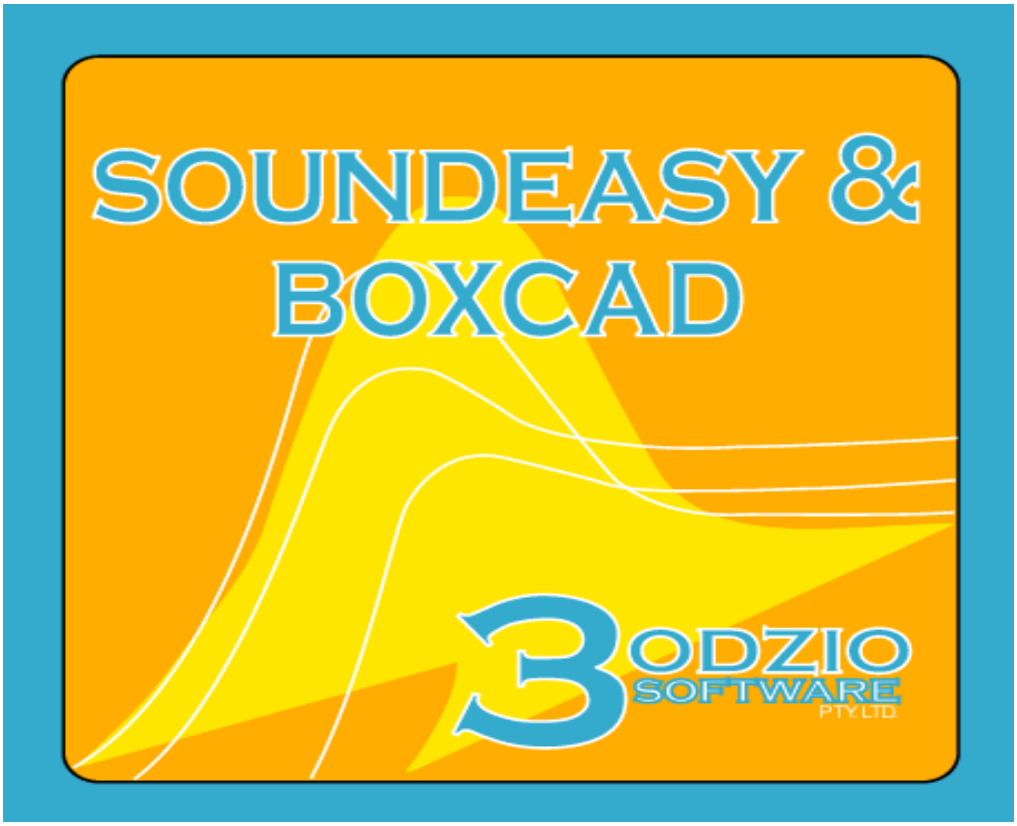

Reference Manual

Version 12.00

Recommended systems: Windows 2000, WindowsXP™

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Part 1

SoundEasy Fundamentals

Chapter 1. Introduction

The theory of the electrodynamic loudspeakers has advanced a great deal over the last four decades. Because of the popularity of multi-driver speaker systems (and their attendant design difficulties) this subject has been the focus of engineering and academic research. As a result, the advantages and limitations of the multi-driver speaker systems (and the various crossover types) have been exhaustively debated and published. Loudspeaker manufacturers, recognising these scientific advances, now provide their products with rather detailed driver specifications. Mathematical models which approximate real drivers and the various types of enclosures have been developed and confirmed in practical applications. **SoundEasy** computer program was developed to consolidate the accumulated knowledge. The resulting computer simulation deals with low, mid and high frequency electrodynamic drivers mounted in a suitable enclosures (one which provides optimum loading for the drivers across the required frequency range). Individual drivers may need to be compensated in time and frequency domains and the selected units eventually mounted in an enclosure to operate as a system. **SoundEasy** offers the designer a set of powerful tools, covering all aspects of system design in one convenient package. Advances in Computational Acoustics enabled us to implement powerful Finite Element Method for the purpose of evaluating room acoustics. We have incorporated two measurement techniques: analog and digital to assist you with data acquisition. **SoundEasy** is aimed at more advanced range of users, but beginners will be surprised how easy it is to evaluate drivers, enclosures and crossovers and produce a working speaker system from them. Professional designers and advanced hobbyists will benefit from the ability to explore the various alternative designs and as a result be able to more quickly choose and optimize both the enclosures and crossovers.

This software release reflects our continual effort to provide you, the User, with a friendly tool and a lot of enjoyment during your design/experimentation effort and is recommended as an upgrade for all **SoundEasy** users.

Program Overview

SoundEasy V12.00 is our fourteenth'th release of the 32-bit version of the program. List of Version 12.00 enhancements over Version 11.00 is shown below:

New CAD system

1. Component placement on nodes and line drawing to/from nodes. Schematics are now much more compact.
2. New component icon graphics. Vertical component pick-up list is now 16-icons high to fit 1024x768 pixels screens.
3. Components and links highlighted during manipulation and editing for easier identification.
4. Re-designed Editing functions: Draw, Erase, Wipe, Move, Cut, Paste, Undo, Redo, Copy, Add.
5. Added function Wipe-Wires. Select this option, and then go and click on all wires you want to be removed.
6. Added Cut-to-HD and Paste-from-HD functions. Allow you to store unlimited number of circuits to Hard Drive. This is the concept of "circuit storage bin", that has been proposed on the User's Group.
7. You can also Copy-to-HD and Paste-from-HD.
8. Differential OPAPMs re-designed for nodal placement.
9. Node numbering ON/OFF switch.
10. Placement nodes ON/OFF switch.
11. Removal of limitation on the number of Super-Component and Mega-Component.
12. Re-worked all filter and crossover built-in templates (wizards)
13. Components being optimized, are shown with different colour on the schematic for easy identification.

Other features

14. DCX2496 shelving filter Q-factor included in the remote flash.
15. Multiple Pen Color screens instances issues resolved – subsequent clicks do not create new instances, only bring the already opened window to front.

16. Number of sound source nodes increased to 9 in FEM system. You can have positive or negative polarity to check dipole loudspeakers in room.
17. Re-designed SPL plotting function in FEM screens. Plotting window scale is controlled the same way as CAD plotting windows scale.
18. DF/DE signal level slider set globally and saved to data file.
19. Added level Input LED indictors for DE/DF system. This feature prevents accidental overdriving of the DE/DF.
20. Added Notched 4-th order HP, LP and 2-way crossover.
21. Atan2 error message fixed.

EasyLab

22. When working with IRs saved to HD, the program automatically reverts to single-shot MLS operation.
23. New - Added RT60 room acoustics measurements using Shroeder Plots – 2 methods of noise cancellation included.
24. Output level for Nonlinear screen I and II is decoupled from the rest of the EasyLab system and is globally set and saved to Preferences file.
25. Sampling rate for Nonlinear screen II decoupled from the rest of the EasyLab system and saved to Preferences file.

Housekeeping

26. Updated Manual.
27. New data files in V12.

SoundEasy is a menu driven software, which requires Windows XP(TM) to be installed on your computer. The program takes advantage of the SVGA graphics capabilities of your computer (min. screen size is 1024x736 pixels) and enables you, the designer, to create, evaluate and then optimize 2, 3, 4 or 5 way loudspeaker systems **prior** to starting the assembly of the enclosure. Most errors or problems can be intercepted and corrected at a very early stage of the design, without cost penalties!. Greater speed and design accuracy can be achieved when designing the enclosure, filters, and complete crossover.

A **Hilbert-Bode Transform** technique is employed which re-creates complex transfer function from the frequency response of a driver. Using this method, the magnitude and phase response of a driver across the desired frequency range can be obtained. You can now take into account phase errors introduced by the driver or drivers at the crossover point. The data imported from the measurement system may be in error. One of the **sources of data errors** can be the noise level in your measurement system (sound cards). This problem may manifest itself as amplitude errors and particularly **phase errors at lower signal levels**. SoundEasy allows you to add small time delay to somewhat compensate for the measurement system errors. Comparison of the imported phase with no acoustic delay with the phase generated by the built-in Hilbert-Bode Transform will tell you if your measured data is correct. One of the most significant sources of errors in speaker system design is **the loading of the crossover** by the complex input impedance of the driver. (Crossover design can be a formidable task when "ideal" loads are replaced by "real" drivers).

Using the built-in **electronic CAD system,** SoundEasy enables you to model the behavior of the crossover when loaded by the driver in an enclosure and observe the dramatic effect on the frequency response curve of the crossover that the driver may have. A suitable design compensation network can be also created on the same screen and the effect on the crossover re-examined. The program allows you to pin-point problems and test possible solutions so that the frequency response of the crossover is as close to the "ideal" as possible.

The effect of the statistical spread (variations within a specific model) of loudspeaker parameters can be quickly evaluated using **parametric family plots**. (This will be of great assistance to loudspeaker manufacturers when setting their quality tolerance levels). In order to provide you with a high degree of flexibility, a **schematic capture** module is provided as one of the SoundEasy tools.

The process supports a 'pick-and-place' technique for creating schematics and is based on **powerful 'Nodal Analysis'** (matrices of complex numbers) techniques, employed by most world-recognized electronic CAD systems. An advanced numerical optimization method called the **'Method of Rotating Coordinates'** is used throughout the program to optimize the enclosure parameters, the individual filters and the complete crossover. The Method of Rotating Coordinates was selected for its speed and accuracy. **Box, crossover** and **system** response to a rectangular pulse can be evaluated in **time domain** and correlated with its frequency response. SoundEasy offers here several tools such as DSP techniques (1024 point **FFT & IFFT**), Lattice Network Calculator, and Network Analysis in Time using Runge-Kutta methods.

The final consideration (prior to the commencement of listening tests) is usually the question of enclosure positioning within your listening environment. This problem can be very complex if all the influencing factors are considered. SoundEasy offers a solid, engineering approach to improve the overall sound quality of the system **in the testing position** using **Ray Summation, Image Summation** and most importantly**, the accurate FEM** methods (see Chapter 15). Current SoundEasy release allows you to actually plot "room contribution" effect vs. frequency for any position of two subwoofers in a **non-rectangular room**. When a high power is applied to the driver for a prolonged period of time, the voice coil heats-up and the performance of the driver deteriorates. **Enclosure Modes are also calculated using the FEM**.

The effects of the **voice coil temperature** on the frequency response of the high power drivers can be evaluated at all power levels. Also, the temperature of the magnetic structure can be calculated if the thermal resistances of the driver are known. Detailed analysis can now be performed on all eleven types of enclosures for any power level between 1mW to 1000W. Current version of the program incorporates **Large-Signal Analysis** based on BL(x) and Cas(x) non-linearities. Both parameters are approximated as a set of curves, which can be edited for the best fit with a known parameters of the driver.

A **Total Harmonic Distortion** (THD) estimator is also built-in. To top it all, **port non-linearities** are also simulated. A complete **off-axis analysis** can be performed in the program and horizontal and vertical SPL plots are available. **Titling drivers** is available and this will come very handy when attempting to design an enclosure with the **front baffle of a complex shape**.

Digital Equalizer provided with the CAD section of the program is capable of "non-minimum phase" equalization of the loudspeaker system, and therefore, correcting amplitude and time domain delays of 5-way loudspeaker system. An extension of this concept is equalization of the "speaker-to-ear" room transfer function – the **Digital Room Equalizer (DRE).** The DRE is a "minimum-phase" equalizer, which in addition, includes a delay function as well.This new feature has been implemented to allow you to "deconvolve" or "equalize" areas of your listening room. The issue of room equalization has been described in many publications, and seven of them are listed below to start your creative thinking process flowing. Possibly start with item 1, as it is mostly descriptive, and then item 4.

SoundEasy also incorporates a measurement system – **EasyLab**. The system has been designed and optimised for analog and digital loudspeaker and room acoustics measurements. **Nine measurement systems** have been incorporated: an analog technique for tone, stepped tone and gated tone, and digital technique (MLS). SPL and input impedance can be measured by both systems. RT60 is the latest addition to EasyLab.

General Program Specifications.

Environment.....…: Windows2000, Windows XP.
User interface.....: Mouse, Keyboard, Windows.
Program control.....: Mouse, Menus, Buttons.
Hard copy.....: Graphics Printer, e.g.: LaserJet 5L.
Screen output.....: 1024x736 pixels min, 256 colours
Processor.....: Pentium 4, 1.7GHz or better is recommended.
RAM.....: 256 Mb **MINIMUM REQUIRED**.
Hard Disk required.....: 30.0 Mb min.

Short Summary of the Main Software functions performed

1. Loudspeaker driver reference library creation.
The program will accept the electro-mechanical (including acoustical center offset) and frequency response parameters of each driver. These parameters can be saved to disk for later editing. Frequency response is entered as SPL (1 meter, 1 watt) for 750 discrete frequencies, separately selected for woofer, upper base, midrange, tweeter and super tweeter. Complex transfer function of each driver is then re-created by the user. Program facilitates importing frequency response and impedance data and manual editing of the imported files. Two **accurate impedance models** are provided.

2. Loudspeaker enclosure design.
The function is implemented by means of entering vital parameters of the enclosure: volume, Q-factor, tuning frequency or TL line length and so on. The software will then plot the frequency response, relative cone excursion, the system (driver + enclosure) impedance for a specified driver, system phase response and group delay (and more curves in total, depending on the enclosure type).

Sealed, vented, double-chamber and triple-chamber, sealed-vented, vented-vented, passive radiator, transmission line, horn enclosures, vented/TL subwoofers, U-Diplose and H-Dipoles (total of 13 enclosures) are supported (both sealed and vented enclosure design functions are provided for midrange and upper bass drivers). Parametric family curves can be obtained by incrementing (within a predetermined range) one of the enclosure parameters. This automatic process will plot the changes in frequency response, cone excursion and system impedance. Isobaric extension is available for sub-woofer system. 'Multiple drivers' variations are also modeled. **BL(x) , Cas(x) , temperature and port non-linearity** and compression modeling are included. You can also print **detailed dimensions of a rectangular enclosure** with bracing and all parasitic elements inside the box accounted for.

3. Loudspeaker enclosure optimization.
This advanced feature also takes advantage of the "Method of Rotating Coordinates". All three enclosure parameters: Fp, Ob and Vb are optimized at the same time as the program attempts to obtain the "optimally flat" alignment for a given loudspeaker. Sealed, vented and band-pass type 2 enclosures can be optimized for the bass driver, upper bass and the midrange driver. Several reference alignments are provided.

4. Compensation of the driver impedance, amplitude peaks or time delays.
This function enables you to :
- Cancel voice coil inductance
- Compensate for the upper bass/midrange/tweeter/super tweeter resonant impedance peak.
- Reduce amplitude irregularities.
- Adjust sensitivities of the higher frequency drivers.
- Insert a time delay network, such as Lattice Network to offset the acoustic center of the driver.

Driver compensation can result in an improvement in the performance of the crossover. The resulting amplitude/phase/impedance/group delay plots are available for all three drivers. Full examination can allow the designer to model the behavior of the designed network. L-type attenuation pads and the interaction of the woofer and midrange drivers with their enclosures can also be evaluated. Active components (OPAMPS) are provided.

5. Crossover filter design.
SoundEasy provides you with a full electronic **CAD GUI for the schematic capture** process. This allows you to design ANY type of circuit or crossover (parallel, series, cascaded and so on…).

There are also several types of circuits and many types of pre-built crossover filters provided**: Bullock, Butterworth, All-Pass, Linkwitz-Riley and Bessel** filters. You can also select four rates of rollover for low-pass, high-pass and band-pass configurations. Amplitude/phase/impedance/group delay plots are available to enable full evaluation of the design. Bullock coefficients for all filter orders are dynamically calculated. As a result, the effect of the crossover on the real driver frequency response can be readily observed. Active filters (OPAMPS) are also provided. The **Add Network function** allows you to quickly put together complex, uneven order crossovers.

The program also includes extensive support for **Transient-Perfect** crossovers and **Digital Equalizer and Filtering** option. The processing power of the CAD system is enhanced by Super-Components and Mega-Components. Finally**, Cut-and-Paste** method combined with the ability to have 3 Projects opened at the same time, allows you to create extremely complex crossover systems. In the concept of Modular Crossover, the filters are defined in terms of cut-off frequencies and slopes. There is also an array of shaping circuitry available. SoundEasy now includes 17 CAD processing functions:

1. **Draw** – For "pick-and-place" components and drawing connections
2. **Erase** – For erasing single connection or component
3. **Overlap** – Changing components without disconnecting or moving them
4. **Undo / Redo -** For "undoing" the last 4 steps, and cancelling the last Undo operation
5. **Wipe** – For disconnecting and erasing the component
6. **Wipe Wire** – For erasing wires
7. **Move** – For disconnecting and moving component around
8. **Add** – For adding a complete, pre-calculated filter or even a complete crossover to your schematic
9. **Cut** – For cutting a section of the schematic
10. **Cut To Hard Disk** – For cutting a section of the schematic and saving it to HD
11. **Copy** – For duplicating a section of the schematic
12. **Copy To Hrd Disk** – For duplicating a section of the schematic and saving it to HD
13. **Paste** – For pasting the Cut section
14. **Paste From Hard Disk** – For pasting the Cut section from HD.
15. **Cad Nodes** – For switching ON/OFF dots (nodes) on CAD area.
16. **Schematic Nodes -** – For switching ON/OFF node numbering on CAD area.
17. In addition, there are over 100 **in-built filters and crossovers** to choose from.

6. Crossover filter optimization.
The optimization process takes into account the driver's complex transfer function, the effect of the enclosure, the input impedance and the compensation network characteristics. As a result of the optimization, alternative low-pass, high-pass or band-pass filter components are calculated. This feature employs the Rotating Coordinates Method and can optimize complete crossover. Target curves can be extensively modified by tilting, stepping, and **adding ripples**.

7. System frequency response evaluation.
This feature enables you to plot the combined frequency response of the complete speaker system taking into account all driver, crossover and enclosure related parameters. SPL amplitude plots are provided, together with number of impedance/phase and delay plots. As well phase reversal effects for all drivers can be observed.

8. System frequency response optimization.
A number of approaches can be tried using this tool. The filter components of any driver can be optimized over a selected frequency range in order to obtain the flattest possible system response. Alternatively, the target frequency response can be heavily distorted, thanks to the **new "Add Ripples" feature**. Once again, the Method of Rotating Coordinates is used in this tool. Optimization can be performed for SPL, phase and Zin constraints.

9. Frequency response of the system "in-room".
The effects of placing the enclosure in the corner of a room on the frequency response can be estimated with this feature. Another screen allows for frequency response evaluation within 10-200Hz in a fully enclosed room of ANY shape.

10. Calculators.
L-pad calculator, Series LRC calculator, Zobel Network calculator, Impedance peak suppressor, Lattice Network Calculator, Vent Calculator, Rectangular Enclosure Details and Enclosure Shapes calculators.

11. Large-Signal analysis.
The program accepts thermal resistances of the driver's voice coil and magnetic structure and enables you to examine all modeling curves and structure temperatures at all input power levels from 1mW to 1000W. Power compression" effect can be clearly observed on the frequency response and all other plots. In addition, **port non-linearities** and non-linearities of the suspension system can be modeled and editing of **BL(x) and Cas(x)** curves is provided. Finally, **Total Harmonic Distortion (THD)** of the driver+enclosure can be estimated over 5Hz-2kHZ frequency range of interest.

12. Polar and "Straight Line" vertical and horizontal plots.
Horizontal, vertical and tilted location of the drivers on the front baffle can be varied and the interaction of the complete system examined at any frequency. This is particularly important at the crossover points. There are three dialogue boxes the implement calculators and plotting functions for "off-axis" response.

13. Time domain responses.
Enclosure response to a rectangular wave can be observed in time domain. This enables the designer to determine the amount of tolerable "ringing" generated by some specific alignments. This analysis is available for all seven enclosure types. Frequency of the rectangular wave can be adjusted between 1 and 500 Hz and FFT and IFFT techniques are employed. Finally, crossover response to a rectangular function can also be observed in time domain. This useful feature enables you to determine if the crossover is of the "constant voltage" type. This feature operates just like a "digital storage oscilloscope" and allows the designer to add up to 5 curves in time domain. Two methods are implemented: (1**)** **FFT Technique** and (2) **Modified Nodal Method** in Time Domain.

14. Importing data.
SoundEasy allows you to import amplitude/phase and impedance/phase from three measurement systems: (1) IMP, (2) LMS, (3) CLIO, (4) LAUD, (5) MLSSA and (6) DAAS32 . Some associated processing functions are also included, so that the imported data can be validated over 10Hz-40kHz frequency range. Small/Thiele parameters can also be imported from CLIO and DAAS32 systems.

15. Low-frequency room analysis.
The low-frequency analysis is performed using the FEM method. The FEM is now capable of analysing objects having irregular vertical shape such as arches and pyramids. 2D and 3D FEM screens are implemented.

16. Subwoofer design.
Active Equalization function is provided in the "Box" module. You can design the passive/active EQ circuit using "Crossover" module and then use the combined (box+EQ) frequency response optimize the subwoofer design and check it under Large-Signal conditions.

17. Enclosure Diffraction Analysis.
This feature allows the user to mount up to 4 drivers (quad-box) on the front panel and analyze resulting diffraction and mutual impedance effects. Diffraction distortion are best analyzed in an anechoic chamber, where the driver radiates into a full space.

18. Implemented Small-Signal Measurement System.
18.1 Measures SPL and Impedance
18.2 Impulse and Step responses
18.3 Two types of Waterfall analysis
18.4 Small/Thiele parameters
18.5 Enclosure stuffing Qb
18.6 Oscilloscope
18.7 Spectrum Analyser

Requires full duplex, 16-bit, stereo sound card.

19. Implemented Non-Linear Measurement / Modeller
19.1 Volterra Series Expansion Method Measurement System
19.2 Non-linear Differential equations Method Measurement System
19.3 Non-linear Differential equations Method Modeller

20. Implemented RLC Meter

21. Room Acoustics – RT60 measurements

22. Digital Filter
If you have a compatible soundcard, you can also employ the crossover schematic as if it was a real filter network, and use it to filter music sourced from an external CD-player. This process is implemented as a DSP function in SoundEasy.

23. Digital Equalizer
You can still take the Digital Filter functionality further, and add equalization function to smooth-out all ripples of SPL curve, together with filtering (as in 21) function. This process is a lot more elaborate and sophisticated. However as a result of it, you will enjoy the acoustic response of the complete driver channel (loudspeaker SPL+crossover filter) looking as ruler flat electrical filter.

The above list is only intended to highlight many of the built-in features of SoundEasy. There is more to be found in the program, while you learn its true potential and flexibility. We suggest, you may want to start with Chapter 18 – which provides you with an overall direction and a "top level" set of hints to manage the program and your design process. Hopefully, you will find the manual informative enough to describe all functions incorporated in the program. We have also strived to provide you with numerous, easy to follow examples, that may provide an additional degree of confidence in your design activities with SoundEasy.

Finally, four fully edited data files (3 driver files and one project file) have been provided with the program, so you can start using the program simply by loading any of the files and immediately exploring SoundEasy processing power.

Installation.

SoundEasy program is supplied on a CD-ROM. Test Files are supplied as well. The complete SoundEasy + BoxCad contains 2 modules plus manuals for SoundEasy and BoxCad. Installation process is intentionally kept very simple and installation of the program is performed in a few steps listed below:

1. Invoke Windows XP
2. Insert CD-ROM disk in the drive E:
3. Select "Start" option from the main menu
3. Select "Run" option from the menu.
5. Type E:\setup and hit "Enter" key.
6. Follow a few simple instructions in message boxes. **We suggest, you DO NOT change the default installation directories**.

7. When finished with the software installation, **connect the Security Key - see page 18.8. SoundEasy ships with the standard Security Key driver for WinXP. For other systems, we have also included a dongle driver installation program that automatically detects your operating system and installs appropriate driver. The program is called "InstDrv.exe". You can run it from your CD-ROM and all you need to do is to follow a couple of simple steps.**

8. SoundEasy and BoxCad Manuals do not have icons created during installation process. Both manuals reside on the CD-ROM as "Adobe Acrobat PDF" files.

Screens As The Processing Tools

The operation of SoundEasy resembles a chain of "production line" activities. The raw driver data, entered at the beginning of the design process is being manipulated, processed and expanded by the rest of the modules. Each module operates on the data provided by the previous module. See Fig 1.1 for the path of data flow.

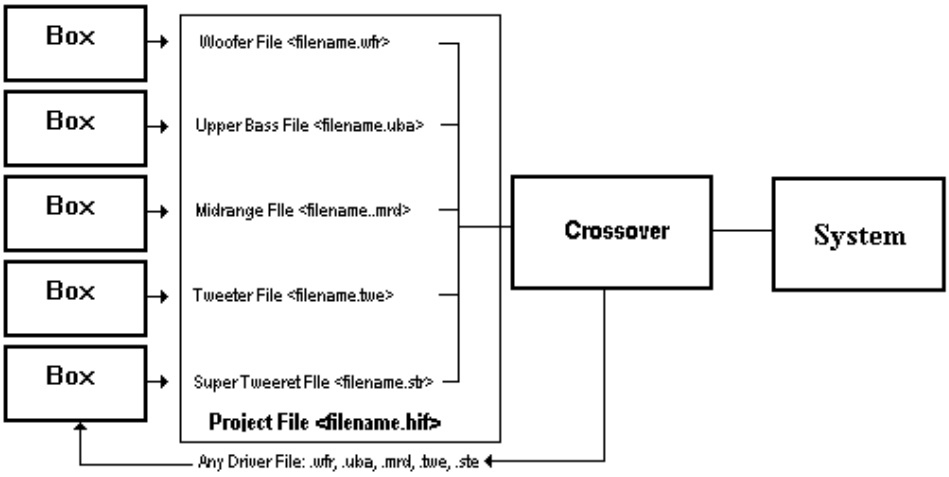


Figure 1.1 Information and data flow within SoundEasy

Data flow is sequential. This process does not allow you to jump a module. For example, you cannot use the summation screen to process data provided by the 'Box' module. The “Crossover” and summation screens requires up to 5 files to be loaded at the same time. To speed up the process, a Project File is being used, where paths and file names can be stored for automatic loading. Passive or active circuits can be created using Crossover Screen and then loaded back into the Box Screen to be included in the modeling of the enclosure frequency response (EQ circuit for subwoofer design). The description of the modules as tools is deliberate. It is essential to the successful use of this program that each module is seen as a tool which the designer uses to work on the available data and arrive at desired results. More exact block diagram of module operation is provided in the following sections of this manual.

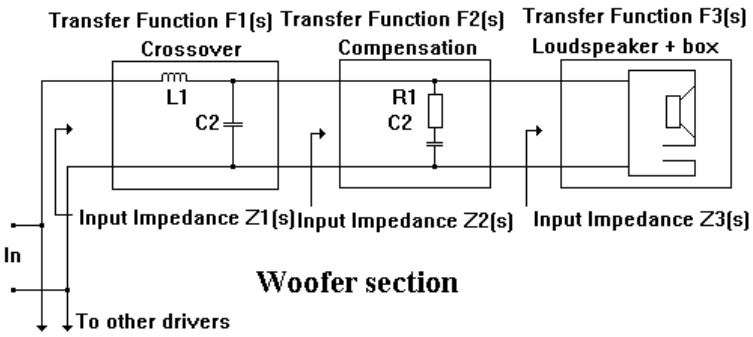


Figure 1.2 Transfer Function example.

In the above example, 'Box' module were used to (1) synthesize driver's input impedance while mounted in the selected box - $Z_3(s)$ and (2) generate transfer function of the driver mounted in the box - $F_3(s)$. $Z_3(s)$ becomes the load impedance for the compensation circuit - that is, if you use loudspeaker symbol in the compensation network developed using 'Crossover' module. Also, total transfer function is now $F_3(s) * F_2(s)$. Finally, $Z_2(s)$ becomes the load impedance for the crossover network, again, if the loudspeaker symbol was used in the crossover network. The total transfer function now equates to $F_3(s) * F_2(s) * F_1(s)$. Input impedance of the woofer section is $Z_1(s)$, $s = j\omega$.

Printing Session Results

All images of the program’s windows within this manual have been developed using the program's graphics capabilities. Printing session results is achieved by selecting the "Print” option from the window's menu. This selection opens SoundEasy printer dialog box (Fig 1.3) and the user may select orientation of the printout, scaling factor and location of the printout.

Some users may give preference to smaller or half-size print-outs. The final dimensions of the print are a combination of the scaling factors entered and the resolution of the printer used. It is therefore recommended to perform a test print-out for the current printer settings and if the results are satisfactory, use the same scaling factor consistently throughout the program. All options selected from the printer dialog box will be saved to the driver's data file. Orientation (Portrait or Landscape) selected from the SoundEasy printer dialogue box must be the same as the one entered in the Standard printer control dialogue box. This is required to avoid graphics being stretched vertically over the whole page by the Standard printer control dialogue box. The "Offset X" and "Offset Y" parameters can be used to move the location of the graph on the printed page independently in X and Y direction. Finally, a small window which appears on the screen and informs the user of the progress of the banded or non-banded print-out.

Printing capabilities of the program were developed for commonly supported printer drivers and tested on number of printers and emulators. If your printer is not operating properly, further customization may be needed, as it is likely, that your printer driver file is supporting only limited functionality or it can not be invoked from within the application program. Also, in case of difficulties, we recommend to experiment with your printer DPI settings and start with low DPI: 75, 150 and then 300. Reduce size of the printout (30%, 50% 80% and 100%) and landscape or portrait orientation.

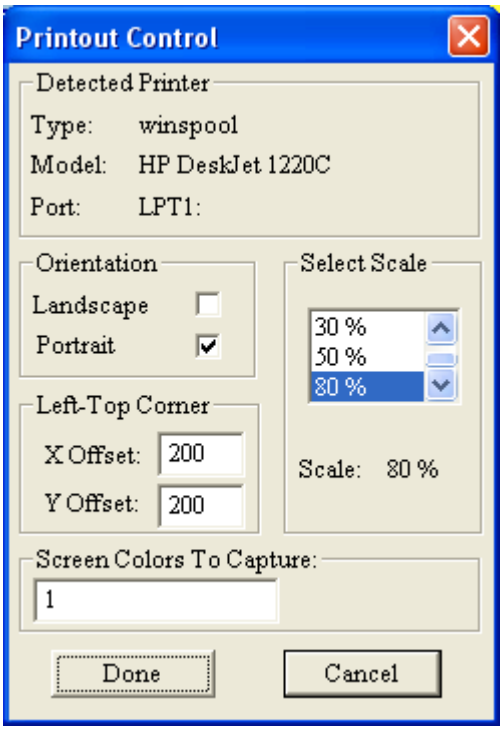


Figure 1.3 "Print" dialogue box

Please consult Windows™ manual for screen settings. Low DPI and smaller printout size will work better on lower memory hardware. Finally, please review your printer's manual and make sure, that your printer is set optimally for complex graphics printouts.

For example: HP LaserJet 5L with standard 1Mb memory needs to be set to Manual/Raster/300dpi setting for optimum screen dumps. All graphics screens presented in this manual were printed on DeskJet1220C

Saving a New Driver

It is recommended that you save intermediate data 3-4 times per design session. To save data select the "Save Driver File" option from the windows menu.

Loading Data File

Loading woofer data file is achieved by using the corresponding menu option. Other driver types follow the same procedure. Once the "Open Driver File" option is activated, user may select the whole path, including drive, directory, sub-directories and finally the name of the file. The dialogue box is a standard WinXP™ box.

File Compatibility

SoundEasy 9.00/10.00/11.0 data files are ONLY PARTIALLY IMPORTABLE into V12.00 release. **CAD schematics are NOT importable**. The recommended way of dealing with the older data files is as follows:

Load every V9.00/10.00/11.00 driver's data file into Editor screen of this program. Please do basic checks for completeness of the parameters and open the Enclosure Design screen and also perform basic checks.

1. Next, open the Crossover CAD screen and confirm, that schematic is OK.
2. Is the above two steps check OK, please save the file, perhaps under a new name.
3. Now load this newly saved file back into this program and confirm, that it loads without "Importing…." Warning message.
4. Proceed the same way with older project files.

Only driver file prepared this way can be used in V12.00 projects.

Preferences Screen

The preference screen is divided into several blocks of common data.

1. Frequency Range.
There are two options available: System and Box. The System option sets the lower and upper frequency limits for all screens with the exception of enclosure design screen. The limits are selected from the provided list boxes and the selected options are saved to driver's data file. We suggest, that you select the most preferred frequency range option and use it consistently within the program. **Please pay particular attention to the fact, that importing SPL and Zin data will be arranged depending on the screen resolution. This has the consequences, that if you import the measurements for one screen resolution and then switch the program to another screen resolution, the imported SPL/Zin data becomes invalid.**

2. Colour Scheme.
Several options for choosing colour scheme for plotting curves is also provided. Adjusting colour is as easy as sliding the "Red", "Green" and "Blue" sliders. Pen colours are also editable from the main menu option.

3. **Audio Devices**.
This section allows you to disable mixer device control by SoundEasy and also invert left and right channels. Also, the **preferred audio devices for input and output can be selected from this screen**.

4. Environment.
There is only one field available for entering the speed of sound you wish to be used for the analogue and digital measurement screens.

5. Non-Linear Screen.

There are several list boxes to select the frequency range of the screen, scaling of the THD vertical axis and two check boxes for enabling F4 and F5 display and finally, enabling the THD plots.

All settings of the “Preferences” screen are saved into the preferences file. This file is stored in the same directory as the main executable code and is automatically loaded when the SoundEasy is invoked. The preferences are saved automatically on exit as well.

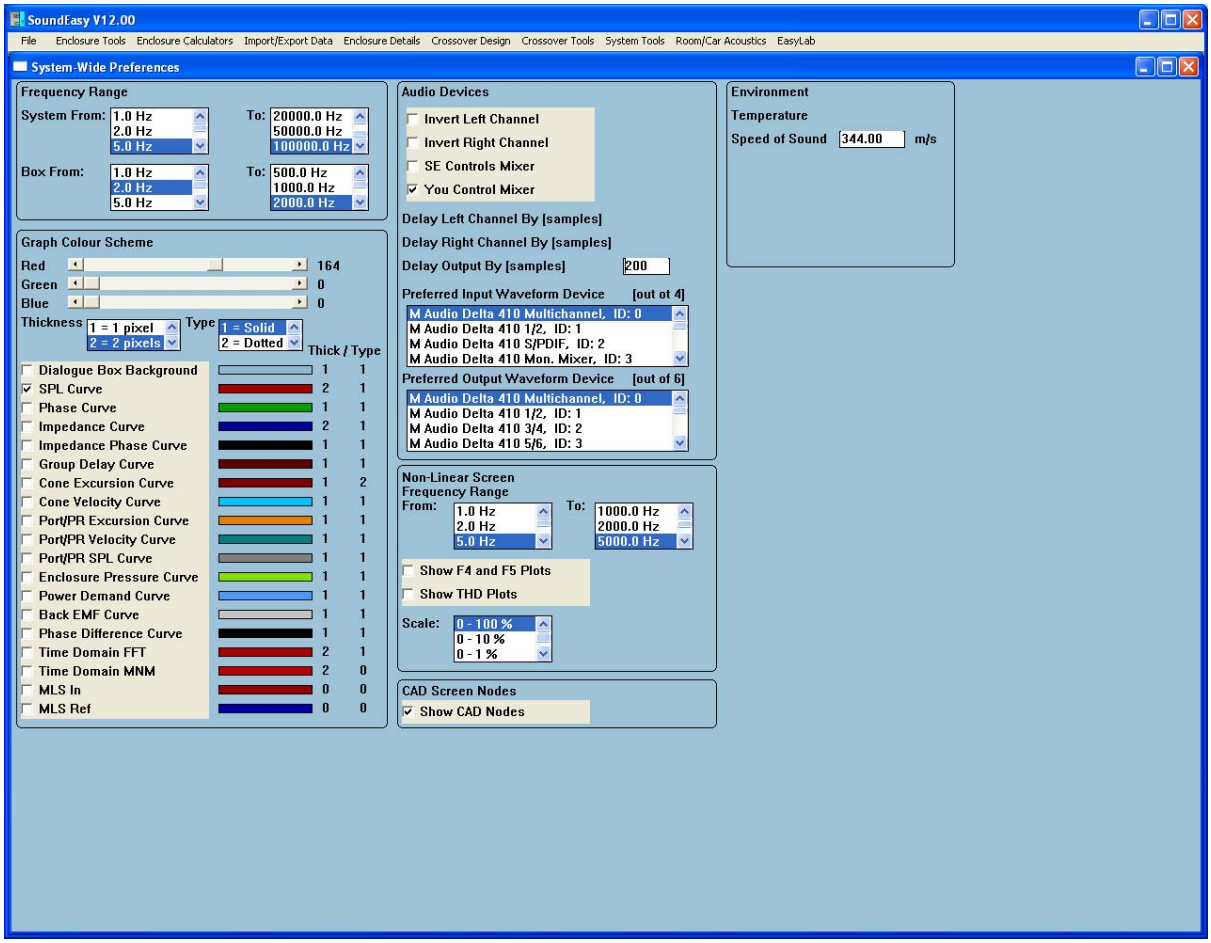


Fig. 1.4 Example of Preferences screen settings

User Interface Customization

The GUI of the program has been designed around standard Windows™ control-display techniques of using dialogue boxes to enter and control of the processing of the data and then using re-sizable windows to display the results. Typical control box is movable and has fixed size, and all plotting windows are re-sizable and movable. SoundEasy "remembers" the last location of the dialogue boxes and the size and location of the plotting windows, so perhaps the first step into the GUI should be the "customization" – meaning, moving around and adjusting the size of SoundEasy screens and dialogue boxes, depending on your PC screen size and resolution.

Minimum recommended screen size is: 1024x768 pixels. You can also use larger sizes: 1280x1024, 1600x1200pixels and so on.

As you may noticed already, the program is divided into several areas of functionality. For instance, when you concentrate on editing of the driver, it is unlikely, that you will need EasyLab screens opened at the same time. Therefore, SoundEasy automatically closes the EasyLab section when you work with the driver, enclosure, CAD or System sections of the program. This approach is applied consistently throughout the program and all unnecessary screens, that do not belong to the current activity are closed. However, be prepared, that many times, you will be working with 3 or 4 screens/control boxes opened at the same time.

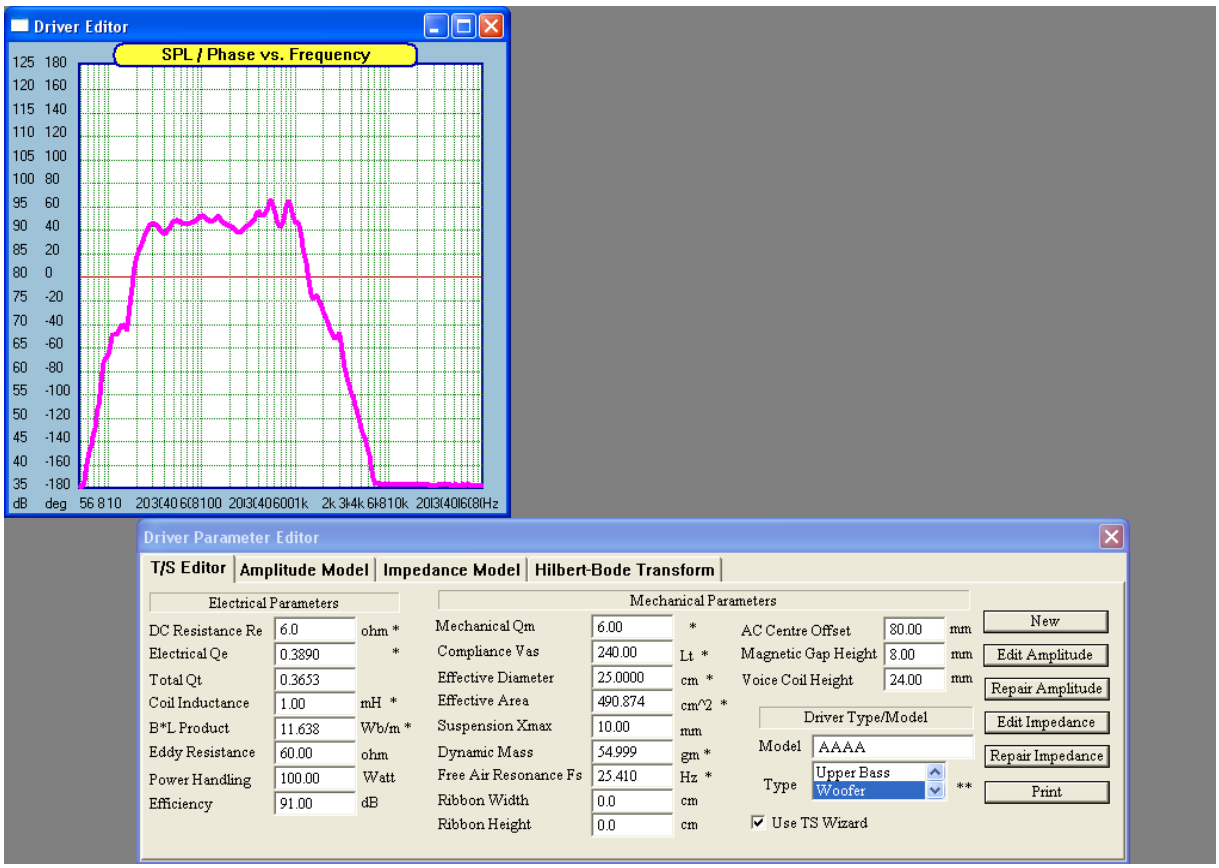


Fig 1.5. Plotting window reduced in size

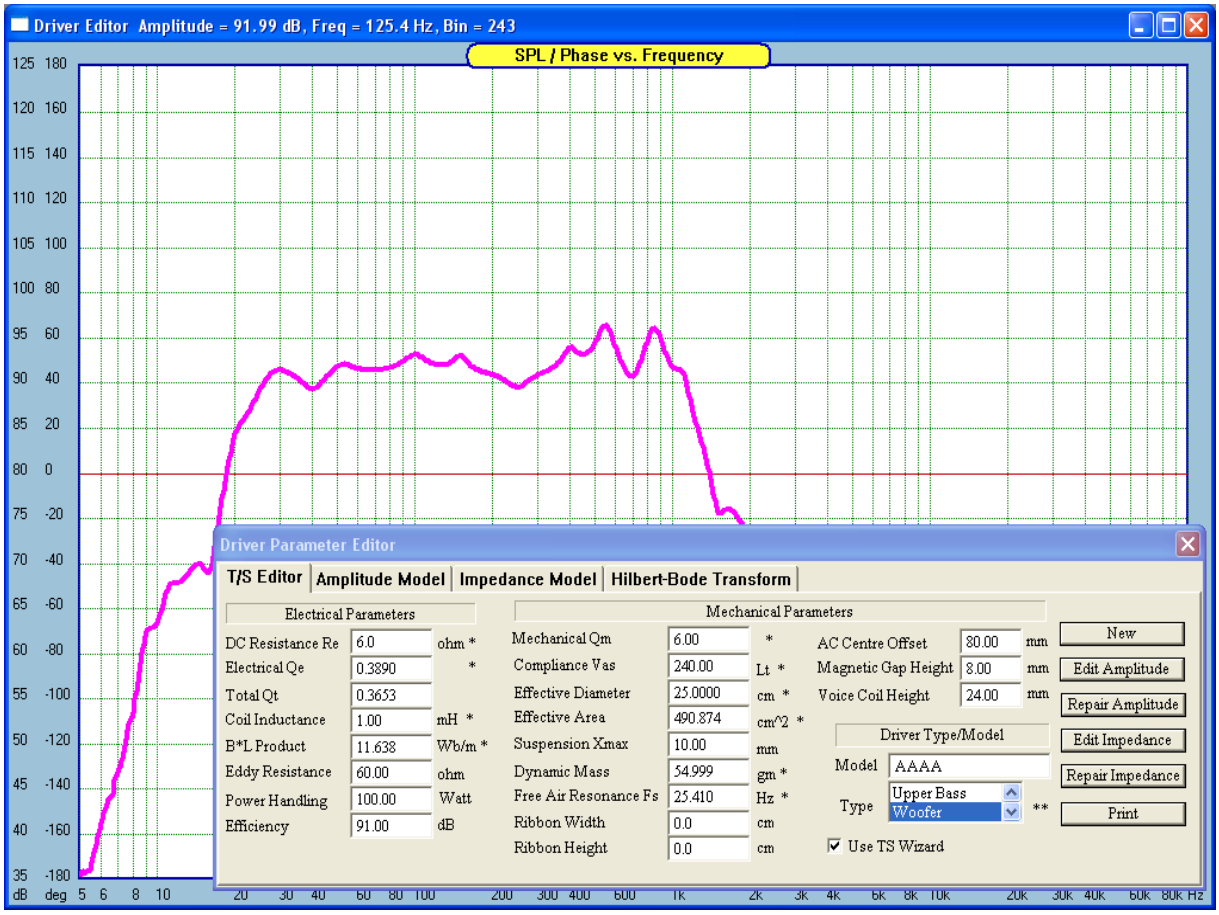


Fig 1.6 Plotting window maximized.

You can customize the look-and-feel of the GUI.

Please follow the main menu of the program and open menu items (screens) as you are moving from left-to-right along the menu. When you start the program for the first time, all windows will be opened with their respective default size and locations. Please feel free to make the necessary adjustments to suit your taste and your monitor. Next time you return to this screen, it will be opened in the same location and same size.

MLS Measurement Screen & Editor Screen

In some instances, it may be beneficial to have uninterrupted access to Editor screen while you make measurements. This would allow you to setup measurement system once, perform an SPL or Zin measurement, save the driver file, open a new driver file obtain measurements for the new driver and save them an so on….

If you need to be working in this mode, please open the measurement screen first and then Driver Editor screen second. Both functions will stay opened now as shown for example on Fig 1.7.

Driver & Project Notes

It is often desirable to remember some unusual details about this particular driver file or a project file. Those detaile can now be convinently entered into the **Notes Windows**. There are two Notes Windows: one associated with Driver File and one associated with Project File. Both Windows can store 1024 characters each.

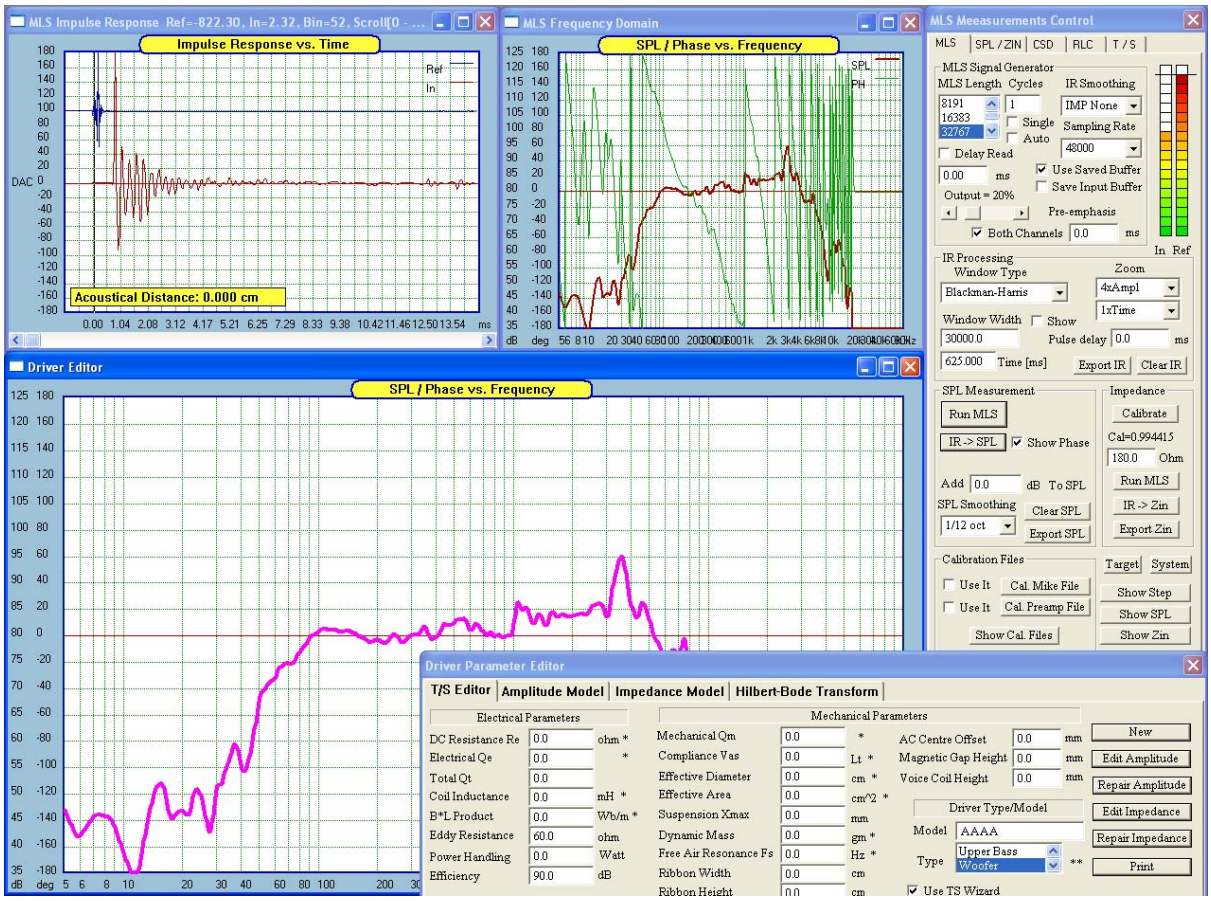


Fig 1.7. MLS Measurement screen opened with Driver Editor screen.

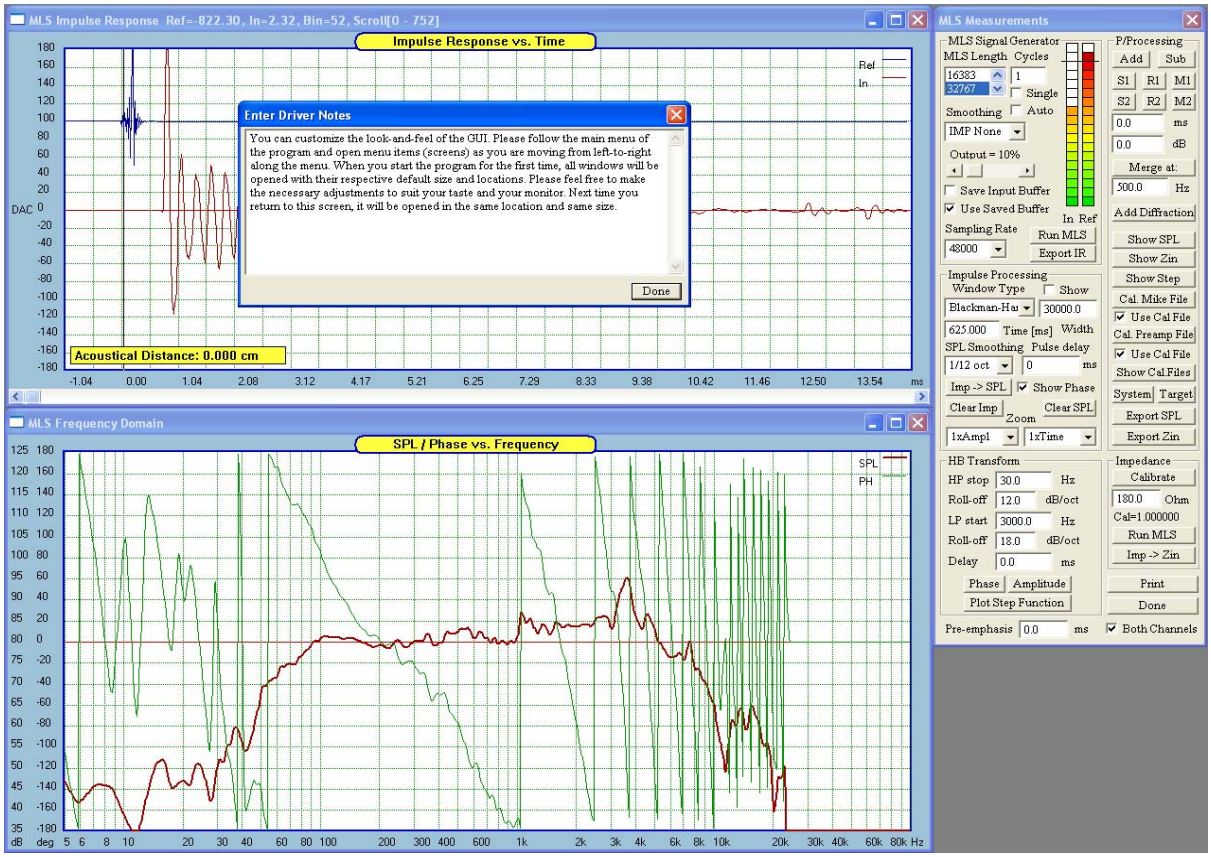


Fig 1.8. Driver Notes window.

Printing Driver and Project Data

Apart from straightforward screen dump, there are three printing options for hard copy output.

1. Print Driver – Selecting this option prints ALL parameters listed on Figure 1.9a,b, SPL/Phase curves and Zin/Phase curves. You may expect 2 pages of printouts.

2. Print Project Drivers - Selecting this option prints ALL parameters listed on Figure 1.9a,b, SPL/Phase curves and Zin/Phase curves for ALL 5 drivers. You may expect 10 pages of printouts.

3. Print CAD + Parts List – Selecting this option prints CAD schematic compressed to asingle A4 page and complete parts list. You may expect the CAD schematic to be rather small. However, this is expected to providing printing capabilities for all home-type of printers with A4 capabilities.

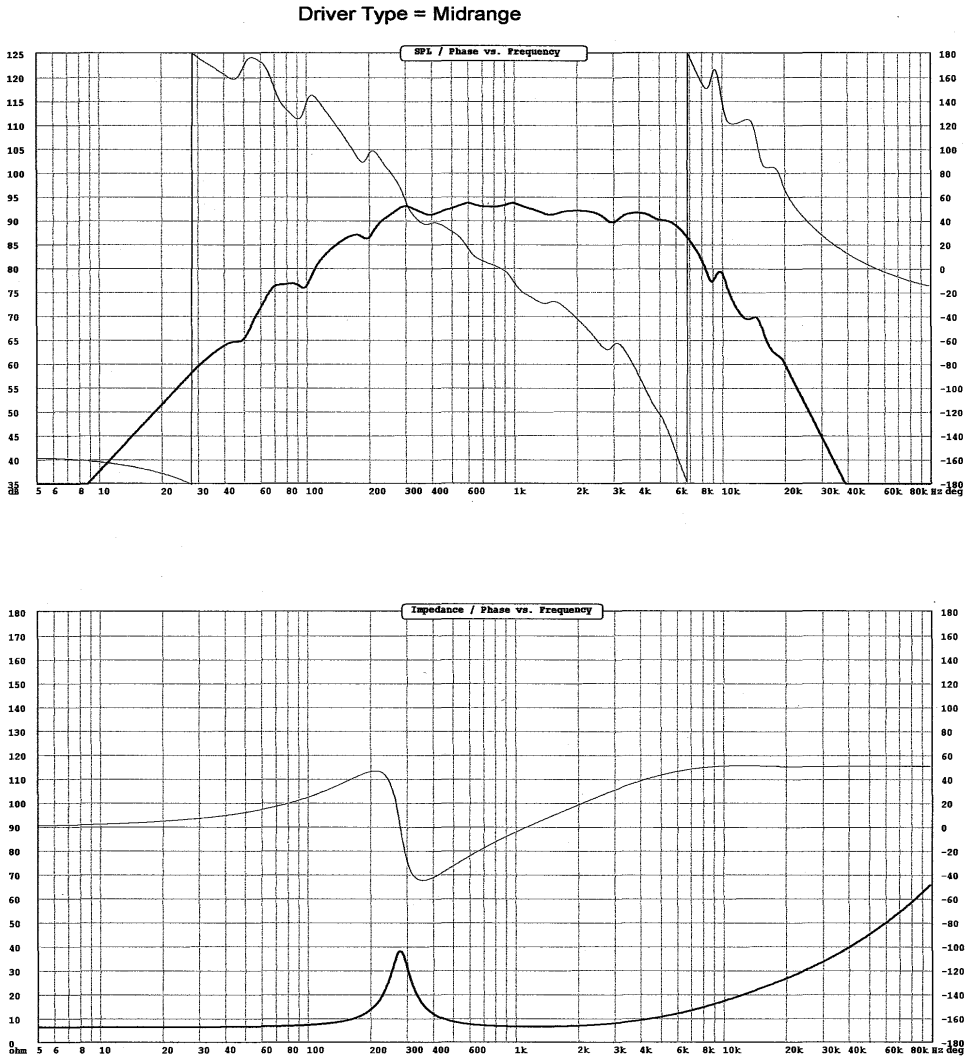


Figure 1.9a. SPL and Zin printout examples.

Driver T/S Parameters

DC Resistance = 5.80 ohm
Electrical Qe = 0.5405
Mechanical Qm = 5.46
Total Qt = 0.4918
VC Inductance = 0.31 mH
Force Factor = 5.000 N/A
Eddy Resistance = 39.00 ohm
Compliance Volume = 1.3 Lt
Nominal Power = 90.00 W
Efficiency 1W/1m = 92.50 dB
Piston Area = 50.265 cm^2
Effective Diameter = 8.0000 cm
Max Linear Excursion = 2.50 mm
Dynamic Mass = 2.000 gm
Free Air Resonance = 185.395 Hz
Dynamic Mass = 2.000 gm
Accoustical Offset = 30.00 mm
Magnetic Gap = 2.00 mm
Voice Coil Height = 5.00 mm
Thermal resistance VC-Magnet = 1.00 ohm
Thermal resistance Magnet-Air = 0.280 ohm
Thermal resistance Total = 1.28000 ohm
VC Thermal Coefficient = 0.00393 mm
Magnet Type = Alnico 5
Z min = 8.00 ohm
Driver Type = Midrange
Driver Model = AAA-85

Driver Volterra Parameters

Nonlinear_BL0 = 4.9
Nonlinear_b1 = 82.0
Nonlinear_b2 = -20000.0
Nonlinear_K0 = 28.8
Nonlinear_k1 = -6067.0
Nonlinear_k2 = 9000000.0
Nonlinear_Le0 = 0.0015
Nonlinear_l1 = 0.0873
Nonlinear_l2 = 10.0
Nonlinear_Rm = 2.6
Nonlinear_m = 0.014
Nonlinear_Eg = 17.0

Non-Linear Diff-EQ Parameters

Asymmetry BL (A1) = 0.00000
Flat/Peak BL (A2) = -0.02000
Wide/Narrow BL (A4) = -0.00050
Shift BL (AS) = 0.00000
Asymmetry Cas (B1) = 0.00000
Flat/Peak Cas (B2) = -0.00002
Wide/Narrow Cas (B4) = -0.00150
Shift Cas (BS) = 0.00000

Figure 1.9b. T/S parameters printout example.

Part 2

Working With SoundEasy Tools

Before progressing any further, please note, that your copy of CD-ROM contains three driver files, one project file and one FEM-type of data file. The files are fully edited and they offer you an excellent way of familiarizing yourself with all program's functions. For instance, when learning about drivers and enclosures, all you need to do, is to load a driver file and use screens and functions related to driver's editing, enclosure editing and modeling and optimization.

Using SoundEasy

Any computer modeling program requires you to provide it with some data to be processed by the program's functions and algorithms. SoundEasy is no exception. Ideally, you would provide the program with all necessary data – in this case, you would be able to utilize all built-in, powerful functionality. In many instances, you may be able to extrapolate some missing data, or use program's default settings.

The design concept, that SoundEasy follows, could be broadly described as shown below:

1. **Create library of drivers.** This should be really your starting point. For sealed/vented enclosure analysis, you can get going with only a minimum of several parameters entered. On the top of this, you will need SPL and impedance curves entered into the program to allow you for crossover analysis – see step 3. The EasyLab should be used extensively at this point to determine T/S parameters.
2. **Design enclosure for appropriate drivers**. At this point of the design process, you should be able to determine physical shape of your enclosure. This is typically accomplished by selecting enclosure internal volume, which corresponds to the SPL of your design goal. For a simple, rectangular enclosures, you can get very good design drawings from SoundEasy as well.
3. **Construct the enclosure and measure SPL/Impedance of the selected drivers**. The enclosure does not to be "commercially finished", but needs to have all sound affecting attributes in-place. At this point, you have all data needed for crossover design. You can use EasyLab at this point of the design process. The SPL and impedance curves need to be measured at this time.
4. **Design crossover**. Use provided CAD screen. You can start with a simple two-way built in crossover and learn from there.
5. **Perform listening tests using Digital Filter function of SoundEasy**. Here you have an opportunity to perform all tweaks and optimizations of the crossover in time and frequency domains.
6. **Built the crossover and perform final system assembly, measurements and optimizations**. Be prepared to spent significant amount of time in Steps 4,5 and 6.
7. **Analyze your listening room using FEM modeling system**. Predict the best placement for bass.
8. **Enjoy your new speaker system in action**.

It needs to be noted, that SoundEasy is capable of a lot more complex tasks than listed above. You will be able to master the program in time and learn quite a few "trick-of-trade" as you move forward.

Create Library of Drivers

You will need to edit all drivers you intend to use in your system. Here, we are talking about entering T/S parameters of the driver, entering the mechanical and non-linear parameters and entering or measuring it's SPL, impedance curves and non-linear parameters. One parameter, that is often considered "esoteric" and difficult to determine is the Acoustic Center Offset (AC-offset) of the driver. The EasyLab section of the Reference Manual shows you one possible method of determining the AC-offset. Having your drivers fully edited is very handy. You will be able to swap drivers in your projects at will, without worry about relative AC-offsets between the drivers. Editing of the driver is grouped into four tasks (four tabs) in "Driver Parameter Editor" control box.

