

LspCAD

Professional Users guide

Dedicated to :
Kristina
Jennifer
and
Julia

© Ingemar Johansson, IJData
Luleå, Sweden
Email: ingemar.johansson@ijdata.com
Homepage: <http://www.ijdata.com>

Unauthorized copying of this manual or the associated software is a violation against applicable copyright laws.

LspCAD Professional
September 2002

Notes

Table of contents

1	INTRODUCTION	1
2	INSTALLATION	3
2.1	COPY PROTECTION AND REGISTRATION	3
2.2	TECHNICAL SUPPORT	5
3	GENERAL COMMENTS	6
3.1	MISCELLANEOUS STUFF	6
3.2	LSPCAD DIRECTORY STRUCTURE	9
3.2.1	<i>Default directories</i>	9
3.3	BACKWARD COMPATIBILITY	10
3.4	LOUD IS WONDERFUL	10
4	THE BOX UTILITY	11
4.1	INTRODUCTION	11
4.2	COMMON MENU PICKS AND DIALOGS	12
4.2.1	<i>Driver unit configuration</i>	15
4.2.2	<i>Driver unit parameters</i>	20
4.2.3	<i>General settings</i>	24
4.2.4	<i>Transient response</i>	25
4.2.5	<i>Room and cabinet impact (Room & Cabinet)</i>	26
4.2.6	<i>Snapshot</i>	31
4.2.7	<i>Yankee tool</i>	33
4.2.8	<i>System optimization</i>	33
4.2.9	<i>Filter</i>	35
4.2.10	<i>EQ</i>	35
4.3	CLOSED BOX	36
4.3.1	<i>Introduction</i>	36
4.3.2	<i>The box dialog</i>	36
4.4	BASSREFLEX BOX	38
4.4.1	<i>Introduction</i>	38
4.4.2	<i>The box dialog</i>	38
4.5	DOUBLE TUNED BASSREFLEX BOX TYPE 1	40
4.5.1	<i>Introduction</i>	40
4.5.2	<i>The box dialog</i>	40
4.6	DOUBLE TUNED BASSREFLEX BOX TYPE 2	41
4.6.1	<i>Introduction</i>	41
4.7	PASSIVE RADIATOR BOX	42
4.7.1	<i>Introduction</i>	42
4.7.2	<i>The box dialog</i>	42
4.8	BANDPASS BOX TYPE 1	44
4.8.1	<i>Introduction</i>	44
4.8.2	<i>The box dialog</i>	44

4.9	BANDPASS BOX TYPE 2	46
4.9.1	<i>Introduction</i>	46
4.9.2	<i>The box dialog</i>	46
4.10	BANDPASS BOX TYPE 3	47
4.10.1	<i>Introduction</i>	47
4.10.2	<i>The box dialog</i>	47
4.11	BANDPASS BOX TYPE 4	48
4.11.1	<i>Introduction</i>	48
4.11.2	<i>The box dialog</i>	48
4.12	WHAT DO THE DIAGRAMS SHOW?	49
4.12.1	<i>Introduction</i>	49
4.12.2	<i>Half space (2pi) SPL at 1m distance</i>	49
4.12.3	<i>Cone excursion</i>	49
4.12.4	<i>Port air speed</i>	50
4.12.5	<i>Impedance</i>	50
4.12.6	<i>Group delay</i>	50
4.12.7	<i>Filter response</i>	50
4.12.8	<i>Equalizer response</i>	50
4.12.9	<i>Impulse- Step- Tone burst response</i>	51
4.12.10	<i>SPL in Room & Cabinet</i>	51
4.12.11	<i>SPL with port standing wave modes</i>	51
4.12.12	<i>Power amp. output voltage</i>	51
4.12.13	<i>Snapshot</i>	51
4.12.14	<i>Information</i>	51
4.13	NONLINEARITY SIMULATIONSON	52
5	THE SIMPLE PASSIVE CROSSOVER UTILITY	53
5.1	INTRODUCTION	53
5.2	MENU PICKS.....	55
5.3	THE NETWORK DIALOGS.....	57
5.4	THE DRIVER DIALOGS	59
5.5	THE GENERAL SETTINGS DIALOG.....	61
5.6	THE MUSIC FILTERING FUNCTION	64
5.7	WHAT DO THE DIAGRAMS SHOW	65
5.7.1	<i>Filter gain</i>	65
5.7.2	<i>Input impedance</i>	65
5.7.3	<i>Summed frequency response</i>	65
5.7.4	<i>Individual phase response</i>	65
5.7.5	<i>Group delay</i>	65
5.7.6	<i>Impulse response</i>	65
5.7.7	<i>Vertical off axis response, surface plot</i>	66
5.7.8	<i>Vertical off axis response, overlay plot</i>	66
5.7.9	<i>Polar plot</i>	66
5.7.10	<i>Snapshot</i>	66
5.8	A FEW HINTS	67

5.8.1	<i>Optimizing frequency/phase response</i>	67
5.8.2	<i>Optimizing the zobel network</i>	67
5.8.3	<i>How to use the notch section</i>	67
6	THE ADVANCED CROSSOVER UTILITY	68
6.1	GENERAL SETTINGS	70
6.2	THE NETWORK DIALOGS	70
6.2.1	<i>The common network</i>	73
6.2.2	<i>The network optimizer</i>	74
6.3	THE SYSTEM OPTIMIZER	78
6.4	THE DRIVER DIALOGS.....	81
6.5	PASSIVE CROSSOVER BRANCHES.....	82
6.5.1	<i>Parallel topology</i>	82
6.5.2	<i>Series topology</i>	83
6.5.3	<i>Circuit elements</i>	84
6.5.4	<i>The HP/LP filter wizard</i>	86
6.6	ACTIVE CROSSOVER NETWORKS	87
6.6.1	<i>Parameter/Component mode</i>	88
6.6.2	<i>Circuit elements</i>	88
6.6.3	<i>The HP/LP filter wizard</i>	91
6.7	WHAT DO THE DIAGRAMS SHOW (CONT.)	91
6.7.1	<i>Input impedance</i>	91
7	THE CROSSOVER EMULATOR	92
7.1	WHAT IS THE GAIN?.....	92
7.2	HOW DOES IT WORK?	92
7.3	HOW EXACT IS IT?	94
7.4	USERS GUIDE	96
7.4.1	<i>System sounds and sounds from other applications</i>	96
7.4.2	<i>Sound cards</i>	96
7.4.3	<i>Error conditions</i>	96
7.4.4	<i>Output configurations (3D soundcards)</i>	97
7.4.5	<i>Output configurations (Multimedia cards with WDM drivers)</i>	97
7.4.6	<i>Output configurations (Multichannel soundcards)</i>	97
7.4.7	<i>Reverse polarity</i>	98
7.4.8	<i>Logfile</i>	98
7.4.9	<i>Entities of the dialog box</i>	99
7.4.10	<i>Output settings</i>	102
7.5	THE PARAMETRIC EQUALIZER.....	104
7.6	KNOWN PROBLEMS AND TRICKS.....	105
7.6.1	<i>Necessary filter order (IIR filter type)</i>	105
7.6.2	<i>More than one driver in a network</i>	105
7.6.3	<i>Poor matching of low-pass filter (IIR filter type)</i>	106
7.6.4	<i>Poor matching of high-pass filters (IIR filter type)</i>	106
7.6.5	<i>Using the sets</i>	106

7.6.6	<i>Overloading output</i>	107
8	OTHER IMPORTANT ISSUES	108
8.1	BUZZ WORDS.....	108
8.2	EXPORT/IMPORT FILE FORMATS.....	110
8.2.1	<i>Export file formats</i>	110
8.2.2	<i>Import file formats</i>	111
9	A SMALL TUTORIAL	112
9.1	LOUDSPEAKER BOXES.....	112
9.1.1	<i>Transient response characteristics</i>	112
9.1.2	<i>Power handling</i>	114
9.1.3	<i>Some construction tips</i>	116
9.1.4	<i>Cabinet edge diffraction</i>	117
9.1.5	<i>Room impact</i>	118
9.2	CROSSOVER NETWORKS.....	119
9.2.1	<i>Filter order and alignment</i>	119
9.2.2	<i>Measuring driver units</i>	121
9.2.3	<i>Constructing zobel networks</i>	123
9.2.4	<i>Constructing the rest of the crossover</i>	124
9.2.5	<i>System optimization</i>	127
10	LITTERATURE	128

1 Introduction

Congratulations to your purchase of LspCAD Professional.

LspCAD is a software that helps you to construct and model loudspeaker boxes and passive crossover networks. The name LspCAD is short for **Loudspeaker Computer Aided Design**.

LspCAD includes four utilities. The **box utility** that works as a box modeling program, the **passive crossover utility** that works as a filter modeling program with passive components and the **advanced passive/active crossover utility** that works as a filter modeling program with passive and active components.

LspCAD does not only present ready to use solutions, a number of diagrams shows virtually all information needed in order to construct passive and active loudspeaker systems. For starters it must be mentioned that this manual covers the program mainly, i.e. the goal is not to be a construct-your-loudspeakers-in-twentyfour-hours book, please feel free to read the tutorial section in chapter 9. This chapter gives a brief introduction how to use LspCAD in the construction work. See chapter 10 for literature recommendations.

The box utility

The box utility manages to model 9 different kinds of loudspeaker boxes with dynamic loudspeaker units, these are:

- Closed box.
- Bass reflex.
- Double tuned bass reflex. Two types.
- ABR (passive radiator).
- Bandpass 1.
- Bandpass 2.
- Bandpass 3.
- Bandpass 4.

The bandpass boxes are named as above due to the lack of decent names for these boxes.

The different box types are described later on in this manual.

LspCAD has a powerful tool for modeling room response and cabinet diffraction effects which will further help you as a user to create good sounding loudspeaker systems.

An active/passive filter (that can be optimized simultaneously with the box parameters) and an equalizer is also included in the box utility.

The simple passive crossover utility

The simple passive crossover utility can model 2-way and 3-way loudspeaker systems. Besides the rudimentary filter calculator existing in most loudspeaker modeling software the user can here import measured SPL and impedance data and thus model a real system. With the help from a unique feature one can filter a music sample through the loudspeaker systems frequency response and listen to the result through a pair of high quality headphones.

The advanced active/passive crossover utility

The advanced passive crossover utility can model 1-way through 4-way loudspeaker systems with up to four loudspeaker units in each network. Modeling capabilities are much more extensive than for the simple passive crossover utility. A major benefit is the capability to optimize the filter components in order to achieve a predetermined transfer function, this makes construction of complex filter structures as easy as a walk in the park.

A novel, "hotdog" feature called crossover emulation makes it possible to emulate the transfer function of the crossover in real time. In other words you can test a number of different crossovers without even touching an inductor. All that is needed is simply put, a four channel (3D) soundcard and an four channel power amplifier. If you are looking for more advanced setups then go for an Echo Darla24 audio card and be able to emulate crossovers for four way stereo systems.

System requirements

The demands on your IBM compatible are:

- At least an intel-Pentium compatible CPU.
- 64Mb RAM, the program consumes up to 40Mb RAM when started.
- About 3 Mb disk space.
- Display resolution 640x480 is minimum, 800x600 or more is recommended for the box utility, for the crossover utility 1024x768 resolution is recommended.
- Math co-processor.
- Mouse or other pointing device, makes life much easier.
- Windows 95/NT/98/2000 (Windows 3.x is not automatically supported as LspCAD is a 32bit application).
- A color printer, this is optional, but it will make printouts easier to read.

2 Installation

The software is easily installed with the enclosed installation program.

Start windows.

1. Insert the CD into the CDROM drive of your PC.
2. Pick **File / Run** in the program manager. Under Windows 95 you pick **Run** in the start menu.
3. On the command line you enter :
[x]:setup.exe followed by OK.
[x] is replaced by the name of your CDROM drive.
4. Follow the steps of the installation program

2.1 Copy protection and registration

LspCAD is equipped with a commercial copy protection software.

To be able to run LspCAD a unique site code is presented to you the first time you start LspCAD. This site code must be emailed to technical support. This is also a good opportunity to register your copy of LspCAD.

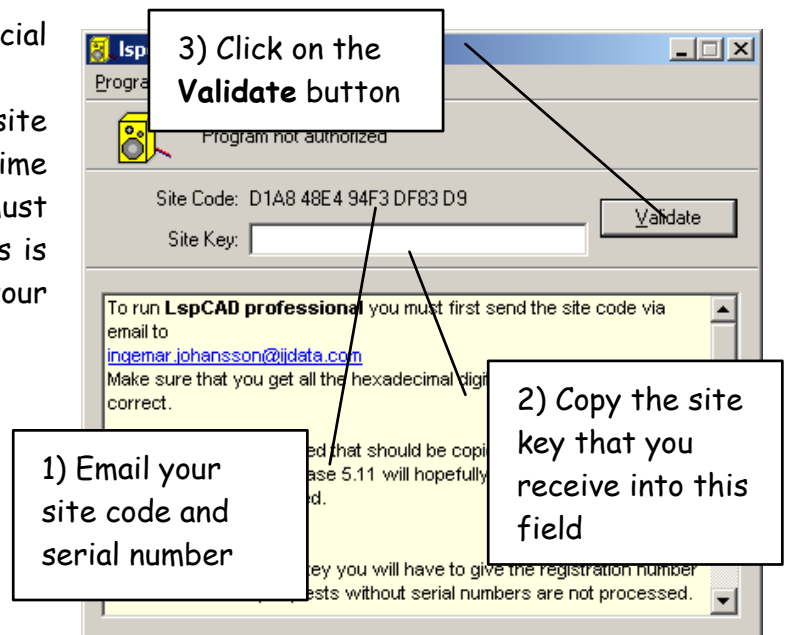
Send a simple email with
(Company name)
Name
Address
Email
Serial number
Site code

All information should be emailed to ingemar.johansson@ijdata.com

This is all that is necessary to register and receive a site key in return.

If you email the registration info, it is not necessary to mail the registration form via snail-mail.

The unique site key that is emailed back should be entered into the empty site key field. After this the **Validate** button is pressed. If the site key was correct, access to LspCAD is granted. Please note that once you have entered the site key, this site key will no longer be valid, this is normal behavior and part of the copy protection software. You should (in normal cases) not need a new site key if you download a new release of LspCAD.



IMPORTANT! Remember to include your serial number in the email, requests without serial number are silently ignored..

The serial number can be found either on the registration card or on the CD The serial number contains of four digits and four letters (eg. 0566-GHKY).

Important about Windows NT/2000/XP:

In order to be able to start LspCAD under Windows NT/2000/XP you need to execute the small program `setupex.exe` that is located in the LspCAD folder. Thus you need to follow the steps below. This is most easily done via the explorer.

- Locate the file `setupex.exe` in the LspCAD folder with the explorer.
- Double click with the mouse on the file so that it will execute.
- Done!, now you should be able to run LspCAD

Licence loss

The license may be lost for a different number of reasons. One surefire way to lose the license is to rename the LspCAD folder. In other words, don't mess with the LspCAD folder name. More sophisticated things such as moving the harddisk from one PC to another will also cause a license loss, note that you will get a new site key but in order to reduce the amount of administrative work (and stupid questions) from me it is recommended to be careful with the handling of the license.

Norton Utilities Speed Disk and Crypkey (the copy protection software)

Speed Disk is the defragmentation utility included in Symantec's Norton utilities.

Unfortunately, Speed Disk moves Crypkey Instant Licensing files, causing licence loss.

Crypkey uses four special licensing files (`.ent`, `.rst`, `.key`, and `.41s`) to control the product licensing. These files are hidden files and reside in the LspCAD folder.

To prevent license loss:

1. Open Speed Disk, and choose File, Options, Customize and Unmovable files.
2. Specify that the `*.ENT`, `*.RST`, `*.KEY`, and `*41S` files cannot be moved.

Moving the license

A license can be moved to another directory or another PC, moving the license to another PC may be good if one for instance one need to upgrade the PC with a new operating system or upgrade the harddisk. For this purpose the following steps should be performed

- Start LspCAD and hit the ENTER key when the first startup window appears. The license configuration window will appear.
- To **move the license to another directory** (folder), click on the *License / Transfer to directory...* menu pick, follow the instructions. Note that it is assumed that you have already installed LspCAD in the directory that you want to move the license to.
- To **move the license to another PC**, click on the *License / Transfer out to another computer...* menu pick and follow the instructions. Like the case above it is assumed that you have already installed LspCAD in the PC that you want to move the license to.

Killing a license

Killing a license is recommended if you don't have another PC to move the license to and wish to do maintenance on the PC that may cause license loss. Click on the *License / Kill license* menu pick. You will be prompted with a question if you really wish to kill the license, if you answer yes a window with a kill code will appear.



Copy the kill confirmation code into an email and email it with your name and serial number to ingemar.johansson@ijdata.com. By doing this you will make sure that you will get a new site key without questions.

2.2 Technical support

Technical support is given only to registered users of LspCAD. Questions from unregistered users are not answered.

When you have problems or questions please send an email to:

ingemar.johansson@ijdata.com

Remember to give your name and the LspCAD serial number in the email even though you have registered your copy earlier. The serial number is your ticket to technical support.

Technical support will **not** be given via snail-mail. This means that you need access to the internet. This requirement should not cause too much trouble these days. Moreover future upgrades will mainly be communicated via internet.


3 General comments

3.1 Miscellaneous stuff

The software for LspCAD for Windows is developed under Borland C++ 5.02. The installation program is used with permission from INSTALLSHIELD.

In order to not be worse than the big companies, it is hereby stated that no responsibility is taken for damages (personal and/or other) caused by using this software.

LspCAD is very straightforward and simple to use, however it is advisable to at least glance through this manual quickly before using the software.

Most interaction with the programs is made through dialogs, the dialogs are covered in detail in this manual, moreover the same and sometimes even additional info is revealed by clicking on the question-mark button that is visible in most dialogs,  then a notepad document with help info will pop up. The extensive use of dialogs is quite neat as one does not need to cram every piece of information onto the screen, the drawback may be that the user only sees a blank screen on start up.

There exist a number of buzz words in the MS-windows world, some of which are covered below. By clicking is meant clicking with the left button on the mouse.

- Dialog

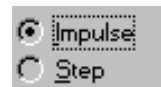
A window that lets you modify numeric fields, check or uncheck checkboxes e.t.c, most of the dialogs in LspCAD are non-modal, i.e they can be left open on the screen. The dialogs are usually accessed via the main menu of the program. As some of the dialogs are very large and tend to occupy a major part of the screen, they can be resized just like ordinary windows. When the dialogs are made smaller scrollbars will show up on the bottom and on the right side.

- Button

Similar to the question mark button above, by clicking on a button one normally initiate a computation or loads a file.

- Radiobutton

Radiobuttons are generally grouped 2 or more and lets the user select between two or more alternatives, only one radiobutton in a group can be selected at a time.



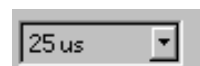
- Checkbox

A checkbox toggles between on (checked) and off (unchecked) when the user clicks on it. A checkbox is used when the user wishes to enable or disable a certain feature.



- Listbox or combobox

A list box allows the user to select items from a list, or to select between alternatives. An interesting feature with many of the listboxes in the advanced crossover utility is that one can modify the values in two ways, either double click and enter a new value, or selecting with the mouse pointer and pressing the arrow left/right keys on the keyboard.



- **Edit box**

An editbox lets the user type in numerical values or strings. There are three types of edit boxes (or fields) in LspCAD. 1) Strings, lets you type in e.g a project name. 2) Integer, lets you type in integer values (e.g 1000 Hz). 3) Floating point, lets you type in floating point values (e.g 2.83V). One unique feature with LspCAD is that the integer and floating point edit boxes can be modified by means of the arrow-up and arrow-down (or arrow-left/arrow-right) keys on the keyboard, each modification of an edit field also initiates an update of the diagrams in LspCAD, this makes it simpler for the user to modify values and see what happens in "real time".
- **Controls**

A common name for the things that constitute a dialog such as edit boxes, buttons and listboxes.
- **Group**

A group is a rectangle that encloses one or more controls in a dialog.
- **Diagrams**

A diagram in LspCAD lets the user view e.g the frequency response of a loudspeaker system, these diagrams follow the rules for MDI-windows in MS-windows with the exception that they cannot be closed. They can, however be rescaled and minimized to icons. A diagram consist of one or more plots. When there is a risk that the plots may be confused, the plots are tagged and explanatory text is shown in the diagram.
- **Drag and drop**

Various files are imported into LspCAD, in the box utility it may be unit (.unt) files while in the crossover utility it may be measured data such as frequency response or impedance data. The normal way to import this data is via e.g the Browse button in the Driver dialog, it is however also possible to import data via a drag and drop procedure where one simply move the file one wish to import, from the Microsoft file explorer into LspCAD. If the file is dropped into the LspCAD main window, it is treated as a project file. If the files are dropped into the dialogs the action depends on where they are dropped in the dialogs. This will be described later.

When LspCAD is started what comes up is only a blank screen. The user must here select what he/she wish to do. The menu pick *File / New / Box...* and

File / New / Filter... lets you select which kind of project you want to start.

Fig. 3.1 shows the looks of the menu picks for the box projects.

For instance if we wish to model a closed box the menu pick *File / New / Box / Closed* will create a closed box project.

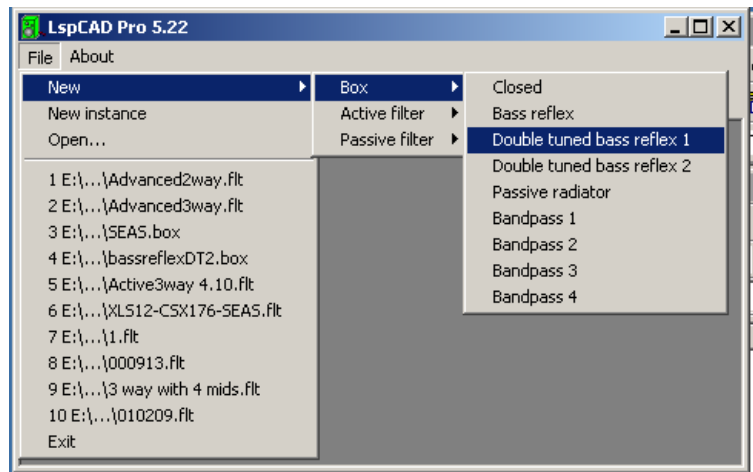


Fig. 3.1

3.2 LspCAD directory structure

When storing driver unit files and measured data it is strongly recommended that you put these data in a subdirectory of the LspCAD directory, the simple reason is that LspCAD then strips away the first part of the path in the path. For instance the path:

C:\PROGRAM\LSPCAD\DRIVER\DYNAUDIO\17W75.UNT

will be truncated to become:

DRIVER\DYNAUDIO\17W75.UNT

This makes it simple to share project files with other LspCAD users.

3.2.1 Default directories

Default directories:

Default directories for import and export functionality can now be set via the environment variables:

LSPCADPROJ

Default project directory, also default directory for snapshot files.

LSPCADIMPORT

Default import directory.

LSPCADEXPORT

Default export directory.

In windows 98 these environment variables can be set in the file autoexec.bat.

An example is

```
set LSPCADPROJ=c:\myprojects\lspcad\proj\
```

```
set LSPCADIMPORT=c:\myprojects\lspcad\import\
```

```
set LSPCADEXPORT=c:\myprojects\lspcad\export\
```

Double quotes may be needed if the directory path contains white space, for example one can write

```
set LSPCADEXPORT="c:\program files\lspcad\export\"
```

In windows NT the environment variables are set from the control panel, under the system properties icon.

It is important that all directory paths have a backslash '\' in the end.

3.3 Backward compatibility

The driver unit files created under LspCAD 2.10 and later can be used in LspCAD Professional.

LspCAD project files created from LspCAD 3.0 and later can be used by LspCAD Professional.

Additional upgrade information can be found at:

<http://hem.passagen.se/ijdata>

3.4 LOUD is wonderful

That is true, question is however *_how_* loud ?. Less than 6 years ago I managed to stand on the dance floor at a discotheque for quite a while. Today I have to leave the dance floor after 5 minutes. Moreover it is virtually impossible to hold a conversation although you are 20m away from the loudspeakers.

I do not know if it is age or not, but fact of the matter is that high sound pressure levels are dangerous to your hearing. If you expose yourself to too much noise (i.e too loud music), then on a sunny day your ears will yell ENOUGH, and if you are unlucky they will continue to yell for the rest of your life. Moreover, you may not be concerned with it today but you might find it nice to hear your grandchildren sing someday.

So, pump of the volume with decency and be aware of the fact that if your ears tell you that it is too loud then it is most likely way too loud.

To all you speaker-building addicts out there, be aware of the risk of too high sound pressure levels and make your friends aware of it too, also consider the fact that while you may enjoy your critical listening at +100dB or so, the experience may not be as pleasant for your neighbors.

Hope you enjoy the software and also that it will contribute to many hours of pleasant listening both for your ears and for your mind.

4 The box utility

4.1 Introduction

The box utility is the part of LspCAD which lets the user model various type of loudspeaker boxes. The main object here is to model the low-frequency part i.e we aim mainly at boxes for the bass region, however boxes for midrange units can be modeled here too.

The project files created by this utility has **.box** as standard extension but one may use other extensions as well.

A multitude of diagrams can be viewed here, they are:

- Frequency and phase response "on axis". This diagram also shows acoustic contribution from ports and driver (loudspeaker) units.
- Cone excursion.
- Air speed in ports.
- Impedance response with phase.
- Group delay.
- Impulse, step or toneburst response.
- Frequency and phase response when things such as baffle diffraction and listening room has been taken into account.
- Frequency response when standing wave modes in ports are taken into consideration (bandpass boxes only).
- Snapshot, a diagram that gives you the opportunity to compare different constructions regarding frequency response on axis and in room/cabinet.
- Filter and equalizer response, displays the transfer function of the active/passive filter and the equalizer.

There also exist an information window that gives an overview of the project.

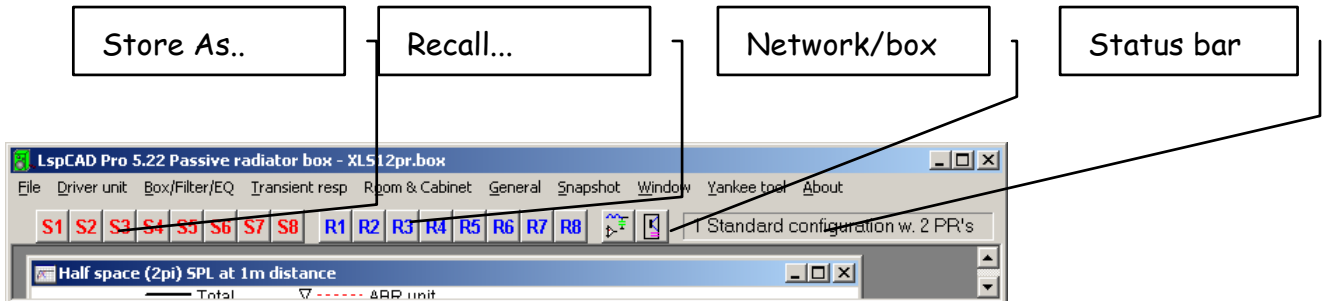
The menus and dialogs that are common are described first after that the different box projects are described. Finally the different box types are described in detail.



When a new box project is created the program will ask you to select a driver unit, this is needed in order to get the program working. The main menu will also be changed then.

4.2 Common menu picks and dialogs

The user interface is arranged such that only the **Box** menu pick differs between the different box types, this makes it easy for the user to get a grip on the user interface very quickly. Besides the menu there exist a button/status bar in the main window.



See below a description of the menu picks that are common for all box types:

- **File / New...**
Create a new project. Already explained in chap 3.
- **File / New instance**
Start a new instance of LspCAD.
- **File / Open project..**
Load a project.
- **File / Save project , File / Save project as...**
Save a project.
- **File / Print / Window..**
Print active window.
- **File / Print / Diagrams..**
A dialog will show up where you can select which diagrams should be printed. The *Snapshot* and *Information* diagrams cannot be selected here. The procedure is simple, just select from the pick list which diagrams should be printed and how many diagrams per page you want and click on the print button. The diagrams are printed in the order they are selected.
- **File / Copy window**
Copy active window, a diagram is copied as a bitmap while the information window is copied in ASCII format.

- **File / Export / ...**
Export diagrams in LAUD (.frd, .zma), LEAP (.gdt), CLIO (.txt) or MLSSA (.fmp, .fm) format (see chap 8.2). The diagrams that can be exported are:
 - *Free air SPL at 1m distance.*
 - *SPL in room/cabinet.*
 - *SPL with standing wave modes (only bandpass boxes).*
 - *Transient response.*
 - *Input Impedance.*
 - *Group delay.*
 - *Filter response*
- **File / Exit**
Terminate the program.
- **Driver unit / Configuration..**
Set number of driver units, electrical connection etc. See chap 4.2.1.
- **Driver unit / Load..**
Load a driver (loudspeaker) unit.
- **Driver unit / Edit / create**
Edit driver unit parameters. See chap 4.2.2
- **Box/Filter/EQ..**
Set the box properties such as Volume, port tuning etc. Described in detail later for each box type. Here one can start the system optimizer and also store/recall sets.
- **Box/Filter/EQ / System optimization**
Starts the system optimizer.
- **Box/Filter/EQ / Store as...**
A nice feature that allows you to store away up to eight different network setups. Fast access to this function is via the **Store** buttons available below the menu bar. The stored sets are saved with the project file.
- **Box/Filter/EQ / Recall...**
Recall a stored setup. Similar to the **Store** buttons a set of Recall buttons are available. If **Prompt on recall** is enabled (see ch. 4.2.3) one have the opportunity to select what items to recall.
- **General**
General settings. See chap 4.2.3.
- **Transient resp.**
Here you choose different setups for the transient response diagram. See chap 4.2.4.
- **Room & Cabinet.**
Here you select the setup for the simulation of the room and cabinet effects. See chap 4.2.5
- **Snapshot.**
Open the snapshot dialog. See chap 4.2.6.

- ***Window / Arrange icons***
Arrange the icons in the client area.
- ***Window / Cascade***
Cascade the windows in the client area.
- ***Window / Tile***
Set the open windows side by side.
- ***Window / Default size***
Set the size of the selected window to a default size.
- ***Window / User note...***
Here you have the option to enter comments (max. 50 characters) in the diagrams, or max. 9 rows or 600 characters of user information in the information window.
- ***Window / Show...***
A curse that comes with applications that use many diagrams is that in the end there are so many windows to keep track of that it is difficult to get an overview over the design. Therefore it is possible to close the diagrams (except snapshot) and bring them back with this menu pick.
- ***Yankee tool***
A simple conversion tool for people who are more acquainted with US standard units. See chap 4.2.7.
- ***justMLS***
A click on this menu pick will start justMLS, this assumes that justMLS.exe is located in the LspCAD folder.
- ***About...***
Show some valuable information about this fantastic software and its author.

4.2.1 Driver unit configuration

In this dialog you can set number of driver units, isobaric configuration and electrical connection of multiple driver units and double voice coil electrical connection. Here you also import measured data.

- **Number of drivers**
Number of driver units (1-12).
- **Extra mass**
Extra mass in grams applied on the loudspeaker units diaphragm in order to lower the resonance frequency.

Isobaric

- **Yes**
Check this checkbox if you want an isobaric configuration of two loudspeaker units that will then constitute a driver unit.
- **Series**
The two loudspeaker units of the driver unit are electrically connected in series.
- **Parallel**
The two loudspeaker units of the driver unit are electrically connected in parallel.
- **Separate sources**
The two loudspeaker units of the driver unit are electrically connected to separate sources.

Double voice coil electrical conn.

If the loudspeaker unit has double voice coils you have here the option to select the electrical connection of the two voice coils.

- **Series**
The two voice coils are electrically connected in series.
- **Parallel**
The two voice coils are electrically connected in parallel.
- **Sep. sources**
The two voice coils are electrically connected to separate sources.
- **Load resistor**
This feature makes it possible to put a load resistor across one voice coil, thus it is possible to tune system total Q in a neat way. A zero ohm resistor will give the lowest Q while an open circuit will give a higher Q.

Additional analysis (also chap 4.13)

- **BL nonlinearity**
Simulate BL (force factor) non linearity.
- **Voice coil heating**
Simulate heating of voice coil.
- **Cms non linearity**
Simulate progressive stiffness in the suspension.
- **Impact from Le**
Simulate the impact that the driver inductance has on the frequency response.
- **Directivity**
Simulate the impact on the on axis response that the directivity factor of the driver gives. This feature can be adjusted between 0.0 (no directivity) and 1.0 (full directivity). The reason to that this feature is adjustable is that there are no general means to simulate this behavior, thus it is better to leave to the user to adjust this to match e.g a measurement of the loudspeaker unit. Fig. 4.1 below displays the simulated frequency response for different settings. The directivity factor simulation has no

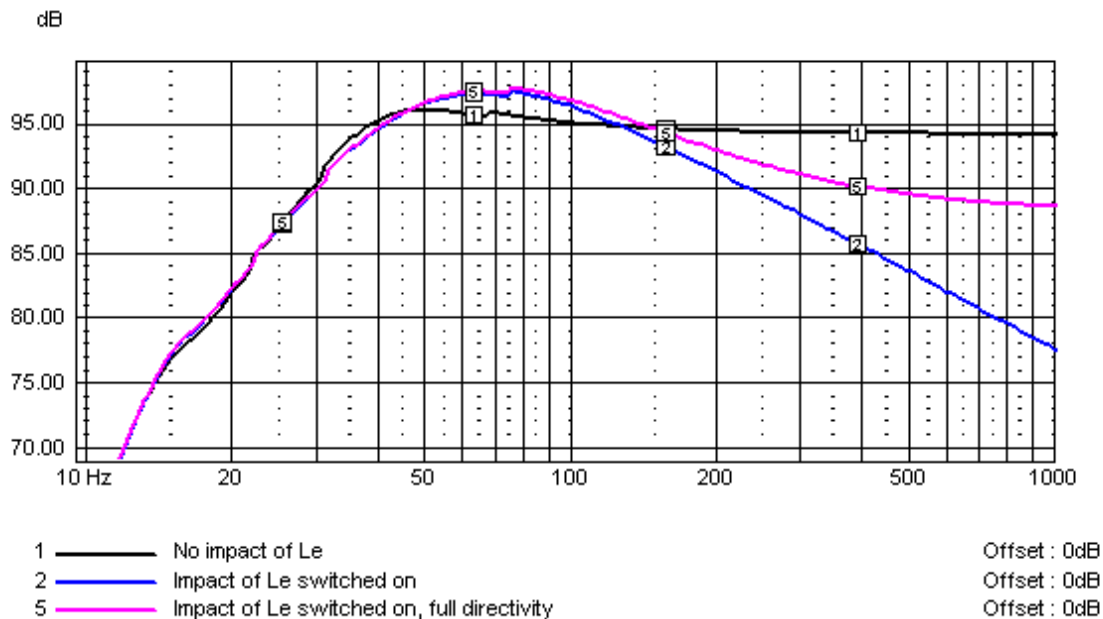


Fig. 4.1 Simulated frequency response for an Adire Audio Shiva in a 48l box tuned to 25Hz with different settings.

meaning for bandpass boxes and is thus disabled for those boxes.

