. See section 6.1 – Folder Structure – Keeping Everything Together & 6.2 – Relative File Locations & Transferring Projects for more info on the recommended folder structure for GLL projects.

2.7 Create New Source Definition, GSS – The Source Editor

The Source Editor is the tool to create, view, and edit Generic Sound Source files (GSS) to be used in GLL loudspeaker models. A GSS describes the sound source in terms of the directivity of its radiation into space (Balloon Data) and its other characteristics such as sensitivity, impedance and maximum allowed voltage (Properties).

Start the Source Editor by selecting File > Source Editor from the menu of ***EASE SpeakerLab*** or by opening any GSS file on your hard disk. You may also load a GLL project and launch the Source Editor by clicking the New File or View File button found on the Edit Source Definition screen under the Source Definitions item on the property tree in ***SpeakerLab***.



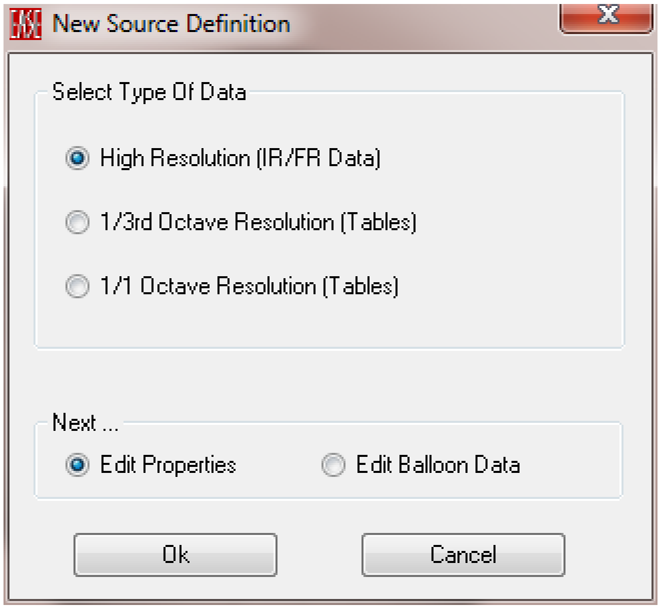
Bandwidth: 1/24th Oc Frequency 1000 Hz

Data Shown: Speaker (Manufacturer) Resolution: High Res. Mer 5° Par 5° Symmetry: Axial
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-16 – GSS source editor dialog box

2.7.1 Defining a New Source (High Resolution Data)

Selecting File > New from the Source Editor main menu opens the New Source Definition dialog seen in Figure 2-17.



New Source Definition

Select Type Of Data

● High Resolution (IR/FR Data)

○ 1/3rd Octave Resolution (Tables)

○ 1/1 Octave Resolution (Tables)

Next ...

● Edit Properties ○ Edit Balloon Data

Figure 2-17 – New Source Definition dialog box

The user is prompted to choose between three data formats for the new GSS file to be created.

- High Resolution (IR/FR Data)
- 1/3 Octave Resolution (Tables)
- 1/1 Octave Resolution (Tables)

The High Resolution data requires measured impulse response (or complex frequency response) data with the desired angular resolution. The two latter choices use a tabular approach where each frequency has a table of attenuation data with columns and rows corresponding to polar angles.

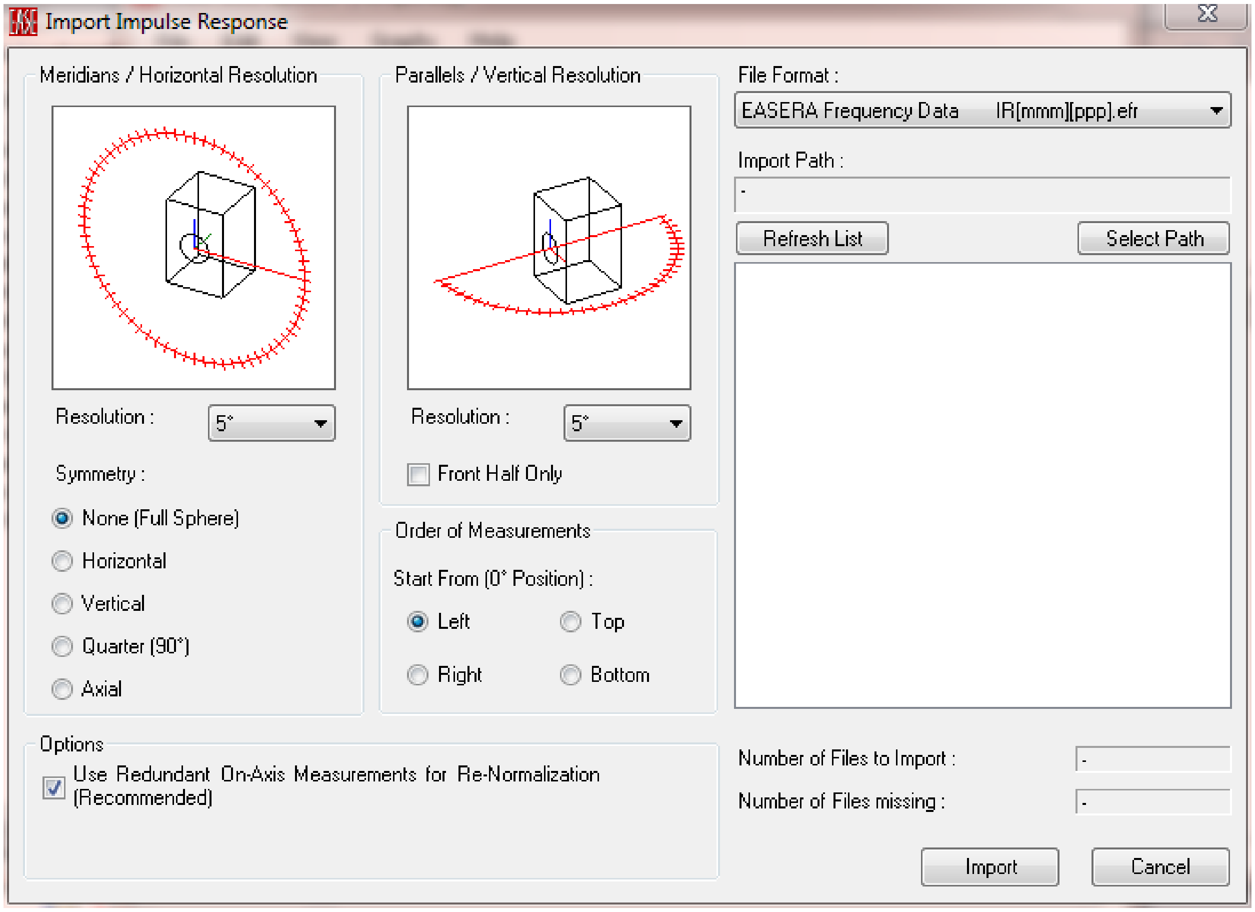
For the development of new loudspeaker models it is recommended that the high-resolution data format be used. This is also the recommendation of the AES standard on source measurements [2] (using an IR resolution of at least 8192 points). The tabular formats can still be employed. However they are indented to be used to import data from legacy low resolution formats such as the ***EASE Speaker Base*** (SPK) or for experimental/research use.

The New Source Definition dialog lets you also decide which operation to perform right after pressing the OK button: Edit Properties or Edit Balloon Data. The default is Edit Balloon Data. This same dialog can be opened by selecting Edit > Balloon Data (F5) from the Source Editor menu. When the High Resolution type of data is selected this will open the Import Impulse Response dialog box (Figure 2-18). If one of the tabular data formats is selected the Edit Table dialog box will be opened.

2.7.2 Importing Directivity Balloon Data

The IMPORT IMPULSE RESPONSE dialog box allows importing a set of directivity measurements in various files formats. The measurement system's file formats that are currently supported in SpeakerLab are shown below. To the right of each of these is the file nomenclature expected by SpeakerLab for properly importing the set of directivity measurements.

EASERA Time Data	IR[mmm][ppp].etm
EASERA Frequency Data	IR[mmm][ppp].efr
EASERA Text Data	IR[mmm][ppp].etx
Wave Time Data	V[mmm]H[ppp].wav
MLSSA Time Data	IR[mmm][ppp].tim
MLSSA Frequency Data	IR[mmm][ppp].frq
LMS Frequency Data	IR[mmm][ppp].txt
MF Frequency Data	V[mmm]H[ppp].dat
MF Time Data	V[mmm]H[ppp].spk
Clio Time Data	IR [m*100] [p*100].txt



Import Impulse Response

Meridians / Horizontal Resolution

Resolution : 5°

Symmetry :

● None (Full Sphere)
○ Horizontal
○ Vertical
○ Quarter (90°)
○ Axial

Parallels / Vertical Resolution

Resolution : 5°

□ Front Half Only

Order of Measurements

Start From (0° Position) :

● Left ○ Top
○ Right ○ Bottom

File Format : EASERA Frequency Data IR[mmm][ppp].efr

Import Path : -

Options

☑ Use Redundant On-Axis Measurements for Re-Normalization (Recommended)

Number of Files to Import : -

Number of Files missing : -

Figure 2-18 – Import Impulse Response dialog box

2.7.2.1 Angular Resolution & File Names

The coordinate system used in ***EASE SpeakerLab*** is basically the same as that used in ***EASE***. The source on-axis direction points along the negative y-axis direction (Figure 2-19).

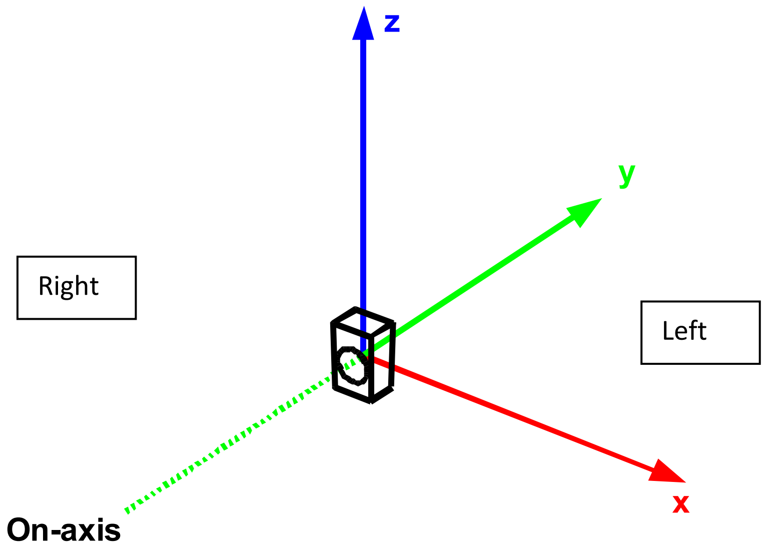


Figure 2-19 – On-axis direction in the EASE SpeakerLab coordinate system

The directivity of a source is determined by taking impulse response measurements over the surface of a sphere surrounding the source. It is possible to think of the source as being placed at the center of the earth. The North Pole represents the on-axis direction of the source. The impulse response is measured over a set of Meridians (Meridians/Horizontal Resolution). Each Meridian contains a set of Parallels (Parallel/Vertical Resolution).

Every measured point on the surface of a sphere surrounding the source can be related to the Meridian (m) and Parallel (p) angle. During the measurement process each measured file needs to be saved with a particular filename that will allow ***EASE SpeakerLab*** to know the measurement point in space based on the filename. As an example, the measurement point shown in Figure 2-20 is at an elevation (Meridian) of 30° and an azimuth (Parallel) of 10°. A file name that would denote this position might be IR030010. The "030" portion of the file name denotes the Meridian, while the "010" portion denotes the Parallel.

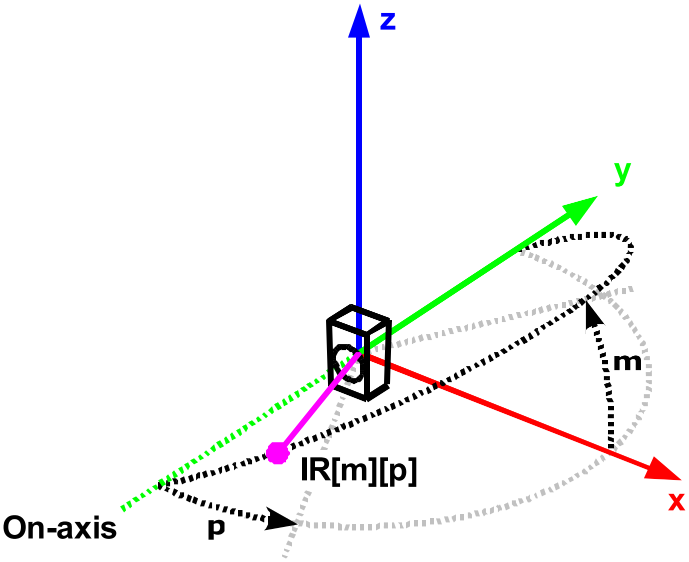


Figure 2-20 – Coordinate system in SpeakerLab showing a measurement point as a function of meridians & parallels

Once the coordinate system used in ***EASE SpeakerLab*** is clearly understood, the IMPORT IMPULSE RESPONSE dialog should be very straightforward.

The angular sampling resolution used during the measurement of the source's directivity must be specified prior to importing the measurements. This is required so that ***SpeakerLab*** will know how many files should be present and what they are named. The angular resolution of the Meridians and the Parallels can be specified individually. The MERIDIANS/HORIZONTAL RESOLUTION can range from 1° to 90° at fixed values. Selecting 90°, for example, would mean that only horizontal and vertical polar measurements were taken.

The PARALLELS/VERTICAL RESOLUTION can range from 1° to 45° at fixed values. There is also check box option to import the FRONT HALF ONLY of the radiation from the measured source (Parallel angles not greater than 90 degrees). This can be is particularly useful when working with ceiling-mounted or in-wall loudspeakers.

2.7.2.2 Symmetry

The MERIDIANS/HORIZONTAL RESOLUTION area allows also defining the SYMMETRY for the source to be imported. If a source should be symmetrical in its directivity response, selecting the corresponding type of Symmetry will allow few measurements to be made. There is no need to measure the redundant, mirror image locations.

Examples of each type of Symmetry are shown in Figure 2-21 – Figure 2-25.

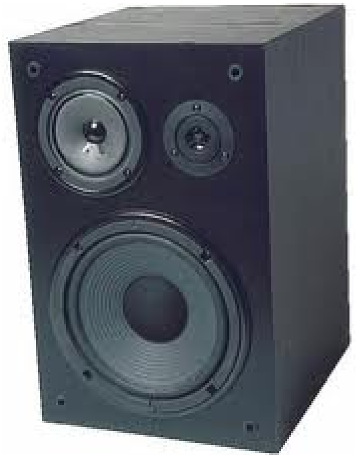


Figure 2-21 – Symmetry = None

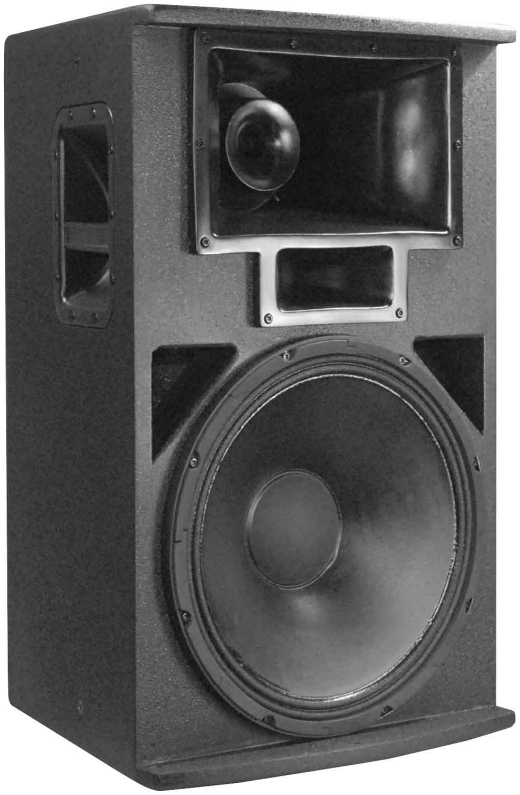


Figure 2-22 – Symmetry = Horizontal

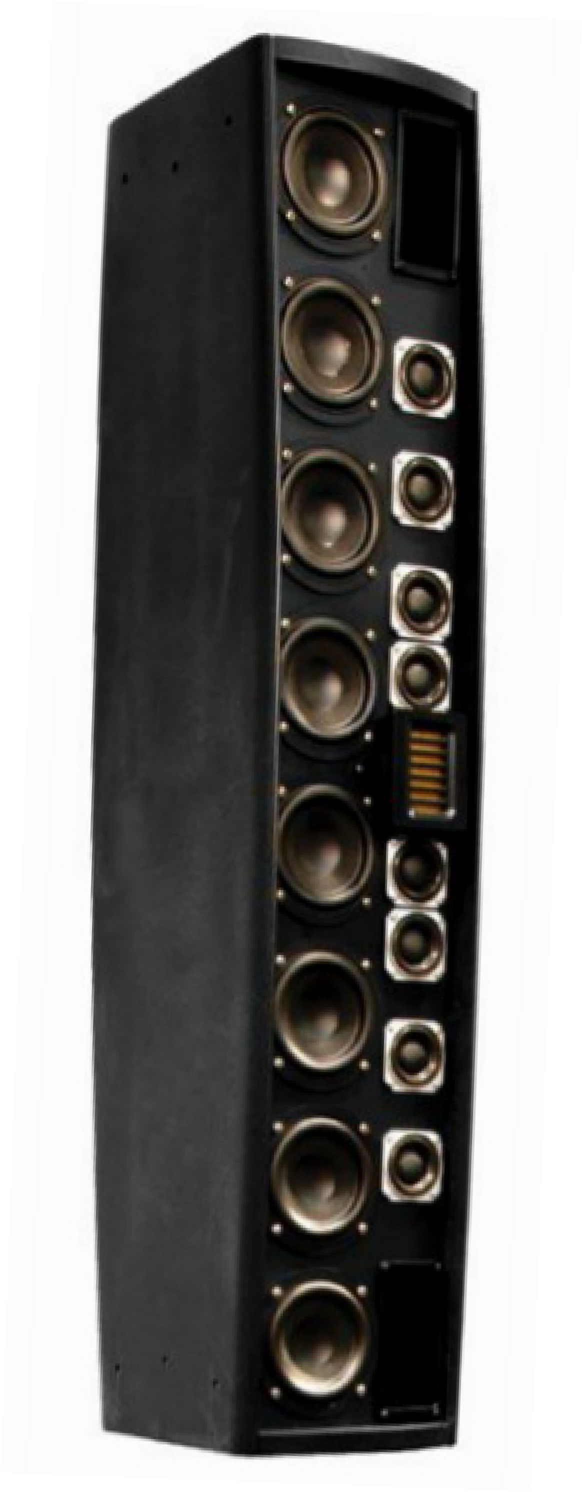


Figure 2-23 – Symmetry = Vertical

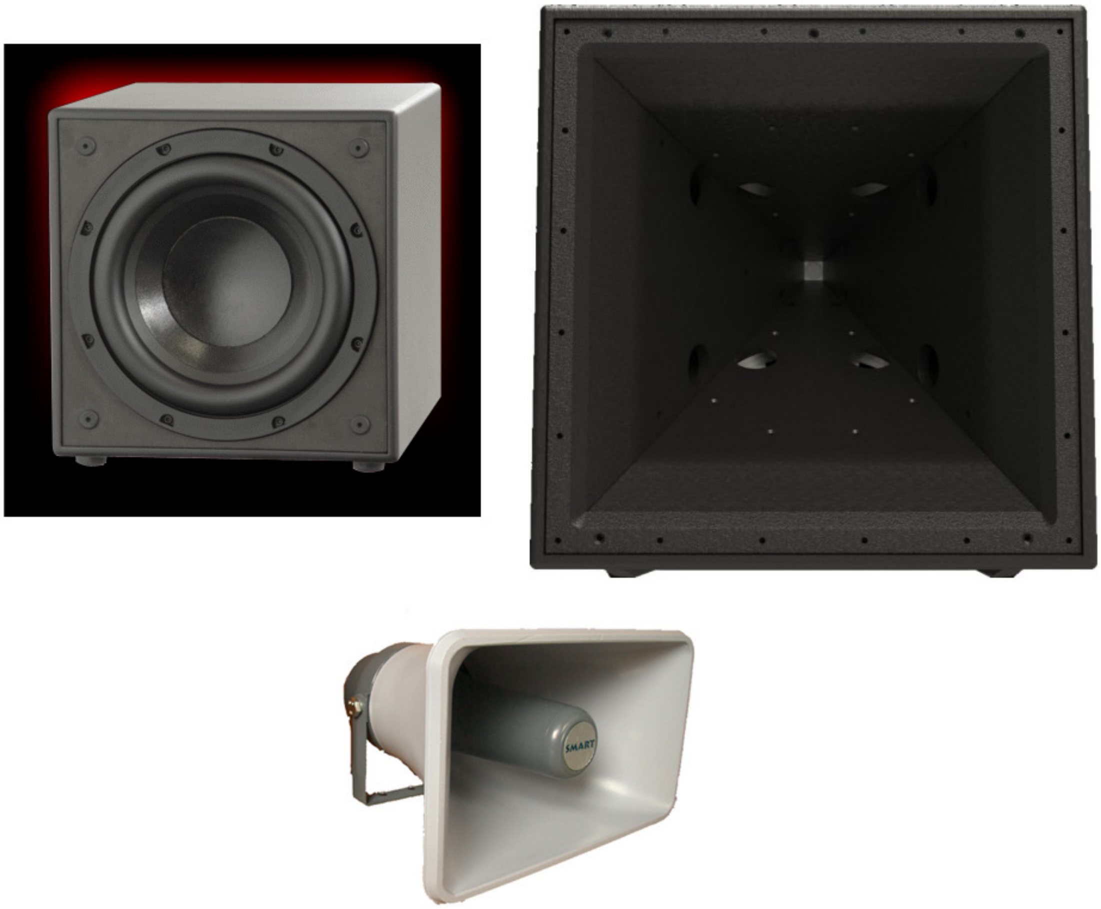


Figure 2-24 – Symmetry = Quarter

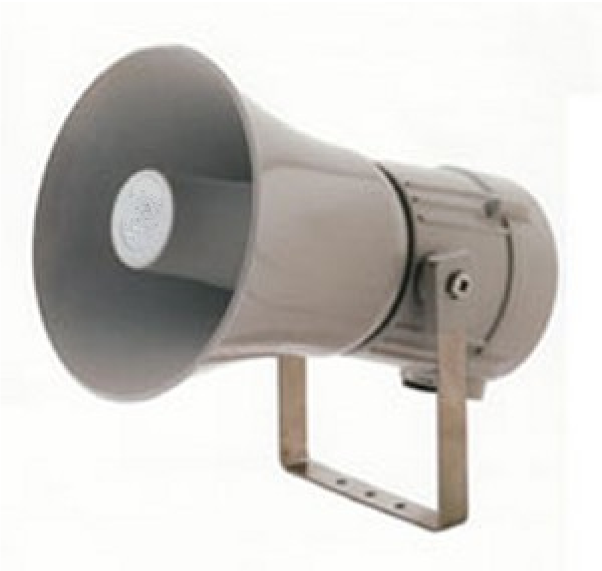


Figure 2-25 – Symmetry = Axial

2.7.2.3 Redundant On-Axis Renormalization

As a result of the coordinate system used in ***SpeakerLab*** the Parallel angle for the on-axis direction will always be 0 and the Parallel angle for the back of the source will always be 180. Because of this there are two redundant measurement points for every Meridian; the on-axis point in front of the source and directly behind the source. The on-axis redundant point can be used as a reference to correct deviations that can occur between measurements due to the drift of the environment conditions during the measurement process. THE USE REDUNDANT ON-AXIS MEASUREMENTS FOR RE-NORMALIZATION checkbox will do this automatically as the measurement files are imported. It is recommended to always select this checkbox.

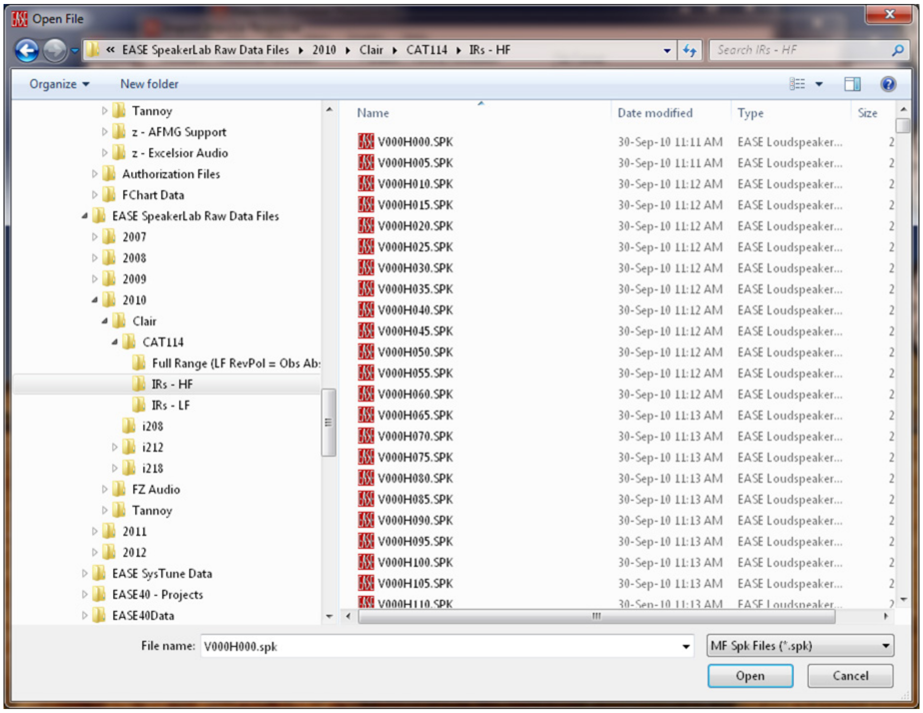
If this option is switched off all directivity balloon data points are normalized to the single on-axis data point measured in the first Meridian. When this option is switched on, all directivity balloon data points in a Meridian are normalized to the on-axis data point measured for that Meridian.

2.7.2.4 Order of Measurements

The ORDER OF MEASUREMENTS can also be specified. The default is LEFT which means the first rotation during the measurements is made so as to measure the parallel locations in the left-half of the horizontal plane. If the first rotation for the measurements is for the right-half of the horizontal plane RIGHT should be selected. Similarly, if the first rotation for the measurements is top-half of the vertical plane TOP should be selected. Finally, if the first rotation for the measurements is bottom-half of the vertical plane BOTTOM should be selected. Choosing an Order of Measurement that is different from the way the measurements were actually made will have the effect of rotating the directivity balloon around the reference axis (-y-axis) in ***SpeakerLab***.

2.7.2.5 Selecting the Files for Import

All of the measured impulse response files should be located in a single folder. Clicking the SELECT PATH button will display an OPEN FILE dialog. Choosing any IR file in the folder where all of the measurement files are located will select the source folder for importing the data files.



Name	Date modified	Type	Size
V000H000.SPK	30-Sep-10 11:11 AM	EASE Loudspeaker...	2
V000H005.SPK	30-Sep-10 11:11 AM	EASE Loudspeaker...	2
V000H010.SPK	30-Sep-10 11:12 AM	EASE Loudspeaker...	2
V000H015.SPK	30-Sep-10 11:12 AM	EASE Loudspeaker...	2
V000H020.SPK	30-Sep-10 11:12 AM	EASE Loudspeaker...	2
V000H025.SPK	30-Sep-10 11:12 AM	EASE Loudspeaker...	2
V000H030.SPK	30-Sep-10 11:12 AM	EASE Loudspeaker...	2
V000H035.SPK	30-Sep-10 11:12 AM	EASE Loudspeaker...	2
V000H040.SPK	30-Sep-10 11:12 AM	EASE Loudspeaker...	2
V000H045.SPK	30-Sep-10 11:12 AM	EASE Loudspeaker...	2
V000H050.SPK	30-Sep-10 11:12 AM	EASE Loudspeaker...	2
V000H055.SPK	30-Sep-10 11:12 AM	EASE Loudspeaker...	2
V000H060.SPK	30-Sep-10 11:12 AM	EASE Loudspeaker...	2
V000H065.SPK	30-Sep-10 11:13 AM	EASE Loudspeaker...	2
V000H070.SPK	30-Sep-10 11:13 AM	EASE Loudspeaker...	2
V000H075.SPK	30-Sep-10 11:13 AM	EASE Loudspeaker...	2
V000H080.SPK	30-Sep-10 11:13 AM	EASE Loudspeaker...	2
V000H085.SPK	30-Sep-10 11:13 AM	EASE Loudspeaker...	2
V000H090.SPK	30-Sep-10 11:13 AM	EASE Loudspeaker...	2
V000H095.SPK	30-Sep-10 11:13 AM	EASE Loudspeaker...	2
V000H100.SPK	30-Sep-10 11:13 AM	EASE Loudspeaker...	2
V000H105.SPK	30-Sep-10 11:13 AM	EASE Loudspeaker...	2
V000H110.SPK	30-Sep-10 11:13 AM	EASE Loudspeaker...	2

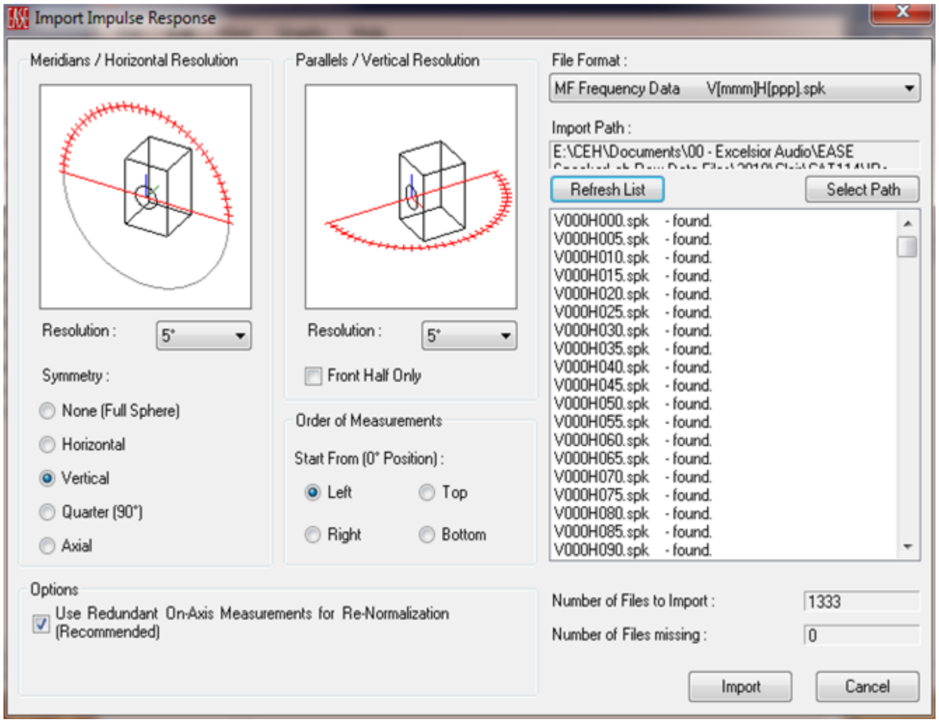
File name: V000H000.spk

MF Spk Files (*.spk)

Figure 2-26 – Select the file in the group to be imported

The software will check all of the files in the specified folder against a list of files that should be present according to all of the Import Impulse Response settings. If any of the required files are missing, the NUMBER OF FILES MISSING will show a number other than 0. The list of all the required files is also shown after the SELECTED PATH has been chosen. Each file will either be shown as found or missing. This can be useful to help understand which files are not in the selected folder or if some of the import settings are incorrect based on the actual measured data and the file names.

If the import settings are modified, files are moved into the selected folder, or files are renamed the required IR data files can be checked again by pressing the REFRESH LIST button.



Meridians / Horizontal Resolution

Resolution : 5°

Symmetry :

○ None (Full Sphere)
○ Horizontal
● Vertical
○ Quarter (90°)
○ Axial

Parallels / Vertical Resolution

Resolution : 5°

□ Front Half Only

Order of Measurements

Start From (0° Position) :

● Left ○ Top
○ Right ○ Bottom

Options

☑ Use Redundant On-Axis Measurements for Re-Normalization (Recommended)

File Format : MF Frequency Data V[mmm]H[ppp].spk

Import Path : E:\CEH\Documents\00 - Excelsior Audio\EASE

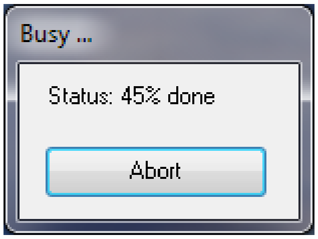
```
V000H000.spk  - found.
V000H005.spk  - found.
V000H010.spk  - found.
V000H015.spk  - found.
V000H020.spk  - found.
V000H025.spk  - found.
V000H030.spk  - found.
V000H035.spk  - found.
V000H040.spk  - found.
V000H045.spk  - found.
V000H050.spk  - found.
V000H055.spk  - found.
V000H060.spk  - found.
V000H065.spk  - found.
V000H070.spk  - found.
V000H075.spk  - found.
V000H080.spk  - found.
V000H085.spk  - found.
V000H090.spk  - found.
```

Number of Files to Import : 1333

Number of Files missing : 0

Figure 2-27 – File selected and all files found

Clicking the IMPORT button will start the import process. This can take a while depending on the number of files to be imported (measurement angular resolution and symmetry) and the number of data points for each measurement file. A progress display with an ABORT button is provided to check and stop the import process if necessary.

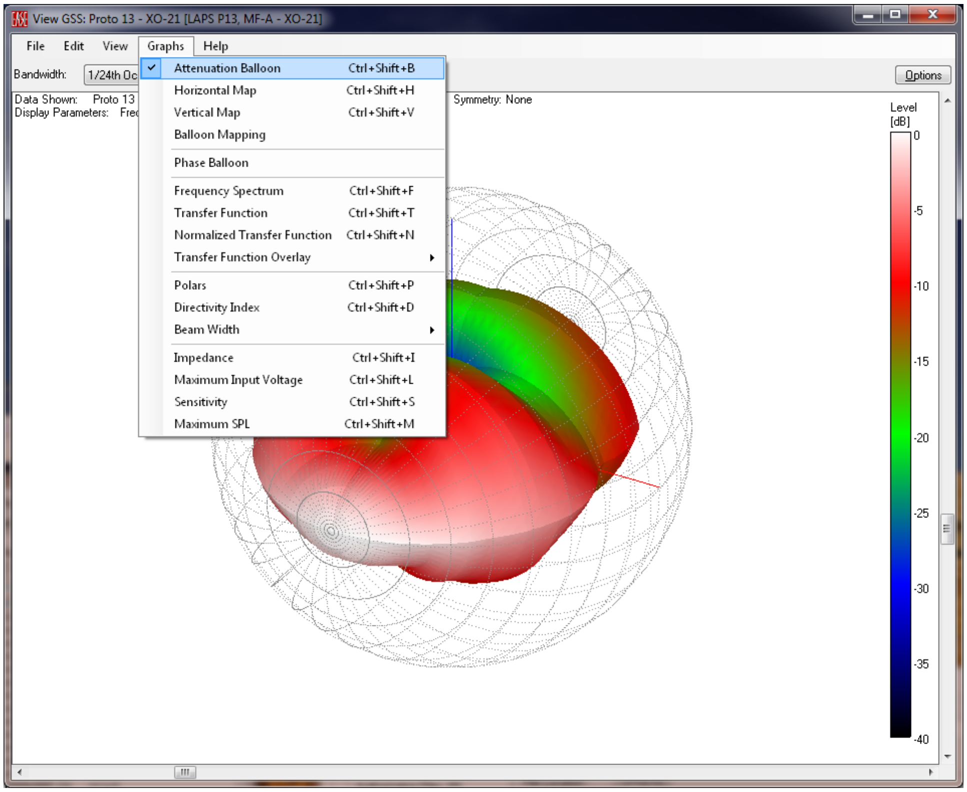


Status: 45% done

Figure 2-28 – Status of import process

2.7.3 Viewing Directivity Data (Brief)

When the import process is complete, the data can be viewed as an attenuation balloon by selecting the GRAPHS > ATTENUATION BALLOON from the menu. Other graphs such as POLARS, BEAMWIDTH, DIRECTIVITY INDEX, HORIZONTAL MAP, or VERTICAL MAP can also be shown by selecting them from GRAPHS on the main menu. Full details on all of the types of graphs and the display options are given in Section 2.7.6 – Viewing the Graphs (Graphs Menu).

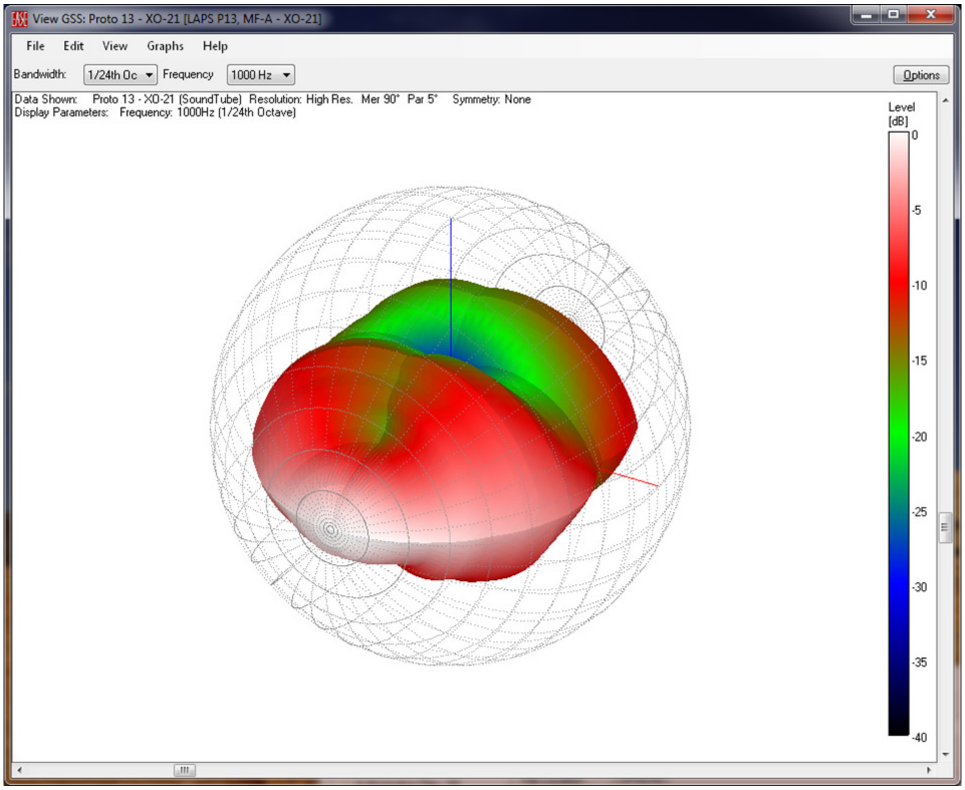

View GSS: Proto 13 - XO-21 [LAPS P13, MF-A - XO-21]

Bandwidth: 1/24th Oc

Data Shown: Proto 13 … Symmetry: None
Display Parameters: Fre…

File
Edit
View
Graphs
Help
✓ Attenuation Balloon Ctrl+Shift+B
Horizontal Map Ctrl+Shift+H
Vertical Map Ctrl+Shift+V
Balloon Mapping
Phase Balloon
Frequency Spectrum Ctrl+Shift+F
Transfer Function Ctrl+Shift+T
Normalized Transfer Function Ctrl+Shift+N
Transfer Function Overlay
Polars Ctrl+Shift+P
Directivity Index Ctrl+Shift+D
Beam Width
Impedance Ctrl+Shift+I
Maximum Input Voltage Ctrl+Shift+L
Sensitivity Ctrl+Shift+S
Maximum SPL Ctrl+Shift+M
Options
Level
[dB]
0
-5
-10
-15
-20
-25
-30
-35
-40

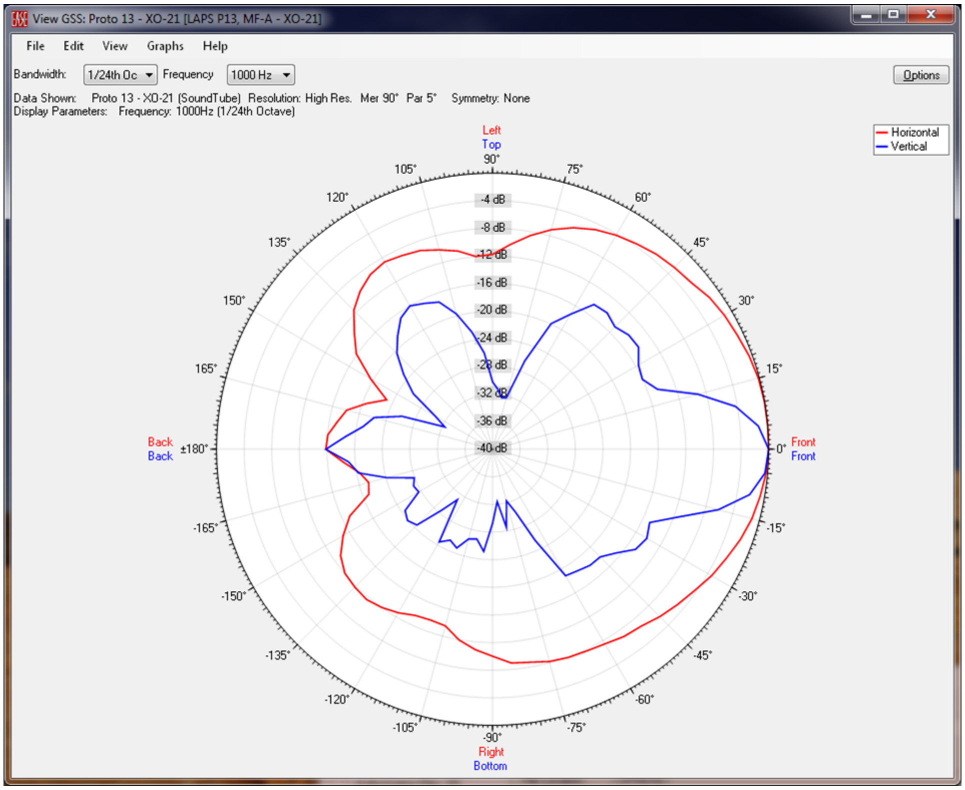
Figure 2-29 – Selection of display graphs in the GSS Editor


View GSS: Proto 13 - XO-21 [LAPS P13, MF-A - XO-21]

Bandwidth: 1/24th Oc Frequency 1000 Hz

Data Shown: Proto 13 - XO-21 (SoundTube) Resolution: High Res. Mer 90° Par 5° Symmetry: None
Display Parameters: Frequency: 1000Hz (1/24th Octave)

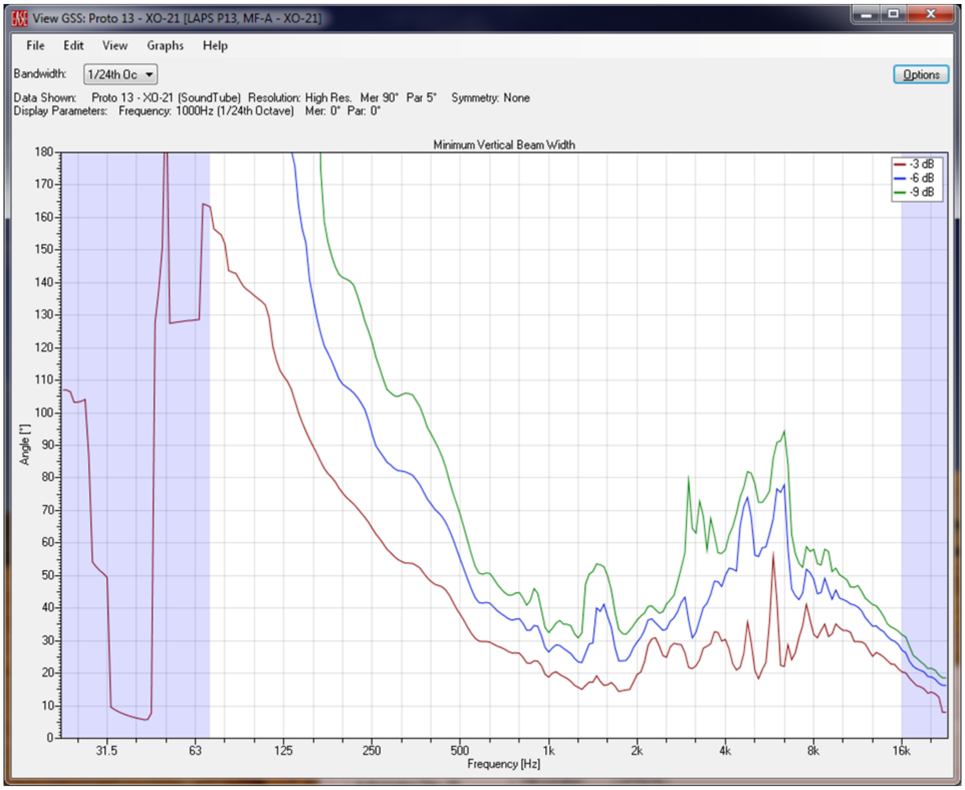
Figure 2-30 – Attenuation balloon


View GSS: Proto 13 - XO-21 [LAPS P13, MF-A - XO-21]

Bandwidth: 1/24th Oc Frequency 1000 Hz

Data Shown: Proto 13 - XO-21 (SoundTube) Resolution: High Res. Mer 90° Par 5° Symmetry: None
Display Parameters: Frequency: 1000Hz (1/24th Octave)

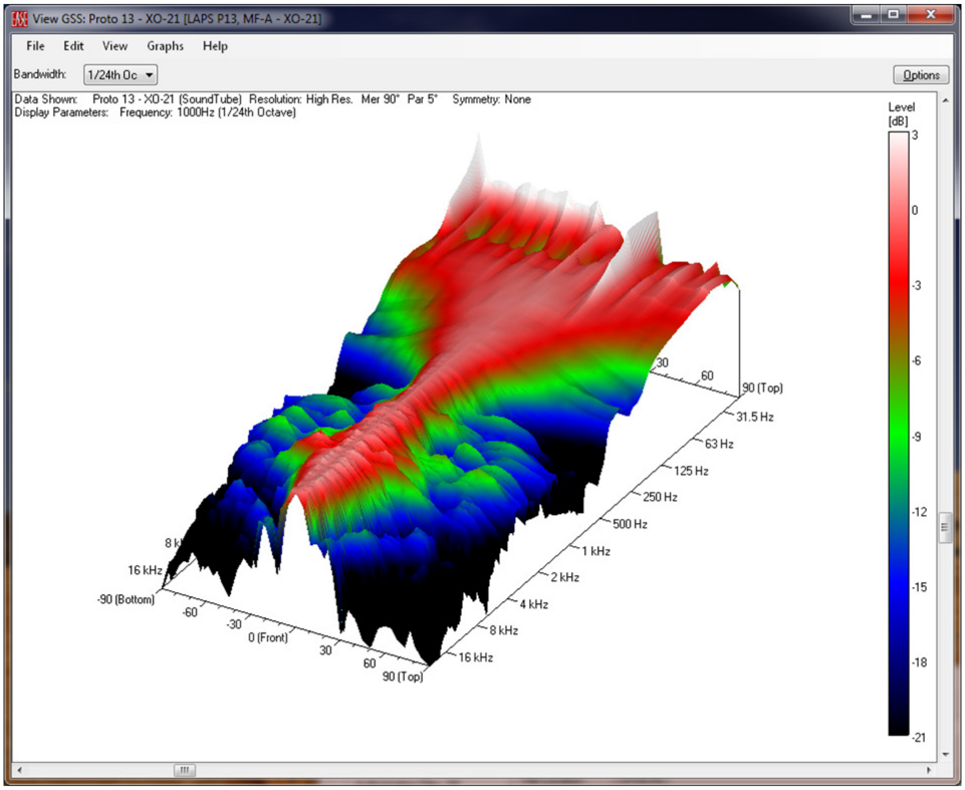
Figure 2-31 – Polar response graph


View GSS: Proto 13 - XO-21 [LAPS P13, MF-A - XO-21]

Bandwidth: 1/24th Oc

Data Shown: Proto 13 - XO-21 (SoundTube) Resolution: High Res. Mer 90° Par 5° Symmetry: None
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-32 – Beamwidth graphs



Data Shown: Proto 13 - XO-21 (SoundTube) Resolution: High Res. Mer 90° Par 5° Symmetry: None
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-33 – Directivity map

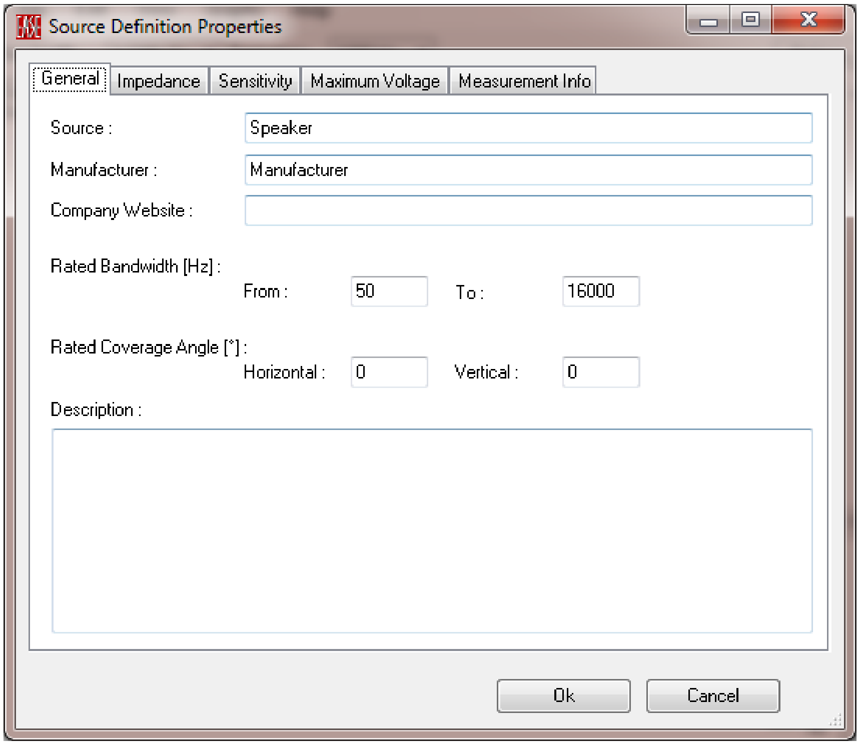
2.7.4 Editing Source Properties

The complete description of the GSS requires entering some other data. This is done in the SOURCE DEFINITIONS PROPERTIES dialog box. The window is accessed by clicking on EDIT > PROPERTIES on the main menu of by pressing F4. This properties dialog box will also be opened automatically if EDIT PROPERTIES is selected in the NEW SOURCE Dialog.

2.7.4.1 General Tab

The GENERAL tab allows entering a name for the source, the manufacturer’s name, and website address into the SOURCE, MANUFACTURER and COMPANY WEBSITE text boxes. These are informational only and have no effect on any of the operational features of the GSS or a GLL in which it is used.

The values entered for the RATED BANDWIDTH are used to limit the frequency range used to calculate the broadband rated sensitivity of the source. The frequency ranges outside of this bandwidth are usually displayed with a light blue background. The RATED COVERAGE ANGLE values are displayed in some applications of the GSS. The DESCRIPTION text box can be used to add information about the GSS source.



Source : Speaker

Manufacturer : Manufacturer

Company Website :

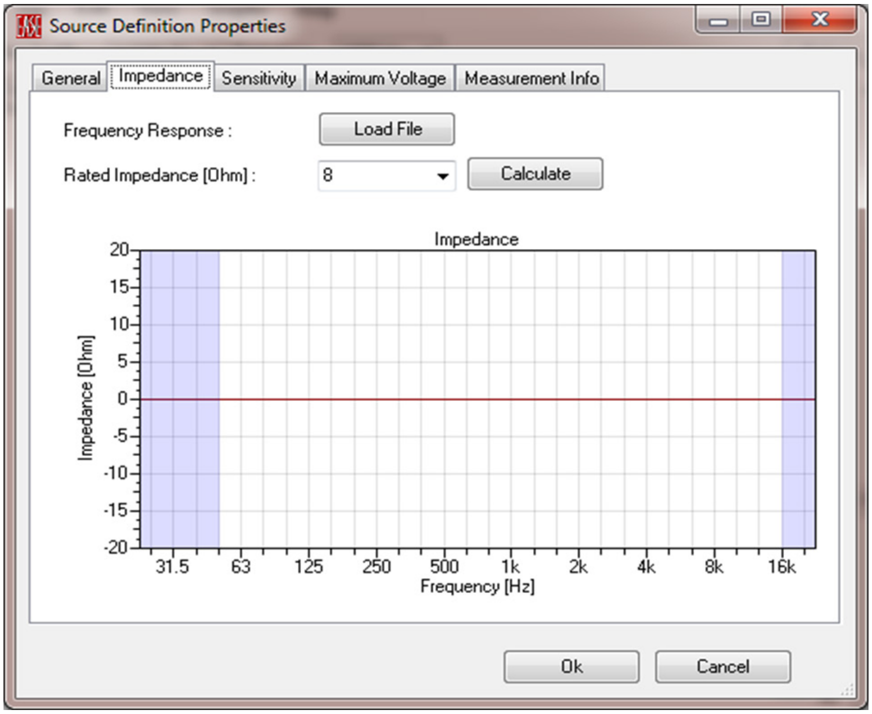
Rated Bandwidth [Hz] :
From : 50 To : 16000

Rated Coverage Angle [°] :
Horizontal : 0 Vertical : 0

Description :

Figure 2-34 – Source Definition Properties General tab

2.7.4.2 Impedance



Frequency Response :

Rated Impedance [Ohm] : 8

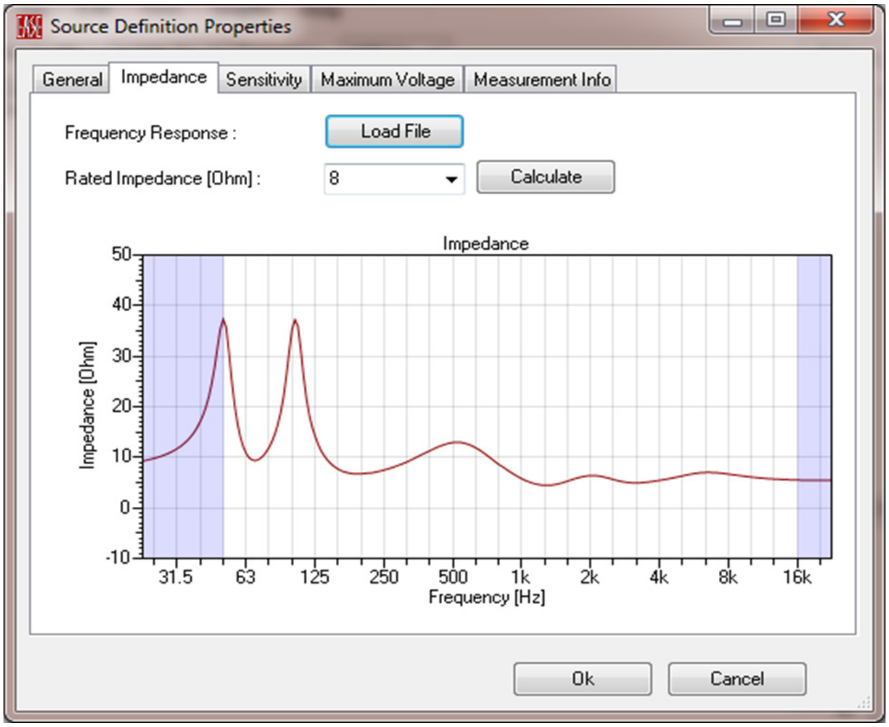
Figure 2-35 – Source Definition Properties Impedance tab

Selecting the IMPEDANCE tab will switch to a display showing an impedance graph. Since we are editing a high resolution source, selecting the LOAD FILE button will show an OPEN FILE dialog box. This can be used to select a file for the measured impedance of the source.

Multiple file formats for several different measurement systems are supported.

- EASERA (.etm) (.efr) and text (.etx) files
- Wave Files (.wav)
- MLSSA (.frq) and (.tim) files
- Monkey Forest (.dat) and (.spk) files
- CLIO time and frequency (.txt) files
- EVI (.pm) files
- TEF Tds (.tds) and Impedance Files (.imp)

After selecting the desired measured impedance file and clicking OK in the OPEN FILE dialog box the graph will show the measured impedance.



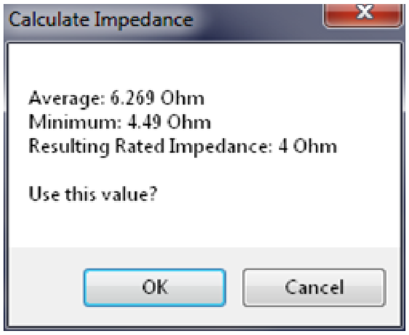
Frequency Response :

Rated Impedance [Ohm] : 8

Figure 2-36 – Source Definition Properties Impedance tab after loading a file

In case of low resolution tabular data, an EDIT TABLE button is provided instead of the LOAD FILE button. This allows the impedance values at fractional octave data points to be entered or edited.

The CALCULATE button opens a message box to display the calculated average and the minimum impedance. A RATED IMPEDANCE is then suggested based on the minimum impedance in compliance with IEC 60268-5 [6]. Clicking OK uses this value for RATE IMPEDANCE.



Average: 6.269 Ohm
Minimum: 4.49 Ohm
Resulting Rated Impedance: 4 Ohm

Use this value?

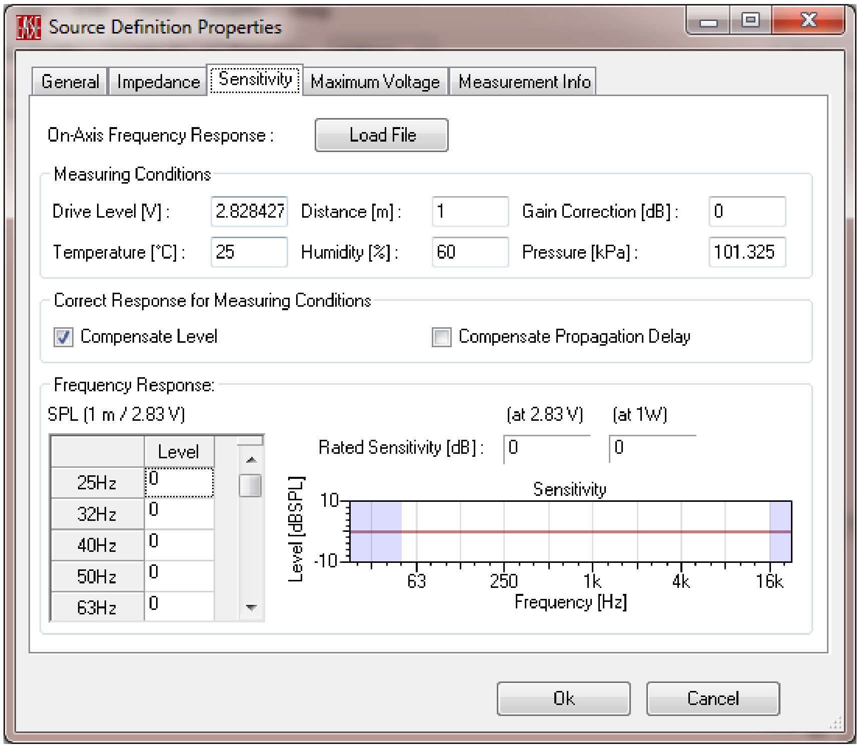
Figure 2-37 – The rated impedance calculated from the imported data file

The RATED IMPEDANCE is for informational purposes only. It is not used in any calculations of a GLL model. The only place the RATED IMPEDANCE value is used is to display an equivalent 1 watt, 1 meter sensitivity number on the SENSITIVITY tab.

The impedance curve itself is only for informational purposes as well.

2.7.4.3 Sensitivity

The SENSITIVITY tab is used to enter data for the ON-AXIS FREQUENCY RESPONSE of the source.



On-Axis Frequency Response :

Measuring Conditions

Drive Level [V] : 2.828427 Distance [m] : 1 Gain Correction [dB] : 0

Temperature [°C] : 25 Humidity [%] : 60 Pressure [kPa] : 101.325

Correct Response for Measuring Conditions

☑ Compensate Level □ Compensate Propagation Delay

Frequency Response:

SPL (1 m / 2.83 V)

	Level
25Hz	0
32Hz	0
40Hz	0
50Hz	0
63Hz	0

	(at 2.83 V)	(at 1W)
Rated Sensitivity [dB] :	0	0

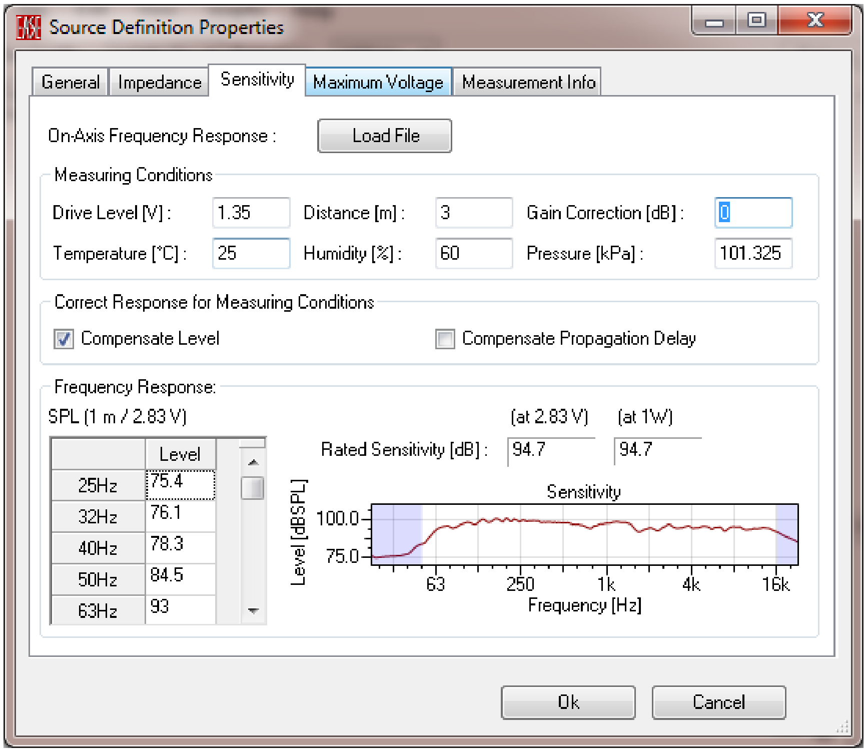
Figure 2-38 – Source Definition Properties Sensitivity tab

Clicking the Load File button will display an Open File dialog box. This can be used to select a file for the measured on-axis response of the source. Again there are multiple file formats supported.

- EASERA (.etm) (.efr) and text (.etx) files
- Wave Files (.wav)
- MLSSA (.frq) and (.tim) files
- LMS Frequency Files (.txt)
- Monkey Forest (.dat) and (.spk) files
- CLIO Time and Frequency (.txt) files
- EVI (.pm) files
- TEF Tds (.tds)

For low resolution data the Edit Table button will open a dialog box for entering or editing the on-axis response curve at fractional octave data points.

Once a file is imported a dialog asks to verify the sensitivity setting. This is a reminder to check that the Rated Sensitivity that is calculated is as expected for the measured source. Click OK to dismiss this box.



On-Axis Frequency Response :

Measuring Conditions

Drive Level [V] : 1.35
Distance [m] : 3
Gain Correction [dB] : 0
Temperature [°C] : 25
Humidity [%] : 60
Pressure [kPa] : 101.325

Correct Response for Measuring Conditions

☑ Compensate Level ☐ Compensate Propagation Delay

Frequency Response:

SPL (1 m / 2.83 V)

	Level
25Hz	75.4
32Hz	76.1
40Hz	78.3
50Hz	84.5
63Hz	93

	(at 2.83 V)	(at 1W)
Rated Sensitivity [dB] :	94.7	94.7

Figure 2-39 – Source Definition Properties Sensitivity tab after entering data

An SPL vs. frequency graph, values calculated at fractional octave intervals (SPL 1m / 2.83 V), and two Rated Sensitivity values (at 2.83 V and 1 W) are shown in the Frequency Response panel. The values calculated here are affected by the values in the Measuring Conditions section, but only if the Compensate Level option is checked in the Correct Response for Measuring Conditions section. The parameters affecting the FREQUENCY RESPONSE section are the DRIVE LEVEL [V], the measuring DISTANCE [M], and the GAIN CORRECTION [DB]. Editing these parameters will result in a change of the FREQUENCY RESPONSE values.

The sensitivity values, along with the maximum voltage PROGRAM VRMS (1MIN) (entered on the MAXIMUM VOLTAGE tab), are used for calculating of the maximum SPL.

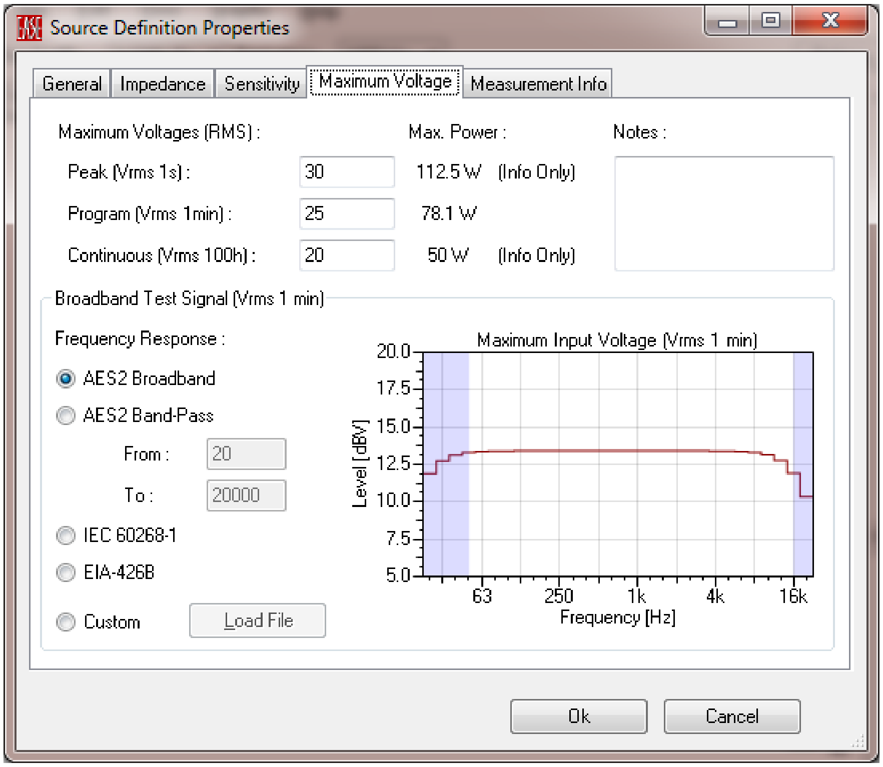
The COMPENSATE PROPAGATION DELAY checkbox will remove the propagation time (time of flight) of the sound from the measured source to the measurement microphone. This will affect the phase response which is not shown in the SENSITIVITY tab. The values affecting the propagation delay are the DISTANCE [M] and the environment conditions TEMPERATURE [°C], HUMIDITY [%], and PRESSURE [KPA]. The environmental conditions are used to calculate the speed of sound when the on-axis response measurement was taken. This is used, along with the DISTANCE, to calculate the time of flight for the sound from the source to the measurement microphone.

It is important to note that the specified DISTANCE should be the distance from the measurement microphone to the same Point of Rotation (POR) used for the measurement of the directivity balloon data. This distance can be different for the on-axis sensitivity measurement and the directivity balloon measurements, but both should use the Point of Rotation as the common reference point.

Note also that all of the directivity balloon measurements imported earlier are normalized to the on-axis response of the imported balloon measurements. As such, omitting the ON-AXIS FREQUENCY RESPONSE data will lead to a GSS with an erroneously flat on-axis frequency response. On the other hand, these clearly separated steps allow you to measure the directional properties at a different time than the sensitivity. This is particular useful if a loudspeaker was already measured in 3D but its equalization was changed later without affecting the radiation patterns.

2.7.4.4 Maximum Voltage

The MAXIMUM VOLTAGE tab allows entering the maximum voltage that can be applied to the source in terms of rms values with three time constants: PEAK VRMS (1S), PROGRAM VRMS (1MIN), and CONTINUOUS VRMS (100H) [6].



Maximum Voltages (RMS) : | Max. Power : | Notes :

Peak (Vrms 1s) : 30 | 112.5 W (Info Only)

Program (Vrms 1min) : 25 | 78.1 W

Continuous (Vrms 100h) : 20 | 50 W (Info Only)

Broadband Test Signal (Vrms 1 min)

Frequency Response :

● AES2 Broadband

○ AES2 Band-Pass

From : 20

To : 20000

○ IEC 60268-1

○ EIA-426B

○ Custom

Figure 2-40 – Source Definition Properties Maximum Voltage tab

Typically testing to determine the maximum input voltage (MIV) is done with signals that have either a 6 dB crest factor or a 12 dB crest factor. The value of the MIV that is used in ***SpeakerLab*** should be based only on the thermal considerations (rms values) of the source and not its mechanical considerations or limitations (peak values).

Each MAXIMUM POWER value shown will be calculated using the RATED IMPEDANCE and displayed for the respective rms voltages. These power values are not used in any GLL calculations. All maximum SPL calculations for the GLL models will use the imported sensitivity measurement, the spectrum of the input signal selected at the time of the GLL calculation, and the PROGRAM (VRMS 1MIN) value. Since the PEAK VRMS (1S) and CONTINUOUS VRMS (100H) values are not used in any GLL calculations these are shown as (INFO ONLY).

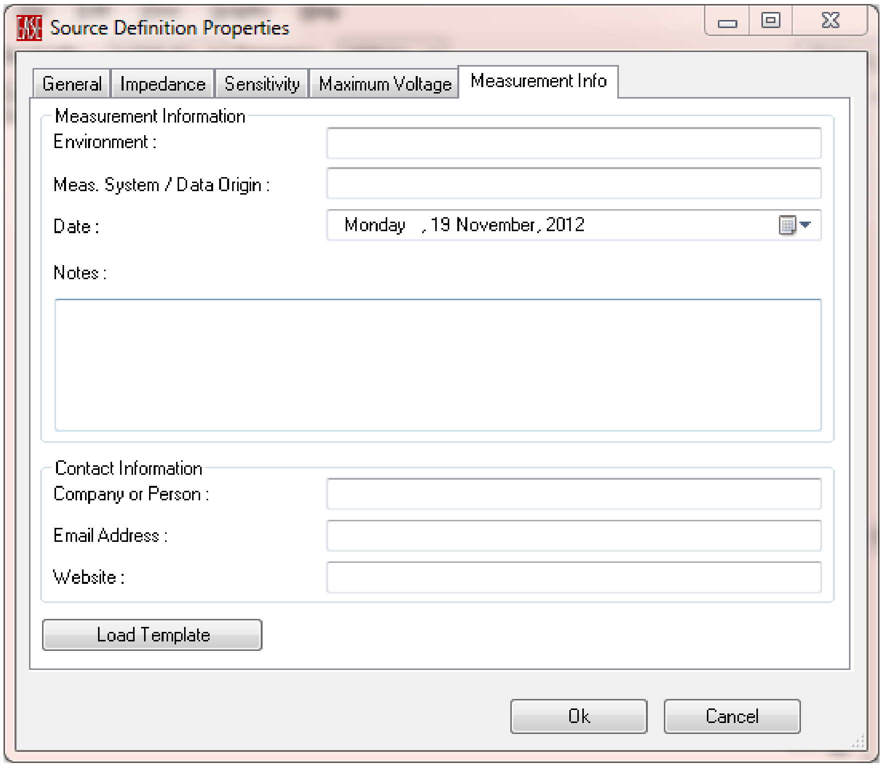
Additional information regarding the voltage measurements can be entered in the NOTES text box.

The Broadband Test Signal (Vrms 1min) section features a set of radio buttons to select spectral content of the test signal that was used to measure the Program (Vrms 1min) voltage. However, this data has only informational character.

The Program (Vrms 1min) voltage value is expected to indicate the maximum input voltage for a source based on thermal limitations. Additional details on how to measure the maximum input voltage for a transducer are given in the section 8.3 – Maximum Input Voltage & SPL Measurement of this manual.

2.7.4.5 Measurement Info

The Measurement Info tab allows additional information on the measurements of the source for the GSS to be included.



Measurement Information

Environment :

Meas. System / Data Origin :

Date : Monday , 19 November, 2012

Notes :

Contact Information

Company or Person :

Email Address :

Website :

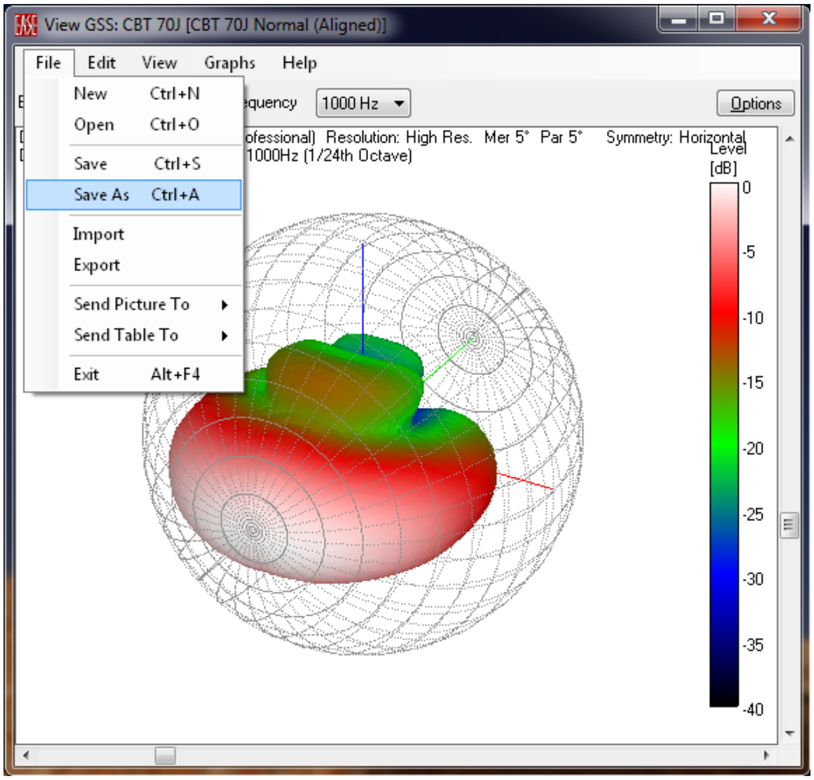
Figure 2-41 – Source Definition Properties Measurement Info tab

The Load Template button allows importing the data from the Measurement Info tab of a saved GSS file. This is particularly useful when multiple GSS files are to be created and they all share the same or similar informational data.

Clicking the OK button closes the Source Definition Properties dialog and switches back to the GSS Source Editor.

2.7.5 Saving & Opening GSS Files

At this point the GSS source is complete (Balloon Data and Properties) and can be saved using FILE > SAVE or FILE > SAVE AS from the main menu.


View GSS: CBT 70J [CBT 70J Normal (Aligned)]

ofessional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
1000Hz (1/24th Octave)

Figure 2-42 – Saving the GSS file

If the SAVE command is selected but the data has not been saved to a file location before, a SAVE SOURCE DEFINITION dialog box will open. This will allow the selection of a destination folder and file name. This same dialog box will open when SAVE AS is selected.

To work with a previously saved GSS file click on FILE > OPEN item from the main menu to select the file with the OPEN SOURCE DEFINITION dialog box.

2.7.6 Viewing the Graphs (Graphs Menu)

Once all of the information for the GSS has been entered it is possible to view the data for the source using the GRAPHS selection on the main menu. This will display a sub-menu showing the different graphs that can be selected.

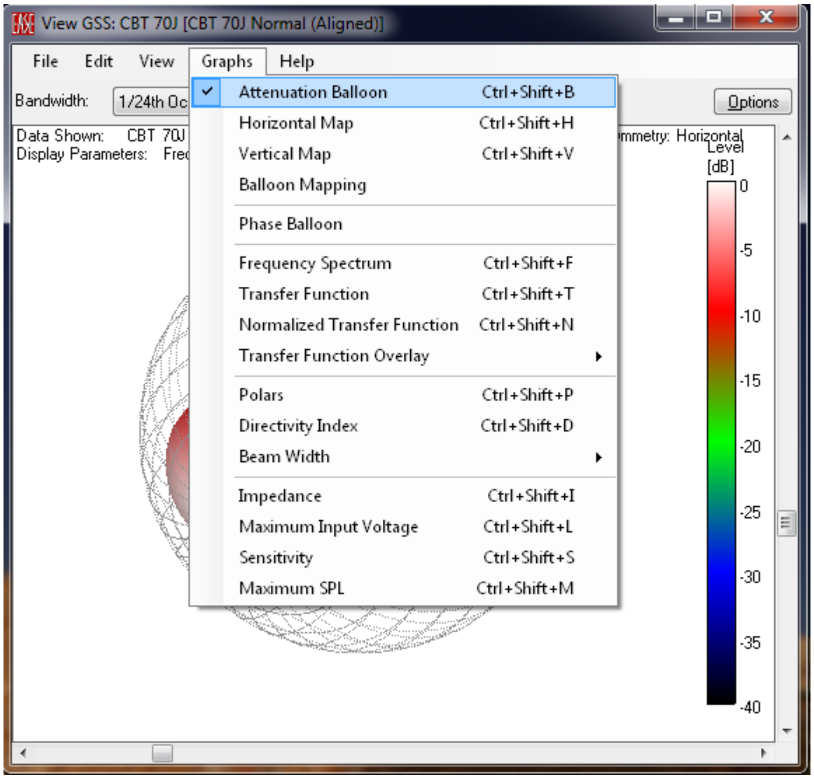
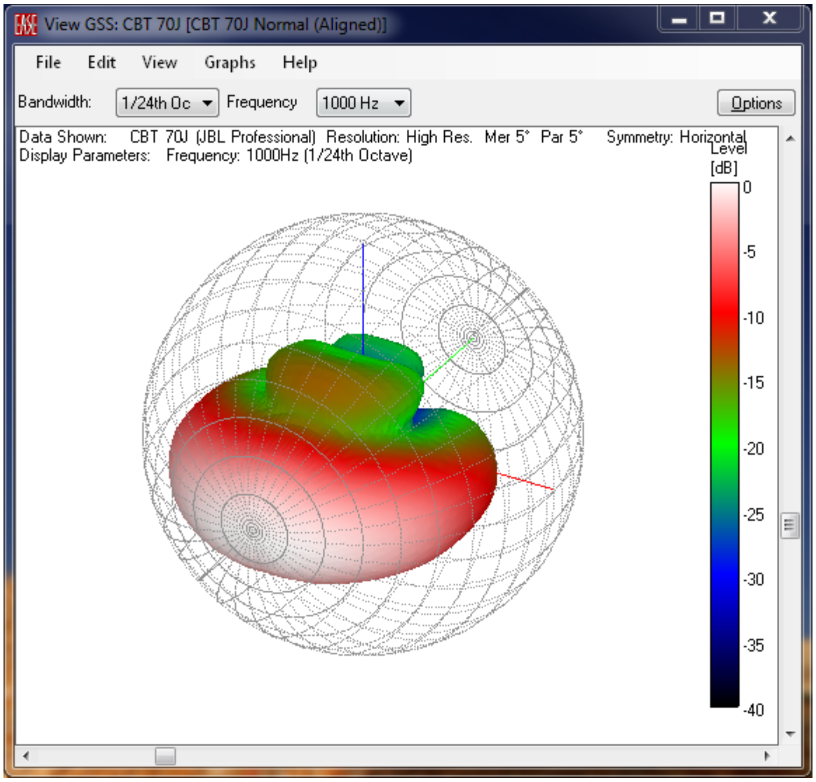

View GSS: CBT 70J [CBT 70J Normal (Aligned)]
File
Edit
View
Graphs
Help
Bandwidth:
1/24th Oc
Options
Data Shown: CBT 70J
Display Parameters: Fre
mmetry: Horizontal
Level
[dB]
0
-5
-10
-15
-20
-25
-30
-35
-40
Attenuation Balloon Ctrl+Shift+B
Horizontal Map Ctrl+Shift+H
Vertical Map Ctrl+Shift+V
Balloon Mapping
Phase Balloon
Frequency Spectrum Ctrl+Shift+F
Transfer Function Ctrl+Shift+T
Normalized Transfer Function Ctrl+Shift+N
Transfer Function Overlay
Polars Ctrl+Shift+P
Directivity Index Ctrl+Shift+D
Beam Width
Impedance Ctrl+Shift+I
Maximum Input Voltage Ctrl+Shift+L
Sensitivity Ctrl+Shift+S
Maximum SPL Ctrl+Shift+M

Figure 2-43 – Graphs available in Source Editor

2.7.6.1 Attenuation Balloon Graph

The Attenuation Balloon shows a 3D balloon denoting level attenuation with respect to the on-axis level at the frequency displayed. This is done since all of the directivity data is normalized to the on-axis SPL. The 3D balloon can be rotated and drawn in different modes using the View menu item. The standard view is the "Colored Contours".

The Attenuation Balloon is shown for the Frequency and Bandwidth selected by the dropdown lists at the top of the window. The Bandwidth selections are 1/24, 1/12, 1/6, 1/3 and 1/1 fractional octaves. This is the bandwidth over which the displayed response is smoothed prior to being rendered. The Frequency selections will change based on the selected Bandwidth. The Frequency selection available will be at the center frequencies for the selected Bandwidth.



View GSS: CBT 70J [CBT 70J Normal (Aligned)]

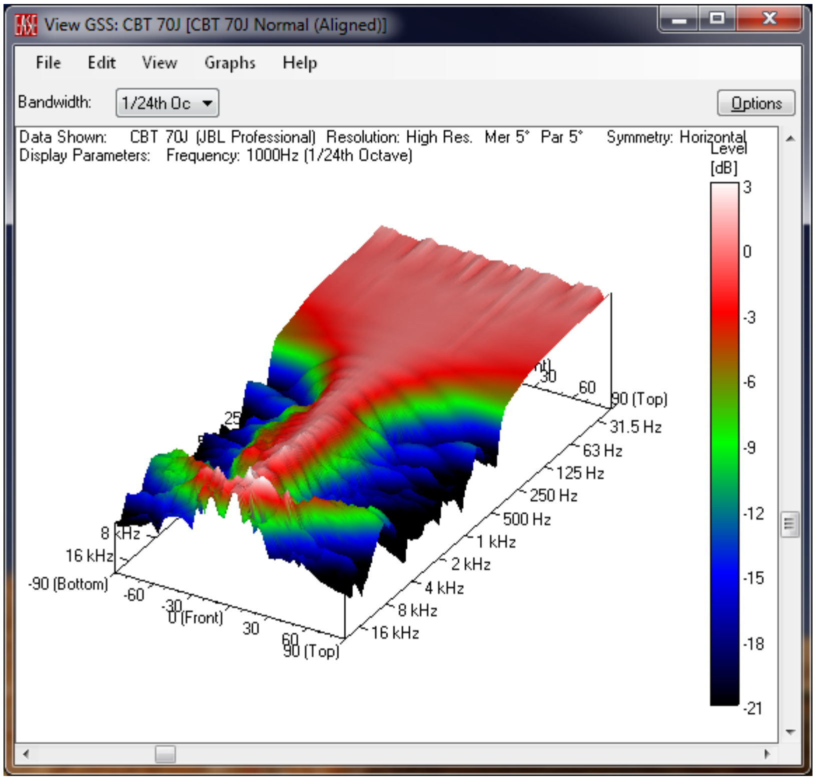
Bandwidth: 1/24th Oc
Frequency 1000 Hz

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-44 – Attenuation balloon

2.7.6.2 Horizontal & Vertical Directivity Maps Graph

The HORIZONTAL MAP and VERTICAL MAP are color directivity maps showing color-coded attenuation level as a function of both frequency and off-axis angle. One of the great advantages of the directivity map display is that the directivity is shown at all frequencies. As such there is no FREQUENCY selection. The BANDWIDTH can be selected to give different smoothing to the display.



View GSS: CBT 70J [CBT 70J Normal (Aligned)]

Bandwidth: 1/24th Oc

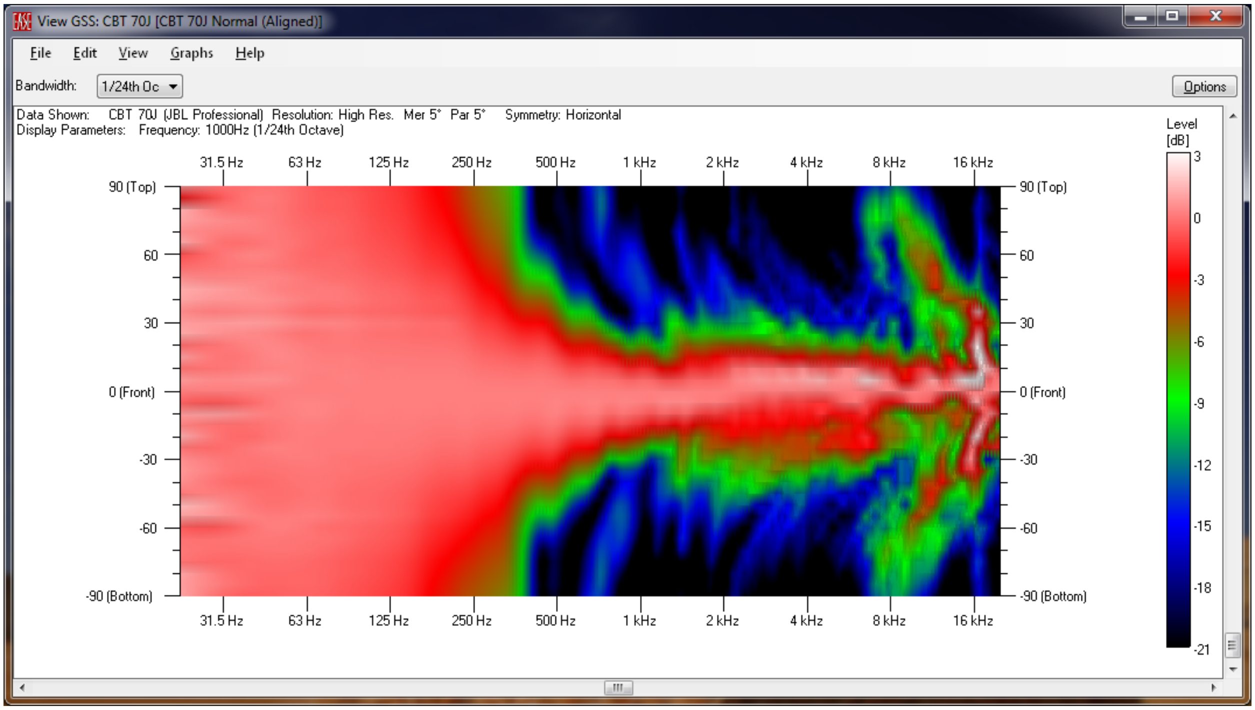
Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-45 – Directivity map

It can be useful to change the viewing perspective to that of an overhead view, looking down on the directivity map, as seen in Figure 2-46. From this view all of the detail to the left and right (or top and bottom) of the source can be seen.

Some additional explanation of the directivity map may be helpful for those who are not familiar with it. Basically, it is a display of the off-axis frequency response of the sound source. All of the off-axis responses have been normalized to the on-axis response. This way, only differences between the on-axis and off-axis response are shown.

If a small left-to-right slice of the directivity map was taken, the result would show the normalized frequency response for the off-axis angle at which the slice was made. Similarly, if a small top-to-bottom slice of the directivity map was taken, the result would show the typical polar response graph for the frequency at which the slice was taken (Figure 2-47).



View GSS: CBT 70J [CBT 70J Normal (Aligned)]

Bandwidth: 1/24th Oc

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-46 – Overhead view of the directivity map

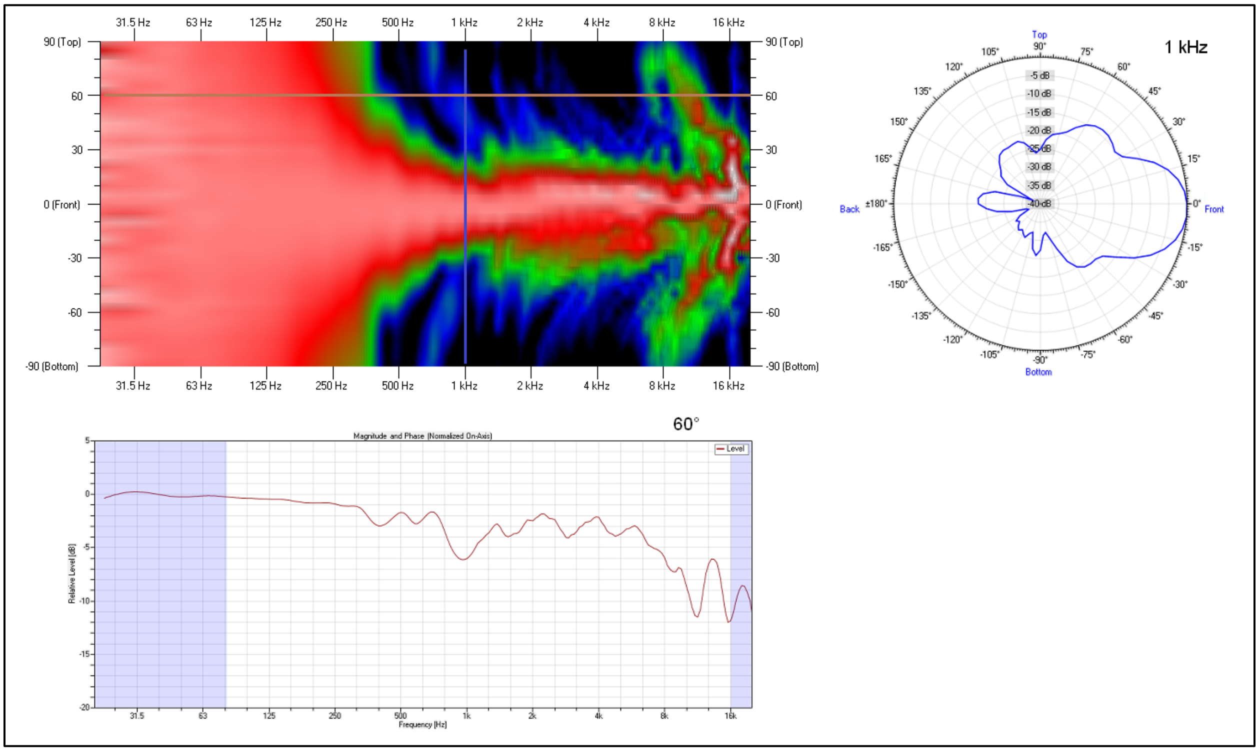


Figure 2-47 – Deciphering the information in the directivity map

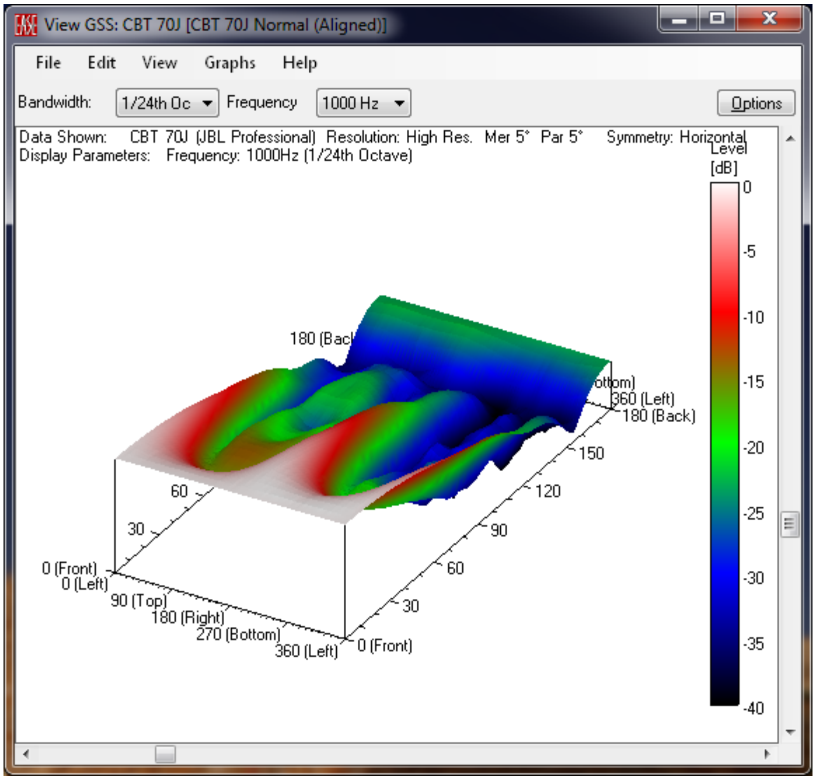
It may be useful to alter the upper and lower limits for the colors denoting level in the directivity maps. Doing this is detailed in section 2.7.7.11 – Directivity Maps Tab. It is not uncommon for an off-axis angle to have slightly higher output than on-axis. Setting the upper limit to +3 dB (instead of the default 0 dB) will help to show these higher output levels. Levels that are higher than specified upper limit relative to on-axis will be shown as grey. The darker the grey appears, the higher the level.

Setting the lower limit to a slightly higher level than the default may allow for better resolution of the displayed level differences. Setting this to -21 dB is useful for many applications. When increased levels of off-axis attenuation need to be monitored this may need to be set to a lower value. One particular advantage of setting the limits at +3 and -21 dB is that the transition from red to green occurs at -6 dB. This makes it very easy to see the -6 dB beamwidth of the source on the directivity map.

2.7.6.3 Balloon Mapping Graph

The BALLOON MAPPING graph is similar to ATTENUATION BALLOON except that it is a projection onto a rectangular surface. This is similar to the way the map on a globe showing the Earth might be unwrapped from the globe and projected onto a flat piece of paper. However, the attenuation level is still shown as a 3D surface plot in addition to being denoted by color. Both the FREQUENCY and BANDWIDTH can be selected for the graph.

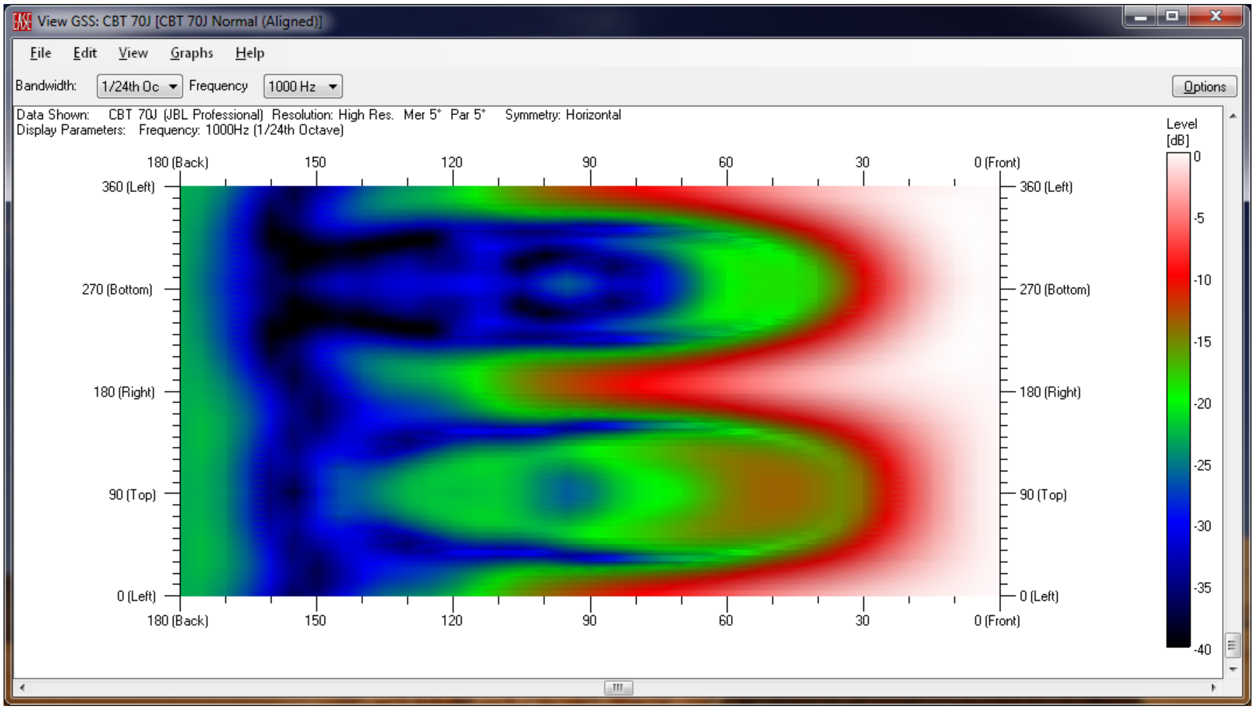
When the graph is oriented as shown in Figure 2-49 the horizontal axis denotes the front-to-back angles (Parallels). The vertical axis denotes the angle of the plane (Meridian) in which the Parallel resides. Thus a horizontal slice through the graph at the 90 (Top) on the vertical axis would give the levels of the top half of the vertical polar response. The bottom half of the vertical polar response would come from a horizontal slice at 270 (Bottom) on the vertical axis. Likewise, horizontal slices at 0 (Left) and 180 (Right) would give the values for the horizontal polar response.



Bandwidth: 1/24th Oc Frequency 1000 Hz

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-48 – Isometric view of the balloon map



Bandwidth: 1/24th Oc Frequency 1000 Hz

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/24th Octave)

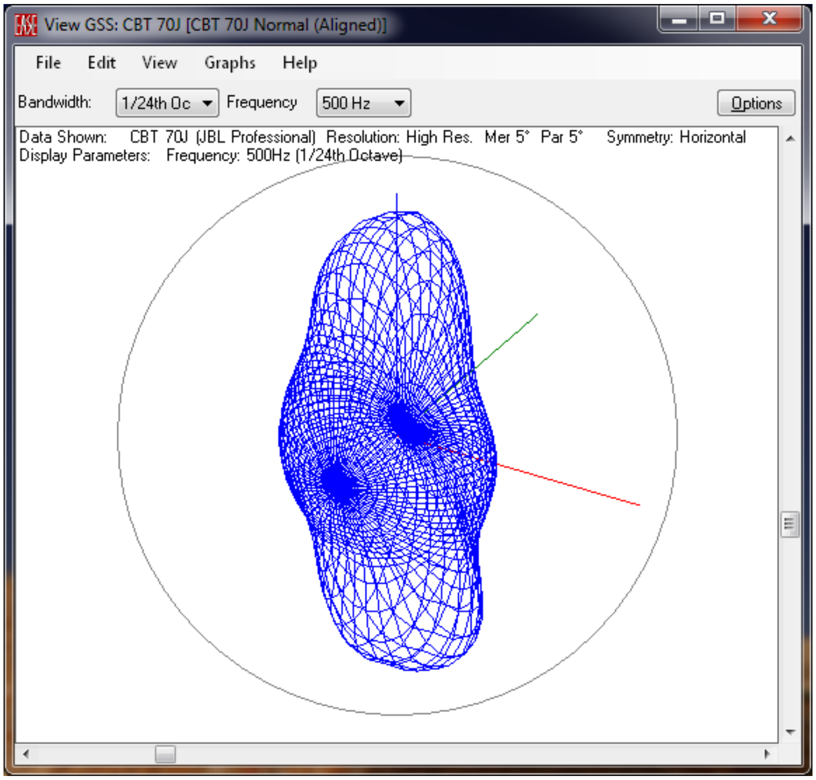
Figure 2-49 – Overhead view of the balloon map

2.7.6.4 Phase Balloon Graph

The PHASE BALLOON graph can be somewhat difficult to try to interpret. Just like the ATTENUATION BALLOON displays the normalized attenuation level of the off-axis sound radiated from the source, the PHASE BALLOON displays the normalized phase of the off-axis sound radiated from the source. The display is not of continuous phase (unwrapped) phase which is why the graph can look so strange. Whenever there is a phase "wrap" from -180° to 180° there is a large discontinuity or jump shown on the graph.

Note that the FREQUENCY can be selected for the graphs. The BANDWIDTH can also be selected but it has no effect on the graph. The phase data are not smoothed or averaged over the selected BANDWIDTH as they are for the other graphs.

Note that at some frequencies the PHASE BALLOON may not appear with wraps and can more easily be interpreted as seen in Figure 2-50. This will be dependent on location of the source that was measured, the POR that was used during the measurement, the accuracy of the measurement, and possibly other factors. In particular, conclusions can be drawn about the location of the source(s) relative to the POR and the symmetry of the arrangement. This is possible because for each direction usually the time of flight is the dominant contribution to the phase balloon.


View GSS: CBT 70J [CBT 70J Normal (Aligned)]

Bandwidth: 1/24th Oc Frequency 500 Hz

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 500Hz (1/24th Octave)

Figure 2-50 – Phase balloon

2.7.6.5 Frequency Spectrum Graph

The FREQUENCY SPECTRUM graph shows the spectral content of the output from the source at the calculated distance using the specified input signal. This is important since an input signal with a different spectral content will result in an output signal with a different spectral content. The output signal is modified by the transfer function of the source for the given radiation angle, and possibly yields a different absolute SPL. The MERIDIAN and PARALLEL for the radiation angle of the source can be selected.

The level of the input signal is at the MAXIMUM INPUT VOLTAGE specified for the source. Similarly, the spectral content of the input signal is the FREQUENCY RESPONSE of the BROADBAND TEST SIGNAL that was specified for the source.

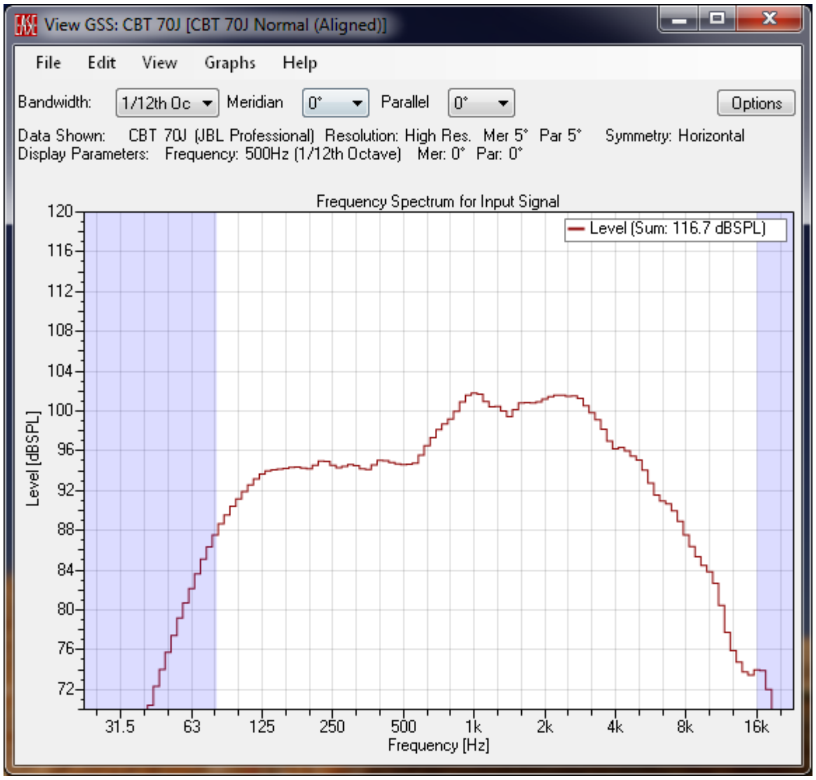
When looking at this same graph for a GLL, instead of a GSS, the spectral content of the input signal is specified prior to the calculations being carried out. The level of the input signal will be the calculated MAXIMUM INPUT VOLTAGE for the entire GLL, not necessarily a single GSS.

For the GSS there is no calculation distance. The frequency response of the source should have been measured in its far-field so that its frequency response does not change with increasing distance. As a result, the displayed level is referenced to 1 meter, as opposed to the actual calculation distance used for a GLL simulation.

The fractional octave BANDWIDTH may also be selected. This is used to determine the intervals over which the energy is summed and displayed for the graph. As larger fractional octave bands are selected the curve will seem to rise on the graph. This is in keeping with how a real sound level meter would respond with the same fractional octave settings. A larger bandwidth (say 1/3 octave instead of 1/12 octave) will contain more energy, and therefore have a greater SPL reading.

In the upper right-hand corner of the graph there is a value given for the total Sum level. Since this is a summation across the entire spectrum it is not affected by the BANDWIDTH setting. This Sum level is what might be expected from a sound level meter set for broadband unweighted readings.

At this point we should explicitly make the distinction between signal-related graphs and transfer function like graphs. Signal-related graphs (Frequency Spectrum, Maximum Input Voltage, and Maximum SPL) show the energy/power content. Because of this the smaller bandwidths are summed together to calculate the energy/power content over the larger bandwidth encompassing these individual smaller bandwidths. Transfer function graphs (Sensitivity, Transfer Function, etc.) do not show energy/power but describe how an input signal passing through a device with a given transfer function is altered to yield the output signal from the device. Transfer function graphs must be averaged, not summed, to calculate their broadband equivalent level.

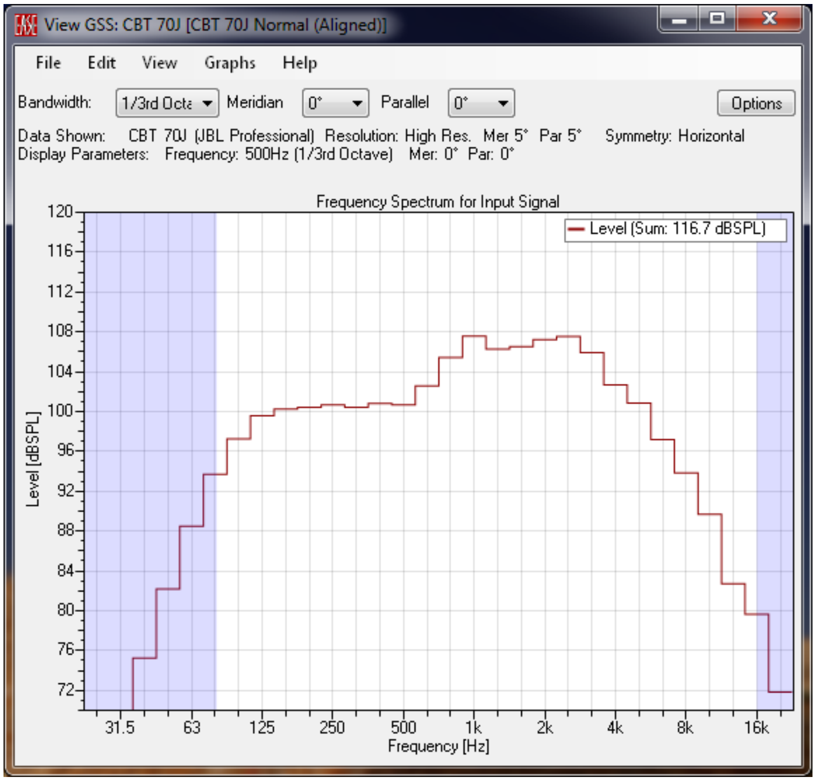


View GSS: CBT 70J [CBT 70J Normal (Aligned)]

Bandwidth: 1/12th Oc Meridian 0° Parallel 0°

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 500Hz (1/12th Octave) Mer: 0° Par: 0°

Figure 2-51 – Frequency spectrum at 1/12 octave



View GSS: CBT 70J [CBT 70J Normal (Aligned)]

Bandwidth: 1/3rd Octa Meridian 0° Parallel 0°

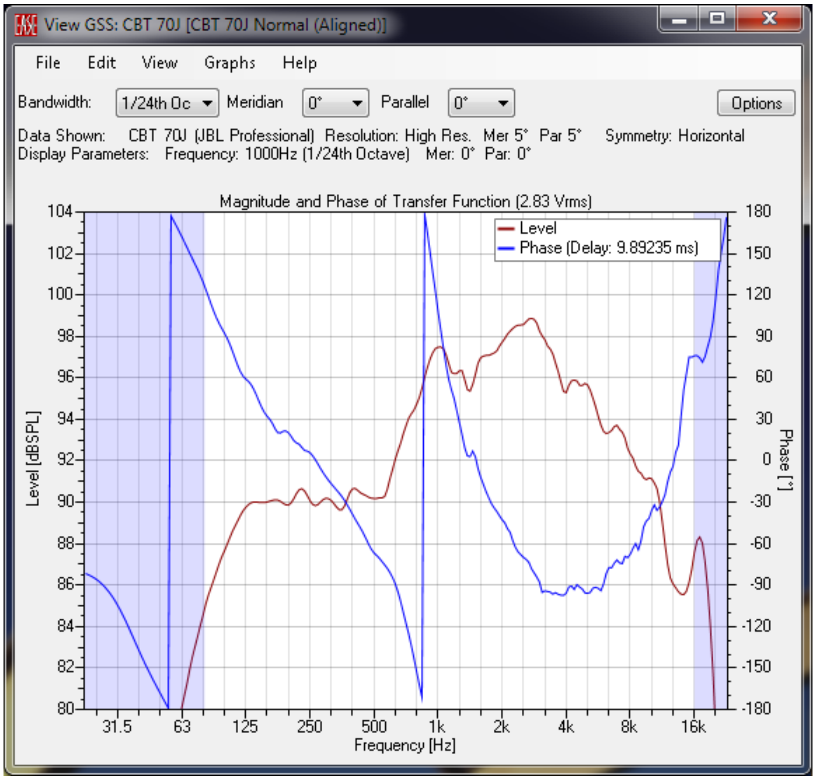
Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 500Hz (1/3rd Octave) Mer: 0° Par: 0°

Figure 2-52 – Frequency spectrum at 1/3 octave

2.7.6.6 Transfer Function Graph

The TRANSFER FUNCTION graph displays the complex transfer function (magnitude and phase) of the source. The level at which this is shown is as if a swept-sine signal with a rms magnitude of 2.83 V was used to excite the source. The SPL is referenced to a distance of 1 meter from the source. The MERIDIAN and PARALLEL for the radiation angle of the source can be selected. The BANDWIDTH can also be selected. This will act to smooth or average the display of magnitude response. The phase response will be unaffected by the BANDWIDTH setting.

The upper right-hand corner of the graph will show a Delay value. This is the time, usually in milliseconds, that was removed from the data to yield the displayed phase response. This is done to minimize "wraps" that may occur (typically in the higher frequency region) in the displayed phase response. If the Compensate Propagation Delay option was not checked, this time represents the amount of time that would need to be removed from the beginning of the measurement to yield a similar phase response on both the measurement system and the TRANSFER FUNCTION graph.


View GSS: CBT 70J [CBT 70J Normal (Aligned)]

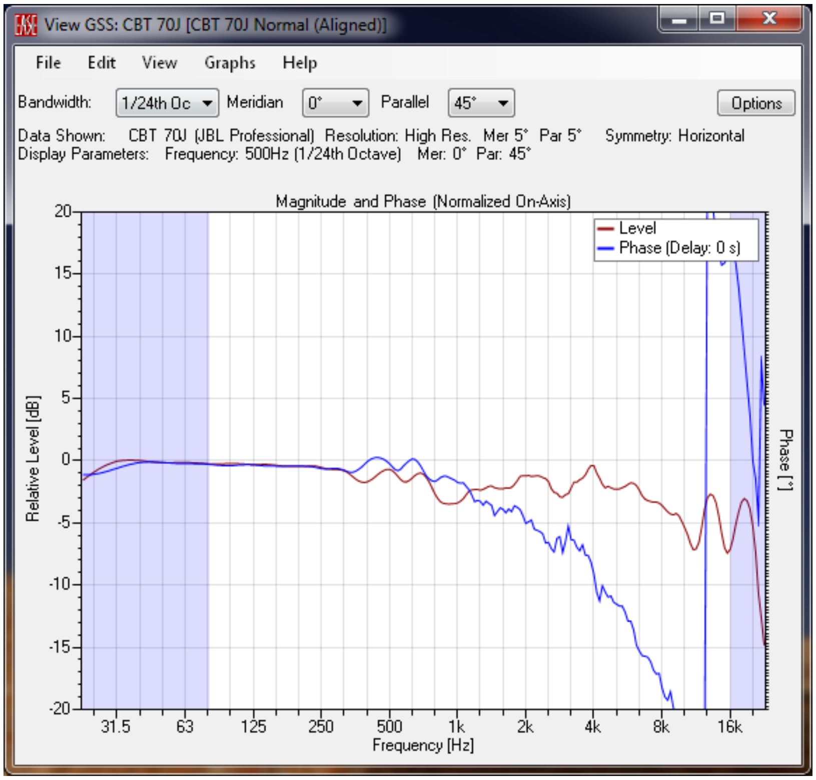
Bandwidth: 1/24th Oc Meridian 0° Parallel 0°

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-53 – Transfer function

2.7.6.7 Normalized Transfer Function Graph

The NORMALIZED TRANSFER FUNCTION graph is similar to the TRANSFER FUNCTION graphs with the exception that the display is normalized to the on-axis transfer function of the source. Thus both the magnitude and phase response displays will only show differences relative to the on-axis response.



View GSS: CBT 70J [CBT 70J Normal (Aligned)]

Bandwidth: 1/24th Oc Meridian 0° Parallel 45°

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 500Hz (1/24th Octave) Mer: 0° Par: 45°

Figure 2-54 – Normalized transfer function

2.7.6.8 Transfer Function Overlay Graph

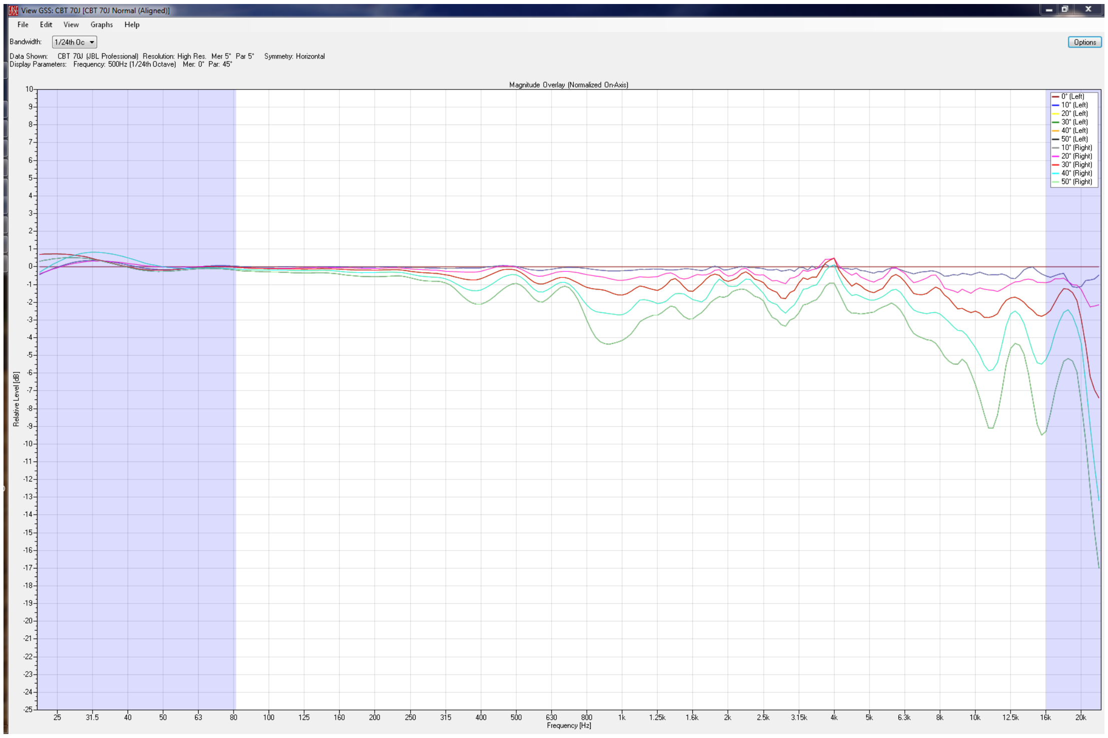
The TRANSFER FUNCTION OVERLAY graphs will display the magnitude response of the multiple off-axis transfer functions normalized to the on-axis magnitude response. This can be very useful to quickly see how well behaved the off-axis response of a source is.

Upon clicking on GRAPHS > TRANSFER FUNCTION OVERLAY there are seven subsequent selections.

- LEFT
- RIGHT
- HORIZONTAL
- TOP
- BOTTOM
- VERTICAL
- USER DEFINED

Each of these will display different data. The LEFT and RIGHT selection will show the data for the left (Meridian = 0°) or right (Meridian = 180°) half-plane of the horizontal polar data. The HORIZONTAL selection will show both the left and right half-planes. This is useful as a comparison of symmetry, provided that both sides of the source were actually measured and not mirrored using one of the SYMMETRY options when importing the data. The TOP (Meridian=90°), BOTTOM (Meridian=270°), and VERTICAL work in a similar manner but displaying the vertical polar data.

The USER DEFINED selection allows the custom selection of the Meridian and Parallel angles for each of the normalized transfer functions in the overlay. The Meridian and Parallel angles a selected in the VIEW OPTIONS dialog box. See section 2.7.7.13 – Transfer Function Overlay Tab for more details on this.



View GSS: CBT 70J [CBT 70J Normal (Aligned)]

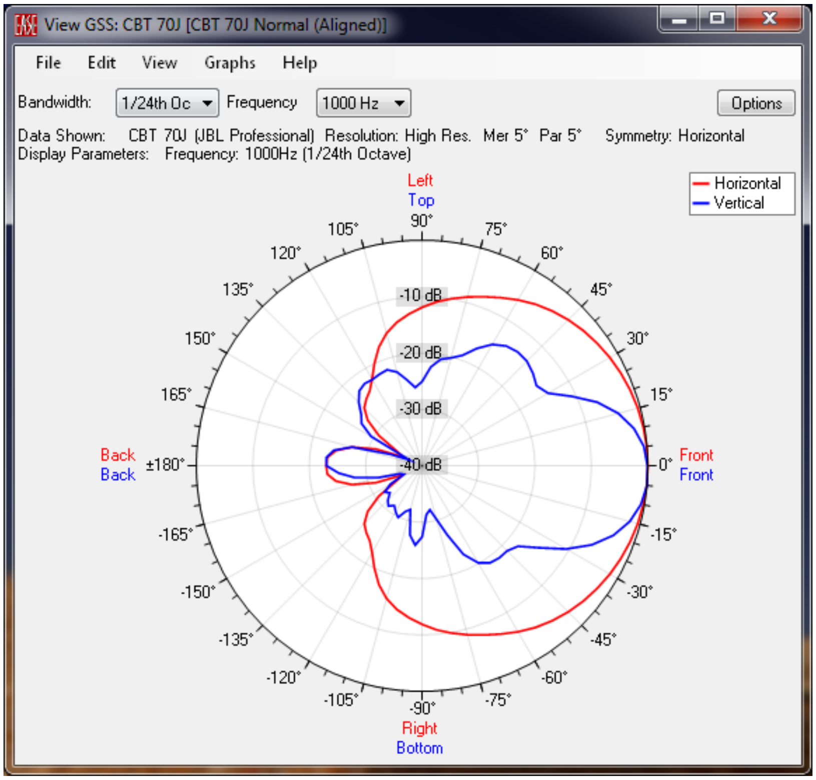
Bandwidth: 1/24th Oc

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 500Hz (1/24th Octave) Mer: 0° Par: 45°

Figure 2-55 – Normalized transfer function overlay

2.7.6.9 Polars Graph

The POLARS graph is a conventional display of the SPL radiated by the source as a function of angle in the horizontal and vertical planes. The horizontal and vertical plots are each normalized, separately, to the highest level within that plane. Both the FREQUENCY and BANDWIDTH can be selected for the graph.