Entering SPL and Impedance Curves

There are basically four ways of entering SPL/Zin curves into the program.

1. Using “Amplitude Model” tab
2. Importing data from measurement file
3. Hand-drawing the curves on the screen
4. Using EasyLab

At times, you will use combination of the methods.

Hand-drawing Data Point on the Screen

Frequency response can also be accomplished with the help of the mouse and cursor keys. The initial frequency response curve is a flat line, that defaults to 90 dB efficiency. Position the pointer arrow at the point with the corresponding coordinates to those on the data sheet frequency response at the same frequency and then click the LEFT mouse button. Then a little gray rectangle will move from the flat '90 dB line to the newly created frequency response plot. The Frequency/Amplitude coordinates, just above of the graph area, indicate the exact coordinates. It is recommended to "fine tune" the position of each data point using cursor keys which move the frequency response current data points by smaller amounts and are very useful after the broad frequency response has been entered. A manual way to edit the frequency response is to position the mouse pointer on the plotting area and press LEFT mouse button. Keep the button depressed and simply trace (draw) the frequency response on the screen. The resulting frequency response will be coarse, but then you can fine tune it using the cursor keys. Use "**Edit Amplitude**" button frequently to clean-up the plotting area and confirm, that none of the data points were left out. This process may seem time consuming. It does, however, need to be performed accurately because errors acquired during this initial phase, must affect all data processing performed by the other tools. It is performed only once per driver, during the creation of the data library. Having edited SPL curve you need to convert it into the driver’s amplitude and phase data. This is called creating a **Transfer Function** and is accomplished with the help of Hilbert-Bode Transform screen.

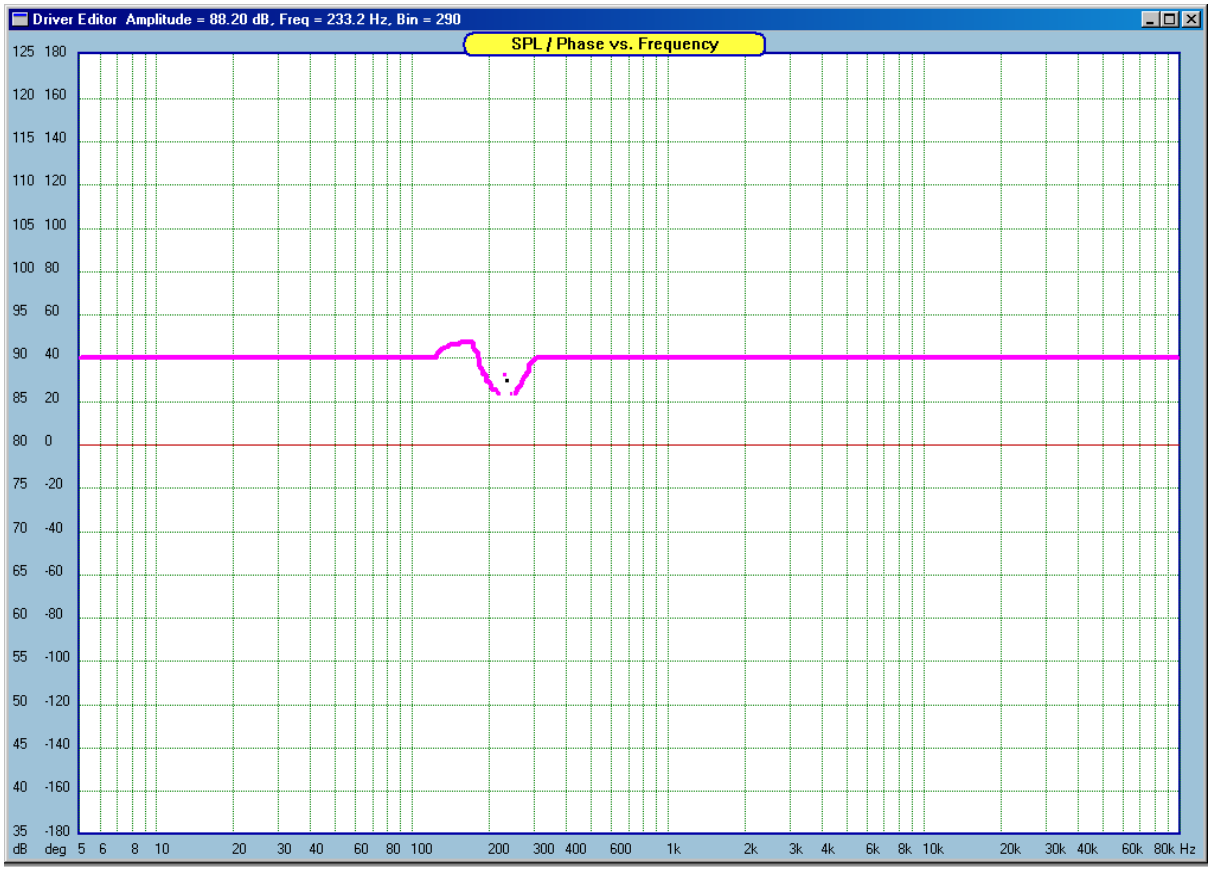


Fig 2.1. Hand-drawing SPL curve on the screen

What to do if you only have SPL graph ?

Driver frequency response plots, as obtained from the manufacturer, specify the test conditions under which the curve was obtained. In the majority of cases the frequency response is a graph representing the sound pressure as a function of frequency when the driver is fed a constant sine wave signal. That implies, the test amplifier was a voltage source, just like your Hi-Fi amplifier. Also, the test enclosure volume and type filling will be given. The test is usually conducted in an anechoic room with the pick-up microphone placed at 1 meter distance. It is almost certain, that the manufacturer's test enclosure type and volume will be different from the one you attempt to design. However, it is all that you have and it does contain valuable information for SoundEasy. It is then recommended, to follow the procedure outlined below. Remember, the goal here is to obtain driver's complex transfer function.

1. Enter all electromechanical parameters of the driver using 'Editor' screen.

2. Invoke 'Enclosure Design' screen and plot the frequency response of the driver selecting exact parameters of the manufacturer's test enclosure.

3. Design the enclosure for your application and plot the resulting frequency response on the same graph as the manufacturer's box frequency response. You now have two graphs on the same plotting area.

4. Then compare the two plots. What is important is the difference between them. It is recommended to have a hard copy of the two plots (printout). You will notice, that for certain frequencies, your box frequency response was drawn above manufacturer's box frequency response (the difference is positive) and for other frequencies, it was drawn below (the difference is negative). Print this screen for reference.

For example:

Frequency	**Manufacturer's box response**	**Your box response**	**Difference**
30.0 Hz	75.2 dB	81.0 dB	+5.8 dB
32.0 Hz	77.0 dB	84.0 dB	+7.0 dB

and so on....

5. Now you can go back to the 'Editor' screen and enter the frequency response required by this module. It is entered as the manufacturer's anechoic room test frequency response plus the difference between their test enclosure and your chosen enclosure (values in column 4 above) That is, for every frequency step offered by the 'Editor' module (occasionally, you may have to approximate your readout from the manufacturer's box frequency response, since SoundEasy offers discrete and fixed frequency steps) you need to enter corresponding value of the manufacturer's anechoic room test frequency response plus the difference above.

6. While entering the final frequency response, you must preserve all the small ripples and irregularities that are evident in the manufacturer's anechoic room test frequency response. This method may require modification if the test microphone is placed at .5 meter instead of 1 meter, but the principle for obtaining the accurate frequency response remains the same. In order to correctly predict the summed magnitude response of the complete speaker system, the magnitude and phase responses of the individual drivers must be known. Using the 'Transfer Function' screen, it is feasible to develop accurate, minimum-phase complex transfer functions of the sound pressure and input impedance of the individual drivers. Previously entered frequency response becomes a reference data for Hilbert-Bode Transform. The transform calculates phase response from the given amplitude response. During this process, SoundEasy automatically develops the complex transfer function for the driver.

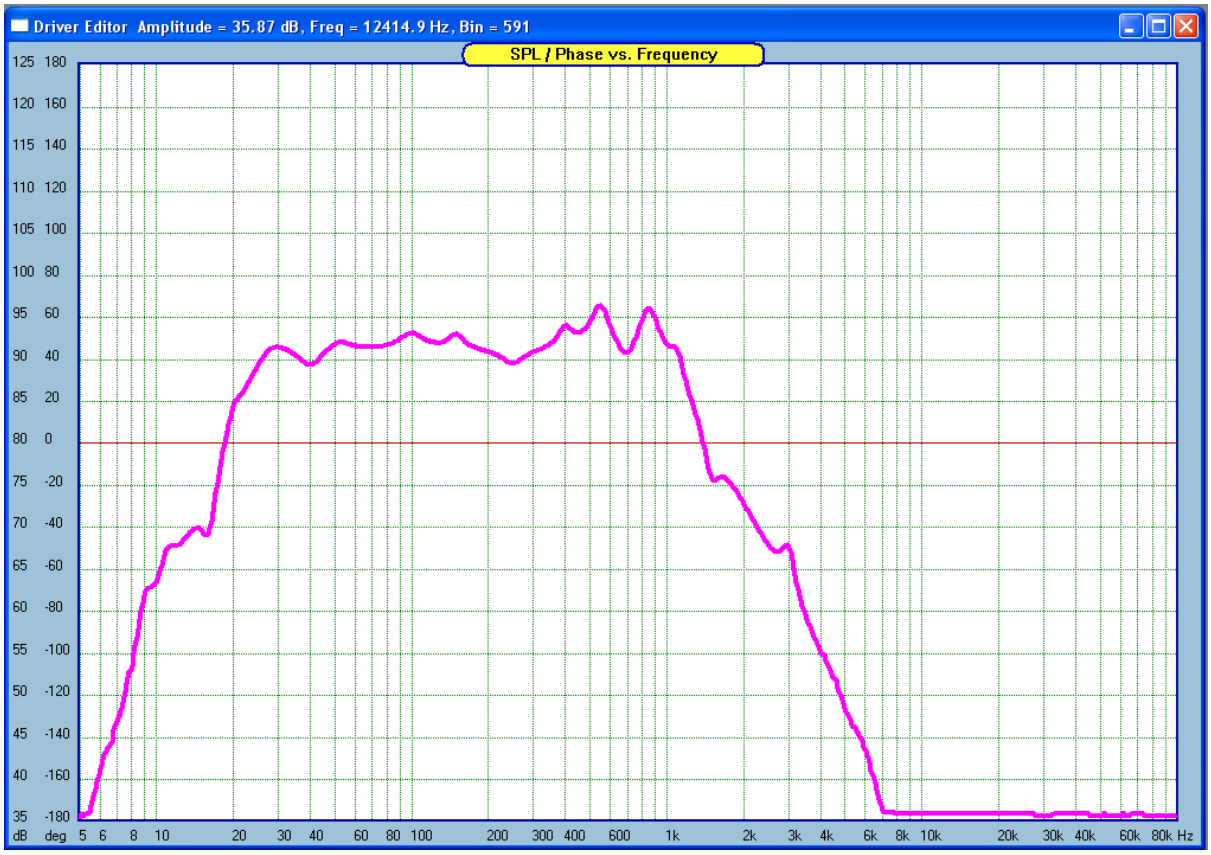


Fig 2.2 Woofer driver – SPL.

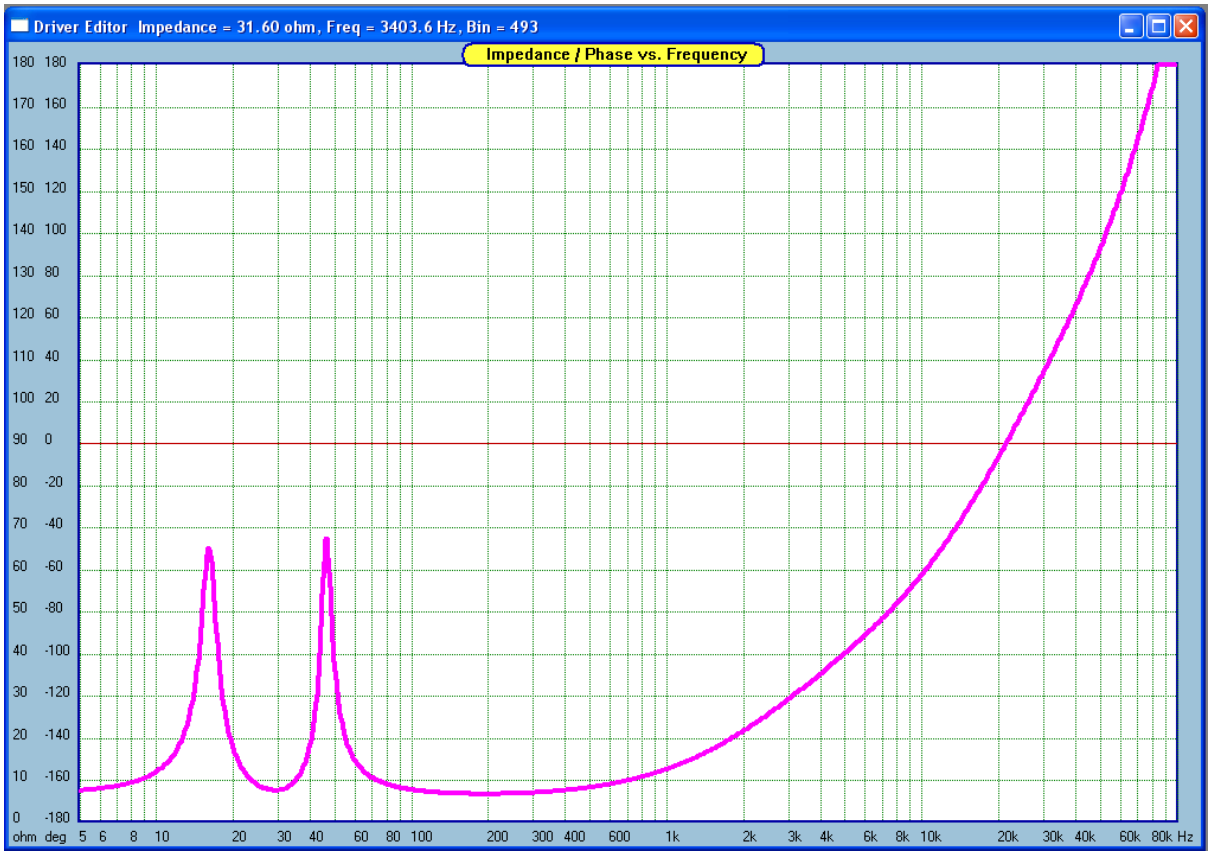


Fig 2.3 Woofer driver – Impedance.

Entering Driver's Data

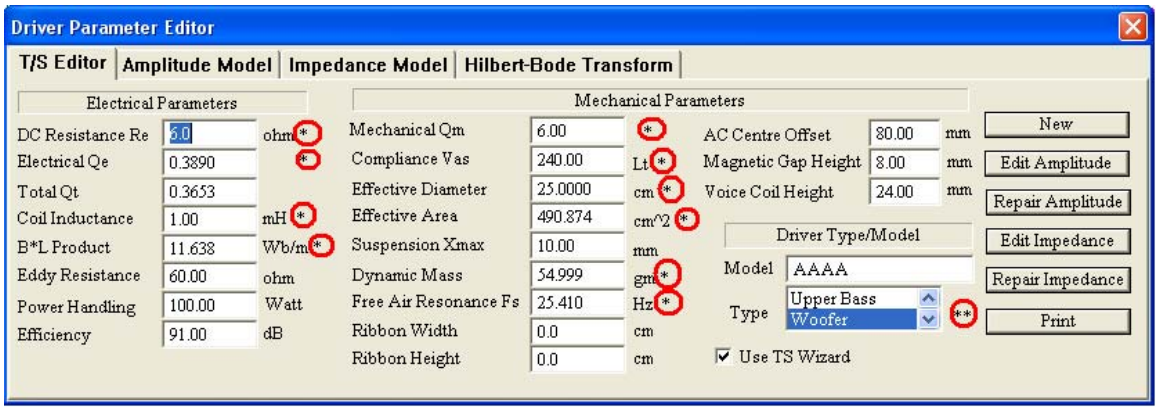


Fig 2.4. Driver Parameter Editor (T/S parameters) dialogue. Parameters with asterisk, " * " are essential.

When the 'Driver Editor' menu option is opened, you will be presented with the window shown in Fig 2.4. The window contains 30 editable data entry fields and a plotting area to enter the frequency response. To activate a data entry field, move the pointer arrow above the field of interest and click LEFT mouse button. A blinking cursor will appear within the data field and you can enter the required parameter. All data entered must be in positive, real numbers. Alphanumeric characters are not allowed. All data must be entered in the units of measure indicated on the right of the data fields.

The dialogue box above shows essential parameters you will need to key-in, in order to plot basic responses of sealed and vented enclosures. Full linear and non-linear analysis of a driver in an enclosure requires you to enter all relevant parameters of the driver and the enclosure(s). Another (recommended) way of acquiring the T/S parameters is to use EasyLab and simply measure them. The T/S parameters are only needed if you intend to perform enclosure design or THD analysis.

Electrical Parameters

1. DC Resistance Re.....DC resistance of the voice coil. Can be measured using an ohmmeter.
2. Electrical Qe.....Electrical Q-factor of the driver. Should be listed in the driver's data chart or can be measured. **NOTE: Qe=2*PI*Fs*Md*Re/(BL*BL)**
3. Total Q.....Total Q-factor being a combination of Qe and Qm.
4. Coil inductance.....Inductance of the voice coil. Should be listed in the driver's data chart or can be measured.
5. B*L Product.....(BL) Also known as a "force factor" of the driver's motor system. Should be listed in the driver's data chart or can be measured. **NOTE: BL=sqrt(2*PI*Fs*Md*Re/Qe)**
6. Eddy resistance.....This parameter may not be listed in the data charts. Eddy resistance represents loss in the pole piece and front plate. User should be able to measure eddy resistance at 20-100kHz by measuring impedance magnitude at 20kHz and subtracting voice coil DC resistance from it.. If in doubt, consult [26].
7. Power handling.....Should be listed in the driver's data chart or can be measured.
8. Efficiency.....Should be listed in the driver's data chart or can be measured. Enter in dB's.

Mechanical Parameters

9. Mechanical Qm.....Mechanical Q-factor of the driver. Should be listed in the driver's data chart or can be measured.
10. Compliance Vas.....Equivalent compliance volume of the river. This is the volume of air that has the same compliance as the driver's suspension system. Should be listed in the driver's data chart or can be measured.

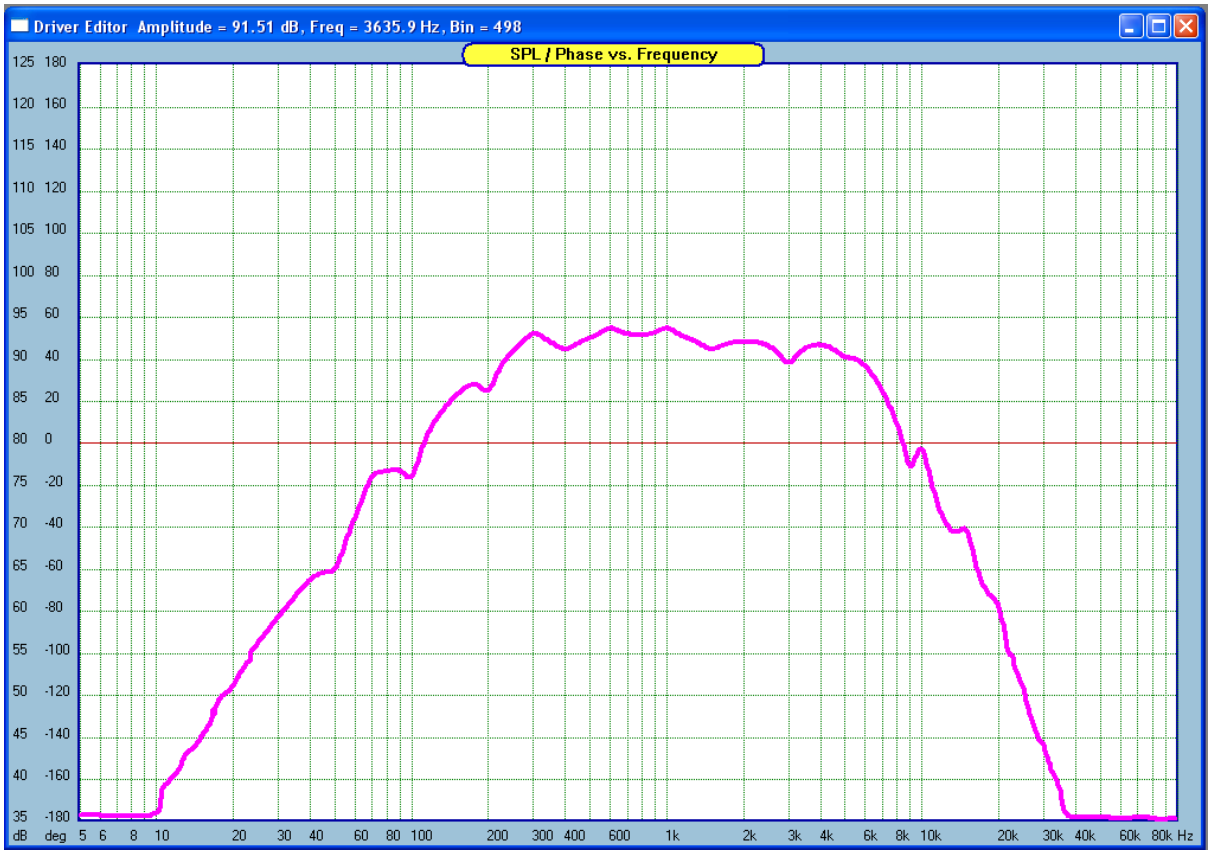


Fig 2.5 Midrange driver – SPL.

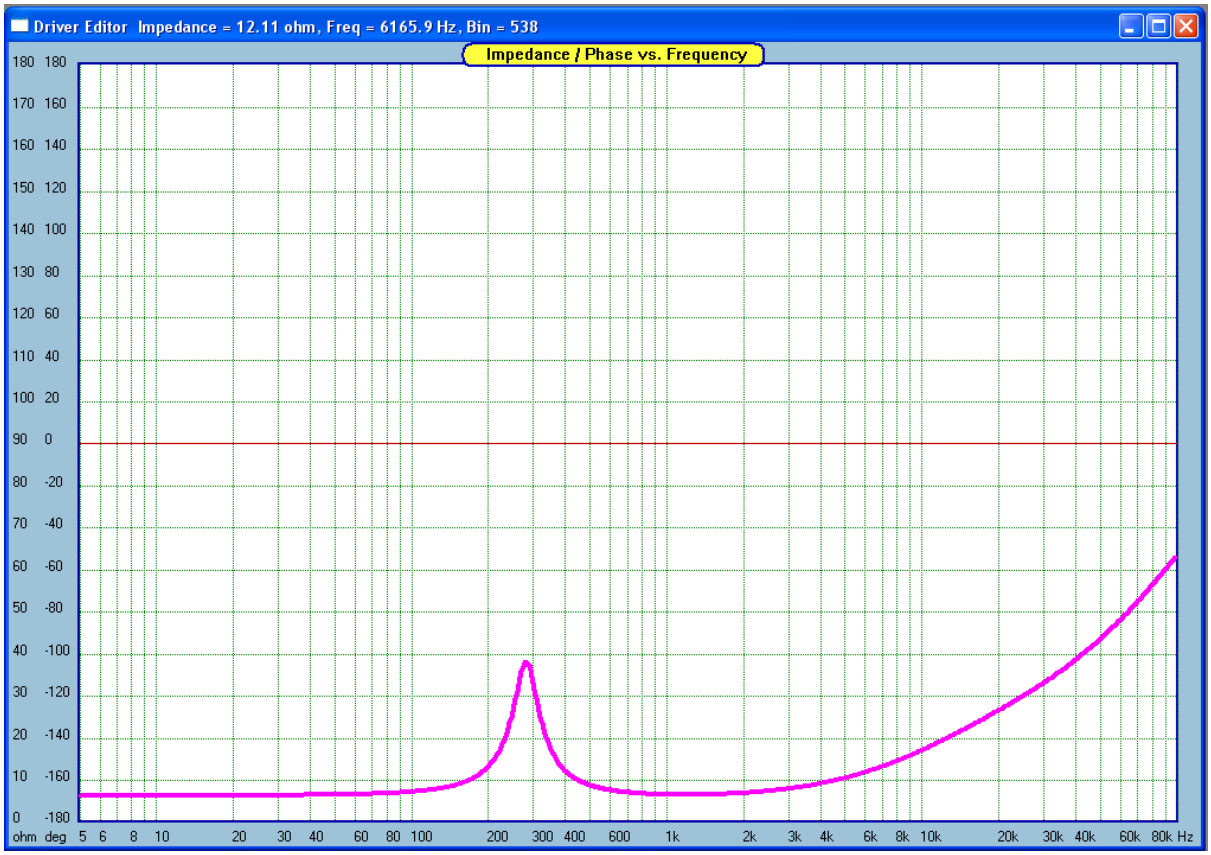


Fig 2.6 Midrange driver – Impedance.

11. Effective Diameter...Usually, the effective diameter is 75-85% of the basket diameter. If you don't know what the effective diameter is, enter this parameter as 80% of the basket diameter. You may notice, that SoundEasy attempts to check the entered data and it will modify other fields is sufficient data has been entered. Later on, when all other parameters are entered, you may need to change this data such a way that Qe and Fs, recalculated by the software, are exactly as listed in the data sheets of the driver.
12. Suspension Xmax.....Maximum linear cone displacement due to spider. See page 2.16 for details.
13. Suspension Type…..Select from the list box. See page 2.16 for details.
14. Dynamic mass Md...(Md) Mass of the whole moving assembly of the driver + "air loading" of the driver.
15. Free air resonance...(Fs) Should be listed in the driver's data chart or can be measured.
NOTE: Fs=Qe*BL*BL/(Re*Md*2*PI)
16. Center offset.....Acoustic center offset, measured from the baffle (mounting) plane of all drivers. An approximation would be the distance from the mounting flange to the voice coil.
17. Gap Height.....…...Magnetic gap height. Should be listed in the driver's data chart.
18. Coil Height..……Height of the voice coil winding. Should be listed in the driver's data chart.
19. Ribbon Width
20. Ribbon Height

Thermal Resistance Data

21. Coil to Magnet Rvc..Thermal resistance between coil and magnet structure.
22. Magnet to air Rmt....Thermal resistance between magnet and air.
23. Total Rtot.....…Calculated by SoundEasy. Rtot=Rvc+Rmt.
24. Coil Temp Coeff.....Resistance increase per deg C.
25. Magnet type.....….Select from "Magnet Type" list box. Determines loss of magnetic flux as follows:
(1) Alnico 5.....loss=2.3% per 100 deg C, (2) Crumax 355.....loss=11.5% per 100 deg C and
(3) Ceramic 5.....loss=17.4% per 100 deg C.
26 Minimum Z.....…..Enter "8.0" for eight ohm driver, "4.0" for four ohm driver and so on...

Model/Make

27 Driver Type.....…...Select from "Driver Type" list box.
28 Driver Model.....…...Enter model name.

Other Data Required

29 Input Power…….Entered from "Power Compression Analyser"
30 Impedance model….**It is essential to select and fully edit impedance model that you wish to use**. Data entered from "Impedance Models" dialogue box.

NOTE 1: Entering the Driver Type (Woofer, Upper Bass, Midrange, Tweeter and Super Tweeter) is particularly important. All other SoundEasy modules will check this field and default according to this simple piece of data.

NOTE 2: If you enter Suspension Xmax = 0.0, the program assumes you do not wish to perform large-signal analysis due to suspension nonlinearities. If you enter Gap/Coil Height = 0.0, the program assumes you do not wish to perform large-signal analysis due to motor system nonlinearities.

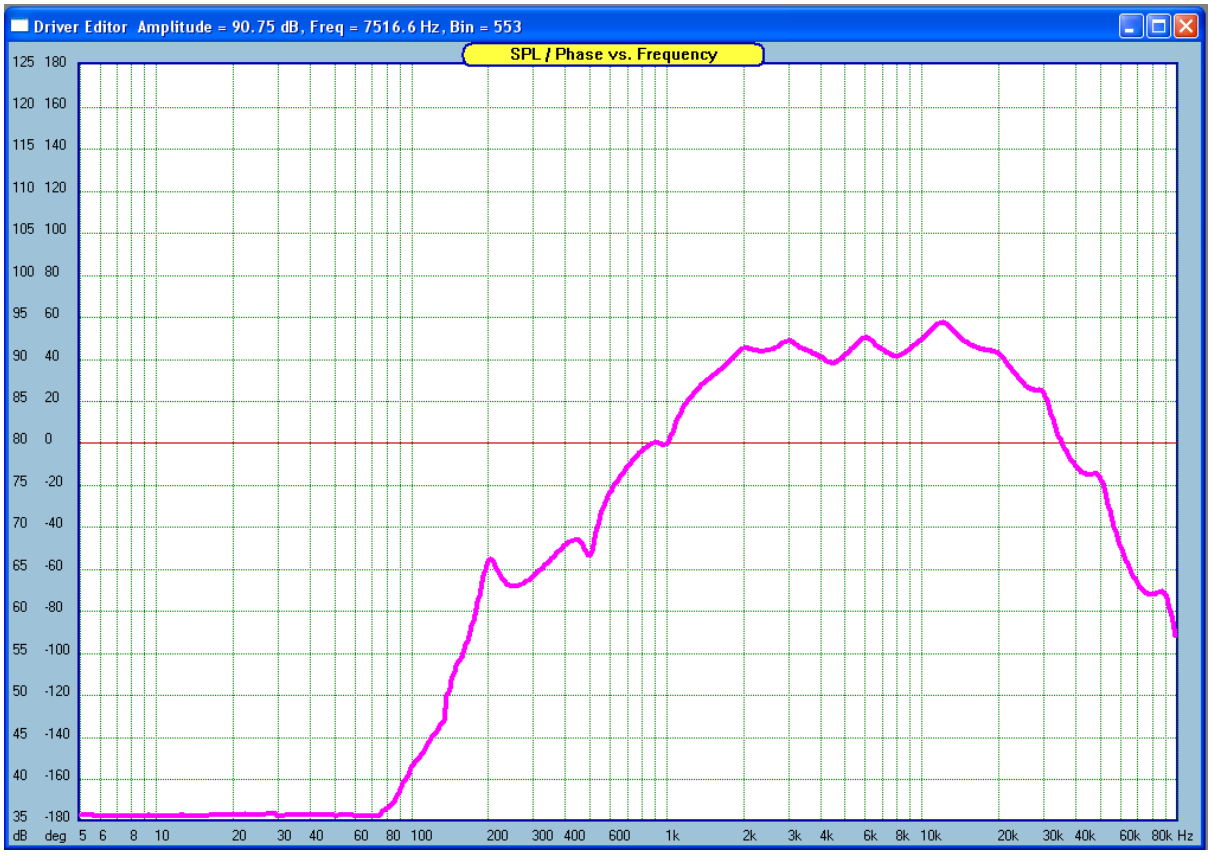


Fig 2.7 Tweeter driver – SPL.

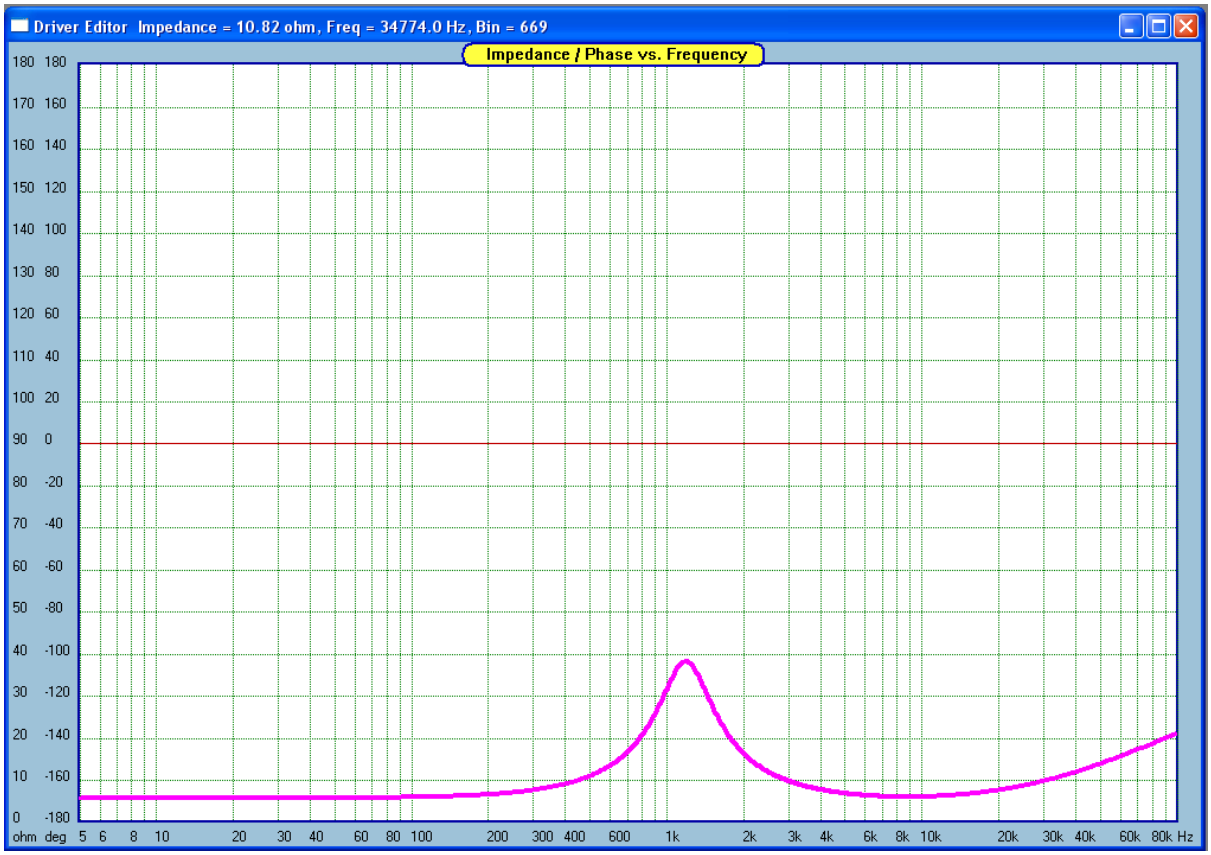


Fig 2.8 Tweeter driver – Impedance.

Entering Frequency Response

In general, frequency response of all electro-dynamic loudspeakers can be compared to a band-pass filter. If all small ripples on the frequency response are disregarded, it is easy to observe the three main regions of the curve: (1) rising slope below the free-air resonant frequency followed by (2) reasonably flat section which is most useful to the designer and finally (3) roll-off section, where the driver loses its ability to maintain cone excursion. With the above in mind, there is a good argument in commencing the modeling of the frequency response as a band-pass filter and then adding all small individual ripples. This method can be very accurate if enough ripples are added. However, for many applications, a good approximation can be accomplished with as little as 10-15 ripples added to a band-pass filter frequency response. The only additional information necessary for the user to supply is the SPL efficiency parameter entered from the main screen. This will determine the overall SPL level of the whole curve. Please enter 90.0 in the "Efficiency" data field now. Entering frequency response is accomplished with the help of "Amplitude Model" tab box. The box is shown on Figure 2.8.

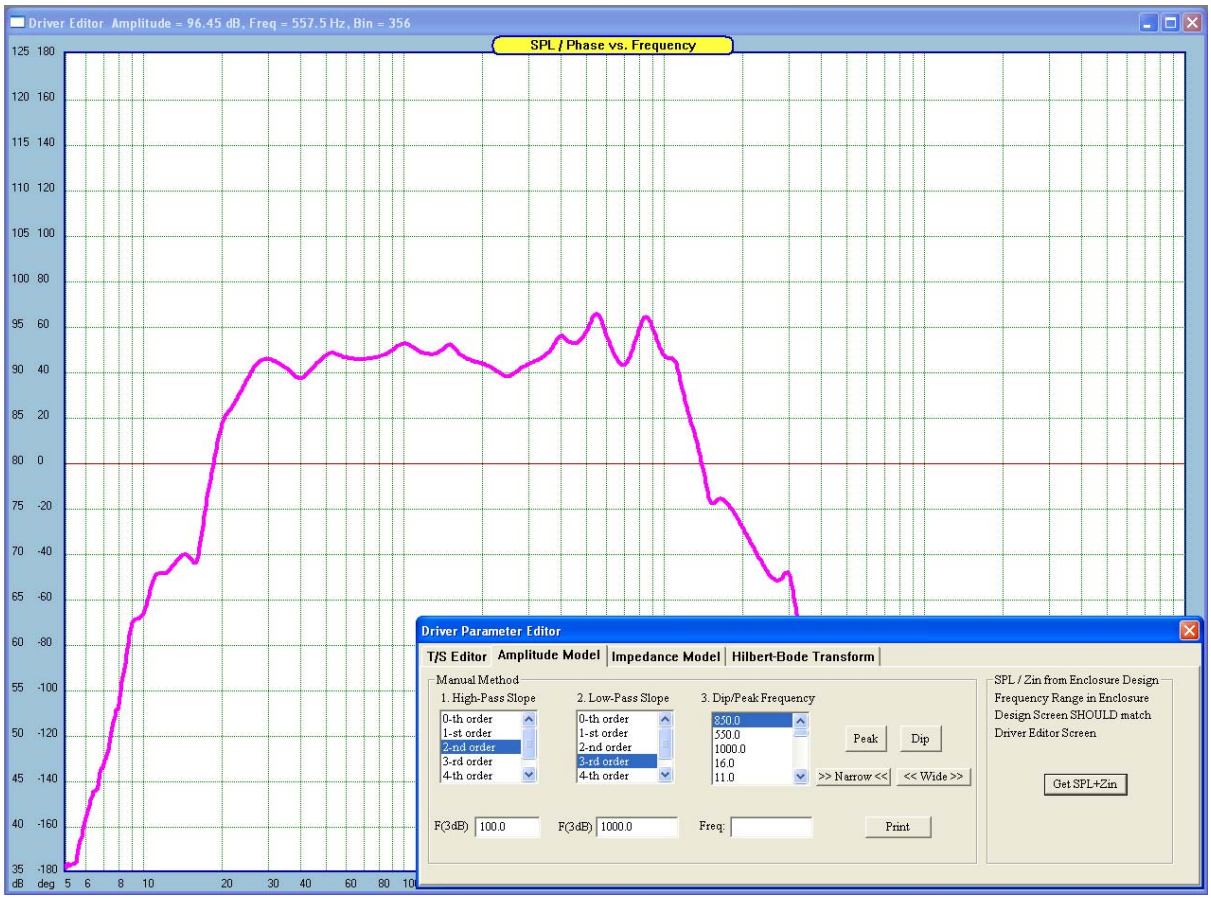


Fig 2.8 Edited SPL with a number of dips and peaks.

After examining the frequency response of the driver, it should be possible to approximately determine the three regions and order of the rising (high-pass) and falling (low-pass) slopes. Having done this, you need to select those orders by clicking left mouse button above the appropriate list box titled "1. High-Pass Slope" and "2.Low-Pass Slope". The associated cut-off frequencies of the filter approximations need to be entered in the associated data fields below the list boxes. Figure 2.8 shows 2-th and 3-rd orders respectively. The frequency response of our band-pass approximating filter can now be displayed by pressing on "→ ←" (means narrower) or "← →" (means wider) buttons. Next step is to add all small ripples onto the band-pass filter curve. The default values for all **50 available** ripples is 1000.0 and are displayed in the "3. Dip/Peak Frequency" list box. Please click on the first list box item – 1000.0 Hz. This will select the frequency to be edited in the data field available below the list box. Now, edit the frequency by entering the right value, say for the most-left ripple – you will see the value in the list box also changes. Now, you have four buttons available to determine the shape of the ripple: (1) "Peak", (2) "Dip", "→ ←" and "← →". Pressing repeatedly on all four buttons will shape the ripple and automatically re-plot the frequency response. Once the shape of the ripple has been finalized, please click on the second item in the list box and edit the second ripple frequency and determine it's shape. Please continue till you are satisfied with the total frequency response

When finished, click on “Done” button. When using the above method, SoundEasy will automatically generate driver’s transfer function with NO ACOUSTIC DELAY – that is driver-microphone distance equal to 0.0[m]. Should you require the standard 1W/m SPL result, please use “Transfer Function” screen and set the microphone distance accordingly. Entering frequency response can also be accomplished with the help of the mouse and cursor keys. The initial frequency response curve is a flat line, that defaults to 90 dB efficiency. The user positions the pointer arrow at the point with the corresponding coordinates to those on the data sheet frequency response at the same frequency and then clicks the LEFT mouse button. Then a little gray rectangle will move from the flat '90 dB line to the newly created frequency response plot. The Frequency/Amplitude coordinates, just to the right of the graph area, indicate the exact coordinates. It is recommended to "fine tune" the position of each data point using cursor keys which move the frequency response current data points by smaller amounts and are very useful after the broad frequency response has been entered. A manual way to edit the frequency response is to position the mouse pointer on the plotting area and press LEFT mouse button. Keep the button depressed and simply trace (draw) the frequency response on the screen. The resulting frequency response will be coarse, but then you can fine tune it using the cursor keys. Use "Edit Amplitude" button frequently to clean-up the plotting area and confirm, that none of the data points were left out. This process may seem time consuming. It does, however, need to be performed accurately because errors acquired during this initial phase, must affect all data processing performed by the other tools. It is performed only once per driver, during the creation of the data library. The better the accuracy of the data the better the design. Finally, amplitude response entered this way, **must be followed by the creating the transfer function**. An example of a fully edited driver is given in Fig 2.2, 2.5 and 2.7. These are the test files included with SoundEasy. The "New" button is used to clear all data fields and the graph plotting area. The Amplitude Model tab allows you to approximate the basic shape of the SPL curve and add up to 50 little ripples onto the curve. You can select High-Pass/Low-Pass slope and corner frequencies to get he basic “band-pass filter” shape and then adjust dips/peaks frequencies and shape using the “Narrow” and “Wide” button. All SPL curves provided in the sample data files were edited using this technique. Having edited SPL curve you need to convert it into the driver’s amplitude and phase data. This is called creating a Transfer Function and is accomplished with the help of Hilbert-Bode Transform screen.

Hilbert-Bode Transform (HBT) Tab

HBT enables to following functions to be implemented in SoundEasy:

1. **Determination of the Acoustic Centre of the driver.** HBT calculates the minimum phase response of the driver that is subsequently used a reference for distance estimation.
2. **Digital Equalizer function.** HBT is used to alter driver’s SPL in a mathematically correct way and consequently eliminate “tails issue”.
3. **MLS system**. HBT calculates minimum-phase response for SPL and Zin. HBT also performs filter-like clean-up of the SPL/Phase responses.
4. **Analog Measurement System**. This system does not account for phase. HBT calculates minimum-phase response for all SPL and Zin measurements.
5. **Importing SPL/Phase data**. HBT is used to repair poorly executed phase measurements.
6. **Importing SPL/Zin data**. HBT is used to extend imported data over any desired frequency range.
7. **Optimization**. HBT calculates phase response for any user defined Target SPL curve.

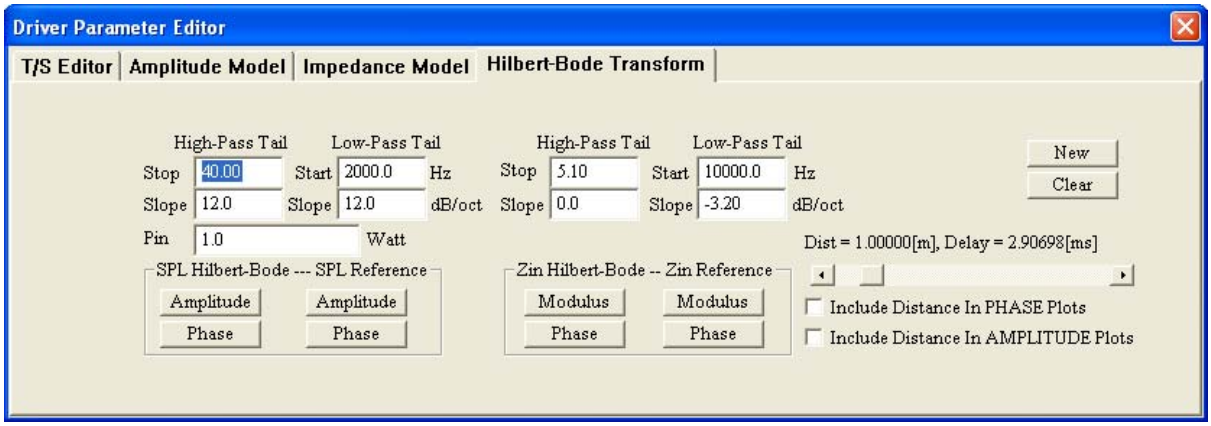


Fig 2.9. Hilbert-Bode Transform Control tab – note typical HBT parameters for Zin.

The “Hilbert-Bode Transform” tab has a built in very powerful algorithm - The Hilbert-Bode transform. It allows you to quickly and accurately recreate phase response from the **available amplitude or input impedance response**. The Hilbert-Bode transform really makes importing of the phase response irrelevant. the process of creating the true acoustic phase response with SoundEasy is instantaneous, this is why the imported phase response may only be used as a template. Also, ANY POST-PROCESSING of the amplitude response on the Editor screen will make the imported phase invalid. The minimum phase systems have a unique relationship between amplitude and phase, this relationship is captured the Hilbert-Bode mechanism. **If you need to extend the measured SPL/phase curves beyond the measurement frequency range**, the tool to use in this case, is the Hilbert-Bode Transform group-boxes provided separately for SPL and Zin Transforms. The transforms are driven by 4 editable parameters and they should be selected to obtain the **best match for phase and amplitude/impedance between measured signal and calculated transform over the widest frequency range**. Typically, good match can be obtained between 20Hz and 20kHz. Generally, you would select the parameters to provide initial match for amplitude response and then fine tune the parameters to provide the match for phase. Please note, that “Roll-off” parameters can be as high as +/- 100dB to +/- 120dB if the measured frequency response is highly irregular (resonant peaks) at the extremes of the measurement range.

Fig 2.10 Woofer transfer function - Hilbert-Bode Transform

The program calculates phase using a Hilbert-Bode transform from any magnitude/impedance curve. This is known as a Minimum Phase response and it does not contain any delay or positional information. Because the Hilbert-Bode Transform requires an infinite integral over all frequencies, any numerical computation of minimum phase is necessarily an approximation. Understanding the characteristics of the phase transform and how it works can be very important if maximum accuracy is to be obtained. For some data curves, the transform can be run directly and will provide excellent results with little attention to the original magnitude data. In other cases, a far better result can be obtained by correcting certain areas of the magnitude curve before running the minimum phase transform. Typically, the regions which can cause problems are **at the ends** of the measurement bandwidth. The transform must project the slope of the magnitude curve beyond the measured bandwidth. It does this by taking the last slope found as the curve reaches the low and high ends of the frequency range. It then assumes that: (1) the slope found at the low end will remain constant down to DC, and (2) the slope found at the high frequency limit will remain constant towards infinity. For this reason it is of fundamental importance that the slope of the magnitude curve at the high and low ends of the frequency range be approaching or already at their final asymptotic values. The Transfer Function [HBT] dialogue box has the following controls:

1. **"New"** – use this button to restore default setting for this screen.
2. **"Clear"** – use this button to clear the screen.
3. **SPL Hilbert-Bode**
- "Amplitude" – use this button to calculate transfer function and display it as amplitude
- "Phase" – use this button to calculate transfer function and display it as phase.
4. **SPL Reference**
- Amplitude" – use this button to display amplitude of the imported data
- "Phase" – use this button to display phase of the imported data
5. **Zin Hilbert-Bode**
- "Modulus" – use this button to calculate transfer function and display it as modulus
- "Phase" – use this button to calculate transfer function and display it as phase.
6. **Zin Reference**
- Modulus" – use this button to display modulus of the imported data
- "Phase" – use this button to display phase of the imported data
7. **"Distance/Delay"** – use this slider to select appropriate distance between the transducers.
8. Check box to include the distance in amplitude calculations.
9. Check box to include the distance in phase calculations.
10. **High-Pass Tail and Low-Pass Tail group boxes** - The low-end slope and stop frequency, the high-end slope and start frequency. Enter the appropriate data as suggested above for amplitude and impedance.

The Hilbert-Bode Transform is used for amplitude as well as for impedance transfer functions. You need to perform tail corrections for both types of curves.

Fig 2.11 Woofer transfer function @ 10cm mike distance – see the "Check" boxes above.

At times, it may be convenient to **add/subtract time delay** to the imported data or scale the whole amplitude response curve up or down by a fixed amount of dB's. The "Import" dialogue box provides the user with two editable fields: (1) "Delay", which accepts the required positive or negative time delay in milliseconds and (2) "Scale", which accepts the amount of positive or negative shift expressed in decibels. Positive delay means moving the driver away from the test microphone and negative delay means moving the driver towards the test microphone. The third editable field in the dialogue box is the reference SPL level (Efficiency).

The "Delay" and "Scale" data are optional, but the "Efficiency" data must be entered - some of the measurement systems only provide RELATIVE to SPL amplitude response data.

Sometimes, your measurement environment will add 6dB to the amplitude response and in those cases, SoundEasy provides you with some helpful tools to rectify the problem. Having accomplished the above, SoundEasy is ready to import the requested file and then convert the imported amplitude and phase templates to Complex Transfer Function (A+jB) format - see also Appendix - Importing Data Files. Driver's complex transfer function does not include any additional filtering. You would probably hesitate before feeding 1 Watt input power at 10Hz into the tweeter or even midrange driver. Yet, the program does require the transfer function to be determined at those extremes. This is at least one compelling reason for providing several tools that enable the user to recover or recreate these extreme data points. In practice, most of the imported files will be incomplete for the purpose of using them with the program. That is why the imported data is not used directly, but treated as templates for Hilbert-Bode transform.

Entering Impedance Data

Manual entry of impedance curve is accomplished much the same as the previously described amplitude response - drawing the curve and then using cursor keys to fine tune the plot. The default impedance curve is plotted at 1 ohm level. At any point of time, the user can flip between frequency response and impedance curve using 'Edit Amplitude' and 'Edit Impedance' buttons. The currently edited curve will be saved and the newly selected one plotted on the screen. Once again, the 'Edit Impedance' button should be used to clean up the display. See Fig 2.12. Once the whole (5Hz - 100kHz) impedance curve is edited, it can be used as a template for recovering complex impedance. Editing impedance curve over the whole frequency range is important. Slope Correction is also provided for the impedance. This is accomplished via 'Hilbert Transform Function' screen, which was also used for recovering amplitude transfer function.

Impedance Transfer Function

As with the amplitude transfer function, it is necessary to develop complex input impedance of the driver (or driver+box). Here are a few ways this can be accomplished:

1. Importing impedance/phase data from one of the supported measurement systems. Ideally, the imported data will cover the required frequency range. If not, the user must manually edit missing parts and recover complex input impedance using Hilbert-Bode Transform.
2. Manual entry of the impedance curve - see previous paragraph.
3. Using provided Impedance Models. Once a model is selected and edited, this information must be saved in the data file.

Complex impedance curve or impedance model data will be used by two functions: (1) 'Enclosure Esign' screen when plotting impedance of a driver in a designated enclosure and (2) 'Crossover CAD' functions, as a load impedance of the compensation circuit. "Crossover' module will first check if an imported impedance data is available and will use it, if provided. It is therefore necessary to provide quality data. In order to develop complex impedance function, you can start from 'Editor' screen by pressing 'Edit Impedance' button. This will bring up impedance curve onto the plotting area. The impedance curve must have been measured, imported or manually edited before. Next step is to make sure, that whole curve is properly edited over the 5Hz - 100kHz frequency range. If so, the template is ready. See Fig 2.3. Hilbert Transform Function' screen should be invoked next and the complex impedance can be produced by pressing 'Impedance' button under 'Impedance Transform' group of buttons. You will notice, that 'Slope Correction' is not provided for the impedance transform as it is not necessary. The two buttons: 'Impedance' and 'Phase' under 'Imported Data' heading can be used to replot imported curves.

Manually Entering Zin Impedance Data

Entry of impedance curve is accomplished much the same as the previously described amplitude response - drawing the curve and then using **cursor keys** to fine tune the plot. The default impedance curve is plotted at 1 ohm level. Once again, the '**Edit Impedance'** button should be used to clean up the display. Once the whole (5Hz - 100kHz) impedance curve is edited, it can be used as a template for recovering complex impedance.

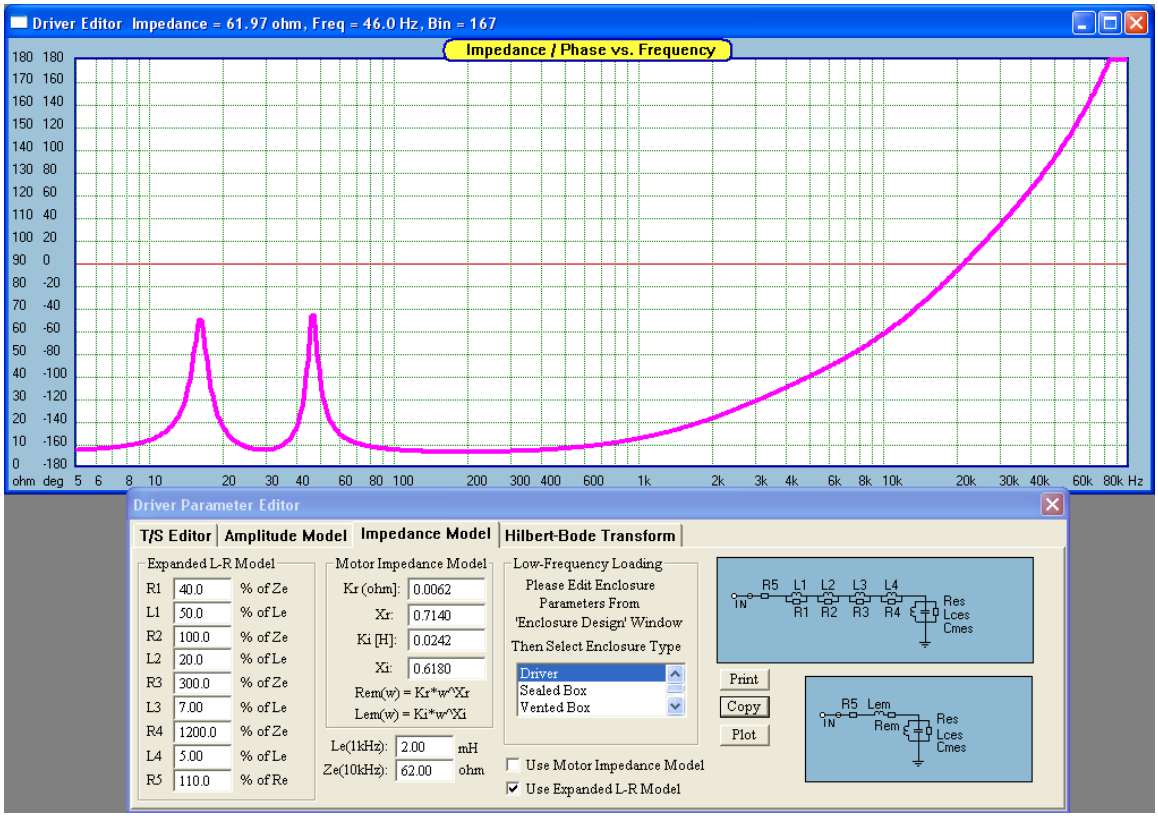


Fig 2.12 Impedance Models

Impedance Models

Two editable impedance models: (1) Quad-LR model and (2) Motor Impedance model, are provided and can be accessed via the main menu. See Fig 2.12. The Quad-LR model requires voice coil inductance as provided by data sheets, often measured at 1kHz and impedance minus DC resistance measured at 20kHz. The four LR sections are then assigned a **percentage** of the values above to fit any given impedance curve. Also R5 in the model can be different from the DC resistance of the driver.

The 'Plot' button plots the curve**. Caution !, at this point of time, the imported impedance data is replaced by the data generated by the model**. Component values in this model should be adjusted for each driver individually. This model is useful, when the impedance curve is actually available on the data sheets. The second model - Motor Impedance model - was developed by J.R. Wright in 1989. This model requires specific parameters to be provided and this is best accomplished by specialized measurement system. The "low end" of the impedance curve is determined by the acoustic loading (enclosure) selected for this particular driver. Enclosure type and parameters can be edited from the same dialogue box. You must provide these parameters in order to obtain correct shape of the impedance curve. Tank circuit elements for the driver alone are calculated for both models as follows: Res = Re*Qm/Qe; Cems = (Ms*Sd*Sd)/(Bl*Bl); Lces = (Cs*Bl*Bl)/(Sd*Sd). If you do not provide data for Le(1kHz) and Ze(10kHz) in the "Impedance Models" dialogue, the program automatically inserts "Coil Inductance" and "Eddy Resistance" values respectively.

If no impedance data is imported, it is essential to select and edit one of the models in order to provide correct data for subsequent modules, secondly, nominating and editing a model is necessary for plotting impedance and power demand curves of ALL enclosures. Using "Exit" button saves entered parameters, but does not replace imported impedance data.

We suggest, that you open the "Impedance Model" dialogue box and select an impedance model for "driver" before entering the enclosure design window. You may also select the Motor Impedance model at this point as the parameters of the model are provided as default values and do not require additional input.

Speaker-Microphone Distance

Driver's transfer function would normally be measured using a microphone placed 1 meter away from the driver. This distance is controlled by the scroll bar at the bottom-right corner of the screen. The actual distance and time delay for the sound to travel this distance is shown next to it. The Phase Response plotted by the module is the phase difference between the signal produced by the signal source (your amplifier) and the acoustical pressure developed by the speaker and measured at the specified distance. The distance is simply included in the phase delay. SoundEasy defaults to 1 meter distance and 1 Watt of the input power to the loudspeaker. The speaker-microphone distance can be varied within 0.0mm to 10.0 meter range. You will notice however, that as the distance gets shorter, the frequency response will be moved higher up on the plotting area. The sound level will increase by 6dB per half the distance. In order to keep the recovered amplitude response over the pink "template" you need to reduce the input power to the loudspeaker, when the microphone is placed closer to the speaker. Fig 2.11 shows a case, where the microphone was placed 10cm away from the speaker and the input power held at 1 Watt. Please note the difference in the amplitude and phase responses. Now, since the distance is 10 times shorter, there can only be a few wavelengths between the microphone and the speaker, as opposed to the 1 meter separation. You can also **INCLUDE/EXCLUDE** the speaker-microphone distance from the transfer function by selecting this option from the two "Check" boxes at the bottom of the screen.

Input Power and Rg

The mechanical structure closest to the voice coil is the magnet system and the thermal interaction between these two determines **Rvc**, the voice coil-to-magnet thermal resistance. Once the magnet heats up, it will radiate the heat directly into the surrounding air. The losses of this process are modeled by **Rmt**, the magnet-to-air thermal resistance. As one can see, the magnet, being larger structure and directly exposed to the air, has lower thermal resistance. The structure of the magnet will lose some of its flux due to the raise of its temperature. This process is reversible, but will cause reduced SPL level. Another factor reducing the SPL is the raise of the voice coil resistance. The resistance increase per degree C is given but the **Ktmp** factor. For copper and aluminum wires **Ktmp**=0.00393 and for a 100.0 deg C temperature raise, there will be almost 40% increase in voice coil resistance (Re), (**Rtot = Rvc+Rmt**). Increased Re will cause less current to be drawn from the amplifier and also increase in electrical Q-factor (Qe). All the data entered on the "Power Compression Analyzer" are real, positive numbers. Magnet type determines the rate of the flux loss. Generator's internal resistance should be entered into the **"Rgen"** field. Input power into the modeled loudspeaker is edited from the **"Pin"** data field. "**Zmin**" is the nominal speaker impedance and should be entered as one of the standard impedances such as: 2 / 4 / 8 / 16 ohms.