- **Measured SPL data**
Here you import measured SPL (frequency response) data. The imported data is merged together with the simulated data so that the imported data occupy the frequency range above the **Transition frequency** in the diagrams. This feature makes it easier to achieve more realistic measurements as the lowest frequency region is often more accurately simulated than measured.

- **Level offset**

Offset in dB to add to the imported data in order to make a seamless transition from simulated to measured data.



If you want to keep the offset of the measured data to a zero value the source level in the *measurement setup* dialog can be modified instead.

- **Transition frequency**

Frequency at which a switch from simulated data to measured data should be made.

- **Phase lag cutoff**

When pasting together simulated and measured data an unnatural jump in phase response often occur that affects the *Group delay* and *Transient response* diagrams negatively. The value in the phase lag cutoff is used to compute the phase response of a fourth order Butterworth LP-filter with the cutoff frequency equal to **Phase lag cutoff**. The phase is added to the phase of the simulated data, thus we are able to make a seamless transition in the phase response.

- **Automatic level offset**

With this checkbox checked the offset will be calculated automatically

- **Disable**

With this checkbox checked the use of imported data will be disabled

- **Browse**

A file open dialog will pop up where you can select an SPL data file for loading.(it is also possible to drop the file inside the **Measured SPL data** box).



When measured SPL data is disabled or not imported the controls in this group has no effect.

Electrical conn...

In the listbox that pops up you have the opportunity to select how two or more driver units are electrically connected, see fig. 4.2. There is always the option to select series, parallel and separate feeding of the driver units.

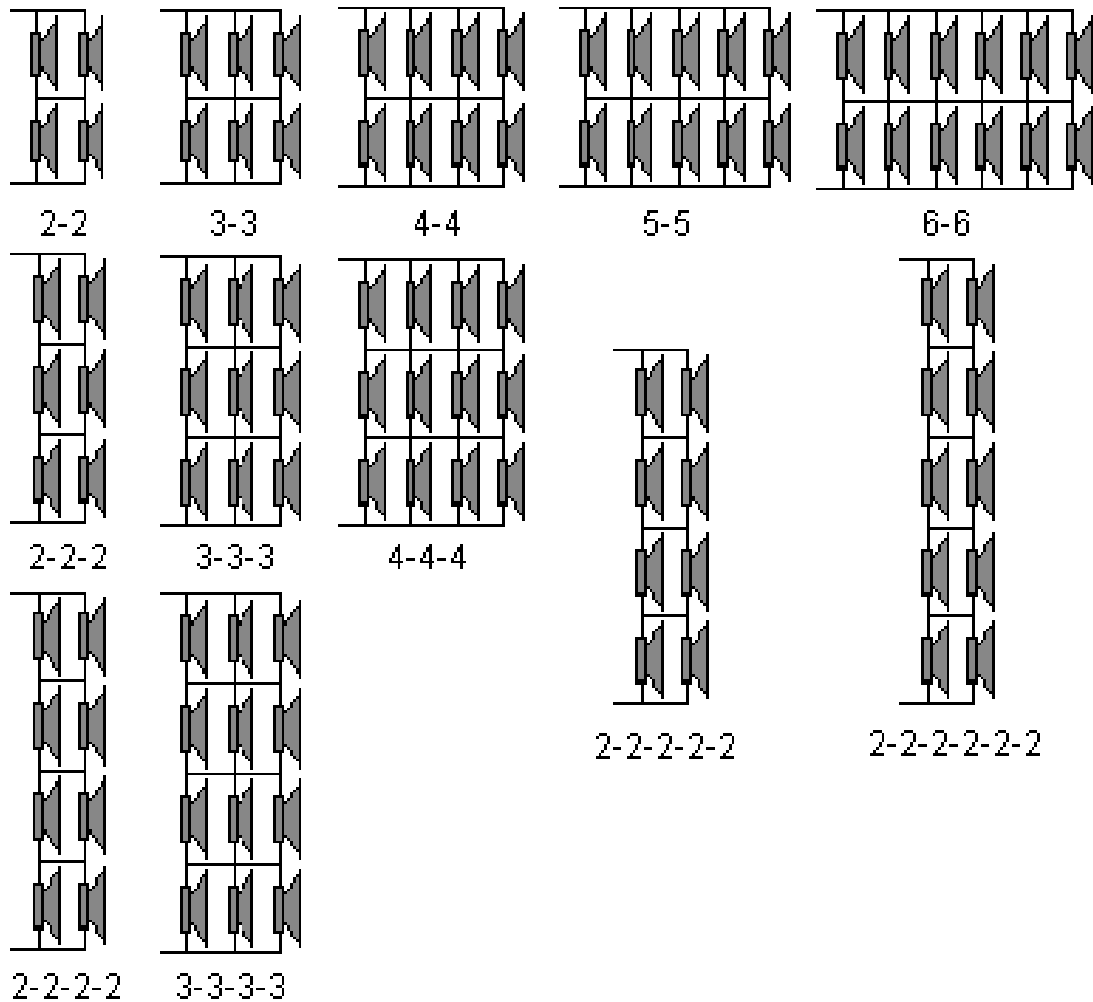


Fig. 4.2 Configuration of four or more driver units

The concept of electrical connection of driver units may sometimes become a tricky matter especially if one also selects nine driver units with isobaric configuration and double voice coils.

Fig. 4.3 shows a case where 4 loudspeaker units with double voice coils are used with configuration 2-2 and the double voice coils are connected for separate sources. In this case the number of driver units is 4.

Fig. 4.4 shows a slightly simpler example where we have 4 loudspeaker units connected in an isobaric configuration two by two. In the **Isobaric** field the **Yes** checkbox should be checked and parallel connection of the isobaric pairs should be selected. The number of driver units in this case is 2 as we use **Isobaric** configuration. The electrical connection (**Electrical conn...**) of the driver units is set to **series** as the driver units are connected in series.

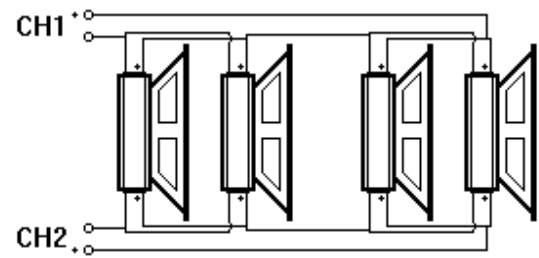


Fig. 4.3

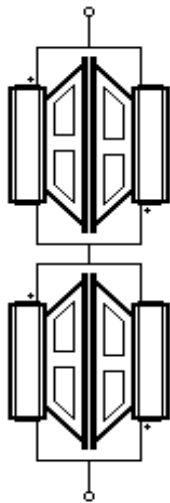


Fig. 4.4

4.2.2 Driver unit parameters

Edit driver unit parameters. You can create your own loudspeaker parameter files that are used by LspCAD. The parameter files use `.unt` as default extension, but nothing stops you from using another extension.

Below follows a description of the various menu picks.

- **File / New**
Erase all information.
- **File / Open...**
Load a parameter file.
- **File / Save, File / Save as...**
Save a parameter file.
- **File / Print data...**
Print the loudspeaker parameters.
- **File / Copy data**
Copy the loudspeaker parameters to windows clipboard.
- **File / User note...**
Here you may enter your own comments, up to 600 characters or 9 rows of information may be entered.
- **File / Exit**
Terminate program.
- **Calc. missing fields.**
Calculate the fields that are missing if possible, this menu pick is grayed if **Crosscalc.** is checked.
- **Bl(x) & Cms(x).**
Enter a $Bl(x)$ and $Cms(x)$ curve for better nonlinearity modeling , requires that the **Use $Bl(x)$ & $Cms(x)$ model** checkbox is checked.

Speaker unit editor

File Calc. missing fields Bl(x) & Cms(x)

Manufacturer: Adire Audio
 Model: Brahma 12"
 Filename: Brahma 12 w Blx Cmsx model 2.unt

Thiele-Small Parameter

Re	1.29	Ohm
Le	2.07	mH
Reb	undef.	Ohm
Leb	undef.	mH
fs	28.1	Hz
Qms	5.06	
Qes	0.99	
Qts	0.83	
Rms	8.00	Ns/m
Mmd	224.20	g
Mms	228.67	g
Cms	1.4e-04	m/N
Cas	2.7e-07	m ⁵ /N
Vas	37.23	l
Sd	436.0	cm ²
Bl	7.25	N/A

Pmax: 800 W
 Xmax: 25.0 mm p-p
 Airgap height: 8.0 mm
 Voicecoil height: 33.0 mm

Type:
 Bass
 Bass/Midrange
 Midrange
 Mid./Treble
 Treble
 Full range
 Passive radiator

Crosscalc Double voice coil
 Use Bl(x) & Cms(x) model

Infinite baffle eff.
 0.08 %
 81.1 dB/1 W/1 m

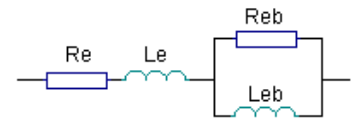
In the dialog window you can set a number of loudspeaker parameters.

- **Manufacturer** Manufacturers name.
- **Model** Model name of the loudspeaker unit.
- **Type** Enter the range that the loudspeaker unit works within. For the moment there is no practical use of this.
- **Re** Voice coil resistance [Ω]
- **Le** Voice coil inductance [mH]
- **Reb** Voice coil extended model parameter [Ω]
- **Leb** Voice coil extended model parameter [mH]

The simple way to model the voice coil inductance is an inductor (L_e) in series with the voice coil DC resistance (R_e).

This model not very useful for highpower long stroke subwoofers where the voice coil inductance is high at low

frequencies and substantially lower and high frequencies. For this purpose an extended model is implemented where the voice coil inductance is modeled by L_e as before plus a resistor (R_{eb}) and an inductor (L_{eb}) in parallel. R_{eb} and L_{eb} can be left unused if wanted, just set R_{eb} and $L_{eb} = 0$ (Undef.), this is default.



- **fs** Resonance frequency in free air [Hz]
- **Qms** Mechanical Q-value
- **Qes** Electrical Q-value
- **Qts** Total Q-value
- **Rms** The loss resistance of the suspension [Ns/m]
- **Mmd** Moving mass excluding surrounding air [g]
- **Mms** Moving mass including surrounding air [g]
- **Cms** The mechanical compliance of the suspension [m/N]
- **Cas** The acoustic compliance of the suspension [m⁵/N]
- **Vas** Equivalent volume [l]
- **Sd** Cone area [cm²]
- **Bl** Force factor [N/A]
- **Pmax** The maximum, thermal limited input power [W]
- **Xmax** The maximum linear excursion, NOTE peak to peak value [mm_{p-p}]
- **Airgap height** Airgap height of the magnet [mm]
- **Voiccoil height** The voice coil winding height [mm].

- **Efficiency** Shows the efficiency of the loudspeaker unit with 1W/8Ω and 1m distance.
- **Crosscalc.** When this checkbox is checked the program will always try to force the parameters to coincide with each other.
- **Use Bl(x) & Cms(x) model.** Enables the user to enter Bl and Cms as a function of displacement. When this checkbox is checked a Bl(x) Cms(x) edit dialog will pop up on the screen. See 4.2.2.1 for more details
- **Double voice coil** Check this checkbox if the loudspeaker unit has double voice coils. **Note that you have to enter the Thiele-Small parameters that you get when one of the terminals are left open, i.e only one voice coil is driven.** Often the DVC drivers are measured with the voice coils either in series or connected in parallel. In this case a few parameters must be modified before they are entered. See table below for a guideline hoe the parameters should be modified before they are entered in the driver unit editor. Qts should not be entered, it is better to let the editor calculate it for you.

	Measured with VC in series	Measured with VC in parallel
Re	*0.5	*2.0
Le	*0.5	*2.0
Qes	*2.0	*2.0
Bl	*0.5	*1.0



The loudspeaker parameters that are used by LspCAD are Sd, Cms, Mmd, Rms, Re, Le, Pmax and Xmax. Make sure therefore that these parameters are correct

Drag and drop. If files are dropped into the driver unit edit dialog, they are treated as LspCAD driver unit (.unt) files.

4.2.2.1 BI(x) Cms(x) edit

This edit window lets the user enter BI and Cms values as a function of cone displacement. This will improve the reliability of the nonlinearity simulations. Values of BI and Cms vs displacement are entered in two different edit windows. Data is entered in two columns where the first column is the displacement values in mm from the resting position and the right column is BI or Cms value

BI(x):

When the BI(x) values are entered one can chose to enter them either as absolute values or a values relative to the BI value when the cone is in the resting position. This is controlled with the **Rel BI at Omm** checkbox. When the validate button is clicked all lines are controlled for sanity and ordered, all lines that have an illegal format are removed.

Displacement [mm]	BI [N/A]	Displacement [mm]	1/Cms
-30.0	3.002	-30.0	2.0000
-20.0	7.000	-22.5	1.5000
-15.0	7.250	-15.0	1.2000
0.0	7.250	-7.5	1.0000
15.0	7.000	0.0	1.0000
22.5	7.000	7.5	1.1000
30.0	4.749	15.0	1.1500
		22.0	1.2000
		30.0	1.3000
		37.5	1.5000

Cms(x):

When Cms(x) values are entered one can also here chose to enter them as absolute values or relative to Cms at the resting position (**Rel Cms at Omm** checkbox). It is also possible to enter 1/Cms values or Kms as they are also referred to (**1/Cms** checkbox).

Besides entering values by hand one can also paste them with the keyboard command **SHIFT+INSERT**.



Remember to set the checkboxes in the correct state before you enter values as the values are automatically recalculated as you check/uncheck any of the checkboxes.

4.2.3 General settings

In this dialog you set general stuff such as input level etc.

Display frequency response

The internal frequency span in the crossover utility is always 1-30000Hz. In order to be able to view a smaller part of the frequency span one may here set the **Lower** and **Upper** limits.

Diagram scaling

The scaling of the diagrams that present impedance, groupdelay filter gain and sound pressure level can be altered here. Moreover it is possible to hide the phase curve in the *Free air SPL at 1m distance* and the *SPL in Room & Cabinet diagrams*

Source characteristics

- **Source level**

Input level from source (i.e. amplifier) in Vrms.

Limit. The maximum output voltage delivered by the power amplifier.

- **Watts into 8 ohm**

Input power from source in Watts over an 8Ω load.

- **Rg (for table lookup)**

The output impedance of the power amplifier or the series resistance of the crossover or connection cable. This is only used in the table lookup box optimization that exist for some box types.

The **Prompt on recall** checkbox is by default checked, this will make a dialog box pop up every time one recall a set, in this dialog one can check what items one wish to recall. If this feature is not desired on can uncheck this option.

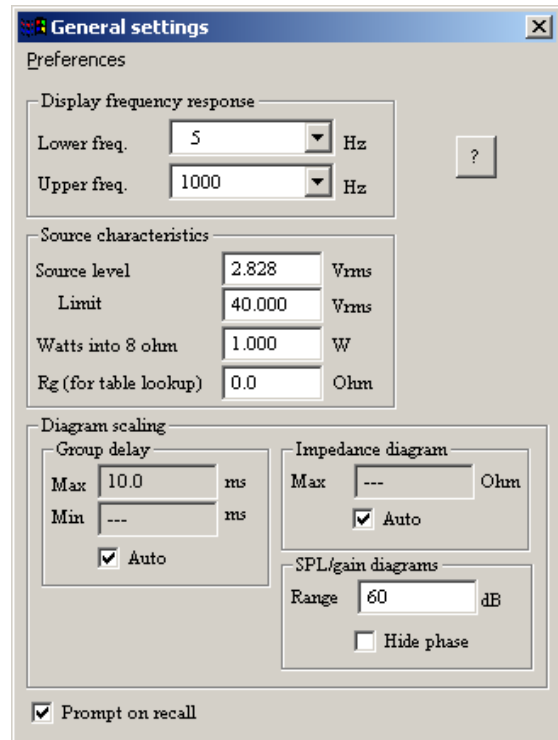
A small menu exist in this dialog:

Preferences / Save

Allows the user to save the preferences so that the next time a project is created, the saved settings will be used from the beginning.

Preferences / Set to default

Will reset all settings.



4.2.4 Transient response

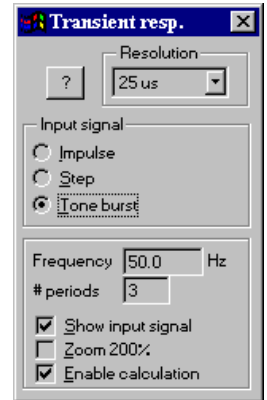
Here you setup the appearance of the transient response diagram. See also chap 4.12.7 and 9.1.1.

Resolution

Select time resolution from 25 μ s to 2 ms.

Input signal

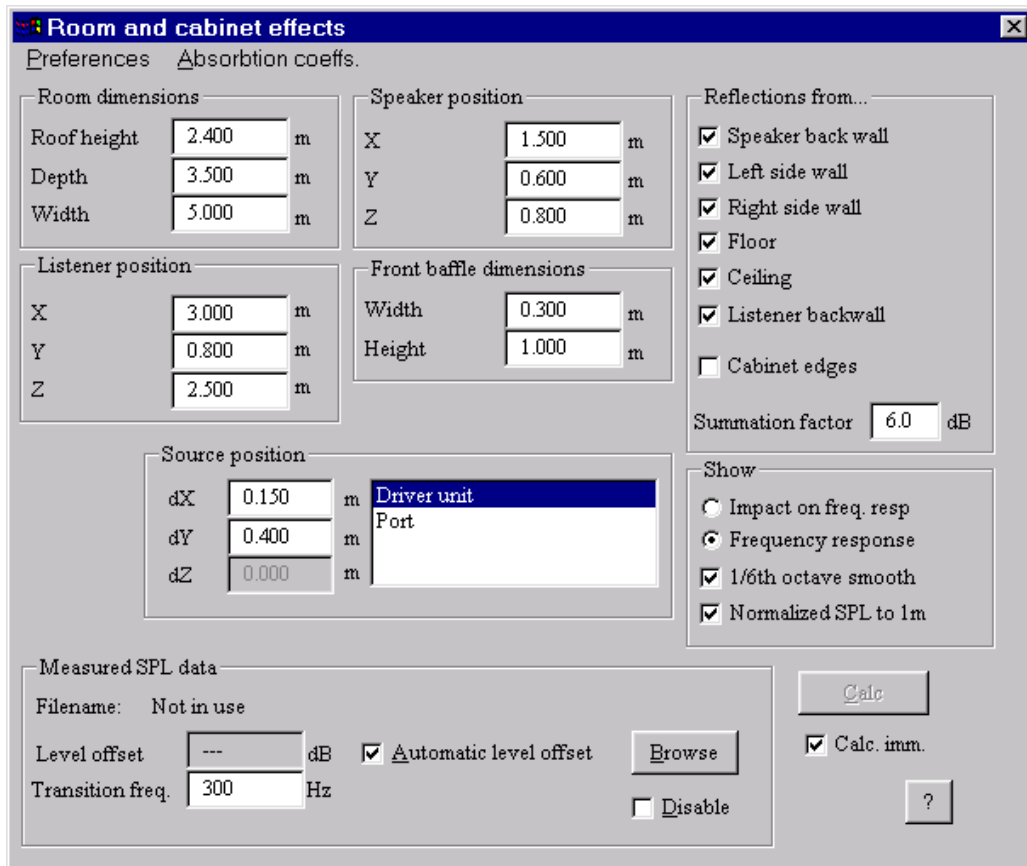
- **Impulse response**
View impulse response.
- **Step response**
View step response.
- **Tone burst**
View tone the response of a tone burst with **#periods** and **Frequency** Hz.
- **Show input signal**
Check this checkbox if you want to view the input signal (tone burst only). The input signal is drawn with a 3 times lower amplitude than the output signal just to improve readability.
- **Zoom 200%**
Check this checkbox if you only want to see the first half of the diagram.



The looks of the the transient response, especially the impulse response and the step response may be heavily distorted if the time resolution is set too coarse, this will show up in that the negative peak gets larger as the resolution gets coarser, therefore keep the time resolution as fine as possible for the box type in question.

4.2.5 Room and cabinet impact (*Room & Cabinet*)

The room and cabinet simulation is controlled by means of a dialog that can be left open on the screen.



The room and cabinet feature gives you the option to watch the loudspeaker system behavior under "real" circumstances, i.e. in a room and with a normal limited-sized cabinet. This gives you the opportunity to simulate cabinet edge diffraction and also the nearest reflections from floor, ceiling and walls. Please note that the intention of this tool is not to model resonances in a room, the objective of this tool is to make it possible to construct loudspeaker systems that can cope with real life conditions in a better way.

The model assumes that the loudspeaker is placed in a room given by the drawing in fig. 4.5.

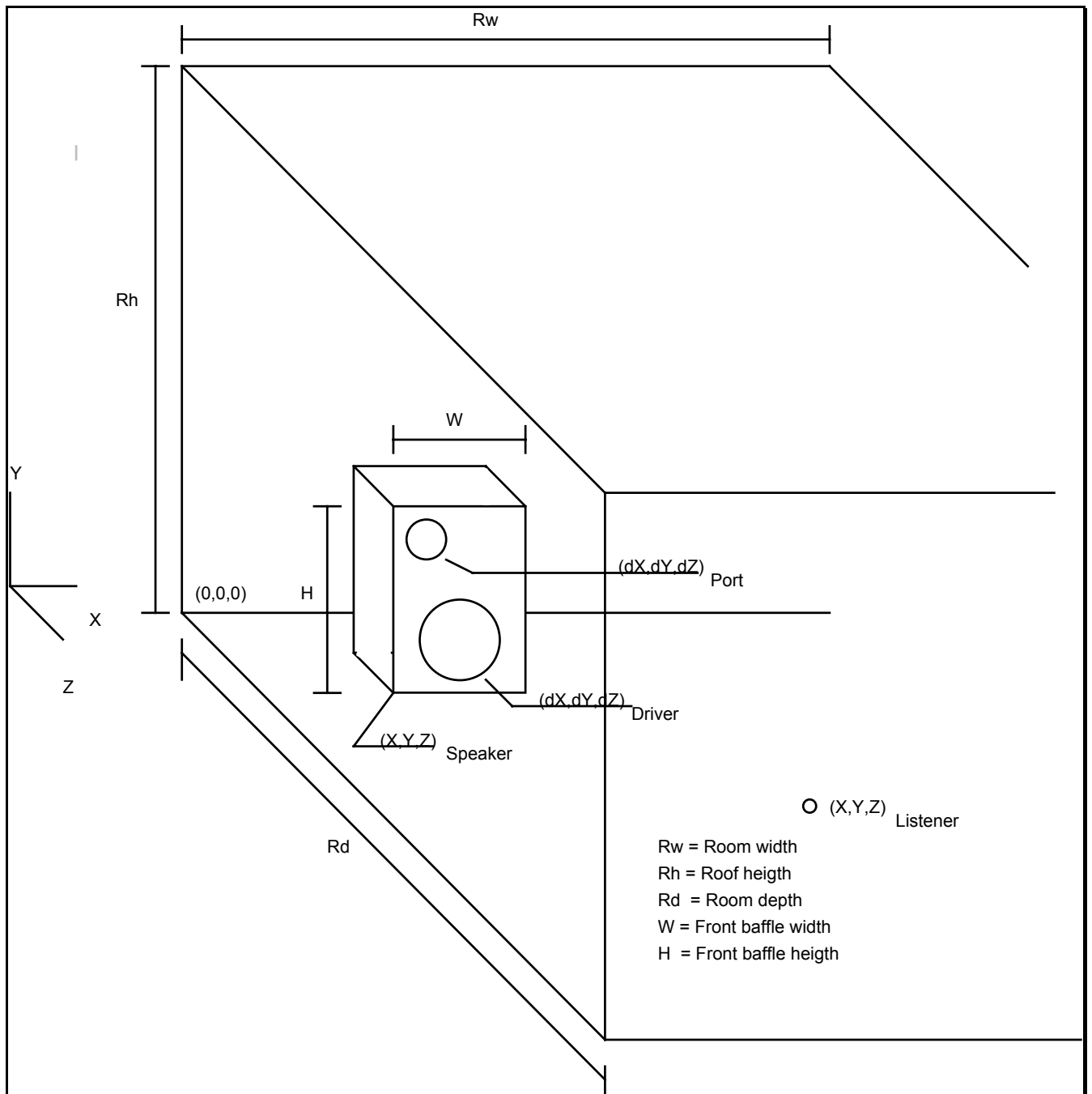


Fig. 4.5 Loudspeaker in listening room

Quite a few parameters can be entered here. The dialog pane is divided in groups. All positions are in a three-dimensional coordinate system where origin is denoted $(0,0,0)$ in fig. 4.5. The x-axis is the direction left to right, the y-axis is the direction down to up and the z-axis is the direction front to rear.

Room dimensions

Set the dimensions of the listening room. The width denotes the size in the x-axis, the height is the size in the y-axis and the depth is the size in the z axis.

Speaker position

Set the position of the lower left corner of the front baffle of the loudspeaker.

Listener position

Set the position of the listener

Front baffle dimensions.

Set the **Width** and **Height** of the front baffle.

Source position

Set the position of the different visible acoustic entities (driver, ports e.t.c) of a box, for a bandpass box, only the ports are visible. The positions (dX, dY, dZ) are relative to the **Speaker position**.



Because the simulation of cabinet edge diffraction only works satisfactorily with the driver unit mounted on the front baffle, the driver units cannot be positioned in the Z-axis (dZ). For the band pass boxes it is not possible to simulate cabinet edge diffraction effects.

Reflections from...

The room reflections that are possible to simulate are:

- **Speaker back wall** (the wall behind the loudspeaker)
- **Left side wall**
- **Right side wall**
- **Floor**
- **Roof**
- **Listener backwall** (the wall behind the listener).

And also double and triple reflections i.e. combinations of the above e.g. **Floor-Roof**, **Side wall-Floor** etc.

- **Cabinet edges**

If checked it allows the user to simulate cabinet edge diffraction. The model assumes a rectangular shaped box, but also works for moderately rounded shaped edges as it has little effect on the 3-6 dB step in the frequency response that stems from the diffraction.

- **Summation factor**

There exist some discussion on how much a reflection in phase rises the sound pressure, here you can select the desired value 3-6 dB that suits your religious beliefs best.

Show

- **1/6 octave smooth Smooth room/cabinet frequency and phase response.**
- **Normalize SPL to 1m**
Normalize the sound pressure to what it would be at 1m distance from the loudspeaker.
- **Impact on freq. resp.**
Enables you to see the impact on the frequency response i.e. how the frequency response is affected by room and cabinet while a click on **Frequency response** enables you to see the frequency response

Calculations can be initiated either automatically (check the **Calc. imm.** checkbox) or by pressing the **CALC**-button.

The menu pick **Absorbtion coeffs.** enables you to set the absorbtion coefficients for the different reflecting areas of the room.

The menu pick **Preferences** lets you save the present setup. When a new project is created the same setup will be used.

Measured SPL data

Here you import measured SPL (frequency response) data. The imported data is merged together with the simulated data so that the imported data occupy the frequency range below the **Transition frequency** in the diagrams.

- **Level offset**
Offset in dB to add to the imported data in order to make a seamless transition from simulated to measured data.
- **Transition frequency**
Frequency at which a switch from measured data to simulated data should be made.
- **Automatic level offset**
With this checkbox checked the offset will be calculated automatically
- **Disable**
With this checkbox checked the use of imported data will be disabled
- **Browse**
A file open dialog will pop up where you can select an SPL data file for loading (it is also possible to drop the file inside the **Measured SPL data** box).



When measured SPL data is disabled or not imported the controls in this group has no effect.

The room reflections are weighted so that reflections up to 20 ms after the direct wave are weighted equally while reflections between 20 ms and 40 ms are gradually de-weighted. Reflections later than 40 ms after the direct wave are not considered at all. The reason to this is twofold.

1. The human ear is more sensitive to early reflections than late reflections.
2. The time window in this implementation is limited to 40 ms.

Fig. 4.6 shows a comparison between a gated measurement and a room simulation. It is obvious that the correlation between the graphs is quite good.



It may be tempting to simulate the response in a car. **Be warned !**, the vibrating walls of a car are not modeled in this program.

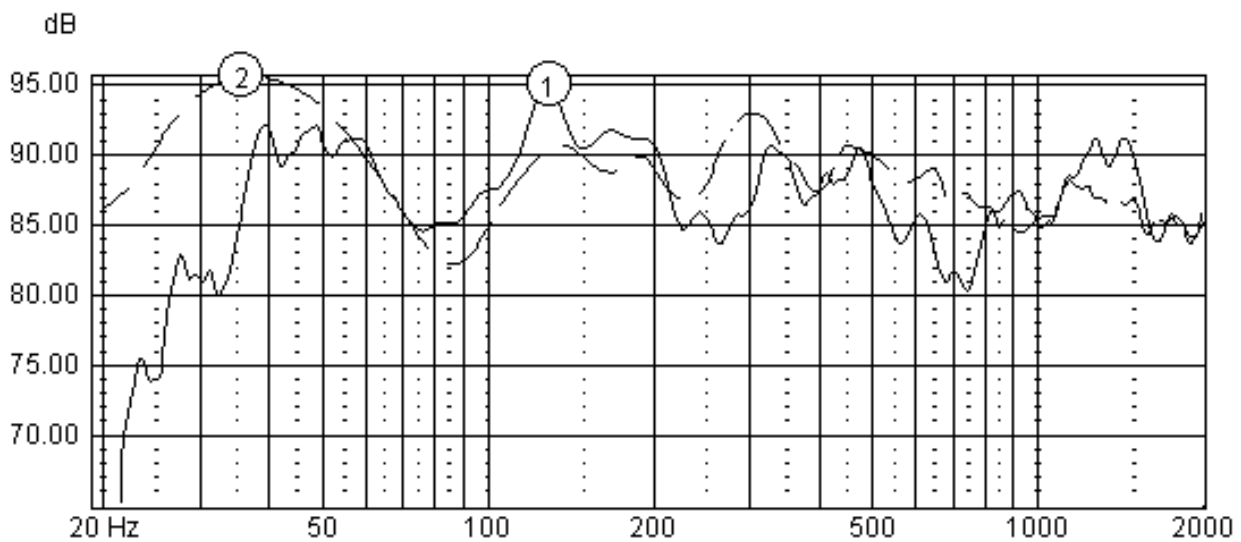


Fig. 4.6 Comparison between measured and modeled frequency response of a loudspeaker in a listening room.

4.2.6 Snapshot

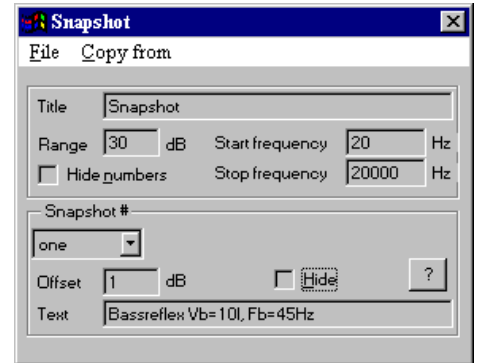
The snapshot feature allows you to save up to five frequency responses on one diagram. It can be used to compare the performance of different constructions and also the frequency response for different power levels. Fig. 4.7 shows an example of how the snapshot diagram may be used.

The snapshot diagrams are saved separately for the simple reason that it should be possible to load them into other projects.

The snapshot dialog controls the appearance of the diagram.

The menu picks of the dialog are:

- **File / Save...** , **File / Save as ...**
Save the snapshots in a file.
- **File / Open...**
Load snapshots.
- **Copy from / Free air SPL at 1m distance**
Copy from the diagram *Free air SPL at 1 m distance* to current plot.
- **Copy from / SPL in room & cabinet**
Copy from the diagram *SPL in room & cabinet* to current plot.



The controls are:

- **Title**
Set the title on the snapshot diagram, default is "Snapshot".
- **Range**
Set the desired dynamic range of the snapshot diagram i.e the difference between the highest and the lowest reading
- **Start frequency**
Start frequency for the diagram.
- **Stop frequency**
Stop frequency for the diagram.
- **Hide numbers**
Hide numbering of the plots.
- **Snapshot #**
Select current plot.
- **Offset**
Set offset for current plot.
- **Text**
Set text for current plot.
- **Hide**
Hide current plot.