View GSS: CBT 70J [CBT 70J Normal (Aligned)]

Bandwidth: 1/24th Oc Frequency 1000 Hz

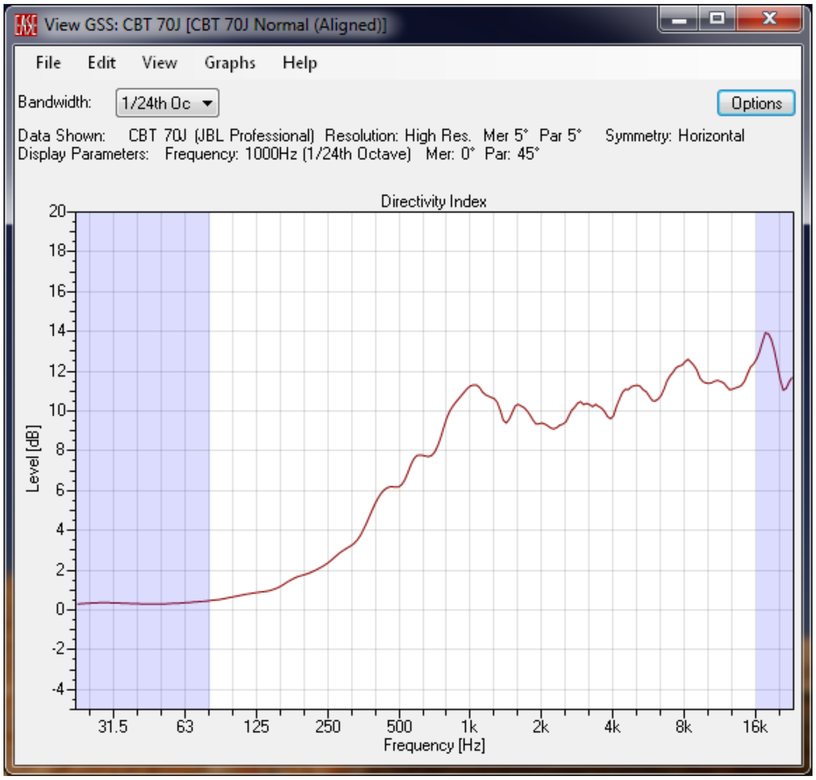
Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-56 – Polar graph

2.7.6.10 Directivity Index Graph

The Directivity Index graph shows the directivity index of the source as a function of frequency. Here the BANDWIDTH can be selected to provide smoothing or averaging of the displayed data.

Directivity Index is a metric of how directional a sound source is in the way it radiates sound. The higher the value of the directivity index is the more directional the source is at radiating its sound. The directivity index is the logarithmic representation (expressed in dB) of the directivity factor, or Q.



View GSS: CBT 70J [CBT 70J Normal (Aligned)]

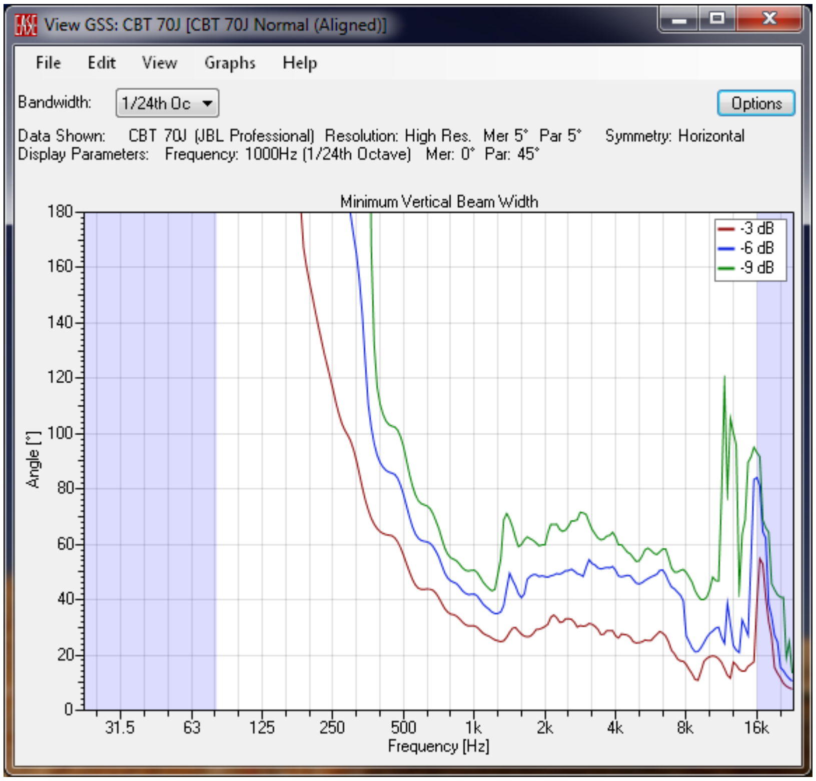
Bandwidth: 1/24th Oc

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 45°

Figure 2-57 – Directivity index

2.7.6.11 Beamwidth Graph

The BEAMWIDTH graphs display the calculated beamwidth for the source. This is calculated as the total included angle between the first occurrences of the specified off-axis attenuation limits on each side of on-axis. The off-axis attenuation limits are -3, -6, and -9 dB. Controlling whether or not to include each of these plots on the graphs is detailed in Section 2.7.7.8.



View GSS: CBT 70J [CBT 70J Normal (Aligned)]

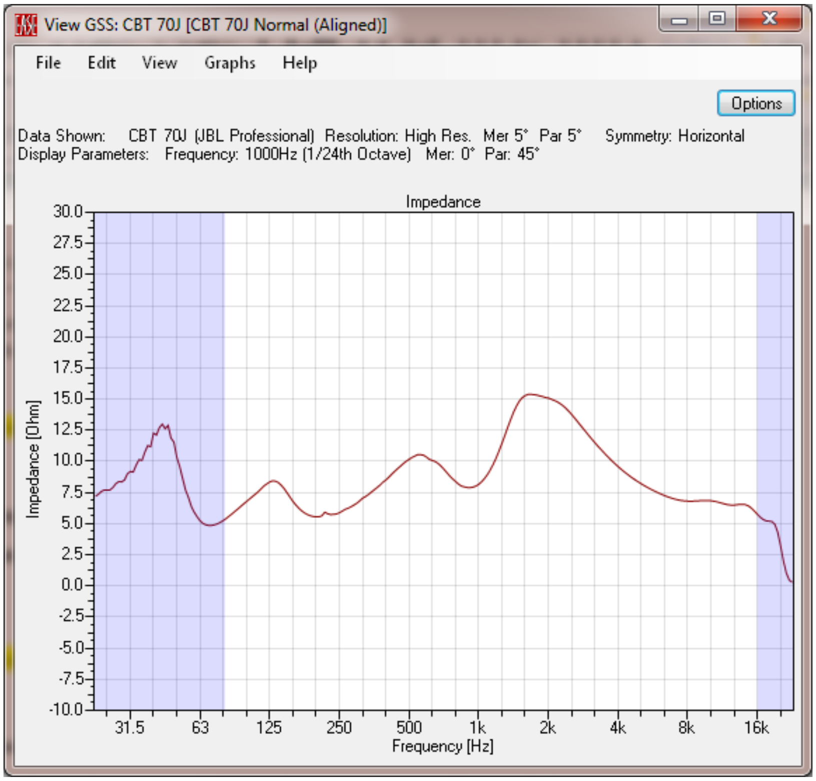
Bandwidth: 1/24th Oc

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 45°

Figure 2-58 – Beamwidth graph

2.7.6.12 Impedance Graph

The IMPEDANCE graph shows the imported impedance data. There are no selections associated with this graph.



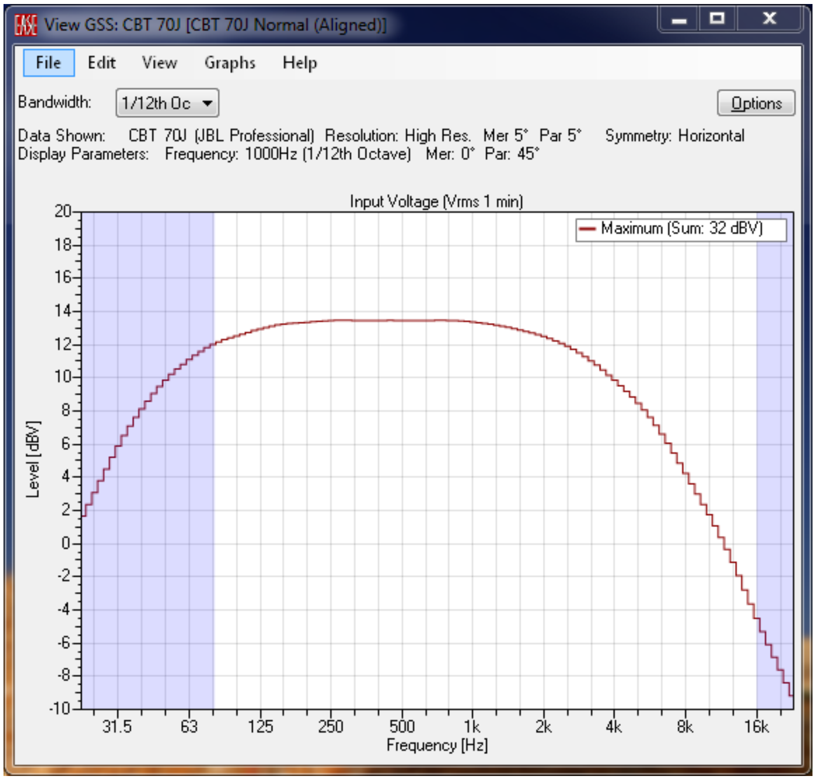
Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 45°

Figure 2-59 – Impedance graph

2.7.6.13 Maximum Input Voltage Graph

The MAXIMUM INPUT VOLTAGE graph displays the spectral content of the specified input signal at its maximum allowed level based on the specified MAXIMUM INPUT VOLTAGE for the source. The BANDWIDTH selection for this graph will change the level of each of the fractional octaves displayed, just as was the case with the FREQUENCY SPECTRUM graph. Again, this is due to the increased energy in each fractional octave as each fractional octave is increased in its bandwidth.

Note also that in the upper right-hand corner of the graph a Sum level is shown. This is the broadband level resulting from the summation across the entire spectrum. If a volt meter was placed at the input of the source when driven at its maximum input voltage, this is the reading one would expect to see.

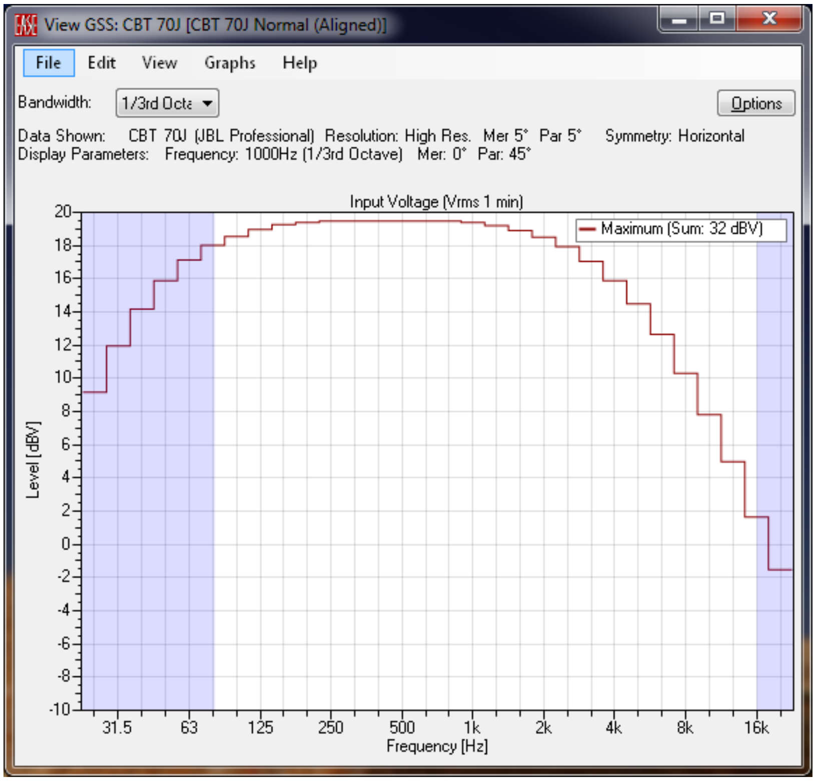


View GSS: CBT 70J [CBT 70J Normal (Aligned)]

Bandwidth: 1/12th Oc

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/12th Octave) Mer: 0° Par: 45°

Figure 2-60 – Maximum input voltage graph at 1/12 octave



View GSS: CBT 70J [CBT 70J Normal (Aligned)]

Bandwidth: 1/3rd Octa

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/3rd Octave) Mer: 0° Par: 45°

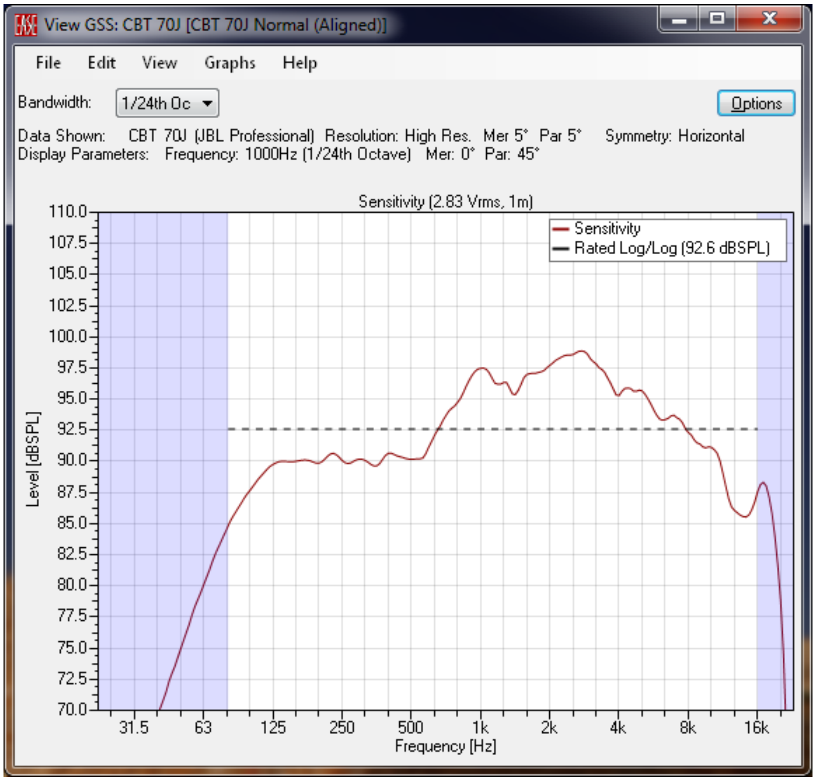
Figure 2-61 Maximum input voltage graph at 1/3 octave

2.7.6.14 Sensitivity Graph

The SENSITIVITY graph is identical to the TRANSFER FUNCTION graph for the on-axis location with two exceptions. First is that the display is of magnitude only. There is no phase data to be shown on this graph. The second is that the level is referenced to 1 meter and an rms input voltage of 2.83 V.

One other difference is that the calculated sensitivity value is shown on this graph. The sensitivity value is calculated over the bandwidth defined by the upper and lower frequency limits entered for the RATED FREQUENCY BANDWIDTH. This is in keeping with the definition of sensitivity in IEC 60268-5 [6] for an rms input of 2.83 V when the Log / Log option is selected (Section 2.7.7.4). This is the default as it corresponds roughly to the results one would get when measuring sensitivity directly with pink noise. For a relatively flat frequency response curve there will be very little difference between a direct measurement of sensitivity with pink noise and the log/log average that is displayed. When the Lin / Lin option is selected a linear (power) calculation is performed instead.

The BANDWIDTH setting will smooth or average the displayed data. It will not affect the calculated sensitivity as the actual data remains unchanged.


View GSS: CBT 70J [CBT 70J Normal (Aligned)]

Bandwidth: 1/24th Oc

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 45°

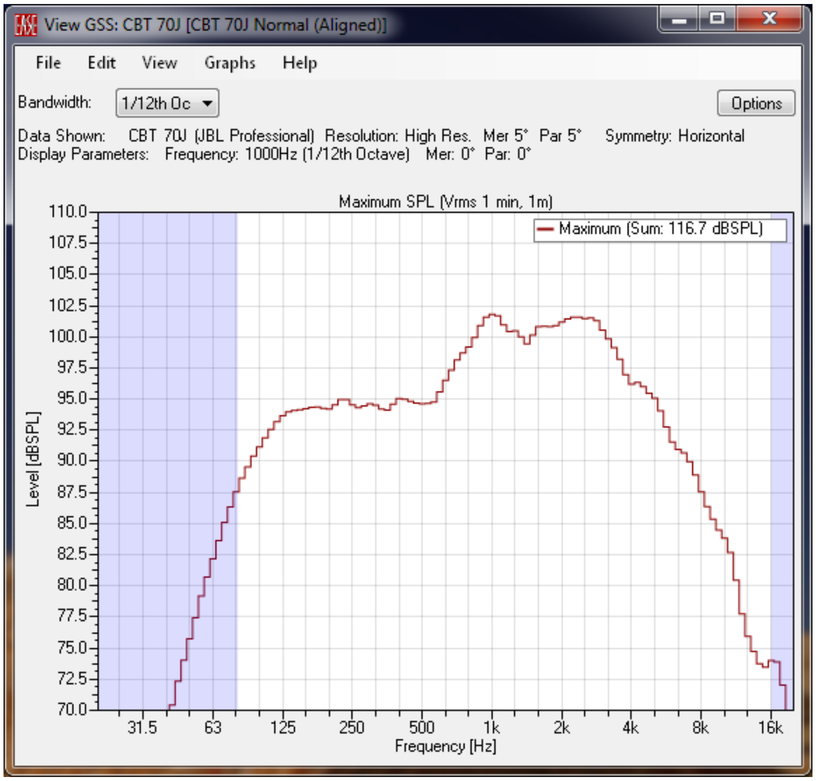
Figure 2-62 – Sensitivity graph

2.7.6.15 Maximum SPL Graph

The MAXIMUM SPL graphs shows the calculated maximum SPL for the source based on the SENSITIVITY data entered, the MAXIMUM INPUT VOLTAGE entered, and the specified INPUT SIGNAL. Note that this is a linear calculation and no level reductions due to non-linearities (power compression) are taken into account. This level is referenced to 1 meter. This graph is otherwise identical to the FREQUENCY SPECTRUM graph at the on-axis location.

As mentioned previously, the fractional octave BANDWIDTH may also be selected. This is used to determine the intervals over which the energy is summed and displayed for the graph. As larger fractional octaves are selected the curve will seem to rise on the graph. This is in keeping with how a real sound level meter would respond with the same factional octave settings. A larger bandwidth (say 1/3 octave instead of 1/12 octave) will contain more energy, and therefore have a greater SPL reading.

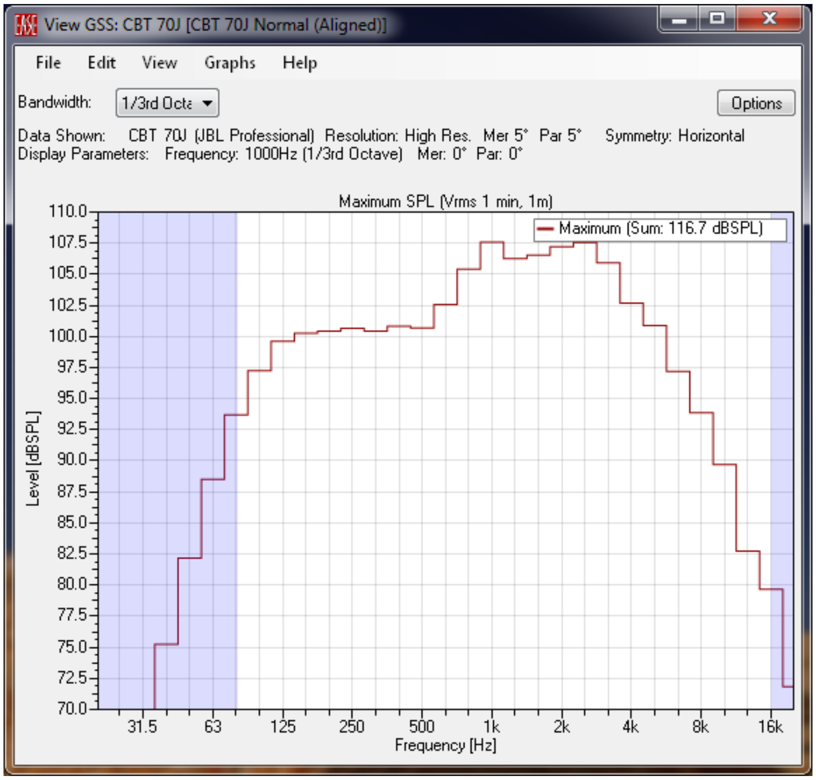
In the upper right-hand corner of the graph there is a value given for the total Sum level. Since this is a summation across the entire spectrum it is not affected by the BANDWIDTH setting. This Sum level is what might be expected from a sound level meter set for broadband unweighted readings.


View GSS: CBT 70J [CBT 70J Normal (Aligned)]

Bandwidth: 1/12th Oc

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/12th Octave) Mer: 0° Par: 0°

Figure 2-63 – Maximum SPL graph at 1/12 octave



View GSS: CBT 70J [CBT 70J Normal (Aligned)]

Bandwidth: 1/3rd Octa

Data Shown: CBT 70J (JBL Professional) Resolution: High Res. Mer 5° Par 5° Symmetry: Horizontal
Display Parameters: Frequency: 1000Hz (1/3rd Octave) Mer: 0° Par: 0°

Figure 2-64 – Maximum SPL graph at 1/3 octave

2.7.7 Graphs Options

Each of the graphs has several display options provided to help tailor it to each user's needs. Clicking on the GRAPH OPTIONS button (found above the upper right-hand area of the graph display area) or right-clicking in the graph display area will open the GRAPH OPTIONS dialog box. There is a tab in this dialog box for each type of graph.

At the bottom left-hand of each tab is a DEFAULT SETTINGS button. Clicking this button will return all of the settings on the selected tab to their default values. The settings on the other tabs will not be affected.

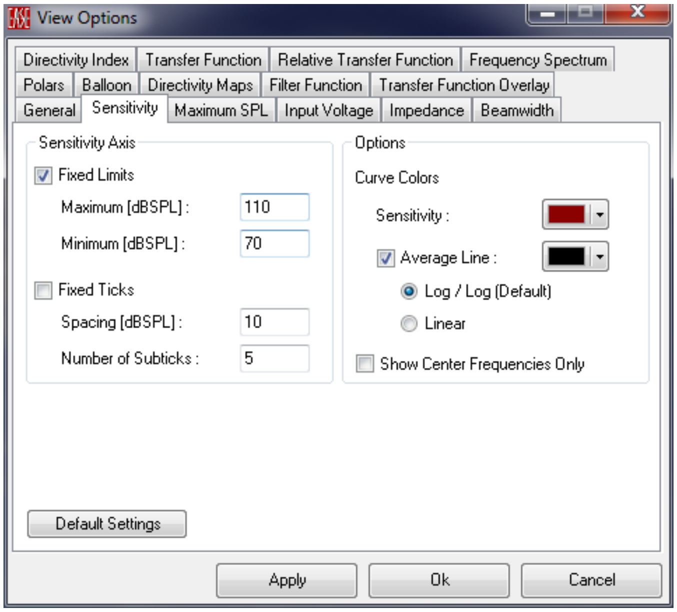
None of the changes to any of the settings will take effect until either the APPLY or the OK buttons are clicked. The APPLY button will immediately apply the changes to the display and will leave the GRAPH OPTIONS dialog box open. Clicking the OK button will apply the changes to the display and close the dialog box. The CANCEL button will cancel any pending changes so that they will not affect the display and close the dialog box. Note that the CANCEL button will not affect any changes previously made to the display by clicking the APPLY button.

2.7.7.1 Display Axis

Every tab (except the Polars, Balloon, Directivity Maps, and General tabs) has a section to control the limits of the vertical display axis. Checking the Fixed Limits checkbox will confine the vertical axis to the values entered in the Maximum [dBSPL] and Minimum [dBSPL] boxes. When this box is left unchecked the vertical axis limits will adjust dynamically to fit the range of data to be displayed.

The Fixed Ticks checkbox, when checked, will force the horizontal SPL lines drawn on the graph at the entered Spacing [dBSPL]. There will be a number tick marks on the vertical axis as well. The number of these is determined by the value entered into the Number of Subticks box.

Some other types of graphs may have other units for the vertical display axis (e.g. dBV, ohms, ° (degree), dB, etc.). These other units will be shown, instead of dBSPL, for controlling their displays.



Sensitivity Axis

☑ Fixed Limits

Maximum [dBSPL] : 110

Minimum [dBSPL] : 70

☐ Fixed Ticks

Spacing [dBSPL] : 10

Number of Subticks : 5

Options

Curve Colors

Sensitivity : 🟥

☑ Average Line : ■

● Log / Log (Default)

○ Linear

☐ Show Center Frequencies Only

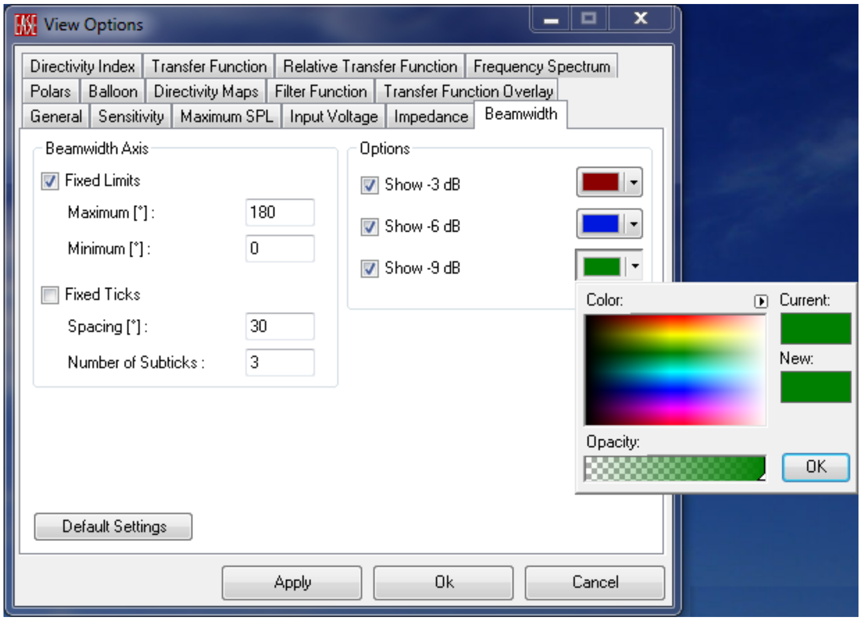
Figure 2-65 – Graph Options dialog box

2.7.7.2 Curve Colors

Almost all of the tabs have an OPTIONS section that can be used to control the CURVE COLOR. The BALLOON and DIRECTIVITY MAPS tab do not have this selection as these are 3 dimensional plots and not 2D curves.

Clicking on the dropdown color box opens a color selection screen. Click anywhere inside the COLOR map to select the desired color. The current (old) color will be shown as well as the new selected color. There is also a button above the upper right-hand corner of the color map. Clicking this button will open a standard WINDOWS COLOR SELECTION dialog box. Predefined colors may be chosen from here.

The OPACITY of the color can also be chosen. Clicking anywhere along the Opacity line will change the amount of transparency for the color. There is also a line that can be dragged to change the transparency.



Beamwidth Axis

☑ Fixed Limits

Maximum [°] : 180

Minimum [°] : 0

□ Fixed Ticks

Spacing [°] : 30

Number of Subticks : 3

Options

☑ Show -3 dB 🟥

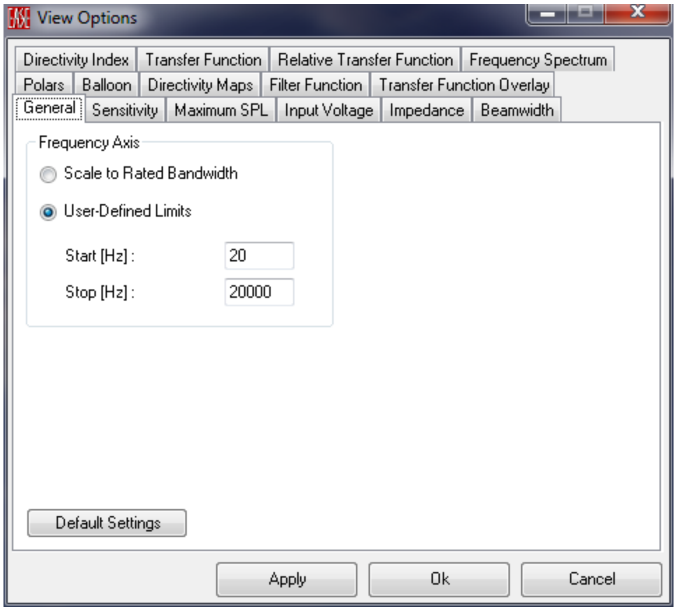
☑ Show -6 dB 🟦

☑ Show -9 dB 🟩

Figure 2-66 – Graph Options dialog box, Beamwidth tab & color selection

2.7.7.3 General Tab

The GENERAL tab allows the upper and lower frequency limits that will be displayed to be set. They can either be set to SCALE TO THE RATED BANDWIDTH of the source or to USER–DEFINED LIMITS.



Frequency Axis

○ Scale to Rated Bandwidth

● User-Defined Limits

Start [Hz] : 20

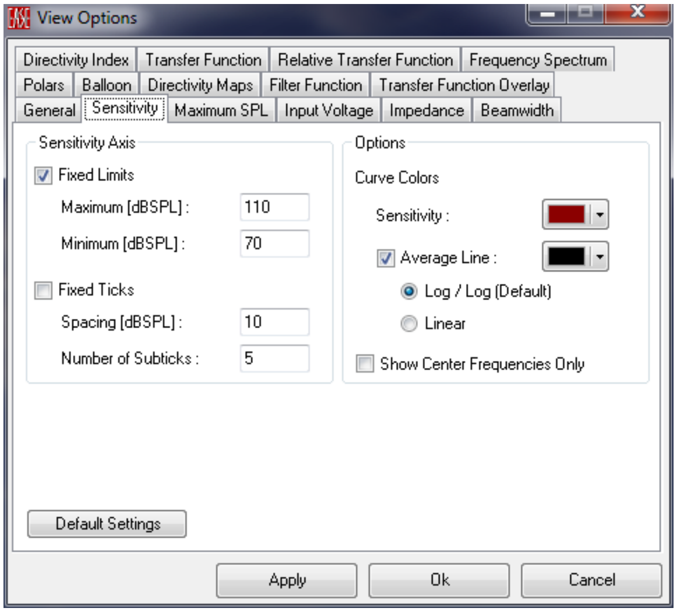
Stop [Hz] : 20000

Figure 2-67– Graph Options dialog box, General tab

2.7.7.4 Sensitivity Tab

The Options section of the Sensitivity tab allows the selection of the color for both the Sensitivity and the Average Line. The Average Line checkbox toggles the rated sensitivity line shown on the graphs off and on. The two radio buttons determine how the rated sensitivity is calculated. The labeling describes weighting applied to the calculation. The default is the Log/Log setting which calculates the average SPL using fractional octave bandwidths. This is equivalent to using pink noise to excite the source. This approximately correlates with the method specified in IEC 60268-5 [6] when the source is excited with an rms input of 2.83 V of pink noise over the rated bandwidth. The Linear setting calculates the power average over linear frequency. This is similar to using white noise to excite the source. This will weight the contribution of the higher frequencies more heavily in the calculation.

The Show Center Frequencies Only checkbox changes the display from a smooth curve to straight line segments between the fractional octave center frequencies. These center frequencies will change based on the Bandwidth selected for the graph.



Sensitivity Axis

☑ Fixed Limits

Maximum [dBSPL] : 110

Minimum [dBSPL] : 70

□ Fixed Ticks

Spacing [dBSPL] : 10

Number of Subticks : 5

Options

Curve Colors

Sensitivity : 🟥

☑ Average Line : ■

● Log / Log (Default)

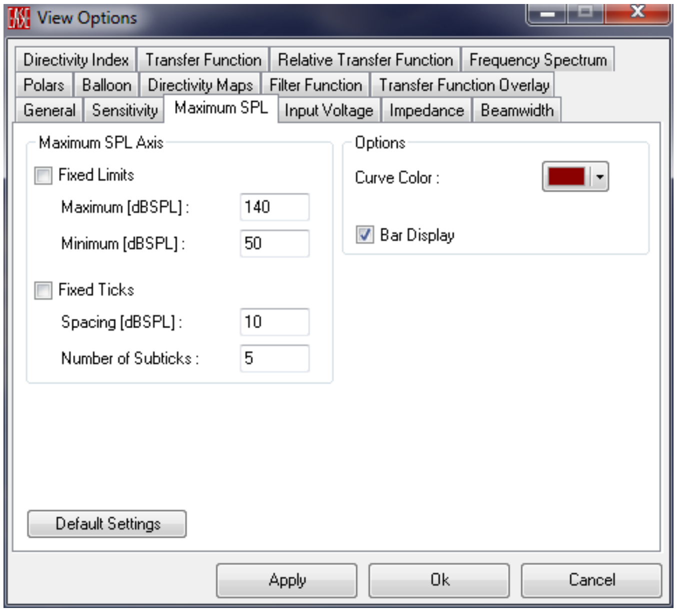
○ Linear

□ Show Center Frequencies Only

Figure 2-68 – Graph Options dialog box, Sensitivity tab

2.7.7.5 Maximum SPL Tab

The MAXIMUM SPL tab is used to set the limits for the MAXIMUM SPL AXIS as well as the color of the curve. When the BAR DISPLAY checkbox is checked a stair-step type bar graph is displayed. When it is unchecked a smooth, continuous curve is displayed. Since the selected fractional octave bandwidth affects the level of each bandwidth (like a sound level meter) it is recommended to always enable the BAR DISPLAY option for the MAXIMUM SPL.



Maximum SPL Axis

□ Fixed Limits

Maximum [dBSPL] : 140

Minimum [dBSPL] : 50

□ Fixed Ticks

Spacing [dBSPL] : 10

Number of Subticks : 5

Options

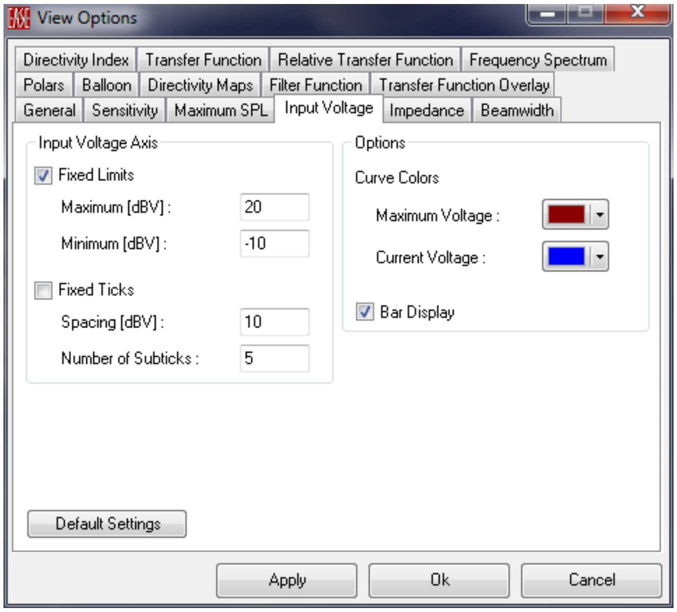
Curve Color : 🟥

☑ Bar Display

Figure 2-69 – Graph Options dialog box, Maximum SPL tab

2.7.7.6 Input Voltage Tab

The Input Voltage tab is used to set the limits for the Input Voltage Axis as well as the color of the curve. When the Bar Display checkbox is checked a stair-step type bar graph is displayed. When it is unchecked a smooth, continuous curve is displayed. Since the selected fractional octave bandwidth affects the level of each bandwidth (like a sound level meter) it is recommended to always enable the Bar Display option for the Input Voltage.



Input Voltage Axis

☑ Fixed Limits

Maximum [dBV] : 20

Minimum [dBV] : -10

☐ Fixed Ticks

Spacing [dBV] : 10

Number of Subticks : 5

Options

Curve Colors

Maximum Voltage : 🟥

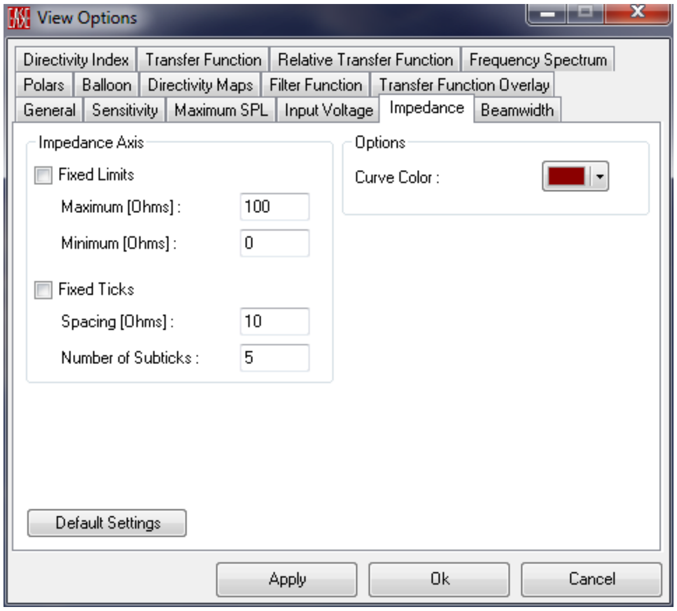
Current Voltage : 🟦

☑ Bar Display

Figure 2-70 – Graph Options dialog box, Input Voltage tab

2.7.7.7 Impedance Tab

The IMPEDANCE tab is used to set the limits for the IMPEDANCE AXIS as well as the color of the curve.



Impedance Axis

□ Fixed Limits

Maximum [Ohms] : 100

Minimum [Ohms] : 0

□ Fixed Ticks

Spacing [Ohms] : 10

Number of Subticks : 5

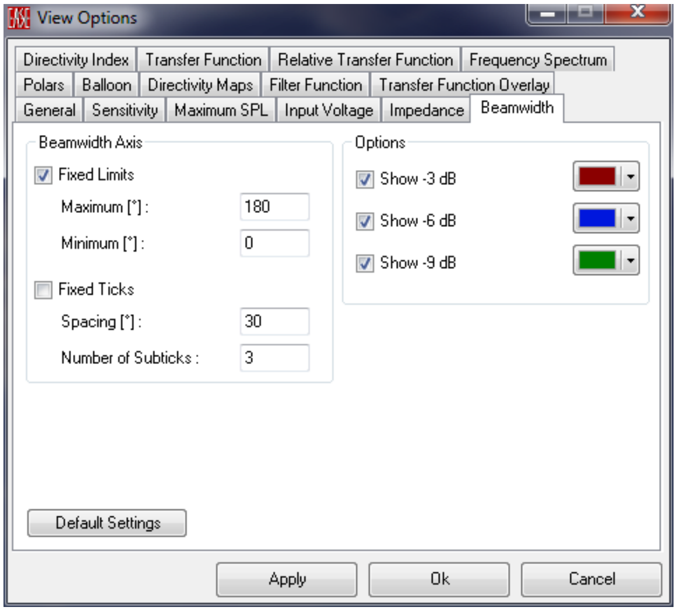
Options

Curve Color : 🟥

Figure 2-71 – Graph Options dialog box, Impedance tab

2.7.7.8 Beamwidth Tab

The Beamwidth tab is used to set the limits for the Beamwidth Axis as well as the color of the curves. The Beamwidth Axis setting affects the display of both the horizontal and vertical beamwidth graphs. In addition to the typical -6 dB beamwidth it is possible to also display beamwidth curves that are calculated for -3 dB and -9 dB points on the polar plot. The Show −x dB checkbox controls whether or not a particular curve is displayed on the graph.



Beamwidth Axis

☑ Fixed Limits

Maximum [°] : 180

Minimum [°] : 0

☐ Fixed Ticks

Spacing [°] : 30

Number of Subticks : 3

Options

☑ Show -3 dB 🟥

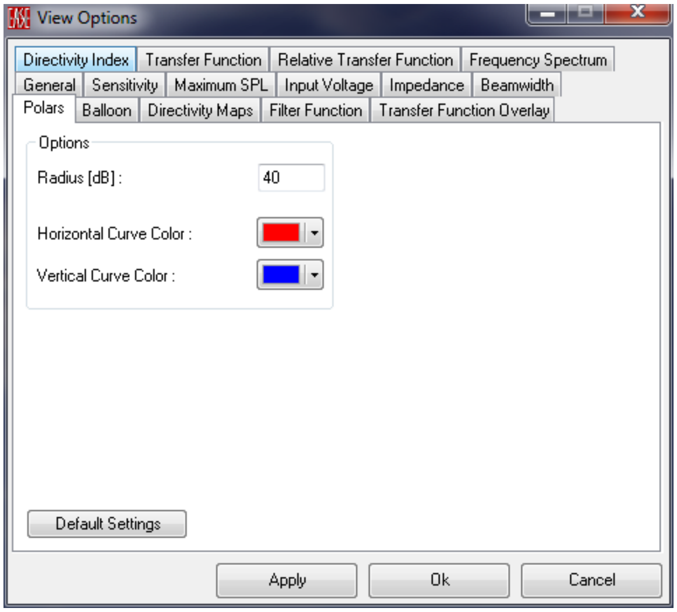
☑ Show -6 dB 🟦

☑ Show -9 dB 🟩

Figure 2-72 - – Graph Options dialog box, Beamwidth tab

2.7.7.9 Polars Tab

Instead of the axis limits settings found on most of the other tabs the POLARS tab has only a RADIUS [DB] setting to control the display limits. This setting determines the total range of the graph from the center to the outer-most circle. The colors of the horizontal and vertical curves can be set independently.



Radius [dB] : 40

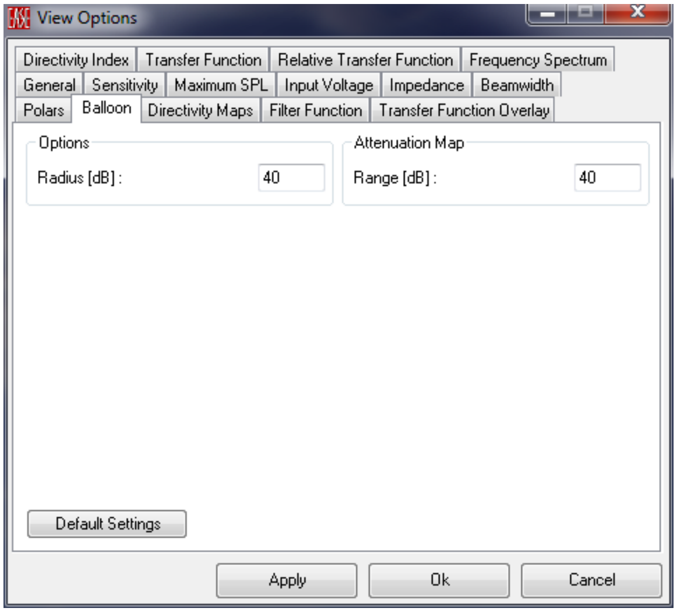
Horizontal Curve Color : 🟥

Vertical Curve Color : 🟦

Figure 2-73 – Graph Options dialog box, Polars tab

2.7.7.10 Balloon Tab

This tab only has a Radius [dB] setting, just like the Polars tab, for the Balloon Graph. There is also a Range [dB] setting that controls the total attenuation range assigned to the RGB color scale for denoting level.



Radius [dB] : 40

Range [dB] : 40

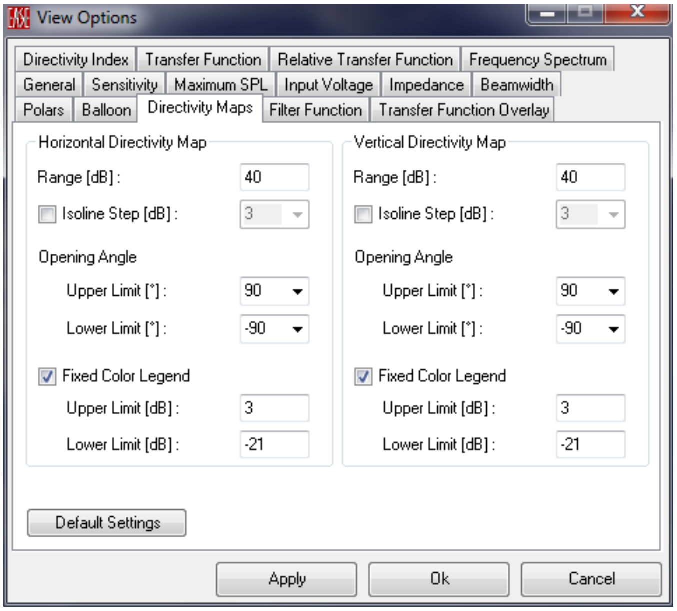
Figure 2-74 – Graph Options dialog box, Balloon tab

2.7.7.11 Directivity Maps Tab

The DIRECTIVITY MAPS tab has a RANGE [DB] setting, similar to the BALLOON tab, which controls the total attenuation range that is displayed in the graph. If the FIXED COLOR LEGEND box is checked the UPPER LIMIT [DB] and LOWER LIMIT [DB] settings will determine the levels assigned to the color display range.

The Opening Angle settings control the off-axis angles shown in the graph. The UPPER LIMIT [°] and LOWER LIMIT [°] can be set separately to better view non-symmetric sources. The dropdown box can be used to select predetermined angles ranging from 10 to 180 degrees.

Note that there are separate areas for the settings for the HORIZONTAL DIRECTIVITY MAP (on the left) and the VERTICAL DIRECTIVITY MAP (on the right).



Horizontal Directivity Map

Range [dB] : 40

□ Isoline Step [dB] : 3

Opening Angle

Upper Limit [°] : 90

Lower Limit [°] : -90

☑ Fixed Color Legend

Upper Limit [dB] : 3

Lower Limit [dB] : -21

Vertical Directivity Map

Range [dB] : 40

□ Isoline Step [dB] : 3

Opening Angle

Upper Limit [°] : 90

Lower Limit [°] : -90

☑ Fixed Color Legend

Upper Limit [dB] : 3

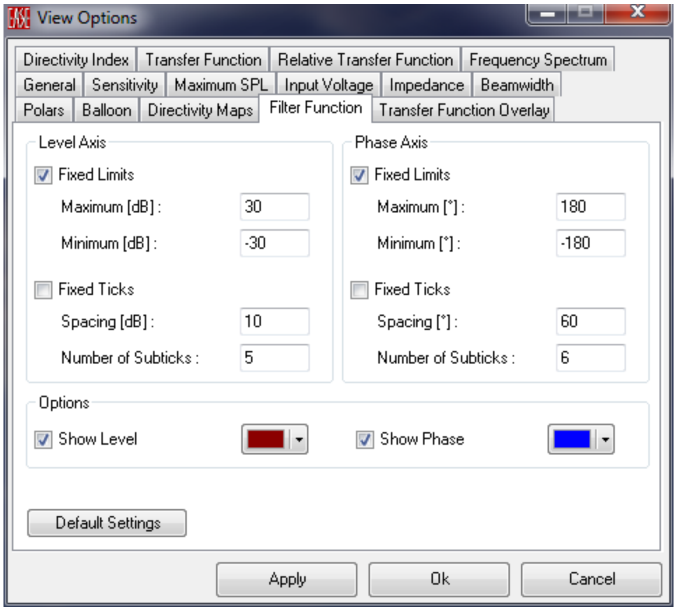
Lower Limit [dB] : -21

Figure 2-75 – Graph Options dialog box, Directivity Maps tab

2.7.7.12 Filter Function Tab

There are no FILTERS in the GSS Source Editor, but in the interest of completeness we will cover this item here anyway. The FILTERS FUNCTION tab controls the display of the complex transfer function of filters that are applied to the sources in a GLL. There are separate sections for the LEVEL AXIS and the PHASE AXIS settings.

Separate checkboxes to enable or disable the SHOW LEVEL and SHOW PHASE are also provided.



Level Axis

☑ Fixed Limits

Maximum [dB] : 30

Minimum [dB] : -30

☐ Fixed Ticks

Spacing [dB] : 10

Number of Subticks : 5

Phase Axis

☑ Fixed Limits

Maximum [°] : 180

Minimum [°] : -180

☐ Fixed Ticks

Spacing [°] : 60

Number of Subticks : 6

Options

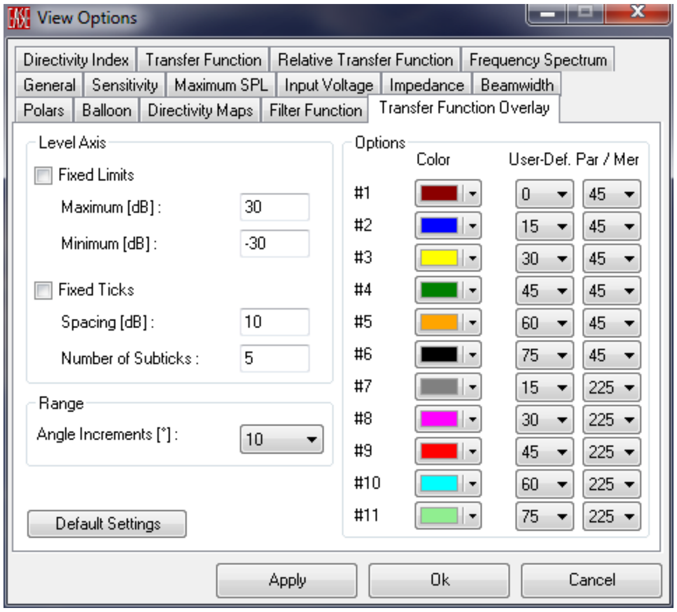
☑ Show Level 🟥

☑ Show Phase 🟦

Figure 2-76 – Graph Options dialog box, Filter Function tab

2.7.7.13 Transfer Function Overlay Tab

The Transfer Function Overlay tab provides a Level Axis section, as most of the tabs do. There is also a Range section that allows the selection for the Angle Increments [°] to be displayed. This is valid for all of the Transfer Function Overlay graphs except the User-Defined graph. The Options settings on the right side of the tab control the settings for the Par (Parallel) and Mer (Meridian) of the off-axis angle to be displayed. Up to eleven different off-axis angles may be specified for the User-Defined graph.



Level Axis

□ Fixed Limits

Maximum [dB] : 30

Minimum [dB] : -30

□ Fixed Ticks

Spacing [dB] : 10

Number of Subticks : 5

Range

Angle Increments [°] : 10

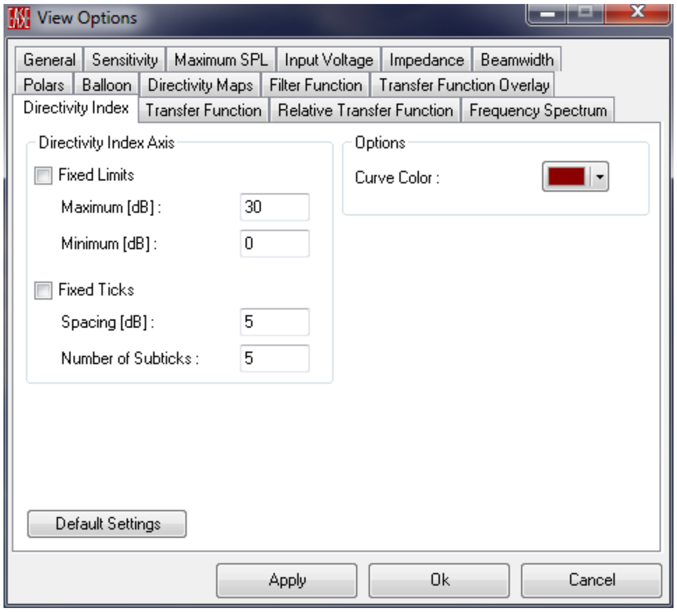
Options

	Color	User-Def. Par	/ Mer
#1	🟫	0	45
#2	🟦	15	45
#3	🟨	30	45
#4	🟩	45	45
#5	🟧	60	45
#6	⬛	75	45
#7	▨	15	225
#8	🟪	30	225
#9	🟥	45	225
#10	🟦	60	225
#11	🟩	75	225

Figure 2-77 – Graph Options dialog box, Transfer Function Overlay tab

2.7.7.14 Directivity Index Tab

The Directivity Index tab is used to set the limits for the Directivity Index Axis as well as the color of the curve.



Directivity Index Axis

□ Fixed Limits

Maximum [dB] : 30

Minimum [dB] : 0

□ Fixed Ticks

Spacing [dB] : 5

Number of Subticks : 5

Options

Curve Color : 🟥

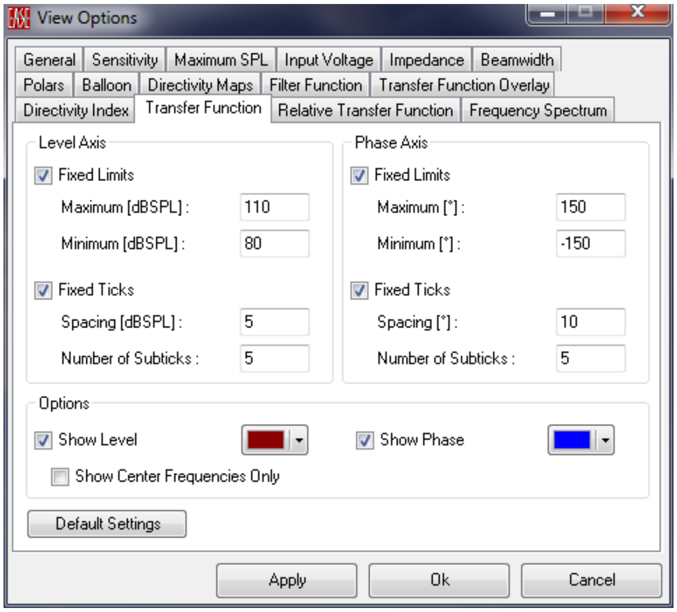
Figure 2-78 – Graph Options dialog box, Directivity Index tab

2.7.7.15 Transfer Function Tab

The TRANSFER FUNCTION tab controls the display of the complex transfer function of the source. There are separate sections for the LEVEL AXIS and the PHASE AXIS settings.

Separate checkboxes to enable or disable the SHOW LEVEL and SHOW PHASE are also provided.

The SHOW CENTER FREQUENCIES ONLY checkbox changes the display from a smooth curve to straight line segments between the fractional octave center frequencies. These center frequencies will change based on the BANDWIDTH selected for the graph.



Level Axis

☑ Fixed Limits

Maximum [dBSPL] : 110

Minimum [dBSPL] : 80

☑ Fixed Ticks

Spacing [dBSPL] : 5

Number of Subticks : 5

Phase Axis

☑ Fixed Limits

Maximum [°] : 150

Minimum [°] : -150

☑ Fixed Ticks

Spacing [°] : 10

Number of Subticks : 5

Options

☑ Show Level 🟥

☑ Show Phase 🟦

□ Show Center Frequencies Only

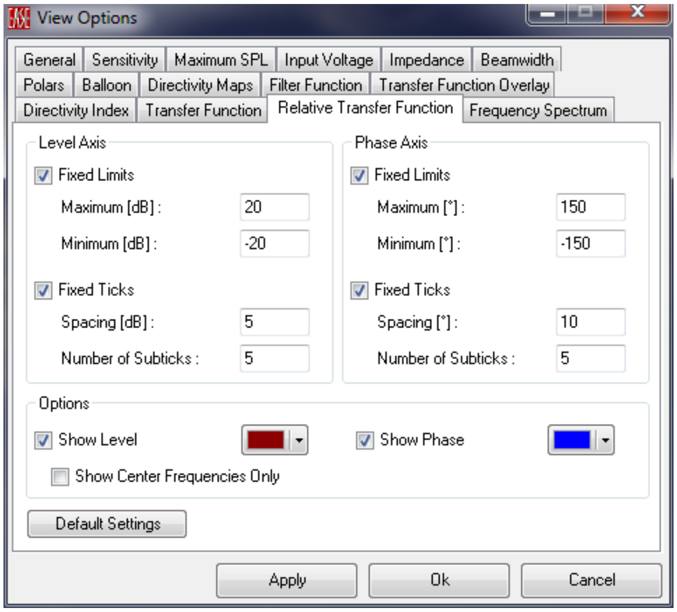
Figure 2-79 – Graph Options dialog box, Transfer Function tab

2.7.7.16 Relative Transfer Function Tab

The Relative Transfer Function tab controls the display of the complex transfer function of the source. There are separate sections for the Level Axis and the Phase Axis settings.

Separate checkboxes to enable or disable the Show Level and Show Phase are also provided.

The Show Center Frequencies Only checkbox changes the display from a smooth curve to straight line segments between the fractional octave center frequencies. These center frequencies will change based on the Bandwidth selected for the graph.


View Options

Level Axis

☑ Fixed Limits

Maximum [dB] : 20

Minimum [dB] : -20

☑ Fixed Ticks

Spacing [dB] : 5

Number of Subticks : 5

Phase Axis

☑ Fixed Limits

Maximum [°] : 150

Minimum [°] : -150

☑ Fixed Ticks

Spacing [°] : 10

Number of Subticks : 5

Options

☑ Show Level 🟥

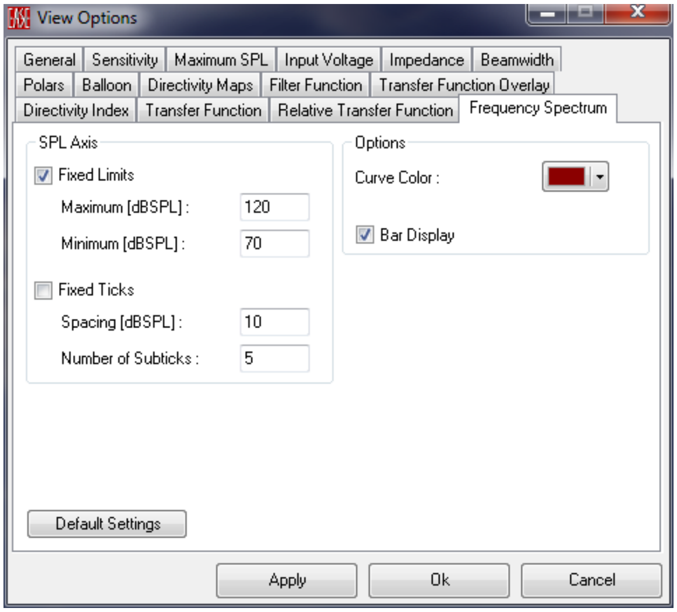
☑ Show Phase 🟦

☐ Show Center Frequencies Only

Figure 2-80 – Graph Options dialog box, Relative Transfer Function tab

2.7.7.17 Frequency Spectrum Tab

The FREQUENCY SPECTRUM tab is used to set the limits for the SPL AXIS as well as the color of the curve. When the BAR DISPLAY checkbox is checked a stair-step type bar graph is displayed. When it is unchecked a smooth, continuous curve is displayed. Since the selected fractional octave bandwidth affects the level of each bandwidth (like a sound level meter) it is recommended to always enable the BAR DISPLAY option for the MAXIMUM SPL.



SPL Axis

☑ Fixed Limits

Maximum [dBSPL] : 120

Minimum [dBSPL] : 70

☐ Fixed Ticks

Spacing [dBSPL] : 10

Number of Subticks : 5

Options

Curve Color : 🟥

☑ Bar Display

Figure 2-81 – Graph Options dialog box, Frequency Spectrum tab

2.7.8 Viewing Options (View Menu)

The View selection on the main menu allows different viewing perspectives for some of the graphs. These selections are 3D Perspective, Side (x), End (y), and Plan (z). They only have an effect on the 3D-type of graphs (Attenuation Balloon, Directivity Maps, Balloon Mapping, and Phase Balloon). These are the first five graphs listed under the Graphs selection from the main menu.

There are also four different Render Option selections; Wireframe, Colored Contours, White Contours, and Flat. These affect only the Attenuation Balloon, Directivity Maps, and Balloon Mapping. The Phase Balloon is always shown as a Wireframe.

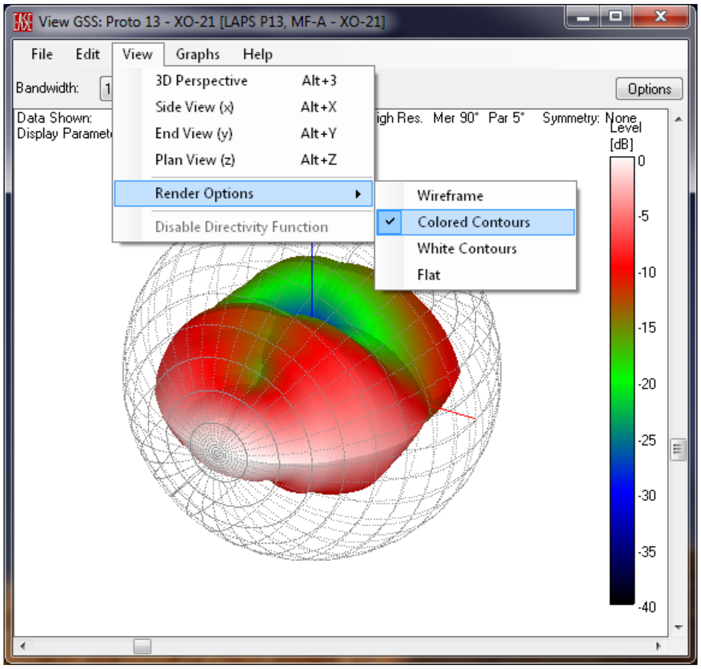
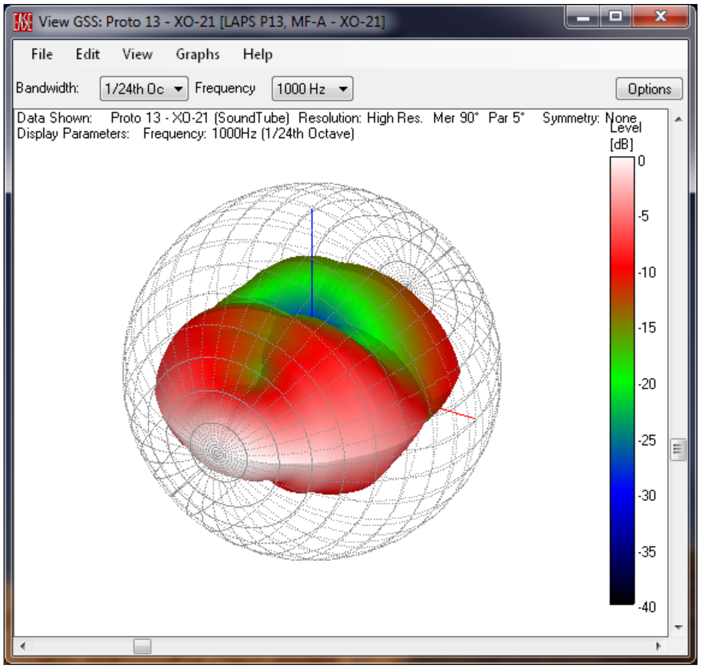


Figure 2-82 – Viewing menu selections

The default selection, COLORED CONTOURS, displays the graph both with colors and contours denoting the attenuation level as shown in Figure 2-83.


View GSS: Proto 13 - XO-21 [LAPS P13, MF-A - XO-21]

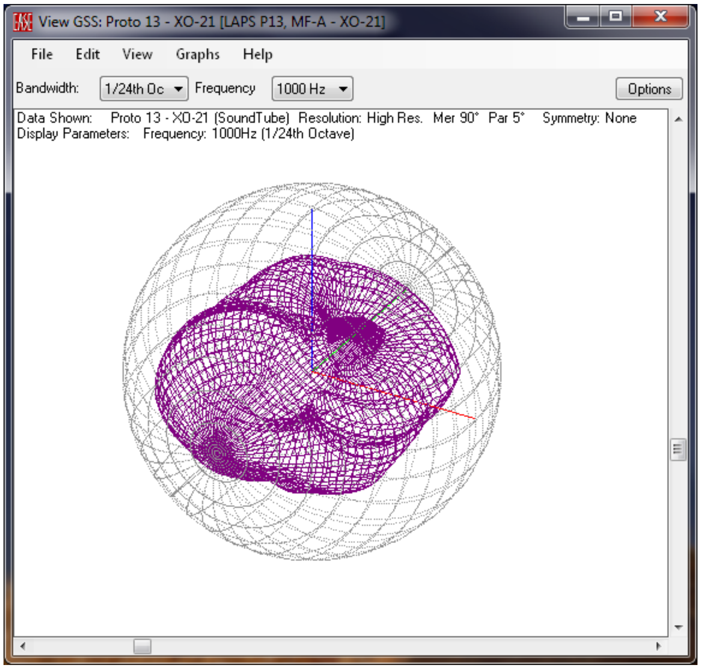
Bandwidth: 1/24th Oc Frequency 1000 Hz

Data Shown: Proto 13 - XO-21 (SoundTube) Resolution: High Res. Mer 90° Par 5° Symmetry: None

Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-83 – Attenuation balloon rendered as colored contours

The WIREFRAME selection displays the graph as a simple wireframe drawing with contours denoting the attenuation level as shown in Figure 2-84.


View GSS: Proto 13 - XO-21 [LAPS P13, MF-A - XO-21]

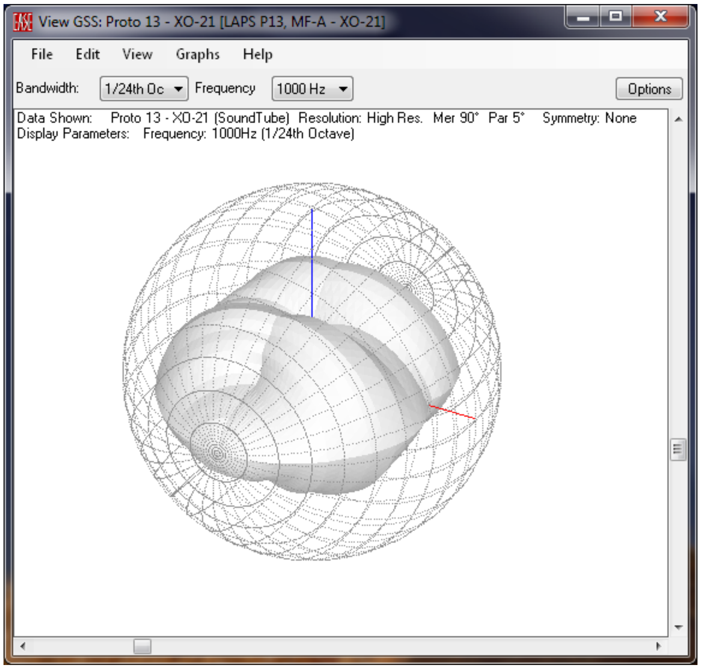
Bandwidth: 1/24th Oc Frequency 1000 Hz

Data Shown: Proto 13 - XO-21 (SoundTube) Resolution: High Res. Mer 90° Par 5° Symmetry: None

Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-84 – Attenuation balloon rendered as wireframe

The WHITE CONTOURS selection displays the graph with contours denoting the attenuation level while the coloring is white with some grey shadowing (3D effect from an apparent light source illuminating the balloon) to help show some of the balloon features.

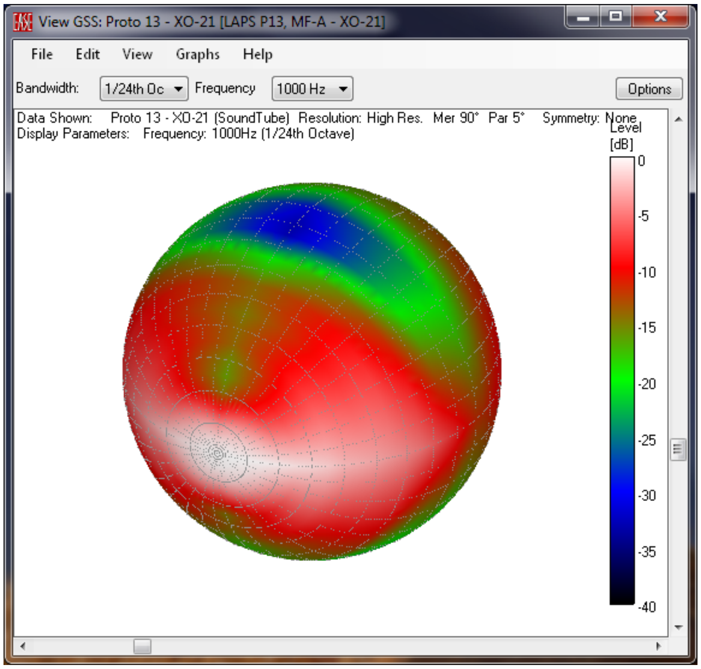

View GSS: Proto 13 - XO-21 [LAPS P13, MF-A - XO-21]

Bandwidth: 1/24th Oc Frequency 1000 Hz

Data Shown: Proto 13 - XO-21 (SoundTube) Resolution: High Res. Mer 90° Par 5° Symmetry: None
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-85 – Attenuation balloon rendered as white contours

The FLAT selection displays the graph as a perfect sphere with only the coloring denoting the attenuation level as shown in Figure 2-86.


View GSS: Proto 13 - XO-21 [LAPS P13, MF-A - XO-21]

Bandwidth: 1/24th Oc Frequency 1000 Hz

Data Shown: Proto 13 - XO-21 (SoundTube) Resolution: High Res. Mer 90° Par 5° Symmetry: None
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-86 – Attenuation balloon rendered as flat

2.7.9 Directivity Function (Pro Version only)

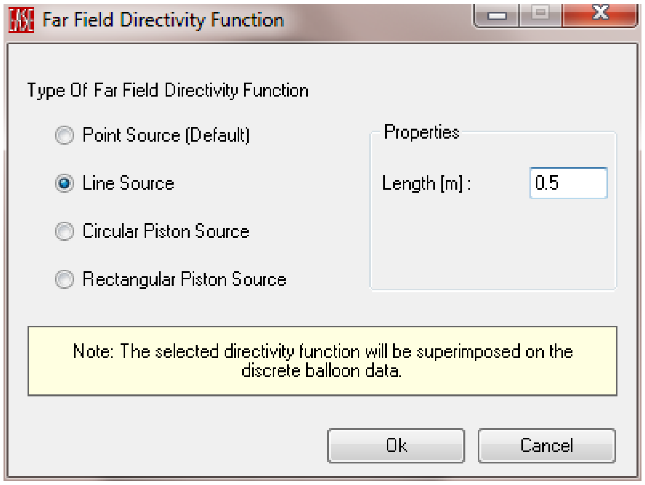
The Pro version of SpeakerLab also features the possibility to superimpose a far field directivity function over a previously defined directivity balloon. These directivity functions can also be used when no data has been imported. This will yield the theoretically ideal directivity behavior for the type of source selected. This can be very useful for initial models at the beginning of a loudspeaker design project.

The Far Field Directivity Function dialog box is opened by clicking Edit > Directivity Function from the main menu.

The available directivity functions are:

- Point Source (Default)
- Line Source
- Circular Piston Source
- Rectangular Piston Source

Once a directivity function is selected, a Properties section will appear in the right part of the dialog box to enter the source dimensions. The dimension(s) entered define the frequency dependent directivity of the source.



Far Field Directivity Function

Type Of Far Field Directivity Function

○ Point Source (Default)
● Line Source
○ Circular Piston Source
○ Rectangular Piston Source

Properties

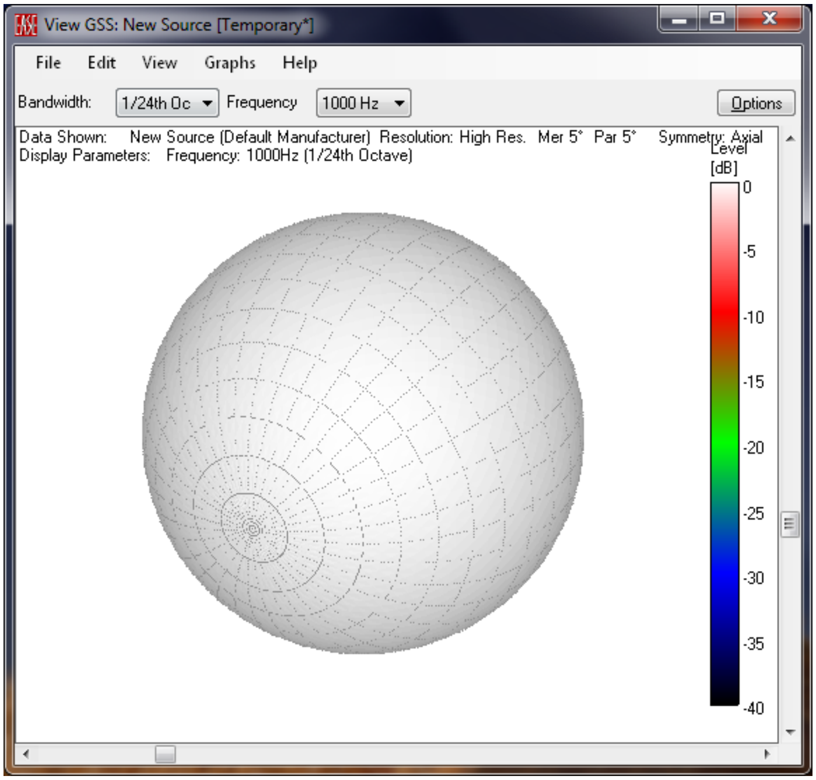
Length [m] : 0.5

Note: The selected directivity function will be superimposed on the discrete balloon data.

Figure 2-87 – Far-Field Directivity Function dialog box

Clicking the OK button will close the Far Field Directivity Function dialog box and switch back to the GSS Source Editor. The original directivity balloon data is not changed and can be recovered at any time by selecting Point Source [Default] as the Far Field Directivity Function. However, the directivity balloon response that will be used for calculations is now the product of the original directivity balloon data and the directivity function.

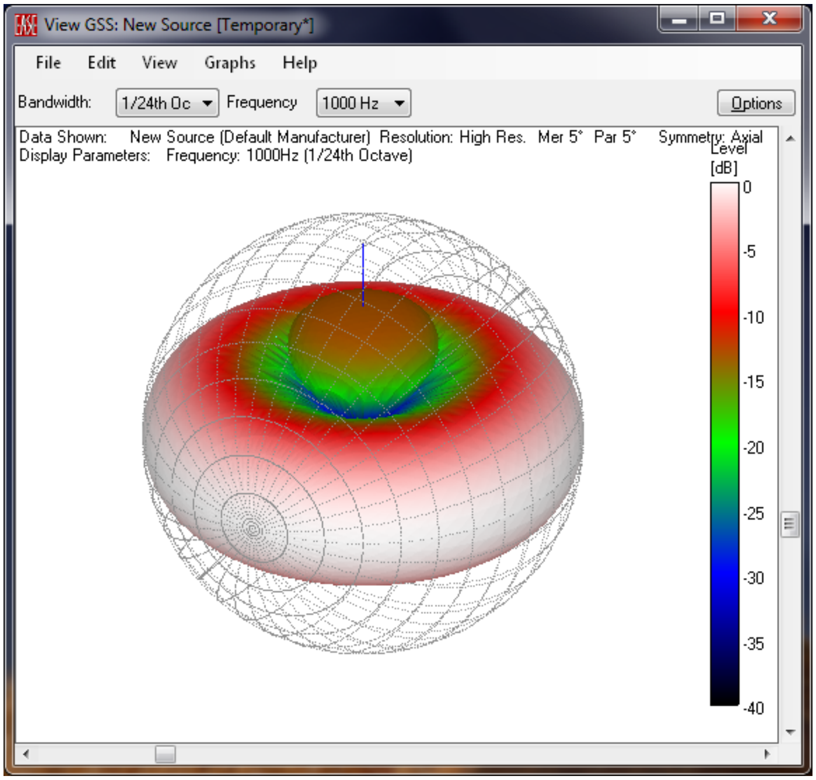
As an example, it is possible to check the effect of the Line Source directivity function superimposed over an omni-directional source.



Bandwidth: 1/24th Oc Frequency 1000 Hz

Data Shown: New Source (Default Manufacturer) Resolution: High Res. Mer 5° Par 5° Symmetry: Axial
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-88 – Point Source



Bandwidth: 1/24th Oc Frequency 1000 Hz

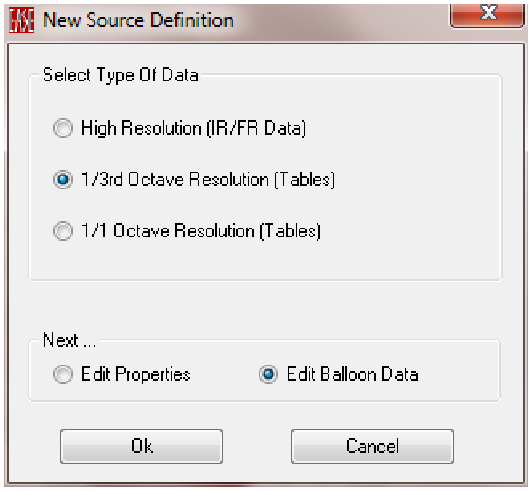
Data Shown: New Source (Default Manufacturer) Resolution: High Res. Mer 5° Par 5° Symmetry: Axial
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-89 – Line Source, 0.5 meters long

2.7.10 Defining a New Source (1/3 Octave Data Table)

As described before, when a new source is defined, the user is prompted to choose between three data formats. The first one, High-Resolution (IR/FR Data), was used for the previous source definition example. Here we briefly describe the tabular approach of the other two available formats: 1/3rd Octave Resolution (Tables) and 1/1 Octave Resolution (Tables).

In the main menu of the Source Editor select File > New, then select the 1/3rd Octave Resolution (Tables) option and Edit Balloon Data in the Next… area.



New Source Definition

Select Type Of Data

○ High Resolution (IR/FR Data)

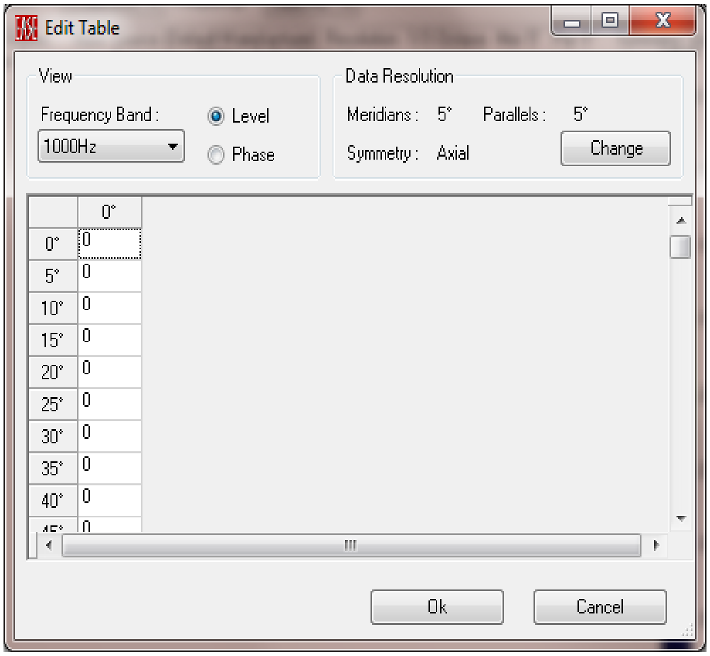
● 1/3rd Octave Resolution (Tables)

○ 1/1 Octave Resolution (Tables)

Next ...

○ Edit Properties ● Edit Balloon Data

Figure 2-90 – New Source Definition data type selection



Edit Table

View

Frequency Band : 1000Hz

● Level

○ Phase

Data Resolution

Meridians : 5° Parallels : 5°

Symmetry : Axial

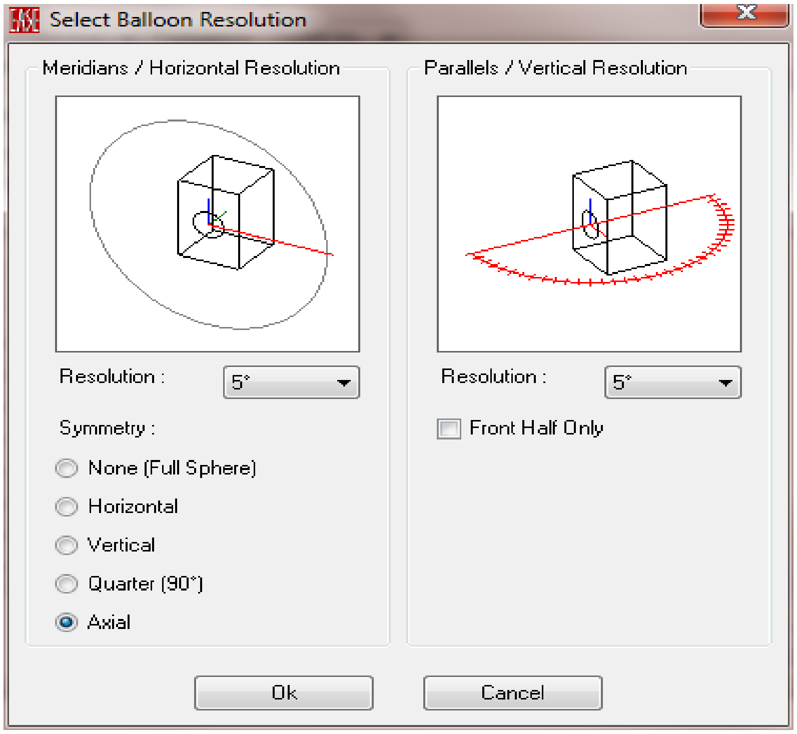
	0°
0°	0
5°	0
10°	0
15°	0
20°	0
25°	0
30°	0
35°	0
40°	0
45°	0

Figure 2-91 – Data table source definition

Clicking the OK button will open the Edit Table dialog box shown in Figure 2-91.

In the tabular directivity source description, the polar data is stored in a set of tables. For each valid frequency band (1/3 octave or 1/1 octave resolution) there are two tables: one for the level attenuation relative to the on-axis magnitude data, and one for the phase difference relative to the on-axis phase data. In each table the columns are the Meridian angles (m) and the rows are the Parallel (p) angles.

By default the Symmetry is Axial and the angular resolution is 5 degrees for both the Meridians and the Parallels. This can be changed by clicking the Change button that will open a dialog box where the Symmetry and the Resolution may be defined. This dialog box is similar to the one previously described for importing the High Resolution impulse responses.


Select Balloon Resolution
Meridians / Horizontal Resolution
Parallels / Vertical Resolution
Ok
Cancel

Meridians / Horizontal Resolution

Resolution : 5°

Symmetry :

○ None (Full Sphere)
○ Horizontal
○ Vertical
○ Quarter (90°)
● Axial

Parallels / Vertical Resolution

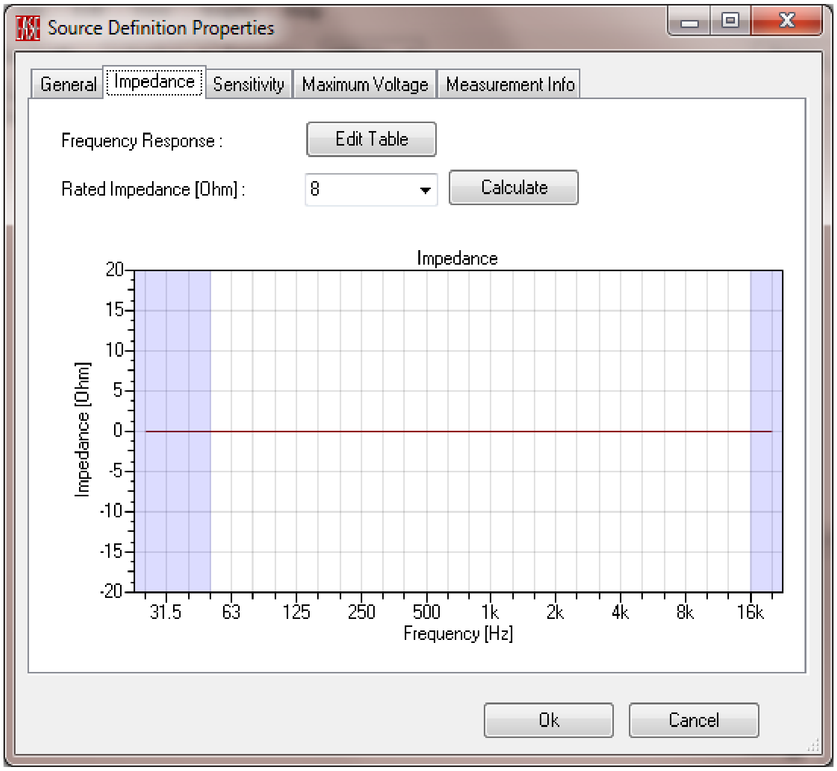
Resolution : 5°

□ Front Half Only

Figure 2-92 – Balloon resolution and symmetry selection

The tables can be directly edited by selecting each cell and typing the attenuation, in dB, or the phase shift, in degrees. Data can also be copied from a different location and pasted into the table like in normal spreadsheets.

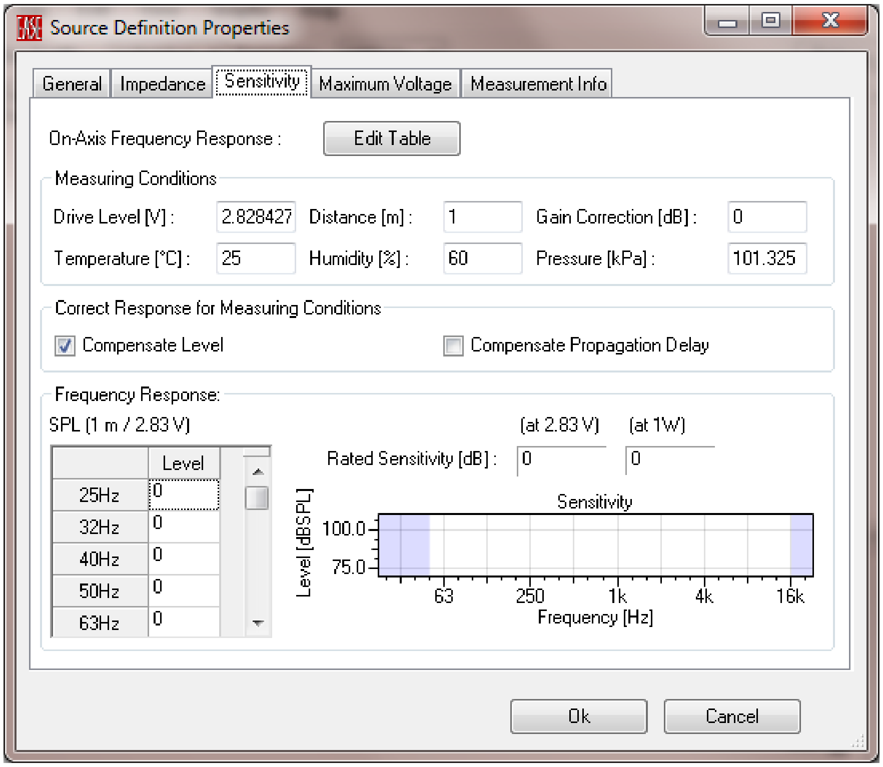
Once the tabular directivity data is edited, it is possible to switch to the Source Definition Properties by selecting Edit > Properties from the main menu or F4. Here again everything is similar to the previously described High Resolution approach, except that the Impedance and Sensitivity tabs each have an Edit Table button instead of a Load File button.



Frequency Response :

Rated Impedance [Ohm] : 8

Figure 2-93 – GSS Impedance tab



On-Axis Frequency Response :

Measuring Conditions

Drive Level [V] : 2.828427 Distance [m] : 1 Gain Correction [dB] : 0

Temperature [°C] : 25 Humidity [%] : 60 Pressure [kPa] : 101.325

Correct Response for Measuring Conditions

☑ Compensate Level ☐ Compensate Propagation Delay

Frequency Response:

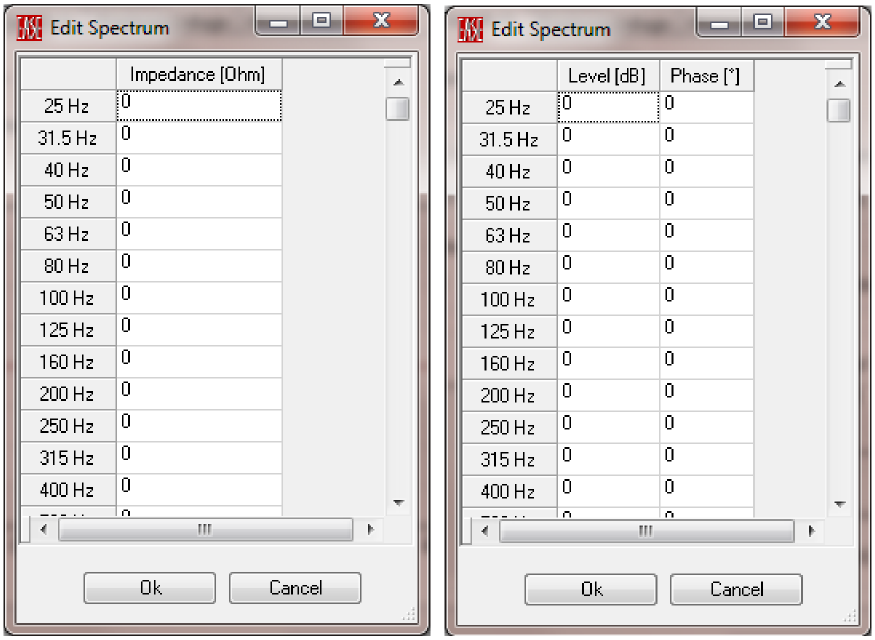
SPL (1 m / 2.83 V)

	Level
25Hz	0
32Hz	0
40Hz	0
50Hz	0
63Hz	0

	(at 2.83 V)	(at 1W)
Rated Sensitivity [dB] :	0	0

Figure 2-94 – GSS Sensitivity tab

Clicking the EDIT TABLE button allows the Impedance and Sensitivity data to be entered into the tables. As with other tables in ***SpeakerLab*** it is possible to copy multiple cells from a spreadsheet and copy the data into the table.



	Impedance [Ohm]
25 Hz	0
31.5 Hz	0
40 Hz	0
50 Hz	0
63 Hz	0
80 Hz	0
100 Hz	0
125 Hz	0
160 Hz	0
200 Hz	0
250 Hz	0
315 Hz	0
400 Hz	0

	Level [dB]	Phase [°]
25 Hz	0	0
31.5 Hz	0	0
40 Hz	0	0
50 Hz	0	0
63 Hz	0	0
80 Hz	0	0
100 Hz	0	0
125 Hz	0	0
160 Hz	0	0
200 Hz	0	0
250 Hz	0	0
315 Hz	0	0
400 Hz	0	0

Figure 2-95 – Data table entry for impedance and sensitivity

While it is possible to define a new source using tabular data, this is not recommended. The tabular data approach can be useful when converting old speaker models, such as the ***EASE 4.2*** and older ***SpeakerBase*** SPK data, into the new Generic Sound Source format. However, this will have all of the modeling accuracy limitations as the older SPK format. A better solution would be to go back to the original source data used to generate the .spk file. If possible, this data should be used to create a new high resolution GSS format source model.

It may also be feasible to use the tabular data format for academic studies where artificial sources or only roughly known sources are to be investigated.

As an example we will import an SPK ***SpeakerBase*** library model into the Source Editor. Click on FILE > IMPORT from the Source Editor main menu.

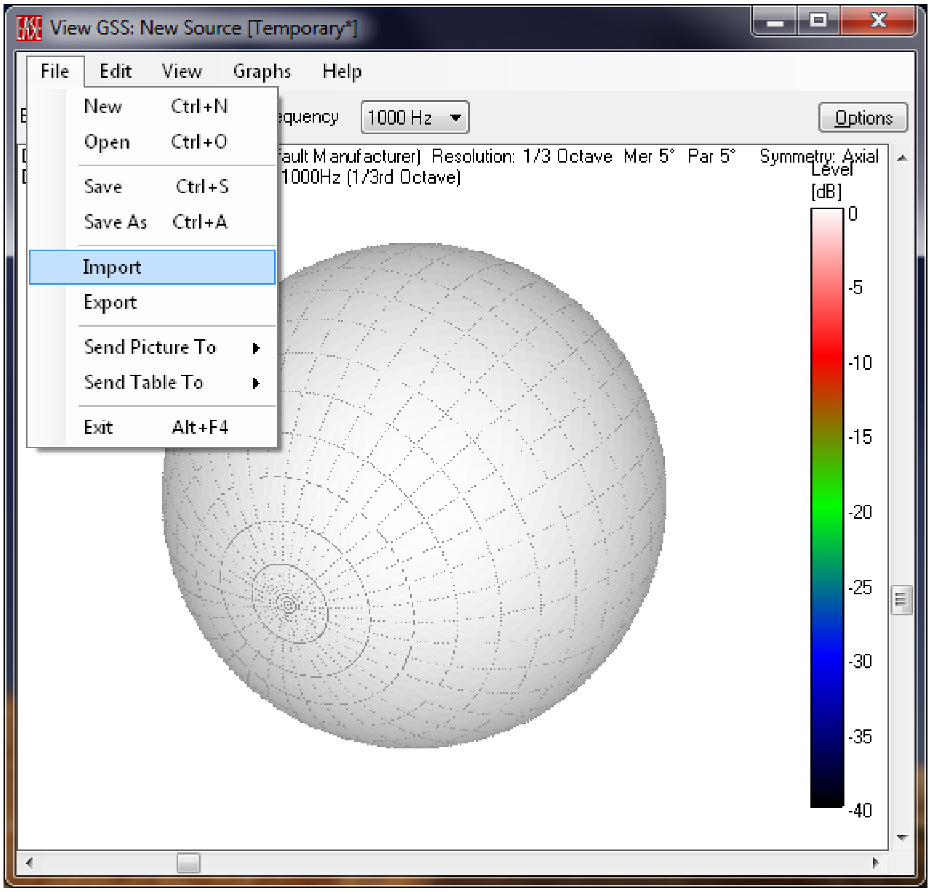


Figure 2-96 – Importing a directivity data file

This will open an IMPORT SOURCE DEFINITION dialog box which is used to select a balloon file.

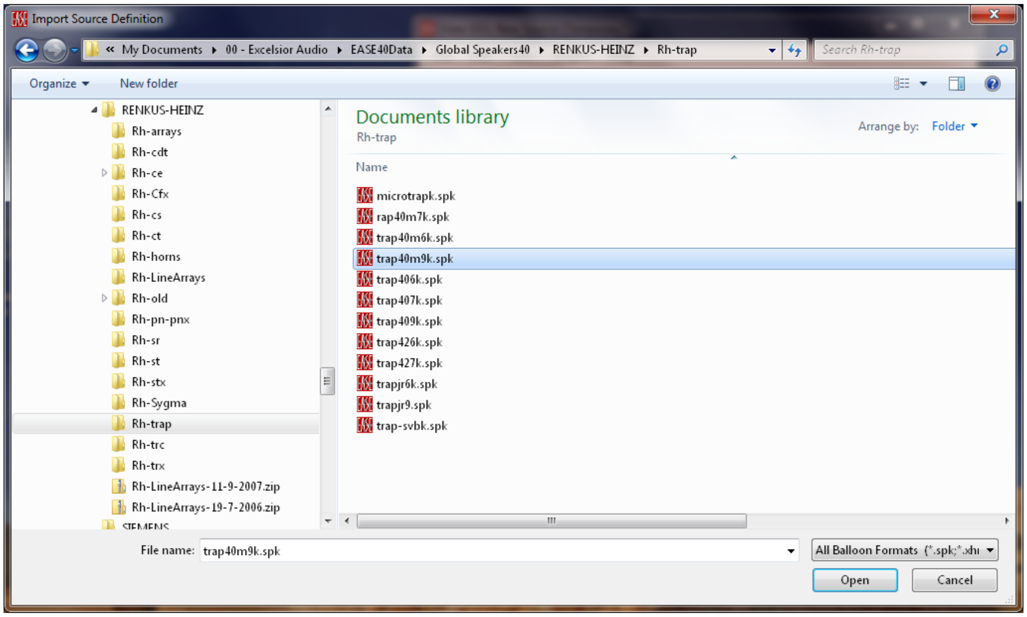
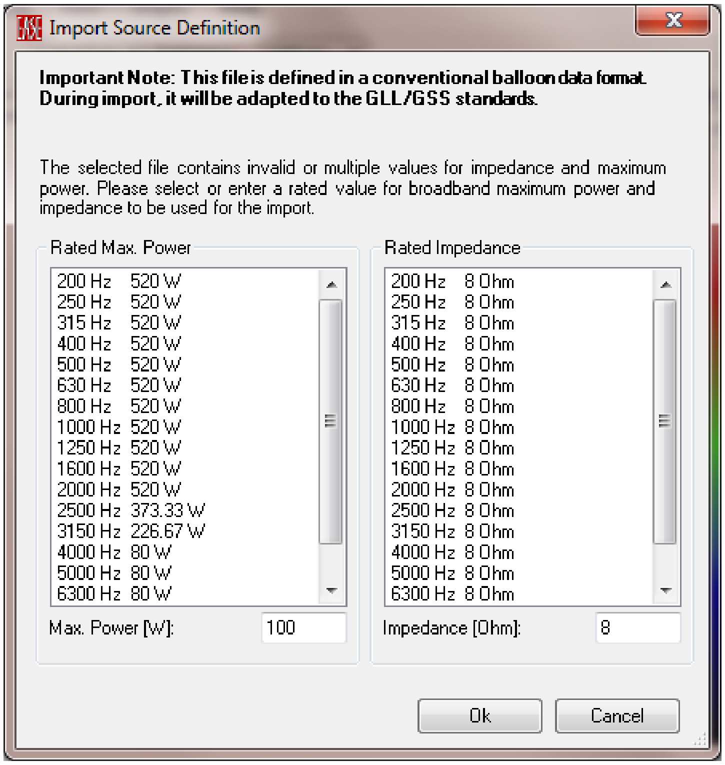


Figure 2-97 – Directivity data file selection

The ***EASE SpeakerLab*** Source Editor can import ***EASE 4.x SpeakerBase*** SPK (.spk) and ASCII XHN (.xhn) loudspeaker data. The ***EASE SpeakerLab*** Pro version also features an import function for other balloon formats: ***CADP2*** GDF (.gdf), ***Ulysses*** UNF (.unf), and ***CLF*** text (.tab) files.

Clicking the Open button will start the import process; an information window may appear if the data parsed from the opened file are not complying with the GSS standard. For the SPK model used in our example the following info is reported.



Important Note: This file is defined in a conventional balloon data format. During import, it will be adapted to the GLL/GSS standards.

The selected file contains invalid or multiple values for impedance and maximum power. Please select or enter a rated value for broadband maximum power and impedance to be used for the import.

Rated Max. Power

200 Hz	520 W
250 Hz	520 W
315 Hz	520 W
400 Hz	520 W
500 Hz	520 W
630 Hz	520 W
800 Hz	520 W
1000 Hz	520 W
1250 Hz	520 W
1600 Hz	520 W
2000 Hz	520 W
2500 Hz	373.33 W
3150 Hz	226.67 W
4000 Hz	80 W
5000 Hz	80 W
6300 Hz	80 W

Max. Power [W]: 100

Rated Impedance

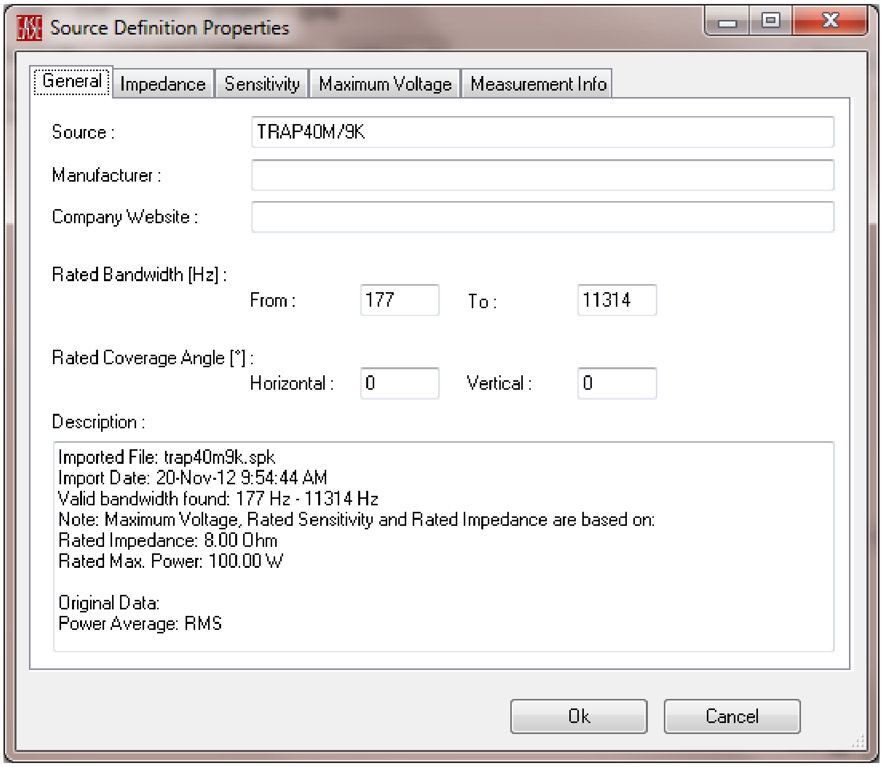
200 Hz	8 Ohm
250 Hz	8 Ohm
315 Hz	8 Ohm
400 Hz	8 Ohm
500 Hz	8 Ohm
630 Hz	8 Ohm
800 Hz	8 Ohm
1000 Hz	8 Ohm
1250 Hz	8 Ohm
1600 Hz	8 Ohm
2000 Hz	8 Ohm
2500 Hz	8 Ohm
3150 Hz	8 Ohm
4000 Hz	8 Ohm
5000 Hz	8 Ohm
6300 Hz	8 Ohm

Impedance [Ohm]: 8

Figure 2-98 – Power handling & impedance for older data file importing

In this specific case the Rated Max. Power was expressed in 1/3 octave bands, while the GSS standard adopts a different approach which specifies a maximum voltage (broadband) and the spectral shape of the signal which is producing this voltage. The Max Power [W] to be used in the GSS should be entered along with the Impedance [Ohm]. These two values will be used to calculate the broadband voltage that will be used for the value of the Program (Vrms 1min) setting in the Maximum Voltages (RMS) section of the GSS.

Once the data is imported the Source Definition Properties dialog box (Edit > Properties or F4) can be used to edit the data, as previously described, and the source model saved as a GSS to be used with GLL loudspeaker models. Each of the tabs contains information from the imported data. The ones shown in Figure 2-99 – Figure 2-102 are representative of importing our .spk file.



Source : TRAP40M/9K

Manufacturer :

Company Website :

Rated Bandwidth [Hz] :
From : 177 To : 11314

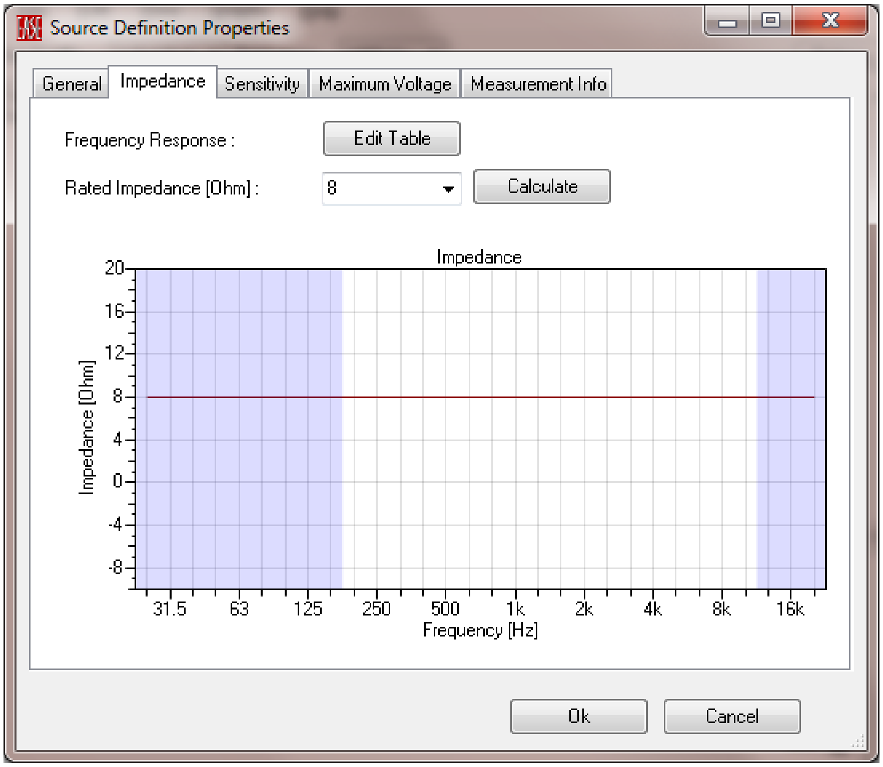
Rated Coverage Angle [°] :
Horizontal : 0 Vertical : 0

Description :

Imported File: trap40m9k.spk
Import Date: 20-Nov-12 9:54:44 AM
Valid bandwidth found: 177 Hz - 11314 Hz
Note: Maximum Voltage, Rated Sensitivity and Rated Impedance are based on:
Rated Impedance: 8.00 Ohm
Rated Max. Power: 100.00 W

Original Data:
Power Average: RMS

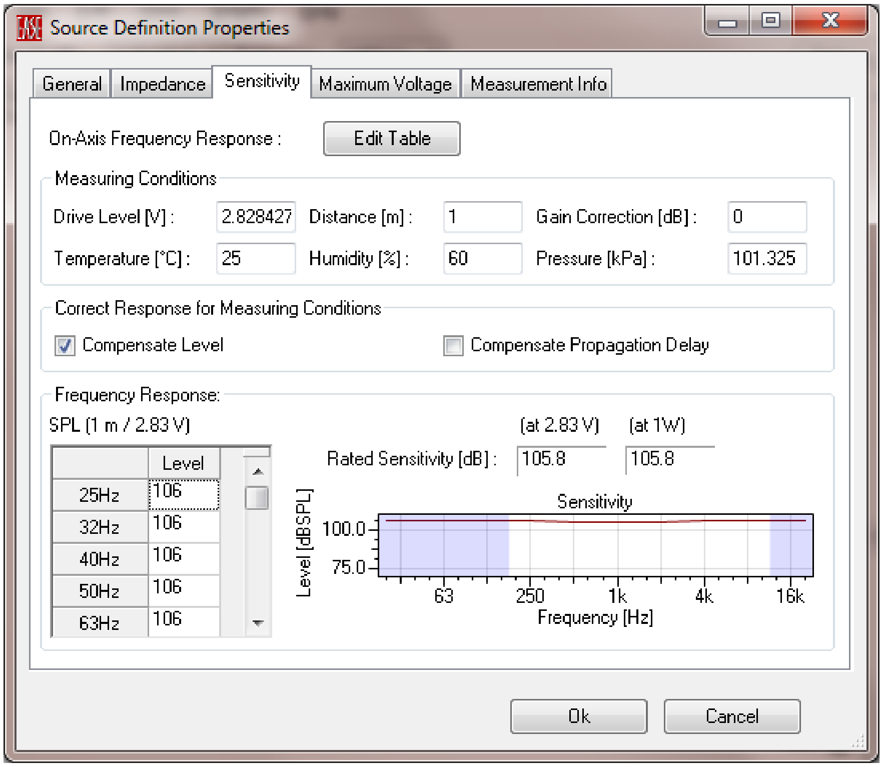
Figure 2-99 – SPK data imported



Frequency Response :

Rated Impedance [Ohm] : 8

Figure 2-100 – SPK data imported



On-Axis Frequency Response :

Measuring Conditions

Drive Level [V] : 2.828427 Distance [m] : 1 Gain Correction [dB] : 0

Temperature [°C] : 25 Humidity [%] : 60 Pressure [kPa] : 101.325

Correct Response for Measuring Conditions

☑ Compensate Level □ Compensate Propagation Delay

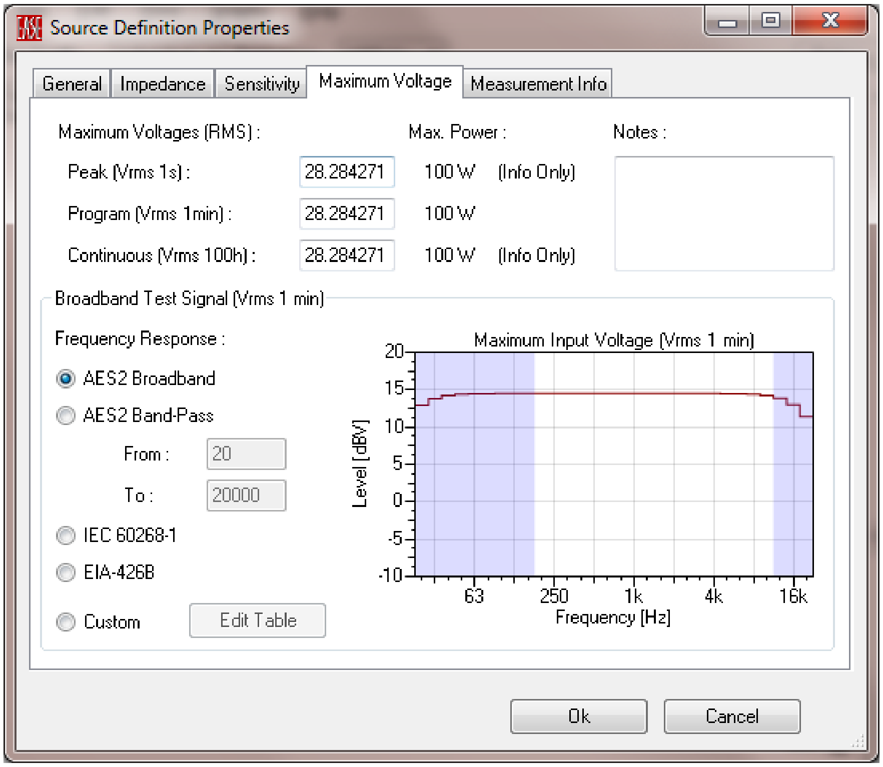
Frequency Response:

SPL (1 m / 2.83 V)

	Level
25Hz	106
32Hz	106
40Hz	106
50Hz	106
63Hz	106

Rated Sensitivity [dB] : (at 2.83 V) 105.8 (at 1W) 105.8

Figure 2-101 – SPK data imported



Maximum Voltages (RMS) : Max. Power : Notes :

Peak (Vrms 1s) : 28.284271 100 W (Info Only)

Program (Vrms 1min) : 28.284271 100 W

Continuous (Vrms 100h) : 28.284271 100 W (Info Only)

Broadband Test Signal (Vrms 1 min)

Frequency Response :

● AES2 Broadband

○ AES2 Band-Pass

From : 20

To : 20000

○ IEC 60268-1

○ EIA-426B

○ Custom

Figure 2-102 – SPK data imported

2.7.11 Exporting Data

The ***EASE SpeakerLab*** Source Editor can export data and graphs in different formats. All of the graphs can be exported as raster and vector images (Pro version only). The 2D graphs (sensitivity, beamwidth, etc.) can also be exported as text files. This can be useful for pasting into spreadsheets for subsequent formatting and comparative analysis. It is also possible to export the directivity data in several different directivity balloon file formats.

Clicking on FILE > SEND PICTURE TO from the main menu will allow an image of the graphs to be sent to the ***Windows*** clipboard, from where it can be pasted into other applications, or saved as a file. When saving a picture the supported file formats are .bmp, .gif, .png, .jpg, and .tif. The vector (Pro version only) .emf format is available only for 2D graphs.

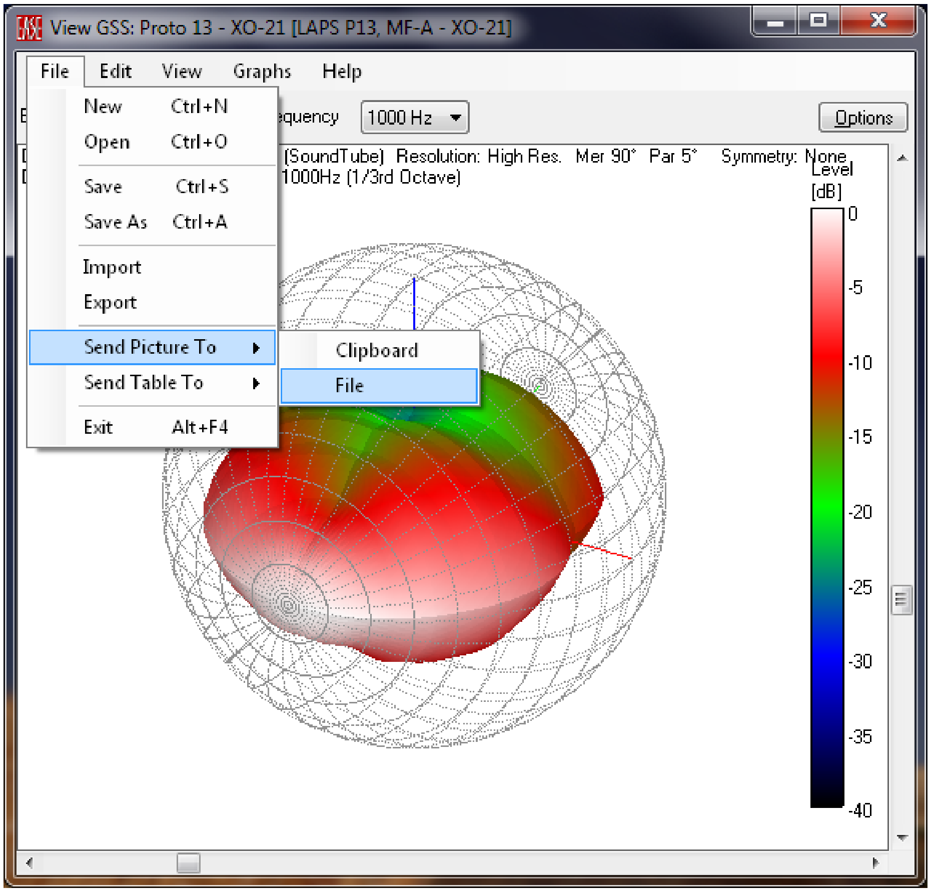


Figure 2-103 – Saving a picture to a file

Clicking on FILE > SEND TABLE TO from the main menu will allow the text data for the 2D graph shown to be sent to the ***Windows*** clipboard or saved as a file. Both simple text (.txt) and binary (.dat) files are supported.

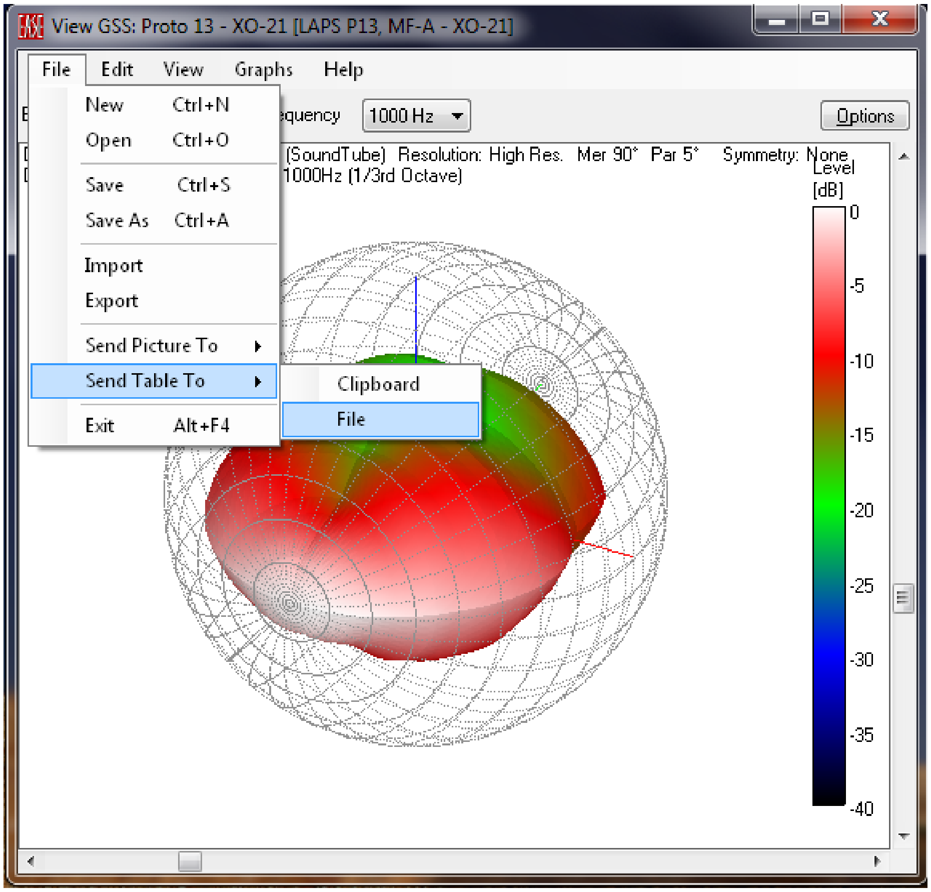


Figure 2-104 – Saving a data table to a file

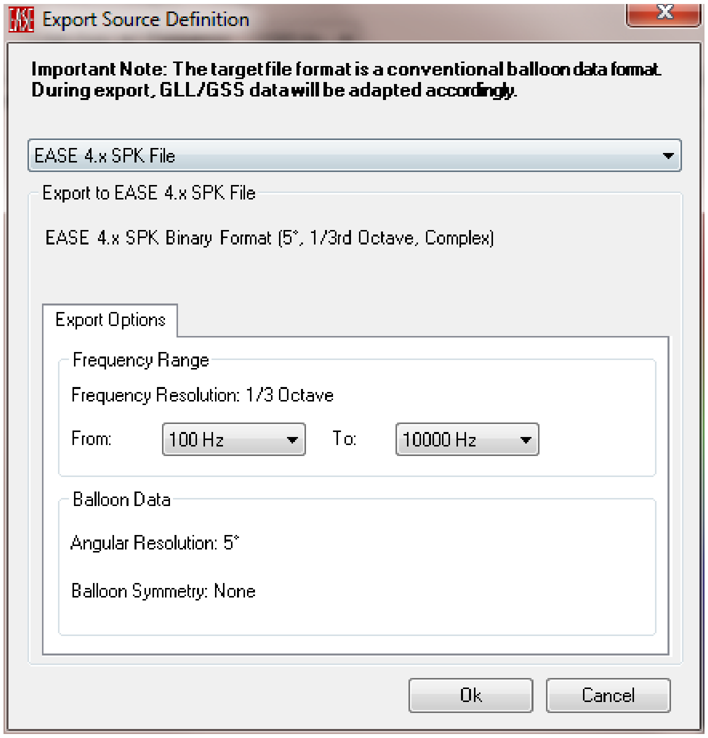
The data in the GSS file can be exported by clicking FILE > EXPORT from the main menu. Note that this is only possible for GSS files that use high resolution data.