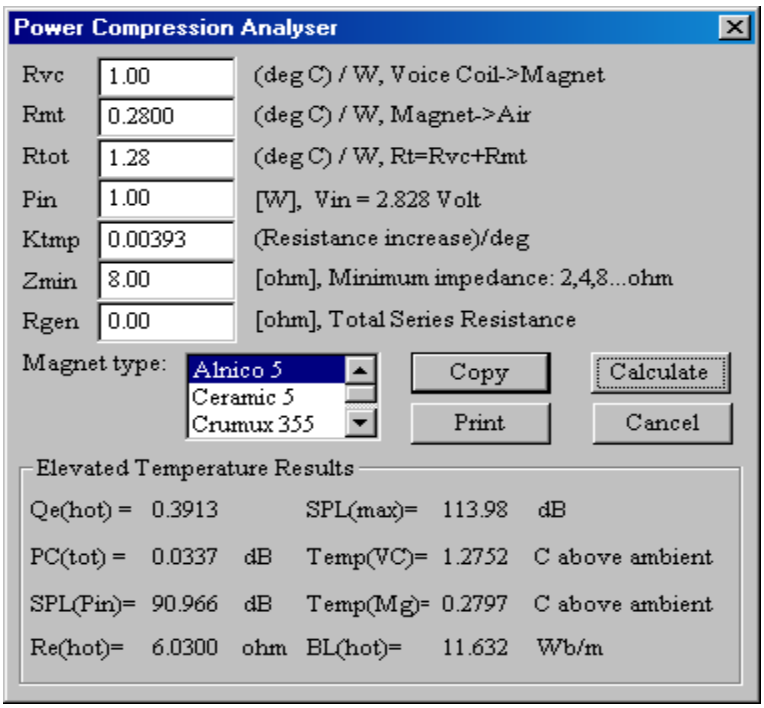
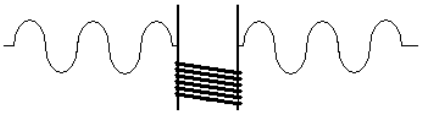


Fig 2.13. Power Compression Calculator

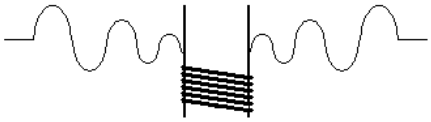
Suspension System - Spiders

Loudspeaker spiders are most often made from resin impregnated cloth. The cloth has corrugation rolls pressed into it, to allow for vertical (ideally) extension of the spider and voice coil movement. There are three major spider profiles in use today.

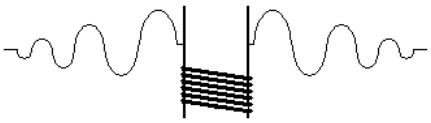
1. Linear spider - This spider has flat profile and has average linearity.



2. Progressive spider - Progressive spider offers "soft clipping" effect. It produces more distortion at low frequencies, but stops the voice coil from "jumping out of the gap" at high power levels.



3. Regressive spider - Regressive spiders offer the best (most extended) linearity



For example: Spiders with **outside radius = 52mm**, **inside radius = 13mm** would offer the following maximum excursions due to spiders:

Standard: **14.4 mm,**
Progressive: **11.0 mm**
Regressive: **21.0 mm**

4. Estimating Xmax due to Cas(x)

If you disregard the actual value of the stiffness of the spider (this is given as 1/Cms anyway), Hutt offers a clever way of estimating spider maximum excursion. The idea is based on Pythagorean Theorem and the formula is given below:

$$X\max = 0.5\sqrt{CL^2 - (Or - Ir)^2}$$

Where: CL = Cloth Length, Or = Outside radius and Ir = Inside radius of the spider. You simply take 50% of the maximum excursion where the cloth length is straight – see Figure 2.14 (the depiction is not perfect, but we are sure, you get the idea).

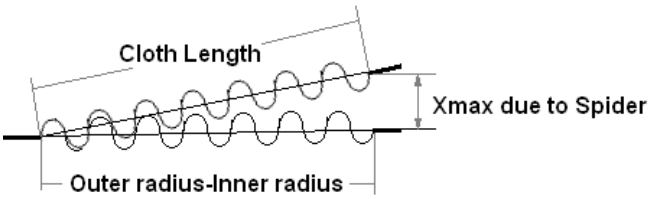


Figure 2.14. Estimation of Spider linearity

NOTE: If you have difficulties assessing Xmax due to Cas(x) please use default value of 10mm for 30cm (12") woofers, 13mm for 38cm (15") woofers and 16mm for 46 cm (18") woofers. These vales are a crude approximations only, but will allow you to perform basic analysis. For midrange drivers the default value would be 2mm.

Enclosure Dimensions

A simple rectangular enclosure calculator is shown on Figure 2.15 below. The calculator is not intended to provide you with a CAD quality drawings of your enclosure. However, dimensions calculated by the dialogue box are accurate enough for you to take this printout to the local hardware store and have the timber cut to size.

In the current implementation, the dialogue is only a simple calculator, and settings are NOT saved to the data file. Furthermore, our carpentry skills are at amateur's level, therefore, we strongly encourage you to treat the drawings as and informative guidance only. We have assumed, that the box will be finished with some form of veneer cover. Secondly, the bottom (and top) panel provide the footprint of the enclosure. Finally**, all joints are braced** with the timber dimensioned by you. The "Rectangular Enclosure Dimensions" dialogue box is invoked from the main menu under "Enclosure Details" option. This dialogue is designed to use as much information as would be already keyed into the program, therefore it "cooperates" with other SoundEasy's screens and data fields. You will notice the following:

1. **"Enclosure Volume"** – editable data field is the same as "Rear Box Vb" on the "Enclosure Design" screen.
2. Port length/volume calculations are based on **"Port Diameter" and "Rear Box Fb"** data fields on the "Enclosure Design" screen.
3. **"Front Baffle Size"** – editable data fields and sliding gadgets are the same data as "Front Baffle Dimensions" on the "Enclosure Diffraction" dialogue box.
4. **"Brace Dimensions"** – editable data fields for entering cross-sectional dimensions of the bracing timber.
5. **"Wall Thickness"** – editable data field for entering thickness of the material used for all panels.
6. **"Driver+Extra Volume"** – editable data field for entering parasitic (that is, volume of ALL objects that will be placed inside the enclosure and must be subtracted from the required enclosure volume).
7. **"Driver's Cut-out Radius"** – editable data field for entering front baffle cut-out for the driver.
8. Number of drivers on the baffle, corresponds to **"Driver Configuration" list box** on the "Enclosure Design" screen.
9. Location of the drivers on the baffle corresponds to **"Location of Driver 1,2,3,4"** on the "Enclosure Diffraction" dialogue box.
10. Location of two ports can be edited from this screen as well.

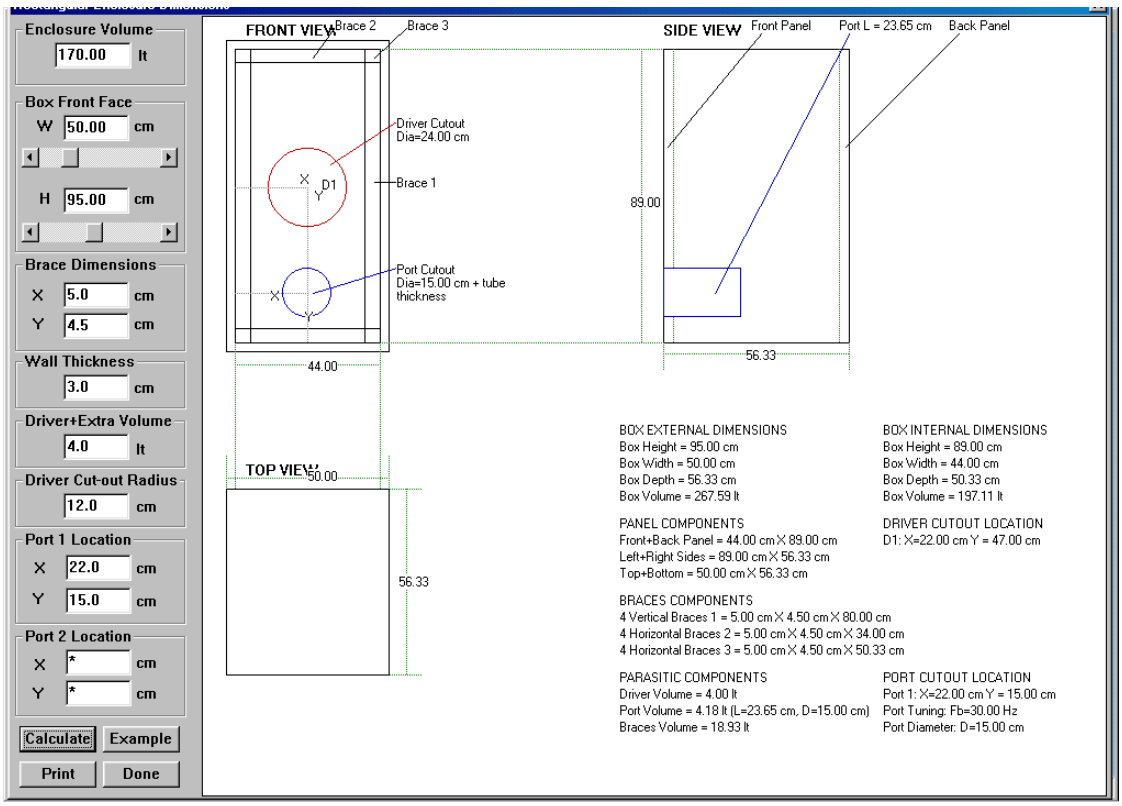


Figure 2.15 Enclosure Dimension

There is no scaling of the graphics in the current implementation of this calculator, therefore "clipping" will occur if your enclosure is large in one or more dimensions. The largest baffle size displayed on the screen would be around 1.0m x 1.0m with total volume of 700liters. The basic job this calculator is to provide you with instant dimensions of all major panels and bracing used for the enclosure, while you make adjustments to the size of the front baffle**. The reference parameter is the required enclosure volume.** Therefore, we suggest, that you do not change the **"Enclosure Volume"** data field provided on this screen. And if you do, please key-in the original value when finished.

Operation of the dialogue is fairly simple. You need to supply the required data to be keyed-in and then you can use sliding gadgets to obtain front baffle dimensions that meet your design goals. You will see all dimensions and the drawings refreshed automatically. There are number of parameters calculated by this dialogue box – see Figure 2.15. There may be instances, where you use really heavy bracing. For instance, bracing also the opposite panels in their midpoints. The dimensions of these braces should be calculated by-hand and their volume lumped together with driver's volume – in the **"Driver+Extra Volume"** data field. The program will automatically increase the internal volume of the box to account for this extra material and more than one driver. The side view of the enclosure shows the size of the port extending into the enclosure from the front panel. This visualization allows you to maintain the recommended minimum distance of the back-end of the port from the back wall ob the enclosure.

The cut-out for the port is located automatically by the program. When you change the data on any of the editable fields, please click on the "calculate" button, or move the sliders to perform calculations and refresh the screen. The example shown on Figure 2.15 indicates, that there will be additional 29liters needed for the enclosure to account for the driver, port and bracing volumes. This figure is the difference between "Enclosure Volume" field (170.0 lt) and "Box Volume" (199.44lt) field calculated under "BOX INTERNAL DIMENSIONS" column.

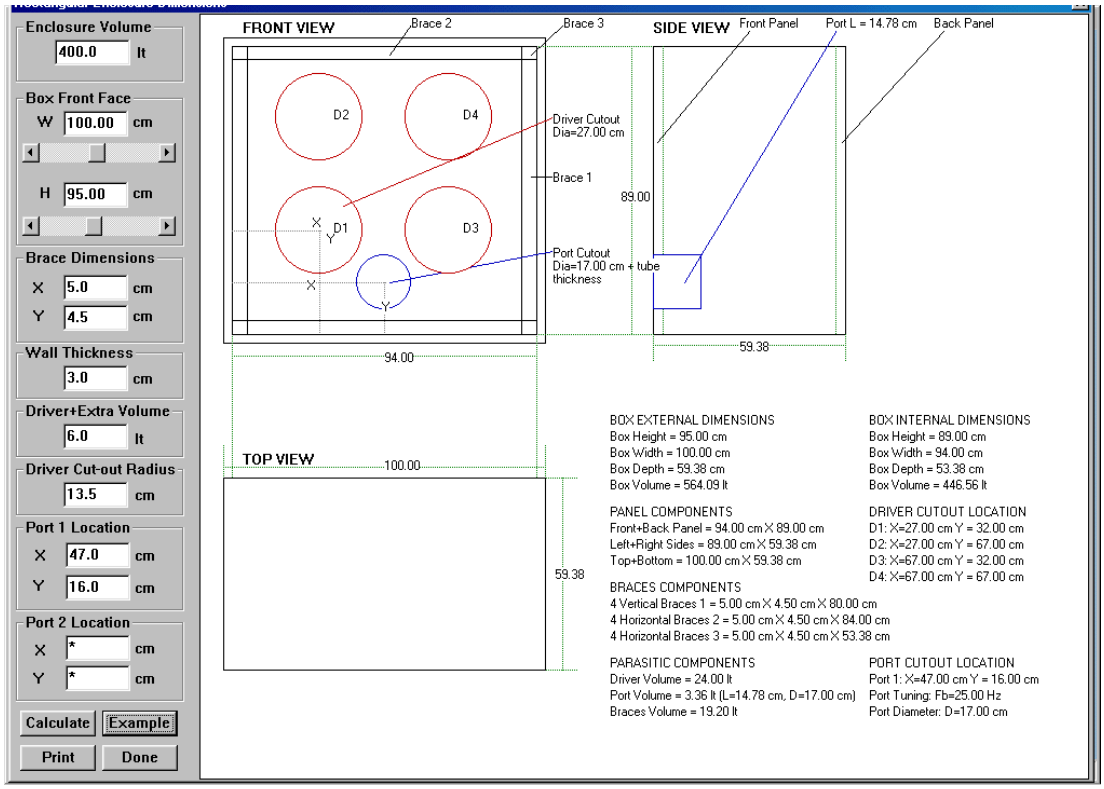


Figure 2.16. "Example" button was pressed.

In the second example, shown on Figure 2.16, is a "quad-box". The box is quite large (400 lt) and would be suitable for 4 x 13" low-frequency drivers intended for a bass guitar application. Bracing used is the same type as in the first example. Please note, that the port required to do the job, is actually quite short (14.78cm) and you could afford to increase the diameter of the port in order to reduce port compression/non-linearity effects.

Finally, when you press the "Example" button, there will be several changes made to the driver's file data. Therefore, you need to go back to the "Enclosure design" screen and re-edit: (1) Rear Box Vb, (2) Rear Box Fb, (3) Port Diameter, and (4) Driver's Configuration.

Quick Assessment of Drivers

There may be a need for a very quick "evaluation" of drivers in sealed or vented enclosure, without detailed design considerations. You may simply want to "plug-in" some basic Small-Thiele parameters and see what the driver is capable of. The program is just capable of doing this. When you open SoundEasy, the program starts in what you may call a **"minimum parameters"** mode.

There are a number of default parameters built-in, that will be set when the program is started. The defaults allow you to make a quick assessment of the driver, but please remember, that for accurate analysis, you must supply all required data **and select and edit one of the available impedance models.** It is becoming more common for the manufacturers these days to provide you with Sd parameter rather than effective diameter (of drivers like 10", 12" or 15" speaker).

SoundEasy provides you with two data fields for entering both: Sd and Effective Diameter. The Sd is only used for enclosure design, while Effective Diameter is used for driver tilting, driver rotating, polar responses and off-axis responses. Both data fields are locked and mutually dependant – entering data in one field causes the other to be calculated.

On the T/S Editor tab, you are only required to enter**: Re, Qe, Qm, Le, BL, Fs,Vas and Sd or Diameter.**
(please note, that Md will be calculated from Fs value).

Parameters like BL could be calculated from other parameters, however, in the interest of cross-checking the data, the program enables you to enter the parameters individually, so they can be checked by the program automatically. You may notice, that the program will modify some parameters while you enter another piece of data.

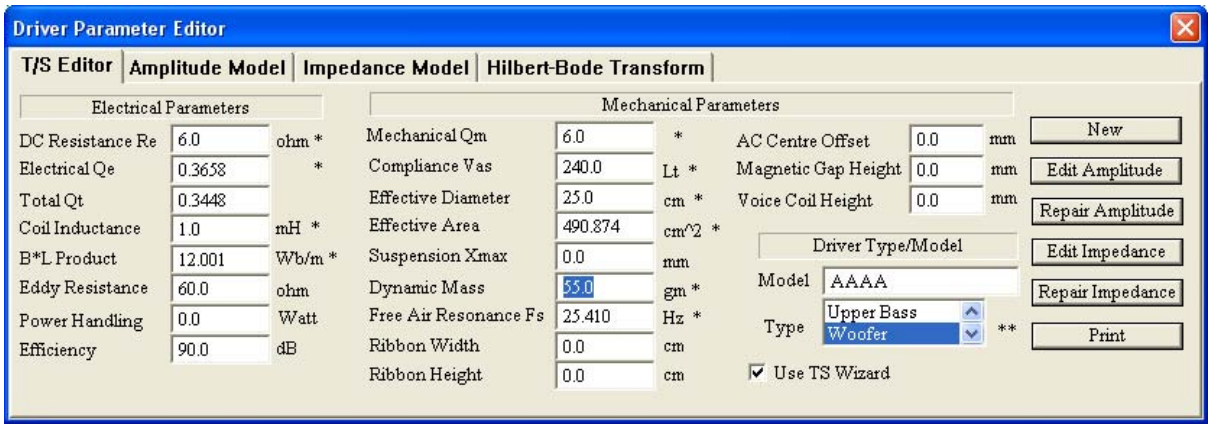


Figure 2.17. Default values with **Re, Qe, Qm, Le, BL, Fs, Vas and Sd.**

If you are missing BL parameter, enter all other parameters (**Re, Qe, Qm, Vas, Sd, Fs – highlighted with asterisks)** and then click on Fs, to get Md calculated and click on Re or Qe to get BL calculated – highlighted in blue. On the Enclosure Design screen, you are only required to enter **Vb for basic sealed box analysis and additionally Fb** (highlighted in red) for basic vented box analysis.

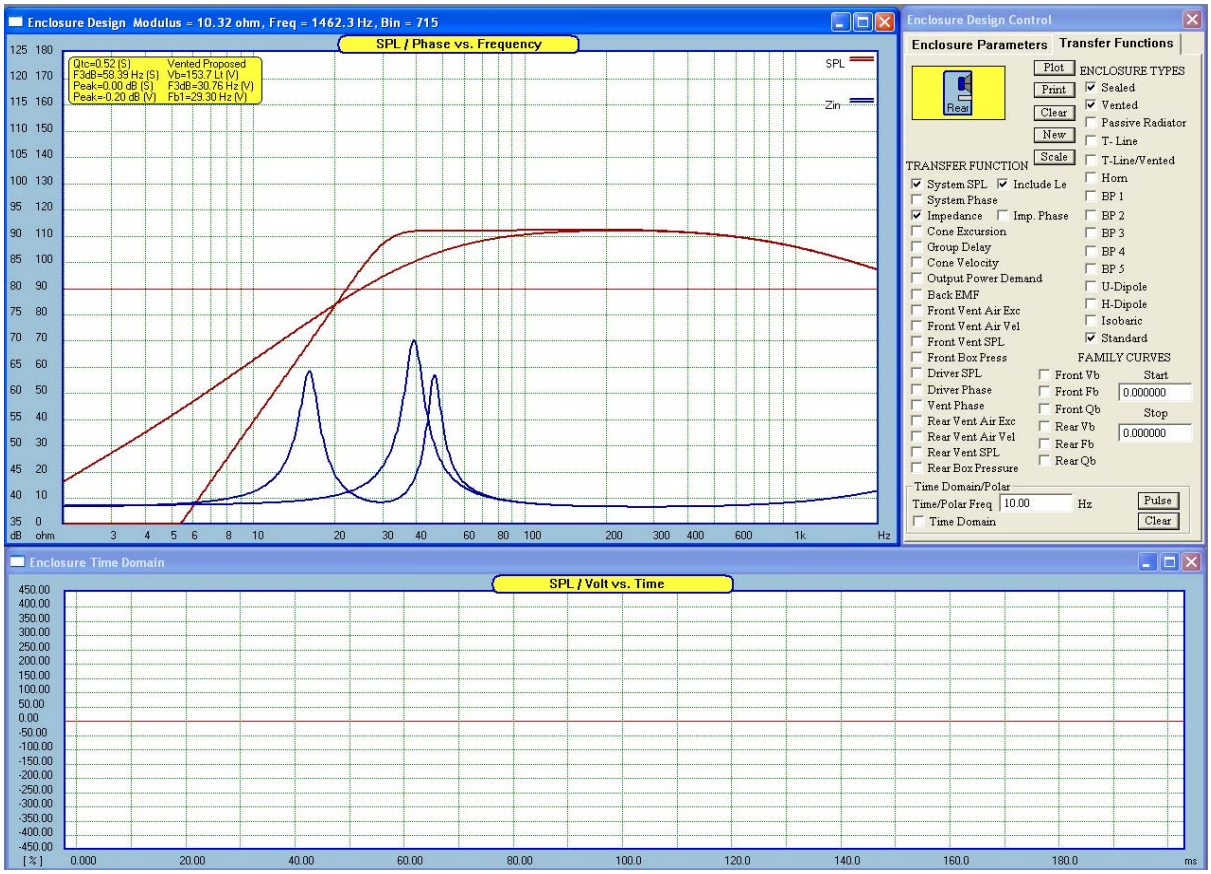


Figure 2.18. Example of Quick Analysis of sealed and vented enclosures.

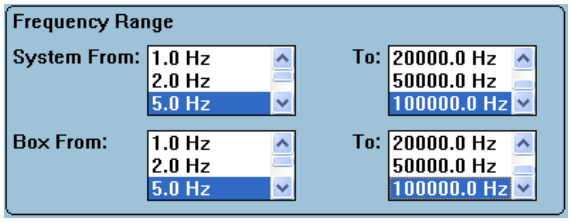
Using Enclosure Simulation as Full Transfer Function

The recommended way of generating an SPL function is to use the Amplitude Model TAB. This way you can incorporate up to 50 dips/peaks onto the "crude" band-pass approximation of the loudspeaker response. The result will be a reasonable approximation of the smoothed driver SPL response. Corresponding input impedance, Zin, can be generated Impedance Model TAB.

At times you may want to quickly generate an SPL and Zin for the purpose of using them as transfer functions in the crossover design. This will still be useful for crossover design. The simplest way to accomplish this is to plot SPL and Zin functions in the Enclosure Design module, with the frequency range of the Box Section set the same as the System Section. Therefore, whatever is calculated in the Enclosure Design section, it will be calculated over the entire frequency range of the CAD and System sections.

Please follow the simple process outlined below:

1. Load your driver file, or enter a minimum of TS + enclosure parameters.
2. Open **Preferences** screen and set Box Frequency Range to be the same as System Frequency Range. Typically the range will be 5-20Hz ……..20-100kHz.



3. Go to **Enclosure Design** section and plot SPL and Zin functions.
4. Go to Driver Editor dialog box and select **Amplitude Model** TAB. There, press "Get SPL + Zin" button. At this stage, whatever was located in your driver's data file for SPL and Zin data vectors, has now been replaced by the data from Enclosure Design SPL and Zin functions.

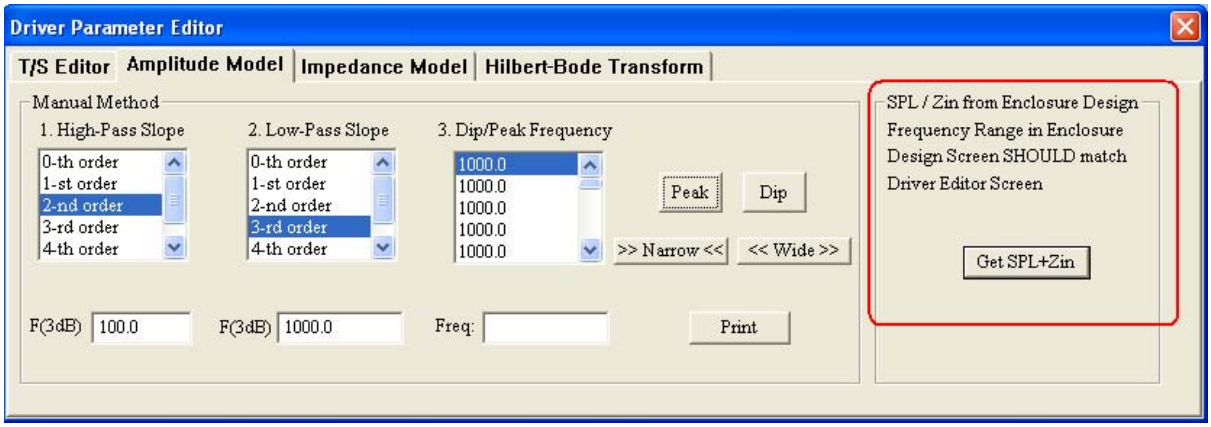


Figure 2.19 Transferring enclosure modelling data to your driver's transfer functions.

5. You can now select **Impedance Model TAB** and review the Zin.
6. Also, you can review box SPL by selecting **Amplitude Model** TAB.
7. Finally, you can preview SPL+phase and Zin+phase plots in the CAD plotting section by selecting "Driver SPL only" list box option on the "Plots" TAB.

Please note, that HBT is not used in this process. The entire transfer function (that is modulus and phase) is calculated by the Enclosure Design screen.

Confirmation of the above is shown in Fig 2.20 below. You can select "Amplitude Model" and "Impedance Model" TABs to review the transferred functions. Finally, the same functions can be plotted in the CAD plotting system – as seen on Figure 2.20

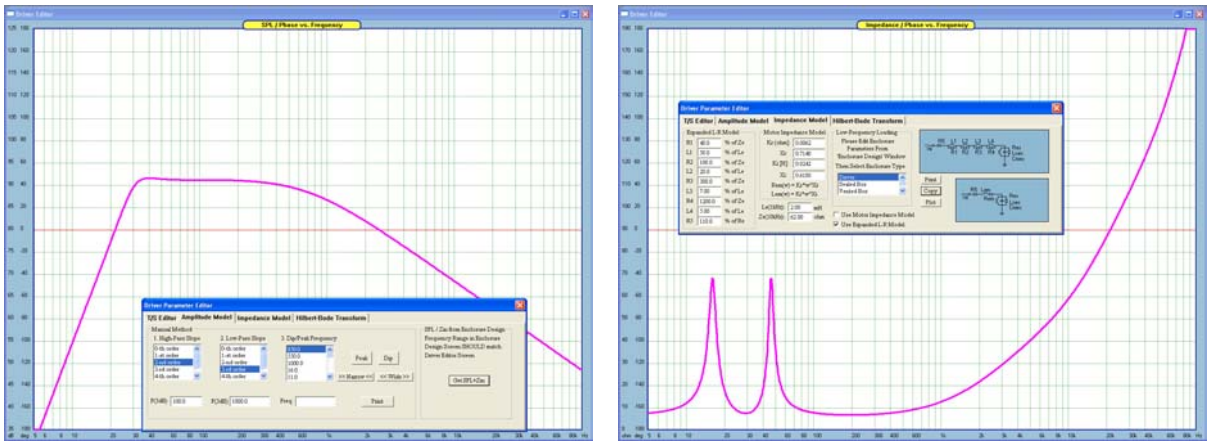


Figure 2.20 SPL + Zin transfer functions generated by the Enclosure Design screen.

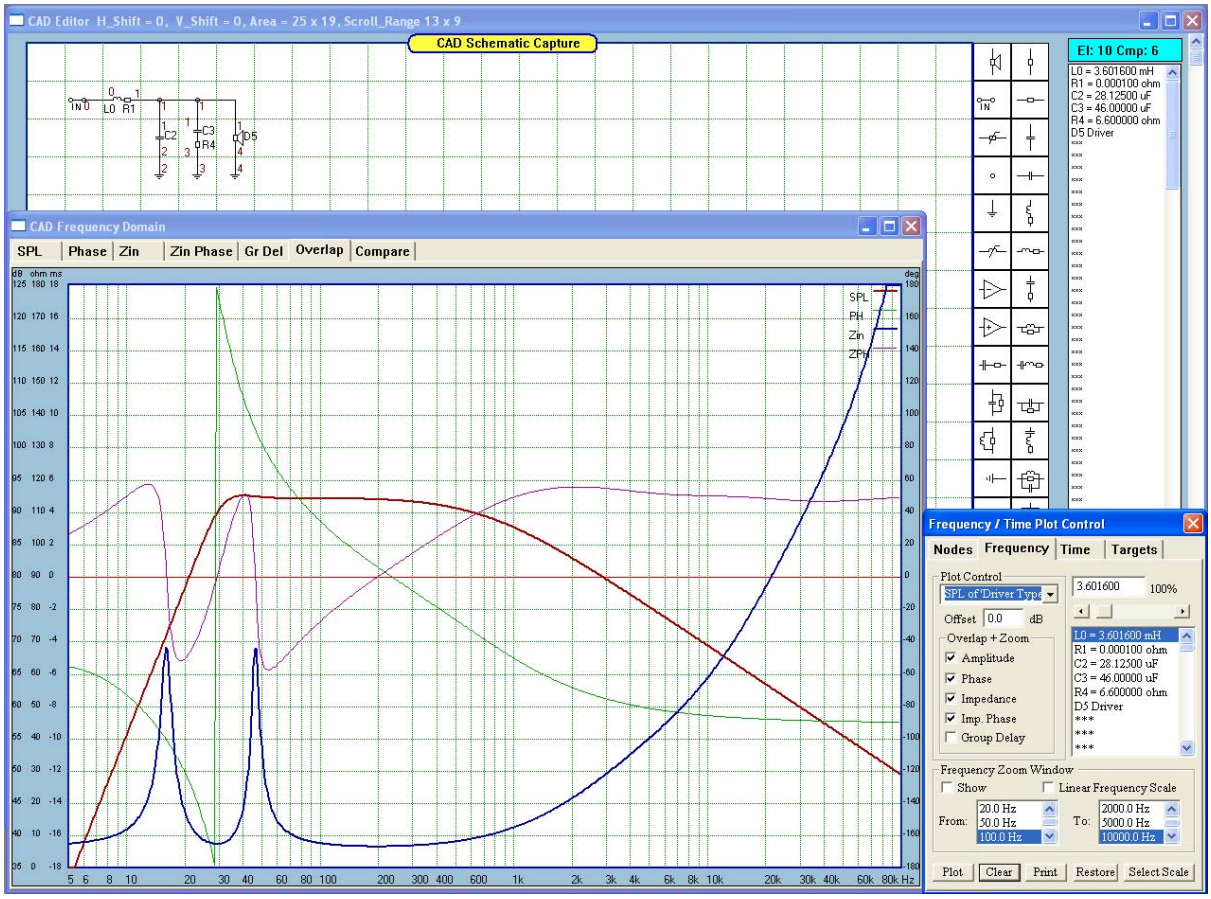


Figure 2.21 Enclosure SPl and Zin used and driver's transfer functions.

Chapter 3. Importing Data Files

Several issues associated with importing files were highlighted in Chapter 2, during our discussion about complex transfer function of the driver. Market competitive forces encourage various organizations to offer attractively priced measurement systems, each one generating its own format of data.

It is therefore very important, that you know exactly what type of data, in what format and under what testing conditions it was recorded. Some examples are discussed below.

Some measurement systems (such as LMS - loudspeaker measurement system by Linear X) are based on the traditional frequency sweep between 10Hz and 100kHz. There is only one measurement microphone used in the tests, so the system records only the amplitude response (or more precisely the modules of the amplitude response). Phase response is not actually being recorded. Phase response is mathematically generated by Hilbert-Bode transform, much like SoundEasy does. The LMS also has the ability to add time delay to the phase response and a function called 'tail correction' to enter the ultimate low-end slope roll-off. All this enables the user to produce and export data files containing amplitude and phase response as if the data was recorded at say, 1 meter distance. The frequency steps are quite uniformly spaced across the whole frequency range and very little post-processing by SoundEasy is required.

Generally speaking, measurement systems based on gated FFT measurements (such as IMP - Impulse Response Measurement and Processing systems) have the ability to compare two pulses: (1) simulation pulse, generated by the amplifier and (2) reproduced and recorded response of the loudspeaker. The recorded time response is then transformed into the frequency domain using FFT process. The time delay between the two pulses will correspond to the distance between the loudspeaker and the test microphone so, the acoustical phase response maybe measured (not calculated). Sampling of loudspeaker response is performed at fixed time intervals. this process is reflected in the frequency domain as fixed frequency steps - amplitude and phase data are available at fixed frequency intervals. In order to attain good accuracy across the whole frequency range, the intervals may have to be different (smaller) at lower end of the frequency range an the two resulting plots combined into one global frequency / phase response. Some of the measurement systems provide the amplitude response data as true SPL level, while others provide this data as relative SPL level. You really need to know what type of data is being imported.

Direct Data Import

Direct data import does not engage Hilbert-Bode Transform function at all. Therefore, you are solely responsible for the quality of data you are going to use. Issues to consider are: is the imported data consistent over the frequency range of interest?, is the phase response measured correctly?, do I require minimum-phase data regardless what acoustic distance was used for measurements?.

The process is actually very simple. The example below (see Figure 3.1a) shows a microphone calibration file containing:

1. No header,
2. The first column is the frequency data,
3. The second column is the amplitude data and
4. The third column is the phase data

The first several lines of the imported file are shown on the right-hand display area of the dialogue box. This field is not editable **– it is intended for visual inspection**, so you can count the number of lines in the header of the file. This is quite important, as the header will be removed during the importing process. Secondly, You will need to nominate which of the data columns you wish to be imported. In the example below, column 1 (frequency), column 2 (amplitude) and column 3 (phase). Finally, please count the total number of columns and enter this number in the “Total Number of Columns” data field.

With the above information you are guiding the importing algorithm accurately. The format of the data can be both: decimal and exponential notation. When the importing algorithm has finished, you will be presented with a screen similar to the Figure 3.2.

Importing SPL Data from Measurement File

The first several lines of the imported file are shown on the right-hand display area of the dialogue box. This field is not editable **– it is intended for visual inspection**, so you can count the number of lines in the header of the file. This is quite important, as the header will be removed during the importing process. Secondly, You will need to nominate which of the data columns you wish to be imported. In the example below, column 1 (frequency), column 2 (amplitude) and column 3 (phase). Finally, please count the total number of columns and enter this number in the "**Total Number of Columns**" data field. With the above information you are guiding the importing algorithm accurately. The format of the data can be both: decimal and exponential notation. More often than not, the imported data file contains gaps between its data points. The missing points have to be mathematically inserted into the curve, so that the lines you see on the screen (and the actual data used by the program) are "continuous". This can be accomplished by pressing the "**Repair Amplitude**" or "**Repair Impedance"** buttons on the tab. The Data Import box has the following controls:

1. **Amplitude** – Use this button to import SPL/Phase data into SoundEasy
2. **Impedance** – Use this button to import Impedance/Phase data into SoundEasy
3. **Cancel** – Closes the dialogue box.
4. **Print** – Simply prints the screen.
5. **Correction Factors Group**
   - **Delay –** Enter additional delay in this field. The amplitude response will remain unchanged, but phase response will have added requested delay.
   - **Scale –** Use this field to add/subtract decibels to the imported data. Typically, when you import SPL data, that was measured at different distance than 1meter reference efficiency, or data that was measured at different power level, you would scale the imported data by a factor, that would display the curve at the reference efficiency level. For instance, the measurements were conducted with 1W power level, but at 0.25m distance. You would expect that measured SPL curve will be +12dB higher than the reference efficiency. For this case, enter –12dB in this field.
   - **Efficiency –** Enter the 1W/1m efficiency here.
6. **Import Small/Thiele from LMS** – Use this button to enter the small-signal parameters measured by LMS system.
7. **Imported Data Format Corrections Group**
   - **Erase Lines** – Enter number of lines, that need to be erased from the start of the file to the first data line. Typically, you would enter the number of lines in the header.
   - **Include Columns** – Check corresponding columns in the data file, that show relevant data.
   - **Total Number of Columns** – Enter total number of columns in the data file.

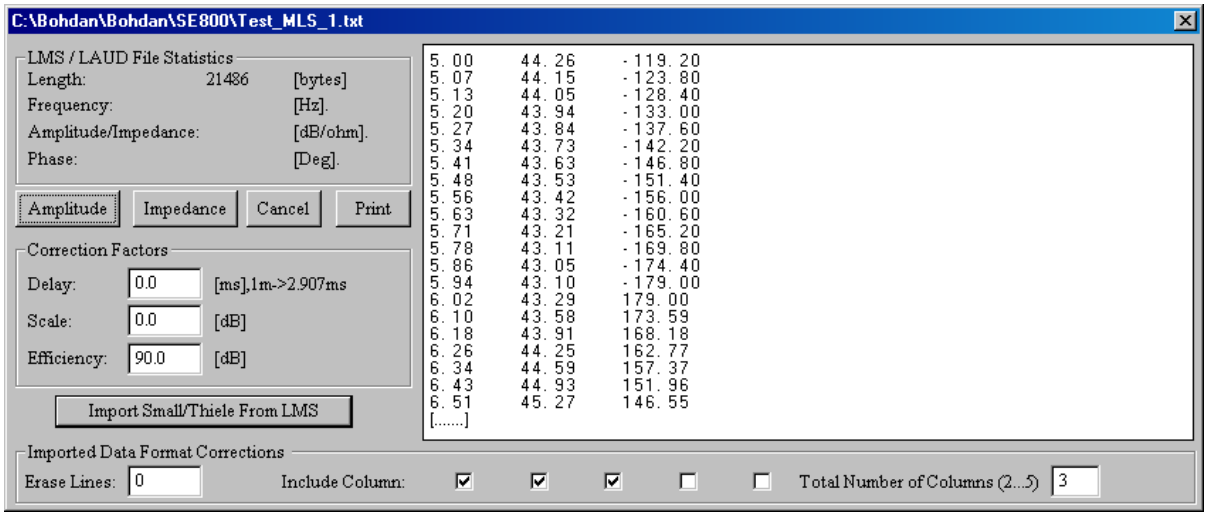


Figure 3.1. File importing dialogue box. The file does not have any headers.

The newly imported file will be displayed on the plotting screen. You may need to press "Repair Amplitude" button to smooth the imported data. **You may decide to rely exclusively on the imported measurement results, OR you may use Hilbert-Bode Transform to clean-up the imported data and create a better quality Transfer Function. We strongly recommend, that you do so.**

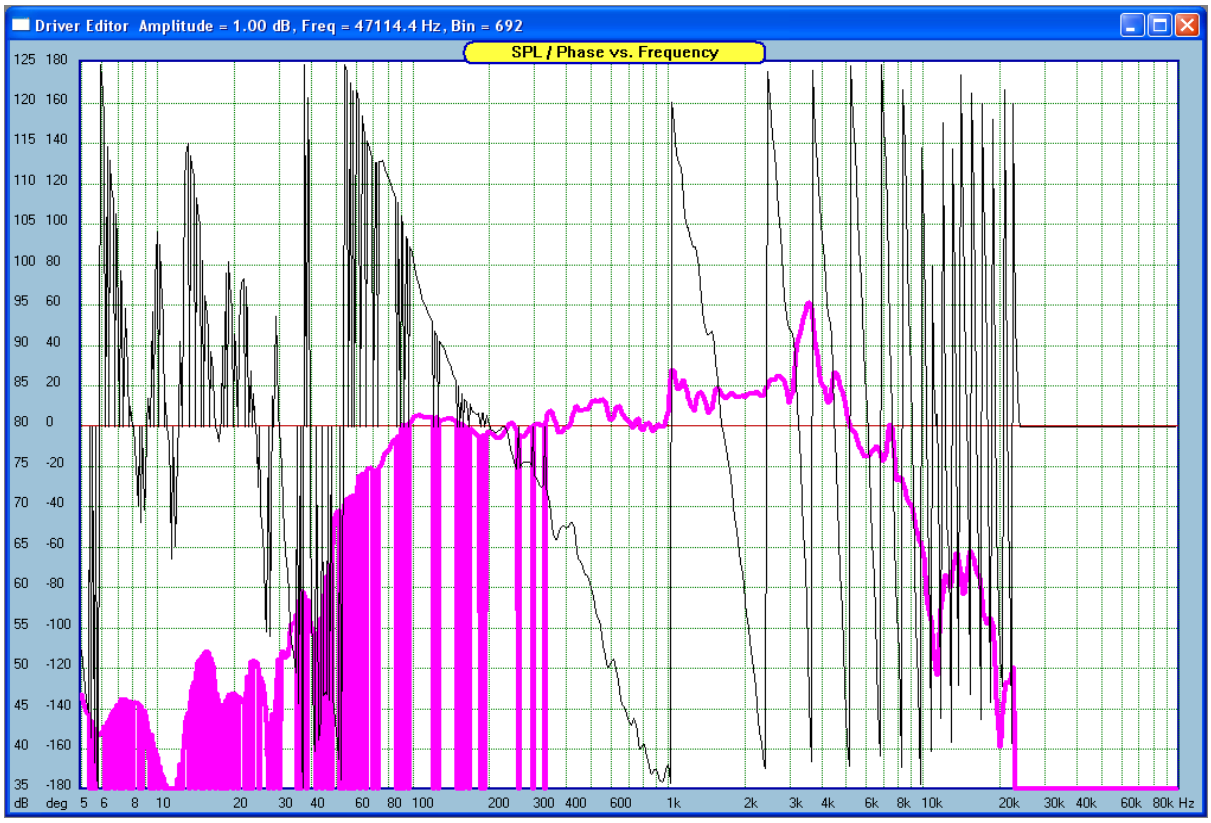


Figure 3.2. Measurement file imported into SoundEasy

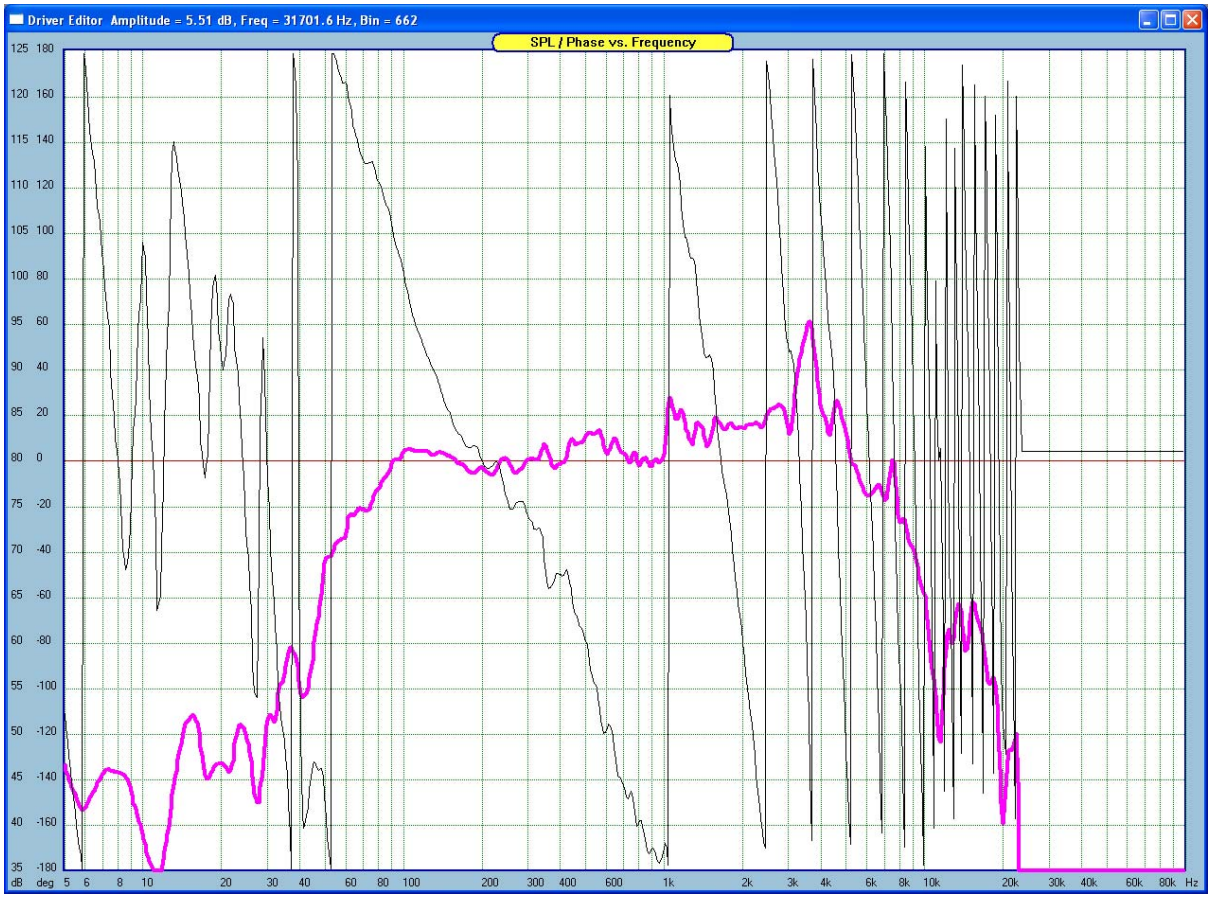


Figure 3.3. Measurement file imported into SoundEasy and **"Repaired".**

More often than not, the imported data file contains gaps between its data points – see Figure 3.2. The missing points have to be mathematically inserted into the curve, so that the lines you see on the screen (and the actual data used by the program) are "continuous". This can be accomplished by pressing the "Repair Amplitude" or "repair Impedance" buttons on the Editor screen – see Figure 3.3. **This is when the program actually calculates Complex Transfer Function from the imported data**. Please remember that measurement data is typically presented as modulus of the amplitude (in dBs) and phase (in deg). What is needed is the complex representation of the above – hence the need to calculate the Complex Transfer Function in rectangular coordinates: A+jB. You are now ready to use the imported data in all other program screens and functions. If you wish to confirm, that the process was indeed successful, please open the frequency response screen in crossover section. You can now plot driver's amplitude and phase as seen on Figure 3.3.

Importing Data with Hilbert-Bode Transform

In order to bring some consistency into the imported data, SoundEasy imports amplitude and phase information, but uses them only as templates. The Complex Transfer Function of the imported driver is derived entirely from the Hilbert-Bode transform. Importing and editing impedance files is identical as the frequency response files. It will therefore not be elaborated here.

The driver section of the program has a built in very powerful algorithm - The Hilbert-Bode transform. It allows you to quickly and accurately recreate phase response from the available amplitude response (this is the way the LMS system creates it's exportable phase data, so it really does not matter where the process is performed). The Hilbert-Bode transform really makes importing of the phase response irrelevant. the process of creating the true acoustic phase response with SoundEasy is instantaneous, this is why the imported phase response is only used as a template. Also, ANY POST-PROCESSING of the amplitude response on the Editor screen will make the imported phase invalid. The minimum phase systems have a unique relationship between amplitude and phase, this relationship is captured the Hilbert-Bode mechanism.

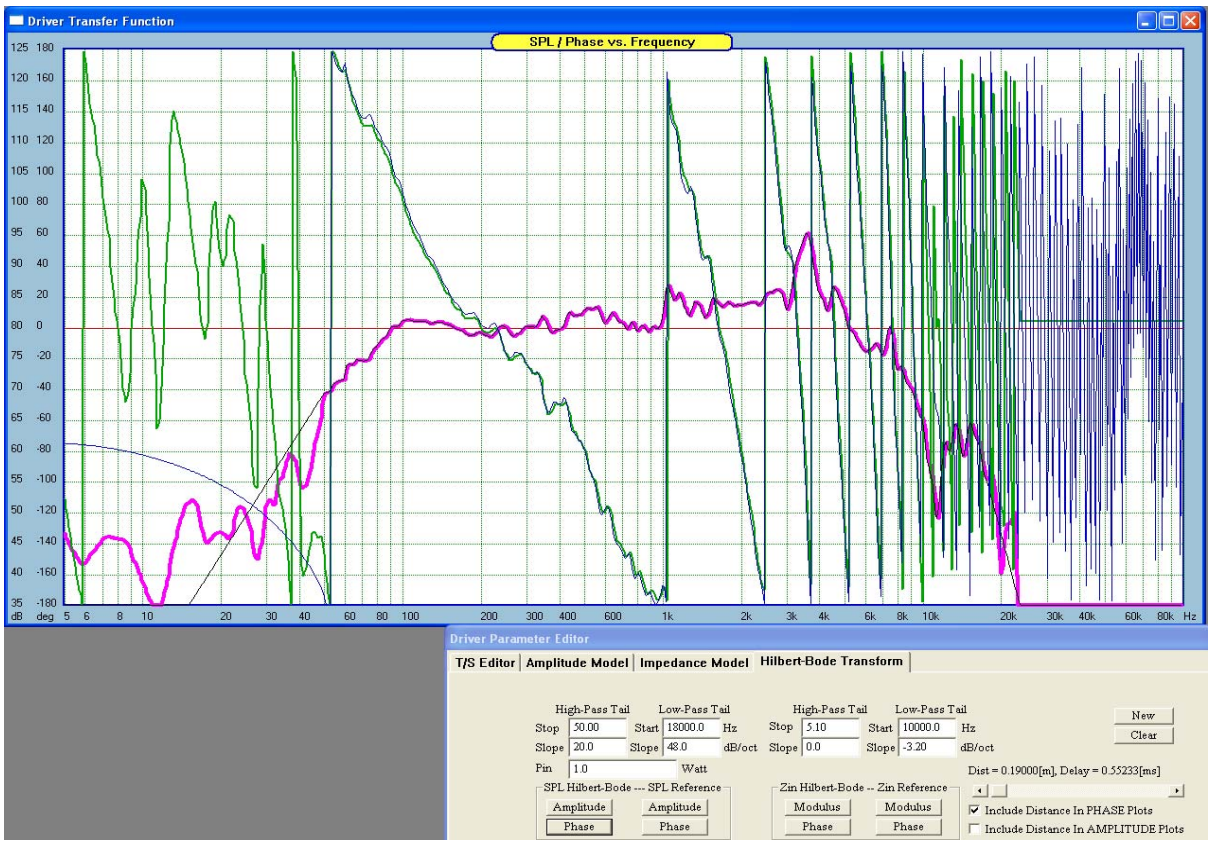


Figure 3.4. Hilbert-Bode Transform parameters

We recommend that you should become familiar with the data structure of the files produced by your measurement system. As the starting point you should examine the content of the file prior to importing it to SoundEasy. SoundEasy will import only ASCII files.

The transform is driven by editable parameters and they should be selected to obtain the **best match for phase and amplitude between measured signal and calculated transform over the widest frequency range**. Typically, good match can be obtained between 20Hz and 20kHz. Generally, you would select the parameters to provide initial match for amplitude response and then fine tune the parameters to provide the match for phase. Please note, that "Roll-off" parameters can be as high as +/- 100dB to +/- 120dB if the measured frequency response is highly irregular (resonant peaks) at the extremes of the measurement range. The program calculates phase using a Hilbert-Bode transform from any magnitude curve. This is known as a Minimum Phase response and it does not contain any delay or positional information. It is a phase response produced solely from the magnitude data curve. This is very convenient for SPL measurements since the distance from the source to the microphone is completely unimportant. The phase curve will have no excess delay. This eliminates the usual problem of removing the arrival delay from the path of the source to microphone. "A minimum phase system is one which is able to transfer input energy to its output in the least amount of time for a given frequency response". Mathematically, you would calculate the phase response of a minimum phase system by taking the Hilbert-Bode Transform of its log magnitude response (i.e. magnitude expressed in decibels). This relationship is also reciprocal: the log magnitude response of a minimum phase system equals the Hilbert-Bode transform of its phase response. Because the Hilbert-Bode Transform requires an infinite integral over all frequencies, however, any numerical computation of minimum phase is necessarily an approximation. Understanding the characteristics of the phase transform and how it works can be very important if maximum accuracy is to be obtained. For some data curves, the transform can be run directly and will provide excellent results with little attention to the original magnitude data. In other cases, a far better result can be obtained by correcting certain areas of the magnitude curve before running the minimum phase transform. Typically, the regions which can cause problems are **at the ends** of the measurement bandwidth.

The transform must project the slope of the magnitude curve beyond the measured bandwidth. It does this by taking the last slope found as the curve reaches the low and high ends of the frequency range. It then assumes that: (1) the slope found at the low end will remain constant down to DC, and (2) the slope found at the high frequency limit will remain constant towards infinity. For this reason it is of fundamental importance that the slope of the magnitude curve at the high and low ends of the frequency range be approaching or already at their final asymptotic values Hilbert-Bode Transform Tab

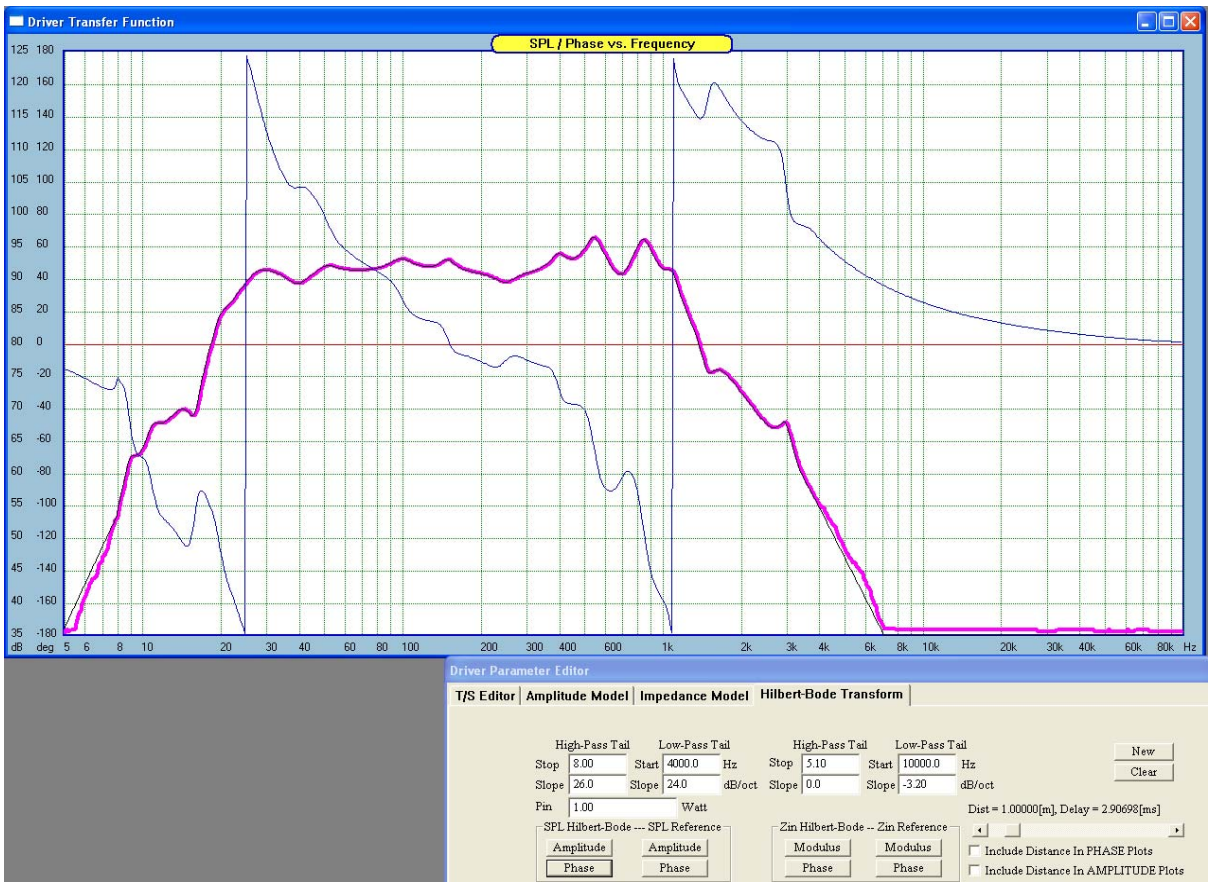


Figure 3.5.Example of Hilbert-Bode Transform

If you need to extend the measured SPL/impedance/phase curves beyond the measurement frequency range, the tool to use in this case, is the Hilbert-Bode. The transforms are driven by 4 editable parameters for each section and they should be selected to obtain the **best match for phase and amplitude between measured signal and calculated transform over the widest frequency range**. Typically, good match can be obtained between 20Hz and 20kHz. Generally, you would select the parameters to provide initial match for amplitude response and then fine tune the parameters to provide the match for phase. Please note, that "Roll-off" parameters can be as high as +/- 100dB to +/- 120dB if the measured frequency response is highly irregular (resonant peaks) at the extremes of the measurement range.

The "Hilbert-Bode Transform" tab has a built in very powerful algorithm - The Hilbert-Bode transform. It allows you to quickly and accurately recreate phase response from the available amplitude response. The Hilbert-Bode transform really makes importing of the phase response irrelevant. the process of creating the true acoustic phase response with SoundEasy is instantaneous, this is why the imported phase response is only used as a template. Also, ANY POST-PROCESSING of the amplitude response on the Editor screen will make the imported phase invalid. The minimum phase systems have a unique relationship between amplitude and phase, this relationship is captured the Hilbert-Bode mechanism. The program calculates phase using a Hilbert-Bode transform from any magnitude curve. This is known as a Minimum Phase response and it does not contain any delay or positional information. Because the Hilbert-Bode Transform requires an infinite integral over all frequencies, however, any numerical computation of minimum phase is necessarily an approximation. Understanding the characteristics of the phase transform and how it works can be very important if maximum accuracy is to be obtained. For some data curves, the transform can be run directly and will provide excellent results with little attention to the original magnitude data. In other cases, a far better result can be obtained by correcting certain areas of the magnitude curve before running the minimum phase transform. Typically, the regions which can cause problems are **at the ends** of the measurement bandwidth. The transform must project the slope of the magnitude curve beyond the measured bandwidth. It does this by taking the last slope found as the curve reaches the low and high ends of the frequency range. It then assumes that: (1) the slope found at the low end will remain constant down to DC, and (2) the slope found at the high frequency limit will remain constant towards infinity. For this reason it is of fundamental importance that the slope of the magnitude curve at the high and low ends of the frequency range be approaching or already at their final asymptotic values. The Transfer Function [HBT] dialogue box has the following controls:

1. **"New"** – use this button to restore default setting for this screen.
2. **"Clear"** – use this button to clear the screen.
3. **SPL Hilbert-Bode**
- "Amplitude" – use this button to calculate transfer function and display it as amplitude
- "Phase" – use this button to calculate transfer function and display it as phase.
4. **SPL Reference**
- - Amplitude" – use this button to display amplitude of the imported data
- "Phase" – use this button to display phase of the imported data
5. **Zin Hilbert-Bode**
- "Modulus" – use this button to calculate transfer function and display it as modulus
- "Phase" – use this button to calculate transfer function and display it as phase.
6. **Zin Reference**
- - Modulus" – use this button to display modulus of the imported data
- "Phase" – use this button to display phase of the imported data
7. **"Distance/Delay"** – use this slider to select appropriate distance between the transducers.
8. Check box to include the distance in amplitude calculations.
9. Check box to include the distance in phase calculations.
10. **High-Pass Tail and Low-Pass Tail group box** - The low-end slope and stop frequency, the high-end slope and start frequency. Enter the appropriate data as suggested above.

The Hilbert-Bode Transform is used for amplitude as well as for impedance transfer functions. You need to perform tail corrections for both types of curves. A very good example of using HBT with impedance curves is given in Chapter 16.

Importing the LMS Files.

One of the measurement systems, that could be used with SoundEasy is the LMS (Loudspeaker Measurement System) package created by Linear X. The LMS system has the ability to export captured data in several different formats. One of the formats, the **'*.dat'** files can be imported by SoundEasy.

The **'*.dat'** files produced by the LMS contain several different type of data. We are only interested in amplitude and phase information captured at frequencies from 1 Hz to 100 kHz. The LMS uses frequency sweep technique to supply test signal to the loudspeaker and the frequency response is recovered by a test microphone, positioned at some distance from the box. The frequency sweep test can be performed at different power levels and different microphone-loudspeaker distances. The common (and recommended standard) set-up would be Pin=1W, Mike Distance=1.0m. However, should your data be captured using some different set-up, the program offers you two editable fields (see Fig 3.6) to enter the parameters of the set-up. Negative "Delay" cancels phase reversals (effect of reducing the speaker-microphone distance). The "Scale" parameter will offset whole amplitude plot up (for positive number) or down (for negative number entered). Also, the reference efficiency of the driver needs to be entered on the main 'Editor' screen prior importing the LMS **'*.dat'** file. The correct usage of the LMS **'*.dat'** (frequency response) files can be very helpful and will reduce the amount of effort necessary to edit the frequency response of your driver. Once again, here is the recommended way:

Example: The task here is to import an LMS data file representing a system that has the following parameters:

1. Reference level SPL is 100.0 dB
2. Power into the system is 1.0 W.
3. Microphone-loudspeaker distance is 0.0 cm.
4. Amplitude data was recorder as SPL level, that is, the data is scattered around 100 dB value.
5. Data is collected using frequency sweep method.
6. The data covers frequency range from 20 Hz to 20 kHz.

When the dialogue box for importing the LMS files is opened, for this particular file, you need to enter the following.

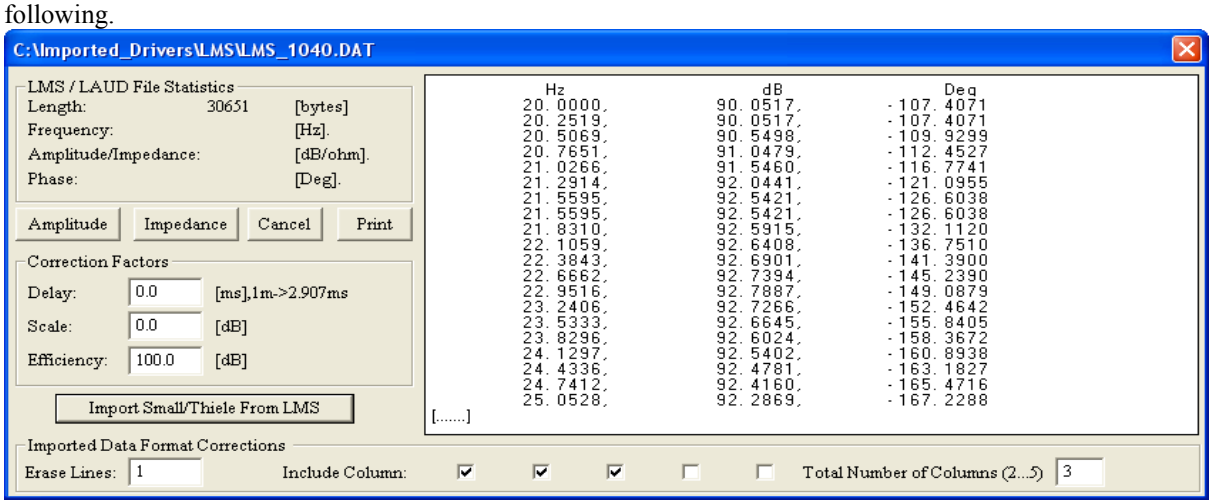


Figure 3.6. Typical import parameters for LMS SPL-type files.