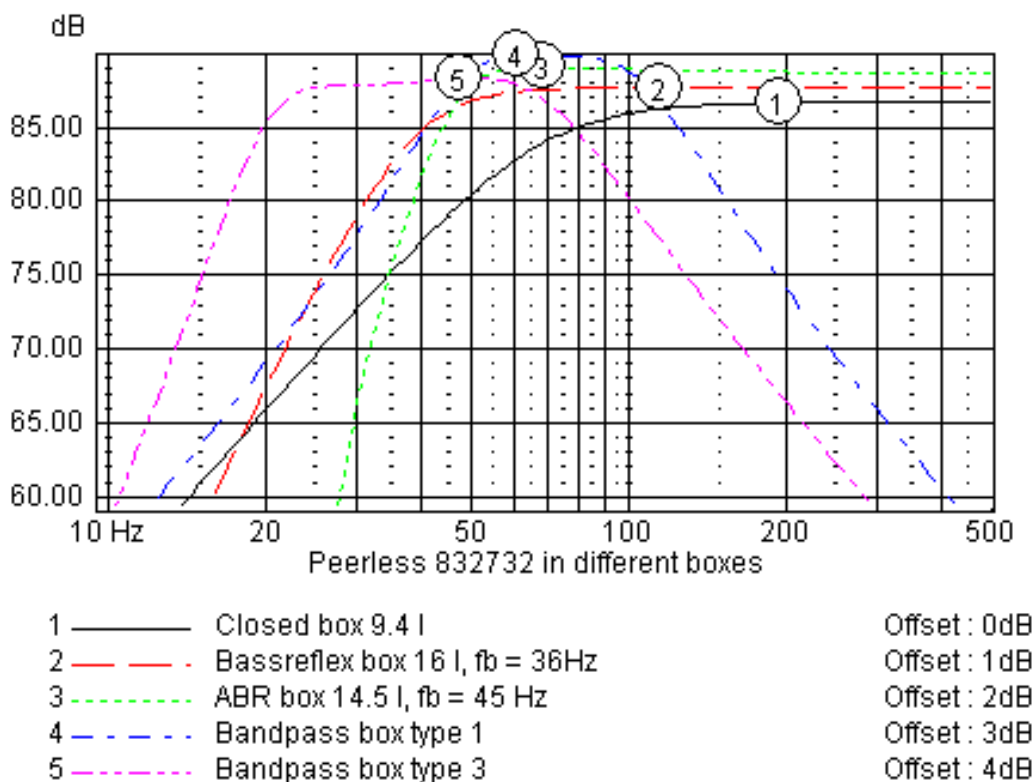
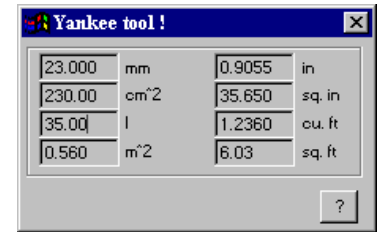


Fig. 4.7 An example on how the snapshot feature is used

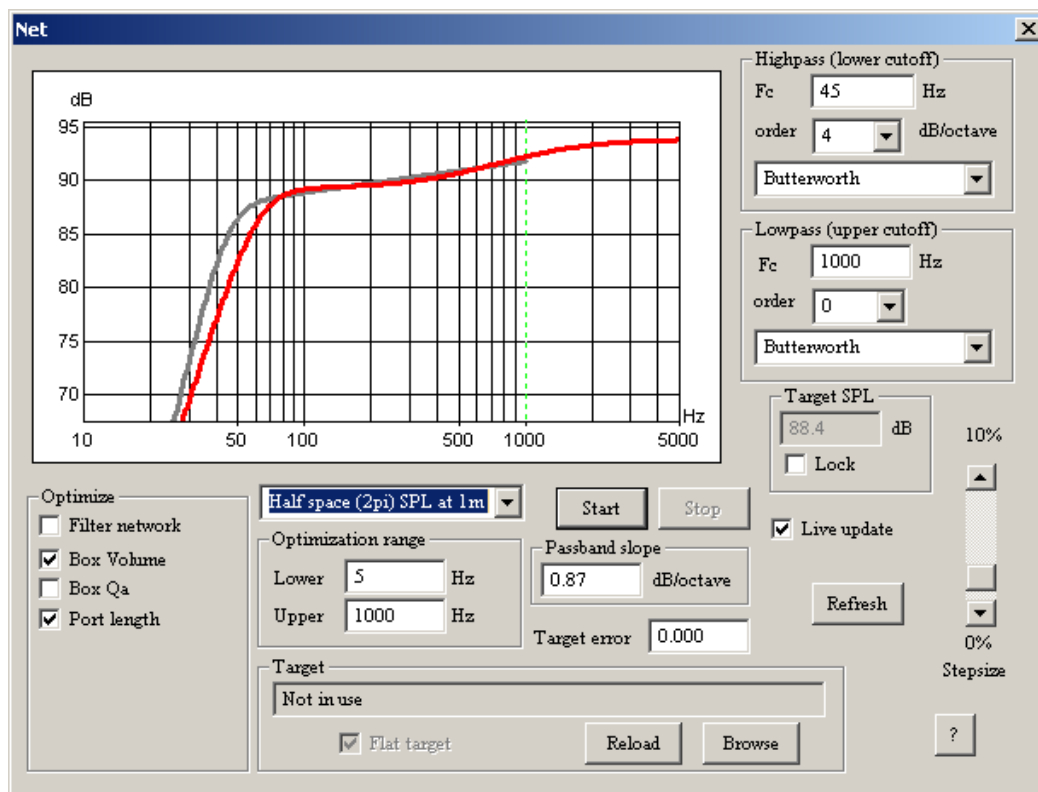
4.2.7 Yankee tool

The Yankee tool is a neat feature, which is NOT intended as a jab at Americans, the yankee tool is a simple conversion tool between e.g mm and inch.



4.2.8 System optimization

The system optimization tool makes it possible to, simultaneously, optimize filters and box parameters such as volume and tuning that match a predetermined target curve. It is possible to optimize the response either in Half space (2pi) or with room reflections included in the simulations



With this tool it is possible to model and construct complex box such as bandpass type 2 boxes with active or passive filters connected

Before the optimization of the filter components can start one must first add the components that need to be optimized to the optimization list (see chap 6.2).

In the diagram two plots are visible:

Solid red (thick): The frequency response of the network+eq+driver unit.

Grey (thick): The target frequency response.

A few controls exist in this dialog.

- **Stepsize**

The stepsize i.e how large steps the optimizer should take for each component, the range is between 0 and 10%.

- **Live update**

Allows for an update of all diagrams during optimization, slows down the process substantially though.

Lowpass

Here one set the properties for the part of the target curve that describes the low pass alignment.

- **Fc**

The cutoff frequency

- **Order**

The filter order (0-8)

- **Butterworth/Linkwitz/Flat delay/Variable slope**

Here one select Butterworth (-3dB @ Fc) or Linkwitz-Riley (-6dB @ Fc) or Flat delay alignment. Which of those one should select depends on how the resulting networks overlap in phase at the crossover point. If the resulting phase difference between two overlapping networks is close to 0 or 180 degrees then Linkwitz-Riley alignment should be selected. If, on the other hand, the phase difference is 90 or 270 degrees the Butterworth alignment is probably a better choice. The flat delay alignment may be selected if one wishes to strive for as small groupdelay distortion as possible. This alignment resembles the Bessel filter alignment.

The variable slope alternative is a special case which allows you to set the cutoff frequency and slope in dB/octave, very useful when optimizing e.g bandpass boxes.

Highpass

Here one set the properties for the part of the target curve that describes the high pass alignment, see above for explanation.

Optimize

Here one select what should be optimized

Optimization range

Here one select over what frequency range the optimizer should compute the difference metric, this is useful as valuable components are otherwise wasted on unimportant parts of the frequency range. A recommendation here is thus to set the optimization range so that the working range is included and also points in the frequency range that are 20-25dB below the peak sound pressure level or higher.

Target SPL

If the lock checkbox is checked the target SPL curve is locked to a specified level so that problems with ever decreasing sensitivity is efficiently avoided. With the checkbox unchecked the target SPL curve will float around the achieved curve.

A target response can be imported with a click on the **Browse** button (also possible to drop inside the **Target** box). The format of the target is a simple ASCII text file with frequency point and response. A simple file with the below contents:

```
20 0
50 0
100 -3
500 -3
```

Will, when imported, force the optimization to achieve a frequency response with a 3dB drop above 50Hz. A click on the **Reload** button will reload the target file. When **Flat target** is checked the target will be forced to become flat. The target data file may contain phase information also but the phase information is not used.

Passband slope

Here one can enter a desired passband slope in the range $-/+2$ dB per octave. Introducing a slope in the frequency response may be desirable if one for instance wish to enhance the level in the lower bass region in a two way system.

Target error (Error)

Here one can set the target mean deviation from the target frequency response, when the start button is pressed this field changes to show the achieved mean deviation from target response. When the target is reached the iteration will stop. In the target is set to zero the optimization will continue forever. The error displayed is the average distance in dB from the target within the optimization range..

4.2.9 Filter

See chapter 6.2-6.5 for a detailed description. The filter dialog is essentially the same as that described in ch. 6.2 with the exception that a few components are removed as they are of no practical use in this context.

4.2.10 EQ

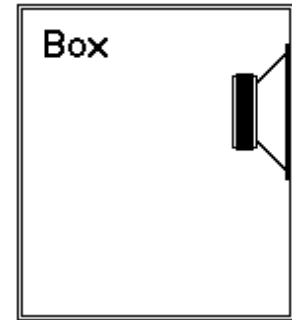
See chapter 7.5 for a detailed description.

4.3 Closed box

4.3.1 Introduction

This is the most simple box type available. One of the shortcomings of this construction is that the output power at low frequencies is very limited unless you use very large or long stroke loudspeaker units.

This box type has no advanced demands on driver unit Qts. However a very low Qts may give problems with a high cutoff frequency. Damping material can be used extensively in this box but should not be placed too close to the driver unit as this may increase distortion.



4.3.2 The box dialog

Here you may enter data about the box, the controls are:

- **Box volume**

Box volume in liters.

- **Box Q**

This box Q value consist of two parts

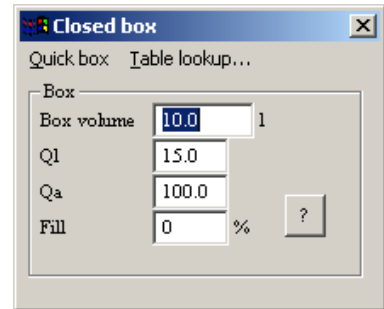
Q_L that is the Q-value due to unwanted leakage from the box.

Q_A that is the Q-value due to damping materials in the box.

The range of Q_L and Q_A is 3-100. In normal cases Q_L is in the range 5-20. A box with moderate lining has a Q_A of about 30. A heavily stuffed box can have a Q_A of 3.

- **Fill**

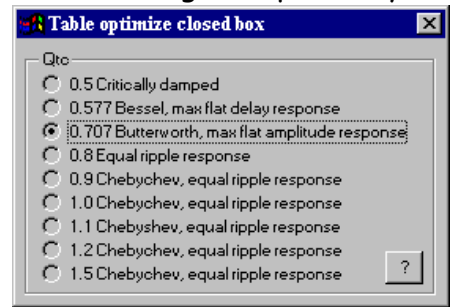
Depicts how much of the box is filled with damping material. If the box is completely stuffed with damping material, the Q_A will decrease, but also will the virtual box volume increase with about 20% i.e. the driver unit will experience a larger box than V_b . Note that you will have to set this parameter to e.g. 100% if you set Q_A to 3. For a Q_A of 7 the fill factor should be set to about 50% while a Q_A of 15-30 will give a fill factor of 0%.



In this dialog you can also select from the menu picks:

- **Table lookup..**

A dialog pops up where you can select between different tuning setups for your selected driver unit. The procedure is simple, just click on the desired Qtc value and the program will calculate the correct box volume for you. A Qtc of 1.5 will give a bump in the frequency response but also a lower volume than a Qtc of 0.5 would give. If the driver unit has a high Qts it is not always possible to select all tuning setups, the impossible setups are then grayed.



- **Quick box**

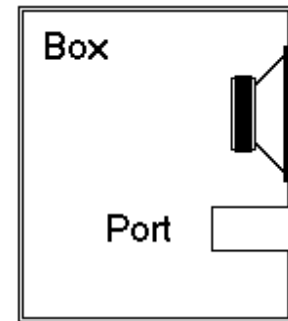
The program calculates a suitable starting valued for the box volume, actually unnecessary for this simple box...

4.4 Bassreflex box

4.4.1 Introduction

This type of box is probably the most common. The possibility to get higher output levels is greater than for a closed box.

The driver units should not have too high Qts. Damping material should be used with restriction and with the objective to reduce standing wave modes within the box. Much damping material may of course be of some help if driver Qts is high.

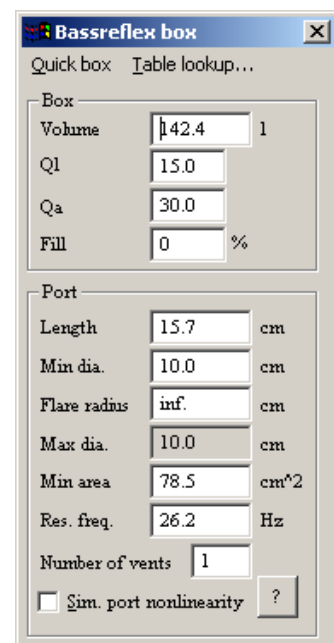


4.4.2 The box dialog

The box volume and damping is described in chap 4.3.

The port is described by (see also fig 12 and chap 9.1):

- **Length**
Vent length in cm. All vents have the same length.
- **Flare radius**
The flare radius of the port. If flared ports are not used, enter a zero value, this will imply an infinite flare radius.
- **Min and Max dia**
Vent min and max diameter in cm. All vents have the same diameter. Observe that a small diameter may give trouble with noticeable chuffing noise at high power levels, a rule of thumb given by R. Small is that the airspeed in the port should be less than 15-20m/s at high power levels. That you can check in the diagram *Port air speed*.
- **Area**
Vent area in cm². Note that it is the area of the vent that counts, the cross sectional shape may be circular or rectangular.
- **Res. freq.**
Resonance frequency in Hz.
- **Number of vents**
Number of vents (1-9). Note that all vents in a port must have the same length and area.
- **Sim. port nonlinearity**
Check this checkbox if you want the program to simulate the non-linearity caused by high air speeds in the port.

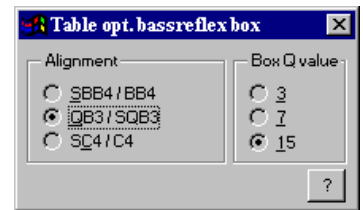


Note that a port can consist of up to 9 vents.

In this dialog you can also select from the menu picks:

- **Table lookup...**

An additional dialog pops up where you can select between different alignments and different stuffing of the box. If your driver unit has a high Q_{ts} , it is not sure that you can select all alignments. The alignment that is considered to give the best transient response is the SBB4/BB4 alignment.



- **Quick box**

The program calculates suitable starting values for box and port.

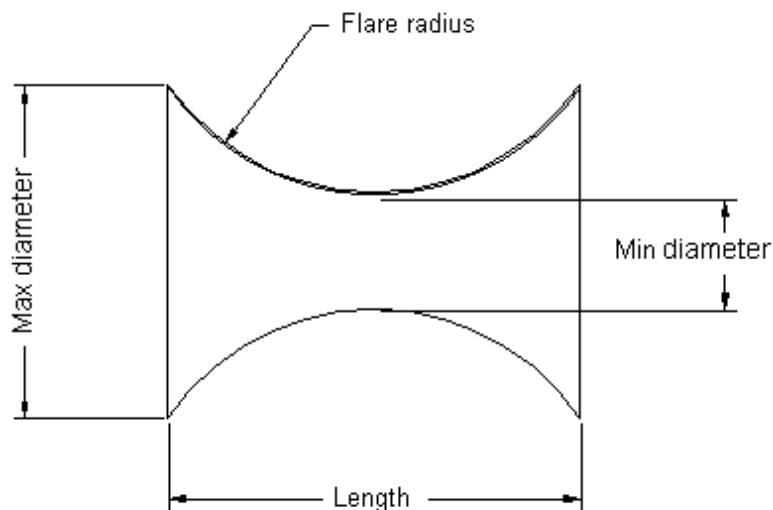


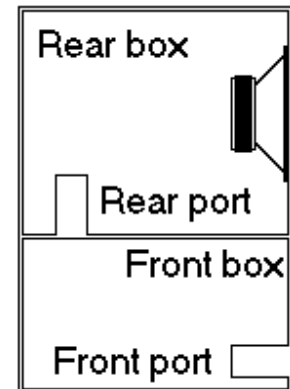
Fig. 4.8 Cross sectional view of a loudspeaker port

4.5 Double tuned bassreflex box type 1

4.5.1 Introduction

This type of box has been described in a book (title unknown). In the description the rear box should have twice the volume of the front box.

It is hard to say whether this box is useful or not. It's extremely difficult to achieve a good frequency response. The idea is probably to widen the port resonance and thus limit the cone excursion over a wider range. In reality this seems to be very difficult, one may however use the front port as a muffler that will block the standing wave modes caused by the rear port. In this case one can tune the rear box as an ordinary bassreflex box and give the front box a relatively small volume and tune it to about 100-200Hz.



4.5.2 The box dialog

The dialog is divided into four major groups.

- Rear box, setup for the rear box. See chap 4.3 for a description of the edit fields.
- Front box, setup for the front box.
- Rear port, setup for the rear port. See chap 4.4 for a description of the edit fields.
- Front port, setup for the front port.

Note that you cannot directly set the resonance frequencies for this type of box. The reason is that the ports are mutually dependent, however you can indirectly set the frequencies by changing e.g. port lengths.

Rear box		Front box	
Volume	59.8	Volume	72.0
Q1	15.0	Q1	15.0
Qa	21.1	Qa	3.8
Fill	0 %	Fill	0 %
Rear port		Front port	
Length	39.6 cm	Length	32.3 cm
Diameter	11.4 cm	Diameter	9.9 cm
Area	101.9 cm ²	Area	76.8 cm ²
Res. freq.	18.9 Hz	Res. freq.	48.8 Hz
Number of vents	1	Number of vents	1

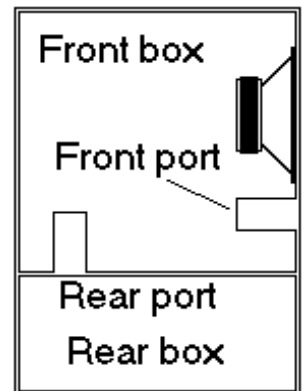
Sim. port nonlinearity

4.6 Double tuned bassreflex box type 2

4.5.1 Introduction

This box type is a variation of the double tuned bassreflex box type 1. It has the advantage that one can achieve a frequency response that helps to reduce the rise in sound pressure toward lower frequencies caused by ordinary listening rooms (see example) The rear box should be filled completely with damping material while the front box should be moderately filled.

When making a construction using this box type one can start from an ordinary bassreflex box with a volume V_b and tuned to a resonance frequency F_b . One third of V_b is assigned to the rear box while two thirds of V_b is assigned to the front box. The front box is tuned to F_b and the rear box is tuned to about 100-120Hz. After these steps one only has to do a little trial and error. One can also use the *Quick box* pick to get a starting point fast.



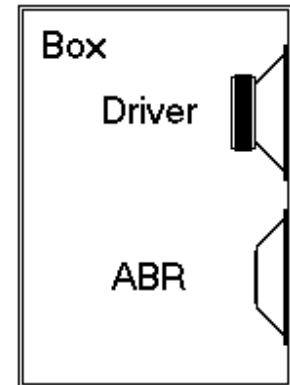
4.7 Passive radiator box

4.7.1 Introduction

The principle behind this box is almost the same as for a bassreflex box. The difference lies in the fact that the vent has been replaced by a loudspeaker without motor and magnet also known as an auxiliary bass radiator (ABR). Alternative names are drone cone or passive radiator.

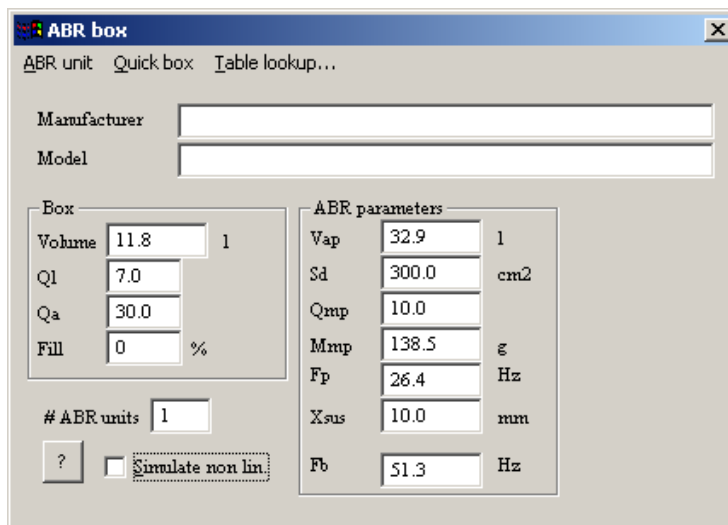
The effective area of the ABR should not be less than that of the driver unit as the former has to move a lot of air at the resonance frequency.

An interesting point is that the frequency response shows a notch below the cutoff frequency. Damping material should be used with care and only to limit standing wave modes in the box.



4.7.2 The box dialog

The dialog is divided into two major groups.



The box volume and damping is described in chap 4.3.

It is possible to save and open ABR data files by means of the **ABR unit | Open / Save / Save as** menu picks the file format is the same as that used for the driver unit data.

The ABR unit is described by:

- **Vap**
Equivalent volume of the ABR [l].
- **Sd**
Cone area [cm²].
- **Mmp**
Moving mass excluding surrounding air [g].
- **Qmp**

Mechanical Q of the ABR

- **Fp**
Resonance frequency in free air [Hz].
- **Fb**
Resonance frequency when mounted in box [Hz].
- **Xsus**
The maximum linear excursion allowed by the suspension, [mmp-p].
NOTE peak to peak value
- **# ABR units**
Number of ABR's in the box [1-9].
- **Simulate non linearity**
Check this checkbox if you want the program to simulate the non-linearity caused by excessive excursion in the ABR.

The menu picks are:

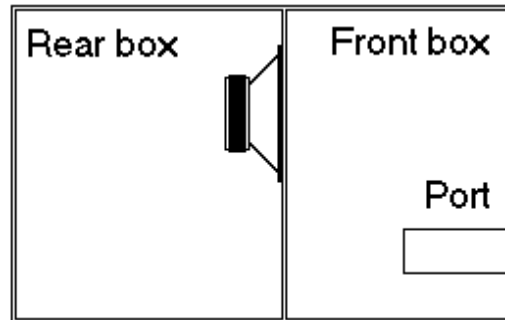
- ***ABR unit / Open***
Open an ABR unit data file
- ***ABR unit / Save /Save as...***
Save to an ABR unit data file
- ***Table lookup***
The program computes optimal values for the box and the ABR.
- ***Quick box***
The program calculates suitable starting values for box and port.

4.8 Bandpass box type 1

4.8.1 Introduction

This type of bandpass box is the simplest. The efficiency depends on the bandwidth in the aspect that the higher the bandwidth the lower the efficiency. The normal bandwidth is 1 to 2 octaves. A benefit of this box type is that it is quite simple to achieve high output at low frequencies, even though the total volume is small. A drawback is that the port is prone to generating standing wave mode resonances, the first at one half wavelength, so the bandpass box is best used in combination with a lowpass filter. The port should have a rather big area as all energy emits through it. The Q_t s of the driver units should not be too high as the required box volume increases with Q_t s. Suitable Q_t s values range from 0.3 to 0.8. Damping material can be used in the rear box but should be used with care in the front box.

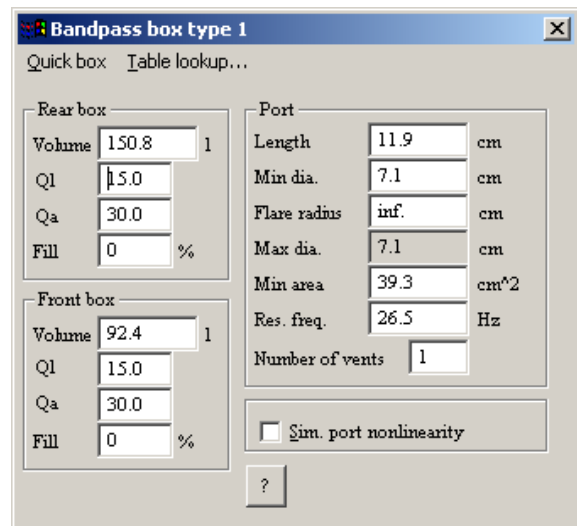
When it comes to iterative frequency fit optimization, it can be said that it is quite hard to fail. Just set the desired upper and lower cutoff frequencies and start optimizing.



4.8.2 The box dialog

The dialog is divided into three major groups.

- Rear box, setup for the rear box. See chap 4.3 for a description of the edit fields.
- Front box, setup for the front box.
- Port, setup for the port. See chap 4.4 for a description of the edit fields.



The menu picks are:

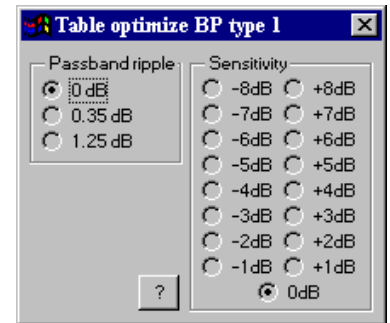
- **Table lookup**

An additional dialog shows up where you can select between different alignments for your system, *Passband ripple* is the irregularity you can allow in the passband, the more ripple you allow the greater bandwidth you get. *Sensitivity* is the sensitivity in the passband relative to that of the same driver in a closed box. The higher sensitivity you demand the narrower the bandwidth you get.

If the driver unit has a high Q ts it may be impossible to select all alignments.

- **Quick box**

Calculate suitable starting values for box and port.

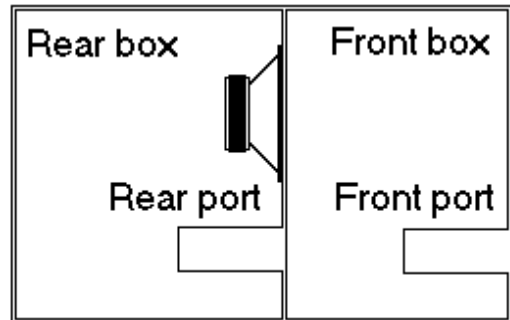


4.9 Bandpass box type 2

4.9.1 Introduction

This boxtype looks like the bandpass box type 1, however an extra port is inserted between the boxes. This port serves to reduce the cone excursion at low frequencies and also to extend the frequency response towards lower frequencies, bandwidths of 2-3 octaves are possible.

The resonance frequency for the rear port should always be lower than the resonance frequency for the front port. Damping material should be used with care and with the objective of preventing standing wave modes in the boxes. A suitable Q ts for the driver units lies between 0.2 and 0.45.



4.9.2 The box dialog

The dialog is divided into four major groups.

- Rear box, setup for the rear box. See chap 4.3 for a description of the edit fields.
- Front box, setup for the front box.
- Rear port, setup for the rear port. See chap 4.4 for a description of the edit fields.
- Front port, setup for the front port.

Note also that you cannot directly set the resonance frequencies for this type of box. The reason is that the ports are mutually dependent, however you can indirectly set the frequencies by changing e.g. port lengths.

The menu picks are

- **Quick box**
Calculate suitable starting values for box and port.

Rear box		Front box	
Volume	144.0	1	72.0
Q1	15.0	1	15.0
Qa	100.0		100.0
Fill	0	%	0
Rear port		Front port	
Length	20.0	cm	10.0
Min dia.	7.3	cm	10.4
Flare radius	inf.	cm	inf.
Max dia.	7.3	cm	10.4
Min area	42.3	cm ²	84.7
Res. freq.	15.8	Hz	53.4
Number of vents	1		1

Sim. port nonlinearity ?

4.10 Bandpass box type 3

4.10.1 Introduction

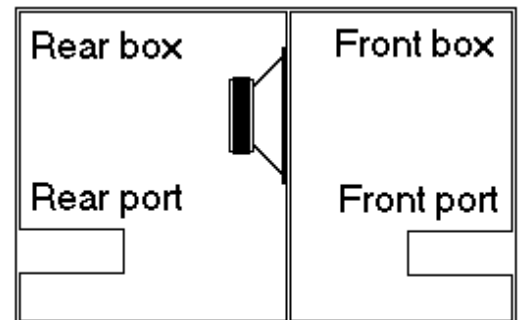
This boxtype looks like the bandpass box type 1, however an extra port emanates from the rear box. This port helps to reduce the cone excursion at low frequencies and to extend the frequency response toward lower frequencies, bandwidths of 2-3 octaves are possible.

An interesting thing with this boxtype is that if both the volume and tuning of the two boxes are identical. The total output will in theory be zero !.

The front port should preferably be tuned to the higher resonance frequency, the acoustic phase will then pass zero in the passband and the signal is reproduced in correct phase.

Damping material should be used with care and only with the objective to reduce standing wave modes in the boxes.

The driver unit should preferably have a Qts in the range 0.2-0.45.



4.10.2 The box dialog

The dialog is divided into four major groups.

- Rear box, setup for the rear box. See chap 4.3 for a description of the edit fields.
- Front box, setup for the front box.
- Rear port, setup for the rear port. See chap 4.4 for a description of the edit fields.
- Front port, setup for the front port.

The menu picks are

- **Table lookup**
The program computes optimal values for the boxes and ports.
- **Quick box**
The program calculates suitable starting values for boxes and ports.

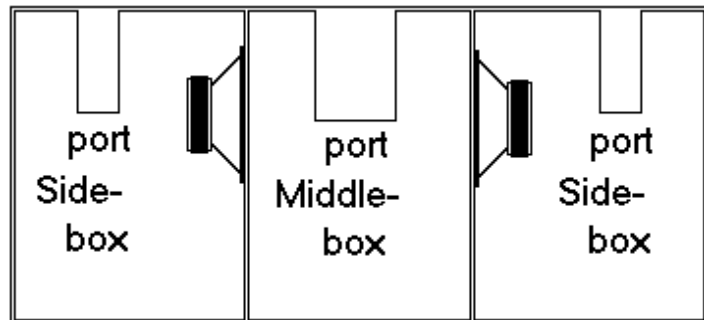
The screenshot shows the 'Bandpass box type 3' dialog box. It has a title bar with a close button. Below the title bar are two menu options: 'Quick box' and 'Table lookup...'. The dialog is divided into four main sections: 'Rear box', 'Front box', 'Rear port', and 'Front port'. Each section contains several input fields with numerical values and units. At the bottom, there is a checkbox for 'Sim. port nonlinearity' and a help button with a question mark.

Parameter	Rear box	Front box	Rear port	Front port
Volume	291.3	63.8		
Q1	15.0	15.0		
Qa	30.0	30.0		
Fill	0 %	0 %		
Length	19.6 cm	16.3 cm	19.6 cm	16.3 cm
Min dia.	8.4 cm	8.4 cm	8.4 cm	8.4 cm
Flare radius	inf. cm	inf. cm	inf. cm	inf. cm
Max dia.	8.4 cm	8.4 cm	8.4 cm	8.4 cm
Min area	56.0 cm ²	56.0 cm ²	56.0 cm ²	56.0 cm ²
Res. freq.	14.5 Hz	33.2 Hz	14.5 Hz	33.2 Hz
Number of vents	1	1	1	1

4.11 Bandpass box type 4

4.11.1 Introduction

This boxtype is actually a variation of bandpass box type 3 with the exception that it is built up with three chambers and has a minimum of 2 driver units, the performance is however the same. Note that you may only use 2, 4, 6 or 8 driver units with this box.



This boxtype is often used in car stereo system, mostly because it generally uses 2 driver units.

The front port should preferably be tuned to the higher resonance frequency, the acoustic phase will then pass zero in the passband and the signal is reproduced in correct phase.

4.11.2 The box dialog

The dialog is divided into four major groups.

- Side boxes, setup for the side boxes. See chap 4.3 for a description of the edit fields.
- Middle box, setup for the middle box.
- Side ports, setup for the side ports. See chap 4.4 for a description of the edit fields.
- Middle port, setup for the middle port.

The menu picks are

- **Table lookup**
The program computes optimal values for the boxes and ports.
- **Quick box**
The program calculates suitable starting values for box and port.

The screenshot shows the 'Bandpass box type 3' dialog box. It has a title bar with a close button. Below the title bar are two menu options: 'Quick box' and 'Table lookup...'. The dialog is divided into four main sections: 'Rear box', 'Front box', 'Rear port', and 'Front port'. Each section contains several input fields with numerical values and units. At the bottom, there is a checkbox for 'Sim. port nonlinearity' and a help button with a question mark.

Parameter	Rear box	Front box
Volume	291.3	63.8
Q1	15.0	15.0
Qa	30.0	30.0
Fill	0 %	0 %

Parameter	Rear port	Front port
Length	19.6 cm	16.3 cm
Min dia.	8.4 cm	8.4 cm
Flare radius	inf. cm	inf. cm
Max dia.	8.4 cm	8.4 cm
Min area	56.0 cm ²	56.0 cm ²
Res. freq.	14.5 Hz	33.2 Hz
Number of vents	1	1

At the bottom, there is a checkbox for 'Sim. port nonlinearity' and a help button with a question mark.

4.12 What do the diagrams show?

4.12.1 Introduction

Speaker box has the diagrams in separate child windows or MDI windows as they are also called. A diagram consists of one or more plots. Diagrams that are iconized may be opened by means of a double-click on the icon. The diagrams can be resized and maximized.

Note that you can select **Window | User note** to insert a line of text at the bottom of the diagrams.

Too many diagrams on the screen?. No problem, click on the close button on each diagram that you don't need to see for the moment. The diagrams are actually hidden. A click on **Window | Show** makes it possible to get the hidden diagrams back again.

4.12.2 Half space (2π) SPL at 1m distance

This diagram shows the sound pressure level at 1m distance as a function of frequency and input power. The speaker is assumed to radiate into 2π or half space field space i.e it is mounted in an infinitely large baffle

The dynamic range i.e. the difference between the lowest and the highest reading on this diagram can be set in the *Measurement setup* dialog (see chap 4.1.3).

This diagram also shows the phase response. This phase may show abrupt +/- 180 degree changes, this is however only caused by the atan2 function used when computing the phase.

4.12.3 Cone excursion

This diagram shows the cone excursion (mm_{p-p}) for each driver unit as a function of frequency and input power.

Remember that the cone excursion in this program is given as a peak to peak value. One important thing is that if the cone excursion exceeds the Xmax value then distortion will increase abruptly and sound pressure will quit rising linearly with input level. This is visible if the simulation of the BL non-linearity is turned on in the *Driver unit config* dialog.

4.12.4 Port air speed

This diagram shows the port air speed in the port(s) (m/s RMS). Important here is that if the port air speed exceeds some 15m/s then problems with chuffing is very likely to occur. If many ports exist in the box, e.g. a bandpass box type 2 then 2 plots are shown which are also tagged to avoid confusion.

If simulation of nonlinearity in the port is turned the port compression is simulated. Non linearities in the port occur when the flow in the port goes from laminar to turbulent flow, this is meant to occur at about 4.5% of the speed of air, in reality this will occur at even lower port air speeds At these high airspeeds chuffing will also occur. High air speeds in the ports should be avoided if possible, not only because of the compression, but also because this chuffing noise excites the organ pipe resonances. The article by Salvatti et. al. (see section 10) describes this in more detail

4.12.5 Impedance

The impedance curve shows the loudspeaker systems total load upon the driving amplifier. Note that the program will automatically compute the impedance depending on the electrical connections of the driver units, even if there are many. This diagram also shows the phase

4.12.6 Group delay

The group delay is defined as $-(\delta\Phi/\delta f)$ i.e. the acoustic phase differentiated with respect to frequency. It may simply be expressed by how tones of specific frequencies are delayed with respect to nearby frequencies. As group delay magnitude is normally much smaller for higher frequencies than for lower frequencies one may have problems in seeing what happens at high frequencies due to the autoscaling function in LspCAD. With the **Page Up** and **Page Down** keys on the keyboard one can set the magnitude scaling of this diagram.

For the inexperienced this diagram may be difficult to understand. In this case the transient response may be easier to understand.

4.12.7 Filter response

The filter response lets you see the transfer function of the filter. The filter response of both the active and passive parts in the filter is shown individually.

4.12.8 Equalizer response

The equalizer response lets you see the transfer function of the equalizer.