Supported formats for export are:

- ***EASE 4.x SpeakerBase*** SPK (.spk) file
- ***EASE*** ASCII XHN (.xhn) file

The Pro version of ***EASE SpeakerLab*** also allows three additional formats:

- ***CADP2*** GDF (.gdf) file
- ***Ulysses*** UNF (.unf) file
- ***CLF*** text V1 and V2 (.tab) file



Export Source Definition

Important Note: The target file format is a conventional balloon data format. During export, GLL/GSS data will be adapted accordingly.

EASE 4.x SPK File

Export to EASE 4.x SPK File

EASE 4.x SPK Binary Format (5°, 1/3rd Octave, Complex)

Export Options

Frequency Range

Frequency Resolution: 1/3 Octave

From: 100 Hz To: 10000 Hz

Balloon Data

Angular Resolution: 5°

Balloon Symmetry: None

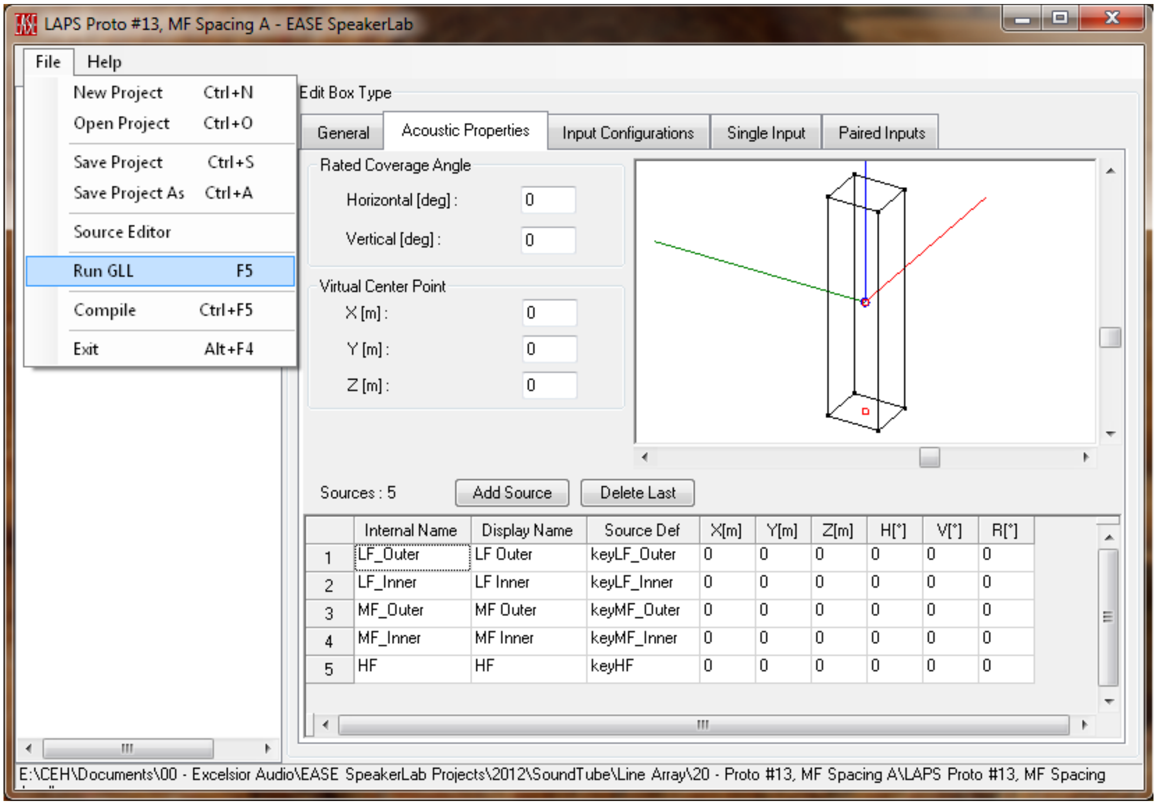
Figure 2-105 – Exporting the data a different file format

The EXPORT SOURCE DEFINITION dialog box lets the user choose the format for the exported file and select export options such as the frequency range. In the case of exporting to the CLF format it is possible to choose between CLF version 1 and 2, and add some auxiliary information.

2.8 Running the GLL

The model can be viewed using FILE > RUN GLL from the main menu or by pressing F5. This will open the ***View GLL*** window with a temporary model, inside of ***SpeakerLab***. It is also possible to open a compiled GLL file outside of ***SpeakerLab*** from the ***EASE GLL Viewer*** directly. Note that if FILE > RUN GLL is used in ***SpeakerLab*** with no project loaded an open file dialog box will appear so a GLL can be selected and loaded.

It is highly recommended to run the GLL prior to closing the XGLL project file. This will check for possible errors in the XGLL project file. Please see section 2.5.1 – Verify the GLL Before Closing for more info.



Rated Coverage Angle

Horizontal [deg] : 0

Vertical [deg] : 0

Virtual Center Point

X [m] : 0

Y [m] : 0

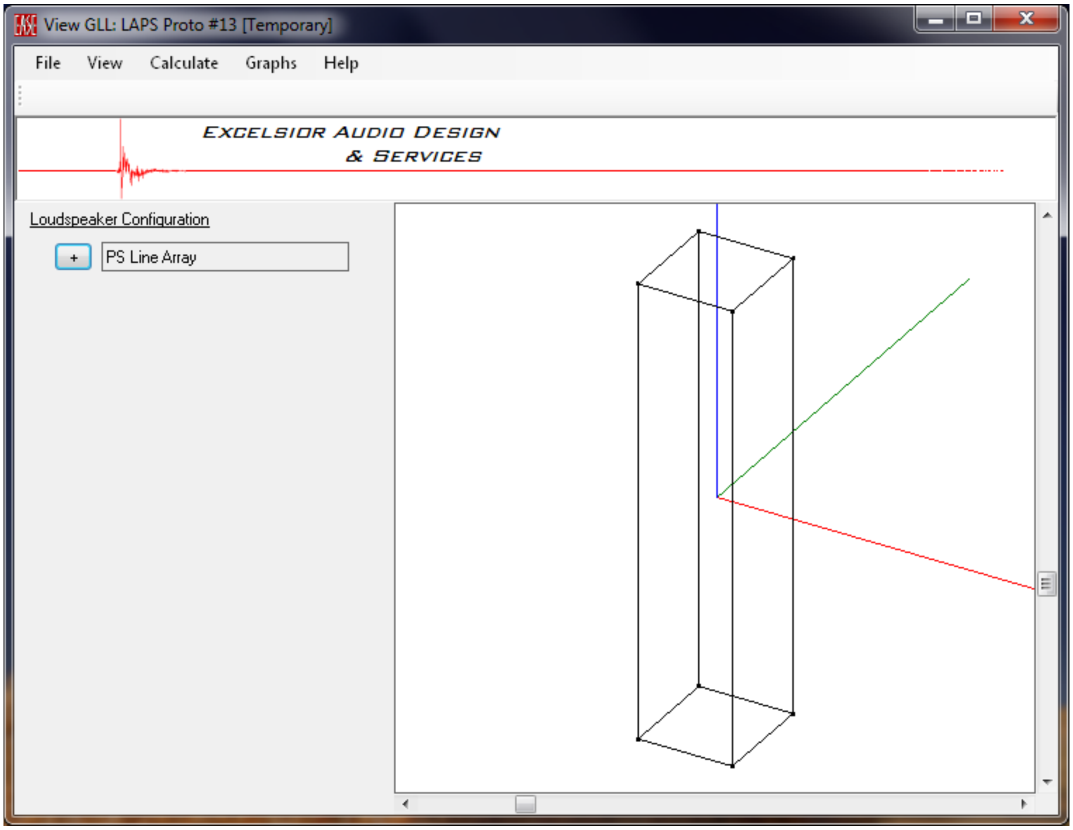
Z [m] : 0

Sources : 5

	Internal Name	Display Name	Source Def	X[m]	Y[m]	Z[m]	H[°]	V[°]	R[°]
1	LF_Outer	LF Outer	keyLF_Outer	0	0	0	0	0	0
2	LF_Inner	LF Inner	keyLF_Inner	0	0	0	0	0	0
3	MF_Outer	MF Outer	keyMF_Outer	0	0	0	0	0	0
4	MF_Inner	MF Inner	keyMF_Inner	0	0	0	0	0	0
5	HF	HF	keyHF	0	0	0	0	0	0

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\SoundTube\Line Array\20 - Proto #13, MF Spacing A\LAPS Proto #13, MF Spacing

Figure 2-106 – Running the GLL


View GLL: LAPS Proto #13 [Temporary]
File
View
Calculate
Graphs
Help
EXCELSIOR AUDIO DESIGN
& SERVICES
Loudspeaker Configuration
+

PS Line Array

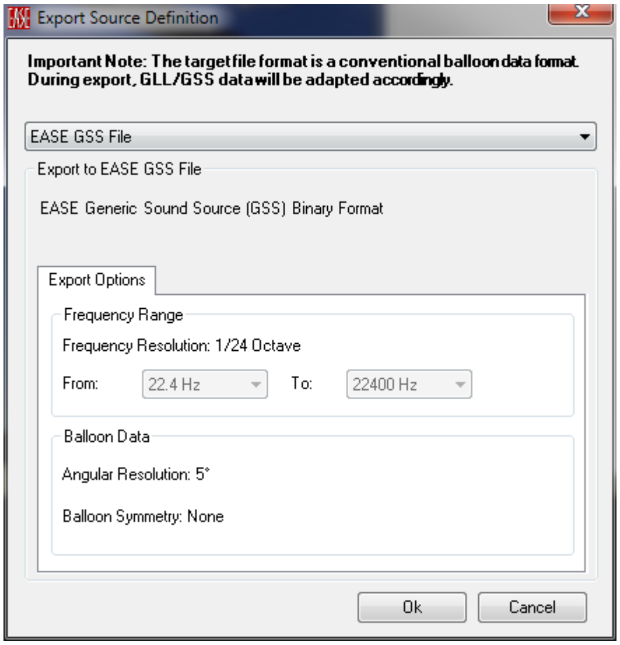
Figure 2-107 – Running the GLL

Running the GLL from within ***SpeakerLab*** allows some additional features that are not available in the stand-alone ***GLL Viewer***. These are described below.

2.8.1 Exporting Results (GSS File Format)

Exporting the calculated results is done just as in the GSS Source Editor. Please see section 2.7.11 – Exporting Data for complete details on this.

However, there is one additional export file format available. When the GLL is run it is possible to also save the results as a GSS (.gss) file. This will allow the calculated directivity data to be used as a single source for other projects. Note however, that the calculated results may only be valid for the specified Calculation Parameters (section 5.1.1) entered. As an example, if the DISTANCE entered is not in the far-field of the loudspeaker system the directivity response and frequency response may change at different distances. It is only in the far field of an acoustical source, or a system of sources, that the on- and off-axis response will no longer change as a function of distance when air attenuation is not considered.


Export Source Definition

Important Note: The target file format is a conventional balloon data format. During export, GLL/GSS data will be adapted accordingly.

EASE GSS File

Export to EASE GSS File

EASE Generic Sound Source (GSS) Binary Format

Export Options

Frequency Range

Frequency Resolution: 1/24 Octave

From: 22.4 Hz To: 22400 Hz

Balloon Data

Angular Resolution: 5°

Balloon Symmetry: None

Figure 2-108 – Exporting the data a different file format

2.8.2 Automatic Recalculation (Pro version only)

The Pro version of ***EASE SpeakerLab*** has a provision to automatically recalculate the GLL when it is run from inside ***SpeakerLab***, and not directly from the ***GLL Viewer***. This can be selected by clicking on CALCULATE > AUTOMATICALLY RECALCULATE from the main menu. Once this is selected, any time there is a change to the GLL parameters (adding a filter, changing the frequency, changing the level, etc.) the GLL will be recalculated and the displayed results updated. This is particularly useful when investigating different setups using various configuration settings which is typical for a loudspeaker design process.

It is recommended that the Calculation Parameters (section 5.1.1) be specified prior to enabling AUTOMATICALLY RECALCULATE.

The Automatically Recalculate feature can be disabled by once again clicking on CALCULATE > AUTOMATICALLY RECALCULATE from the main menu.

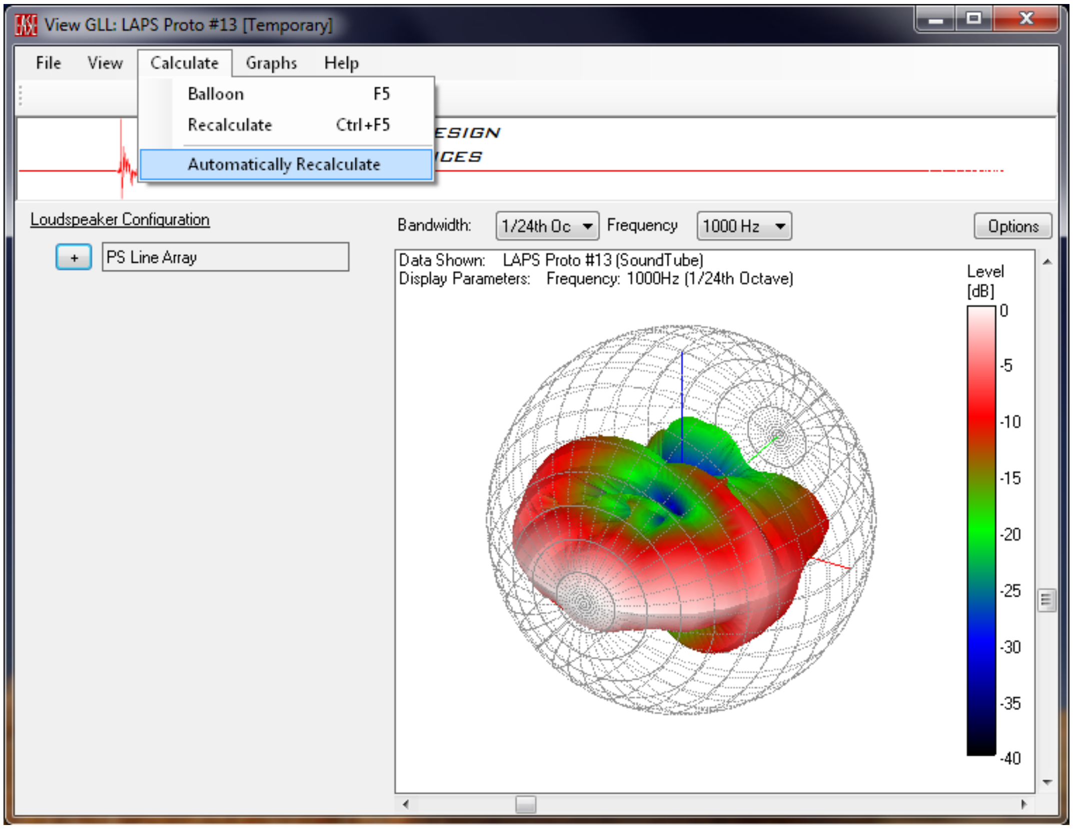

View GLL: LAPS Proto #13 [Temporary]
File
View
Calculate
Graphs
Help
Balloon F5
Recalculate Ctrl+F5
Automatically Recalculate
ESIGN
ICES
Loudspeaker Configuration
+
PS Line Array
Bandwidth: 1/24th Oc
Frequency 1000 Hz
Options
Data Shown: LAPS Proto #13 (SoundTube)
Display Parameters: Frequency: 1000Hz (1/24th Octave)
Level [dB]
0
-5
-10
-15
-20
-25
-30
-35
-40

Figure 2-109 – Automatically recalculating the GLL

2.9 General Data, Part 2

We will now return to some of the other GENERAL DATA items in the property tree of an XGLL project file. These items are not critical for the compiled GLL to run properly. However, they can greatly increase the versatility and improve the experience of those using the GLL.

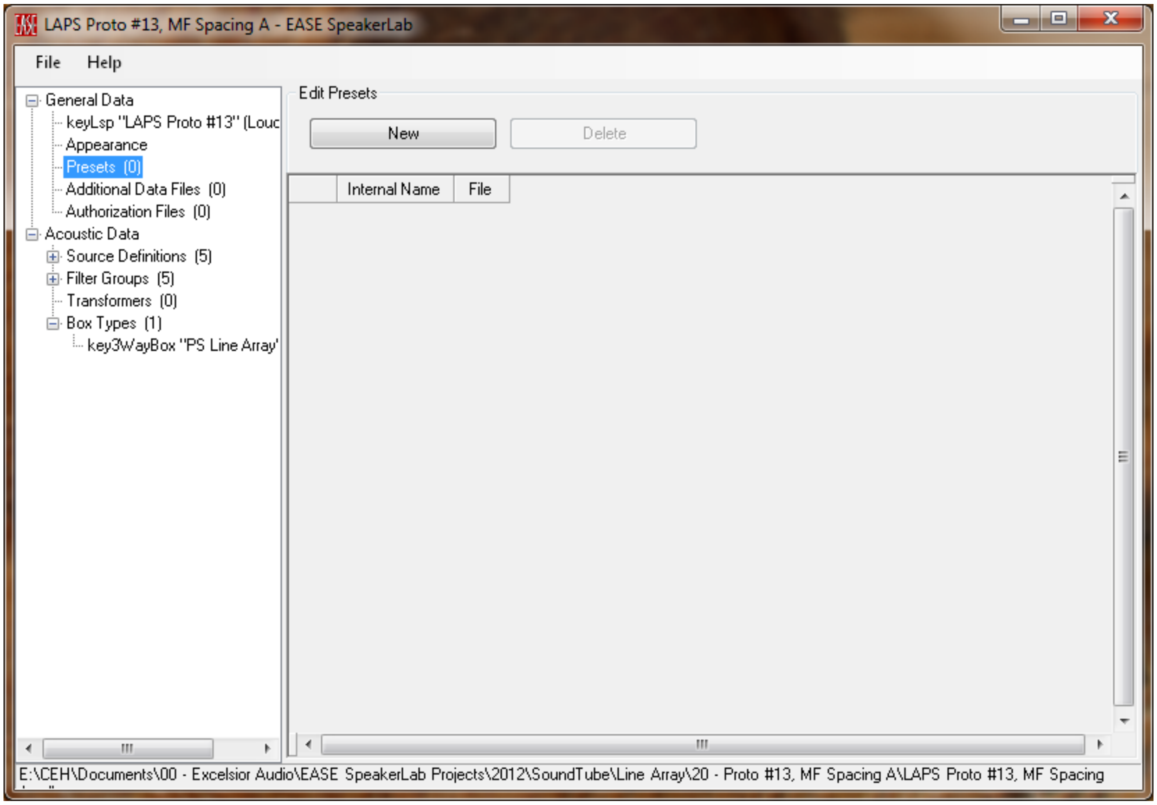
2.9.1 Presets

Presets are an easy way for users to recall system configurations. A preset is basically a configuration file that is embedded into the GLL itself.

To add a preset to the XGLL project, click on Preset in the property tree. The page on the right will show the Edit Preset page. Clicking on the New button will add a slot in which a preset can be specified. Notice also that the New button has also changed to a Duplicate button after there is at least one preset slot.

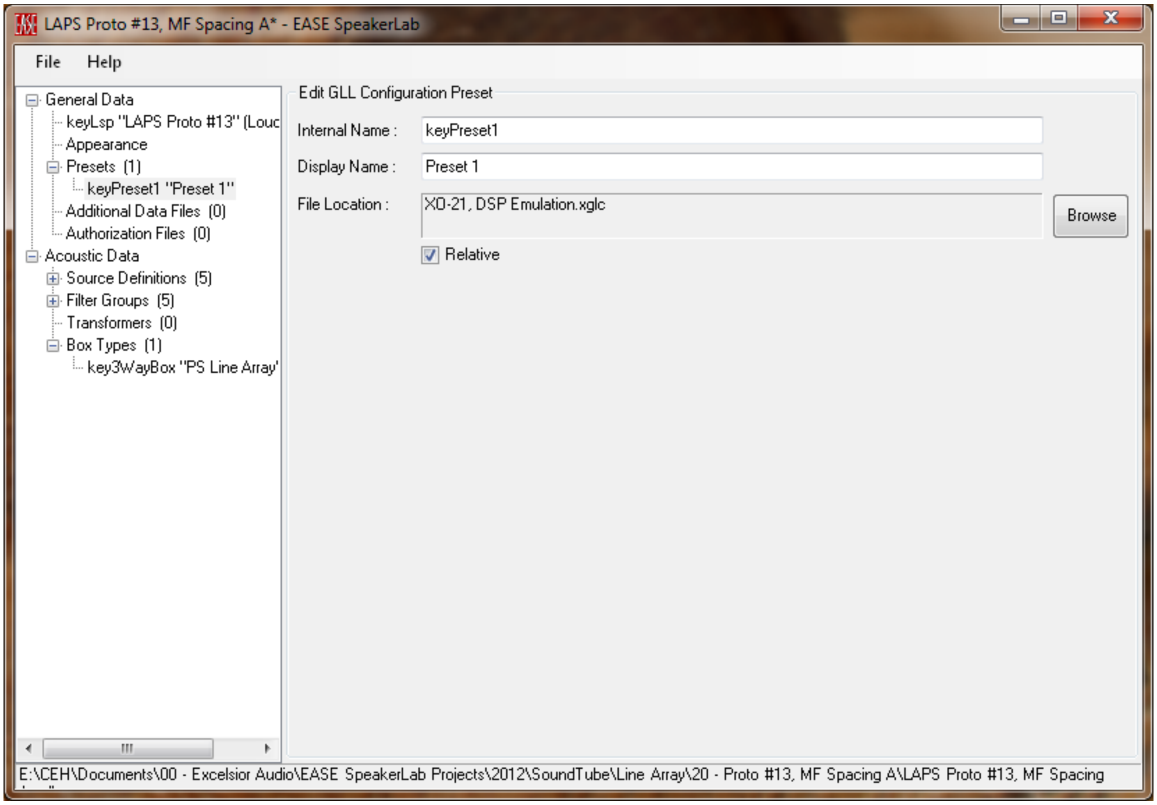
After clicking the New button select keyPreset1 “Preset 1” in the property tree. This will show the Edit GLL Configuration Preset page in the right display area. Here the Internal Name and Display Name can be entered. The Display Name should be descriptive of the preset, but not too long. The Internal Name must be a unique name among all the different presets for the XGLL project.

Clicking on the Browse button opens a Select GLL Configuration File dialog box. This allows a previously saved configuration file (.xglc) to be selected for the preset. Select the desired file and click OK. The complete path to the selected file is shown in the File Location box. Checking the Relative box will change this from an absolute path to a relative path. If the guidelines in Section 6.1 – Folder Structure – Keeping Everything Together are followed it is recommended to always use relative paths.



E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\SoundTube\Line Array\20 - Proto #13, MF Spacing A\LAPS Proto #13, MF Spacing

Figure 2-110 – Entering presets for the GLL



Edit GLL Configuration Preset

Internal Name : keyPreset1

Display Name : Preset 1

File Location : XO-21, DSP Emulation.xglc

☑ Relative

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\SoundTube\Line Array\20 - Proto #13, MF Spacing A\LAPS Proto #13, MF Spacing

Figure 2-111 – Naming the preset & specifying the configuration file for the preset

When a preset is included in a GLL the first preset is the default configuration for the GLL when it is first opened or used. If more than one preset is included in the GLL the others must be selected by the end user (Section 5.1.8 – Selecting Presets) to have the system configuration changed to that preset.

More than one preset can be included in the GLL. To add more presets an existing preset must first be duplicated. This is done basically to add another slot in which to put the new preset. Select the PRESETS item in the property tree on the left. In the EDIT PRESETS page on the right select the preset to be duplicated. It really doesn’t matter which one is used. Now select the newly added preset in the property tree on the left and change the names and the selected file in the EDIT GLL CONFIGURATION PRESET on the right as described above.

Presets can also be deleted if desired. To delete a preset select PRESETS from the property tree. In the table on the EDIT PRESETS page select the preset to be deleted. Click on the DELETE button to delete it. All of the presets can be deleted if desired.

If more than one preset has been added to the XGLL project but the first one is not the one that should be the initial default configuration for the GLL there is a way to reorder the presets. This must be done in a text editor, outside of ***SpeakerLab***, however. Open the .xgll file in a text editor and cut the preset that should be the default. Copy this preset to the first one. See Appendix A – XGLL Project File – Text File Organization for complete details on the format of the .xgll file format.

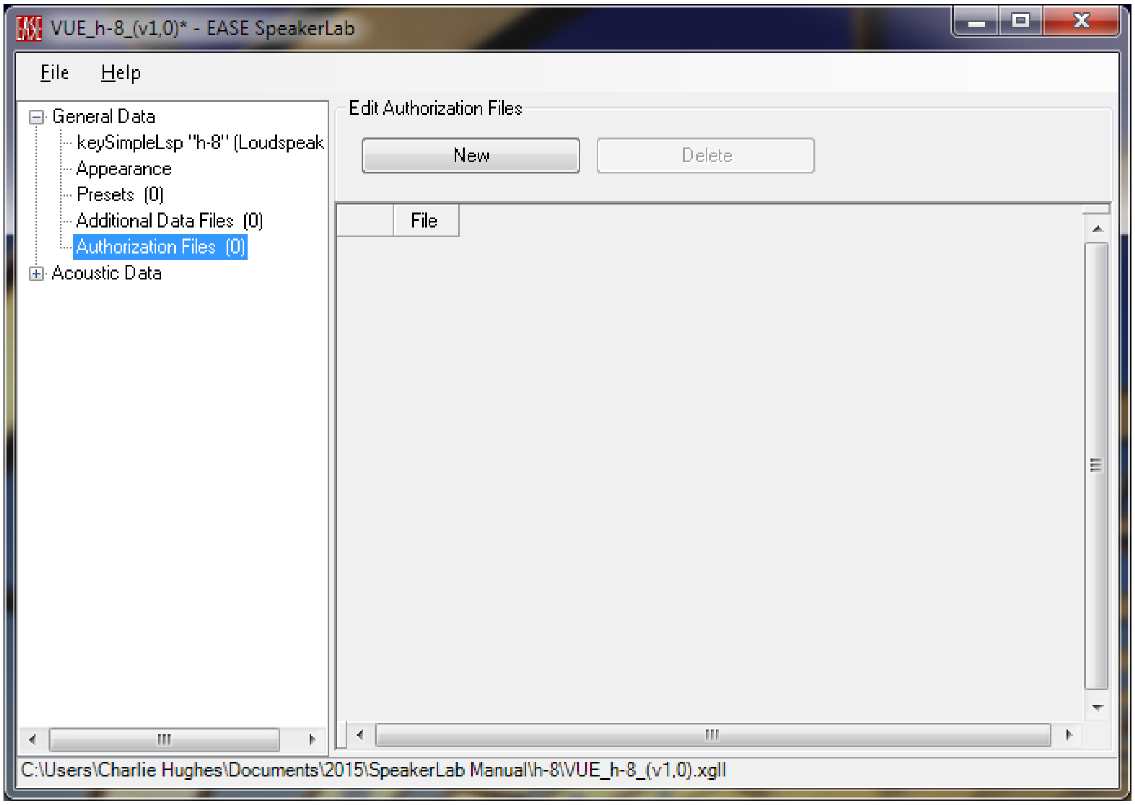
2.9.2 Data Authorization – the .author file

The AUTHORIZATION FILES item on the property tree allows manufacturers, as well as independent test labs and GLL creation services, to add their .author file to the XGLL project. When a manufacturer's .author file is included in the GLL it will show as authorized in ***EASE*** and other AFMG programs. Multiple .author files can be included in a GLL. For example a measurement lab and the GLL author may have included their .author files along with the manufacturer's.

It is important to understand that in order for the GLL to show as authorized the name entered in the MANUFACTURER box on the EDIT GENERAL SYSTEM PROPERTIES page (Section 2.4 – General Data) must match exactly the name in the .author file issued to the manufacturer.

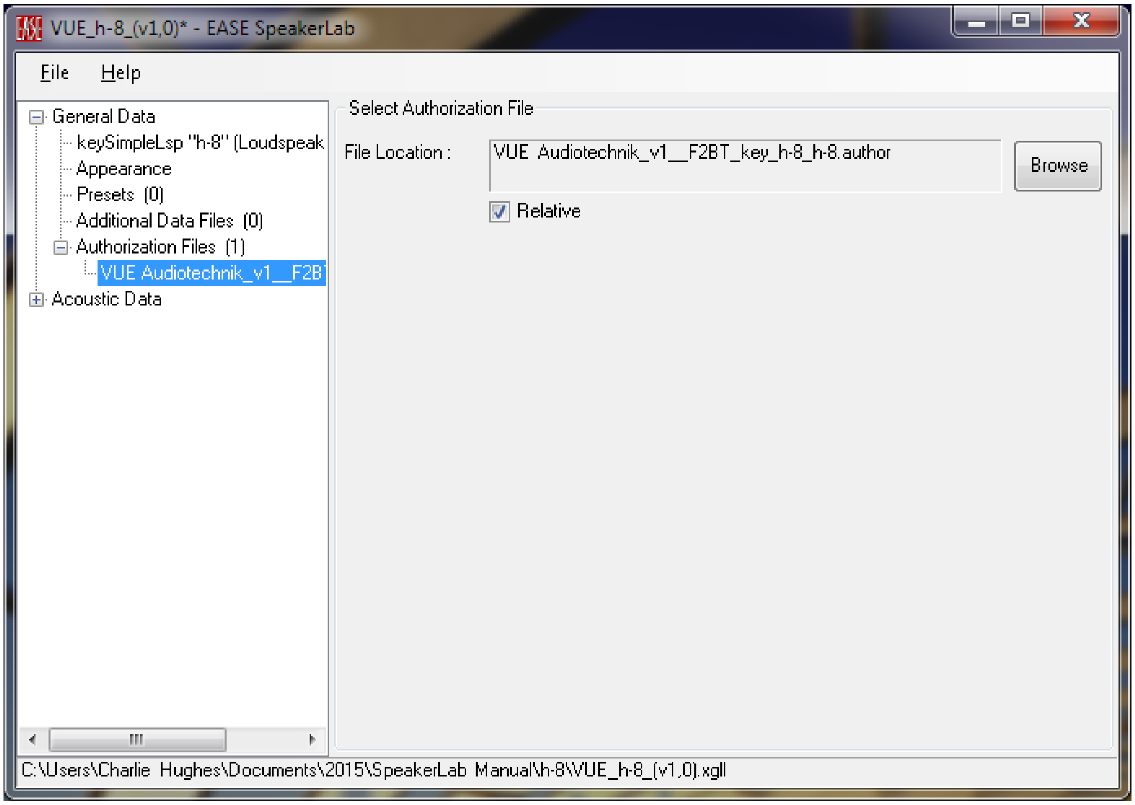
Those wanting to obtain an .author file should contact AFMG Support at support@AFMG.eu to request the file. When requesting the .author file please include the name exactly as it should appear in the GLL. An .author file is available at no cost.

To add an authorization file select AUTHORIZATION FILES from the property tree and then click the NEW button in the EDIT AUTHORIZATION FILES page. Next, click the + to the right of the AUTHORIZATION FILES in the property tree and select UNDEFINED. This will show the SELECT AUTHORIZATION FILE page on the right side of the display. Click on the BROWSE button to open a SELECT AUTHORIZATION FILE TO INCLUDE dialog box. Browse to the desired .author file and select it. Clicking the OK button will close the dialog box and show the absolute path to the selected .author file in the display box to the left of the BROWSE button. Checking the RELATIVE box will change this from an absolute path to a relative path. If the guidelines in Section 6.1 – Folder Structure – Keeping Everything Together are followed it is recommended to always use relative paths.



C:\Users\Charlie Hughes\Documents\2015\SpeakerLab Manual\h-8\VUE_h-8_(v1.0).xgll

Figure 2-112 – Authorization File in the property tree

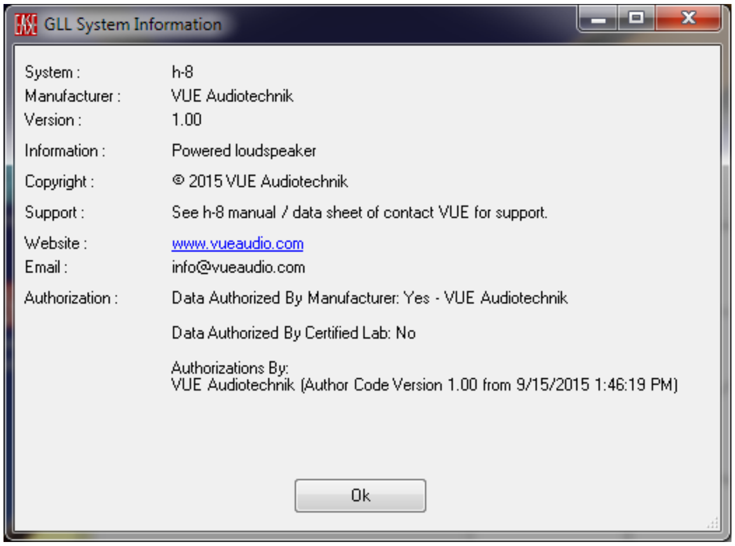

VUE_h-8_(v1.0)* - EASE SpeakerLab
File
Help
General Data
keySimpleLsp "h-8" (Loudspeak
Appearance
Presets (0)
Additional Data Files (0)
Authorization Files (1)
VUE Audiotechnik_v1__F2B
Acoustic Data
Select Authorization File
Browse

File Location : VUE Audiotechnik_v1__F2BT_key_h-8_h-8.author

☑ Relative

C:\Users\Charlie Hughes\Documents\2015\SpeakerLab Manual\h-8\VUE_h-8_(v1.0).xgll

Figure 2-113 – Specifying the authorization file



System :	h-8
Manufacturer :	VUE Audiotechnik
Version :	1.00
Information :	Powered loudspeaker
Copyright :	© 2015 VUE Audiotechnik
Support :	See h-8 manual / data sheet of contact VUE for support.
Website :	www.vueaudio.com
Email :	info@vueaudio.com
Authorization :	Data Authorized By Manufacturer: Yes - VUE Audiotechnik Data Authorized By Certified Lab: No Authorizations By: VUE Audiotechnik (Author Code Version 1.00 from 9/15/2015 1:46:19 PM)

Figure 2-114 – Authorization file details shown in the Info dialog box

Authorization files can also be deleted if desired. To delete an authorization file select Authorization Files from the property tree. In the table on the Edit Authorization Files page select the authorization file to be deleted. Click on the Delete button to delete it. All of the authorization files can be deleted if desired.

The authorization file also serves another purpose. Some AFMG programs require the manufacturer to purchase a license in order to be able to use their GLL in a specific program. This is the case with both ***EASE Address*** and ***EASE Focus***. These programs are free to the end user to download and install. As such the manufacturer helps to pay the cost of these programs. Other programs, such as FIRmaker, also require that the manufacturer purchase a special license. However, these programs are not freely downloadable by end users. Still other programs, such as ***EASE*** and ***EASE Evac***, are paid for by each end user. These programs do not require manufacturers to purchase a license in order to use their GLLs with them.

To obtain information about licensing GLLs for use with ***EASE Address*** or ***EASE Focus*** please contact AFMG by emailing support@AFMG.eu.

2.9.3 Appearance

Selecting APPEARANCE on the property tree changes the right side of the display to show the EDIT APPEARANCE DETAILS page. Here it is possible to specify the BACKGROUND COLOR for the artwork heading of the GLL as well as select the image files to be displayed. To select the image files click on the BROWSE button for either the MAIN LOGO or SECONDARY LOGO. This will open A SELECT LOGO FILE dialog box. Browse to the location of the image file and select it. Clicking on OK will close the dialog box and the path to the file will be shown in the text box to the left of the BROWSE button.

The RELATIVE checkbox will change the path from an absolute path to a path relative to the directory in which the .xgll project file is located.

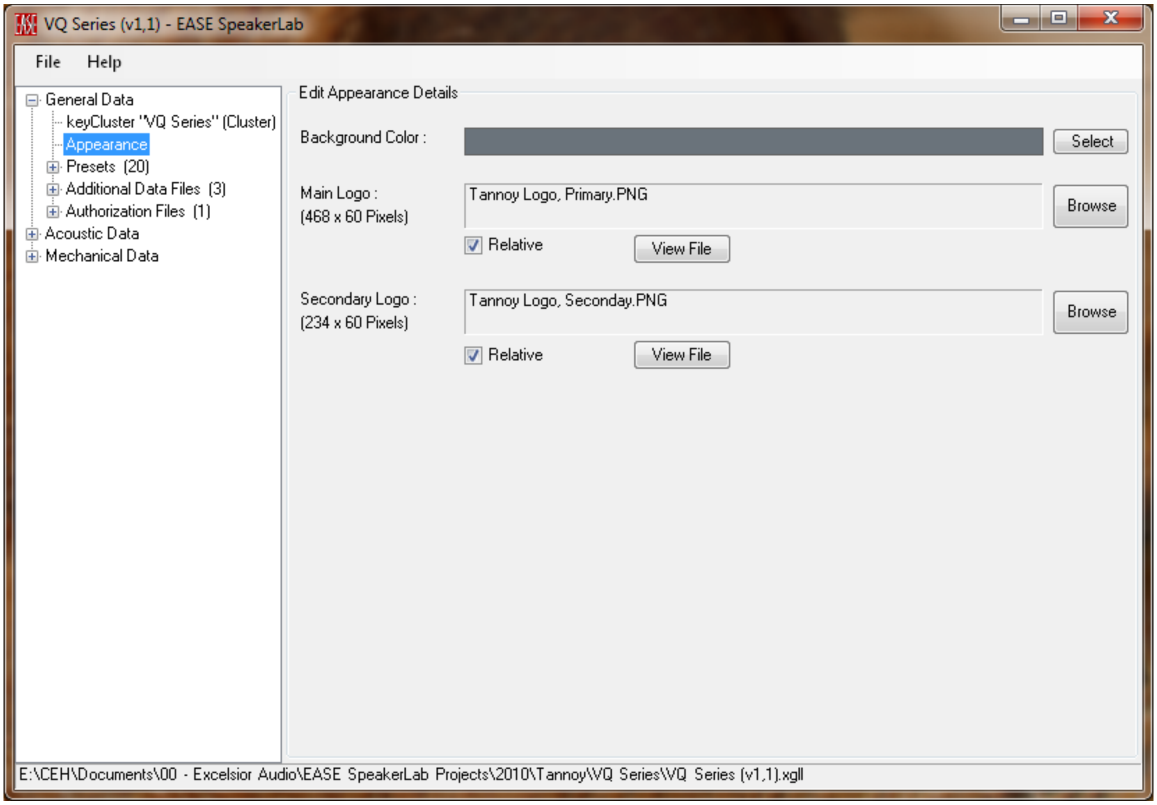
The VIEW FILE button will open the selected image file for viewing.

The Background Color can be chosen by clicking the SELECT button which will open a standard WINDOWS COLOR SELECTION dialog box. Select the color desired or click on the DEFINE CUSTOM COLORS button. The will expand the box to show a color map and luminance bar. There are also text boxes where the Hue, Saturation, and Luminance or the RGB values can be entered directly. This can be very useful for matching the background color in the Secondary Logo file to give the appearance of a seamless transition. Click the OK button once the desired color is selected. The BACKGROUND COLOR bar will be updated to show the selected color.

The Main Logo is displayed on the far left-hand side of the GLL artwork heading. It is limited to a maximum size of 468 x 60 pixels. The Secondary Logo is displayed immediately to the right of the Main Logo area of the GLL artwork heading. If the Main Logo is less than 468 pixels wide there will be a gap between the Main Logo and the Secondary Logo. The Secondary Logo is limited to a maximum size of 234 x 60 pixels. The supported image files are .gif, .jpg, and .png. If the logos are left blank, or if they do not completely fill the pixel area reserved for them, the GLL artwork heading will be white in these areas.

The selected Background Color will be displayed immediately to the right of the Secondary Logo.

A logo wider than 468 pixels can be used with a bit of extra work. The wider logo can be split into two separate files so that one can be used for the Main Logo and the other for the Secondary Logo. An image editing program should be used to split the logo image file. This will allow an image up to a maximum size of 702 x 60 pixels to be split and used.



Background Color :

Main Logo : (468 x 60 Pixels) Tannoy Logo, Primary.PNG

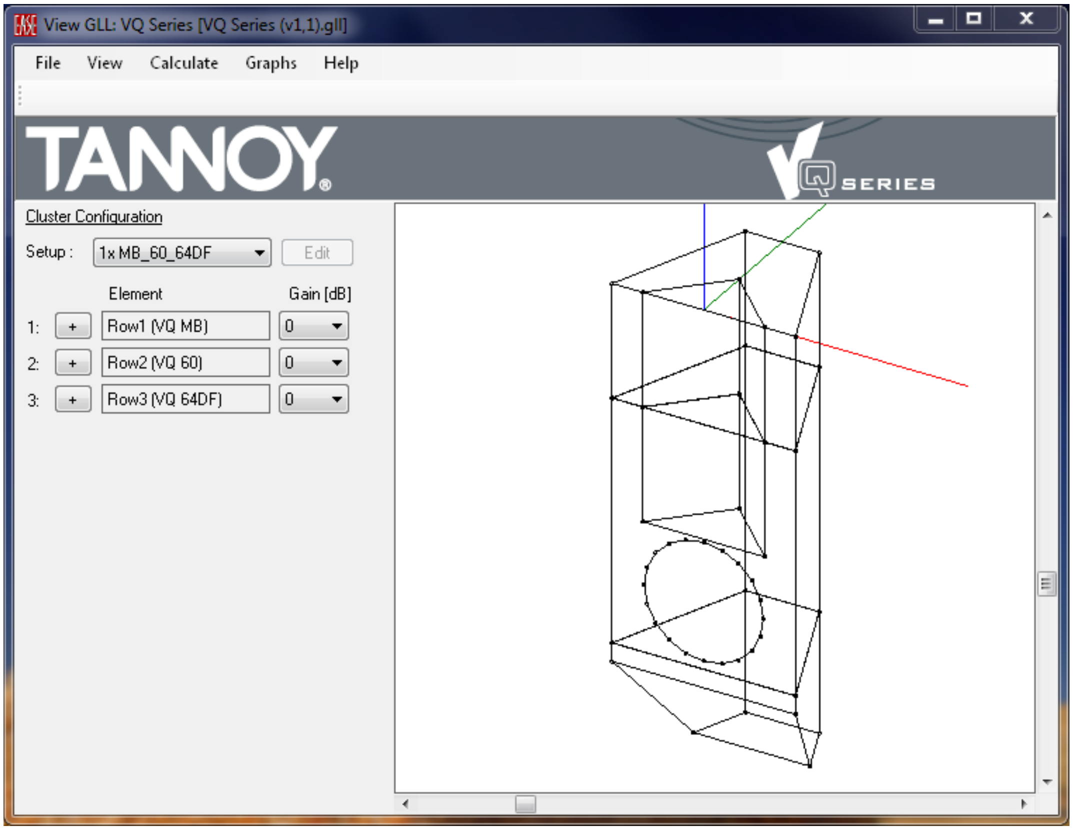
☑ Relative

Secondary Logo : (234 x 60 Pixels) Tannoy Logo, Seconday.PNG

☑ Relative

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2010\Tannoy\VQ Series\VQ Series (v1,1).xgll

Figure 2-115 – Specifying the artwork logos and the background color for the top banner of the GLL



View GLL: VQ Series [VQ Series (v1,1).gll]
File View Calculate Graphs Help
TANNOY
SERIES
Cluster Configuration
Edit
Element
Gain [dB]

Setup : 1x MB_60_64DF

1: + Row1 (VQ MB) 0

2: + Row2 (VQ 60) 0

3: + Row3 (VQ 64DF) 0

Figure 2-116 – Running GLL showing the banner at the top with the specified artwork logos and background color

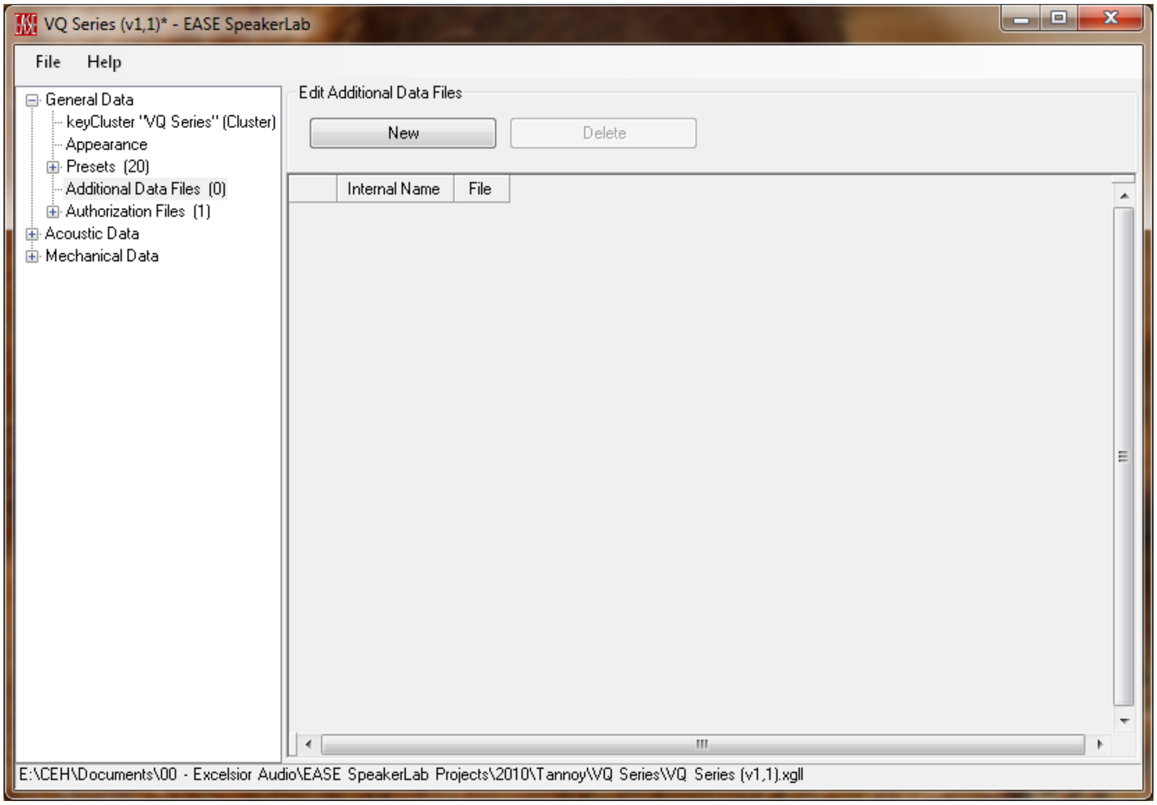
2.9.4 Additional Data Files – Embedding Files

The last item in the GENERAL DATA section of the property tree we will cover is ADDITIONAL DATA FILES. This can be a very useful feature of the GLL. It allows any file to be embedded inside the GLL. An end user can extract any of the embedded files. A PDF file for a user's manual or spec sheet, instructions for using the GLL, photos or line drawings, even 3D CAD files can be embedded. Remember that the size of the embedded files will correspondingly increase the size of the GLL once it is compiled.

To add an additional data file to the XGLL project, select ADDITIONAL DATA FILES in the property tree. The right side of the display area will show the SELECT ADDITIONAL DATA FILE page. Clicking on the NEW button will add a slot in which an additional data file can be specified. Notice also that the NEW button has also changed to a DUPLICATE button after there is at least one additional data file slot.

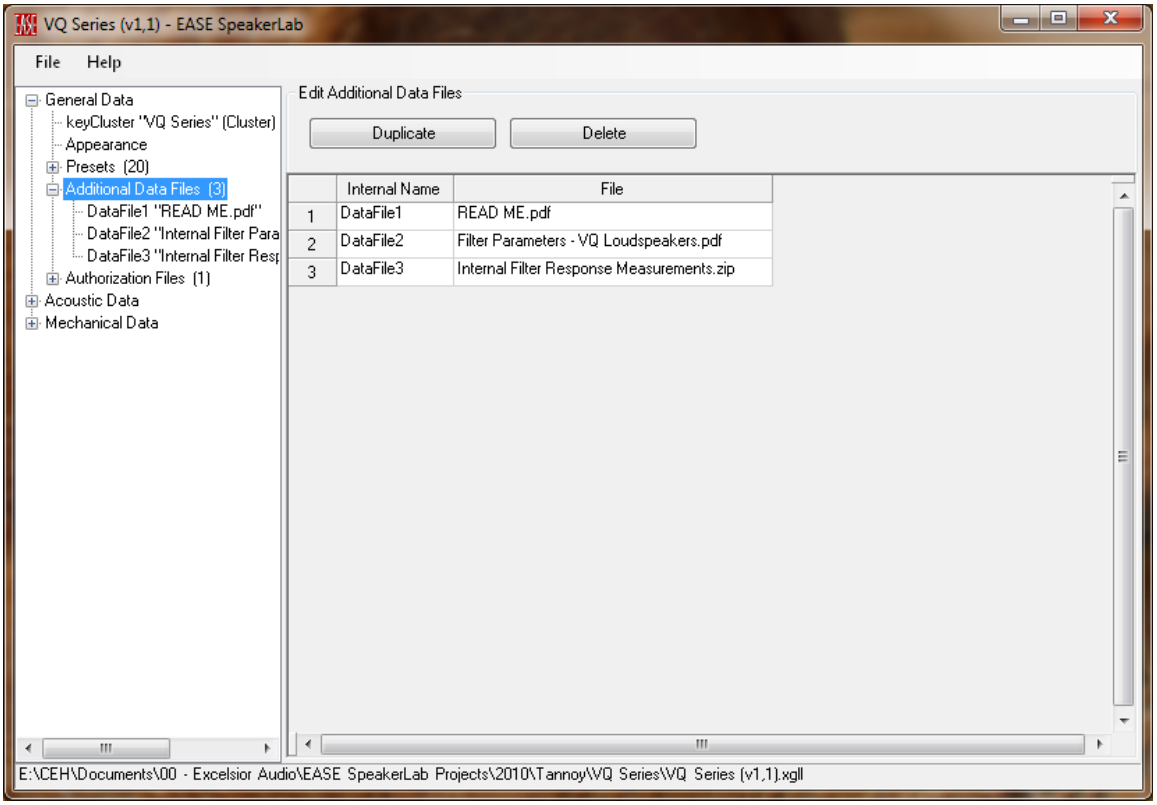
After clicking the NEW button select DATAFILE1 "DATA FILE 1" in the property tree. This will show the SELECT ADDITIONAL DATA FILE page in the right display area. Here the INTERNAL NAME and DISPLAY NAME can be entered. The DISPLAY NAME should be descriptive of the additional data file so that end users will know what it is. The INTERNAL NAME must be a unique name among all the different additional data files for the XGLL project. It can be left as DataFile*X* if desired*.*

Clicking on the BROWSE button opens a SELECT FILE TO BE INCLUDED dialog box. This allows any file to be selected for embedding. Select the desired file and click OK. The complete path to the selected file is shown in the FILE LOCATION box. Checking the RELATIVE box will change this from an absolute path to a relative path. If the guidelines in section 6.1 – Folder Structure – Keeping Everything Together are followed it is recommended to always use relative paths.



E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2010\Tannoy\VQ Series\VQ Series (v1,1).xgll

Figure 2-117 – Additional data files can be added


VQ Series (v1,1) - EASE SpeakerLab
File
Help
General Data
keyCluster "VQ Series" (Cluster)
Appearance
Presets (20)
Additional Data Files (3)
DataFile1 "READ ME.pdf"
DataFile2 "Internal Filter Para
DataFile3 "Internal Filter Resp
Authorization Files (1)
Acoustic Data
Mechanical Data
Edit Additional Data Files
Duplicate
Delete

	Internal Name	File
1	DataFile1	READ ME.pdf
2	DataFile2	Filter Parameters - VQ Loudspeakers.pdf
3	DataFile3	Internal Filter Response Measurements.zip

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2010\Tannoy\VQ Series\VQ Series (v1,1).xgll

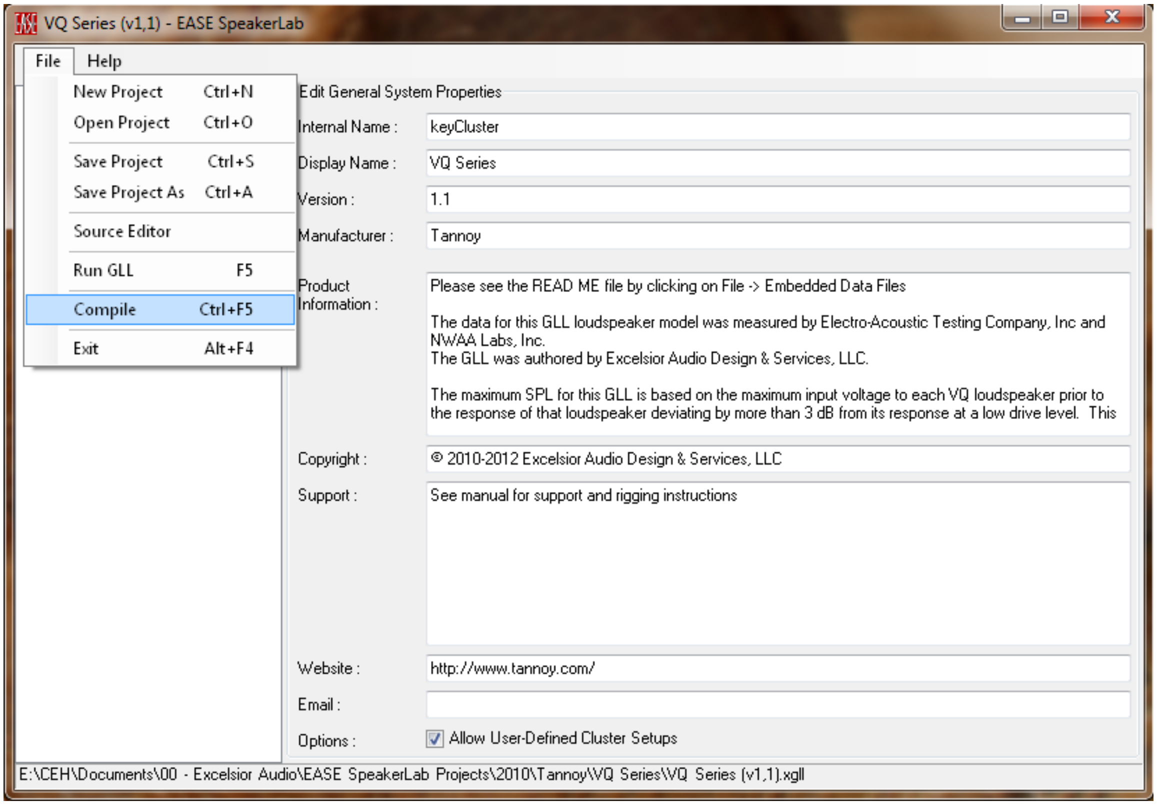
Figure 2-118 – Additional data files added to the GLL project file

More than one additional data file can be included in the GLL. To add more additional data files an existing additional data file must first be duplicated. This is done basically to add another slot in which to put the new additional data file. Select the Additional Data Files item in the property tree on the left. In the Edit Additional Data Files page on the right select the data file to be duplicated. It really doesn’t matter which one is used. Now select the newly added Data File in the property tree on the left and change the Display Name and select the file in the Edit Additional data files on the right as described above.

Additional data files can also be deleted if desired. To delete an additional data file select Additional data files from the property tree. In the table on the Edit Additional Data Files page select the data file to be deleted. Click on the Delete button to delete it. All of the additional data files can be deleted if desired.

2.10 Compiling the GLL

To redistribute the GLL model for use in ***EASE***, ***EASE Address***, ***EASE Evac***, and ***EASE Focus*** the project must be compiled in a GLL binary file. To do this click on File > Compile from the menu or use the Ctrl+F5 shortcut.



Edit General System Properties

Internal Name : keyCluster

Display Name : VQ Series

Version : 1.1

Manufacturer : Tannoy

Product Information : Please see the READ ME file by clicking on File -> Embedded Data Files

The data for this GLL loudspeaker model was measured by Electro-Acoustic Testing Company, Inc and NWAA Labs, Inc.
The GLL was authored by Excelsior Audio Design & Services, LLC.

The maximum SPL for this GLL is based on the maximum input voltage to each VQ loudspeaker prior to the response of that loudspeaker deviating by more than 3 dB from its response at a low drive level. This

Copyright : © 2010-2012 Excelsior Audio Design & Services, LLC

Support : See manual for support and rigging instructions

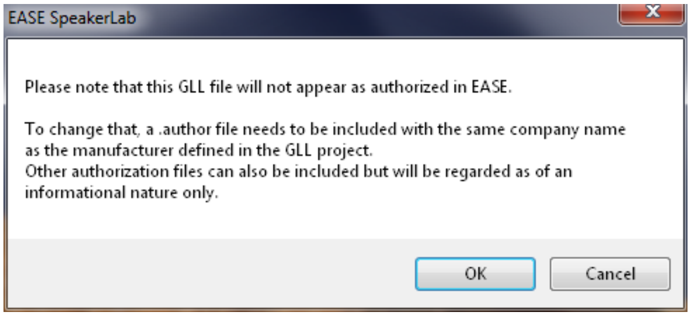
Website : http://www.tannoy.com/

Email :

Options : ☑ Allow User-Defined Cluster Setups

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2010\Tannoy\VQ Series\VQ Series (v1,1).xgll

Figure 2-119 – Compiling the GLL



Please note that this GLL file will not appear as authorized in EASE.

To change that, a .author file needs to be included with the same company name as the manufacturer defined in the GLL project.
Other authorization files can also be included but will be regarded as of an informational nature only.

Figure 2-120 – Authorization warning if no author file is specified in the GLL project

If there is no authorization file included with the XGLL project file a message box indicating that the GLL will not appear as authorized in ***EASE***.

To obtain an .author file a manufacturer needs to contact AFMG support at support@afmg.eu. The .author file is available at no cost to manufacturers. Once the .author file is received it can be added to the project by creating another entry in the AUTHORIZATION FILES area under the GENERAL DATA section of the property tree.

2.11 Single-Source GLL with Filters

2.11.1 Introduction

This is a slight variation of the previous model of a GLL using a single source. Here we will also use only a single source but we also want to include a filter applied between the electrical input to the loudspeaker and the single acoustical source. In this way we can model filters that have been updated in the design process or on-site, such as a high pass (low cut) filter, without the need to re-measure the whole loudspeaker system response. This is possible due to the fact that electro-acoustic data for the source (GSS) are implemented separately from filters.

The acoustical source is defined by a GSS file that must be measured without the loudspeaker system filter (this is easily possible if the box is composed of a single electro-acoustic transducer).

2.11.2 GLL Overview as a Passive Loudspeaker

At this point it will be helpful to present a general overview of how the GLL functions in the way it handles signals, routing them to the sources, and calculating certain items such as sensitivity, maximum SPL, and input voltage. The easiest way to do this is to think of the GLL model of the loudspeaker system as if it is a passive loudspeaker. This includes the input filters (high pass, EQ, or others). Even if these input filters are in a DSP or other active electronics, for the purposes of the GLL think of them as being placed in the signal path ***after*** the output of the power amplifier.

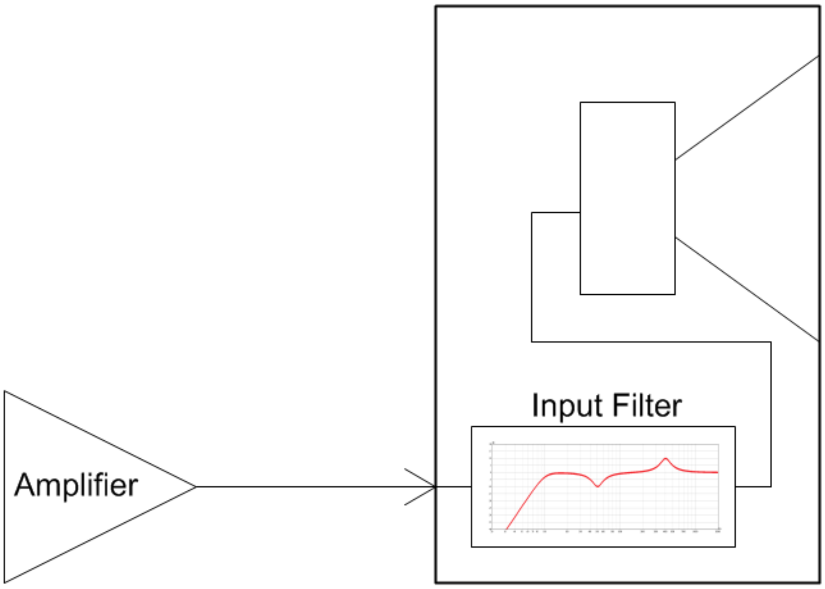


Figure 2-121 – Conceptual signal flow of a single-source GLL

Picturing the GLL in this way is necessary because the effect of the input filters is to change the spectral content and/or level of the signal before it is sent to the source (GSS). The sensitivity for the GLL is calculated using an rms input of 2.83 V at the input to the GLL. Since the filter is between the input and the GSS it will affect the sensitivity. Let's say the GSS we are using has a sensitivity of 97 dB. We will apply a filter that is a simple 6 dB boost across the entire frequency band. This will increase the calculated sensitivity of the loudspeaker system (GLL) to 103 dB. Our filter boosts the level of the signal 6 dB so that the GSS has 5.66 V at its input. This is why the system sensitivity is 103 dB.

Note also that every GSS has a Maximum Input Voltage (MIV) associated with it. In the GLL the level of the signal at the input of the GSS is not allowed to exceed its MIV. Let us say our GSS has a MIV of 28 V. This means when 28 V (approx. 29 dBV) is applied to the source (GSS) it should have a maximum SPL of approximately 117 dB.

With our 6 dB boost filter in place the MIV of 28 V for the GSS is reached when only 14 V (approx. 23 dBV) is applied to the input of the GLL. The GLL will not allow the voltage at its input to go any higher. If it did the voltage at the input of the GSS would exceed its MIV limit. This is not allowed. So even though the sensitivity of the system has increased its maximum SPL does not.

It should also be mentioned that the GLL is a steady-state model of a loudspeaker system. That is to say, all filters based on complex transfer functions are basically infinitely long. These are not to be confused with filter measurements (sampled and limited by sample rate), constructed digital filters (morphed to sampled frequency domain), or FIR filters (finite length) which all have a finite length.

2.11.3 Adding Filters

Let's start by creating a new project. From the main menu click on FILE > NEW PROJECT and then select SIMPLE LOUDSPEAKER on the LOUDSPEAKER tab. Make sure to select the desired DESTINATION path as in the previous example.

Expand the ACOUSTIC DATA item in the property tree. Select the FILTER GROUPS item. This will show the Edit Filer Groups page on the right side of the display.

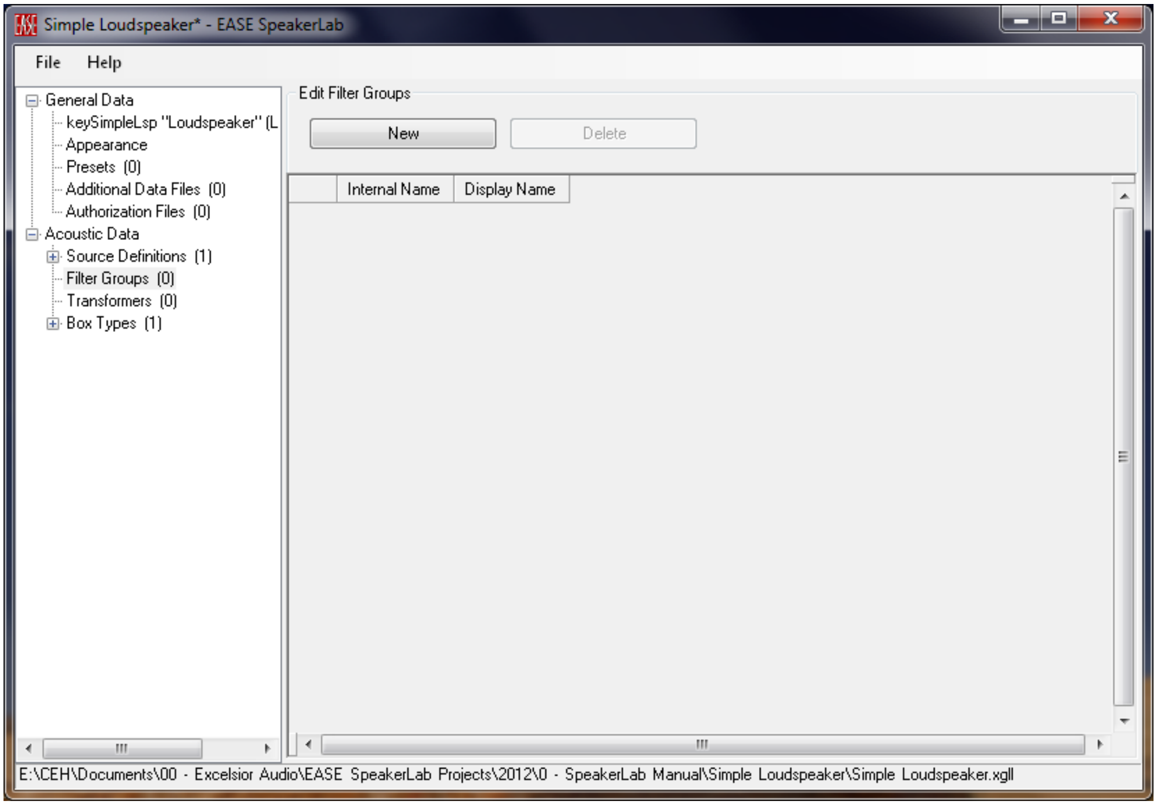


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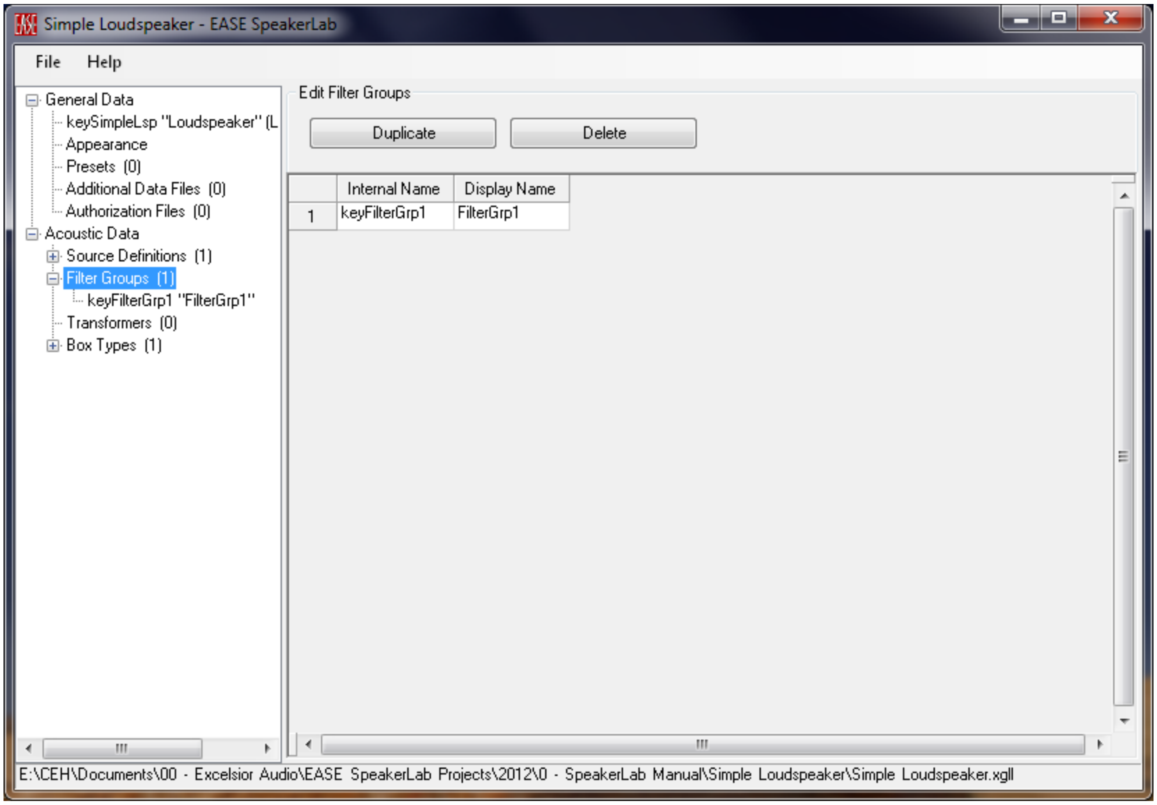
Figure 2-122 – Adding filter groups

Clicking the NEW button will add a Filter Group with the default INTERNAL NAME "KeyFilterGroup1" and DISPLAY NAME "FilterGroup1" to the project.

A Filter Group is a set of filters that can be placed between GLL system inputs and GSS acoustic source inputs. The signal routing for a Filter Group is defined by the BOX TYPE INPUT CONFIGURATION as we will see later.



	Internal Name	Display Name
1	keyFilterGrp1	FilterGrp1

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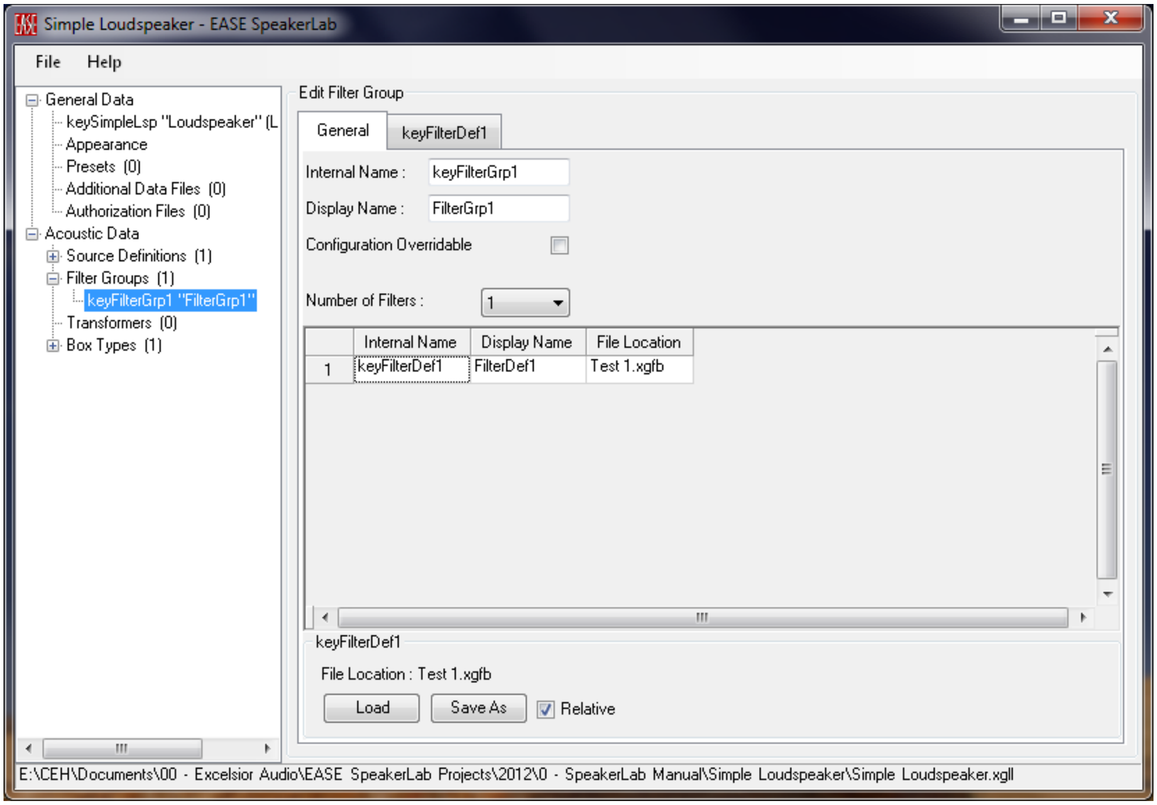
Figure 2-123 – One filter group added to the GLL project

Click on the + next to Filter Groups in the property tree to expand the Filter Groups. Selecting the filter group name from the property tree will show the Edit Filter Group page on the right. This page is used to specify the Internal Name and the Display Name of the Filter Group.

The Configuration Overridable check box allows the filter to be edited by the GLL user if it is checked. If it is not checked, the filters included with the GLL cannot be modified except for switching between predefined configurations.



Internal Name : keyFilterGrp1

Display Name : FilterGrp1

Configuration Overridable ☐

Number of Filters : 1

	Internal Name	Display Name	File Location
1	keyFilterDef1	FilterDef1	Test 1.xgfb

keyFilterDef1

File Location : Test 1.xgfb

☑ Relative

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Figure 2-124 – Editing a Filter Group, General tab

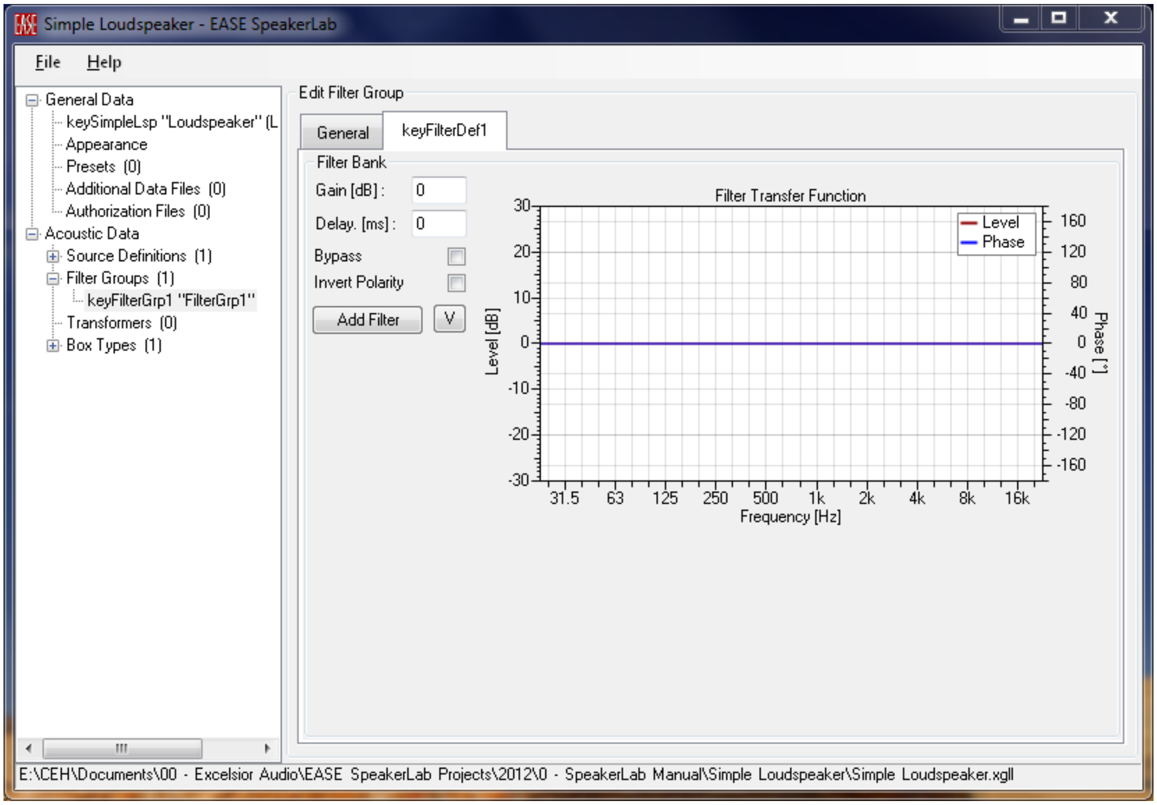
Each Filter Group can contain from 0 up to 99 filter definitions. If the Filter Group is composed of more than one filter definition the user will be able to select between them when using the GLL (***GLL Viewer***, ***EASE***, ***EASE Focus***). This allows the user to select different processing schemes, if available. An example might be a user adjustable high frequency shelving filter to select a flat response or to attenuate the HF level of the loudspeaker.

To add or delete filter definitions use the NUMBER OF FILTER DEFINITIONS dropdown box to select the number desired. Each filter definition is shown in a table where it is possible to edit the filter INTERNAL NAME and DISPLAY NAME.

Each filter definition, either after it is loaded from another file or defined in the Filter Bank, must be saved to a file. The location for this file is shown in the FILE LOCATION cell.

Each filter definition in the table also has a tab. The name of the tab is the filter definition's Internal Name.

For this example we will select only one filter definition and edit it.


Simple Loudspeaker - EASE SpeakerLab

Edit Filter Group

Filter Bank

Gain [dB] : 0

Delay. [ms] : 0

Bypass □

Invert Polarity □

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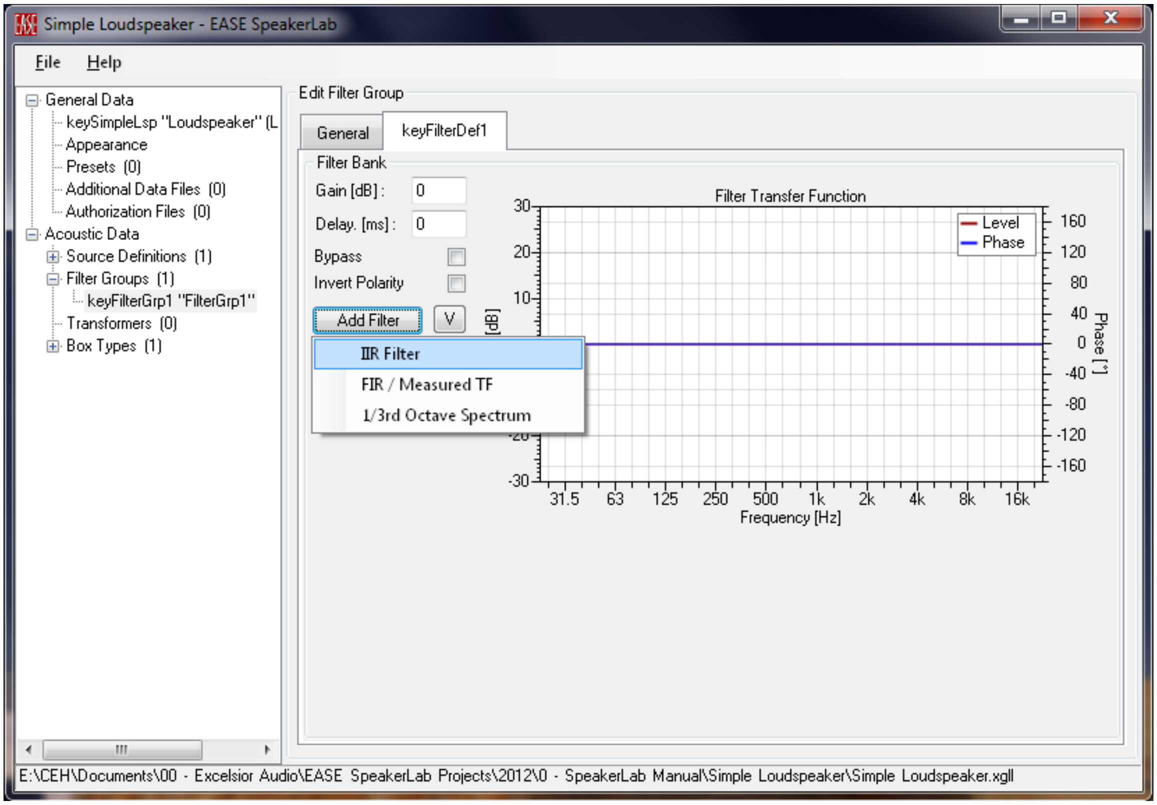
Figure 2-125 – Editing the Filter Bank of a filter group

Selecting the FILTER DEFINITION tab (keyFilterDef1 in our example) shows the Filter Bank. This is where some of the general filter settings are entered. These are GAIN [DB] in dB and DELAY [MS] in milliseconds. The BYPASS and INVERT POLARITY check boxes are self-explanatory.

The FILTER TRANSFER FUNCTION graph shows the combined (summed) transfer function of all the applied filters in the Filter Bank.



Gain [dB]: 0
Delay. [ms]: 0
Bypass ☐
Invert Polarity ☐

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Figure 2-126 – Adding an IIR filter to the filter bank

To insert a filter in the Filter Bank click the ADD FILTER button. This will display a context menu to select one of three ways to specify a filter.

Three ways to input a filter:

- IIR filter
- FIR/Measured TF
- 1/3 Octave Spectrum

It is possible to cascade filters by adding more filters to the Filter Bank. There is a limit of 20 filters that can be added to each Filter Bank.

Each filter also has an individual BYPASS checkbox. This allows a single filter to be bypassed, as opposed to the BYPASS checkbox for the entire Filter Bank. Clicking the DELETE button for a filter will remove that individual filter from the Filter Bank. There is no confirmation prompt or an undo. Once the filter is deleted it is gone. A new filter can be added to replace it but the filter parameters will have to be entered.

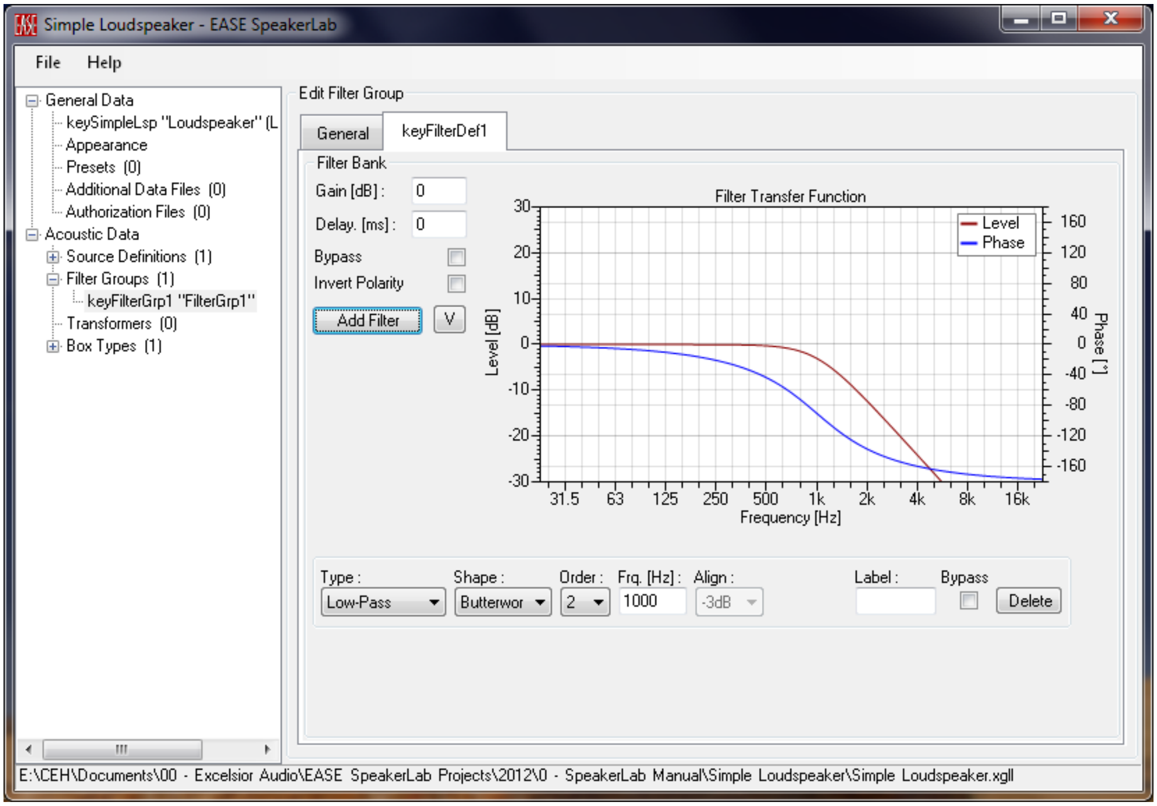
There is also a LABEL provided for each filter. This can be used to name the filter or write some short notes about it.

2.11.3.1 IIR Filter

The IIR (Infinite Impulse Response) filter is a typical analog-style filter. After adding an IIR filter it is possible to edit the filter parameters very much like doing so with a modern DSP unit. There are multiple types of filters from which to choose.

IIR filter types available:

- Low Pass
- High Pass
- All Pass
- Peak
- Peak Sym
- Low Shelf
- High Shelf


Simple Loudspeaker - EASE SpeakerLab

- General Data
  - keySimpleLsp "Loudspeaker" (L
  - Appearance
  - Presets (0)
  - Additional Data Files (0)
  - Authorization Files (0)
- Acoustic Data
  - Source Definitions (1)
  - Filter Groups (1)
    - keyFilterGrp1 "FilterGrp1"
  - Transformers (0)
  - Box Types (1)

Edit Filter Group

General | keyFilterDef1

Filter Bank

Gain [dB] : 0

Delay. [ms] : 0

Bypass □

Invert Polarity □

Type : Low-Pass

Shape : Butterwor

Order : 2

Frq. [Hz] : 1000

Align : -3dB

Label :

Bypass □

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Figure 2-127 – Filter bank with a low pass filter added

Different alignment (Shape) such as Butterworth, Linkwitz-Riley, and Bessel can be selected for the low pass, high pass, and all pass filters. The all pass filter also allows a Sallen-Key selection. When the Order is set to 2 an additional Q parameter can be set.

The low pass and high pass filters can be set from 1st up to 8th order (48 dB/octave) using the Order dropdown box.

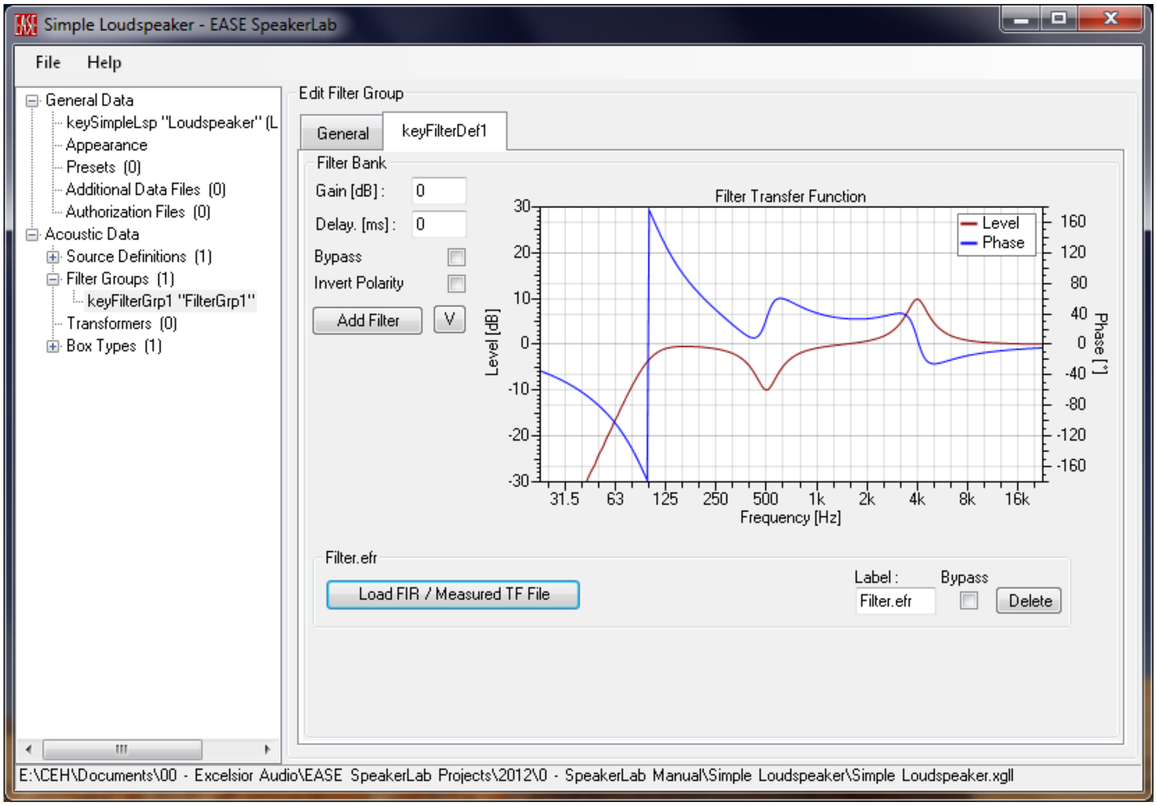
The Peak and Peak Sym filters are typical parametric EQ filters. They allow the frequency (FRQ. [Hz]), Q, and GAIN [dB] to be set.

The Low-Shelf and High-Shelf filters are shelving filters. They have a setting for the frequency (FRQ. [Hz]) and GAIN [dB].

2.11.3.2 FIR / Measured Transfer Function

Another option is to measure the electrical response of the filter and import the measured transfer function. The supported file formats for importing are listed below.

- ***EASERA*** (.etm, .efr) and text (.etx) files
- Wave Files (.wav)
- ***MLSSA*** (.frq) and (.tim) files
- ***LMS*** Frequency Files (.txt)
- ***Monkey Forest*** (.dat) and (.spk) files
- ***CLIO*** Time and Frequency (.txt) files
- ***EVI*** (.prn) files
- ***TEF*** Tds (.tds)


Simple Loudspeaker - EASE SpeakerLab

Gain [dB] : 0
Delay. [ms] : 0
Bypass □
Invert Polarity □

Filter.efr
Label : Filter.efr
Bypass □

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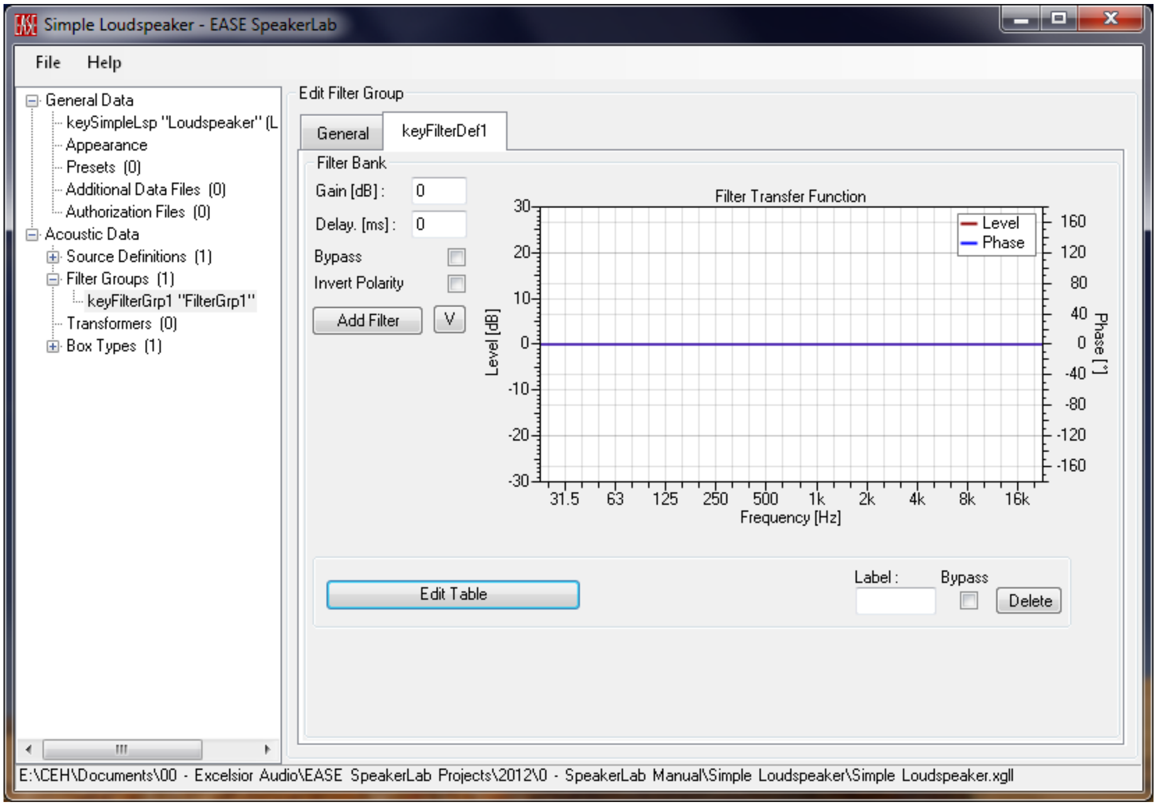
Figure 2-128 – Filter bank with a measurement of a filter added

It should be noted that when importing the measured transfer function of a filter the size of the imported measurement file can increase the size of the compiled GLL file. If many large transfer function measurement files are imported this can cause the compiled GLL to become quite large. This can lead to increased load times for the GLL when opening it or using it in other programs. It is recommended that care be taken to not make the measurement files have greater frequency resolution (and more data points) than necessary to adequately capture the response of the filter.

2.11.3.3 1/3 Octave Spectrum

The last option is to create a filter based on a 1/3 octave table of level and phase values.



Gain [dB] : 0

Delay. [ms] : 0

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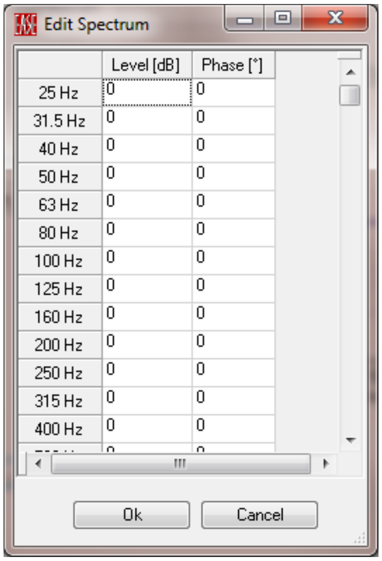
Figure 2-129 – A table-based filter in the filter bank

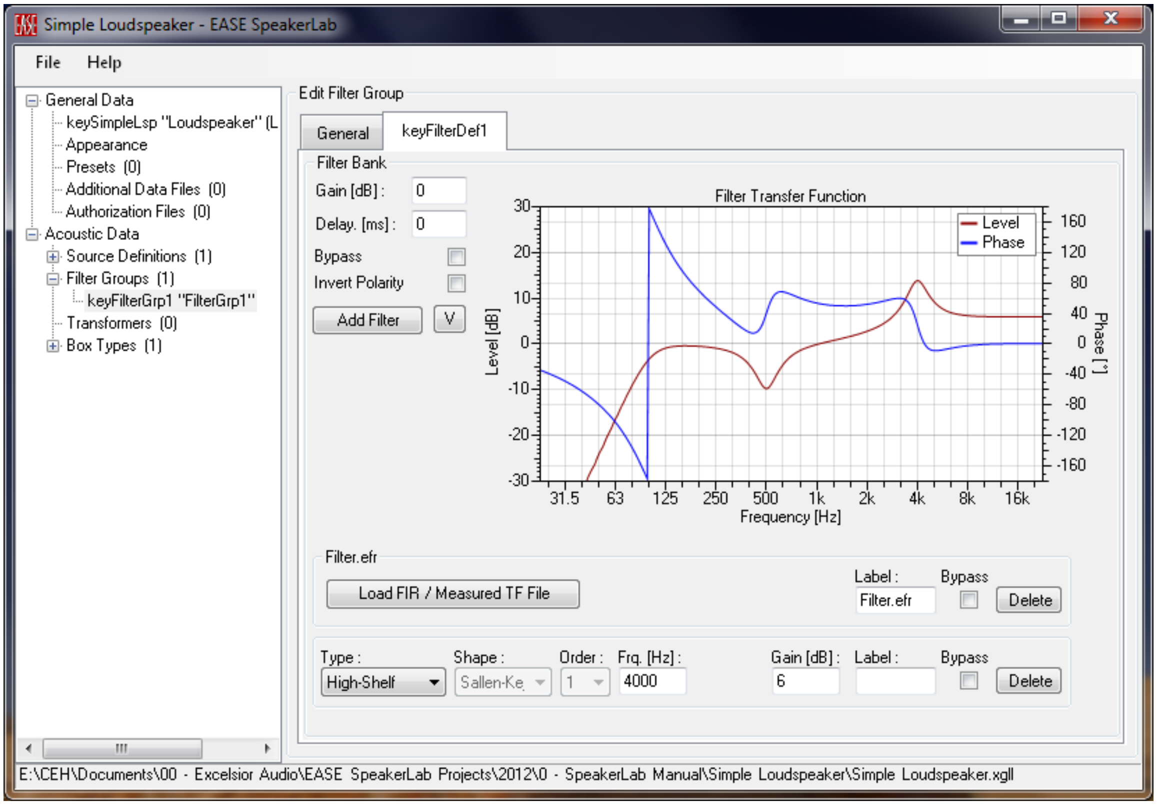


	Level [dB]	Phase [°]
25 Hz	0	0
31.5 Hz	0	0
40 Hz	0	0
50 Hz	0	0
63 Hz	0	0
80 Hz	0	0
100 Hz	0	0
125 Hz	0	0
160 Hz	0	0
200 Hz	0	0
250 Hz	0	0
315 Hz	0	0
400 Hz	0	0

Figure 2-130 – Filter table

2.11.3.4 Comments On Filters

A note should be made about FIR filters and measured transfer functions.



Gain [dB] : 0
Delay. [ms] : 0
Bypass □
Invert Polarity □

Label : Filter.efr Bypass □

Type : High-Shelf Shape : Sallen-Ke Order : 1 Frq. [Hz] : 4000 Gain [dB] : 6 Label : Bypass □

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Figure 2-131 – Filter bank with a measurement of a filter and a high pass filter added

The ability to import the measured response of a filter is very important in practice because in this way it is possible to capture the "real" behavior of the filter for the loudspeaker model. It must be noted that there is no standardization in the definition of Q in IIR filter implementations. Therefore the implementation of the IIR filter parameters in ***EASE SpeakerLab*** can differ from those in commercially available DSP units, causing differences in the response of the filters. This, in turn, can lead to errors in the loudspeaker modeling. The measurement of the electrical response of the filter avoids this potential source of error.

2.11.3.5 Comments On Filter Measurements

In case of measuring passive filters, great care must be taken during the measurement since the filter's response can be greatly affected by the load placed on the filter. The polarity of the transfer function must be maintained during the measurement. There is also the issue of electronic components tolerance that must be correctly addressed in order for the response of the filter to be representative of the average unit.

There are also items that must be handled properly when measuring active filters. As with passive filters, the issue of component tolerance and the conservation of polarity are present. If the active device is a DSP unit component tolerances are no longer an issue, but others take its place. As mentioned above there is no standardized definition for Q or bandwidth for EQ filters. The response of EQ filters, shelving filters, and even some high pass or low pass filters will be different when measuring DSP units from different manufacturers with the same front panel parameters. It is even possible that this will occur with different DSP units from the same manufacturer!

The latency of a DSP unit may also need to be removed from the measurement if the intent is to model the filter and not a specific DSP. For some cases the latency may be better left as part of the measurement (specific FIR filters).

See [7] and also section 8.4 – Filter Measurement for more details about measuring filters.

2.11.3.6 Importing/Exporting a Filter Bank

Entire Filter Banks can be imported or exported by clicking the V button (located to the right of the ADD FILTER button). The IMPORT and EXPORT functions are based on the Generic Filter Bank (.xgfb) files. Clicking on IMPORT displays an OPEN GENERIC FILTER BANK dialog box. This is used to select the desired filter bank (.xgfb) file to import. Similarly, clicking EXPORT will allow the current Filter Bank to be saved as a Generic Filter Bank (.xgfb) file.

Exporting and importing an entire filter can be very useful, if for example the identical or a very similar set of filters needs to be used on more than one Filter Definition or Filter Group.

2.11.3.7 Saving a Filter's Transfer Function

The V button (located to the right of the ADD FILTER button) also allows the transfer function of all the filters in the Filter Bank to be saved in one of the supported file formats (the same as for the LOAD FIR/MEASURED TF FILE). This can be very useful to compare the transfer function of the filters used in the GLL model to the measured response of the filters actually implemented in the DSP that will be used with the loudspeaker. This may be the only way to verify that the responses of the filters match.

2.11.3.8 Saving the Filter

When editing of the filter definition is complete, it is necessary to save the filter as a Generic Filter Bank (.xgfb) file. To do this click on the General tab and select the row containing the filter to be saved in the table. At the bottom of the EDIT FILTER GROUP page click the SAVE AS button. This will open a SAVE FILE AS dialog box which is used to select the file name and folder location for the file. Clicking on OK will save the file. The path to the file will be shown both in the FILE LOCATION cell in the table as well as the FILE LOCATION at the bottom of the page.

The RELATIVE checkbox allows the file location to be a relative path instead of an absolute path. If the guidelines in section 6.1 – Folder Structure – Keeping Everything Together are followed it is recommended to always use relative paths.

Note that if saving the XGLL project is attempted before saving the filter definitions, ***SpeakerLab*** will prompt you to save the filter definitions first. ***SpeakerLab*** lab will not allow the project to be saved until after the filters definitions are saved.

2.11.4 Filter Hierarchy (Why All The Different Filter Grouping)

It may seem a bit confusing why there are Filter Groups, Filter Definitions, Filter Banks, and Filters. We will try to explain the differences in these as well as the necessity for them.

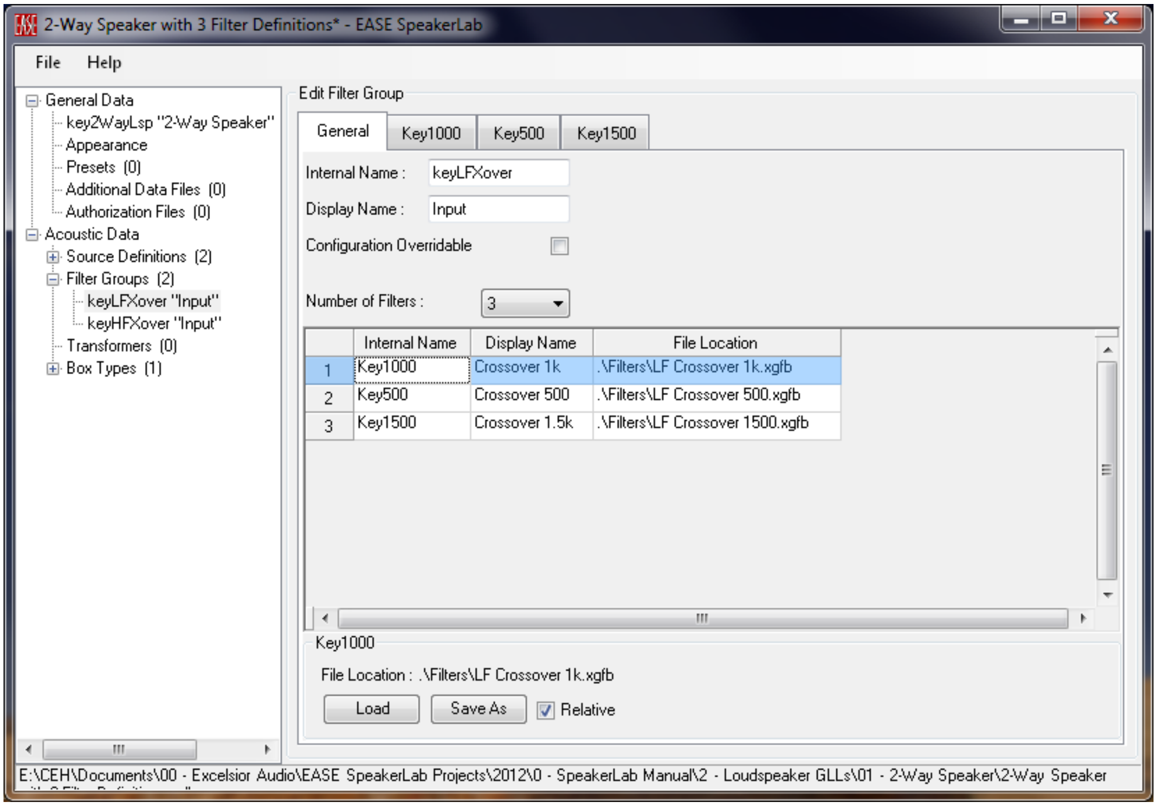
2.11.4.1 Filter Groups

Filter Groups are used to hold one or more Filter Definitions. A Filter Group can be applied to the input of one or more sources. As an example, one Filter Group may be used for the low pass section of a two-way loudspeaker system while a second Filter Group may be used for the high pass section. This will be detailed more in section 2.12 – Two-Source GLL.

2.11.4.2 Filter Definitions

Multiple Filter Definitions within the Filter Groups allows for alternate crossover or processing options for a loudspeaker system. This is most applicable to GLLs with two or more sources, but we shall cover the concept here. Let's say we have a two-way, single-input loudspeaker which is modeled using a two-source GLL. As described above we will have two Filter Groups. We will call them keyGroupHF and keyGroupLF. Each Filter Group will have three different Filter Definitions. This will allow the end user to select which set of Filter Definitions is to be used.

It is important to note that for the selection to be able to properly take place in the GLL there must be some sort of connection between the different Filter Definitions in the different Filter Groups. The INTERNAL NAME of the Filter Definition serves as this connection. The Filter Definition INTERNAL NAMEs shown in Figure 2-132 are used in both the keyGroupHF and keyGroupLF Filter Groups. This links the Filter Definitions together across the Filter Groups within (internally) the GLL.



Internal Name : keyLFXover

Display Name : Input

Configuration Overridable ☐

Number of Filters : 3

	Internal Name	Display Name	File Location
1	Key1000	Crossover 1k	.\Filters\LF Crossover 1k.xgfb
2	Key500	Crossover 500	.\Filters\LF Crossover 500.xgfb
3	Key1500	Crossover 1.5k	.\Filters\LF Crossover 1500.xgfb

Key1000

File Location : .\Filters\LF Crossover 1k.xgfb

☑ Relative

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\2 - Loudspeaker GLLs\01 - 2-Way Speaker\2-Way Speaker

Figure 2-132 – Two filter groups with three filters in the groups

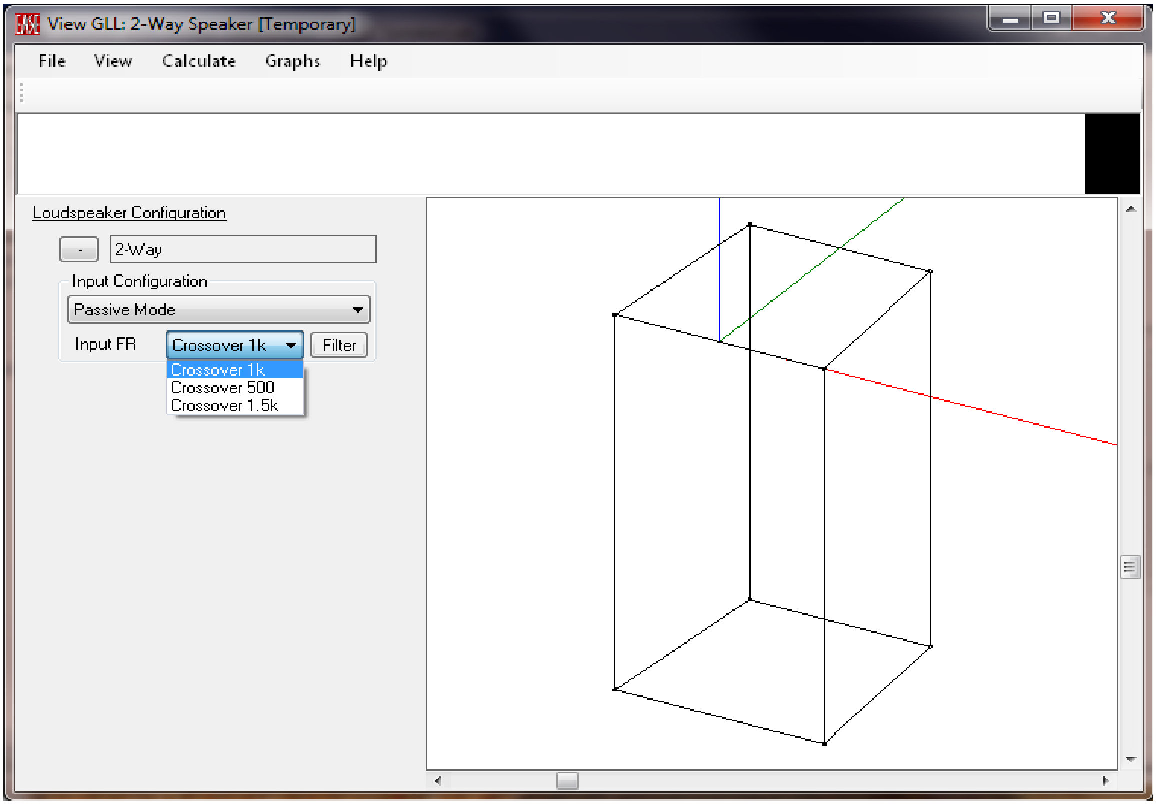


Figure 2-133 – Running GLL showing the selection of the different filters shown in Figure 2-132

2.11.4.3 Filter Banks

Each Filter Definition refers to one Filter Bank, which is shown on the tab for the Filter Definition. From a hierarchal point of view there is not much practical difference between the Filter Definition and the Filter Bank. The only real difference is that the Filter Definition is the element in the XGLL project that links to the Filter Bank. The Filter Bank is stored as a Generic Filter Bank (.xgfb) file separately from the .xgll file. This is similar to the Generic Sound Source (.gss) file being stored separately from the .xgll file.

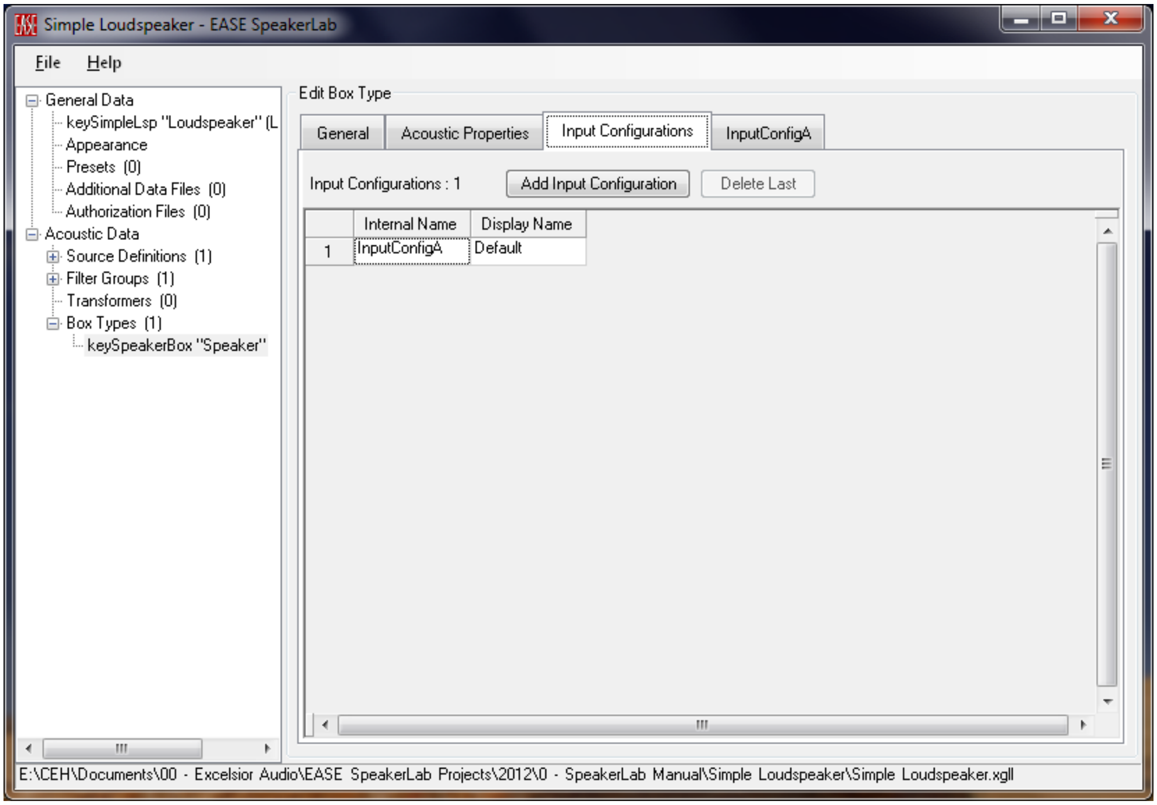
2.11.4.4 Filters

A Filter Bank can contain many different individual filters, such as high pass filters, low pass filters, parametric EQ filters, as well as actual measured transfer functions. This allows almost any required filtering scheme necessary for a loudspeaker system to be incorporated into the GLL model. It should be noted that the current internal structure of an XGLL project does not allow a hierarchy of filters (i.e. global input EQ, filter grouping, etc.) to be defined. This does not preclude the modeling of such a system, however. To get the effect of global input EQ, one must simply apply the same filters to each output. The net effect is the same.

2.11.5 Assigning Filters to Input Configurations

At this point the filters are specified, but it is still necessary to assign the filter inputs to the system input (or inputs) and the filter outputs to the acoustical sources of the loudspeaker box model. This is done using Box Type INPUT CONFIGURATIONS tab.

To access the INPUT CONFIGURATIONS tab expand the BOX TYPES item on the property tree, if it is not already expanded. Under BOX TYPES the different keys of the defined box types are listed. In this case since the model is a loudspeaker-type GLL only a single item is present. Select that item to edit the box type and choose the INPUT CONFIGURATIONS tab.

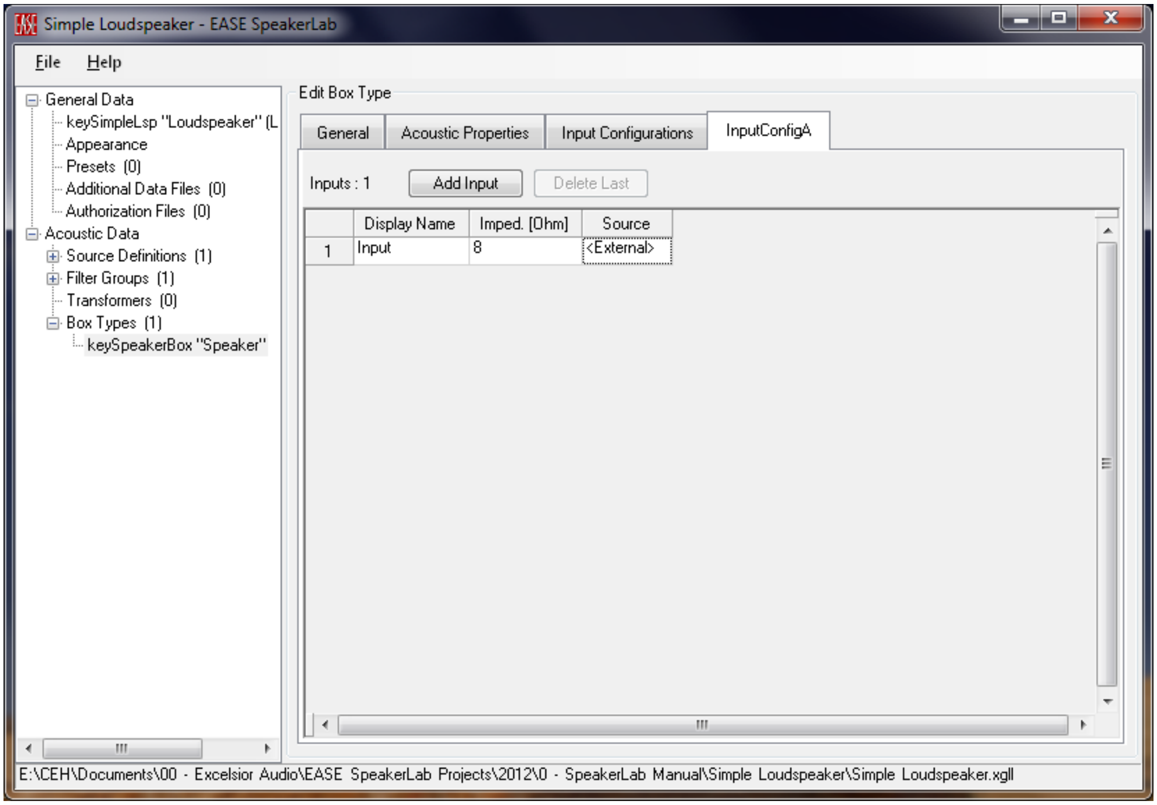


Input Configurations : 1

	Internal Name	Display Name
1	InputConfigA	Default

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\Simple Loudspeaker\Simple Loudspeaker.xgll

Figure 2-134 – Input configurations tab of Edit Box Type page



Inputs : 1

	Display Name	Imped. [Ohm]	Source
1	Input	8	<External>

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\Simple Loudspeaker\Simple Loudspeaker.xgll

Figure 2-135 – Input Configuration tab

In this example we have a single configuration. To edit the configuration select the INPUT CONFIGURATION tab. Note that the name of this tab will be whatever its INTERNAL NAME is on the INPUT CONFIGURATIONS tab.

It is possible to look at the Box Input Configuration as an input-to-source connection matrix which uses the Filter Groups as the links. Every Input Configuration defines a connection structure between the box inputs, filters, and acoustical sources.

In this case we have a model like this:

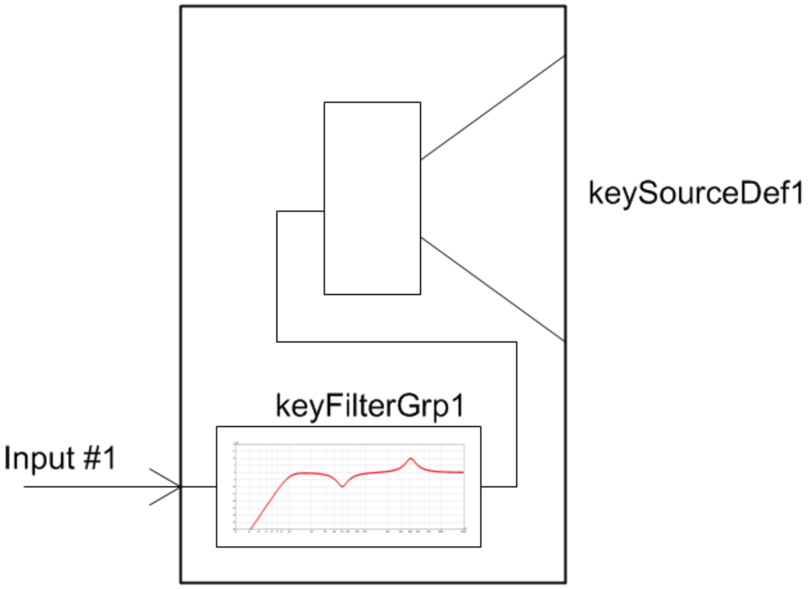
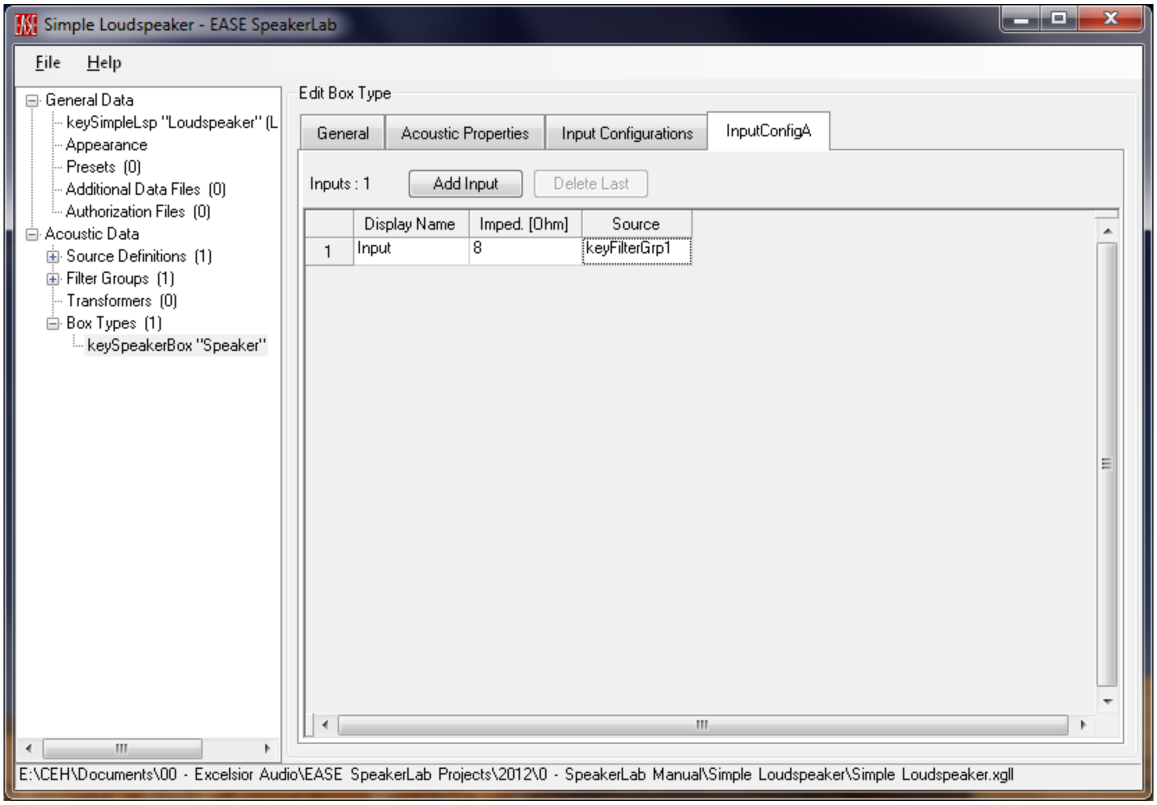


Figure 2-136



Inputs : 1

	Display Name	Imped. [Ohm]	Source
1	Input	8	keyFilterGrp1

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\Simple Loudspeaker\Simple Loudspeaker.xgll

Figure 2-137 – Input Configuration tab showing routing of the input signal through a filter group

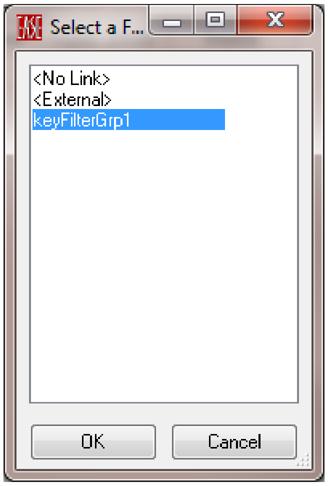


Figure 2-138 – Select a Filter Group dialog box

Thus we need to connect our acoustical source (keySourceDef1) to the electrical input Input #1 using the set of filters defined in keyFilterGrp1. This is done by editing the InputConfigA table to choose the desired Filter Group. Clicking in a cell in the Source column opens a SELECT A FILTER GROUP dialog box. Select the desired Filter Group from the dialog box and click OK. This will close the dialog box and the selected Filter Group will be displayed in the cell.