Pressing the "Amplitude" button will activate the process and when finished, press “Repair Amplitude” button to clear the plots. The imported file is shown on Fig 3.8

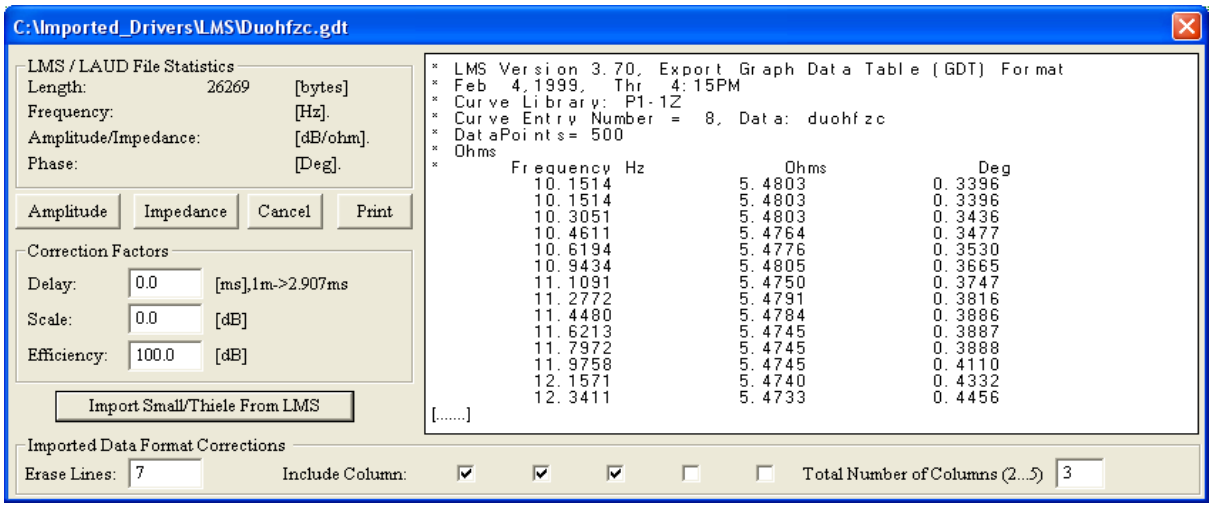


Figure 3.7. Typical import parameters for LMS, impedance-type files.

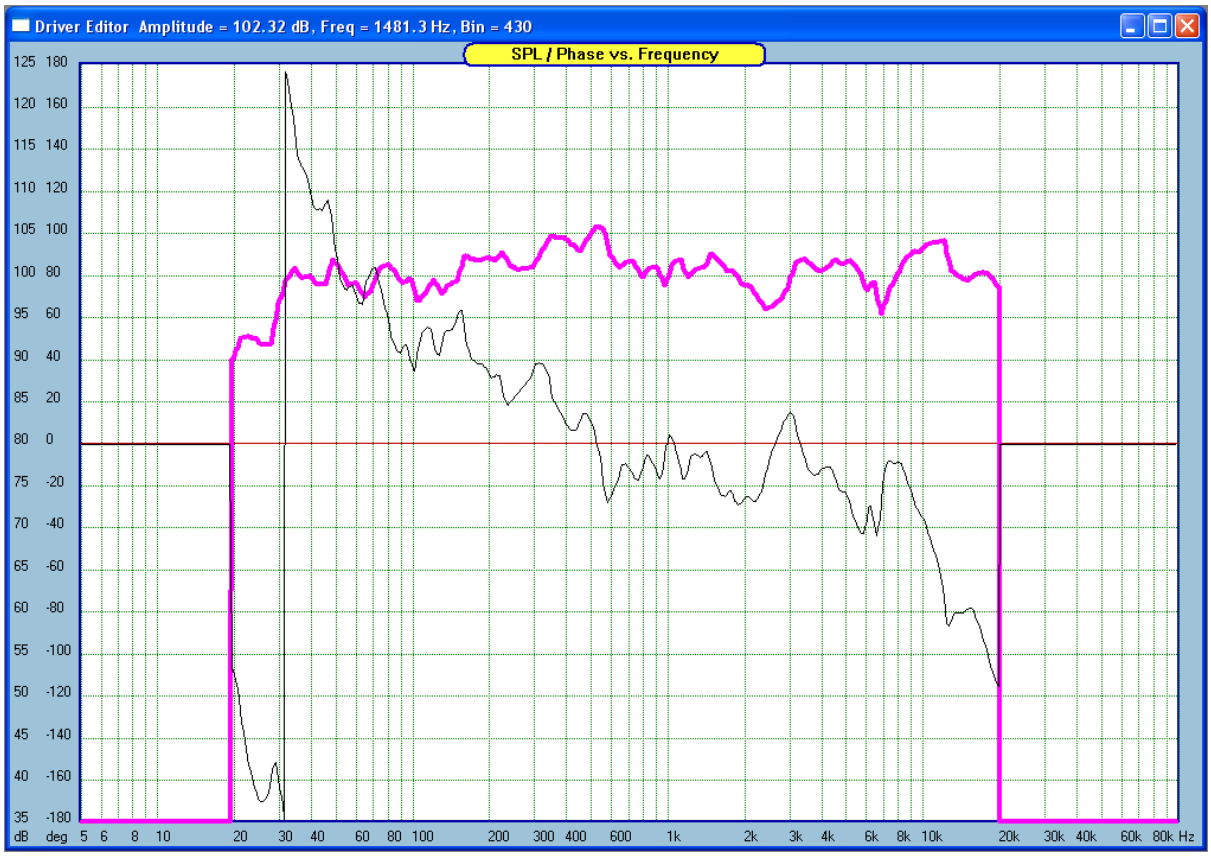


Fig 3.8. Imported LMS SPL file after “Repair Amplitude”.

Fig 3.9. Imported LMS Zin file after “Repair Amplitude”.

Importing the IMP file

If you have access to the Impulse Response Measurement and Processing system (IMP), you could make use of SoundEasy file importing capabilities and obtain the magnitude of the frequency response of the driver, as measured by the IMP system.

Example: The task here is to import an IMP data file representing a system that has the following parameters:

1. Reference level SPL is 90.0 dB
2. Power into the system is 0.065 W.
3. Microphone-loudspeaker distance is 25 cm.
4. Amplitude data was recorder as **SPL relative level**, that is, the data is scattered around 0 dB value – see below.
5. Data is collected using gated FFT method.
6. The data covers frequency range from 30 Hz to 20 kHz.

```
30.0007  -0.83  -130
45.0010  0.609  147.
60.0014  1.546  110.
75.0018  2.232  62.7
…..
255.006  1.110  -24.............
```

As one can see, there is quite a lot of information, that is not included in the data file, but is necessary to be taken into account. Power into the system is 12dB below 1W and the test distance is 1/4 of the recommended 1 meter. Please remember, that the reference SPL (that is 1W/1m conditions) is 90dB.

Therefore, you would expect, that if, on the "Transfer Function" screen, the power is set to 1W and speaker-microphone distance to 1 meter, you would see the amplitude response plotted at the reference SPL level, regardless of the imported data test conditions. Also, if the power is set to 0.065W and microphone distance to 25cm, you would expect the plotted amplitude and phase curves to overlap the imported data on the "Transfer Function" screen. When the dialogue box for importing the IMP files is opened, for this particular file, you need to enter:

1. Delay = 0.00,
2. 2. Scale = 0.00 dB (do not scale this file),
3. 3. Efficiency = 90 dB (reference SPL level)

You could enter Delay = -0.7352 ms (because the microphone distance is 25 cm). This would reduce number of phase reversals on the imported curve, as you would cancel the acoustic path delay. The resulting phase curve would not be entirely smooth, due to digitization and different discrete frequency steps used by the IMP and SoundEasy. Pressing the "Amplitude" button will activate the process. You may notice, that at the lower end of the frequency range, the data is incomplete. To repair this section of the plot, press the "Repair" button. This function scans the data file starting from 10 Hz up, looking for the first non-zero data point. It will then link this data point with the second non-zero data point interpolating all of the program's data between the adjacent, imported data points. The recovery process is however still not completed. The next step is to recreate data below 30Hz and above 20kHz.

If you decide to use Hilbert-Bode Transform you may consider the following

In order to avoid rapid and unrealistic amplitude/phase changes at 30Hz and 20kHz, you should edit the amplitude response (using Editor screen) around those frequencies by extending the pink curve 3-4 frequency steps below and above the currently "repaired" curve. Try to create a round "knee" in this transition areas, so that when the algorithm extends the amplitude response automatically, the imported and recreated sections of the amplitude response add smoothly. The imported data has now been prepared sufficiently to be processed by the "Transfer Function" screen. SoundEasy and IMP files are not directly compatible. The sample frequencies of both systems are also different. SoundEasy, in order to obtain high accuracy of the design, requires recovering of the driver complex transfer function within 1Hz to 100kHz.

Importing CLIO Files.

Third measurement system, that could be used with SoundEasy is the CLIO system. Much of what has been discussed previously is also applicable to importing the CLIO **"*.txt"** files. **You need to make sure, that the correct type of data is exported from the CLIO system - see below.**

Freq	dB spl	Phase
205.18	55.87	-33.86
206.94	55.98	-34.02
208.71	56.08	-34.11
210.50	56.17	-34.19
212.30	56.24	-34.43
214.12	56.29	-34.69
215.96	56.33	-35.22

The file converter dialogue box offers the same data fields and as before, you must enter SPL level. The other two fields (Delay and Scale) are optional - see Fig 3.1. Negative "Delay" cancels phase reversals (effect of reducing the speaker-microphone distance). The "Scale" parameter will offset whole amplitude plot up (for positive number) or down (for negative number entered). When the dialogue box for importing the CLIO files is opened, for this particular file, you need to enter:

1. Delay = 0.00,
2. Scale = 20.00 dB (this file was scaled up),
3. Efficiency = 90 dB (reference SPL level).

Importing S/T parameters from CLIO

In addition to importing amplitude and impedance data from CLIO measurement system, SoundEasy is also capable of importing Small/Thiele parameters. The CLIO import dialogue box includes additional button, “CLIO TS”, which activates importing process. An example of CLIO file with these parameters is given below. Currently, the following parameters are imported: Re, Fs, Qms, Qes, Qts, Bl, Vas, DbSpl, D (diameter), L1K (inductance at 1kHz), Ms, L10K (inductance at 10kHz). These are sufficient to invoke “Enclosure Design” screen and model the enclosure. The process will only be successful if the correct type of file is imported. Fig 3.10

MANUFACTURER : EMINENCE MODEL : PS15CV

- IMPEDANCE -

Freq	Ohm	Phase
10.0	7.14	36.7
11.2	7.56	39.6
12.6	8.08	42.3
……..		
15848.9	73.68	54.5
17782.8	79.22	54.6
19952.6	85.22	54.7

- PARAMETERS -

Re : 5.08 [ohm]
Fs : 43.40 [Hz]
F1 : 28.18 [Hz]
F2 : 64.94 [Hz]
Zm : 65.21 [ohm]
D : 330.00 [mm]
Qms : 4.23
Qes : 0.36
Qts : 0.33
Bl : 21.99 [N/A]
L1K : 2.04 [mH]
L10K : 0.88 [mH]

Ms : 124.76 [g]
Vas : 110.10 [l]
DbSpl: 97.94 [dB]
Cms : 0.11 [mm/N]
Ma : 0.00 [g]
FsMa : 0.00 [Hz]
V : 124.60 [l]
FsKnV: 59.57 [Hz]
Ras : 1.10 [Kohm]
Rat : 13.02 [Kohm]
Cas : 7.88E-07 [m^5/N]
Mas : 17.05 [Kg/m^4]

Fig 3.10 - CLIO S/T data file

Importing data from DAAS 32 System

Importing data files generated by DAAS-32 system is accomplished by selecting "Import" + "DAAS 32" menu options from the main screen menu. Please note, that there will be 5 files required to fully characterize the driver. They are :

1. .AMP – file containing SPL data
2. .PHA – (acoustical phase) file containing SPL phase data
3. .IMP – file containing input impedance modulus
4. .PHE – (electrical phase) file containing impedance phase data and
5. .TS – file containing a subset of Small/Thiele data.

The DAAS 32 dialogue box shown on Figure 3.11 has 5 buttons (2 in SPL box, 2 in Impedance box and 1 for TS parameters) that need to be activated each time you import appropriate file. The only other parameter required to be entered in the DAAS 32 import dialogue box is the "Efficiency" data. SoundEasy will automatically store imported data in its memory. As before, you can modify imported data during import by entering "Delay" or "Scale" parameters in the dialogue box.

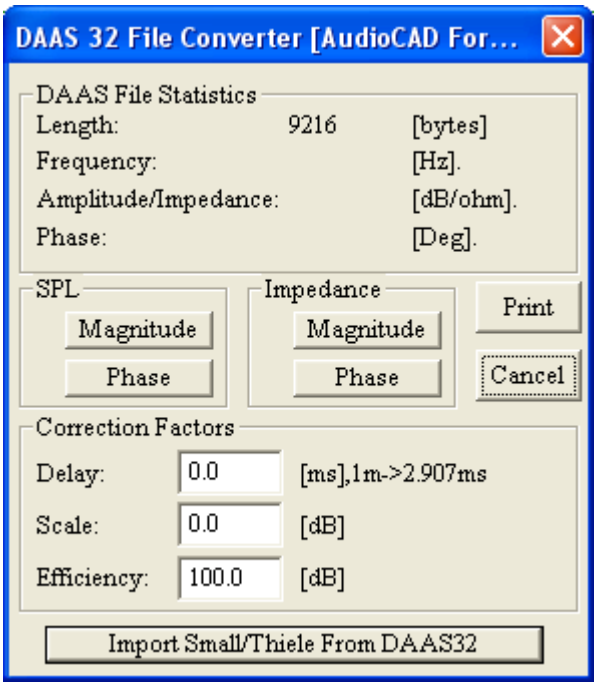


Figure 3.11 DAAS file importing dialogue box.

The process is simple and when completed, you need to run Transfer Function process to generate a full set of driver functions needed for other processing modules. Once again, the imported phase data should be used as a template for generating drivers Complex Transfer Function.

Exporting SPL/Impedance data files

SoundEasy can be used to export SPL/Phase and Impedance/Phase data after the Hilbert-Bode transform has been performed. The SPL file has file extension ".txt" and the first column is the frequency, followed by corresponding SPL value and finally phase value – see below.

```
5.00    39.84    44.92
5.07    40.29    44.80
5.13    40.75    43.64
5.20    41.20    42.48
5.27    41.65    41.30
5.34    42.10    40.11
5.41    42.56    38.91
5.48    43.01    37.70
```

The Impedance file has file extension ".imp" and the first column is the frequency, followed by corresponding impedance modulus value and finally phase value – see below.

```
5.00    7.97    31.37
5.07    8.01    31.74
5.13    8.05    32.12
5.20    8.09    32.50
5.27    8.14    32.88
5.34    8.18    33.27
5.41    8.22    33.66
5.48    8.27    34.05
5.78    8.47    35.66
5.86    8.53    36.07
```

There is no header in those files.

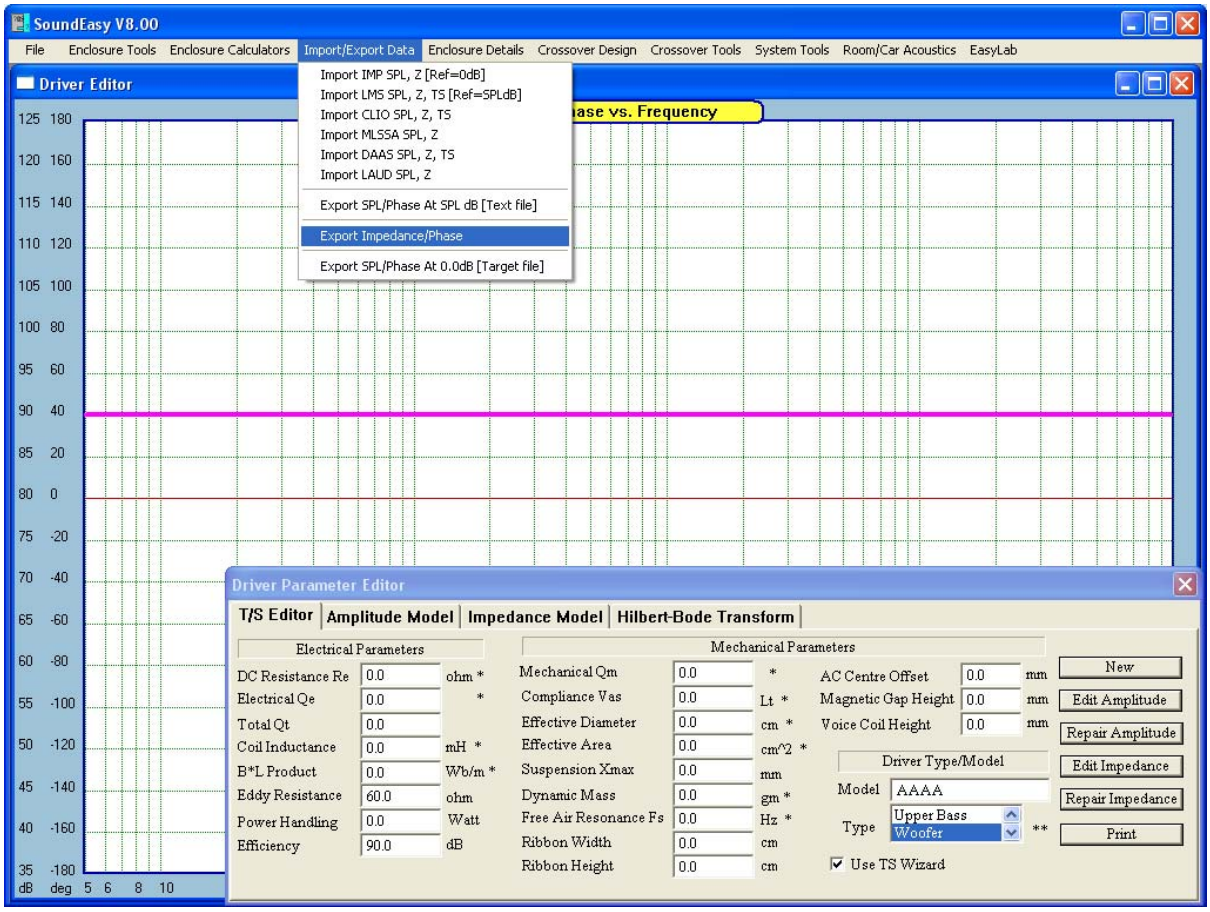


Figure 3.12. File Export Menu.

Exporting Data

SoundEasy exports SPL and impedance data into ASCII (or text) files. The SPL data can be exported in two ways:

1. **With SPL normalized to 0.0dB** – This option requires, that you select "**Export SPL/Phase At 0.0dB [Target file .trg]"** from the main menu. Target-type files can be later used as target response for Optimizers and Super-Components.

2. **With SPL exported at SPL level** – This option requires, that you select **"Export SPL/Phase At SPL dB [Text file .txt]"** from the main menu. This type of file will be also exported from measurement system.

You can export the following curves:

1. SPL / Phase and Zin / Phase from Hilbert-Bode Transform.
2. SPL / Phase and Zin / Phase from CAD frequency domain plotting screen.
3. SPL / Phase and Zin / Phase from System frequency domain plotting screen.
4. SPL / Phase and Zin / Phase from Enclosure Design frequency response plotting screen.

Before a curve can be exported, it must be obviously plotted first. Therefore, you would first open, for instance, CAD frequency domain plotting screen. Then you need to plot SPL / Phase curves and finally, select exporting option: export as a target OR export as a text file.

In the ".trg" SPL file has file the first column is the frequency, followed by corresponding SPL value and finally phase value – see below.

```
5.00    -53.43    -36.65
5.07    -52.97    -36.93
5.13    -52.52    -37.21
5.20    -52.06    -37.50
5.27    -51.61    -37.79
5.34    -51.15    -38.09
5.41    -50.70    -38.40
5.48    -50.24    -38.71
5.56    -49.79    -39.03
5.63    -49.33    -39.34
5.71    -48.88    -39.67
5.78    -48.42    -39.99
5.86    -47.97    -40.32
5.94    -47.51    -40.64
6.02    -47.06    -40.97
```

Chapter 4. Enclosure Design Function

SoundEasy enables the designer to thoroughly evaluate **thirteen enclosure types, Isobarik extension of each enclosure and multiple drivers configurations**.

1. Sealed enclosure,
2. Vented enclosure,
3. Double chamber (bandpass subwoofer Type 1) sealed-vented enclosure,
4. Double chamber (bandpass subwoofer Type 2) vented-vented box ,
5. Double chamber (bandpass subwoofer Type 3) vented-vented box.,
6. Triple chamber (bandpass subwoofer Type 4) sealed-vented-sealed box and
7. Triple chamber (bandpass subwoofer Type 5) vented-vented-vented box.
8. Passive Radiator.
9. Transmission line (TL).
10. Horn system.
11. TL/Vented system.
12. U-Dipole.
13. H-Dipole.

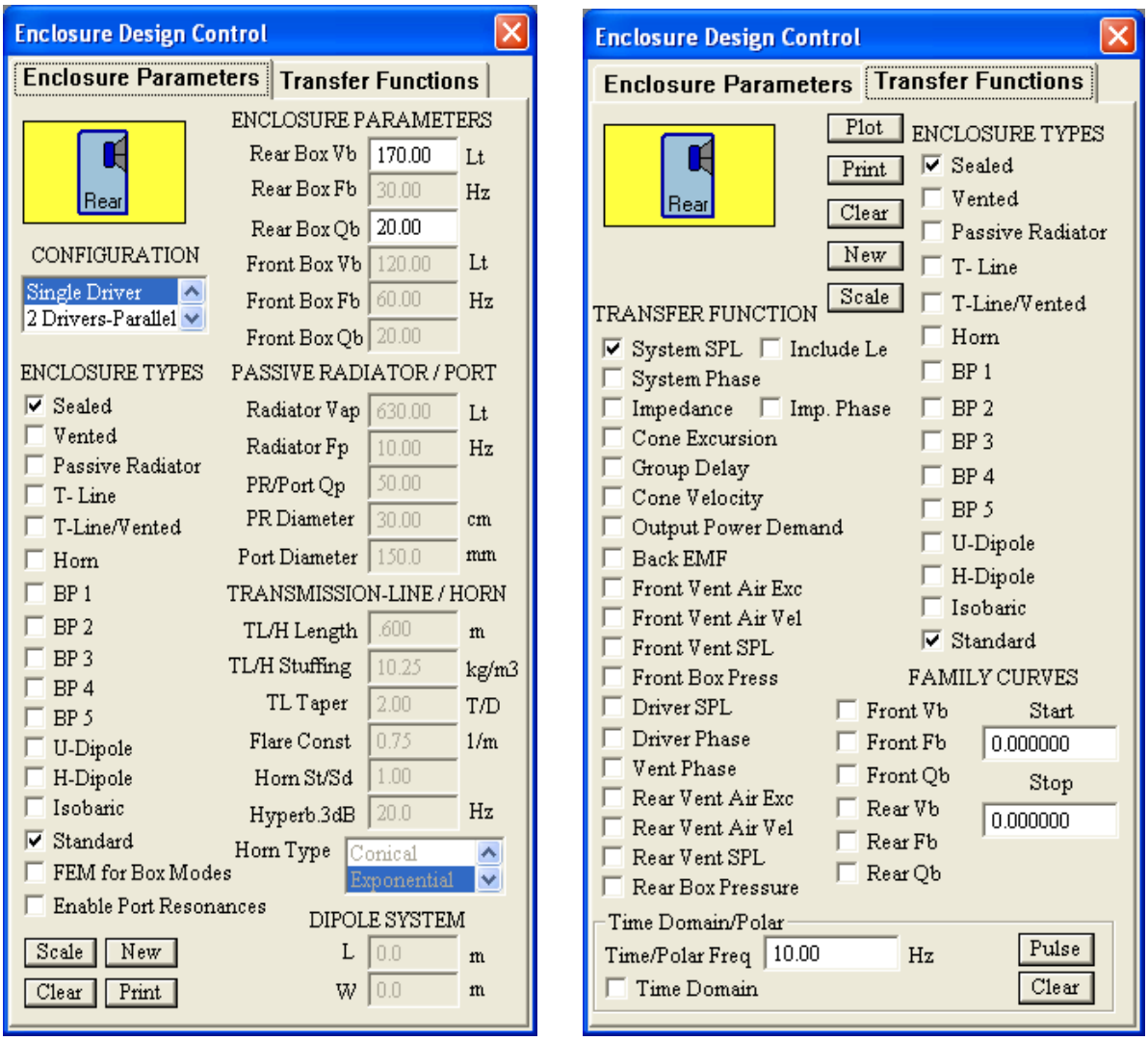


Figure 4.1. Enclosure Modeling Dialogue Box with 2 TABs.

Enclosure modelling is controlled by two TABs: "**Enclosure Parameters**" and "**Transfer Function**". Enclosure parameters are keyed-in to the first TAB and the cesond controls what get plotted. Labelling of all functions and data fields is quite obvious, therefore, will not be expanded in the text.

Enclosure design is achieved with the 'Enclosure Design' screen provided with SoundEasy and accessible via “Tools” main menu option. The recommended design method is based on a family curves approach. The 'Enclosure Design' screen enables the designer to plot a family of box responses with one parameter of the three, "Vb", being variable within predetermined boundaries. The maximally flat curve is then selected and the corresponding parameter fixed at the value that produced the best curve. Then the next parameter, affecting the enclosure "Fb" is varied and later made fixed as the flattest curve was recognized. Lastly, Qb is varied. If you have selected a vented enclosure, these three family plots will usually be sufficient to establish a good estimation of the box dimensions and tuning.

Enclosure Modeling/Review

To gain an insight into this tool please load the Test***.wfr data file and follow the example below. The Thiele/Small parameters of the test driver indicate that the driver is more suitable for a vented system. The total Q factor (Qt) is lower then 0.38 and this indicates that the enclosure volume should be less than driver's equivalent compliance volume (Vas). You could attempt to design a sealed box, but the -3db cut-off frequency of the system (Fs) will rise significantly compared to driver's free air resonance frequency (Fo). In order to model the enclosure for this woofer, the "family curves" feature of the 'Enclosure Design' screen will be explored. Start by clicking on **"Rear Vb Family"** checkbox. The resulting plots will show a family of frequency responses for enclosures having different volumes. The initial (start), and final (stop) volume values for the boxes to be examined can be entered in the provided datafields. Enter the following numbers for the requested values: Start value = **40** (lt), Stop value = **200** (lt), then check the **"System SPL"** box and then click on the **"Plot"** button. 5 curves will be plotted on the screen representing frequency responses for an enclosure with it's volume is incremented from 40 to 200 liters in 40 liter steps. The first curve plotted corresponds to 40 liter box, the second curve corresponds to 80 liter box and so on. The resulting plots would be similar to the ones shown in Fig 4.2.

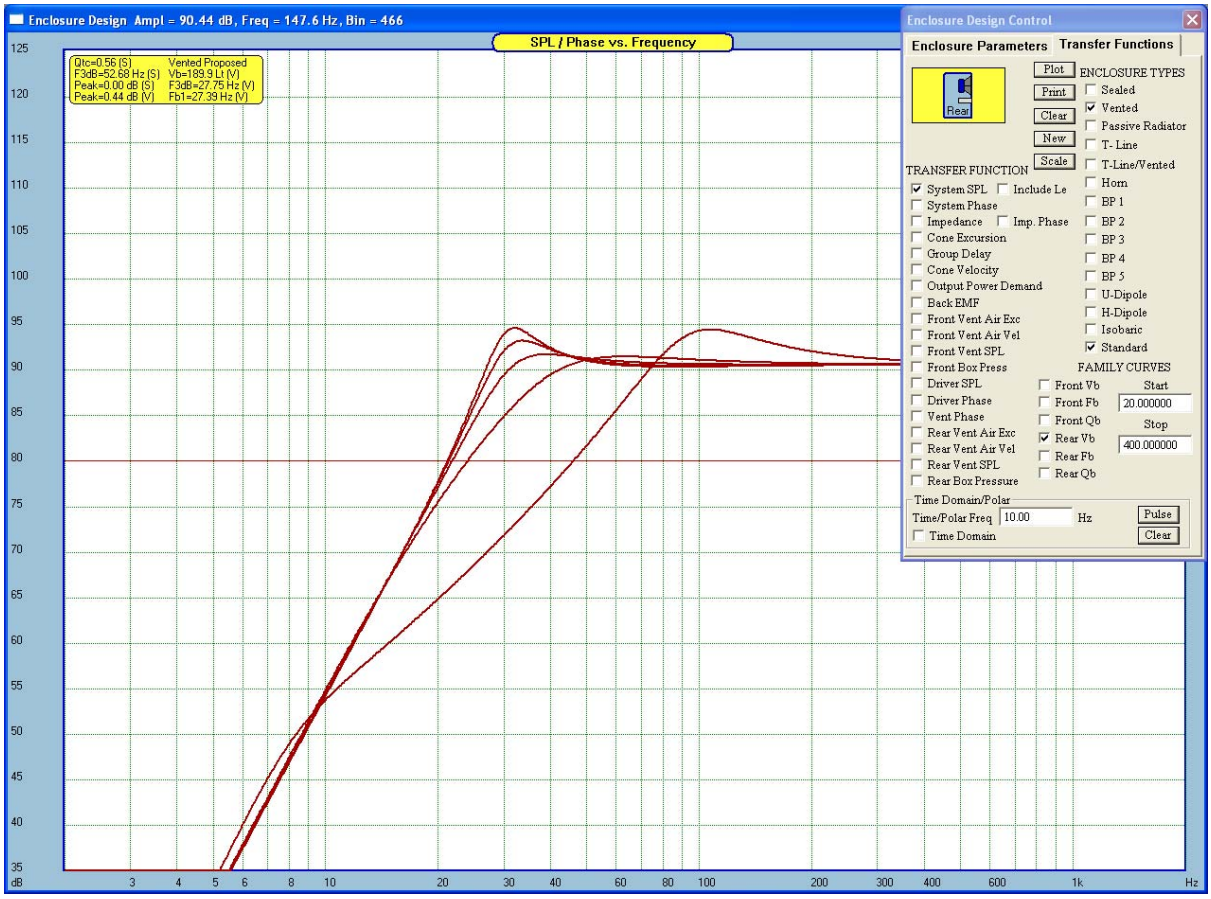


Fig 4.2 Family curves for Different Vb values.

Once the plots are examined it is observable that there is little difference between curves corresponding to box volume 120 liters and greater. In this case, one would enter 140 liters in "Rear Box Vb" data field. This is a good compromise between 120 and 200 liters and is a good starting value for selecting other parameters. Clear the screen by clicking the "Clear" button, unless of course you wish to obtain a hard copy of the plots.

Click again but this time select **"Rear Fb family**" option from the family option area. The resulting plots will show a family of frequency responses for enclosures having different tuning frequencies. Enter the following numbers for the requested values: Start value = **20** (Hz), Stop value = **60** (Hz) and then check the **"System SPL"** box. This time, 5 curves will be plotted on the screen representing frequency responses for the enclosure when tuning is incremented from 20 to 60 Hz in 4 Hz steps. The first curve plotted corresponds to 20 Hz tuning, the second curve corresponds to 24 Hz tuning and so on. The resulting plots would be similar to those shown in Fig 4.3.Examination of these curves will lead to the conclusion, that curves corresponding to 28-32 Hz box tuning, offer the flattest frequency response.

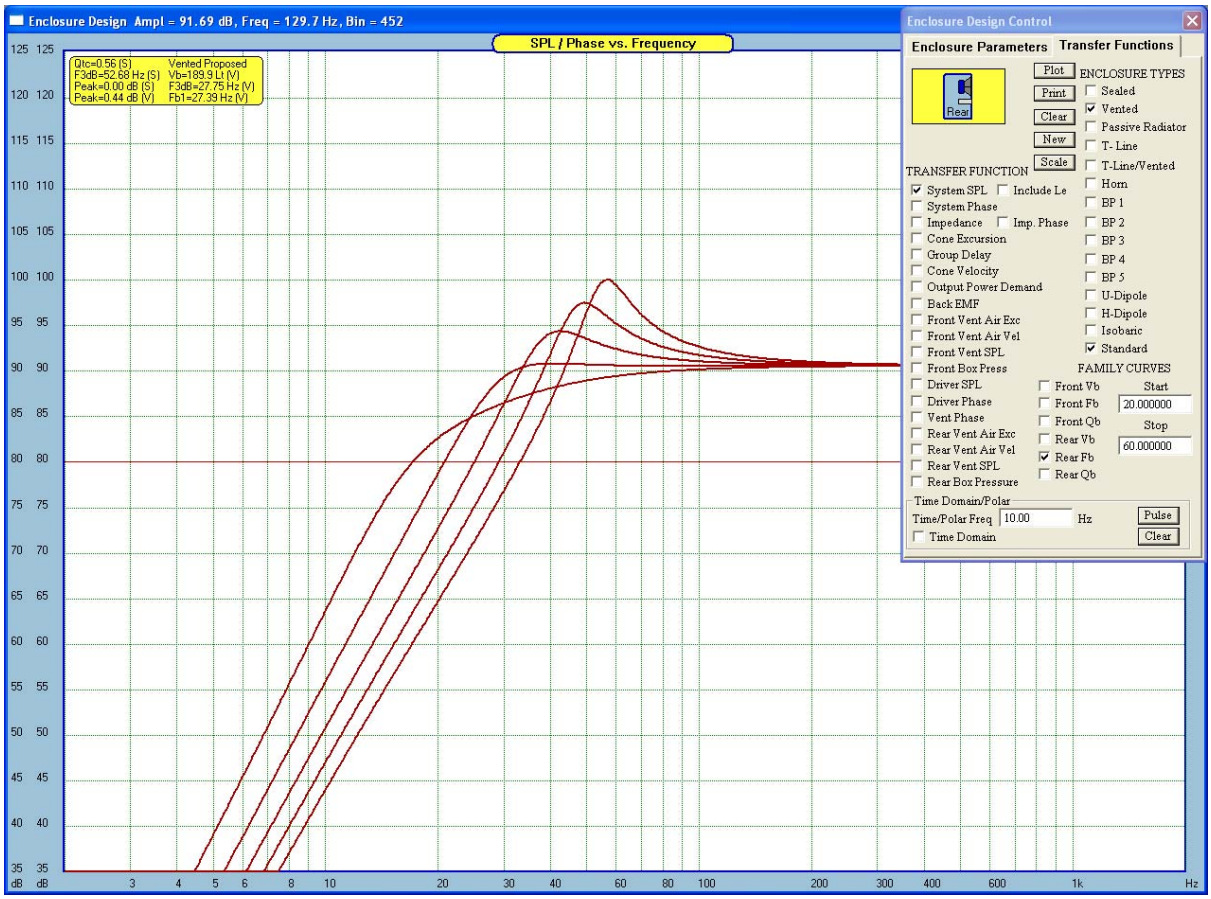


Fig 4.3 Family curves for Different Fb values

In this case, 32 Hz is the right choice and should be entered in the "Rear Box Fb" data field. One remaining parameter is left to be examined. Clear the screen by clicking the "Clear" button. Select the **"Rear Qb family**" option from the family options area. The resulting plots will show a family of frequency responses for enclosures which have different Qb factors. Enter the following numbers for the requested values: Start value = **4,** Stop value = **20,** then check the **"System SPL"** box. 5 curves will be plotted on the screen representing frequency responses for the enclosure with it's Qb incremented from 4 to 20 in steps of 5. The first curve plotted corresponds to Qb=4, the second curve corresponds to Qb=8 and so on. The resulting plots would be similar again to those shown in Fig 4.4.Examination of these curves will lead to the conclusion, that curves corresponding to Qb=8 or greater, offer the flattest frequency response. In this case, Qb=10 is the right choice and should be entered in the "Rear Box Qb" data field. It can also be observed, that the system is relatively insensitive to changes in Qb above the value of 8.0. In summary, the enclosure has the following parameters: Volume = **140** liters, Tuning frequency = **30** Hz, Q-factor = **10.0**

This design procedure offers the user an opportunity to review a number of plots for each parameter varied. The designer already has instant answers to a significant number of "What if..." scenarios. The family curves approach gives the user the choice and an assurance that your choice was a correct one. It is always recommended that the impedance curve of the driver in the enclosure and the cone excursion plot is examined. Plotting the vented system impedance curve reveals the three characteristic frequencies for the vented system. The first peak to the left is called the "anti-resonant" frequency of the system. This is the frequency at which the cone moves in opposite phase to the air in the vent.

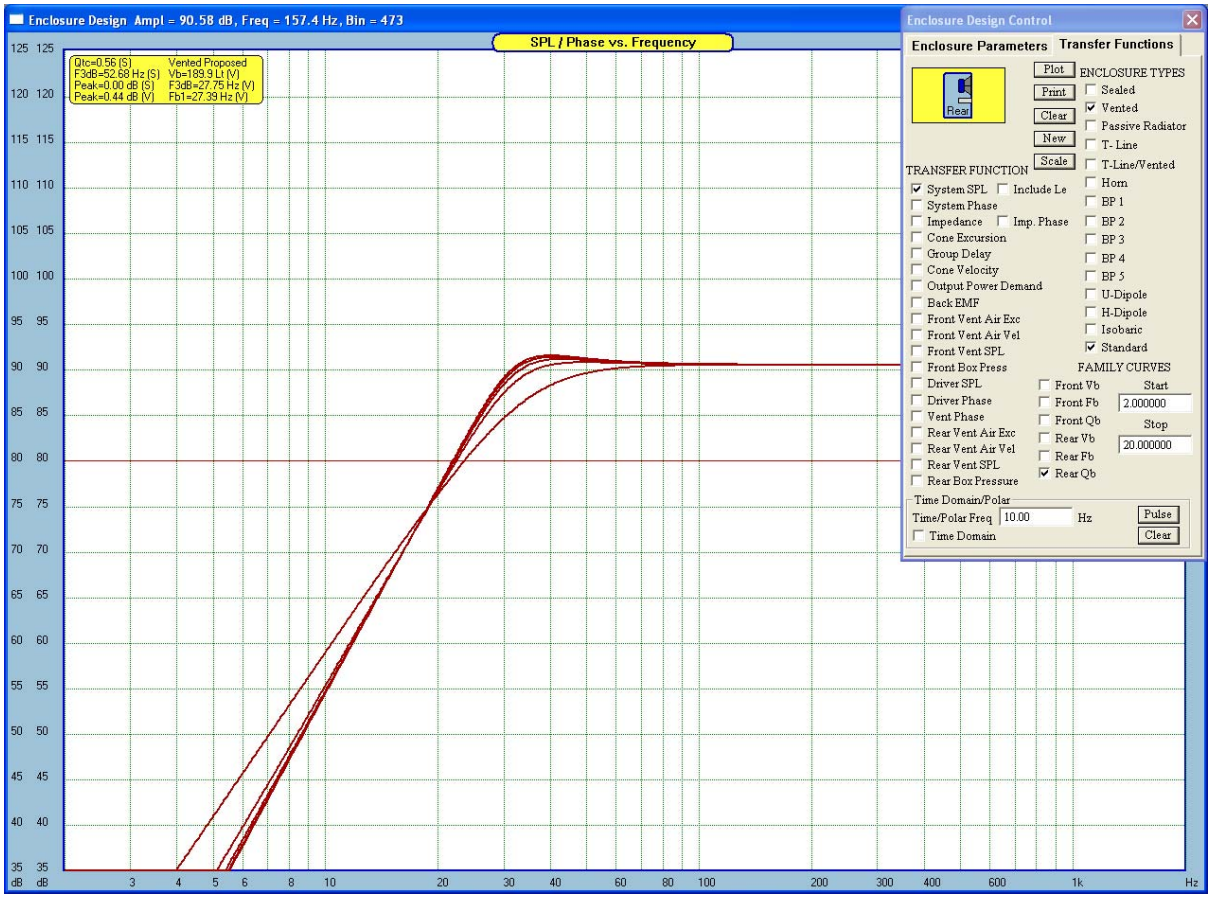


Fig 4.4 Family curves for different Qb values.

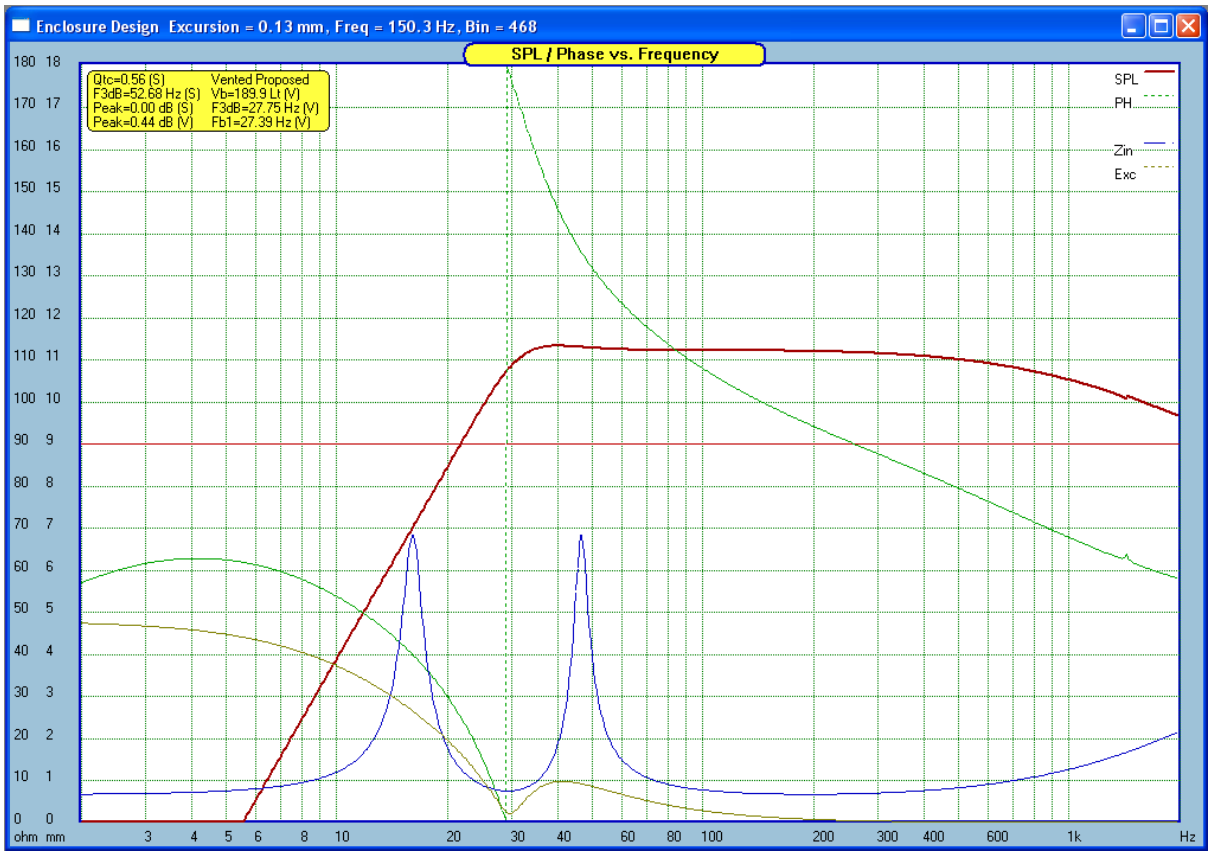


Fig 4.5 Various vented box plots

There is the pressure cancellation effect and in spite of the fact that the cone moves a long way, there is little acoustic output from the system. The middle frequency (the valley between the two peaks) is called the "enclosure frequency". An interesting fact regarding this frequency is that if the Qb could be made infinite, the cone would stop moving at this frequency. In practice this will not happen, but if you plot the cone excursion curve superimposed on the impedance curve, you will notice that the cone really moves very little at this frequency and most of the acoustic output from the system is produced by the vent. The right-most peak is the frequency at which the cone moves "in-phase" with the air in the vent. This is the frequency where the bump occurs if your driver has a total Q-factor (Qt) which is too high for the vented enclosure.

In order to plot the input impedance curve, check the **"Impedance"** and then press **"Plot"** button. Plotting the predicted cone excursion curve will show the potentially dangerous frequency range where the cone moves further then it's allowed maximum linear excursion range. For example: change "Mechanical Q" to 9.0, "Electrical Q" to 1.3 and "Total Q" to 1.2. Now, plot all three curves. The frequency response is bumpy, but suppose you could live with this. The cone excursion curve, however, shows that the cone oscillates excessively around 12-16 Hz. You could expect a turntable to rumble at those frequencies and you may well need to review your design idea in order to save the woofer from being damaged. This case is a drastic one, but it does illustrate the purpose of the cone excursion plot. In order to plot the cone excursion curve, check **"Cone Excursion"** box and then press **"Plot"** button. An example of the available plots is shown on Fig 4.4. Other plots are selected the same way. The design process for the other enclosure types are similar. The sealed enclosure is the simplest because there are only two variables: Vb and Qb. The double chamber vented enclosure (also called "bandpass sub-woofer") are the most difficult with six parameters to consider, three for the front box and three for the rear. Please refer to the relevant professional literature for a detailed analysis of the physics of this type of enclosures [3].

There is a wealth of information contained in these graphs. Users are strongly encouraged to experiment with all the parameters of the boxes and drivers. Your design goals may vary from "optimally flat alignment" type of frequency response to enclosure "no bigger then...." and so on. SoundEasy will enable you to evaluate the most esoteric driver.

Isobarik Extension

The Isobarik (constant pressure) system has the following characteristics as compared with a single woofer configuration:
1. Qts will be the same as a single woofer driver.
2. Fs will be the same as a single woofer driver.
3. Vas will be half that of a single driver.
4. Impedance will be half that of a single driver for parallel connection.
5. Sensitivity (efficiency) will be the same as that of a single driver.

The major advantage of the Isobarik system is an enclosure volume that is half that of a single driver. SoundEasy recalculates all the above data for you from a single driver specification. Thus you would enter all electrical and mechanical parameters of a driver and then press "Isobarik" button to get the resulting compound driver parameters. The information field "Standard" will now change to "Isobarik". You can always go back to the single-driver configuration by pressing the "Standard" button. User should not confuse Isobarik extension with the triple-enclosure, dual driver configuration. Isobarik extension of the triple chamber enclosure will require four drivers.

Vent Calculator

The Vent Calculator (see Fig 4.6) can be selected from the main menu. It requires four parameters: box volume (Vb), box tuning frequency (Fb), the internal diameter of the vent (Dia) and XY dimensions of the rectangular vent. When the "Calculate" button is pressed, the length of the vent is calculated. Additionally, if the edited driver file had been loaded, minimum vent diameter (Dmin) and port occupied volume (Vol) are calculated for a given driver. The Dmin parameter relates to the effective cone area of the driver, so you need to provide this data first. A hard copy of the calculations can be obtained by pressing the "Print" button from the Calculator dialogue box and "Copy" function works the same as for the main screen.

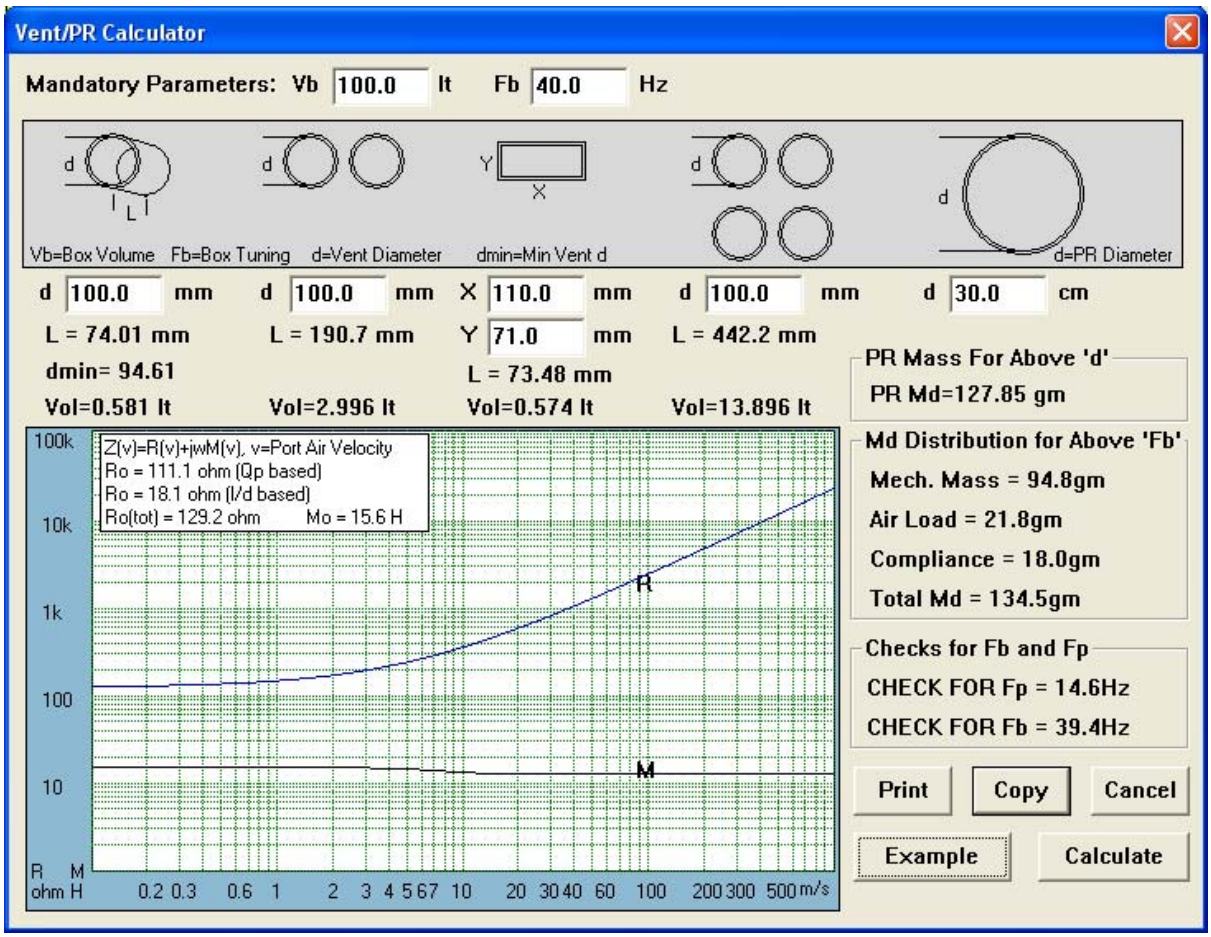


Figure 4.6 Vent Calculator

Enclosure Shapes

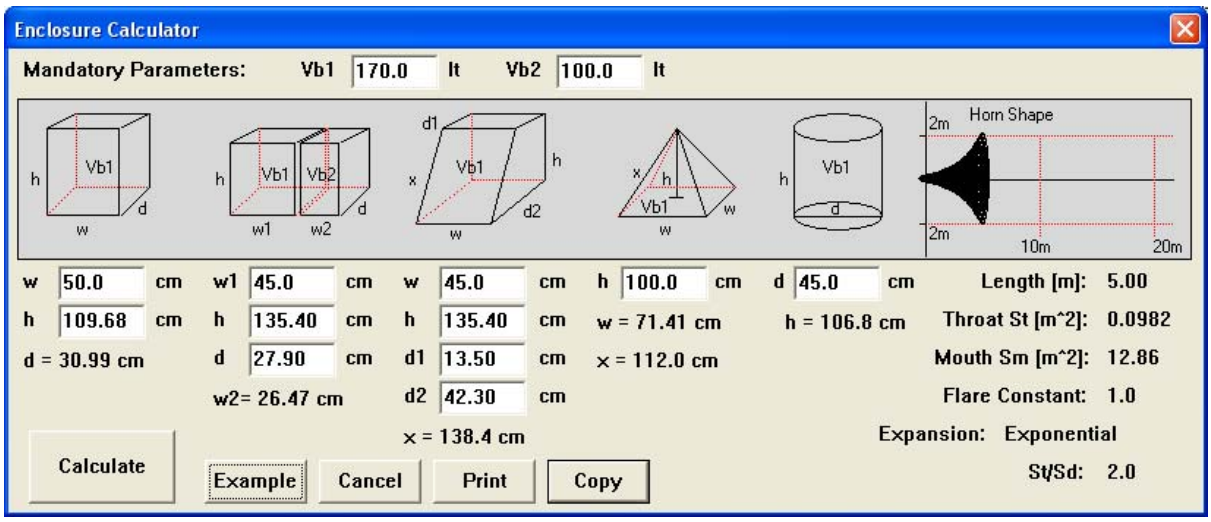


Figure 4.7. Enclosure schapes calculator

The Enclosure Shapes Calculator (see Fig 4.6) can be accessed from the main menu as well. It requires two mandatory parameters: (1) rear enclosure volume and (2) front enclosure volume - if used. Also, before you press “Calculate” button, the length of one of the sides must be entered to give the program a starting point. If you disagree with the second and/or third dimension calculated by the program, please enter your own value into the second dimension data field and press “Calculate” button. The program will accept your number and will recalculate the third dimension.

All parameters provided by the Enclosure Shapes Calculator are the internal dimensions of the enclosure. It is recommended to increase the required volume by 10% to accommodate for bracing, crossover, volume of space occupied by the drivers and the vent(s). “Copy” to Clipboard function is also provided.

Adjusting Vertical Scales

At times, it may be beneficial to increase/reduce resolution of the vertical scales. The adjustment can be accomplished from a dialogue box which pops up when selecting “Calculators” -> “Scale Resolution” option from the main menu. If you agree with the default values being displayed, just click on “Done” button. Otherwise, edit the required resolution and then press “Done” button. Please note, that amplitude response resolution must be entered as an integer number. Please refer to Fig 4.8

1. Curve on the left was plotted using Centre = 90, Resolution = 5.
2. Curve on the right was plotted using Centre = 90, Resolution = 1 settings.

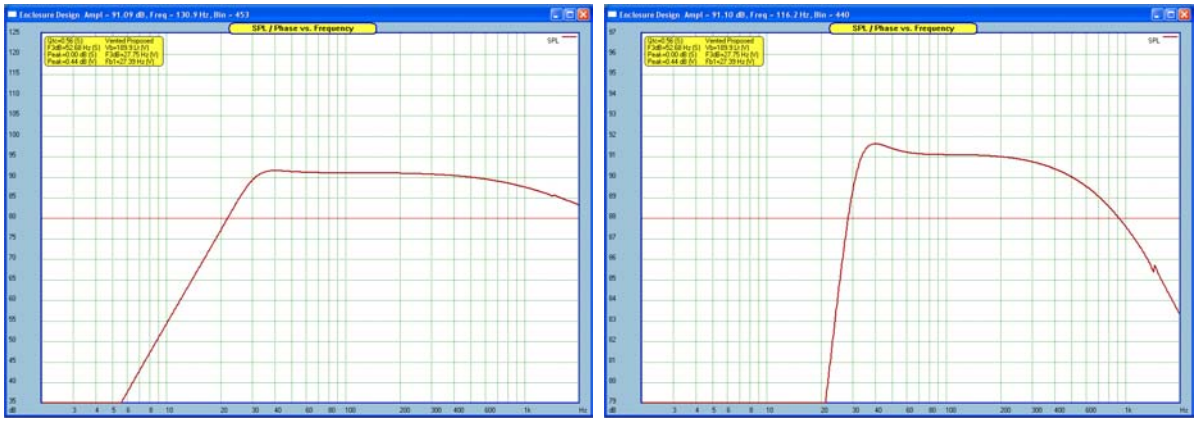


Figure 4.8. Adjusting vertical resolution.

Multiple drivers

There are four types of driver configuration available to the user :

1. **Single driver**.
2. **Two drivers connected in parallel**. In this configuration, the efficiency of combined woofer drivers is 6dB higher than that of a single driver. This figure splits into 3dB gain from parallel connection - each driver receives the same power equal to that of a single driver and 3dB gain due to mutual radiation impedance. Input impedance will be halved and Vas(total) is twice of the single driver. Power handling is also doubled.
3. **Two drivers connected in series**. Here, the efficiency of combined woofer drivers is the same as for single driver. Although there is 3dB gain from mutual radiation impedance, this will be cancelled by halving of the electrical power delivered to each driver. Input impedance will be twice of the single driver and Vas(total) and power handling is also twice of the single driver.
4. **Four drivers connected in series/parallel** (Quad Box). In this configuration, the efficiency of combined woofer drivers is 6dB higher than that of single driver. Vas(total) and power handling are four times that of the single driver. Input impedance is the same as the single driver’s impedance Please note, that midrange drivers may have sealed pots already attached, hence Vas considerations may not be relevant. Also, mutual radiation impedance may not be applicable to midrange drivers. Configuration can be changed from a list box provided on the right side of the ‘Enclosure Design’ screen and user’s choice will be saved in the data file. The effect of mutual radiation impedance will be more evident at lower frequencies, resulting in the 3dB efficiency gain mentioned above. Fig 4.10 shows, that for a separation of 1 meter between the drivers, the 3dB gain was maintained up to 100Hz for this particular sealed system with two drivers connected in parallel. The weighting factor used for modelling the mutual radiation impedance is expressed below.

$f(kd) = (1 + \sin(kd)/(kd))$
where $k = (2*3.142*f)/c$, $c = 3.44$[m/s], f = frequency [Hz], d = separation [m].

Screen Cursor

Screen cursor, incorporated in the "Enclosure Design" screen, has been implemented as an "Automatic Pointer" on the plotting area of the screen. The cursor is activated by clicking LEFT mouse button above the plotting area and X-axis cursor movements point to the desired frequency, at which you want the SPL, Zin and other data to be displayed..

Active Equalisation

In a generalised case of tunable enclosures, it can be observed that the system's frequency response is already down by 3-4dB at the enclosure tuning frequency. Without equalisation, this would be the specified system -3dB cut-off frequency. Also, cone excursion curve superimposed on the amplitude response plot reveals that cone excursion has a dip at the enclosure tuning frequency. One can take advantage of this and feed more power into the driver while still avoiding mechanical failure and any significant increase in distortion. Obviously, the driver must be able to handle increased electrical power. One can take this concept even further and deliberately tune the enclosure to lower frequency than that which the optimum flat response would require. Any irregularities introduced in the frequency response this way can be electronically equalised with the added advantage that the cut-off 3dB frequency would be still lower. It is this particular advantage of tuned enclosures that makes them particularly suitable for active equalisation. Here, one must assure the peak of the equalisation matches the enclosure tuning frequency, where the dip of the cone excursion resides.

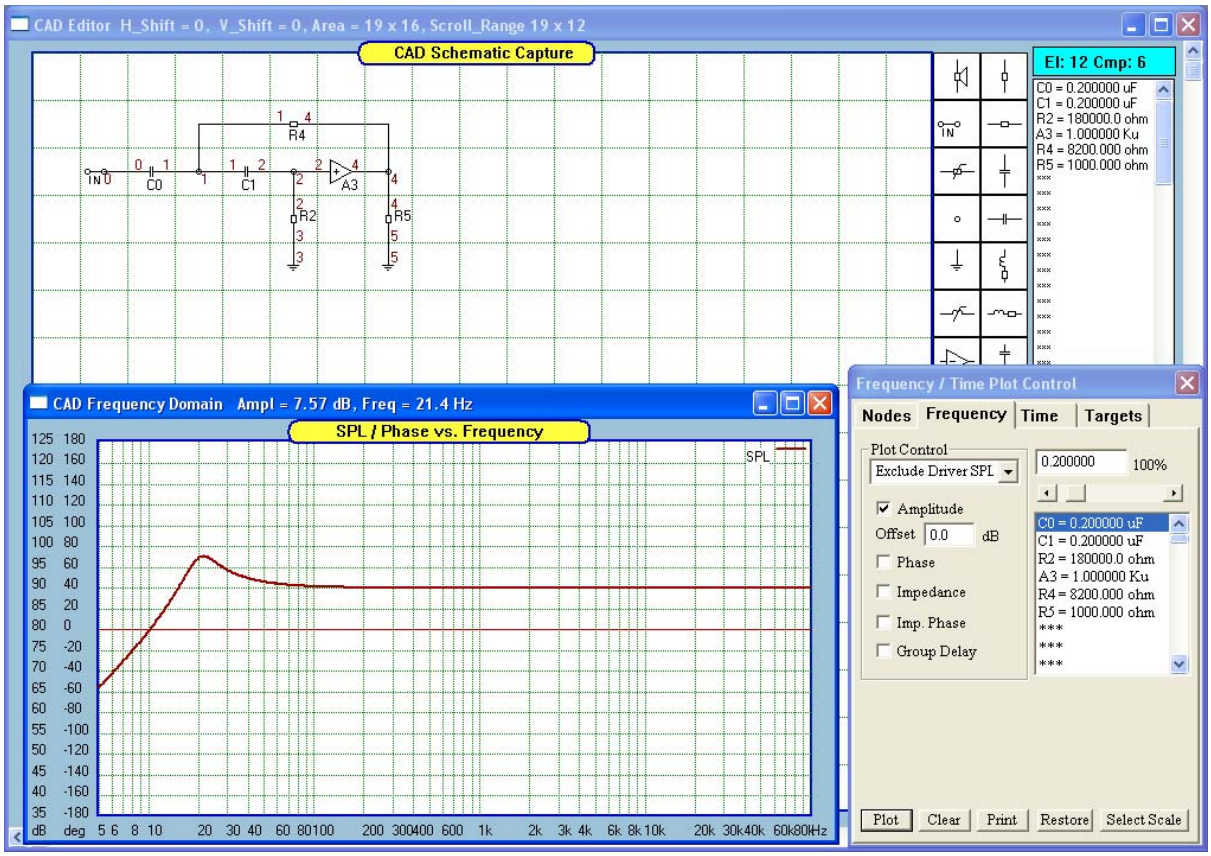


Figure 4.9 Active EQ circuit.