4.12.9 Impulse- Step- Tone burst response

This diagram shows the loudspeaker performance in the time domain.

In the dialog that controls the appearance of this diagram one can choose between impulse, step or tone burst response.

The step response shows the response if you apply a fixed voltage to the loudspeaker terminals and measure the response with a microphone and an oscilloscope.

The impulse response shows the response if you apply a very short pulse to the loudspeaker terminals

The tone burst response shows what happens if we apply a tone burst with a specified frequency and specified number of periods at the loudspeaker terminals. This shows what happens when the loudspeaker tries to reproduce a hit on e.g. a kettle drum.

4.12.10 SPL in Room & Cabinet

See chapter 4.2.5 for further description of this. This diagram also shows the phase.

4.12.11 SPL with port standing wave modes

This diagram only exists for the bandpass boxes and is intended to make the constructor aware of the problem that may arise with bandpass boxes. See also chap 9.1, the section about the organ pipe resonances.

4.12.12 Power amp. output voltage

Displays the output voltage from the power amplifier. This is equal to the generator source level if no active filter or equalizer is in use.

4.12.13 Snapshot

See chap 4.2.6 for further description of this.

4.12.14 Information

The information window is not a diagram. It is a document that shows all setups in the program and also some extra information such as -3 dB frequencies. This document may be printed out and copied as the can diagrams. If the diagram is copied the contents of the clipboard will be in ASCII format, the reason is that it should be simple to import the data to your own documents.

4.13 Nonlinearity simulationson

The so called $j\omega$ method is used in this software, this assumes that we are dealing with linear relationships, this is the fact at low to moderate power levels. There are some possibilities to model non-linear relationships in LspCAD, they are described below. The models are not 100% proof but anyway they give a quite good idea of what happens at high power levels.

- **BL non linearity:**
Simulates the limitation in cone excursion that occurs if the cone excursion exceeds **Xmax** and the voice coil leaves the air gap partly or completely. This function cannot be used if **Xmax** is not defined.
- **Voice coil heating:**
Simulates the heating of the voice coil at high power levels, which gives rise to an increased **Re**, which in turn has tremendous impact on the resulting frequency response. This is important to consider when constructing PA systems as they generally operate at high power levels.
The model assumes that the model has a temperature of +250° C at **Pmax** and that the voice coil is made of copper. The function cannot be used if **Pmax** is not defined.
- **Cms non linearity:**
Most high quality loudspeaker units, especially those suited for bass reflex boxes are equipped with progressive stiffness in the suspension. This means that the suspension makes more and more resistance the more it is displaced from its origin.
This feature is at best only commented in the product sheet, and it is virtually never mentioned what the stiffness curve looks like.
The model in this program assumes that the stiffness (1/Cms) starts to increase at **Xmax/2** and is doubled at **Xmax**.
This function cannot be used if **Xmax** is not defined.
- **Port air speed non linearity (port compression):**
Simulates the non linearity that occur in a port at higher power levels, which in turn leads to turbulent airflow which will obstruct the flow in the port. It is important to note here that the effect on the impedance should be regarded as an average effect over a sinusoidal cycle. In reality the impedance curve (and everything else that depends on it) is modulated by the input signal, this fact also applies for the Cms and the BL non linearity simulation.
- **ABR non linearity:**
Simulates the non linearity in the suspension caused by excessive excursion of the ABR unit.

If $Bl(x)$ and $Cms(x)$ data is provided in the .unt files the nonlinearity simulations will follow reality to a better degree than with the simple assumptions given above. Still it is important to know that making nonlinearity simulations with a linear model like that in LspCAD won't give you the exact truth, for that purpose tools that solve nonlinear differential equations are needed.

5 The simple passive crossover utility

5.1 Introduction

The simple passive crossover utility is a simple filter calculator with the ability to import measured SPL and impedance data, thus one can simulate and model "real" 2-way and 3-way passive crossover networks.

For the moderately skilled designer it is recommended to just skim through this section and start with advanced filters (see ch. 6) instead.

The crossover networks consist of 2 or 3 networks for the bass, mid and treble region. Each network can model up to fourth order highpass and lowpass filters, furthermore the bass and mid networks support baffle diffraction compensation (a resistor in parallel with an inductor). All networks have support for attenuators and zobel networks.

The number of analysis points is 500 in the frequency range 1-30000 Hz.

The project files created by this utility has .flt as standard extension but one may use other extensions as well.

There exists a schematic window that shows the looks of the passive crossover. The complete schematic for the mid network with all components in use is shown in fig. 5.1 below.

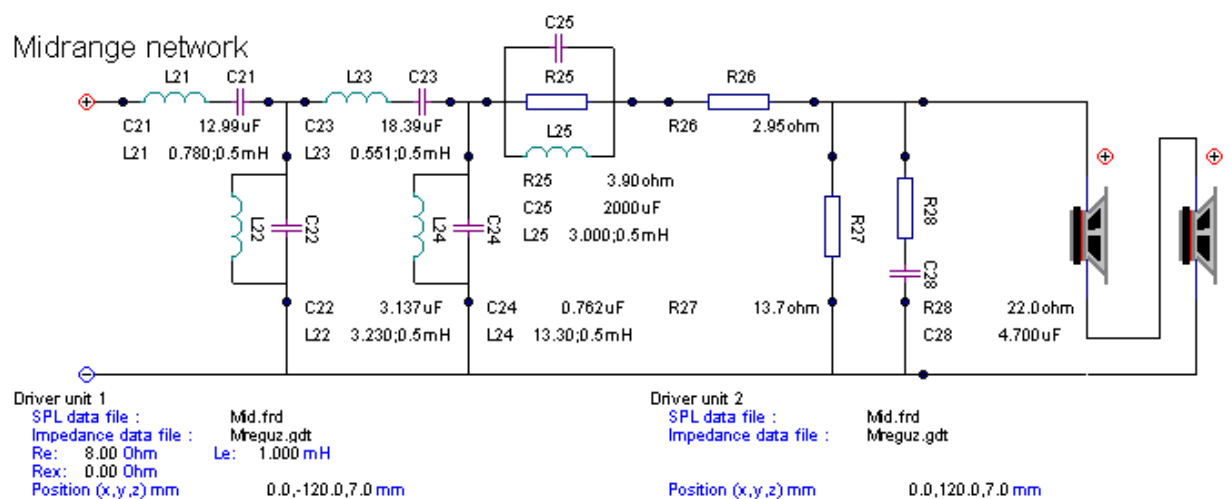


Fig. 5.1 Complete schematic of the midrange network with all components defined

The component values in the schematic are hidden if they are set to zero (undef.) in the network dialogs. Note that the first digit in the component names is **(1)** for the bass network, **(2)** for the mid network and **(3)** for the treble network.

The user can view 9 diagrams, they are:

- Filter gain
- Snapshot
- Input impedance
- Summed frequency response
- Individual phase response
- Group delay
- Impulse response
- Vertical off axis response, surface plot
- Vertical off axis response, overlay plot



As long as you do not import measured data the 7 last diagrams are quite uninteresting, as they all rely on imported data.

About off axis simulations:

LspCAD models the off axis simulations behavior with the radiation pattern of a stiff piston mounted at the end of a long tube.

It must be kept in mind that this is after all a model and while models are of great help to approximate reality they won't give you the exact truth. What this means is that one gets a fair view of how a driver behaves off axis, but it is not the truth. However the model gets you closer to the real thing than no off axis model at all.

Not only the magnitude is affected when a driver is modeled off axis, the phase is modified in LspCAD with a simple equation that adds a phase lag when a driver is modeled off axis. Fig. 5.3 displays both the magnitude and phase of a 100mm radius driver with flat on axis response modeled 30 degrees off axis.

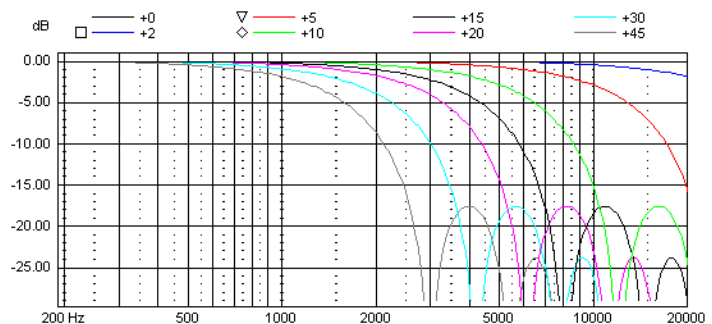


Fig. 5.2 Simulated off axis behavior for a driver with 100mm piston radius and a flat on axis response.

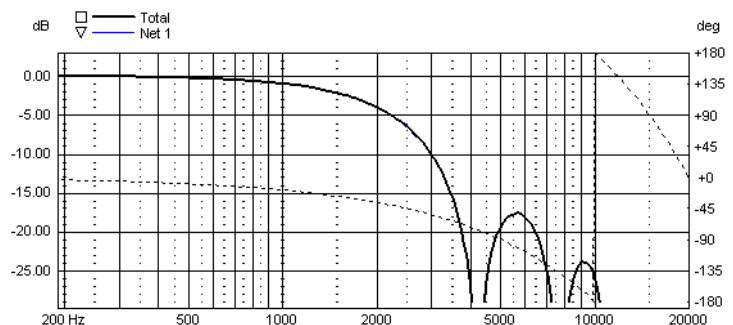


Fig. 5.3 Magnitude and phase at 30 degrees off axis for an driver with 100mm piston radius and flat on axis response (simulation).

5.2 Menu picks

Below are described the menu picks for the filter utility is described.

- **File / New...**
Create a new project. Already explained in chap 3.
- **File / New instance**
Start a new instance of LspCAD.
- **File / Open project...**
Load a project.
- **File / Save project , File / Save project as...**
Save a project.
- **File / Print / Window...**
Print active window.
- **File / Print / Diagrams..**
A dialog will pop up where you can select which diagrams should be printed. *Schematic* can not be selected here. The procedure is simple, just select from the pick list which diagrams should be printed and how many diagrams per page you want and click on the print button. The diagrams are printed in the order they are selected.
- **File / Copy window**
Copy the active diagram or drawing in bitmap format.
- **File / Export / ...**
Export diagrams in LAUD (.frd, .zma), LEAP (.gdt), CLIO (.txt) or MLLSA (.fmp, .fm) format (see ch 8.2). The diagrams that can be exported are:
 - *SPL* (total in different angles or on axis for each individual network)
 - *Filter gain* for each individual network
 - *Input Impedance* (total or for each individual network)
 - *Group delay* (total or for each individual network)
 - *Impulse response*
- **File / Exit**
Terminate the program.
- **Network / Bass**
Edit the parameters for the bass network
- **Network / Mid**
Edit the parameters for the mid network (not available for 2-way systems)
- **Network / Treble**
Edit the parameters for the treble network
- **Driver / Bass**
Edit the parameters for the bass driver unit
- **Driver / Mid**
Edit the parameters for the mid driver unit (not available for 2-way systems)
- **Driver / Treble**
Edit the parameters for the treble driver unit

- **General**
Edit the general settings such as sample rate and length for the impulse response and select the angles for the off axis plots.
- **Filter music**
Strange name for this function, could not find out any better. Allows the user to filter music samples through the loudspeakers frequency response and listen to the result with a pair of high quality headphones.
- **Window / Arrange icons**
Arrange the icons in the client area.
- **Window / Cascade**
Cascade the windows in the client area.
- **Window / Tile**
Set the open windows side by side.
- **Window / Default size**
Set the size of the selected window to a default size.
- **Window / User note...**
Here you have the option to enter comments (max. 50 characters) in the diagrams.
- **Window / Show...**
A curse that comes with applications that use many diagrams is that in the end there are so many windows to keep track of that it is difficult to get an overview over the design. Therefore it is possible to close the diagrams (except snapshot) and bring them back with this menu pick.
- **justMLS**
A click on this menu pick will start justMLS, this assumes that justMLS.exe is located in the LspCAD folder.
- **About...**
Show some valuable information about this software.

5.3 The network dialogs

The network dialogs serve as placeholders for filter parameters, the network dialog for the mid network is described in this chapter, the dialogs for the bass and treble networks are similar

The dialog is divided into a number of groups:

Attenuator

Attenuator circuit

- **Att**
Sets the desired attenuation in dB.
- **Calc**
A click on this button and R26 and R27 will be computed based on Re given in the *driver dialog* and Att.
- **R26 and R27**
Two resistors that will form the attenuator

LP filter section

Lowpass filter section

- **Filter order**
Select the filter order for the lowpass filter section.
- **Cutoff**
Cutoff frequency for lowpass filter
- **Calc**
A click on this button and L21, C22, L23 and C24 will be computed based on the filter order, cutoff frequency and Re. Note that if the filter order is lower than N=4 some of the components will be undefined. The component values are that of common Butterworth filters.
- **L21, C22, L23, C24**
The components that constitute the lowpass filter section

Notch section

- **R25 and L25 and C25**

Notch filter section, when combined they will form a parallel resonance circuit, that can be used to notch out peaks in the frequency response. The resonance frequency is given by

$$f_c = \frac{1}{2\pi\sqrt{LC}}$$

HP filter section

Highpass filter section

- **Filter order**

Select the filter order for the highpass filter section.

- **Cutoff**

Cutoff frequency for highpass filter

- **Calc**

A click on this button and C21, L22, C23 and L24 will be computed based on the filter order, cutoff frequency and Re.

- **C21, L22, C23, L24**

The components that constitute the highpass filter section

Disconnect

Disconnect this network, actually a very large resistor ($10^{12} \Omega$) is inserted in series with the input terminals.

Zobel network

Impedance equalization

- **Calc**

A click on this button and R28 and C28 will be computed based on Re and Le given in the *driver dialog*.

- **R28 and C28**

The components that constitute the zobel network

Important to see is that the inductance values consist of 2 values separated by semicolon, the left value is the inductance in mH while the right value is the associated internal DC-resistance of the inductance in Ω . For instance a reading of **0.780;0.5** in the **L21** field means that the inductance is 0.780mH and that the internal DC-resistance is 0.5 Ω .

5.4 The driver dialogs

The screenshot shows the 'Bass driver' dialog box. It has a title bar with a close button. The main area is divided into several sections:

- Unit Selection:** A dropdown menu showing '1 driver unit'.
- Invert polarity:** A checkbox that is currently unchecked.
- Electrical Parameters:** Three input fields: 'Re' with value 4.0 and unit 'ohm', 'Le' with value 1.000 and unit 'mH', and 'Rex' with value 0.0 and unit 'ohm'. A question mark button is to the right.
- Driver parameters:** A section with four input fields: 'dX' (0.0 mm), 'dY' (-400.0 mm), 'dZ' (30.0 mm), and 'Piston radius' (75.0 mm).
- SPL data:** A section with a 'Filename' field containing 'Examples\Bass.frd' and a 'Browse' button, and a 'Scaling' field with value 0.0 and unit 'dB'.
- Impedance data:** A section with a 'Filename' field containing 'Examples\Bass.zma' and a 'Browse' button, and a 'Scaling' field with value 1.00.

The driver dialogs serve as placeholders for driver parameters, the driver dialog for the mid network is described in this chapter, the dialogs for the bass and treble drivers are identical

The dialog is divided into a number of groups:

- **Invert polarity**
Shift the polarity of the speaker unit. The polarity is visible in the drawing window.
- **Rex**
Additional increase of R_e due to voice coil heating, this helps the user to simulate what happens to the total frequency response at high power levels.
- **Re**
The DC-resistance of the voice coil
- **Le**
The voice coil inductance



The R_e and L_e values are in practice used only when computing filter components, the effect on frequency response and total impedance exist only as long as you do not import measured impedance data. Note also that R_e and L_e does not fully describe the impedance response of the driver unit.

- **1 driver unit / 2 driver units**
Here the number of driver units is selected, if 2 driver units they can be connected either in series or in parallel.

Driver parameters

In this group one set the parameters for each driver unit, if only 1 driver unit is in use the one can only set the parameters for **Driver unit 1**.

- **dX, dY, dZ**

The relative position of the driver unit from an acoustic point of view relative an imaginary reference point. dX denote the left-right direction and dY denote the down-up direction, a positive dZ is farther away from the listener than a negative dZ. For practical reasons one can set the position for e.g. the treble unit to dX=0,dY=0,dZ=0 and use that as a reference point for the other units. The settings of these parameters affect how the different driver units sum up and therefore the total frequency/phase response.

- **Piston radius**

The effective radius of the cone in the driver unit. This affects the off axis simulations.

- **SPL data**

In this subgroup the filename of the SPL data file is given, when the Browse button is clicked a file open dialog pops up where one may select a desired data file (also possible to drop files inside the **SPL data** box). The Scaling field allows you to offset the values in the SPL data file by a desired amount of dB. See chap 8.2.2 for a description of the import file formats.

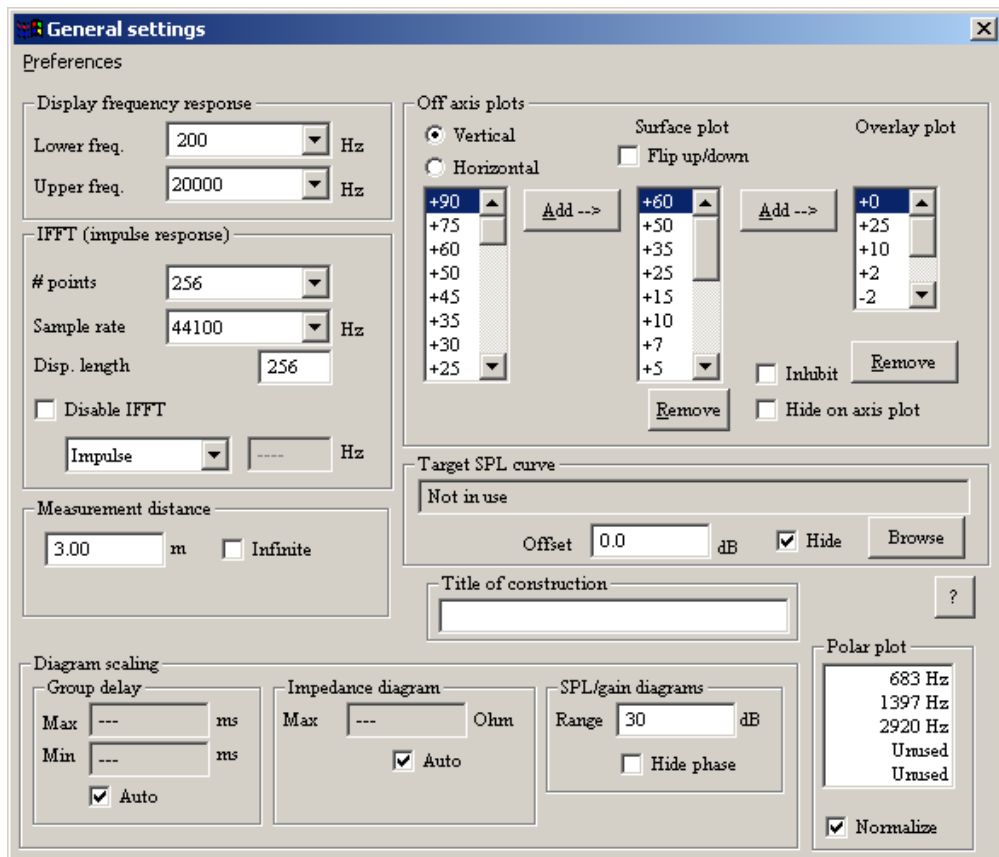
- **Impedance data**

In this subgroup the filename of the impedance data file is given. The Scaling field allows for scaling of the impedance values. For instance a value of 0.5 in the Scaling field will half the impedance values. Besides the Browse button to import measured data it is also possible to drop files inside the **Impedance data** box.

Note that if no filename is given for the second driver unit (if any), the SPL and impedance data of driver unit one will be used.

5.5 The general settings dialog

In the general settings dialog various parameters are set. This dialog consists of a few groups.



Display frequency response

The internal frequency span in the crossover utility is always 1-30000Hz. In order to be able to view a smaller part of the frequency span one may here set the **Lower** and **Upper** limits.

IFFT (impulse response)

In this group one set the number of points and the sample rate of the impulse response calculation. One can also disable the calculation of the impulse response just to save time. Besides impulse response it is possible to select step response or square wave response. The **Disp. length** allows to see the first part in detail.

Measurement distance

One can here choose how the frequency response should be calculated, either at an **Infinite** distance or at a predefined distance from the loudspeakers reference point. In all cases the measurement is along the Z axis in the coordinate system.

Off axis plots

Here one can select which vertical off axis angles to measure from. For the surface plot one can select up to 16 angles besides the 0 degree angle which is always default. For the overlay plot one can select up to 7 angles besides the 0 degree angle. The angles for the surface plot is selected from the leftmost list, while the angles for the overlay plot are selected from the surface plot list. With the **Flip up/down** checkbox the user can select if the positive angles should be upper or the lower half of the surface plot. The **inhibit** check box will cause the plot of the off axis diagrams to be inhibited when it is checked, this is valuable if one wish to e.g optimize the networks without having to deselect all rays. The **Hide on axis plot** checkbox makes it possible to hide the on axis plot in the off axis diagrams. The user can select in what direction (**vertical/horizontal**) the off axis modeling should be performed. Positive angles are always up (vertical) or right (horizontal) while negative angles are down (vertical) or left (horizontal). This is controlled by the measurement **Angle** setting.

Polar plot

This feature allows the user to set the features of the polar plot. Up to five different frequencies can be selected. The frequency values can be modified either by pressing the Arrow-left/right keys on the keyboard or by a double-click on one element in the listbox. Note that the frequencies are snapped to the nearest frequency point on the internal frequency vector, e.g 1000Hz is changed to 992Hz. The polar plot can be viewed at up to five different frequencies simultaneously. In the figure on previous page the frequencies 557Hz, 992Hz and 2003Hz are selected. The resulting polar plot is shown in fig 5.4. The polar plot can be either normalized or absolute.

Diagram scaling

The scaling of the diagrams that present impedance, groupdelay filter gain and sound pressure level can be altered here. Moreover it is possible to hide the phase curve in the *Summed frequency response* diagram

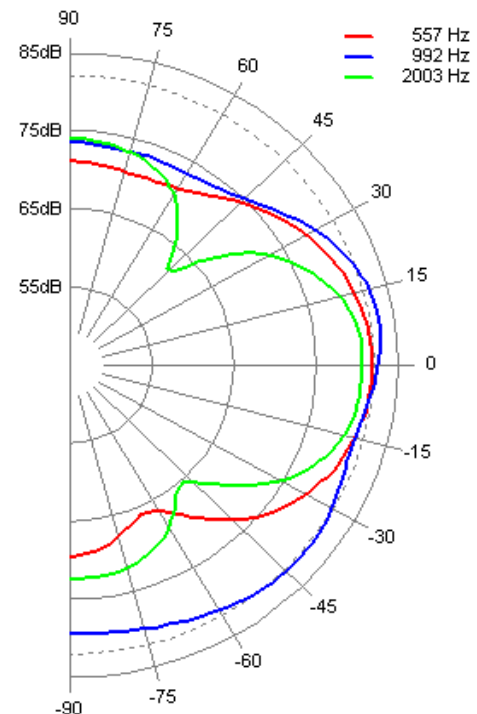


Fig. 5.4 A polar plot

Target SPL curve

It is possible to add an overlay target SPL curve to the *Summed frequency response* diagram. This target curve can be used in many applications, one is when one wish to match the dZ of measured units (see chap. 9.2.2). The target curve is imported with a click on the **Browse** button (it is also possible to drop the file inside the **Target SPL curve** box). An offset can be set so that target curve is located at the optimum level relative to the simulated SPL. The **Hide** checkbox is checked if one wish to hide the target SPL curve.

A small menu exist in this dialog:

Preferences / Save

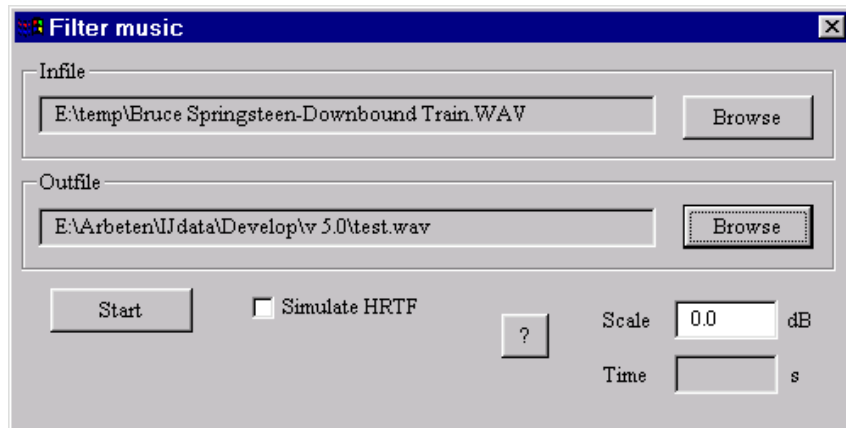
Allows the user to save the preferences so that the next time a project is created, the saved settings will be used from the beginning.

Preferences / Set to default

Will reset all settings.

5.6 The music filtering function

This function has been the most difficult one to name. The purpose is, however to filter a music sample through the frequency response of the loudspeaker. After this operation one can listen to the result with a pair of high quality headphones. The file format should be Windows WAV format.



The **Browse** buttons in the **Infile** and **Outfile** groups are used to select the in/out files. The **Start/Stop** is finally used to start and stop the filtering function. If the output level is too high one can reduce the level by means of the **Scale** edit field. Note that the filtering must be stopped before the scaling is changed. The **Simulate HRTF** makes it possible to simulate the head related transfer function when checked. The effect is that the perceived sound image is more alike a loudspeaker sound image although one listens with headphones.

For optimum speed a block filtering function using overlap-add FFT/IFFT operations is used. Therefore the **sample rate** in the IFFT group in the *general setup dialog* must be set correctly. Very short IFFT lengths may give an audible block effect, therefore it is recommended that the **#points** field in the *general setup dialog* is not less than 1024. The block length in the FFT operations is twice the **#points**, i.e. if **#points** is 1024 then the block length is 2048. Note that this function is quite slow, filtering 1 minute of CD music takes about 5 minutes on a Pentium 200MHz PC.

For optimum performance one should use high quality soundcards in combination with this function.

Also a couple of high quality headphones with a flat frequency response is a must.

5.7 What do the diagrams show

5.7.1 Filter gain

The filter gain lets you see the transfer function of each network.

5.7.2 Input impedance

The input impedance diagram shows the load on the driving amplifier, the individual impedance curves for each network is shown as is also the total impedance and the phase thereof.

5.7.3 Summed frequency response

The summed frequency response on axis is shown. Also the individual contributions and the total phase is shown.

5.7.4 Individual phase response

This diagram shows the phase response for each network, including driver phase response. This diagram is good to look at when optimizing for the best group delay characteristics.

5.7.5 Group delay

This diagram shows the group delay for each network including driver and the total group delay. Interesting to see here is that if e.g the bass and mid regions does not overlap well at the crossover frequency the group delay will peak at this frequency. As group delay magnitude is normally much smaller for higher frequencies than for lower frequencies one may have problem in seeing what happens at high frequencies due to the autoscaling function in LspCAD. With the **Page Up** and **Page Down** keys on the keyboard one can set the magnitude scaling of this diagram.

5.7.6 Impulse response

Impulse response for the total system. The length and sample rate is set in the general setup dialog. This diagram can also show the step response or the square wave response depending on the setting in the general dialog. When the square wave response is viewed the caption shows the actual frequency that depends on the number of points in the IFFT response and the sampling frequency. There is also the option to see the Energy time response, i.e. how the energy (dB) of the impulse response decays as a function of time.

5.7.7 Vertical off axis response, surface plot

This diagram looks much like a carpet when many viewing angles are selected. Here one can view how uniform the loudspeaker is in performance for other angles than on axis. Redrawing of this diagram is quite time consuming, therefore this diagram should be minimized when not in use.

5.7.8 Vertical off axis response, overlay plot

This diagram allows the user to see the frequency response for a limited set of viewing angles.

5.7.9 Polar plot

This diagram can display the polar response at up to five different frequencies, the frequency points are set in the general settings dialog. Note here that the polar response of a driver is simulated in LspCAD by a model that assumes a stiff piston placed at the end of a long tube, while this may hold for moderate off axis situations it is recommended to not pay too much attention to the behavior beyond +/-30 degrees. It is possible to view the polar plot either normalized or as an absolute reading.

5.7.10 Snapshot

See chap 4.2.6 for further description of this.

5.8 A few hints

5.8.1 Optimizing frequency/phase response

For the beginner it is advisable to start with a first or second order filters as higher order filters are quite tricky to optimize by hand. When optimizing two regions, for example bass and mid, it is recommended that the phase response of the two regions are overlapping i.e that the phase values are nearly the same at the crossover frequency. If the above proves difficult then one might consider moving the crossover frequency up or down or change the filter order of one or both networks, note that badly overlapping phase response will instantly show up as bad group delay characteristics.

5.8.2 Optimizing the zobel network

The simplest way to optimize the zobel network is type in **Re** and **Le** and then click on **Calc** in the zobel network group. This gives fairly good results but of course this can be done better. The steps below shows how:

1. Undefine all filter components and diffraction compensation links and attenuators in the network of interest, disconnect all other networks.
2. Open the Impedance diagram and look at the impedance for the network of interest while modifying the components of the zobel network. The optimum is when the impedance is flat and the phase of the impedance is close to zero over an as large frequency range as possible, especially at the crossover frequency.

5.8.3 How to use the notch section

The notch section can be used as a notch section if all three components R15, L15, C15 (for the bass network) are defined.

If only R15 and L15 are defined one can use it as a diffraction rise compensation link. A diffraction rise compensation link can be used when one wish to compensate for the step-like increase in sound pressure between 300-1500Hz that generally occurs due to baffle diffraction. The steps below shows how to utilize this feature in the best ways:

1. Undefine all filter components
2. Make sure that you have a good zobel network
3. Set the resistor part of the diffraction rise compensator (**R15** for bass network or **R25** for the mid network) to the same value as **Re** value as a starting point.
4. Set the inductor part (**L15** or **L25**) to about 1-2mH. You will now hopefully see that the frequency response has become flatter
5. Adjust the resistor and inductor until you are satisfied
6. Compute a suitable filter

6 The advanced crossover utility

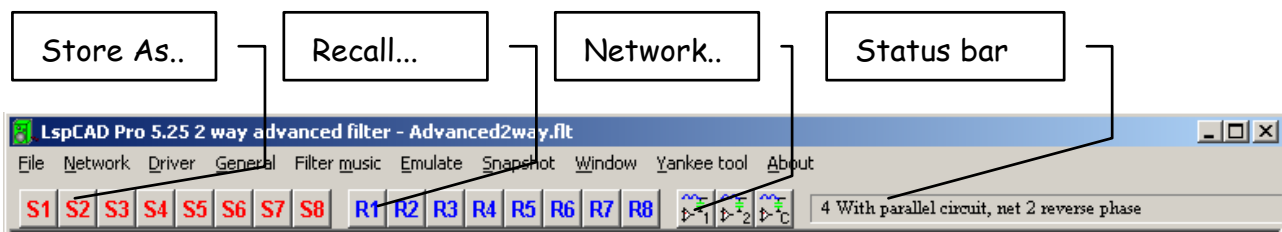
The advanced crossover utility allows you to model and construct a mixed blend of passive and active crossover networks.

The passive crossover networks can be either parallel or series topology networks. Up to four way systems (four networks) can be modeled, each network can consist of up to four driver units. The simple crossover utility is a good start for the one who feels that he or she is less experienced in this area.

The advanced crossover however makes no assumptions about what is bass, midrange or treble. Instead for each network one can select from the same set of components with a mix of both active and passive components. In the following text an active branch means that the branch contains active components i.e an OP amp.

The major advantage of the advanced passive crossover is that one can optimize the components of each network in order to achieve a predetermined frequency response.

The main window contains both a menu bar and a button/status bar.



The menu picks that differ from the simple passive crossover utility:

- **Network / Net 1**
Edit the parameters for network one.
- **Network / Common net**
Edit the parameters for a common network
- **Network / System optimization**
The system optimization dialog which allows you to optimize the frequency response against a predetermined target will appear
- **Driver / Net 1**
Edit the parameters for the drivers for network one.

The similar applies for network 2, 3 and 4.

- **Network / Store as...**
A nice feature that allows you to store away up to eight different network setups. Fast access to this function is via the **Store** buttons available below the menu bar. The stored sets are saved with the project file.
- **Network / Recall...**
Recall a stored setup. Similar to the **Store** buttons a set of **Recall** buttons are available. If **Prompt on recall** is enabled (see ch. 6.1) one have the opportunity to select what items to recall.

- **Network / Equalizer**
The control panel of the five band digital parametric shows up on screen. See section 7.5 for a detailed description.
- **Driver / Store as...**
Similar to **Network / Store as...**, can be used e.g to test different loudspeaker units or different relative placements of the loudspeaker units. This can also be of use if one wish to use real off axis measurements.
- **Driver / Recall...**
Recall a stored setup.
- **Emulate**
The crossover emulator dialog appears.

The status bar will contain the same text that you type in when you store network setup. If you later change e.g. a component value, the text will be prepended with a "[Modified]" text.

The schematics diagram displays the circuit schematic for all networks.

Fig 6.1 shows the overview of the crossover. The schematic shows a four way crossover with an active subwoofer and a common network that feeds a 3way series crossover network. For a detailed view of e.g network 2 one simply click on the **Net 2** button.

It will become obvious that one can mix active and passive crossover in many ways. For instance one can implement a passive crossover network with a 2way parallel topology part in parallel with a 2way series topology part. To put it strict, it actually takes at least two series topology networks to form a series topology crossover.