Note that the default connection is EXTERNAL. This means that the external input to the box is connected directly to the source. If no Filter Group has been defined or was to be used this could be left as EXTERNAL.

The table for the connection matrix will have as many columns as the number of sources (defined in the ACOUSTIC PROPERTIES tab). If another input is added, for example as with a biamplified two-way loudspeaker, another row will be added to the table. The DISPLAY NAME for each input can be edited if desired. Double-click the text in the cell of the Display Name to be changed and type in the new name.

There is also a column for the IMPEDANCE of each input. Since SpeakerLab was not designed to model the impedance of passive filter networks it is required that the rated impedance for the loudspeaker system be entered manually in the XGLL project. The impedance value entered for the first input (i.e. the top row) serves an important function in a GLL that uses transformers. Since it is most common to specify a transformer tap setting by its power, it is necessary to know the impedance of the loudspeaker connected to the output of the transformer in order to calculate the voltage applied across the loudspeaker. It is this impedance entered in the first input that is used to determine the voltage at the output of a selected power tap of a transformer in the GLL. For programs such as EASE Address and EASE Evac that recognize transformers within a GLL, it is this voltage that will drive the loudspeaker.

To maintain consistency within an XGLL project it is recommended that the value for the impedance entered for this first input is also the value entered for the LOUDSPEAKER IMPEDANCE [OHMS] when specifying a transformer in the XGLL project. See 2.15.2 – Transformers for additional information on transformers.

2.11.6 Example

We will now show an example of adding a filter to the Simple Loudspeaker XGLL project. If this project has not been created, please create it now and follow along with this example (see 2.2 – New GLL Project: Simple Loudspeaker for details on creating a new XGLL project).

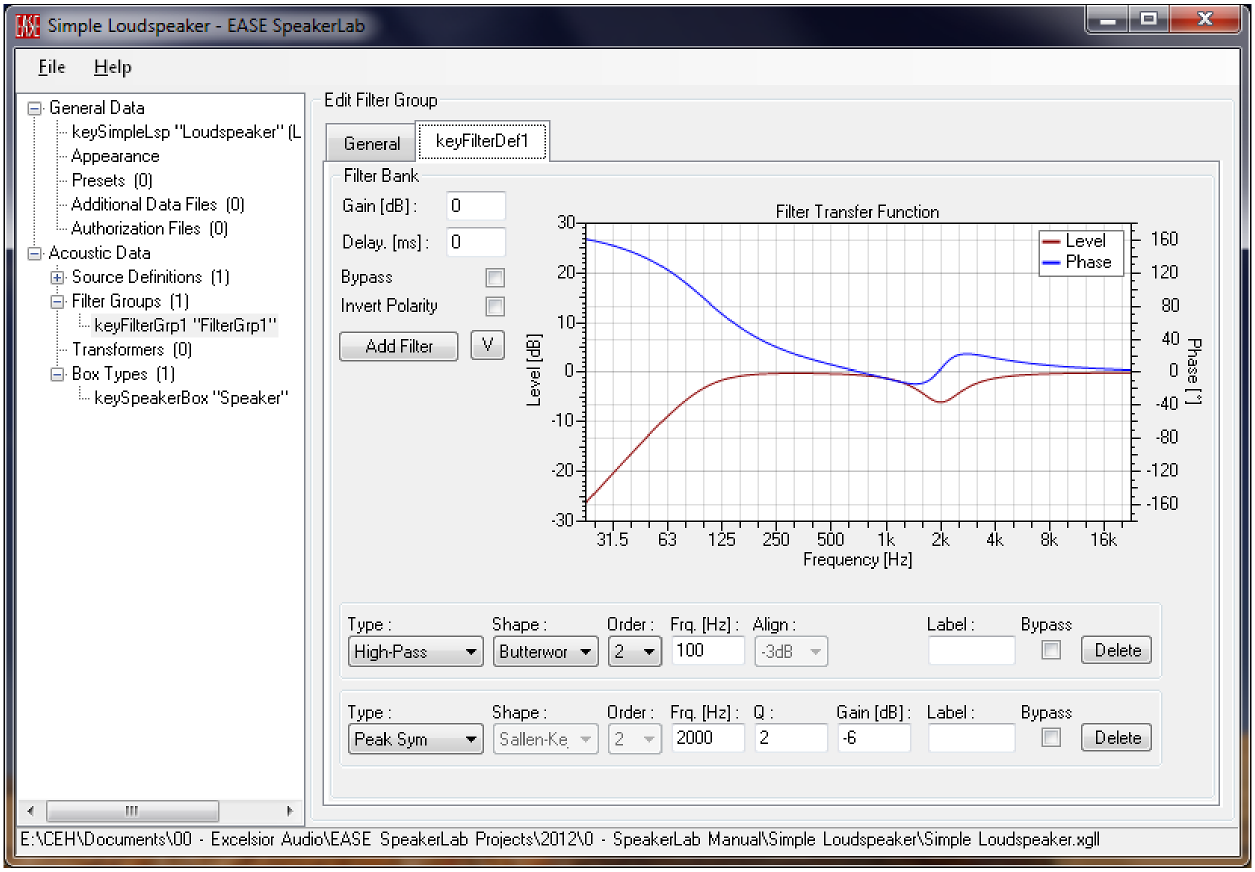
1. Expand the ACOUSTIC DATA item in the property tree and select FILTER GROUPS.
2. Click the NEW button on the EDIT FILTER GROUPS page to add a new Filter Group to the project.
3. Expand the FILTER GROUPS item in the property tree and select the newly added Filter Group (KEYFILTERGRP1).
4. On the EDIT FILTER GROUP page change the NUMBER OF FILTERS from 0 to "1".
5. Select the KEYFILTERDEF1 tab.
6. Click the ADD FILTER button in the FILTER BANK section.
7. Select IIR FILTER.

There should now be a low pass filter shown in the Filter Bank. Let's change this to a high pass filter at 100 Hz.

1. Click the TYPE dropdown and select HIGH-PASS.
2. Change the FRQ. [HZ] box from 1000 to "100". Just click in the box & change the value.

We will also add a second filter and configure it as a parametric EQ at 2 kHz with a Q of 2 and 6 dB of attenuation. When a new filter is added its default values will be identical to the last filter in the listing. The new filter will be placed at the bottom of the filter listing.

1. Click on the ADD FILTER button.
2. Select IIR FILTER from the list.
3. Use the TYPE drop to select PEAK SYM.
4. Change the frequency to "2000".
5. Change the Q to "2".
6. Change the Gain to "-6".


Simple Loudspeaker - EASE SpeakerLab

File Help

- General Data
  - keySimpleLsp "Loudspeaker" (L
  - Appearance
  - Presets (0)
  - Additional Data Files (0)
  - Authorization Files (0)
- Acoustic Data
  - Source Definitions (1)
  - Filter Groups (1)
    - keyFilterGrp1 "FilterGrp1"
  - Transformers (0)
  - Box Types (1)
    - keySpeakerBox "Speaker"

Edit Filter Group

General | keyFilterDef1

Filter Bank

Gain [dB] : 0

Delay. [ms] : 0

Bypass □

Invert Polarity □

Type :	Shape :	Order :	Frq. [Hz] :	Align :		Label :	Bypass	
High-Pass	Butterwor	2	100	-3dB			□	Delete

Type :	Shape :	Order :	Frq. [Hz] :	Q :	Gain [dB] :	Label :	Bypass	
Peak Sym	Sallen-Ke	2	2000	2	-6		□	Delete

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\Simple Loudspeaker\Simple Loudspeaker.xgll

Figure 2-139 – Filter bank with the two filters added

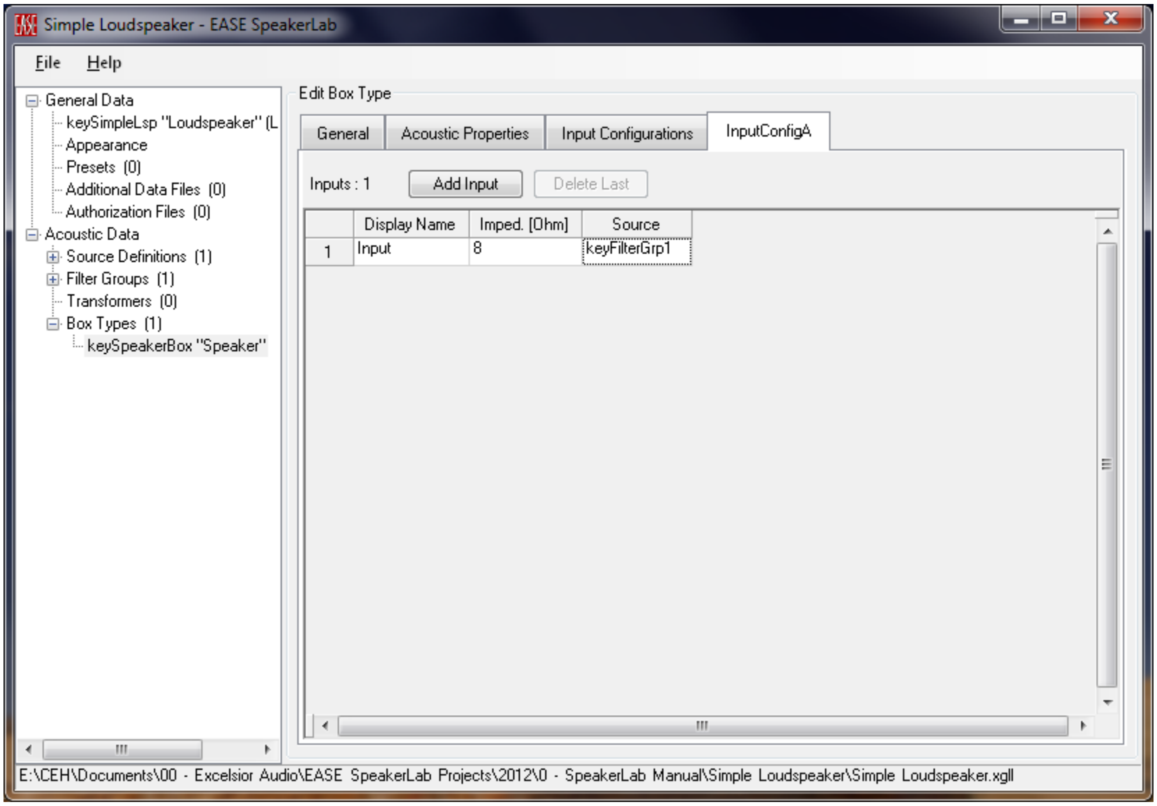
We must now save the Filter Bank we have created so that it can be referenced by the XGLL project.

1. Click on the GENERAL tab.
2. Click anywhere in row one of the filter table.
3. Click the SAVE AS button at the bottom of the page.
4. Select the folder into which the filter bank is to be saved.
5. Enter a name for the filter bank file ("Test 1" for example) and click SAVE.
6. Check the RELATIVE checkbox.

We should now also save the XGLL project by clicking FILE > SAVE PROJECT from the main menu.

Now we must tell the XGLL to route the signal through the filter we have created. To do this click on the + next to the BOX TYPES item in the property tree, if it is not already expanded. Select the KEYSPEAKERBOX item under BOX TYPES. Click on the INPUTCONFIGA tab. This is the default name for the INPUT CONFIGURATION. Here we see that the signal at the feeding the Source will be <EXTERNAL>; that is to say it will come directly from the input without going through any filters that are internal to loudspeaker system (GLL). We will change this so that the signal will be routed through our filter before it goes to the source.

1. Click on <EXTERNAL> in the SOURCE column.
2. Select KEYFILTERGRP1 from the list.
3. Click the OK button.



	Display Name	Imped. [Ohm]	Source
1	Input	8	keyFilterGrp1

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\Simple Loudspeaker\Simple Loudspeaker.xgll

Figure 2-140 – Input Configuration tab showing routing of the input signal through a filter group

Once again, save the XGLL project by clicking FILE > SAVE PROJECT from the main menu.

Before we run the GLL to demonstrate the effects of the filter we should modify the acoustical source that is being used in the project. We will set the sensitivity of the source to be 100 dB SPL with 2.83 V at 1 meter. We will also set the maximum voltage of the source to 28.3 V. This will make it easier to understand how the filters are affecting the system.

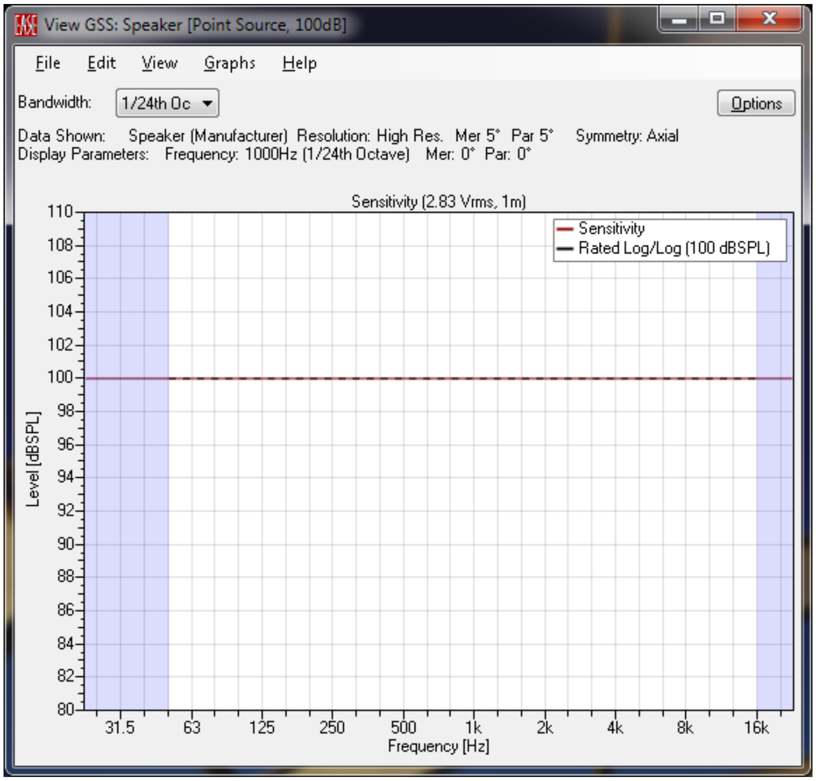
1. Click the + to the left of SOURCE DEFINITIONS item in the property tree to expand the list of sources.
2. Click on the KEYSOURCEDEF that is now shown under the SOURCE DEFINITIONS in the property tree.
3. On the EDIT SOURCE DEFINITION page in the right the IDEAL SPHERE radio button is selected and the <SPHERE> object is shown in the FILE LOCATION box. Select the DATA FILE (GSS) radio button and then click the NEW FILE button.
4. In the VIEW GSS dialog box that appear click on EDIT > PROPERTIES from the main menu.

5. Click on the SENSITIVITY tab.
6. In the GAIN CORRECTION [DB] text box change the value from 0 to "100".
7. Click on the MAXIMUM VOLTAGE tab.
8. In the PROGRAM (VRMS 1MIN) box change the value from 25 to "28.3".
9. Click on OK.

Note that we set the GAIN CORRECTION to 100 dB so that the sensitivity of our new source would be 100 dB SPL.

We should now save the GSS source file by clicking on FILE > SAVE AS in the main menu. Let's name this file "Point Source, 100dB". Enter this as the file name and click SAVE.

Let's verify the sensitivity and maximum SPL of the source before we close the dialog box and return to the XGLL project. Click on GRAPHS > SENSITIVITY from the main menu. A straight line at 100 dB should be displayed. If it can't be seen adjust the Graph OPTIONS so that it is (section 2.7.7). The RATED SENSITIVITY shown in the upper right corner of the graph should also show 100 dB.

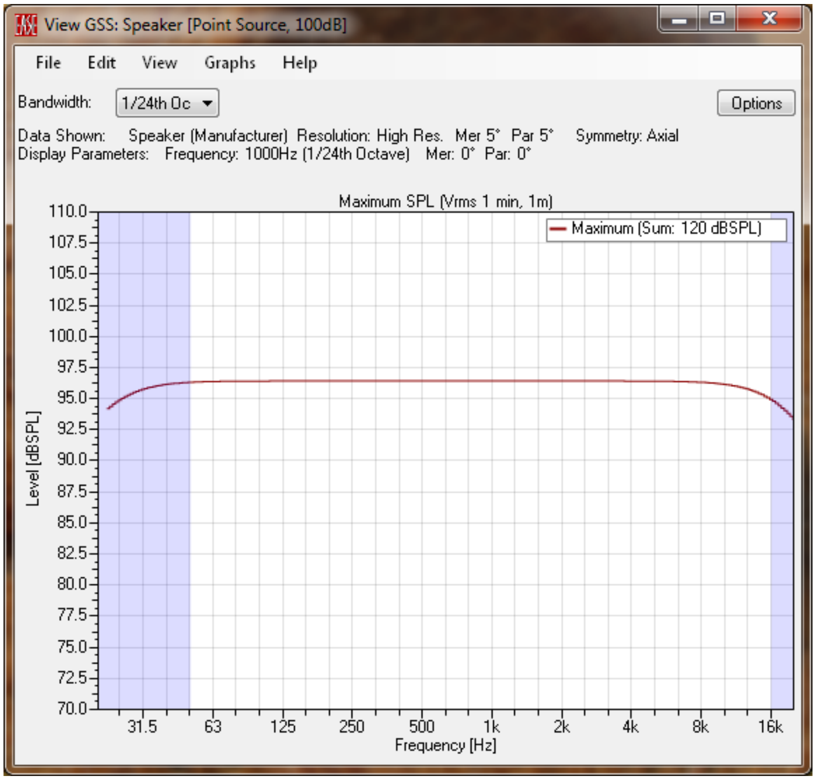

View GSS: Speaker [Point Source, 100dB]

Bandwidth: 1/24th Oc

Data Shown: Speaker (Manufacturer) Resolution: High Res. Mer 5° Par 5° Symmetry: Axial
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-141 – Sensitivity of the 100 dB point source GSS

Clicking on GRAPHS > MAXIMUM SPL will show a curve at just above 96 dB, provided the BANDWIDTH is set to 1/24 octave. Greater bandwidth settings will raise the level of the curve. The MAXIMUM SPL SUM shown in the upper right corner of the graph is 120 dB. This is exactly what we would expect from a source with a sensitivity of 100 dB when 28.3 V (or +20 dB relative to 2.83 V) is applied to it.

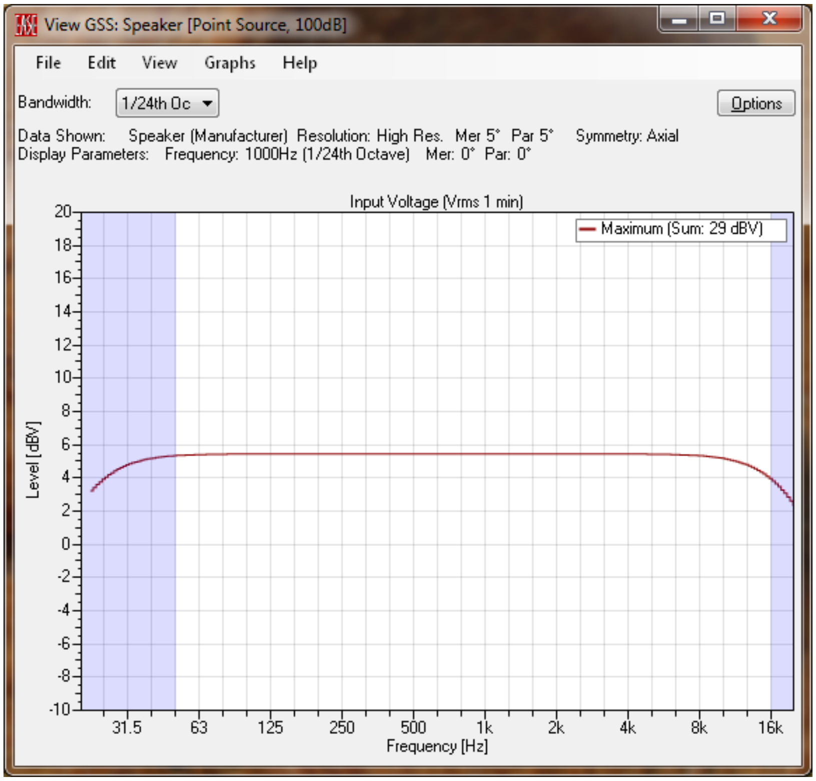


Bandwidth: 1/24th Oc

Data Shown: Speaker (Manufacturer) Resolution: High Res. Mer 5° Par 5° Symmetry: Axial
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-142 – Maximum SPL of the 100 dB point source GSS

The input level can also be examined by clicking on Graphs > Maximum Input Voltage. Here we see the Maximum Level Sum (in the upper right corner of the graph) is 29 dBV. This is the level corresponding to an rms voltage of 28.3 V.


View GSS: Speaker [Point Source, 100dB]

Bandwidth: 1/24th Oc

Data Shown: Speaker (Manufacturer) Resolution: High Res. Mer 5° Par 5° Symmetry: Axial
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-143 – Maximum input voltage of the 100 dB point source GSS

We can now close the View GSS dialog box & return to the XGLL project. After doing this we should change the source in the project to use the GSS we just saved.

1. Click on Browse.
2. Select the "*Point Source, 100dB*" gss file that was just saved.
3. Click the Open button.
4. Check the Relative checkbox.
5. Save the XGLL project.

2.11.7 Running the GLL

We should now run the GLL to demonstrate the effect of the filters we added in the previous section. Click on FILE > RUN GLL (or hit F5). In the VIEW GLL dialog box that appears click on the + button next to the SPEAKER to expand show the INPUT CONFIGURATION settings. It can be seen that the input to the GLL is routed to the FILTERDEF1 Filter Definition. The filters in the associated Filter Bank internal to the GLL will affect the response of the GLL.

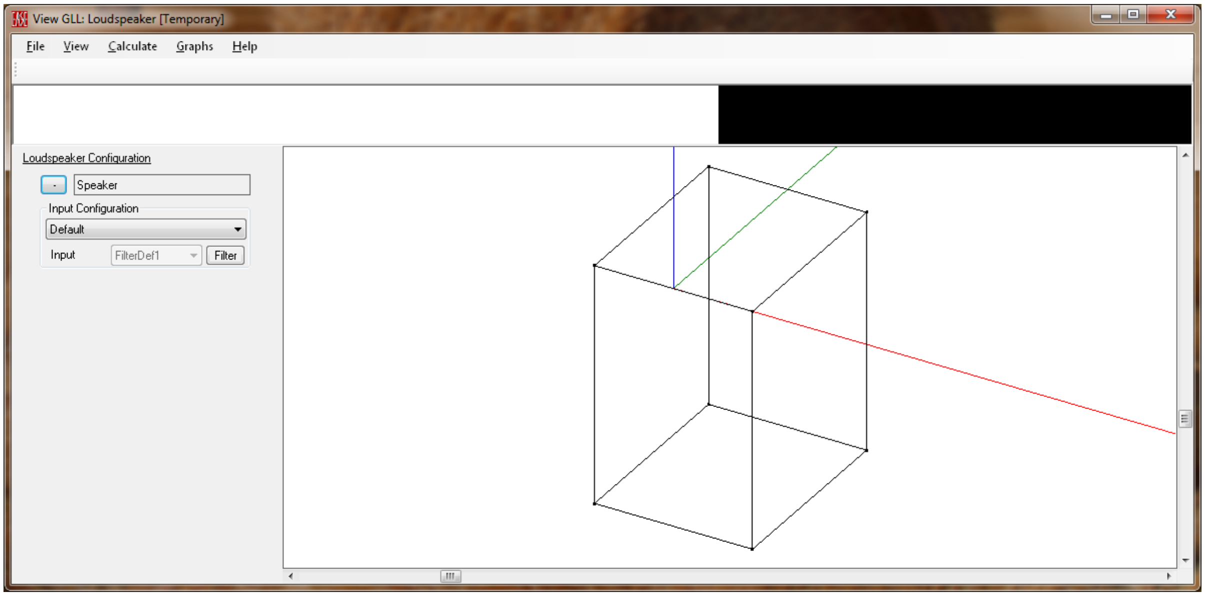
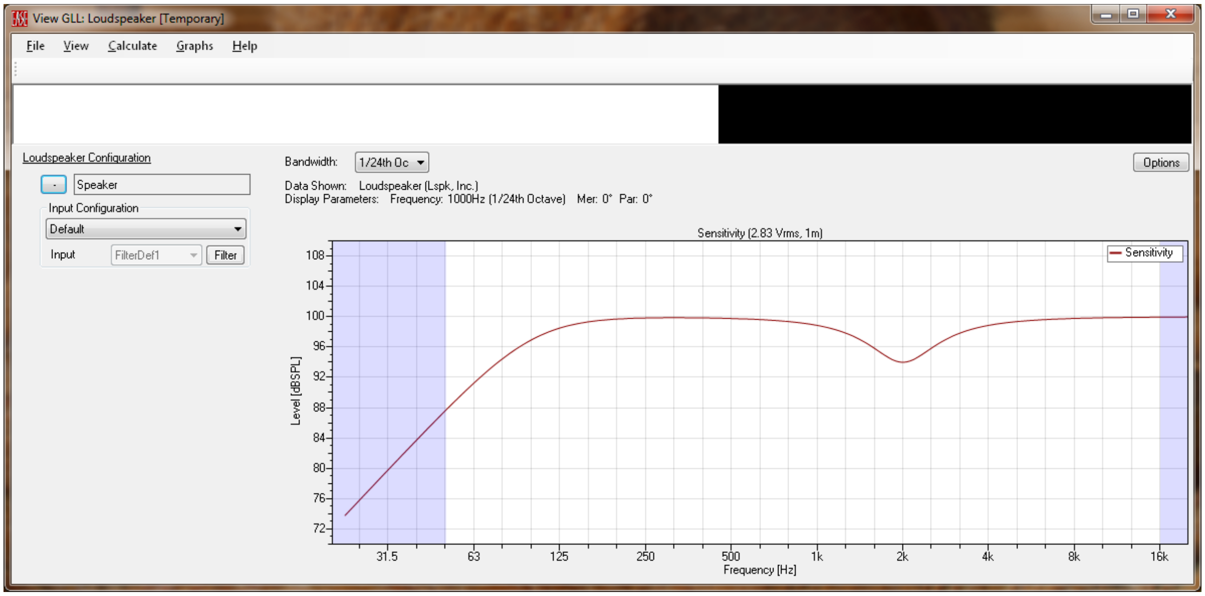


Figure 2-144 – Running GLL

Let's calculate the response of the GLL and see how this compares to the response of the source used in the GLL. Click on CALCULATE > BALLOON (or hit F5) so that we can specify the CALCULATION PARAMETERS. We will keep all of the default parameters except for one. We will change the INPUT SIGNAL from IEC 60268-1 to AES Broadband. The IEC 60268 signal is the default for the GLL. We only change the GLL input signal so that we will be able to directly compare the maximum SPL of the GLL with that of the GSS (the GSS had the maximum input voltage defined with the AES Broadband input signal spectrum). Click on OK to calculate the response.

Select GRAPHS > SENSITIVITY to show the sensitivity. We see that the perfectly flat frequency response of our source at 100 dB is no longer flat. The high pass and EQ filters have changed the response. Note that at 300 – 400 Hz and above 10 kHz, where the filters have no effect, the response is still at 100 dB. The RATED SENSITIVITY is now calculated as 98 dB instead of 100 dB. This occurs because we have 2.83 V at the input to the system, but the filters attenuate this level by about 2 dB, on a broadband basis.



Speaker

Input Configuration: Default

Input: FilterDef1

Bandwidth: 1/24th Oc

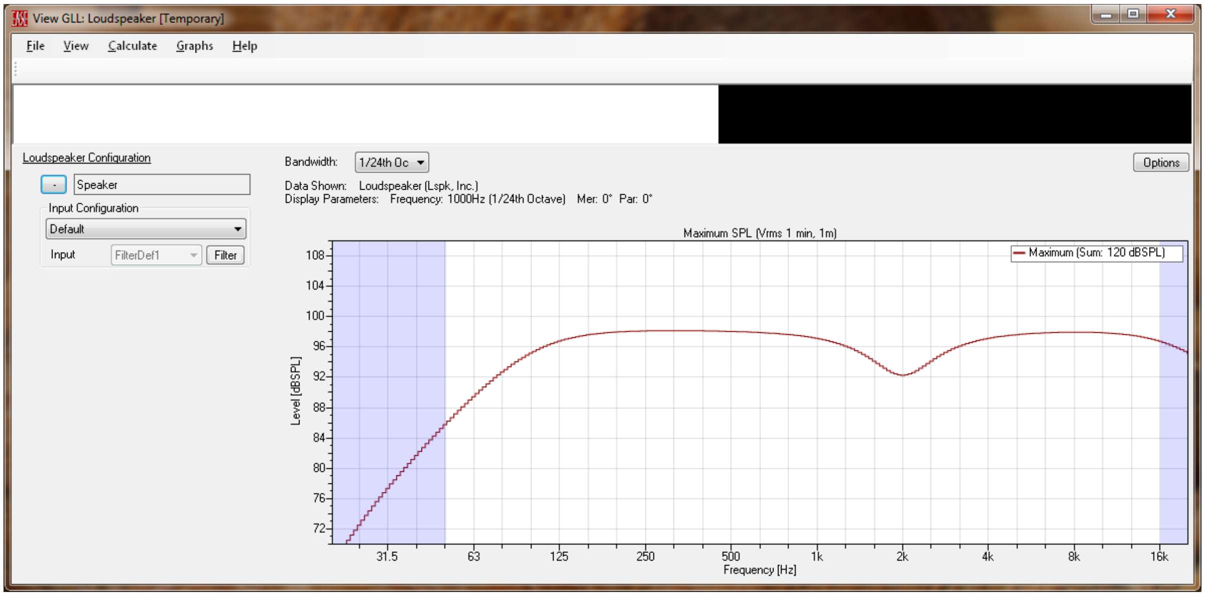
Data Shown: Loudspeaker (Lspk, Inc.)

Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-145 – Calculated sensitivity of the GLL

Now let's look at the maximum SPL. Click on GRAPHS > MAXIMUM SPL. Here we also see the effects of the high pass and EQ filters on the response of the system. However, the MAXIMUM SPL SUM is 120 dB, just as it was for the GSS. The maximum SPL was not reduced as a result of the filters. This is because the filters attenuated the signal by about 2 dB before the signal reached the source. Therefore the input to the GLL could be driven about 2 dB greater than the input to the GSS.

We can verify this by looking at the maximum input level to the GLL (GRAPHS > INPUT LEVEL). Here we see the spectral content of the signal as being relatively flat with -3 dB points at 20 Hz and 20 kHz (AES Broadband pink noise). The MAXIMUM INPUT SUM level is 30.9 dBV. This is almost 2 dB greater than the 29 dBV MAXIMUM INPUT SUM level for the GSS.



Speaker

Input Configuration: Default

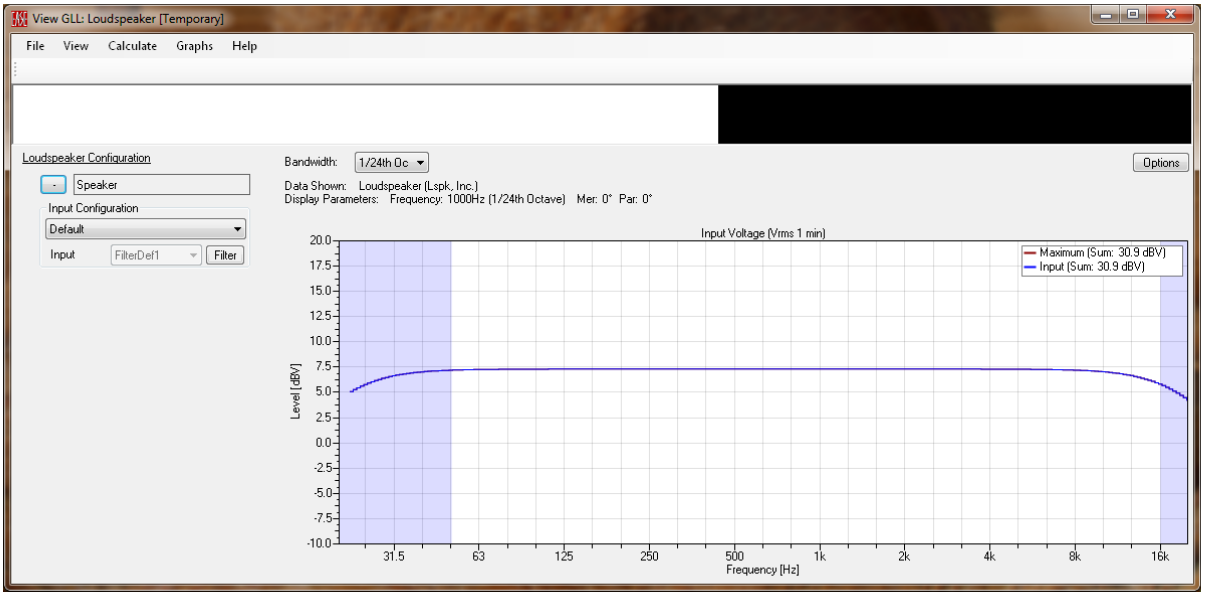
Input: FilterDef1

Bandwidth: 1/24th Oc

Data Shown: Loudspeaker (Lspk, Inc.)

Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-146 – Calculated maximum SPL of the GLL



Speaker

Input Configuration: Default

Input: FilterDef1

Bandwidth: 1/24th Oc

Data Shown: Loudspeaker (Lspk, Inc.)

Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-147 – Calculated maximum input voltage of the GLL

2.11.8 Internal Filters vs. Filters in Configuration File

Continuing from the running GLL in the example above, there is a FILTER button to the right of the Input label. Clicking this button will open a Filter Bank in the GLL. This is not the internal Filter Bank of the GLL. It is an external Filter Bank. This can be used to apply filtering before the input of the loudspeaker, like an external DSP for example. Even though this is an external filter the GLL still sees this as part of the loudspeaker system, as if it comes after the output of the power amplifier. See section 2.11.2 – GLL Overview as a Passive Loudspeaker for details about this.

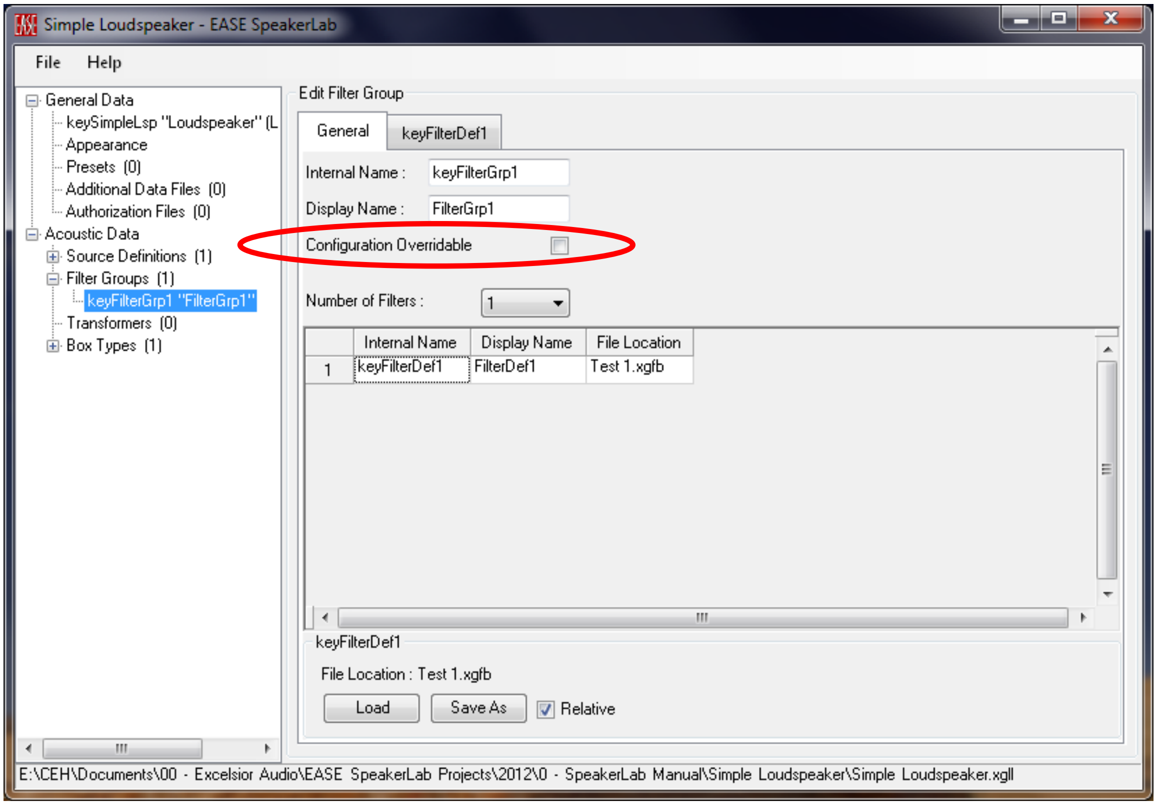
The separation of these Filter Banks may seem a bit confusing and unnecessary. However, they each can be very useful. The internal filter bank can be used to contain processing that is neither accessible to nor can it be changed by the end user of the loudspeaker system. We will call this locked processing. Examples of locked processing may be crossover filters for multi-source GLL (as we will explore later). Whether these are passive crossover, active filters, or DSP-based crossover, delay, and EQ filters the manufacturer may decide to not allow modification or user access to the parameters for modification.

For DSP-based loudspeaker systems the manufacturer may have front-end EQ that the end user is allowed to modify. The external filters for the GLL can be used to implement this processing. As was covered in section 5.1.7 – Configurations, the default settings for a loudspeaker may be entered into this external filter bank and then saved as a configuration. This configuration can then be included in the XGLL project as a preset.

2.11.9 Overriding Internal Filters

It is possible to have a configuration file that will override the internal filters in a GLL. One of the main reasons for overriding internal filters is to allow an otherwise "locked" internal filter to be changed. This may be necessary for a GLL to model the capabilities of an electronically steered line array or column loudspeaker. In this case the filter settings and delays that feed each driver (or groups of drivers) may need to change to yield the overall beam steering shape that is desired.

For this to occur, the CONFIGURATION OVERRIDABLE checkbox on the General tab of the EDIT FILTER GROUP page must be checked prior to compiling the GLL.



Internal Name : keyFilterGrp1

Display Name : FilterGrp1

Configuration Overridable □

Number of Filters : 1

	Internal Name	Display Name	File Location
1	keyFilterDef1	FilterDef1	Test 1.xgfb

keyFilterDef1

File Location : Test 1.xgfb

☑ Relative

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\Simple Loudspeaker\Simple Loudspeaker.xgll

Figure 2-148 – Edit Filter Group page showing the Configuration Overridable checkbox

The configuration file (XGLC file) must be formatted properly so that it will be accepted by the GLL and override the internal filter. An example of a properly formatted XGLC file is shown in Figure 2-149. Comment lines are preceded with a semicolon and are not required. They are there only to help make the syntax of the file more understandable.

```
"GLC"
"Format", "GLL Configuration"
"FormatVersion", "1.01"

"Configuration"
  "Used System", "Loudspeaker", "keySimpleLsp", "LS", 1.00, "Lspk, Inc."
  "Angle", 0.00
  "Elements", 1
    "Element"
      "Box Type", "keySpeakerBox"
      "Gain", 0.00
      "Input Config", "InputConfigA"
        "Inputs", 1
          "Input"

; The next line specifies the Internal Name of the internal filter to override
            "Internal Filter", "keyFilterDef1"

; The next line specifies the Display Name that will appear for the overriding filter
              "Override Setting", "*New Filter"

; The next line specifies the number of internal filters that will be overridden
              "Internal Filters To Override", 1

; The next line specifies the Internal Name of the Source (on the Acoustic Properties tab) to which the overriding filter specified immediately below will be applied
                "Override Internal Filter", "Source"

; The lines below specify the parameters of the overriding filer
                  "GFB"
                  "Format", "General Lspk"
                  "FormatVersion", "1.0"

                  "Bypass", "No"
                  "Gain", 0.00
                  "Delay", 0.00
                  "Invert Polarity", "No"
                  "Filters", 1
                    "Filter", "", ""
                    "Kind of Filter", "IIR Filter"
                    "Bypass", "No"
                    "Gain", 0.00
                    "Delay", 0.00
                    "Type", "High-Pass"
                    "Shape", "Bessel"
                    "Order", 3
                    "Frequency", 500.00
                    "Alignment", "-3dB"
                    "Q Factor", 1.00
                    "Parametric Gain", 10.00
```

```
             "External Filter"
               "GFB"
               "Format", "General Lspk"
               "FormatVersion", "1.0"

               "Bypass", "No"
               "Gain", 0.00
               "Delay", 0.00
               "Invert Polarity", "No"

               "Filters", 0
```

Figure 2-149 – Syntax for a properly formatted configuration file that will override the specified filters

2.12 Two-Source GLL

2.12.1 Overview: Multiple Sources per Box

With a two-way loudspeaker model (two-source GLL) the power and flexibility of the GLL approach begins to become clear. Here we have two separate acoustic sources defined by GSS files that can be connected to an internal crossover (or not for external biamp operation) with a given Input Configuration. In the event that the crossover filters should need to be modified to make changes to the system there is no need to re-measure the entire balloon response of the sources. It is sufficient to only update the filters.

There are many benefits to the separation between sources and filters, as well as the connection between them. All transducers are treated coherently, giving the model a better match to reality. With the old ***EASE SpeakerBase*** models it was very difficult to understand how to treat maximum input power in multi- way systems. The possibility to measure the electro-acoustic performance of the source independently, and the creation of specific GSS files for every source, allows for a better resolution of the whole box response.

We will create a two-source GLL model that has two different configurations from which the end user may select. The first is a 2-way passive system with an internal crossover (Figure 2-150). The second is a 2-way biamped system using external DSP-based crossover filters (Figure 2-151). This is a typical situation encountered with two-way loudspeaker systems that can be used with its internal passive crossover or biamped with external processing.

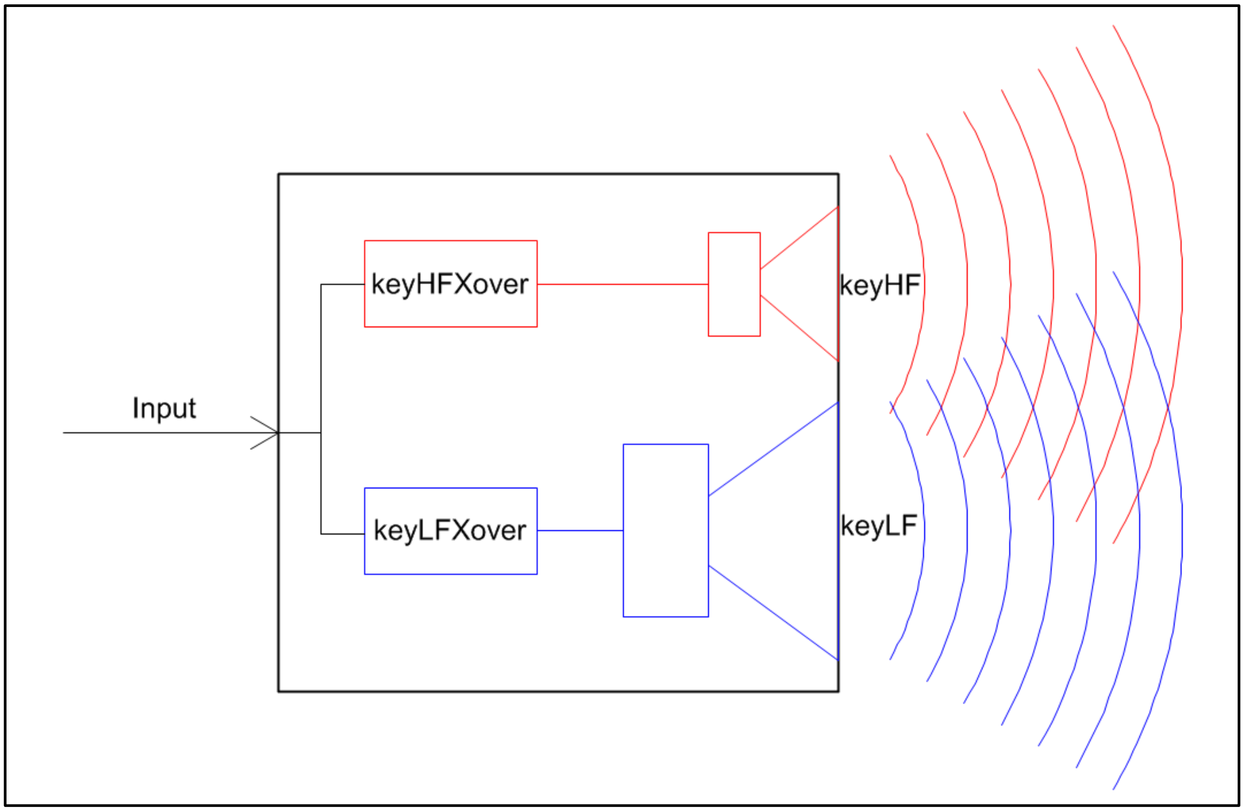


Figure 2-150 – Two-source GLL with a passive crossover

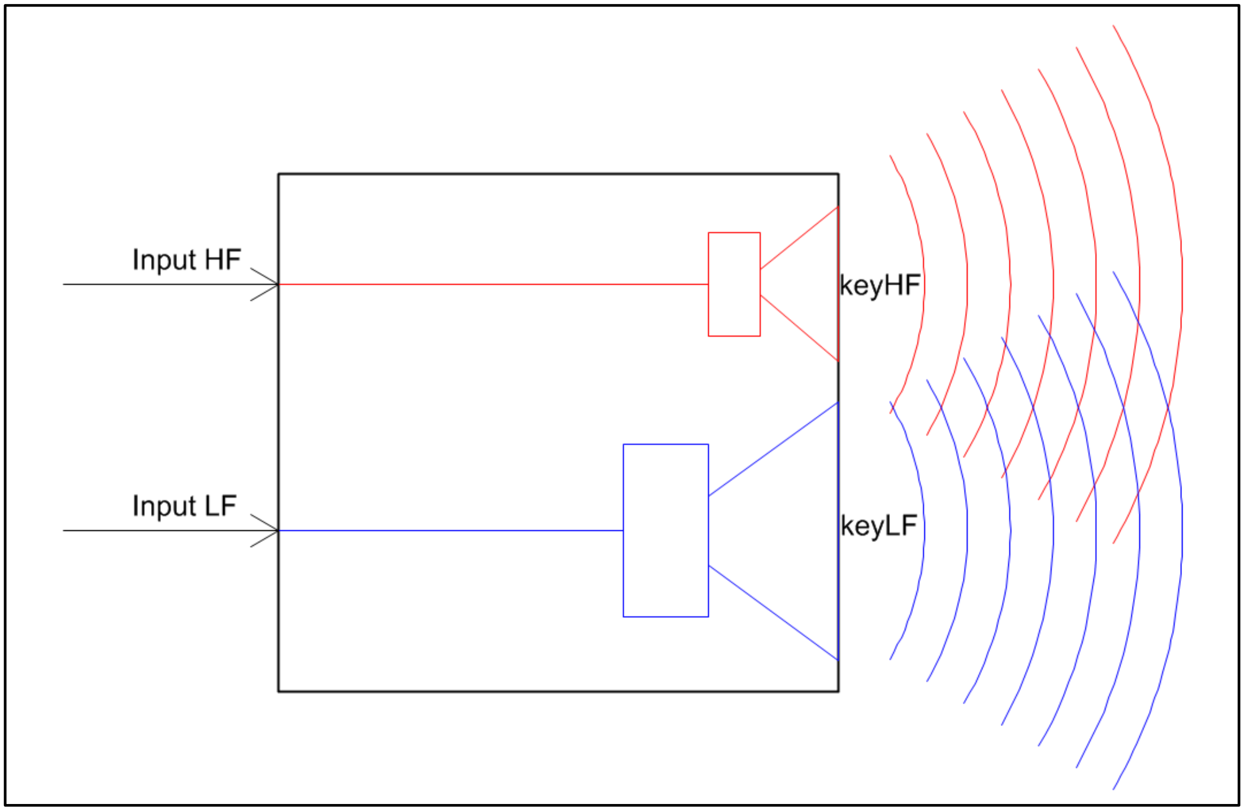


Figure 2-151 – Two-source GLL with an external crossover for biamp operation

Here again it is important to think about signal flow and functionality of the XGLL project as a passive loudspeaker. Please review Section 2.11.2 – GLL Overview as a Passive Loudspeaker if this is not already familiar.

2.12.2 Measured Data

We should have measured data for our model so we will first give a brief overview of this. Each actual radiating source that makes up the loudspeaker system needs to be accounted for separately in the model. When we say actual radiating source here, we mean a source which radiates into the listening environment. This should not be confused with multiple sources which may radiate into a common horn for example. It is this common horn that is the actual radiating source into the listening space.

For our example here we will use a loudspeaker with a direct radiating low frequency driver in a vented enclosure (LF pass band) and a compression driver coupled to a horn (HF pass band). This is in keeping with Figure 2-150 and Figure 2-151 shown above. The directivity balloon of each pass band must be measured separately, with no contribution from the other pass band. It is also critical that these measurements be made with the actual enclosure in which the pass band drivers/radiators will be used. Errors will result if attempting to use measured data from an LF pass band in some generic enclosure or baffle. Similarly, errors will result if using measured data of the HF horn alone (no enclosure) or in a generic enclosure. The acoustical loading and boundary conditions of the enclosure will affect the on- and off-axis response of the individual pass bands. Changing these boundary conditions will change the response.

When measuring one pass band the input terminals of the other pass band should be shorted together with as low of an impedance as possible. This will minimize the effect of the non-measured pass band on the measurement. The low impedance connection across the input terminals will minimize the diaphragm motion due to the self-damping of the motor system (voice coil and magnet).

Directivity balloon measurement with sufficient angular resolution to accurately characterize each pass band should be made. A sensitivity measurement of each pass band is also needed. These sensitivity measurements can be on-axis measurements from the directivity balloon data set, or they can be completely different measurements. It is important that the sensitivity measurement of each pass band be accurately referenced to the Point of Rotation (POR) for the directivity balloon measurements for that pass band. This serves as the linkage between the two within the GLL. The POR used for each pass band should also be noted. It will be needed when entering data into the XGLL project.

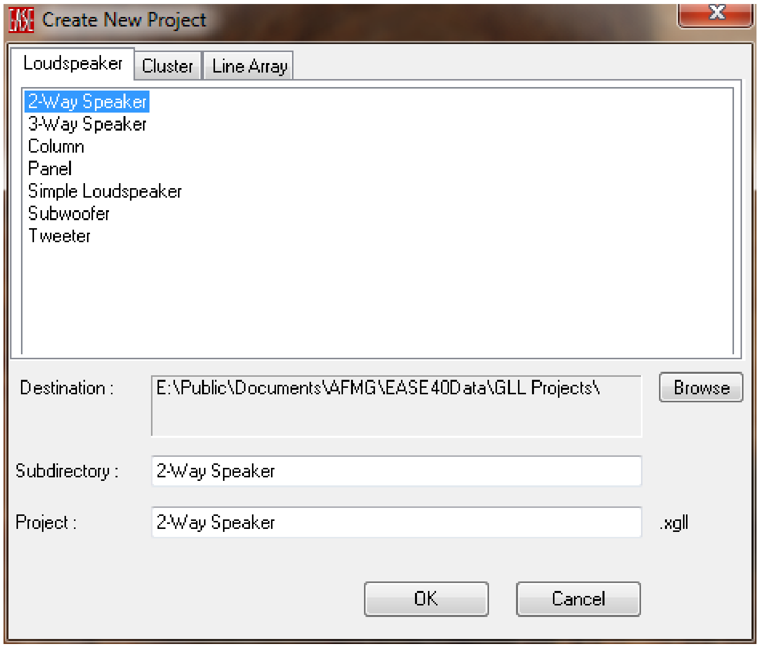
If at all possible there should be no filters used when measuring each of the pass bands. This may mean that small excitation signal levels need to used, particularly with high frequency drivers or small drivers with limited input capabilities. The objective with the directivity balloon and sensitivity measurements is to characterize the linear response of the pass bands.

The maximum input voltage should also be measured for each pass band. For these measurements it may be necessary to use a high pass filter to protect a driver from damaging voltages at frequencies outside of its intended pass band. A low pass filter may also be implemented here if necessary.

For additional details on measurements please see sections 8.1 – Definition of POR, 8.2 – Measurements and Uncertainties, 8.3 – Maximum Input Voltage & SPL Measurement, and 8.5 – Measurement Considerations, as well as [1], [2], [3], [4], [5], & [7].

2.12.3 Example: GLL With Passive Crossover

We will start from scratch by creating a new two-way loudspeaker project. Select File > New Project in the main menu. Next, select 2-Way Speaker in the Loudspeaker tab. Select the Destination folder if one different than the default is desired. Finally, click on OK.



Destination : E:\Public\Documents\AFMG\EASE40Data\GLL Projects\

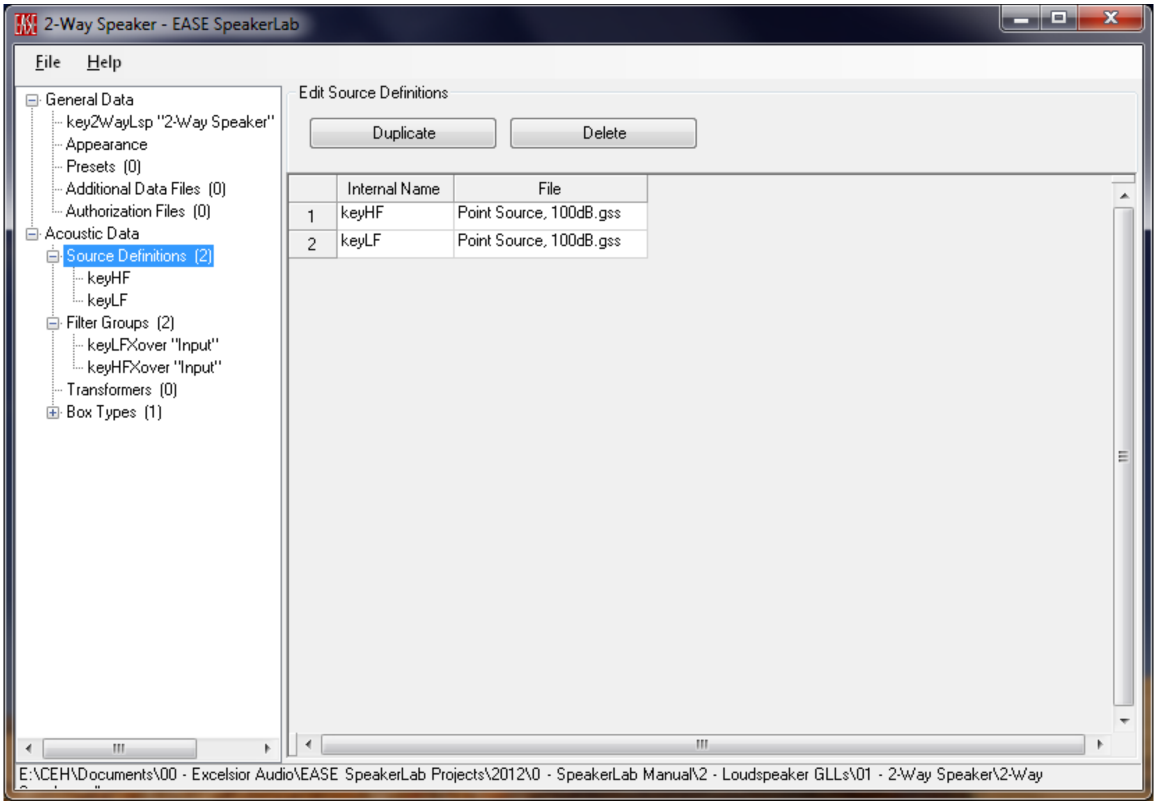
Subdirectory : 2-Way Speaker

Project : 2-Way Speaker .xgll

Figure 2-152 – Create New Project dialog box

We will skip over the General Data and proceed directly to entering the Acoustic Data. The first step is to define the acoustic sources, so expand the Acoustic Data item from the property tree. Selecting the Source Definitions item in the property tree shows the table on the Edit Source Definitions page already contains with two rows corresponding to the two sources. The HF source has the Internal Name keyHF and the LF source with Internal Name keyLF. As the default for the two-way loudspeaker template the sources are set to <SPHERE>. To use the measured data of real sources it is necessary to create and use two GSS files, one for the LF section and one for the HF section of the loudspeaker.

For our example here we will use the `Point Source, 100dB.gss` file we created in section 2.11.6. Please copy this GSS file from its folder into the folder of this current example. After doing this select this GSS file as the Source Definition for both the keyHF and keyLF sources.



- General Data
  - key2WayLsp "2-Way Speaker"
  - Appearance
  - Presets (0)
  - Additional Data Files (0)
  - Authorization Files (0)
- Acoustic Data
  - Source Definitions (2)
    - keyHF
    - keyLF
  - Filter Groups (2)
    - keyLFXover "Input"
    - keyHFXover "Input"
  - Transformers (0)
  - Box Types (1)

	Internal Name	File
1	keyHF	Point Source, 100dB.gss
2	keyLF	Point Source, 100dB.gss

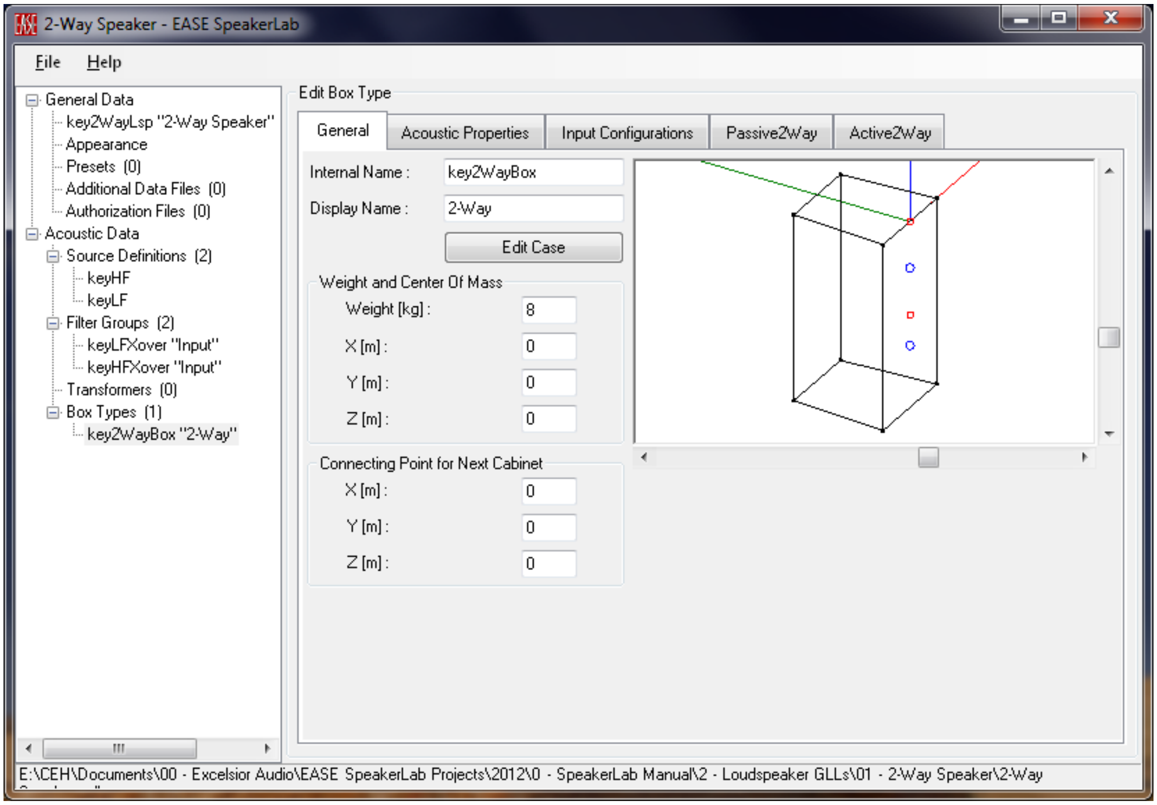
E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\2 - Loudspeaker GLLs\01 - 2-Way Speaker\2-Way

Figure 2-153 – Edit Source Definitions page with sources specified

2.12.3.1 Source Positions

The geometrical position of the two sources relative to the coordinate origin of the loudspeaker model can be specified in the Edit Box Type page. Expand the Box Types item in the property tree and select the Key2Way Box.

On the General tab it's possible to edit some general properties of the box such as the Names, Weight, location for the Center of Mass, etc. We will use the default loudspeaker case drawing for this example. For now set all of the coordinates for the Center of Mass to "0". This will move it to the origin and be out of the way for the rest of this example. As the coordinates are updated the small red circle noting its position will move.



2-Way Speaker - EASE SpeakerLab

File Help

- General Data
  - key2WayLsp "2-Way Speaker"
  - Appearance
  - Presets (0)
  - Additional Data Files (0)
  - Authorization Files (0)
- Acoustic Data
  - Source Definitions (2)
    - keyHF
    - keyLF
  - Filter Groups (2)
    - keyLFXover "Input"
    - keyHFXover "Input"
  - Transformers (0)
  - Box Types (1)
    - key2WayBox "2-Way"

Edit Box Type

General | Acoustic Properties | Input Configurations | Passive2Way | Active2Way

Internal Name : key2WayBox

Display Name : 2-Way

Edit Case

Weight and Center Of Mass

Weight [kg] : 8

X [m] : 0

Y [m] : 0

Z [m] : 0

Connecting Point for Next Cabinet

X [m] : 0

Y [m] : 0

Z [m] : 0

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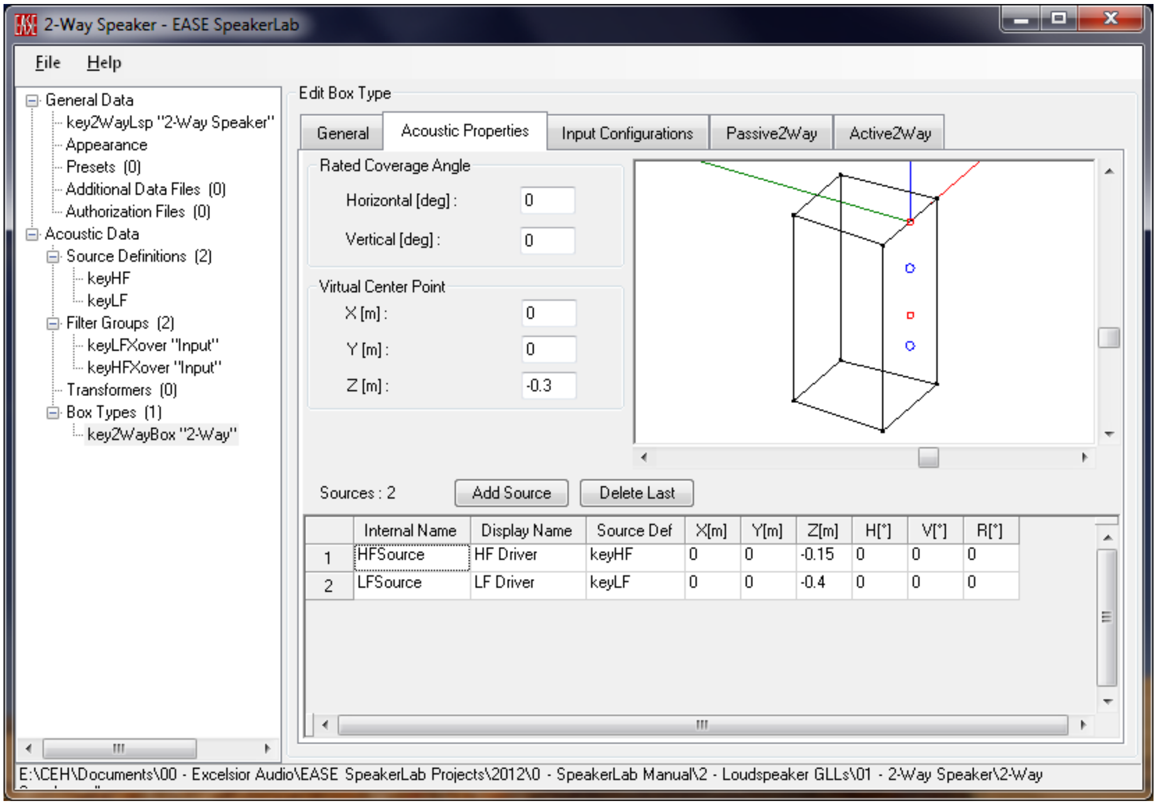
Figure 2-154 – General tab on the Edit Box Type page



2-Way Speaker - EASE SpeakerLab

File Help

- General Data
  - key2WayLsp "2-Way Speaker"
  - Appearance
  - Presets (0)
  - Additional Data Files (0)
  - Authorization Files (0)
- Acoustic Data
  - Source Definitions (2)
    - keyHF
    - keyLF
  - Filter Groups (2)
    - keyLFXover "Input"
    - keyHFXover "Input"
  - Transformers (0)
  - Box Types (1)
    - key2WayBox "2-Way"

Edit Box Type

General | Acoustic Properties | Input Configurations | Passive2Way | Active2Way

Rated Coverage Angle

Horizontal [deg] : 0

Vertical [deg] : 0

Virtual Center Point

X [m] : 0

Y [m] : 0

Z [m] : -0.3

Sources : 2 Add Source Delete Last

	Internal Name	Display Name	Source Def	X[m]	Y[m]	Z[m]	H[°]	V[°]	R[°]
1	HFSource	HF Driver	keyHF	0	0	-0.15	0	0	0
2	LFSource	LF Driver	keyLF	0	0	-0.4	0	0	0

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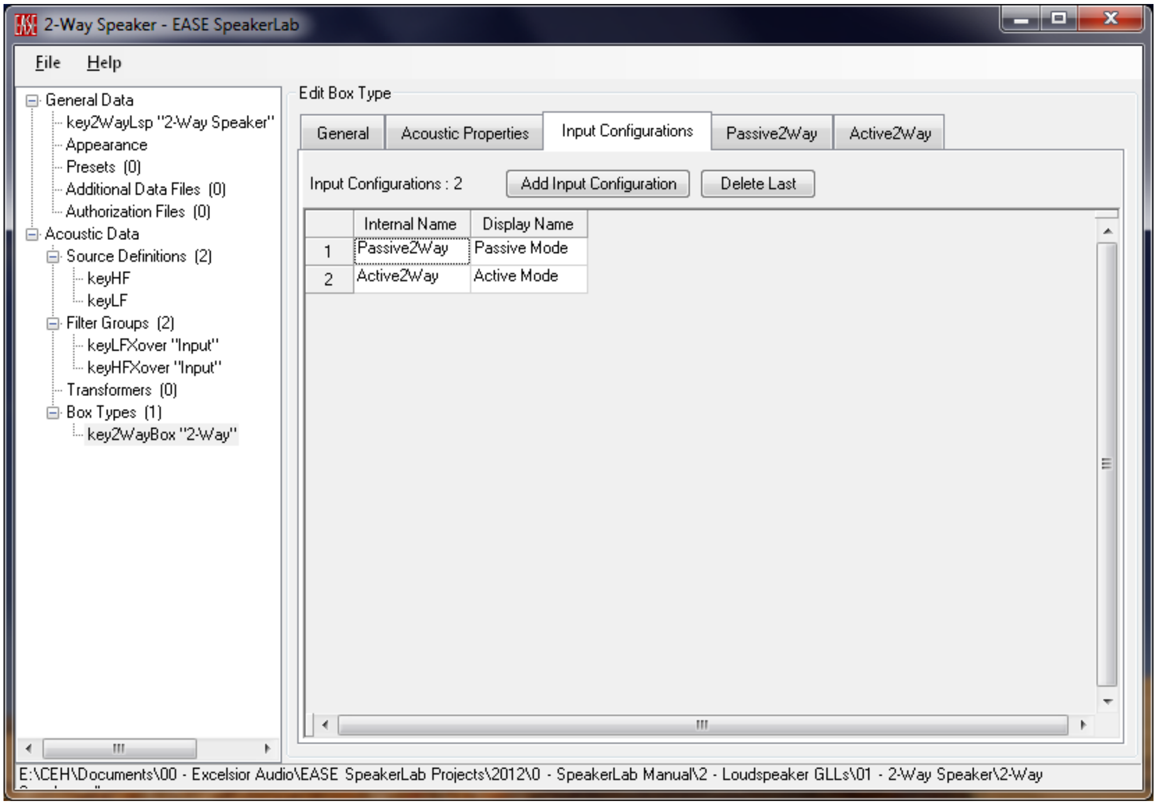
Figure 2-155 – Acoustic Properties tab on the Edit Box Type page

The Acoustic Properties tab is where the sources and their positions are specified. In this example we have two sources, named HFSource and LFSource. The sources are not located at the same position. Some care must be taken defining the source position relative to the coordinate system of the loudspeaker, i.e. relative to the enclosure. The source position must be the Point of Rotation adopted during the directivity balloon response measurement. Let’s move the HFSource up just a bit. Double-click on its value in the Z [m] column and change it from -0.2 to “-0.15”.

2.12.3.2 Connection of Filters & Sources

Once the sources are defined and placed into the box, it is necessary to define the connections between the electrical inputs to the loudspeaker box and acoustical outputs. As described previously this is managed from the Input Configurations tab.

In this template there are two default configurations Passive2Way and Active2Way. These Internal Names can be changed if desired, as can the Display Names. We will leave them as they are for now.



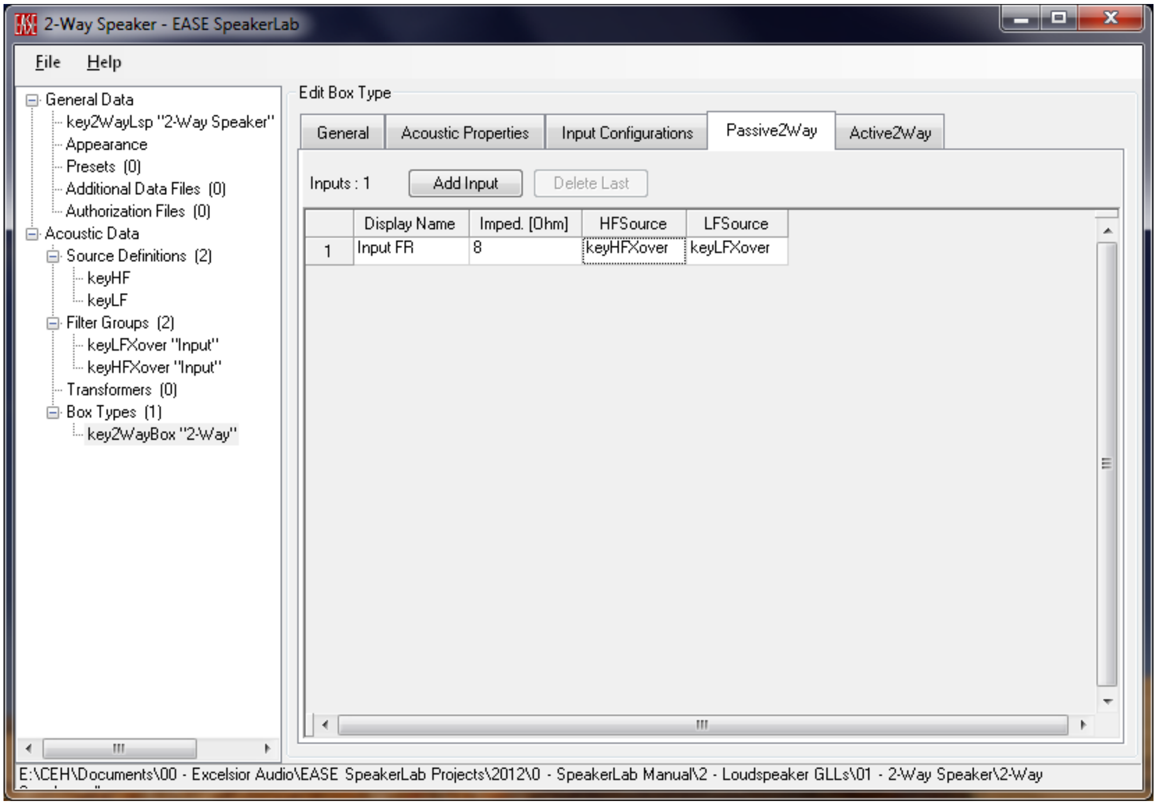
Input Configurations : 2

	Internal Name	Display Name
1	Passive2Way	Passive Mode
2	Active2Way	Active Mode

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\2 - Loudspeaker GLLs\01 - 2-Way Speaker\2-Way

Figure 2-156 – Input Configurations tab on the Edit Box Type page

Selecting the Passive2Way tab shows the connection matrix. This table only has one row, meaning that there is only a single input to the loudspeaker system for this configuration. A single input is all that is needed for a passive 2-way loudspeaker system. There are two columns, one for each of the sources in the XGLL project. The matrix shows that the signal is routed through the keyHFXover filter before going to the HFSource. Similarly, the signal is routed through the keyLFXover filter before reaching the LFSource.

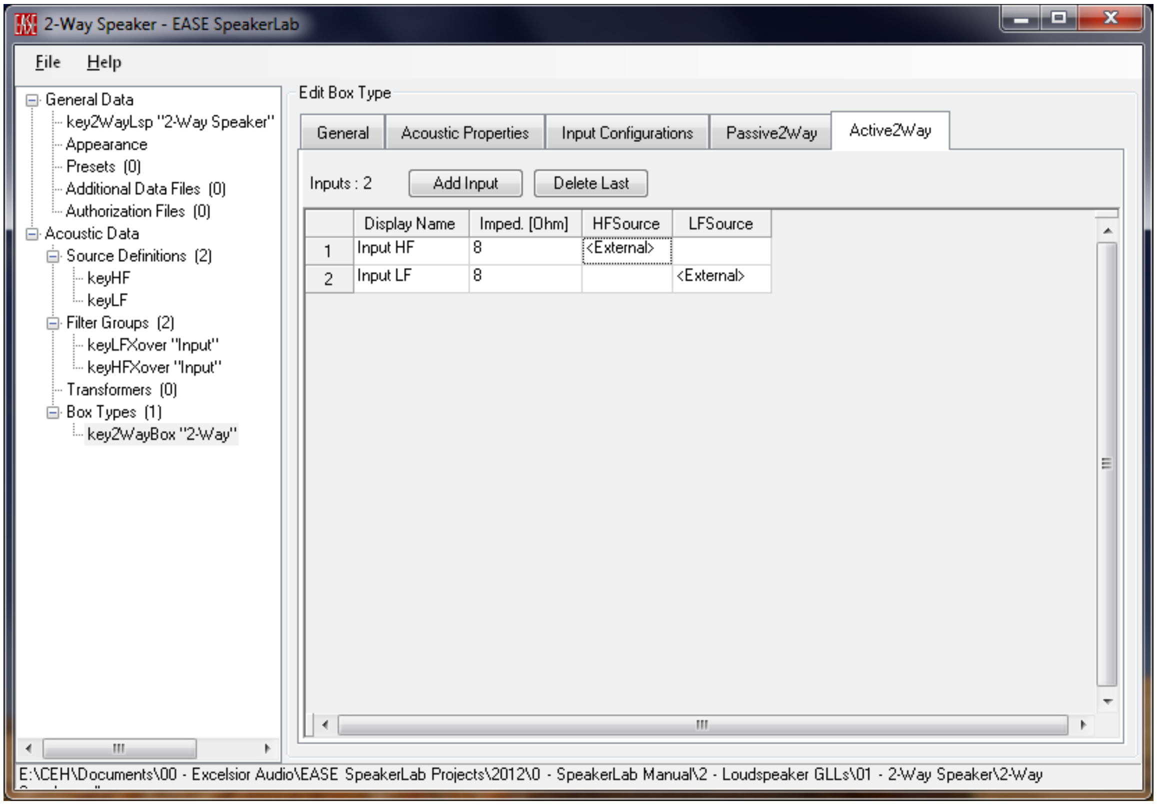


	Display Name	Imped. [Ohm]	HFSource	LFSource
1	Input FR	8	keyHFXover	keyLFXover

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\2 - Loudspeaker GLLs\01 - 2-Way Speaker\2-Way

Figure 2-157 – Input Configuration tab (Passive2Way) on the Edit Box Type page

Selecting the Active2Way tab shows that this input configuration uses two separate inputs. One input is connected directly to the HFSource. The other input is connected directly to the LFSource. The direct connections, without any filters, are accomplished by using the <External> selection. This is the signal routing typically used for biamp operation of a loudspeaker system. Empty slots (cells) in the connection matrix indicate that there is no connection between a given input and a sound source. It is possible for one input to feed more than one sound source if this is required to accurately model the functionality of a loudspeaker system in the GLL. In any case, each source should be connected to exactly one input.



Inputs : 2

	Display Name	Imped. [Ohm]	HFSource	LFSource
1	Input HF	8	<External>	
2	Input LF	8		<External>

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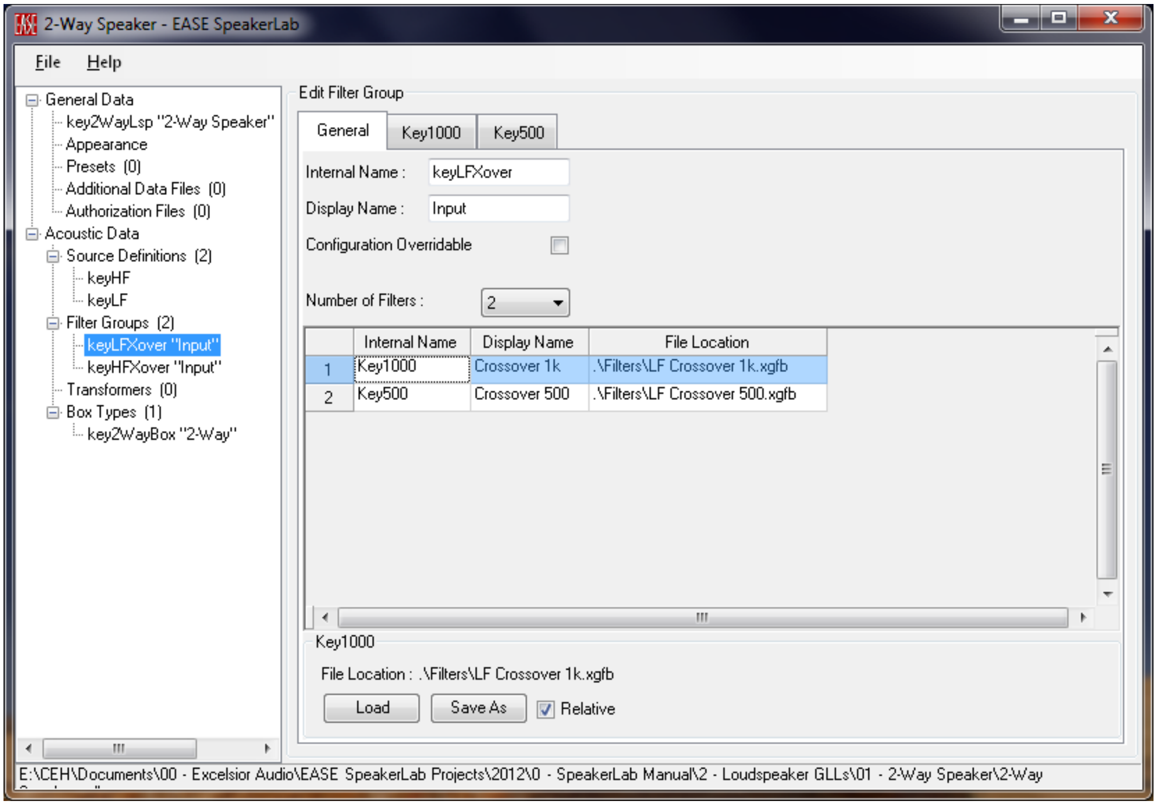
Figure 2-158 – Input Configuration tab (Active2Way) on the Edit Box Type page

2.12.3.3 Filter Groups

At this point we must define these filters. Expand the Filter Groups item from the property tree. Select the keyLFXover item. The Edit Filter Group page shows that this template has two different Filter Definitions already assigned to each Filter Group. Select the keyHFXover item from the property tree. Notice that it also has two Filter Definitions. It is required that each Filter Group have the same number of Filter Definitions when they are to be switched together, i.e. used for the same input. Note also that the Internal Name of the Filter Definitions is the same for both the keyLFXover and keyHFXover Filter Groups. This is also required to link the selection of the different Filter Definitions across the Filter Groups of the same input.



Internal Name : keyLFXover

Display Name : Input

Configuration Overridable □

Number of Filters : 2

	Internal Name	Display Name	File Location
1	Key1000	Crossover 1k	.\Filters\LF Crossover 1k.xgfb
2	Key500	Crossover 500	.\Filters\LF Crossover 500.xgfb

Key1000

File Location : .\Filters\LF Crossover 1k.xgfb

☑ Relative

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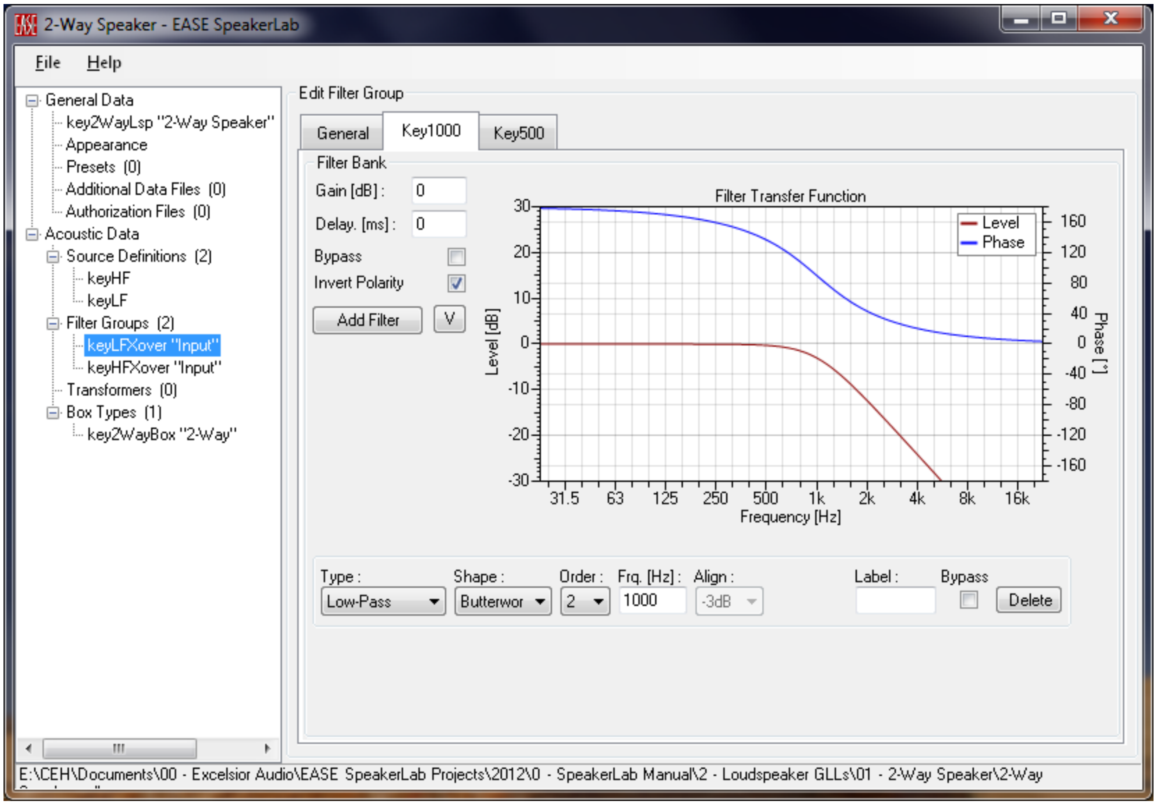
Figure 2-159 – Two filters in the LF filter group



Gain [dB] : 0

Delay. [ms] : 0

Bypass □

Invert Polarity ☑

Type :	Shape :	Order :	Frq. [Hz] :	Align :	Label :	Bypass
Low-Pass	Butterwor	2	1000	-3dB		□

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Figure 2-160 – The 1,000 Hz low pass filter

Select the KEY1000 tab. Here we see that the filter in this template has been defined using a 1/3 octave data table. We will change this to use an IIR FILTER; second order Butterworth at 1 kHz with inverted polarity.

1. Click the DELETE button for the filter table.
2. Click the ADD FILTER button & select IIR FILTER.
3. Check the INVERT POLARITY box.

Now let's change the 500 Hz filter in a similar manner. Select the KEY500 tab and do the following.

1. Click the DELETE button for the filter table.
2. Click the ADD FILTER button & select IIR FILTER.
3. Change the filter frequency from 1000 to "500".
4. Check the INVERT POLARITY box.

We now must save these filters.

1. Select the GENERAL tab.
2. Click anywhere in the top (KEY1000) row.
3. At the bottom of the page click the SAVE AS button and then click SAVE in the SAVE FILE AS dialog box that is opened.
4. A confirmation dialog box will appear asking if you want to replace the existing file. Click YES.
5. Finally, check the RELATIVE box.

Repeat the above steps for the KEY500 row.

We must also change the filters for the high frequency section of the loudspeaker model. Select the KEYHFXOVER item from the property tree. Select the KEY1000 tab on the EDIT FILTER GROUP page and perform the following steps.

1. Click the DELETE button for the filter table.
2. Click the ADD FILTER button & select IIR FILTER.
3. Change the TYPE from LOW-PASS to HIGH-PASS.

Select the KEY500 tab ad do the following.

1. Click the DELETE button for the filter table.
2. Click the ADD FILTER button & select IIR FILTER.
3. Change the TYPE from LOW-PASS to HIGH-PASS.
4. Change the filter frequency from 1000 to "500".

Finally, we must save the high frequency section filters we just changed.

1. Select the GENERAL tab.
2. Click anywhere in the top (KEY1000) row.
3. At the bottom of the page click the SAVE AS button and then click SAVE in the SAVE FILE AS dialog box that is opened.
4. A confirmation dialog box will appear asking if you want to replace the existing file. Click YES.
5. Finally, check the RELATIVE box.

Repeat the above steps for the KEY500 row.

Now, save the XGLL project file by clicking on FILE > SAVE PROJECT. These filters will emulate our internal passive crossover network.

2.12.3.4 Running the GLL

Let's run the GLL now to see how these configurations affect the functionality of the GLL for the end users. Click on FILE > RUN GLL (or hit F5). Click the + to the left of the 2-Way label under LOUDSPEAKER CONFIGURATION. We can see that the INPUT CONFIGURATION dropdown box now allows two options, PASSIVE MODE and ACTIVE MODE. The default, Passive Mode (because it is first the first Input Configuration in the XGLL project) has a single full-range input (Input FR). We can also see that the dropdown selection box for this input has two options, CROSSOVER 1K and CROSSOVER 500.

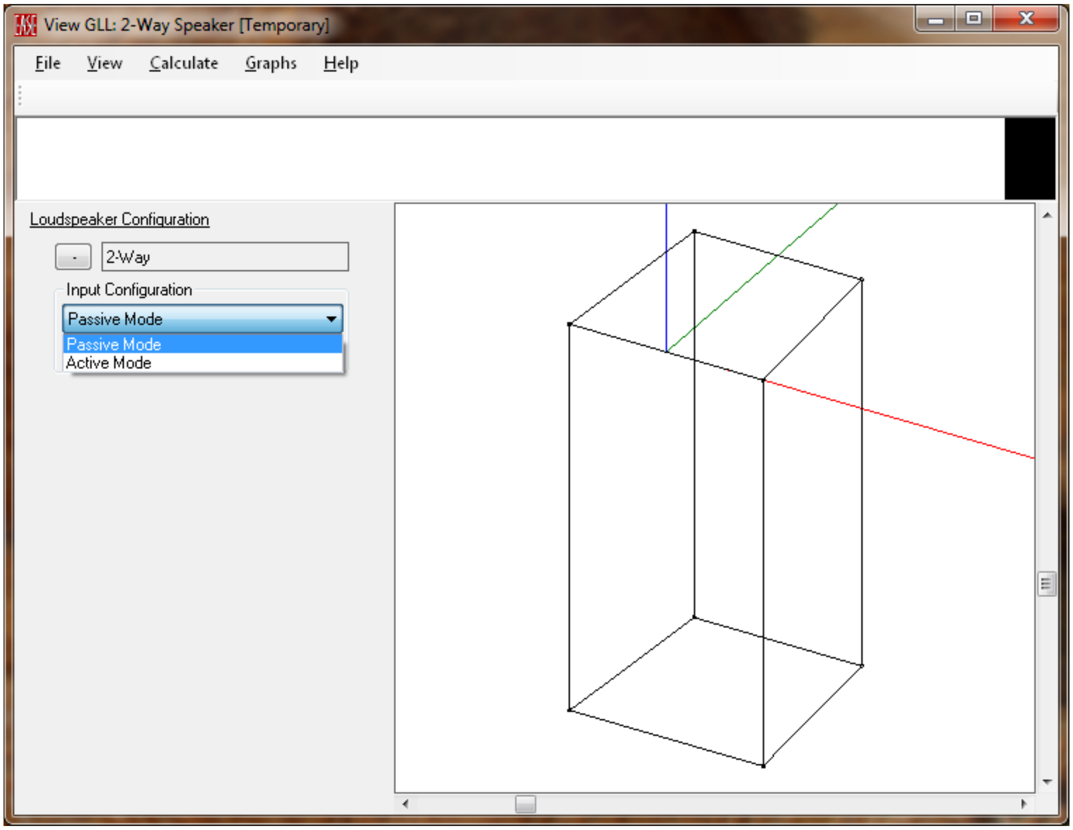


Figure 2-161 – Running GLL showing Input Configuration selections

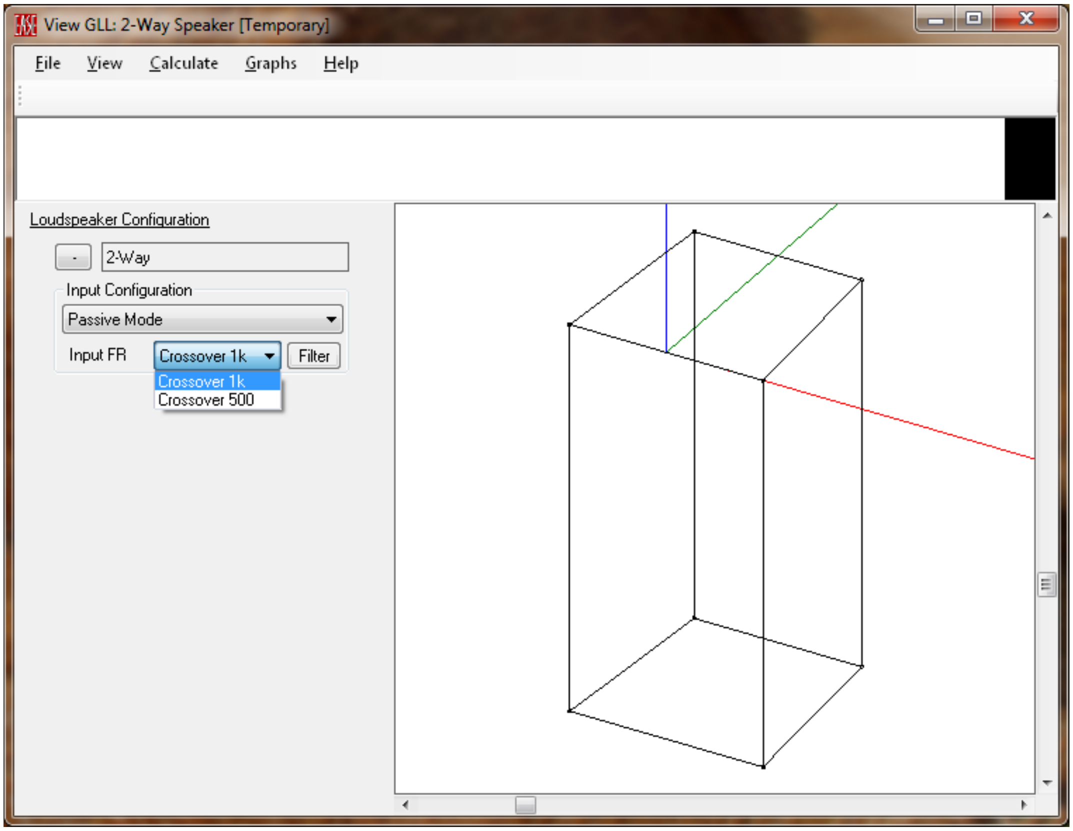
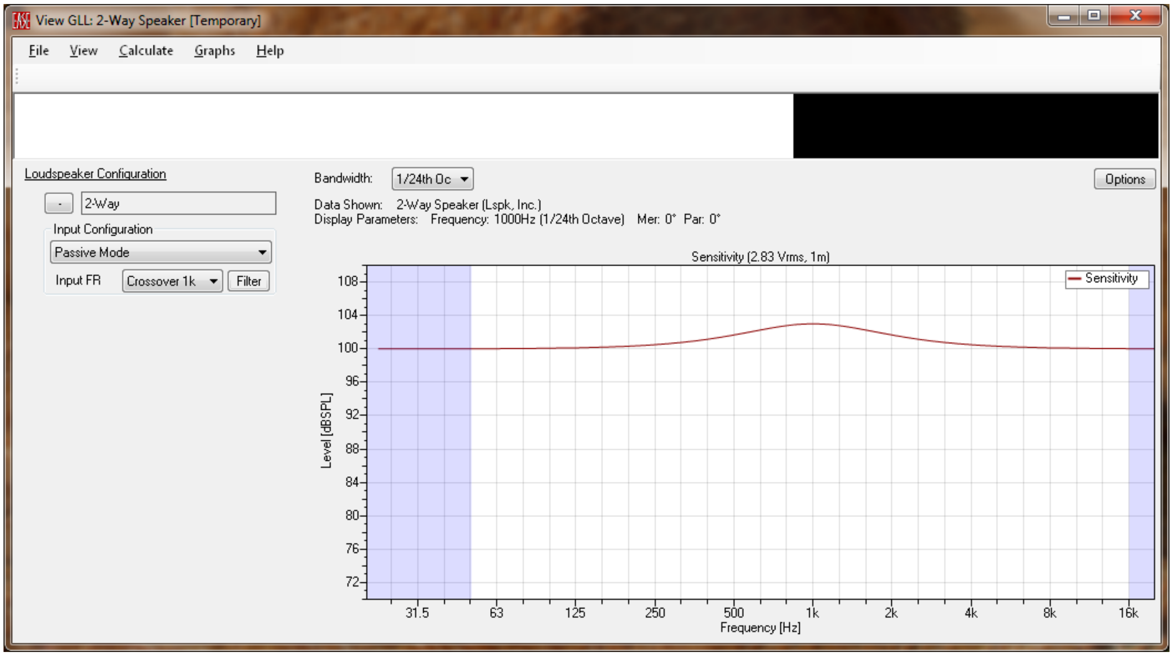


Figure 2-162 – Running GLL showing internal crossover/filter selections

Let's calculate the response of the GLL using the 1 kHz crossover. Click on CALCULATE > BALLOON (or hit F5). In the CALCULATION PARAMETERS dialog box change the RESOLUTION from 10° to 5 °. All of the other parameters can stay at their default settings. Click on OK to begin the calculation. When it is complete a directivity balloon at 1 kHz should be displayed. Let's look instead at the sensitivity (GRAPHS > SENSITIVITY) for a moment. Here we see that the response is relatively flat with a 3 dB peak at the crossover frequency; exactly what we would expect for second order Butterworth filters with one of them having inverted polarity.



2-Way

Input Configuration: Passive Mode

Input FR: Crossover 1k

Bandwidth: 1/24th Oc

Data Shown: 2-Way Speaker (Lspk, Inc.)
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

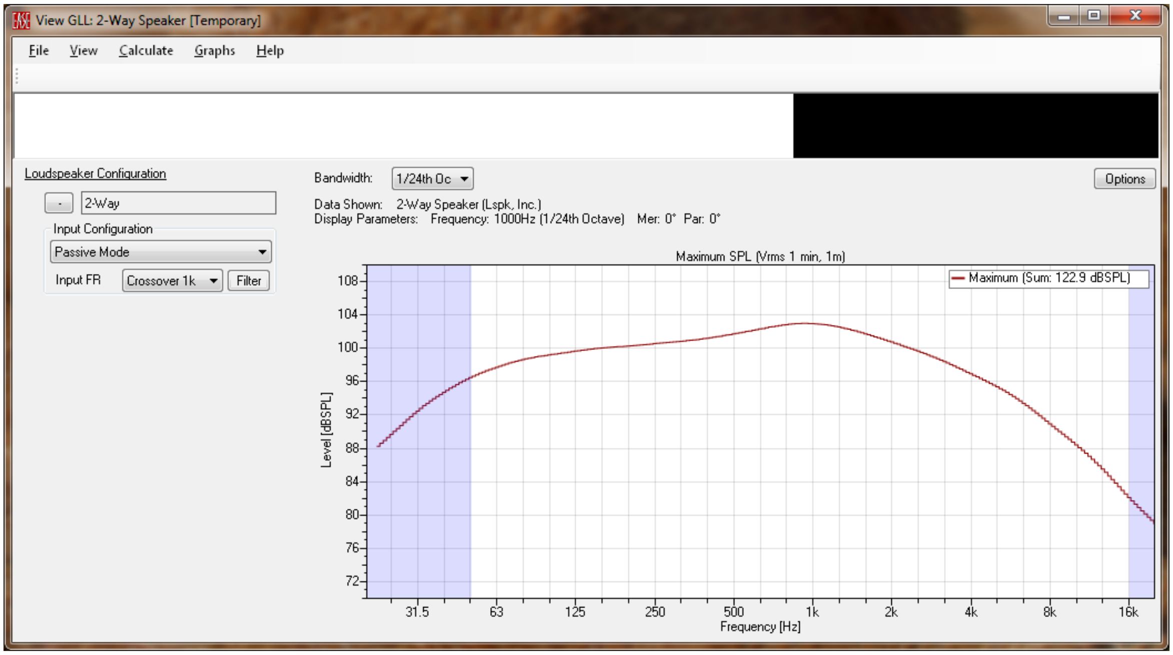
Sensitivity (2.83 Vrms, 1m)

Figure 2-163 – Running GLL showing the calculated sensitivity with the 1 kHz crossover

Now lets' examine the maximum SPL (GRAPHS > MAXIMUM SPL). Here we see the spectral content of the IEC 60268 noise along with the Maximum SPL Sum of 122.9 dB. Change the crossover from Crossover 1k to Crossover 500 and recalculate the response (CALCULATE > RECALCULATE or CTRL+F5). Notice that the Maximum SPL Sum changed to 124.2 dB. This occurred because the LF source was limiting the maximum input voltage to the system. With the 1 kHz crossover the maximum input voltage (MIV) for the LF source was being reached before the MIV for the HF source. Moving the crossover frequency down to 500 Hz sends less energy to the LF and more to the HF. The result is that the MIV for the system is now slightly higher and the system can get slightly louder.

Note that this occurs in this example only because the same GSS file (with an MIV of 28.3 V) is being used for both the LF and HF source. Real HF drivers rarely have the same MIV as LF drivers. Typically moving the crossover frequency from 1 kHz to 500 Hz will not result in higher system SPL.

However, this example demonstrates a very powerful feature of SpeakerLab and the GLL format. When multi-source GLLs are used they can more accurately predict the maximum SPL and the maximum input voltage for the loudspeaker system model. A single-source GLL, or other loudspeaker model formats that treat the loudspeaker as a single-source, cannot do this.



Loudspeaker Configuration: 2-Way

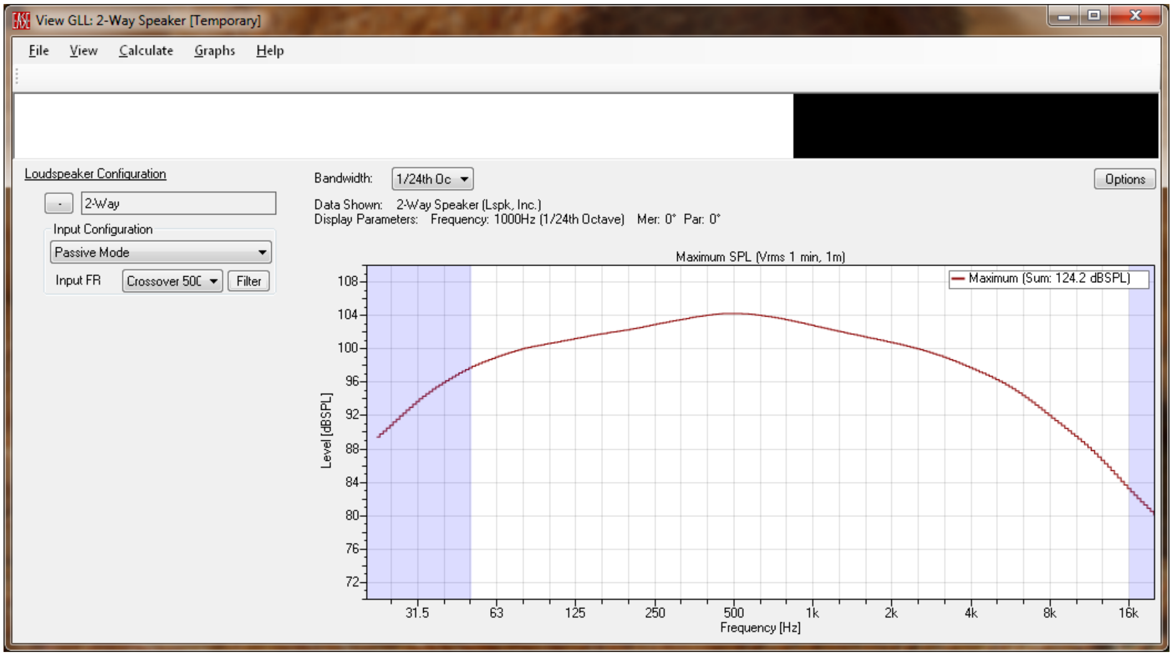
Input Configuration: Passive Mode

Input FR: Crossover 1k

Bandwidth: 1/24th Oc

Data Shown: 2-Way Speaker (Lspk, Inc.)
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-164 – Calculated maximum SPL with the 1 kHz crossover



Loudspeaker Configuration: 2-Way

Input Configuration: Passive Mode

Input FR: Crossover 50C

Bandwidth: 1/24th Oc

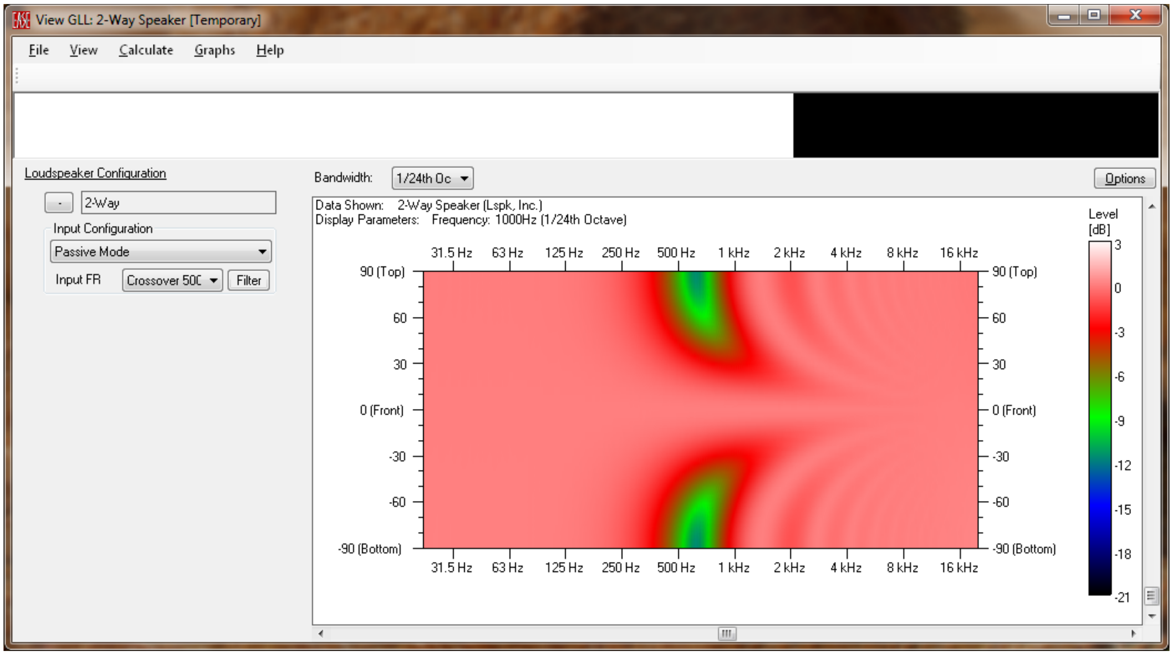
Data Shown: 2-Way Speaker (Lspk, Inc.)
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-165 – Calculated maximum SPL with the 500 Hz crossover

Before we move on let us also examine the vertical directivity map (GRAPHS > VERTICAL MAP) for this GLL for each crossover frequency. The initial view will show a 3D perspective of the directivity map. This can be changed to an overhead (plan) view by clicking VIEW > PLAN VIEW (Z) (or ALT+Z three times). Each time the ALT+Z command is issued it will rotate the map 90°. It takes three rotations until lower frequencies are at the left and higher frequencies are at the right. The map should look something like Figure 2-166. To make it look exactly like Figure 2-166 click on OPTIONS and check the FIXED COLOR LEGEND box and change the UPPER LIMIT [DB] to 3 and the LOWER LIMIT [DB] to -21. Then click OK.

Notice the off-axis cancellations denoted by the green coloring just above the 500 Hz crossover frequency. Now change the crossover to Crossover 1k and recalculate the response (CALCULATE > RECALCULATE or CTRL+F5). Notice how the cancellations not only move a bit higher in frequency but also got deeper (blue color) and now occur at angles closer to on-axis.

This type of data visualization and analysis can be extremely useful to help in optimizing the directivity response of a loudspeaker system through the crossover region.



View GLL: 2-Way Speaker [Temporary]

Loudspeaker Configuration: 2-Way

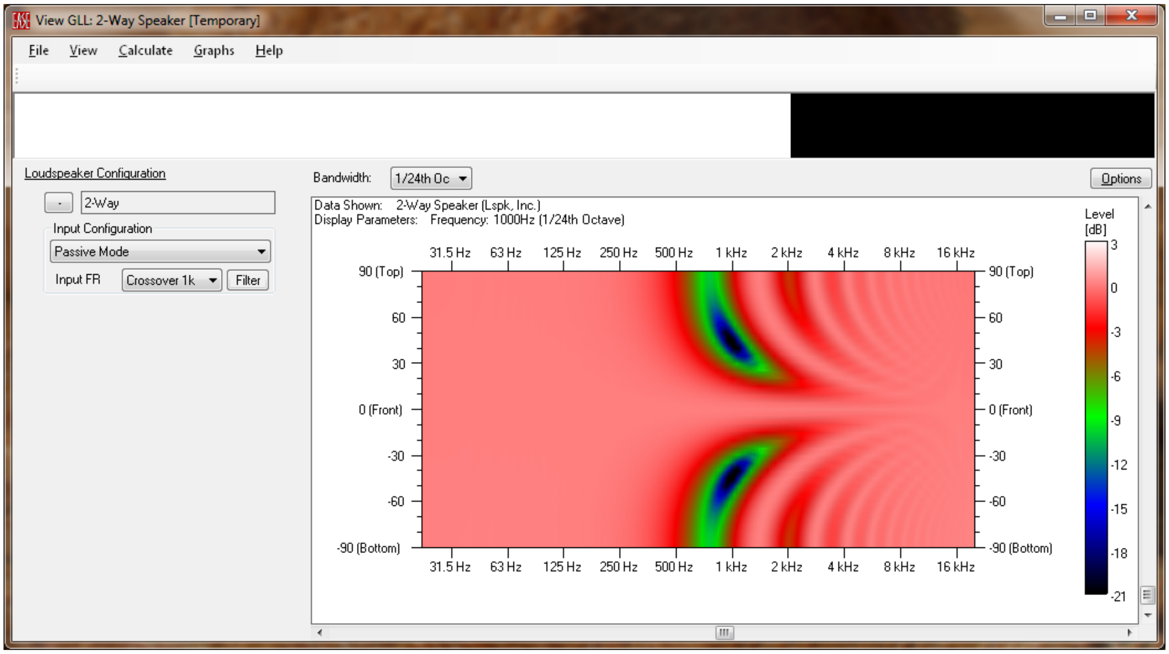
Input Configuration: Passive Mode

Input FR: Crossover 50C

Bandwidth: 1/24th Oc

Data Shown: 2-Way Speaker (Lspk, Inc.)
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-166 – Vertical directivity map with the 500 Hz crossover



View GLL: 2-Way Speaker [Temporary]

Loudspeaker Configuration: 2-Way

Input Configuration: Passive Mode

Input FR: Crossover 1k

Bandwidth: 1/24th Oc

Data Shown: 2-Way Speaker (Lspk, Inc.)
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-167 – Vertical directivity map with the 1 kHz crossover

2.12.4 Example: Different Input Options with Filter Groups (Internal Filter Switching)

The example given here is useful for modeling filters internal to the loudspeaker system. Whether the loudspeaker is passive or powered with internal filters and amplifiers makes no difference. The filters are internal and not accessible to the end user. No modifications can be made to the internal filter in the real loudspeaker but the end user can select between different options for the internal filters.

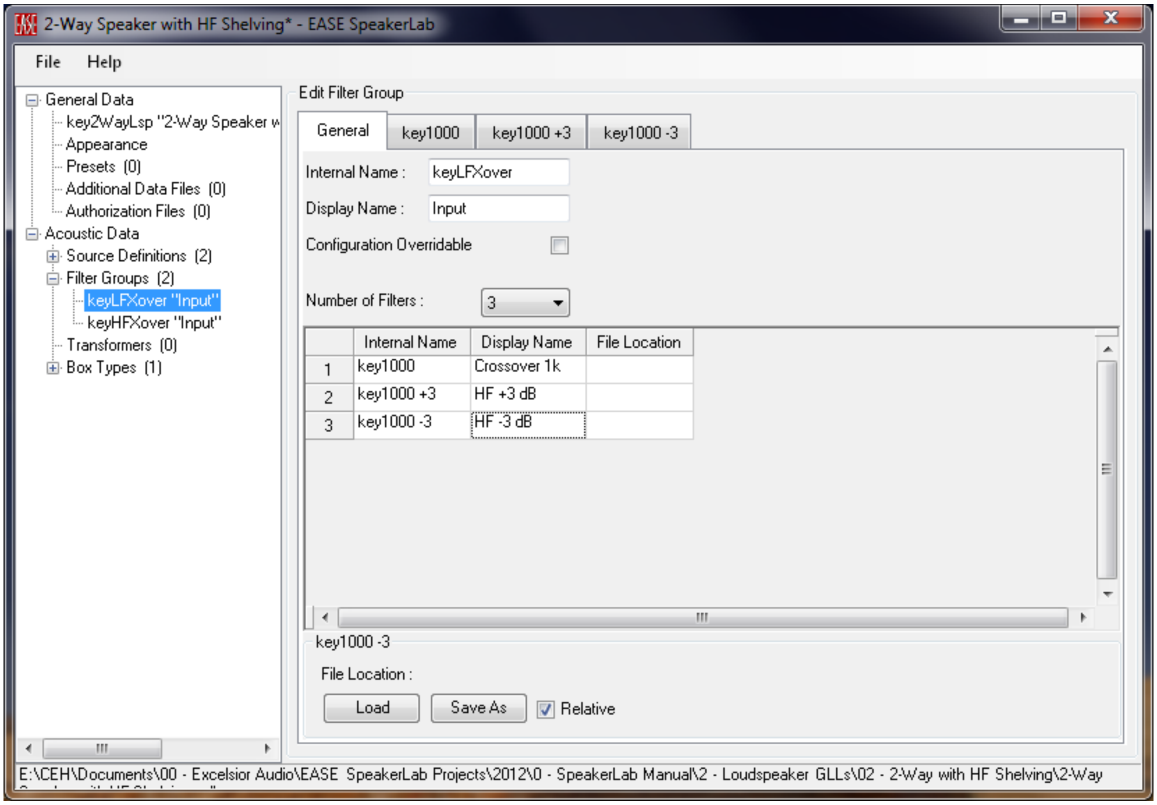
We will continue working on the project file from the previous section. This example will also use the passive crossover at 1 kHz but will incorporate a high frequency shelving filter to tailor the HF output level to better suit different listener preferences. This will emulate a switch on the loudspeaker to give a 3 dB boost or cut to the high frequency part of the spectrum.

So that we will have a flat sensitivity as a base line let's change the crossover filters to fourth order Linkwitz-Riley filters. Select KEYLFXOVER "INPUT" in the property tree. In the EDIT FILTER GROUP page on the right select the KEY1000 tab. Change the SHAPE of the filter by selecting LINKWITZ-RILEY using the dropdown box. Change the filter ORDER by selecting "4" its dropdown box. We must now save the filter. Select the GENERAL tab then click anywhere in row 1 of the filter table. Click the SAVE AS button at the bottom of the page. Let's name this file "LF - 1k, LR4.xgfb". Finally, since we will not need the 500 Hz crossover any longer we can delete it. Change the NUMBER OF FILTERS from 2 to "1" using the dropdown box. Save the XGLL project (FILE > SAVE PROJECT or CTRL+S).

Repeat these same steps for the KEYHFXOVER. Save this file as "HF - 1k, LR4.xgfb". Save the XGLL project (FILE > SAVE PROJECT or CTRL+S).

We will now add the two other filter entries to both the LF and HF crossover filter groups so that we can include options for a 3 dB boost and a 3 dB cut in the high frequency region of the loudspeakers response. The same number of filter entries must be present in each filter table of each filter group. The naming of each filter entry in the filter table must also match. It is the matching of the INTERNAL NAME for the different filter entries that enables ***SpeakerLab*** to use the correct filters from each of the filter groups when switching between the different filter entries.

Select the KEYHFXOVER "INPUT" in the property tree and then the GENERAL tab on the EDIT FILTER GROUP page on the right. Change the NUMBER OF FILTERS from 1 to 3 using the dropdown box. Let's change the naming of the second filter in the filter table. Select the INTERNAL NAME and change if from KEY1001 to "Key1000 +3". Now change the DISPLAY NAME for this filter "HF +3 dB". Change the Internal Name of the third filter from KEY1002 to "Key1000 -3" and the DISPLAY NAME to "HF -3 dB".



Internal Name : keyLFXover

Display Name : Input

Configuration Overridable □

Number of Filters : 3

	Internal Name	Display Name	File Location
1	key1000	Crossover 1k	
2	key1000 +3	HF +3 dB	
3	key1000 -3	HF -3 dB	

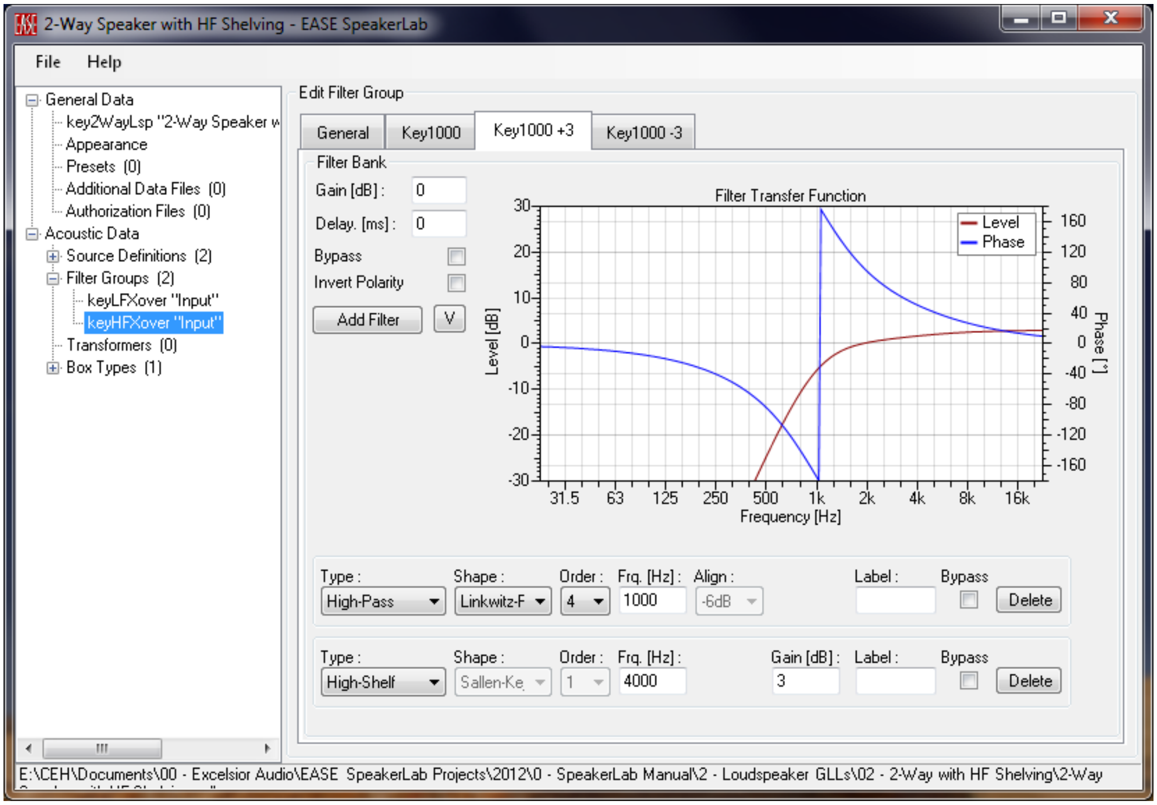
File Location :

☑ Relative

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\2 - Loudspeaker GLLs\02 - 2-Way with HF Shelving\2-Way

Figure 2-168 – LF filter group with 3 filters in the group

We now need to modify the filter groups to include the 3 dB shelving filters to give us the different responses from which the end user may choose. Select the KEY1000 +3 tab. Click the ADD FILTER button and then select IIR FILTER from the context menu that appears. For the new filter that has just been added change the TYPE to HIGH-SHELF. Change the FRQ [HZ] to "4000" and the GAIN [DB] to "3".