The second factor taken into account is a need for "mirror image" curves for active boost and passive driver frequency roll-off. The third factor is matching power handling of the driver with electrical peak power required. Securing the above three conditions protects the driver and provides optimum bass extension. SoundEasy facilitates modeling of the above concept. As we see later, this idea is highly recommended for reliable and excellent results when attempting to extend your system response for maximum low frequency output. The starting point is to create an active compensation circuit using CAD screen in the "Crossover" module and re-load this file back into the "Box" module for modeling of combined behavior of enclosure + active equaliser. The active circuit is design in a usual way described in the following sections of the manual with one limitation: the circuit diagram **must NOT include loudspeaker symbol**.

If the loudspeaker symbol was included in the schematic, SoundEasy would automatically attempt to use driver's input impedance values, which may or may not have been created by CAD plotting screen. Now, even if the impedance curve was plotted, it would be fixed for a particular type of enclosure over 10-40000Hz frequency range, so its data points would not match with 5-2000Hz frequency range of the Enclosure Design screen and various types of enclosures available.

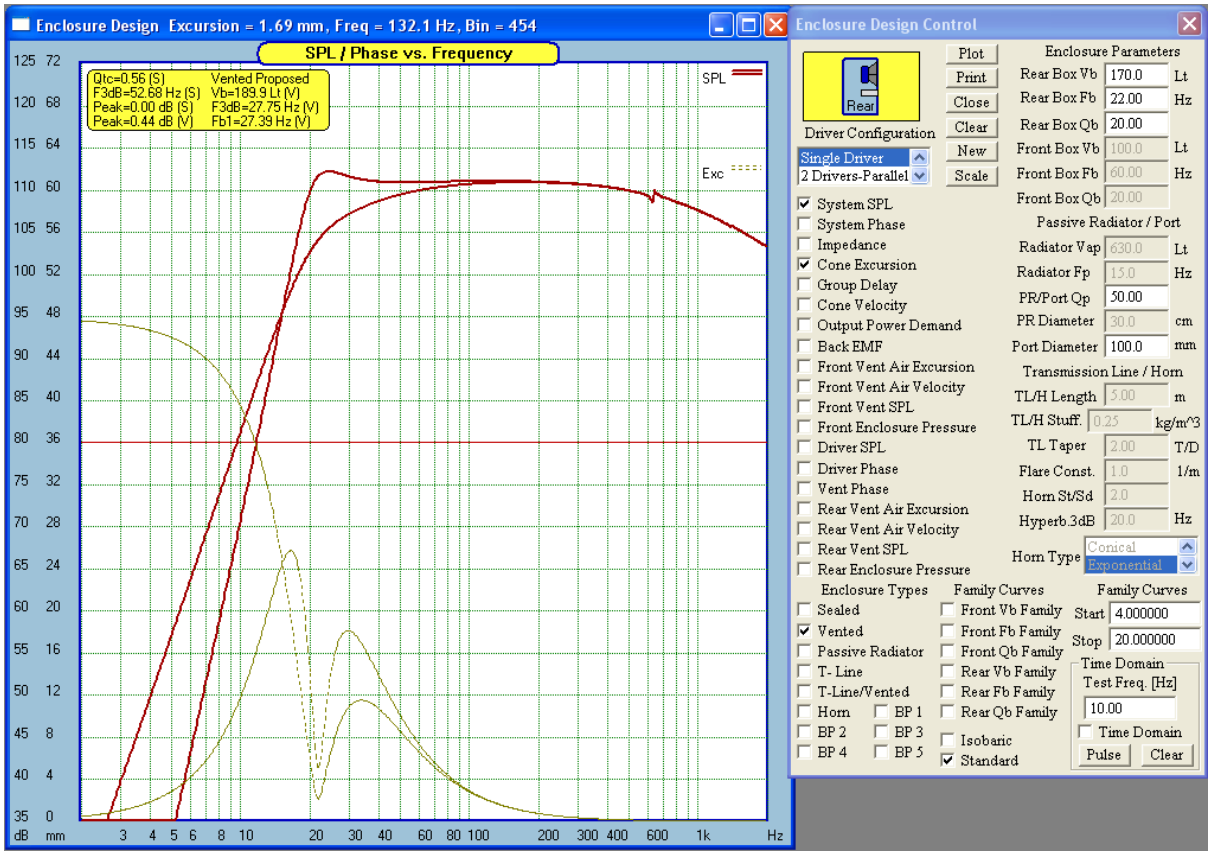


Fig 4.10 Vented enclosure "Unassisted" and with EQ Circuit.

The process is now explained with the help of the following example. The active equaliser used in the model is a second order high-pass filter (Sallen and Key type). Rather than selecting a more complex and sophisticated equaliser, this (possibly the simplest) active equaliser has been chosen deliberately to illustrate the main issue at hand – demonstrating the process and avoiding pitfalls of active equalisation. Filter values and schematic are shown on Figure 4.9. The filter has the following main characteristics:

1. The amount of boost is around 7.5dB at 21Hz.
2. For 100W active subwoofer, the system would require 562W peak electrical power.
3. 12dB/oct high-pass filter protects the system against excessive infrasonic frequencies.
4. Amplifier voltage gain is equal to 1.

The filter can be built using any popular, low noise operational amplifier and can be inserted in the audio path just prior the main power amplifier. The all-important system responses are evaluated at 100W input power fed into the loudspeaker. As previously explained, plotted curves represent system performance after the temperature of all components has stabilized. Figure 4.10 shows the relevant transfer functions. It is immediately observable that even at the equalization peak (562W input power), the cone excursion is well within the safe operating range, at around 24mm maximum linear displacement for this long-throw woofer. The -3dB low-end cut-off point of the combined system moved down to 17Hz (from 30Hz for un-equalized system), which is better than was expected. Usage of the high-pass filter has visibly dramatic effect on the cone excursion at very low frequencies, providing excellent protection for the driver. Generally, this system would pass as a very good subwoofer. As far as input power is concerned it would be advisable to check if one could operate the model system at 560W of power (100W system or 560W peak).

It may be useful to note that several well known manufacturers offer home Hi-Fi class loudspeaker drivers capable of 1000W and more electrical impulse power. At this stage, careful analysis of driver's electrical power ratings and thermal analysis are recommended.

If thermal resistances of voice coil-to-magnet and magnet-to-air are known, it will be possible to determine power compression factors for any level of continuous input power. For comparison, Fig 4.11 shows the same system with the vent blocked. This would convert vented enclosure to a sealed box. Although the cone excursion of the passive system is significantly lower below 20Hz, the active equalization causes cone excursion to exceed the 25mm limit at 21Hz by up to 50% (!). Also, despite the 7.5dB of equalization, the amplitude response of the sealed system is still 6.0dB below vented system at 20Hz and began to roll-off at 40.0Hz. Another 6.0dB of active boost would be required, which would make the cone excursion and THD problem even worse. It needs to be noted that sealed enclosure of the same size as vented enclosure is not suitable for this driver. However, it does illustrate cone excursion problems when a sealed boxes is actively equalized. Analysis of the two cases would indicate that tunable enclosure would be the preferred choice when active equalization is considered. The combined vented system would provide excellent low frequency extension, and, if used with properly designed high-pass filter, mechanical integrity of the driver can be assured. Combined vented system is sensitive to misalignment between the following parameters: (1) enclosure tuning, (2) amount of low frequency boost required, (3) frequency of maximum boost. Mismatch in the above conditions seems to be the most commonly encountered difficulty when attempting active equalization. In the context of active equalization, tuning of the vented enclosure could be viewed from a different angle.

Traditionally, box tuning frequency, Fb, would be chosen with the optimally flat frequency response of the passive system in mind. With the help of an active network, box tuning frequency can be selected as the lowest frequency for which the "detuned" vented enclosure can still be equalized, taking into account all power and cone displacement related issues. Evaluation of several transfer functions, cone excursion being one of the most important, is needed for successful design outcome. Overlapping all relevant curves greatly improves understanding of correlation between all vital parameters and trade-offs being made. The accuracy of the analysis was further enhanced by including thermal/power compression effects.

Fig 4.11 Vented enclosure (upper curve), Sealed enclosure (lower curve) + EQ circuit

Loudspeaker Diffraction Distortion and Radiation Impedance

Diffraction distortion (or diffraction loss) appears to be one of those subjects, that easily attract a number of "pro" and "against" arguments. When considering a typical domestic listening environment, where some sound reflections are inevitable anyway, one may stop to ponder - should we make a big fuss about it or not.

There are a few facts to consider first. (1) Even a medium size front baffle (60cm x 60 cm) produces diffraction distortion in order of +9dB at 400Hz (accordingly to Olson [62]). This is well within woofer operating frequency range for 2-way and 3-way systems. The diffraction loss is the easiest to observe in the anechoic chamber, but it exists in most situations where the speakers are not radiating into half-space. (2) Contemporary dome midrange and tweeter drivers are specifically designed to have wide radiation angle (omnidirectional) and therefore be prone to diffraction distortion. Diffraction effect will typically add +6dB at higher frequencies for asymmetrically mounted drivers. (3) Many currently available loudspeaker systems are placed sufficiently far away from the walls of the listening room to be subjected to diffraction distortion of varying degree. (4) Listening rooms of today are being acoustically treated for the best balance of the direct and reflected sound. This may be even unintentional, as simply having a carpet, drapes and some soft furniture in the room. In fact, Christensen [63] for example, develops several useful rules for improving listening pleasure in typical room, with the emphasis on reducing 1-st reflections. Therefore, typical listening environment is nowhere near the "reverberant room", which reflects all the sound produced inside it and makes the diffraction distortion difficult to separate. Finally, there is the issue of consistency in the design approach. Here is one point of view. It is known, that room modal response at low frequencies (standing waves) causes 30dB variations in the low end of the frequency response of the loudspeaker, but this has NOT deterred anybody from using technically sound approach (Small/Thiele parameters) for proper design of the enclosure.

Why should this approach be limited only to the low end of the audio spectrum ?. In the mid to high frequency range room reflections will also distort the frequency response of the loudspeakers, so should the design methodology be relaxed and the diffraction distortion neglected in this frequency range ?. We would advocate careful consideration of the "whole picture".

At least as long as loudspeakers are tested and compared in anechoic chambers and placed away from the walls in well designed listening rooms. You see, at some point of the design process, you need to gain the confidence, that you are creating the best source of sound you can come up with. And if it still does not sound right to you, it must be the listening room environment, which needs corrections - NOT the speakers. And if you ever even think about having your speakers evaluated in an anechoic chamber, diffraction distortion should become your daily bread. Situation becomes even more complex if several identical drivers are placed on the front baffle. Mutual radiation impedance effect becomes evident strongly at lower frequencies with the cut-off frequency determined by the geometry of the design.

SoundEasy presents one possible way of modeling the combined effect of diffraction distortion and radiation impedance for multi-driver designs operating in anechoic environment.

Diffraction distortion

Loudspeaker diffraction loss has been studied and discussed by engineers and researchers extensively over the years. Available test results support the scientific findings and several models have been proposed to adequately quantify the phenomenon. Current discussions revolve around selecting the fastest and most accurate algorithm modeling the phenomenon on currently available computer hardware.

Plainly speaking, when testing in the anechoic chamber, at high frequencies the speaker is radiating into "half space" (**2π**) i.e. it is only radiating into the forward hemisphere. At low frequencies the speaker radiates into "full space", (4**π**) exhibiting a loss of bass when implemented in typical speaker enclosures. The difference in the SPL is 6dB and is referred to as the "6 dB baffle step" or the enclosure's "diffraction loss". The location of the "baffle step" on the frequency scale is dependent on the dimensions of the baffle. The smaller the baffle the higher the transition frequency. Diffraction Loss modeling performed for this article is based on the GTD (Geometric Theory of Diffraction). Figure 4.12 below explains basic idea of GTD. The loudspeaker is mounted on a baffle and it generates certain SPL level at the "Observer" location. The observer receives a combination of direct (A) rays and diffracted (B) rays.

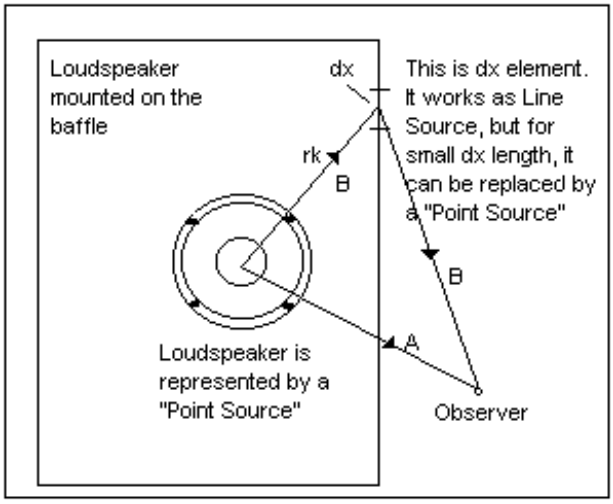


Fig 4.12. GTD concept.

Loudspeaker enclosure or baffle and the placement of the driver contribute up to +10dB to the frequency response of the system. The GTD using ray model is applied to determine the exact amount of SPL deviation due to the diffraction. In this model, sound rays, B, propagate along the surface of the baffle and are scattered when encounter the edge of the baffle. This secondary sources combine with the direct rays, A, produced by the loudspeaker and the resulting frequency response is far from being flat. To calculate the contribution of baffle edge, total length of the baffle edge is quantised into a number of sections of length ***dx***. The average distance from each section to the "point source" representing the loudspeaker is ***rk***. If ***dx*** is made sufficiently small, it can be replaced with a "diffraction point source".

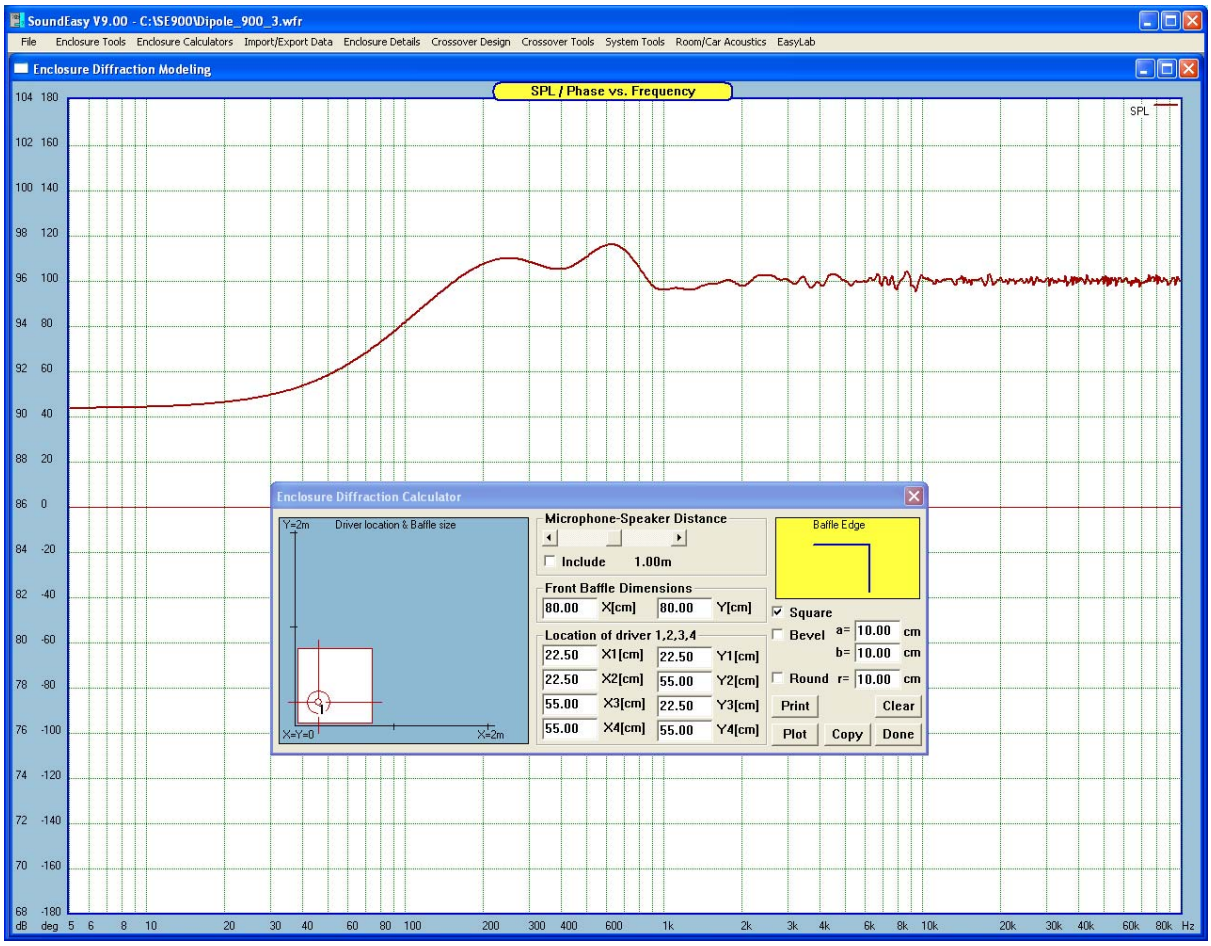


Figure 4.13. Single driver diffraction – 2dB/div, 90dB Ref.

Knowing the SPL of each "diffraction point source" and the distance to the observer, it is possible to predict the total SPL from the driver and all of the diffraction sources. We have selected 48 ***dx*** line elements to represent top and bottom edge of the baffle and 96 ***dx*** line elements to represent the left and right side of the baffle. Loudspeaker driver is modeled as 8 "point sources" contributing to the direct SPL. For the purpose of modeling the diffraction loss, we also assume, the reference distance to the observer to be the standard 1 meter. This situation is depicted on Figure 4.13. You can also change loudspeaker location on the front baffle, the dimensions of the baffle and microphone distance. **Asymmetrical mounting of the loudspeaker will result in "smoother" frequency response**. The enclosure diffraction effect is much less pronounced when the test microphone is placed much closer to the loudspeaker. The distance that can be preset the "direct" sound (ray A) has to travel is only 1cm, but the "diffracted" (ray B) wave has to travel to the edge of the enclosure and back, therefore it will arrive at the test microphone at much lower level. This situation is depicted on Figure 4.14. Here, the level picked up by the microphone is much higher due to close proximity to the speaker, and at the same time, the ripples due to enclosure diffraction are almost non-existent. What does this mean for you at home situation ?. One possible approach to modeling and testing loudspeakers is using 2π (half-space) approach advocated by LMS "pit measurement" technique to obtain frequency response of all drivers used in the system. This is good approach and guarantees, that all drivers (including woofer) radiate into half space, so their respected SPL levels are +6dB over the 4π technique. This is not a problem, as loudspeaker modeling software facilitates shifting the SPL curves up or down. However, if the system design is based on those measurements and tested in anechoic chamber, the diffraction distortion will manifest itself quite clearly. Now the system radiates into 4π space and the enclosure diffraction loss comes into play producing relative loss of bass below certain frequency determined by the enclosure geometry. The **tweeter will continue to see 2π radiating space** thanks to the enclosure diffraction. A simple solution to this problem, that has been around for some time, is to pre-distort the frequency response of the woofer crossover to account for the enclosure loss (or gain to be more precise) and attenuate tweeter accordingly.

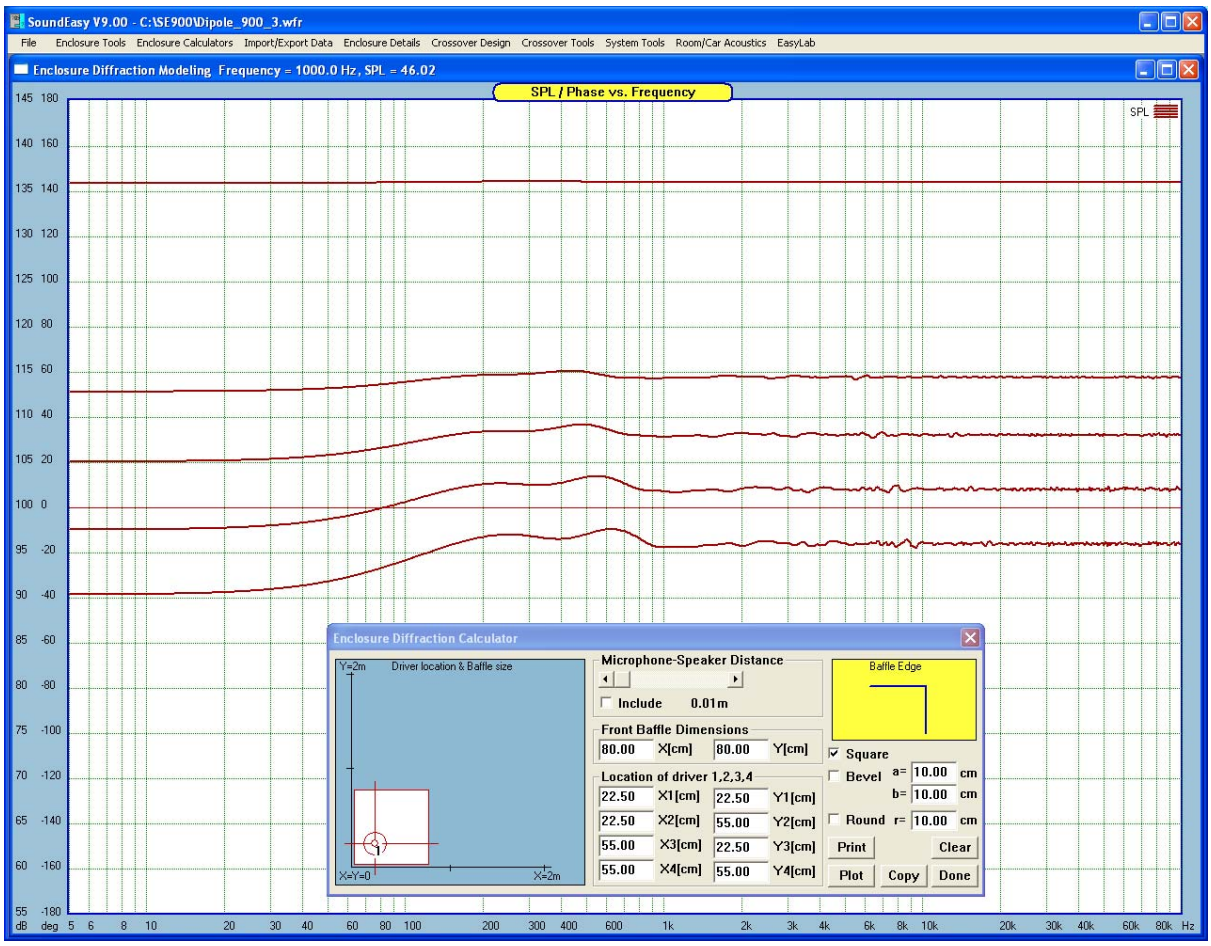


Figure 4.14. Near-field diffraction – 5dB/div, 110dB Ref.

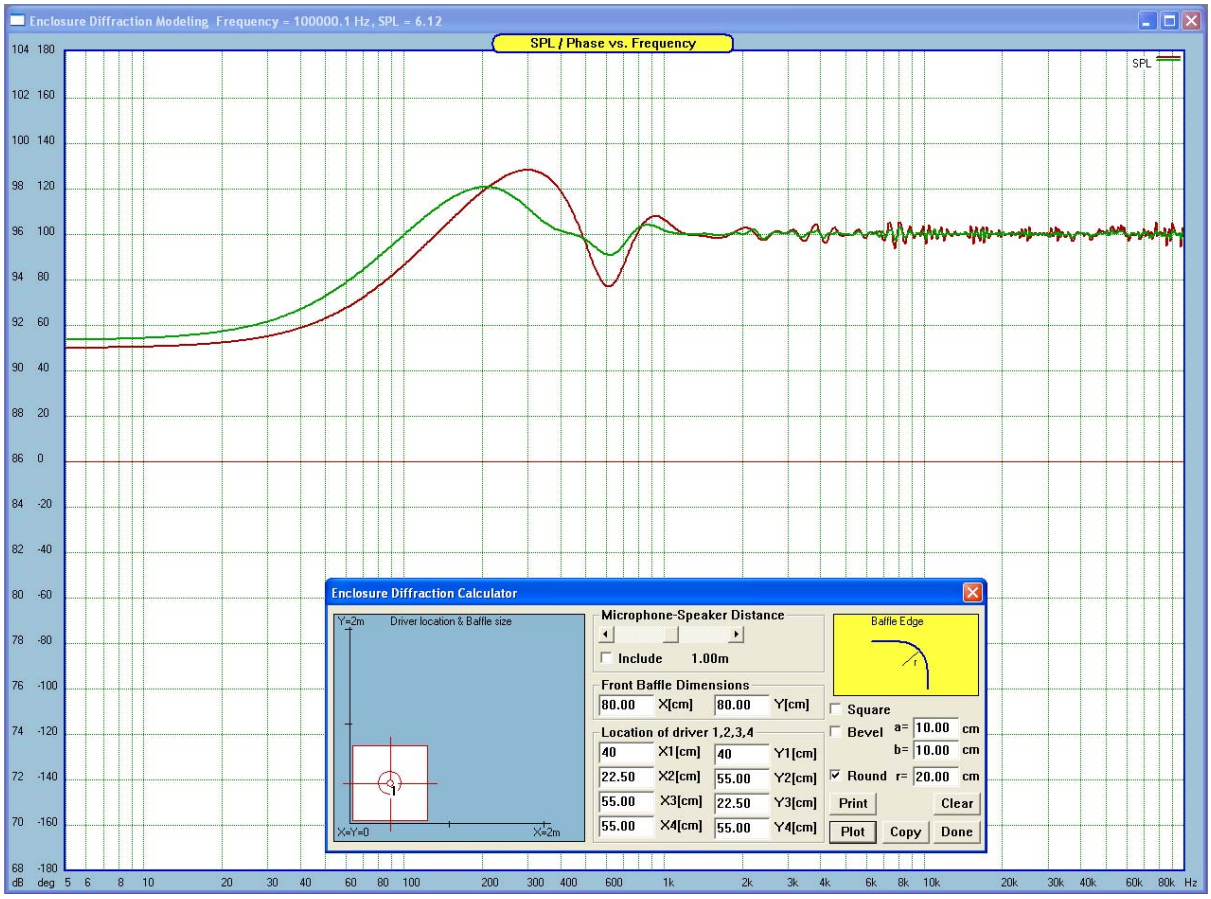


Figure 4.15. Trying different edge finish – Square (brown) and round (green).

Significant reduction in diffraction ripples can be accomplish by simply rounding the edges of the enclosure. There are three options for enclosure edges:

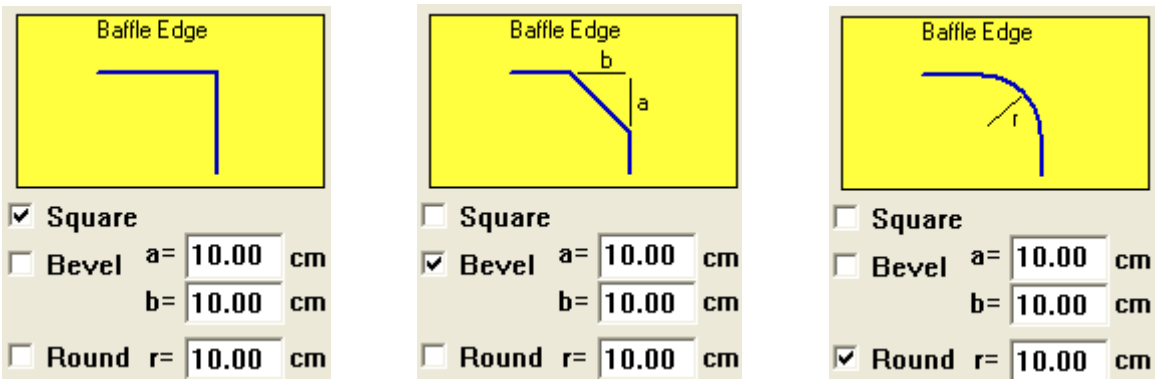


Figure 4.16. Three enclosure edges: square, bevel and round.

1. Square – there is no extra parameters to enter.
2. Bevel-type – there are two extra dimensions to enter: a and b as shown on Figure 4.16
3. Round-type – there is only the radius, r, to be determined – see Figure 4.16

Mutual Radiation Impedance

When two loudspeakers are mounted on the same baffle and fed the same signal, one driver starts to produce additional pressure on the other, increasing its radiation impedance. The next logical step is therefore to determine power radiated by two sources mounted on the same baffle.

Vanderkooy and Lipshitz [58] examined a simple case of two pistons mounted in an infinite baffle and proposed an elegant formula for expressing radiated power into the far-field taking into account self and mutual radiation impedance of source1 (piston1) coming from itself and from piston 2 as:

$$W = [1 + \frac{\sin \omega ad / c}{\omega ad / c}] * \frac{4\pi p_1^2}{\rho c} = k * \frac{4\pi p_1^2}{\rho c}$$

where d, is the distance between the pistons and a, is the radius, identical for both pistons and ω=2πf, with f being the frequency of the applied signal. For low frequencies, the above formula reduces to:

$$W = \frac{8\pi p_1^2}{\rho c}$$

The above result is four times (or 6dB SPL) the single source result. Factor k, plotted for single driver vs. frequency (Figure 4.15) exhibits 3dB raise at low end of the spectrum and 0dB at the high end of the frequency range. For two drivers, the curve is up by 3dB and it can be observed, that 3dB gain in SPL is attributed to doubling the electrical power supplied to two drivers connected in parallel. Additional 3dB gain in SPL is due to mutual radiation impedance effect. Engebretson in [59] indicated, that this additional increase in efficiency will hold to a frequency above which the diaphragms no longer "couple". This phenomenon has been experimentally verified by Gander and Eargle in [60]. They have performed comparative measurements on single subwoofer loudspeaker vs. an array of 8 subwoofers noting increase in SPL at 30Hz as 21dB. Of this gain, they attributed 9dB to 8-fold increase in input power (8 = 2x2x2 = 3dB+3dB+3dB) and 9dB to mutual coupling increasing 3dB per doubling of units. Additional 3dB gain was due to slight increase in directivity index of this large array. Also, Keele [61], investigating the performance of Bessel Arrays concluded that an array of two loudspeakers simply connected in-parallel, exhibits maximum SPL increase of 6dB, but only up to a frequency where the sources are about 1/4 wavelength apart.

Signal Summation Strategies

Power Summation and Phasor Summation strategies have been the two main methods used in prediction programs. The Power Summation method assumes, that phase interaction between arriving signals can be ignored and only mean square pressures are added. Using this convention, the SPL of two equal signals would increase by 3dB. Summation example for N identical sources p, is shown below.

$$p_{rms}^2 = \sum_{i=1}^{N} p_i^2$$

The Power Summation technique will sometimes produce different result from the Phasor Summation technique. However, it works well in the case of mutual radiation impedance, because the drivers, are closely coupled and are driven from the same source (amplifier), so they can be considered coherent (radiating signal with same phase). Therefore, the SPL increase is the same (+6dB) as would be if the Phasor Summation method was used. This assumption holds only up to certain frequency and is dependent on the geometry of the system.

The Phasor Summation method computes the phase and magnitude of each arriving signal and sums them as vectors (complex addition). Using this technique, the predicted SPL for two equal signals will by 6dB greater than the SPL produced by single source. Again, summation example for N sources p, driven by a common signal is shown below.

$$p_{rms}^2 = \left[\sum_{i=1}^{N} p_i^2 \cos \phi_i \right]^2 + \left[\sum_{i=1}^{N} p_i^2 \sin \phi_i \right]^2$$

The Phasor Summation technique was used in calculating diffraction from the enclosure edges. As you may recall, we specifically looked at each signal's path length and added all arrivals as vectors. It is fair to point out, that Phasor Summation method may produce results that are mathematically quite correct, but would appear worse than listening tests would indicate.

This disparity is associated with the ear's critical bandwidth and suggests, that reproduced sound should not be evaluated based on single frequencies, but rather on 1/3 octave bands. This is rather interesting, as the optimum modeling concept seems to be a combination of Power Summation and Phasor Summation techniques. In any case, for a truly representative modeling the sound pressure magnitude and phase needs to be included in the prediction process.

Combining All the Above

We are now in good position to review the SPL gains due to:
1. enclosure diffraction,
2. mutual radiation impedance and
3. multiple drivers in the same enclosure.

We assume that: (1) our amplifier is an ideal voltage source - this is the case of most currently available amplifiers. The amplifier will deliver 2.83VRMS to the load. (2) all drivers are identical and have real impedance of 8ohm. For the input voltage of 2.83VRMS, the loudspeaker will deliver 90dB SPL. Finally, (3) for the purpose of evaluating SPL levels for different combination of drivers, we will keep the amplifier output constant (no change in volume level).

Single driver.

This is our reference case. The amplifier will deliver 1.0W electrical power to the speaker (U*U/R = 2.83*2.83/8 = 1 W) and the loudspeaker will now generate 90dB SPL. The diffraction effect will add +6dB in the upper end of the operating frequency range and the SPL gains curve will look as on Figure 4.13.

Two Drivers Connected in-series.

Being the ideal voltage source, the amplifier will also cope well with the 8ohm load impedance being now presented to it (two 4ohm loudspeakers connected in-series). Each driver will now receive only half of the 10VRMS voltage generated by the amplifier. With this in mind, each speaker will receive only 6.25W of electrical power (U*U/R = 5*5/4 = 6.25 W) Each driver will now generate only 94dB SPL, so that total SPL of the system is now 97dB. The diffraction effect will add +6dB in the upper end of the operating frequency range and the mutual radiation impedance effect will add +3dB in the lower end of the operating frequency range. The electrical power delivered to the system is now only a quarter (12.5W) of the single driver configuration.

Two drivers connected in-parallel.

Being the ideal voltage source, the amplifier will cope well with the 4ohm load impedance being now presented to it (two 4ohm loudspeakers connected in-parallel). Each driver will generate 90dB SPL, so that total SPL of the system is now 93dB. The diffraction effect will add on the top of it +6dB in the upper end of the operating frequency range and the mutual radiation impedance effect will add +3dB in the lower end of the operating frequency range. The final SPL gains curve is shown on Figure 4.17. It is worth noticing, that electrical power delivered to the system is now twice (2.0W) of the single driver configuration.

Four drivers connected in-series and in-parallel.

This type of configuration results in the system input impedance equal to that of single driver (8ohm). Therefore the electrical power delivered to the system is now 1.0W. The SPL gains curve can now be constructed from two SPL levels representing drivers connected in-series (the whole curve will raise by +3dB) and again, the mutual radiation impedance effect, which will add +3dB in the lower end of the operating frequency range. . The final SPL gains curve is shown on Figure 4.18. In the lower end of the frequency range, the SPL level is +6dB over the single driver configuration and in the high end the levels are identical (also +6dB). Each driver receives only 0.25W of electrical power, which is 1/4 of the single driver configuration. However, combination of all three factors mentioned before, produces fairly bumpy +6dB SPL gain from the "quad box".

Operation of the Diffraction Distortion Dialogue Box

The "Enclosure Diffraction" dialogue box can be opened from the main menu, under the "Calculators" option. The Calculator plots **enclosure diffraction component that will be added to the SPL over and above anechoic chamber SPL (or free-field) measurements**. The edges of your enclosure simply act as additional sources of sound, increasing total, on-axis SPL. The number of drivers mounted on the front baffle is selected from the "Enclosure Design" screen list box and the following choices are available: (1) Single Driver, (2) 2 Drivers Parallel, (3) 2 Drivers Series and (4) 4 Drivers Parallel/Series. Activated "Enclosure Diffraction" box is shown on above. There are several parameters editable by the user:

1. Microphone-Speaker Distance. The distance can be adjusted between 1mm and 2000mm (2meters) using horizontal sliding bar.
2. Dimensions of the front baffle. The x-coordinate and y-coordinate must be entered in centimeters.
3. Location of all four drivers. This parameter is entered as the location of the center of each driver, eg: X1,Y1 is the location of the center of the first driver and so on.
4. Check-box to include the Speaker-Microphone distance and driver positioning in diffraction modeling of multiple drivers.

The five buttons activate the following functions:

1. **Clear** - clears the plotting area.
2. **Copy** - enables copying the graphics of the dialogue box to the Windows™ clipboard for later inclusion in the project documentation. The "Copy" function was described earlier in the manual.
3. **Cancel** - closes the dialogue box.
4. **Print** - prints the screen for immediate hard copy.
5. **Plot** - calculates diffraction, low frequency coupling and plots the results.

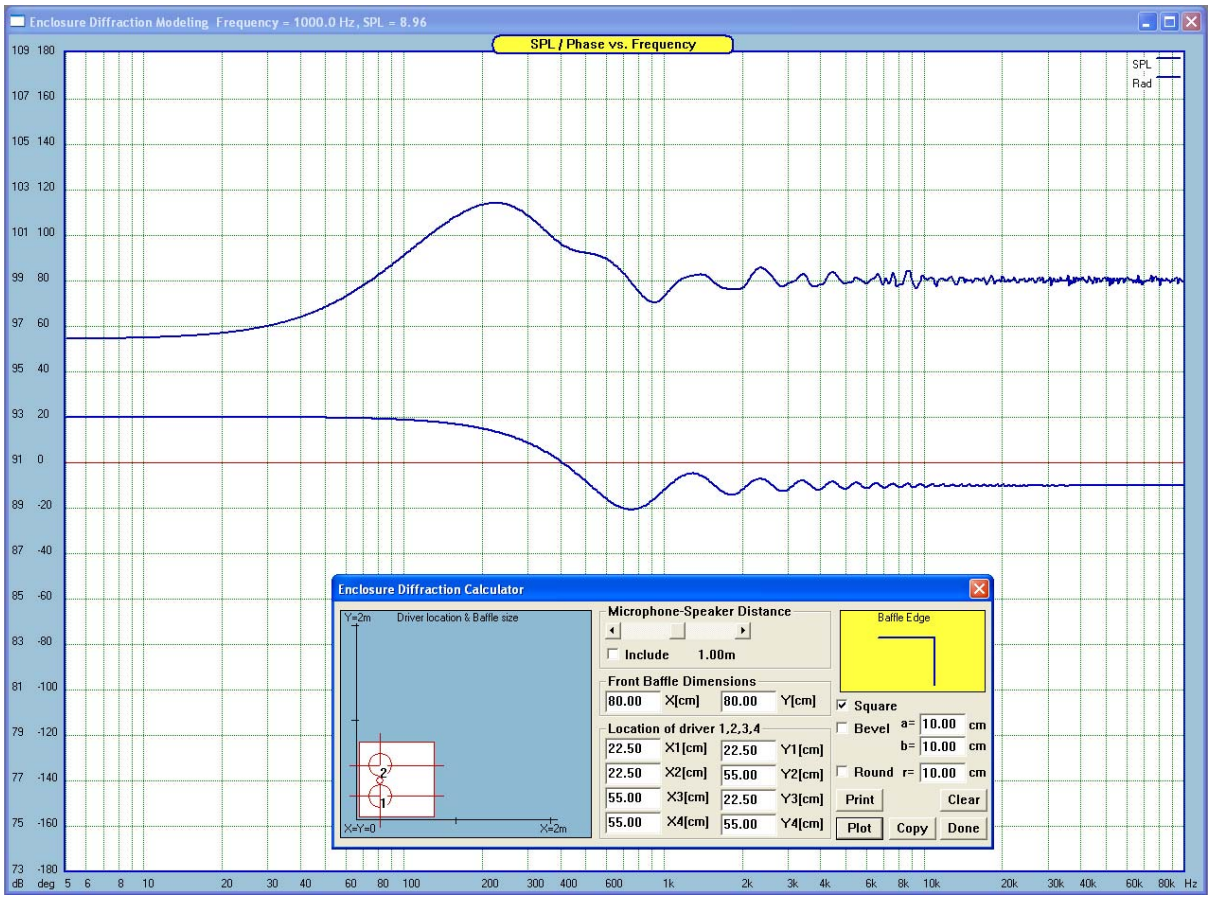


Fig 4.17 Two drivers connected in-parallel

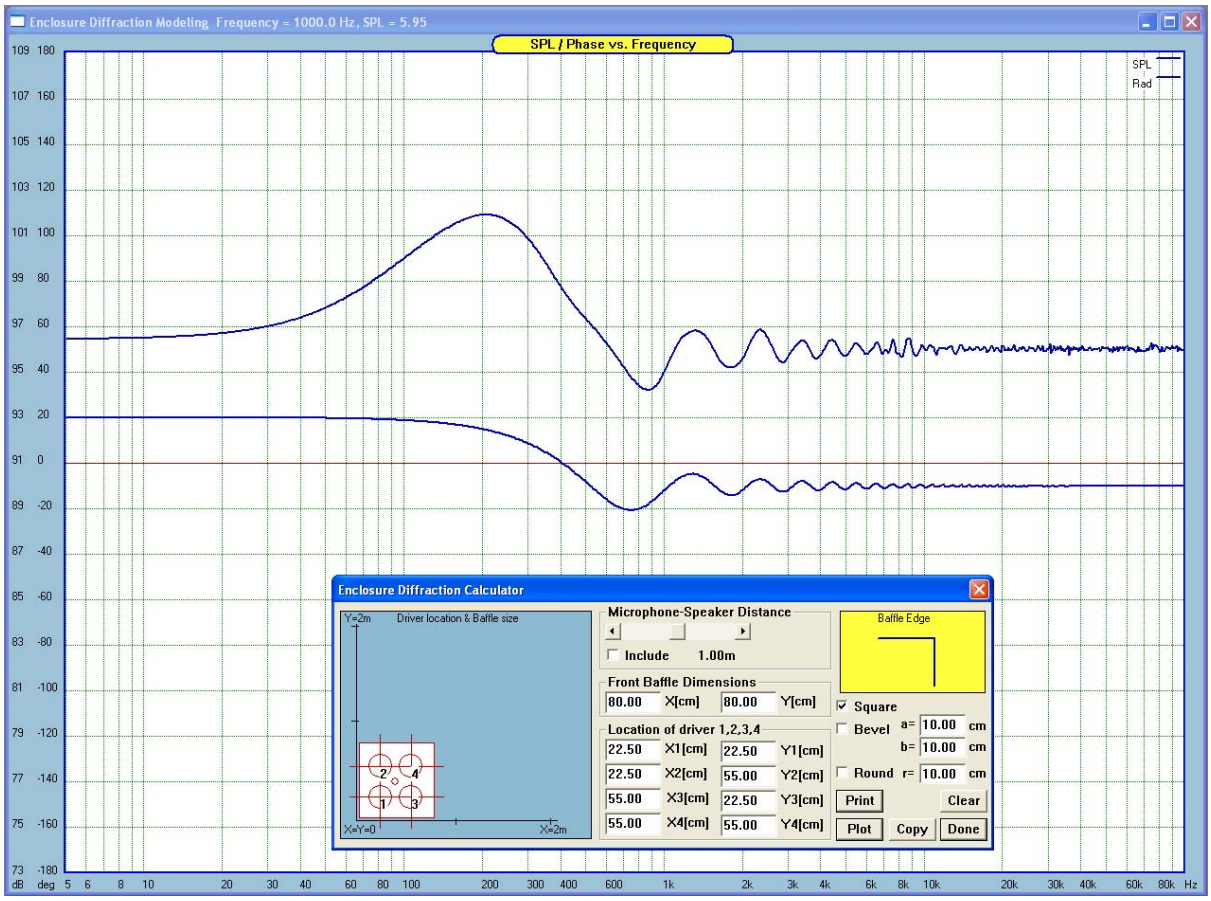


Fig 4.18 Four drivers connected in-series and in-parallel

Passive Radiator Systems

Passive radiator (drone cone) systems eliminate vent noises and are more practical for smaller enclosure, for which the length of the vent would be prohibitive. They also exhibit steeper low-end slope response and a notch in the frequency response. The notch is shown in Fig 4.19. Here, for the same diameter of the driver and radiator, the driver's cone excursion and the radiator cone excursion are equal at 15Hz (PR tuning frequency). However, the phase difference between those two (pink curve #3) is 180deg, leading to sound cancellation – hence the notch.

Tuning The PR

A simple calculator is provided for calculating mechanical mass necessary for tuning your passive radiator to the required "box frequency" Fb. The Fb is understood to be the frequency at which the cone excursion is minimum – just like it happens for the vented box. In fact, Fb is the frequency at which box compliance resonates with PR mass.

It is observable, that dynamic mass of the passive radiator Md, is always greater than its mechanical mass alone. There are three factors that affect the difference: (1) mechanical mass of the PR, (2) air load on both sides of the PR and (3) additional mass needed to nullify the compliance at the given frequency. Fig 4.20 shows an example of passive radiator (Vap=630lt, Fp=15Hz, Diameter=30cm) which tunes the system to Fb=32.58Hz. The Md required for this to happen is Md=128grams. Approximate mass provided by the calculator is Md=130.8grams.

Fig 4.19 Passive radiator (PR) system.

The Calculator also provides some insight into how this mass was apportioned:

1. Mechanical mass, Mm = 89.1 grams, Air Load, Ma = 21.8 grams
2. Additional mass Mc, to cancel PR compliance at Fb is Mc = 27.1 grams
3. Total moving mass Md = Mm + Ma + Mc = 130.8 grams and Mm + Mc = 116.2 grams

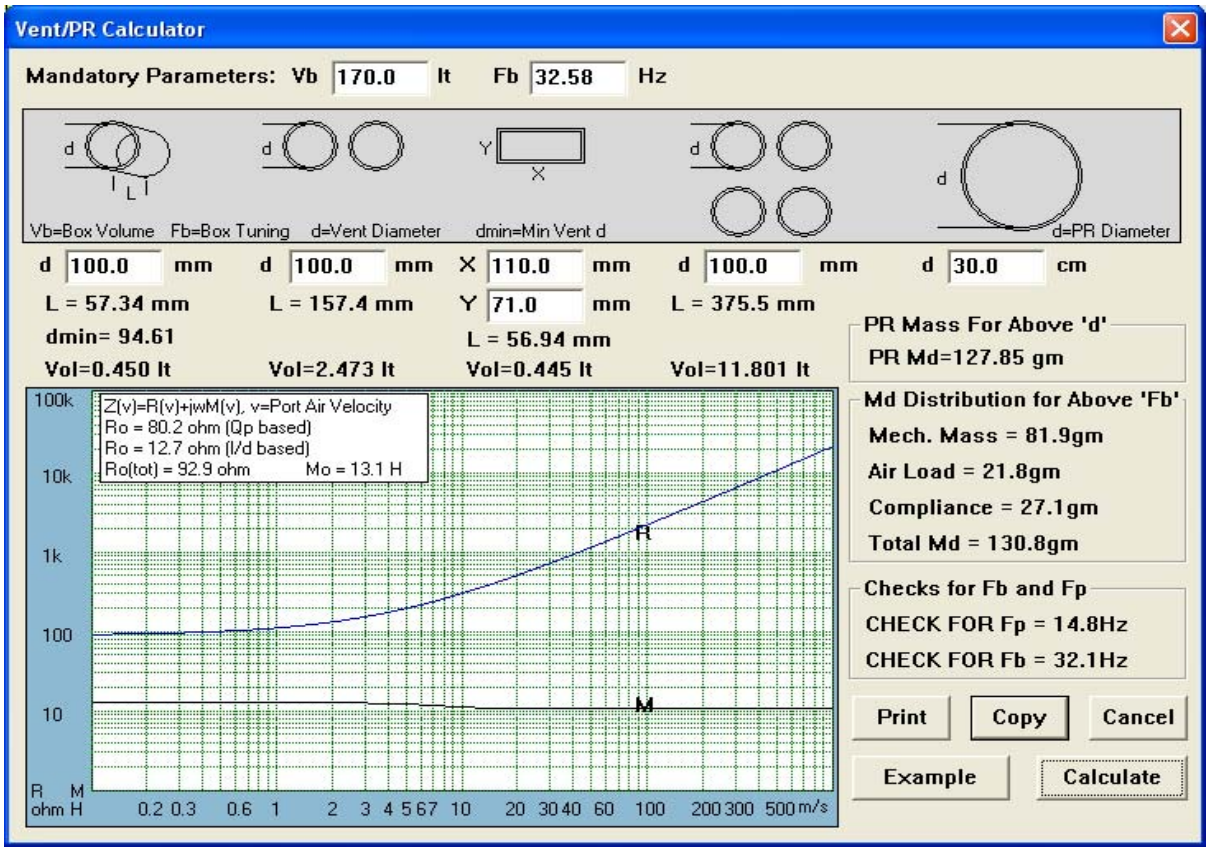


Fig 4.20 Tuning PR

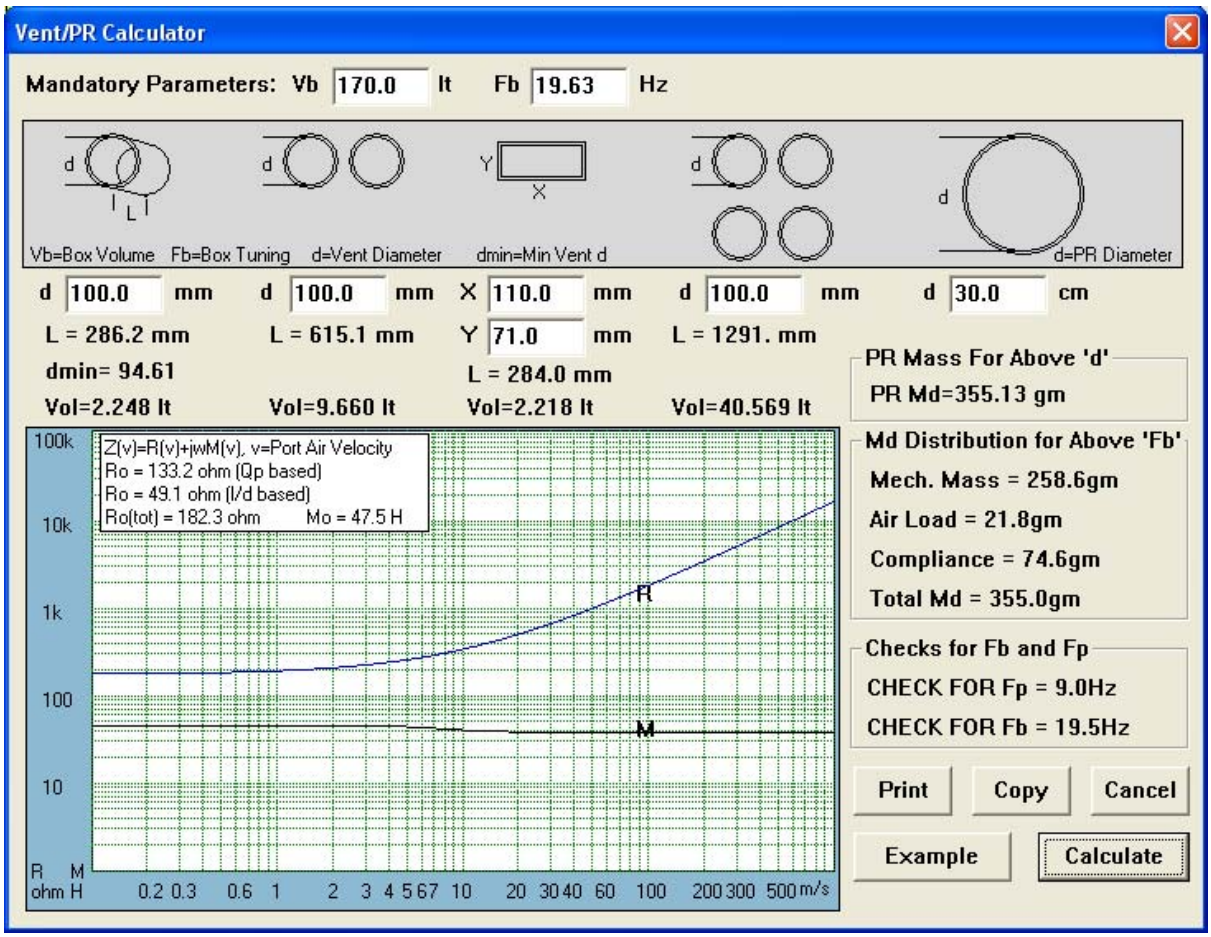


Fig 4.21 Re-Tuning the PR - continued

In order to re-tune this PR to **Fb = 19.63Hz**, the natural (or free air) resonant frequency of the PR would have to be reduced to 9Hz – see Fig 4.21. This can be accomplished by adding extra mass to the PR, so that total Md of the "new" PR is Md = 355.13grams. Mechanical mass, Mm = 258.6 grams

1 Air Load, Ma = 21.8 grams
2 Additional mass Mc, to cancel PR compliance at Fb is Mc = 74.6 grams
3 Total moving mass Md = Mm + Ma + Mc = 355.0 grams and Mm + Mc = 333.2 grams.

This represents an increase of 217 grams in required mass.

Transmission Line (TL) Enclosure

Transmission Line systems incorporate at least one acoustical wave tube attached to the front or rear of the loudspeaker driver. The length of the tube needs to be comparable with the wavelength of the lowest frequency the system is intended to reproduce. Typically, but not necessarily, the quarter-wave resonance of the tube would be selected close the driver's Fs. The line is stuffed with an absorbent material to eliminate (or significantly reduce) mid-frequency ripple introduced by the line's resonances. Parameters that determine the properties of the TL tube are:

1. **Length in meters** (m),
2. **Line stuffing** (absorbent density) in kg per cubic meter (kg/m^3) and
3. **Taper coefficient** – TR. The TR value is defined as the ratio of terminus (mouth) area to the driver area. Therefore if the mouth is larger than driver, the TR is > 1.0.

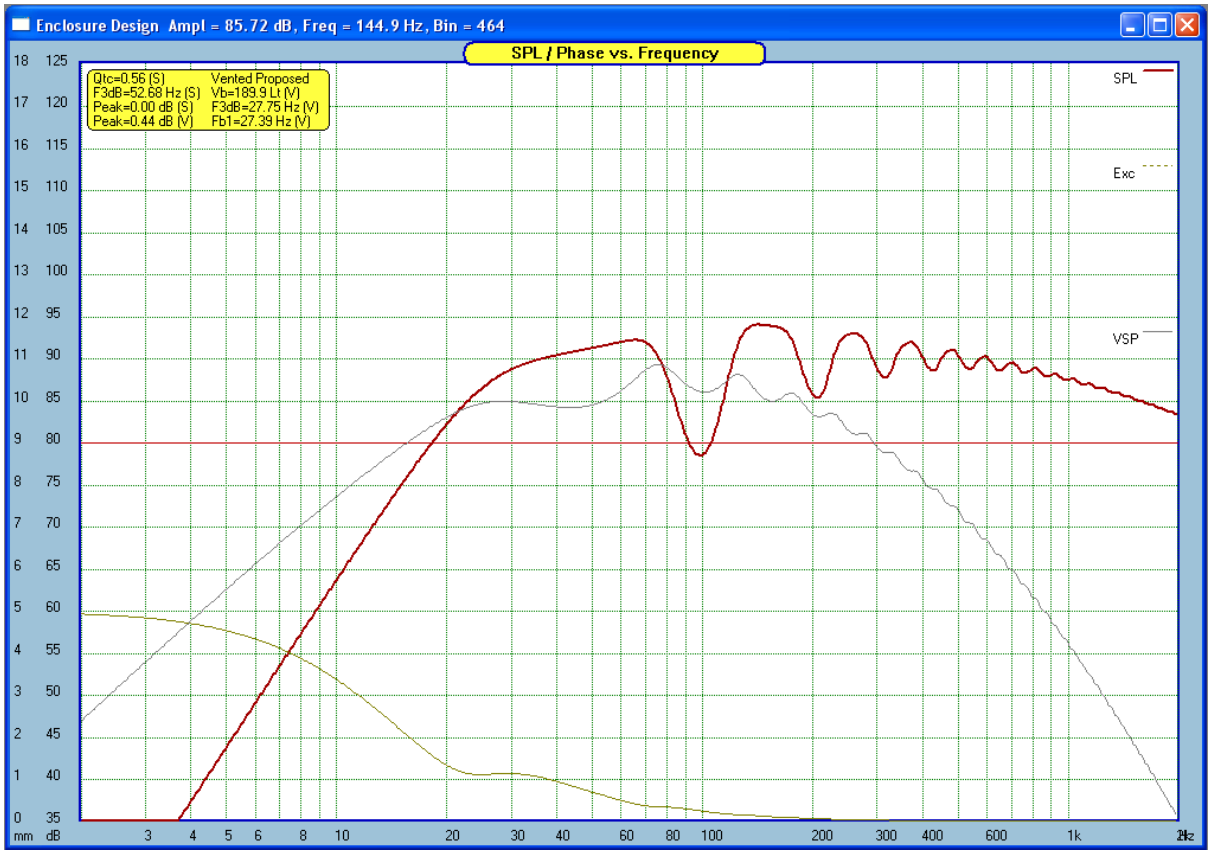


Fig 4.22a Transmission Line basic characteristics

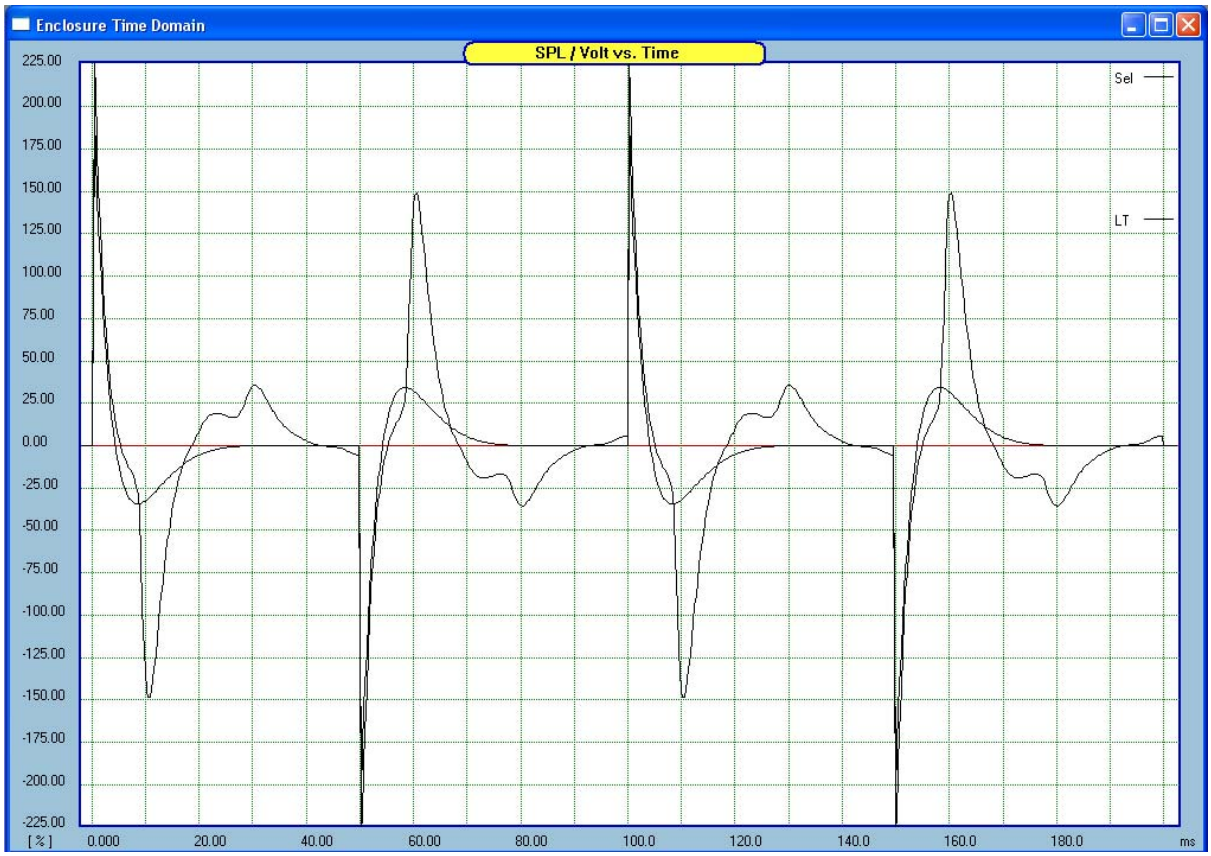


Fig 4.22b. TL and Sealed box system transient response

The stuffing densities can vary from 0.0 to 90 kg/m^3 and line length can be as long as 5 meters, but greater than zero. An example of the TL system responses is shown on Fig 4.22a. The system SPL is marked as curve “0” and labeled “TL Amp”. Terminus SPL is marked as curve “1” and labeled “Air Amp” and finally, cone excursion is marked as curve 2 and labeled “TL-Exc”. Fig 4.22b shows the TL system’s transient response for 5Hz “square wave” signal. It is observable, that the output pulse from the terminus arrives delayed in time in respect to the radiation from the cone.