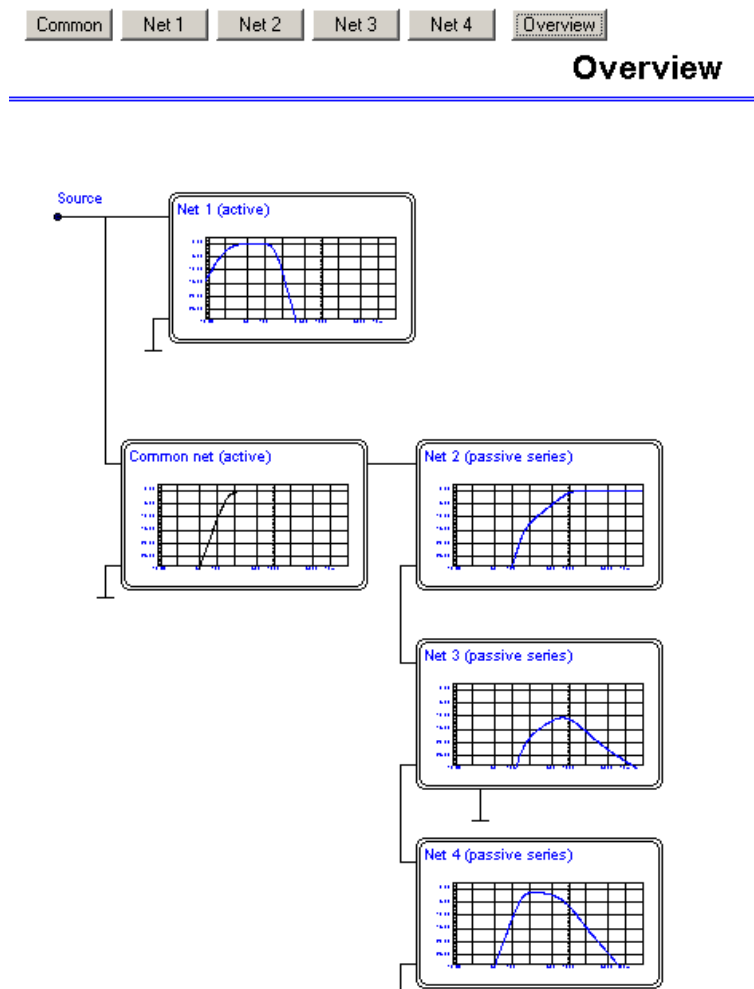


Fig 6.1 An overview of the crossover.

6.1 General settings

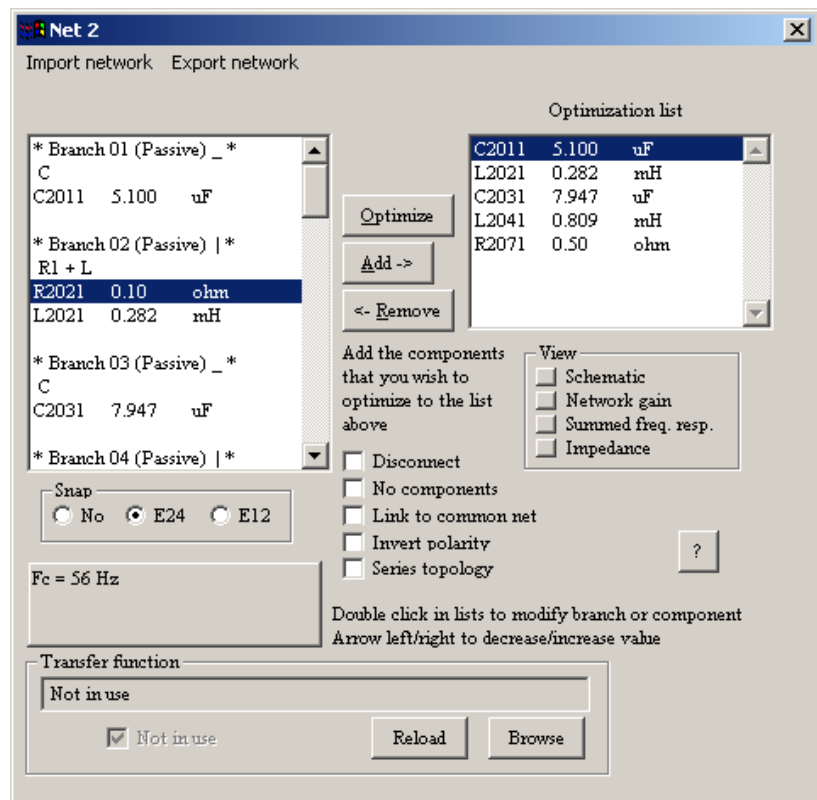
The general settings dialog is similar to the dialog found in the simple crossover utility with the addition of a measurement angle that will put the "measurement microphone" on some other angle than on axis, this is useful if one e.g. wish to view the square wave response of a loudspeaker off axis. The "Driver impedance load" will enable to view the load that the driver units and the shunt branches (branches 21..24) act on the end of the filter network (i.e. after branch 20). This is useful if one wish to see if a network is "sucking level" by lowering the impedance. If this checkbox is checked the driver load will be visible as dotted lines in the impedance diagram. Note that branches 21..24 are incorporated in the driver impedance load. The **Prompt on recall** checkbox is by default checked, this will make a dialog box pop up every time one recall a network set, in this dialog one can check what items one wish to recall. If this feature is not desired one can uncheck this option.

6.2 The network dialogs

The network dialogs differ from the simple passive crossover utility in that one can select how the network should look like by altering the branch types. For instance if one double clicks on **Branch 03** in the left listbox a popup menu shows up (see info later in this section) where one may alter the branch type and also copy/paste branches.

If one double clicks on a component in the list box a smaller dialog pops up that allows the user to enter a new value. One may also alter the component values by selecting a component (single click) and then press arrow left/right to alter the values 5% up or down. The snap option makes the component values snap to either E24 or E12 values when turned on, note that only components that are modified are snapped.

The network dialog box dialog box consist of two listboxes, the left is the main listbox where one can alter the branchtype and the component values. The right listbox is the optimization listbox. In this listbox one can only modify component values.



All components that one want the network optimizer to optimize are selected from the left listbox and added to the right (optimize) listbox by clicking on the **Add** button, to remove components from the optimize list one select the component in the optimize listbox and click on the **Remove** button.

The box below the snap group box shows information about the circuit element, for instance the Q-value and the resonance frequency of a series resonance circuit. In the figure above the box displays the center frequency and Q value of a parallel resonance circuit that serves as a notch filter.

Transfer function. Transfer functions can be imported into LspCAD, they should be of the format described in section 8.2.2. The need to import a transfer function may occur if one wish to incorporate an already constructed subwoofer with fixed filters into the simulations. Another use is if one wish to test crossovers devised with other crossover modeling software in the emulator.

If one for some reason wish to disconnect the entire network one can check the **disconnect** checkbox. **No components** is checked whenever one wish to disconnect all components but still have the driver unit connected. Note that all components are still visible in the network drawing although **No components** is checked.

The polarity of the driver unit(s) connected to this network is inverted if the **Invert polarity** checkbox is checked.

Link to common network is selected when one wish to feed this network from the common network instead of directly (see chap. 6.2.1 about the common network). Be aware that it makes very little sense to link a network with active branches to a common network with passive branches

For each branch one can select if it should be either active or passive. A rule to remember here is though that branches with active components should always precede branches with passive components (pretty obvious isn't it?)

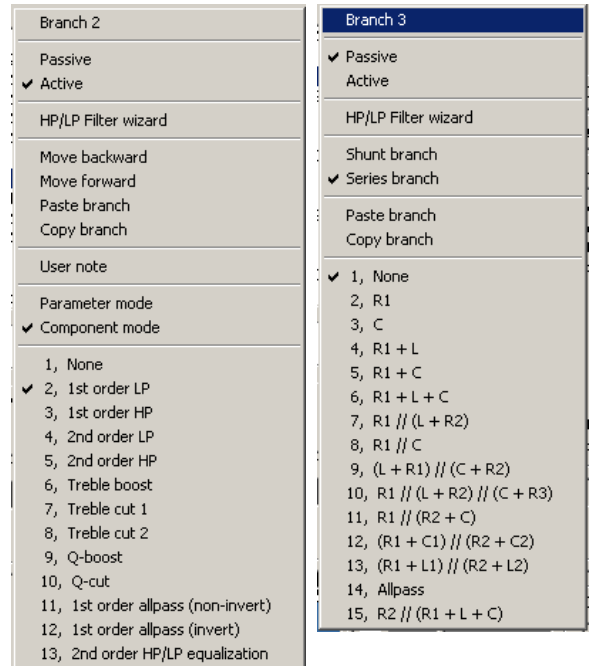
Completely passive networks (i.e no branches with active components) can be either parallel or series topology. If series topology is desired, the **Series topology** checkbox is checked. The difference between series topology and parallel topology networks is described later. Note that a series topology network can not contain active branches, therefore the **series topology** checkbox is hidden if a branch in the network is active.

The **View** group contains a number of buttons that, when clicked will cause the selected diagram to be brought on top. For instance if one click on the button next to "Summed freq. resp." the Summed frequency response diagram will be brought on top and visible for the user.

When a branch in the branchlist is double clicked a popup menu appears that allows one to move branches forward/backward, copy/paste, set the branchtype and also select if the branch should be active or passive. The looks of the popup menu depends if the branch is active or passive. Note that in some cases some menu picks are grayed out or invisible, for instance one cannot place a passive branch before an active branch.

The HP/LP Filter wizard allows to quickly construct initial valued passive or active filters.

Active branches can be modified in either Component mode or Parameter mode, in the latter case one can control the things such as the cutoff frequencies and Q values instead of the component values, this mode makes filter prototyping faster (see ch 6.6.1). For active branches it is possible to set a user note for each filter section via the *User note* menu pick.



The passive branches can be selected to be either series or shunt branches. By default the branches in a passive network are series-shunt-series-shunt, this can be modified to reflect the desired looks of the crossover.

This dialog contains a small menu with two menu picks:

Export network

Makes it possible to save the network so that it can be imported in another project. The format is a native LspCAD format.

Import network

Import a network.

The export / import may be useful if one for instance wish to copy entire networks between crossover projects.

Drag and drop. The action given by a drag and drop operation into the network dialog depends on where the file is dropped. If the file is dropped inside the **Transfer function** box the dropped file will be treated as transfer function data, otherwise it will be treated as a network file.

6.2.1 The common network

The common network can be used for instance if one have the scenario below:

A three-way system with two power amplifiers per channel.

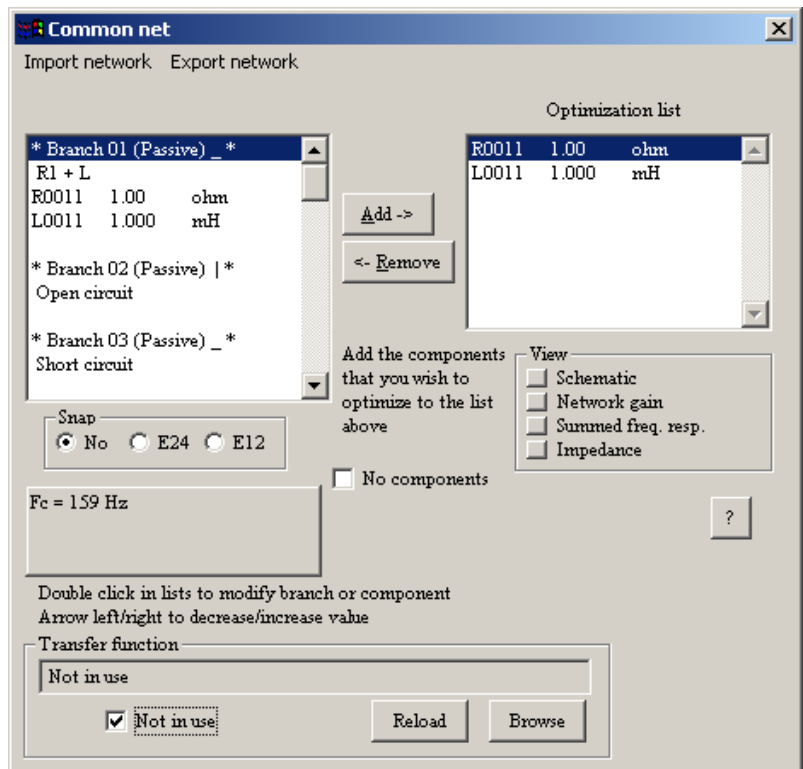
The network for the bass is active, therefore the bass unit is fed by a separate power amplifier where the crossover is preferably active.

The midrange and the treble units share the same power amplifier, therefore the crossover between mid and treble must be taken care of by a passive network with a highpass filter for the treble unit and a lowpass filter for the midrange. The highpass

filter for the midrange can be active. Here the common network plays a crucial role.

This common network can contain both passive and active.

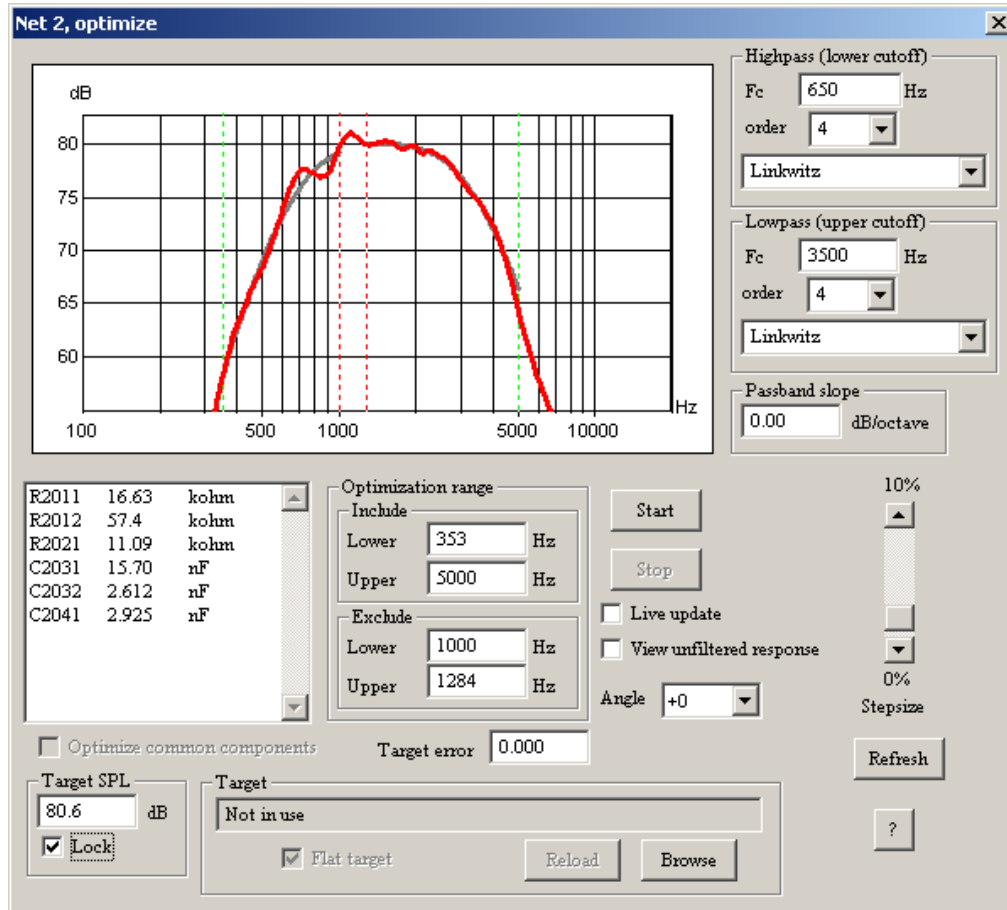
The looks of the dialog box for the common network is the same as for the other network dialogs, components that are to be optimized are put in the optimization list. There is no Optimize button available however. The idea here is that the components of the common network is optimized in the optimize dialog of the networks that are linked to the common network. Complicated ?, it will become more clear later.



6.2.2 The network optimizer

The optimization tool makes it possible to compute filters that match a predetermined target curve.

With this tool it is possible to model and construct complex structures such as 4th order bandpass filter alignments, a task which would take weeks or even months to do by hand.



Network optimization is not possible with the passive series topology networks.

The optimization tool is accessed by a click on the **Optimize** button in the network dialog. Before the optimization session can start one must first add the components that need to be optimized to the optimization list (see chap 6.2).

The components that are subject to optimization are available in the listbox on the left of the optimization dialog. In the diagram up to three plots are available:

- Solid red (thick):** The frequency response of the network+driver unit.
- Grey (thick):** The target frequency response.
- Red dotted:** The unfiltered frequency response.

A few controls exist in this dialog.

- **Stepsize**

The stepsize i.e how large steps the optimizer should take for each component, the range is between 0 and 10%.

- **Live update**
Allows for an update of all diagrams during optimization, slows down the process substantially though.
- **View unfiltered response**
Allows the user to view the unfiltered response.
- **Angle**
Provided that you have selected rays for off axis plots you can here select in which direction you wish to optimize. Please make sure that you optimize in the correct direction each time you open the optimization dialog.

Lowpass

Here one set the properties for the part of the target curve that describes the low pass alignment.

- **Fc**
The cutoff frequency
- **Order**
The filter order (0-4)
- **Butterworth/Linkwitz/Flat delay/2nd order variable Q**
Here one select Butterworth (-3dB @ Fc) or Linkwitz-Riley (-6dB @ Fc) or Flat delay alignment. Which of those one should select depends on how the resulting networks overlap in phase at the crossover point. If the resulting phase difference between two overlapping networks is close to 0 or 180 degrees then Linkwitz-Riley alignment should be selected. If, on the other hand, the phase difference is 90 or 270 degrees the Butterworth alignment is probably a better choice. The flat delay alignment may be selected if one wishes to strive for as small groupdelay distortion as possible. This alignment resembles the Bessel filter alignment.
The 2nd order variable Q is a special case that allows you to optimize against highpass or lowpass 2nd order transfer function with a given Q value.

Highpass

Here one set the properties for the part of the target curve that describes the low pass alignment, see above for explanation.

Optimization range

Here one select over what frequency range the optimizer should compute the difference metric, this is useful as valuable components are otherwise wasted on unimportant parts of the frequency range. A recommendation here is thus to set the optimization range so that the working range is included and also points in the frequency range that are 20-25dB below the peak sound pressure level or higher. Two ranges can be given.

Include: gives the frequency range over which the optimization should work.

Exclude: an exclude range could be given in order to exclude e.g. a bump in the frequency range that happens to be within the include range. In the example above the range 1000-1284 Hz is excluded so that the bump in that region wont disturb the optimization.

Target SPL

If the lock checkbox is checked the target SPL curve is locked to a specified level so that problems with ever decreasing sensitivity is efficiently avoided. With the checkbox unchecked the target SPL curve will float around the achieved curve.

A target response can be imported with a click on the **Browse** button (also possible to drop inside the **Target** box). The format of the target is a simple ASCII text file with frequency point and response. A simple file with the below contents:

```
20 0
500 0
2000 -3
3000 -3
5000 0
20000 0
```

Will, when imported, force the optimization to achieve a frequency response with a 3dB drop between 2000Hz and 3000Hz. A click on the **Reload** button will reload the target file. When **Flat target** is checked the target will be forced to become flat. The target data file may contain phase information also but the phase information is not used.

Passband slope

Here one can enter a desired passband slope in the range ± 2 dB per octave. Introducing a slope in the frequency response may be desirable if one for instance wish to recess the level in the midrange in a two way system.

Target error (Error)

Here one can set the target mean deviation from the target frequency response, when the start button is pressed this field changes to show the achieved mean deviation from target response. When the target is reached the iteration will stop. In the target is set to zero the optimization will continue forever (well, not exactly as we deal with a Microsoft operative system). The error displayed is the average distance in dB from the target within the optimization range..

Optimize common components

If this checkbox is checked the common components that are selected for optimization also appear in the list. Note that this option is only available if **Link to common network** is checked in the network dialog.

A few words...

With the optimization feature the construction of complex filter structures is really simple. One problem that one may face is the case where the drivers for the different networks are not acoustically aligned.

By acoustically aligned is meant that the sound from each driver should arrive at the listener at the same time. Often the sound from the bass unit will arrive later at the listener than the sound from the treble unit. The reason is that the sound from the bass unit is "created" in the middle of the driver at the dust cap at high frequencies, as the piston of the driver is shaped as a cone the result is thus that the sound from the bass unit is delayed more.

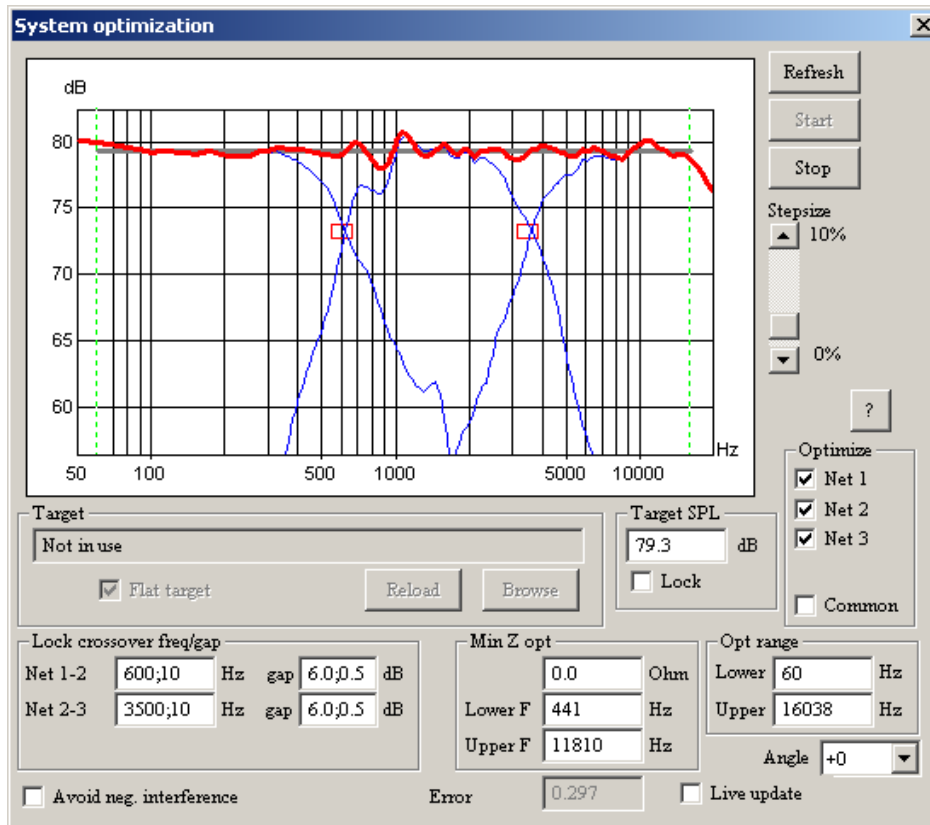
The obvious solution is to place the bass unit closer to the listener by about 2-4 cm depending on the units involved.

In the case where the drivers are not acoustically aligned the minimum phase property that makes it possible to optimize each network to a certain target response and then "glue" together the result will fail to various extents depending on how badly aligned the driver units are. This problem is larger in the treble region. There are many reasons to why the driver units should be acoustically aligned, the main reason is that the sound image fits together much better if the drivers are aligned.

Selecting the appropriate filter order and alignment can be a cumbersome task, often it is quite easy to achieve the target frequency response when optimizing. Problem often may come when it is time to combine the two adjacent networks in the frequency range and achieve a flat response. One simple rule is that one should strive for the same slope in the target response on both sides of the crossover frequency. Also the "Loudspeaker design cookbook" by Vance Dickason or any other book that covers this area is recommended.

6.3 The system optimizer

As the name reveals this tool helps to optimize the frequency response of the whole system towards either a flat target or an imported target within a given frequency interval. The system optimization is preferably the last step in the optimization round after all individual networks have been fixed but in some cases it is possible to get good results by just jumping right to the system optimizer bypassing the network optimization (see tutorial section).



The system optimizer dialog looks almost like the network optimization dialogs, the **opt range**, **angle**, **start/stop**, **Target SPL** and the **stepsize** controls are used in the same way. The same applies to the **Target error / error**, note however that the error value here will not only depend on the deviation from the target response but also on the deviation from the desired crossover frequencies (if set) and the amount of negative interference (if checked). Therefore the error value may reach values as high as 100000 or more, don't let that disturb you, it will most likely go down as the optimization continues.

Note that the optimization range should be set sensibly, for instance the rolloff in the bass and treble end is inevitable and is better omitted. In the figure above the response of the treble unit drops above 16000Hz, for this reason the Upper limit is set to 16038Hz.

The components that are about to be optimized are selected in the individual network dialogs. It is recommended to select all components for this type of optimization, possible exceptions are impedance equalization networks such as zobel and series resonance circuits that are used for the sole purpose of straightening the impedance response of the drivers.

In certain cases (not too seldom) one may end up with a ruler-flat response, the impedance response may however reach very low levels. If the **Min Z** value is set to a value, e.g. 7.0Ω the optimization algorithm will optimize the frequency response with the extra rule that the impedance will not get below **Min Z** within the given range **Lower F** to **Upper F**.

When using this feature it is possible that the response might get worse temporarily, don't let that discourage you, let it work and you'll see that it will most likely converge again. The **Min Z** value should not be set too high, instead increase it gradually by 0.5Ω at a time and let it converge in between the increments.

In the **Optimize networks** field it is possible to select which networks should be optimized.

Lock crossover freq.

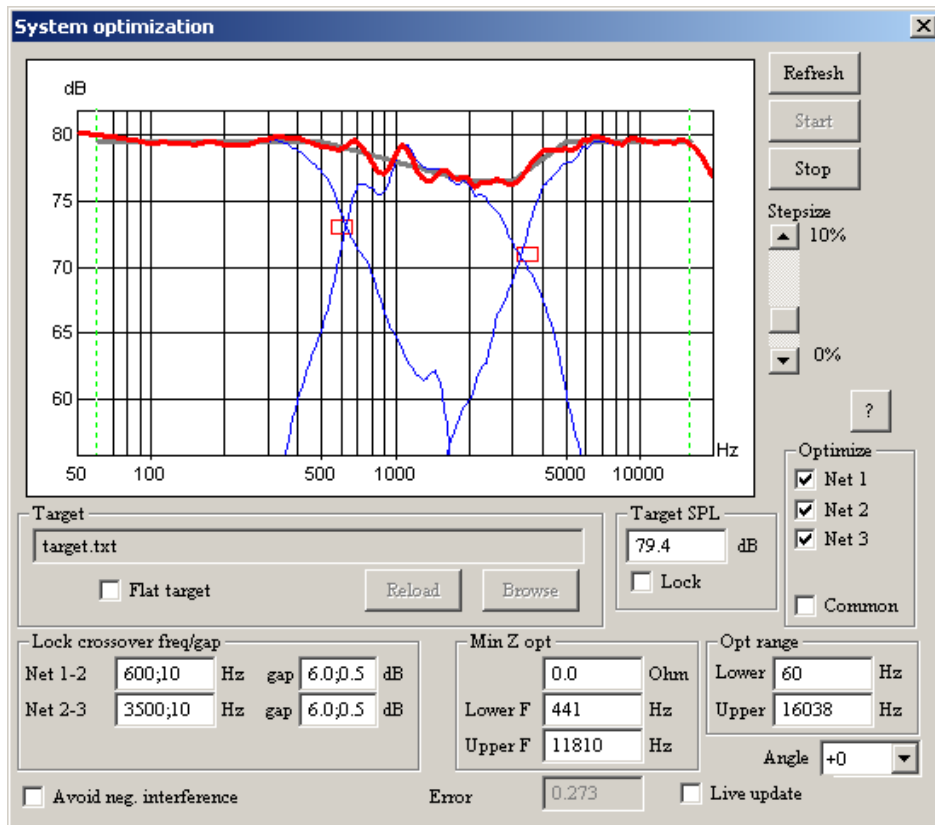
Often it is desirable to lock the crossover frequencies to certain regions during the optimization phase. This can be done via the controls in the **Lock crossover freq** group. In the above example the crossover frequencies have been locked to $600\text{Hz} \pm 10\%$ and with a gap between the target curve and the individual contributions of $6\text{dB} \pm 1\text{dB}$ at the crossover frequency. If locking is not desired one simply enter a zero value in the controls. The controls contain two values separated by a semicolon (;). The first value is the crossover frequency (or the gap) and the other value is the allowed deviation expressed as a percent value for the crossover frequency and a dB value for the gap.

Avoid neg. interference will, when checked cause the optimizer to avoid results that would cause the drivers to counteract one another, in other words out of phase problems close to crossover frequencies will be avoided as much as possible.



It is important to remember here that as one locks a crossover frequency or put a lower limit on the impedance one also add more restrictions and reduce the ability for the optimizer to reach a target. Also the **Avoid neg. interference** option might the reduce the freedom for the optimizer.

A target response can be imported with a click on the **Browse** button (also possible to drop inside the **Target** box).



The format of the target is a simple ASCII text file with frequency point and response. A simple file with the below contents:

```
20 0
500 0
2000 -3
3000 -3
5000 0
20000 0
```

Will, when imported, force the optimization to achieve a frequency response with a 3dB drop between 2000Hz and 3000Hz. A click on the **Reload** button will reload the target file. When **Flat target** is checked the target will be forced to become flat. The target data file may contain phase information also but the phase information is not used. The figure shows the result when the target -3dB between 2000Hz and 3000Hz has been achieved.

6.4 The driver dialogs

The driver dialogs are almost the same as for the simple passive crossover utility. (see section 5.4)

When driver units 2 and up are selected two extra checkboxes appear with the caption "Same as driver 1". The idea behind this is that one checks this checkbox if one wish to use the data that is imported for driver unit 1. Thus one avoids the need to import the same data over and over again.

These are by default checked and will be unchecked automatically as soon as one imports a file.

Moreover the Scaling input fields are dimmed out to signify that the scaling properties of driver 1 apply if the checkbox is checked.

Smoothing of imported frequency response data can be applied, the available options are: No smoothing, 1/24th, 1/12th, 1/6th and 1/3rd octave smoothing. In the figure above 1/12th octave smoothing is applied. This can be especially useful when modeling with crossovers for e.g. horn loudspeakers.

It is possible to remove or add delay to the imported SPL data by means of the **delay** control.

Each driver can be oriented individually both in x and y axis in a +/- 90 degree range. This is accomplished by setting the orientation in the **Orientation** field to the right of the **Rel. location** field. Positive angles are always up (y-axis) or right (x-axis) while negative angles are down (y-axis) or left (x-axis).

The shape of the driver can be either **Circular** or **Rectangular**. If the shape is circular then the piston **Radius** is set, if the shape is rectangular then the **Height** and **Width** is set instead. This setting is used to predict the off axis response of the driver. Note here that it may be necessary to set the dimensions of the driver to other than the physical dimensions to get a good match with the measured off axis response of the driver.

6.5 Passive crossover branches

As mentioned earlier a network can contain both active and passive branches, the passive branches are described in this section.

For those who are familiar with the **IMP3wVR** software by G.R Koonce it can be said here that many ideas from that program are stolen with pride (and also with kind permission). In order to get meaningful results, both impedance and SPL data should be imported for all passive networks.

The topology of the network is such that each network can consist of up to 24 branches connected after one another (of them the four last branches are across the driver units and are always passive). When a new advanced passive crossover project is created the user select whether the crossover should have series or parallel topology. Later this setting can be modified, it is ever possible to change one network to become active.

The use of the branches differs somewhat depending on whether or not the network is part of a series topology network or a parallel topology network.

6.5.1 Parallel topology

By default, when a project is created, odd numbered branches are series (in series with the loudspeaker load) and even numbered branches are shunt (parallel with the loudspeaker load). The last branch(es) (numbered 21-24) are always shunt branches as they are connected across the loudspeaker(s).

Fig 6.2 below shows the midrange section of a three way system

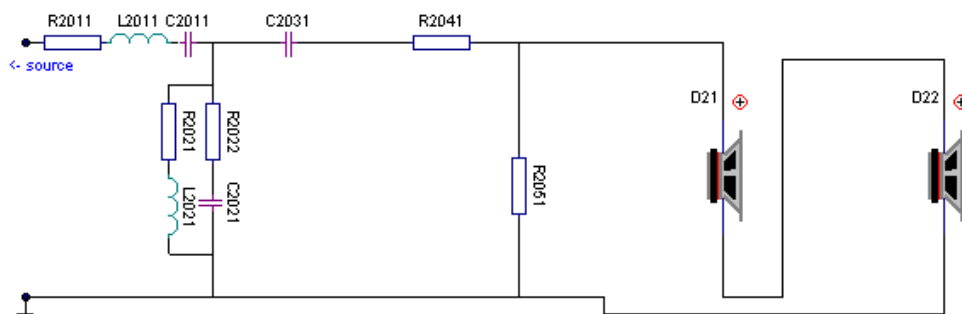


Fig. 6.2

6.5.2 Series topology

A network can only be a part of a series topology crossover if it only contains passive branches. By default, when a project is created, even numbered branches are series (in series with the loudspeaker load) and odd numbered branches are shunt (parallel with the loudspeaker load).

The last branch(es) (numbered 21-24) are always shunt branches as they are connected across the loudspeaker(s).

Fig. 6.3 shows network 3 in a series crossover (same crossover as fig. 6.1). Note the connection to networks 2 and 4. Note also the special connection of branch 0 (C3001) directly to ground, this arrangement is necessary for the networks that are in the middle in 3-way or 4-way series crossover.

It is not an exaggeration to say that crossovers built on series topology networks are much more difficult to model than crossovers built on parallel topology networks. The reason is of course that if you modify components in one series network, not only that network will be affected but also the other series network will be affected.

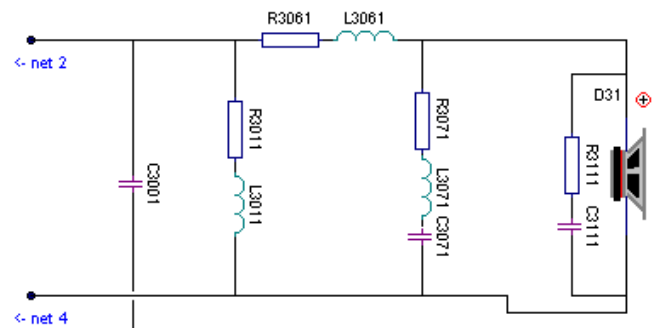


Fig. 6.3

6.5.3 Circuit elements

The circuit elements consists of only passive components.

The naming of the components follows the convention

XIJJK where:

X is the component type, C for capacitor, L for inductor and R for resistor.

I is the network number (1..4)

JJ is the branch number (1..20,21..24)

K is either 1 or 2, in cases where 2 resistors are used in a branch one resistor will have suffix 2.

One example:

R3082 is the second resistor of the eighth branch of the third network.

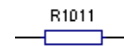
Note that if the value of a resistor is set to zero (actually a value smaller than $10^{-8} \Omega$) the component will be hidden in the schematic.

It is possible to select from 13 different branch types, they should cover most needs.

Some of the more complex circuit elements are equipped with a "pop-up" wizard that will help you to select the appropriate values

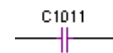
6.5.3.1 R1

A resistor



6.5.3.2 C

A capacitor



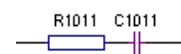
6.5.3.3 R1+L

An inductance with (internal) resistance in series



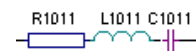
6.5.3.4 R1+C

A capacitor and a resistor in series



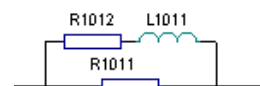
6.5.3.5 R1+L+C

A series resonance circuit, a wizard helps you to get the correct values.