Gain [dB] : 0
Delay. [ms] : 0
Bypass □
Invert Polarity □

Type :	Shape :	Order :	Frq. [Hz] :	Align :	Gain [dB] :	Label :	Bypass
High-Pass	Linkwitz-F	4	1000	-6dB			□
High-Shelf	Sallen-Ke	1	4000		3		□

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Figure 2-169 – HF filter group showing the filter with the +3 dB high shelf filter added to the high pass filter

Select the Key1000 -3 tab and repeat these steps to add the high shelf filter. For this one the gain should be -3 dB.

We must now save these filters. Select the General tab and then click anywhere in row 2 of the filter table to select this filter. Click the Save As button at the bottom of the page. Let's name this file "HF - 1k, LR4, +3dB.xgfb". Remember to check the Relative checkbox. Repeat this for the filter in row 3 but name it "HF - 1k, LR4, -3dB.xgfb".

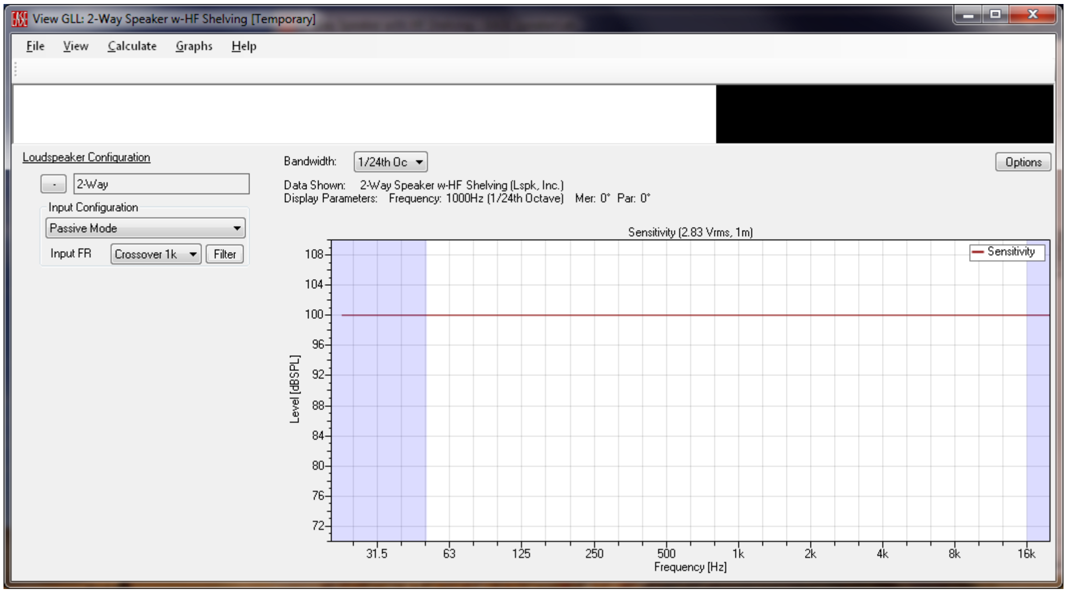
In order to fulfill the requirement of having the same number of filter entries for each filter group we must also have 3 filter entries in the keyLFXover "Input" filter group. Select this filter group in the property tree. On the General tab of the Edit Filter Group page on the right change the Number of Filters from 1 to 3 using the dropdown box. We must also change the naming of the filter in the filter table to match that of the keyHFXover "Input" filter group. In row 2 select the Internal Name and change it from Key1001 to "Key1000 +3". Now change the Display Name for this filter "HF +3 dB". Change the Internal Name of the third filter from Key1002 to "Key1000 -3" and the Display Name to "HF -3 dB".

There will be no difference in the LF crossover filters for these filter entries so we can load the previously saved filter for them. Select row 2 and click on the Load button at the bottom of the page.

Select the file *LF - 1k, LR4.xgfb*. Remember to check the RELATIVE checkbox. Repeat this for the third filter.

We can now save the XGLL project (FILE > SAVE PROJECT or CTRL+S).

Let's see how these changes work in the GLL. Run the GLL (FILE > RUN GLL or F5). Calculate the GLL (CALCULATE > BALLOON or F5) and change the display to show the sensitivity graph (GRAPHS > SENSITIVITY or CTRL+SHIFT+S). Click on the + symbol under LOUDSPEAKER CONFIGURATION. The display should now look like Figure 2-170. We can see that this is our flat, base line sensitivity for the 1 kHz crossover using the 4th order Linkwitz-Riley we specified in the filter definition.


View GLL: 2-Way Speaker w-HF Shelving [Temporary]

Loudspeaker Configuration: 2-Way

Input Configuration: Passive Mode

Input FR: Crossover 1k

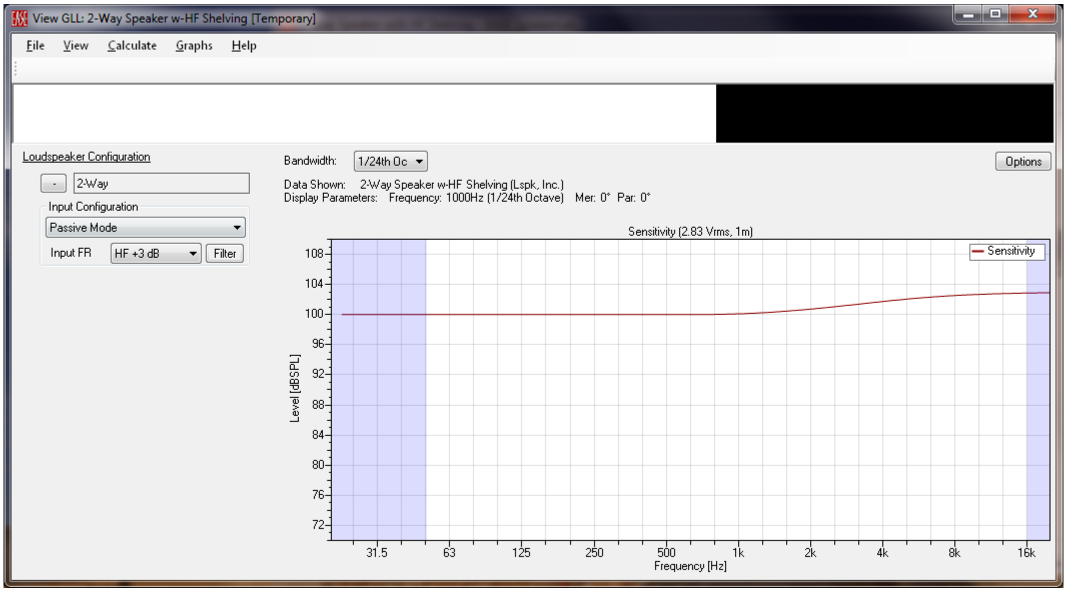
Bandwidth: 1/24th Oc

Data Shown: 2-Way Speaker w-HF Shelving (Lspk, Inc.)

Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-170 – Running GLL showing the calculated sensitivity using the Crossover 1k input

Use the dropdown box of the input labeled INPUT FR to select the HF +3 DB internal filter settings and recalculate the GLL (CALCULATE > RECALCULATE or CTRL+F5). The sensitivity curve has now changed to show the effects of the high frequency shelving filter boost (Figure 2-171). Try selecting the HF -3 DB input option and recalculating.



2-Way

Input Configuration: Passive Mode

Input FR: HF +3 dB

Bandwidth: 1/24th Oc

Data Shown: 2-Way Speaker w-HF Shelving (Lspk, Inc.)

Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-171 – Running GLL showing the calculated sensitivity using the HF +3 dB input

2.12.5 Example: Configuration Files (Changing External Filters / Biamping)

This example will show how to setup a GLL to use external filters as is typical for biamping a two-way system, i.e. using a DSP for the crossover and EQ. We will also modify the GSS files used for the low frequency and high frequency devices so that they will more closely resemble actual drivers. We will continue working with the 2-Way Speaker XGLL project by making these modifications.

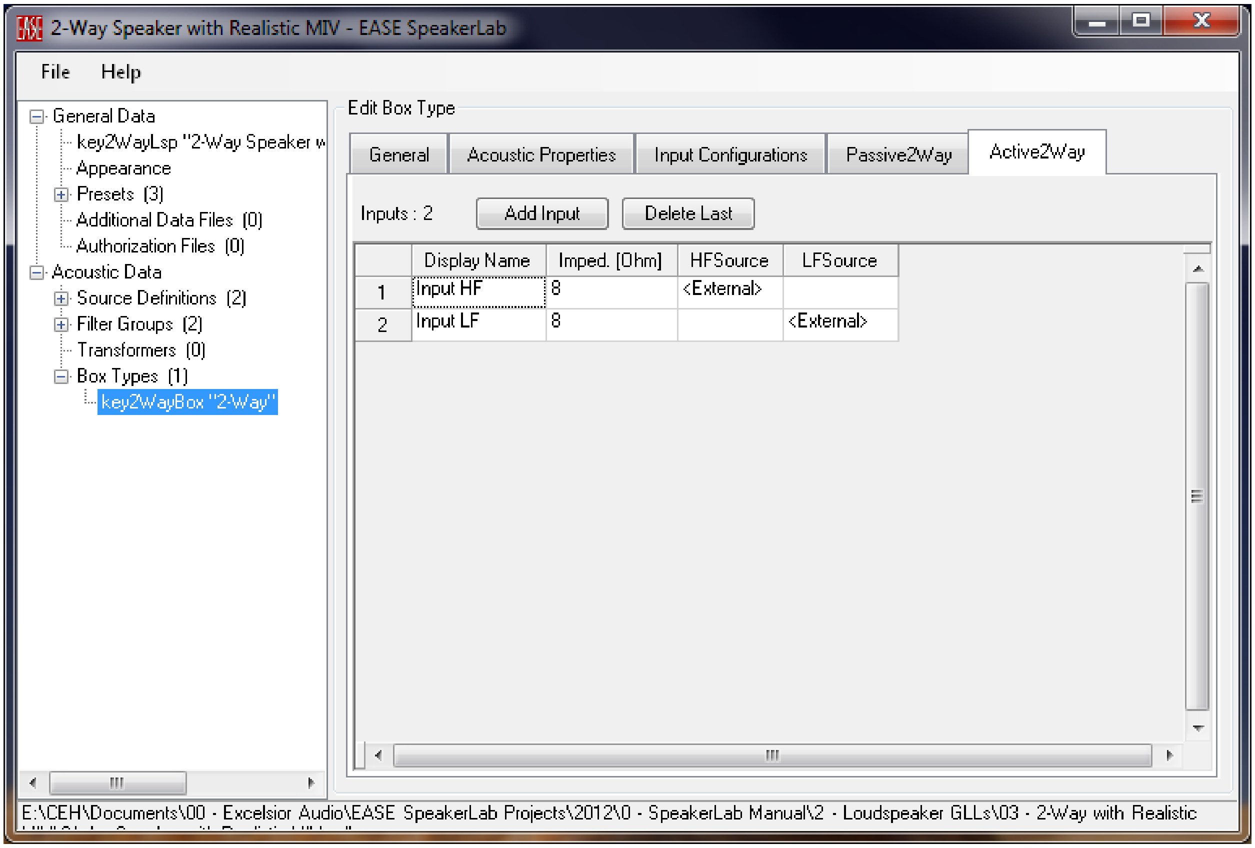
Run the ***Source Editor*** (FILE > SOURCE EDITOR). Then from the ***View GSS*** window open the *Point Source, 100dB.gss* file (FILE > OPEN). Now save this file with a new name of "Point Source, HF, 110dB.gss" (FILE > SAVE AS).

We will now change some of the properties of this new GSS file to make it more closely match that of a typical HF compression driver (higher sensitivity and lower MIV), but it will not be exact. Click on EDIT > PROPERTIES (F4) then click on the SENSITIVITY tab. In the GAIN CORRECTION box change 100 to "110" so that we have a higher sensitivity. Next, click on THE MAXIMUM VOLTAGE tab. In THE PROGRAM (VRMS 1MIN) box enter "15" so that we have a lower MIV. Now click the OK button. Finally, save the GSS file (FILE > SAVE or CTRL+S).

Now save the GSS file as "Point Source, LF, 98dB.gss" (FILE > SAVE AS). Click on EDIT > PROPERTIES (F4) then click on the SENSITIVITY tab. In THE GAIN CORRECTION box change 110 to "98". Next, click on THE MAXIMUM VOLTAGE tab. In THE PROGRAM (VRMS 1MIN) box enter "40". Now click the OK button. Finally, save the GSS file (FILE > SAVE or CTRL+S) and close the ***View GSS*** window.

Now let us use these new sources in our XGLL project. Go to the property tree and select KEYHF under ACOUSTIC DATA > SOURCE DEFINITIONS. In the EDIT SOURCE DEFINITIONS page on the right click the Browse button and select the *Point Source, HF, 110dB.gss* file. Be sure to check the Relative checkbox. Save the XGLL project file. Next, select keyLF under Acoustic Data > Source Definitions. In the Edit Source Definitions page on the right click the Browse button and select the *Point Source, LF, 98dB.gss* file. Be sure to check the Relative checkbox. Save the XGLL project file.

Expand Box Types in the property tree if it isn't already by clicking the + button on its left. Select key2WayBox "2-Way" underneath Box Types. Click the Active2Way tab on the Edit Box Type page on the right. We can see that the proper signal routing for biamping using two separate inputs is already setup (Figure 2-172). This was part of the default layout for the initial project file that was created using the wizard in Section 2.12.3. The HF input for the loudspeaker routes its signal directly to the HFSource and the LF input routes its signal directly to the LFSource. This completely bypasses any of the internal filters that were setup within the XGLL project.



Inputs : 2

	Display Name	Imped. [Ohm]	HFSource	LFSource
1	Input HF	8	<External>	
2	Input LF	8		<External>

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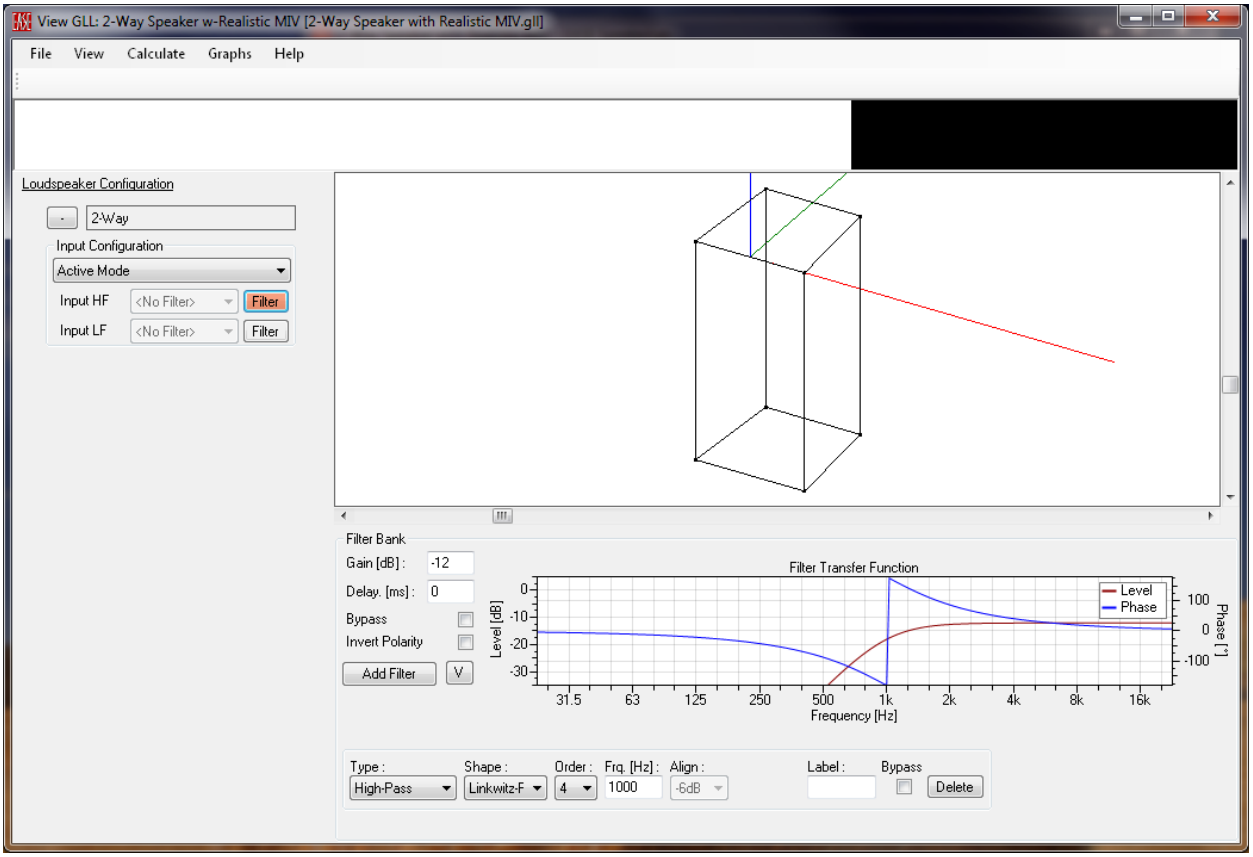
Figure 2-172 – Input Configuration tab showing the input signal routed directly to the acoustical sources (no internal filters)

Let's now run the GLL (File > Run GLL or F5) and build some external filters to use for biamping our loudspeaker system. Click on the + symbol under Loudspeaker Configuration. Use the dropdown box under Input Configuration to change from the Passive Mode to Active Mode. The separate HF and LF inputs should now be visible. Click on the Filter button to the right of the Input HF. This will show the Filter Bank for the HF input. Click the Add Filter button and then click the IIR Filter selection in the context menu that appears. Change the filter parameters so that it is a high pass filter, Linkwitz-Riley, 4th order at "1,000" Hz. Also change the gain for the Filter Bank to "-12" dB. This is required due to the sensitivity of our HF device being 12 dB greater than the LF device.

The gain of this filter will reduce the level of the signal that feeds the HF device. This is advantageous since the HF device has a lower MIV than the LF device. This gain adjustment, along with the frequency dependent roll-off of the filter, will help to make the MIV at the input of this filter close to the same as the MIV at the input of the filter that feeds the LF device.



View GLL: 2-Way Speaker w-Realistic MIV [2-Way Speaker with Realistic MIV.gll]

Loudspeaker Configuration

2-Way

Input Configuration

Active Mode

Input HF <No Filter> Filter

Input LF <No Filter> Filter

Filter Bank

Gain [dB] : -12

Delay. [ms] : 0

Bypass □

Invert Polarity □

Add Filter V

Type : High-Pass

Shape : Linkwitz-F

Order : 4

Frq. [Hz] : 1000

Align : -6dB

Label :

Bypass □

Delete

Figure 2-173 – Running GLL showing a high pass filter added to the HF input

Click on the Filter button to the right of the Input LF. This will show the Filter Bank. The filter bank for the LF input is now displayed. Click the Add Filter button and then click the IIR Filter selection. Change the filter parameters so that it is a low pass filter, Linkwitz-Riley, 4th order at 1,000 Hz. Run the GLL if you would like. The sensitivity should show a straight line at 98 dB.

We should save these filter settings in a configuration file by clicking on File > Save Configuration (CTRL+S). The first time a configuration file is saved a Save File dialog box will appear so the file can be named and the folder in which it is to be stored specified. Let's name this file "LR4, 1kHz, Flat.xglc". This file can be recalled at any time to reset the configuration of this GLL to the same state it has now. As we will see shortly this can be very useful.

Let's also save two other configuration files. Change the filter frequency for both the LF and HF filter banks from 1,000 Hz to "500" Hz. Save this configurations files as "LR4, 500Hz, Flat.xglc". Now add another IIR filter to the filter bank for the LF input. Change the parameters of this filter so that it is a low Shelf at "60" Hz with a gain of "-6" dB. This will be useful if the loudspeaker is used very close to walls or hung against the ceiling. Save this configuration file as "LR4, 500Hz, LF Shelf.xglc".

Now open one of the previously saved configuration files (FILE > OPEN CONFIGURATION) to see that the GLL is completely reconfigured per the saved file. The GLL will have to be recalculated (CALCULATE > RECALCULATE or CTRL+F5) to see the results in any of the graphs, but the filter parameters can be seen immediately by simply clicking on the FILTER buttons (Note the 2-WAY LOUDSPEAKER CONFIGURATION will need to be expanded first).

The configuration files can be sent to anyone using the GLL for which the configuration files are specifically saved. This can be a very effective way to distribute new crossover filter settings for loudspeaker system or to try new filter settings in other models using the same GLL.

2.12.6 Example: Presets (Changing External Filters / Biamping)

The configuration files that were saved can also be used with the GLL. If the ***View GLL*** window is still open, close it and go back to the main window in ***SpeakerLab***. We will now add these configuration files to define presets within the GLL.

Select the PRESETS property under GENERAL DATA in the property tree. Click the NEW button in the EDIT PRESETS page on the right. You'll notice that several things happened. A table is now present on the EDIT PRESETS page and there is now a + button to the left of the PRESETS on the property tree. The NEW button changed to become a DUPLICATE button. Click anywhere in row one of the table and then click the DUPLICATE button. There should now be two preset slots in the table. Click it one more time to add a third slot.

Now expand the PRESETS property (+ button to the left of PRESETS) in the property tree. The three preset slots should be shown displayed in the property tree. Select KEYPRESET1. The EDIT GLL CONFIGURATION page is now shown on the right. Click the BROWSE button and select the *LR4, 1kHz, Flat.xglc* file. Change the DISPLAY NAME to "1 KHZ CROSSOVER, FLAT". This will be displayed in the GLL for users to select so it should be something fairly simple but descriptive. Check the RELATIVE checkbox and save the XGLL project file. We should note that the first preset will be the default whenever the GLL is run. This is the configuration that will be used unless the user selects another preset or loads a different configuration file.

We will now specify the second and third presets. Select KEYPRESET2 from the property tree. In the EDIT GLL CONFIGURATION page on the right click the Browse button and select the *LR4, 500Hz, Flat.xglc* file. Change the DISPLAY NAME to "500 HZ CROSSOVER, FLAT" and check the

RELATIVE checkbox. Select KEYPRESET3 from the property tree. In the EDIT GLL CONFIGURATION page on the right click the BROWSE button and select the *LR4, 500Hz, LF Shelf.xglc* file. Change the DISPLAY NAME to "500 Hz Crossover, LF Shelf", check the RELATIVE checkbox, and save the XGLL project file.

Run the GLL (FILE > RUN GLL or F5) to see how the presets can be selected in the GLL. Expand the 2-WAY LOUDSPEAKER CONFIGURATION and notice that the GLL is already in ACTIVE MODE. Clicking the FILTER buttons also shows that the 1 kHz crossover filters are in place since they were placed in the first preset slot.

Change to another preset by clicking on FILE > SELECT PRESET. This will open a Select Preset dialog box. Select the 500 HZ CROSSOVER, FLAT preset and click OK. Take a look at the filters to verify that they are now set to 500 Hz.

This can be a very useful way to include multiple configurations for a loudspeaker system within the GLL itself. This functionality becomes even more powerful when used with Cluster and Line Array type GLLs. Entire array configurations (both the physical placement of loudspeaker boxes and the processing for them) can be saved as a preset for distribution with the GLL.

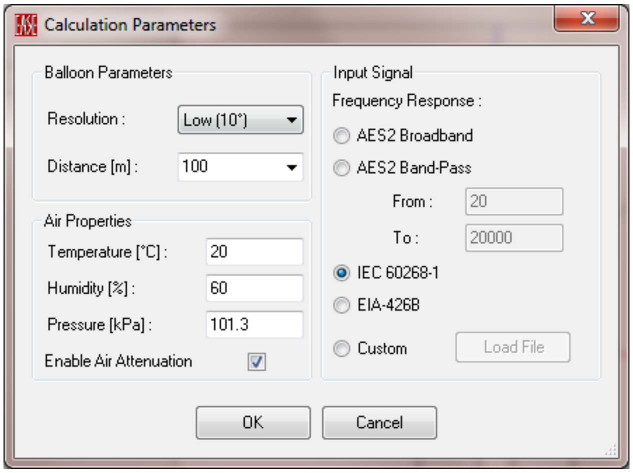
2.12.7 Example: System Sensitivity, Maximum SPL, and Maximum Input Voltage

Let's have a brief look at how the GLL calculates the sensitivity, maximum SPL, and the maximum input voltage for the model of the loudspeaker system. Regardless of whether a GLL is modeling a loudspeaker with a passive crossover, an active loudspeaker driven by external crossover and EQ filters and external power amplifiers, or a powered loudspeaker with internal processing and power amps, the GLL model should always be visualized as a system containing all of the parts of the loudspeaker system, including the processing and power amplifiers that may be external to the loudspeaker box. The reason for this is that this systemic view of the loudspeaker has only a single input, whether it is the input to the passive crossover, the input to the external crossover/DSP feeding the external power amplifier, or the input to the active loudspeaker box feeding the internal crossover/DSP and power amplifier. It is this single input that is important when seeking to determine the sensitivity of the system, the maximum input voltage to the system, or maximum SPL the system can produce.

In light of this we once again reiterate the recommendation to view the GLL model of a loudspeaker system as if it is a passive loudspeaker (Section 2.11.2) or even a powered loudspeaker with a single input. The use of any filters in the GLL, whether they are internal or external to the GLL, will affect the sensitivity, maximum input voltage, and maximum SPL calculated by the GLL.

The calculation of sensitivity is relatively straight-forward; the input to the GLL is presented with an rms input of 2.83 V at each frequency. This is similar to a swept-sine signal of 2.83 V used for testing the sensitivity of loudspeakers. It should not be confused with an input signal having a broad-band magnitude of 2.83 V which would give a lower value for the sensitivity. It should be understood that a rigorous definition of frequency dependent sensitivity for a loudspeaker system is the transfer function of the loudspeaker system. As such the signal used to measure this transfer function is not overly important as, by definition for a transfer function, the output of the loudspeaker system is divided by the input to it.

The calculation of the maximum input voltage for the system is different than for the sensitivity. For the maximum input voltage the spectral content of the input signal must first be defined. This is done when selecting the Frequency Response of the Input Signal on the Calculation Parameters dialog box (Figure 2-174). The GLL then increases the broad-band voltage of this signal at the input of the GLL until one of the GSS files used in the GLL reaches its MIV after the input signal has been passed through all of the filters in the signal path defined in the GLL. The input voltage for the GLL is not allowed to exceed this magnitude. This is the maximum input voltage for the GLL.



Calculation Parameters

Balloon Parameters

Resolution : Low (10°)

Distance [m] : 100

Air Properties

Temperature [°C] : 20

Humidity [%] : 60

Pressure [kPa] : 101.3

Enable Air Attenuation ☑

Input Signal

Frequency Response :

○ AES2 Broadband

○ AES2 Band-Pass

From : 20

To : 20000

● IEC 60268-1

○ EIA-426B

○ Custom

Figure 2-174 – Calculation Parameters dialog box

Calculating the maximum SPL is now fairly easy. The specified input signal (with its defined spectral content) at the maximum input voltage is combined with the sensitivity curve of the calculated GLL. This yields a maximum SPL curve. The sound pressure for each fractional octave bandwidth is summed together to yield the broad-band sound pressure and the broad-band SPL is then calculated.

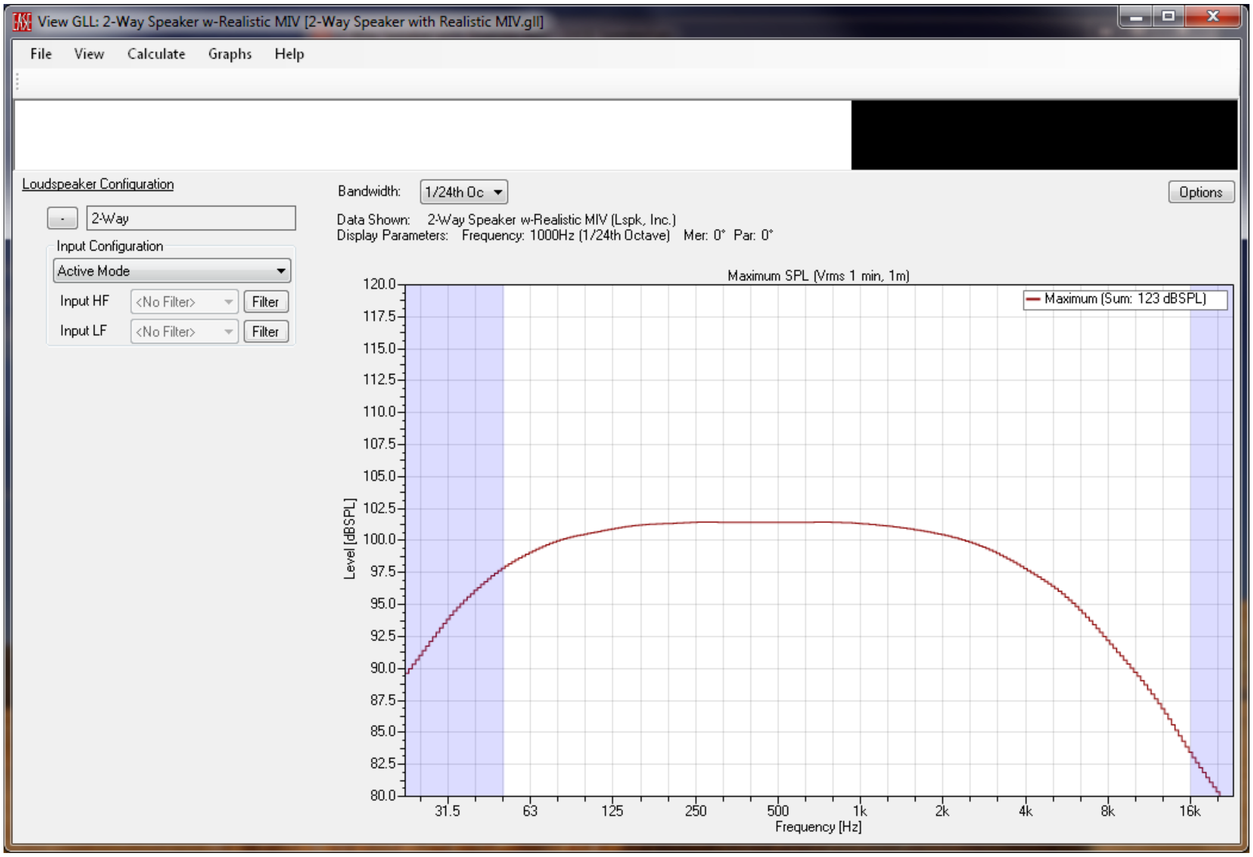
It should be explicitly pointed out that with this method of calculating the maximum SPL, amplitude non-linearity (whether due to heating of the driver or other causes) is not accounted for in the GLL. The GLL only calculates the linear increase in SPL output for a loudspeaker system. For this reason great care should be taken when specifying the MIV for each individual GSS. They should not be the equivalent voltage of the full rated power handling for the device. Special, non-destructive, testing (detailed in Section 8.3) can be easily carried out to determine a more appropriate MIV to be used with the GSS.

Let's now step through an example of these calculations with a GLL. For this we will continue to use the project file from the previous section. Run this GLL and calculate its response using the default preset.

When setting the CALCULATION PARAMETERS select the IEC 60268-1 FREQUENCY RESPONSE for the INPUT SIGNAL.

Examine the sensitivity of the GLL (GRAPHS > SENSITIVITY or CTRL+SHIFT+S). It should be a flat line at 98 dB.

Look at the maximum SPL (GRAPHS > MAXIMUM SPL or CTRL+SHIFT+M). This shows a curve of the SPL at fractional octave spacings selected by the BANDWIDTH dropdown box (Figure 2-175). The total, broad-band sum of all of the fractional octaves is shown in the MAXIMUM (SUM: xxx dBSPL) in the upper right corner of the graph. It is this SPL shown as the MAXIMUM SUM that is the maximum broadband SPL calculated by the GLL.


View GLL: 2-Way Speaker w-Realistic MIV [2-Way Speaker with Realistic MIV.gll]

Loudspeaker Configuration: 2-Way

Input Configuration: Active Mode

Input HF: <No Filter>

Input LF: <No Filter>

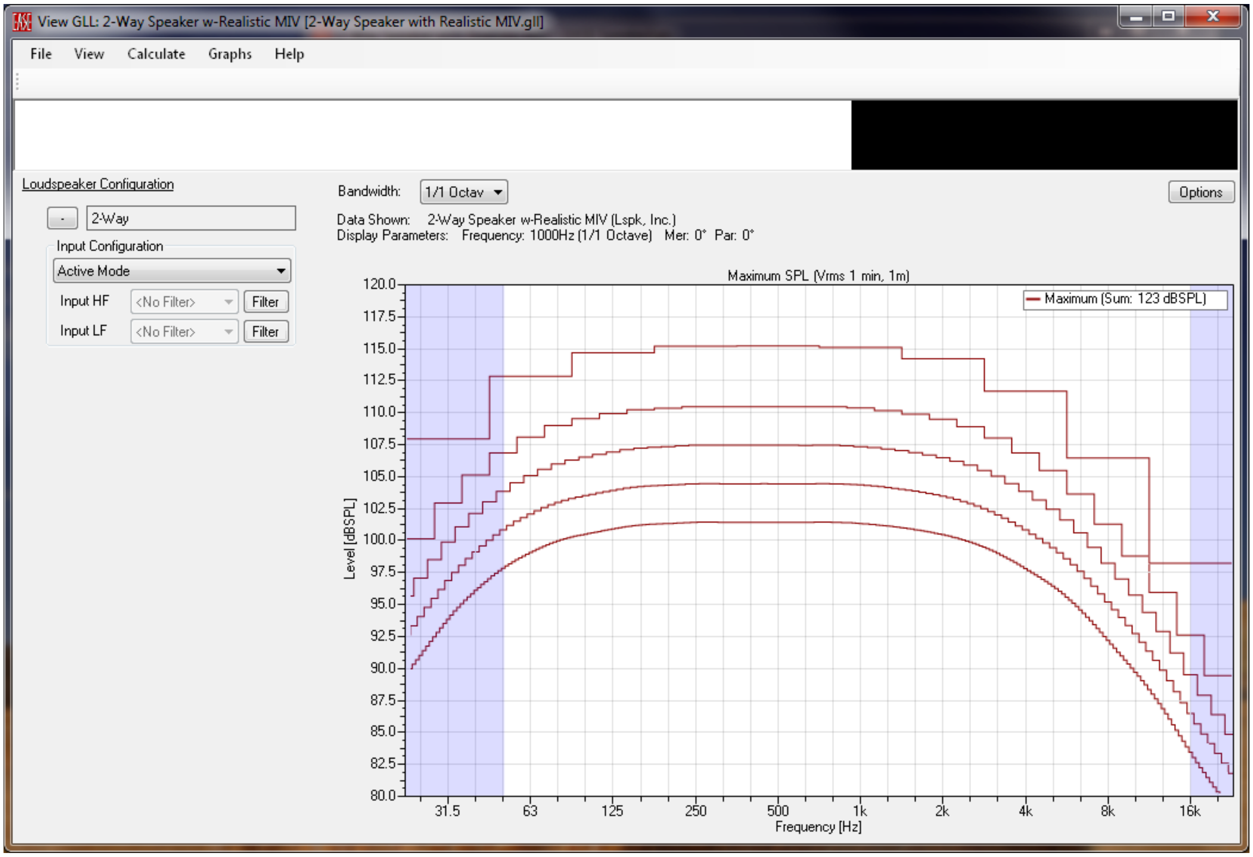
Bandwidth: 1/24th Oc

Data Shown: 2-Way Speaker w-Realistic MIV (Lspk, Inc.)

Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-175 – Running GLL showing the calculated maximum SPL for the IEC 60268 input signal

Note that changing the BANDWIDTH of the display will move the curve up and down in level on the graph. This is the same as changing the bandwidth setting on an SPL meter. For a given signal, a larger selected bandwidth will result in a higher SPL reading. This is the result of more energy entering the SPL meter due to the large bandwidth. This is shown in Figure 2-176. This display is not possible, but was created by overlaying multiple screen shots at different bandwidth settings.


View GLL: 2-Way Speaker w-Realistic MIV [2-Way Speaker with Realistic MIV.gll]

Loudspeaker Configuration: 2-Way

Input Configuration: Active Mode

Input HF: <No Filter>

Input LF: <No Filter>

Bandwidth: 1/1 Octav

Data Shown: 2-Way Speaker w-Realistic MIV (Lspk, Inc.)

Display Parameters: Frequency: 1000Hz (1/1 Octave) Mer: 0° Par: 0°

Maximum (Sum: 123 dBSPL)

Figure 2-176 – Illustration of different selected bandwidths on the maximum SPL display

Look at the maximum input voltage (GRAPHS > INPUT VOLTAGE or CTRL+SHIFT+L). This shows a curve of the voltage level at fractional octave spacings selected by the BANDWIDTH dropdown box. The total, broad-band sum of all of the fractional octaves is shown in the MAXIMUM (SUM: XX DBV) in the upper right corner of the graph. Just as with the maximum SPL, the bandwidth setting will change the level of each of the fractional octaves that make up the curve. It will have no effect on the overall broad-band level shown as the MAXIMUM SUM.

Let's once again look at the maximum SPL (GRAPHS > MAXIMUM SPL or CTRL+SHIFT+M). We see that the maximum SPL is 123 dB referenced to 1 meter.

Load the 500 HZ CROSSOVER, FLAT preset (FILE > SELECT PRESET) and recalculate the GLL (CALCULATE > RECALCULATE or F5). Notice that the maximum SPL has now changed to 124.3 dB. This occurred because the low frequency pass band (the LF GSS) reached its MIV before the high frequency pass band (HF GSS) did. When we moved the crossover filters from 1 kHz down to 500 Hz less energy from the input signal is sent to the LF and more is sent to the HF. This means that the input signal can now be increased until one of the GSS devices again reaches its MIV.

Check the maximum input voltage (GRAPHS > INPUT VOLTAGE or CTRL+SHIFT+L). Notice that it increased from 34 dBV to 35.4 dBV. Since we have ideal, perfectly flat outputs for our GSS files the maximum SPL increase by the almost same amount. This 0.1 dB difference is the result of rounding internal to the GLL. It is negligible and can be ignored.

Let's try the 500 HZ CROSSOVER, LF SHELF preset. The maximum input voltage for this preset is 36.2 dBV. The maximum SPL is 124.8 dB. This again tells us that it was the LF GSS that was the limiting the maximum input voltage for the GLL. But why did the maximum input voltage increase 0.8 dB while the maximum SPL only increased by 0.5 dB compared to the 500 Hz Crossover, Flat preset? The answer is that with the LF shelf the frequency response of the GLL is no longer flat. There is less output in the low frequency region. Notice the broadband sensitivity is 97.7 dB now. With the 500 Hz Crossover, Flat preset it is 98 dB. There is the 0.3 dB difference.

Load the 1 KHZ CROSSOVER, FLAT preset once again and calculate the GLL. Look at the maximum SPL. It should be 123 dB. Now click CALCULATE > BALLOON or F5, do not recalculate as we want to change the FREQUENCY RESPONSE of the INPUT SIGNAL. Select the AES2 BROADBAND signal. The maximum SPL should now be 123.8 dB. The increase is due to the additional high frequency content of the AES2 Broadband signal compared to the IEC 60268-1 signal.

It should be understood that the choice of the input signal used to calculate the response of a GLL will have an effect on the spectral content (frequency response) of the output as well as the maximum SPL. The input signal selected will not have an effect on the sensitivity or any other items derived from the transfer function of the loudspeaker GLL, as the transfer function is independent of the excitation signal (input signal).

2.13 Multi-Source GLL

A multi-source GLL is no different than a two-source GLL. All of the items covered in Section 2.12 still apply. There are a couple of items that may benefit from some additional explanation and examples.

Suppose that we have a three-way loudspeaker system. This loudspeaker system can be run with three different input schemes or input configurations. The first is to use its internal passive crossover to handle the crossover functions between all three pass bands. The second is to run it in biamp mode which will require external filters for the crossover between the low frequency (LF) and mid frequency (MF) pass bands. The crossover between the mid frequency (MF) and high frequency (HF) pass bands will use a portion of the internal passive crossover. The third is to run the loudspeaker system in triamp mode. This will require external filter for the crossover between the LF and MF as well as the MF and HF pass bands.

Begin a new project file by clicking FILE > NEW PROJECT. In the CREATE NEW PROJECT dialog box select 3-WAY SPEAKER under the LOUDSPEAKER tab. Enter the desired name and the folder location in which to save this new project and then click the OK button.

2.13.1 Filter Groups

We should now create the filter groups that will be used for the internal passive crossover. Even if we do not yet know what the exact filters will be for all of these crossovers, we know that they need to exist so we should make provisions for them in our XGLL project file. This must be done before we can define the signal routing through these filters.

Expand the ACOUSTIC DATA portion of the property tree and select the FILTER GROUPS property. Click the NEW button on the EDIT FILTER GROUPS page on the right. Select row 1 of the filter groups table and insert two additional rows (click the DUPLICATE button two times). This gives us the three filter groups we need for our 3-way loudspeaker system.

Select the KEYFILTERGRP1 property from the property tree. In the EDIT FILTER GROUP page on the right change the filter group INTERNAL NAME to "keyFilterHF", change the filter group DISPLAY NAME to "HF Passive" and change the NUMBER OF FILTERS to "1". Do the same for KEYFILTERGRP2 but name this one "keyFilterMF" (Internal) and "MF Passive" (Display). Repeat for KEYFILTERGRP3 calling it "keyFilterLF" (Internal) and "LF Passive" (Display).

Now we will put some temporary filters in place until we have the measured response of the actual passive filters that will be used. (Note that we won't actually change to the real measured filter response in this section. For details on this see Section 8.4 – Filter Measurement)

Select the KEYFILTERHF property from the property tree. In the EDIT FILTER GROUP page on the right click the KEYFILTERDEF1 tab. Add an IIR filter. Change the parameters of this filter so that it is a high pass, Linkwitz-Riley, second order, at "3000" Hz. Click on the General tab and save this filter. You can call it anything you like, but something such as "HF – LR2, 3kHz" is both short and fully describes the filter.

It may also be useful to create a new folder inside the XGLL project folder called FILTERS. This can be a handy place to keep all of the filters that are created for a project and not clutter the root project folder.

Select the KEYFILTERMF property from the property tree. In the EDIT FILTER GROUP page on the right click the KEYFILTERDEF1 tab. Add an IIR filter. Change the parameters of this filter so that it is a high pass, Linkwitz-Riley, second order, at "400" Hz. Add a second IIR filter. Note that the existing filter is duplicated. The default behavior of ***SpeakerLab*** is to duplicate the last filter in the listing when adding a new filter.

Change the parameters of the second filter so that it is a low pass, Linkwitz-Riley, second order, at "3000" Hz. Invert the polarity of this filter bank. Click on the General tab and save this filter. You can call it anything you like, but something such as "MF – LR2, 400-3kHz" is both short and fully describes the filter.

Select the KEYFILTERLF property from the property tree. In the EDIT FILTER GROUP page on the right click the KEYFILTERDEF1 tab. Add an IIR filter. Change the parameters of this filter so that it is a low pass, Linkwitz-Riley, second order, at "400" Hz. Click on the General tab and save this filter. You can call it anything you like, but something such as "LF – LR2, 400Hz" is both short and fully describes the filter.

Remember to check the Relative checkbox for the file location of all three of these filters. Save the XGLL project file.

We now have our preliminary filters in place for our passive crossover. However, we need to add some slightly modified filters to use for our biamp mode.

Click on the FILTER GROUPS property in the property tree. In the EDIT FILTER GROUPS page on the right select the row 1 of the table and then click the DUPLICATE button. Now, select row 2 and duplicate it.

Click on the KEYFILTERHF1 property in the property tree. In the EDIT FILTER GROUPS page on the right change the DISPLAY NAME for this filter group to "HF Biamp". Change the DISPLAY NAME of the filter in the table to "Biamp XO". Click the LOAD button at the bottom of the page and select the *HF – LR2, 3kHz.xgfb* file, or whatever you named it when you saved it.

Click on the KEYFILTERMF1 property in the property tree. In the EDIT FILTER GROUPS page on the right change the DISPLAY NAME for this filter group to "MF Biamp". Change the DISPLAY NAME of the filter in the table to "Biamp XO".

Click the KEYFILTERDEF1 tab. For biamp operation we need to disable only the high pass filter in this filter bank. Check the BYPASS checkbox on the right side of the filter that is defined as the high pass filter. Alternatively, we could have deleted this high pass filter.

Click on the GENERAL tab so that we can save this filter bank. Click anywhere in filter row 2 to select it. Click the SAVE AS button at the bottom of the page. Name the file "MF - LR2, 3kHz.xgfb" or something you wish. Do not name it the same as the filter bank file used for the passive mode as this will overwrite that file.

Make sure that the Relative checkboxes were checked when saving these new filter bank files. Save the XGLL project file.

2.13.2 Input Configurations

We must still setup the signal routing through the appropriate filters for our different input configurations. Expand the BOX TYPES property in the property tree and select KEY3WAYBOX. Select the INPUT CONFIGURATIONS tab in the EDIT BOX TYPE page on the right. In row 1 of the input configurations table change the Internal Name to "keyPassive" and the Display name to "Passive". Add two new input configurations by clicking the Add Input Configuration button twice.

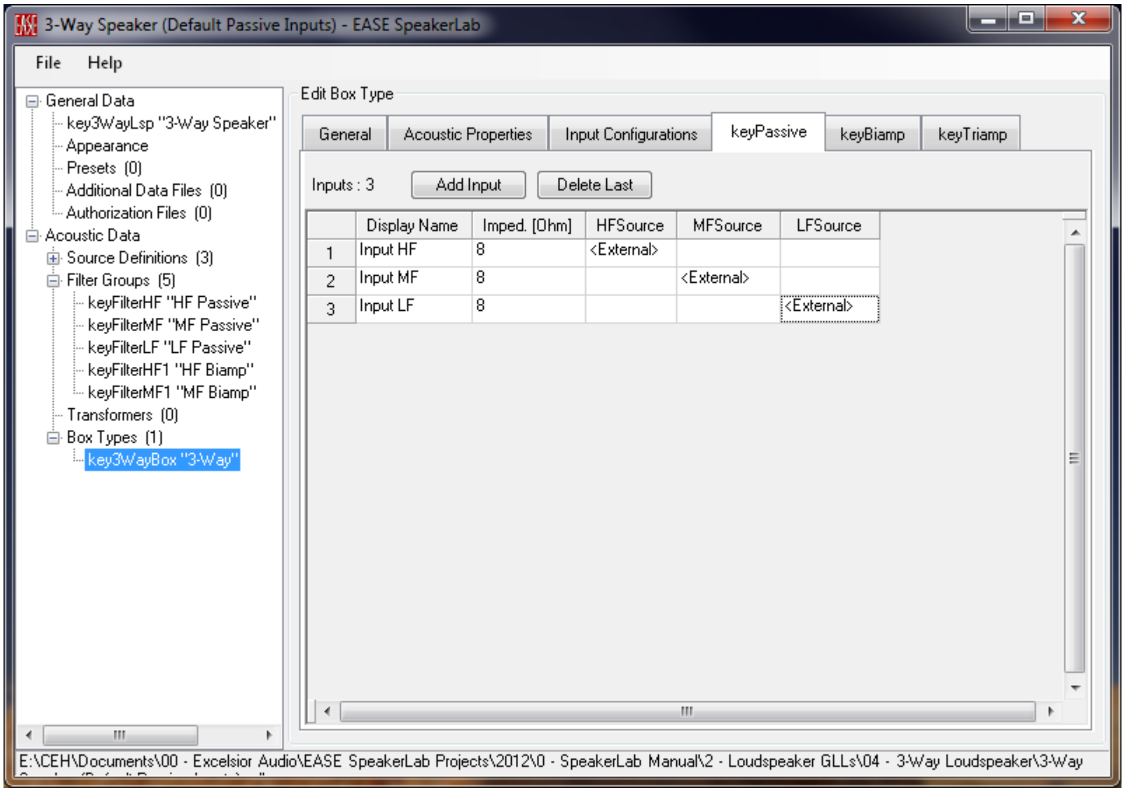
In row 2 of the table change the Internal Name to "keyBiamp" and the Display name to "Biamp". In row 3 of the table change the Internal Name to "keyTriamp" and the Display name to "Triamp".

The reason we changed the first row to be the input for the passive crossover is that the GLL will use the input configuration in the first row by default, unless there is a preset specified which will define the complete configuration of the GLL. If an input configuration other than the passive input is desired it should be used in the first row.

Save the XGLL project file.

2.13.2.1 Passive Mode – Fully Passive Crossover

Click on the keyPassive tab. We see that there are three inputs (Figure 2-177). We only need one input for use with the internal passive crossover. Delete two of the inputs using the Delete Last button.



Inputs : 3

	Display Name	Imped. [Ohm]	HFSource	MFSource	LFSource
1	Input HF	8	<External>		
2	Input MF	8		<External>	
3	Input LF	8			<External>

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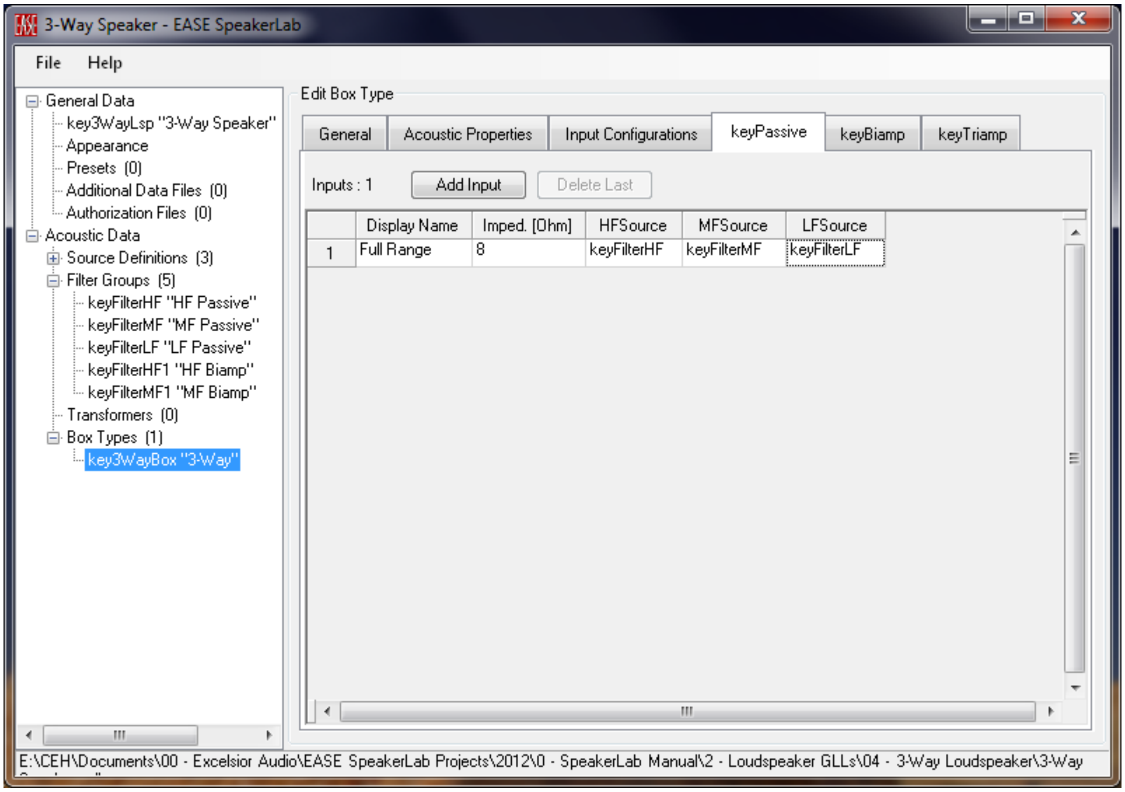
Figure 2-177 – Input Configuration tab showing 3 inputs

We should rename the Display Name for this input to something like "Full Range" to help prevent confusion.

We now need to define the signal routing through the appropriate filters. This is the same as connecting the output wires of the passive crossover to the correct drivers. If this is not done properly the response calculated by the GLL will not yield the expected results.

Click the cell in the HFSource column. A Select Filter Group dialog box will appear with a list of choices. This is the filter feeding the HF source so we should select keyFilterHF. This is the Internal Name of HF passive filter. Click the cell in the MFSource column and select the keyFilterMF. Finally, click in the LFSource column and then select keyFilterLF. The screen should look like Figure 2-178. From this we should see that the signal at the Full Range input is routed through the selected filters and then onto their respective sources.

Save the XGLL project file.



- General Data
  - key3WayLsp "3-Way Speaker"
  - Appearance
  - Presets (0)
  - Additional Data Files (0)
  - Authorization Files (0)
- Acoustic Data
  - Source Definitions (3)
  - Filter Groups (5)
    - keyFilterHF "HF Passive"
    - keyFilterMF "MF Passive"
    - keyFilterLF "LF Passive"
    - keyFilterHF1 "HF Biamp"
    - keyFilterMF1 "MF Biamp"
  - Transformers (0)
  - Box Types (1)
    - key3WayBox "3-Way"

Inputs : 1

	Display Name	Imped. [Ohm]	HFSource	MFSource	LFSource
1	Full Range	8	keyFilterHF	keyFilterMF	keyFilterLF

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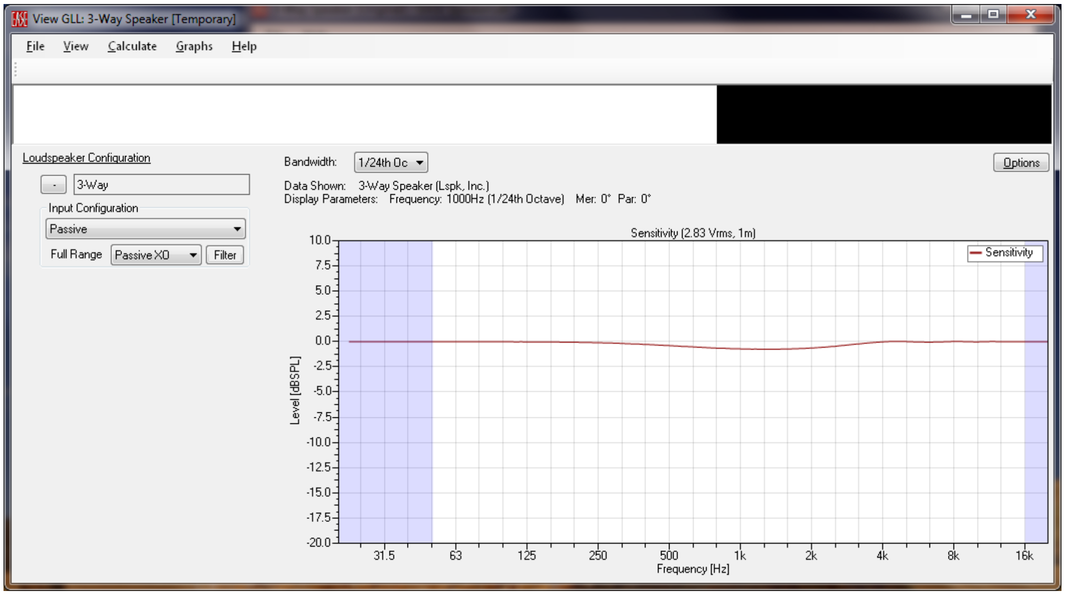
Figure 2-178 – Input Configuration tab showing 1 input and the signal routed through the appropriate filters before going to the acoustic sources

Run the GLL to see how the passive mode works. Expand the 3-Way Loudspeaker Configuration. Notice that the default Input Configuration is Passive. Calculate the GLL and look at the sensitivity. It should be an almost flat line and look like Figure 2-179.



3-Way

Input Configuration: Passive

Full Range: PassiveXO

Bandwidth: 1/24th Oc

Data Shown: 3-Way Speaker (Lspk, Inc.)

Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-179 – Running GLL showing the calculated sensitivity of the passive 3-way crossover

2.13.2.2 Biamp Mode – External LF to MF and Passive MF to HF

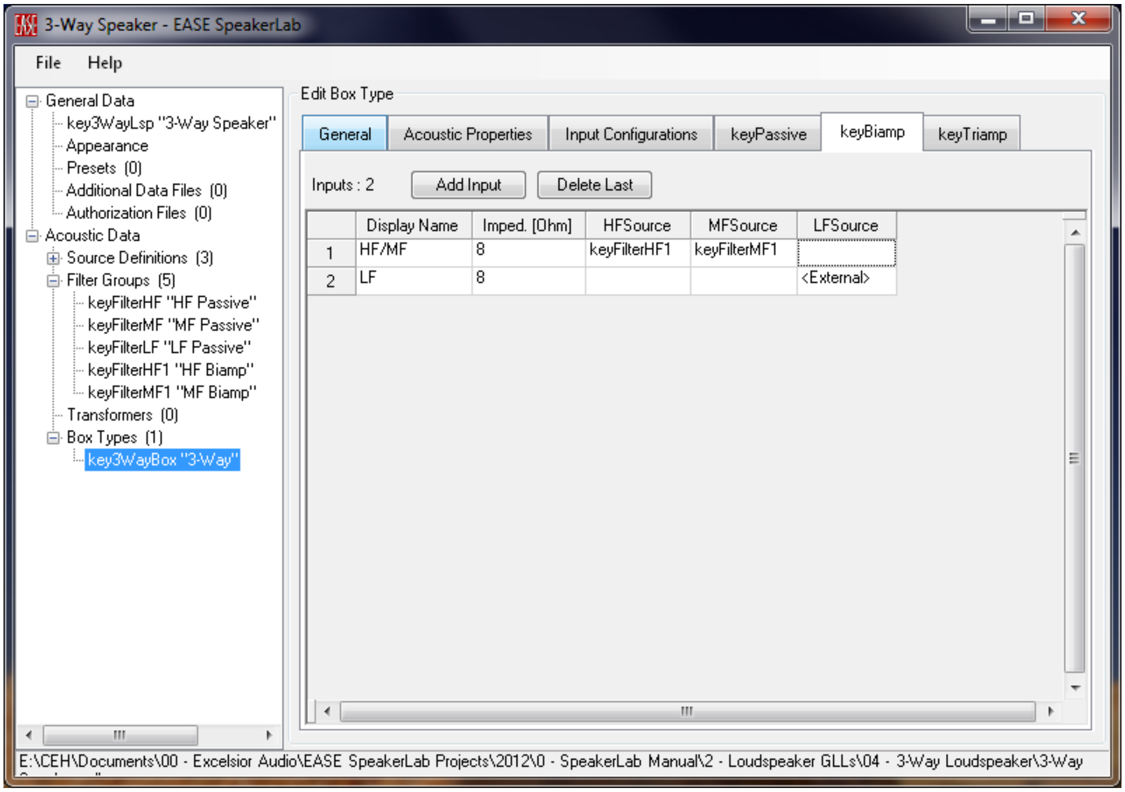
Select the ACOUSTIC DATA > BOXTYPES > KEY3WAYBOX property on the property tree. Click the KEYBIAMP tab on the EDIT BOX TYPE page. We only need two inputs to the loudspeaker system for biamp mode so delete the last input.

Let's change the Display name for row 1 to "HF/MF" and for row 2 to "LF".

To route the signal from the MF/HF input to the passive HF filter and then to the HFSOURCE click in the cell for row 1 and the HFSOURCE column. Select the KEYFILTERHF1 and click the OK button. Similarly, to route the signal from the MF/HF input to the passive MF filter and then to the MFSOURCE click in the cell for row 1 and the MFSOURCE column. Select the KEYFILTERMF1 and click the OK button. It is important to select these filters, the ones for biamp operation, and not the other HF and MF filters.

Routing the signal from the LF input to the LFSource requires that we click the cell for row 2 and the LFSource column. Select <External> and click the OK button. This will route the external signal directly to the intended acoustical source without any internal filters being applied. The external filtering will have to be specified when the GLL is run. This is analogous to the external crossover or DSP having to be used and its parameters properly set if the real loudspeaker is to be used in biamp mode.

Once the biamp signal routing is setup the ***SpeakerLab*** window should look like Figure 2-180. Save the XGLL project file.



Inputs : 2

	Display Name	Imped. [Ohm]	HFSource	MFSource	LFSource
1	HF/MF	8	keyFilterHF1	keyFilterMF1	
2	LF	8			<External>

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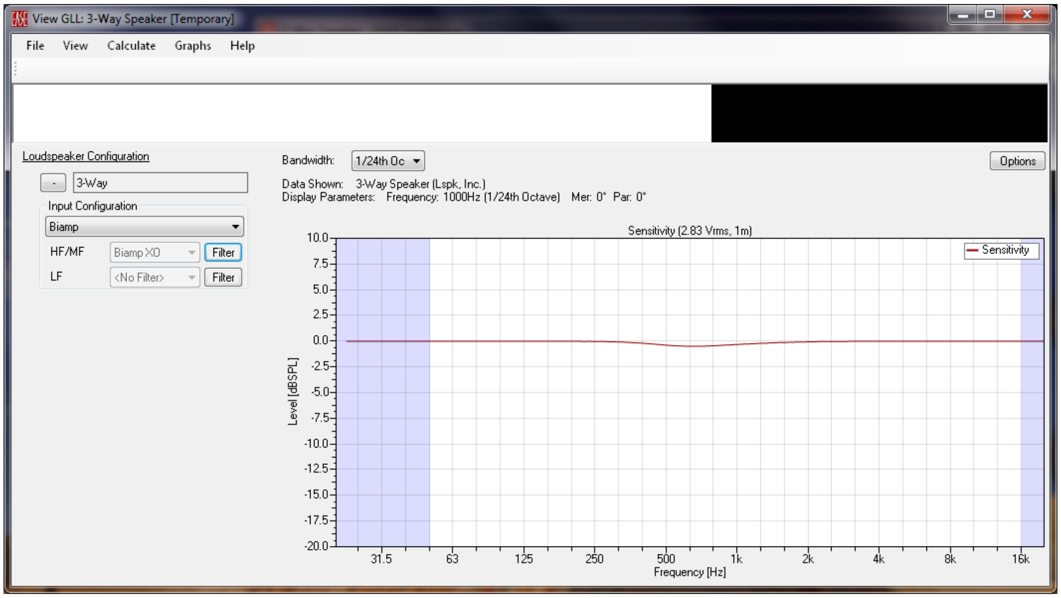
Figure 2-180 – Input Configuration tab showing 2 inputs and one of the signals routed through the appropriate filters before going to the acoustic sources

Run the GLL to see how the biamp mode works. Expand the 3-Way Loudspeaker Configuration. Notice that the default Input Configuration is Passive mode. Use the dropdown to select the Biamp mode.

Before we calculate the GLL we should setup the appropriate external filters. Click the Filter button for the HF/MF input. Click the Add Filter button in the Filter Bank and select IIR Filter. Set the parameters for a high pass, Linkwitz-Riley, fourth order, at "500" Hz. Click the Filter button for the LF input. Add an IIR filter and set the parameters for a low pass, Linkwitz-Riley, fourth order, at "500" Hz.

Recall that our passive filter for the MF had reversed polarity due to the second order passive filters that were used. We may want to reverse to polarity of either the filter for the LF or the HF/MF.

Calculate the GLL and look at the sensitivity. It should be an almost flat line and look like Figure 2-181. The external filters used for this configuration can be saved in a configuration file if desired. This will make it easier to recall the external filter settings than having to manually add them and change the setting next time.



Loudspeaker Configuration

3-Way

Input Configuration

Biamp

HF/MF Biamp XO

LF <No Filter>

Bandwidth: 1/24th Oc

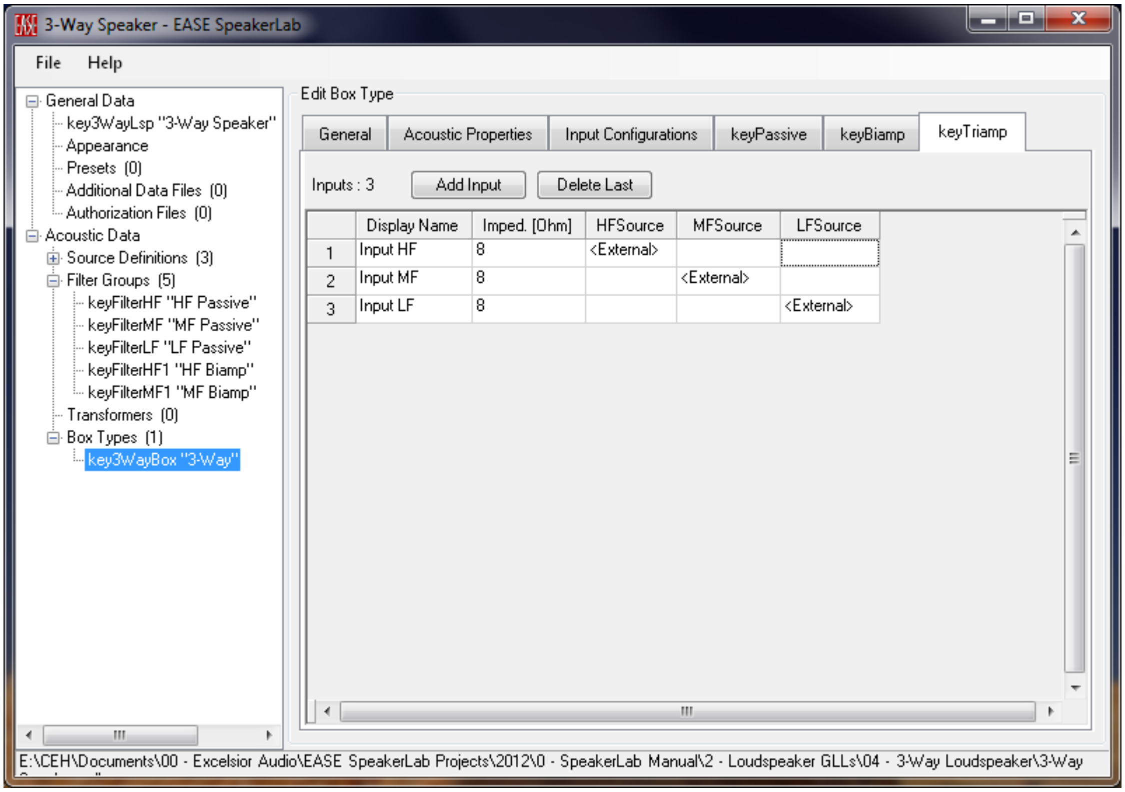
Data Shown: 3-Way Speaker (Lspk, Inc.)

Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-181 – Running GLL showing the calculated sensitivity in biamp mode

2.13.2.3 Triamp Mode – All External Filters

Click on the keyTriamp tab. This should already be setup properly with the three different inputs, each being routed directly to their respective acoustical sources. If the SpeakerLab window does not look like Figure 2-182 please make the appropriate changes so that it does. Save the XGLL project file.



- General Data
  - key3WayLsp "3-Way Speaker"
  - Appearance
  - Presets (0)
  - Additional Data Files (0)
  - Authorization Files (0)
- Acoustic Data
  - Source Definitions (3)
  - Filter Groups (5)
    - keyFilterHF "HF Passive"
    - keyFilterMF "MF Passive"
    - keyFilterLF "LF Passive"
    - keyFilterHF1 "HF Biamp"
    - keyFilterMF1 "MF Biamp"
  - Transformers (0)
  - Box Types (1)
    - key3WayBox "3-Way"

Inputs : 3

	Display Name	Imped. [Ohm]	HFSource	MFSource	LFSource
1	Input HF	8	<External>		
2	Input MF	8		<External>	
3	Input LF	8			<External>

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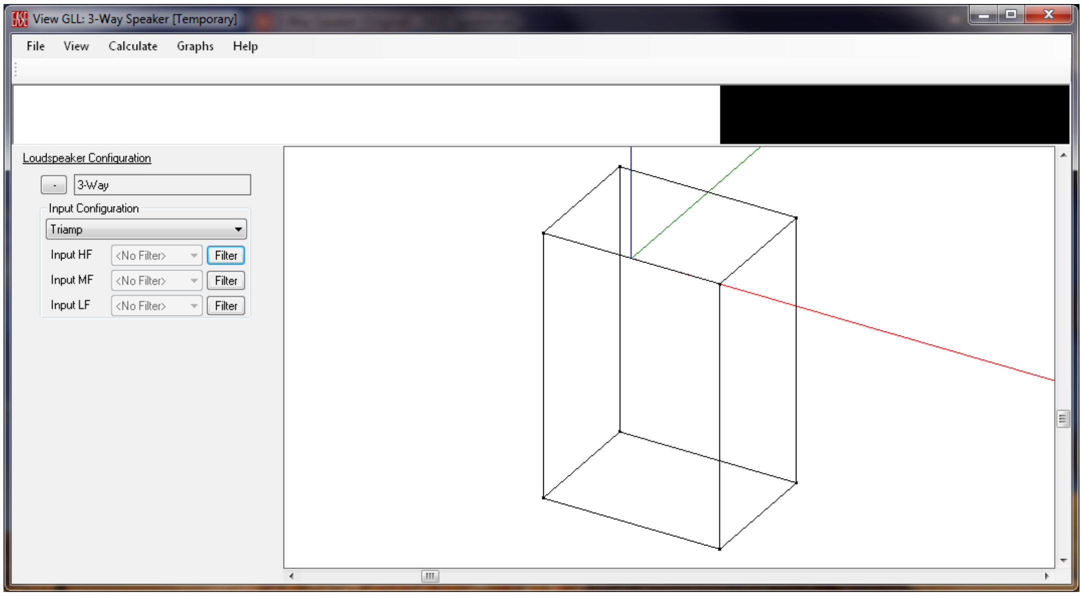
Figure 2-182 – Input Configuration tab showing 2 inputs with the signals routed directly to the acoustic sources

Run the GLL, expand the 3-WAY LOUDSPEAKER CONFIGURATION and select the TRIAMP under INPUT CONFIGURATION. All three external filters must be specified now. Once the filters are entered and the GLL is calculated the sensitivity response might look like that shown in Figure 2-184.



View GLL: 3-Way Speaker [Temporary]

Loudspeaker Configuration: 3-Way

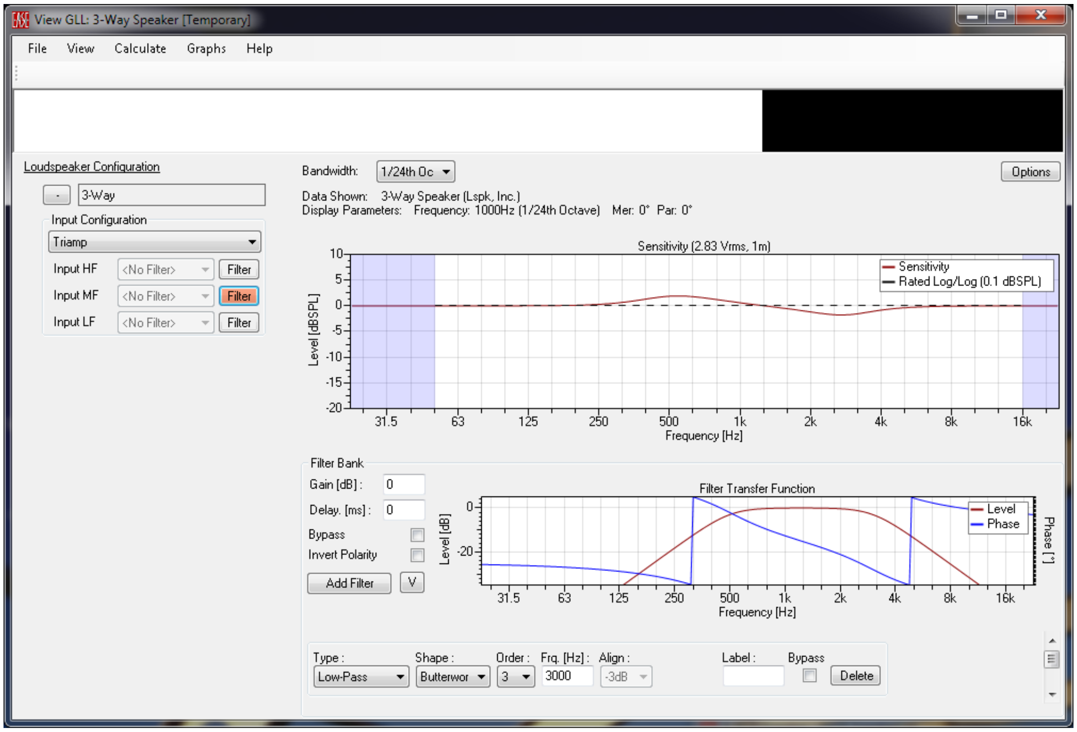
Input Configuration: Triamp

Input HF: <No Filter>

Input MF: <No Filter>

Input LF: <No Filter>

Figure 2-183 – Running the GLL with the Triamp input configuration selected



View GLL: 3-Way Speaker [Temporary]

Loudspeaker Configuration: 3-Way

Input Configuration: Triamp

Input HF: <No Filter>

Input MF: <No Filter>

Input LF: <No Filter>

Bandwidth: 1/24th Oc

Data Shown: 3-Way Speaker (Lspk, Inc.)

Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Filter Bank

Gain [dB] : 0

Delay. [ms] : 0

Bypass □

Invert Polarity □

Type : Low-Pass

Shape : Butterwor

Order : 3

Frq. [Hz] : 3000

Align : -3dB

Label :

Bypass □

Figure 2-184 – GLL with Triamp configuration selected showing the sensitivity with some external filters used

2.14 Column Loudspeakers

Column loudspeakers are fixed line arrays. They are typically intended to be used with only one box at a time. If multiple boxes are to be used together it is better to use either a cluster-type or a line array-type GLL to model the loudspeaker system. The loudspeaker may be either passive or active and can include multiple settings for altering its directivity.

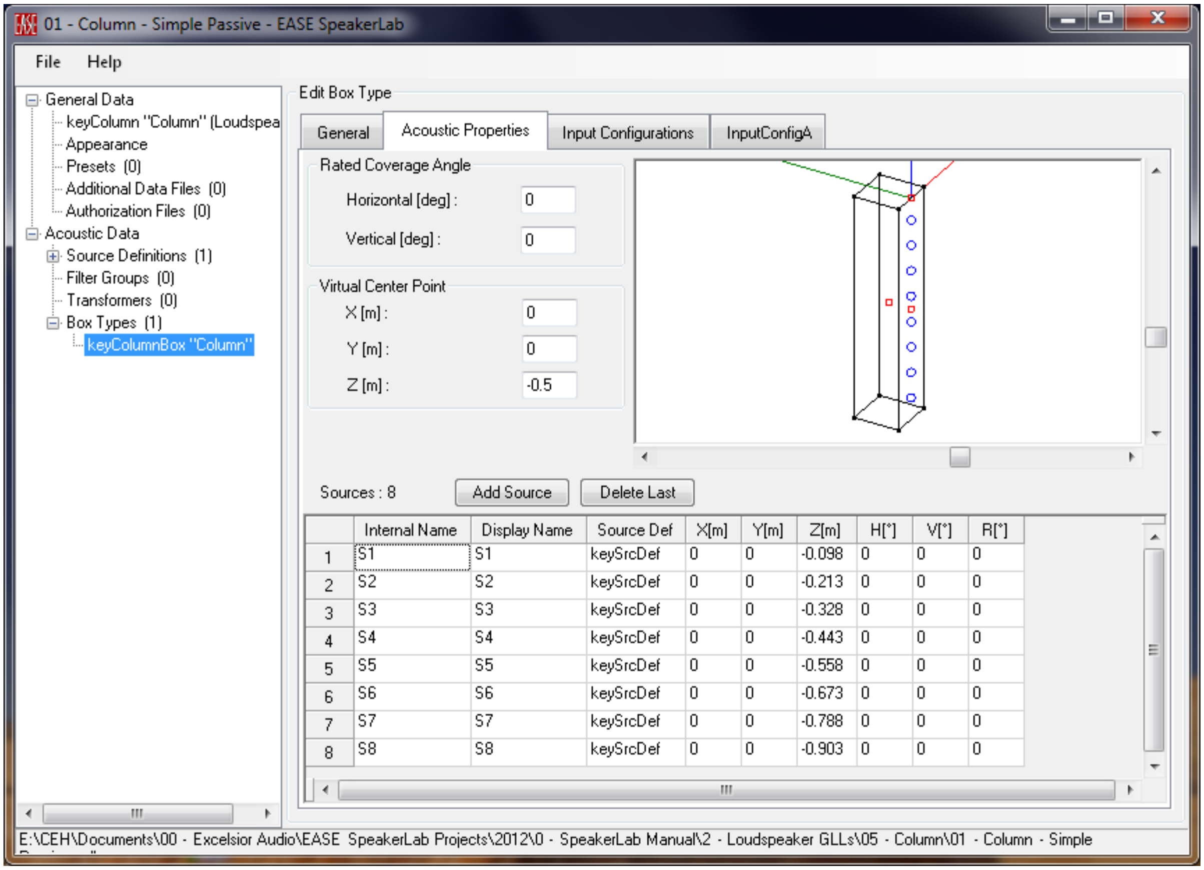
2.14.1 Simple Passive Column

Let's model a very simple passive column loudspeaker consisting of 8 full-range drivers. We shall use omni-directional point sources to model these drivers. This can be an effective design aid in the initial design stages of the loudspeaker system to model and visualize the directivity of a particular driver layout/positioning.

Create a new project. Select COLUMN from the LOUDSPEAKER tab. Choose the name for the XGLL project and the folder in which it is to be saved and click the OK button.

Expand the ACOUSTIC DATA > SOURCE DEFINITIONS property in the property tree and then click on the KEYSRCDEF property. Note that in the Edit Source Definition page for the DATA TYPE, the IDEAL SPHERE is already selected.

Select the BOX TYPES > KEYCOLUMNBOX property in the property tree and then click on the ACOUSTIC PROPERTIES tab in the EDIT BOX TYPE page. The vertical spacing between the sources (drivers) is 100 mm. This is a bit too close for the 4.5 inch (115 mm) drivers we want to model for our loudspeaker. Change the values in the Z[M] column of the source table so that they match the ones shown in Figure 2-185.



Rated Coverage Angle

Horizontal [deg] : 0

Vertical [deg] : 0

Virtual Center Point

X [m] : 0

Y [m] : 0

Z [m] : -0.5

Sources : 8

	Internal Name	Display Name	Source Def	X[m]	Y[m]	Z[m]	H[°]	V[°]	R[°]
1	S1	S1	keySrcDef	0	0	-0.098	0	0	0
2	S2	S2	keySrcDef	0	0	-0.213	0	0	0
3	S3	S3	keySrcDef	0	0	-0.328	0	0	0
4	S4	S4	keySrcDef	0	0	-0.443	0	0	0
5	S5	S5	keySrcDef	0	0	-0.558	0	0	0
6	S6	S6	keySrcDef	0	0	-0.673	0	0	0
7	S7	S7	keySrcDef	0	0	-0.788	0	0	0
8	S8	S8	keySrcDef	0	0	-0.903	0	0	0

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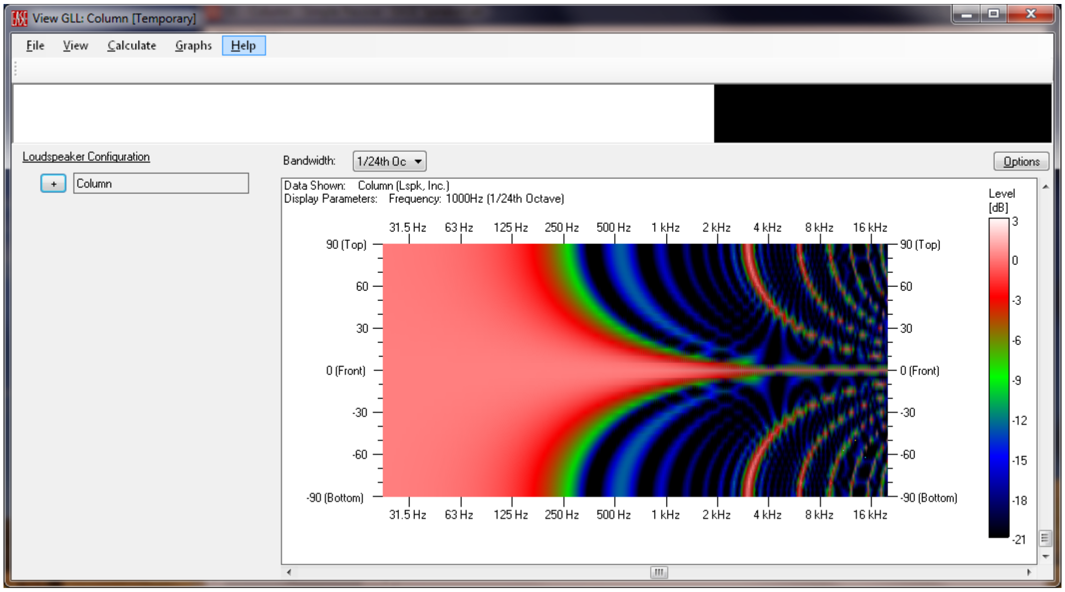
Figure 2-185 – Acoustic Properties tab showing the vertical spacing of the sources

Click the InputConfigA tab. Notice that there is a single input with the external signal being routed directly to all 8 of the sources. There are no internal filters being used.

Now run the GLL and calculate its response. In the Calculation Parameters dialog box change the Balloon Parameters Resolution to Intermediate (5°). Look at the vertical directivity map (Graphs > Vertical Map or Ctrl+Shirt+V). Change the plan view (View > Plan View (z) or Alt+Z). This will need to be done three times to have the display match Figure 2-186.



Bandwidth: 1/24th Oc

Data Shown: Column (Lspk, Inc.)
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-186 – Running GLL showing the vertical directivity map

Here we can see the vertical directivity of the column loudspeaker with the driver spacing we entered. Notice the very strong lobes 90° off-axis at 3 kHz. Our driver spacing of 115 mm is a full wavelength at almost 3 kHz. This is the reason for these lobes.

2.14.2 Active - Filter Groups for Beam Steering

We can use the internal filtering of the GLL to enable some beam steering options for our column loudspeaker. The first two examples, shown below, use delay. They will only be possible if the loudspeaker system is an active system with DSP on-board the loudspeaker. The third example (Frequency Shading) can be accomplished with either active or passive filters.

2.14.2.1 Example: Delay Steering (Down Tilt)

Suppose that we want to mount our column loudspeaker straight against a wall. For cosmetic reasons the loudspeaker box itself is not allowed to be tilted down. This may cause a problem in covering the audience area. We can apply delay to each individual driver to tilt the beam radiated from the column. The beam can be tilted either up or down depending on what is desired. In this example we will enter a value to tilt the beam down by 15° and 30°.

In SpeakerLab select the ACOUSTIC DATA > FILTER GROUPS property in the property tree. We will need a separate filter group for each driver so add 8 filter groups in the EDIT FILTER GROUPS page.

Select ACOUSTIC DATA > FILTER GROUPS > KEYFILTERGRP1 property in the property tree. In the EDIT FILTER GROUP page change the NUMBER OF FILTERS to 2. Change the Display Name of the filter in row 1 to "15°" and row 2 to "30°".

FILTER GROUPS > KEYFILTERGRP2 property in the property tree. In the EDIT FILTER GROUP page change the DISPLAY NAME to "FilterGrp2" and change the NUMBER OF FILTERS to "2". Make the Display Name for these 2 filters "15°" and "30°", respectively. Repeat this for the other 6 filters with the appropriate display names for the filter group.

For tilting the main lobe down driver at the top does not need to be delayed, only the drivers below it. Select the KEYFILTERGRP1 property in the property tree and on the General tab save the filter in row 1 as "01 - Delay 0ms.xgfb". Load this same filter for the filter in row 2. Select the remaining keyFilterGrp item in the property tree and enter the delay values shown in Table 2-1 on each of the keyFilterDef tabs for each filter group.

Table 2-1

	KEYFILTERDEF1 tab	**KEYFILTERDEF2 tab**
KEYFILTERGRP	**Delay for 15°**	**Delay for 30°**
1	0.000	0.000
2	0.090	0.170
3	0.179	0.341
4	0.269	0.511
5	0.358	0.681
6	0.448	0.852
7	0.537	1.022
8	0.627	1.192

Click on the KEYFILTERGRP2 property in the property tree and then on the General tab. Save both filters in filter group number 2. Use a filename such as "YY – Delay ZZ°.xgfb" where YY is the filter group number (with a leading zero) and ZZ is the tilt angle (either 15 or 30). Remember we already saved the filters for filter group 1. Click on the KEYFILTERGRP3 property in the property tree and then on the General tab. Save both filters in filter group number 3. Repeat this until both filters in all 8 filter groups have been saved. Remember to check the RELATIVE checkbox after saving each filter. Save the XGLL project file.

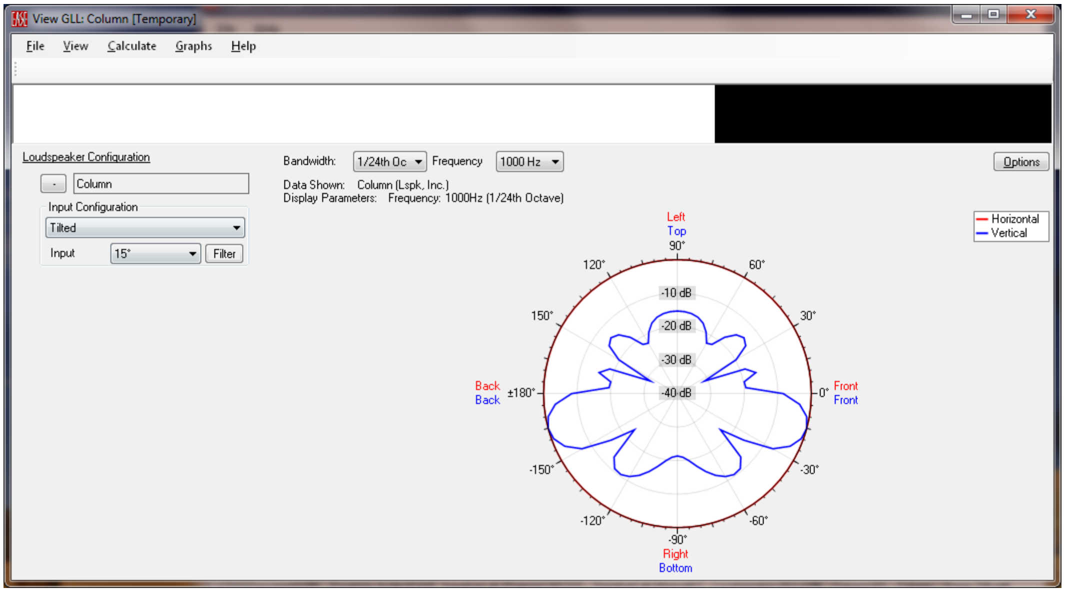
We must now assign the signal routing through the appropriate filters to enable the beam steering. Select ACOUSTIC DATA > BOX TYPES > KEYCOLUMNBOX in the property tree and then the INPUT CONFIGURATIONS tab in the Edit Box Type page. Let's change the name DISPLAY NAME for the input configuration in row 1 to "Straight Ahead". This will be the existing signal routing that does not use any of the filters with delay we just setup.

Add an input configuration. Change its INTERNAL NAME to "InputConfigTilted" and its DISPLAY NAME to "Tilted". Click on the InputConfigTilted tab. Assign the signal routing so that S1 use KEYFILTERGRP1, S2 use KEYFILTERGRP2, S3 use KEYFILTERGRP3, etc. until all 8 filter groups are assigned to the proper sources.

Save the XGLL project file.

We can now run the GLL. Expand the Loudspeaker Configuration of the Column. We can see that the Input Configuration uses the Straight Ahead configuration. Calculate the response of the GLL using the Intermediate (5°) resolution. The vertical directivity map should look the same as Figure 2-186.

Change the Input Configuration to Tilted using the dropdown box. Notice that the Input dropdown box became active and changed to the default of 15°. Recalculate the response of the GLL. The vertical directivity map gets a bit messy since the main lobe is no longer directly on axis. Lets' look at the polar graph instead (Graphs > Polars or Ctrl+Shift+P). If all of the delays were entered correctly and the filters assigned to the proper drivers the display should match Figure 2-187. We can see that the main lobe is indeed tilted down 15°. Scroll through different frequencies to see how the directivity changes with frequency.


View GLL: Column [Temporary]

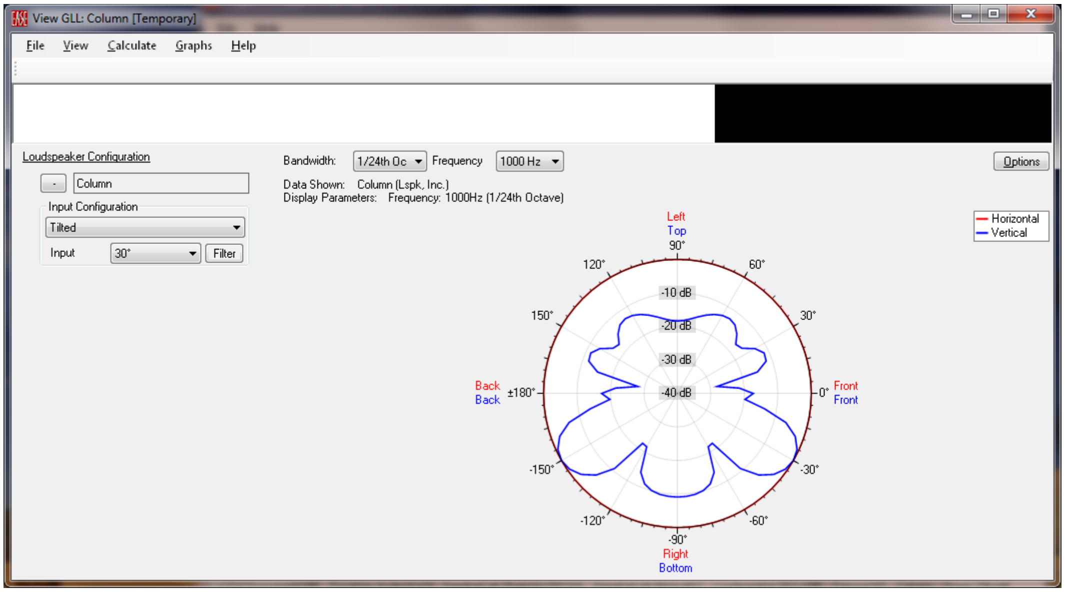
Loudspeaker Configuration
Column
Input Configuration: Tilted
Input: 15°

Bandwidth: 1/24th Oc
Frequency: 1000 Hz

Data Shown: Column (Lspk, Inc.)
Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-187 – Running GLL showing the polar graph at 1 kHz for the Tilted input configuration and the 15° filters selected

Change the Input from 15° to 30° using the dropdown box and recalculate the GLL. The display should look like Figure 2-188 if the delays and signal routed were set correctly.



View GLL: Column [Temporary]

Loudspeaker Configuration

Column

Input Configuration: Tilted

Input: 30°

Bandwidth: 1/24th Oc

Frequency: 1000 Hz

Data Shown: Column (Lspk, Inc.)

Display Parameters: Frequency: 1000Hz (1/24th Octave)

Figure 2-188 – Running GLL showing the polar graph at 1 kHz for the Tilted input configuration and the 30° filters selected

2.14.2.2 Example: Delay Tapering (Increasing Beamwidth)

The main lobe of our example GLL has a fairly narrow beamwidth above about 400 - 500 Hz (Figure 2-186). Above about 1 kHz it is very narrow and continues to decrease as frequency increases. We can open up the beamwidth somewhat by appropriately delaying the signal to the drivers. This requires that the drivers farther from the center of the array be delayed progressively more. The goal is to delay the drivers so that the virtual position of each one is on the arc of a circle whose radius is defined by the center of the array and the outer-most driver. The desired coverage angle is also defined by the position of the outer-most drivers in the array.

An example layout is shown in Figure 2-189. Here we can see the delay for each driver is given by the required rearward spacing y_n, where the subscript n is for a given driver. It has a value given by the following equation [8].

$$y_n = \frac{L}{\sin\theta} - \sqrt{\left(\frac{L}{\sin\theta}\right)^2 - {x_n}^2}$$

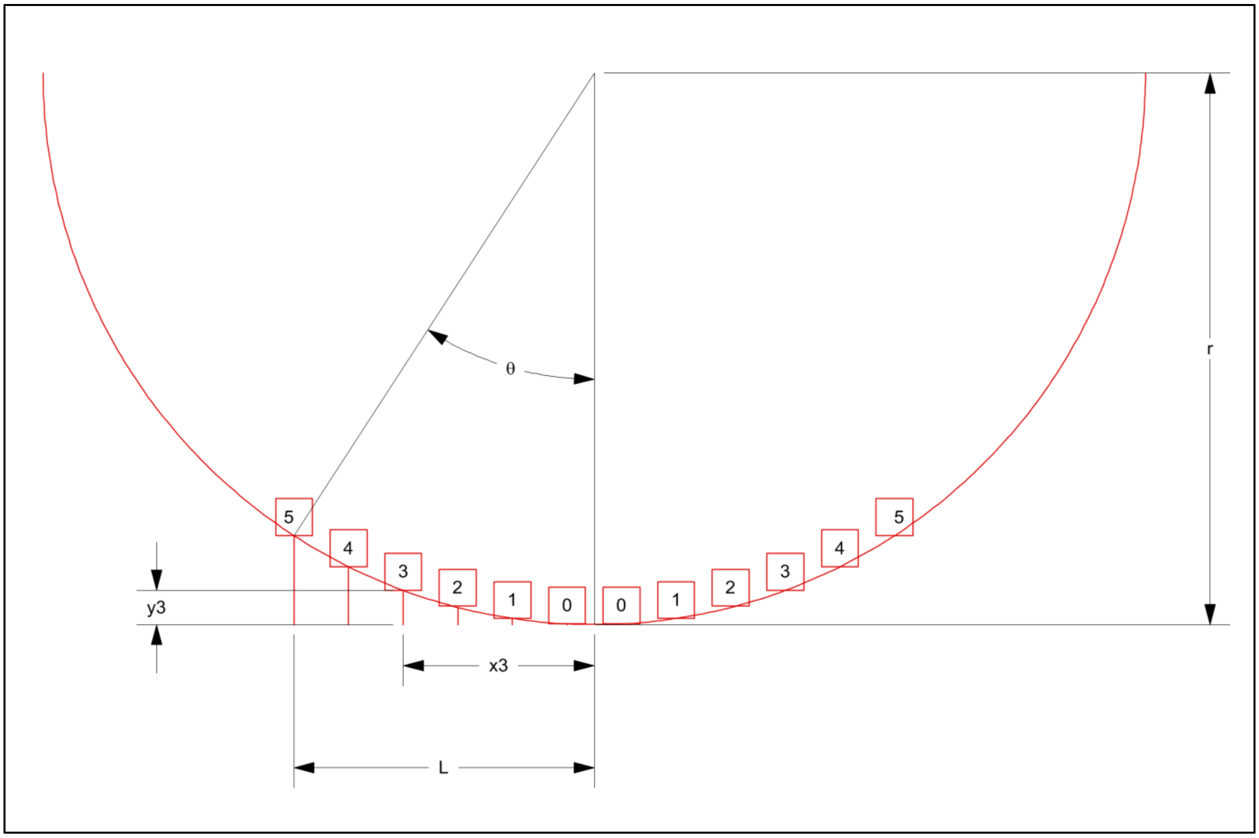


Figure 2-189 – Geometry of an arc of acoustical sources

The center drivers will have the shortest delay. For arrays with an even number of drivers the delay to all of the driver pairs can be reduced by the delay to the center driver pair. This is not the case for arrays with an odd number of drivers, as the center driver will require no delay itself.

With the above information and our column of 8 drivers with a spacing of 115 mm between them we can calculate the required delay values for each driver pair provided we also know the desired opening angle. We will use an opening angle of 70°. The value of θ in the equation is half of this, 35°. The required delay times for each driver pair are shown in Table 2-2.

Table 2-2

	θ = 35°
Driver Pair	**Delay (ms)**
0	0.000
1	0.056
2	0.172
3	0.362

We need to add a new set of filters to the XGLL project to implement these delay settings. We will add 4 completely new filter groups to do this. This allows the existing filter groups for beam steering the main lobe to remain unaffected.

In ***SpeakerLab*** select the ACOUSTIC DATA > FILTER GROUPS property. On the EDIT FILTER GROUPS page duplicate any one of the filter groups just one time. Select the new filter group in the property tree. In the EDIT FILTER GROUP page change the INTERNAL NAME to "keyFilterGrp10" and the DISPLAY NAME to "FilterGrp10". For the filter in row 1 of the filter table change the DISPLAY NAME to "Open 70°".

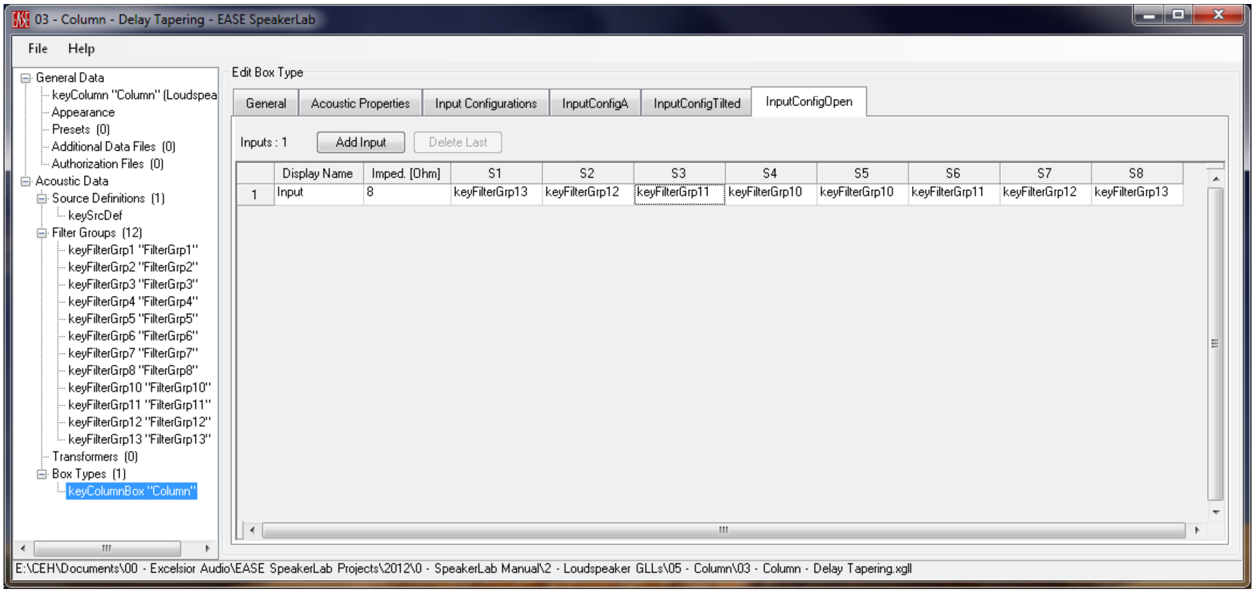
Once again select the ACOUSTIC DATA > FILTER GROUPS property. On the EDIT FILTER GROUPS page select the filter in row 9. This should be the KEYFILTERGRP10 filter group that we just added. Duplicate this filter group 3 times. The number appended to the Internal Name of these filter groups should automatically be incremented so that these new filters groups are keyFilterGrp10 – keyFilterGrp13. You may want to change the Display Name of each of these filter groups to correspond to the numbering of the Internal Name, but this is not required.

Enter the delay times from Table 2-2 in the Delay box for each of the new filter groups. Save each filter using a unique name, using the Relative path. A file name such as "Open 70° - X.xgfb", where X is the driver pair number, may work well.

We must now setup the proper signal routing through these new filters to enable the delay tapering of the drivers in the column to be utilized. Select ACOUSTIC DATA > BOX TYPES > KEYCOLUMNBOX in the property tree. In the EDIT CONFIGURATIONS page add another input configuration to the two that already exist. Change the INTERNAL NAME to "InputConfigOpen" and the DISPLAY NAME to "Open Beam". Click on the InputConfigOpen tab and assign the appropriate filters to their corresponding sources. Once this is complete the display should look like Figure 2-190. Save the XGLL project.

We can now run the GLL to see how the delay tapering, and the functionality of the GLL in selecting the various processing options works. Expand the LOUDSPEAKER CONFIGURATION for the COLUMN. The default for the INPUT CONFIGURATIONS is STRAIGHT AHEAD. This is the option with no filtering at all. Go ahead and calculate the response of the GLL for the Straight Ahead Input Configuration. Change the graph to the vertical directivity map. Change the view to the plan view.

Use the INPUT CONFIGURATION dropdown box to select the OPEN BEAM configuration. This one is now in addition to the other two that we previously had. Once this is selected we can see that the INPUT has changed to OPEN 70°. Recall that this was the Display Name given to the filters in row 1 of the newly added filter groups. If we wanted to include additional opening angles other than 70° we could have added more rows of filter in these filter groups with the proper delay times to yield the desired opening angles. The Input dropdown box would then be activated to allow the selection of these other choices.


03 - Column - Delay Tapering - EASE SpeakerLab
File Help
General Data
keyColumn "Column" (Loudspea
Appearance
Presets (0)
Additional Data Files (0)
Authorization Files (0)
Acoustic Data
Source Definitions (1)
keySrcDef
Filter Groups (12)
keyFilterGrp1 "FilterGrp1"
keyFilterGrp2 "FilterGrp2"
keyFilterGrp3 "FilterGrp3"
keyFilterGrp4 "FilterGrp4"
keyFilterGrp5 "FilterGrp5"
keyFilterGrp6 "FilterGrp6"
keyFilterGrp7 "FilterGrp7"
keyFilterGrp8 "FilterGrp8"
keyFilterGrp10 "FilterGrp10"
keyFilterGrp11 "FilterGrp11"
keyFilterGrp12 "FilterGrp12"
keyFilterGrp13 "FilterGrp13"
Transformers (0)
Box Types (1)
keyColumnBox "Column"
Edit Box Type
General
Acoustic Properties
Input Configurations
InputConfigA
InputConfigTilted
InputConfigOpen
Add Input
Delete Last

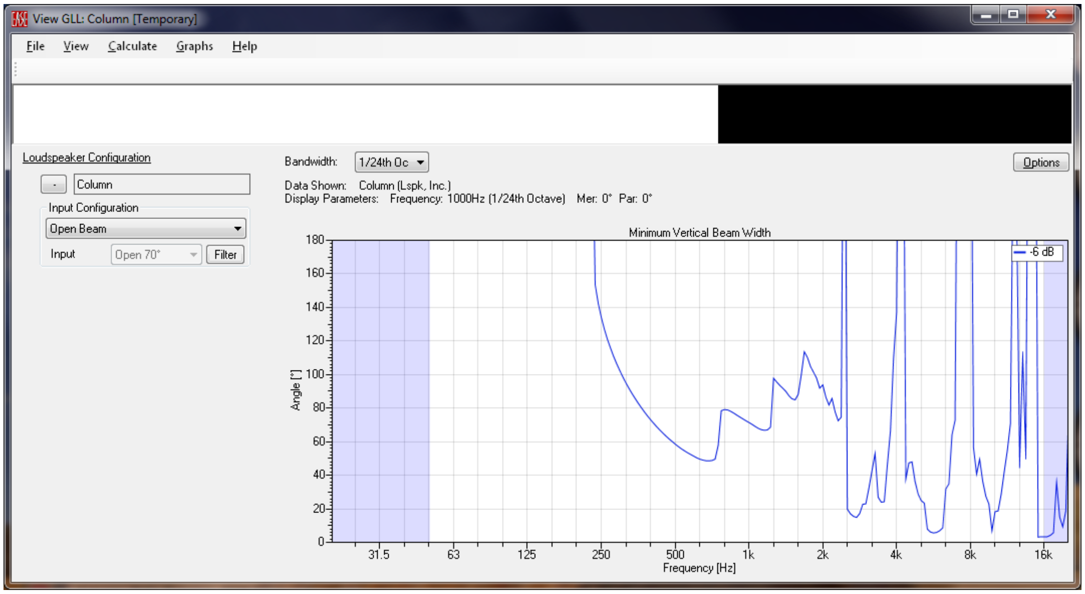
Inputs : 1

	Display Name	Imped. [Ohm]	S1	S2	S3	S4	S5	S6	S7	S8
1	Input	8	keyFilterGrp13	keyFilterGrp12	keyFilterGrp11	keyFilterGrp10	keyFilterGrp10	keyFilterGrp11	keyFilterGrp12	keyFilterGrp13

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\2 - Loudspeaker GLLs\05 - Column\03 - Column - Delay Tapering.xgll

Figure 2-190 – Input Configuration tab showing the signal routing to each of the acoustic sources

Recalculate the response of the GLL. We should see that the vertical beamwidth has increased. This is also shown in the vertical beamwidth graph (GRAPHS > BEAMWIDTH > VERTICAL or CTRL+ATL+V) of Figure 2-191. The opening of the beamwidth falls apart above about 1.5 kHz. This is because the spacing between the drivers is about half of a wavelength at this frequency.


View GLL: Column [Temporary]
File View Calculate Graphs Help
Loudspeaker Configuration
Options
Filter
Minimum Vertical Beam Width
-6 dB
180
160
140
120
100
80
60
40
20
0
Angle [°]
31.5
63
125
250
500
1k
2k
4k
8k
16k
Frequency [Hz]

Column
Input Configuration: Open Beam
Input: Open 70°
Bandwidth: 1/24th Oc
Data Shown: Column (Lspk, Inc.)
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-191 – Running GLL showing the vertical beamwidth of the array using the 70° opening angle filters

2.14.2.3 Example: Frequency Shading (Constant Directivity & Line Array Size)

Another way to increase the beamwidth in the higher frequency region for an array of drivers used in a column loudspeaker is to employ frequency shading. This method seeks to keep the ratio of the overall length of the line array of drivers to the wavelength being radiated constant over as large a frequency range as possible. This means that at high frequencies (shorter wavelengths) the length of the array needs to decrease. Doing this effectively can yield a fairly constant-directivity loudspeaker system.

To accomplish this we need to apply low pass filters to the pairs of drivers that are farther away from the center of the array. The farther away from the center the drivers are located the lower in frequency the low pass filter will need to be set. This will constrain the higher frequencies to be radiated only from the center drivers in the array. Another way to think about this is that for every doubling of frequency (octave) the height of the array needs to be cut in half.

Let's start at about 500 Hz. At this frequency we will begin to decrease the height by a factor of 2 by applying a first order Butterworth low pass filter to the outer 4 drivers. This will make the array only 4 drivers tall, instead of 8, above about 500 Hz. At 1 kHz we need to cut the height in half again, so we'll use another first order Butterworth low pass filter at 1 kHz to the pair of drivers next to the center pair. At 2 kHz we again need to cut the height in half so a final first order Butterworth low pass filter at 2 kHz to one of the center drivers will accomplish this. This makes 3 different low pass filters. We'll also need one for the other center driver, so that a filter group is used for all the drivers, for a total of 4 filter groups.

In ***SpeakerLab*** select the ACOUSTIC DATA > FILTER GROUPS property. On the EDIT FILTER GROUPS page duplicate any one of the filter groups just one time. Select the new filter group in the property tree. In the EDIT FILTER GROUP page change the INTERNAL NAME to "keyFilterGrp20" and the DISPLAY NAME to "FilterGrp20". For the filter in row 1 of the filter table change the DISPLAY NAME to "Freq. Shading". Click the KEYFILTERDEF1 tab. Delete any filters that may be present in the filter bank. Make sure the DELAY [MS] setting is "0". Click on the General tab and save this filter. This is the "blank" filter that will be used for one of the center drivers.

Once again select the ACOUSTIC DATA > FILTER GROUPS property. On the EDIT FILTER GROUPS page select the filter group in row 13. This should be the KEYFILTERGRP20 filter group that we just added. Duplicate this filter group 3 times. The number appended to the Internal Name of these filter groups should automatically be incremented so that these new filters groups are keyFilterGrp20 – keyFilterGrp23. You may want to change the Display Name of each of these filters to correspond to the numbering of the Internal Name, but this is not required.

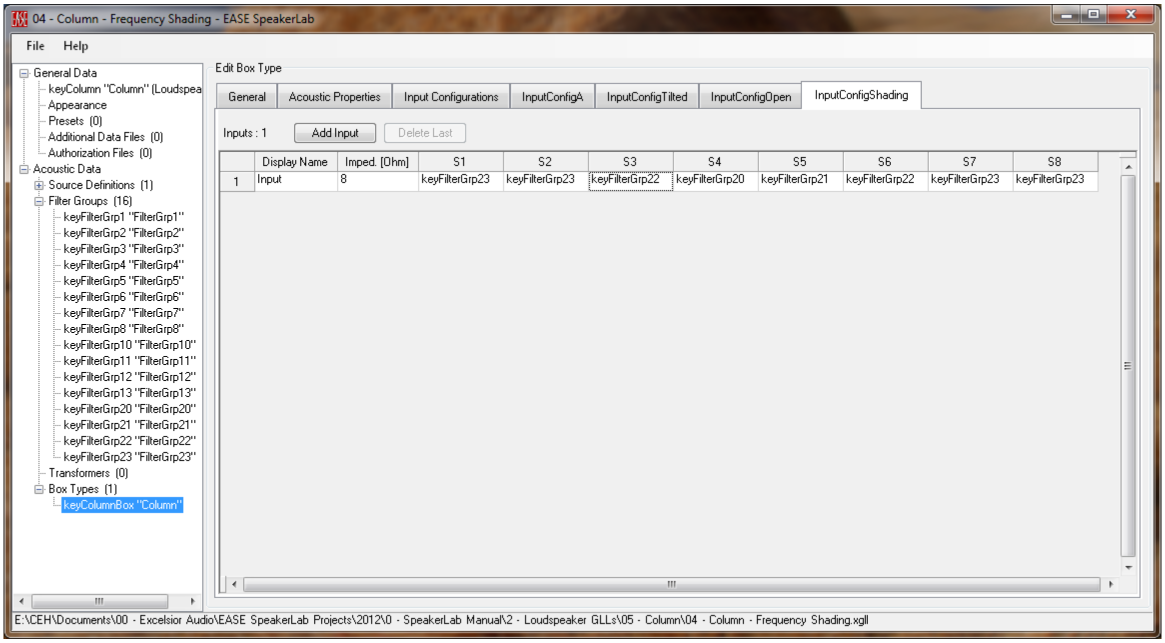
Select the ACOUSTIC DATA > FILTER GROUPS > KEYFILTERGRP21 property. On the EDIT FILTER GROUP page click the KEYFILTERDEF1 tab and add an IIR filter. Change the settings for this filter to a low pass, Butterworth, first order, at "2000" Hz. Save this filter. This will be the filter applied to the other center driver in the column.

Select the ACOUSTIC DATA > FILTER GROUPS > KEYFILTERGRP22 property. On the EDIT FILTER GROUP page click the KEYFILTERDEF1 tab and add an IIR filter. Change the settings for this filter to a low pass, Butterworth, first order, at "1000" Hz. Save this filter. This will be the filter applied to the pair of drivers immediately adjacent to the pair of center drivers.

Select the ACOUSTIC DATA > FILTER GROUPS > KEYFILTERGRP23 property. On the EDIT FILTER GROUP page click the KEYFILTERDEF1 tab and add an IIR filter. Change the settings for this filter to a low pass, Butterworth, first order, at "500" Hz. Save this filter. This will be the filter applied to the outer 4 drivers in the column.

Save the XGLL project file.

Now we must setup the proper signal routing through the low pass filters to enable the frequency shading drivers in the column. Select ACOUSTIC DATA > BOX TYPES > KEYCOLUMNBOX in the property tree. In the EDIT CONFIGURATIONS page add another input configuration to the three that already exist. Change the INTERNAL NAME to "InputConfigShading" and the DISPLAY NAME to "Frequency Shading". Click on the INPUTCONFIGSHADING tab and assign the appropriate filters to their corresponding sources. Once this is complete the display should look like Figure 2-192. Save the XGLL project.


04 - Column - Frequency Shading - EASE SpeakerLab
File
Help
General Data
keyColumn "Column" (Loudspea
Appearance
Presets (0)
Additional Data Files (0)
Authorization Files (0)
Acoustic Data
Source Definitions (1)
Filter Groups (16)
keyFilterGrp1 "FilterGrp1"
keyFilterGrp2 "FilterGrp2"
keyFilterGrp3 "FilterGrp3"
keyFilterGrp4 "FilterGrp4"
keyFilterGrp5 "FilterGrp5"
keyFilterGrp6 "FilterGrp6"
keyFilterGrp7 "FilterGrp7"
keyFilterGrp8 "FilterGrp8"
keyFilterGrp10 "FilterGrp10"
keyFilterGrp11 "FilterGrp11"
keyFilterGrp12 "FilterGrp12"
keyFilterGrp13 "FilterGrp13"
keyFilterGrp20 "FilterGrp20"
keyFilterGrp21 "FilterGrp21"
keyFilterGrp22 "FilterGrp22"
keyFilterGrp23 "FilterGrp23"
Transformers (0)
Box Types (1)
keyColumnBox "Column"
Edit Box Type
General
Acoustic Properties
Input Configurations
InputConfigA
InputConfigTilted
InputConfigOpen
InputConfigShading
Add Input
Delete Last

Inputs : 1

	Display Name	Imped. [Ohm]	S1	S2	S3	S4	S5	S6	S7	S8
1	Input	8	keyFilterGrp23	keyFilterGrp23	keyFilterGrp22	keyFilterGrp20	keyFilterGrp21	keyFilterGrp22	keyFilterGrp23	keyFilterGrp23

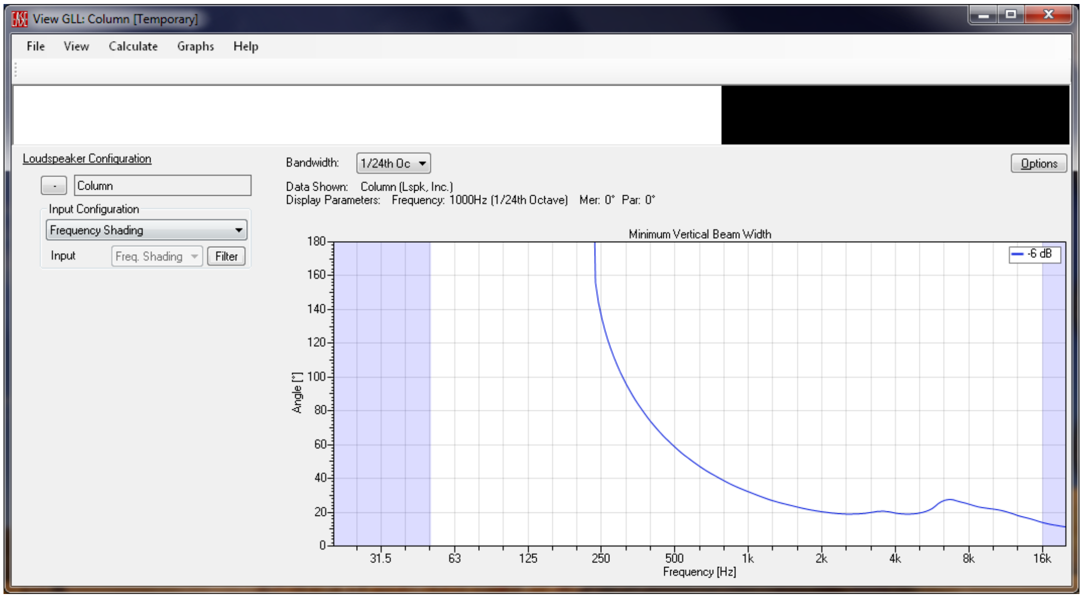
E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\2 - Loudspeaker GLLs\05 - Column\04 - Column - Frequency Shading.xgll

Figure 2-192 – Input Configuration tab showing the signal routing to each of the acoustic sources

We can now run the GLL to see how the frequency shading works. Expand the LOUDSPEAKER CONFIGURATION for the COLUMN. The default for the INPUT CONFIGURATIONS is STRAIGHT AHEAD. This is the option with no filtering at all. Go ahead and calculate the response of the GLL for the Straight Ahead Input Configuration. Change the graph to the vertical beamwidth.

Use the Input Configuration dropdown box to select the Frequency Shading configuration. Once this is selected we can see that the Input has changed to Freq. Shading, the Display Name given to the filters in row 1 of the newly added filter groups. If we wanted to include other frequency shading schemes we could have added more rows of filter in the filter groups with the desired filter settings. The Input dropdown box would then be activated to allow the selection of these other choices.

Recalculate the response of the GLL. We should see that the vertical beamwidth has increased in the mid and high frequency regions (Figure 2-193). It should also be more consistent with respect to frequency, but it isn't exactly constant. If measured data from real drivers, or theoretical piston sources, were used the beamwidth would be much closer to constant directivity over a larger bandwidth, particularly in the higher frequency region. Using more drivers to increase the height of the column would tend to make the directivity more constant in the lower frequency region.


View GLL: Column [Temporary]

Loudspeaker Configuration: Column

Input Configuration: Frequency Shading

Input: Freq. Shading

Bandwidth: 1/24th Oc

Data Shown: Column (Lspk, Inc.)

Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-193 – Running GLL showing the vertical beamwidth of the array using the frequency shading filters

2.14.3 Overridable Internal Filters

It is possible to override the internal filters within the GLL. However, to do this the GLL must allow the internal filters to be overridden. This safety makes sure that the author of the GLL files intended for override access to be granted. One of the main reasons for overriding internal filters is to allow an otherwise "locked" internal filter to be changed. This may be necessary for a GLL to model the capabilities of an electronically steered line array or column loudspeaker. In this case the filter settings and delays that feed each driver (or groups of drivers) may need to change to yield the overall beam steering shape that is desired.

When internal filters are overridden they are temporarily replaced with filters specified in the configuration file that is used to override them. The internal name of the input configuration, within the

GLL, in which the filters to be overridden are used must be known and called out in the configuration file. The internal name of each source must also be known and called out in the configuration file. Knowing these items is trivial for the author of the GLL (the one who created the XGLL project and compiled it into a GLL). However, it would be very difficult for anyone else to know these items without being told. This further protects the internal filters in the GLL from being overridden in an unintended manner.

The first step in allowing internal filters to be overridden is to explicitly flag them as being overridable. Do this by checking the CONFIGURATION OVERRIDABLE check box on the EDIT FILTER GROUP page. Let's do this with our frequency shading example. Select ACOUSTIC DATA > FILTER GROUPS > KEYFILTERGRP20 in the property tree and check the box in the page on the right. Repeat this for keyFilterGrp21 – keyFilterGrp23. Now these four filter groups can be overridden. However, none of the other filter groups in this XGLL project can be.