The delay is proportional to the length of the line and the speed of sound in the stuffed tube. The terminus pulse is more rounded than its direct counterpart because of the high-frequency attenuation of the TL tube. Plotting of the TL system functions is accomplished the same way as for all others enclosures. The “TL System” button on the “Enclosure Design” screen activates a floating menu with an array of options to choose from. The TL systems attract some attention as they offer the highest SPL at really low frequencies (10-20Hz) in comparison to sealed or vented systems – see Fig 4.23. In addition, the TL systems run at much lower air speeds coming from the terminus. Therefore they do not suffer from the vent-noise problems related to the turbulent air flow in vented enclosures. When experimenting with the transmission line systems please note, that absorbent density will need to be changed for different line lengths. Longer tubes will need less dense absorbent to avoid overstuffing of the line. Also, increasing the TR coefficient helps boost low end output of the system. Finally, the above discussion assumes that the driver and terminus are located close to each other (line is folded as it would in typical design). Otherwise, additional delay would result between the driver and terminus outputs.

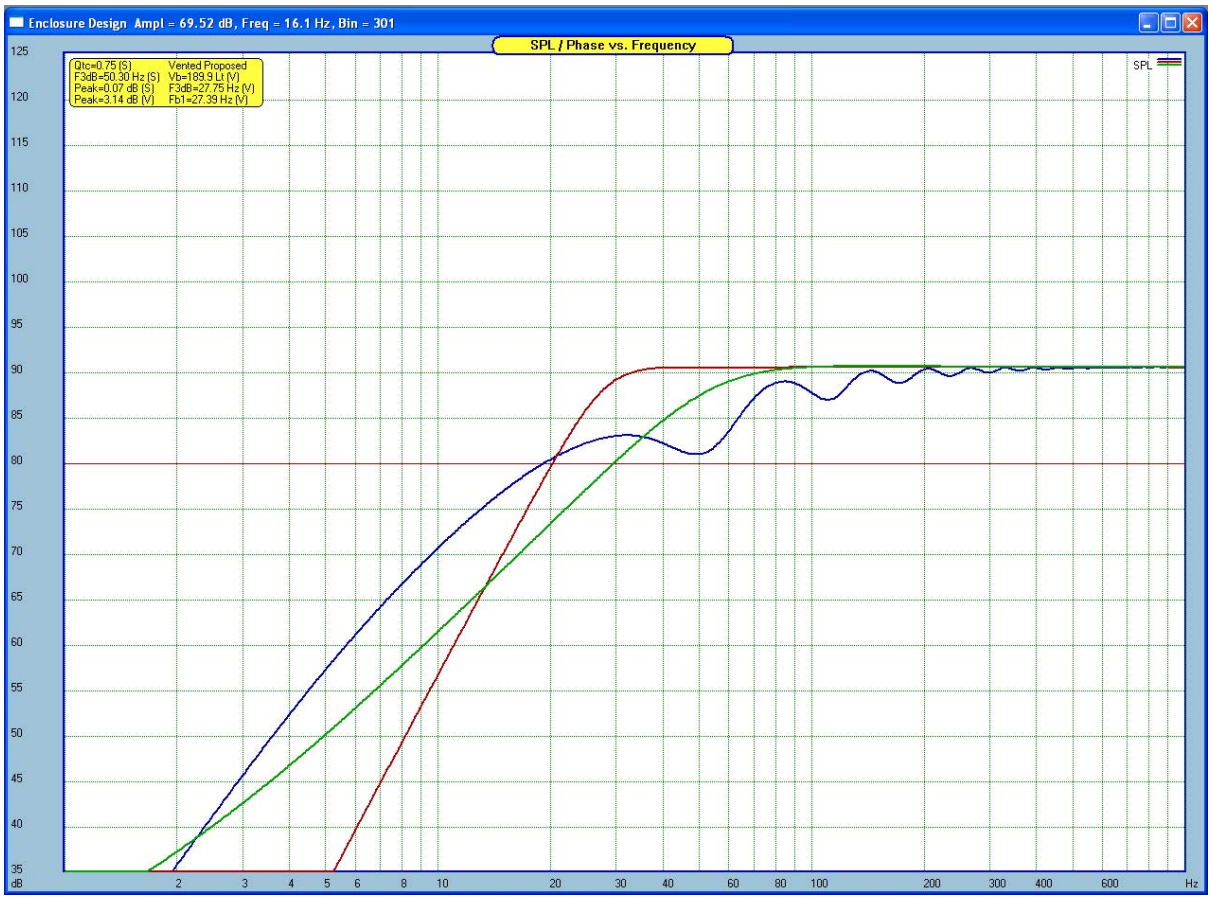


Figure 4.23 Sealed (green), Vented (brown) and TL (blue) system SPL

Horn Loudspeakers

The horn element may be viewed as an acoustic impedance transformer. Here is how it works. When a diaphragm vibrates, pressure waves are created in front of it. This is the sound we hear. Coupling the motion of the paper to the air is not an easy thing to do due to the very **different densities of the vibrating diaphragm and air**. This is can be viewed as an impedance mismatch. Impedance mismatch always results in reduction (and back reflection) of the sound trying to pass through.

Sound travels better in high density materials than low density materials, and in a speaker system, the diaphragm is the high density (high impedance) medium and air the low density (low impedance) medium. The horn creates the solid-air impedance transformation effect by acting as an intermediate transition medium. It simply creates a higher acoustic impedance for the transducer to work into thus allowing more power to be transferred to the air. The impedance mismatch is significantly reduced. The horn does its job by employing a tube whose cross-section increases exponentially/hyperbolically . The narrow end (where the transducer is mounted) is called the throat and the wide end is called the mouth. When the diaphragm moves near the throat, we have a high pressure with a small amplitude in a small area. As the pressure wave moves towards the mouth, the pressure decreases along the way and the amplitude increases. We have a natural efficient amplification effect taking place here. The length of the horn and the size of the mouth are selected for the desired low-end cut-off frequency, below which the horn output drops rapidly.

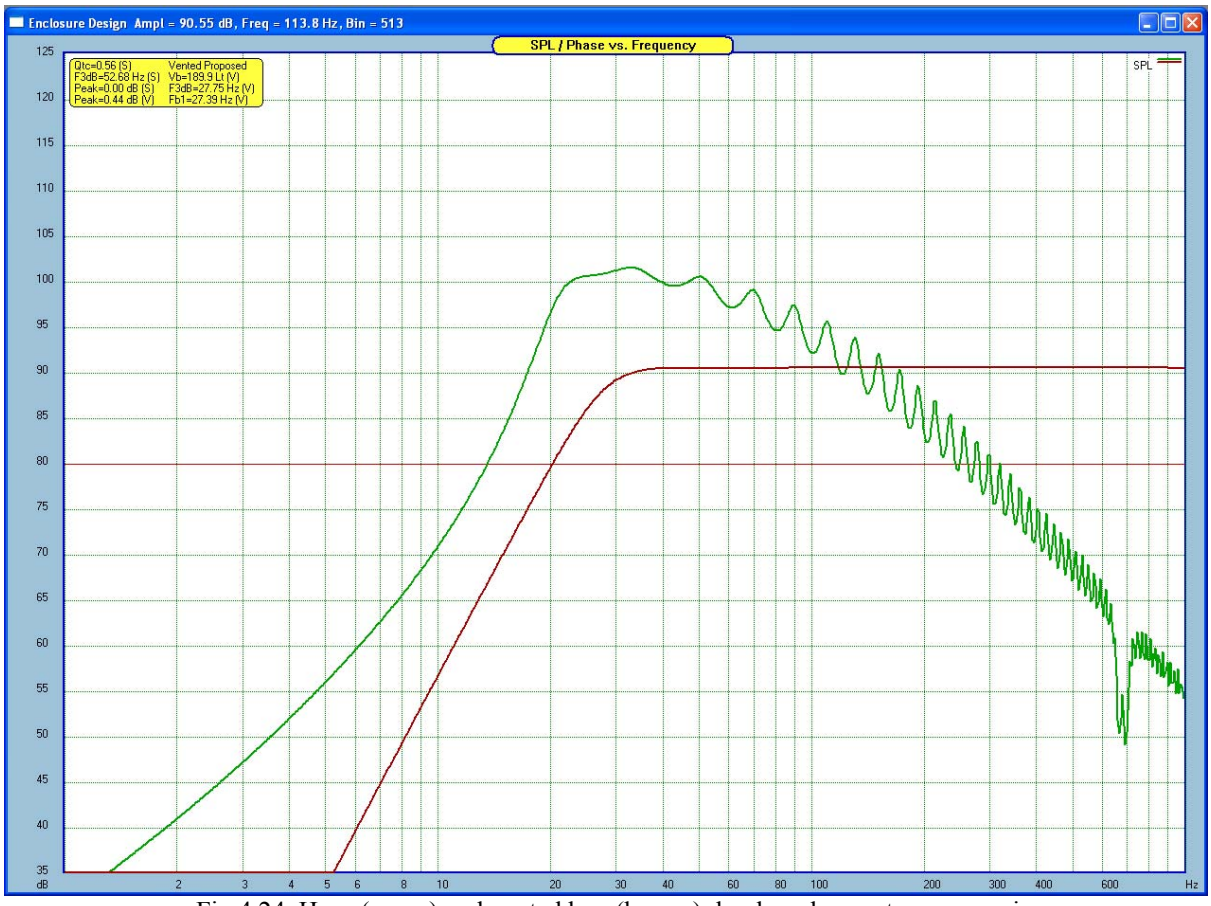


Fig 4.24. Horn (green) and vented box (brown) loudspeaker system comparison

The horn loudspeaker is essentially an acoustic transformer matching the impedance of the air to that of the piston (driver). This is particularly evident at low frequencies – see Figure 4.24. Here, we have a driver with 91dB SPL efficiency generating 97dB SPL with only 1W of input power (!) at 20Hz. The horn is large: 8m long and mouth area is 17 square meters. The brown depicts SPL of a 170Lt vented box for comparison. Model of the horn element implemented in SoundEasy consists of 40 short cylindrical sections of transmission line with varying (increasing) radius. If S_T is the throat area and S_L is the mouth area, their relationship along x-axis for each type of horn, is described by the following equations:

Conical Horn $$S_L = S_T * (1+x) * (1+x)$$

Exponential Horn $$S_L = S_T * e^{(x*FC)}$$

Hyperbolic Horn $$S_L = S_T * [(\cosh(x) + FC * \sinh(x)]^2$$

where x, is the distance from the throat and FC, is the Flare Constant. Please note, that horn and TL systems share the same data fields on the "Enclosure Design" screen. Data necessary for modelling horn enclosure includes:

1. Horn Length in meters.
2. Horn Stuffing in kg per cubic meters. Horn stuffing is rarely required, however, data field called "TL/H Stuffing" is provided for entering stuffing material density. Sometimes, a vary small amount of dumping material helps smoothing resonance peaks of the horn.
3. Flare Constant in 1/meter.
4. Horn St/Sd ratio. This is throat area to driver area ratio.

Horn impedance and input power demand can be plotted only after selecting and editing one of the built-in impedance models. As before, please make sure, that input power level, Pin is also provided.

Voice Coil Inductance

When modeling VC inductance of open radiator type (Sealed, Vented, Passive Radiator and Transmission Line) loudspeakers you will notice, that the characteristics of the modeled system change quite dramatically when the voice coil inductance is changed - this is depicted on Figure 23 below. Voice coil inductance cause the SPL curve to roll-off towards the higher end of the plot.

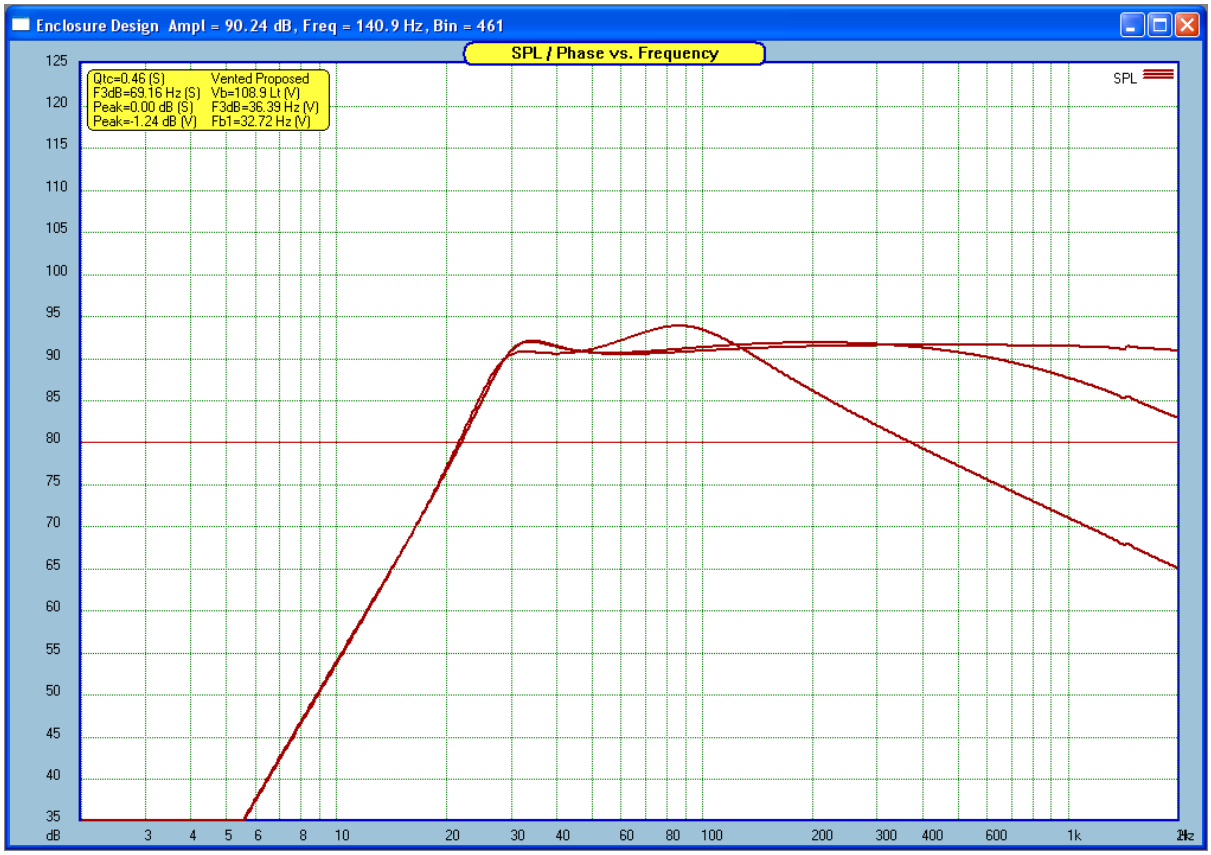


Figure 4.25. SPL comparison foe VC=0.2mH, 1.2mH and VC = 10.0 mH.

U-Dipole Enclosure

Dipolar enclosures are often described by its "directional bass" characteristics. Primary reason for this characteristics is the interaction (cancellation) between the acoustic radiation from the front and the back of the loudspeaker. The two most popular dipole systems are U-Dipole and H-Dipole. The U-Dipole system is simply "flat baffle" with the sides folded back, so that an open-back cabinet is created.

The most obvious feature of such arrangement is the length of the side-wall, as it determines the frequency**, Fc**, at which front-to-back cancellation occurs. Moving further towards lower frequencies, below the Fc, you will find system frequency resonance**, Fs.** Typically, system SPl will roll-off with –6dB/oct between Fs and Fc.

The relatively slow roll-off could be equalized by a simple, first-order circuit. Below the Fs, the system SPL response falls with –18dB/oct and can not be equalized effectively.

In the example shown below, the U-Dipole has 0.6m long side wall, therefore:

Fc = (334m/s) / (1.2m) = 278Hz.

Indeed, you can observe a dip in the SPL response at round 270Hz – see Fig 4.26. Polar plot at 20Hz clearly indicates directional characteristics of this enclosure.

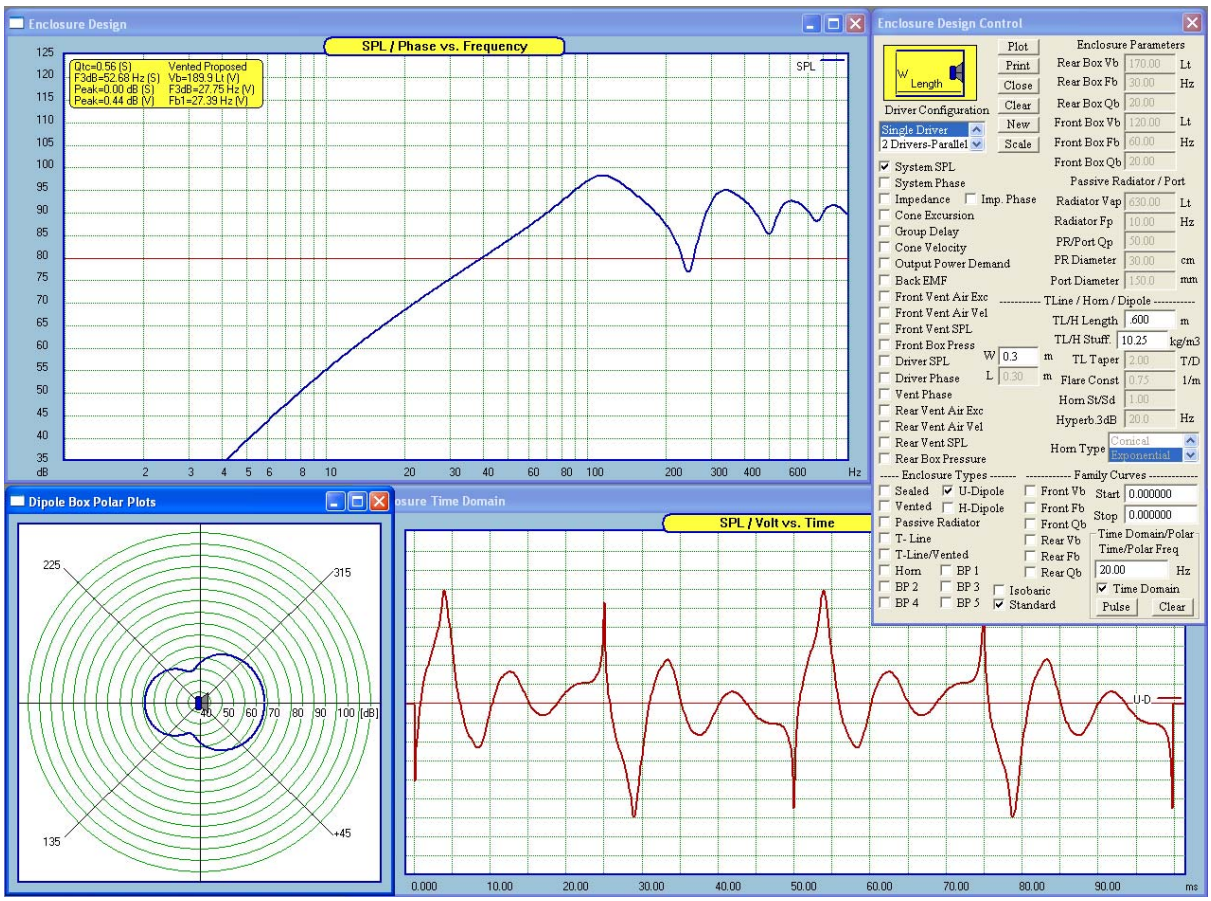


Fig 4.26. U-Dipole SPL, time domain response and polar plot.

Modeling of the U-Dipole involves three parameters: (1) length of the side wall, (2) stuffing and (3) width of the front baffle.

H-Dipole Enclosure

H-Dipole is a close brother of the U-Dipole arrangement. Typical H-Dipole would have a driver mounted in the middle of the "H" frame. In this case, lengths of the walls in fron and the back of the frame would be equal. Should you arrange your system as described above, you may expect that the polar plot of this system will resemble figure "8".

It may be noticeable, that a dipole system requires greater available excursion for a given performance than all other common system types. It is unfortunate, but a substantial amount of driver excursion is used in front-to-back cancellation rather then supplying useful output. This is perhaps where the most of the controversy associated with the dipole system is concentrated. Directivity of the dipole system results from a process of mutual cancellation of the sound pressure from the two sides of the diaphgram. This bass-directivity, is often argued, to prevent the dipole system from axciting all room modes, and therefore add less "room coloration" to the overall sound spectrum.

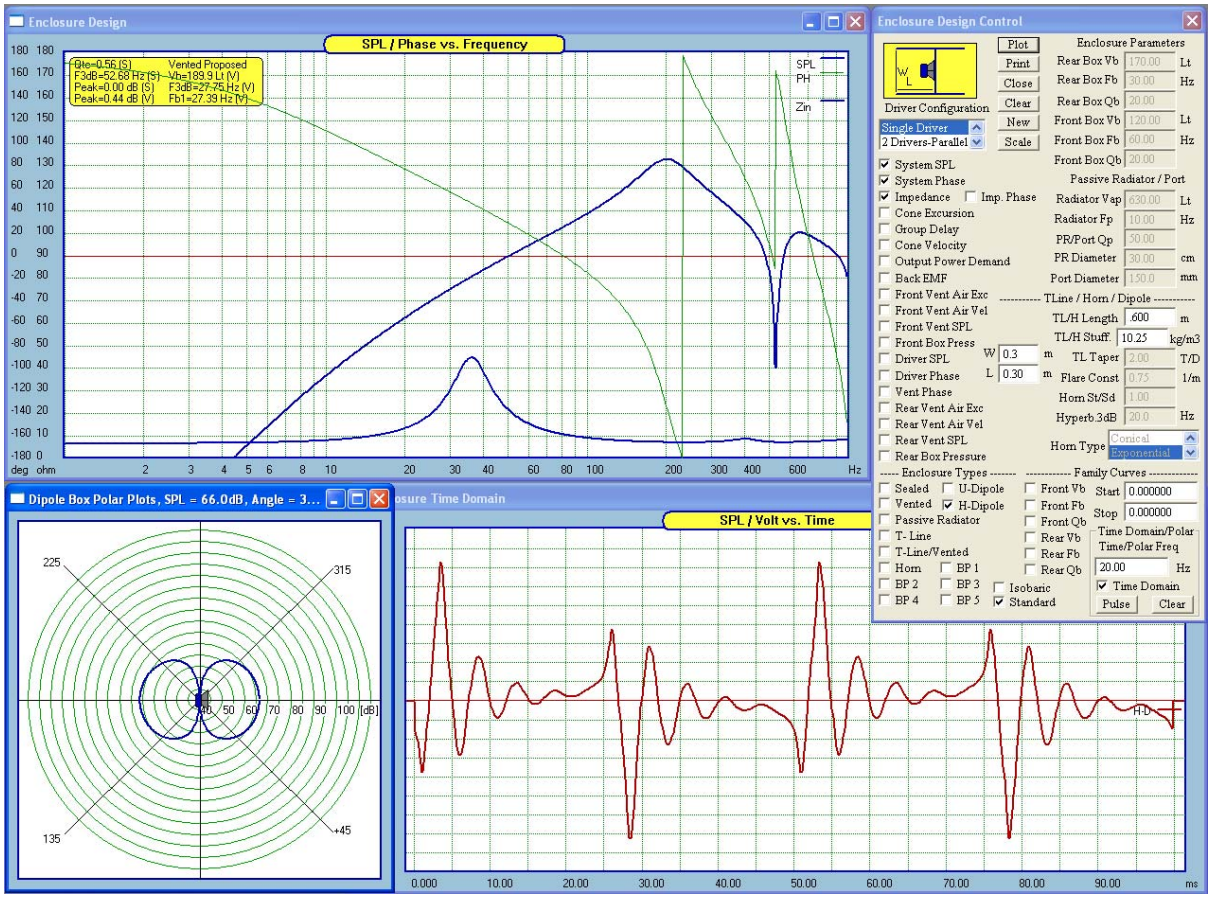


Fig 4.27. H-Dipole – note very characteristic polar plot.

Modeling of the H-Dipole involves four parameters: (1) total length of the frame wall, (2) stuffing, (3) width of the front baffle, **W**, and (4) internal length of the wall **- L**.

Getting more out of Dipole plots

The H-Dipole model implemented in the program, allows for the internal section of the H-frame to be be moved backward-and-forward. This particular ability converts the H-Dipole to U-Dipole at the extremes of the internal section locations. Please remember, that all physical parameters of the enclosures must be greater than zero, therefore, for the frame length of 0.60meter, you should not move the internal H-section more than 0.599meters in one direction or 0.001meters in the other.

Taking advantage of the above possibility, you can plot U-Dipole front SPL and rear SPL at the same time. Simply "check " both types of dipoles for SPL plotting and select L=0.001meter for the H-frame dipole. This will have the effect of moving the center section right to the back of the frame – making it effectively a U-Dipole.

Resulting SPL plots are very useful, as they show clearly the difference in front and rear SPL levels., which is the design objective. This feature is perhaps more beneficial to have than just the polar plots. It allows you to see the different SPLs over the whole frequency range of interest – see Fig 4.28

Please note, that polar plots frequency is entered from the same data field as the Time Domain test frequency – left bottom corner of the control box.

Finally, The polar plot screen is re-sizable like all other plotting screens. You can close this screen and when re-opened, it will display polar plots correctly drawn.

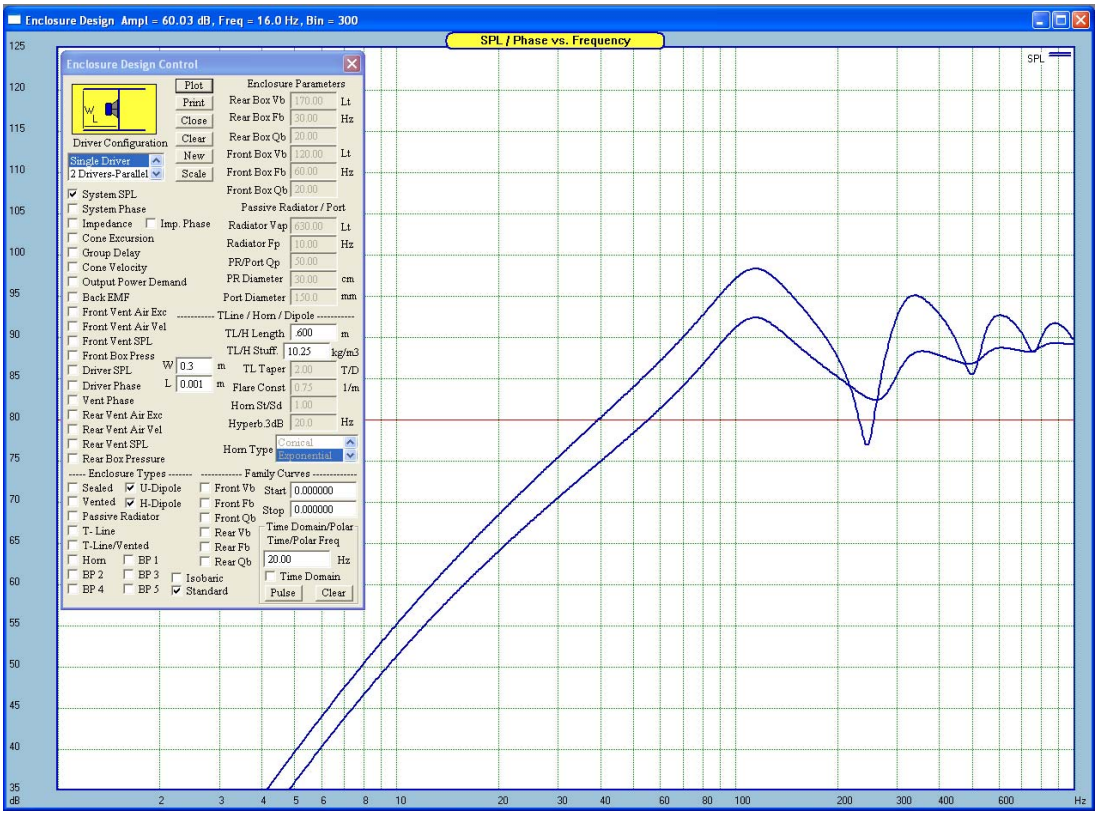


Fig 4.28. Converting H-Dipole into U-Dipole.

Polar plots vertical offset for H / U Frame dipoles

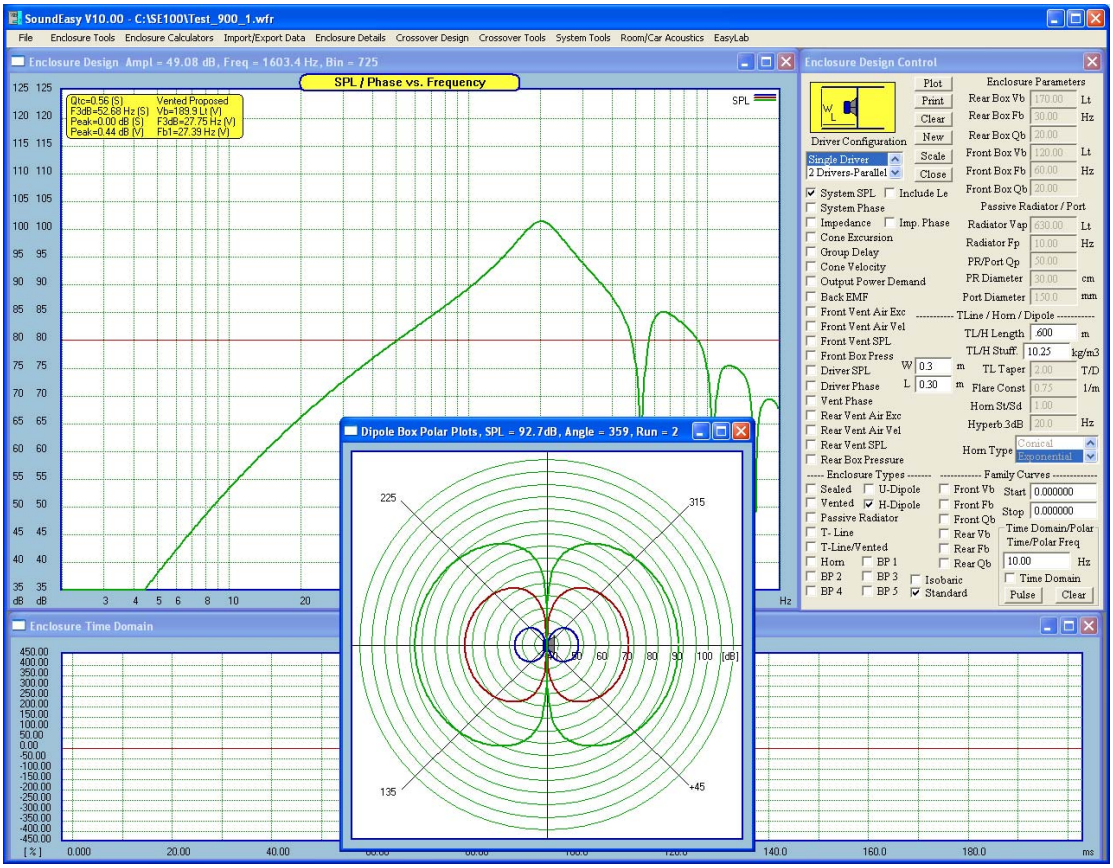


Fig 4.29. H-Frame dipole loudspeaker plotted to three offsets in 20dB steps

Dipole enclosures exhibit interesting polar characteristics. However, the details and more defined shapes of the polar plots are often more clearly visible at the lowest frequency range, where the actual SPL is quite low already.

In order to improve presentation of polar patterns at such low SPL levels, you can add several decibels to the SPL polar plots. Curves modified this way are much better presented. Figure 4.29 above shows the same SPL polar plot of a H-Frame dipole loudspeaker plotted to three offsets in 20dB steps.

The offsets are controlled from “Enclosure Calculators” ->”Scale Resolution+Pin”->”Polar Plot Offsets” field.

Added Le ON/OFF switch in plots

It is sometimes desirable to disregard the effect of driver’s voice coil inductance when plotting it’s transfer functions on the screen. By doing so, you may be able to better visualize the influence of varying enclosure parameters on the transfer functions. You can easily switch the voice coil semi-inductance, Le, by using the check box provided on the Enclosure Designer control box. Example of plots with / without Le is shown below.

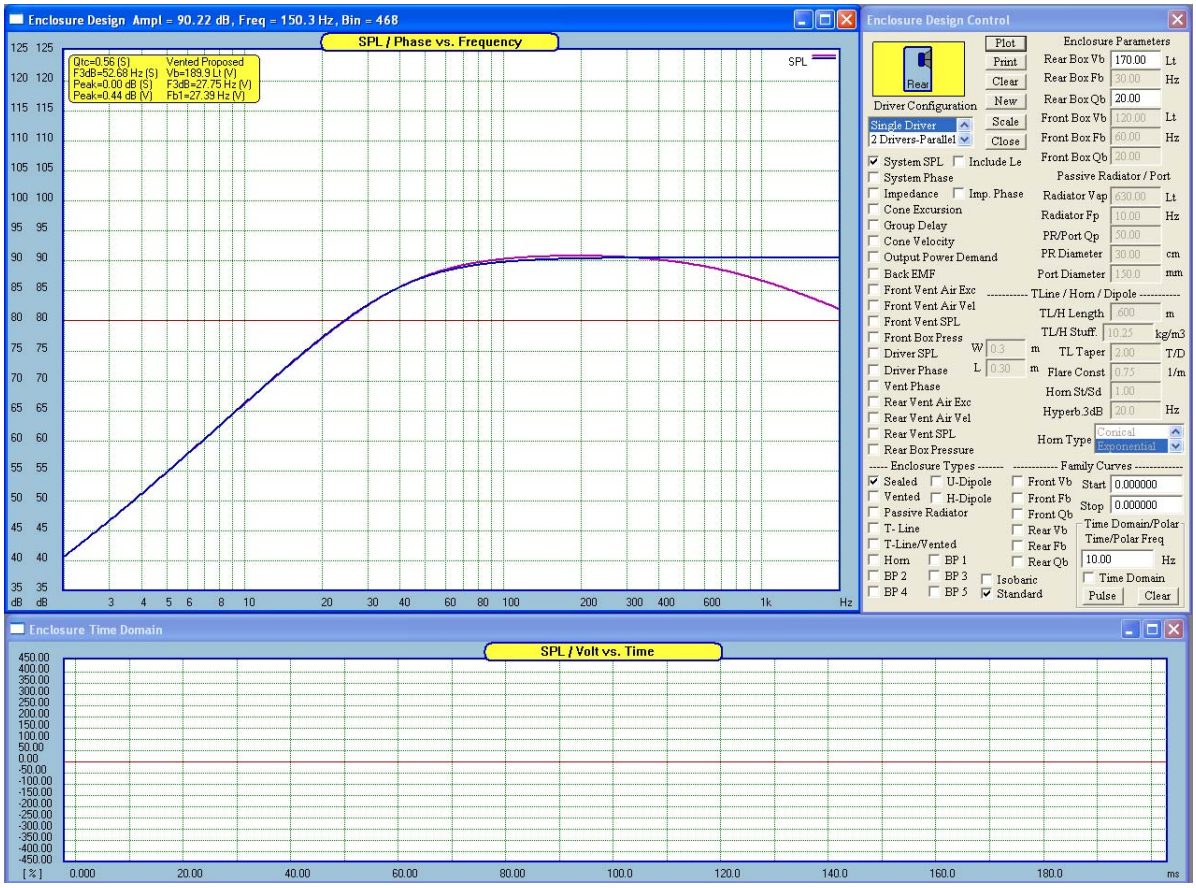


Fig 4.30 Example of plots with / without Le.

Large-Signal Analysis

Thermal Analysis

When a high level signal (input power) is applied to the loudspeaker for a prolonged period of time, the temperature of the voice coil will raise significantly. This is mainly due to the low efficiency of the loudspeaker itself, which in most cases falls between 1-5% for a typical HiFi driver. The remaining 95% of the energy provided by the amplifier is wasted as a heat in the voice coil. The raise in the temperature of the voice coil has a detrimental effect on the driver's performance. The final temperature of the voice coil depends on the input power of the applied signal over time and driver's mechanical design. Various mechanical solutions have been proposed for the motor system of the driver to minimize the voice coil temperature. All these solutions have common elements, which for the purpose of analysis, are translated into a thermal model characterized by heat flow and appropriate thermal resistances.

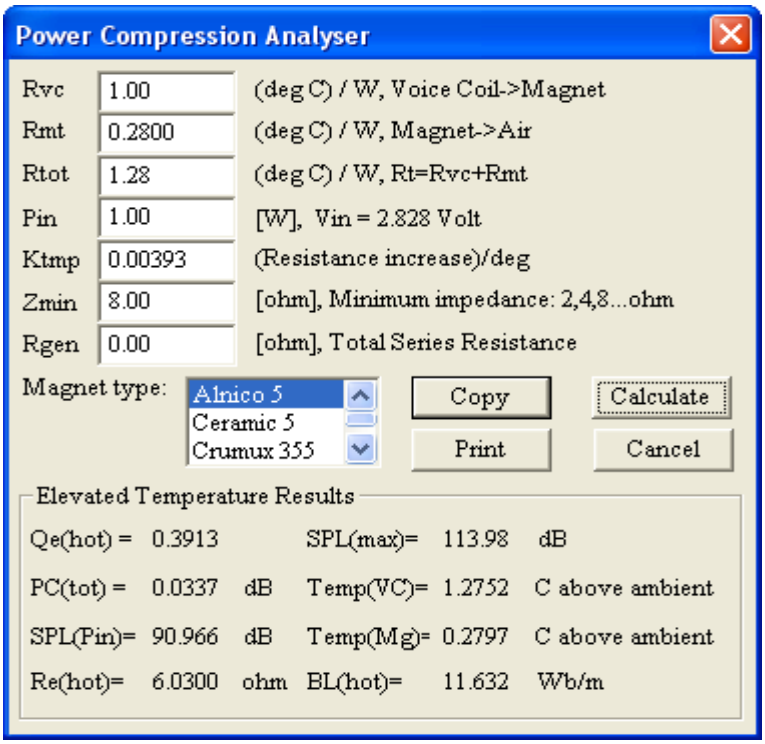


Fig 4.31 Thermal Compression analyser.

The mechanical structure closest to the voice coil is the magnet system and the thermal interaction between these two determines **Rvc**, the voice coil-to-magnet thermal resistance. Once the magnet heats up, it will radiate the heat directly into the surrounding air. The losses of this process are modeled by **Rmt**, the magnet-to-air thermal resistance. As one can see, the magnet, being larger structure and directly exposed to the air, has lower thermal resistance. The structure of the magnet will lose some of its flux due to the raise of its temperature. This process is reversible, but will cause reduction in SPL level. Another factor reducing the SPL is the raise of the voice coil resistance. The resistance increase per degree C is given by the **Ktmp** factor. For copper and aluminum wires **Ktmp**=0.00393 and for a 100.0 deg C temperature raise, there will be almost 40% increase in voice coil resistance (Re), (**Rtot = Rvc+Rmt**). Increased Re will cause less current to be drawn from the amplifier and also increase in electrical Q-factor (Qe).

All the data entered on the "Power Compression Analyzer" are real, positive numbers. Magnet type determines the rate of the flux loss. Generator's internal resistance should be entered into the **"Rgen"** field. Input power into the modeled loudspeaker is edited from the **"Pin"** data field. "**Zmin**" is the nominal speaker impedance and should be entered as one of the standard impedances such as: 2 / 4 / 8 / 16 ohms. Once the frequency response curves are plotted for 1, 10, 100 and 1000 watts of the input power, it can be noticed, that the curves do not raise by 10 dB, but less and the shape of the plots is changing due to modified Qe values (see Fig 4.31). All the data entered on the "Power Compression Analyzer" are real, positive numbers. Magnet type determines the rate of the flux loss and was given in Chapter 2.

If your driver uses totally different magnet material, select the flux loss, which is closest to your drive's magnet material (using the "Editor" screen) or contact Bodzio Software for ***free*** customized version of the program. The maximum SPL level is calculated for the voice coil reaching 200 deg C. This is the maximum temperature level for the cone assembly before the glue starts to deteriorate. The "Power Compression Analyzer" is shown on Fig 4.31. A hard copy of the "Power Compression Analyzer" calculations can be obtained by pressing the **"Print"** button from the Analyzer dialog box.

How to determine Thermal Resistances ?.

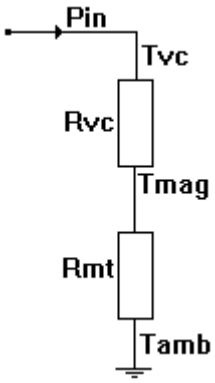
The thermal resistance model used by SoundEasy takes into account three major elements: (1) loss of electro-motor force due to heating up of the magnet assembly, (2) raise of the Qe factor and (3) raise of the Re (DC resistance of the voice coil) due to rising temperature of the voice coil. Combination of these factors is known as "Thermal Compression". The first factor will be reflected as reduction in SPL. The second factor will cause the Qt to raise and then changes in the amplitude response of the driver, particularly around the box tuning frequency will be observed.

Fig 4.32 SPL and Cone Excursion at various Pin: 1, 10,100 and 1000W – BL(x) and Cas(x) NOT included

The third factor will cause less current being drawn from the amplifier (assuming constant voltage source) and as a consequence, further reduction in SPL. Several very useful thermal models of the loudspeaker have been proposed and Bodzio Software acknowledges here the works of C.A. Henricksen and D.J. Button. The thermal resistance model consists of two elements:

1. Rvc - Voice coil to magnet thermal resistance. This element lumps together several associated thermal resistances.
2. Rmt - Magnet to air thermal resistance.

The simplification is quite deliberate and because of this, both components can be measured quite simply. The model thus looks as follows:



Where:
Tvc - Temperature of the voice coil.
Tmag - Temperature of the magnet assembly.
Tamb - Temperature of the surrounding air, usually 20 Deg C.
Pin - Input power to the loudspeaker applied for min 1 h.

$$R_{VC} = \frac{T_{vc} - T_{mag}}{P_{in}} \qquad R_{mt} = \frac{T_{mag} - T_{amb}}{P_{in}}$$

Example: Tvc = 100 deg C.
Tmag = 35 deg C.
Tamb = 20 deg C.
Pin = 60 Watt.

$$R_{vc} = \frac{100-35}{60}[\text{deg}/\,watt] = 1.08[\text{deg}/\,watt]$$

$$R_{mt} = \frac{35-20}{60}[\text{deg}/\,watt] = 0.25[\text{deg}/\,watt]$$

$$R_{tot} = R_{vc} + R_{mt} = 1.33[\text{deg}/\,watt]$$

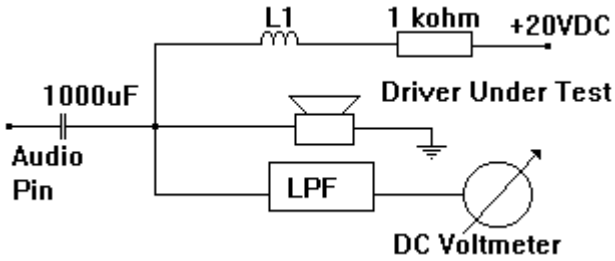
It is clear, that in order to calculate the Rvc and Rmt, the temperature of the voice coil and magnet assembly must be known. There should be no problem with measuring the temperature of the magnet. A good, accurate thermal probe (such as the DVM thermal probe) attached to the magnet, with the help of some thermal compound should suffice. Measuring the temperature of the voice coil is a little trickier. The method proposed here is similar to the work of C.A. Henricksen (JAES. Vol 35, No 10, 1987 October). Essentially, two signals are fed to a loudspeaker: (1) an AC power, to heat up the voice coil and (2) a small DC current, which us used to determine the resistance of the hot voice coil. The DC current will cause a small voltage drop across the loudspeaker's DC resistance (Re). The voltage is sufficiently small, that it will not cause any extra heating or mechanical offsetting the voice coil. When the voice coil is cold, the DC resistance of the loudspeaker is Re (known), and causes a DC voltage Vcold, to appear across the loudspeaker terminals. The AC power is generated by an ordinary audio power amplifier and should be kept at about half of the maximum loudspeaker's rated power. When the AC power is applied to the loudspeaker, the voice coil starts to heat up. After 1-2 hours, of continues operation, the temperature in the system starts to stabilize. The DC resistance of the voice coil has increased and now the DC voltage across the loudspeaker reads Vhot. Temperature of the magnet assembly, as measured by the thermometer, is now Tmag. The AC power is now turned off and the loudspeaker disconnected from the system and replaced by a variable resistor (you may use several smaller resistors of different values). The variable resistor is that set to such value (Rhot), that produces the same voltage (Vhot) on the DC voltmeter. The Rhot is the final DC resistance of the hot voice coil. From the formula:

$$R_{hot} = R_e * (1 + 0.00393 * (T_{vc} - T_{amb}))$$

valid for copper and aluminum voice coils, one may calculate:

$$T_{vc} = \frac{R_{hot} - R_e}{0.00393 * R_e} + T_{amb}$$

All the values: Tvc, Tmag, Tamb=20deg, and Pin are known and Rvc and Rmt can be calculated from the formulas shown before. Block diagram presented below shows typical test arrangement. Audio power is fed to the loudspeaker through a large capacitor (1000 uF), to block the DC current flowing back to the amplifier. The large coil (L1) prevents the audio power from going back to the DC voltage source. The 1kohm resistor converts the 20V DC voltage source into a 20mA current source. The LPF prevents the audio power to reach the DC voltmeter and degrade the accuracy of the readings. The LPF should have at least 100dB attenuation at the audio test frequency (300Hz). You may decide to use a combination of passive and active filtering.



Method for measuring voice-coil temperature of a loudspeaker.

Example:

1. Re = 6 ohm (from driver's data sheet)
2. DC voltage source = 20 VDC, in series with 1kohm resistor produces 20mA current.
3. Voltage Vcold = 6ohm*20mA = 120mV
4. Vhot = 157.7mV, and Rhot = 7.88ohm, as measured using substitution method.
5. Tvc = (7.88 - 6.0)/(6.0*0.00393)+20 = 79.9 deg + 20 deg = 100 deg C.
6. Tmag = 35 deg C, as measured using thermometer.
7. Pin = 60 watt.
8. Rvc = (100-35)/60 = 1.08 [deg/watt]
9. Rmt = (35-20)/60 = 0.25 [deg/watt]

Port Nonlinearities and Compression

Ported enclosures are known to extend the low frequency output of the loudspeaker system by exploiting Helmholtz resonator created by the compliance of the air inside the enclosure and inertance of the air in the port. Acoustic transformer created this way has its own resonant frequency, Fb, at which most (or all, if there was no losses) of the system acoustic output comes from the port. A fairly obvious implication of this is significant velocity associated with the air flow through the port. This in turn, causes all sorts of non-musical noises to be generated by the port and also distortion and acoustic compression. Depending on port geometry, and required low frequency SPL of the system, the issue can be quite significant.

The problem described above belongs to rather complex field of fluid flow. Assuming incompressible flow, some sophisticated FEM programs would be able to model air turbulence and associated vortex shedding in more detail[3], but this is well beyond the scope of my article. Fortunately, existing research results, design material and tests results enable us to formulate an approximate macro view of the problem and look at the acoustic impedance of the port under high air velocity. It needs to be stressed, that the approach taken here is a significant simplification of the physics of the problem. However, the resulting model is quite useful and finds confirmation in practical tests.

Intuitive approach to port non-linearity

We are all familiar with the need for good carpentry skills when building speaker boxes. Accuracy of joints and sealing the box is essential for proper operation of all types of boxes, be it sealed, vented, passive radiators and so on. Sealed box means exactly this – the air inside the box is trapped and sealed from the external world. There is no parasitic or accidental leakage from the box, so that mathematical model developed for the enclosure continues to be accurate. Consequently, the box Qb factor is controlled by the designer and NOT by sloppy workmanship. Vented enclosure also needs to be “air tight”, of course with the exception of the purposefully introduced opening – port. Just as for the sealed box, the cabinet needs to be sealed, so that no accidental air leakage from the box can occur. Properly executed design would include all sorts of seals and gaskets to ensure, that connector boards or drivers themselves do not cause air leakage. Assuming you have your perfect vented box build, you may expect that the SPL curve and input impedance curve will look as on Figure 4.33.

The impedance curve is very familiar and has two characteristic peaks with the dip between them. The dip is located exactly on the box tuning frequency, Fb = 25Hz. My design is a QB3 type with system parameters as shown on Figure 4.33, and we have also assumed, that Qb = Qp = 1000, so we have a very low-loss design. Port diameter in this example is 15cm or 6 inches and the input power to the system is 1watt. Now, assume, that we start reducing port diameter. Initially, the picture will not change by much. However, when the port eventually becomes very small, intuitively, there should be very little difference between the vented box with a "very small port" and sealed box with a "large leakage" problem. In this case, you would expect than the SPL curve will resemble that of a leaky sealed box and the input impedance curve will loose the lower peak and will become a "single peak" curve just like the sealed boxes have. Between those two extremes, you may expect problems commonly labeled as "port non-linearity".

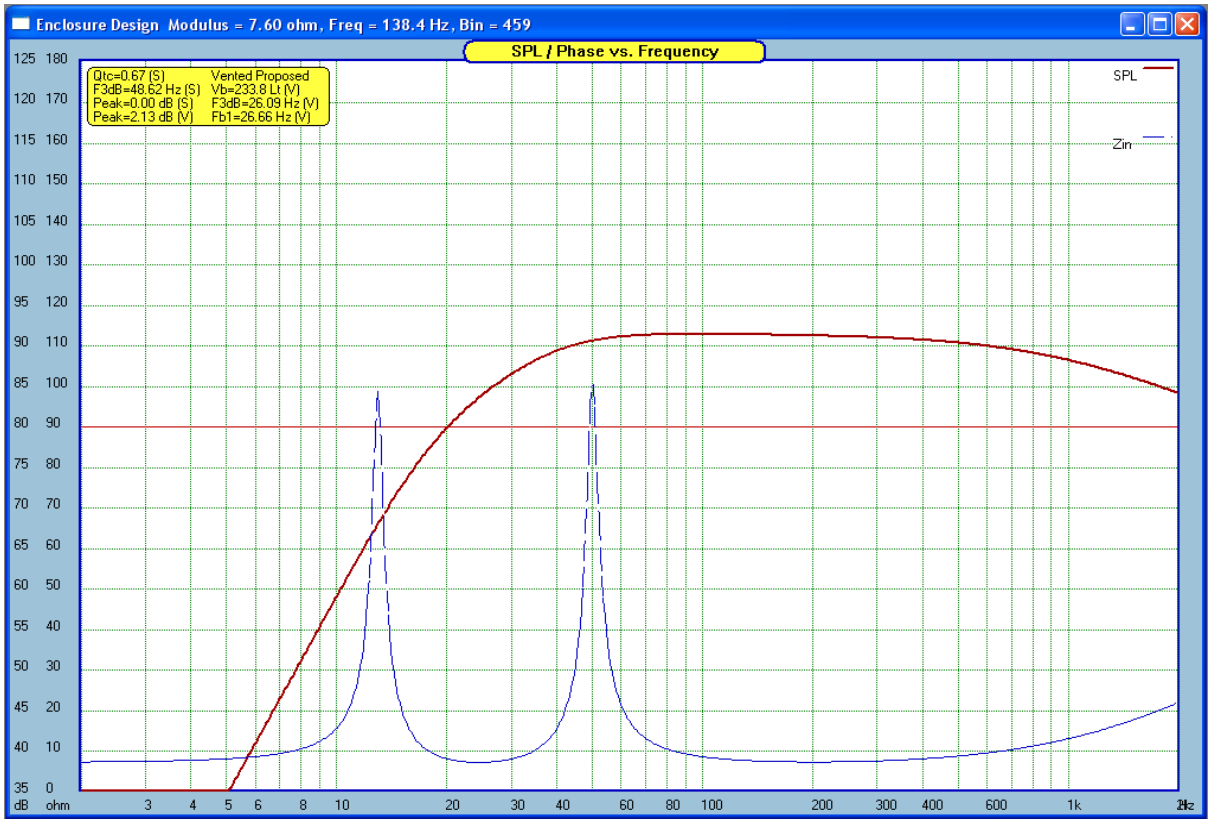


Figure 4.33 SPL and input impedance @ 1watt, port diameter = 15cm.

Now, assume, that we start reducing port diameter. Initially, the picture will not change by much. However, when the port eventually becomes very small, intuitively, there should be very little difference between the vented box with a "very small port" and sealed box with a "large leakage" problem. In this case, you would expect than the SPL curve will resemble that of a leaky sealed box and the input impedance curve will loose the lower peak and will become a "single peak" curve just like the sealed boxes have. Between those two extremes, you may expect problems commonly labeled as "port non-linearity".

Historical research on fluids in tubes

Bies and Wilson actually experimented with an orifice (more like a decent port) 9cm (3.5") in diameter and 2.9cm (1.13") long. They found, that the acoustic resistance, R, of the tube varies with the particle velocity in a similar way as shown on Figure 4.34.Thurston, working with fluid flow through circular tubes found that analogous acoustic resistance, R, and inductance, M, again vary in similar way as shown on Figure 4.34. A landmark paper by Ingard offers an empirical formula for the non-linear component of the acoustic resistance of a tube. Later on, this work has been expanded by Ingard and Ising who offered more complete mathematical treatment of orifice behavior. Backman, experimenting with port non-linearity, plotted driver's input impedance for various input voltages and determined, that the most sensitive to the amplitude variations is the magnitude of lower impedance peak. He pointed out, that the behavior of the vented enclosure approaches a sealed enclosure at high levels. The inner diameter of the port was 102mm (≈4") and the length of the port was 168mm (≈6.6"). Vanderkooy measured and analyzed ports performance at high levels and proposed a simplified nonlinear model of a port. The comprehensive analysis included jet formation, acoustic compression, turbulence noise and distortion.

Simplified view of the problem

Typically, the area of the cone is many times larger than the area of the port. Since the volume of the air pushed via the port needs to be approximately equal to the volume displacement of the cone, the only way to accomplish this, is to push the air in the port much faster. At low SPL levels, the volume displacement of the cone is small and consequently, the cone velocity is low as well. This, in turn, stipulates low air velocity in the port or what is known – a laminar air flow. The process is characterized by a low Reynolds number, Re<2000, and there is no turbulence in the air flow.

$$Re = (r*V_{max}*\rho)/\mu$$

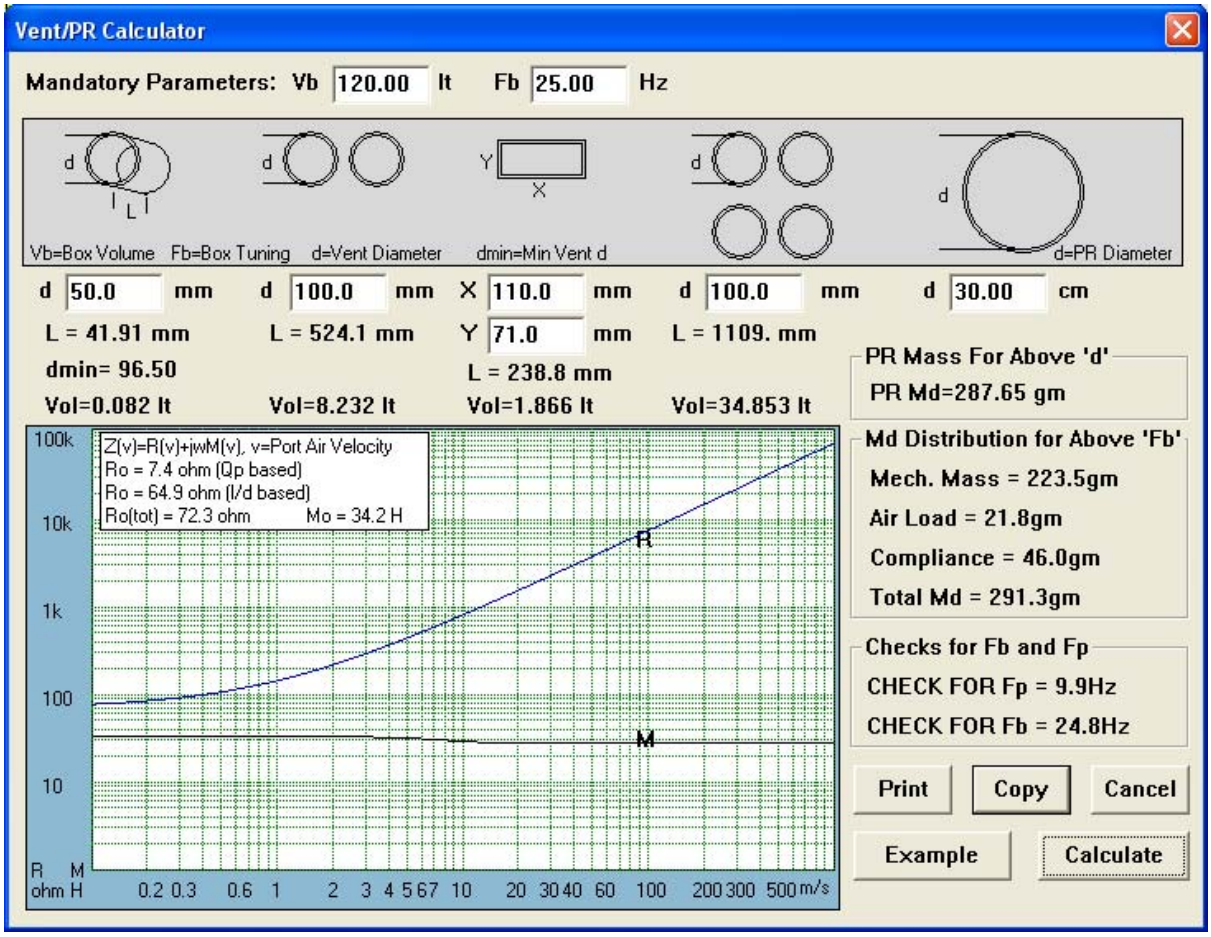


Figure 4.34. Acoustic impedance R+jωM of a short port.

Where V_{max} is the peak flow rate, ρ is the density of, μ is the viscosity of air, and r is the radius of the port. At this point of time, the acoustic resistance of the port, Rp, (port loss) is linear and is relatively low (50-200ohm). Also, the mass of air in the port, Mp, is constant. Remember, that mass of the air in the port resonates at Fb (box tuning frequency) with the compliance of the air inside the box, halting the movement of the cone. If there were no losses, at resonance, Fb, all acoustic output of the system would come from the port. Now, lets start reducing the port diameter. As we can guess, the volume displacement of the cone remains the same, so the air in the port must move faster and faster. Soon, the turbulence occurs in the air flow, jets of air are forming and the process is no longer linear. Reynolds number corresponding to the onset of turbulence is Re ≈ 20000. When the Reynolds number hits 50000, your vent is compressing. It is now more and more difficult to push the air in the vent. The acoustic output of the port is below the expected level (if the process continued to be linear) and you begin to experience "port compression" effect. Port impedance, Zp, has now additional component relating to turbulent air flow and depending on the air velocity, Rp(v). Also, the Mp has changed as well, drifting towards 60-70% of its original value and becoming velocity dependant. The Rp(v) is an interesting element. If plotted in frequency domain, it would resemble the curve depicting port air velocity, shown on Figure 4.35. This should be of no surprise, as Rp(v) is so heavily dependant on the port air velocity. In the frequency domain, the Rp(v) will quickly reach its peak not far away from the lower peak of the input impedance curve. Depending on the geometry of the port and the volume displacement of the driver, the air velocity in the port may reach 50-100m/s. Such a high values of the air flow result in Rp(v) reaching 5-10kohm levels.

The “double-peak” impedance curve is a clear result of the port action. The port air velocity curve plotted on Figure 3 clearly indicates, that the lower impedance peak will be much more affected by the port non-linearity than the upper impedance peak. Therefore, the Rp(v) needs to be incorporated in the driver’s model.

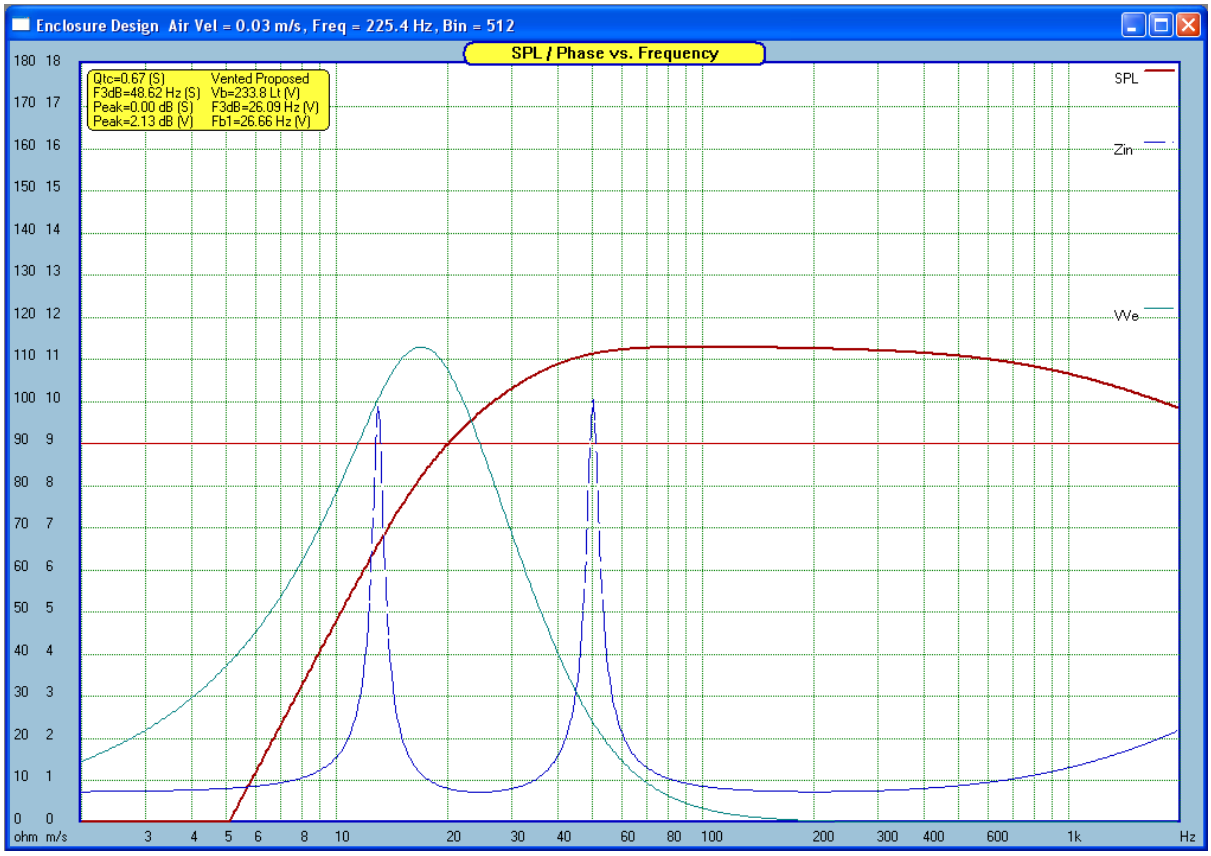


Figure 4.35. Port air velocity (curve 1) for port diameter = 5cm.