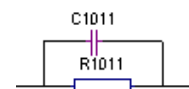
6.5.3.6 R1 // (L+R2)

An inductor in parallel with a resistor.



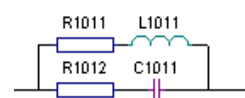
6.5.3.7 R1 // C

A capacitor in parallel with a resistor



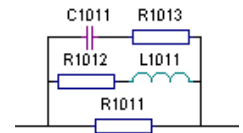
6.5.3.8 (L + R1) // (C + R2)

A capacitor in parallel with an inductor

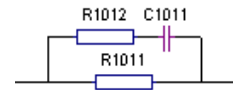


6.5.3.9 $R1 // (L + R2) // (C + R3)$

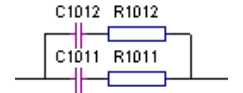
A parallel resonance circuit, a wizard helps you to get the correct values.

**6.5.3.10 $R1 // (R2 + C)$**

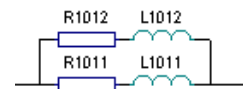
A capacitor in parallel with a resistor (variation of $R1 // C$)

**6.5.3.11 $(R1 + C1) // (R2 + C2)$**

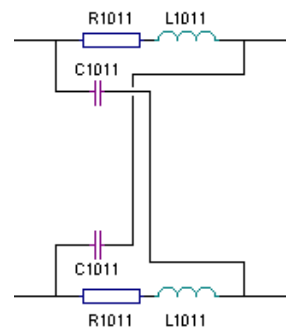
Two capacitors in parallel

**6.5.3.12 $(R1 + L1) // (R2 + L2)$**

Two inductors in parallel

**6.5.3.13 Allpass filter**

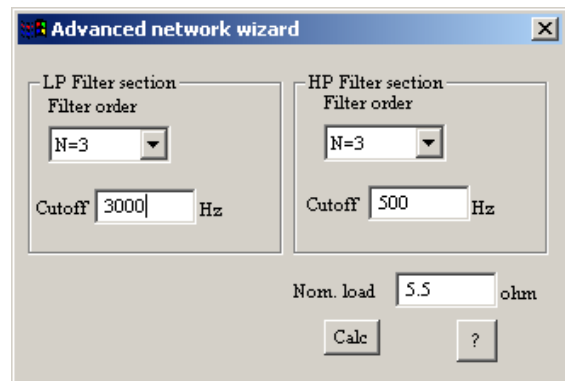
An allpass filter i.e a delay network. A wizard pops up that gives you the option to set Z_{nom} , the nominal load (e.g 8Ω) and T_d , the desired timedelay, the components are then calculated to meet the requirements

**6.5.3.14 $R2 // (R1+L+C)$**

A series resonance circuit in parallel with a resistor.

6.5.4 The HP/LP filter wizard

Not exactly a new Houdini but anyway the Wizard might help the user to get a first starting point. The wizard is accessed via the menu pick *HP/LP filter wizard* in the menu that pops up when one double click on a branch in the branchlist.



Here the user makes a first wild guess of the components in the network. In the example given in the figure a filter will be computed that would achieve a 3rd Butterworth bandpass alignment with cutoff frequencies 500 and 3000 Hz. The nominal load (here 5.5 Ω) is set so that the match with the loudspeaker load is the best possible.

The insertion point is where you double clicked.

Note that if only a low pass filter is wanted the order of the highpass filter (HP) is simply set to zero. The opposite applies if only a high pass filter is wanted.

Very quickly one will notice that the filter components that the Wizard propose does not match a real loudspeaker unit very well, especially as the driver units frequency response need to be compensated for, a fact that the wizard does not care the least about. That is why we need the optimization tool that was described in a previous chapter.

6.6 Active crossover networks

An active crossover network contains mainly active components. The topology of the network is such that each network can consist of up to 24 branches, each connected after the other.

Branch 1 through 20 are series branches i.e they are in series with the signal to the power amplifier. Each of these branches are equipped with an operational amplifier. Please note that each branch assumes that it is fed with a low impedance source, this especially applies to the first branch.

Branch 21 through 24 are passive parallel (shunt) branches and they are placed after the power amplifier. The naming of the components follow the same rules as given for the advanced passive networks. In order to get meaningful results, SPL data (and possibly also impedance data. should be imported for all active networks.

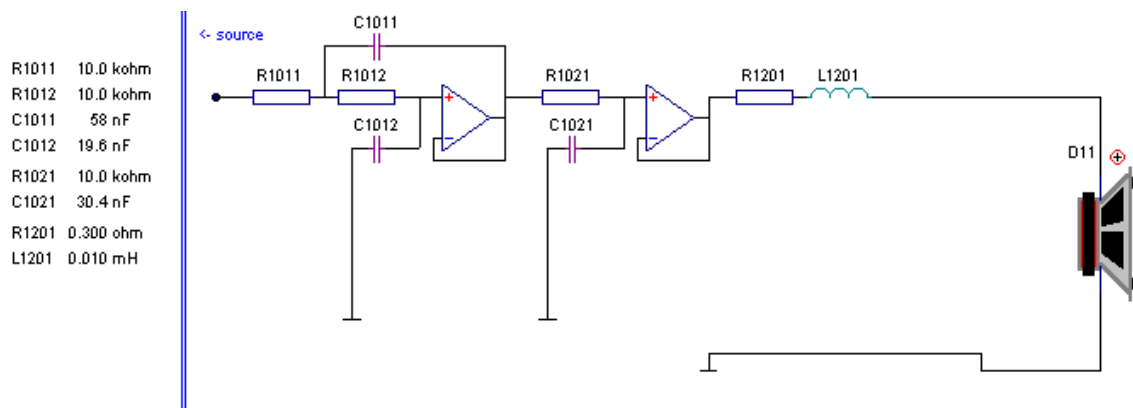


Fig. 6.4 A bitmap copy of the schematic for an active crossover unit, note that it is always assumed that the circuit that feeds the first active filter has a low output impedance. Note also the additional passive branch 20 consisting of a resistor (R1201) and an inductor (L1201) connected in series with the loudspeaker load to simulate the cable between the power amplifier and the loudspeaker unit.

6.6.1 Parameter/Component mode

By default all active branches are in component mode, which means that the transfer function is modified by changing the components. This may be a tedious way to tweak a complete filter with several 2nd order filter links.

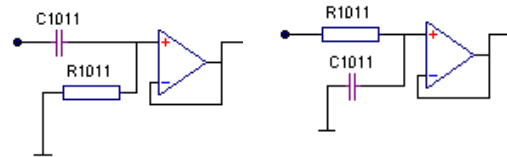
In the parameter mode the user has the opportunity to set the parameters that control the transfer function directly. As an example we have a 2nd order lowpass filter section, it is controlled by the cutoff frequency (f_0), the Q value (Q) and possibly also the gain (K). In the parameter mode the components in the network listboxes will be replaced with the parameters instead. For instance the parameters for the 2nd order LP filter above may be named F1011, Q1011 and K1011 where the numbering follow the same principles as described in ch 6.5.3.

6.6.2 Circuit elements

The circuit elements for the branches 1 to 20 that are active consists of an operational amplifier and a few passive component. Below follows a description.

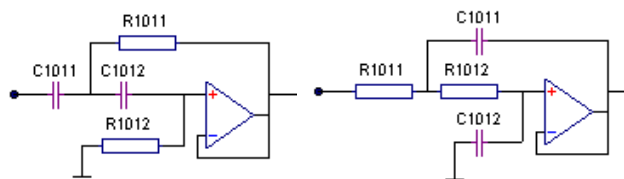
6.6.2.1 1st order HP and LP filter

The first order highpass filters has a slope of 6dB/octave.



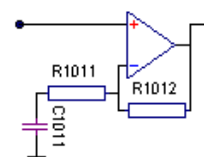
6.6.2.2 2nd order HP and LP filter

The second order highpass filter has a slope of 12dB/octave. For higher order slopes one can cascade with either first or second order filter elements. It is possible to use equal component filters, then two additional resistors (Rxxx3 and Rxxx4) are present that allows to achieve the desired Q value.



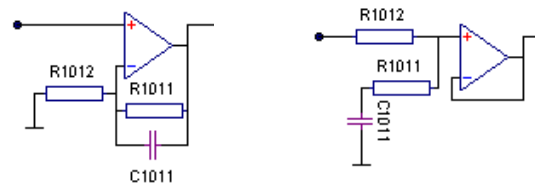
6.6.2.3 Treble boost

A treble boost circuit makes it possible to achieve a raised level in the treble region.



6.6.2.4 Treble cut

A treble cut circuit makes it possible to achieve a suppressed level in the treble region, two types of treble cut circuits exists.



6.6.2.5 Q-boost and Q-cut

A Q-cut and a Q-boost circuit creates a notch or a peak in the frequency response. A wizard will help you to select the appropriate values for the components.

When using the Q boost and Q cut one may quickly end up with enormously large inductors. The trick is here to use a gyrator circuit as shown in fig 6.5. With the use of a gyrator circuit one gets a simulated inductance that eliminates the need for a real inductance.

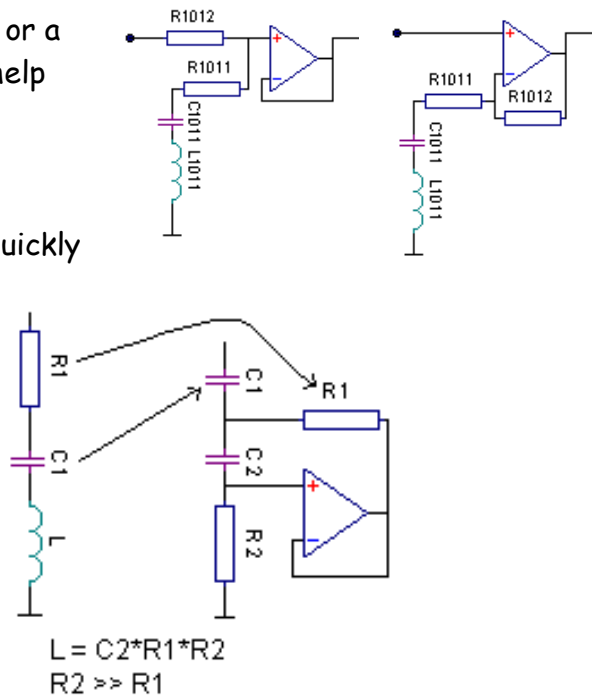
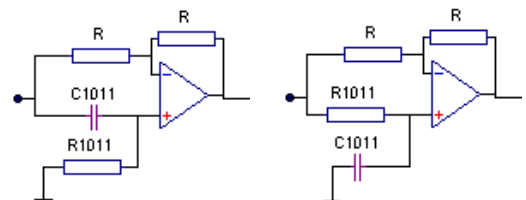


Fig. 6.5 A gyrator circuit that replaces an inductor.

6.6.2.6 Allpass filters

An allpass filter can be used to add an extra delay for e.g a treble unit. Two types of allpass filters are possible, one that is inverting at low frequencies and one that is non-inverting. The selection of the values for the resistors "R" is arbitrary and may be for instance 10kΩ.



6.6.2.7 2nd order highpass equalization

This circuit is also known as the Linkwitz transform circuit. It makes it possible to equalize the response of a closed box. When this circuit element is selected, a wizard will pop up that allows one to set the necessary parameters for the actual and desired cutoff frequencies and Q values. Figure 6.6 below shows the relationship between actual and desired F_0 and Q_0 .

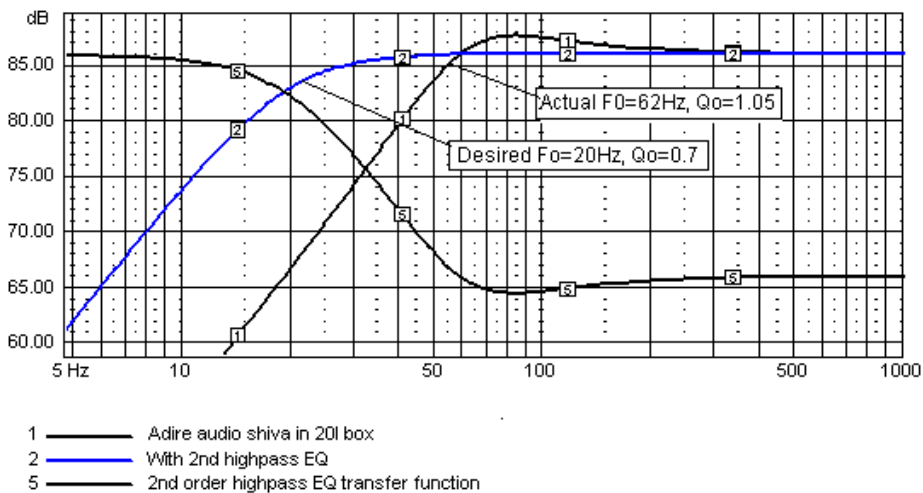
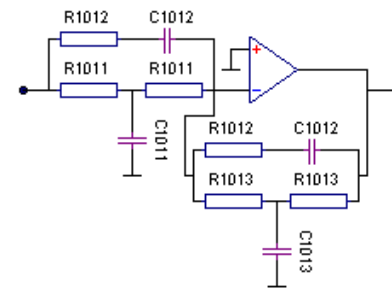


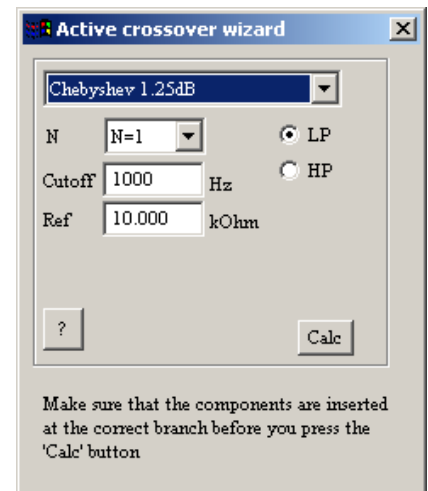
Fig 6.6 The relationship between actual and desired F_0 and Q_0

In parameter mode one can use this filter section as a 2nd order lowpass equalizer, it is however then not possible to switch back to component mode as the 2nd order lowpass equalizer is not supported in component mode.

6.6.3 The HP/LP filter wizard

The wizard is accessed via the menu pick *HP/LP filter wizard* in the menu that pops up when one double click on a branch in the branchlist.

With this wizard one can calculate component values for up to 8th order active highpass or lowpass filters. The reference value Ref is a resistor value (LP filters) or a capacitor value (HP filters) on which the other component values are computed.



6.7 What do the diagrams show (cont.)

This is a complement to section 5.7.

6.7.1 Input impedance

The input impedance diagram shows the load on the driving amplifier, the individual impedance curves for each network is shown, moreover also the total impedance (thick black) and the phase thereof is shown.

If the common network is being used the impedance load from that is shown in gray, the gray line is thick if the common network contains active components, this implies that the common network has its own power amplifier and will thus not load the driving amplifier (the source), instead the thick gray line is the load on the power amplifier for the common network. If no active components exist in the common network, the gray line is thin. This implies the load that the passive common network acts on the driving amplifier (source).

If **Driver impedance load** is checked in the *General settings* dialog a set of extra graphs (with dotted lines) will appear in the impedance diagram. These graphs show the load that the driver units (and the shunt branches 21..24 cause on the other end of the crossover. With this information it is possible to see if the rest of the crossover is "sucking level" by lowering the impedance.

7 The crossover emulator

When working with the design of crossover networks for loudspeakers, computer software gives the developer great help to pave the way towards the ultimate design.

We know software such as LEAP and CALSOD and these days also LspCAD gives great opportunities to optimize crossovers, and that at a quite small cost in terms of spent man-hours.

One thing that still is missing in this aspect is the ability to directly listen to how it sounds without having to build up the complete crossover.

The **Emulate**-function gives the user the option to emulate the behavior of the crossover networks inside the computer and that possible in real time. What is needed is a high quality 3D soundcard with at least 4-channel (2 stereo) output. Better yet is an Echo Audio multi-channel audio card. The Echo cards that can be used with the emulator is the Darla/Gina /Layla or Darla24/Gina24/Layla24. See <http://www.echoaudio.com> for details. It is possible that in the future other multi channel audio cards will be supported, the requirement is that the multi-channel card supports open and playback of several channels in synch (synchstart). An updated list of supported soundcards will be present at <http://www.ijdata.com>.

7.1 What is the gain?

With real-time emulation of loudspeaker crossovers units one has the ability to test a number of tweaks in order to reach the perfect configuration.

- **Different filter orders.**
- **Different crossover frequencies.**
- **Move individual loudspeaker units forward and backward.**

With the above mentioned possibilities it is easy to tailor a loudspeaker for a specific placement in a listening room. Moreover it is a well-known fact that some music sounds better on some loudspeakers, with this tool one can easily adapt the behavior of a loudspeaker to a specific kind of music.

7.2 How does it work?

The filter emulate function calculates a set of digital filters for each crossover network, with the same transfer function i.e. the frequency response of each filter (both magnitude and phase). Moreover the user selects the favorite music piece that he or she wishes to listen during the evaluation of the particular crossover. The music signal is then split up in several frequency bands, for instance for a two-way stereo system one gets the following configuration.

Bass, left channel	Output 1
Treble, left channel	Output 2
Bass, right channel	Output 3
Treble, right channel	Output 4

Thus, for a two-way stereo system a four-channel soundcard such as SoundBlaster Live Player 1024 is needed. Moreover a four-channel power amplifier and some cabling will be necessary. Figure 21 below shows a block diagram that visualize LspCAD's workflow inside the PC.

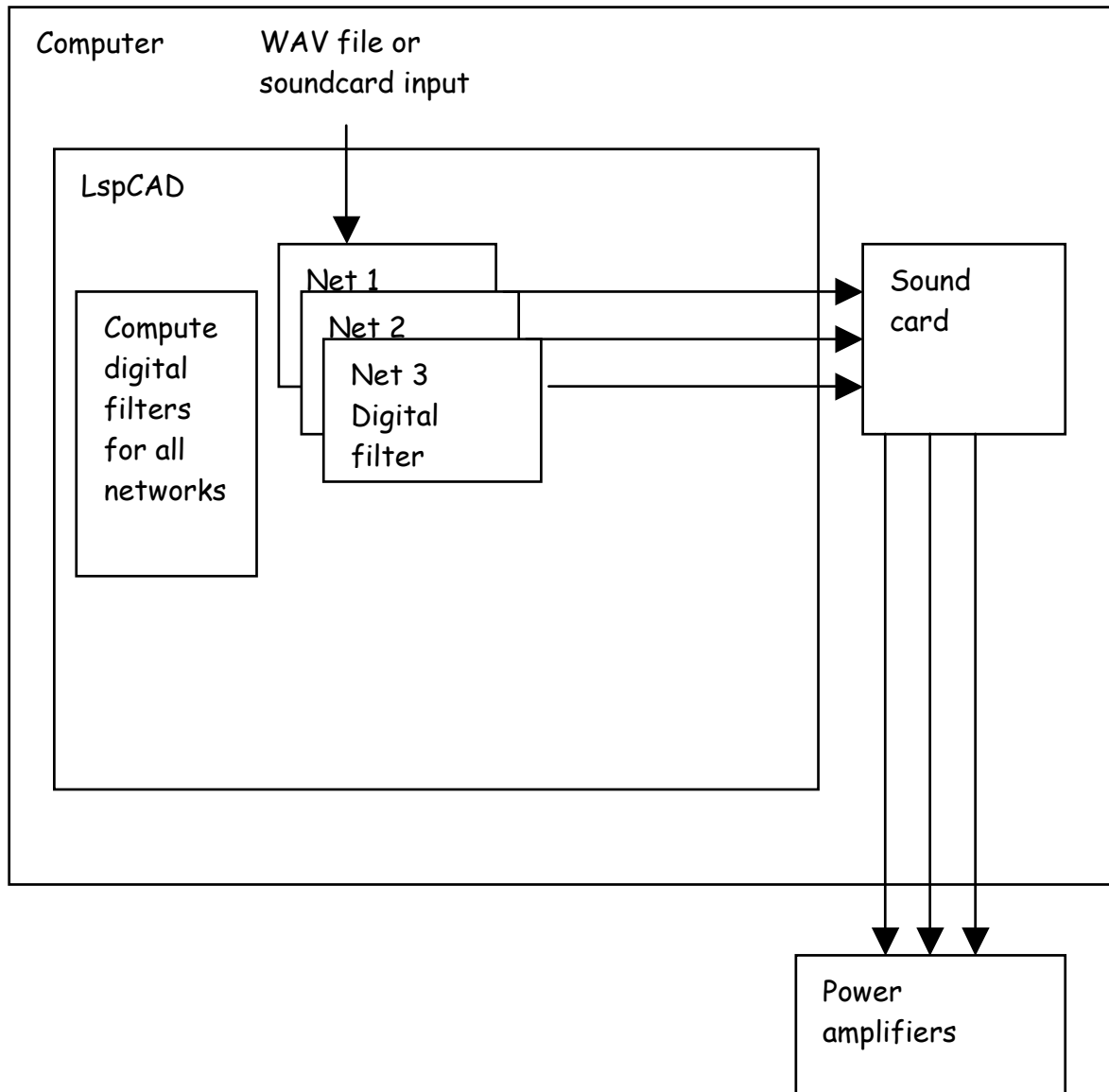


Fig. 7.1 Block diagram showing the role LspCAD has in the crossover emulation work

7.3 How exact is it?

The exactness of this function depends of how well magnitude and phase of the achieved transfer functions of the digital filters match the target transfer function.

Fig. 7.2 shows the magnitude part of the transfer function for the two networks in a two way crossover example..

Note that both the target and achieved transfer function is shown for the two networks and that it is almost impossible tell the difference between them. The difference between target and achieved transfer function for Net 1 is below 0.2dB within the working range. Above 2500Hz the transfer functions for Net 1 starts to diverge but that is no problem as the driver unit itself cuts off those frequencies even more. As a protective capacitor is used for the treble unit the resulting transfer function looks a little bit odd, reason is of course that this protective capacitor is compensated for internally in the emulate function.

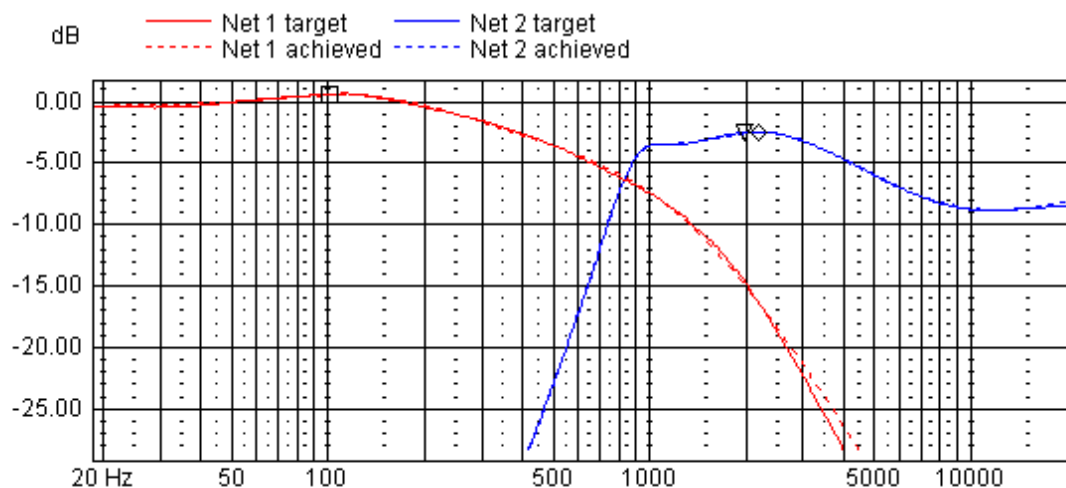


Fig. 7.2 Target and achieved transfer functions (magnitude).

Fig. 7.3 shows the phase part of the transfer functions for the two networks, important here is that the phase is correct within the working range. One may object that the phase is not equal to the phase response of the network especially in the treble region, the

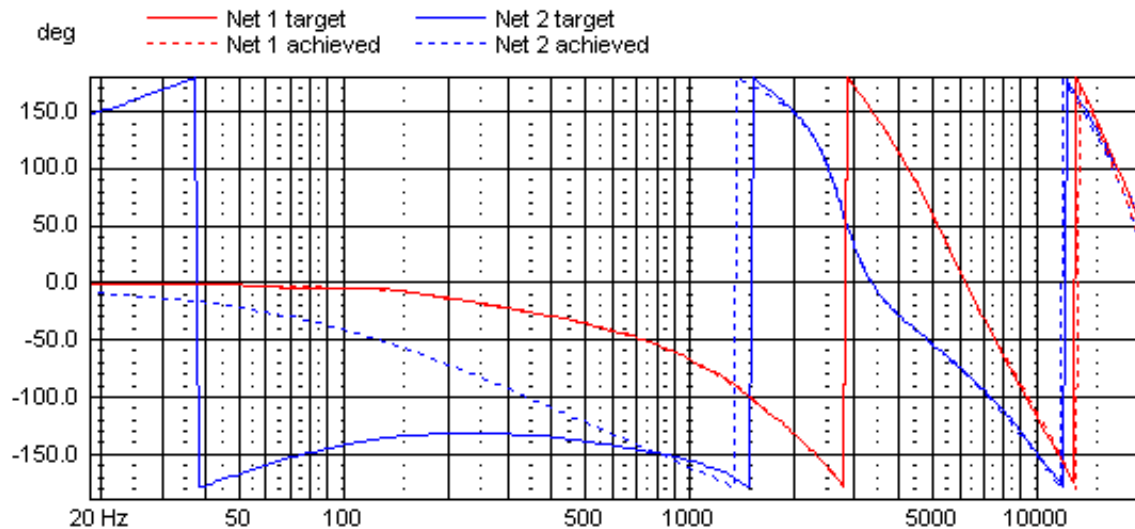


Fig. 7.3 Target and achieved transfer functions (phase)

reason here is that a two sample delay is added to the phase response to make sure that the digital filters are causal, this has no impact on quality as all frequencies will be delayed by two samples

It is very important (for passive networks) that the impedance response is correctly measured and that it is measured over the whole frequency range.

There are two things that are not emulated correctly (for passive networks):

- Temperature dependency of voice coil resistance. At high power levels the voice coil resistance will increase. This will affect the transfer function of a real filter, but not the digital filter that is used by LspCAD. This problem can however be alleviated by modifying the Rex value in the driver dialog, for instance, if one has reason to believe that the voice coil resistance is increased by 1.5Ω at operating power then one simply enters the value 1.5 for Rex.
- Modulation of voice coil inductance (mainly bass units). At high cone excursions the value of the voice coil inductance varies. This will also affect the transfer function of the real filter. One means to limit the impact of this feature is the use of zobel networks.
- Excessive cone excursions. For large cone excursions (bass units) the force factor will drop dramatically when the voice coil leaves its linear excursion limit. This is reflected in the impedance curve of the driver and will of course affect the transfer function of the real passive filter.

Apart from the three exceptions above the filter emulation function will provide a real time saver when it comes to do the final tweaking of loudspeaker systems in order to get the best out of them.

7.4 Users guide

This section describes how to use the emulator function in LspCAD it is highly recommended that you read through this section thoroughly.

7.4.1 System sounds and sounds from other applications

If possible all system sounds should be turned off, also it is recommended to shut down applications such as MS-word/excel that might play sounds. This requirement can be relaxed if a protective capacitor is used for the treble unit.

7.4.2 Sound cards

The emulator function is designed to work with 2- or 4-channel soundcards. Examples of 4-channel (or 3D) soundcards are e.g the Soundblaster Live player 1024. It is important that the driver routines for the soundcard supports interleaved MME, this may be tested with the stand alone 4 channel test program that comes with this distribution. Multi-channel soundcards with synchstart such as the Echo family cards does not have this problems as they are accessed in another way. Presently the soundcards are accessed using the windows MME. ASIO support is not implemented for the moment. Multimedia soundcards such as 5.1ch soundcards with WDM can also be used with the emulator.

7.4.3 Error conditions

Two error conditions may occur during the work.

- The playback starts to hiccup, after a while a Message box will pop up with the text "Too slow playback": The reason is simply that the PC is too slow. You will relatively fast realize what is possible for your PC. A 300MHz Pentium III is enough to make most realizations of a two-way stereo system. Generally speaking a doubled filter order costs twice the CPU power.
- A message box appears with the text "Unstable filter for Net xx". This may have two reasons. The first reason may be that the filter has a very high amplification factor. If this is not the case then it is an abnormal condition that should not occur.

7.4.4 Output configurations (3D soundcards)

The output of the soundcard will be organized according to the tables below.

Table 1 Output configurations for a 2-channel soundcard

Configuration	Channel 1 (left)	Channel 2 (right)
1-way	Net 1 (left)	Net1 (right)
2-way	Net 1 (left)	Net2 (left)

Table 2 Output configurations for a 4-channel sound card

Configuration	Channel 1 (front left)	Channel 2 (front right)	Channel 3 (rear left)	Channel 4 (rear right)
2-way	Net 1 (left)	Net 2(left)	Net 1 (right)	Net 2 (right)
3-way	Net 1 (left)	Net 2(left)	Net 3 (left)	
4-way	Net 1 (left)	Net 2(left)	Net 3 (left)	Net 4 (Left)

Please note that it is not likely, but still possible that the soundcards may interpret the channels differently, in other words, hook up the bass units first and listen.

A last word about 4-channel audio cards, they must be configured for 4-speaker output in order to work. Also it is important that all special effects are turned off.

7.4.5 Output configurations (Multimedia cards with WDM drivers)

If WDM drivers are supplied with the soundcard the channel mapping is selected from the output configuration dialog (see ch 7.4.10).

7.4.6 Output configurations (Multichannel soundcards)

The output of the multi-channel soundcards is organized according to the table below.

Table 3 Output configurations for a multi-channel soundcard

Channel	Network
1	Net 1 (left)
2	Net 2 (left)
3	Net 3 (left)
4	Net 4 (left)
5	Net 1 (right)
6	Net 2 (right)
7	Net 3 (right)
8	Net 4 (right)

7.4.7 Reverse polarity

When working with the emulator, always connect all driver units with correct polarity i.e. always the positive terminal to the positive output of the power amplifier. The emulator will automatically invert the signal for a network if reverse polarity is checked in the network dialog.

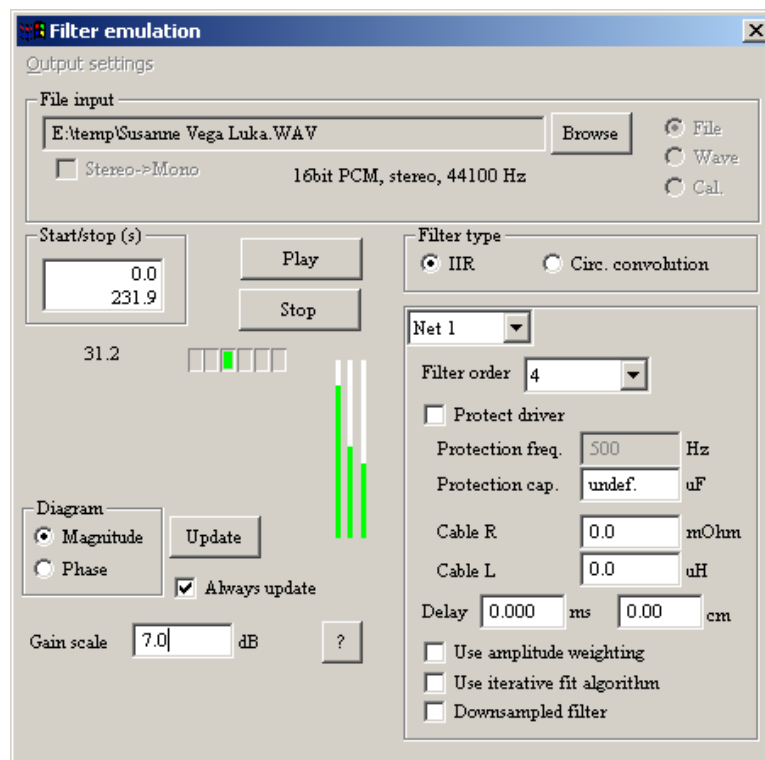
7.4.8 Logfile

For the mathematically interested person LspCAD creates one logfile for each network. These files can be found in the LspCAD directory and are named "Filter coeffs for net1.txt" and so on. Please note that these files are only updated when the digital filters are recalculated, i.e. they are not updated due to a **Network | Recall...** command.

7.4.9 Entities of the dialog box

The dialog box is shown below.

The dialog frame is divided into a number of groups that will be described here.



File input/Wave input

As input it is possible to select either File input or Wave input. This is selected on the File/Wave/Cal. radio buttons.

File input:

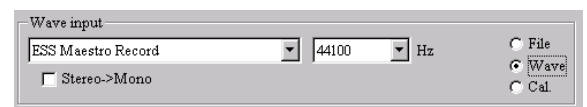
Enter the filename of a WAV file by pressing the **Browse** button (also possible to drop inside the **File input** box). The file should be in Windows WAV format. Also it is required that the file is in stereo 16 bit format. If the **Stereo->Mono** checkbox is checked then the left and right channels are mixed. In the Start/Stop box one can set the endpoints in the input file between which the playback should occur.

Wave input:

If wave input is selected then the dialog changes the looks slightly.

It is possible to select the input device.

Note that the input device must be capable of handling 16bit stereo input of the given sampling rate. If this requirement is not fulfilled an error message is displayed.



Calibrate:

If calibration is selected a 1000Hz tone (-20dBOV) is output to all output channels when the play button is pressed. This can be used to calibrate the output levels.

Playback is started and stopped with the **Play** and **Stop** buttons.

Filter type

The filter type can be either IIR filter or circular convolution.

The **IIR** filter type makes it possible to emulate with very low latency. Moreover the computational complexity is low. The drawback is that one need to set the filter order for each filter in order to get a good match between the digital filter response and the target response and this may be a tedious task in the beginning before one learn the do's and don't's about it.

The **circular convolution** filter type is based on the Fast Fourier Transform (FFT) with an FFT length of 16384 samples. The buffer size needed for this is 8192 samples which means that the latency is higher than the IIR filtering approach described above. The benefit with circular convolution is that it is simple to use, no need to set the filter order or any other special tricks that is the case with IIR filtering. There is a catch however, the filter is not time invariant, the impulse response has a maximum length of 16384 samples at best, in the worst case the length is only 8192 samples. For most applications this should not be any serious problem as the energy of the impulse response most often is very low after 8192 samples. Another problem is frequency resolution. At 44100Hz sample rate the frequency resolution is 2.7Hz. The computational complexity of the circular convolution is about 3-5 times that of the IIR filter type.