The next thing to be done is to create a configuration file that specifies which filters to override and the parameters of the new, overriding filters. A brief example of one of these configuration files was shown in Figure 2-149. It may be a good idea to review this briefly.

We will show a configuration file that will replace the internal filters with filters that are one octave lower than the internal filters. The results will be that the frequency shading will occur at a lower frequency and the beamwidth of the column should increase.

The complete configuration file for overriding the internal filters in the XGLL project on which we have been working is listed in Figure 2-194 through Figure 2-201. The content of this file is broken up into multiple figures to help explain it better. To create this file yourself, copy the text in each figure and paste it into a single file in a text editor. Save the file with the .XGLC extension.

Note that the configuration file will only work if the internal names for the GLL, the filter groups, and the input configuration are exactly as specified in this manual to match the names in the text for the example configuration file shown below.

```
"GLC"
"Format", "GLL Configuration"
"FormatVersion", "1.01"

"Configuration"
  "Used System", "Column", "keyColumn", "LS", 1.00, "Lspk, Inc."
  "Angle", 0.00
  "Elements", 1
    "Element"
      "Box Type", "keyColumnBox"
      "Gain", 0.00
      "Input Config", "InputConfigShading"
        "Inputs", 1
          "Input"

; The next line specifies the Internal Name of the internal filter to
override
            "Internal Filter", "keyFilterDef1"

; The next line specifies the Display Name that will appear for the
overriding filter
              "Override Setting", "*Octave Lower"

; The next line specifies the number of internal filters that will be
overridden
              "Internal Filters To Override", 8
```

Figure 2-194 – Example configuration file to override internal filters (part 1)

The items highlighted in Figure 2-194 must match exactly the names of the items in the GLL which the configuration will override. Note also that the number of inputs used in the specified input configuration must also match. If these items do not match an error will be returned when trying to open the configuration file with the GLL.

The Override Setting specifies the name which will appear in the Input dropdown box in the GLL. In this example ***Octave Lower** will be displayed. An asterisk is used at the beginning of the name to denote that this is a temporary override configuration. The name specified after Override Setting can be anything. However, the dropdown box in which it is shown in the GLL has only a limited width so long names should be avoided.

The INTERNAL FILTERS TO OVERRIDE setting specifies how many filters will be specified in the configuration file. There must be a separate filter specified for each source that is to use a filter. Even if two sources use the same filter, when specifying overriding filters a separate filter must be duplicated so it can be assigned to the desired source.

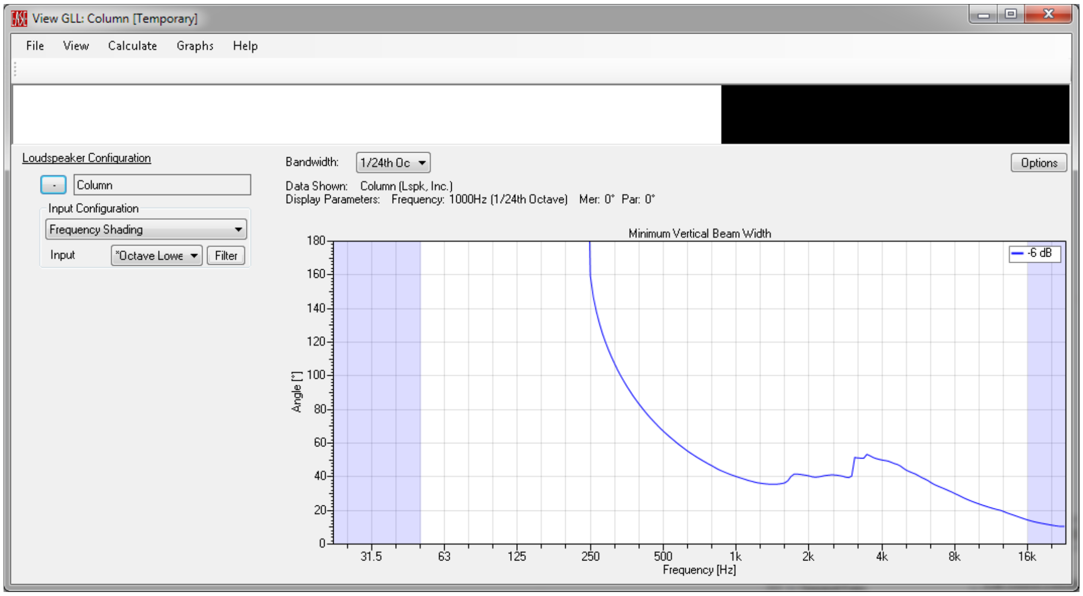
```
; The next line specifies the Internal Name of the Source (on the Acoustic
Properties tab) to which the overriding filter specified immediately below
will be applied
                "Override Internal Filter", "S1"

; The lines below specify the parameters of the overriding filter
                  "GFB"
                  "Format", "General Lspk"
                  "FormatVersion", "1.0"

                  "Bypass", "No"
                  "Gain", 0.00
                  "Delay", 0.00
                  "Invert Polarity", "No"

                  "Filters", 1
                    "Filter", "", ""
                    "Kind of Filter", "IIR Filter"
                    "Bypass", "No"
                    "Gain", 0.00
                    "Delay", 0.00
                    "Type", "Low-Pass"
                    "Shape", "Butterworth"
                    "Order", 1
                    "Frequency", 250.00
                    "Alignment", "-3dB"
                    "Q Factor", 1.00
                    "Parametric Gain", 10.00
```

Figure 2-195 – Example configuration file to override internal filters (part 2)

The Override Internal Filter setting specifies the internal name of the source in the GLL to which the filter specified immediately following this setting is to be applied. There must be the same number of these entries as the Internal Filters to Override setting.

For our frequency shading GLL this 250 Hz low pass filter will be applied to S1, which is the top driver.

For the rest of the filters specified in this configuration file sources S2, S7, and S8 will also be assigned an identical 250 Hz low pass filter. Sources S3 and S6 will be assigned a 500 Hz low pass filter. Source S5 will be assigned a 1000 Hz low pass filter. Source S4 will be assigned a filter that is bypassed. The low pass filter frequencies for these filters are all one octave lower than the internal filters for the GLL. A vertical beamwidth plot using these filters to override the internal filters is shown in Figure 2-196. Compared to the beamwidth using the internal filters (Figure 2-193), this beamwidth is about twice as wide from about 1 kHz to 8 kHz. This is to be expected since the overriding filters are 1 octave lower in frequency. They broaden the beamwidth.



Column

Input Configuration: Frequency Shading

Input: "Octave Lowe [Filter]

Bandwidth: 1/24th Oc

Data Shown: Column (Lspk, Inc.)
Display Parameters: Frequency: 1000Hz (1/24th Octave) Mer: 0° Par: 0°

Figure 2-196 – Running GLL showing the vertical beamwidth of the array with the internal filters overridden

```
; The next line specifies the Internal Name of the Source (on the Acoustic
Properties tab) to which the overriding filter specified immediately below
will be applied
                "Override Internal Filter", "S2"

; The lines below specify the parameters of the overriding filter
                  "GFB"
                  "Format", "General Lspk"
                  "FormatVersion", "1.0"

                  "Bypass", "No"
                  "Gain", 0.00
                  "Delay", 0.00
                  "Invert Polarity", "No"

                  "Filters", 1
                    "Filter", "", ""
                    "Kind of Filter", "IIR Filter"
                    "Bypass", "No"
                    "Gain", 0.00
                    "Delay", 0.00
                    "Type", "Low-Pass"
                    "Shape", "Butterworth"
                    "Order", 1
                    "Frequency", 250.00
                    "Alignment", "-3dB"
                    "Q Factor", 1.00
                    "Parametric Gain", 10.00


; The next line specifies the Internal Name of the Source (on the Acoustic
Properties tab) to which the overriding filter specified immediately below
will be applied
                "Override Internal Filter", "S3"

; The lines below specify the parameters of the overriding filter
                  "GFB"
                  "Format", "General Lspk"
                  "FormatVersion", "1.0"

                  "Bypass", "No"
                  "Gain", 0.00
                  "Delay", 0.00
                  "Invert Polarity", "No"
                  "Filters", 1
                    "Filter", "", ""
                    "Kind of Filter", "IIR Filter"
                    "Bypass", "No"
```

Figure 2-197 – Example configuration file to override internal filters (part 3)

```
                    "Gain", 0.00
                    "Delay", 0.00
                    "Type", "Low-Pass"
                    "Shape", "Butterworth"
                    "Order", 1
                    "Frequency", 500.00
                    "Alignment", "-3dB"
                    "Q Factor", 1.00
                    "Parametric Gain", 10.00


; The next line specifies the Internal Name of the Source (on the Acoustic
Properties tab) to which the overriding filter specified immediately below
will be applied
                "Override Internal Filter", "S4"

; The lines below specify the parameters of the overriding filter
                  "GFB"
                  "Format", "General Lspk"
                  "FormatVersion", "1.0"

                  "Bypass", "Yes"
                  "Gain", 0.00
                  "Delay", 0.00
                  "Invert Polarity", "No"

                  "Filters", 1
                    "Filter", "", ""
                    "Kind of Filter", "IIR Filter"
                    "Bypass", "No"
                    "Gain", 0.00
                    "Delay", 0.00
                    "Type", "Low-Pass"
                    "Shape", "Butterworth"
                    "Order", 1
                    "Frequency", 1000.00
                    "Alignment", "-3dB"
                    "Q Factor", 1.00
                    "Parametric Gain", 10.00


; The next line specifies the Internal Name of the Source (on the Acoustic
Properties tab) to which the overriding filter specified immediately below
will be applied
                "Override Internal Filter", "S5"

; The lines below specify the parameters of the overriding filter
                  "GFB"
                  "Format", "General Lspk"
```

Figure 2-198 – Example configuration file to override internal filters (part 4)

```
                  "FormatVersion", "1.0"

                  "Bypass", "No"
                  "Gain", 0.00
                  "Delay", 0.00
                  "Invert Polarity", "No"

                  "Filters", 1
                    "Filter", "", ""
                    "Kind of Filter", "IIR Filter"
                    "Bypass", "No"
                    "Gain", 0.00
                    "Delay", 0.00
                    "Type", "Low-Pass"
                    "Shape", "Butterworth"
                    "Order", 1
                    "Frequency", 1000.00
                    "Alignment", "-3dB"
                    "Q Factor", 1.00
                    "Parametric Gain", 10.00


; The next line specifies the Internal Name of the Source (on the Acoustic
Properties tab) to which the overriding filter specified immediately below
will be applied
                "Override Internal Filter", "S6"

; The lines below specify the parameters of the overriding filter
                  "GFB"
                  "Format", "General Lspk"
                  "FormatVersion", "1.0"

                  "Bypass", "No"
                  "Gain", 0.00
                  "Delay", 0.00
                  "Invert Polarity", "No"

                  "Filters", 1
                    "Filter", "", ""
                    "Kind of Filter", "IIR Filter"
                    "Bypass", "No"
                    "Gain", 0.00
                    "Delay", 0.00
                    "Type", "Low-Pass"
                    "Shape", "Butterworth"
                    "Order", 1
                    "Frequency", 500.00
                    "Alignment", "-3dB"
                    "Q Factor", 1.00
```

Figure 2-199 – Example configuration file to override internal filters (part 5)

```
                    "Parametric Gain", 10.00


; The next line specifies the Internal Name of the Source (on the Acoustic
Properties tab) to which the overriding filter specified immediately below
will be applied
                "Override Internal Filter", "S7"

; The lines below specify the parameters of the overriding filter
                  "GFB"
                  "Format", "General Lspk"
                  "FormatVersion", "1.0"

                  "Bypass", "No"
                  "Gain", 0.00
                  "Delay", 0.00
                  "Invert Polarity", "No"

                  "Filters", 1
                    "Filter", "", ""
                    "Kind of Filter", "IIR Filter"
                    "Bypass", "No"
                    "Gain", 0.00
                    "Delay", 0.00
                    "Type", "Low-Pass"
                    "Shape", "Butterworth"
                    "Order", 1
                    "Frequency", 250.00
                    "Alignment", "-3dB"
                    "Q Factor", 1.00
                    "Parametric Gain", 10.00


; The next line specifies the Internal Name of the Source (on the Acoustic
Properties tab) to which the overriding filter specified immediately below
will be applied
                "Override Internal Filter", "S8"

; The lines below specify the parameters of the overriding filter
                  "GFB"
                  "Format", "General Lspk"
                  "FormatVersion", "1.0"

                  "Bypass", "No"
                  "Gain", 0.00
                  "Delay", 0.00
                  "Invert Polarity", "No"

                  "Filters", 1
```

Figure 2-200 – Example configuration file to override internal filters (part 6)

```
                    "Filter", "", ""
                    "Kind of Filter", "IIR Filter"
                    "Bypass", "No"
                    "Gain", 0.00
                    "Delay", 0.00
                    "Type", "Low-Pass"
                    "Shape", "Butterworth"
                    "Order", 1
                    "Frequency", 250.00
                    "Alignment", "-3dB"
                    "Q Factor", 1.00
                    "Parametric Gain", 10.00


; The lines below specify the parameters of the external filter at the
input of the GLL
            "External Filter"
              "GFB"
              "Format", "General Lspk"
              "FormatVersion", "1.0"

              "Bypass", "No"
              "Gain", 0.00
              "Delay", 0.00
              "Invert Polarity", "No"

              "Filters", 0
```

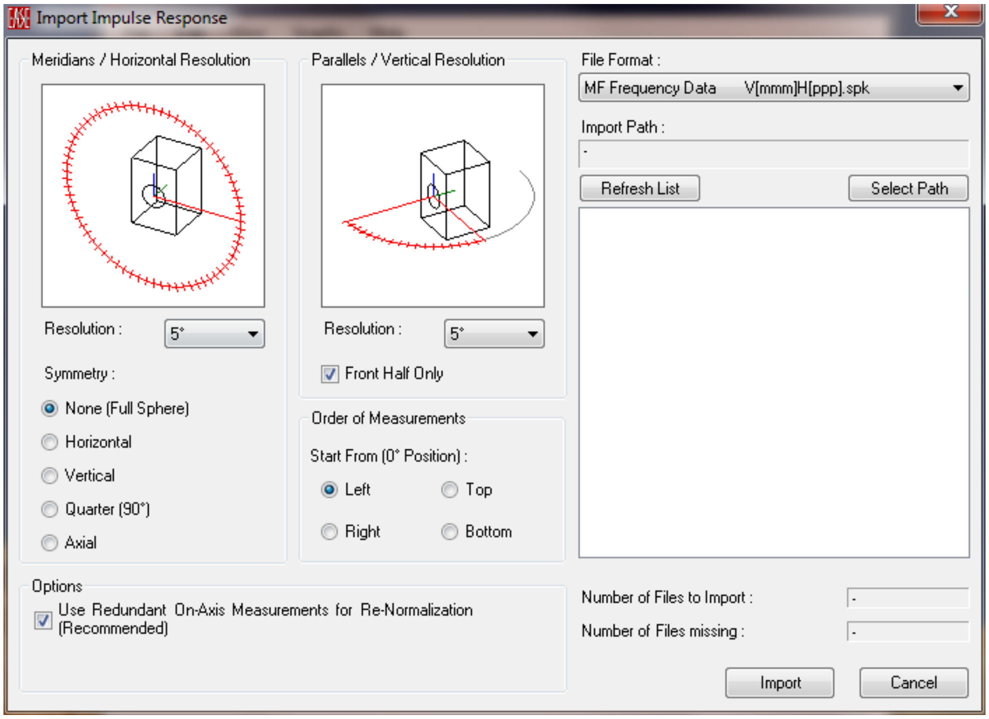
Figure 2-201– Example configuration file to override internal filters (part 7)

2.15 Ceiling & Wall Mount Loudspeakers

Creating GLLs for ceiling and wall mount loudspeaker is not much different from creating a GLL for other types of loudspeakers. All the items covered in this manual for single-source and multi-source GLL also apply to ceiling and wall mount loudspeakers. There are two typical differences of which one should be aware for ceiling and wall mount units. The first item is the collection of the directivity balloon data and entering it into the GSS. The second is specifying the parameters for transformers that are generally used with ceiling and wall mount loudspeakers.

2.15.1 Front Hemisphere of Directivity Balloon

Since by definition, ceiling and wall mount loudspeakers do not have rearward radiation of sound into a listening space their directivity balloon only needs to be entered for the front hemisphere. In the Import Impulse Response dialog box of the Source Editor there is a Front Half Only checkbox (Figure 2-202). When this box is checked only the measurement data files for the front half of the directivity balloon will be imported for the GSS. The rear side of the directivity balloon will not exist.


Import Impulse Response
Meridians / Horizontal Resolution
Parallels / Vertical Resolution
Refresh List
Select Path
Import
Cancel

File Format : MF Frequency Data V[mmm]H[ppp].spk

Import Path : -

Resolution : 5°

Symmetry :

● None (Full Sphere)

○ Horizontal

○ Vertical

○ Quarter (90°)

○ Axial

Resolution : 5°

☑ Front Half Only

Order of Measurements

Start From (0° Position) :

● Left ○ Top

○ Right ○ Bottom

Options

☑ Use Redundant On-Axis Measurements for Re-Normalization (Recommended)

Number of Files to Import : -

Number of Files missing : -

Figure 2-202 – Import Impulse Response dialog box

2.15.2 Transformers

When measuring a loudspeaker that is used with a transformer, it is typically best to measure only the loudspeaker and bypass the transformer. If this is not possible it is recommended to measure the loudspeaker system with the transformer at its highest power tap setting. This should be the lowest impedance tap setting and the closest to not having the transformer in the signal path.

When loading the sensitivity measurement into the GSS for a loudspeaker measured with a transformer, the level of the measurement will have to be referenced to the voltage across the output terminals of the transformer, not the input to the transformer. This is required to make sure the overall level (SPL) for the sensitivity is based on only the loudspeaker and not the transformer.

Specifying the parameters to describe a transformer in a GLL is relatively straight forward. There are just a few items that need to be known and entered. The first item is the network voltage for the transformer. Typically this is 100 V, 70.7 V, 50 V, or 25 V. There are other possibilities for the network voltage, but these are the most common. The second item is the maximum power rating for the transformer. This will be the highest power tap setting for a given network voltage on which the transformer is used. Of course the other tap settings need to be known, too. The final item that must be known is the loudspeaker’s rated impedance for the transformer.

Neither ***SpeakerLab***, the GLL, ***EASE***, nor any other AFMG program performs any calculations based on power. All of the calculations are based on voltage, and for very good reasons. However, since transformer taps are most often given in terms of power, it is convenient to specify them this way in the GLL. By knowing the rated load impedance for the power taps of a transformer the voltage that will appear across the output of the transformer can be calculated. It is this voltage that is ultimately used for all subsequent calculations calling on the GLL.

Let’s look at how we would actually enter the parameters for a transformer into an XGLL project. If you have an XGLL project open it can be used. If not, please create a new project. We will enter the parameter for a 70.7 V transformer that has 5 taps at 32 W, 16 W, 8 W, 4 W, and 2 W into an 8 ohm load.

Select the ACOUSTIC DATA > TRANSFORMER property in the property tree. Click on the NEW button to add a transformer. Expand the TRANSFORMER property and select the XFMR1 property in the property tree. The General tab on the Edit Transformer page should now be displayed on the right.

The INTERNAL NAME for the transformer must be unique for each transformer in the XGLL project. This is generally taken care of automatically as ***SpeakerLab*** will increment the number at the end of the internal name each time a new transformer is added.

The DISPLAY NAME for the transformer can be anything. It is displayed as part of the tap selection in certain programs, such as ***EASE Evac***. It is recommended to keep the name relatively short. A good choice for the display name is the network voltage for the transformer. In other words, if we are defining a 70.7 V transformer a good display name would be “70 V".

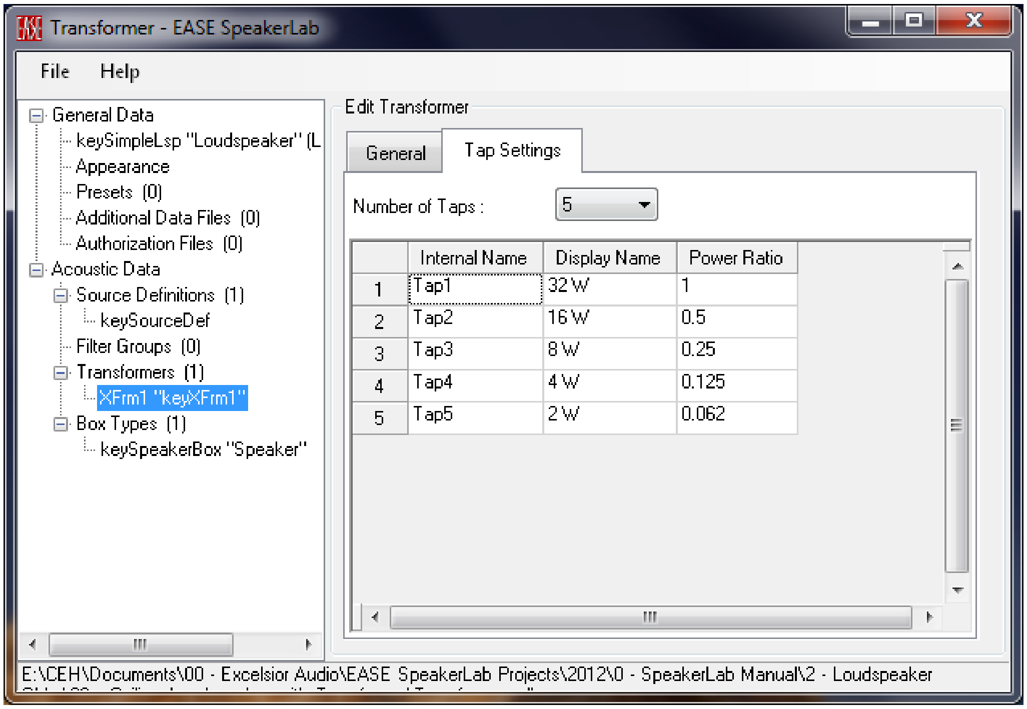
The POWER RATING area is where the MAXIMUM POWER is entered. This is the maximum power for the transformer, not the loudspeaker driver with which it is used. For the transformer we are specifying we should enter “32”. This is not a choice in the dropdown box but it can be typed in directly. The NETWORK

Voltage should be specified at "70.7". Since this 32 W maximum power is valid only for an 8 ohm load we will enter "8" for the Loudspeaker Impedance.

We must now define all of the tap settings for this transformer. On the Tap Setting tab use the Number of Taps dropdown to select 5, since we have 5 taps on our transformer. The transformer tap table will then be populated with 5 rows. The Internal Name column is used to specify a unique name for each tap for this particular transformer. Typically these do not need to be altered as ***SpeakerLab*** should keep the names unique automatically. The Display Name column is used to enter the name to be shown for each tap. It is recommended to use the same name as the tap on the real transformer. For some transformers this will be the power for the tap setting. For others the taps may be named A, B, C, D, and E. It really doesn't matter as long as the end user can correlate the tap setting in the GLL with the tap setting on the real loudspeaker system.

The Power Ratio column is used to specify the power for each tap. This is the ratio of the power for the specified tap to the maximum power tap. For our transformer the power ratios are "1", "0.5", "0.25", "0.125", and "0.0625". The power ratios cannot exceed 1.

Once all the entries have been made for the tap settings it should look something like Figure 2-203. Note that the entries do not have to be in any order. They can be ascending, descending, or scrambled. It is best to use some logic when ordering the entries. Perhaps it should be similar to the layout order on the actual device.



Number of Taps : 5

	Internal Name	Display Name	Power Ratio
1	Tap1	32 W	1
2	Tap2	16 W	0.5
3	Tap3	8 W	0.25
4	Tap4	4 W	0.125
5	Tap5	2 W	0.062

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\2 - Loudspeaker

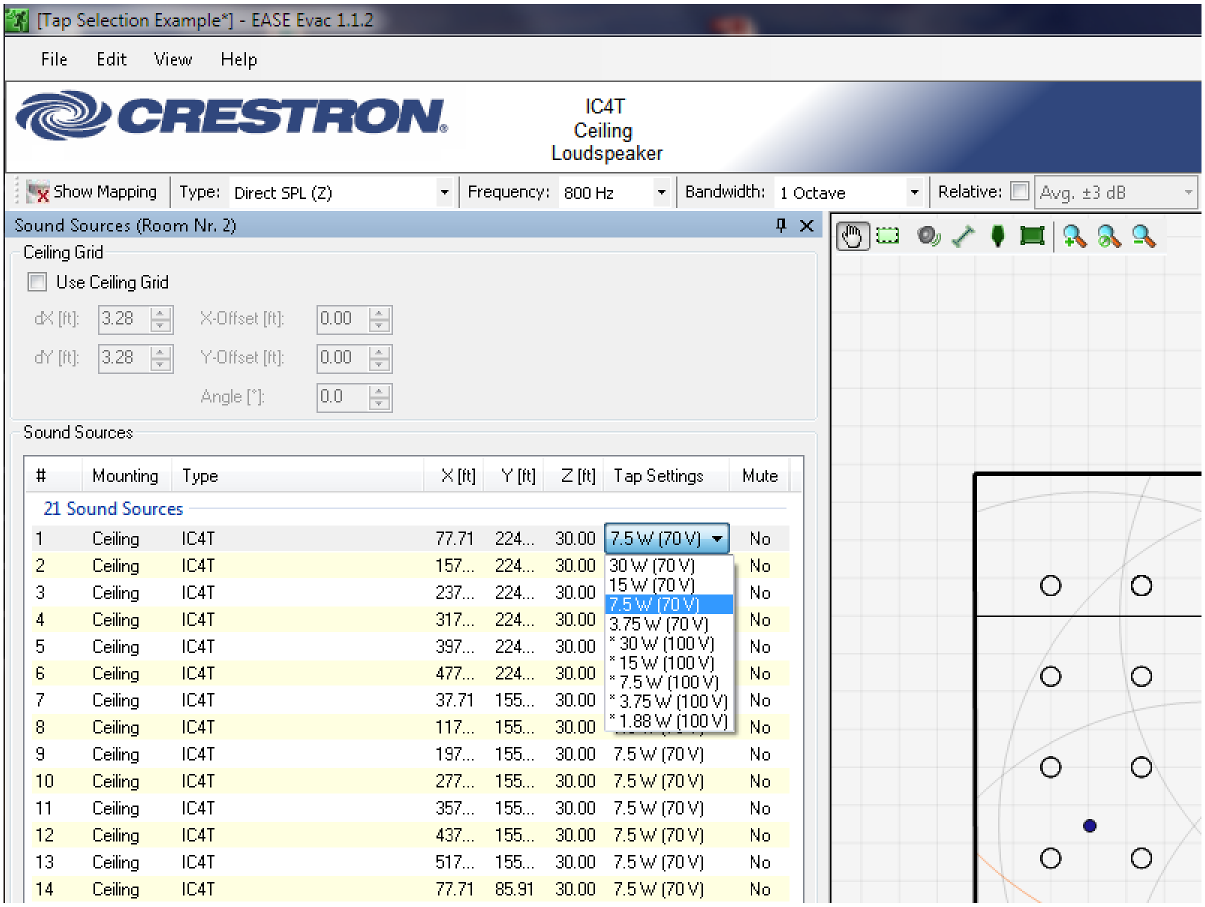
Figure 2-203 – Edit Transformer page showing 5 tap being specified

It is quite common in the real world for a single transformer to have windings for more than one network voltage on the same core. In SpeakerLab we must add a new transformer for each different network voltage that a real transformer may have.

The insertion loss of transformers is not currently handled in SpeakerLab, nor is the frequency response of the transformer itself.

We should note that currently only ***EASE Address*** and ***EASE Evac*** utilize the transformer functionality of the GLL. It is not implemented in the current release version of ***EASE*** but will be part of a future update.

It is also not possible to access the transformer tap when the GLL is run in the ***GLL Viewer***.


[Tap Selection Example*] - EASE Evac 1.1.2
File
Edit
View
Help
CRESTRON
Show Mapping
Sound Sources (Room Nr. 2)

IC4T
Ceiling
Loudspeaker

Type: Direct SPL (Z)
Frequency: 800 Hz
Bandwidth: 1 Octave
Relative: □ Avg. ±3 dB

Ceiling Grid

□ Use Ceiling Grid

dX [ft]: 3.28 X-Offset [ft]: 0.00
dY [ft]: 3.28 Y-Offset [ft]: 0.00
Angle [°]: 0.0

Sound Sources

21 Sound Sources

#	Mounting	Type	X [ft]	Y [ft]	Z [ft]	Tap Settings	Mute
1	Ceiling	IC4T	77.71	224...	30.00	7.5 W (70 V)	No
2	Ceiling	IC4T	157...	224...	30.00		No
3	Ceiling	IC4T	237...	224...	30.00		No
4	Ceiling	IC4T	317...	224...	30.00		No
5	Ceiling	IC4T	397...	224...	30.00		No
6	Ceiling	IC4T	477...	224...	30.00		No
7	Ceiling	IC4T	37.71	155...	30.00		No
8	Ceiling	IC4T	117...	155...	30.00		No
9	Ceiling	IC4T	197...	155...	30.00	7.5 W (70 V)	No
10	Ceiling	IC4T	277...	155...	30.00	7.5 W (70 V)	No
11	Ceiling	IC4T	357...	155...	30.00	7.5 W (70 V)	No
12	Ceiling	IC4T	437...	155...	30.00	7.5 W (70 V)	No
13	Ceiling	IC4T	517...	155...	30.00	7.5 W (70 V)	No
14	Ceiling	IC4T	77.71	85.91	30.00	7.5 W (70 V)	No

30 W (70 V)
15 W (70 V)
7.5 W (70 V)
3.75 W (70 V)
* 30 W (100 V)
* 15 W (100 V)
* 7.5 W (100 V)
* 3.75 W (100 V)
* 1.88 W (100 V)

Figure 2-204 – EASE Evac showing transformer tap selection

2.15.3 Rated Coverage Angle

It is particularly important for a GLL that is intended to be used with either ***EASE Address*** or ***EASE Evac*** to have the RATED COVERAGE ANGLE specified. Leaving this blank could lead to problems when attempting to use a GLL in these programs. For the recommended method of calculating the RATED COVERAGE ANGLE for ceiling and wall mounted loudspeakers please see Section 2.6.1.2.

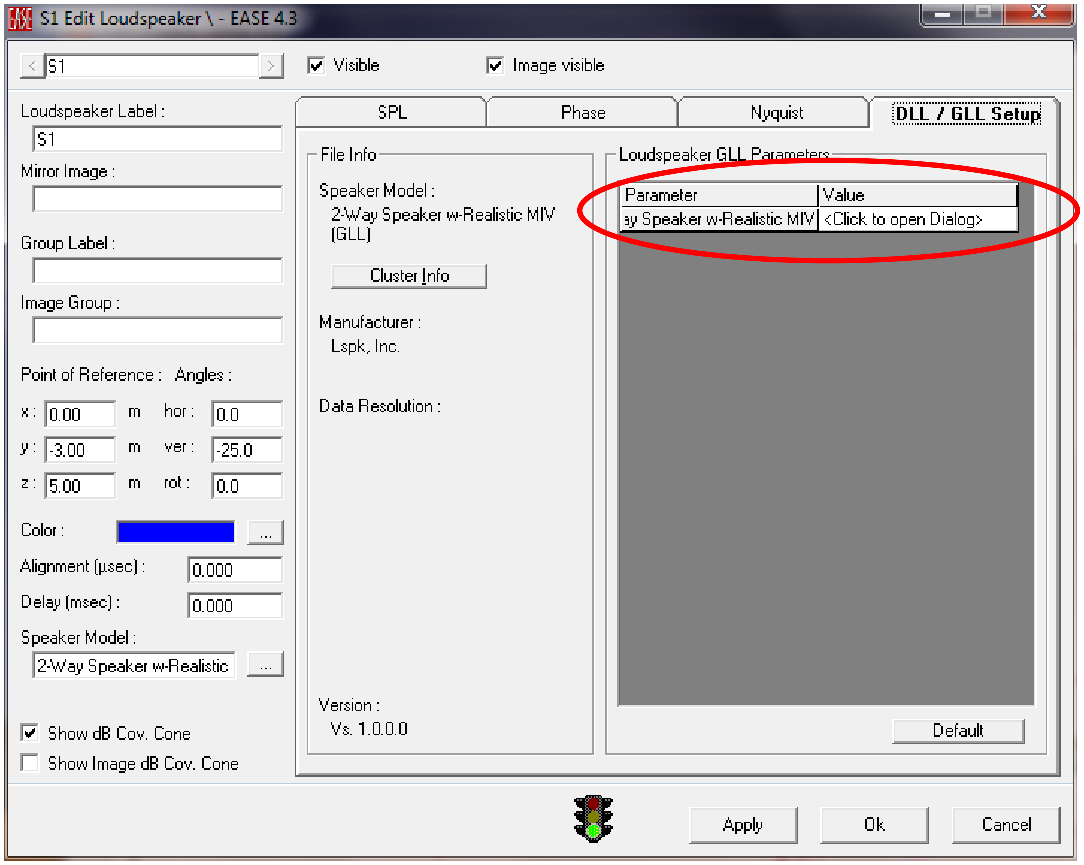
2.16 Application in EASE

Inserting a GLL data file for a loudspeaker in ***EASE*** is no different than inserting an older SPK data file. However, once the loudspeaker is inserted in the model, using the GLL in ***EASE*** is a little bit different than the older SPK data files. The GLL file format allows for many more options that can be configured. The way the GLL is driven (or "powered") is a little bit different as well.

One item of which users should be aware is that the case drawing for GLL files used in ***EASE*** will not be visible unless Room Symmetry is turned off for the EASE model. In addition to this the display of edges in the ***EASE*** model must be switched on.

2.16.1 Configuring the GLL

Once the GLL has been selected and inserted into the ***EASE*** model the DLL / GLL SETUP tab in the EDIT LOUDSPEAKER PROPERTIES dialog box can be selected. On the right side of this tab there is a LOUDSPEAKER GLL PARAMETERS area (Figure 2-205). There is a <CLICK TO OPEN DIALOG> cell in this area. Clicking this cell will open the GLL CONFIGURATION dialog box.



S1 Edit Loudspeaker \ - EASE 4.3

S1 ☑ Visible ☑ Image visible

Loudspeaker Label : S1

Mirror Image :

Group Label :

Image Group :

Point of Reference : Angles :

x : 0.00 m hor : 0.0

y : -3.00 m ver : -25.0

z : 5.00 m rot : 0.0

Color : 🟦

Alignment (µsec) : 0.000

Delay (msec) : 0.000

Speaker Model : 2-Way Speaker w-Realistic

☑ Show dB Cov. Cone

☐ Show Image dB Cov. Cone

File Info

Speaker Model :
2-Way Speaker w-Realistic MIV (GLL)

Manufacturer :
Lspk, Inc.

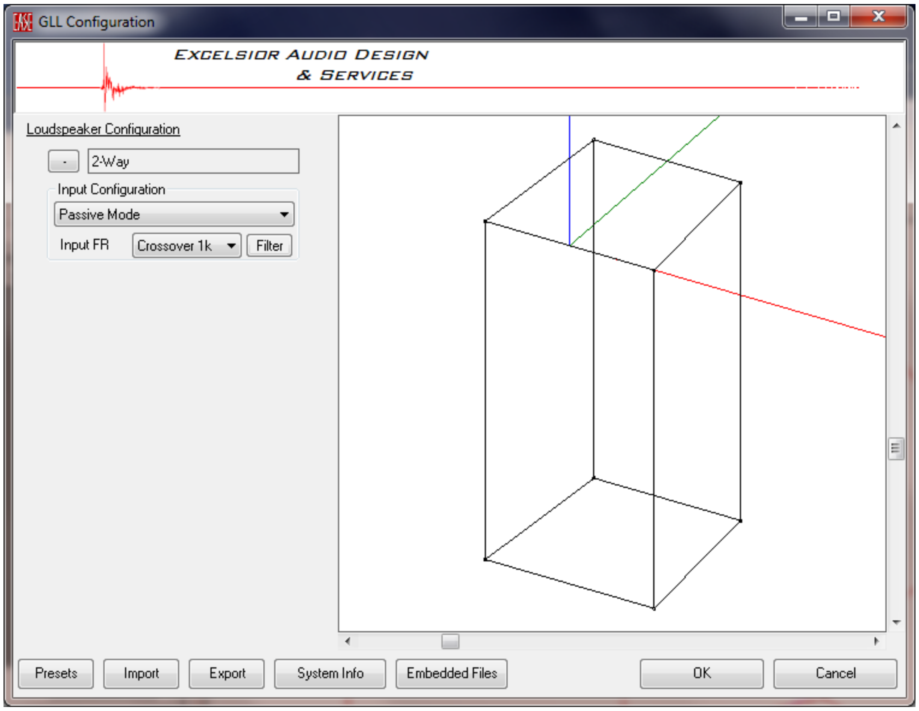
Data Resolution :

Version :
Vs. 1.0.0.0

Loudspeaker GLL Parameters

Parameter	Value
ay Speaker w-Realistic MIV	<Click to open Dialog>

Figure 2-205 – EASE Edit Loudspeaker dialog box showing the edit GLL configuration button



2-Way

Input Configuration: Passive Mode

Input FR: Crossover 1k | Filter

Figure 2-206 – Edit GLL Configuration dialog box in EASE (similar to the GLL Viewer)

The GLL Configuration dialog box (Figure 2-206) is almost identical to the View GLL dialog box we used earlier when running the GLL from inside ***SpeakerLab***. The only difference is that the GLL Configuration dialog box does not have the menu selections at the top. Instead it has a row of buttons at the bottom. The 5 buttons on the left, detailed below, are used for setting up the GLL configuration or getting additional information.

The Presets button opens the Select Presets dialog box. This is used to select one of the presets included in the GLL. If no presets are included in the GLL a message stating such will be displayed.

The Import button functions exactly like the Open Configuration command in the View GLL window. It allows a configuration file that is compatible with the GLL to be opened. This will configure the GLL. If the XGLC configuration file is not compatible with the GLL, meaning that it is not designed for this particular GLL or is formatted incorrectly, an error will be displayed.

The Export button functions exactly like the Save Configuration As command in the View GLL window. It allows a configuration file for this particular GLL to be saved.

The System Info button opens the GLL System Information dialog box. This displays information about the GLL such as the manufacturer of the loudspeaker system, the version/revision number of the GLL, whether or not the GLL is authorized by the manufacturer, and possibly some additional information about the GLL the manufacturer may have decided to include. This dialog box can be resized (enlarged) by grabbing a corner or edge with the mouse and pulling it.

The EMBEDDED FILES button opens a SELECT EMBEDDED FILE dialog box, if any embedded files are included with the GLL. If not, a message stating such is displayed. The author of the GLL may have included a specification sheet or instruction documents, drawings, measurement data, crossover transfer function measurements, etc. for the loudspeaker system embedded in the GLL. These files can be selected, extracted from the GLL, and saved to a folder where they can be opened by the appropriate application.

The two buttons on the right are to accept the configuration (OK) or cancel the changes (CANCEL). Both of these buttons close the GLL CONFIGURATION and return to the EDIT LOUDSPEAKER PROPERTIES dialog box in ***EASE***.

The Loudspeaker Configuration can be expanded, just as in the VIEW GLL dialog box, to change the input configuration (if more than one exists), to change the internal filtering for an input (if more than one exists), and to add external filters to each input.

2.16.2 Driving (or "Powering") the GLL

Once the GLL is configured in ***EASE*** it will be driven to its maximum input voltage allowed as described in Section 2.12.7. In EASE the spectrum for the input signal used to drive the GLL is AES2 pink noise (band limited from 20 Hz to 20 kHz). If a spectrum other than pink noise is desired an external filter applied to the input(s) of the GLL can be used to yield the desired spectrum. Alternatively, this can perhaps be accomplished more easily on the SPL tab of the EDIT LOUDSPEAKER PROPERTIES dialog box.

On this tab there is a column labeled SPL (1M). When setting the properties for a GLL, this column will not display the SPL at the specific 1/3 octave frequency band. Instead the SPL (1M) column is used to set the drive level to the GLL. The Save File and Load File buttons can be useful to save and recall settings that are used often. If the use of IEC 60268-1 noise is desired in ***EASE*** the levels in Table 2-3 can be used for their corresponding frequencies.

Table 2-3

Frequency (Hz)	IEC 60268-1 Spectrum (dB)
100	-0.9
125	-0.5
160	-0.2
200	-0.1
250	0.0
315	0.0
400	0.0
500	0.0
630	0.0
800	0.0
1,000	-0.1
1,250	-0.3
1,600	-0.6
2,000	-1.0
2,500	-1.6
3,150	-2.5
4,000	-3.7
5,000	-5.1
6,300	-7.0
8,000	-9.4
10,000	-11.9

The levels entered in this SPL (1M) column also serve another very useful purpose. This is the only place in ***EASE*** where it is possible to turn down the level of a GLL-based loudspeaker. The default levels in the SPL (1M) column are 0 dB. This means the drive level to this GLL will be at 0 dB relative to its maximum input voltage. If the level of this GLL needs to be decreased the −1 DB and −10 DB buttons in the LEVEL area just to the right of the SPL (1M) column can be used to quickly and easily change the level. The +1 DB and +10 DB buttons can be used to increase the level. Note, however, that the level cannot exceed 0 dB. It should also be mentioned that any filters used within the GLL, front-end EQ for example, are relative within the GLL. A broadband gain or attenuation by a filter within the GLL will have no effect on the SPL within EASE.

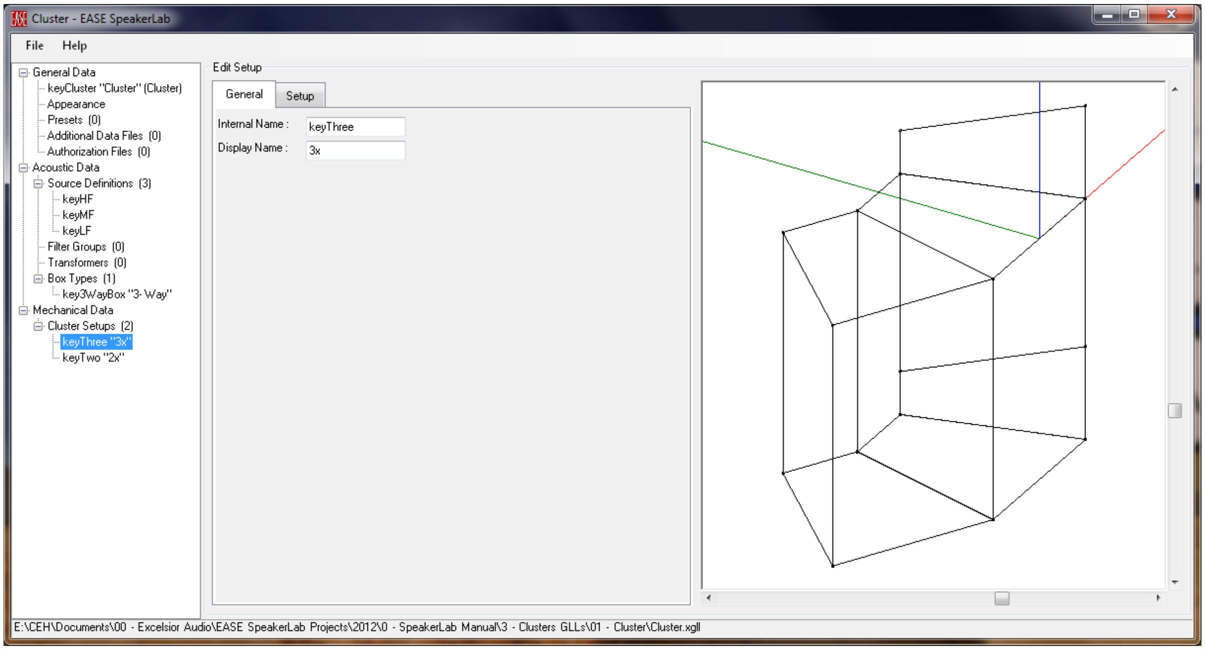
3 Cluster-Type GLLs

3.1 General Overview

The cluster-type GLL differs from the loudspeaker-type GLL in that multiple loudspeaker boxes can be configured within the GLL to make a cluster. This can be multiple instances of the same loudspeaker or it can be one or more instances of multiple different loudspeakers. All of the sound sources used in all of the loudspeaker boxes within the cluster are treated as coherent sources when the GLL calculations are carried out.

Because multiple boxes can be positioned relative to each other the cluster-type XGLL project file has an additional branch in the property tree. This is the MECHANICAL DATA property. Selecting the CLUSTER SETUPS property, under MECHANICAL DATA, displays the Edit Cluster Setups page on the right. This is used to add or delete different cluster setups. Each of the cluster setups that are added are shown below the CLUSTER SETUPS property in the property tree. The default cluster GLL (using the CLUSTER tab of the CREATE NEW PROJECTS dialog box) has two of these. One for a three box cluster and one for a two box cluster.

Selecting one of the cluster setup properties, KEYTHREE for example, displays the EDIT SETUP page on the right (Figure 3-1). There are two tabs on this page, GENERAL and SETUP. The current placement of the loudspeakers used in the setup is shown in the display area at the right side of the window. The GENERAL tab also has text boxes to specify an INTERNAL NAME and a DISPLAY NAME for the setup. The INTERNAL NAME must be unique from all of the other internal names for the cluster setups in this GLL. The DISPLAY NAME can be whatever is desired. This is what users of the GLL will see when selecting the setup.



Internal Name : keyThree

Display Name : 3x

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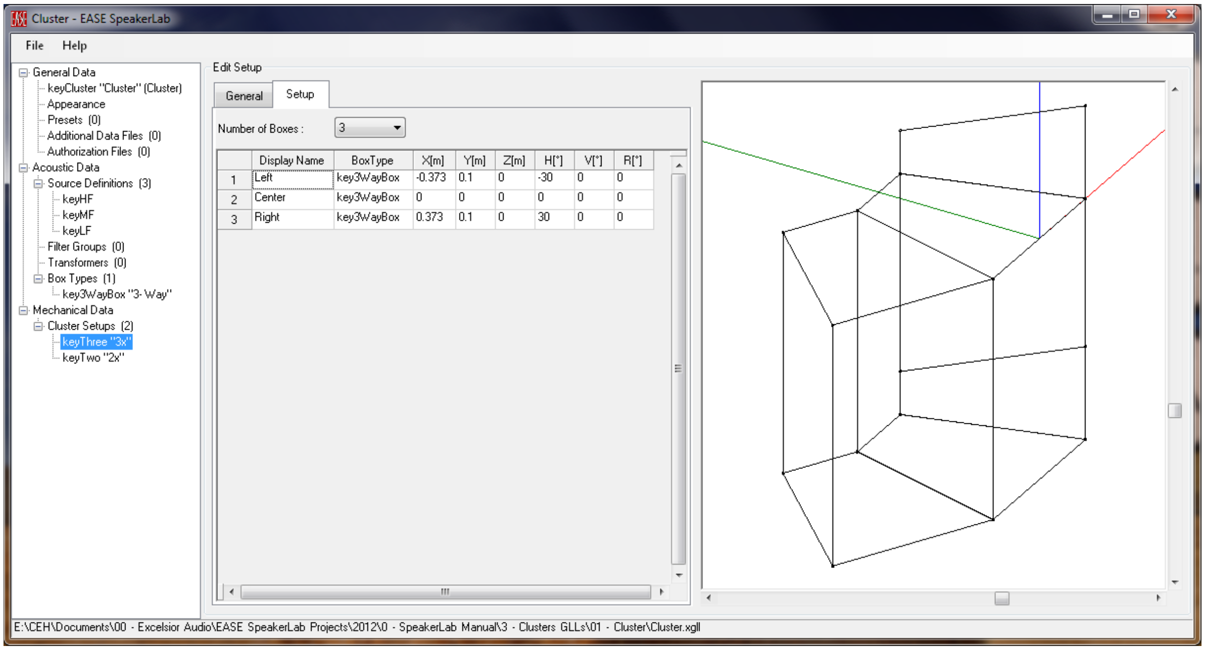
Figure 3-1 – Edit Setup page, General tab for a cluster-type GLL



Number of Boxes : 3

	Display Name	BoxType	X[m]	Y[m]	Z[m]	H[°]	V[°]	R[°]
1	Left	key3WayBox	-0.373	0.1	0	-30	0	0
2	Center	key3WayBox	0	0	0	0	0	0
3	Right	key3WayBox	0.373	0.1	0	30	0	0

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\3 - Clusters GLLs\01 - Cluster\Cluster.xgll

Figure 3-2 – Edit Setup page, Setup tab showing the number of boxes selected and their locations specified

The Setup tab (Figure 3-2) has a NUMBER OF BOXES dropdown box used to specify how many boxes (loudspeakers) are used in the setup. Each box has its own row in the table. The table has several different columns to fully specify the cluster.

The DISPLAY NAME column is used to identify a particular loudspeaker location to the user. This can be any name desired. However, the space for the text is somewhat limited when the GLL is run so it is a good idea to keep the name relatively short.

The BOXTYPE column is used to select the box type (loudspeaker) used in the specified location. Clicking a cell in this column will open the SELECT A BOX TYPE dialog box. This will allow a box type to be selected from those currently defined in the XGLL project.

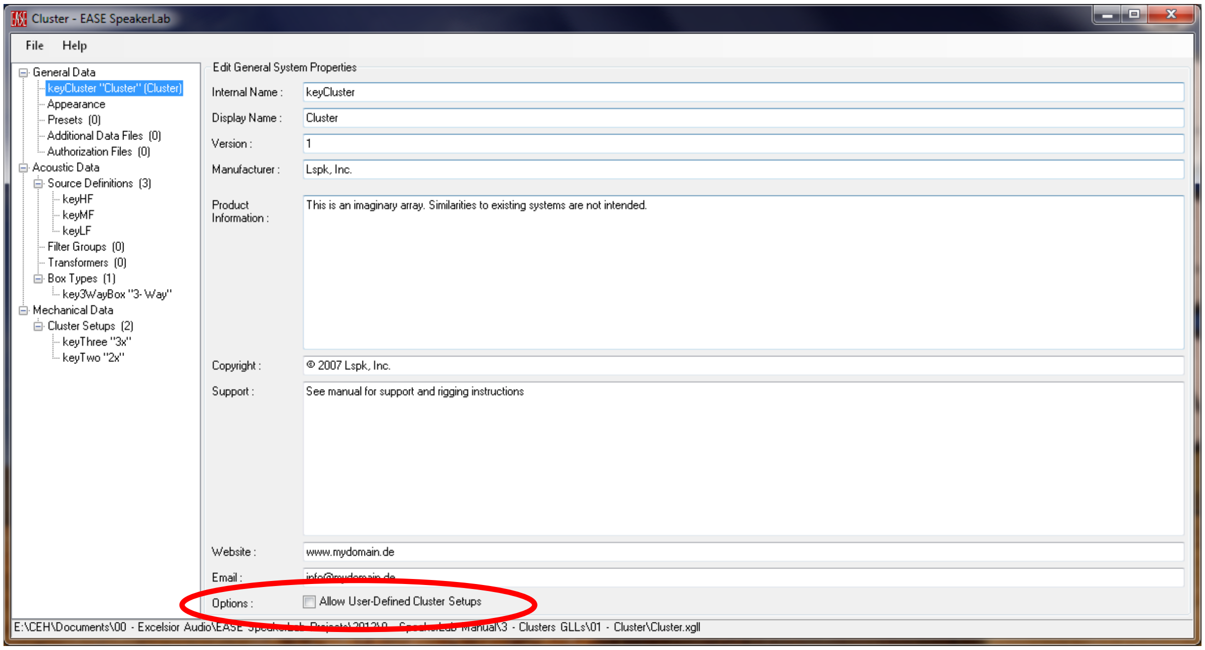
The X[M], Y[M], and Z[M] columns are used to specify the x, y, and z coordinates for the location of the loudspeaker in the cluster. The origin (0,0,0) point of the loudspeaker will be placed at the x, y, z, coordinates entered.

The H[°] column is used to specify the horizontal aiming angle (azimuth or yaw) of the loudspeaker.

The V[°] column is used to specify the vertical aiming angle (elevation or pitch) of the loudspeaker.

The R[°] column is used to specify the rotation angle (roll) of the loudspeaker.

It is possible for the user to define cluster setups if this is enabled by the author of the GLL. This can be set on the EDIT GENERAL PROPERTIES page (GENERAL DATA > KEYCLUSTER). At the bottom of this page (Figure 3-3) there is an ALLOW USER–DEFINED CLUSTER SETUPS checkbox. If this is checked the user will be able to define cluster setups. This can be very useful for allowing custom clusters to be built, limited only by the imagination of the end users. There may be some instances where a manufacturer may want to intentionally limit the clusters to only those defined in the GLL. Unchecking this checkbox is a way to disallow users from building custom clusters.



Edit General System Properties

Internal Name : keyCluster

Display Name : Cluster

Version : 1

Manufacturer : Lspk, Inc.

Product Information : This is an imaginary array. Similarities to existing systems are not intended.

Copyright : © 2007 Lspk, Inc.

Support : See manual for support and rigging instructions

Website : www.mydomain.de

Email : info@mydomain.de

Options : □ Allow User-Defined Cluster Setups

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\3 - Clusters GLLs\01 - Cluster\Cluster.xgll

Figure 3-3 – Edit General System Properties page showing the Allow User-Defined Cluster Setups checkbox

3.2 Adding Another Box Type

In order to use different loudspeaker boxes in a cluster setup these different loudspeaker boxes must be defined within the XGLL project. It is relatively straight forward to add more box types to the project. By selecting ACOUSTIC DATA > BOX TYPES the EDIT BOX TYPES page is shown in the right. Select the row in the table corresponding to the loudspeaker box that should be duplicated. There may only be one box currently in the table so select it. Clicking the Duplicate button will add another loudspeaker box type.

The newly added box type can be selected in the property tree and its properties will be displayed in the Edit Box Type page on the right. Refer to Section 2.6.1 – Box Types for details on these individual properties used to specify the new loudspeaker system.

3.3 Cluster Setups

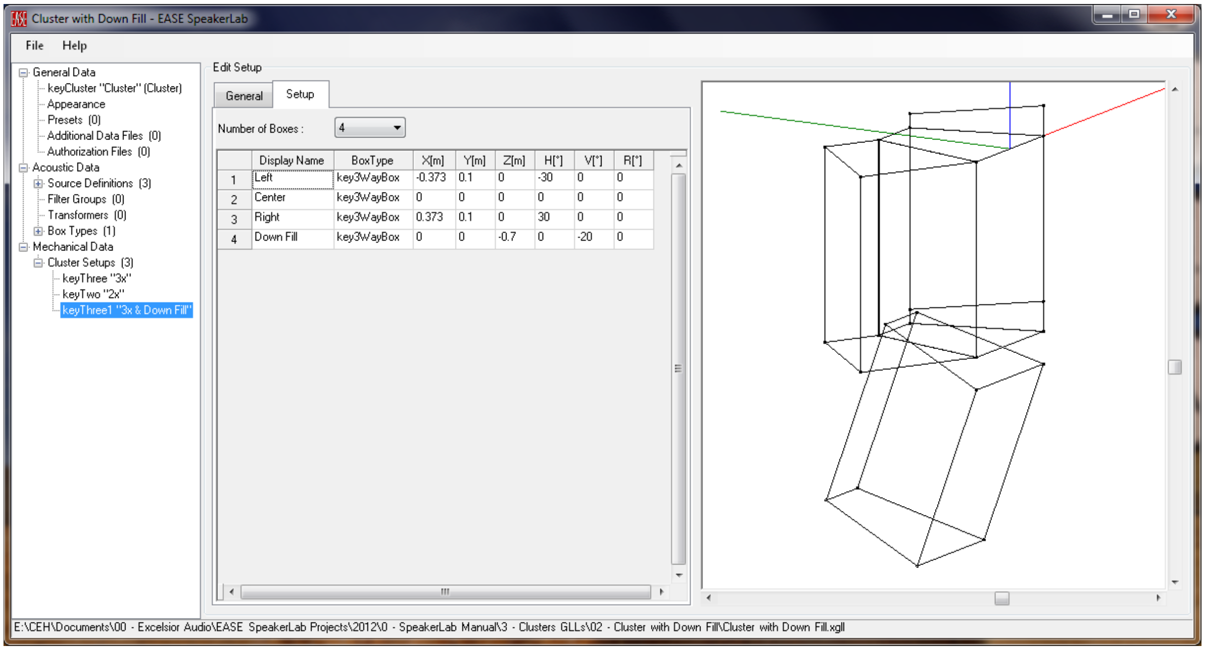
Let's add a new cluster setup to the default cluster XGLL project. We can see that the current setups use a 2-wide and a 3-wide cluster. We will add a 3-wide cluster with a down fill loudspeaker. For this example we will use the same loudspeaker box for the down fill as for the 3-wide array. While this would not be advisable in the real world, it will help to show how to configure a cluster setup. We would like the new down fill box to be centered in the cluster, directly below the center box, and aiming down about 20° compared to the top row of boxes.

If you have not already created a new cluster project, please do so. Select MECHANICAL DATA > CLUSTER SETUPS. In the EDIT CLUSTER SETUPS page on the right select the top row of the table (keyThree) and click the DUPLICATE button. This will add a third entry to the table. In the property tree on the left select this newly added setup. On the GENERAL tab of the EDIT SETUP page on the right change the DISPLAY NAME to "3x & Down Fill". This is the name that will be displayed for the user to choose between the available cluster setups.

We now need to add the down fill loudspeaker to the cluster. Click the SETUP tab and change the NUMBER OF BOXES from 3 to 4. This will duplicate the last row of the table to the newly added row. Select the DISPLAY NAME in row 4 and change this to "Down Fill".

Now that we have added the box to the cluster we need to properly position it where we want it to be. Since the down fill is to be centered directly below the center box in the top row, the x and y coordinates of the down fill box should be the same as the center box (0,0). To get the z coordinate we would need to know the height of the other boxes, which is 0.6 m. Let's enter "-0.7 m" for the z coordinate of the down fill box. Hopefully this will allow the clearance we need for our down tilt aiming angle.

We should now see the down fill box below the top row of boxes. However, it's aimed to one side and not straight ahead. The horizontal aiming angle, [H°], needs to be changed from 30 to "0". We also need to change the vertical aiming angle, [V°], from 0 to "-20". Once we have done this we should see that there is interference between the down fill box and the center box in the top row (Figure 3-4). We need to lower the down fill box a bit. Changing the z coordinate form -0.7 to "-0.75" should eliminate this problem.



Number of Boxes : 4

	Display Name	BoxType	X[m]	Y[m]	Z[m]	H[°]	V[°]	R[°]
1	Left	key3WayBox	-0.373	0.1	0	-30	0	0
2	Center	key3WayBox	0	0	0	0	0	0
3	Right	key3WayBox	0.373	0.1	0	30	0	0
4	Down Fill	key3WayBox	0	0	-0.7	0	-20	0

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2012\0 - SpeakerLab Manual\3 - Clusters GLLs\02 - Cluster with Down Fill\Cluster with Down Fill.xgll

Figure 3-4 – Setup tab on Edit Setup page showing the number of boxes selected and their locations specified. Notice that the down fill needs to be lowered still.

Note that ***SpeakerLab*** will not give a warning for cabinet overlap resulting in this type of physical interference between boxes. The author of the GLL must explicitly check to make sure the cluster setup can physically be built in reality.

3.4 Examples

3.4.1 Cluster Switchable Between 2-Wide, 3-Wide, and 3-Wide with Down Fill

Let's now run the GLL we just modified in Section 3.3. We see that the default CLUSTER CONFIGURATION used in the GLL is the 3x, as this is the first one in the EDIT CLUSTER SETUPS table. Use the Setup dropdown box under Cluster Configurations to change the setup to 2x. The cluster immediately changes to the corresponding setup. Now try the 3x & Down Fill setup.

Note that each element (box) in the cluster can be expanded to show its individual INPUT CONFIGURATION. We did not change these in the default cluster XGLL project so each box only has an

Active 3-Way input configuration. This can be changed in the XGLL project as detailed in Sections 2.12 and 2.13.

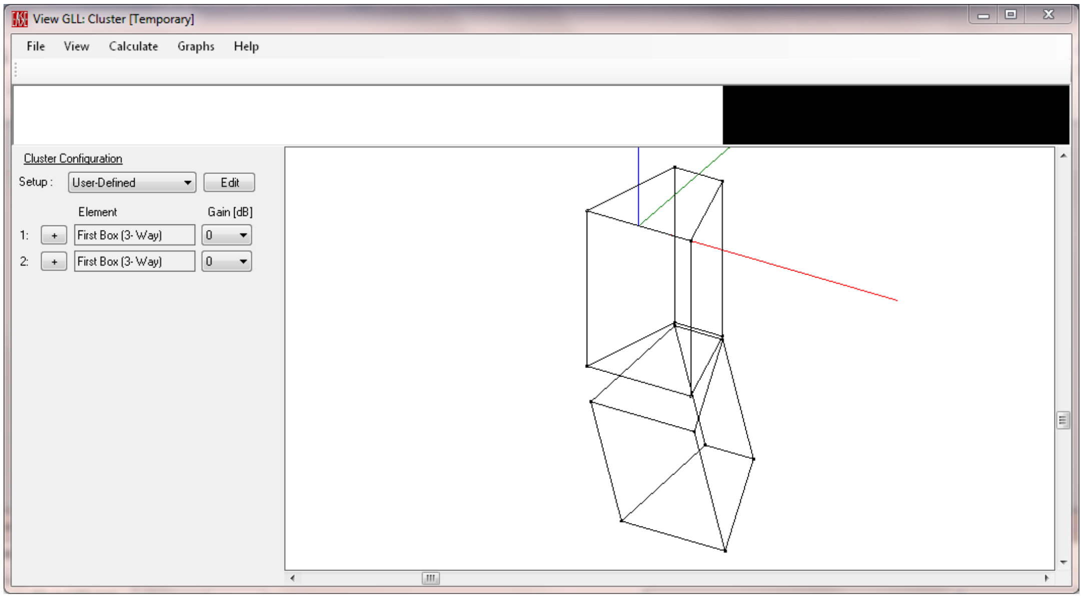
3.4.2 User Defined Cluster Setups

Note that in the Cluster Configuration Setup dropdown box there is no selection for User Defined. Let’s change this in the XGLL project and see how it affects the running GLL.

Close the View GLL window to return to ***SpeakerLab***. Select General Data > keyCluster to see the Edit General System Properties page on the right. At the very bottom of the page check the Allow User-Defined Cluster Setups checkbox. Save the XGLL project and then run the GLL.

We now see that the Cluster Configuration Setup includes the User-Defined option. When this is selected the cluster setup changes immediately to a single box located at the coordinate system origin. The Edit button to the right of the Setup dropdown is now active. Click this button to define or edit the cluster setup using the Define User-Defined Cluster dialog box. Upon the initial display of this dialog box it will probably be too small to adequately display all of the parameters in the Edit Setup table. Grab an edge or corner of the box with the mouse and drag it to make it larger.

We can use this setup table exactly like the one in SpeakerLab. We’ll add a second box and position it below the first one to create a typical 2 box, over/under cluster (Figure 3-5).



Cluster Configuration

Setup : User-Defined

		Element	Gain [dB]
1:	+	First Box (3-Way)	0
2:	+	First Box (3-Way)	0

Figure 3-5 – Running GLL showing a user-defined cluster that was specified manually

This is a fairly simple cluster and easy to define for its setup. However, larger clusters with many boxes may take considerable time to setup. If larger clusters are defined with the User-Defined setup it may be a good idea to save the setup as a configuration file. This way it can be recalled later with no additional effort.

3.5 Cluster Setups vs. Presets and Configuration Files

Both authors and users of GLLs need to be keenly aware of the difference between cluster setups and presets. The difference is simple but important. As we have seen the cluster setup defines only the number and type of boxes and their location in the cluster. A preset, as well as a saved configuration file, contains all of the information of the setup, and additionally all of the filtering specified for the loudspeaker boxes used in the cluster. If a cluster has been designed with special signal processing (via external filters) for each of the boxes, or groups of boxes, it is much better to have these saved as a preset or a configuration file.

It is recommended that only loudspeakers that are designed to be arrayed together in a cluster or that have a high likelihood that end users will do so should be included in a cluster-type GLL. This keeps the GLL file size as small as possible while still facilitating the desired functionality when modeling the performance of the loudspeakers in a clustered array. Including an entire product catalog of loudspeakers in a cluster-type GLL is not recommended. This would results in excess time spent transferring data between the different program modules in ***EASE***, potentially causing lengthy delays for the end user.

4 Line Array-Type GLLs

4.1 General Overview

The line array-type GLL differs from the loudspeaker-type GLL in that multiple loudspeaker boxes can be configured within the GLL to make a vertical array of loudspeakers. This can be multiple instances of the same loudspeaker or it can be one or more instances of multiple different loudspeakers. All of the sound sources used in all of the loudspeaker boxes within the line array are treated as coherent sources when the GLL calculations are carried out.

Like the cluster-type GLL, the line array-type GLL also has a MECHANICAL DATA branch of the property tree. However, the branches below it are different than the cluster-type GLL. This is because the rigging systems used with line arrays are also typically very different from those used with clusters.

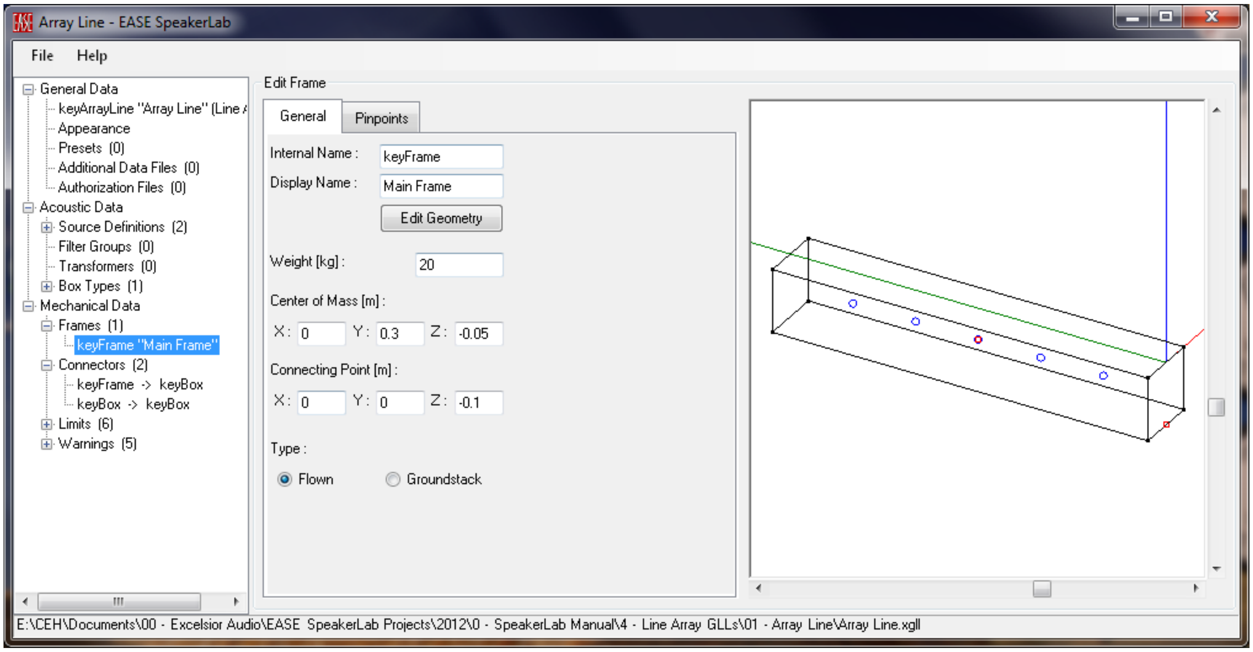
These mechanical data properties of Frames and Connectors allow the line array cabinet box types to be assembled just as they would be in the real world. Perhaps more importantly, they can disallow the joining of certain boxes to each other and/or certain boxes to certain frames. By strictly defining within the XGLL project only the connecting hardware that exists in the real world for the line array system, only arrays that actually can be built in the real world will be able to be assembled with the GLL.

4.2 Additional Items

4.2.1 Frames

The fly bars and fly frames, which are typically used to suspend line array systems, are modeled in ***SpeakerLab*** via the Frame element. A line array-type GLL has one frame element by default. Additional frames may be added, if needed, by using the Duplicate button on the Edit Frames page (Mechanical Data > Frames). If a previously added frame is no longer needed it can be selected in the table and deleted with the Delete button. There must always be at least one frame in a line array-type GLL.

A frame has several parameters that must be specified in order for it to function properly with the line array loudspeaker cabinet (box type). Most of these are found on the General tab of the Edit Frame page (Mechanical Data > Frames > keyFrame).



- General Data
  - keyArrayLine "Array Line" (Line A
  - Appearance
  - Presets (0)
  - Additional Data Files (0)
  - Authorization Files (0)
- Acoustic Data
  - Source Definitions (2)
  - Filter Groups (0)
  - Transformers (0)
  - Box Types (1)
- Mechanical Data
  - Frames (1)
    - keyFrame "Main Frame"
  - Connectors (2)
    - keyFrame -> keyBox
    - keyBox -> keyBox
  - Limits (6)
  - Warnings (5)

Internal Name : keyFrame
Display Name : Main Frame

Weight [kg] : 20

Center of Mass [m] :
X: 0 Y: 0.3 Z: -0.05

Connecting Point [m] :
X: 0 Y: 0 Z: -0.1

Type :
● Flown ○ Groundstack

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Figure 4-1 – General tab on the Edit Frame page for a line array-type GLL

Like all other items in the XGLL project file, a Frame has an Internal Name and a Display Name to differentiate multiple frames internally as well as for the user to select the desired one for building an array.

A geometry file (very similar to a case drawing for a loudspeaker cabinet) is used to be able to display the shape of the frame. It is for display purposes only and has no impact on the actual working of the GLL. Clicking on the Edit Geometry button will allow a geometry file (.XED or .FED) to be used for a frame. Please see Section 2.6.1.1 for more information on the details of these geometry files.

The mass or weight of a frame must be entered in the WEIGHT [KG] box, along with the location of the center of mass of a frame in the X, Y, and Z coordinate boxes under CENTER OF MASS [M] (in meters). This is required so that programs using the GLL can reasonably calculate an estimate for the center of mass for the entire array as well as the total weight. Without this data it is not possible to determine how the array will hang in the air.

Note: It should be mentioned that the GLL does not consider the weight of the cables used for an array. These can sometimes be a factor in determining the center of mass for an array. For this reason we say the calculations for the GLL are only a very good estimate. The overall weight and center of mass for an actual array can be slightly different.

The point on the frame where the loudspeaker cabinet attaches to the frame must also be specified in the X, Y, and Z coordinate boxes under CONNECTING POINT [M]. The x-coordinate of the connecting point is typically 0. Only a single point is defined and most fly systems have symmetry about the yz-plane (x=0).

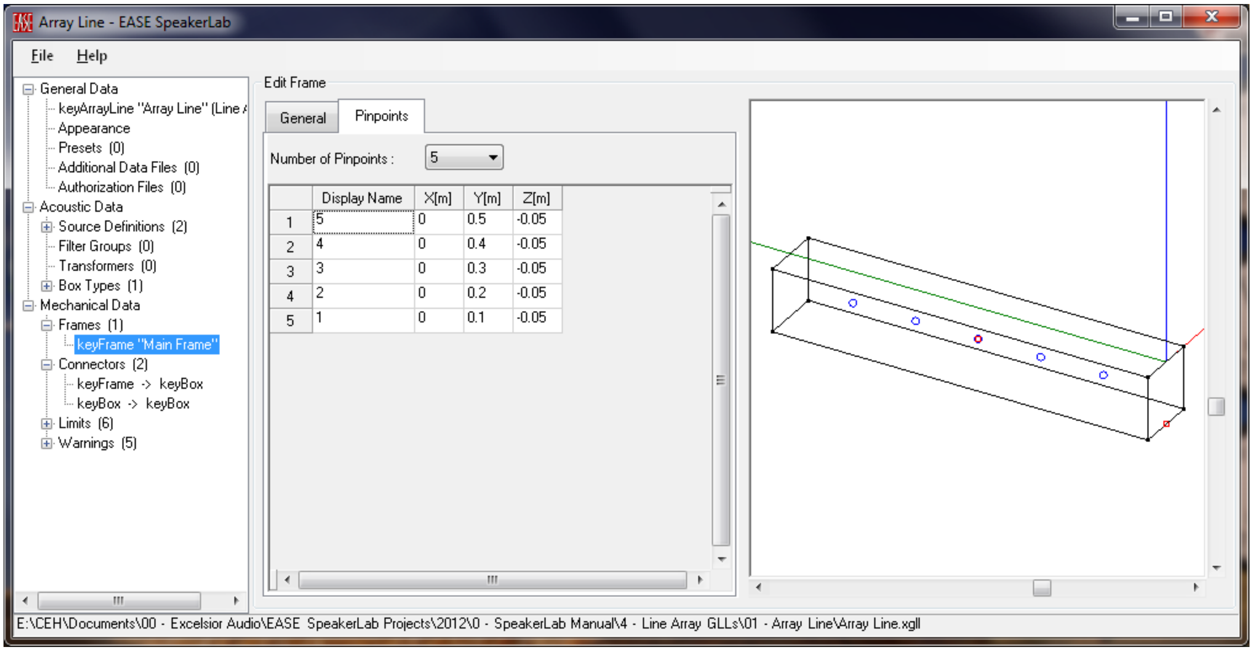
We should also note that if the loudspeaker is allowed to have multiple splay angles with respect to the frame the pivot point is at this connecting point. This also means that the pivot point is at the insertion point of the loudspeaker cabinet that is attached to the frame. This insertion point is the origin (0,0,0) of the coordinate system for the case drawing of the loudspeaker. This should be kept in mind when laying out the case drawing to define the cabinet (box type) used in the XGLL project.

The frame TYPE must also be selected as whether it is for FLOWN or GROUNDSTACK applications. Because of the way the GLL handles loudspeaker boxes we must specify whether a frame is to be used for flying an array or ground stacking it. It doesn't matter if the same frame can be used for both in the real world. We must specify two different frames inside the GLL, even if they are otherwise identical. We will discuss this further in Section 4.5 – Ground Stack Example.

The suspension points used to hang the fly frame in the air are specified on the PINPOINTS tab. The NUMBER OF PINPOINTS dropdown box is used to specify how many pin points are on the fly frame. Changing this number will alter the number of rows in the pin points table immediately below the dropdown box.

This table has four columns. The first is the DISPLAY NAME. The value of a cell in this column is what the user will see to select the various pin points. It is recommended that the labeling on the actual fly frame be used for the DISPLAY NAME.

The other three columns are for the x, y, and z coordinates (X[M], Y[M], Z[M]) of the pin points. Here again, the x-coordinate of the pin points are typically 0 as most fly systems have symmetry about the yz-plane (x=0).



Number of Pinpoints : 5

	Display Name	X[m]	Y[m]	Z[m]
1	5	0	0.5	-0.05
2	4	0	0.4	-0.05
3	3	0	0.3	-0.05
4	2	0	0.2	-0.05
5	1	0	0.1	-0.05

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Figure 4-2 – Pinpoints tab on the Edit Frame page for a line array-type GLL

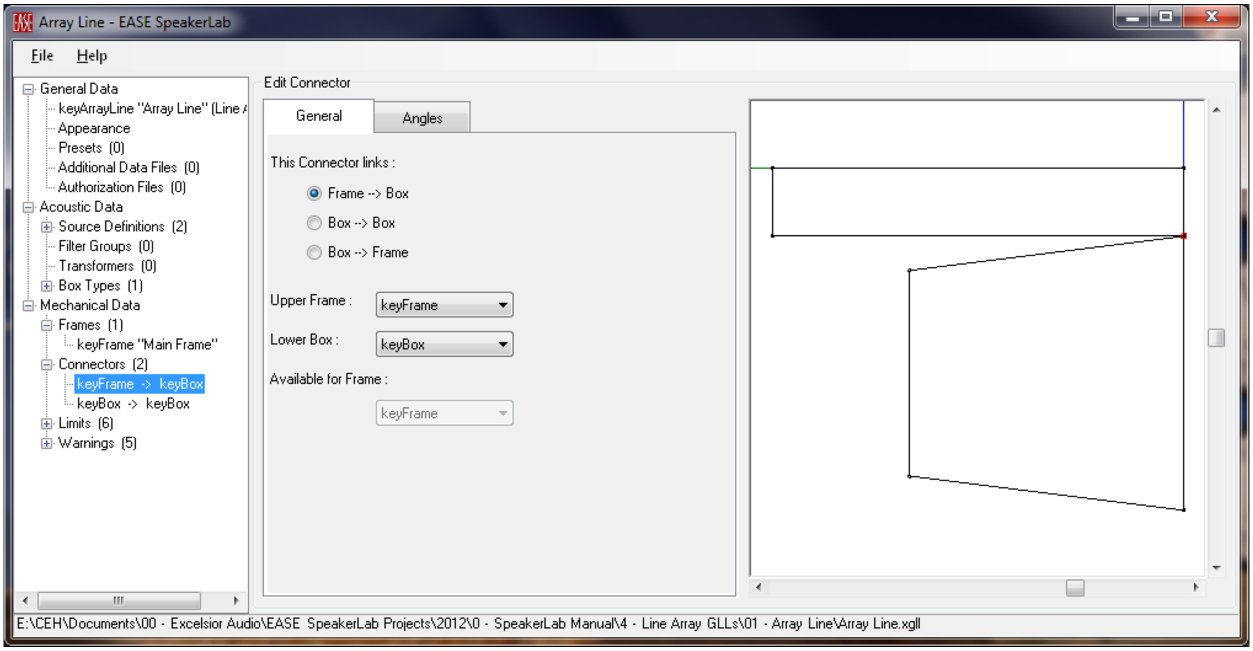
4.2.2 Connectors

A connector is the element used to join a box to a frame or a box to another box in an array. If a connector is not defined to join two boxes together they will not be allowed to be connected in the GLL. Additional connectors may be added, if needed, by using the Duplicate button on the Edit Connectors page (Mechanical Data > Connectors). If a previously added connector is no longer needed it can be selected in the table and deleted with the Delete button. There must always be at least two connectors in a line array-type GLL; one to join the box to the frame and the other to join two boxes together. Without these it is not possible to make an array.

The General tab on the Edit Connector page (Mechanical Data > Connectors > keyFrame or keyBox) is used to specify the box(es) and/or frame(s) that can be joined together by a particular connector. For the case of a flown frame a connector can be used to link a Frame to a Box (Frame --> Box) or a Box to a Box (Box --> Box). For the case of a ground stacked frame a connector can be used to link a Box to a Frame (Box --> Frame) or a Box to a Box (Box --> Box).

Two dropdown boxes allow the upper element (Upper Frame or Upper Box) and the lower element (Lower Box or Lower Frame) to be selected. Only frames or boxes already defined in the XGLL project will be available for selection. When the connector is used to connect two boxes together the bottom Available for Frame dropdown box is enabled. This can be used to specify if the connector is allowed to be used for all frames defined in the XGLL project or only one of the frames.

As these items are changed the display on the right of the page will be updated. This display shows the upper and lower item connected together just as they will be when using the GLL. The connection point is shown as a small red circle. This can be very useful when troubleshooting connections that don't seem to work as expected.



This Connector links :

● Frame --> Box
○ Box --> Box
○ Box --> Frame

Upper Frame : keyFrame

Lower Box : keyBox

Available for Frame : keyFrame

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Figure 4-3 – General tab on the Edit Connector page showing a frame to box connector

The ANGLES tab is used to specify the number of discrete splay angles allowed for the connector, as well as the splay angle itself for each selection. The POSSIBLE ANGLES dropdown determines the number of rows in the splay angles table. This table has two columns; DISPLAY NAME and ANGLE. The DISPLAY NAME is what the users see when making their selection. The ANGLE is the actual value of the splay angle between the two elements joined by the connector.

At the bottom of the page is a SHOW ANGLE dropdown box. This allows any of the entries in the table above to be selected. The display on the right of the page will change to show the select angle. This, too, can be useful for troubleshooting.

As an example, let's say we have a line array system with one frame and three different box types; a subwoofer, a long throw box, and a down fill box. The manufacturer allows either the subwoofer or the long throw box to be connected directly to the frame. Long throw boxes can be hung below a subwoofer. Down fill boxes can be hung below long-throw boxes, but not below a subwoofer. Subwoofers can be hung below each other. Long-throw boxes can be hung below each other. Down fill boxes can also be hung below each other. The following seven connectors will need to be defined to accomplish this.

Connector Links	Upper Item	Lower Item
Frame to Box	Frame	Subwoofer
Frame to Box	Frame	Long-Throw
Box to Box	Subwoofer	Subwoofer
Box to Box	Long-Throw	Long-Throw
Box to Box	Down Fill	Down Fill
Box to Box	Subwoofer	Long-Throw
Box to Box	Long-Throw	Down Fill

4.2.3 Limits

It is possible to define limits in the XGLL project. Some, but not all, of the limits defined in the XGLL are strictly enforced in AFMG software products. It is possible for users to exceed some of the non-enforced limits. When a particular limit it exceeded an error is typically shown in the software using the GLL.

An XGLL project is not required to have any limits. They may all be deleted with the Delete button on the EDIT LIMITS page (MECHANICAL DATA > LIMITS) if desired. If no limits exist clicking the NEW button will add one. If more are needed, select one of the limits in the table and click the DUPLICATE button.

There are six different types of limits that can be used in an XGLL project. Each is detailed below.

4.2.3.1 Maximum Number of Cabinets

This specifies the maximum number of cabinets that can be used in an array. It can be specified for all frames or one specific frame. Multiple instances of this limit can be present in an XGLL project, each applying to a different frame. This type of limit can be useful if, for example, a frame should not have more than 18 boxes suspended from it.

4.2.3.2 Maximum Number of Cabinet of This Type

This specifies the maximum number of cabinets of a particular box type that can be used in an array. It must be specified for a particular box type. It can be specified for all frames or one specific frame. Multiple instances of this limit can be present in an XGLL project, each applying to a different frame. This limit can be useful if, for example, both subwoofers and full-range boxes are specified in a GLL. Perhaps no more than 4 subs should be used on a frame.

4.2.3.3 Maximum Weight

This specifies the maximum weight of an array. This weight includes the weight of the frame and the cabinets. Since no weight is assigned to connectors they are not included in this maximum weight. It can be specified for all frames or one specific frame. Multiple instances of this limit can be present in an XGLL project, each applying to a different frame.

4.2.3.4 Maximum Tilt Angle

This specifies the maximum tilt angle of an array. This is the angle of the frame with respect to the horizon. It can be specified for all frames or one specific frame. Multiple instances of this limit can be present in an XGLL project, each applying to a different frame. It is quite possible that the mechanical design of a frame requires that it not be suspended at an angle that exceeds its safety rating. This type of limit can be used to specify this condition.

4.2.3.5 Minimum Tilt Angle

This specifies the minimum tilt angle of an array. This is the angle of the frame with respect to the horizon. It can be specified for all frames or one specific frame. Multiple instances of this limit can be present in an XGLL project, each applying to a different frame.

4.2.3.6 Minimum Number of Cabinets

This specifies the minimum number of cabinets that can be used in an array. It can be specified for all frames or one specific frame. Multiple instances of this limit can be present in an XGLL project, each applying to a different frame. A minimum number of cabinets can be useful to assure that the array is sufficiently long to yield good directivity control.

4.2.4 Warnings

Warnings to be given to the user can also be defined in the XGLL project. When the limit value for a particular warning is exceeded the specified text is typically shown in the software using the GLL. An XGLL project is not required to have any warnings. They may all be deleted with the DELETE button on the EDIT WARNINGS page (MECHANICAL DATA > WARNINGS) if desired. If no warnings exist clicking the NEW button will add one. If more are needed, select one of the warnings in the table and click the DUPLICATE button.

There are five different types of warnings that can be used in an XGLL project. They are listed below. Each is identical to the same type of Limit.

- Maximum Number of Cabinets
- Minimum Number of Cabinets
- Maximum Weight
- Maximum Tilt Angle
- Minimum Tilt Angle

The main difference between a limit and a warning is that for a warning the message to be displayed to the user can be specified in the XGLL project. However, the limit values for warnings are not enforced and therefore the associated warning messages are only informational for the end user.

4.3 Rated Coverage Angle

It is highly recommended that the Rated Coverage Angle data be entered for line array cabinets. This info is used to show the rated horizontal coverage for an array in ***EASE Focus*** prior to calculating the coverage map.

4.4 Example

4.4.1 Box Type

The geometry used for the box types must be approached in a particular manner. Specifically, the origin of the drawing must be on the line between the points on the left and right sides of the loudspeaker where the device (loudspeaker or frame) immediately above the loudspeaker connects to it. This is required because this defines the pivot point for the loudspeaker as the splay angle is changed.

All locations for the loudspeaker should be defined relative to this pivot point which is one of the reasons it must be located at the origin of the drawing.

Use ***EASE*** to draw the line array box shown in Figure 4-4 or create an .xed file for it. This will be used in our example. The rigging hardware for this line array places the attachment point 25 mm above the box and 100 mm behind the front edge of the box. This is the origin of the drawing. The attachment points for the next box (Connecting Point for Next Cabinet) are denoted by the small grey circles. These are 175 mm below the top of the box (200 mm below the origin).

The center of mass for the loudspeaker with its rigging hardware attached is 175 mm behind the front of the box and 100 mm below the top of the box. This is 75 mm behind and 125 mm below the origin. The point is denoted by a small red circle.

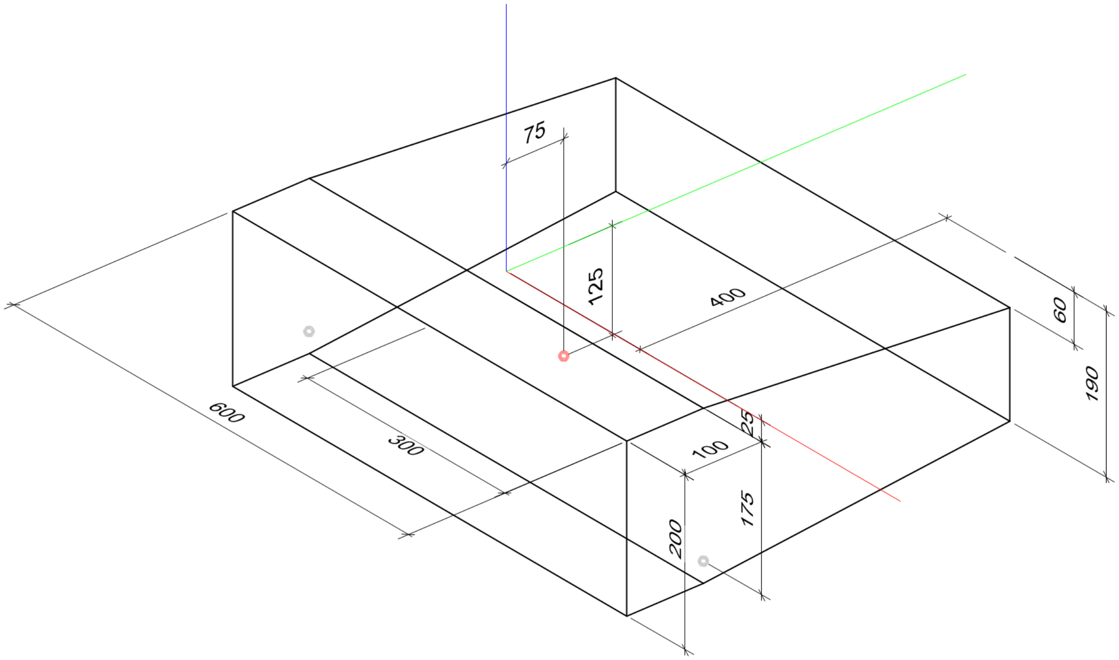


Figure 4-4 – Isometric view of line array loudspeaker (dimensions in mm)

The Connecting Point for Next Cabinet must be entered for each box type. It should be understood that the location of this point may not be on the loudspeaker. It may be below the loudspeaker. This will depend on the design of the loudspeaker and the rigging system.

4.4.2 Frames

The geometry for a flown frame used at the top of an array is not required to locate the origin at a particular point. However, all locations for the frame are relative to the origin so some care in choosing its location may simplify the specification of these other locations for the GLL. It is recommended to consider how everything fits together when selecting the origin. It may help to have an assembly drawing of the frame and the loudspeaker(s) to assist with this.

The geometry for a ground stacked frame used at the bottom of an array is required to have its origin located at the point where the bottom cabinet will be attached to the frame.

The drawing in Figure 4-5 shows the frame for our example line array. Use ***EASE*** to draw this frame or create an .xed file for it also. The origin for this frame drawing is at the top, front edge of the frame, centered left to right. The center of mass (denoted by the small green circle) is 250 mm behind the front and 35 mm below the top of the frame. The Connecting Point (denoted by the light grey circles) is 50 mm behind the front and 50 mm below the top of the frame.

The holes, or pin points, in the center strut of the frame do not need to be drawn. However, their locations do need to be specified on the PIN POINTS tab. The first one is 25 mm behind the front of the frame. The center-to-center space for each of them is 25 mm. The entire row of 15 holes is centered 40 mm below the top of the frame.

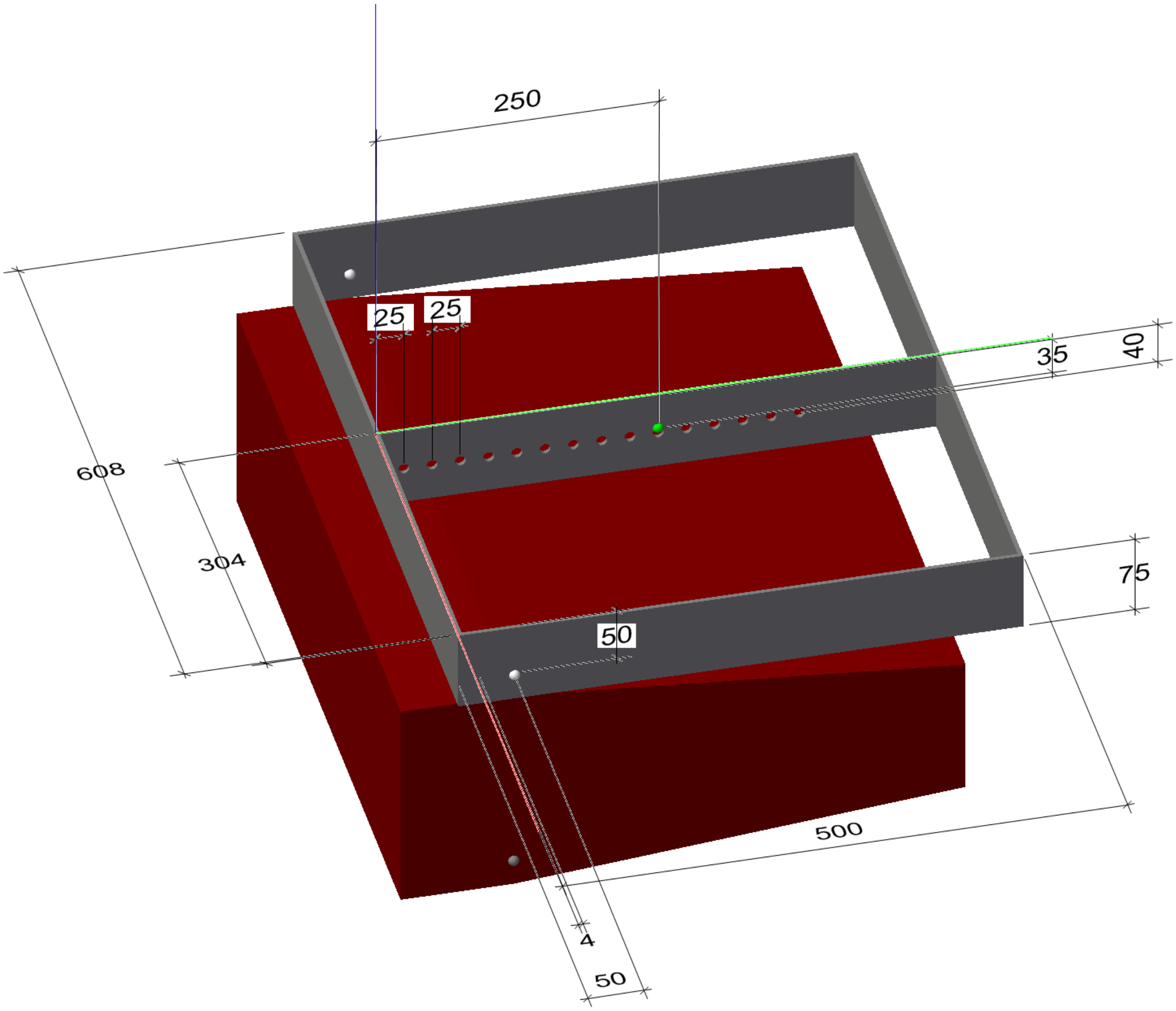
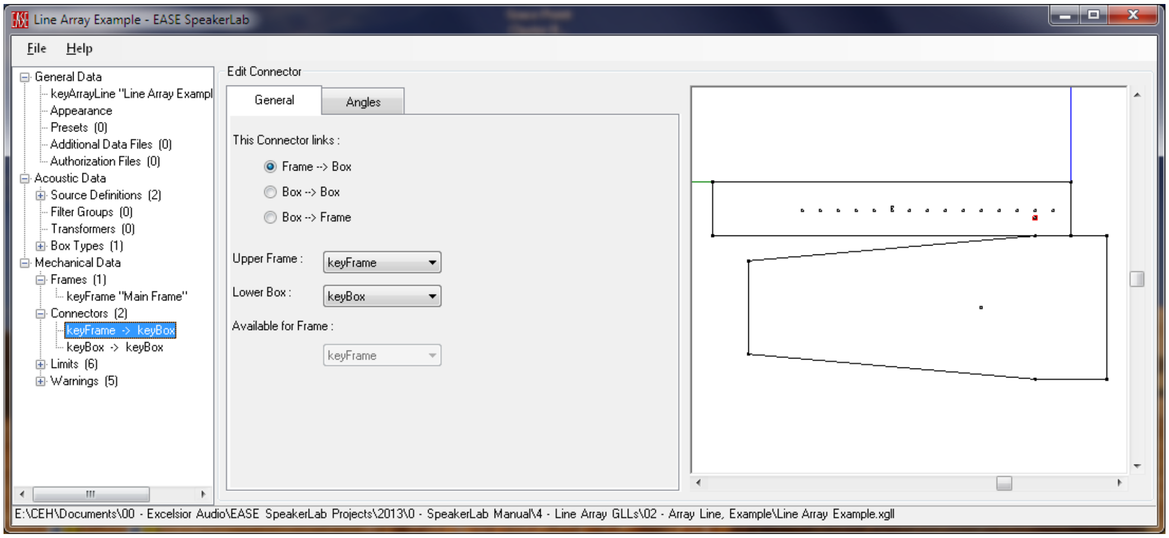


Figure 4-5 – Isometric view of line array frame and loudspeaker (dimensions in mm)

4.4.3 Connectors

Since our example line array system has only one frame and one box we will need only two connectors. One connector will allow a box to be connected to the frame. The other connector will allow two boxes to be connected together. By default these two connectors are part of every line array system. As mentioned previously, other connectors can be added as needed.

Selecting the KEYFRAME -> KEYBOX (MECHANICAL DATA > CONNECTORS) will show the EDIT CONNECTOR page on the right (Figure 4-6). Select the ANGLES tab to specify the angles allowed between the frame and the line array. For the POSSIBLE ANGLES select 6. In the table the values in the ANGLE column should be from 0 to 5. The corresponding DISPLAY ANGLE is what the user sees when selecting this angle within the GLL. It is recommend to use the same display here as the markings on the rigging system for the array. The SHOW ANGLE dropdown box at the bottom can be used to display a selected angle.



This Connector links :

● Frame --> Box
○ Box --> Box
○ Box --> Frame

Upper Frame : keyFrame

Lower Box : keyBox

Available for Frame : keyFrame

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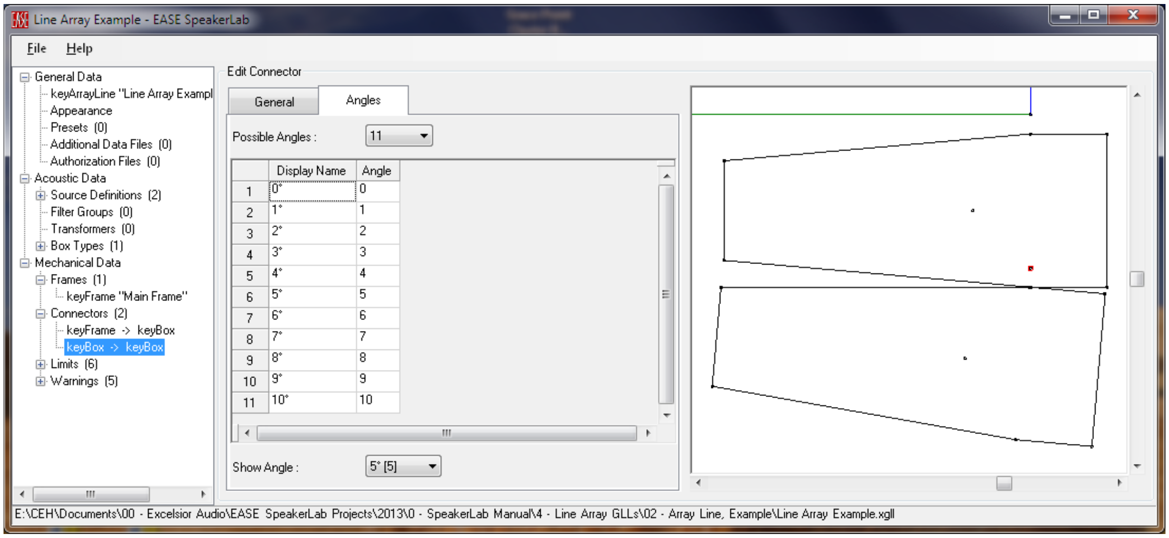
Figure 4-6 – Edit Connector page, General tab for a frame to box connector

Selecting the KEYBOX -> KEYBOX will show the EDIT CONNECTOR page on the right for this connector (Figure 4-7). Select the ANGLES tab to specify the angles allowed between the two line array boxes. For the POSSIBLE ANGLES select 11. In the table the values in the ANGLE column should be from 0 to 10. The corresponding DISPLAY ANGLE is what the user sees when selecting this angle within the GLL. It is recommend to use the same display here as the markings on the rigging system for the array. The SHOW ANGLE dropdown box at the bottom can be used to display a selected angle. Select several different angles to see the lower box rotate relative to the upper box.



Possible Angles : 11

	Display Name	Angle
1	0°	0
2	1°	1
3	2°	2
4	3°	3
5	4°	4
6	5°	5
7	6°	6
8	7°	7
9	8°	8
10	9°	9
11	10°	10

Show Angle : 5° [5]

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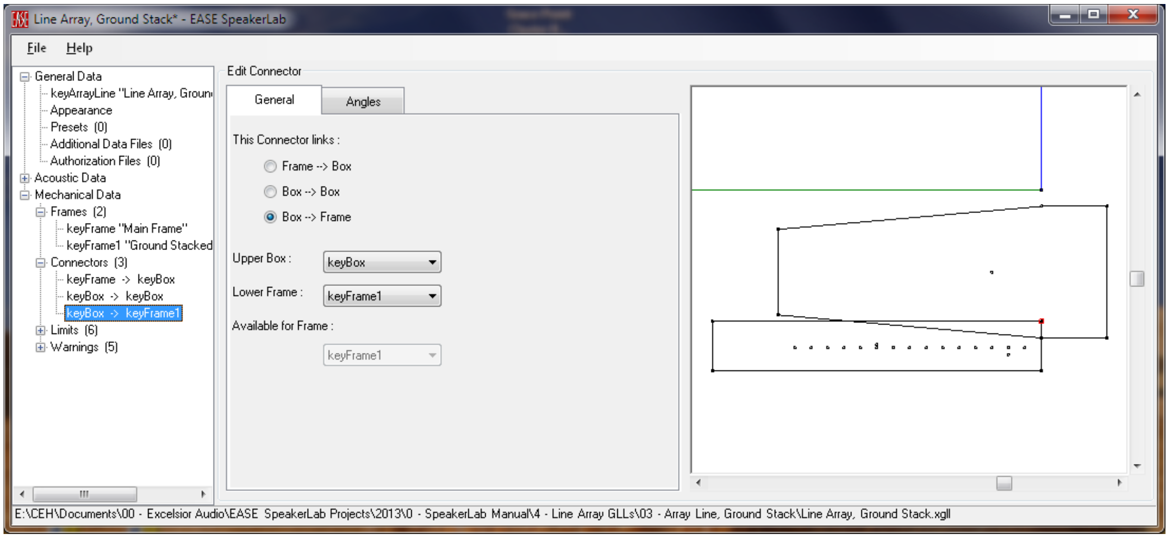
Figure 4-7 – Edit Connector page, Angles tab for a box to box connector

4.4.4 Run GLL

The GLL can now be run if desired. Different array configurations can be set-up and the performance investigated.

4.5 Ground Stack Example

Frames used at the bottom of an array (i.e., ground stacked) have one very important difference compared to a frame used at the top of an array. This difference is that the origin of the geometry drawing for the frame is the point at which the frame attaches to the loudspeaker above it. For the flown frame it was convenient to use the top of the front edge as the origin. This will not work for the ground stack frame. Using the above example select the KEYFRAME -> KEYBOX (MECHANICAL DATA > CONNECTORS) and of the GENERAL tab change the CONNECTOR LINKS from FRAME --> BOX to BOX --> FRAME. This will place the frame below the box. It can be seen that the box is not oriented properly with respect to the frame (Figure 4-8). Change the CONNECTOR LINKS back to FRAME --> BOX so that the flown frame will function properly.


Line Array, Ground Stack* - EASE SpeakerLab

- General Data
  - keyArrayLine "Line Array, Groun
  - Appearance
  - Presets (0)
  - Additional Data Files (0)
  - Authorization Files (0)
- Acoustic Data
- Mechanical Data
  - Frames (2)
    - keyFrame "Main Frame"
    - keyFrame1 "Ground Stacked
  - Connectors (3)
    - keyFrame -> keyBox
    - keyBox -> keyBox
    - keyBox -> keyFrame1
  - Limits (6)
  - Warnings (5)

Edit Connector

This Connector links :

○ Frame --> Box
○ Box --> Box
● Box --> Frame

Upper Box : keyBox

Lower Frame : keyFrame1

Available for Frame : keyFrame1

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Figure 4-8 – Ground stack box to frame connector with incorrect origin of the frame drawing

For this same frame in the real world to work as a ground stack frame in the GLL there are two things that must be changed. We don't want to change the flown frame used at the top of an array so we must create a second frame in the GLL and then make these changes to that frame. The end user of the GLL will never know about the two frames internal to the GLL. They will only see that they are selecting between a flown or a ground stacked type of application.

Since our frame is not symmetrical top-to-bottom (the connection point for the loudspeaker is below the center line) we must flip the frame over, that is turn it upside down. This makes sense because in actual use in the real world the frame would have to be flipped upside down to be used as a ground stacked frame.

The other change that must be made to the frame geometry is that the origin must be on the line between the points on the left and right sides of the frame where the loudspeaker above the frame connects to the frame. For the upside-down frame this would be 50 mm behind the front of the frame and 25 mm above the top of the frame. This point should be the origin of the drawing for the ground stacked frame.

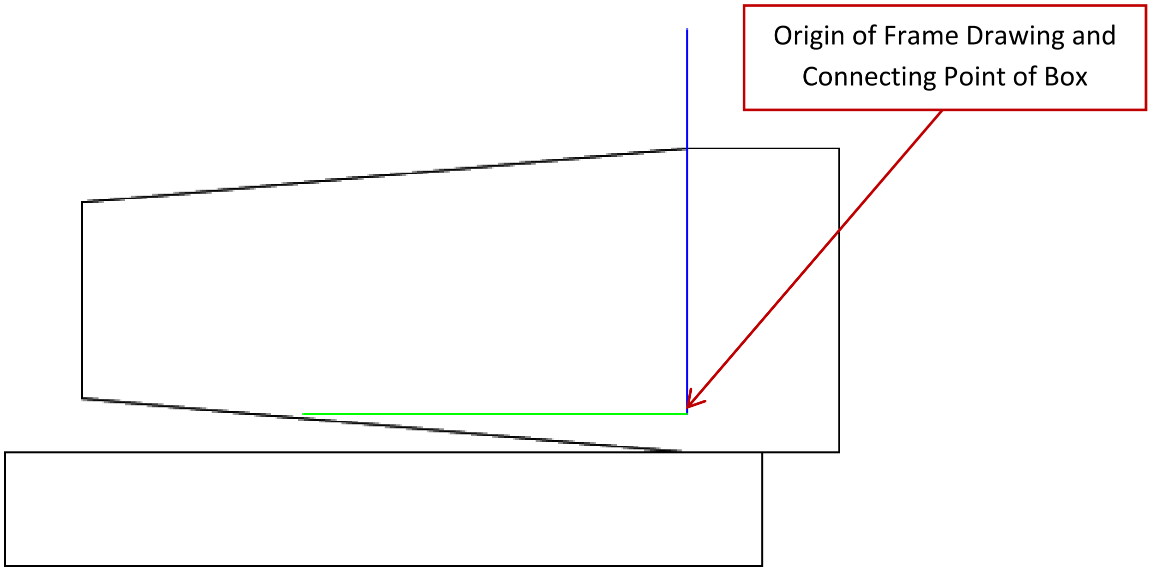


Figure 4-9 – Example of a ground stacked line array showing frame and box

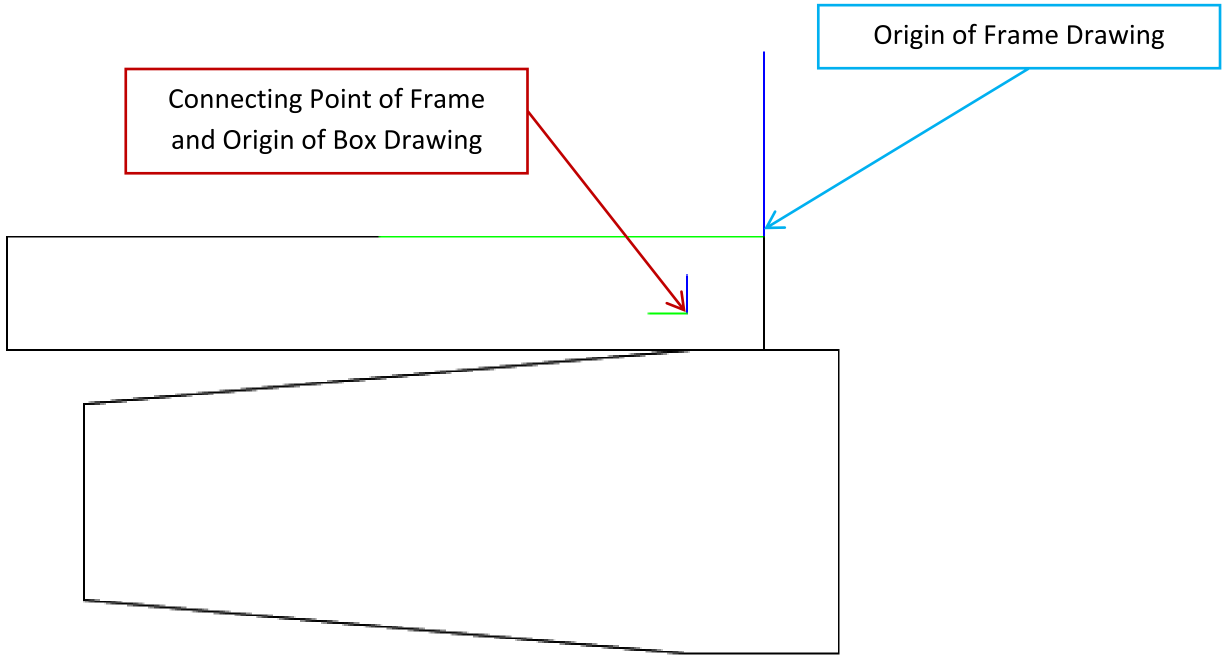


Figure 4-10 – Example of a flown line array showing frame and box

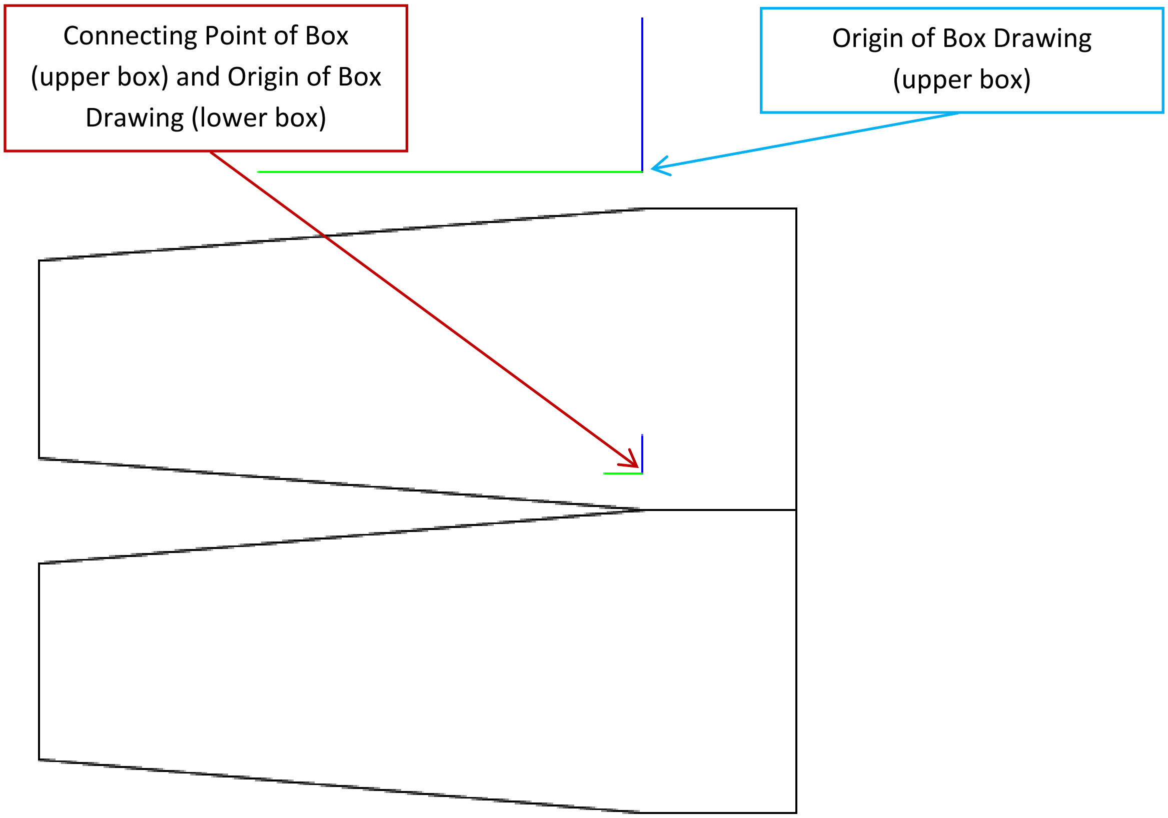

Connecting Point of Box (upper box) and Origin of Box Drawing (lower box)
Origin of Box Drawing (upper box)

Figure 4-11 – Example of a line array showing adjacent boxes

After a new frame drawing has been created with these changes it should be added to the GLL. Select FRAME (under MECHANICAL DATA) in the property tree. On the EDIT FRAMES page select row 1 of the table and click the DUPLICATE button. In the property tree select KEYFRAME1 that was just added. On the EDIT FRAME page change the DISPLAY NAME to "Ground Stacked". Click the EDIT GEOMETRY button and select the new drawing with the changes. Notice that once the geometry file is changed neither the Pin Points, the Center of Mass, nor the Connecting Point are in the correct places. Since the frame was turned upside-down and the origin relocated the location of these should also be updated.

The location of the origin was moved 50 mm (0.050 m) along the y-axis and 25 mm (0.025m) along the z-axis. Using these values the new Center of Mass will be at (0, 0.200, 0.060).

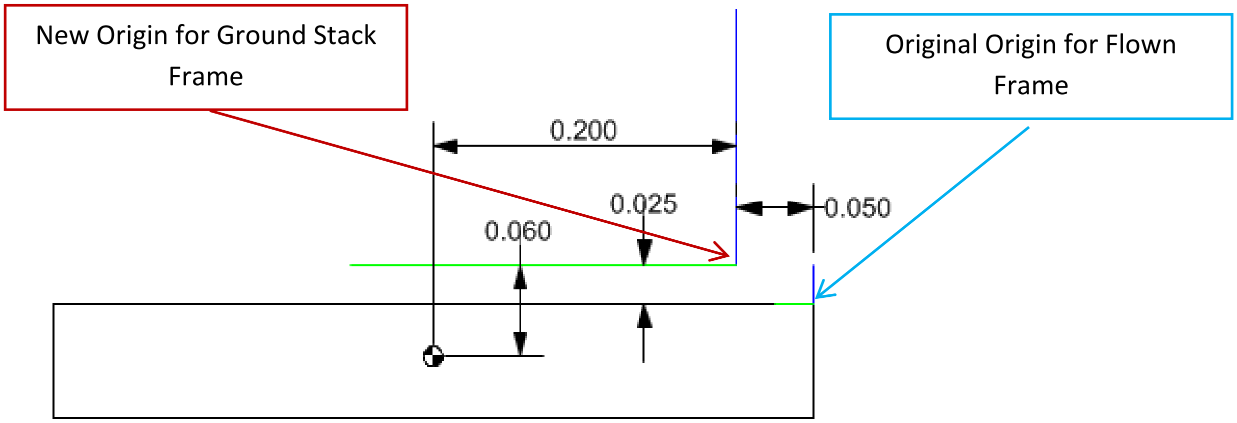
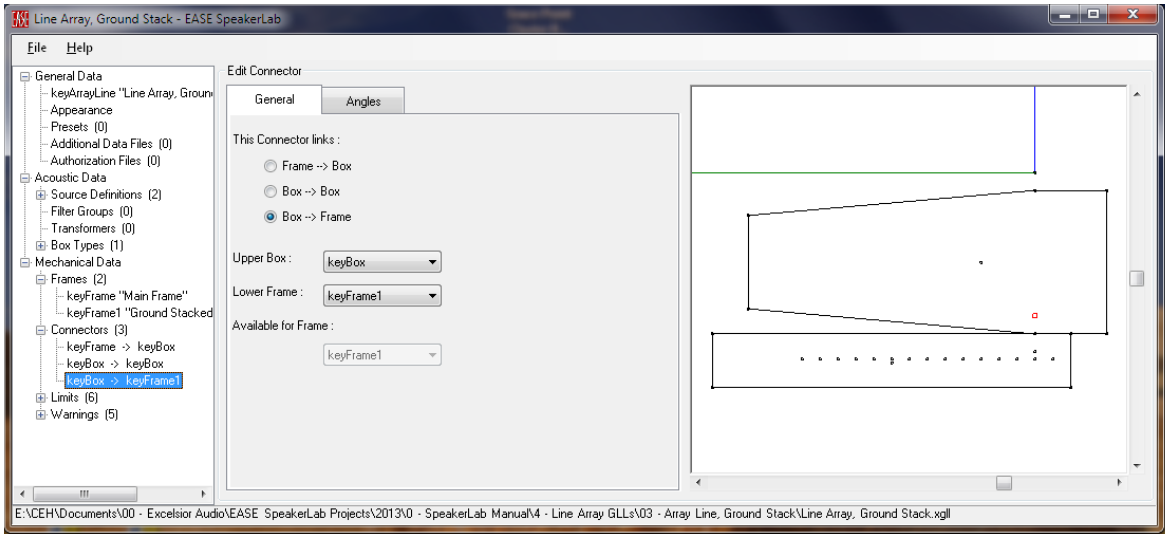

New Origin for Ground Stack Frame
Original Origin for Flown Frame
0.200
0.025
0.060
0.050

Figure 4-12 – Frame drawing showing the change in the location of the origin and new coordinates for the center of mass

Since this is a ground stacked array there will not be a box connected below this frame. The Connecting Point will never be used and does not need to be specified. This can be changed to the origin if desired. Similarly, the ground stack frame will never be used to hang an array so none of the Pin Points need to be specified. The NUMBER OF PINPOINTS can be changed to 0.

We must now add a new connector to allow the line array box to be connected to the ground stacked frame. Select CONNECTORS (under MECHANICAL DATA) in the property tree. Select row 1 in the table and click the DUPLICATE button. Select the KEYBOX -> KEYFRAME1 that was just added. Change the CONNECTOR LINKS from FRAME --> BOX to BOX --> FRAME. This should now have the line array box and the ground stacked frame oriented correctly.



This Connector links :

○ Frame --> Box
○ Box --> Box
● Box --> Frame

Upper Box : keyBox

Lower Frame : keyFrame1

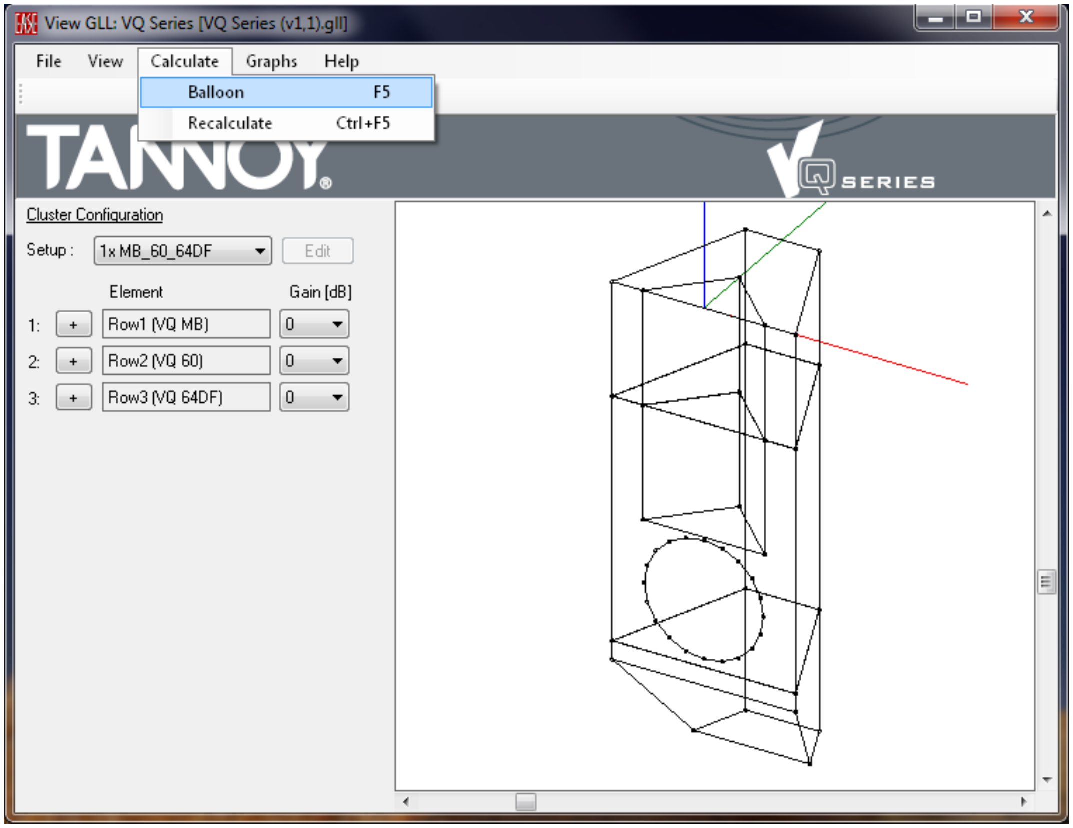
Available for Frame : keyFrame1

E:\CEH\Documents\00 - Excelsior Audio\EASE SpeakerLab Projects\2013\0 - SpeakerLab Manual\4 - Line Array GLLs\03 - Array Line, Ground Stack\Line Array, Ground Stack.xgll

Figure 4-13 – Ground stack box to frame connector with the correct origin of the frame drawing

5 The GLL Viewer

The ***EASE GLL Viewer*** is a program that allows users to open GLL models and perform response calculations of the modeled loudspeaker system. It is included with every copy of ***EASE*** or ***EASE SpeakerLab*** and is also available as a free download from the AFMG website at http://ease.afmg.eu/index.php/ea-tools-en.html.