Changes in driver’s model

Vented enclosure shown of Figure 4.36, provides different loading for the back of the diaphragm, as compared to the sealed box. The vibrating system and front loading of the diaphragm are represented on the mechanical mobility circuit the same way as for the sealed enclosure. Introduction of the vent adds several more components such as: (1) mass of the air in the port Mmp and its losses Rmp and (2) radiation impedance of the port represented by Rmrp and Mmrp. The air in the port is treated as a mass because of its small volume and more importantly, because it is incompressible. Particles of air will move on both sides of the vent with the same velocity. The air compressed in the box by the back side of the diaphragm has only one path to escape - pushing the air mass trough the vent. Therefore, the pressure path consists of series connection of Cmb, representing compliant air in the box and the four elements of the port. Since the air in the port is incompressible, the immediate layer of air in front of the box (radiation impedance) will be connected to the same velocity line as the entry to the port inside the box. The other end of the masses is connected to the U=0, or reference velocity as required in mechanical mobility circuits. The mechanics of the above process can be easily demonstrated on a physical model of a vented box. Connecting a small (1.5V) battery across vented box terminals, we can displace the cone in or out of the box. Small air-flow detecting device (candle) positioned in front of the port will show significant air movements, in the direction opposite to the diaphragm. The volume of air displaced by the cone should be similar to the volume of air leaving the port. If the difference is significant, than leakage losses are significant. The above experiment clearly shows the pressure (current) path in the mechanical mobility model, so it should now be easy to explain why the compliance of the box is connected in-series with port elements. It is observable, that Cmb and Mmp+Mmrp form series resonant circuit in the mechanical mobility representation. The circuit will act as a “selective short circuit” for the volume velocity Uc, shorting it to U=0 (ground) at the circuit resonant frequency. Because of the circuit losses, the short is not perfect, but velocity Uc will be much reduced. In the practical system this situation translates into much reduced cone excursion at the box resonant frequency.

Acoustical impedance representation shows Cas and Map+Marp forming parallel resonant circuit. Electrical circuit theory advocates that very little energy (current) needs to be fed into the circuit for it to resonate and for the current (volume velocity) in the resonant circuit to be still very high. Therefore, volume velocity in the "feeding" branch, which contains diaphragm output will be very small and volume velocity in the resonant circuit containing port will be high. This effect, although the strongest on the resonant frequency Fb, will extend over some narrow frequency range and on the low-end side creates extended system output. It is the enclosure/port resonance effect, that is being exploited here to augment system output at low frequency. Figure 4.29 shows mechanical mobility (top diagram) and acoustical impedance (bottom diagram) representation adopted for the vented enclosure model. The components are:

Cas,	equivalent compliance volume Vas transformed to acoustical side.
Mad,	mass of the vibrating system Mms transformed to acoustical side.
Ras,	vibrating assembly loss Rms transformed to acoustical side.
Mar+Mab,	air radiation of the front side of the diaphragm + air load of the back side of the diaphragm.
Rar,	air radiation of the front side of the diaphragm.
Cab,	enclosure compliance Vab transformed to acoustical side.
Rab,	absorption losses of the enclosure transformed to acoustical side.
Marp, Rarp	port radiation.
Map,	mass of the air in the port.
Rap,	frictional losses in the port.

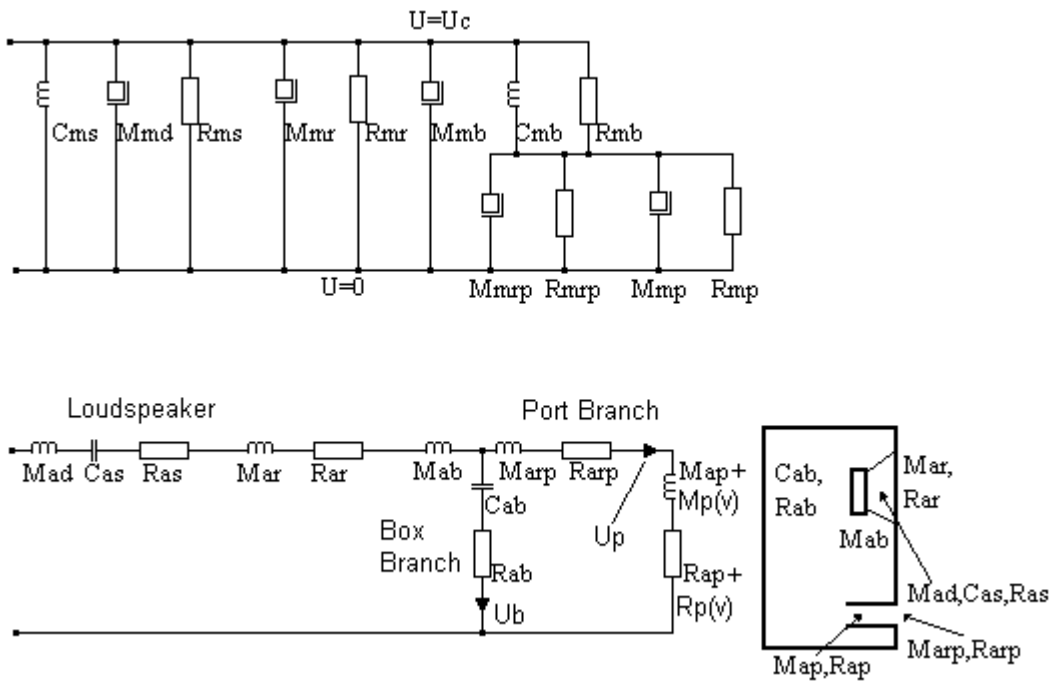


Figure 4.36. Modified model of the vented box including Zp(v) = Rp(v) +jωMp(v)

The non-linear port impedance was implemented as Zp(v) = Rp(v) +jωMp(v) in the port branch. Please note, that Map+Mp(v) will exhibit slight reduction in value as the air speed increases and Rap+Rp(v) will exhibit significant increase in value as the air velocity in the port increases.

Resulting performance

In order to gain some insight into system performance affected by port non-linearity problems, was plotted the SPL for a port of 5cm in diameter for 1W (curve 0), 10W (curve 1) and 100W (curve 2) input power – see Figure 4.37. As we can see, with the increased input power, there is a sort of "saddle" developing on the SPL curve around the box tuning frequency of 25-30Hz. This is exactly the frequency range, where we would expect the port to contribute most to the system SPL. Our small port is clearly not performing as anticipated. Next, we modeled SPL for the same power levels, but this time, we used larger port, 15cm in diameter. The resulting plots on Figure 4.38 do not exhibit the "saddle" any more for 100W power level. Clearly, as the input power is increased, the SPL curves go up, maintaining the approximate shape acquired at 1W power level. It is also easy to observe, that the SPL curves now have resonant peaks above 200Hz, not seen on Figure 4.37. This is the result of enlarging the diameter of the port.

Figure 4.37. SPL for port diameter of 3cm, @1W, 10W and 100W input power

You may remember, that larger port must also be longer if tuned to the same frequency. Now the length of the port is such, that the self-resonances of the port tube fall into much lower frequency range – just to be displayed on the screen. In order to estimate "port compression" effect, you can plot the SPL of the two ports on the same screen – see Figure 4.39. Visual inspection of the graphs shows about 5dB of port compression at Fb = 25Hz. Well, we have just about lost our vented box performance gains if we was to use the smaller port. In the next step, it would be interesting to compare the input impedance plots to see if the lower impedance peak, characteristic for the vented enclosure, indeed disappeared from the plots. The answer is clearly evident on Figure 4.40, where the input impedance for three ports is plotted at 100W power level. The 15cm vent exhibits the expected "double peaks" curve, however the compression is still registered on the impedance plot. Ideally, the port should still be larger. The Reynolds number calculated for these conditions is Re = 68000. Therefore, the port is indeed compressing slightly. However, the 10cm port has the lower impedance peak significantly reduced. This is a sure sign, that this port is too small for the job. The severely undersized 5cm port produces "single peak" impedance curve for 100W input power. This port would also prove to be inadequate for 1 watt of input power. The Reynolds number calculated for 1W conditions is Re = 20000, which is a clear indication, that the port becomes turbulent. Indeed, the corresponding input impedance plot shown on Figure 4.41, is a clear indication of the non-linearity problem @ 1 watt for this port.

Remedies

First and foremost, the problem is related to air velocity in the port. As we know, air velocity in the port, Vp, depends on volume velocity Up, via the port branch divided by the port's area, Sp.

$$\mathbf{Vp = Up/Sp}$$

Since the area $Sp = \pi r^2$, it is easy to observe, that air velocity in the port depends on the inverse squared of the port's radius. It seems rather obvious, that port radius should be kept as large as possible. Dickason recommends 15cm (6") ports for 15" woofer as a minimum and 10cm (4") ports as a minimum for 12" woofers. Salvatti offers a number of recommendations relating to port geometry. This includes balancing inlet and exit flows by using different tapers for inlet and exit. Roozen[3] advocates port geometry based on 6 degrees diverging contour towards the ends of the port. The inlet and outlet are rounded with relatively small curvature. This brief analysis of the vent performance clearly indicates, that nearly all practical size vents will compress at higher SPL levels. Loudspeakers intended for high power stage applications should be give very careful consideration regarding port non-linearity. For this type of application, port compression problem will tend to further degrade system SPL, already affected by the thermal compression during prolonged stage performance.

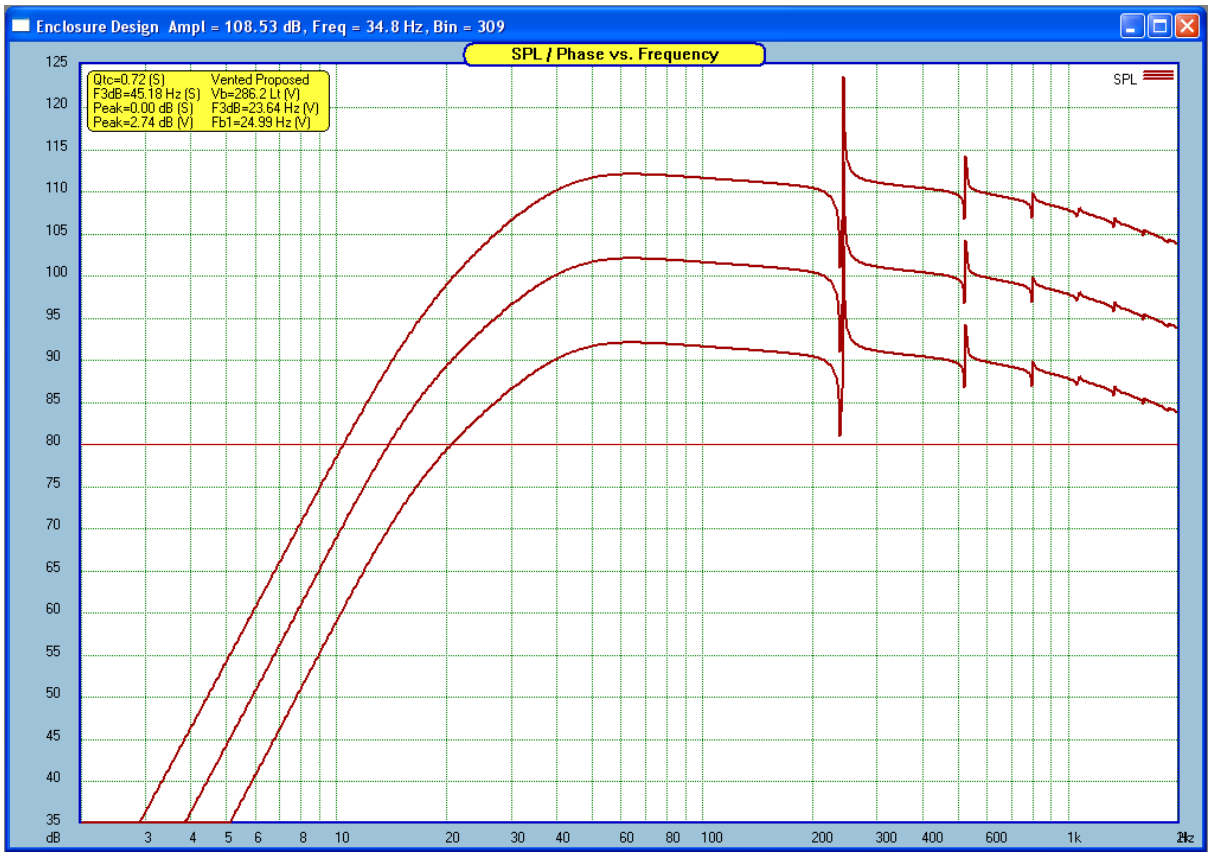


Figure 4.38. SPL for port diameter of 15cm, @1W, 10W and 100W input power

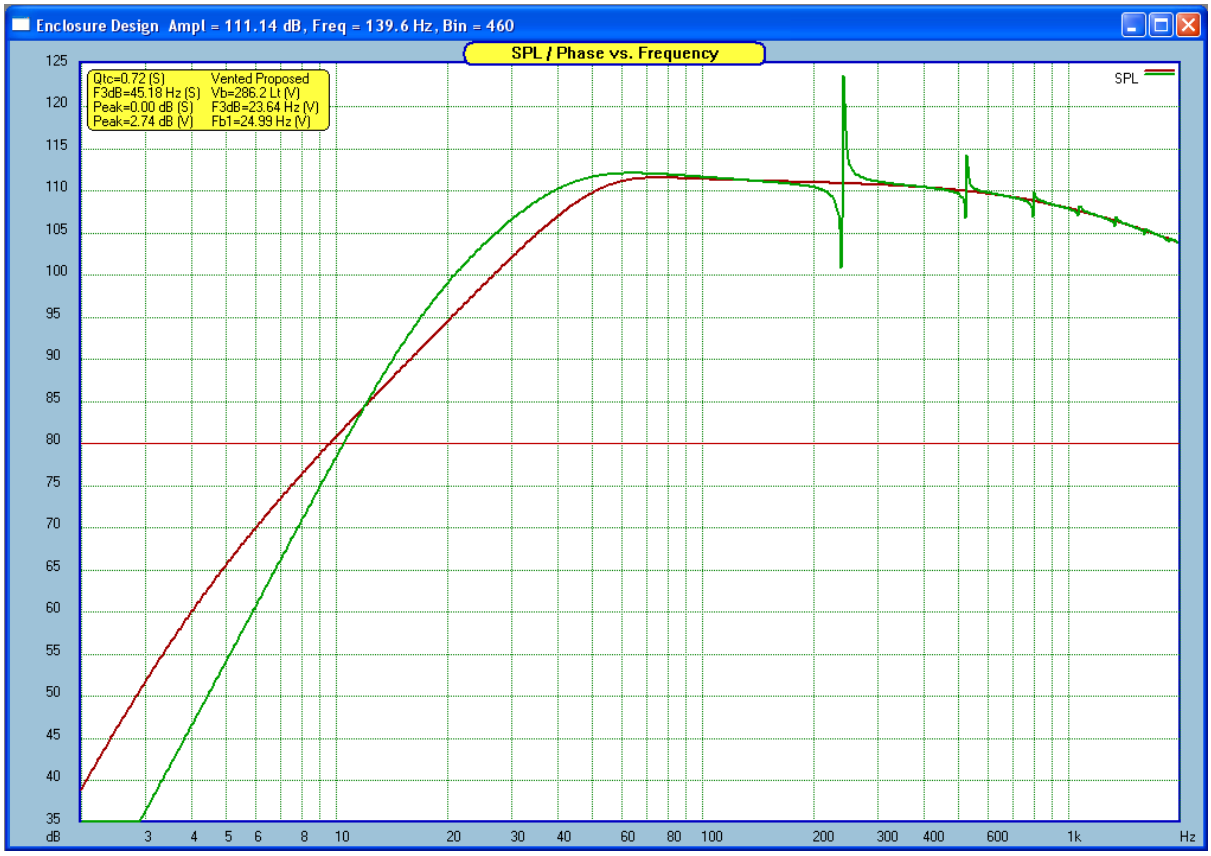


Figure 4.39. SPL for port diameter of 3cm (brown curve), and 15cm (green curve) @100W

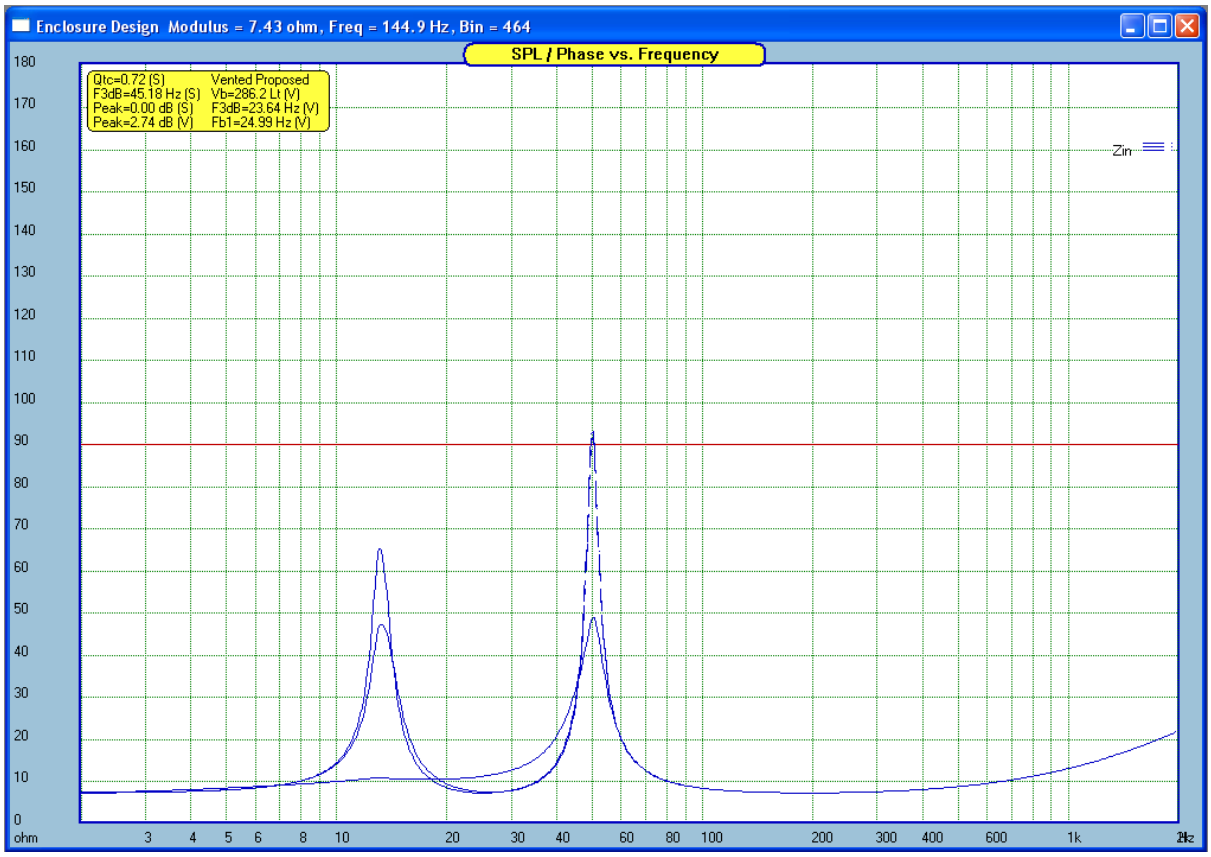


Figure 4.40. Impedance @100W for port diameter of 15cm,10cm and 3cm.

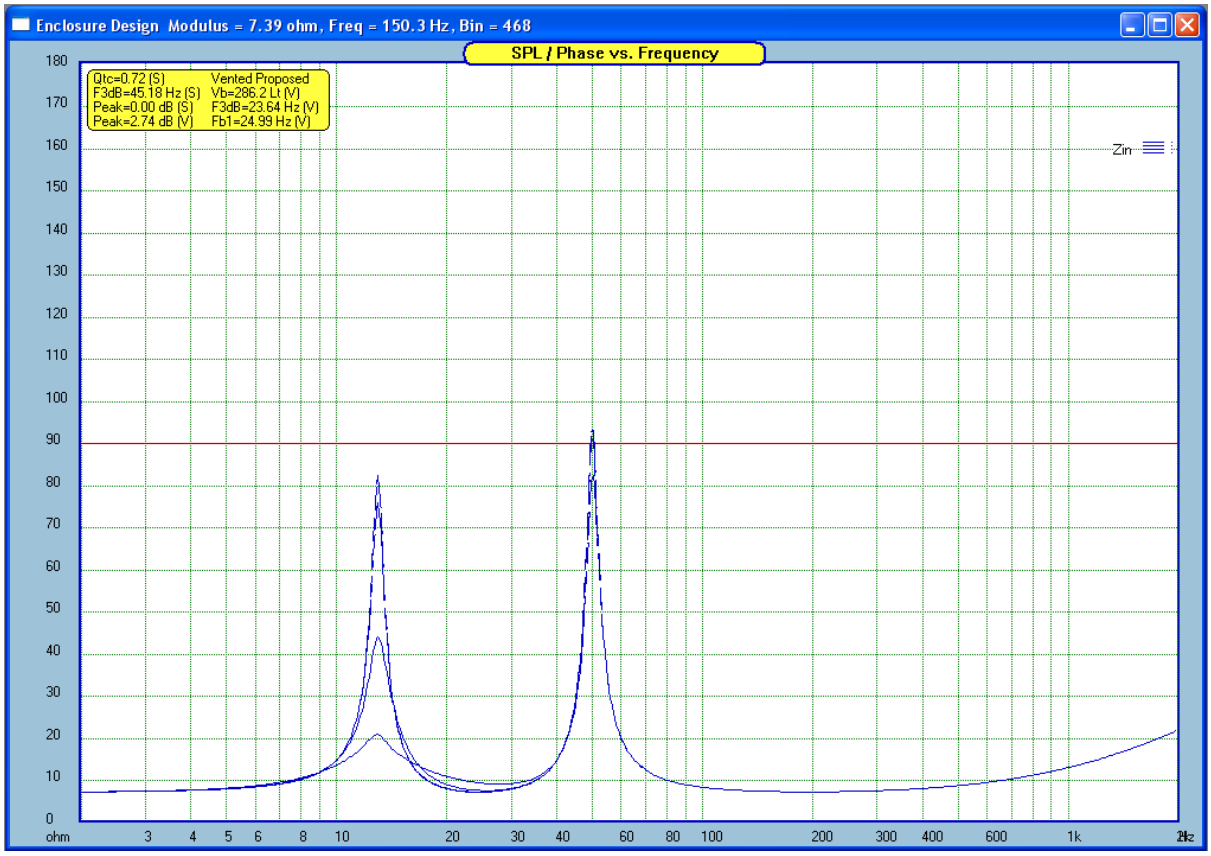


Figure 4.41. Input impedance @1W for port diameter of 15cm, 10cm, 5.0cm and 3.0cm.

BL(x) and Cas(x) analysis

Small-signal model of the loudspeaker and associated enclosure has been increasingly evolving to accommodate non-linear effects inherent in the design of the magnetic motor systems and cone suspension systems. It appears however, that we are in the early stages of the process, as the commercially available drivers are still not fully specified for their large-signal performance.

Indeed, it is very important to characterize the driver for small- and large-signal conditions, as two drivers having similar small-signal data will often perform quite differently under large-signal conditions. In our simplified approach to the problem, we assume, that the large-signal performance of the loudspeaker is mainly governed by non-constant "force factor" – BL(x) and non-constant compliance – Cas(x). Both these quantities are quite dependant on cone excursion – variable "x".

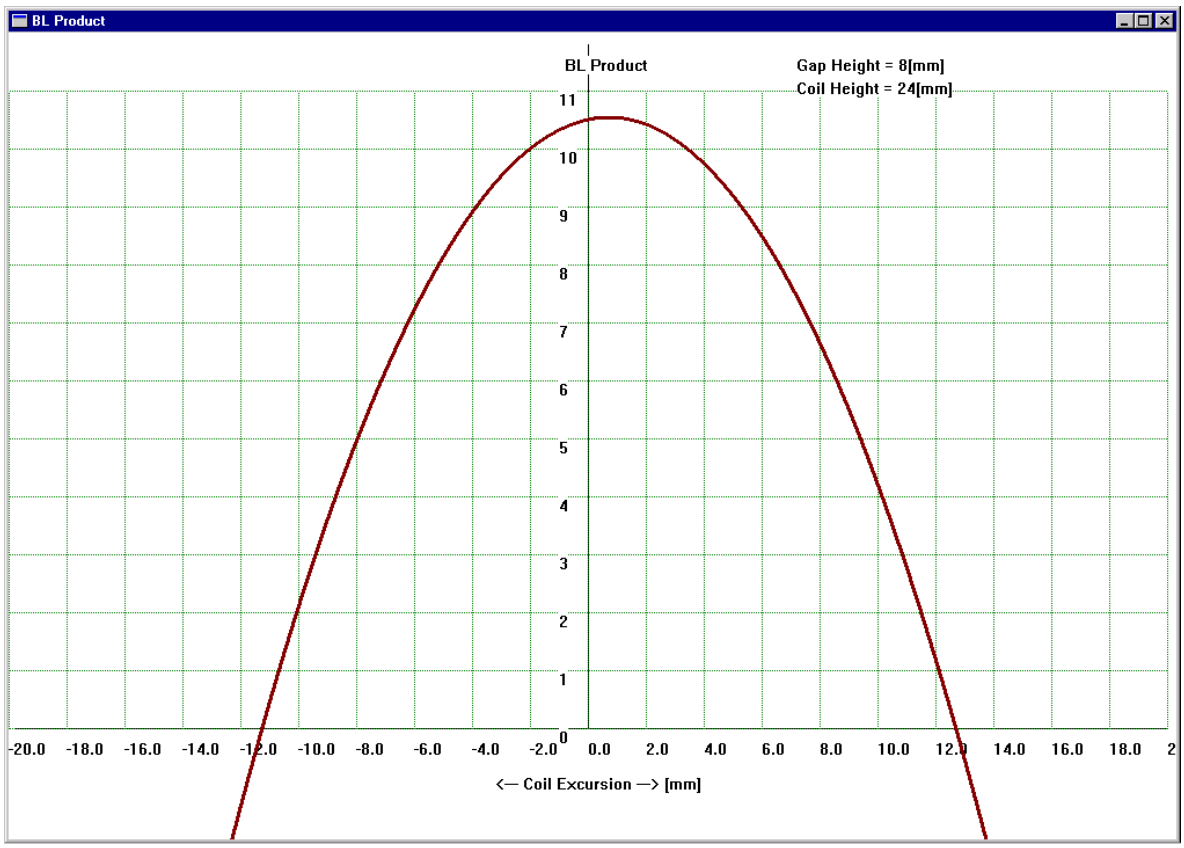


Figure 4.42. Truncated power series - BL(x) becoming negative for larger excursions, BL(0) =10.0

Recent research work focuses on measuring loudspeaker non-linear parameters and presenting them to the user as a power-series coefficients for the force-factor and stiffness. One can only hope, that loudspeaker manufacturing industry will finally embrace this idea and will offer non-linear parameters as a standard set of driver's data. Currently, we seem to some distance away from this point.

Truncated series of power coefficients (first and second order coefficients) approach suffers from several problems. For example: y = BL * (1.0 + 0.01*x - 0.007*x*x) plotted above shows that:

1. BL(x) does not fade gradually for larger excursions and can become negative.
2. Many magnetic "underhung" motor systems do actually have a flat (linear) section around the x=0 excursion and this can not be approximated by the "two coefficient", parabola approach.
3. Many loudspeakers have nearly constant stiffness (1/Cas(x)) over a wide range of excursions and then seep increase at larger excursions. Parabola does not approximate this very well.

The accuracy of power series coefficient method can be much increased by including higher order coefficients.

Possibly the biggest problem for you (and us) is to actually find out what the values of the coefficients are. Please refer to Chapter 17 for more information on large-signal measurement and analysis.

Is there "the next best thing"?.

There is a possibility extracting as much information from the visual inspection of the driver and data sheets as possible. We will end up with an estimation of the large-signal performance, but this is better than nothing.

Loudspeaker driver components are designed with a specific task in mind. Depending on the materials used and the geometry of the design, these components will perform differently in the final design. What we are looking for, is the relationship between design and the intended performance. For example: if you make the voice coil longer, you may expect larger maximum cone excursion, but also some drop in the SPL. If you also use the spider with more corrugation rolls, you would expect higher compliance and again, larger maximum cone excursion – this would be the "long throw woofer".

Now, these are some of the clues to which we need to assign some numerical values.

Estimating Xmax due to BL(x)

Xmax is usually termed as the **"maximum linear excursion". Practical measurements have shown, that this excursion corresponds to 3 – 10% harmonic distortion or 65 – 75% of the drop in BL(x) from its maximum value.** Please keep these figures in mind, as they are my "sanity check" reference.

Formulas for Xmax given below are due to Small and also Gender found good correlation between Xmax and distortion. The formulas link the voice coil length and magnetic gap height and offer an approximated expression for Xmax. The Xmax is expressed in mm.

For "overhung" voice coils:

$$BL(x) = BL_0 \frac{1.0 - \frac{0.005}{X\max} x^{\frac{5.0}{\sqrt{X\max + 10.0}}}}{1.0 + \frac{0.005}{X\max} x^{\frac{18.0}{\sqrt{X\max + 10.0}}}}$$

$$where \quad X\max = \frac{CoilHeight - MagneticGap}{2.0}$$

For "underhung" and "equal" voice coils:

$$BL(x) = BL_0 \frac{1.0 - \frac{0.000005}{X\max} x^{\frac{5.0}{\sqrt{X\max + 10.0}}}}{1.0 + \frac{0.005}{X\max} x^{\frac{12.0}{\sqrt{X\max + 10.0}}}}$$

$$where \quad X\max = \frac{MagneticGap - CoilHeight}{2.0} + 0.15 CoilHeight$$

Formulas for BL(x) are empirical. The formulas are "rough" and could be improved by adjusting the constants but as you will see later, they offer some insight into the difficult issues of driver's non-linearity. More importantly, they link the well known Xmax formulas with my goal - to generate BL(x) curves. Please note, that BL(x) curves shown on Figure 4.43 and Figure 4.44, do not suffer from the problems associated with the "two coefficient" power expansion approach. Also, these curves are symmetrical , and therefore will approximate odd order non-linearities better than even order. You would expect a well designed driver to be the odd order limited, particularly in the suspension area.

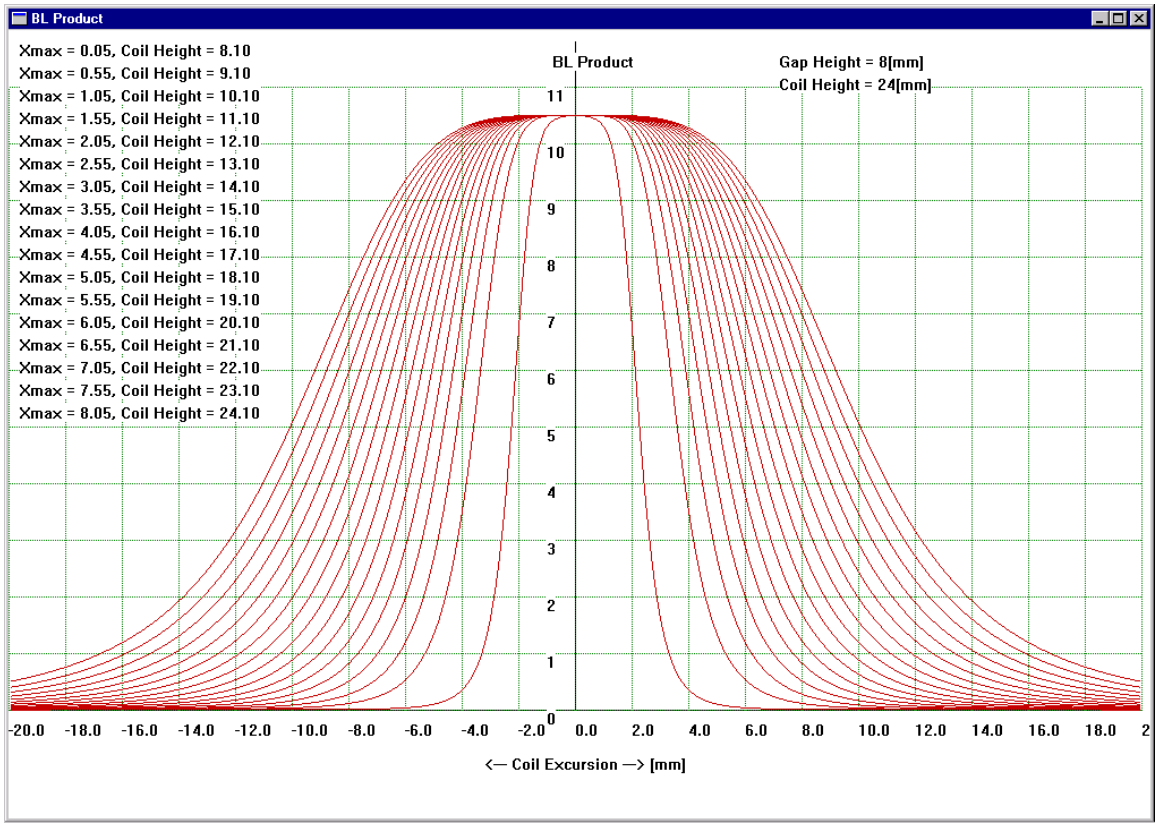


Figure 4.43. Example of the family of overhung Voice Coils

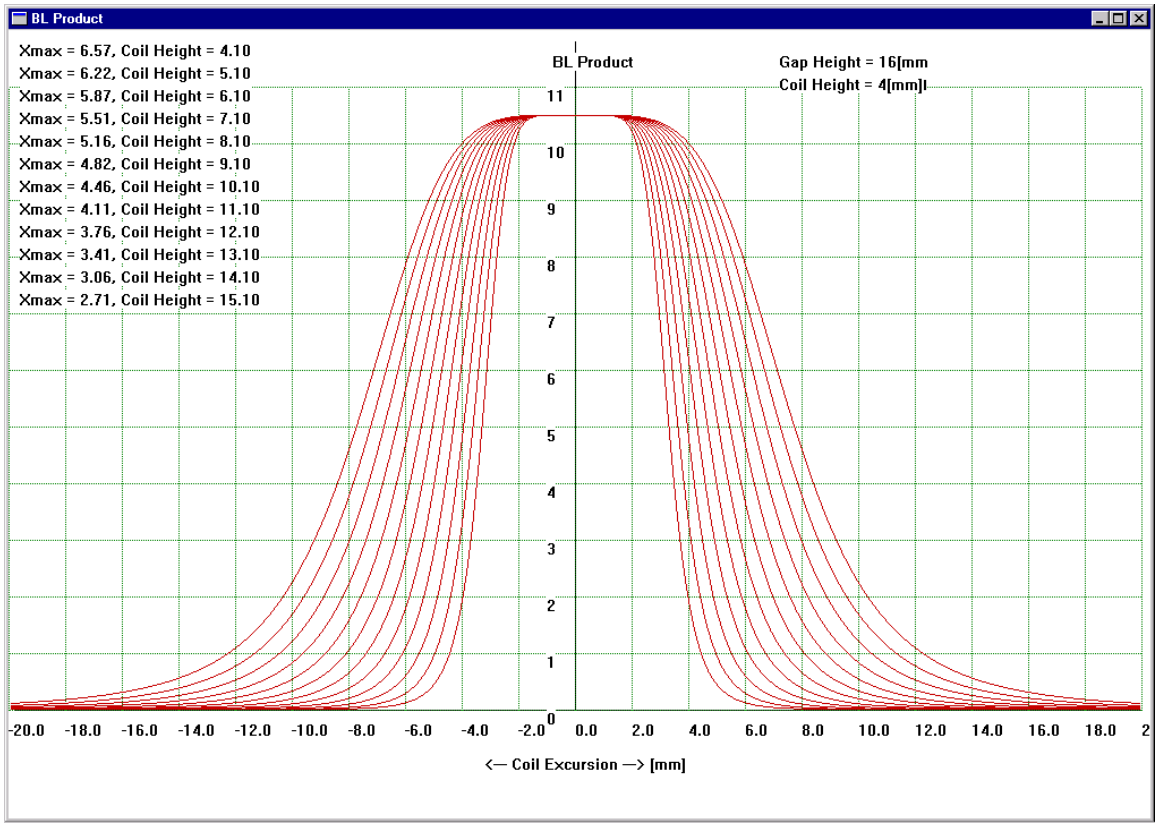


Figure 4.44. Example of the family of underhung Voice Coils

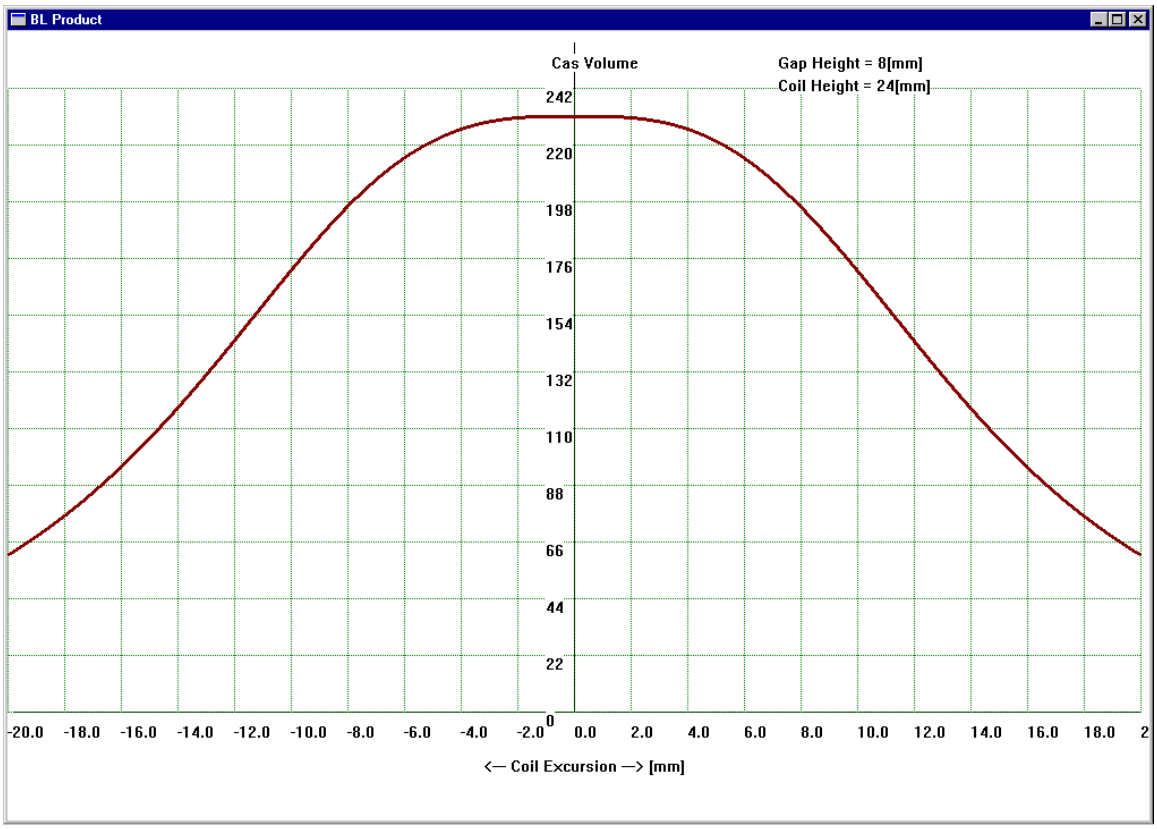


Figure 4.45. Estimating Cas(x) non-linearity. Assumed Xmax = 10.0mm

Estimating Xmax due to Cas(x)

Expression for Xmax due to the spider was developed in Chapter 2.

Sanity check

Having estimated Xmax due to BL motor strength and also Xmax due to stiffness of the suspension, it would be sensible to check if the computer model based on these assumptions was indeed of any practical use. Fortunately, we have a good reference point provided by the relationship between the cone excursion Xpeak, and SPL of the driver, widely developed in the literature:

$$X_{peak} = \frac{(1.18*10^3)10^{SPL/20}}{f^2 a^2}$$

Where: "f" is the test frequency in Hz and "a" is the radius of the driver in mm.

In this example a 25cm driver (125mm radius) at 52Watt input power and 50Hz test frequency, develops 105dB SPL, and cone excursion is ~5.6mm – the theoretical formula gives 5.37mm. We are **within 5%** from the desired value.

We would like to point out, that we are now in a good position to discard the option of not performing the large-signal analysis at all, as we do have a crude model, working with 15% accuracy. In summary:

1. We have examined physical design of the driver, looking at the voice coil length and magnetic gap height. These parameters are likely to be included in your driver specifications.
2. With the help of some crude mathematics, we have built my BL(x) curve, taking into account available and measurable references, eg: BL drops to 65-75% at Xmax (due to BL).
3. We have also visually inspected the design of the spider and from there, we calculated my Xmax due to the spider. Then, we developed my Cas(x) curve again, assuming, that just like for BL(x) curve, the 65-75% drop in Cas(x) value would correspond to cone excursion = Xmax (due to Spider), as shown on Figure 4.45.

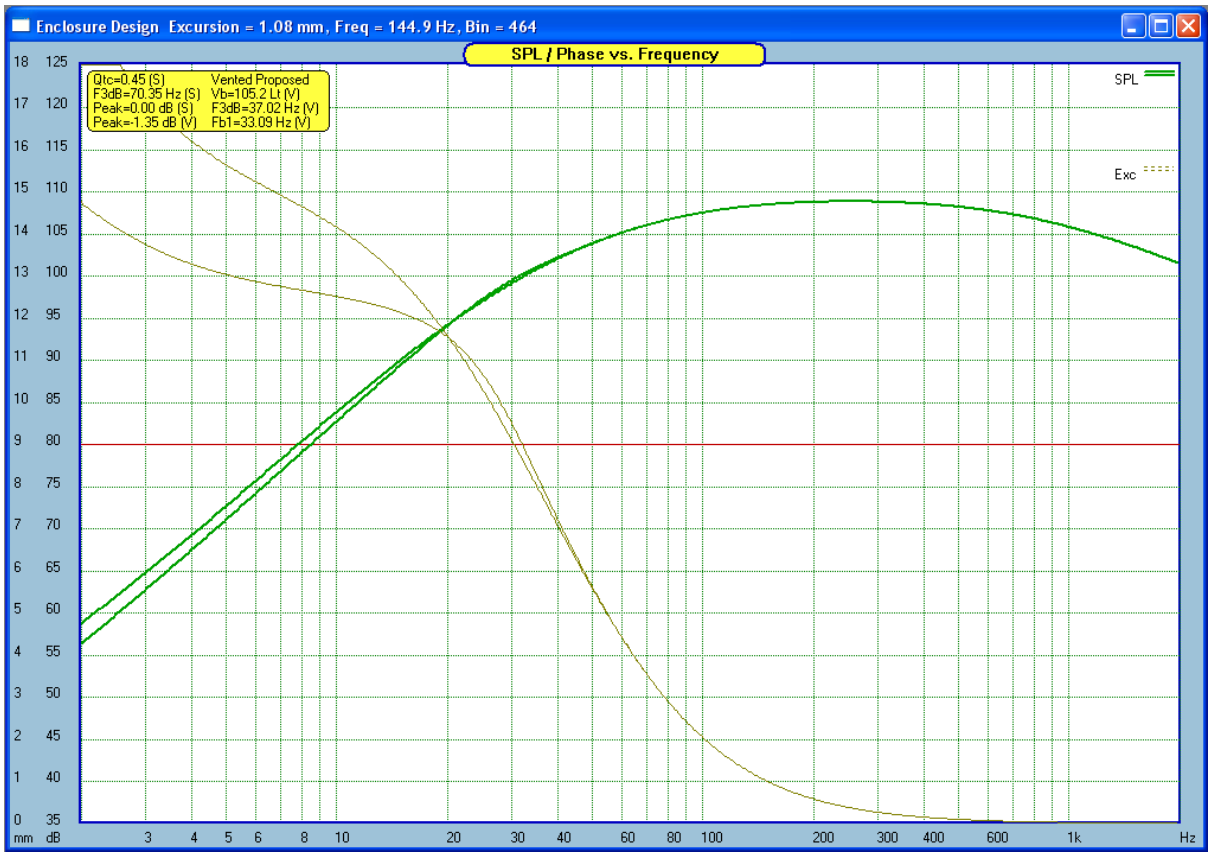


Figure 4.46. Checking the model of the example driver for accuracy.

How useful is this simple model ?. In order to evaluate this, we would like to take a look **at "suspension instability" issue and associated with it, "soft clipping"**. This problem would be more related to vented enclosures rather than sealed boxes, because sealed enclosure automatically offers "stiffer" suspension due to the air trapped inside the box. We would like to start with a quick remainder about spiders.

Suspension Instability

When a driver is operated at input power levels leading to excessive cone excursion (cone excursion > Xmax) there is a distinct possibility, that the voice coil will eventually "jump out" of the magnetic gap when approaching system resonance. When the coils is moved into much reduced BL(x) region, there is a dramatic increase in Qes of the loudspeaker. Remember, that electrical dumping (Qes) is the primary means that the amplifier uses to control the driver. Drivers with low Qes will be tightly controlled, while drivers with high Qes will exhibit very poor and "ringing" behavior in their impulse response. So we are now in a situation, where the amplifier looses the control over the driver excursion and if something is not done to prevent it, the voice coil will leave the magnetic motor system permanently. This situation is depicted on the Figure 4.47 below. The driver has 24mm long voice coil and 8mm magnetic gap. This gives us:

Xmax = (24mm – 8mm)/2 = 8mm due to BL(x)

It is also easy to observe, that the voice coil will be leaving the magnetic motor system at X > 16mm. There is some magnetic field extending above and below the top plate, so at X>16mm, the coil is above the top plate, but is still in some residual magnetic field. You would expect, that for high input power applied to the speaker, this field may still be able to push the coil a mm or so further out. Ideally, our simple model should be able to show this behavior. Now please consider two cases:

Case 1

The suspension system and motor systems are designed with the characteristics as shown on Figure 4.48. Here, the BL(x) is much more non-linear than Cas(x). In fact, this design would be characterised by distortions generated mainly due to BL(x) non-linearity.

The Cas(x) curve is much wider than BL(x) curve. It is likely, that a “long throw” woofer would be designed along these lines and chosen for this application.

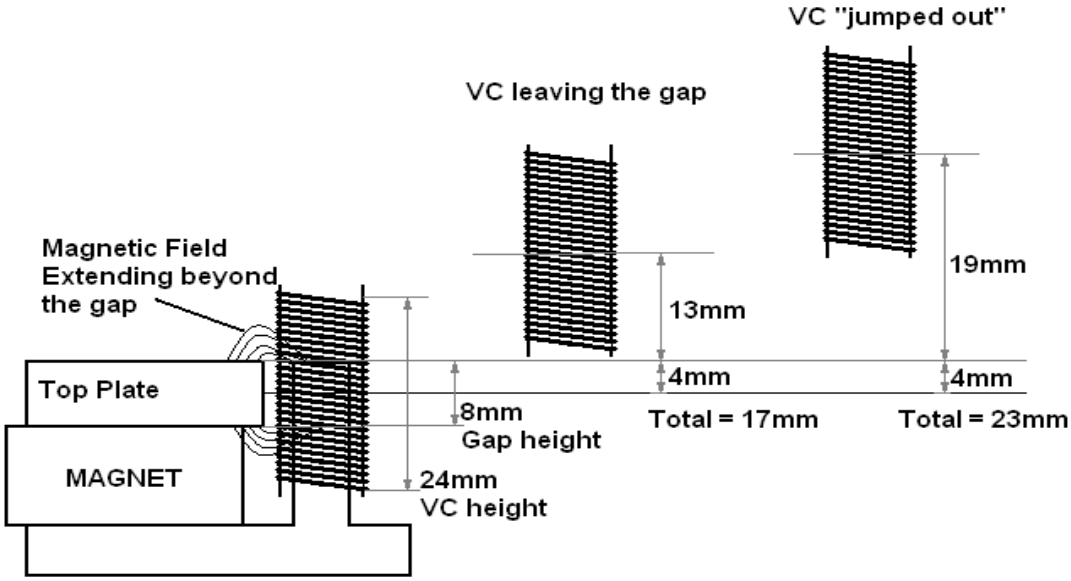


Figure 4.47. Excessive VC excursion leading to suspension instability

Case 2

The suspension system and motor systems are designed with the characteristics as shown on Figure 4.49. Here, the Cas(x) is more non-linear than BL(x). In fact, this design would be characterised by distortions generated mainly due to Cas(x) non-linearity. The Cas(x) curve is providing “soft clipping”.

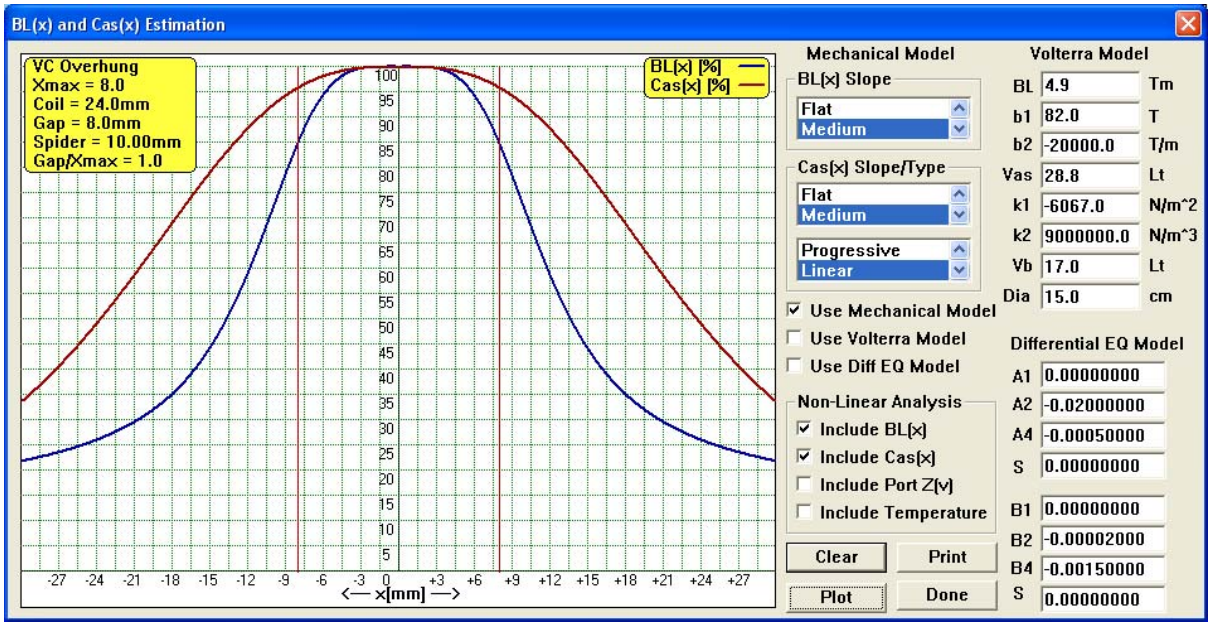


Figure 4.48. Very linear Cas(x) characteristics.

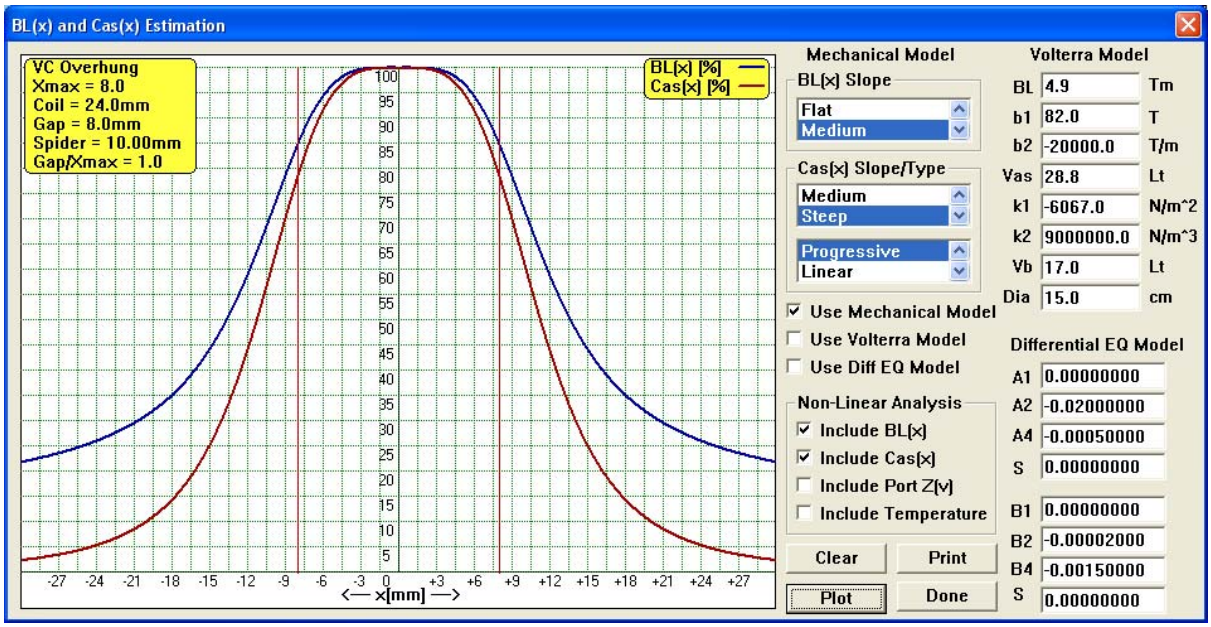


Figure 4.49. Cas(x) implemented as "soft clipping".

Soft Clipping

Figure 4.48 shows the "high compliance" and very linear suspension case, leading to voice coil "lock out". Initially, (curve "light brown") the voice coil moves only as much as the combined effect of the relaxed suspension and BL(x) non-linearity allows for. At frequencies around 15-18Hz, the cone is approaching the lower resonant peak of the impedance (one of the system resonances) and the movement begins to be controlled by the mechanical losses in the system. The control is much poorer and this leads to further increase of excursions and the voice coil lands outside of the magnetic system. Figure 4.50 shows this situation, where curve labeled "light brown" is reaching 23mm of excursion and corresponds to Figure 4.47 – VC "jumped out" of the motor system.

As a remedy to this problem, the "progressive" suspension has been proposed. Again, Figure 4.50 shows clearly the effect of "soft clipping" provided by the "progressive" suspension system. This design adopted different characteristics for Cas(x). Now the spider does not allows the cone to move too far. The Cas(x) curve is much narrower and has stepper slopes designed to prevent the voice coil from traveling beyond the magnetic gap. Indeed, if a design shown on Figure 4.49 was implemented in the same driver, the cone excursion would be quite flat, as the voice coil is now heavily restricted from leaving the magnetic gap at any frequency. There is only a small rise at 15-18Hz, as opposed to the previous peak. It is easy to observe, that 12-13mm is just about as far as the spider and the rim can stretch for this input power level. Acoustic output monitored by a microphone would show symmetrical clipping.

Interesting observation can be made about the shape of the cone excursion curves shown on Figure 4.50. Cone excursion curve due to BL(x) looks quite different from the cone excursion due to the Cas(x) non-linearity. The BL(x) exhibits a sharp and pronounced peak around the system resonance frequency for the reason explained above. The cone excursion curve controlled mostly by the stiffness or Cas(x) is significantly flatter. This was to be anticipated, as the "progressive" spider with the characteristics shown on Figure 4.49 continues to be the dominant, movement restrictive element in the driver design.

Figure 4.50. Cone excursions. "light brown"- Cas(x) nearly linear, "green" – Cas(x) soft clipping, "red" – BL(x) and Cas(x) completely linear..

BL(x) and Cas(x) Dialog Box

As discussed previously, current release of the program provides you with an alternative method of estimating BL(x) and Cas(x) curves. Only three parameters are required to enable the large-signal evaluation:

1. Voice coil height
2. Magnetic gap height and
3. Estimation of the spider linearity – Xmax due to spider.

(1) and (2) are likely to be specified in the driver's data sheets. The spider's linearity, (3), can be estimated from the geometry of the spider. However, there can be additional information available, which provides you with more accurate estimation of the non-linear behavior of the driver.

For example: if the center pole of the motor system is extended above the top plate, it is likely, that linearity Xmax, due to BL(x) will be improved. The amount of the center pole extension will determine the improvement in linearity. One way of reflecting this additional data in the analysis, is to use provided dialogue box – "BL(x) and Cas(x) Estimation".

This box allows you to "expand" or "shrink" the BL(x) and Cas(x) curves. Option "Medium" from the "BL Slope" and Cas(x) list boxes will typically correspond to the BL(x) and Cas(x) curves dropping to 65-75% at x = Xmax.

An example of the suspension system and motor systems are shown on Figure 4.48, and Figure 4.49 The width of the BL(x) curve is calculated from coil and motor parameters. Therefore, another simple way of adjusting the width of the BL(x) curve is to simply modify the voice coil height and magnetic gap height. These two values are used only for calculating the Xmax due to magnetic system and can be easily tuned to obtain the best approximation of your BL(x) curve. The controls on the dialogue box are really self-explanatory. Simply select the settings that approximate your BL(x) and Cas(x) curves and continue with the analysis.

Harmonic Distortion

Total Harmonic Distortion (THD) coefficient is calculated as per formula given below. The nominator contains less summation factors, because we do not include the fundamental frequency (F_1) in the sum. Summation is performed over the first 64 harmonics. The Fi, denotes the level of each harmonic.

$$THD = \sqrt{\frac{\sum_{i=2}^{64} F_i^2}{\sum_{i=1}^{64} F_i^2}} * 100\%$$

The dialogue box simulating the THD is very simple and self-explanatory. All you need to do, is to **enter the desired power** level, Pin (Watts) and select enclosure type from the provided list box. Then click on the "Plot" button and you can watch the THD curve being plotted together with the harmonic content of the distorted signal.

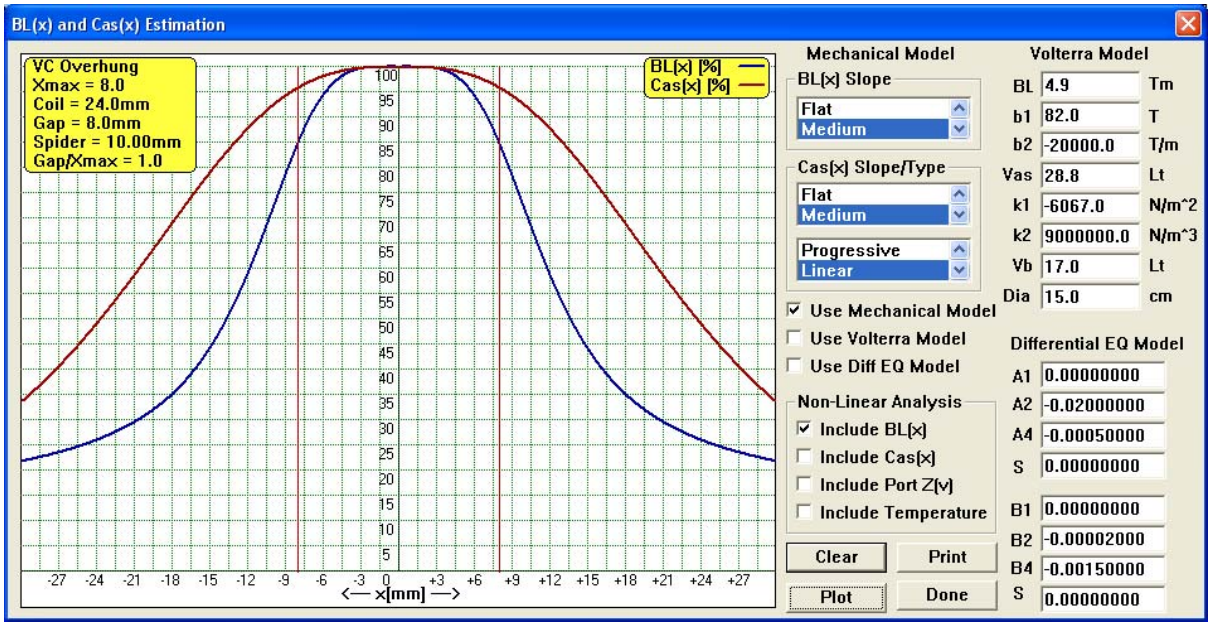


Figure 4.51. Example of BL(x) curve highly non-linear at low cone excursions.