Diagram

When the emulate dialog is opened an extra diagram shows up. It is called "Target and achieved transfer functions". On this diagram one can select to see either the **Magnitude** or the **Phase** of the transfer function. In the circular convolution mode the worst case magnitude and phase response is shown.

Update/Always update

A click on the **Update** button makes for an update of all digital filters. If **Always update** is checked an update will be done whenever a single component value is changed, this consumes a lot of processing power, thus the **Always update** checkbox ought to be unchecked on slower computers. It is recommended that you have this checkbox checked.

Gain scale

If the output on one of the channels is too high (clipping) one can set the overall gain here so that clipping can be avoided.

Net x

For each net there exists a group, for two-way systems both networks can be modified. For three- and four-way systems only the settings of one network at a time is visible. There are a number of options to play with here.

If you work with a fragile treble unit it is recommended to check the **Protect driver** checkbox. When this is checked an extra check will be performed so that the magnitude of the transfer function is more than 20dB below the maximum at the **Protection frequency**. Given the transfer function for the treble unit in figure 7.2 an appropriate protection frequency is thus 500Hz.

A physical **Protection capacitor** can be inserted in series with e.g. the treble unit, if this is done then the value of that capacitor must be entered. A zero value in this field will remove the capacitor (undef.). Note that this assumes that that correct impedance data is used.

The **Cable R** and **Cable L** lets the user enter the actual series resistance and inductance values of the cables between the power amplifiers and the loudspeaker units. As is the

case with the protection capacitor the correct impedance data is used, this is especially important if one model and emulate active filters where impedance data is normally not needed.

The **Delay** can be entered either in cm or in ms. This makes it possible to fine-tune the correct placements of the driver units in the axis perpendicular to the baffle plane (the Z-axis). Please note that a change of this value does not affect the dZ value in the Driver dialog and vice versa.

For the IIR filter type a few extra controls exist.

The **Filter order** determines the filter order or the model, generally speaking a sharper slope in the transfer function demands a higher filter order. It is possible to select filter orders between 2 and 24.

Use amplitude weighting adds a weighting function so that the response of the digital filters may be somewhat improved.

Use iterative fit algorithm is almost always needed, the only exception seems to be low-pass transfer functions (for bass units). For band-pass and high-pass transfer functions it is most likely always a must. It is possible that the magnitude transfer function might look good without this option, however this is not the case for the phase transfer function.

Downsampled filter This option can be very helpful when one wish to emulate crossovers for subwoofers.

7.4.10 Output settings

Output settings is accessed by means of the *Output settings* menu pick.

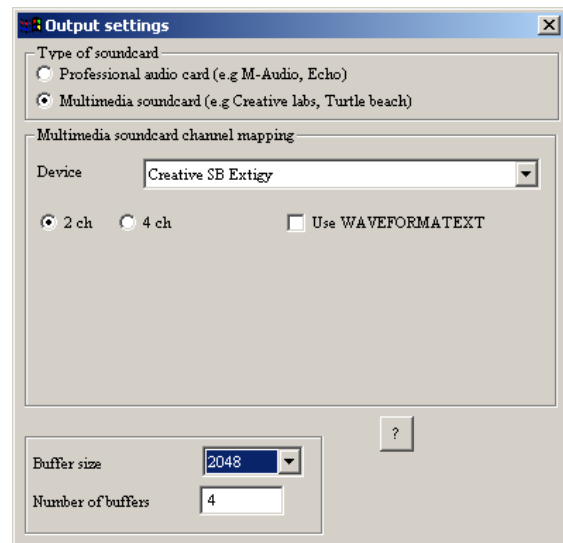
Here the soundcard that should be used for output is selected. One can essentially work in two modes. The number of buffers and also the buffer size is selected so that annoying hiccups is avoided. A rule is that the larger the buffer size and the number of buffers, the larger playback latency. On the other hand the risk of getting hick-up's in the playback is reduced. If circular convolution filter mode is used the buffer size will be locked to 8192 samples.

Soundcard modes

There exist four different ways to access the soundcard for audio output with LspCAD.

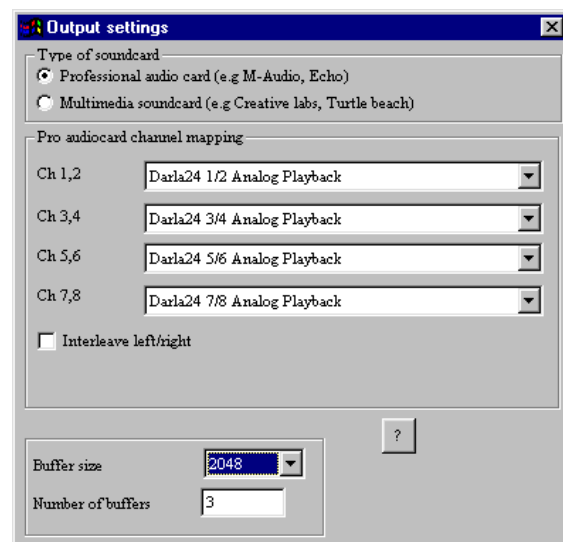
Normal MME soundcard mode

The number of output channels is selected with the **2ch** and **4ch** radiobuttons. Note that 4ch output must be supported by the soundcard. If this is not the case an error message will be displayed.



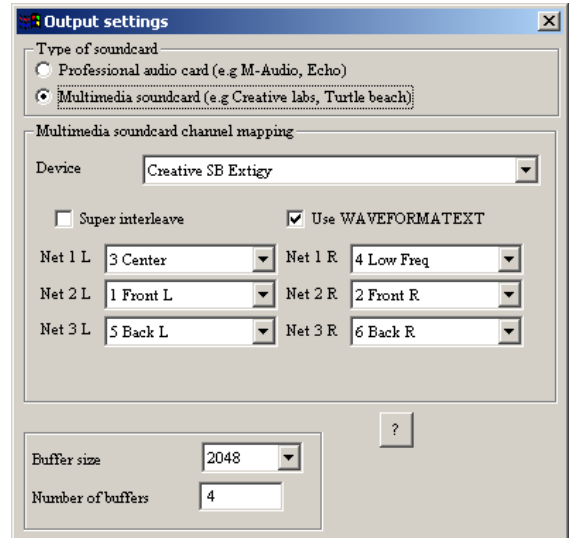
MME synchstart mode

In this mode one have to select the appropriate mapping to the multi-channel soundcard output channels. MME synchstart is supported e.g. by the Echo Audio and Maudio soundcards.

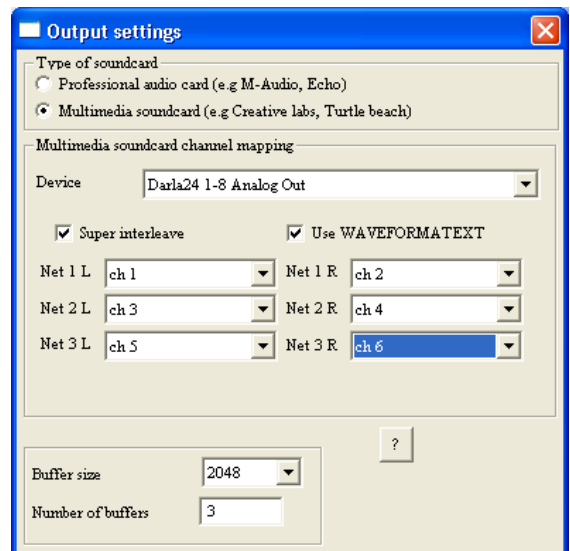


Multimedia WDM mode

In this mode it is possible to use a multimedia soundcard such as the Creative Labs Soundblaster Extigy. Each network output is mapped to a specific output. It is possible to map outputs from several networks to one soundcard output (the outputs will then be added). This mode requires WDM drivers to work.

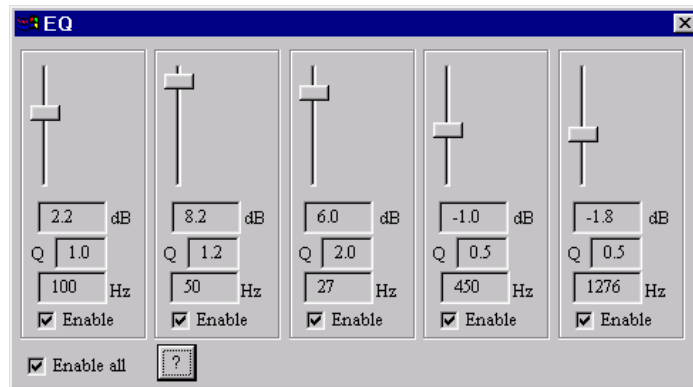
**Super interleave (WDM) mode**

A special mode used by e.g. Echo Audio. Similar to the Multimedia WDM mode the network outputs are mapped to soundcard outputs. This mode requires WDM drivers to work.



7.5 The parametric equalizer

The 5 band parametric equalizer is available through the menu pick *Network / Equalizer* and makes it possible to adjust the frequency response and see immediately how the amplitude response, phase and group-delay is affected, moreover it is possible to hear the result immediately during playback (apart from the playback latency).



Note that this is a digital equalizer (read time discrete), this essentially means that the response will differ from an analog equalizer at high frequencies, this is easily seen if the center frequency is set above 10kHz with 44.1 kHz sampling rate. For the technically interested the filter coefficients are derived from the transfer function of an analog equalizer by bilinear transform with pre-warping so that the accuracy is the highest around the center frequency.

Initially the sampling rate of this equalizer is set to 48Khz, when the emulator is started the filter coefficients are recalculated for the sampling rate that is used.

The settings of this equalizer is not affected by the *Network / Store as / Recall* commands. The handling of this equalizer should not cause any serious problem, each band has its own setting for gain, Q, and center frequency. The enable checkbox enables the specific band.

7.6 Known problems and tricks

This section explains some of the problems that may arise and also the solution to them.

7.6.1 Necessary filter order (IIR filter type)

When the filter order is selected for a network is good practice to not choose too high orders as this may only turn things worse. Instead it is always recommended to work from a low order and up till one does not get a better match anymore. Generally speaking one can say that one order is needed per 6dB/octave slope. For instance a 4th order filter probably is necessary to model a 24dB low-pass slope. A notch will need a 2nd order filter. Passive filters needs a higher filter order than an active filter, the reason is simply that the transfer function of the passive filter is dependent on the impedance of the driver unit.

7.6.2 More than one driver in a network

Generally this is no problem. LspCAD will match the transfer function "measured" across all driver units. If however one plan to make a design with two driver units in series in a network and were one driver is clamped with a capacitor, then this capacitor must be physically inserted across the real driver unit as well. Figure 7.4 gives an example where driver D11 is clamped with a capacitor. This capacitor C1111 must be put across the real driver also.

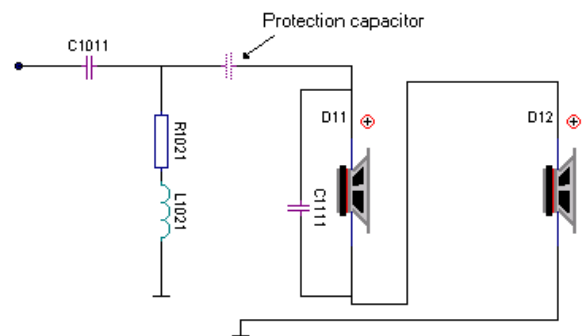


Fig. 7.4 An example of clamped drivers

7.6.3 Poor matching of low-pass filter (IIR filter type)

In some cases the achieved transfer for the bass unit will not match the target transfer function no matter how high filter order it selected. This problem has been observed mostly only for passive filters. The cause of this problem is that the impedance data does not contain data points for low frequencies. For instance one mistake is to only measure the impedance curve between 100Hz and 20000Hz as it is more than enough for crossover modeling in general. It is therefore **very** recommended to measure the impedance of the driver over the whole frequency range, preferably from 20Hz to half the sampling rate.

For filters with very low cutoff frequency (read subwoofers) especially those that also incorporate a highpass filter one can get problems with filter matching. In this case it is beneficial to check the **Downsampled filter** checkbox. One then have to test what setting of the **Use iterative fit algorithm** gives the best results in terms of magnitude and phase matching. It is very important here to check the phase response and not only the magnitude response in this case

7.6.4 Poor matching of high-pass filters (IIR filter type)

As in the case of the low-pass filters one may get into trouble with high-pass filters also. The reason is basically that it is not possible to match an analog filter and a digital filter perfectly in magnitude and phase. The effects of this problem shows up close to half the sampling rate and it is presently not solved. The root of the problem is basically that analog and digital frequencies are not the same at high frequencies, thus it is very likely that one gets problems with this for highpass filters aimed for treble units.

The simple solution provided here is to use a protection capacitor in series with the treble unit (and possibly also midrange unit). LspCAD will compensate for the impact of the capacitor. Besides protecting the driver units, it is likely that the filter matching problems will vanish. For a treble unit one may use e.g. a 2.2 μ F or 4.7 μ F capacitor, while a 10 μ F or 22 μ F capacitor may be better for midrange units. Please note that impedance data is needed also for an active network if the protection capacitor is used. Figure 24 shows where the protection capacitor should be physically placed. Please note that this capacitor is only used with the emulator and should not be used in the final "real" implementation.

7.6.5 Using the sets

If the settings of a crossover are stored away (**Network / Store as...**), all settings of the digital filters will also be stored away. When the settings are retrieved later (**Network / recall...**) the digital filters does not need to be recalculated provided that the it was up to date when it was stored away, this is a very handy feature as it allows for a faster switch between filters. The only latency is then due to the playback. Note that the log files are not updated when **Network / recall...** command is issued. The hot keys **Control-1** through **Control-4** makes it possible to recall the settings 1 through 4 without the need to click on the menu.

7.6.6 Overloading output

..or clipping should be avoided. This is especially important for the output from the highpass filters. The reason is that the distortion components above half the sampling rate will be folded back and will then be audible as aliasing distortion. If overflow occurs a warning indicator will tell how much overload (in dB) there is, adjust the gain scaling so that overflow does not occur.

8 Other important issues

8.1 Buzz words..

Below is a short description of some strange words that show up every now and then.

- **Isobaric :**

Isobaric or compound mounting is when two loudspeaker units are mounted face to face as in fig. 8.1. Note that the loudspeaker units may also be mounted face to back. These two loudspeaker units form a driver unit. The benefit of this is that the box volume may be halved, the drawback is that the efficiency in terms of dB/W is decreased.

It's important that the mounting doesn't leak and that the volume in the cavity between the loudspeaker units is as small as possible.

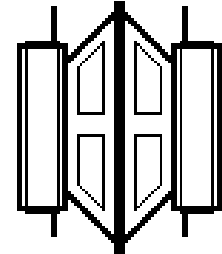


Fig. 8.1 Isobarik

- **Loudspeaker unit :**

An ordinary loudspeaker unit.

- **Driver unit :**

A conception that is used in this program just to avoid confusion when isobaric mounting is used.

If isobaric mounting is **not** used then a *Driver unit* is synonymous to a *Loudspeaker unit*.

If isobaric mounting is used then two *Loudspeaker units* form a *Driver unit*

- **Diffraction:**

A phenomenon that is best described that sound waves that encounter an obstacle are dispersed. This effect leads to that the sound pressure will increase with 3-6 dB between 100 Hz and 1000 Hz for normal sized boxes. If the box has sharp edges it will also cause a more disturbed frequency response than should the edges be rounded.

- **Organ pipe resonances:**

Standing wave modes in a port occur when the length of the port is a multiple of a half wavelength. For example if we have a 20 cm long port, then we will get standing wave modes at 850 Hz, 1700 Hz, 2550 Hz and so on. The effect of the above may be remedied somewhat with the use of crossovers or a dense cloth that obscures the port from the driver unit(s), the cloth must not, however obstruct the airflow in the box. As the problem with this phenomenon is more serious for the bandpass box, a separate diagram shows the frequency response with organ pipe resonances for the latter.

- **Flared ports**

Flared ports are often used for circumstances where a high sound pressure level is desired in the lower bass region. Flared ports allow for more output in the bass region. A study by A. Salvatti et. al. has covered this topic quite extensively (see literature list). A general recommendation is to have the flare radius about equal to the port length. Also is recommended that the rear part of the port is terminated by a small baffle, this will make the flow resistance more equal regardless of the direction of the air in the port

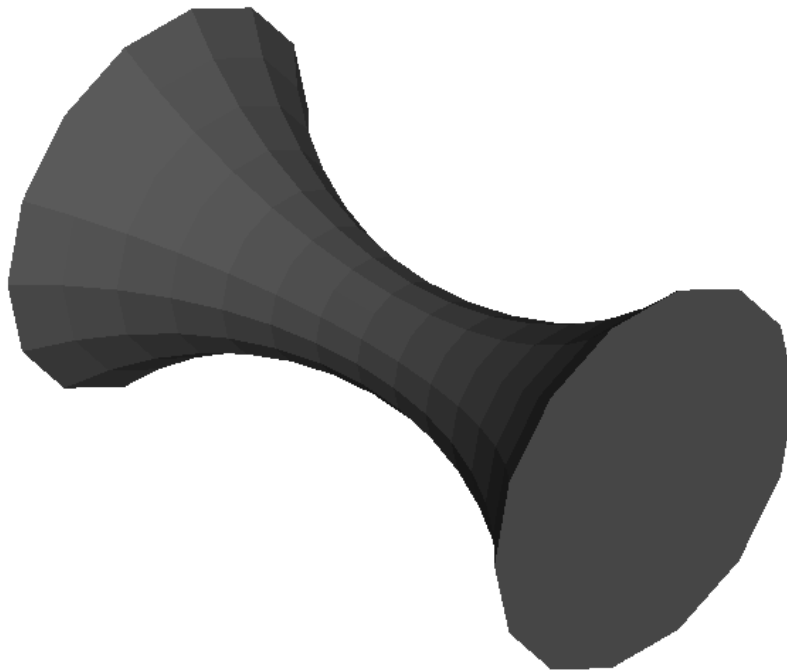


Fig. 8.2 A flared port

8.2 Export/import file formats

8.2.1 Export file formats

The file formats that can be exported are

LAUD (.frd),

Frequency response data, no header exist for this format.

Each line is of the form

```
< freq >                < dB >                < phase >
```

LAUD (.zma)

Impedance data, no header exist for this format.

Each line is of the form

```
< freq >                < mag >                < phase >
```

LEAP (.gdt)

A file exported in LEAP format looks something like:

- LEAP Version 4.00, Export Graph Data table (GDT) Format
- Mon Dec 01 12:58:00 1997
- DGL Library Name : LspCAD
- DGL Entry Number : 1, Name : ---
- Number of Points : 500
- Graph 1 > Acoustic on Axis Free Air Response
- Hz dB deg
- 10.000000 40.656863 -63.670400
- 10.139395 41.104935 -64.348630
- 10.280732 41.552318 -65.034261
- ..
- ..
- End of LEAP GDT File.

MLSSA (.fmp)

MLSSA (.fmp) has a one line header

```
Acoustic on Axis Free Air Response
      Hz                dB                deg
10.000000            40.656863           -63.670400
10.139395            41.104935           -64.348630
10.280732            41.552318           -65.034261
```

MLSSA Standard (.fm)

MLSSA (.fm) has a one line header.

```
Acoustic on Axis Free Air Response
      Hz                dB
10.000000            40.656863
10.139395            41.104935
10.280732            41.552318
```

If there is no use for the third column e.g. when exporting group delay data the elements of the third column are set to zero.

8.2.2 Import file formats

LspCAD can import files generated by LAUD, Praxis, IMP, CLIO, LEAP, LMS, DLSAPro, LspLAB, justMLS and MLSSA, the format is assumed to be of the format

```
< freq >                < dB >                < phase >
```

For the frequency resonance. As an example the the line

```
100                86.2                178.2
```

is valid.

The impedance data files are assume to be of the format

```
< freq >                < mag >                < phase >
```

If a line starts with a non-numerical word that line is skipped and the import procedure continues until end of file is reached.



Important to remember is that when files are imported into LspCAD it is only the reference to the file (i.e the filename) that is stored in the project file. In other words if you remove the files that you have imported the project file will not open successfully.

9 A small tutorial

This section is a very small tutorial about different aspects of loudspeaker construction. As explained before the intention is not to explain everything. Instead the idea is to only to give a brief introduction to the fantastic world of loudspeaker building. Also the level of this tutorial demands that you are not totally new to this subject.

9.1 Loudspeaker boxes

The figure below shows the frequency response when the PEERLESS 6.5" driver unit 832732.unt available in the examples subdirectory is put into different boxes. In all cases the table lookup optimization in LspCAD has been used.

The resulting total volumes are quite moderate except for bandpass type 3 where the total volume is 60l, something that definitely is not acceptable for a 6.5" driver unit. On the other hand one can see that this box extends the frequency response to very low frequencies, it must be stated here also that the transient response characteristics of this box are very poor. A compromise is the bandpass type 1 box. This box extends the frequency response a bit, however the efficiency is lower as a relatively wide bandwidth (35-150Hz) was selected.

9.1.1 Transient response characteristics

The difference between the closed box and the bassreflex box is that the bassreflex box can reproduce lower frequencies at the cost of a sharper cutoff slope. As the closed

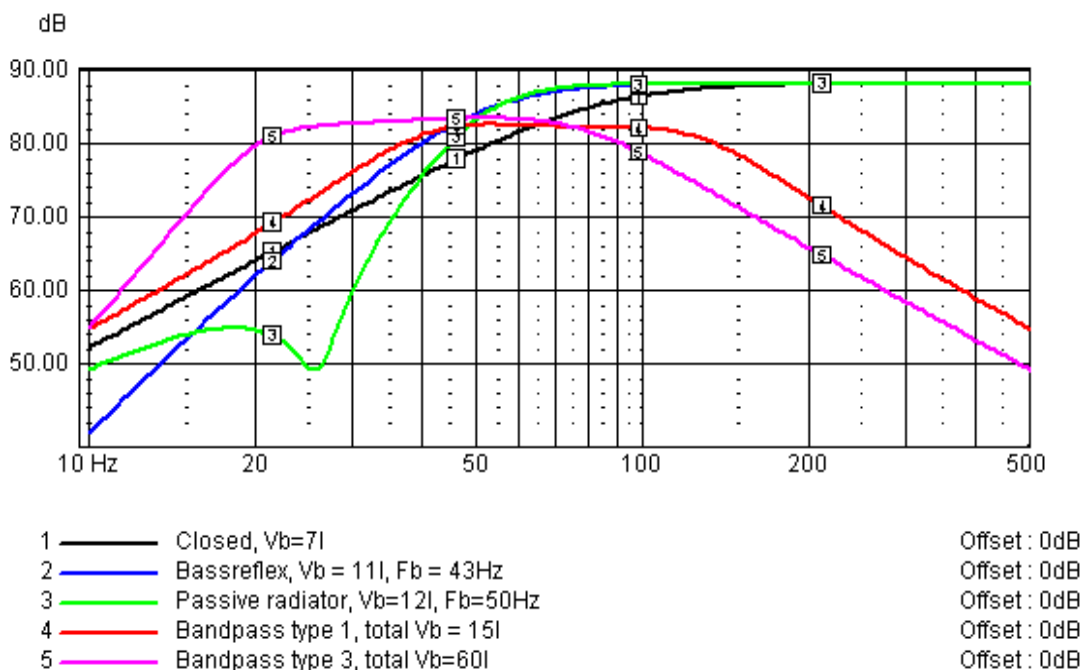


Fig. 9.1 A Peerless 832732 driver unit in different boxes

box is a 2nd order system the slope below the cutoff frequency is 12dB/octave. The bassreflex box is a 4th order system, as a consequence the slope in this case is 24dB/octave. Consequently as the bassreflex box is a higher order system the transient response characteristics are worse than for a closed box.

The passive radiator (or ABR) box has an interesting notch in the frequency response below the cutoff frequency. This notch appears at the free air resonance frequency (F_p) of the passive radiator unit, in this case 26Hz. The passive radiator box is a 6th order system (due to the notch), thus one can expect the passive radiator box to have worse transient response characteristics. In figure 9.2 one can clearly see that this is indeed the case. The passive radiator box shows a considerable longer tail after the toneburst is interrupted.

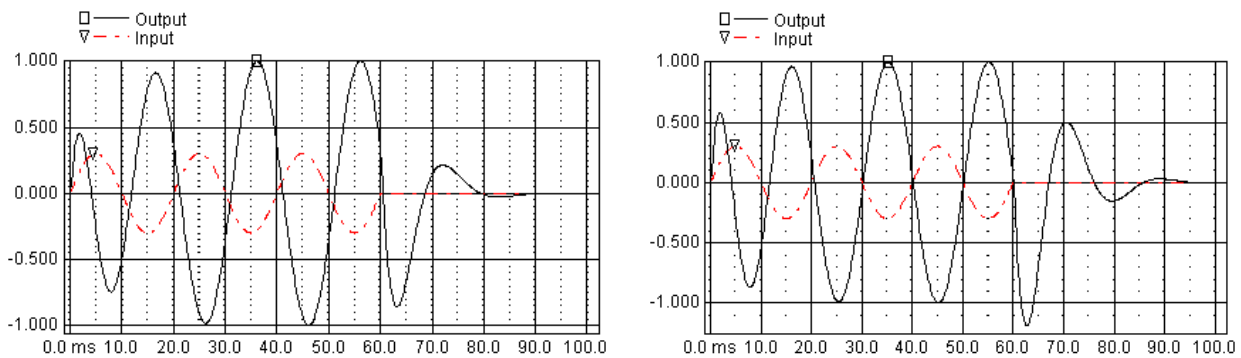


Fig. 9.2 Tone burst response for a bassreflex and a passive radiator box, input signal is shown dashed. Frequency 50Hz, 3 periods.

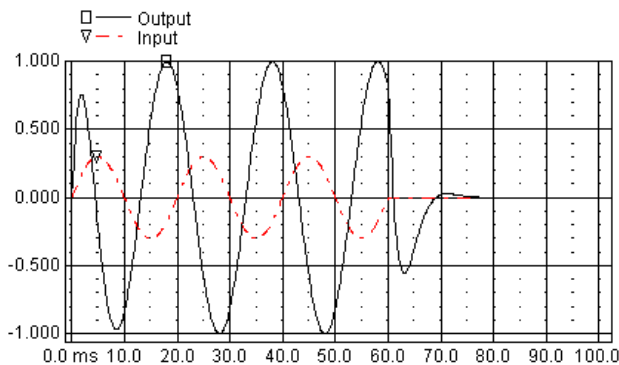


Fig. 9.3 Tone burst response for a closed box.

Speaking in general one can say that the higher the system order, the worse the transient response. With this at hand we can make a list of transient response quality with closed box being the best:

- 2nd order systems
Closed box
- 4th order systems
Bass reflex, band pass type 1

- 6th order systems
Passive radiator, all other bandpass boxes, double tuned bass reflex.

One interesting thing is that a bass reflex box with a bump in the frequency response may in some cases show a better step response than a bass reflex box with a flat response. A tone burst should be built up fast and die off fast. A closed box builds up and stops this tone burst very quickly while e.g. a bandpass box type 3 may show a very slow behavior.

How does a good transient response sound ?

The answer may be that a good transient response should not be heard at all. The major problem with boxes with poor transient properties is that the punch in the transients is smoothed up. The ringing after the transients are not an equally big problem as the ear is not as sensitive to this due to temporal masking effects in the auditory system.

9.1.2 Power handling

Another aspect, equally important when the type of box is selected is the power handling. One can compare the cone excursion between a closed box and a bassreflex box. While the cone excursion for a closed box increases up to a certain limit as frequency gets lower the bassreflex box shows a notch in cone excursion at the port resonance frequency. This has implications on both power handling and distortion and makes the bassreflex box able of handling higher power levels than a closed box. The above statement applies only to steady state sinewave signals however. If a toneburst signal of the same frequency as the port resonance is applied to a bass reflex box, the cone excursion will initially be quite high and will reduce to a small value after one half to a couple of periods. This is due to the fact that the port-box system can be viewed as a spring with a weight connected to it. At first we need quite an effort to get things going, but after a while the system will oscillate by itself. With this in respect we realize that for transient rich bass sounds the cone excursion is initially equal to or sometimes greater than for a closed box!. Also if the bassreflexbox is subject to inputs with frequencies lower than the port resonance (e.g. rumble from turntables) the driver is very easily overloaded with large cone excursions even at low power levels.

Fig. 9.4 below shows the cone excursion for the closed and the bassreflex box, in an ideal situation the cone excursion would increase indefinitely for the bassreflex box as the frequency decreases, this is not the case in fig. 9.4 The reason is that nonlinear simulation is turned on. In this case it is the progressive stiffness of the driver unit that limits the cone excursion.

One drawback with ported boxes are that the vents are prone to producing standing wave

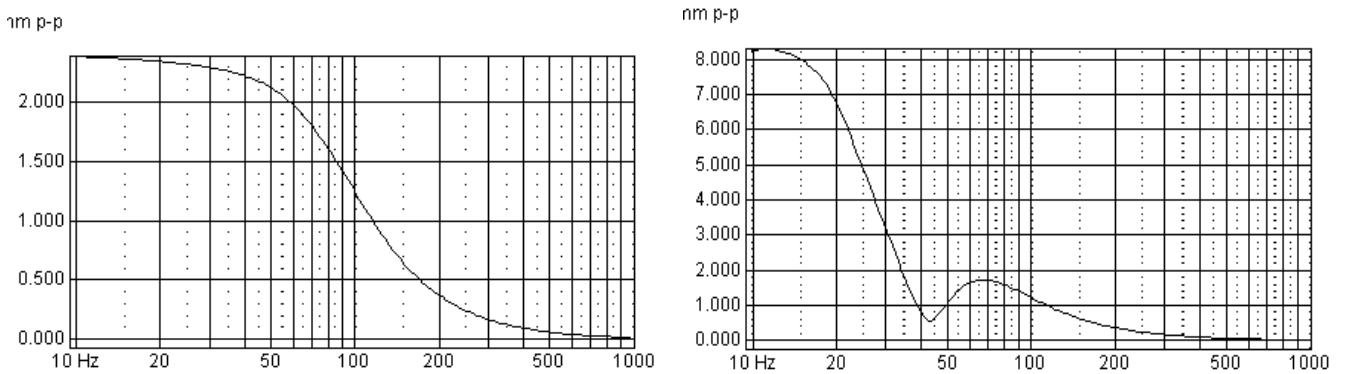


Fig. 9.4 Cone excursion for the closed and the bassreflex box

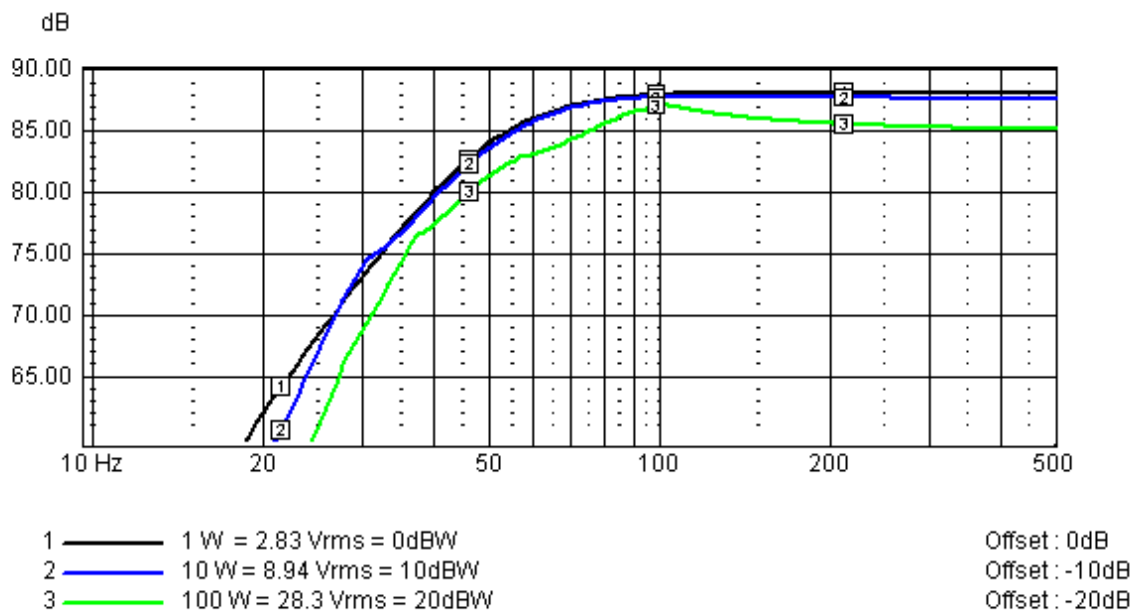


Fig. 9.5. Sound pressure level for different input levels, note that the curves are offset by a value corresponding to the input level. The level for curve (3) is actually 20dB higher than the reading.

resonances and also chuffing noise at high power levels. Chuffing noise at working levels is solved by increasing the area of the port. A rule of thumb is that the port air speed should not exceed 15m/s.

Interesting thing is the handling of high power levels. Fig. 9.5 shows the sound pressure level for different input levels for the bassreflex box described above. The curves are normalized, i.e. the reading for curve (3) is actually 20dB higher.

From the figure one can see that at 100W input the output level is compressed about 3dB at higher frequencies. This compression is due to the heating of the voice coil. At lower frequencies the compression is about 2-5dB and depends also on the non linearity in force factor and compliance. Note that this is only a model that depends on the values X_{max} and P_{max} given in the driver unit editor.

9.1.3 Some construction tips

Simulation and modeling is one side of the coin, in the end one also need to construct the boxes. For the cabinet or enclosure it is very good to use Medium Density Fiberboard (MDF). MDF is easy to work with and has very good acoustic properties.

For the front baffle it is good to use double 19 mm board for medium sized boxes. Remember though to cut the loudspeaker hole of the inner board bigger than the mounting hole on the outer board. This will limit the risk that the mounting hole forms a long tube that may cause resonances.

It's generally better to use a high and narrow box rather than a square shaped box as the medium and high frequency range is better reproduced if the driver units of the latter is higher above the floor. Of course one does not need to follow this advice if the loudspeaker is mounted on speaker stands.

The most common damping material is fiberglass. The main purpose of damping material is to limit standing wave mode resonances in the box which occur when one inner size of the box is equal to a multiple of a half wavelength. For example if the inner height of a box is 1m then one gets the first mode as low as 170 Hz !. This mode is actually best dampened out with heavy stuffing of damping material halfway up the box as the particle velocity is highest at this point for this resonance mode. Stuffing material along the inner walls seldom does much for the damping of low frequency standing wave modes inside a box.