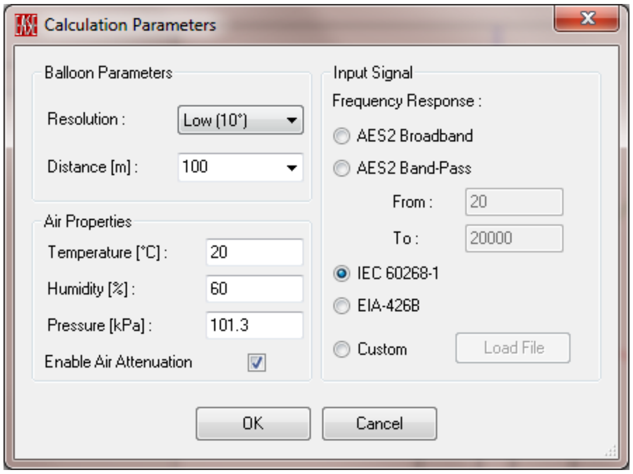
Setup : 1x MB_60_64DF

		Element	Gain [dB]
1:	+	Row1 (VQ MB)	0
2:	+	Row2 (VQ 60)	0
3:	+	Row3 (VQ 64DF)	0

Figure 5-1 – Calculate the GLL

5.1.1 Calculating the GLL

To calculate the response of the GLL loudspeaker model click on CALCULATE > BALLOON. The CALCULATION PARAMETERS dialog box will appear.



Calculation Parameters

Balloon Parameters

Resolution : Low (10°)

Distance [m] : 100

Air Properties

Temperature [°C] : 20

Humidity [%] : 60

Pressure [kPa] : 101.3

Enable Air Attenuation ☑

Input Signal

Frequency Response :

○ AES2 Broadband

○ AES2 Band-Pass

From : 20

To : 20000

● IEC 60268-1

○ EIA-426B

○ Custom

Figure 5-2 – Calculation parameters

The Balloon Parameters panel lets the user choose the angular Resolution of the calculation and the Distance from the coordinate origin of the loudspeaker system. Note that the coordinate origin may not be the center of a loudspeaker cabinet or an array. It is the origin defined for the system. By inspecting the case drawings of the loudspeaker (Graphs > Case Drawings) it is possible to see the coordinate axes and the origin in relation to the loudspeaker cabinet or array. The Distance is an important value for the calculations as the directivity of a multi-source loudspeaker model (as well as a real multi-source loudspeaker) will not be the same at all distances until the observation location is in the far field of the loudspeaker.

The Input Signal panel defines the spectral content of the excitation signal. The importance of this selection relates to multi-source loudspeaker models as well. The calculated maximum SPL and maximum input voltage will depend on the spectral content of the input signal, just as it would in the real world.

The Air Properties panel defines the environmental conditions. The Enable Air Attenuation checkbox includes the effects of high frequency attenuation due to the Air Properties in the calculated results. This calculation is in accordance with ISO 9613 [9].

5.1.2 Graphs Menu

Once the calculation is complete a set of graphs are available for viewing. The graphs available in the ***GLL Viewer*** are identical to those in GSS Source Editor. Please refer to the section 2.7.6 – Viewing the Graphs (Graphs Menu) of this manual for a complete description of the graphs and visualization features.

The graphs in the ***GLL Viewer*** have the same display options as those in the GSS Source Editor as well. Please see section 2.7.7 – Graphs Options for a complete description of these options.

5.1.3 View Menu

As with the GRAPHS menu, the VIEW menu of the ***GLL Viewer*** is the same as the GSS Source Editor. Please see section 2.7.8 – Viewing Options (View Menu) for complete details of the View menu.

5.1.4 GLL System Information

The information entered in the GENERAL SYSTEM PROPERTIES of the xgll project file is shown after clicking on FILE > INFO from the main menu of the ***GLL Viewer***. Here the user can see information the GLL author has included such as the system name, manufacturer, and contact info. The Information area can be particularly useful for conveying important items to the users.

Note that each GLL is different and will contain different information here depending on what the author of the GLL has chosen to include.

Clicking the OK button will dismiss the GLL SYSTEM INFORMATION dialog box.

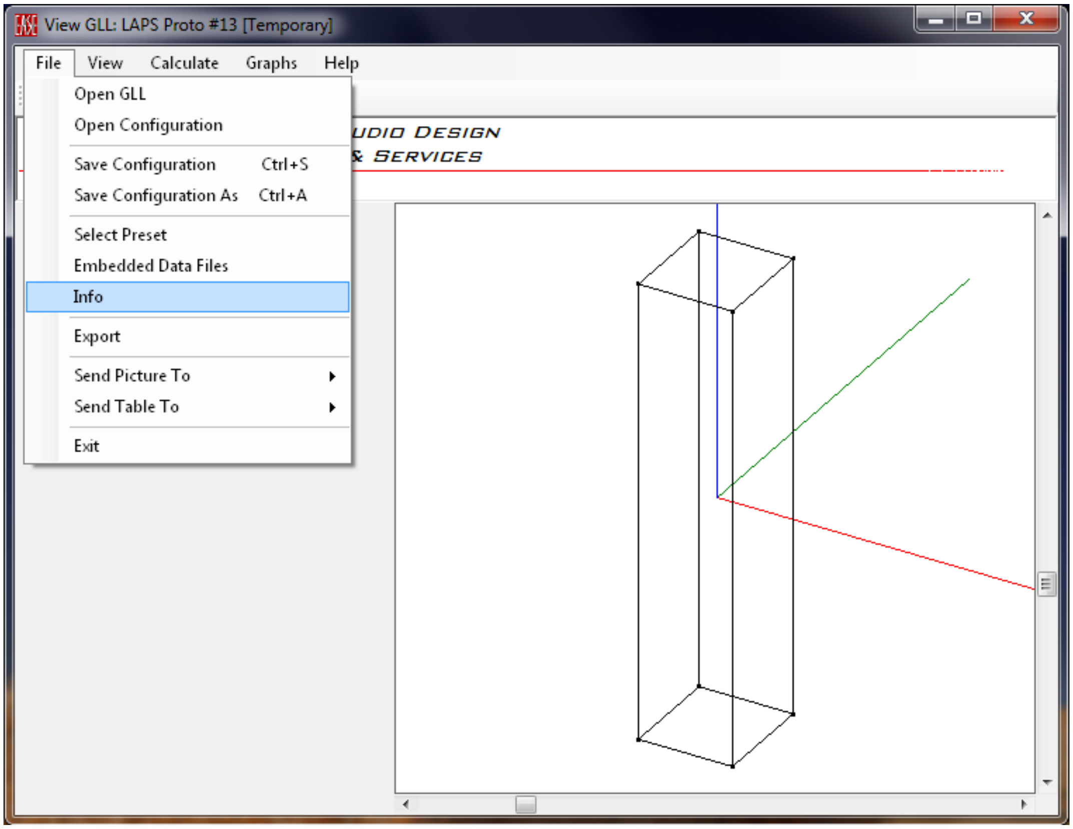
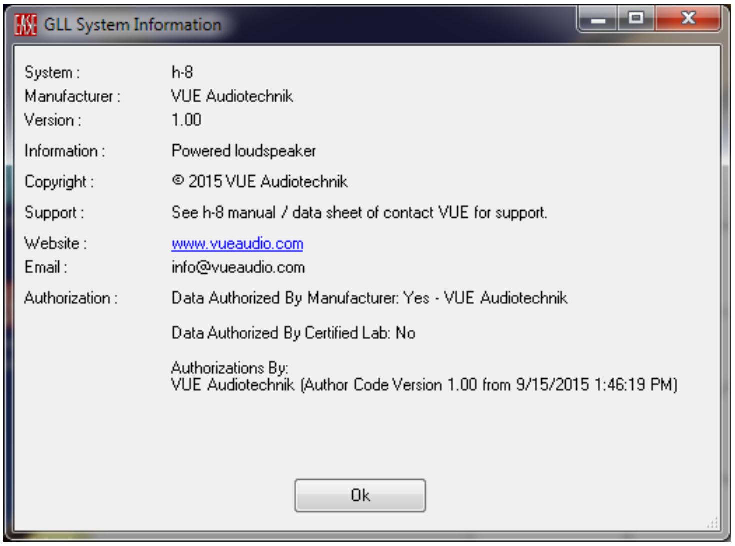


Figure 5-3 – Calling the Info dialog box for the GLL



System : h-8
Manufacturer : VUE Audiotechnik
Version : 1.00
Information : Powered loudspeaker
Copyright : © 2015 VUE Audiotechnik
Support : See h-8 manual / data sheet of contact VUE for support.
Website : www.vueaudio.com
Email : info@vueaudio.com
Authorization : Data Authorized By Manufacturer: Yes - VUE Audiotechnik

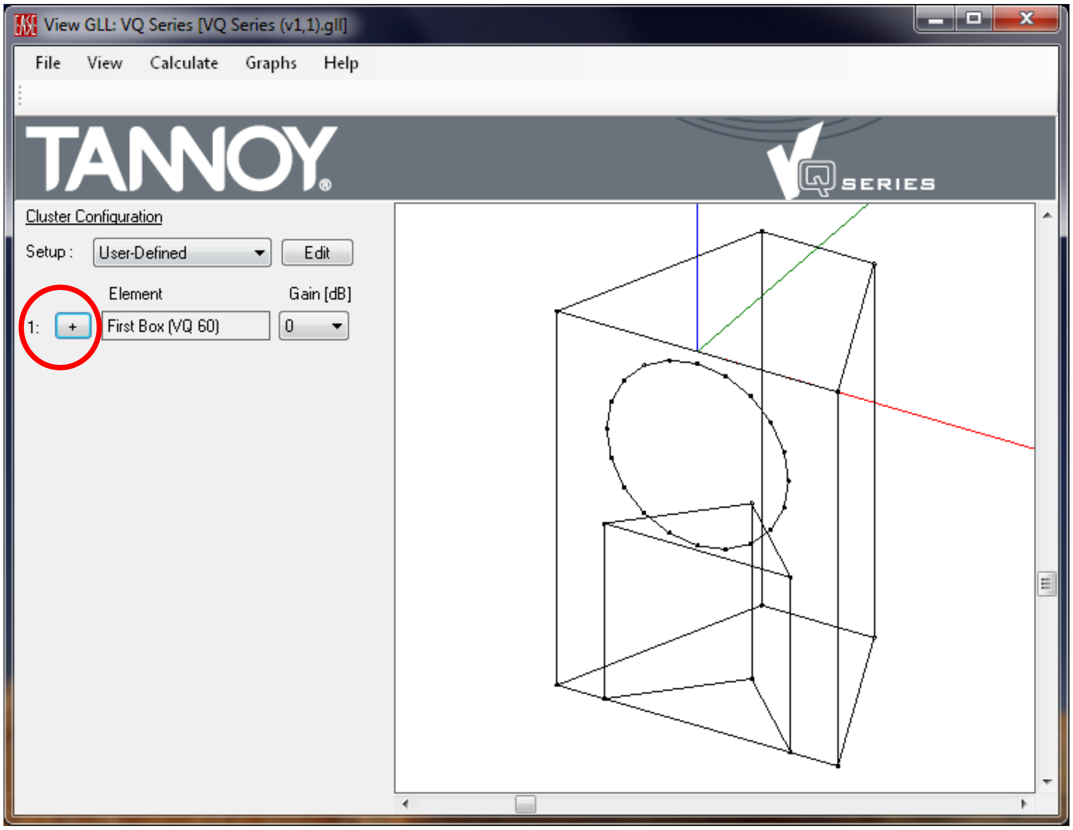
Data Authorized By Certified Lab: No

Authorizations By:
VUE Audiotechnik (Author Code Version 1.00 from 9/15/2015 1:46:19 PM)

Figure 5-4 – Info dialog box

5.1.5 Selecting the Input Configuration

Some loudspeaker models may have more than one INPUT CONFIGURATION. If this is the case, the different configurations available can be chosen by expanding the properties tree for the loudspeaker (+ sign to the left of the loudspeaker's name). This will reveal the INPUT CONFIGURATION area with a dropdown box for selecting the available configuration, as well as the available INPUT (there may be more than one) and a FILTER button for each INPUT.

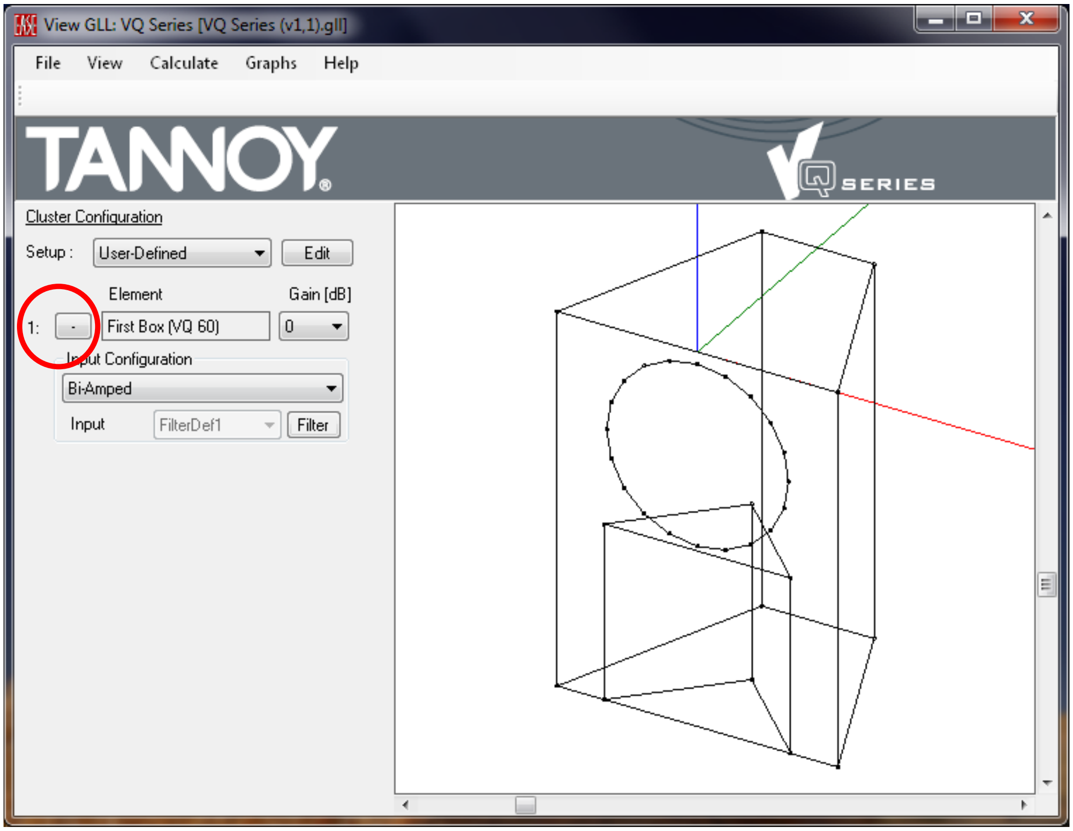


Setup : User-Defined

Element: 1: First Box (VQ 60)

Gain [dB]: 0

Figure 5-5 – Running GLL showing the expand button for the Input Configuration circled



Setup : User-Defined

Element: 1: First Box (VQ 60)

Gain [dB]: 0

Input Configuration: Bi-Amped

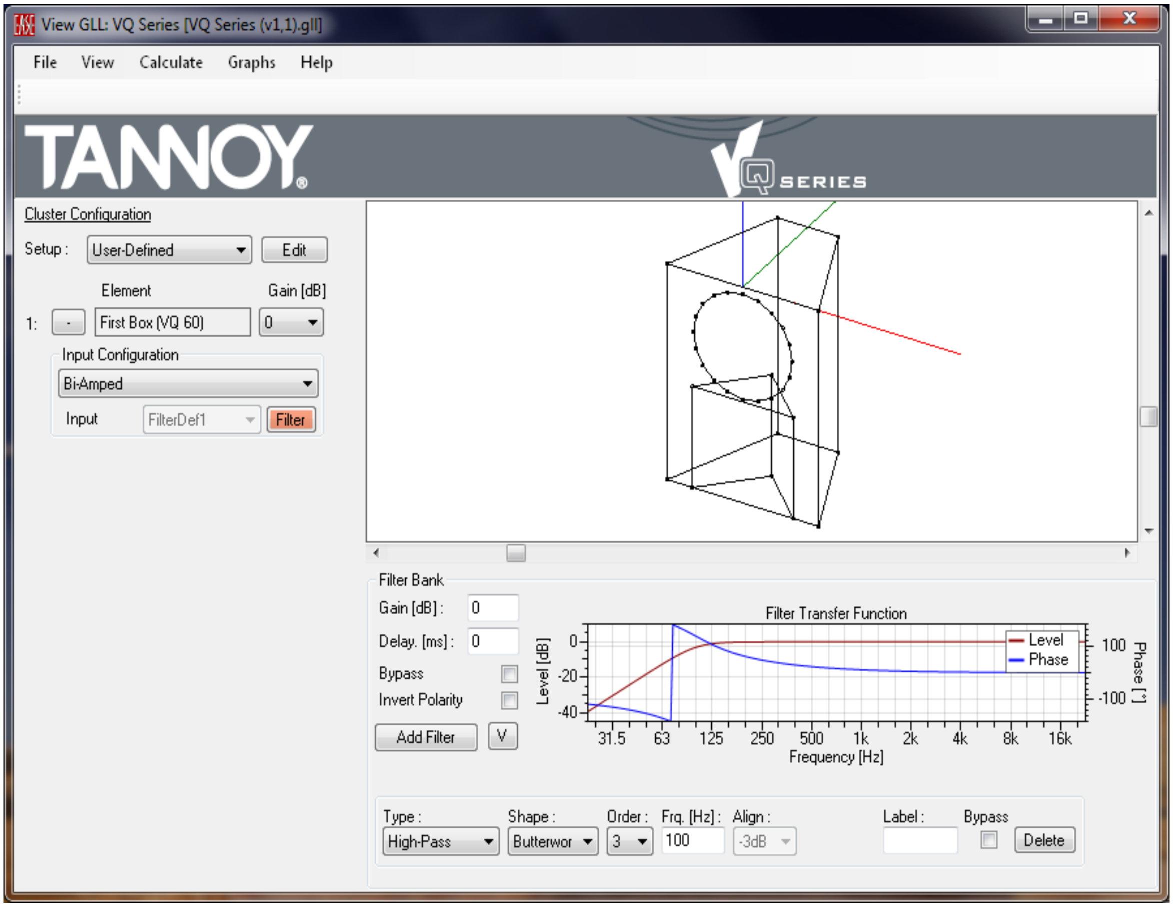
Input: FilterDef1

Figure 5-6 – Running GLL showing the Input Configuration expanded

Let's say that we have a GLL for a passive loudspeaker that has an included passive crossover. The passive crossover can be bypassed so that the loudspeaker can be biamped using an external crossover (DSP) and separate amplifier channels. This GLL should have two different Input Configuration, one for Passive or Full-Range (or whatever the GLL author decided to call it) and one for BiAmp (or whatever the GLL author decided to call it).

When the BiAmp configuration is selected we see that there are two inputs to the loudspeaker, just like there would be for a real loudspeaker when it will be biamped. Clicking the Filter button displays a Filter Bank area on the right side of the ***View GLL*** window just under the main display. This Filter Bank is very much like a modern DSP unit. There are controls for the Gain [], Delay [], and Invert Polarity. There is also a provision to Bypass all of the settings of the Filter Bank; essentially setting them all to zero.

Note that this Filter Bank is for the selected Input only. Each Input has its own, separate Filter Bank.



View GLL: VQ Series [VQ Series (v1,1).gll]

Setup : User-Defined

1: First Box (VQ 60) 0

Input Configuration: Bi-Amped

Input FilterDef1

Filter Bank

Gain [dB] : 0

Delay. [ms] : 0

Bypass □

Invert Polarity □

Type : High-Pass

Shape : Butterwor

Order : 3

Frq. [Hz] : 100

Align : -3dB

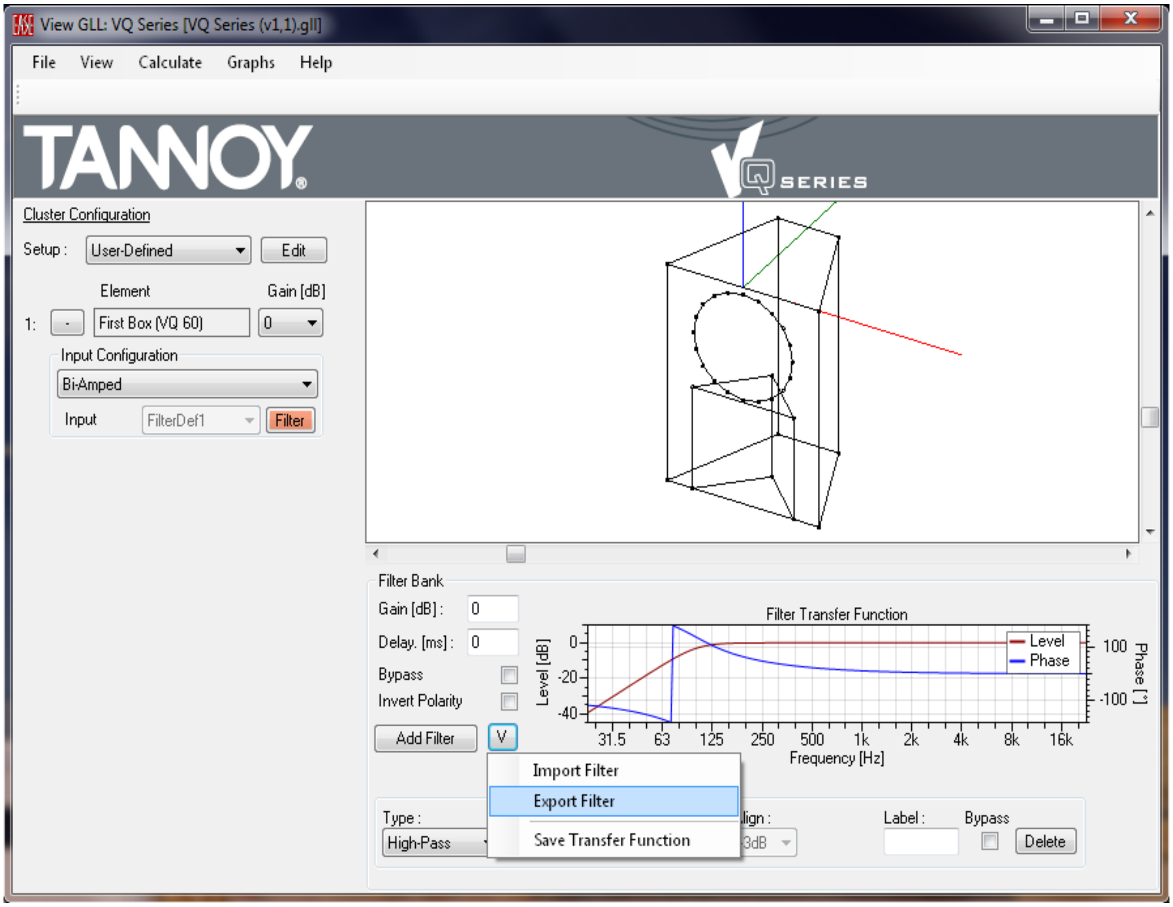
Label :

Bypass □

Figure 5-7 – Running GLL showing the filter bank

Clicking the ADD FILTER button will add another filter to a FILTER BANK. There is no limit on the number of filters that may be added. For complete details on adding the different types of filters and editing their settings please see Section 2.11.3 – Adding Filters.

The V button allows a complete FILTER BANK to be either imported or exported. This can be very useful if an extensive set of filters needs to be copied to multiple INPUTS. Clicking on V > EXPORT FILTER will open an EXPORT GENERIC FILTER dialog box which is used to specify the location to store the Filter Bank (.xgfb) file which contains the filter parameters. Any Filter Bank (.xgfb) file can be imported by clicking on V > IMPORT FILTER and choosing the desired .xgfb file.



View GLL: VQ Series [VQ Series (v1,1).gll]

Cluster Configuration

Setup : User-Defined

1: First Box (VQ 60) Gain [dB]: 0

Input Configuration: Bi-Amped

Input: FilterDef1

Filter Bank

Gain [dB] : 0

Delay. [ms] : 0

Bypass □

Invert Polarity □

Type : High-Pass

Label :

Bypass □

Figure 5-8 – Export a filter from the filter bank

The transfer function of the FILTER BANK can also be saved. This can be very useful for comparing the transfer function of the filter in the GLL model to the transfer function of the crossover or DSP actually being used. Please see Section 8.4 – Filter Measurement for more details on this important feature.

5.1.6 Exporting Graphs

Exporting the graphs of the calculated results is done in the same way as for the GSS Source Editor. Please see section 2.7.11 – Exporting Data for details.

5.1.7 Configurations

Configurations are a very useful feature of both ***SpeakerLab*** and the ***GLL Viewer***. They allow every setting associated with the set-up / configuration of the GLL model to be saved to a single file (.xglc). This makes it possible to work on optimizing the crossover filters or EQ settings of a loudspeaker system, for example, and save the results to be recalled later. It is also possible to optimize an array using ***SpeakerLab*** or the ***GLL Viewer*** and save the configuration file (.xglc). This configuration file can then be loaded into the same GLL used in ***EASE***. This can be very helpful so that multiple setting don't have to be reentered manually. The configuration file can also be sent to others that are using the same GLL so that they can open the configuration file and have the exact same settings for the GLL.

The configuration file saves the following settings.

- Number and types of boxes (Cluster & Line Array type GLLs, sections 3 & 4)
- Aiming angles of each box
- Filter Bank settings for each box

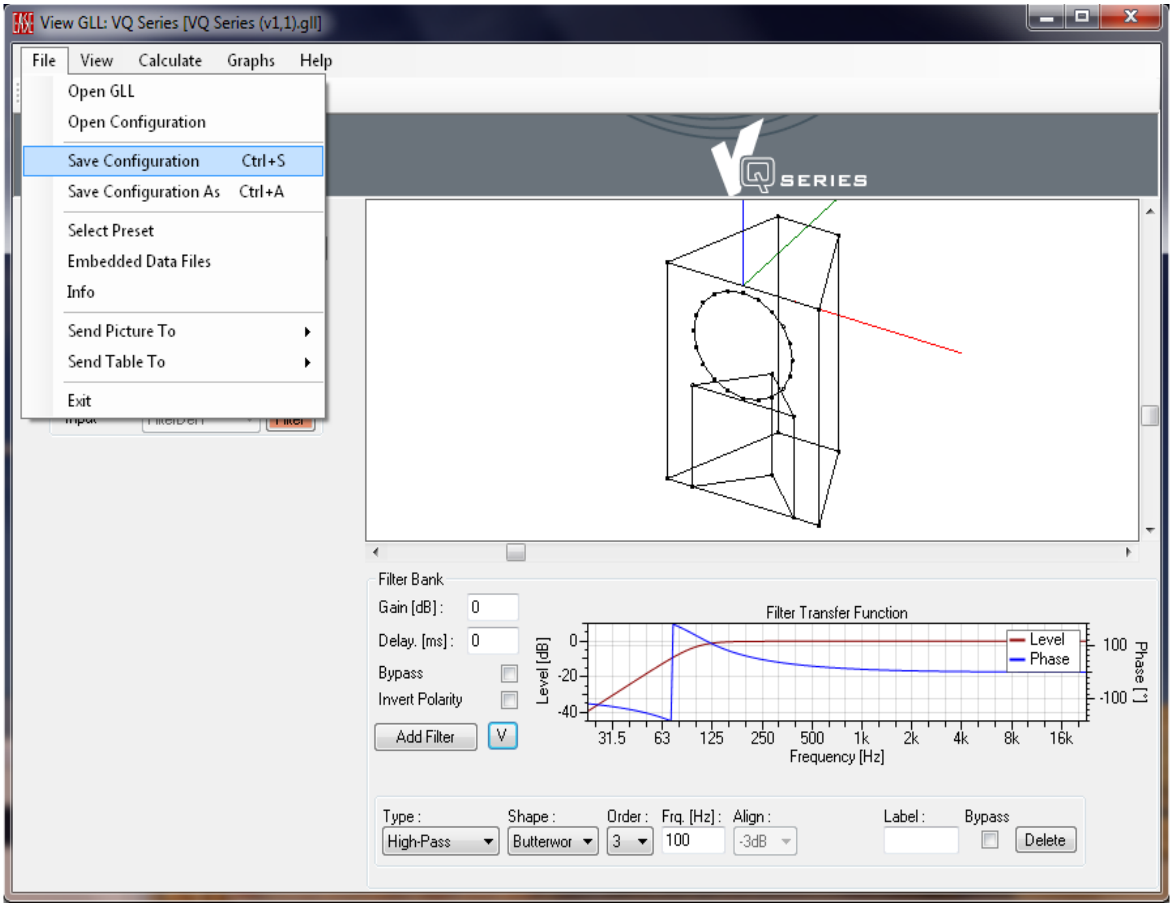
Note that the Calculation Parameters are not saved as part of the configuration file. These must be specified each time the GLL is calculated, or CTRL+F5 can be used after the first calculation to recalculate with the same parameters.

5.1.7.1 Saving a Configuration File

A configuration file can be saved by simply clicking on FILE > SAVE CONFIGURATION or FILE > SAVE CONFIGURATION AS from the main menu. The first time SAVE CONFIGURATION is used a SAVE GLL CONFIGURATION FILE dialog box is opened. This allows the file name and location for the file to be entered. This also occurs every time SAVE CONFIGURATION AS is used.

When SAVE CONFIGURATION is used after the first time or after SAVE CONFIGURATION AS has been used the current setting are saved to the previously named configuration file, thus overwriting the file.

The SAVE CONFIGURATION AS is useful if new settings are to be saved without overwriting the existing configuration file.



Gain [dB] : 0

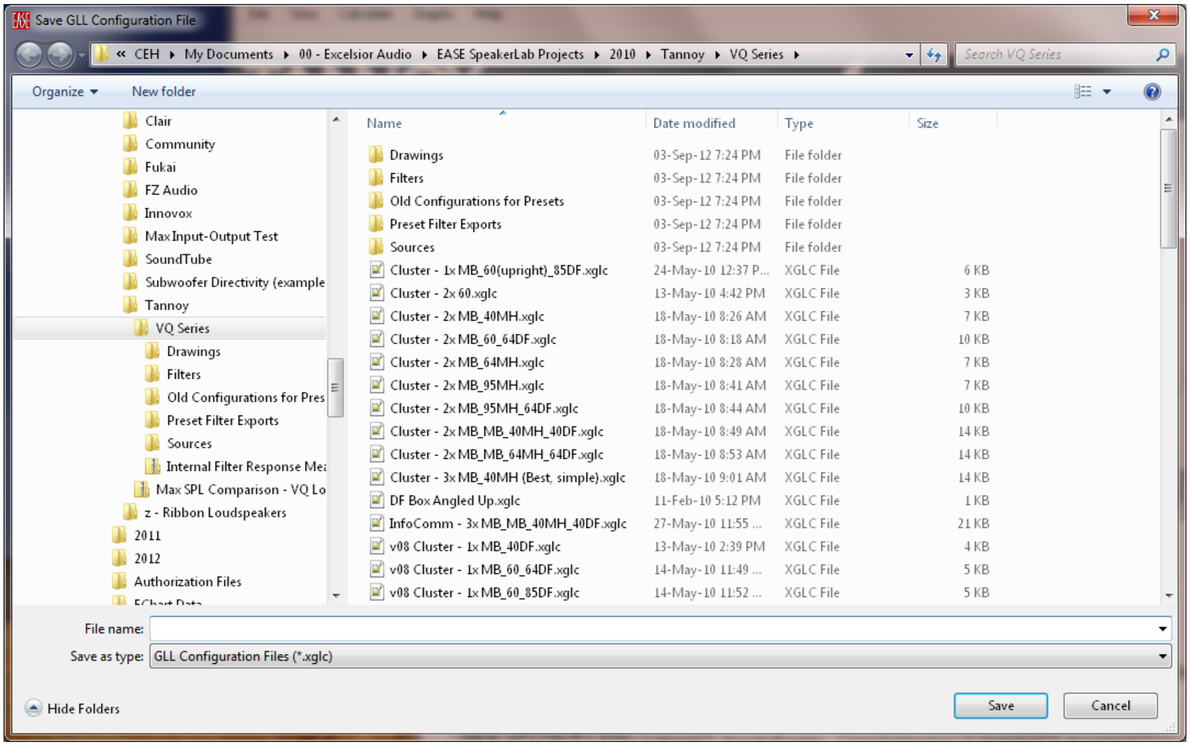
Delay. [ms] : 0

Bypass □

Invert Polarity □

Type : High-Pass | Shape : Butterwor | Order : 3 | Frq. [Hz] : 100 | Align : -3dB | Label : | Bypass □

Figure 5-9 – Save a configuration file



Name	Date modified	Type	Size
Drawings	03-Sep-12 7:24 PM	File folder	
Filters	03-Sep-12 7:24 PM	File folder	
Old Configurations for Presets	03-Sep-12 7:24 PM	File folder	
Preset Filter Exports	03-Sep-12 7:24 PM	File folder	
Sources	03-Sep-12 7:24 PM	File folder	
Cluster - 1x MB_60(upright)_85DF.xglc	24-May-10 12:37 P...	XGLC File	6 KB
Cluster - 2x 60.xglc	13-May-10 4:42 PM	XGLC File	3 KB
Cluster - 2x MB_40MH.xglc	18-May-10 8:26 AM	XGLC File	7 KB
Cluster - 2x MB_60_64DF.xglc	18-May-10 8:18 AM	XGLC File	10 KB
Cluster - 2x MB_64MH.xglc	18-May-10 8:28 AM	XGLC File	7 KB
Cluster - 2x MB_95MH.xglc	18-May-10 8:41 AM	XGLC File	7 KB
Cluster - 2x MB_95MH_64DF.xglc	18-May-10 8:44 AM	XGLC File	10 KB
Cluster - 2x MB_MB_40MH_40DF.xglc	18-May-10 8:49 AM	XGLC File	14 KB
Cluster - 2x MB_MB_64MH_64DF.xglc	18-May-10 8:53 AM	XGLC File	14 KB
Cluster - 3x MB_40MH (Best, simple).xglc	18-May-10 9:01 AM	XGLC File	14 KB
DF Box Angled Up.xglc	11-Feb-10 5:12 PM	XGLC File	1 KB
InfoComm - 3x MB_MB_40MH_40DF.xglc	27-May-10 11:55 ...	XGLC File	21 KB
v08 Cluster - 1x MB_40DF.xglc	13-May-10 2:39 PM	XGLC File	4 KB
v08 Cluster - 1x MB_60_64DF.xglc	14-May-10 11:49 ...	XGLC File	5 KB
v08 Cluster - 1x MB_60_85DF.xglc	14-May-10 11:52 ...	XGLC File	5 KB

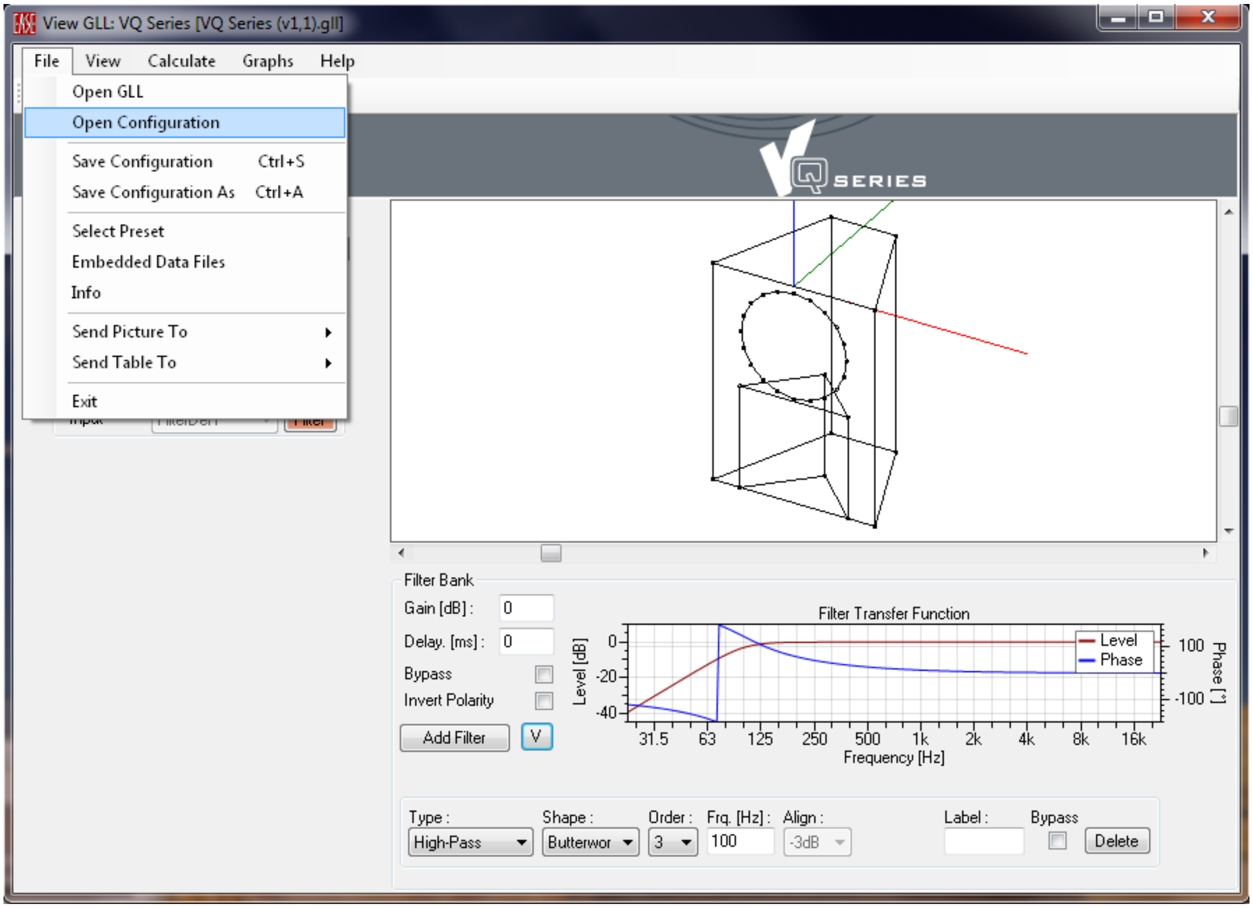
File name:

Save as type: GLL Configuration Files (*.xglc)

Figure 5-10 – Select the folder & name for the configuration file

5.1.7.2 Opening a Configuration File

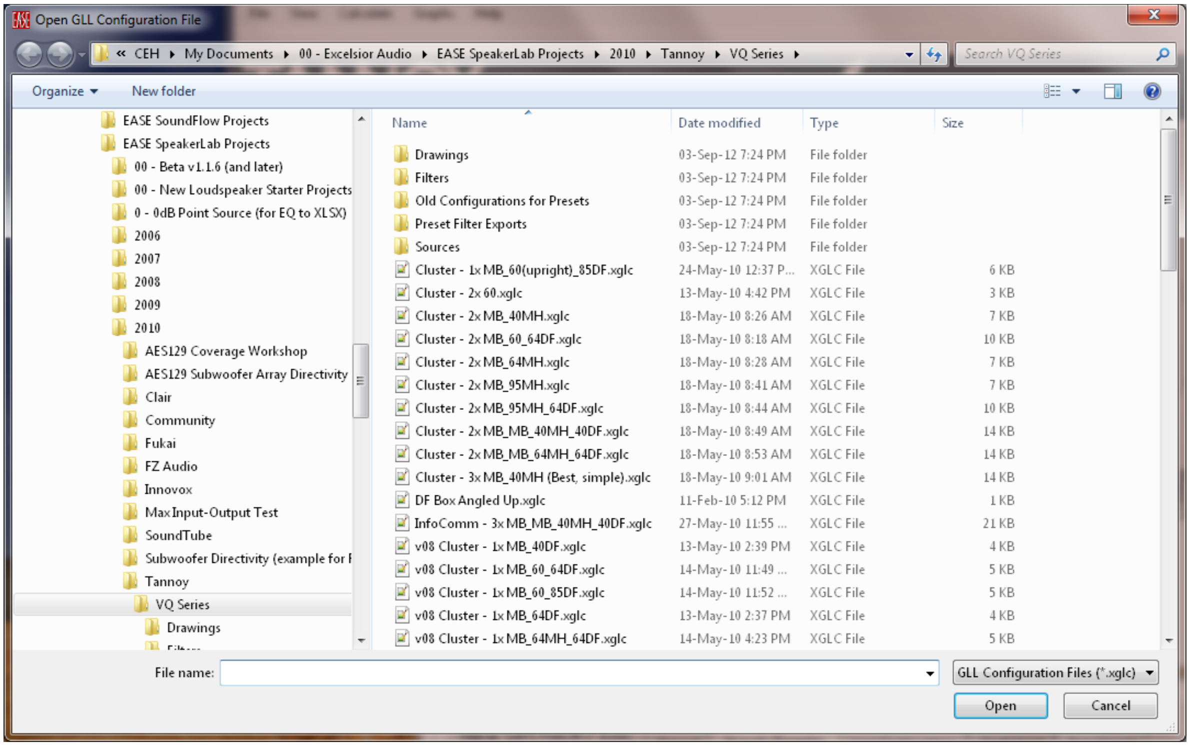
Loading the settings from a previously saved configuration file is accomplished by clicking on FILE > OPEN CONFIGURATION. This will display an OPEN GLL CONFIGURATION FILE dialog box to select the file to be opened. Select the desired file and click on OK to load the settings into the GLL.



Gain [dB] : 0
Delay. [ms] : 0
Bypass □
Invert Polarity □

Type :	Shape :	Order :	Frq. [Hz] :	Align :	Label :	Bypass
High-Pass	Butterwor	3	100	-3dB		□

Figure 5-11 – Opening a configuration file


Open GLL Configuration File
« CEH ▸ My Documents ▸ 00 - Excelsior Audio ▸ EASE SpeakerLab Projects ▸ 2010 ▸ Tannoy ▸ VQ Series ▸
Search VQ Series
Organize ▾
New folder
EASE SoundFlow Projects
EASE SpeakerLab Projects
00 - Beta v1.1.6 (and later)
00 - New Loudspeaker Starter Projects
0 - 0dB Point Source (for EQ to XLSX)
2006
2007
2008
2009
2010
AES129 Coverage Workshop
AES129 Subwoofer Array Directivity
Clair
Community
Fukai
FZ Audio
Innovox
Max Input-Output Test
SoundTube
Subwoofer Directivity (example for I
Tannoy
VQ Series
Drawings
File name:
GLL Configuration Files (*.xglc)
Open
Cancel

Name	Date modified	Type	Size
Drawings	03-Sep-12 7:24 PM	File folder	
Filters	03-Sep-12 7:24 PM	File folder	
Old Configurations for Presets	03-Sep-12 7:24 PM	File folder	
Preset Filter Exports	03-Sep-12 7:24 PM	File folder	
Sources	03-Sep-12 7:24 PM	File folder	
Cluster - 1x MB_60(upright)_85DF.xglc	24-May-10 12:37 P...	XGLC File	6 KB
Cluster - 2x 60.xglc	13-May-10 4:42 PM	XGLC File	3 KB
Cluster - 2x MB_40MH.xglc	18-May-10 8:26 AM	XGLC File	7 KB
Cluster - 2x MB_60_64DF.xglc	18-May-10 8:18 AM	XGLC File	10 KB
Cluster - 2x MB_64MH.xglc	18-May-10 8:28 AM	XGLC File	7 KB
Cluster - 2x MB_95MH.xglc	18-May-10 8:41 AM	XGLC File	7 KB
Cluster - 2x MB_95MH_64DF.xglc	18-May-10 8:44 AM	XGLC File	10 KB
Cluster - 2x MB_MB_40MH_40DF.xglc	18-May-10 8:49 AM	XGLC File	14 KB
Cluster - 2x MB_MB_64MH_64DF.xglc	18-May-10 8:53 AM	XGLC File	14 KB
Cluster - 3x MB_40MH (Best, simple).xglc	18-May-10 9:01 AM	XGLC File	14 KB
DF Box Angled Up.xglc	11-Feb-10 5:12 PM	XGLC File	1 KB
InfoComm - 3x MB_MB_40MH_40DF.xglc	27-May-10 11:55 ...	XGLC File	21 KB
v08 Cluster - 1x MB_40DF.xglc	13-May-10 2:39 PM	XGLC File	4 KB
v08 Cluster - 1x MB_60_64DF.xglc	14-May-10 11:49 ...	XGLC File	5 KB
v08 Cluster - 1x MB_60_85DF.xglc	14-May-10 11:52 ...	XGLC File	5 KB
v08 Cluster - 1x MB_64DF.xglc	13-May-10 2:37 PM	XGLC File	4 KB
v08 Cluster - 1x MB_64MH_64DF.xglc	14-May-10 4:23 PM	XGLC File	5 KB

Figure 5-12 – Select the configuration file to open

5.1.7.3 XGLC File Format

The .xglc file is an ASCII text file that contains all of the settings entered for a GLL when it was saved. The file can be edited manually using any text editor. There is a specific hierarchy for the information in the XGLC file (Appendix B).

5.1.8 Selecting Presets

If one or more presets have been included in a GLL file they can be selected by clicking on FILE > SELECT PRESET. This will open a SELECT PRESET dialog box. Selecting the desired preset and clicking the OK button will configure the loudspeaker system exactly as the preset specifies. This preset is nothing more than an XGLC configuration file embedded in the GLL.

6 Tips & Recommendations

6.1 Folder Structure – Keeping Everything Together

It is suggested that all of the project files associated with an XGLL project be kept together in the same folder or in subfolders of the main project file folder. This way all of the items required for the project (image files, drawing files, authorization files, configuration files, etc.) stay together in a centralized location. One possible folder structure is listed below.

```
Main Project Folder
     Drawings
     Filters
     Sources
```

6.2 Relative File Locations & Transferring Projects

It is highly recommended to always use the RELATIVE checkbox when saving or loading project files within the XGLL project. This keeps the paths stored in the XGLL project file relative to the folder location of the XGLL project file. If the relative checkbox is not checked for a particular project file the absolute path for that file is stored. If the XGLL project file and all the associated project files are copied to a different folder or a different computer the absolute paths will no longer be correct and the XGLL project file will not open in ***SpeakerLab***. If this happens it is possible to manually edit the XGLL project file using a text editor to change the path locations so that it can be opened by ***SpeakerLab.***

6.3 Compare GLL Calculation To Measured Response

When working with multi-source GLLs it is recommended to compare the on-axis measured response of the loudspeaker system with the on-axis response calculated by the GLL for a single box. This can help to assure that signal routing within the GLL, source positions, levels, etc. are correct in the GLL.

6.4 Modeling Powered Loudspeakers

There are several different ways to handle the modeling of power loudspeaker systems in ***SpeakerLab***. The two most advisable would be to either match the sensitivity of the GLL to the sensitivity of the drivers used in the loudspeaker or to match the maximum input level of the GLL to the maximum input level of the powered loudspeaker system. It is not possible to do both with powered loudspeakers in ***SpeakerLab*** because of where the gain for the power amplifier(s) must be located in the signal flow through the system model. Whichever of these methods one chooses to use will depend on how one wants the GLL to present its sensitivity and maximum input level to its users.

6.4.1 Matched Sensitivity Method

For this method the gain of the power amplifier(s) used in the loudspeaker system are ***not*** accounted for in the GLL. The power amplifier(s) are seen as being before the input to the GLL. This yields a level for the system sensitivity, at 2.83 V, that should be relatively close to the level of one of the drivers used for the system. The maximum SPL should be as accurate as the data used in the GSS files. However, the maximum input level to the GLL will be far greater than the line-level input the powered loudspeaker system can take.

6.4.1.1 Multi-way Loudspeaker Modeled as a Single-Source GLL

The transfer function of the front-end filtering and EQ used in the loudspeaker system should be examined. The pass band filtering and any EQ in the signal path after the crossover should be examined also. A frequency region that is at or close to 0 dB (no boost or cut) through the combination of filtering should be identified.

Setup a measurement system to drive the input of the loudspeaker system with a swept-sine wave so that the amplifier being driven by these combined transfer functions outputs an rms voltage of 2.83 V at the frequency where the combined transfer functions are at 0 dB. Measure the response of the loudspeaker system using a swept-sine wave at this same fixed level from the measurement system to get the sensitivity measurement to use in the GSS file.

6.4.1.2 Multi-way Loudspeaker Modeled as a Multi-Source GLL

The transfer function for each pass band of the loudspeaker system should be examined separately. The crossover filtering and any EQ in the signal path after the crossover should reveal a frequency region that is at or close to 0 dB (no boost or cut) through the filtering. Set the gain in the filter bank for each pass band so that the transfer function of each pass band is at 0 dB at the appropriate frequency.

6.4.2 Matched Maximum Input Level Method

For this method the gain of the power amplifier(s) used in the loudspeaker system ***are*** accounted for in the GLL. The power amplifier(s) are essentially internal to the GLL, just as with the real loudspeaker. This yields a very high SPL for the system sensitivity, which is calculated for a 2.83 V input to the system. Typically an rms input of 2.83 V is close to, or in excess of, the clipping point of the input for many powered loudspeaker systems.

The maximum SPL should be as accurate as the data used in the GSS files. The maximum input level to the GLL should also be fairly accurate, when compared to the maximum input level of the powered loudspeaker itself, in relation to the drivers reaching their thermal limits or the voltage limit of the amplifier driving them.

6.4.2.1 Multi-way Loudspeaker Modeled as a Single-Source GLL

It is only possible to accurately use the matched maximum input level method to model a multi-way loudspeaker as a single-source GLL if there is a single power amplifier in the loudspeaker system. If multiple amplifiers are used (i.e. biamp or triamp operation) in the loudspeaker then a multi-source GLL should be used for the model.

Even when a single power amplifier is used with a multi-way loudspeaker the maximum input level may change based on the spectral content of the input signal. A single-source GLL is less accurate in calculating this since the maximum input level of each pass band is unknown to the GLL.

For the maximum input voltage (MIV) of the GSS file for the loudspeaker system, enter the lower of the maximum input voltage (MIV) for the loudspeaker system or the maximum rms voltage of the power amplifier driving the loudspeaker system.

In the GAIN box of the filter bank for the XGLL project, enter the level gain of the power amplifier, as well as any other gain present in the signal chain before the amplifier.

6.4.2.2 Multi-way Loudspeaker Modeled as a Multi-Source GLL

In the GSS file for each driver (or pass band) of the loudspeaker system, enter the lower of the maximum input voltage (MIV) for that driver (pass band) or the maximum rms voltage of the power amplifier driving that driver (pass band) of the loudspeaker system.

In the GAIN box of the filter bank for each pass band start off with the gain set to the level gain of the power amplifier for that pass band, as well as any other gain present in the signal chain that affects all pass bands of the loudspeaker system.

Offset the gain for the filter bank of each pass band by the broadband gain used for each pass band in the loudspeaker system. This will account for any level differences required between pass bands to yield the desired frequency response.

7 Case Studies

7.1 Optimization of a Two-Way Loudspeaker Crossover

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.

7.1.1 Optimization Steps

A two-way loudspeaker system 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. The directivity balloons and on-axis sensitivity of the woofer (LF) and the horn (HF) are measured in the enclosure. A GLL model is created with this measured data of each individual source. This GLL is then used to simulate the application of different crossover and equalization filters to these sources. The resulting system response is then calculated using the GLL.

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 is applied to the LF section and constant directivity horn EQ to the HF section. Fourth order Linkwitz-Riley low pass and high pass filters at 1.6 kHz are applied to the LF and HF sections, respectively. The effects of this filtering are shown in Figure 7-1, Figure 7-2, and Figure 7-3.

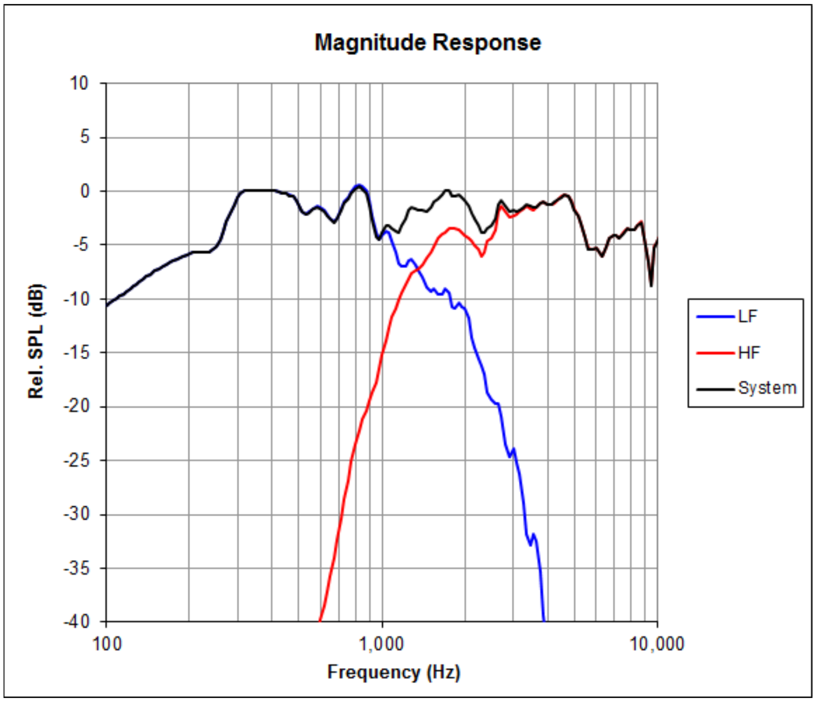


Figure 7-1 – Normalized magnitude response of woofer, horn, and combined system using initial Linkwitz-Riley filtering

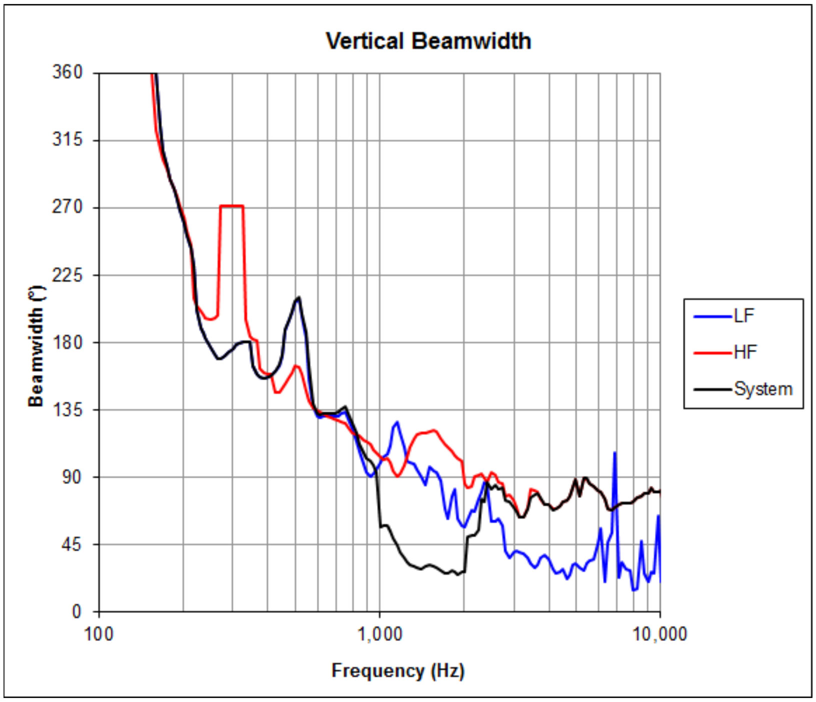


Figure 7-2 – Vertical beamwidth of woofer, horn, and combined system using initial Linkwitz-Riley filtering (1/24 octave)

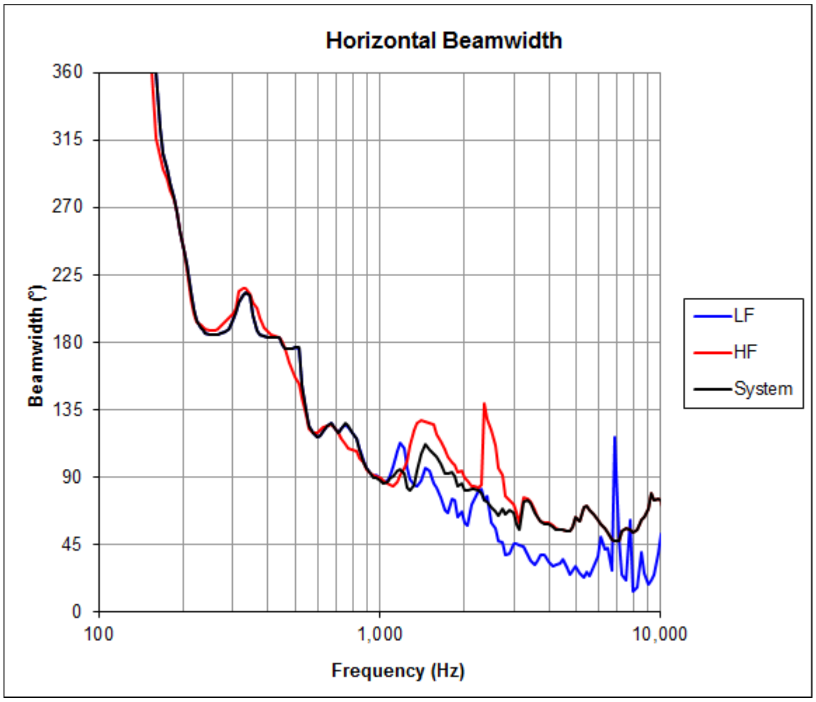


Figure 7-3 – Horizontal beamwidth of woofer, horn, and combined system using initial Linkwitz-Riley filtering (1/24 octave)

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 1 – 2 kHz shown in Figure 7-2. 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 low pass and high pass filter functions can be chosen. These new optimized filter functions better complement the acoustic response of the transducers to yield a better overall system response as shown in Figure 7-4, Figure 7-5, and Figure 7-6. The new filters are a fourth order Butterworth low pass at 1.6 kHz and a fifth order Butterworth high pass at 2.0 kHz. Some additional minor equalization is also employed for each of the pass bands.

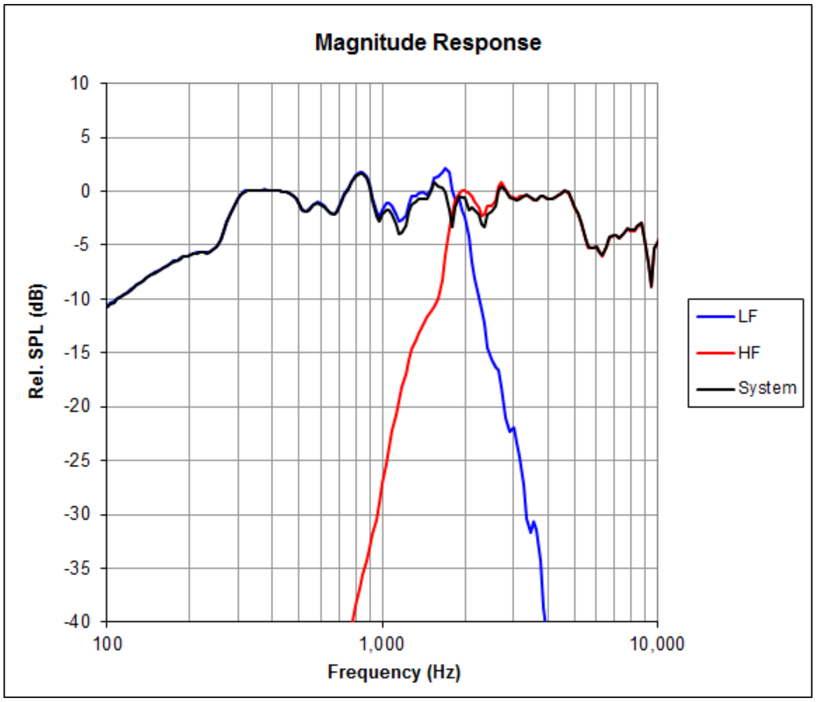


Figure 7-4 – Normalized magnitude response of woofer, horn, and combined system using optimized filtering

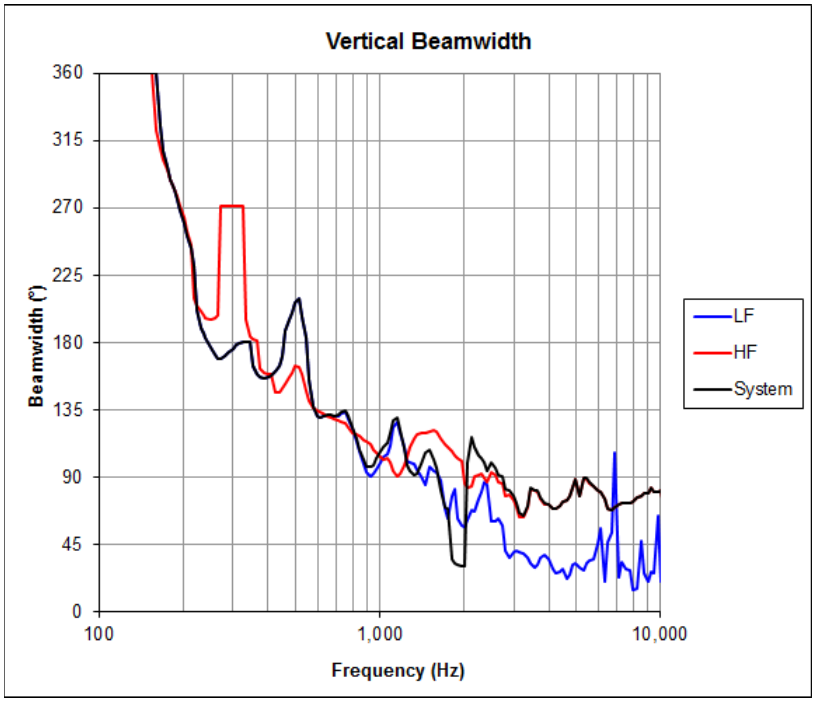


Figure 7-5 – Vertical beamwidth of woofer, horn, and combined system using optimized filtering (1/24 octave)

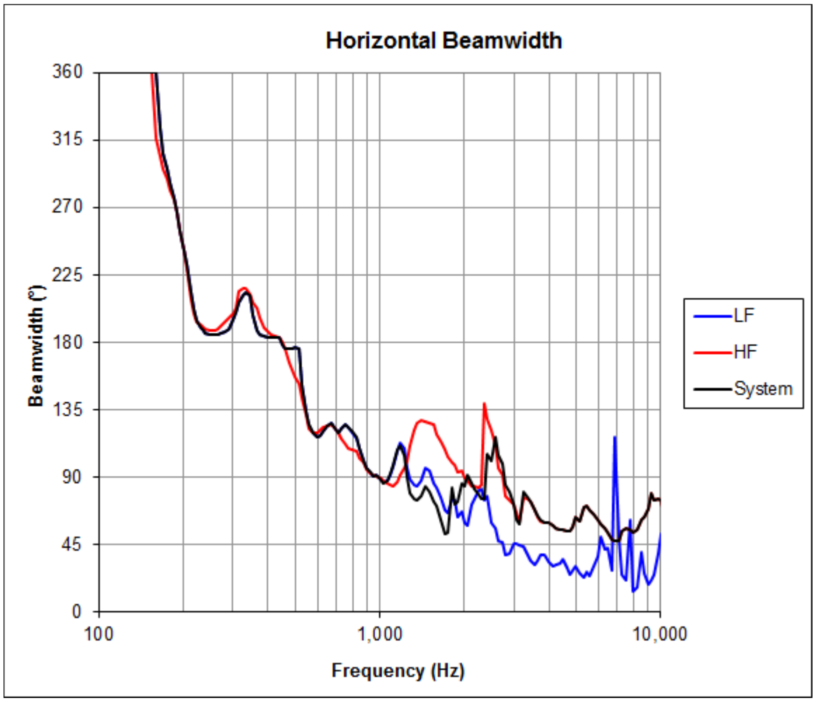


Figure 7-6 – Horizontal beamwidth of woofer, horn, and combined system using optimized filtering (1/24 octave)

The vertical beamwidth through the crossover region is now much more consistent. The bandwidth over which the coverage angle decreases has been greatly reduced; 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.

A comparison of the loudspeaker system's response with each set of filters is shown in Figure 7-7 through Figure 7-11. 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.

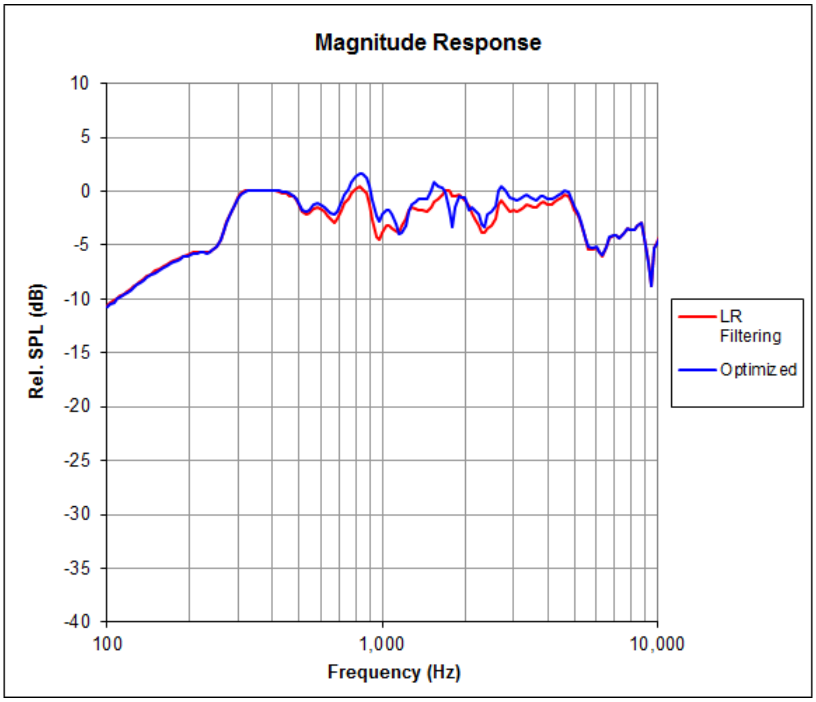


Figure 7-7 – Normalized magnitude response of the systems using the Linkwitz-Riley filters and the optimized filtering

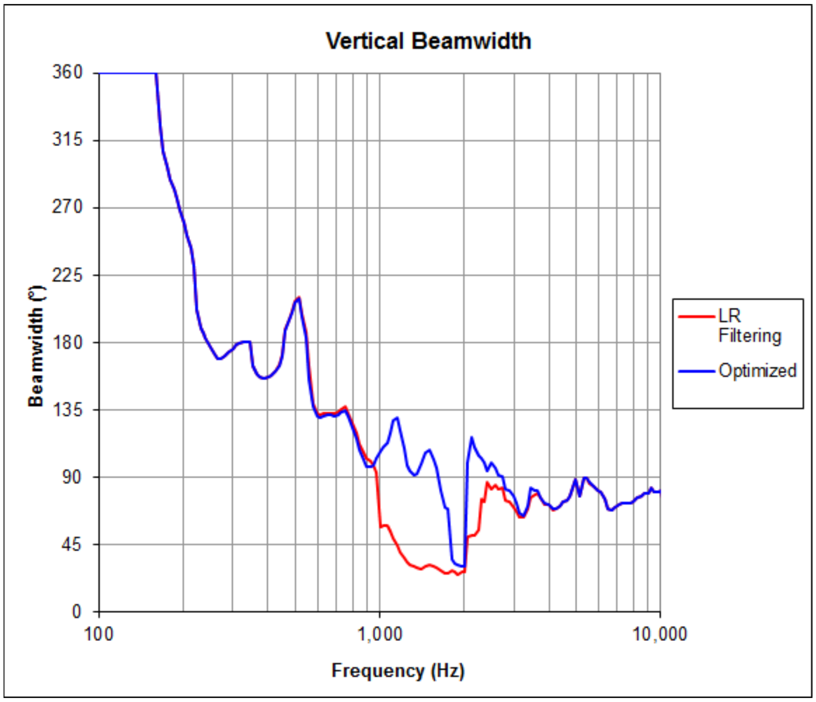


Figure 7-8 – Vertical beamwidth of the systems using the Linkwitz-Riley filters and the optimized filtering (1/24 octave)

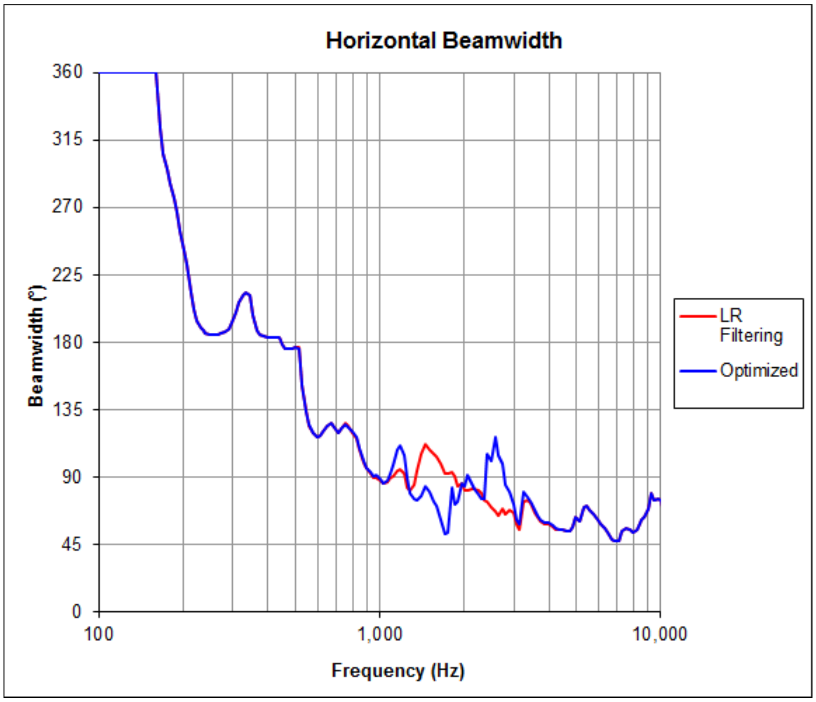


Figure 7-9 – Horizontal beamwidth of the systems using the Linkwitz-Riley filters and the optimized filtering (1/24 octave)

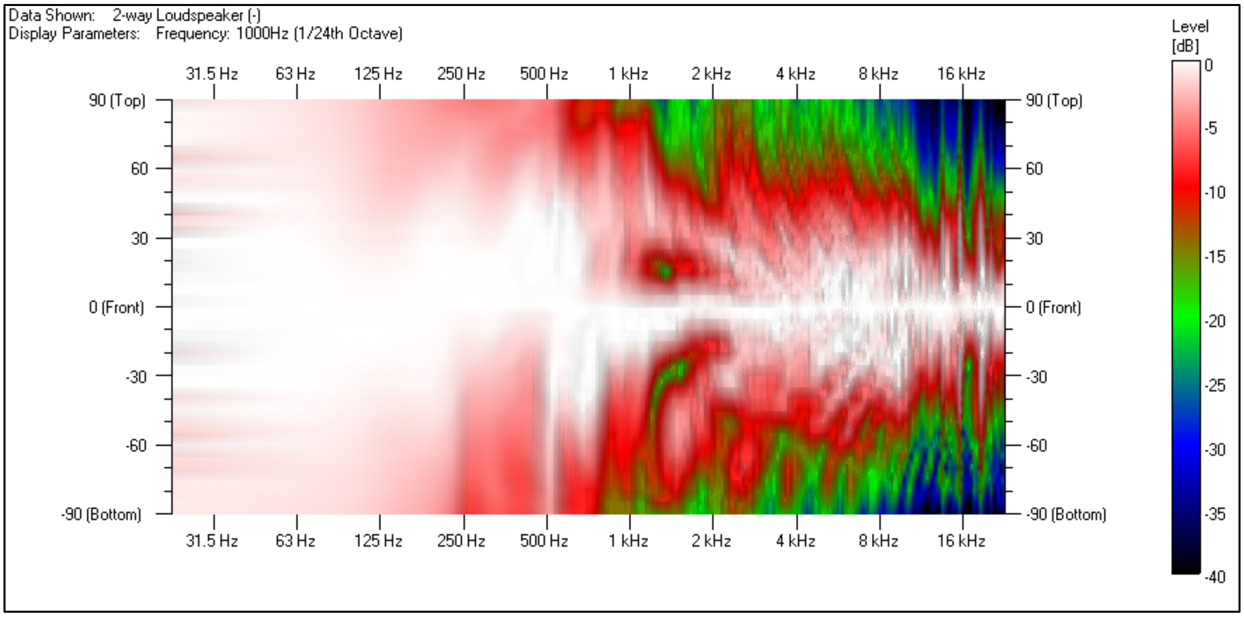


Figure 7-10 – Vertical directivity map of system response using initial Linkwitz-Riley filtering (1/24 octave)

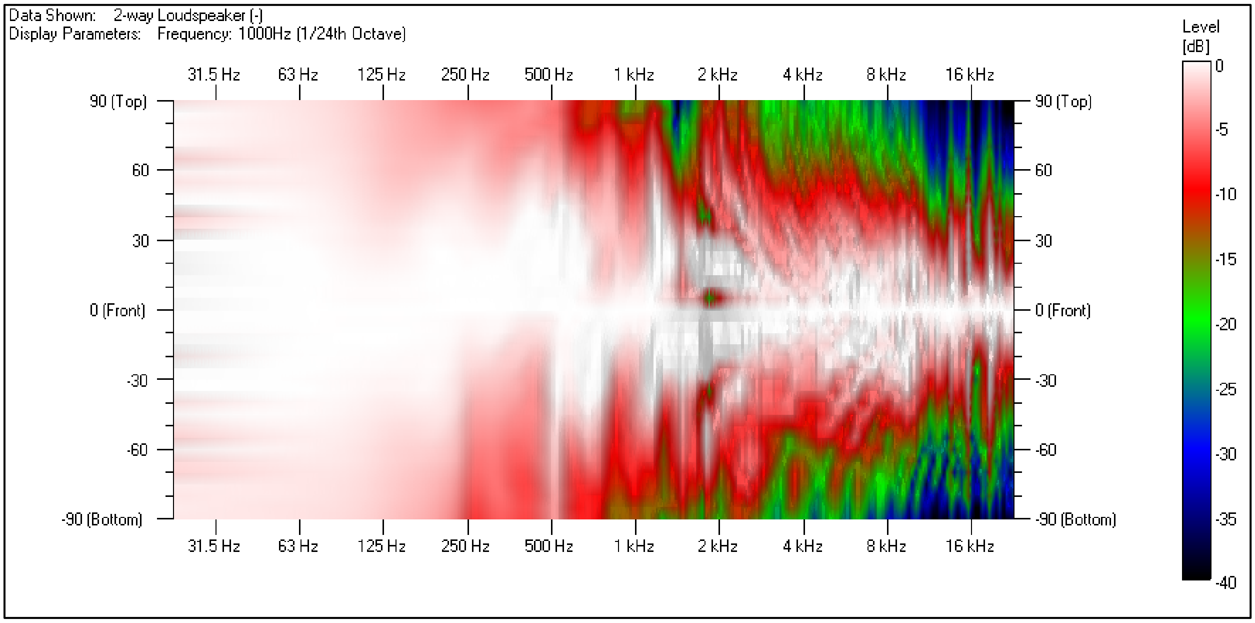


Figure 7-11 – Vertical directivity map of system response using optimized asymmetrical filtering (1/24 octave)

7.1.2 Measured Comparison

To verify the prediction, the optimized directivity filters (asymmetrical Butterworth) are implemented on a readily available DSP unit (Biamp Audia). This DSP is used to drive two identical amplifier channels that power the LF and HF sections of the loudspeaker system. The polar plots of the GLL and the measured system are shown in Figure 7-12 and Figure 7-13. These are at the crossover frequency of 1.9 kHz. This shows that there is good agreement between the prediction and the measurement.

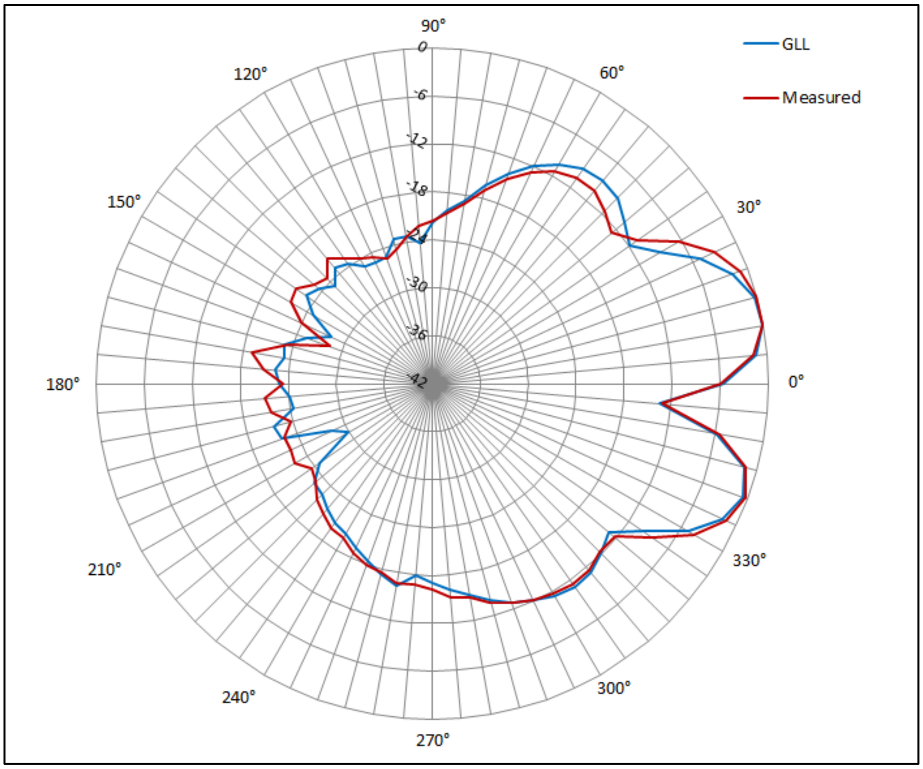


Figure 7-12 – Prediction and measurement for system response using optimized asymmetrical filtering, vertical polar at 1.9 kHz (1/3 octave bandwidth)

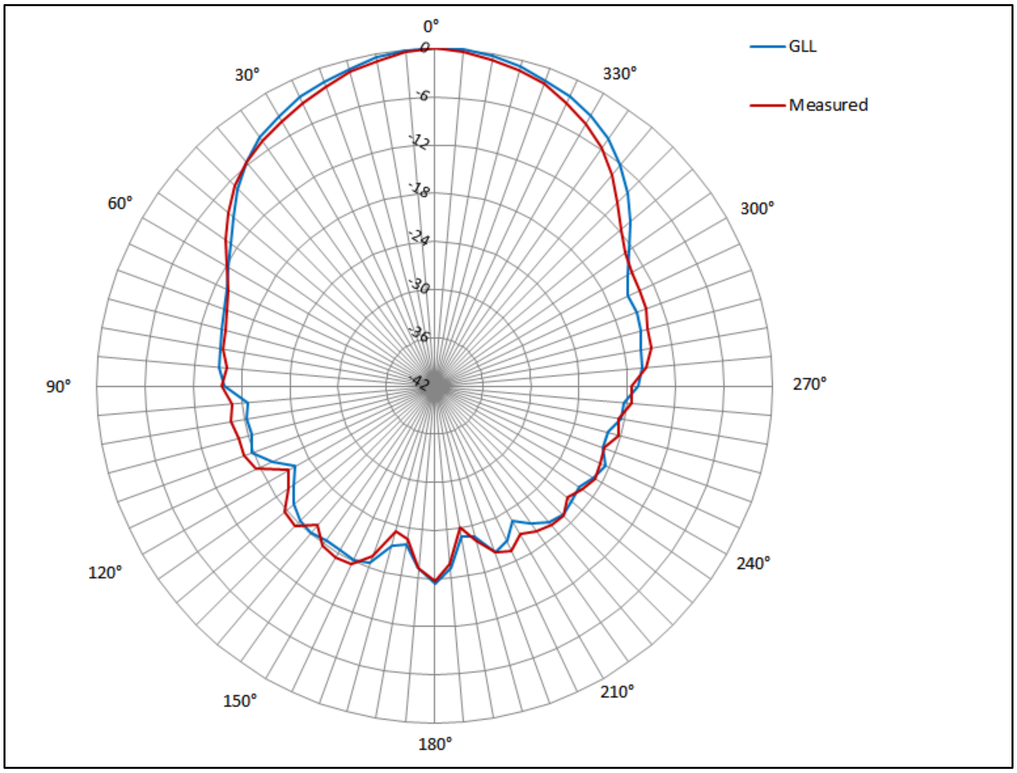


Figure 7-13 – Prediction and measurement for system response using optimized asymmetrical filtering, horizontal polar at 1.9 kHz (1/3 octave bandwidth)

7.1.3 Passive Filters

It is also quite possible to use a GLL model of a loudspeaker system 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 (Figure 7-14 and Figure 7-15). They can also be opened in measurement software for comparison to the measured response of the passive circuit.

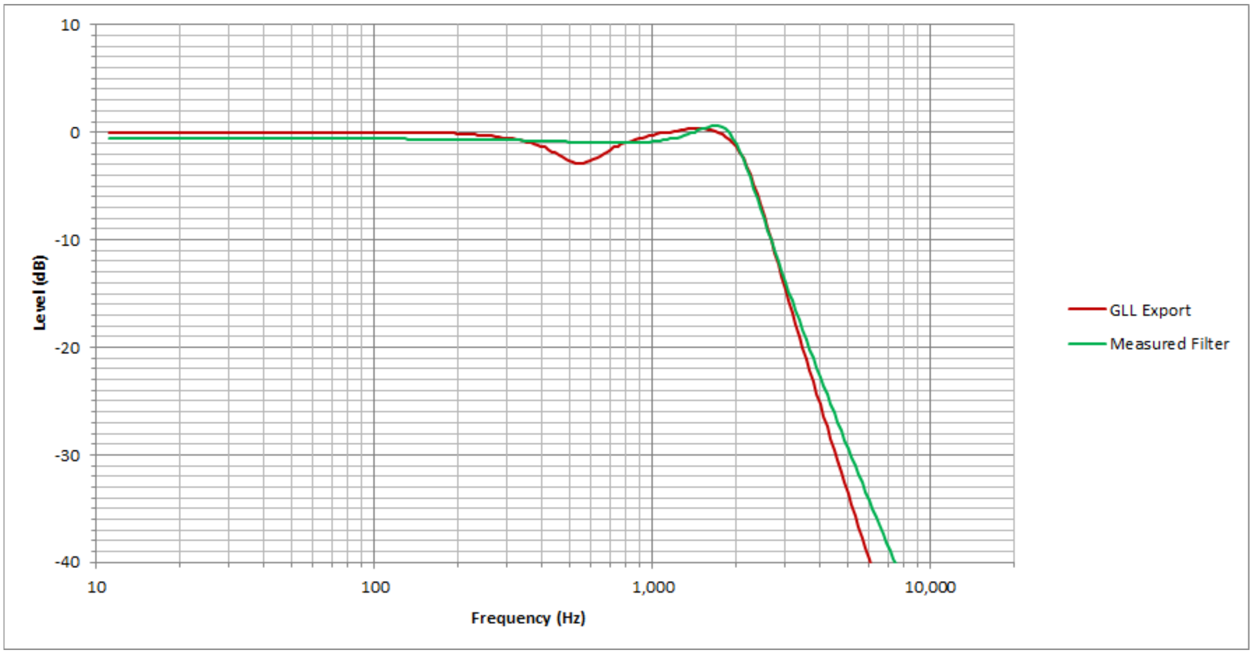


Figure 7-14 – Low pass filter transfer function for GLL export and modeled/measured passive circuit

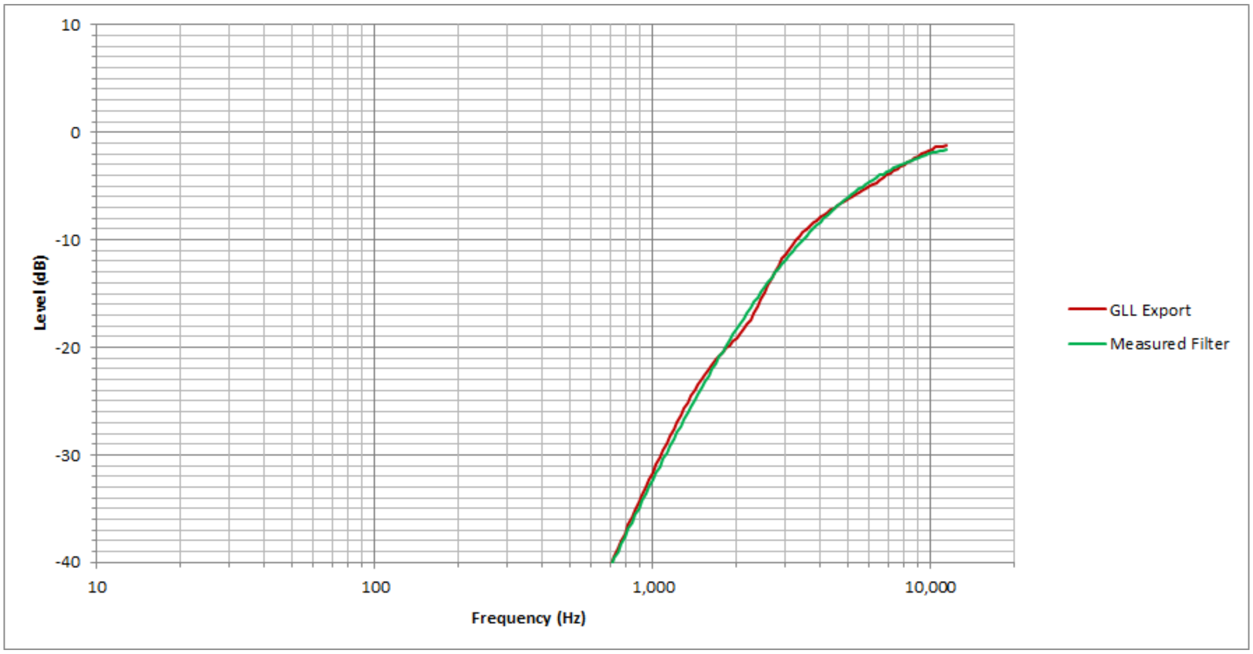


Figure 7-15 – High pass filter transfer function for GLL export and modeled/measured passive circuit

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 maps using the initial GLL filters and the final passive filters is shown in Figure 7-16 and Figure 7-17. 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.

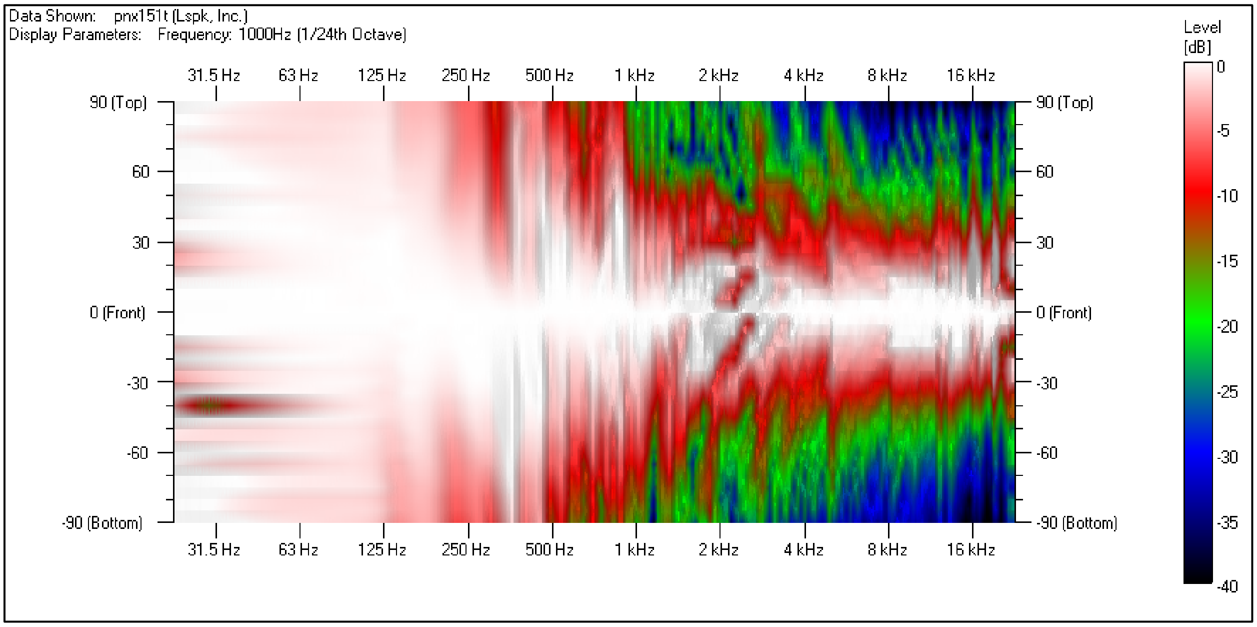


Figure 7-16 – Vertical directivity map of system response using GLL filters to develop a passive crossover

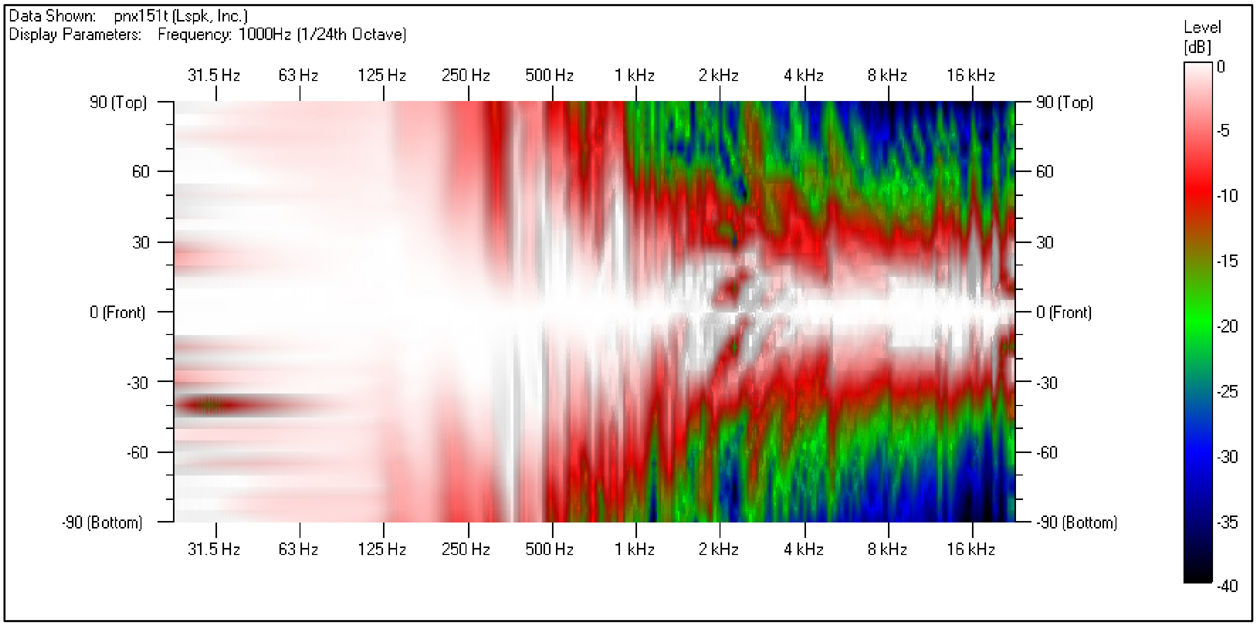


Figure 7-17 – Vertical directivity map of system response using measured passive filters

7.1.4 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.

8 Background

8.1 Definition of POR

When measuring the directivity balloon of a loudspeaker, or a pass band used within a loudspeaker, the point of rotation (POR) is the point around which the loudspeaker is rotated. The POR is important as this is the only stationary location relative to the loudspeaker during all of the measurements. As such, the POR will be used as the geometrical reference point for this data.

When the GSS file that uses the measured directivity balloon is placed into a GLL project, the location point for the GSS should be the POR for the measurement, not the actual location of the driver, relative to the cabinet drawing of the loudspeaker. This is a very important item. Failure to correctly specify the location of the GSS within the GLL project as the POR used during the measurements will result in errors in the GLL model of the loudspeaker.

It should be noted that the POR does not need to be centered on a particular driver. However, it should be within a certain distance of the acoustical center of the driver or pass band for the frequency region of interest radiated by that pass band [4]. See section 8.5.2 -Critical Frequency (Measurement Distance, POR & "Acoustic Center") for details about this.

In addition to the POR being used as the geometrical reference point for the measured data, it also needs to be the time origin for the measured data. Again, this is because the POR is the only stationary point during the measurement process.

The arrival time of the on-axis measurement used for the sensitivity data for the GSS should be referenced to the POR. In other words, the added delay in the arrival of the impulse response due to the time of flight between the POR and the measurement microphone should be removed from the measurement. If the POR is in front of a particular driver (Figure 8-1) then not all of the time of flight from the actual measurement might be removed (Figure 8-2).

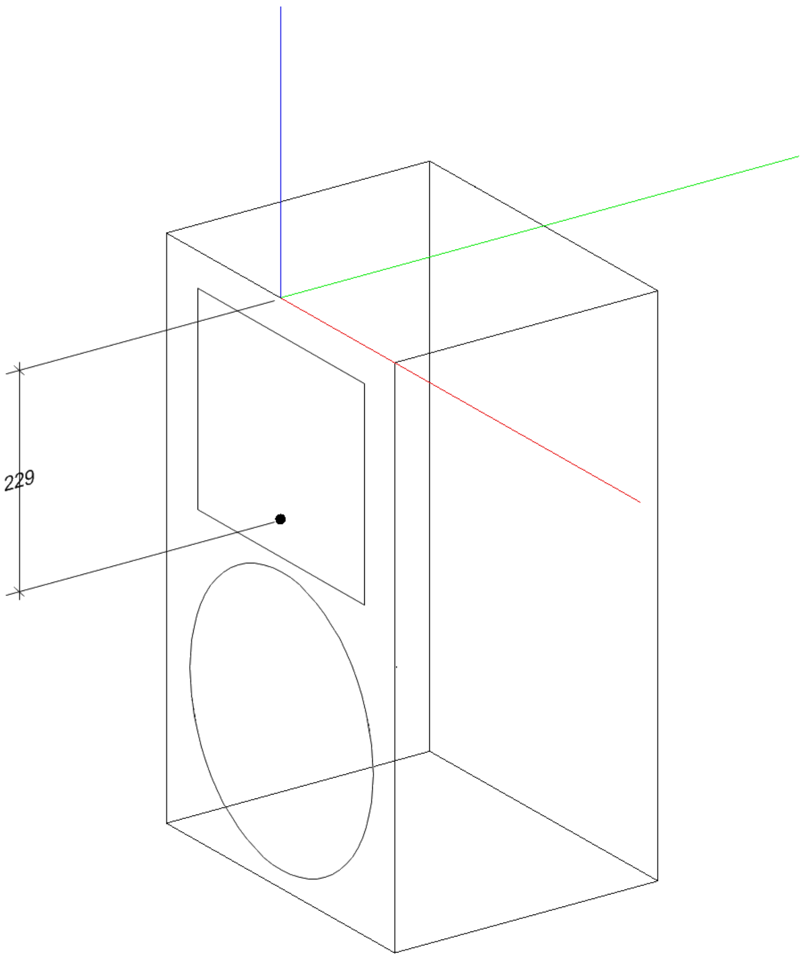


Figure 8-1 – Loudspeaker with the POR for both the LF and HF sections 229 mm below the top of the cabinet, centered left-to-right on the front of the baffle

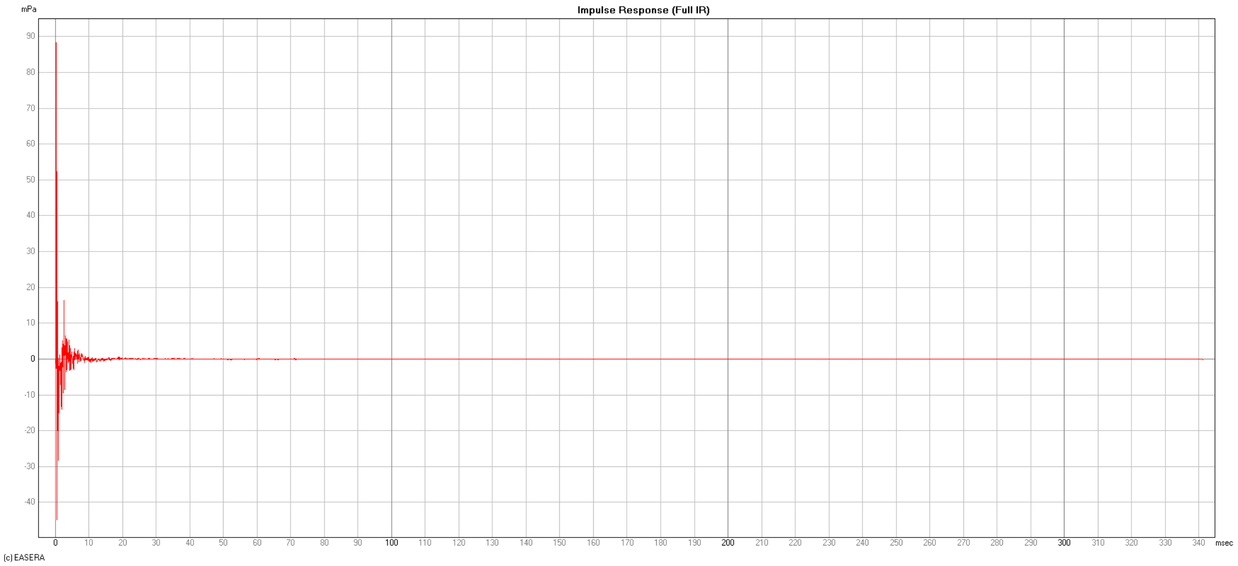


Figure 8-2 – Impulse response of woofer measured at 4 m referenced to the POR shown in Figure 8-1 in front of the driver

On the other hand, if the POR is behind a particular driver, as shown in Figure 8-3, then more time than the time of flight from the actual measurement might be removed. For most FFT-based measurement systems this would lead to the impulse response wrapping around to the end of the time record for the measurement (Figure 8-4). For a case such as this one must be extremely careful about how the impulse response windowing is setup. It is recommended that the windowing be done before the time of flight is removed. This way there should be very little chance of the window attenuating the desired part of the impulse response that has wrapped to the end of the time record.

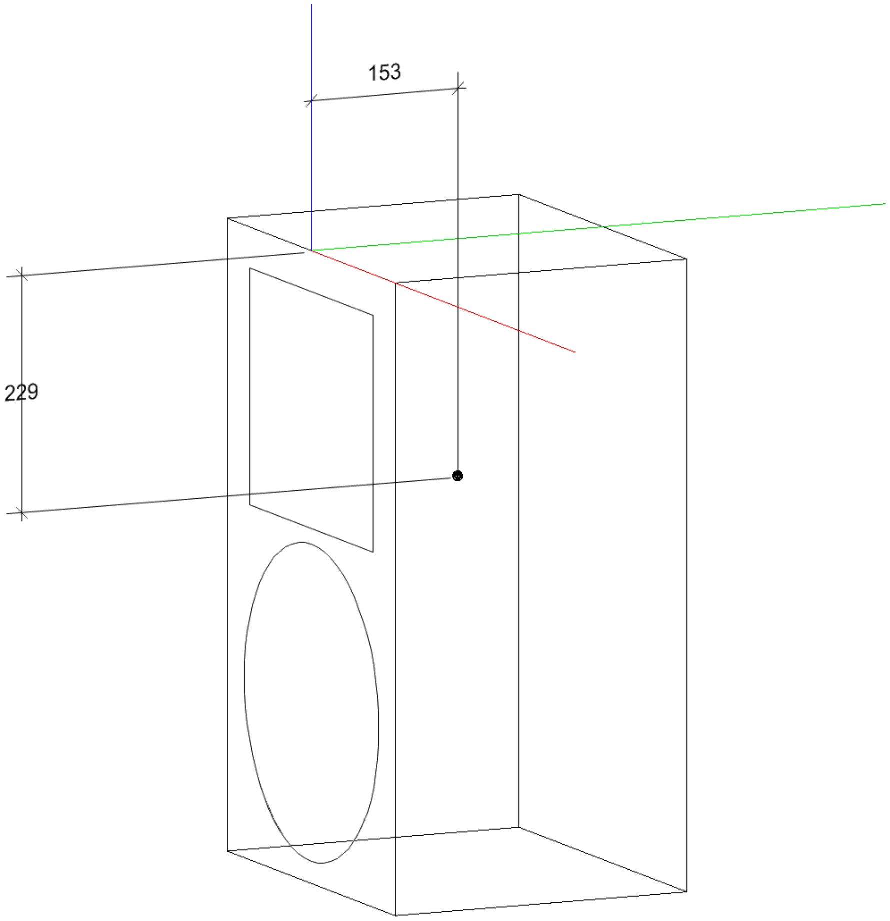


Figure 8-3 – Loudspeaker with the POR located 229 mm below the top and 153 mm behind the front of the cabinet

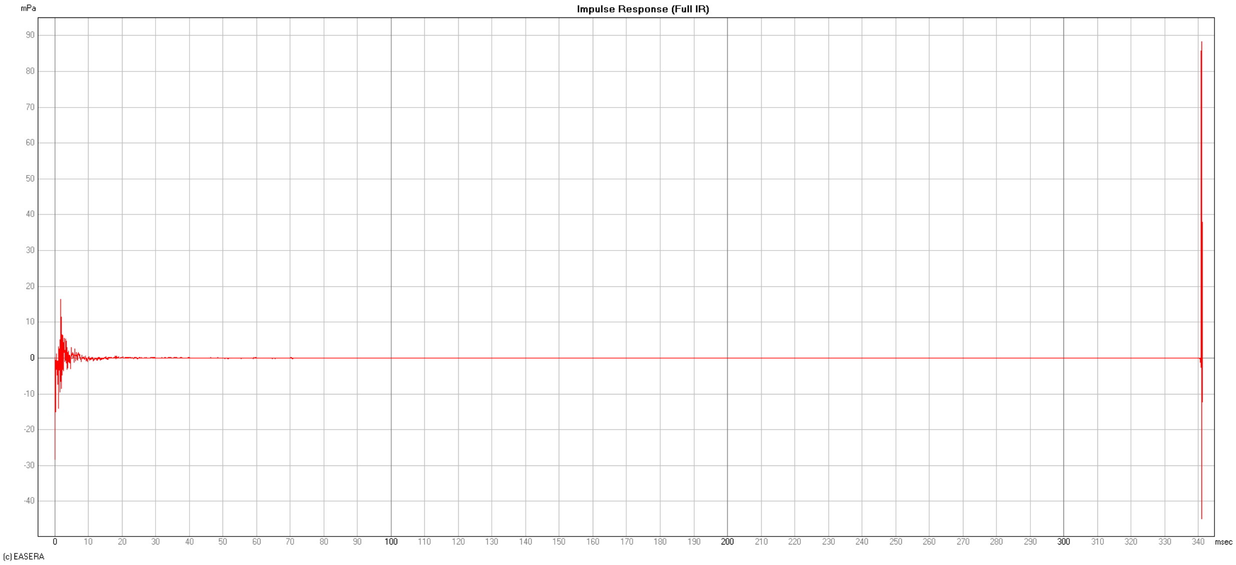


Figure 8-4 – Impulse response of woofer measured at 4 m referenced to the POR shown in Figure 8-3 behind the driver

Failure to remove the time of flight delay from the sensitivity measurement so that the arrival is properly referenced to the POR is essentially the same as adding delay to the measurement with a DSP. As long as all of the data used for all of the GSS files in the same GLL project have the same amount of delay, the result should be a correct response calculation, but with additional delay (as if it was applied to the input of the entire loudspeaker system).

However, if each of the GSS files are not referenced to their respective POR or have different amounts of delay the calculated response of the loudspeaker model will not match the real loudspeaker.

8.2 Measurements and Uncertainties

When measuring loudspeaker systems to acquire data for subsequent use in modeling it is important to take reasonable precautions to minimize the possibility of errors in the data. Errors in the amplitude and phase response can be introduced by many different sources. The guidelines below are intended to help minimize the occurrence of errors. Since it is impossible to foresee all possible measurement scenarios, following these guidelines does not guarantee good measurement data. However, not following these guidelines will almost certainly guarantee questionable measurement data.

8.2.1 Measurement Setup

We should strongly emphasize that measurements for use in a GLL and SpeakerLab must be taken on the loudspeaker system as a physical whole, and not on separate, individual pieces of the loudspeaker system. This is to say that one should not attempt to measure a high frequency horn having no loudspeaker enclosure with the expectation that this data can be combined with measurement data of a woofer to yield the correct system response. The loudspeaker enclosure, and the external boundary conditions it presents to the acoustical elements (i.e. HF horn and woofer), will affect the response of these elements both on and off axis. Measurements should always be carried out using the full loudspeaker enclosure in which the separate devices are intended to be used.

This should not be confused with separately measuring the response of an HF horn and a woofer which are both installed in their intended enclosure. This is certainly possible and the recommended method for use in a multi-source GLL.

8.2.1.1 Measurement Distance Requirements

In order for the measurement data to be valid for use in modeling the measurements must be made in the far-field of the device under test (DUT). The measurement distance also affects the accuracy of the measured complex data as detailed in Section 8.5.2 – Critical Frequency (Measurement Distance, POR & “Acoustic Center”). At distances in the near-field and the transition region of the DUT, the frequency response of the DUT may change as a function of distance. This means the shape of the directivity balloon may also change within the near-field of the DUT.

The beginning of the far-field for a device is dependent on both the acoustical size of the device and the wavelength (frequency) of interest being produced by the device. The beginning of the far-field *d* can be estimated using Equation 8-1 [10], where *S* is the surface area of the DUT, *f* is the highest frequency of interest being radiated by the DUT, and *c* is the speed of sound.

$$d = S * f/c$$

Equation 8-1

When determining the surface area for the DUT consideration should be given to not only the active radiating element (a woofer for example) but also to the size of the front baffle or the enclosure in which it is housed. This diffraction from the edges of the enclosure may have a significant contribution to the directivity of the DUT within its intended pass band of operation.

The beginning of the far-field for a line source is governed by a slightly different equation. For a line source the distance *d* can be estimated by Equation 8-2, where *h* is the height of the line source [11].

$$d = h^2 * f/2c$$

Equation 8-2

8.2.1.2 Point Of Rotation Location Requirements

The point of rotation (POR) used for measuring the directivity balloon of the DUT should not be chosen arbitrarily. When measuring complex data for use in modeling, the location of the POR relative to the so-called acoustical center of the DUT is important [4]. The maximum distance *r* in 3D space between these locations is given by Equation 8-3, where *d* is the distance from the POR to the measurement microphone, f_{crit} is the critical frequency for the distances *r* and *d*, and *c* is the speed of sound.

$$r = \sqrt{c * d \,/\, (4 * f_{crit})}$$

Equation 8-3

At this distance the maximum error in the phase response could be as much as 45° at the upper frequency limit f_{crit}. It is recommend that the value of f_{crit} used for these calculations be at least one octave above the highest frequency of interest being radiated by the DUT. Please see section 8.5.2 for additional information.

8.2.1.3 Environmental Conditions

It is important that the measurement environment remains reasonably unchanged throughout the duration of the measurements. The ability of SpeakerLab to normalize each half-plane of measurement data can help to reduce the effects of inconsistencies due to temperature change during the measurements. However, care should be taken to ensure that changes of temperature and/or temperature gradients are minimized [12].

8.2.1.4 Reflections, Windowing, and Frequency Resolution

When measuring loudspeakers and not using an anechoic chamber, the size of the measurement room is important. It directly affects the allowable frequency resolution of the measured data as well as the low frequency limit to which the data is valid. The impulse response (IR) measurements of the DUT should be windowed appropriately in order to eliminate the energy from reflection of sound off of the wall, floor, and ceiling from the data used for the GLL. The arrival time difference between the direct sound from the DUT and the first reflection that must be windowed out will determine the potential frequency resolution of the data.

As an example, if the direct arrival of sound from the DUT occurs at 12 ms and the first reflection occurs at 22 ms, a rectangular window truncating the data at 22 ms and beyond will yield a frequency resolution of 100 Hz. If a window function other than rectangular is used the data will suffer from a decrease in frequency resolution (larger number). As an example, if a Blackman window is used instead of a rectangular window in the example above the frequency resolution will be approximately 300 Hz. If an Hann window is used the frequency resolution will be approximately 200 Hz.

The decreased frequency resolution of these and other windowing functions is due to the shape of the window attenuating more energy in the time domain than a rectangular window does.

It should also be mentioned that the time span of the measurement data (FFT block size) should be selected in relation to the size of the measurement room. When measuring in anechoic rooms FFT lengths on the order of 250 ms can work well. If measurements are being made in reflective environments the FFT length may need to be larger. This is necessary as the reflections in the room will need to decay sufficiently prior to the beginning of the subsequent FFT block. If the reflections have not decayed they will contaminate the data in the subsequent FFT block(s). Of course, the FFT length that is used will be a decisive factor in the overall time required for a directivity balloon measurement.

Appropriate selection of the measurement time span can help to keep the file size of each measurement optimized. With hundreds, or even thousands, of measurements required for each DUT the size of the data files stored on hard drives can become significant over time. When long FFT blocks are required it may be possible to truncate the impulse response to a time length after is has been windowed to eliminate the first reflection. It is not of much use to have an impulse response that is much longer than when the first reflection occurred. Past this time the windowed impulse response has a value of zero and contributes no additional information.

8.2.2 Mounting – Orientation & Direction of Rotation

Mounting of the DUT and its direction of rotation while conducting the measurements are important. If the DUT is not rotated correctly, or the file names not corrected for the measurement angles ***SpeakerLab*** expects, the data may be imported, and subsequently displayed, as a mirror image of what it should be.

The rotation of the DUT within each measurement plane (or half-plane), i.e. where the measurement direction moves along a meridian, should be clockwise as viewed from above the DUT (Figure 8-5). The rotation of the DUT between each measurement plane (or half-plane), i.e. where the measurement direction moves along a parallel, should be counter-clockwise as viewed from the rear of the DUT (Figure 8-6).

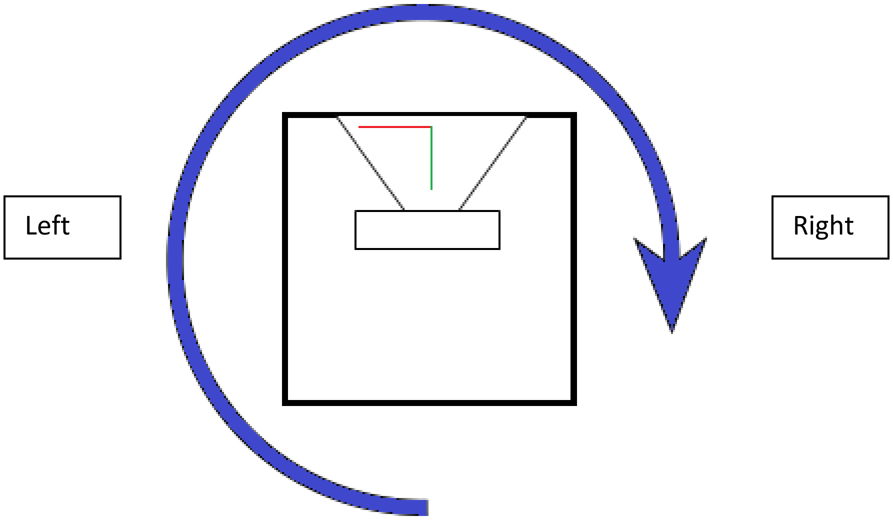


Figure 8-5 – Rotation of the DUT within each measurement plane (view from top)

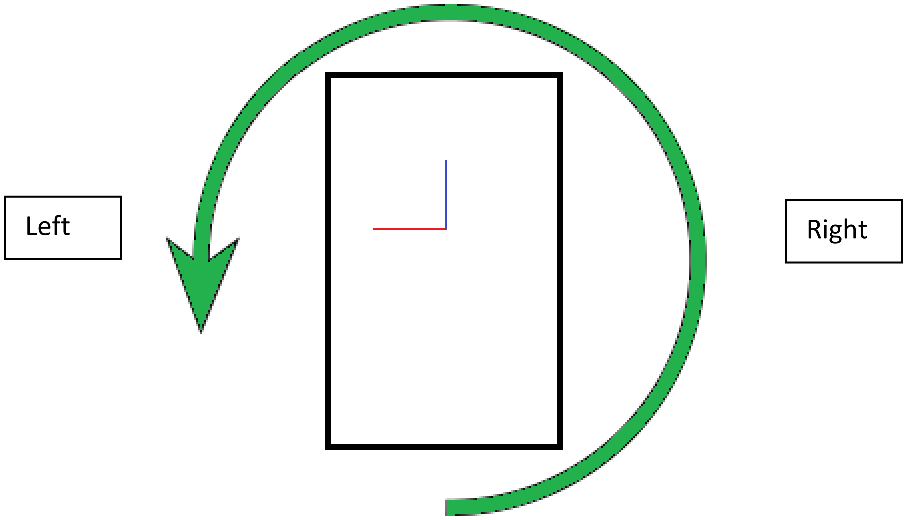


Figure 8-6 – Rotation of the DUT between each measurement plane (view from rear)

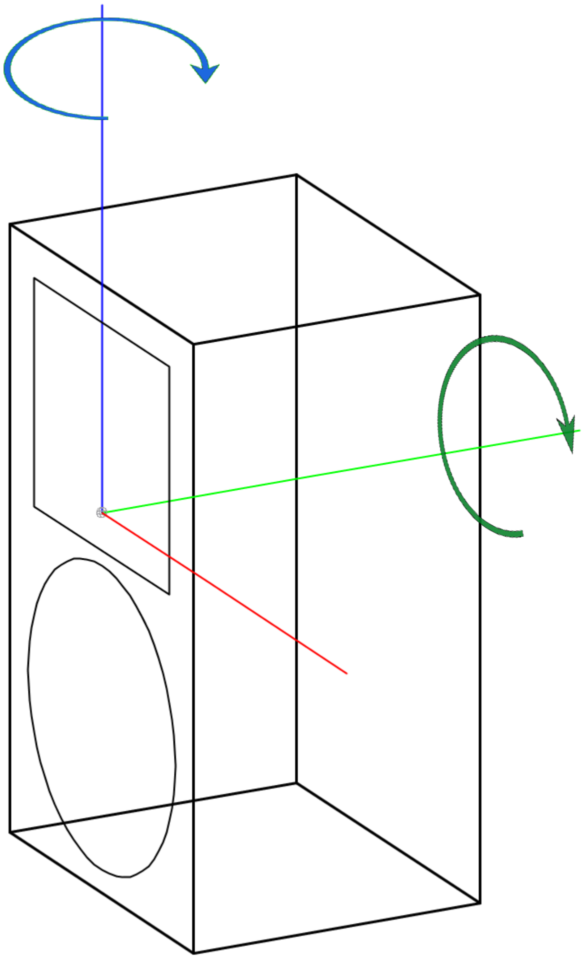


Figure 8-7 – Isometric view showing the rotation of the DUT

8.2.3 Sample Variation

The sample unit chosen for measurement can have an impact on how closely the simulations correlate with the actual performance. No two loudspeakers are identical. It should not be expected that simulations using a model based on measurement data from one sample of a loudspeaker will exactly match the performance of a different sample of the same type of loudspeaker.

The differences between the simulation and the actual performance of a randomly chosen sample can be minimized. This requires that the transfer function (frequency response) of the sample unit chosen for measurement be as close as possible in its performance to the statistical average of all production units. The responses should match closely across the frequency range for which the driver is used in the DUT. This includes the crossover region and slightly beyond. If the manufacturer of the transducers tracks this data it may be possible to acquire sample drivers that are within 1.0 dB (+/-0.5 dB) of the production average. If drivers within 1.0 dB of the production average cannot be acquired perhaps drivers that are within 2.0 dB (+/-1.0 dB) can be acquired. Loading the sample DUT with drivers that are as close to the production average as possible should help to assure the modeling data used for the simulations is not skewed too far towards one end of the tolerance margin for the production units.

If the DUT uses a passive crossover network the impedance response of the sample driver(s) should also be matched closely to the production average. This will help to insure that the passive crossover is loaded in a similar manner to the average of production units.

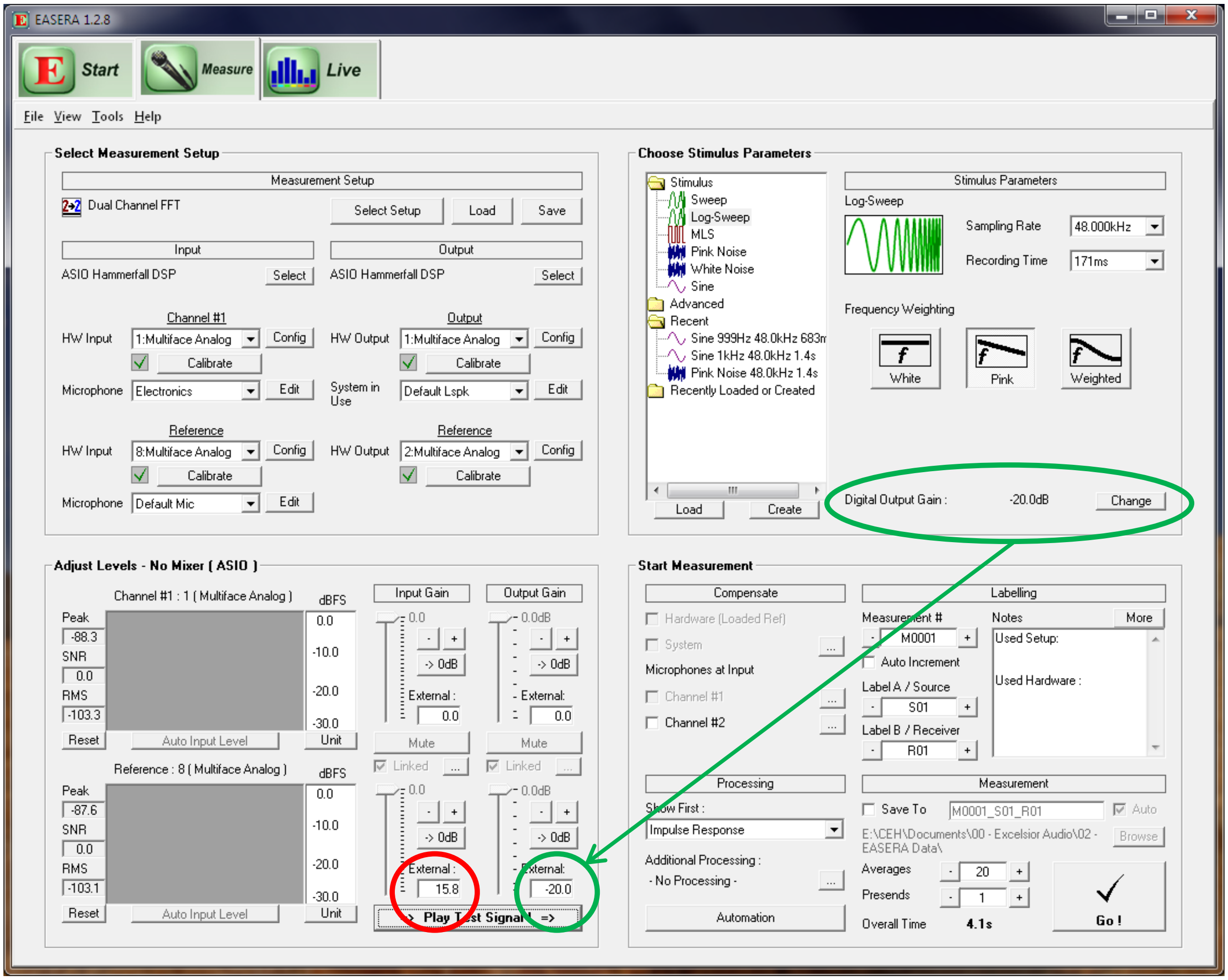
When measuring loudspeakers using passive crossovers, or measuring the passive crossovers themselves, the issue of parts variation should also be considered. It is recommended that the components used for the passive crossover to be measured are hand selected so that they are within 1% of their rated value. The transfer function of the crossover should then be close to the average of all production units.

When multiple loudspeaker units are used together, as in the case of line arrays, the issue of sample variation can itself average out in the performance of the whole array compared to an individual unit [13]. This will tend to have the performance of the array converge towards the statistical average of the units used in the array. This should hopefully be closer to the overall statistical average of production units. However, if the sample unit selected to be measured for the modeling data is not close to this statistical average systematic errors will be made and there will be a greater likelihood of differences between the predictions using modeling and the end result in reality.

8.2.4 Measuring Absolute SPL in EASERA

When measuring the directional response of loudspeakers it is not required to measure the absolute SPL. However, measuring the sensitivity using the on-axis position does require that the absolute SPL be measured. If ***AFMG***'s ***EASERA*** software is used for measuring the data it should be understood that EASERA does not inherently record the absolute SPL of a measurement. Instead, it automatically normalizes the SPL as if the DUT was excited with a Dirac stimulus signal of 1.0 volt (peak).

There is a trick that can be used to get EASERA to show the absolute SPL relative to a different reference voltage [14]. It requires that EASERA is setup to function in dual-channel FFT mode. The value for the EXTERNAL INPUT GAIN of the reference channel must be set to the equivalent maximum rms output level of the audio interface (sound card) being used. The value of the EXTERNAL OUTPUT GAIN for the reference channel must be set to the same value at the DIGITAL OUTPUT GAIN used for the measurement. For the settings shown in Figure 8-8 the maximum rms output voltage of the sound card is 6.2 V, which is equivalent to 15.8 dBV. The DIGITAL OUTPUT GAIN is set to -20 dB.



EASERA 1.2.8

Select Measurement Setup

Measurement Setup: Dual Channel FFT

Input: ASIO Hammerfall DSP

Output: ASIO Hammerfall DSP

Channel #1 – HW Input: 1:Multiface Analog ☑ Calibrate – Microphone: Electronics

Output – HW Output: 1:Multiface Analog ☑ Calibrate – System in Use: Default Lspk

Reference – HW Input: 8:Multiface Analog ☑ Calibrate – Microphone: Default Mic

Reference – HW Output: 2:Multiface Analog ☑ Calibrate

Choose Stimulus Parameters

Stimulus: Sweep, Log-Sweep, MLS, Pink Noise, White Noise, Sine; Advanced; Recent: Sine 999Hz 48.0kHz 683n, Sine 1kHz 48.0kHz 1.4s, Pink Noise 48.0kHz 1.4s; Recently Loaded or Created

Stimulus Parameters: Log-Sweep

Sampling Rate: 48.000kHz

Recording Time: 171ms

Frequency Weighting: White / Pink / Weighted

Digital Output Gain : -20.0dB Change

Adjust Levels - No Mixer (ASIO)

Channel #1 : 1 (Multiface Analog) – Peak -88.3, SNR 0.0, RMS -103.3 (dBFS 0.0, -10.0, -20.0, -30.0)

Reference : 8 (Multiface Analog) – Peak -87.6, SNR 0.0, RMS -103.1 (dBFS 0.0, -10.0, -20.0, -30.0)

Input Gain: 0.0, External : 0.0 / 0.0, External : 15.8

Output Gain: 0.0dB, External: 0.0 / 0.0dB, External: -20.0

Start Measurement

Compensate: □ Hardware (Loaded Ref) □ System

Microphones at Input: □ Channel #1 □ Channel #2

Labelling: Measurement # M0001, □ Auto Increment, Label A / Source S01, Label B / Receiver R01

Notes: Used Setup: Used Hardware :

Processing: Show First : Impulse Response; Additional Processing : - No Processing -

Measurement: □ Save To M0001_S01_R01 ☑ Auto; E:\CEH\Documents\00 - Excelsior Audio\02 - EASERA Data\

Averages 20; Presends 1; Overall Time 4.1s

Figure 8-8 – EASERA measurement set-up tab showing input & output external gain settings for measuring absolute levels

8.3 Maximum Input Voltage & SPL Measurement

The Maximum Input Voltage (MIV) should not be taken as the equivalent voltage for the rated power handling of a loudspeaker system or a driver. The MIV should be measured in a different manner [15]. Details on this measurement are given below. For full information on this please review the standard document AES2-2012, section 5.3.

A shaped noise signal is used to excite the DUT at a relatively low level and the frequency response of the DUT recorded. This frequency response is used as the reference to which subsequent frequency response measurements are compared. The voltage to the DUT is increased in defined increments at defined intervals. The frequency response is recorded at the end of each interval. When the frequency response has changed by 3 dB with no more than 1/3 octave smoothing, compared to the reference frequency response, the test is terminated. The rms input voltage that resulted in the 3 dB change is the MIV.

To summarize, the output of the DUT should be monitored closely during the test. The rms input voltage to the DUT that results in the frequency response of the DUT changing by 3 dB, compared to its frequency response at a low input level, is the MIV.

It should be mentioned that ***AFMG***'s ***SysTune*** software works very well for performing this type of measurement.

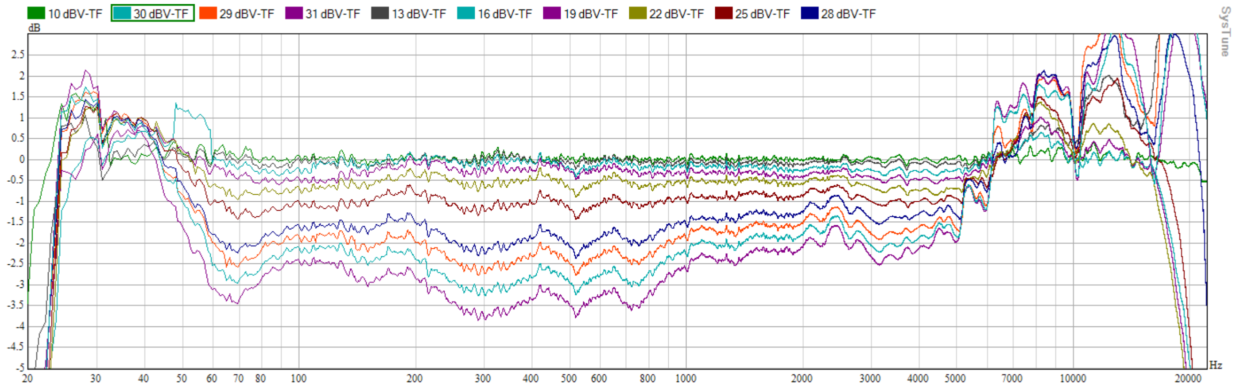


Figure 8-9 – Overlay of normalized response curves for an MIV test of a 12 inch woofer

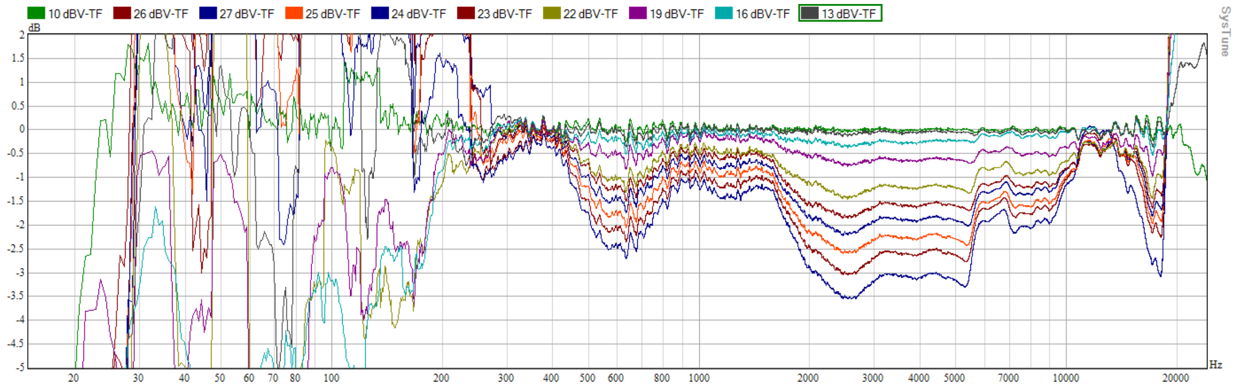


Figure 8-10 – Overlay of normalized response curves for an MIV test of a high frequency compression driver on a horn

8.4 Filter Measurement

It can be very useful to import the measured transfer function of a filter for use in a GLL. The measurement of these filters should be done carefully to assure that all of the necessary attributes of the filter are accurately captured in the measurement. A few specific examples are cited below. In all cases it is strongly recommended that a balanced input be used for these measurements. Use of an unbalanced (single-ended) input can result in measurement errors, and in some cases damage to the DUT and/or the measurement hardware.

8.4.1 Active Filters, DSP Latency, and Front Panel Setting

The measurement of active filters is fairly straight forward. However, when measuring DSP-based filters there are items that should be considered. All digital signal processor (DSP) units have some amount of latency. Latency is defined as the time required for the signal to transit from the input of the unit to the output of the unit. Typically this is taken with no processing applied to the signal other than passing it through the DSP unit itself.

If a particular DSP unit will always be used with the loudspeaker system being modeled by the GLL (such as the DSP being internal to the loudspeaker system) the latency should not be removed from the measurement. This is because the latency will always be a part of the real loudspeaker system.

If, on the other hand, the DSP unit being measured will not always be used with loudspeaker system being modeled by the GLL (such as the DSP being external to the loudspeaker system) the latency probably should be removed from the measurement. This is because this particular latency will not always be a part of the real loudspeaker system. The end user can measure the latency of the DSP they will be using and include this as a common delay if needed.

While this may seem a trivial item, DSP units with a large latency that are used for measurement but not in the loudspeaker can adversely affect the arrival time analysis in ***EASE*** and ***EASE Focus*** models using GLLs that don't have this latency removed.

It should be mentioned that the same parameters entered into different DSP units from different manufacturers may not yield the same transfer functions. Similarly, entering the parameters used in the GLL filters into a real DSP unit may not yield the same transfer functions. It is strongly recommend to always measure the transfer function of a DSP unit to verify the transfer function matches the intended transfer function.

When matching transfer function measurements it is important to verify not only the magnitude response in the frequency domain, but also the phase response. In some cases delay is applied to some pass bands. This delay will have no effect on the magnitude response of the filters. The delay can only be seen in the time domain or by inspecting the phase response in the frequency domain.

8.4.2 Measurement System Latency

If the measurement system being used is a digitally-based measurement system it will have a latency of its own. This latency should always be removed from all measurements. This includes loudspeaker measurements as well as filter measurements.

A loop-back (output connected to input) of the measurement system can usually indicate the latency of a measurement system.

8.4.3 Passive Filters, Loading, and Polarity

When measuring the transfer function of a passive filter it is critical that certain procedures be followed [7]. One of these is that the passive circuit must be loaded with the actual loudspeaker driver with which the filter will be used. Any other load on the passive circuit could result in changes to the transfer function of the circuit.

Another consideration is that the test leads for the input channel of the measurement system should ideally be connected to the input terminals of the loudspeaker driver that is fed by the output of the passive filter. It is important that the hot or positive (+) lead of the measurement input channel be connected to the positive (+) terminal of the loudspeaker driver and the cold or negative (-) lead of the measurement input channel be connected to the negative (-) terminal of the loudspeaker driver. Only by using this connection method will the polarity of the transfer function be maintained.

It is very important to understand that this connection method should only be used with a measurement system which has a balanced input circuit. The ground lead for the input is never used and should be left floating (Figure 8-11). If the ground lead of the input is connected to the negative (-) terminal of the loudspeaker driver and there is a polarity reversal in the passive filter for that output the result will be that the filter output will be shorted to ground through the connection of the measurement system input. This can result in severe damage to the measurement system.

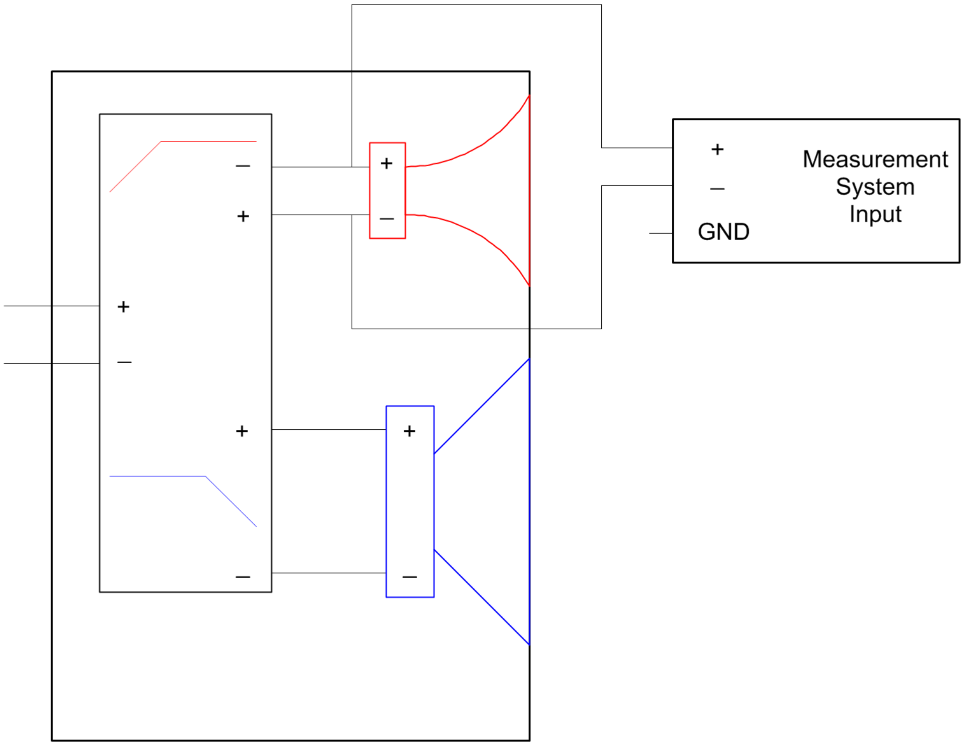


Figure 8-11 – Representative schematic of measuring the output of passive crossover filter

8.5 Measurement Considerations

There are several items that should always be considered when making measurements of loudspeaker systems. Some of these are detailed below.

8.5.1 Use of Complex Data

The use of complex data (i.e. magnitude and phase) for modeling loudspeaker systems and their interaction with other loudspeaker systems greatly improves the accuracy of the modeled results by at least on order of magnitude [4]. It is for this reason that complex data should always be measured and used for loudspeaker modeling.

8.5.2 Critical Frequency (Measurement Distance, POR & "Acoustic Center")

There exists a so-called "critical frequency" for measuring loudspeaker systems [4]. This is the frequency which denotes that data at higher frequencies may not be free of errors. Data at this frequency and higher should not be used for loudspeaker modeling.

The critical frequency is a function of the measurement distance (from the Point Of Rotation and the measurement microphone) and the distance from the POR and the acoustic center of the DUT (device under test). Stated simply, the distance between the POR and the acoustic center of the DUT must be kept relatively small to avoid errors. How "small" is related to the overall measurement distance.

The maximum distance, in 3D space, from the POR to the estimated acoustic center of the DUT is given by the equation below.

$$r = \sqrt{c * d \;/\; (4 * f_{crit})}$$

Where,
c is the speed of sound,
d is the measurement distance (POR to measurement microphone), and
f_{crit} is the critical frequency.

The criterion for this equation is that f_{crit} is the frequency at which the maximum phase error in the measurement is no greater than 45° (or 1/8 wavelength). As this is still a rather large potential error, it is recommended that in practice the value of f_{crit} used in this equation to determine r be at least one octave higher than the highest frequency of interest for a particular device.

This equation can be written another way.

$$f_{crit} = \frac{c * d}{4 * r^2}$$

This tells us that for larger values of d (the larger the measurement distance) the critical frequency is increased. Conversely, for larger values of r (the larger the distance from the POR to the acoustic center) the critical frequency is decreased.

Since it is often impossible to know the exact location of the acoustic center of a DUT, it is best to make conservative assumptions and not make the POR too far from the assumed acoustic center.

8.5.3 Angular Rotation of Horns and the POR

When measuring loudspeaker horns it can be very important to choose an appropriate POR. For most instances the POR should be located at, or very close to, the plane in which the mouth of the horn is located [4], [16], [17]. Using this location has been shown to minimize errors when using the data for modeling. Of course, it is also required that the measurement distance be sufficiently great so that the far field response of the horn is being measured.

Another reason for choosing the POR to be in the plane of the mouth is to keep the acoustical path length from both the mouth of the horn and the driver to the measurement microphone as close to being constant as possible. The illustrations below show examples of a horn with the POR in the plane of the mouth and closer to the driver. As can be seen in Figure 8-12, when the POR is at the mouth the acoustical path length the sound travels from the driver to the measurement microphone changes minimally. However, as shown in Figure 8-13 the acoustical path length changes greatly when the POR is moved away from the horn mouth and closer to the driver. This can have an effect in two different ways.

Perhaps the more important of these effects is that the increased path length relative to the POR shown in Figure 8-13 results in an increase in the rate of change (i.e. the slope of) for the phase response. This increased slope, or steepness, of the phase response requires a correspondingly higher frequency resolution for the measurement data (see Section 8.5.5). Thus the data may be less accurate at high frequencies when used later for modeling the measured source with others.

The second item of which one should be aware is that of a shifting arrival time. If a time window is applied to the measured impulse response data it becomes critical to keep the change in the acoustical path length to a minimum as this directly affects the arrival time of the impulse response (Figure 8-14 & Figure 8-15). Having this arrival time vary must be taken into account when setting the window parameters so that none of the needed data from the impulse response is inadvertently attenuated by the time window.

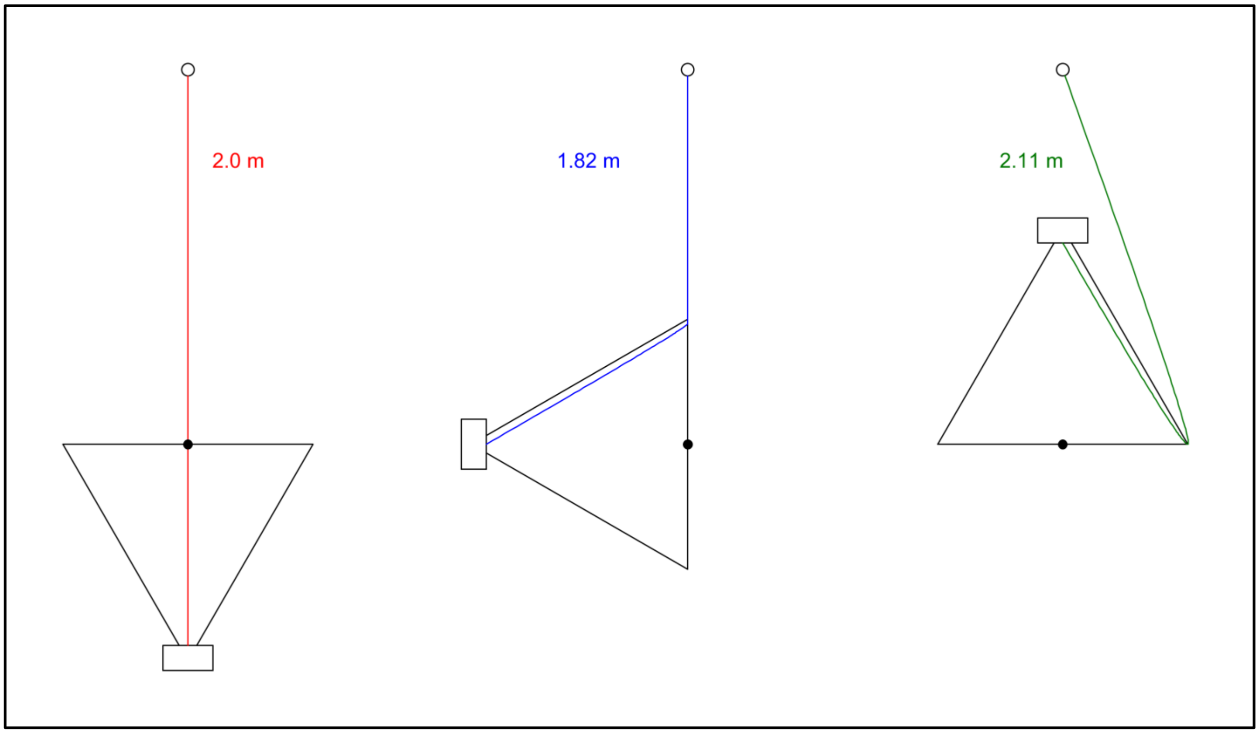


Figure 8-12 – Example path lengths for the POR located at the horn mouth (not to scale)

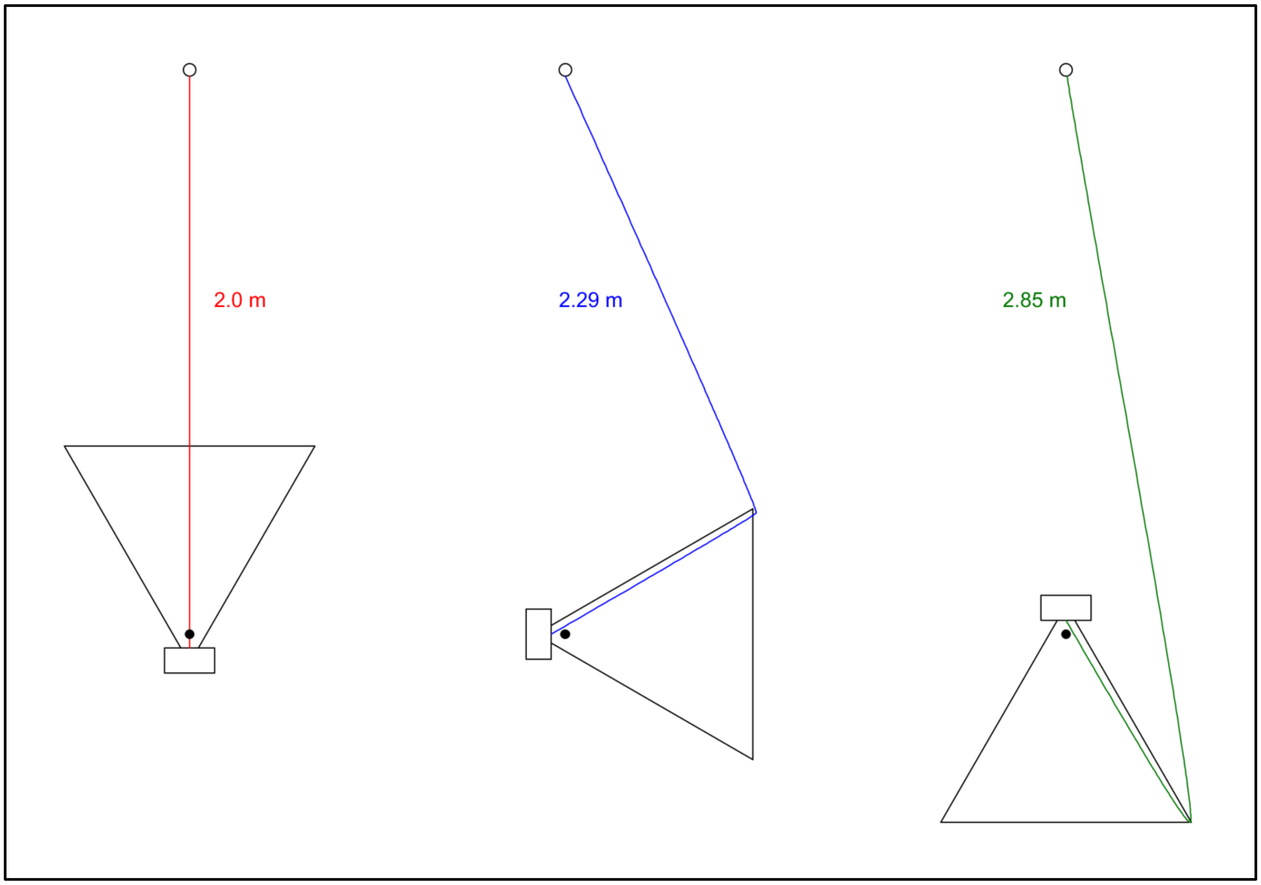


Figure 8-13 – Example path lengths for the POR located near the horn throat (not to scale)

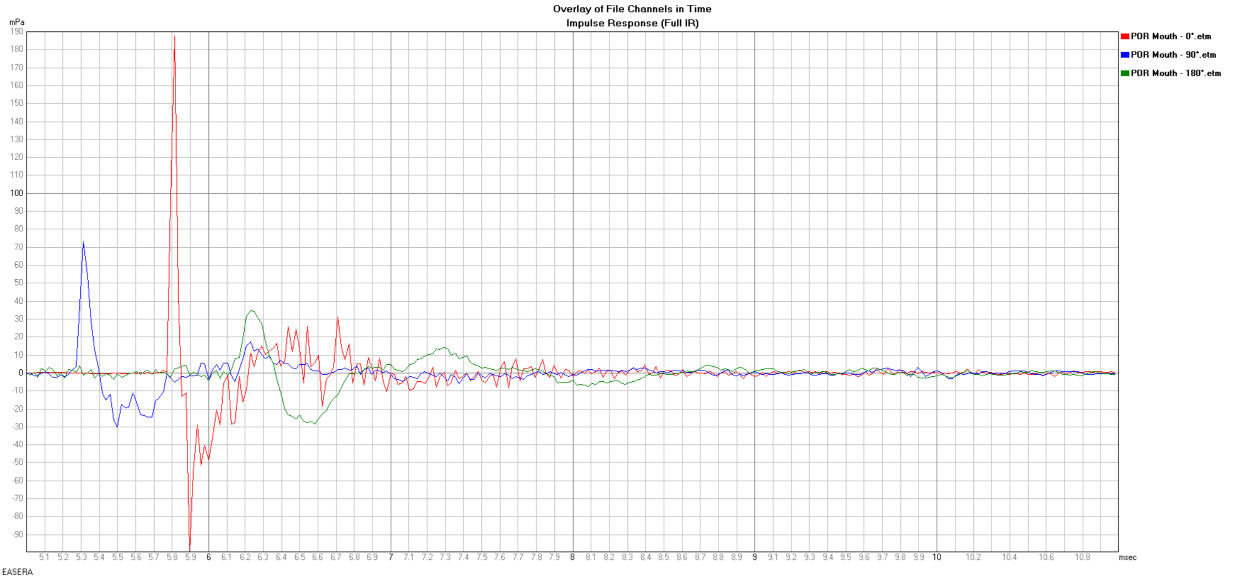


Figure 8-14 – IR arrival times for the POR shown in Figure 8-12

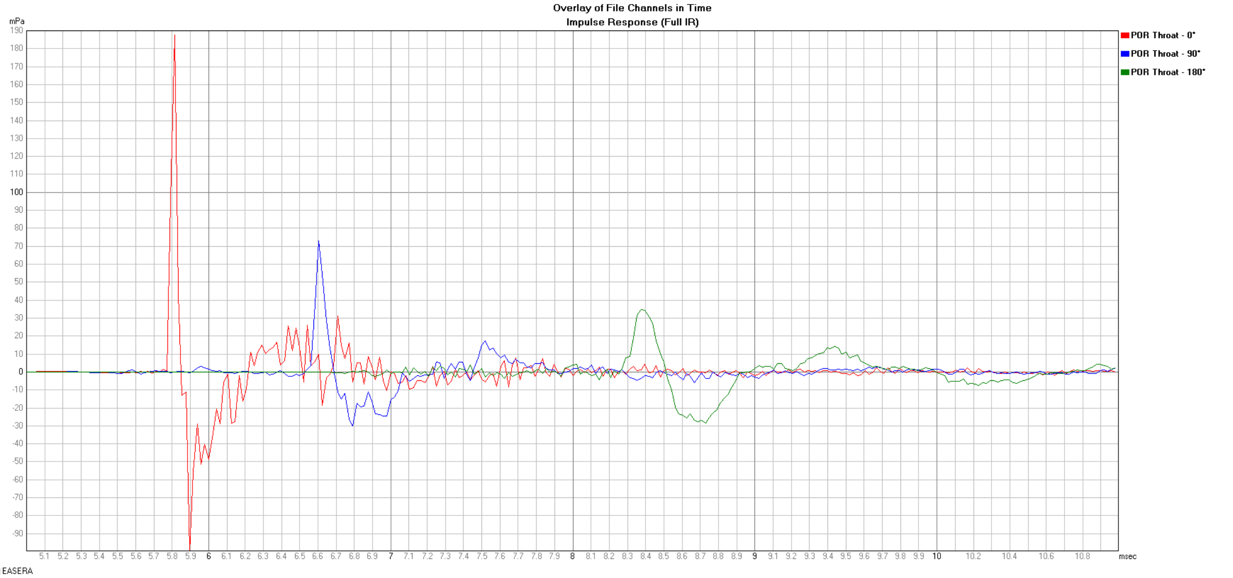


Figure 8-15 – IR arrival times for the POR shown in Figure 8-13

8.5.4 Angular Sampling Issues

Angular sampling is the process of measuring the response of a loudspeaker at different angular increments between measurements in order to characterize the directivity balloon of the loudspeaker. Just as the Nyquist theorem states that in order to accurately characterize a signal when sampling it, the sampling must occur at a rate that is at least twice the frequency of the highest frequency in the signal, the same holds true for angular sampling of the directivity balloon for a loudspeaker [5]. In order to accurately measure the shapes and locations of the lobes and nulls in the radiation pattern of a loudspeaker the angular sampling (measurement points) must occur at certain intervals compared to the lobes or nulls in the radiation pattern. Having the angular sampling at half the angular value between adjacent nulls seems to work well. In other words, if there are nulls in the radiation pattern that are 20° apart an angular sampling of at least 10° is required to characterize them.

This issue can become very important for multiple sources that are at different locations radiating sound in the same frequency region, or for continuous ribbon loudspeaker drivers. This is particularly true if the wavelengths being radiated are small compared to the spacing between the sources. This situation can lead to many nulls and lobes in the radiation pattern. A higher angular sampling rate (i.e. smaller angular increment between measurements) is required.

There is another requirement for the angular sampling or angular resolution of the measurement. This relates to the highest frequency of interest and the distance between the POR and the assumed acoustic center of the device being measured. The phase difference between two adjacent measurement points (angular samples) must be less than $\pi/2$ (90°) at any frequency of interest to be measured [1]. To satisfy this constraint the angular sampling, $\Delta\theta$, must satisfy the following equation.

$$\Delta\theta = \sin^{-1}\left(\frac{c}{4 * f * r}\right)$$

Where,
c is the speed of sound,
f is the highest frequency of interest to be measured, and
r is the distance from the POR to acoustic center of the DUT.

For additional information on angular resolution requirements please see section 8.5.6.1 as well as [1] and [5].

8.5.5 Frequency Resolution (Phase Interpolation, POR, and Smoothing)

Just as there is a relationship between the angular resolution, POR, and the acoustical center of the DUT, there is also a relationship between the frequency resolution, POR, and the acoustical center of the DUT. This, too, is determined as result of the constraint that the phase difference between two adjacent frequency data points in the final data set must be less that $\pi/2$ (90°) at any frequency of interest to be represented by the data set [1].

To satisfy this criterion the distance between the POR and the acoustical center of the DUT, *r*, must be observed.

$$r = c \, / \, (4 * \Delta f)$$

Where,
c is the speed of sound,
Δf is the data point spacing at the highest frequency of interest to be measured.

The data point spacing in the GLL file format is 1/24 octave. This and the frequency of interest define the spacing between adjacent data points. As an example, at 8 kHz the frequency spacing between adjacent data points is about 230 Hz. This would require the distance between the POR and the acoustic center of the DUT to be no more than 0.37 m for valid modeling data up to 8 kHz. At higher frequencies Δf would be larger, forcing *r* to be smaller.

It should be noted that, in an effort to increase the accuracy of the GLL modeling data while not increasing frequency resolution or data storage requirements, propagation delays are stored separately from the phase data. This helps to make the slope of the phase not as steep as if this delay was not removed and stored separately.

8.5.6 Line Sources

When measuring line sources, or an array of sources to approximate a line source, the measurement must be made in the far-field of the source. This holds true for measuring all sources when the intent is for the measured data to be used for modeling. In addition to this and the other requirements already mentioned in this section, there are some additional requirements that should be followed when measuring line sources or an array of sources to approximate a line source.

8.5.6.1 Angular Resolution

Line sources and line arrays of sources often have lobes and nulls in their radiation pattern. This is especially true in the higher frequency region of these devices. As previously mentioned it is important to accurately capture this lobe and null structure of the radiation in the measurement of the directivity balloon. Nulls in the off-axis response will occur at angles θ_i that satisfy the equation below.

$$\theta_i = \sin^{-1}\left(\pm i * \frac{c}{f * l}\right)$$

Where,
c is the speed of sound,
f is a particular frequency,
l is the length of the line source, and
i is any integer value greater than 0 (i.e. 1, 2, …).

To make sure that our angular resolution is at least twice that of the nulls we must have measurement points at angular increments, $\Delta\theta$, that are half the angular spacing of adjacent nulls [5].

$$\Delta\theta = \frac{1}{2} * \left[\sin^{-1}\left(2 * \frac{c}{f * l}\right) - \sin^{-1}\left(\frac{c}{f * l}\right)\right]$$

The required values for angular resolution as a function of the length of the line source, *l*, are shown in Table 8-1. Note that this is for a fixed frequency of 10 kHz. Different frequencies may have different angular resolution requirements.

Table 8-1

	Frequency	10 kHz	
Source Length, l (m)	**Null Angle #1**	**Null Angle #2**	**Required Angular Resolution**
0.10	20.12	43.47	11.5
0.15	13.26	27.30	7.0
0.20	9.90	20.12	5.0
0.25	7.91	15.97	4.0
0.30	6.58	13.26	3.5
0.35	5.64	11.34	3.0
0.40	4.93	9.90	2.5
0.45	4.38	8.79	2.0
0.50	3.95	7.91	2.0
0.60	3.29	6.58	1.5
0.70	2.82	5.64	1.5
0.90	2.19	4.38	1.0
0.90	2.19	4.38	1.0
1.00	1.97	3.95	1.0
1.10	1.79	3.59	1.0
1.20	1.64	3.29	1.0
1.30	1.52	3.03	1.0
1.40	1.41	2.82	0.5
1.50	1.31	2.63	0.5

Some line sources and line arrays can be very directional; meaning the beamwidth of the main lobe may be very small. It is critical when measuring such devices that the on-axis (0°) measurement actually be centered on the main lobe. If this measurement, which serves as the reference measurement for all of the off-axis data, is not centered on the main lobe a lower SPL will be measured. This is due to the rate of change for the SPL being high for a narrow main lobe. Any errors in the measurement of the on-axis level will artificially increase the level of the side lobes, as well as decrease the depth of the off-axis nulls. These errors will propagate though all of the results for the models that use this erroneously measured data.

8.5.6.2 Frequency Resolution and Smoothing

At off-axis angles line sources will produce comb filtering. The frequencies, f_i, at which the nulls (notches) occur in the response are easily predicted based on the length of the line, l, and the off-axis angle, θ.

$$f_i = i * \frac{c}{l * \sin\theta}$$

This worst case for the closest spacing of the nulls occurs when θ = 90° and the sine of θ is 1.

Having frequency data points at least twice the frequency of the spectral information to be measured allows us to accurately characterize the true frequency response. Thus, we must have at least 2 frequency data points within the bandwidth of the nulls.

$$\Delta f = \frac{1}{2} * \frac{c}{l}$$

Even when the measurement data is gathered in accordance with the above requirements, there can be potential errors generated when attempts are made to smooth, or average, the frequency response [5]. This is due to the spacing between the center points for the smoothed data being too large to capture the response changes of the measurement. The frequency region where these smoothing errors may occur will depend on the spectral structure of the measured response and the fractional octave bandwidth, *b*, for the smoothing. The lower frequency limit of this region is determined by the half spacing of the comb filter minima, Δf, and is given by the following equation.

$$f_{lower} = \frac{\Delta f}{2^{b/2} - 2^{-b/2}}$$

Where Δf is the more general form that includes the off-axis angle dependency.

$$\Delta f = \frac{1}{2} * \frac{c}{l * \sin\theta}$$

The upper frequency limit of this region is determined by the twice the spacing of the comb filter minima, Δf, and is given by the following equation.

$$f_{upper} = \frac{4\Delta f}{2^{b/2} - 2^{-b/2}} = 4 * f_{lower}$$

In this frequency region from f_{lower} to f_{upper}, called the "forbidden range", the smoothed data point spacing is too coarse and will generate aliasing errors due to the fast rate of change in the response curve.

An example is shown in Figure 8-16 for a 1/3 octave smoothing (green curve) with a source of 0.2 m in length at 90° off-axis. The blue markers denote the f_{lower} and f_{upper} limits of the forbidden range. It can be seen that through much of this range the magnitude features of the smoothed curve are actually opposite the magnitude features of the unsmoothed (red) curve. In other words, when the unsmoothed curve has a peak the smoothed curve has a dip, and vice-versa.

For these reasons smoothing or averaging should only be done for display purposes. Smoothed or averaged data should not be used in subsequent calculations. It should also be noted that we have only discussed the implications of smoothing or averaging magnitude data. It is much more difficult to smooth or average phase data in a meaningful way.

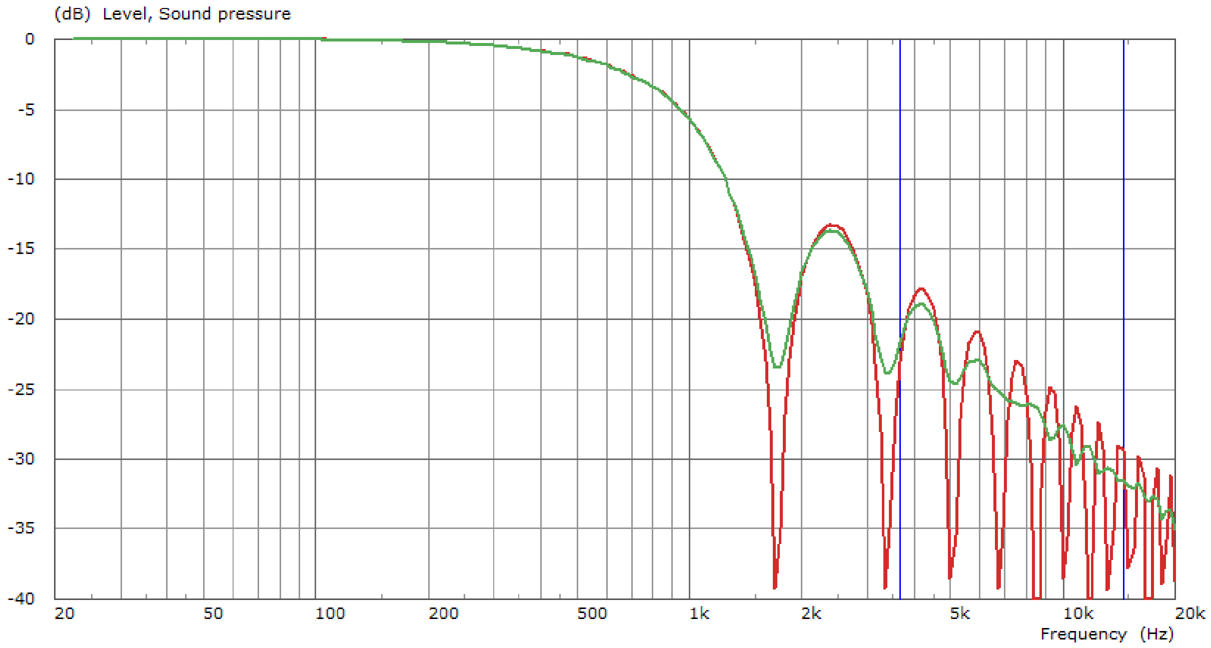


Figure 8-16 – Errors in the smoothed response within the "forbidden range"

8.5.6.3 Application to Other Extended Sources

The potential for measurement and data presentation errors detailed in this section on line arrays is not intended to imply that they can only occur with line sources or line arrays. These sources of errors are equally applicable to other types of sources and arrays or sources. Care should be taken when measuring sources to make sure that the errors detailed in this section are eliminated by properly setting up the measurements.

8.6 Weight and COG Measurement

For the calculation of the weight and center of mass of an entire line array system the weight and center of mass for each component (loudspeaker and frame) must be accurate. Any errors entering the correct data for the GLL will be compounded when multiple boxes are used in an array. It is recommended that the center of mass for the line array box include the loudspeaker itself as well as the rigging hardware that will be attached to it when it is used in an array.

There are different ways to determine the center of mass of a loudspeaker. One is to suspend the loudspeaker from a single point. The location of this point needs to be able to be varied while the loudspeaker is suspended. The location of the suspension point is changed, relative to the loudspeaker, along a given dimension (either x, y, or z) until the loudspeaker is perfectly level along that dimension. The suspension point is then directly above the center of mass for that dimension. This process can be repeated for the other dimensions until all 3 coordinates for the center of mass are known.

It should be mentioned that only the weights that are entered into the GLL are accounted for in determining the total weight of the array and its overall center of mass. Typically this does not include the weight of electrical wire or cables used for the array. The weight and location of these cables may cause the actual array to hang slightly differently than the calculated by programs that use the GLL model. The total weight of an assembled array should always be measured to assure that safe working loads are not exceeded. The final aiming angle of the actual assembled array should also be checked for possible differences compared to any calculations.

9 References

1. **Feistel S. and Ahnert, W.** Modeling of Loudspeaker Systems Using High-Resolution Data. *JAES.* July 2007, Vol. 55, 7/8, pp. 571-597.

2. **AES56-2008.** Sound source modeling - Loudspeaker polar radiation measurements. 2008.

3. **Struck, C. and Temme, S.** Simulated Free Field Measurements. *JAES.* June 1994, Vol. 46, 6, pp. 467-482.

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Appendix A. XGLL Project File – Text File Organization

A.1 General Description

The GLL text file describes geometric and acoustic properties of a loudspeaker system, cluster, and array or of a similar system. It is also used to define the way the user can configure the system, such as possible splay angles, filters, weight limits, etc. This file uses the file extension .xgll. ***EASE SpeakerLab*** will allow a manufacturer or user to compile this file. This compiled version can be used in other programs (***EASE***, ***Evac***, ***EASE Focus***, etc.) and uses the file extension .gll. ***EASE SpeakerLab*** will also allow the text file or the compiled file to be used in RUN mode, which will allow you to interact with the loudspeaker system and view the results of setting the various parameters and settings.

The file consists of several mandatory blocks in defined sequence:

- A **File Header Block**, which contains only information about the format of the file.
- A **System Header Block**, which holds general information about the system and the manufacturer.
- A **Layout Block**, which defines the logos to be used for display.
- A **Data Block**, which consists of several individual blocks depending on the type of system selected, that provide information about the geometry of the cabinets and rigging frames, the acoustical parameters of the sources, the modifications to the physical and acoustical parameters, and predefined setups.

General format conventions:

- In general, each item or list of items of the format is preceded by a keyword in quotation marks.
- The keyword identifies the data that follows.
- Usually, an entity, such as a cabinet, frame, filter, or setup has an internal name used to refer to it by the software and by other entities and a display name which is displayed to the user.
- Text items, such as display and internal names, are also enclosed in quotation marks. Text items cannot contain additional quotation marks.
- All internal names for cabinets, sources, frames etc. must be unique.
- All coordinates are defined relative to the origin of the drawing for the cabinet or the frame.
- All measures are in metric units.
- All text items, display names, and internal names are limited to 250 characters.
- The only decimal separator is the period (".").
- Adjacent two items in a line are separated by comma (", ").
- Description lines must be continuous; they cannot be continued on a consecutive line.
- Different lines are separated by a carriage return (CR, ASCII 13) or carriage return and line feed (CR and LF, ASCII 13 and ASCII 10).
- Lines are to contain only characters of ASCII values from 32 to 127 (standard letters).
- Comment lines are allowed and must begin with a semicolon in quotes (";").
- The file can contain any number of empty lines or comment lines between description lines.

File locations can be absolute or relative to the GLL project file:

- "C:\Datafiles\Logo1.jpg" – an absolute file location.
- "Logo1.jpg" – a relative file location, pointing to a file in the same directory.
- ".\Datafiles\Logo1.jpg" – a relative file location, pointing to a subdirectory.
- "..\Datafiles\Logo1.jpg" – a relative file location, pointing to a directory in the same root directory.

In the description below, each mandatory keyword is listed, followed by one or more data items. The file should hold all of the blocks; they should be located in the proper order and include all sub-blocks.

A.2 File Header Block

The file must begin with a line containing the string token "GLL". This is used by ***EASE SpeakerLab*** to determine the type of file. The next line should state the format of the XGLL file. The only type currently allowed is "3D". The third line in the file should be the version number of the XGLL format. The current version is "1.0".

```
GLL"
"Format", "3D"
"FormatVersion", "1.0"
```

A.3 System Header Block

This describes the type of loudspeaker system and contains general information about the company that manufactured the loudspeaker system.

```
"System", "SphereLine19", "SL19", "LA"
"SystemVersion", 1.00
"Company", "SDA"
"InfoText", "This is an imaginary array. Similarities to existing systems
               are not intended."
"CopyrightText", "© 2007 SDA Software Design Ahnert GmbH"
"SupportText", "See manual for support and rigging instructions"
"WebsiteText", "www.sda.de"
"EmailText", "info@sda.de"
```

System *Display Name, Internal Name, Type*.

The **System** keyword has these items:

Item	Description
• *Display Name*	Label used for display to the user, does not need to be unique.
• *Internal Name*	Used for identification of this System, during the upgrade process, for example, must be unique.
• *Type*	Determines the blocks of data that are available. See **Settings** below.

Settings Currently there are 3 types defined.

The settings for *Type* are:

Type	Description
• LA	Line Array with Configurable Boxes – This type allows the use of the data blocks for Frames, Connectors, Limits, and Warnings. Typically these are used for Line Array type devices that use individual boxes.
• CL	Array or Cluster of Fixed Boxes – This type allows the use of the data blocks for Cluster Setups. Typically these are used for Cluster type arrangements of multiple boxes.
• LS	Loudspeaker, Single Box – This type is used for any system that is in a single box, such as a single loudspeaker, a 2-way or 3-Way loudspeaker, a column loudspeaker, or a flat panel loudspeaker.

System Version Version number for the **System**.

The version number is used to detect whether a system definition is older or newer and therefore allows for an upgrade. It should be incremented whenever the definition file is changed.

Manufacturer Name of the manufacturer of the loudspeaker system.

This text is displayed in the Info window of the compiled GLL file.

Info Text holding some information about the system.

This text is displayed in the Info window of the compiled GLL file.

Copyright Text holding some copyright information about the system.

This text is displayed in the Info window of the compiled GLL file.

Support Text holding some support information about the system.

This text is displayed in the Info window of the compiled GLL file.

Website Text holding a link to the manufacturer's website or the website for the system.
This text is displayed in the Info window of the compiled GLL file. The link can be accessed by clicking the logo bar.

Email Text holding the email address for support questions regarding the system.
This text is displayed in the Info window of the compiled GLL file.

A.4 Layout Block

This block contains information about the appearance of the GLL when run in the ***GLL Viewer*** or other programs such as ***EASE***. If either Logo item is left blank then the BackgroundColor will be used.

```
"Logo1", ".\DataFiles\Logo1_TestA.png"
"Logo2", ".\DataFiles\Logo2_TestA.png"
"BackgroundColor", 0, 0, 255
```

Logo1 File location for the left logo to be displayed in the logo bar of ***EASE SpeakerLab***.
The picture size must be 468x60 pixels, and the format should be png, jpg, or gif. It is recommended that the graphics for the system be provided here.

Logo2 File location for the right logo to be displayed in the logo bar of ***SpeakerLab***.
The picture size must be 234x60 pixels, and the format should be png, jpg, or gif. It is recommended that the graphics for the manufacturer be provided here.

Background Color The 3 color values for the background color of the logo bar.
The color consists of the three values Red, Green, Blue in a range of 0 to 255. This color is used for the space to the right of Logo2.

A.5 Data Blocks

A.5.1 Box Types

This data block contains the geometric information about the loudspeaker boxes available for this system. Each BoxType is commonly referred to as a cabinet or loudspeaker box. Each cabinet can use one or more acoustic sources for the directivity data.

```
"BoxTypes", 4
    "BoxType", "Sub", "LSub"
          "Sources", 1
                "Source", "Sub", "LF", 0.0, 0.3, -0.3, 0.0, 0.0, 0.0, "DSub"
          "Input Configurations", 1
                "Input Configuration", "Input Config", "InputConfigA"
                      "Inputs", 1
                      "Input", "Input LF"
                      "Links", 1
                      "Link", "LF", "SubFilterGrp"
          "GeometryFiles", 1
                "GeometryFile", "CaseSubwoofer.xed"
          "ReferencePoint", 0.0, 0.3, -0.3
          "NextPivot", 0.0, 0.0, -0.6
          "CenterOfMass", 0.0, 0.3, -0.3
          "Weight", 30

"BoxType", "Big", "LBig"
…

"BoxType", "Small", "LSmall"
…

"BoxType", "DownFill", "LDownFill"
…
```

BoxTypes *Number* of types of cabinets available for the system. This is followed by an equivalent number of BoxType sub-blocks.

BoxType *Display Name, Internal Name*.

This sub-block contains information on one of the BoxTypes available for use in this system.

The BoxType keyword has these items:

Item	Description
• *Display Name*	Label used for display to the user, does not need to be unique.
• *Internal Name*	Used for identification of this BoxType, during the upgrade process, for example, or for later reference by other entities, must be unique.

Sources *Number* of acoustic sources used to model this cabinet. This is followed by an equivalent number of Source keywords.

In the simplest case a cabinet has one point where the acoustic directivity data is referred to. A source may be a particular driver or a particular point of a wave front model.

Source *Display Name, Internal Name, X[m], Y[m], Z[m], H[°], V[°], R[°], Source Definition*

This keyword contains information on one of the acoustic sources used for this BoxType.

The Source keyword has these items:

Item	Description
• *Display Name*	Label used for display to the user, does not need to be unique. Normally the compiled GLL file will show the display name of the Input that this Source is assigned to via an Input Configuration.
• *Internal Name*	Used for identification of this Source, during the upgrade process, for example, or for later reference by other entities, must be unique.
• *X[m], Y[m], Z[m]*	3D location of the acoustic directivity data that this source uses.
• *H[°], V[°], R[°]*	Aiming angle of the acoustic directivity data that this source uses.
• *Source Definition*	The SourceDefinition to use for this source. This internal name must be included in the Source Definitions data block below.

Input Configurations *Number*

This is the number of ways the input connections can be configured for the system. This is followed by an equivalent number of Input Configurations sub-blocks.

Input Configuration *Display Name, Internal Name*

This sub-block contains information on one of the Input Configurations available for use in this BoxType.

The Input Configuration keyword has these items:

Item	Description
• *Display Name*	Label used for display to the user, does not need to be unique.
• *Internal Name*	Used for identification of this Input Configuration, must be unique. (In ***SpeakerLab*** a tab will be dynamically created using this name and added to the Edit Box Type page).

Inputs *Number* of input connections used in this Input Configuration. This is followed by an equivalent number of Input keywords.

In the simplest case a cabinet has one input that is defined by a single input configuration. An input usually corresponds to an input connection.

Input *Display Name*

This keyword contains information on one of the inputs used for this Input Configuration.

The Input keyword has this item:

Item	Description
• *Display Name*	Label used for display to the user, does not need to be unique.

Links *Number* of mappings between this input and the acoustic sources. This is followed by an equivalent number of Link keywords. This link uses a Filter Group or only an external filter for the mapping.

In the simplest case, a loudspeaker box modeled as a single-source GLL with a single input has its input mapped directly to the only source in the XGLL using the <EXTERNAL> link.

For other cases, such as a three-way loudspeaker modeled as multi-source GLL, there may be three separate inputs such that the loudspeaker may be triamplified. There may be a different configuration that uses a single input and routes this input signal through filters internal to the GLL and then to each of the three sources used in the GLL.

Link *Source, Filter Group*

This sub-block contains information on one of the Links available for use with this input.

The Link keyword has these items:

Item	Description
• *Source*	The Source to use for this link. This internal name must be included in the Sources data block above.
• *Filter Group*	The Filter Group to use for this link. This internal name must be included in the Filter Groups data block below. If empty, only an external filter can be applied. In either case multiple Sources linked to an Input are displayed to the user in a combined way as a single Input.

GeometryFiles *Number* of geometry files available for this BoxType. This is followed by an equivalent number of GeometryFile keywords.

GeometryFile File location for this geometry file. This should point to an ***EASE*** Edge (.fed) or ASCII Edge (.xed) file to show lines for the geometry. See Figure 2-11 in section 2.6.1.1 for documentation on the ASCII format. Use ***EASE*** to convert an ***AutoCAD*** DXF file into the necessary format(s).

If you specify .fed files there must be ***EASE*** Room Data (.frd) and Vertex (.fvt) files that define the ends of the lines.

Reference Point *X[m], Y[m], Z[m].*

3D location of the Virtual Center Point. It is used for several auxiliary calculations relating to the box as a whole. If there is just a single source, the Virtual Center Point should have the location of this source. Otherwise the geometric center of all sources or of the box should be used.

NextPivot *X[m], Y[m], Z[m]*

3D location of the Connecting Point. This is essential information for the rigging structure. The next pivot point is the point around which the next cabinet is rotated. This information is relevant only for line array type GLLs.

CenterOfMass *X[m], Y[m], Z[m]*

3D location of the Center of Mass for this BoxType. This is part of the basic information needed to correctly compute the center of mass for the whole system and allow for proper tilt angles.

Weight *Weight [kg]*

Weight for this BoxType. This is part of the basic information needed to correctly compute the center of mass for the whole system and allow for proper tilt angles.

A.5.2 Frames (only for line array type GLL)

This data block contains the geometric information about each of the support frames available for this system.

```
"Frames", 3
   "Frame", "Down", "FDown"
       "Type", "Flown"
       "GeometryFiles", 1
          "GeometryFile", "FrameDown.xed"
       "NextPivot", 0.0, 0.0, -0.1
       "CenterOfMass", 0.0, 0.4, 0.0
       "Weight", 20
       "PinPoints", 7
          "PinPoint", "1", 0.0, 0.1, 0.0
          "PinPoint", "2", 0.0, 0.2, 0.0
          "PinPoint", "3", 0.0, 0.3, 0.0
          "PinPoint", "4", 0.0, 0.4, 0.0
          "PinPoint", "5", 0.0, 0.5, 0.0
          "PinPoint", "6", 0.0, 0.6, 0.0
          "PinPoint", "7", 0.0, 0.7, 0.0
   "Frame", "Up", "FUp"
      …
   "Frame", "Stacked", "FStacked"
      …
```

Frames *Number* of rigging frames or setup versions available for the system. This is followed by an equivalent number of Frame sub-blocks.

Frame *Display Name, Internal Name*

This sub-block contains information on one of the rigging frames available for use in this system.

The Frame keyword has these items:

Item	Description
• *Display Name*	Label used for display to the user, does not need to be unique.
• *Internal Name*	Used for identification of this Frame, during the upgrade process, for example, or for later reference by other entities, must be unique.

Type This keyword is for internally distinguishing between the Flown and Groundstack **s**etups.

Settings

The settings for *Type* are:

Type	Description
• *Flown*	The Frame is intended for use at the top of the system.
• *Groundstack*	The Frame is intended for use at the bottom of the system.

GeometryFiles *Number* of geometry files available for this Frame. This is followed by an equivalent number of GeometryFile keywords.

GeometryFile File location for this geometry file. This should point to an ***EASE*** Edge (.fed) or ASCII Edge (.xed) file to show lines for the geometry. See Figure 2-11 in section 2.6.1.1 for documentation on the ASCII format. Use ***EASE*** to convert an ***AutoCAD*** DXF file into the necessary format(s).

If you specify .fed files there must be ***EASE*** Room Data (.frd) and Vertex (.fvt) files that define the ends of the lines.

NextPivot *X[m], Y[m], Z[m]*

3D location of the Connecting Point. This is essential information for the rigging structure. The next pivot point is the point around which the attached first cabinet in the array is rotated.

CenterOfMass *X[m], Y[m], Z[m]*

3D location of the Center of Mass for this Frame. This is part of the basic information needed to correctly compute the center of mass for the whole system and allow for proper tilt angles.

Weight *Weight [kg]*

Weight for this Frame. This is part of the basic information needed to correctly compute the center of mass for the whole system and allow for proper tilt angles.

PinPoints *Number* of pin points (or pick points) available for hanging the frame. This is followed by an equivalent number of PinPoint keywords.

A flown version should have at least 1 pin point, or pick point, if there are no pins, while ground-stack versions must have 0.

PINPOINT *Display Name, X[m], Y[m], Z[m]*

This keyword contains information on one of the PinPoints used for hanging the Frame.

The PinPoint keyword has these items:

Item	Description
• *Display Name*	Label used for display to the user, does not need to be unique.
• *X[m], Y[m], Z[m]*	3D location of the pin point.

A.5.3 Connectors (only for line array type GLL)

This data block contains the geometric information about how each of the cabinets and/or frames can be attached to each other.

```
"Connectors", 25

   "Connector", "FUp", "LBig", ""
       "Angles", 1
          "Angle", "0°", 0.0

   "Connector", "LSmall", "LSmall", "FUp"
       "Angles", 4
          "Angle", "0°", 0.0
          "Angle", "1°", 1.0
          "Angle", "2°", 2.0
          "Angle", "3°", 3.0
       …
```

Connectors *Number* of connectors available for the system.

A Connector is an element that defines how any two cabinets or how a frame and a cabinet are linked. This is followed by an equivalent number of Connector keywords.

Connector *Upper, Lower, Frame*

This keyword contains information on one of the Connectors used for attaching Frames and BoxTypes to each other.

The Connector keyword has these items:

Item	Description
• *Upper*	The Internal Name for the upper BoxType or Frame to be used with this Connector. This internal name must be included in the BoxTypes or Frames data blocks above.
• *Lower*	The Internal Name for the upper BoxType or Frame to be used with this Connector. This internal name must be included in the BoxTypes or Frames data blocks above.
• *Frame*	The Internal Name for the Frame setup that this connector can be used with. This Internal Name must be included in the Frames data block above. For Connectors between a Frame and BoxType, this item is not needed, and an empty text field “” should be used in place of the Internal Name.

Connectors for ground-stack versions are specified the same way as for flown versions. Here the Frame must be referred to as the second element, because the order is always top-down. The same applies to Connectors between BoxTypes.

If a combination of BoxTypes is not allowed, the Connector must not be defined.

Angles *Number* of splay angles provided by the connector. This is followed by an equivalent number of Angle keywords.

Angle *Display Name, Angle*

This keyword contains information on one of the possible Angles available for this connector.

The Angle keyword has these items:

Item	Description
• *Display Name*	Label used for display to the user, does not need to be unique. This may even be a display name such as “Hole #2”.
• *Angle*	The rotational angle is the true angle in degrees for the lower BoxType used for this setting.

Specified angles should always be sorted according to their values or to the actual order appearing for/at the hardware.

A.5.4 Limits (only for line array type GLL)

This data block contains the information about the physical limits for various parts of the system.

```
"Limits", 12
    "Limit", "MaxBoxes", "FDown", 16
    "Limit", "MaxBoxesOfType", "FDown", "LSub", 3
    "Limit", "MaxTiltAngle", "FUp", 16
    "Limit", "MaxTiltAngle", "FDown", 12
    …
```

Limits *Number* of physical limits applying to the System. This is followed by an equivalent number of Limit keywords.

A limit is a restriction to the system configuration as defined by the manufacturer. The most important limits are the weight and the overall number of cabinets.

Limit *Type, {Frame}, {Box Type}, Limit*

This keyword contains information on one of the possible Limits defined for this System.

The Limit keyword has these items:

Item	Description
• *Type*	Identifier for the limit type. See **Settings** below.
• *Frame* ***{Optional}***	The Frame that this limit applies to. This internal name must be included in the Frames data block above. The limit can apply to only one Frame, omitting this value will let the limit be applied to all frames.
• *Box Type* ***{Optional}***	The BoxType that this limit applies to. This internal name must be included in the BoxTypes data block above. The limit can apply to only one BoxType, omitting this value will let the limit be applied to all box types.
• *Limit*	The numerical value for this type of limit.

Settings

The type of limit defines the condition under which the restriction is enforced by the software.

The settings for *Type* are:

Type	Description
• *MaxBoxes*	This is the allowed maximum number of BoxTypes for the System.
• *MaxBoxesOfType*	This is the allowed maximum number of BoxTypes of this type, the internal name for the type of box must be included.
• *MaxWeight*	This is the allowed maximum weight of all BoxTypes of the System.

• *MinTiltAngle*	This is the allowed minimum tilt angle for the System.
• *MaxTiltAngle*	This is the allowed maximum tilt angle for the System.

Note: A few limits may not be enforced directly. Instead the program will display a message.

A.5.5 Warnings (only for line array type GLL)

This data block contains the information about warnings provided to the user of the system.

```
"Warnings", 4
   "Warning", "MaxWeight", "FUp", 350, "Too heavy!"
   "Warning", "MaxBoxes", "FUp", 12, "Consult manual for that many
             cabinets!"
   …
```

Warnings *Number* of warnings defined by the manufacturer. This is followed by an equivalent number of Warning keywords.

A Warning is a text message to be displayed if a certain condition for the system configuration is fulfilled. A Warning does not affect the system configuration. Its only purpose is providing more information to the user under defined circumstances.

Warning *Type, Limit, {Frame}, Text*

This keyword contains information on one of the possible Warning conditions defined for this System.

The Warning keyword has these items:

Item	Description
• *Type*	Identifier for the warning type. See Settings below.
• *Limit*	The limit value.
• *Frame*	The Internal Name of the Frame that this limit applies to. This internal name must be included in the Frames data block above. The condition can apply to only one Frame, omitting this value will let the warning be applied to all frames.
• *Text*	The warning text displayed to the user.

Settings

The type of warning defines under which condition the warning is displayed by the software.

The settings for *Type* are:

Type	Description
• *MaxBoxes*	This is the threshold maximum number of BoxTypes for the System.
• *MinBoxes*	This is the threshold minimum number of BoxTypes for the System.
• *MaxWeight*	This is the threshold maximum weight of all Box-Types of the System.
• *MinTiltAngle*	This is the threshold minimum tilt angle for the System.
• *MaxTiltAngle*	This is the threshold maximum tilt angle for the System.

A.5.6 Source Definitions

This data block contains the information about each of the sources used in the cabinets listed above.

```
"SourceDefinitions", 3
   "SourceDefinition", "DSmall"
      "File", "<SPHERE>"

   "SourceDefinition", "DBig"
      "File", "<SPHERE>"

   "SourceDefinition", "DSub"
      "File", "<SPHERE>"
```

SourceDefinitions *Number* of source definitions. This is followed by an equivalent number of SourceDefinition keywords.

SourceDefinition *Internal Name*

This is the internal name referred to by the BoxType in the Sources keyword in the data block for BoxTypes.

File File location for this source definition. This should point to an ***EASE SpeakerLab*** Generic Sound Source (*.gss) file. This file can be created from IR/FR data or from ***EASE*** .spk files using the GSS Source Editor. To use the default sphere source, enter <SPHERE> in place of the file name.

A.5.7 Cluster Setups (only for cluster type GLL)

This data block contains the geometric information about each of the cluster setups available for this system.

```
"ClusterSetups", 2
   "ClusterSetup", "3x", "keyThree"
      "ClusterBoxes", 3
         "ClusterBox", "Left", 0.0, 0.0, -0.02, 0.0, 0.0, 0.0, "keyBox"
         "ClusterBox", "Center", 0.0, 0.0, -0.04, 0.0, 0.0, 0.0, "keyBox"
         "ClusterBox", "Right", 0.0, 0.0, -0.04, 0.0, 0.0, 0.0, "keyBox"
         …
"ClusterSetup", "2x", "keyTwo"
   "ClusterBoxes", 3
   …
```

ClusterSetups *Number* of cluster setups available for the system. This is followed by an equivalent number of ClusterSetup sub-blocks.

ClusterSetup *Display Name, Internal Name*

This sub-block contains information for one of the assemblies or layouts of the cabinets available for use in this system.

The ClusterSetup keyword has these items:

Item	Description
• Display Name	Label used for display to the user, does not need to be unique.
• Internal Name	Used for identification of this ClusterSetup during the upgrade process, for example, or for later reference by other entities, must be unique.

ClusterBoxes *Number* of boxes defined for this cluster setup. This is followed by an equivalent number of ClusterBoxes keywords.

ClusterBox *Display Name, X[m], Y[m], Z[m], H[°], V[°], R[°], BoxType*

This keyword contains information on one of the boxes used in this cluster.

The ClusterBox keyword has these items:

Item	Description
• Display Name	Label used for display to the user, does not need to be unique.
• X[m], Y[m], Z[m]	3D location of the cabinet.
• H[°], V[°], R[°]	Aiming angle of the cabinet.
• BoxType	The BoxType to use for this cabinet. This internal name must be included in the BoxTypes data block above.

A.5.8 Filter Groups

This data block contains the information about each of the filter groups that are available.

```
"FilterGroups", 3 "FilterGroup", "Input", "keyLFXover"
    "FilterDefinitions", 2
        "FilterDefinition", "Crossover 1k", "Key1000"
            "File", "LF_1k.xgfb"
         "FilterDefinition", "Crossover 500", "Key500"
            "File", "LF_500.xgfb"
         …
```

FilterGroups *Number* of filter groups available for this system. This is followed by an equivalent number of FilterGroup keywords.

FilterGroup *Display Name, Internal Name*

This sub-block contains information on one of the filter groups available for this system.

The FilterGroup keyword has these items:

Item	Description
• *Display Name*	Label used for display to the user, does not need to be unique.
• *Internal Name*	Used for identification of this FilterGroup, must be unique.

FilterDefinitions *Number* of filter definitions available for this FilterGroup. This is followed by an equivalent number of FilterDefinition keywords.

FilterDefinition *Display Name, Internal Name*

This sub-block contains the information about one of the filter definitions that is included in this filter group.

The FilterDefinition keyword has these items:

Item	Description
• *Display Name*	Label used for display to the user, does not need to be unique.
• *Internal Name*	Used for Identification of this FilterDefinition, must be unique in this FilterGroup. (In SpeakerLab a tab will be dynamically created using this name and added to the Edit Filter Group page). When two different Filter Groups are assigned to the same Input (see Box Types and Input Configurations above) each FilterDefinition must have the same internal name as a FilterDefinition from the other Filter Group. Then these will be used together to apply filters from an input to separate sources. Typically used for internal crossovers: 2 Filter groups with different filters inside (at least one) are combined using the same internal names for the filter definitions.

File File location for this filter definition. This should point to an ***EASE SpeakerLab*** Generic Filter Bank (.xgfb) file. (See Appendix C – XGFB Filter Bank File - Text File Organization for documentation.)

Examples

Following is the header portion of a system definition for a Simple Loudspeaker:

```
"GLL"
"Format", "3D"
"FormatVersion", "1.0"

"System", "Loud Speaker", "keySimpleLsp", "LS”
"SystemVersion", 1.0
"Company", "SDA"
```

Following is the header portion of a system definition for a Column Loudspeaker:

```
"GLL"
"Format", "3D"
"FormatVersion", "1.0"

"System", "Column 8Ch", "keyCol8Ch", "LS"
"SystemVersion", 1.0
"Company", "SDA"
```

Following is the header portion of a system definition for a Line Array:

```
"GLL"
"Format", "3D"
"FormatVersion", "1.0"

"System", "Array Line", "keyArrayLine", "LA"
"SystemVersion", 1.0
"Company", "SDA"
```

Following is the header portion of a system definition for a Cluster Loudspeaker:

```
"GLL"
"Format", "3D"
"FormatVersion", "1.0"

"System", "Cluster of 3", "keyClusterOf3", "CL"
"SystemVersion", 1.0
"Company", "SDA"
```

Appendix B. XGLC Configuration File - Text File Organization

B.1 General Description

The GLC (Generic Loudspeaker Configuration) text file describes a specific configuration to be applied to a loudspeaker or loudspeaker system defined in the GLL format. This configuration can include mechanical data, gain settings and filters. Overriding is allowed for external filters and, as of version 1.01, for internal filters, too. If allowed by the GLL, internal filters can also be replaced and user-defined cluster configurations loaded.

This file uses the file extension **.xglc**. A configuration for a specific GLL file can be created, loaded and saved with ***EASE SpeakerLab*** when the GLL file is being viewed in *Run* Mode, with the ***GLL Viewer***, or from within the GLL configuration interface of ***EASE*** as well as from within ***EASE Focus***.

The file consists of two mandatory blocks in defined sequence:

- A **File Header Block**, which contains only information about the format of the file.
- A **Data Block**, which consists of several individual blocks depending on the type of configuration selected, that provide information about the filters bank groups, input groups and cluster setup.

General format conventions:

- In general, each item or list of items of the format is preceded by a keyword in quotation marks.
- The keyword identifies the data that follows.
- Each filter has an optional display name which is displayed to the user.
- Text items, such as display names, are also enclosed in quotation marks. Text items cannot contain additional quotation marks.
- All text items, and display names are limited to 250 characters.
- The only decimal separator is the period (“.”).
- Adjacent two items in a line are separated by comma (“, ”).
- Description lines must be continuous; they cannot be continued on a consecutive line.
- Different lines are separated by a carriage return (CR, ASCII 13) or carriage return and line feed (CR and LF, ASCII 13 and ASCII 10).
- Lines are to contain only characters of ASCII values from 32 to 127 (standard letters).
- Comment lines are allowed and must begin with a semicolon in quotes (“;”).
- The file can contain any number of empty lines or comment lines between description lines.

In the description below, each mandatory keyword is listed, followed by one or more data items. The file should hold all of the blocks; they should be located in proper order and include all sub-blocks.

B.2 File Header Block

The file must begin with a line containing the string token "GLC". This is used by ***EASE SpeakerLab*** to determine the type of file. The next line should state the format of the XGLC file. The only type currently allowed is "GLL Configuration". The third line in the file should be the version number of the XGLC format. The current version is either "1.0" or "1.01" depending on what attributes are used (e.g. user defined cluster setups and filter overrides).

```
"GLC"
"Format", "GLL Configuration"
"FormatVersion", "1.0"
```

B.3 Data Block

B.3.1 Configuration

This data block contains the information about the loudspeaker configuration.

```
"Used System", "Tapered Column", "keyCol8Ch", "LS", 1.00, "Lspk, Inc."
"Angle", 0.00
"Elements", 1
   "Element"
   ...
```

Used System *Display Name, Internal Name, Type, Version, Company*

Information about the GLL file to which the configuration applies. *Display Name, Internal Name* and *Type* must be identical to the corresponding attributes of the **System** in the GLL file.

Version and *Company* refer to the **System Version** and **Company** attributes in the GLL, respectively (please refer to the GLL file format documentation, Appendix A).

Cluster (Cluster type GLL only) Specifies the internal name of the cluster setup to which the configuration applies.

Angle Defines the vertical, mechanical angle of the entire system. Currently not in use.

Elements *Number* of elements used in this configuration.

B.3.2 Cluster (only for cluster type GLL)

Describes the configuration of the cluster setup. This section is only applicable to GLLs of type Cluster. Note also that the format version of the XGLC used to specify user-defined cluster setups must be "1.01".

```
"Cluster", ""
   "Boxes", 2
      "Box", "Bottom", 0.00, 0.00, 0.00, 0.00, 0.00, 0.00, "key4WayBox"
      "Box", "Top", 0.00, 0.00, 0.60, 0.00, 0.00, 0.00, "key4WayBox"
```

Boxes *Number* of boxes used in this configuration.

Box *Display Name, X, Y, Z, Horizontal Angle, Vertical Angle, Rotation Angle, Type.* Information about each box used in the cluster.

Item	Description
Display Name	Box label displayed to the user.
X	X Coordinate of the box in meters.
Y	Y Coordinate of the box in meters.
Z	Z Coordinate of the box in meters.
Horizontal Angle	Horizontal angle of the box in degrees.
Vertical Angle	Vertical angle of the box in degrees.
Rotation Angle	Rotation angle of the box in degrees.
Type	Internal name of box type as defined in GLL.

B.3.3 Element

This part contains information on a single element or box.

```
"Element"
   "Box Type", "key4WayBox"
   "Gain", 0.00
   "Input Config", "InputConfig"
      ...
```

Box Type Internal name of box type as defined in GLL.

Gain Element gain in dB. The value of this gain cannot be positive (i.e. > 0).

Input Config Sub-block containing details of the input configuration.

B.3.4 Input Config

Details of the input configuration.

```
"Input Config", "InputConfig"
   "Inputs", 2
      "Input"
         ...
```

Input Config Internal name of active input configuration, as defined in GLL.

Inputs *Number* of inputs defined in this configuration.

Input Sub-block containing the input definition.

B.3.5 Input

Details about the configuration of a specific input.

```
"Input"
   "Internal Filter", "XO1k"
   "External Filter"
      "GFB"
         ...
```

Internal Filter Name of the internal filter definition to be used. If overriding is allowed, this attribute may be followed by details about the filter to be used for overriding.

External Filter Sub-block defining the external filter directly in GFB format (see Appendix C – XGFB Filter Bank File - Text File Organization for reference).

B.3.6 Internal Filter

If overriding is allowed and intended, the attribute **Internal Filter** may be followed by other attributes. Note that overriding internal filters is only possible if there is already a filter group in the original GLL and that filter group is assigned to the sources of the box. Note also that the format version of the XGLC used to override the internal filters must be "1.01".

```
"Internal Filter", "XO1k"
   "Override Setting", "*Crossover 250 Hz"
   "Internal Filters To Override", 2
      "Override Internal Filter", "keyHF"
         "GFB"
         ...
      "Override Internal Filter", "keyLF"
         "GFB"
         ...
```

Override Setting Name of the new filter configuration as displayed to the end user. This will replace the display name of the original filter group temporarily.

Internal Filters to Override

Number of the internal filters defined in this group.

Override Internal Filter

Internal name of the source whose internally associated filter is to be overridden. This is followed by a sub-block defining the internal filter directly in GFB format (see Appendix C – XGFB Filter Bank File - Text File Organization for reference).

B.4 Examples

Following is an example of a very simple configuration of a loudspeaker GLL:

```
"GLC"
"Format", "GLL Configuration"
"FormatVersion", "1.0"

"Configuration"
   "Used System", "Tapered Column", "keyCol8Ch", "LS", 1.00, "Lspk, Inc."
   "Angle", 0.00
   "Elements", 1
      "Element"
      "Box Type", "keyColumn"
      "Gain", 0.00
      "Input Config", "SteeredMode"
         "Inputs", 1
            "Input"
               "Internal Filter", "keyFilterDef1"
               "External Filter"
                  "GFB"
                  "Format", "General Lspk"
                  "FormatVersion", "1.0"

                  "Bypass", "No"
                  "Gain", 0.00
                  "Delay", 0.00
                  "Invert Polarity", "No"

               "Filters", 0
```

Following is a more complex configuration, with a two-box cluster and overriding of internal crossover filters:

```
GLC"
"Format", "GLL Configuration"
"FormatVersion", "1.01"

"Configuration"
   "Used System", "Four-Way Library", "keyClusterFourWay", "CL", 1.00,
      "Lspk, Inc."
   "Cluster", ""
      "Boxes", 2
         "Box", "Bottom", 0.00, 0.00, 0.00, 0.00, 0.00, 0.00, "key4WayBox"
         "Box", "Top", 0.00, 0.00, 0.60, 0.00, 0.00, 0.00, "key4WayBox"

   "Angle", 0.00
   "Elements", 2
      "Element"
         "Box Type", "key4WayBox"
         "Gain", 0.00
         "Input Config", "InputConfig"
            "Inputs", 2
               "Input"
                  "Internal Filter", "XO1k"
                  "External Filter"
                     "GFB"
                     "Format", "General Lspk"
                     "FormatVersion", "1.0"

                     "Bypass", "No"
                     "Gain", 0.00
                     "Delay", 0.00
                     "Invert Polarity", "No"

                     "Filters", 0

               "Input"
                  "Internal Filter", "XO1k"
                  "External Filter"
                     "GFB"
                     "Format", "General Lspk"
                     "FormatVersion", "1.0"

                     "Bypass", "No"
                     "Gain", 0.00
                     "Delay", 0.00
                     "Invert Polarity", "No"

                     "Filters", 0
      "Element"
         "Box Type", "key4WayBox"
         "Gain", 0.00
         "Input Config", "InputConfig"
            "Inputs", 2
               "Input"
                  "Internal Filter", "XO1k"
                  "Override Setting", "*Crossover 250 Hz"
                  "Internal Filters To Override", 2
                     "Override Internal Filter", "keyHF"
                        "GFB"
                        "Format", "General Lspk"
                        "FormatVersion", "1.0"

                        "Bypass", "No"
                        "Gain", 0.00
                        "Delay", 0.00
                        "Invert Polarity", "No"

                        "Filters", 1
                           "Filter", "", ""
                              "Kind of Filter", "1/3rd Spectrum"
                              "Bypass", "No"
                              "Gain", 0.00
                              "Delay", 0.00
                              "Samples", "Log", "FR", 30
                                 "Sample", 25, -60.00, 0.00
                                 "Sample", 31, -54.00, 0.00
                                 "Sample", 40, -48.00, 0.00
                                 "Sample", 50, -42.00, 0.00
                                 "Sample", 63, -36.00, 0.00
                                 "Sample", 80, -30.00, 0.00
                                 "Sample", 100, -24.00, 0.00
                                 "Sample", 125, -18.00, 0.00
                                 "Sample", 160, -12.00, 0.00
                                 "Sample", 200, -6.00, 0.00
                                 "Sample", 250, -3.00, 0.00
                                 "Sample", 315, 0.00, 0.00
                                 "Sample", 400, 0.00, 0.00
                                 "Sample", 500, 0.00, 0.00
                                 "Sample", 630, 0.00, 0.00
                                 "Sample", 800, 0.00, 0.00
                                 "Sample", 1000, 0.00, 0.00
                                 "Sample", 1250, 0.00, 0.00
                                 "Sample", 1600, 0.00, 0.00
                                 "Sample", 2000, 0.00, 0.00
                                 "Sample", 2500, 0.00, 0.00
                                 "Sample", 3150, 0.00, 0.00
                                 "Sample", 4000, 0.00, 0.00
                                 "Sample", 5000, 0.00, 0.00
                                 "Sample", 6300, 0.00, 0.00
                                 "Sample", 8000, 0.00, 0.00
                                 "Sample", 10000, 0.00, 0.00
                                 "Sample", 12500, 0.00, 0.00
                                 "Sample", 16000, 0.00, 0.00
                                 "Sample", 20000, 0.00, 0.00

                     "Override Internal Filter", "keyLF"
                        "GFB"
                        "Format", "General Lspk"
                        "FormatVersion", "1.0"

                        "Bypass", "No"
                        "Gain", 0.00
                        "Delay", 0.00
                        "Invert Polarity", "No"

                        "Filters", 1
                           "Filter", "", ""
                              "Kind of Filter", "1/3rd Spectrum"
                              "Bypass", "No"
                              "Gain", 0.00
                              "Delay", 0.00
                              "Samples", "Log", "FR", 30
                                 "Sample", 25, 0.00, 0.00
                                 "Sample", 31, 0.00, 0.00
                                 "Sample", 40, 0.00, 0.00
                                 "Sample", 50, 0.00, 0.00
                                 "Sample", 63, 0.00, 0.00
                                 "Sample", 80, 0.00, 0.00
                                 "Sample", 100, 0.00, 0.00
                                 "Sample", 125, 0.00, 0.00
                                 "Sample", 160, 0.00, 0.00
                                 "Sample", 200, 0.00, 0.00
                                 "Sample", 250, -3.00, 0.00
                                 "Sample", 315, -6.00, 0.00
                                 "Sample", 400, -12.00, 0.00
                                 "Sample", 500, -18.00, 0.00
                                 "Sample", 630, -24.00, 0.00
                                 "Sample", 800, -30.00, 0.00
                                 "Sample", 1000, -36.00, 0.00
                                 "Sample", 1250, -42.00, 0.00
                                 "Sample", 1600, -48.00, 0.00
                                 "Sample", 2000, -54.00, 0.00
                                 "Sample", 2500, -60.00, 0.00
                                 "Sample", 3150, -66.00, 0.00
                                 "Sample", 4000, -72.00, 0.00
                                 "Sample", 5000, -78.00, 0.00
                                 "Sample", 6300, -84.00, 0.00
                                 "Sample", 8000, -90.00, 0.00
                                 "Sample", 10000, -96.00, 0.00
                                 "Sample", 12500, -102.00, 0.00
                                 "Sample", 16000, -108.00, 0.00
                                 "Sample", 20000, -114.00, 0.00
```

```
         "External Filter"
            "GFB"
            "Format", "General Lspk"
            "FormatVersion", "1.0"

            "Bypass", "No"
            "Gain", 0.00
            "Delay", 0.00
            "Invert Polarity", "No"

            "Filters", 0

      "Input"
         "Internal Filter", "XO1k"
         "External Filter"
            "GFB"
            "Format", "General Lspk"
            "FormatVersion", "1.0"

            "Bypass", "No"
            "Gain", 0.00
            "Delay", 0.00
            "Invert Polarity", "No"

            "Filters", 0
```

Finally, the following is a configuration based on the same GLL as the first example, but with internal filters used to generate a beam directed 20 degrees below the horizontal plane (sound speed is assumed as 340 meters per second, the distance between individual sources is 10 cm).

```
"GLC"
"Format", "GLL Configuration"
"FormatVersion", "1.01"

"Configuration"
   "Used System", "Tapered Column Overridable", "keyCol8Ch", "LS", 1.00,
      "Lspk, Inc."
   "Angle", 0.00
   "Elements", 1
      "Element"
         "Box Type", "keyColumn"
         "Gain", 0.00
         "Input Config", "SteeredMode"
            "Inputs", 1
               "Input"
                  "Internal Filter", "keyFilterDef_0"
                     "Override Setting", "Down 20 deg"
                     "Internal Filters To Override", 8
                        "Override Internal Filter", "1"
                           "GFB"
                           "Format", "General Lspk"
                           "FormatVersion", "1.0"

                           "Bypass", "No"
                           "Gain", 0.00
                           "Delay", 0.00
                           "Invert Polarity", "No"

                           "Filters", 0

                        "Override Internal Filter", "2"
                           "GFB"
                           "Format", "General Lspk"
                           "FormatVersion", "1.0"

                           "Bypass", "No"
                           "Gain", 0.00
                           "Delay", 0.10059416
                           "Invert Polarity", "No"

                           "Filters", 0
                        "Override Internal Filter", "3"
                           "GFB"
                           "Format", "General Lspk"
                           "FormatVersion", "1.0"

                           "Bypass", "No"
                           "Gain", 0.00
                           "Delay", 0.20118832
                           "Invert Polarity", "No"

                           "Filters", 0

                        "Override Internal Filter", "4"
                           "GFB"
                           "Format", "General Lspk"
                           "FormatVersion", "1.0"

                           "Bypass", "No"
                           "Gain", 0.00
                           "Delay", 0.301782479
                           "Invert Polarity", "No"

                           "Filters", 0

                        "Override Internal Filter", "5"
                           "GFB"
                           "Format", "General Lspk"
                           "FormatVersion", "1.0"

                           "Bypass", "No"
                           "Gain", 0.00
                           "Delay", 0.402376639
                           "Invert Polarity", "No"

                           "Filters", 0

                        "Override Internal Filter", "6"
                           "GFB"
                           "Format", "General Lspk"
                           "FormatVersion", "1.0"

                           "Bypass", "No"
                           "Gain", 0.00
                           "Delay", 0.502970799
                           "Invert Polarity", "No"

                           "Filters", 0
                        "Override Internal Filter", "7"
                           "GFB"
                           "Format", "General Lspk"
                           "FormatVersion", "1.0"

                           "Bypass", "No"
                           "Gain", 0.00
                           "Delay", 0.603564959
                           "Invert Polarity", "No"

                           "Filters", 0
                        "Override Internal Filter", "8"
                           "GFB"
                           "Format", "General Lspk"
                           "FormatVersion", "1.0"

                           "Bypass", "No"
                           "Gain", 0.00
                           "Delay", 0.704159119
                           "Invert Polarity", "No"

                           "Filters", 0
               "External Filter"
                  "GFB"
                  "Format", "General Lspk"
                  "FormatVersion", "1.0"

                  "Bypass", "No"
                  "Gain", 0.00
                  "Delay", 0.00
                  "Invert Polarity", "No"

                  "Filters", 0
```

Appendix C. XGFB Filter Bank File - Text File Organization

C.1 General Description

The GFB (Generic Filter Bank) text file describes the properties of a set of filters, e.g. a filter bank. It allows for three different types of filters in any combination. IIR Filters consist of common crossover and parametric filter types. FIR / Measured TF filters are defined using an Impulse Response (IR) or complex Frequency Response (FR) file. The third possibility defines the filter as a 1/3rd octave spectrum.
This file uses the file extension **.xgfb**. ***EASE SpeakerLab*** will allow a manufacturer to provide this file compiled into the GLL file, and/or allow the user to import or export this file for external filters that are user adjustable. When used as an external filter bank in a GLL, the filter bank transfer function can be viewed. ***EASE SpeakerLab*** will also allow the filter bank to be used in *Run* mode, which will allow you to view the results of setting the various filters.

The file consists of several mandatory blocks in defined sequence:

- A **File Header Block**, which contains only information about the format of the file.
- A **Filter Header Block**, which holds general information that applies to this filter bank.
- A **Data Block**, which consists of several individual blocks depending on the type of filters selected, that provide information about the filters used in the filter bank and the parameter settings for each of the filters.

General format conventions:

- In general, each item or list of items of the format is preceded by a keyword in quotation marks.
- The keyword identifies the data that follows.
- Each filter has an optional display name which is displayed to the user.
- Text items, such as display names, are also enclosed in quotation marks. Text items cannot contain additional quotation marks.
- All text items, and display names are limited to 250 characters.
- The only decimal separator is the period (".").
- Adjacent two items in a line are separated by comma (", ").
- Description lines must be continuous; they cannot be continued on a consecutive line.
- Different lines are separated by a carriage return (CR, ASCII 13) or carriage return and line feed (CR and LF, ASCII 13 and ASCII 10).
- Lines are to contain only characters of ASCII values from 32 to 127 (standard letters).
- Comment lines are allowed and must begin with a semicolon in quotes (";").
- The file can contain any number of empty lines or comment lines between description lines.

In the description below, each mandatory keyword is listed, followed by one or more data items. The file should hold all of the blocks; they should be located in proper order and include all sub-blocks.

C.2 File Header Block

The file must begin with a line containing the string token "GFB". This is used by ***EASE SpeakerLab*** to determine the type of file. The next line should state the format of the GFB file. The only type currently allowed is "General Lspk". The third line in the file should be the GFB version number. The current version is "1.0".

```
"GFB"
"Format", "General Lspk"
"FormatVersion", "1.0"
```

C.3 Filter Header Block

This describes the general information that applies to this filter bank.

```
"Bypass", "No"
"Gain", 0.00
"Delay", 0.00
"Invert Polarity", "No"
```

Bypass

A Boolean expression specifying whether the entire Filter Bank is enabled. See **Settings** below.

Settings

The settings for *Bypass* are:

Setting	Description
• *No*	This Filter Bank is enabled.
• *Yes*	This Filter Bank is bypassed.

Gain

Level[dB].

Defines a broadband gain for this filter bank in relative dB.

Delay

Time[ms].

Defines a delay time added to this filter bank in milliseconds.

Invert Polarity

A Boolean expression specifying whether the entire Filter Bank has inverted polarity. See **Settings** below.

Settings

The settings for *Invert Polarity* are:

Setting	Description
• *No*	This Filter Bank has normal polarity.
• *Yes*	This Filter Bank has inverted polarity.

C.4 Data Blocks

C.4.1 Filters

This data block contains the information about each of the filters that are available.

```
"Filters", 3
  "Filter", "First IIR Filter", ""
      "Kind of Filter", "IIR Filter"
      "Bypass", "No"
      "Gain", 0.00
  "Delay", 0.00
```

Filters

Number of filters contained in this filter bank. This is followed by an equivalent number of **Filter** sub-blocks.

Filter

Display Name, Internal Name.

This sub-block contains information on one of the filters available for this filter bank.

The **Filter** keyword has these items

Item	Description
• *Display Name*	Label used for display to the user, does not need to be unique.
• *Internal Name*	The internal name is currently not required and is not in use. An empty text field “” should be used in place of the *Internal Name*.

Kind of Filter *Kind*.

The *Kind* is a text field specifying what kind of filter is to be used.

Settings

Currently there are 3 kinds of filters defined. The settings for *Kind* are:

Setting	Description
• *IIR Filter*	Crossover and Parametric Filters.
• *FIR Filter*	Filter defined using an IR or complex FR.
• *1/3rd Spectrum*	Filter defined using one-third octave complex values.

Bypass A Boolean expression specifying whether the entire Filter Bank is enabled. See **Settings** below.

Settings

The settings for *Bypass* are:

Setting	Description
• *No*	This Filter Bank is enabled.
• *Yes*	This Filter Bank is bypassed.

Gain *Level[dB]*.

Level of this filter in relative dB.

Delay *Time[ms]*.

Delay time of this filter in milliseconds.

C.4.2 Kind of Filter

This additional information is best managed using the user interface provided by ***EASE SpeakerLab***. The information provided below is for advanced implementations of filters only.

C.4.2.a IIR Filters

This set of keywords contains the additional information for an Infinite Impulse Response filter.

```
"Type", "Low-Pass"
"Shape", "Linkwitz-Riley"
"Order", 4
"Frequency", 1000.00
"Alignment", "-6dB"
"Q Factor", 1.00
"Parametric Gain", 10.00
```

Type

Type.

The *Type* is a text field specifying the type of filter to be used.

Settings

Currently there are 7 types of filters defined. The settings for *Type* are:

Setting	Description
• *Low-Pass*	A filter that passes low frequencies and attenuates high frequencies.
• *High-Pass*	A filter that passes high frequencies and attenuates low frequencies.
• *All-Pass*	A filter that passes all frequencies equally, but changes the phase relationship between various frequencies. It does this by varying its propagation delay with frequency. Generally, the filter is described by the frequency at which the phase shift has reached half of the overall phase shift (180° per order).
• *Peak*	A filter that makes a peak or a notch in the frequency response.
• *Peak Sym*	A filter that makes a peak or a dip in the frequency response. The peak response is symmetrical with the dip response.
• *Low-Shelf*	A filter that passes all frequencies, but boosts or cuts frequencies below the cut-off frequency by a specified amount.
• *High-Shelf*	A filter that passes all frequencies, but boosts or cuts frequencies above the cut-off frequency by a specified amount.

Shape

Shape.

The *Shape* is a text field specifying what shape of filter is to be used.

Settings

Currently there are 4 shapes of filters defined. The settings for *Shape* are:

Setting	Description
• *Butterworth (HP, LP & AP only)*	A linear filter with a maximally flat amplitude response in the pass band.
• *Linkwitz-Riley (HP, LP & AP only)*	A linear filter used for crossovers that is also known as a Butterworth squared filter, named after its inventors Siegfried Linkwitz and Russ Riley. The advantage compared to a Butterworth filter is that the gain at the crossover frequency will be 0 dB (when a low-pass and a high-pass of same frequency are combined), so the crossover behaves like an all-pass filter, having a flat amplitude response with a smoothly changing phase response.
• *Bessel (HP, LP & AP only)*	A linear filter with a maximally flat group delay (linear phase response). Bessel filters are often used in audio crossover systems. Analog Bessel filters are characterized by almost constant group delay across the entire pass band, thus preserving the wave shape of filtered signals in the pass band.
• *Sallen-Key (AP, Peak, Peak Sym, Low-Shelf & High-Shelf only)*	A linear filter topology that is also known as a voltage controlled voltage source (VCVS) filter. It was introduced by R.P. Sallen and E. L. Key of MIT's Lincoln Laboratory in 1955. This topology is used to provide a variety of different filter shapes and types. Their steepness is usually only controlled by a band-width parameter or Q factor. These filters are also specifically known as Parametric or Active filters.

Order

Order

Sets the slope of the frequency response.

The allowed values for *Order* are:

Filter Shape	Orders Allowed
• *Butterworth, Bessel*	1, 2, 3, 4, 5, 6, 7, 8
• *Linkwitz-Riley*	2, 4, 6, 8
• *Sallen-Key*	1, 2

Frequency

Frequency[Hz]

Cut-off frequency of the filter. Uses the **Alignment** setting below.

Alignment *Alignment*.

The *Alignment* is a text field specifying what frequency alignment is to be used for the filter.

Settings

Currently there are 4 settings for alignments of filters defined. The settings for *Alignment* are:

Setting	Description
• *-3dB*	Butterworth, Bessel only
• *-6dB*	Linkwitz-Riley, Bessel only
• *Phase*	Bessel only
• *None*	Sallen-Key only

Q Factor *Q Factor*

For Sallen-Key filters. A measure of the filter's sharpness. The Q of a band pass filter is the ratio of the center frequency to the difference between the frequencies where the signal is attenuated 3 dB (the 3 dB bandwidth)

Parametric Gain *Gain[dB]*
Level of the boost or cut (negative value) in dB for Sallen-Key filters.

C.4.2.b FIR / Measured TF Filters

This set of keywords contains the additional information for a complex transfer function or a Finite Impulse Response filter. In this version of the format, data can only be given in the frequency domain.

```
"Frequency Samples", 8193
"Time Samples", 16384
"Sample Rate", 44100.00
"Sample", 0.00, 0.00, 0.00
 …
```

Frequency Samples *Number*.

The **Frequency Samples** keyword has these items

Item	Description
• *Number*	*Number* of samples in the frequency domain available for this filter definition. This is followed by an equivalent number of **Sample** keywords below.

Time Samples *Number*.

The **Time Samples** keyword has these items:

Item	Description
• *Number*	*Number* of samples in the time domain used for this filter definition. This value is used to be able to reconstruct the exact time length from the samples in the frequency domain. The number of frequency samples is not unique for this purpose.

Sample Rate *Number*.

The **Sample Rate** keyword has these items:

Item	Description
• *Number*	The *Number* of samples per second used for this filter definition. The sample rate is used to relate the following sample values to frequencies.

Sample *f[Hz], Real, Imaginary*.

This keyword contains information on one of the frequency samples used in this filter definition.

The **Sample** keyword has these items:

Item	Description
• *f[Hz]*	Frequency (Informational).
• *Real*	The Real part of the complex number for the (Fourier) Coefficient. Normalized to 1, as 1+j0 is the ideal All-Pass.
• *Imaginary*	The Imaginary part of the complex number for the (Fourier) Coefficient. Normalized to 1, as 1+j0 is the ideal All-Pass.

C.4.2.c 1/3 Spectrum Filters

This set of keywords contains the additional information for a one-third octave spectrum filter.

```
"Samples", "Log", "FR", 30
"Sample", 25, 0.00, 0.00
"Sample", 31, 0.00, 0.00
"Sample", 40, 0.00, 0.00
"Sample", 50, 0.00, 0.00
"Sample", 63, 0.00, 0.00
"Sample", 80, 0.00, 0.00
"Sample", 100, 0.00, 0.00
"Sample", 125, 0.00, 0.00
"Sample", 160, 0.00, 0.00
"Sample", 200, 0.00, 0.00
"Sample", 250, 0.00, 0.00
"Sample", 315, 0.00, 0.00
"Sample", 400, 0.00, 0.00
"Sample", 500, 0.00, 0.00
"Sample", 630, 0.00, 0.00
"Sample", 800, 0.00, 0.00
"Sample", 1000, 0.00, 0.00
"Sample", 1250, 0.00, 0.00
"Sample", 1600, 0.00, 0.00
"Sample", 2000, 0.00, 0.00
"Sample", 2500, 0.00, 0.00
"Sample", 3150, 0.00, 0.00
"Sample", 4000, 0.00, 0.00
"Sample", 5000, 0.00, 0.00
"Sample", 6300, 0.00, 0.00
"Sample", 8000, 0.00, 0.00
"Sample", 10000, 0.00, 0.00
"Sample", 12500, 0.00, 0.00
"Sample", 16000, 0.00, 0.00
"Sample", 20000, 0.00, 0.00
```

Samples *Spacing, Domain, Number*.

This keyword contains information on one of the filters in this filter definition.

The **Samples** keyword has these items:

Item	Description
• *Spacing*	Selects a logarithmic or linear spacing between the frequency values. See **Settings** below. Currently the only valid value is Log.
• *Domain*	Selects whether the filters are in the Time or Frequency domain. See **Settings** below. Currently the only valid value is FR.
• *Number*	*Number* of samples available for this filter definition. Currently the only valid value is 30. This is followed by an equivalent number of **Sample** keywords.

Settings

The settings for *Spacing* are:

Setting	Description
• *Lin*	Use linear frequency spacing.
• *Log*	Use logarithmic frequency spacing.

The settings for *Domain* are:

Setting	Description
• *IR*	The samples are in the time domain.
• *FR*	The samples are in the frequency domain.

Sample *f[Hz], Level, Phase*.

This keyword contains information on one of the filter bands used in this filter definition.

The **Sample** keyword has these items:

Item	Description
• *f[Hz]*	Frequency band (Informational).
• *Level*	Level in relative dB
• *Phase*	Phase in relative degrees.

C.5 Examples

Following is the header portion of a system definition for an IIR Filter:

```
"GFB"
"Format", "General Lspk"
"FormatVersion", "1.0"

"Bypass", "No"
"Gain", 0.00
"Delay", 0.00
"Invert Polarity", "No"

"Filters", 1
   "Filter", "", ""
      "Kind of Filter", "IIR Filter"
      "Bypass", "No"
      "Gain", 0.00
      "Delay", 0.00
      "Type", "Low-Pass"
      "Shape", "Linkwitz-Riley"
      "Order", 4
      "Frequency", 1000.00
      "Alignment", "-6dB"
      "Q Factor", 1.00
      "Parametric Gain", 10.00
```

Following is the header portion of a system definition for a 1/3 Spectrum:

```
"GFB"
"Format", "General Lspk"
"FormatVersion", "1.0"

"Bypass", "No"
"Gain", 0.00
"Delay", 0.00
"Invert Polarity", "No"

"Filters", 1
   "Filter", "", ""
      "Kind of Filter", "1/3rd Spectrum"
      "Bypass", "No"
      "Gain", 0.00
      "Delay", 0.00
      "Samples", "Log", "FR", 30
         "Sample", 25, -96.00, 0.00
         "Sample", 31, -90.00, 0.00
         "Sample", 40, -84.00, 0.00
         …
         "Sample", 12500, 0.00, 0.00
         "Sample", 16000, 0.00, 0.00
         "Sample", 20000, 0.00, 0.00
```

Following is the header portion of a system definition for a FIR Filter:

```
"GFB"
"Format", "General Lspk"
"FormatVersion", "1.0"

"Bypass", "No"
"Gain", 0.00
"Delay", 0.00
"Invert Polarity", "No"

"Filters", 1
   "Filter", "", ""
      "Kind of Filter", "FIR Filter"
      "Bypass", "No"
      "Gain", 0.00
      "Delay", 0.00
      "Frequency Samples", 8193
      "Time Samples", 16384
      "Sample Rate", 44100.00
         "Sample", 0.00, 0.00, 0.00
         "Sample", 2.691650391, 0.00, 0.00
         "Sample", 5.383300781, 0.00, 0.00
         "Sample", 8.074951172, 0.00, 0.00
         "Sample", 10.766601563, 0.00, 0.00
         "Sample", 13.458251953, -0.11425779, 0.054010473
         "Sample", 16.149902344, -0.106140673, 0.106281586
         "Sample", 18.841552734, -0.072427325, 0.15859884
         "Sample", 21.533203125, -0.013504324, 0.197131649
         "Sample", 24.224853516, 0.065054119, 0.210836411
         "Sample", 26.916503906, 0.151111603, 0.19115743
         "Sample", 29.608154297, 0.229500294, 0.133966684
         "Sample", 32.299804688, 0.283934027, 0.043895174
         …
         "Sample", 22050.00, 0.00, 0.00
```

Appendix D. EASE SpeakerLab Installation and Licensing

D.1 Installation Instructions

D.1.1 Microsoft .NET Framework 3.5

Please note that you need to have .NET Framework 3.5 installed before installing ***EASE SpeakerLab***: https://www.microsoft.com/en-us/download/details.aspx?id=21

D.1.2 EASE SpeakerLab Startup

First, please run the Setup file to begin the ***EASE SpeakerLab*** Startup application. Follow the instructions on the screen to install both ***EASE SpeakerLab*** and the ***AFMG Licence Manager***.

D.1.2.a EASE SpeakerLab

This will create a folder for the program and place an ***EASE SpeakerLab*** icon on the Windows Desktop. The default location for this folder is (*C:\Program Files\AFMG\EASE SpeakerLab*) for 32-bit and (*C:\Program Files (x86)\AFMG\EASE SpeakerLab*) for 64-bit.

D.1.2.b AFMG Licence Manager

This will create a folder for the program and place an ***AFMG Licence Manager*** icon on the Windows Desktop. The default location for this folder is (*C:\Program Files\AFMG\AFMG Licence Manager*) for 32-bit and (*C:\Program Files (x86)\AFMG\AFMG Licence Manager*) for 64-bit.

D.1.3 EASE SpeakerLab User Files

Finally, run the ***EASE SpeakerLab*** User Certificate Setup file. Select TYPICAL to install with the preferred settings. This will make the software Licence available to all users of this computer. Select CUSTOMIZE to change this preset to a different location. The installer will then create a folder for the licence files and allow the ***AFMG Licence Manager*** installed above to register a Licence for the software.

D.1.4 Licensing the Software

EASE SpeakerLab is licensed via ***AFMG Licence Manager***. It can be opened using the desktop icon or the Windows Start menu. After it has opened, just click on ***EASE SpeakerLab*** and then on DOWNLOAD LICENCE. Wait a few seconds for the confirmation message and it is done.

D.2 Licensing Instructions

D.2.1 Online Licensing

To further improve your comfort we have created a Licensing web application to run on our web server. It allows you to easily download an ***EASE SpeakerLab*** Licence via internet (being online with your ***EASE SpeakerLab*** computer assumed). This means the software sends the computer's reference information to our web application, which creates Licence information on our server. This information is then automatically downloaded and installed. So with a single button push on DOWNLOAD LICENCE you can unlock ***EASE SpeakerLab***.

By subscribing to the Licence agreement you are entitled to install the program on one computer. After that installation, additional Licences must be purchased. Please see AFMG or your distributor for prices.

If you intend to uninstall ***EASE SpeakerLab*** from the original computer then please upload the Licence information from that computer by clicking on UPLOAD LICENCE. This will allow you to download this Licence again and then unlock ***EASE SpeakerLab*** on a different computer.

D.2.2 Licensing by File

You should only use this option if you are not able to use the online Licensing functions.

D.2.2.a Reference File

The Reference File is a file generated by the ***AFMG Licence Manager*** program and placed in the *My Documents* folder. This file is different for each installation. If you have more than one computer each will have its own Reference File. To order a Licence you must send the Reference File to AFMG (support@afmg.eu) by E-Mail.

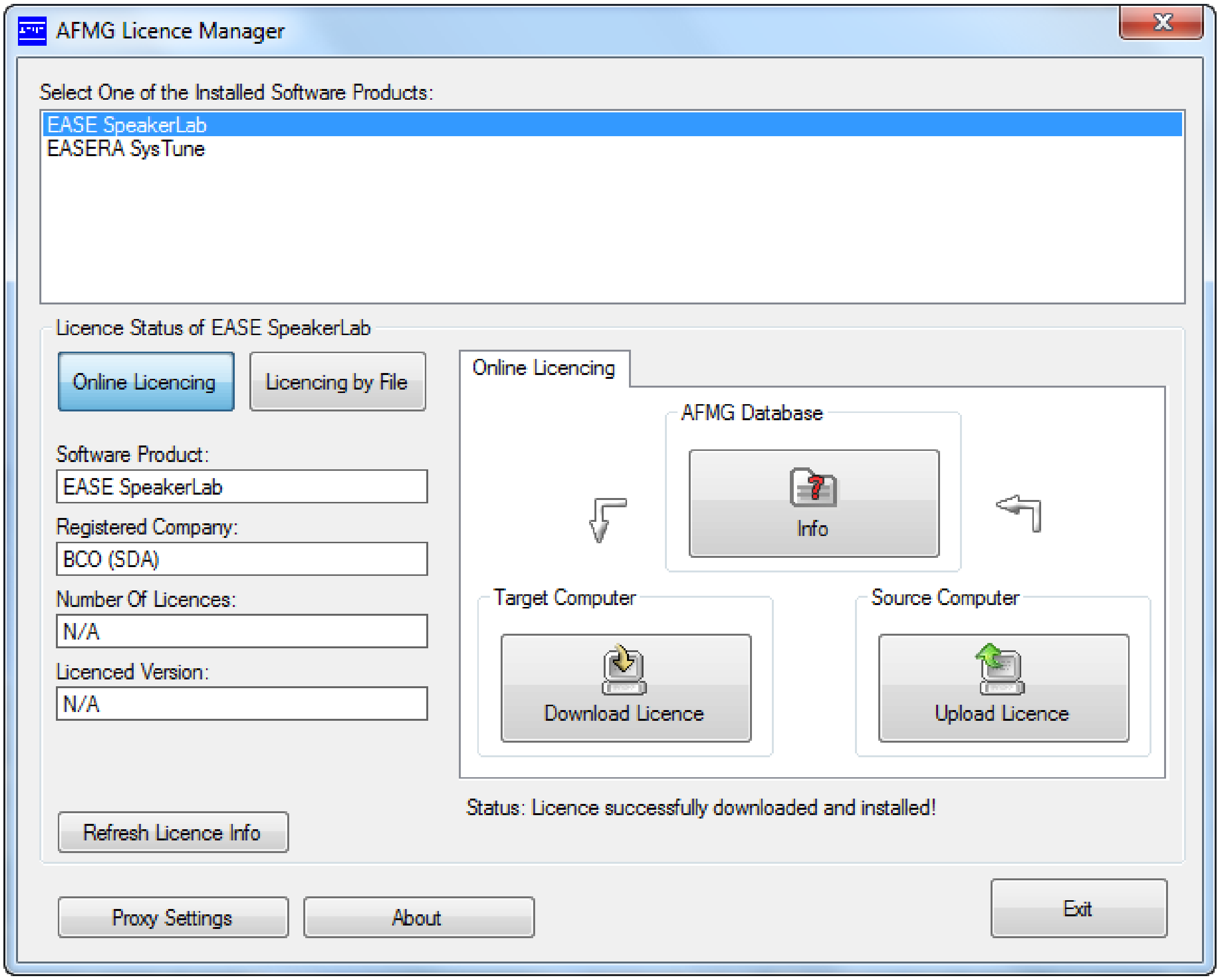
D.2.2.b Licence File

The Licence File is a file generated by AFMG, which is linked to the Reference File. The Licence File is supplied to you by E-Mail. Loading this file into the ***AFMG Licence Manager*** with INSTALL LICENCE unlocks the particular ***EASE SpeakerLab*** version purchased.

If you intend to uninstall ***EASE SpeakerLab*** from the original computer then remove the Licence information from that computer before by using the TERMINATION. TERMINATE LICENCE creates a Termination File which you must send to AFMG (support@afmg.eu) by E-Mail. This will allow you to order a new Licence for the terminated one and unlock ***EASE SpeakerLab*** on a different computer or computers. There can be only one operational program at the same time without additional Licences.

D.3 AFMG Licence Manager Program

This program allows the Licensing of ***EASE SpeakerLab***.



Select One of the Installed Software Products:
EASE SpeakerLab
EASERA SysTune

Licence Status of EASE SpeakerLab

Software Product: EASE SpeakerLab

Registered Company: BCO (SDA)

Number Of Licences: N/A

Licenced Version: N/A

Status: Licence successfully downloaded and installed!

- PROXY SETTINGS: Opens the Proxy Server Configuration window. It allows adjusting proxy server settings for online Licensing through a proxy server.
  See also:

Proxy Server Configuration Window

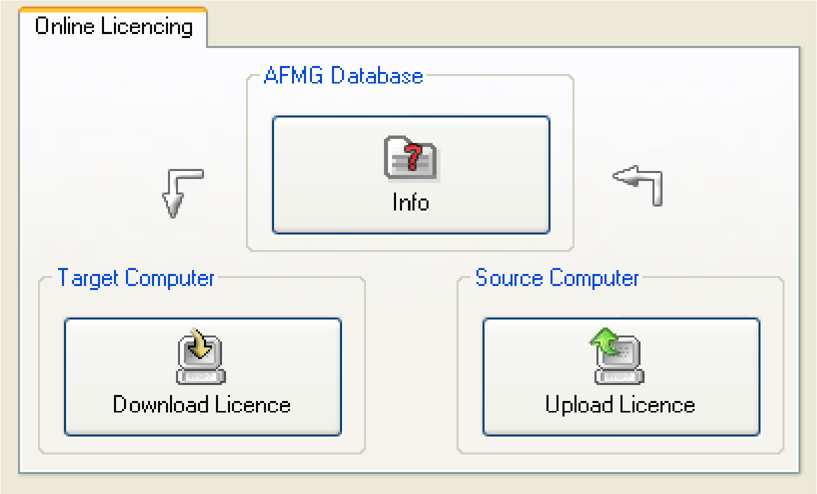
- ABOUT: Shows information about the currently installed ***AFMG Licence Manager***.
- EXIT: Closes the ***AFMG Licence Manager*** window.

D.3.1 Licence Status

- ONLINE LICENSING: Shows the Online Licensing tab in the right part of the ***AFMG Licence Manager***.
- LICENSING BY FILE: If it is not possible for you to be online with your computer an ***EASE SpeakerLab*** Licence can be ordered via email instead. To do that, generate a reference file and send this file to AFMG (support@afmg.eu). As a response you will receive a Licence file from AFMG which is needed to unlock ***EASE SpeakerLab***.

  This button enables three tabs in the right part of the ***AFMG Licence Manager*** labeled LICENCE, TERMINATE, and IMPORT/EXPORT. See below for instructions on how to use these tabs.
- SOFTWARE PRODUCT: This field shows the AFMG software product selected for Licensing.
- REGISTERED COMPANY: This field shows the company name for which the installed Licence is registered.
- NUMBER OF LICENCES: This field shows how many Licences are available on this computer.
- LICENCED VERSION: This field shows which software version is unlocked.
- REFRESH LICENCE INFO: Reloads the Licence information.

D.3.2 Online Licensing



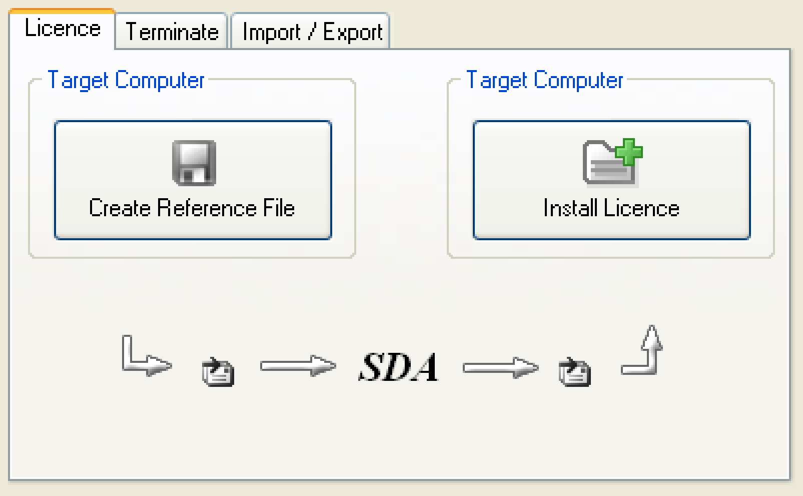
- INFO: Checks the ***EASE SpeakerLab*** Licence database on the AFMG web server and downloads information about the user registration, the purchased version and the number of Licences still available.
- DOWNLOAD LICENCE: Downloads one Licence from the AFMG web server. With ***EASE SpeakerLab*** it is not possible to have more than one licence on the computer.

- UPLOAD LICENCE: Terminates all available Licences. The Licences will be uploaded to the AFMG web server and will be available for new downloads later. If you intend hard disk manipulations or to buy a new computer you should use this procedure to prevent a Licence being lost. It just means a Licence backup for a certain time. After this procedure ***EASE SpeakerLab*** will reset to an unlicensed mode.

D.3.3 Licensing by File

D.3.3.a Licensing by File – Licence Tab

This tab allows a Licence to be installed on this computer. You should only use this option if this computer is not online and a Licence download is not possible.



- CREATE REFERENCE FILE: Creates a Reference File (*.erf format) to send to AFMG. To use the command:
  - Click on CREATE REFERENCE FILE.
  - This opens a SAVE REFERENCE FILE window after a confirmation message.
  - Use the SAVE IN portion of the window to select the folder where you would like to save the Reference File.
  - Click on the SAVE button.

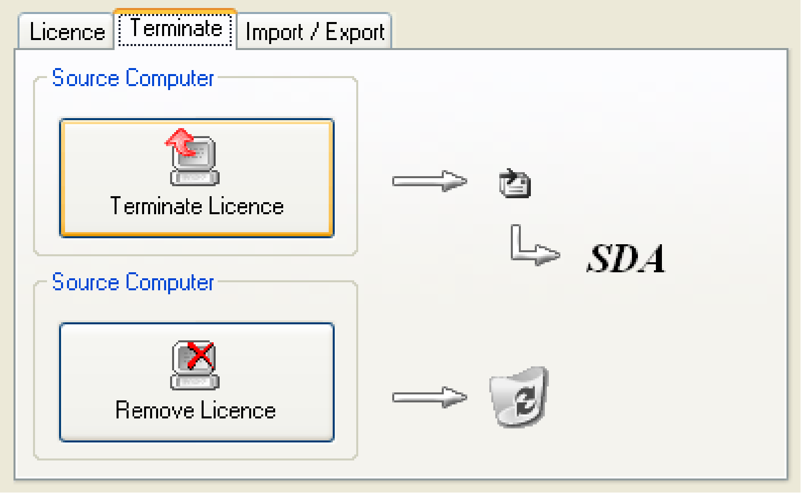
  After saving the file, a "Send Email now?" prompt appears. Click on YES TO automatically send a Licence order with the reference file as an attachment using an installed email client (e.g. MS Outlook or MS Outlook Express) to AFMG. It is also possible to save the file and to mail it later attached to a Licence order to AFMG.

- INSTALL LICENCE: Click to load a Licence File (*.elf format) and install a Licence for ***EASE SpeakerLab***. To use this command: Click on INSTALL LICENCE.
  - This opens an OPEN LICENCE FILE window.
  - Use the LOOK IN portion of the window to find the folder containing the Licence File.
  - Click on the Licence File name.
  - Click on the OPEN button.

  If the reference signature in the Licence File matches this computer, the Licence will be installed. The Licence information and parameters will be shown in the Licence Status frame.

D.3.3.b Licensing by File – Terminate Tab

This tab allows a Licence to be uninstalled or removed from this computer. You should only use this option if this computer is not online and a Licence upload is not possible.



- TERMINATE LICENCE: Creates a Termination File (*.etf format) to send to AFMG. To use the command:
  - Click on TERMINATE LICENCE.
  - This opens a SAVE TERMINATION FILE window after a confirmation message.
  - Use the SAVE IN portion of the window to select the folder where you would like to save the Termination File.
  - Click on the SAVE button.

  After saving the file, a "Send Email now?" prompt appears. Click on YES to automatically send the termination file as an attachment using installed email client (e.g. MS Outlook or MS Outlook Express) to AFMG. It is also possible to save the file and to mail it later to AFMG. There it will be verified and if it is correct you can order a replacement Licence any time for the terminated one. After this procedure EASE ***SpeakerLab*** will be reset to an unlicensed mode.

- REMOVE LICENCE: Click to remove all traces of the Licence on this computer.

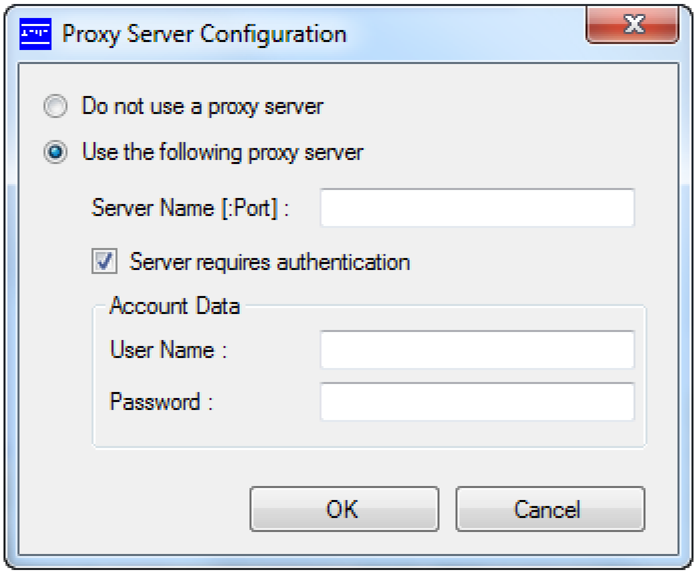
  Note: Be very careful with this button! All Licence information will be deleted from the computer. This option should only be used in case of general Licensing problems due to software or hardware errors. Please contact AFMG before using this command or your ***EASE SpeakerLab*** Licence may be lost completely.

D.3.3.c Licensing by File – Import / Export Tab

This functionality is not supported by ***EASE SpeakerLab***.

D.3.4 Proxy Server Configuration Window

If you are running ***EASE SpeakerLab*** in a secured intranet, access to the internet for online Licensing may be blocked by a firewall, depending on your network's configuration. Configuring ***AFMG Licence Manager*** to use the local proxy server on your network may resolve this issue. If you are unsure of the appropriate proxy server settings, please consult your network administrator.



Proxy Server Configuration

○ Do not use a proxy server

● Use the following proxy server

Server Name [:Port] :

☑ Server requires authentication

Account Data

User Name :

Password :

`Use the following proxy server`

- Server Name [:Port]: Proxy server name or IP address and port number for internet access.
- Server requires authentication: Check if the proxy server needs an additional authentication.
- Account Data: User Name and Password to authenticate on the proxy server.
- OK: Accepts configuration settings and closes the window.
- Cancel: Discards configuration settings and closes the window.