9.1.4 Cabinet edge diffraction

One property that almost always shows up when a driver unit is mounted in a box is something called cabinet edge diffraction. Without going too deep into the theory about this effect it can be said that the results of it is that the sound pressure level increases 6dB from low to high frequencies. This is referred to as the baffle diffraction step and there is no way of avoiding this effect other than mounting the driver units into the walls of the living room. The worst placement for a driver unit is actually in the middle of the baffle. Fig. 9.6 shows the baffle diffraction step for a worst case and a best case placement of a 6.5" driver in a normal-sized baffle. It is obvious that one can gain a lot by placing the driver unit correctly. This experiment can be performed by means of the room/cabinet function in LspCAD.

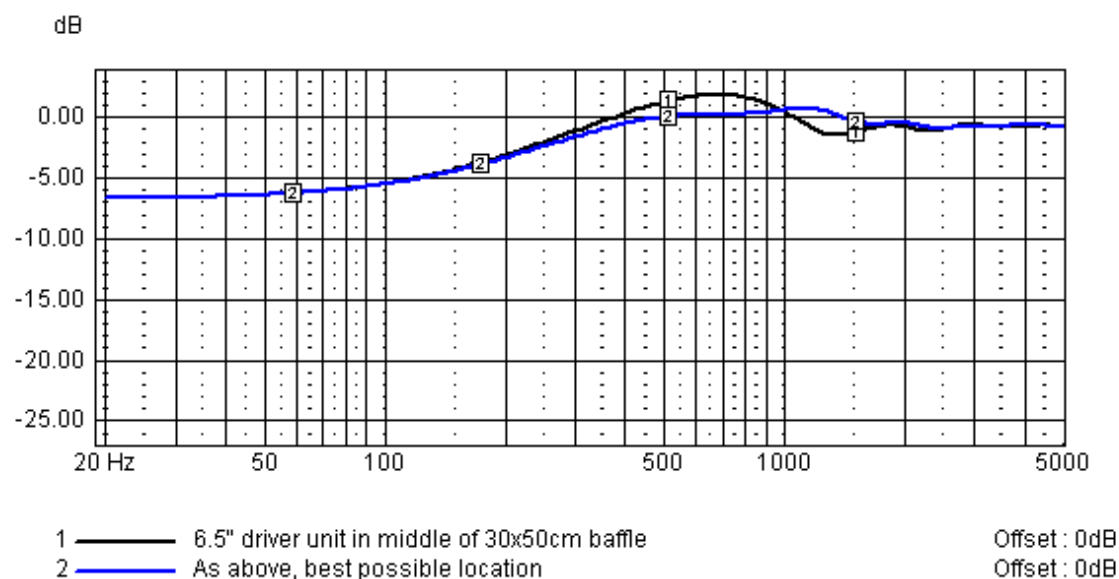


Fig. 9.6 Baffle diffraction step for a 6.5" driver unit mounted on a 30x50cm baffle, both worst case and best possible case placement is shown.

9.1.5 Room impact

A loudspeaker is intended to be placed in a listening room, and this fact may be good to take into consideration. Fig. 9.7 shows the simulated frequency response of a loudspeaker "measured" in free field (far away from any boundaries) and when placed in a living room. The most noticeable effect is that the low frequencies are amplified about 10-15dB. Does it then sound good if one strive for a flat response in a living room. Too often the answer

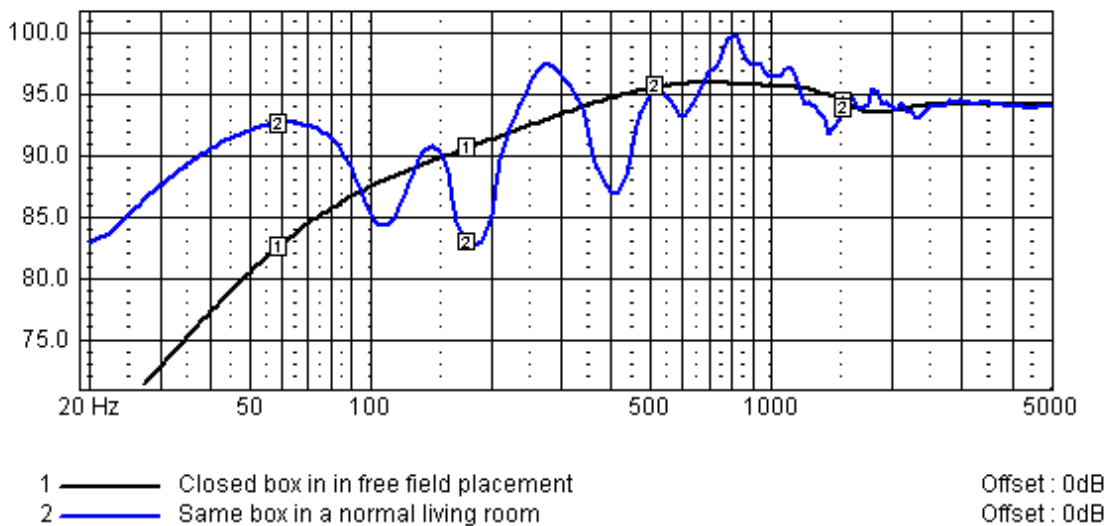


Fig. 9.7 Simulated loudspeaker response, both free air and in living room.

is, no!. I (the author of this text) have the feeling that, for much modern music the producers seem to be aware of this "room transfer function" and they consequently mix the music according to this. Most likely however classic music or live recordings of Jazz music is reproduced better if the room is taken into account when the loudspeakers are constructed.

9.2 Crossover networks

In the beginning there was the "one crossover for any combination of loudspeakers". At least this was what the DIY (do it yourselves) had to resort to when a computer was something that almost nobody could afford. With computers came measurement systems that made it possible to import measured data to crossover design programs and thus the path was open for DIY's to construct crossovers for loudspeaker. Before we continue, we may as well state that it is necessary to have a measurement system so that one can measure the driver units of a loudspeaker separately, preferably mounted in the box that they are intended for. In the below text it is assumed that the advanced active or passive crossover utilities are used.

9.2.1 Filter order and alignment

When selecting filter order and alignment one must bear in mind that the individual driver units of a loudspeaker sum up in a way that the total frequency response is as flat as possible (not always true). Important is that the driver units overlap well at the crossover point. In many crossover software one can select from a great number of alignments, Chebyshev, Butterworth, Linkwitz-Riley, Bessel, Elliptic with their own magnitude and phase properties just to mention a few. What is easily forgotten is that the individual driver units has its own characteristics. Fig. 9.8 shows the magnitude and phase response of a treble unit mounted in a box. One can see that not only is the magnitude variant but also is the phase.

Many crossover design software tend to ignore this fact, and that is like constructing a

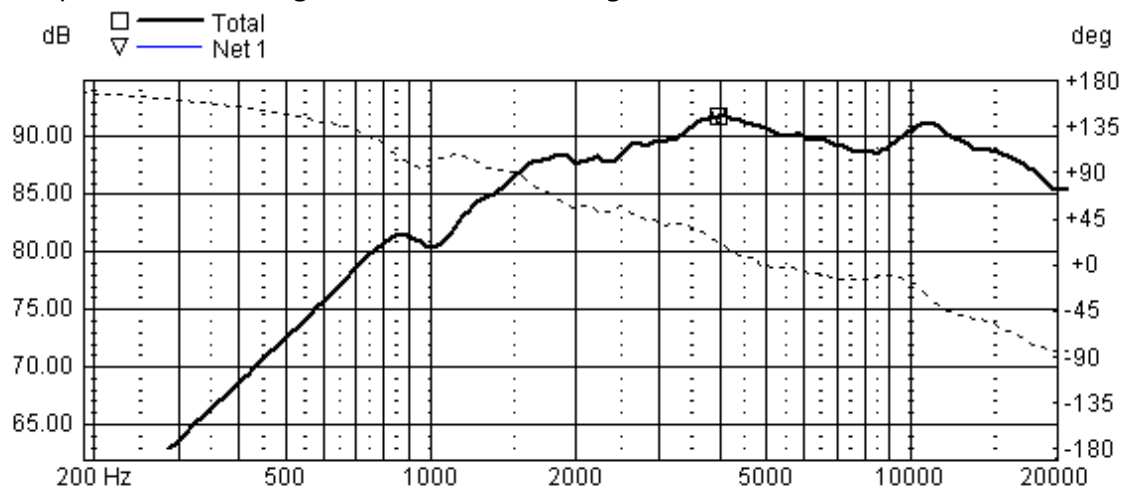


Fig. 9.8 Magnitude (solid) and phase (dashed) of treble unit Dynaudio T-330D mounted in a box

car without steering wheel assuming that the road is straight with no curves!. The conclusion is thus that the important thing is not the response of the crossover, what is important is that the driver units overlap. Fig. 9.9 shows the frequency response of the crossover network and the driver unit+crossover when LspCAD has optimized it for a 4th order Linkwitz-Riley alignment with cutoff frequency 1500Hz

Note that while the total frequency response (driver unit+crossover) follows the Linkwitz-Riley target (-6dB and almost -180degree phase at 1500Hz) the crossover

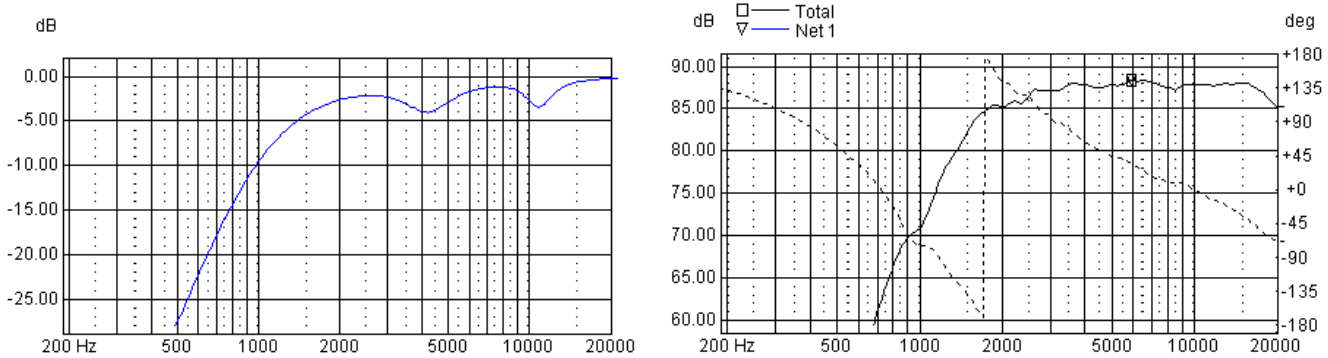


Fig. 9.9 Frequency response of crossover and crossover+driver unit.

network definitely does not do it. The above solution was realized using an active crossover, this means that the impedance data is not needed as the driver unit is fed by a low impedance power amplifier. Should a passive crossover have been used, then of course impedance data would have been needed also. Conclusion is then that driver unit data is needed, period.

As the driver unit characteristics is an important part of the total result it is now stated that when referring to the term slope order in the optimization phase (and in the text below) is meant the order of the total slope of the driver+crossover. It may well be the case that it order to realize a 4th order slope one may settle for a 2nd order filter as the driver units response display a 2nd order slope itself.

The optimizer in LspCAD can achieve both Linkwitz-Riley and Butterworth alignment of order 1-8. Also it is possible to achieve "flat groupdelay" target that resembles Bessel alignment. The phase properties of the different alignments and orders are shown in the table below. The Linkwitz-Riley alignment is -6dB down at the crossover frequency and the Butterworth alignment is 3dB down at the crossover frequency then steepness of the slope is "6dB per order" i.e. a 3rd order slope has a steepness of 18dB/octave. The phase-lag at the crossover frequency is dependent on the order according to the relation

phase = N*45 degrees,

where N is the order. For example a 2nd order slope has a phase-lag of 90degrees at the crossover frequency.

One cannot combine orders and alignments freely. It is recommended to use the same slope order on both sides of the crossover frequency. Note that it is possible to use different slope orders but it is more complicated to succeed with this and not

recommended for the beginner. As the Butterworth alignment only is 3dB down, it is demanded that the difference in phase between the highpass and lowpass is 90degrees in order to get a flat response. In case the Linkwitz-Riley alignment is selected the difference in phase should be either 0 or 180 degrees. The table below displays recommended alignments for different orders. In some cases one of the loudspeaker terminals must also be inverted.

Order	Alignment	Invert
1	Butterworth	No
2	Linkwitz-Riley	Yes
3	Butterworth	No
4	Linkwitz-Riley	No
5	Butterworth	No
6	Linkwitz-Riley	Yes
7	Butterworth	No
8	Linkwitz-Riley	No

9.2.2 Measuring driver units

Measuring the driver units is a bit complicated, especially for DIY's with limited access to anechoic rooms. Luckily enough many loudspeaker measurement systems makes it possible to measure either with gated measurements or with time windows.

FFT based systems such as LAUD, CLIO, LspLAB or ETF makes it possible to get the absolute phase. In other systems such as LMS the phase response is extracted from the magnitude data by means of minimum phase (Hilbert) transformation.

In order to make for accurate modeling in LspCAD (and other crossover modeling software for that matter), the following method is recommended to use.

1. Start with measuring the impedance of the individual driver units, note that both magnitude and phase is needed in LspCAD.
2. When measuring the frequency response of the driver units. The microphone must be placed at a distance from the driver unit, directly on axis. This is especially important for midrange and treble units as the near field and far field response are very different due to cabinet edge diffraction effects. It is recommended to use a distance of about 1m, in some cases one can use a distance of 60-70cm. It is recommended to use a microphone stand so that the microphone can be raised and lowered so that it is always in front of the driver unit being tested. Remember to keep the distance between the microphone and the **baffle plane** constant throughout the measurements.
3. If FFT based systems are used, the phase is already available, remember though that the start of the analysis window must be kept constant. It is good practice to put the start of the analysis window so that the phase-lag that depends on the distance is removed. If minimum phase extraction is to be used, remember to do tail correction before.

4. The measured frequency response must be verified in some way. A method commonly used is to do a final measurement with all driver units connected together and the microphone placed in front of the treble unit (at some distance).

The measured response will look awful but don't let that scare you off. Create an advanced passive crossover and import both the measured frequency response and impedance data. Remember also to set the positions of all the drivers to the positions relative to the location of the treble unit, let $dZ=0$ for all driver units. This of course means that the position of the treble unit is $dX=dY=dZ=0$. Moreover set the position the microphone, that is the distance between the microphone and the baffle.

Now take a look at the *Summed frequency response* diagram, does it look similar to the real measurement. If yes then we can continue with the crossover construction. If no, then we have to set the dZ values for some of the driver units. This should not be needed if absolute phase measurements from an FFT based measurement system is in use. If minimum phase data is used (e.g. LMS data) then we need to put a dZ value of about 2-3cm for the bass units. The simple reason is that the sound from a bass unit mainly comes from the middle of the driver unit at higher frequencies, therefore this sound will be delayed a little bit extra due to the cone shape. Often it is possible to get dZ by measuring the depth of the driver cone i.e how recessed the dust cap is relative to the rim of the cone.

Afraid of destroying your treble units with this exercise? Put a capacitor in series with the treble unit, remember though to add the capacitor in LspCAD as well.

9.2.3 Constructing zobel networks

The first thing to do (probably), before starting with the real crossover work, is to make the impedance curve more apt for the rest of the crossover, this is not needed for active crossovers. The way to deal with the series inductance L_e that cause the impedance to increase at higher frequencies is by means of a zobel network, i.e. a resistor and a capacitor in series. It can be stated that it is not always necessary to use zobel networks, but it is experienced that the network optimizer (at least in LspCAD) performs better when a zobel is used. The best way to get an optimal zobel network is to create an advanced passive crossover in LspCAD and import the impedance data. The network dialog for one of the networks is opened and Branch 11 is changed to an R1+C type. As starting value for R1 is selected a value that is about 20% higher than the minimum impedance before impedance correction.

After these initial values it is only to trial and error change the values of both R1 and C. It is good practice at this time to use the arrow keys and change the values while looking at the impedance diagram. This manual method is quite fast, about 2 minutes later one have a good zobel network. Note that it is possible that during the optimize phase one can change the values of the zobel network in order to improve the total frequency response.

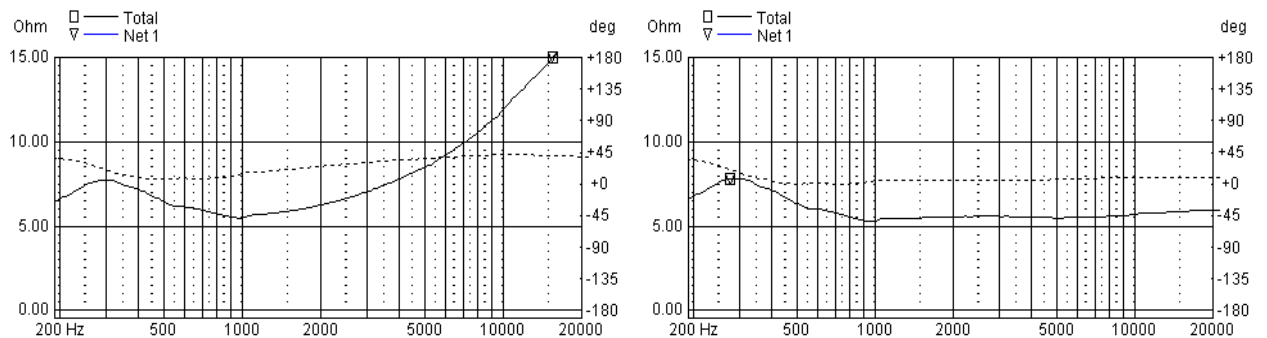


Fig. 9.10 Impedance curve for a midrange unit before (left) and after (right) impedance correction.

9.2.4 Constructing the rest of the crossover

After the zobel networks are created it is only to start with the rest of the crossover network. Below is the frequency response of the bass and treble unit for a commercial loudspeaker.

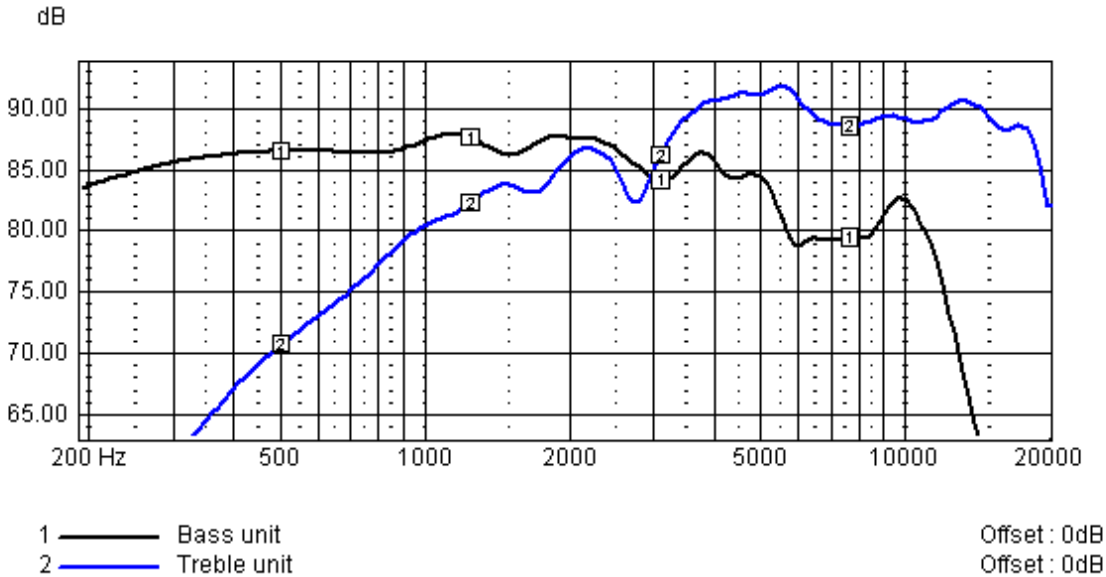


Fig. 9.11 Frequency response of bass and treble unit

The goal is here to create a crossover that makes for an as flat response as possible in the in the working range. From the frequency response one can see that the optimal crossover point is set at about 4000Hz, this may change during the optimization process.

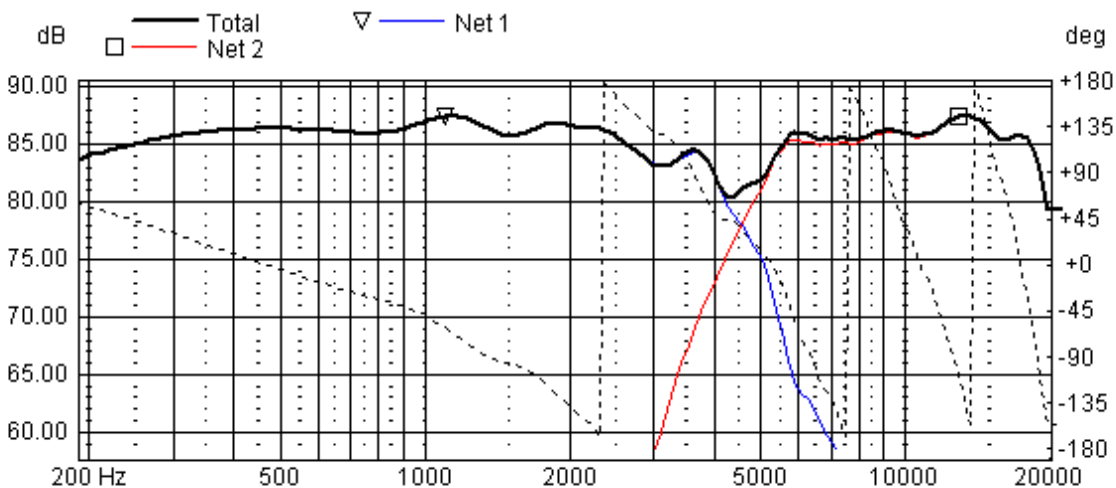


Fig. 9.12 Frequency response of system when starting values from wizard has been retrieved.

For convenience a 4th order alignment seems to be good. Also a series resistor for the treble unit seems to me needed. The network dialogs are opened and the wizard is asked for appropriate starting values.

Fig. 9.12 shows the results after the wizard has done its work. The results are quite OK, but not perfect. The next step is to start the optimizer.

In the optimizer the desired alignment and slope order is entered. In this case a 4th order slope and Linkwitz-Riley alignment was selected. Also it is important to select the include range so that the error metric of the optimization is only computed over the frequency response of interest. Consequently the include range for the bass network is set to 300-7000Hz and the include range for the treble unit is set to 2500-18000Hz.

In fig. 9.13 the resulting frequency response and impedance curve is shown. The results are quite good. One thing that is important to observe though is that the impedance does not get too low. In this particular case the impedance got very low when the optimization was allowed to run, in particular it was the crossover for the treble unit that caused the

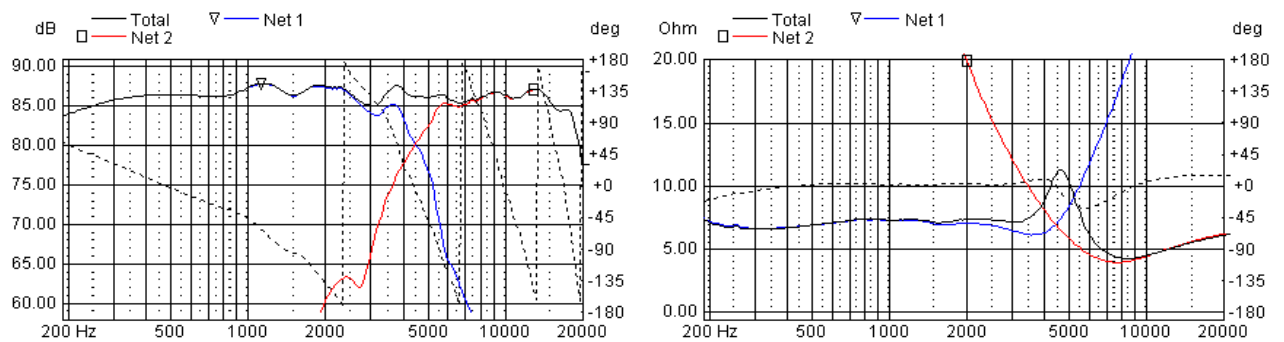


Fig. 9.13 Frequency response and impedance curve after optimization and some additional tweaking .

problems. Therefore some additional manual tweaking was needed in order to get rid of these problems. These impedance problems are quite common when parallel topology networks are used.

Last but not least important, some words about power handling. In fig. 9.14 is shown two frequency responses, one for low power levels and one for high power levels. The latter is

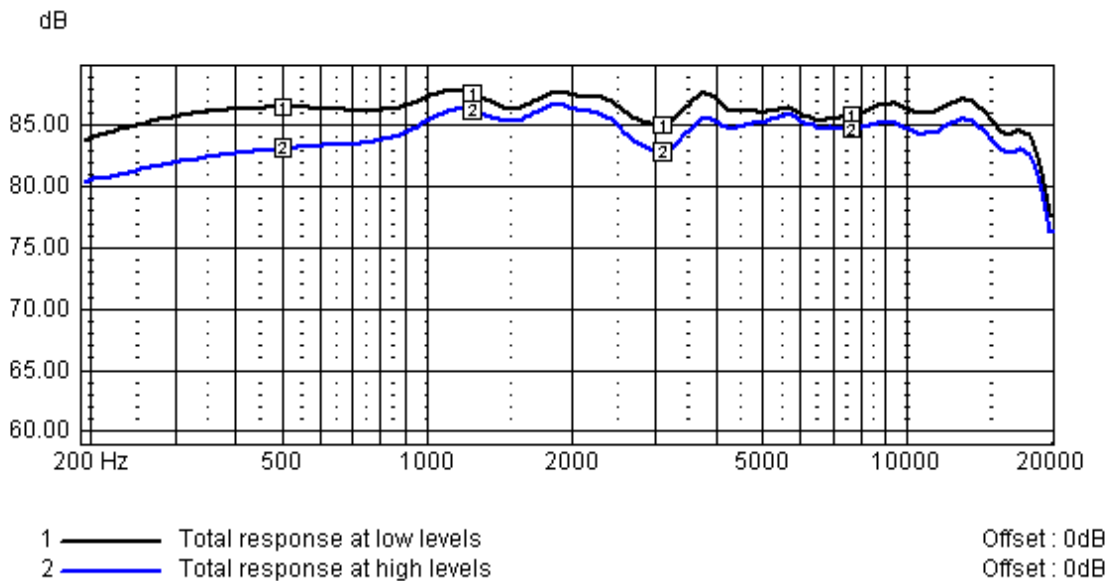


Fig. 9.14 Simulated (normalized) frequency response for low and high power levels.

achieved in LspCAD by setting a value other than zero in the Rex field in the driver unit dialogs. For the bass unit an increase of the voice coil resistance of 50% was simulated, while a 25% increase was simulated for the treble unit. The result that can be observed is that the response around 6000Hz is raised. This may give a yelling character at high sound pressure levels. If the loudspeakers are intended for PA use, then it is recommended to do the optimization for the working levels and not for low levels. This is the end of this tutorial, much longer than originally intended, hope this helps you as a beginner to get started with the loudspeaker building.

9.2.5 System optimization

The system optimizer is often an efficient tool to achieve a flat or a predefined frequency response. Fig. 9.15 displays the frequency response and impedance of a 3way system after the wizard has been used and L-pads have been inserted for the mid and

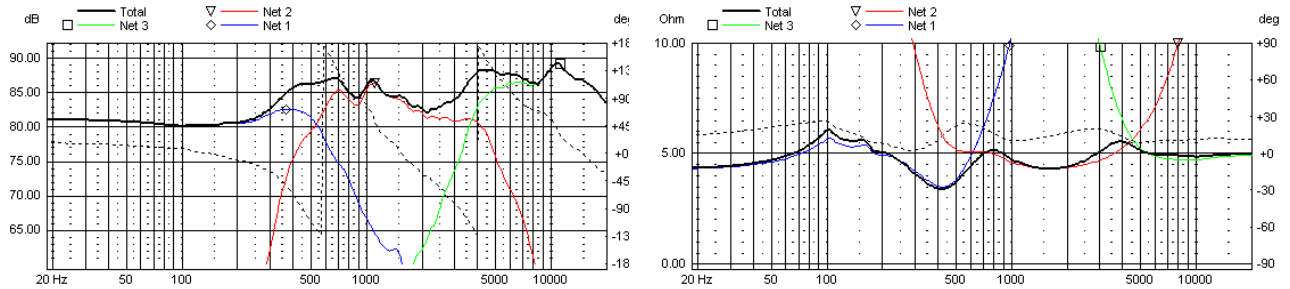


Fig. 9.15 Frequency response and impedance curve after first attempt with wizard and L-pads (manual tweaking)

treble networks.

The frequency response in fig. 9.15 is not very impressive. Fig. 9.16 shows the result when

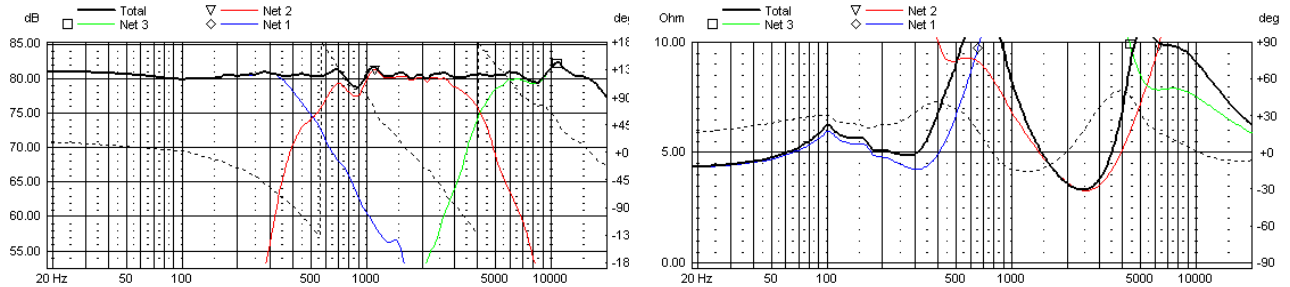


Fig. 9.16 Frequency response and impedance curve after system optimization.

the system optimizer has done its job.

This is getting closer to something acceptable, one problem here is that the impedance is only 3.2Ω at 2400Hz . If we set the minimum impedance to 4.2Ω and run the optimizer

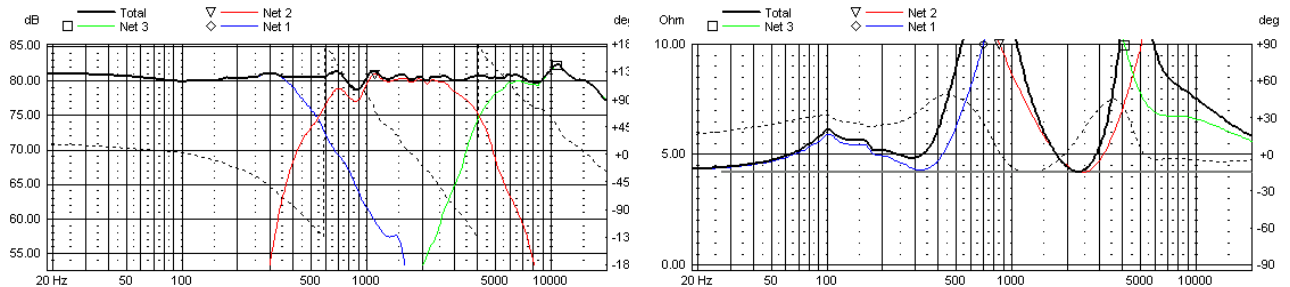


Fig. 9.17 Frequency response and impedance curve after system optimization with Z_{\min} limited to 4.2Ω .

again we get the result in fig. 9.17.

As we can see the minimum impedance is now 4.2Ω , still the frequency response is almost the same. With this option it is possible to get a fix to loudspeakers that are difficult power amplifier loads without making them perform worse.

10 Litterature

There exist a lot of good literature in this area.

Below is a list of some that I recommend for the interested.

JAES = JOURNAL OF THE AUDIO ENGINEERING SOCIETY.

Acoustics ; Leo L Beranek ; McGraw-Hill ; ISBN 07-004835-5

Almost a bible in this area, handles everything from the wave equation to room acoustics. However things have happened since this book was written.

Testing Loudspeakers; Joseph D'Appolito Audio Amateur Press 1998 ; ISBN 1-882580-17-6

Highly recommended for the DIY. Among other things it deals with the problem of making reliable measurements in far from ideal conditions.

High performance loudspeakers ; 3rd edition ; Martin Colloms ; Prentech Press ISBN 0-7273-0806-8

More intended for the newbie than the previous book.

The loudspeaker design cookbook, 4th edition ; Vance Dickason ; Audio Amateur Press 1991 ;

ISBN 0-9624-191-7-5.

A very good book, both for the newbie and the one who consider himself as a pro. Is highly recommended.

Great sound stereo speaker manual ; 2nd edition; David B. Weems, G.R. Koonce ; McGraw-Hill ISBN 0-07-134874-3

A good book for those new to the subject of speaker building, includes a lot of building projects with complete drawings and crossovers.

The influence of room boundaries on loudspeaker power output ;

JAES June 1974, vol 22, no 5, p 314-320.

Treats the placement of loudspeakers in the listening room.

A simple theory of cabinet edge diffraction; John Vanderkooy; JAES Dec 1991, vol 39, No 12, p 923-933.

Cabinet edge diffraction theory, part of the results is used in this software.

Loudspeaker Enclosure Survey ; Wireless world, November 1972, p 552-558.

Maximizing Performance from Loudspeaker Ports; A. Salvatti et. al.; Preprint 105th AES convention, San Fransisco, CA.

Empirical formulas and an extensive study of flared ports and their behavior at high SPL.

Work by Richard H. Small :

Direct-Radiator Loudspeaker System Analysis;

JAES June 1972,vol 20,no 5,p 383-395

Closed-Box Loudspeaker Systems. Part I : Analysis.

JAES Dec 1972,vol 20,no 10,p 798-808

Closed-Box Loudspeaker Systems. Part II : Synthesis.

JAES Jan/Feb 1973,vol 21,no 1,p 11-18

Vented-Box Loudspeaker Systems. Part I : Analysis.

JAES Jan/Feb 1973,vol 21,no 5,p 363-372

Vented-Box Loudspeaker Systems. Part II : Large-Signal Analysis.

JAES July/Aug 1973,vol 21,no 6,p 438-444

Vented-Box Loudspeaker Systems. Part III : Synthesis.

JAES Sept 1973,vol 21,no 7,p 549-554

Vented-Box Loudspeaker Systems. Part IV : Appendices.

JAES Oct 1973,vol 21,no 8,p 635-639

Passive-Radiator Loudspeaker Systems. Part I : Analysis.

JAES Oct 1974,vol 22,no 8,p 592-601

Passive-Radiator Loudspeaker Systems. Part II : Synthesis.

JAES Nov 1974,vol 22,no 9,p 683-689

Notes