The THD performance of the driver is highly dependant on the characteristics of the BL(x) and Cas(x) curves. The BL(x) curve exhibiting high non-linearity at low cone excursions and dropping more gradually (see Figure 4.51) will produce higher level of THD at low to moderate cone excursions (10% at 42Hz) and lower THD levels at higher cone excusrions (70% at 2Hz) – see Figure 4.52. Conversly, the BL(x) curve as shown on Figure 4.46, will produce lower level of THD at low to moderate cone excursions (5% at 42Hz) and higher THD levels at higher cone excusrions (83% at 2Hz) – see Figure 4.54. Quite similar analysis could be conducted for Cas(x) curves. The "soft clipping" issue discussed previously can only implemented at the expense of significant increase of harmonic distortion.

The THD dialogue box has also **"Fourier Analyser"** built-in. This unique function allows you to observe the level of the first 32 harmonics of the sweeping sine wave as it is reproduced by the non-linear driver. The test sine wave will be distroted accordingly to the BL(x) and Cas(x) nonlinearities. The symmetric nature of the BL(x) and Cas(x) curves will produce dominant odd-harmonics.

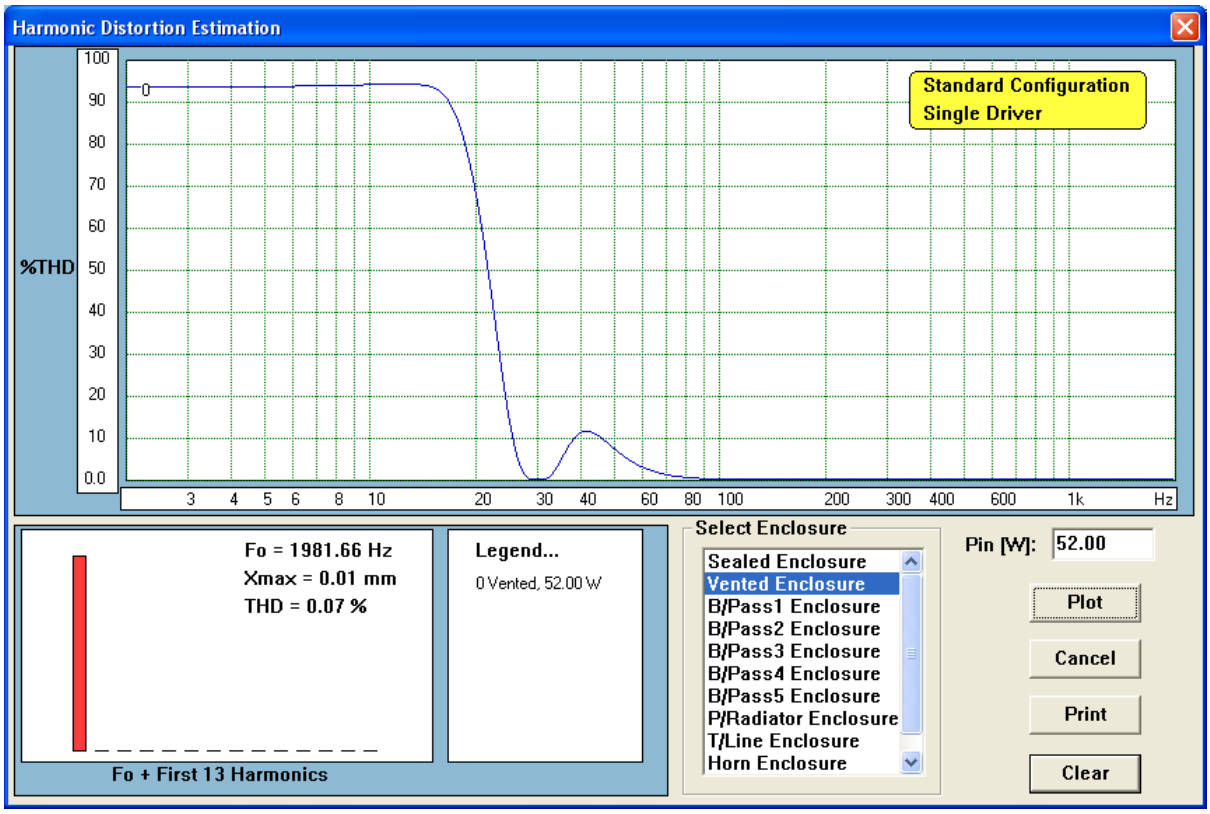


Figure 4.52. THD corresponding the BL(x) design shown on Figure 4.44

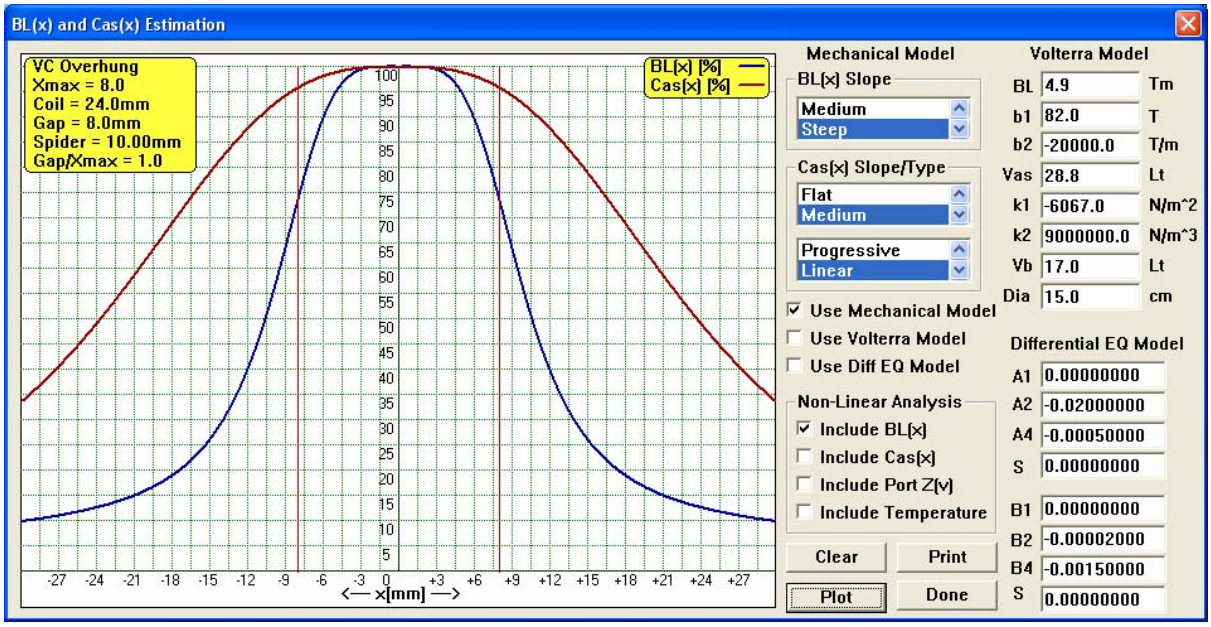


Figure 4.53. Example of BL(x) curve highly non-linear at high cone excursions.

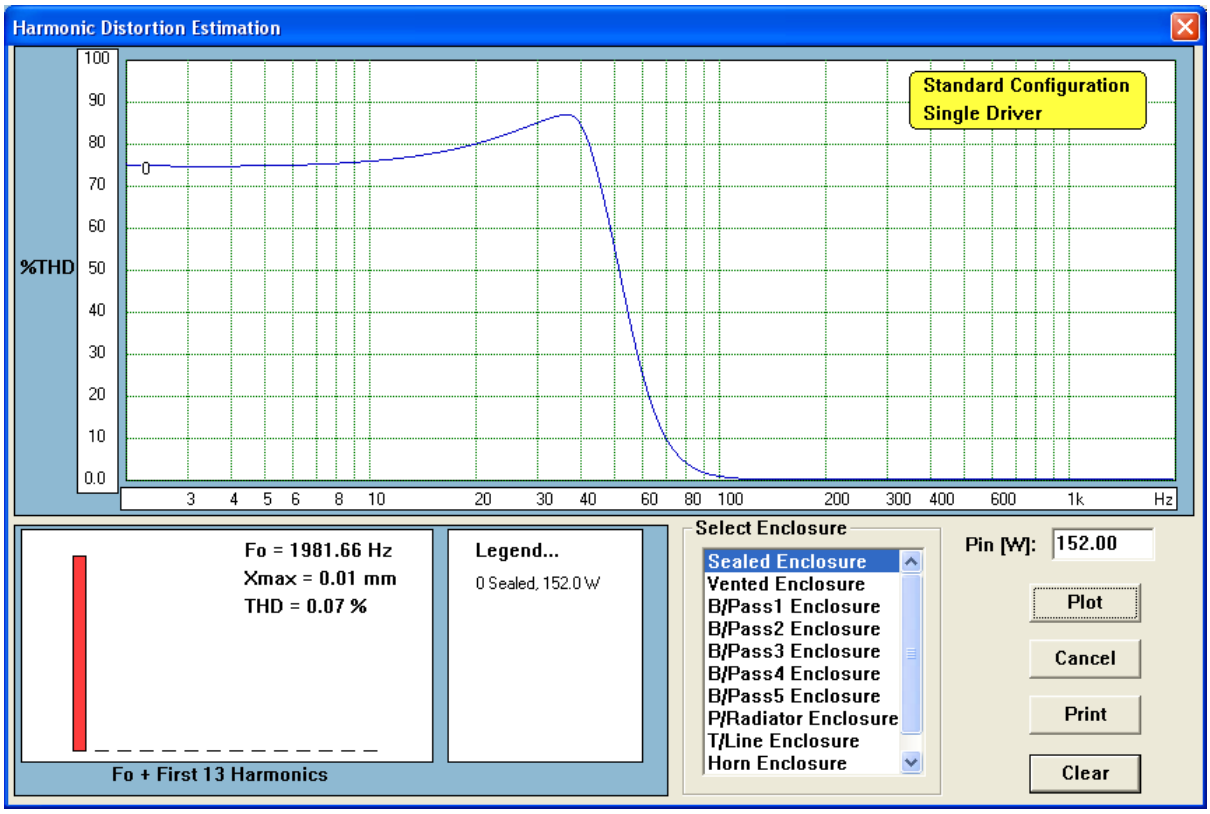


Figure 4.54 Example of harmonic distortion plot for 152W and sealed enclosure

Measuring BL(x) curve

The BL(x) curve can be measured quite simply and with minimum hardware. Here is the basic idea:

1. Place your woofer vertically up on a flat surface.
2. Apply mass M1, sufficient to displace the cone by -10mm to –12mm
3. Apply DC current I1 to the driver, to return the cone back to x = -1mm.
4. Apply different mass M2 (<M1) to displace the cone by -7mm to –9mm.
5. Apply DC current I2 to the driver, to return the cone back to x = –1mm.
6. Calculate:

$$BL(x) = BL(x = -1mm) = g\frac{M_1 - M_2}{I_1 - I_2}, \qquad g = 9.81\frac{m}{s^2}$$

You have now your first data point for BL(x = -1mm). The idea of returning the cone back to the same deflection x, allows you to nullify the influence of the stiffness in the BL(x) formula. Numerical value of stiffness will be the same for both masses, only if the cone returns to the same x-position.

Re-apply mass M1, but now apply enough DC current I1, to return the cone to x = -2mm. Then again, re-apply mass M2 and only enough DC current I2, to return the cone to x = -2mm. Once again, apply formula given in step (6) to calculate BL(x = - 2mm).

Continue on, until the BL(x) is completed within say, +/-10mm. You may have to turn the speaker upside-down and suspend the weights from the speaker to obtain BL curve for positive displacements. The deflections for M1 and M2 were given only as a guide. For "long-throw" woofers you may opt to use larger displacements – simply apply common sense here. After plotting all calculated BL(x) values, you can easily extrapolate the actual BL(x) curve. The next step is to select appropriate parameters in SoundEasy that approximate the measured BL(x) curve best.

Unfortunately, the same procedure can not be applied to stiffness (or 1/Cas(x)) curve. Unless, you have a hard copy of the Cas(x) curve (or stiffness plot) you will need to rely on the Cas(x) estimation from spider's geometry. You may also be able to measure Cas(x) curve – as shown in Chapter 16.

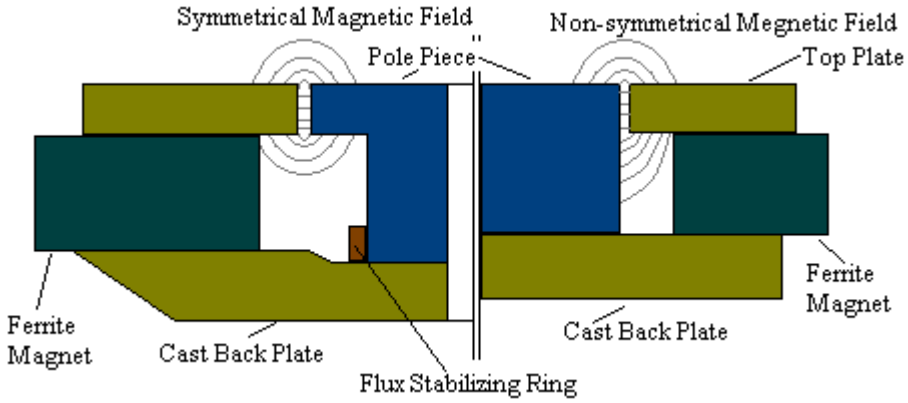


Figure 4.55. Example of a symmetrical magnetic field motor design.

Mechanical Model and Volterra Model

Measuring and modeling nonlinear behavior of a loudspeaker is a difficult process. We have taken the "step-by-step" approach to introducing these features into the program.

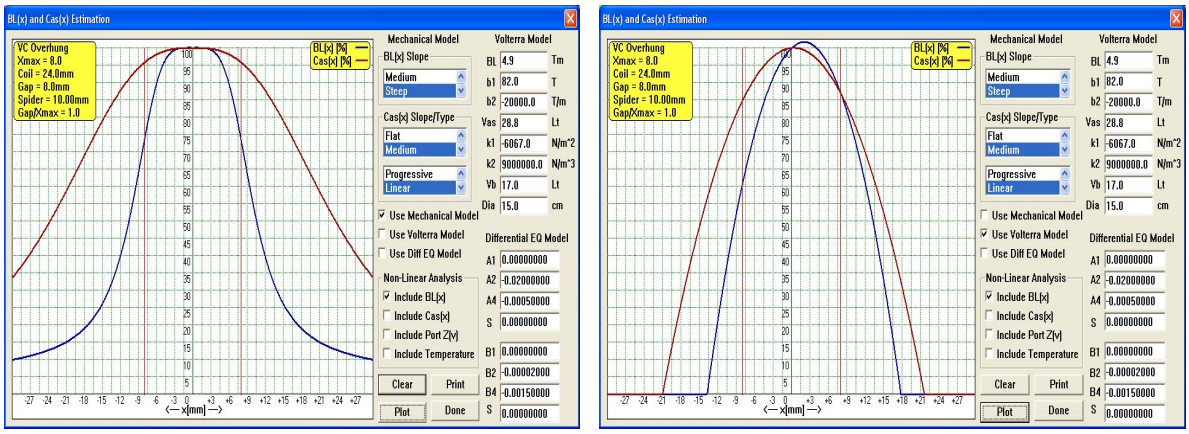


Figure 4.56. Mechanical and Volterra BL(x) and Cas(x) models.

Mechanical model of loudspeaker was our "first step" into the area of nonlinear operation of the loudspeaker. The model was simple and was able to relate easily visible and measurable mechanical parameters of the loudspeaker to it's performance at high input power. In this program release, we have implemented measurement system (refer to Chapter 16) for extracting non-linear parameters suitable for Volterra model of loudspeaker non-linearity. It can be seen clearly, that the two models produce different BL(x) and Cas(x) curves, particularly for larger cone excursion. This will have impact on all performance curves plotted on the "Enclosure design" screen for higher power levels.

The built-in truncated Volterra model has the advantage of providing a set of coefficients extracted and calculated from loudspeaker under actual operating conditions. This is important and would indicate, that at least the initial section of BL(x), Cas(x) and Le(x) curves is approximated reasonably well for -Xmax < x < +Xmax. For cone excursion values higher than Xmax, the truncation becomes quite evident. Also, loudspeakers with flatter section of the BL(x) and Cas(x) curves around -Xmax < x < +Xmax are not approximated well by the parabola-type of curves. The mechanical model holds a little better in this area and also provides more gradual and realistic "decay" of the BL(x) and Cas(x) curves for higher cone deflections. On the disadvantage side, this model is purely based on the mechanical properties of the loudspeaker. Also, the model is symmetrical for positive and negative cone deflections. There are several other non-linear models and measurement techniques proposed over the years by the scientific community. Some of them exploit MLS properties and others stay in time domain and propose to solve a set of differential equations in order to obtain a set of non-linear coefficients. We will continue to evaluate these methods and implement other techniques gradually into the program as well.

You can switch between the models by selecting **"Use Mechanical Model" or "Use Volterra Model" check boxes**. Volterra model parameters, that are directly affecting the shape of the curves are highlighted in red.

Differential Equation Model.

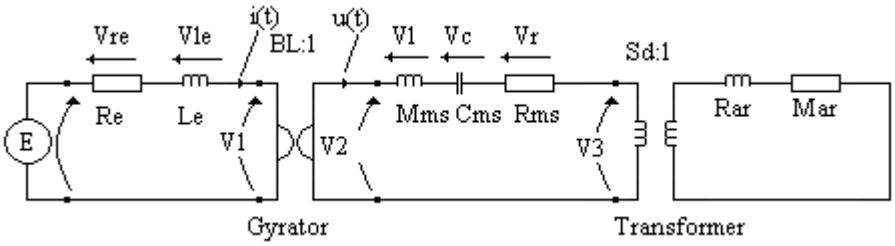


Figure 4.57 **Mechanical impedance** representation of a loudspeaker.

System equations can be derived starting with the basic of the electro-mechanical transduction.

$$F = Bli \quad and \quad e = Blv$$

Where: v = velocity of the conductor and i = current through the conductor

We can then introduce the nonlinear functions for BL and Cms. As far as the shape of the BL(x) and Cms(x) functions is concerned, one can employ an exponential function of the following prescription:

$$BL(x) = BL_0 e^{A4(x-S)^4 + A3(x-S)^3 + A2(x-S)^2 + A1(x-S)}$$

$$Cms(x) = Cms_0 e^{A4(x-S)^4 + A3(x-S)^3 + A2(x-S)^2 + A1(x-S)}$$

BLo = Static force factor
Cms_0 = Static mechanical compliance
A1 = Asymmetry about x = 0
A2 = Flat (low value) or Peaked (high value) around x = 0
A3 = Not used, but can be useful as absolute value, A3 = 0
A4 = Wide (low value) or narrow (high value) of the curve as it falls to y = 0
S = Left / Right shift of the whole curve

There are several advantages of using this function rather than simple quadratic approximation of the BL and Cms curves. The quadratic approximation (or polynomial approximation) tends to positive or negative infinite value for large excursions. The BL factor will clearly go to zero as the excursion increases. The exponential approximation is more realistic in following the BL(x) and Cms(x) shape of real loudspeakers.

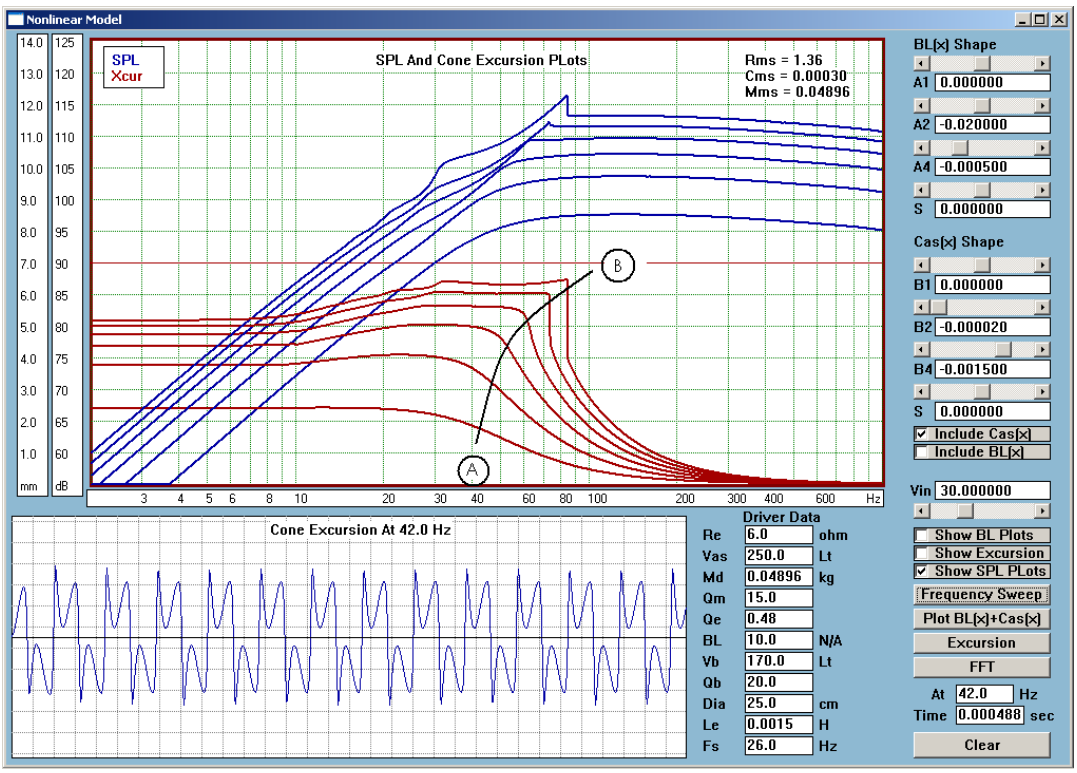


Figure 4.58. SPL and cone excursions for different input voltages – Cas(x) non-linearity only.

We can now consider the SPL and cone excursion of the loudspeaker for different input voltage levels in the frequency domain. It is clearly observable, that the resonant frequency of the system increases with the applied amplitude of the input voltage. This phenomenon is the result of increasing stiffness of the practical suspension systems, as the cone moves further and further away from the central position. As in the example above, at the extreme input voltages, one can observe the system resonance shift (moving from A to B) and finally, the vibrating system "jumping into another state" – the vertical portion of the cone excursion curve.

Differential Equation Model seems to be the easiest to understand and use. We recommend this model for your modeling work.

Enclosure Self-Resonance

The Thiele-Small approach to electro-mechanical behavior of a loudspeaker is based on "equivalent circuit" consisting solely of resistors, capacitors and inductors. This is also known as the "lumped element" model. The sound produced by the loudspeaker can then be obtained via a relatively simple circuit analysis. In the lumped element model, it is assumed, that driver's enclosure is represented by pure capacitor (or acoustic compliance Cab).

In real life, this is unfortunately not the case. Loudspeaker enclosure – just like any other enclosed volume of air, will exhibit it's own resonant frequencies – or modes. For the purpose of increasing the accuracy of modeling of the loudspeaker system, it would be very beneficial to incorporate modal behavior of the enclosure into the standard model.

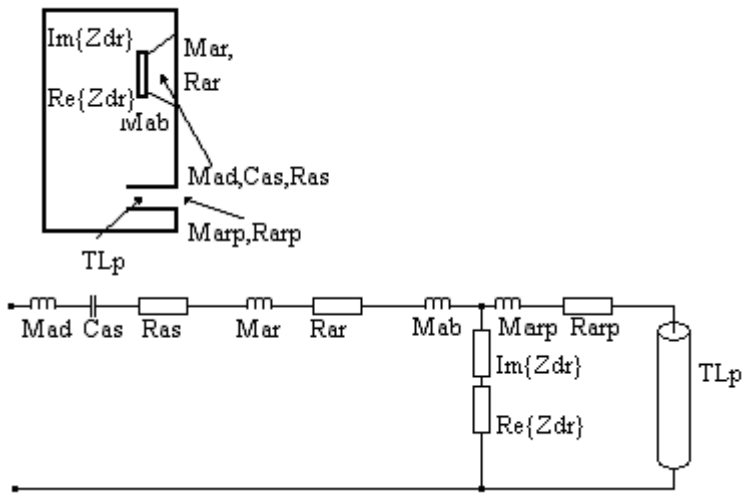


Fig 4.59 New acoustical impedance representation adopted for vented enclosure model.

The components are:

Cas = equivalent compliance volume Vas transformed to acoustical side.
Mad = mass of the vibrating system Mms transformed to acoustical side.
Ras = vibrating assembly loss Rms transformed to acoustical side.
Mar+Mab = air radiation of the front side of the diaphragm + air load of the back side of the diaphragm.
Rar = air radiation of the front side of the diaphragm.
Im{Zrd} = enclosure reactance (former compliance Vab) transformed to acoustical side.
Re{Zdr} = enclosure resistance (former absorption losses of the enclosure) transformed to acoustical side.
Marp = port radiation.
Rarp = port radiation.
TLp = short transmission line tube representing the port.

Possibly the best mathematical tool for evaluating modal behavior of fluids (air too) is **Finite Element Method.** More precisely, the **FEM** will be used to calculate driving point impedance, **Zdr**, that the driver's cone is exerting on the air in the enclosure.

First thing about FEM that one will notice, will be the long duration of these processes. Therefore, it is sensible to strive for a reasonable compromise between the accuracy of the analysis (number of nodes involved) and the length of the process. You will notice, that number of nodes in a vicinity of ~400, will give you acceptable accuracy up to 600Hz.

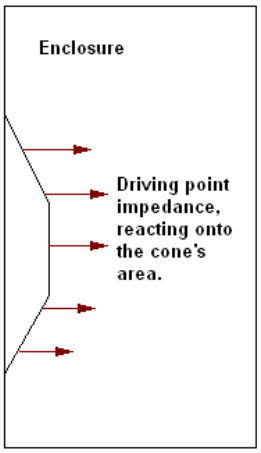


Figure 4.60 FEM allows you to replace enclosure reactance (**ω*Vb**) with **Zdr = A(ω) +jB(ω).**

Note: Before we proceed with explaining the FEM approach, please note, that the loudspeaker model implemented into SoundEasy incorporates the following:

1. Enclosure T/S analysis
2. Port resonance and non-linearity (compression)
3. Diffraction effects
4. BL(x), Cms(x) and Le(x) non-linearity
5. Enclosure modes (Zdr) analysis
6. Thermal power compression analysis

Diffraction effects and Enclosure modal analysis are position-dependent phenomena, therefore, a high level of attention must be paid to make sure, that the overall model is coherent, and all parameters are in agreement with each other. For instance, baffle location of the driver for Diffraction Analysis must correspond to the location of nodes in Enclosure Modal Analysis.

In this software release, you have a choice of using FEM with "brick elements" or FEM with Delaunay triangulation (or wedge elements) for sealed and vented enclosures. Future releases of the program will implement this method for dual-chamber enclosures as well, with brick elements for one chamber and wedge elements for the other. Wedge elements are often a better choice for modeling tilted enclosures, or enclosures with sharp edges.

A brief over view of the process is given first.

To implement either the brick or Delaunay method the following windows must be accessed: First, the driver file must be opened or conventional T/S data must be entered in the Driver Editor screen. Then the Preferences screen should be opened to set the frequency range for the Box and FEM analysis. Then, you must set-up the enclosure parameters in the FEM system. Now, go back to the Enclosure Design system and select the desired FEM option and enable the port resonance, if it is a ported box is to be computed. Then you must select the correct enclosure type, sealed or vented, and be sure System SPL is checked. Only one enclosure type should be checked. Then enter the desired Qb and port properties for a vented box and click plot. The process takes a long time. If your computer has minimum memory and/or speed the screen may appear to lock up. However, allow the code to run and when completed the screen will refresh correctly. After the frequency response has been plotted you can change Qb and/or port properties and produce a second frequency response plot without returning to the FEM screens. If you which to compare the result to the conventional response plot open the Vent/PR Calculator screen and uncheck the "Use FEM for Enclosure Modes" check box. Be sure that the correct box volume is entered in the Enclosure Design Dialogue and then click plot. First, we are going to take a look at the FEM with brick elements.

Enclosure Modes in FEM Brick-Element Model

Here is a simple description of how to use the built-in enclosure model.

1. Load a test driver file. The example driver will use 170Lt box volume and Rear Box Qb=20.

2. Go to **Preferences** screen and select frequency range for Box – type of screens.
   For example 1Hz – 1000Hz. This will set the frequency range of the FEM analysis.

3. Open **Room/Car 3D** FEM system – FEM Process Control TAB box will appear..

4. Select "**Fr/Response**" TAB and click on "**My Enclosure**" check box. This will ensure two things: (1) the wire mesh will display nodal numbers on the front panel, so you can easily select which nodes will represent driver's cone area, and (2) the enclosure modeling process will use brick element FEM representation of the enclosure.

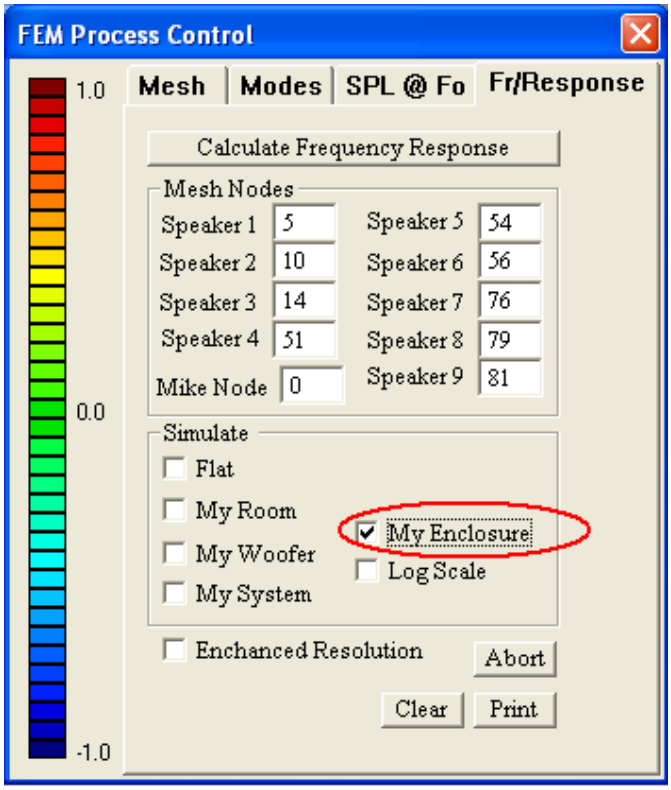


Fig 4.61 FEM brick element control box

5. Go to "**Mesh**" TAB and select "**Loudspeaker 275 nodes**" built-in option from "**Load Models**" list box. This is done for your convenience at this stage. Normally, you would built your enclosure from "brick elements", and then allocate nodes mimicking driver location. You can have 4 OR 9-nodes ONLY to represent the driver.

This model is a simple 173Lt rectangular enclosure. The size of the brick element is X=12cm, Y=12cm and Z=7.5cm. There is 160 elements in this model and 275 nodes. The footprint of the enclosure is 4 x 4 elements, and there are 10 layers vertically.

This model will suffice till about 500Hz, but will run faster than more complicated model. If you need more accurate analysis at higher frequencies, you will need to built a more dense model. In this case, the footprint may be as big as 6 x 6 elements or similar.

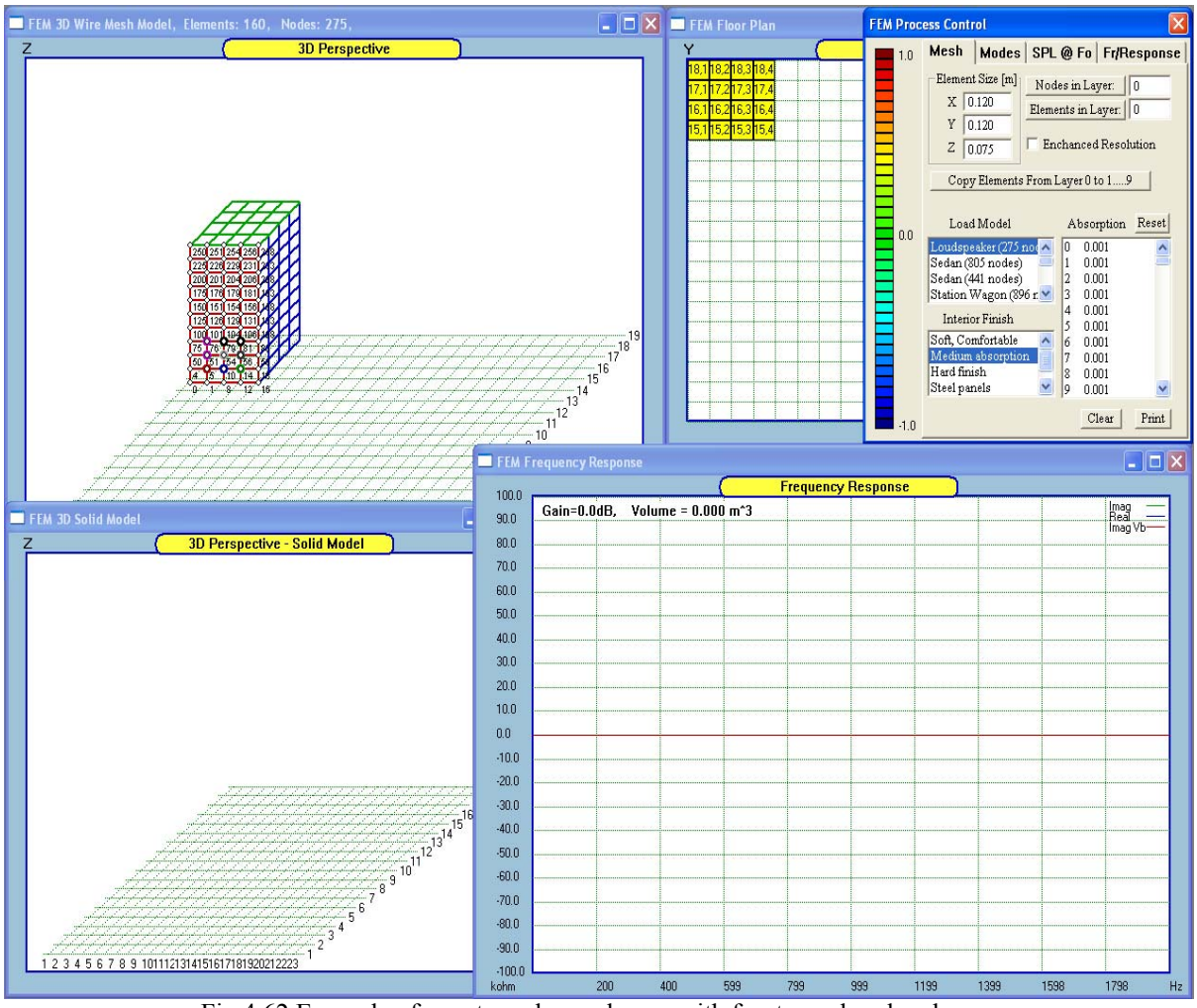


Fig 4.62 Example of a rectangular enclosure with front panel nodes shown.

6. Go back to "**Fr/Response**" TAB and enter node numbers you wish to use. For the built-in model, all nodes are already entered for you. Also, box volume will be calculated, based on brick element size. As you can see, the example enclosure volume is 0.173m^3 = 173Lt. This is close enough.

   If you need to change the enclosure volume or dimensions, at this stage, just go back to "**Mesh**" TAB and enter slightly different Element Size (X, Y or Z). When you go back to "**Fr/Response**" TAB, you will see the new enclosure volume re-calculated.

7. **IMPORTANT**: "**My Enclosure**" check box will enable FEM brick Element model during enclosure frequency response calculation.

8. In the next step, go to "**Enclosure Design Control**" and select on the "**Enclosure Parameters**" TAB:

   "**Sealed**" enclosure,
   "**Standard**" check box,
   "**Rear Box Qb**" – enter desired value
   "**FEM For Box Modes**" check box

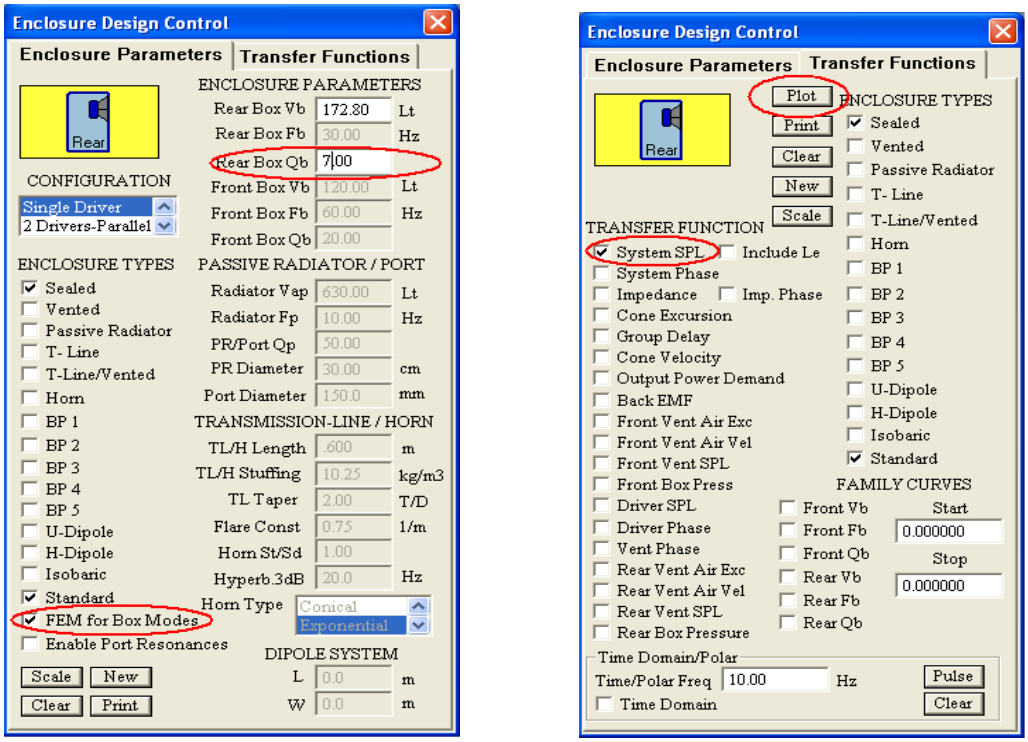


Fig 4.63 Enclosure Design box with marked controls for FEM analysis

9. Now select "**Transfer Functions**" TAB and check the "**System SPL**" box only. Then press "**Plot**" button. You will notice a progress bar will pop-up and indicate status. The process is LONG.

You will notice, that the enclosure volume calculated by the FEM screen will be transferred automatically to the driver's file.

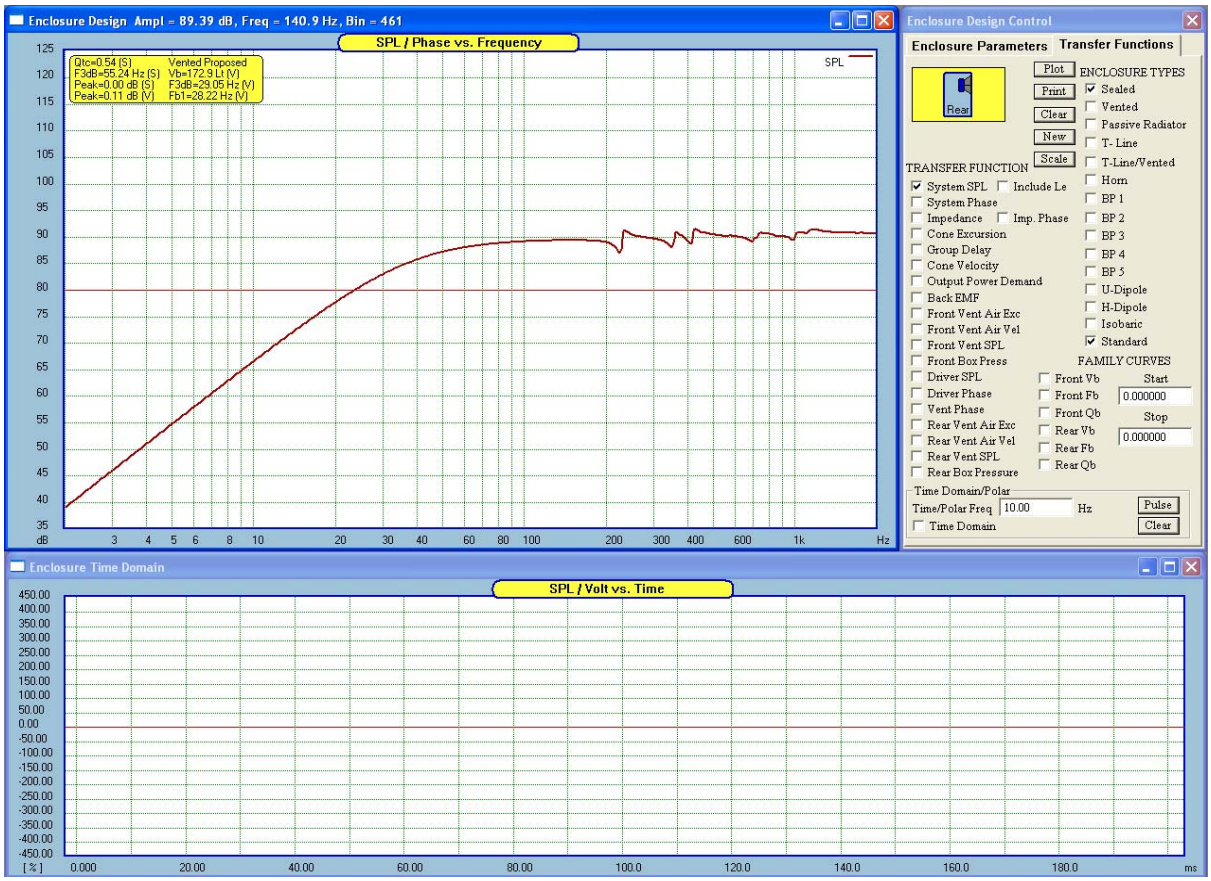


Fig 4.64 Example of a sealed enclosure SPL with FEM modal analysis

10. Finally, you can go back to the FEM system, and on the "**Fr/Response**" TAB press "**Calculate Frequency Response**" button to calculate the driving point impedance of the enclosure. Picture below was plotted for Qb=20. Simulation of "**My Enclosure**" on the FEM system predicts driver's driving point acoustical impedance, Zdr, looking into the enclosure. The Q-factor (Qb) is selected from the Enclosure Design Control box.

Blue plot = Real part of Zdr.
Green plot = Imaginary part of Zdr (reactance).
Red plot = w*Cab, reactance, if the box was pure compliance (as in the standard T/S model).

It is easily observable, that Zdr deviates from the T/S model well below the first modal frequency of the enclosure.

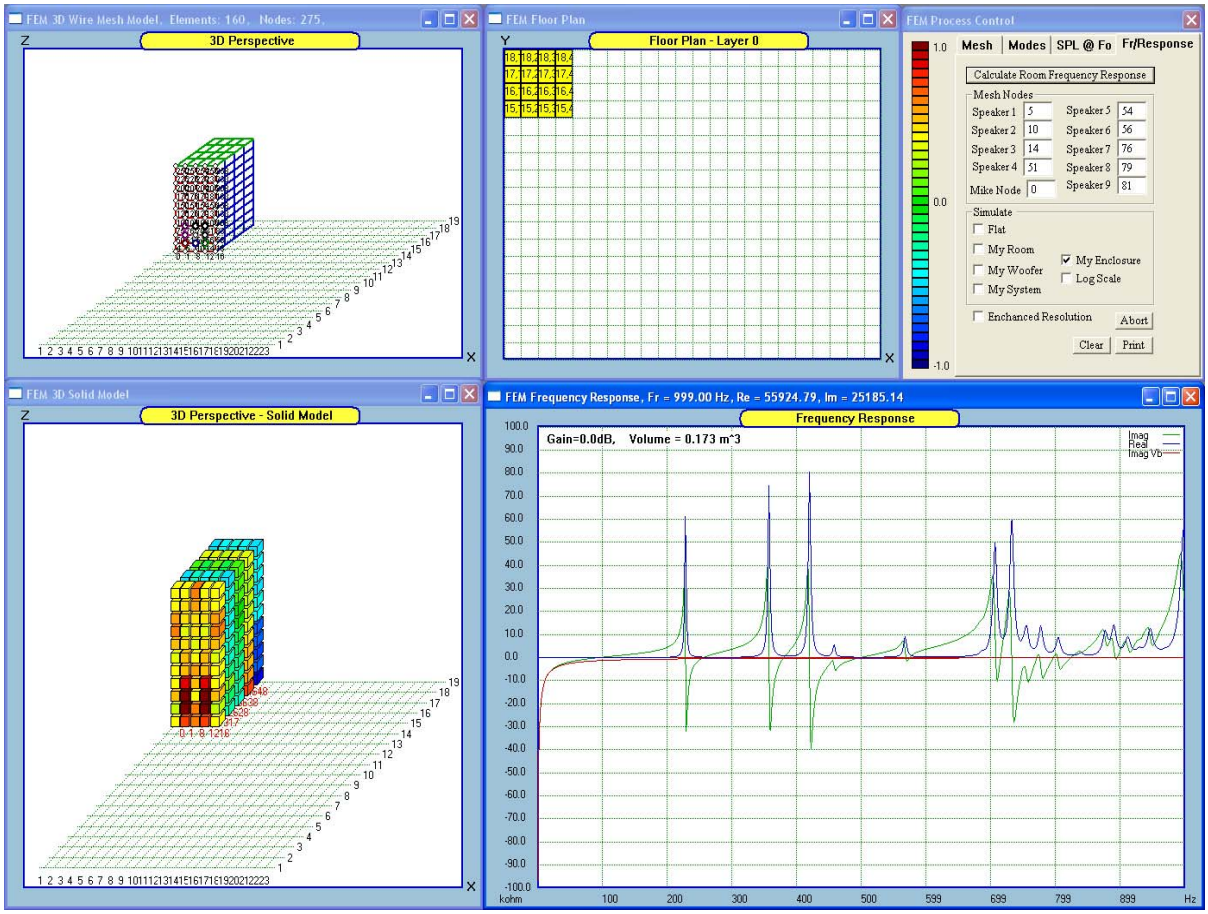


Fig 4.65 Driving point impedance of a simple rectangular enclosure

There is a need to locate the driver on the front baffle for diffraction effects and select corresponding nodes in the FEM box model. Also, the size and shape of the front baffle, used for diffraction analysis needs to be the same as the front baffle resulting from element dimensions and used for Fem modal analysis. Currently, it is user's responsibility to make sure, driver location and node locations fit together.

The built-in enclosure is actually very useful. You can adjust the volume and shape of the enclosure by selecting different element X, Y, and Z dimensions. Certainly, if more extended frequency range is desired, or perhaps mode accurate determination of enclosure modes is necessary, then you will need to construct more accurate (finer) mesh. As a consequence, you will experience longer calculation times.

Constructing the mesh from "brick" and "wedge" elements, isn't complicated, is given in Chapter 15, therefore will not be introduced here.

Example of an SPL plot for a sealed enclosure, for Qb = 7 is shown below. During the run of the FEM enclosure modal analysis, you will notice, that the plotting window title bar shows progressive calculations of Zdr, and is indicative, that this rather complex process is running properly.

Important Note: The FEM analysis involves long and complex mathematical processes. In order to speed-up calculation time, some sections of the program were written in assembler (machine code) language. Therefore, when running the FEM analysis, the CPU usage will be 100%. We strongly suggest, that you let the process run to completion, without attempting to use your PC for anything else.

Port Resonance and Enclosure Modes

The SPL plot below, were generated for the same enclosure (Vb=170Lt, Qb=7, Fb=30Hz) , the only difference being – the second picture has port diameter set to 150mm (instead of 100mm, as on the first picture).

It is observable, that resonating port, interacts significantly with self-resonating enclosure, and produce a substantial level of SPL. Since the port subtracts from the cone's output, the higher the level from the port, the more irregular the system SPL becomes.

Secondly, larger ports will be much longer, therefore, the fundamental resonant frequency of the port, will easily fall into the audio range designated for the woofer driver – causing undesirable SPL variations. This is quite unfortunate, since it is desirable, to have as large port as possible, in order to minimize undesirable port compression and non-linearity at higher port output. It seems, that there are conflicting requirements for the port dimensions. Therefore, it may be beneficial to evaluate vented woofer systems not only for the "piston" range of the driver, but also taking into account resonant modes of the port and the enclosure.

For instance, Figure 4.66 below clearly shows, that enclosure reactance become positive above ~96Hz (green curve), and the resistive part of Zdr is not a flat (enclosure loss, or pure resistor in T/S model) curve either. However the standard "lumped element" model obviously continues to behave as the capacitive reactance (red curve). This discrepancy would suggest, that the system designer needs to be aware of the possible weaknesses, when working with the standard "lumped element" model.

Guided by the plots below, one could draw the conclusion, that larger ports will require lower cut-off frequencies of their LP system filters, so that the driver is not forced into the frequency range, where the port-enclosure interaction begins to affect the smoothness of the SPL.

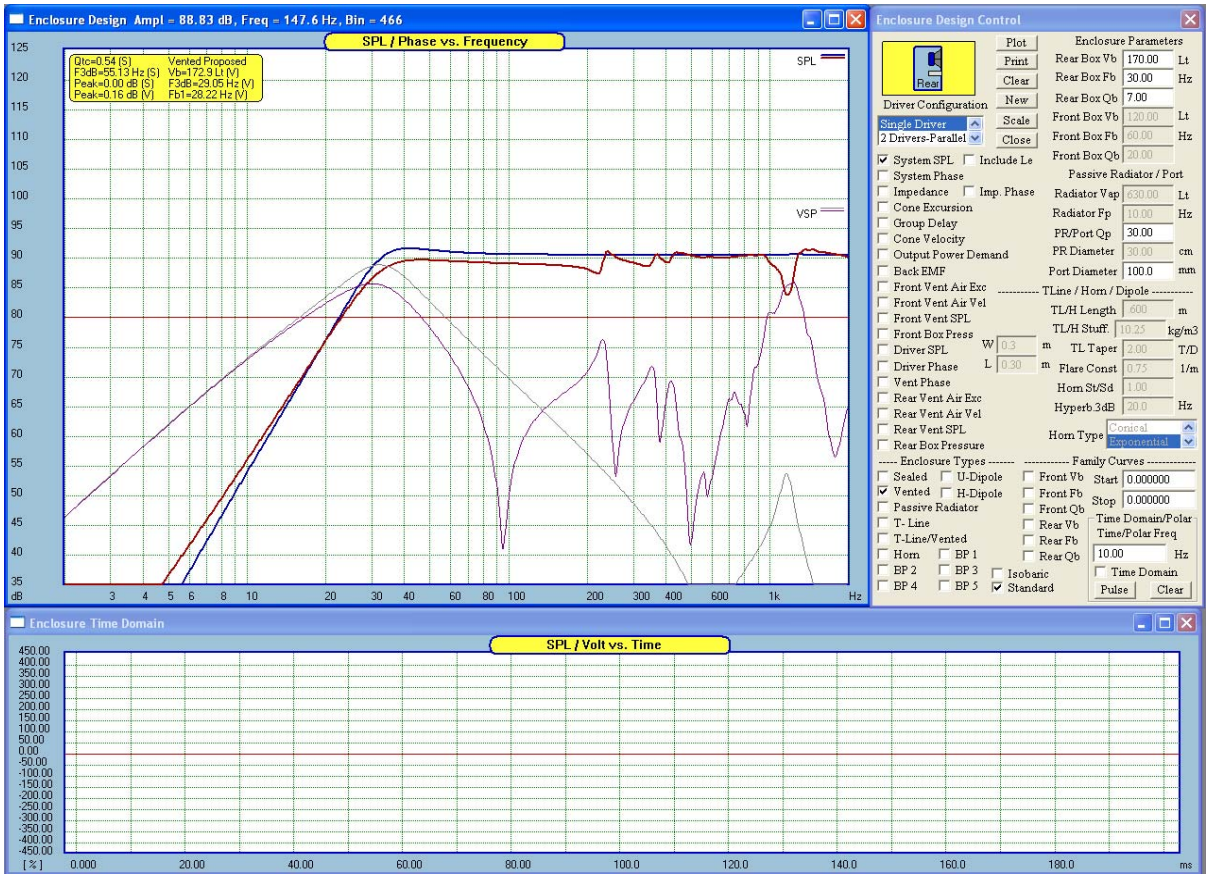


Fig 4.66 Vented enclosure SPL with Qb = 7 and port diameter = 100mm.

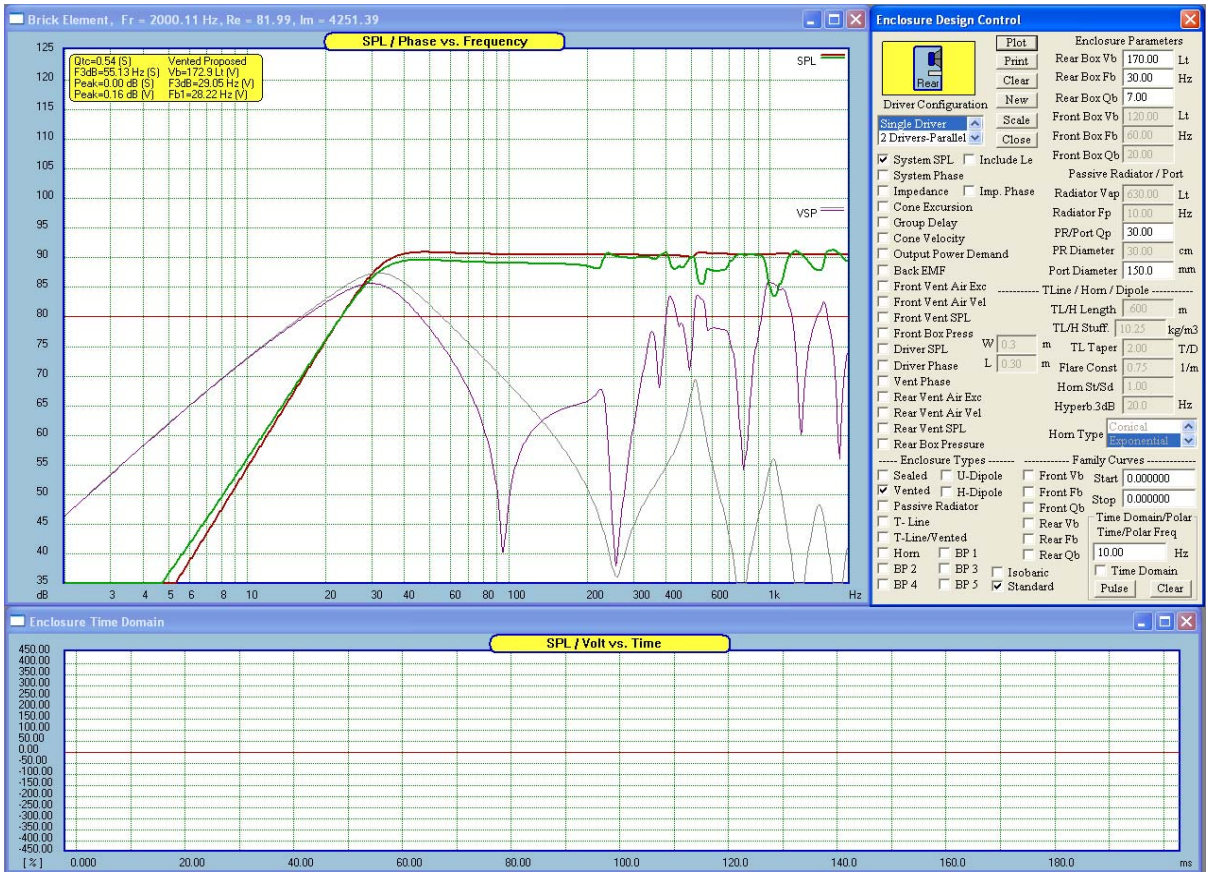


Fig 4.67 Vented enclosure SPL with Qb = 7 and port diameter = 150mm.

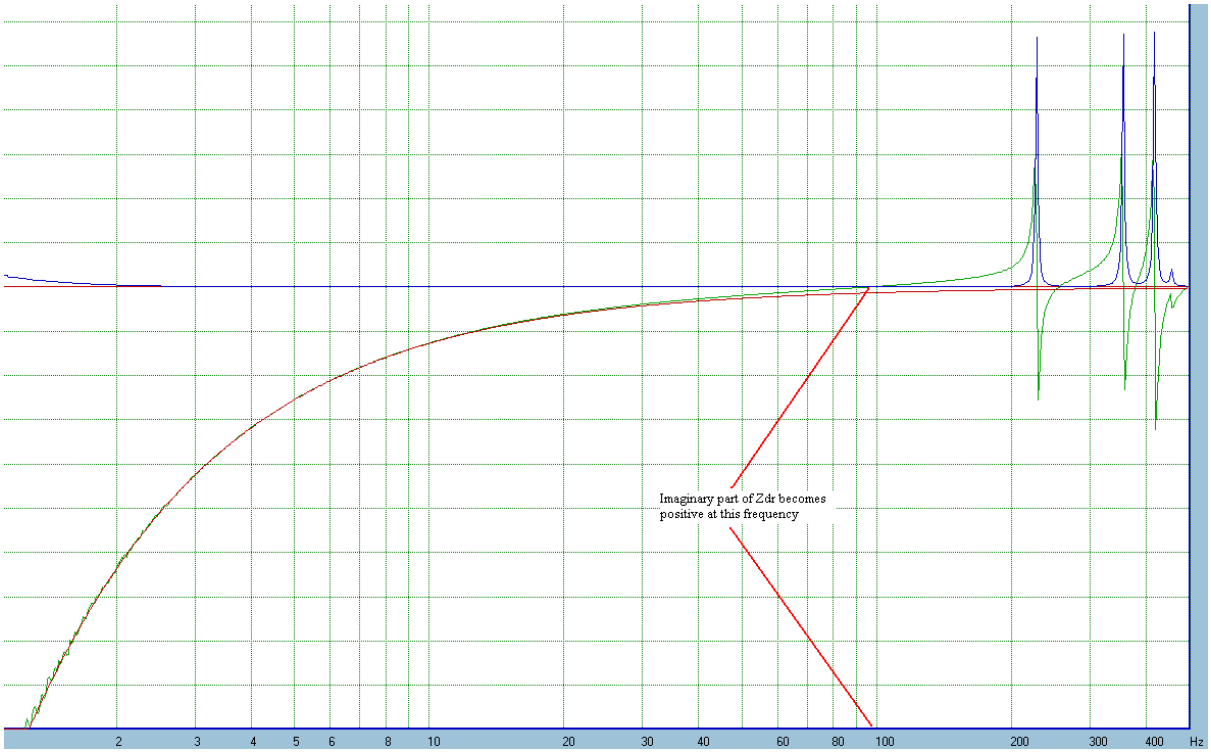


Fig 4.68 Transition frequency above which the “lumped element” model deviates from FEM enclosure model.

Enclosure Modes in FEM Wedge-Element Model

Here is another simple description of using built-in enclosure model. This time, it is "wedge element" based FEM, and we will only calculate the Zdr.

1. Load a test driver file. Our example driver will use 170Lt box volume and Rear Box Qb=20.
2. Go to **Preferences** screen and select frequency range for Box – type of screens.
   For example 1Hz – 1000Hz. This will set the frequency range of the FEM analysis.
3. Open **Room/House DEL 3D** FEM system.

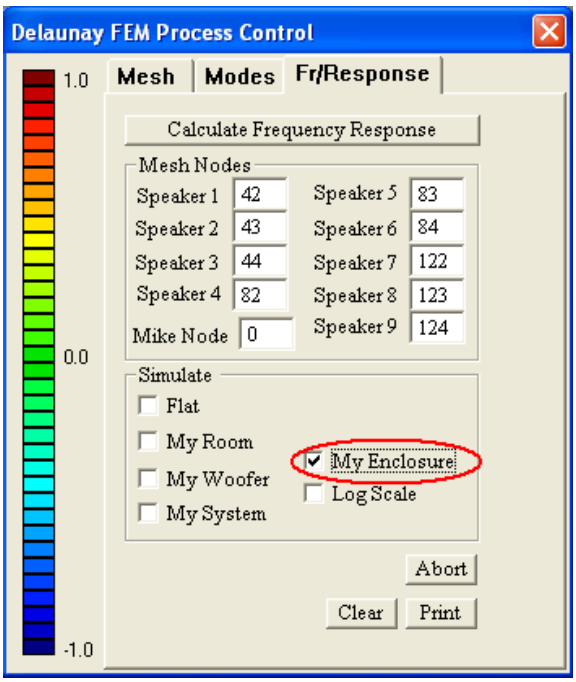


Fig 4.61 "Wedge" element control box

Select "**Fr/Response**" TAB and click on "**My Enclosure**" check box. This will ensure two things: (1) the wire mesh will display nodal numbers, so you can easily select which nodes will represent driver's cone area, and (2) the enclosure modeling process will use wedge element FEM representation of the enclosure.

4. Select "**Mesh**" TAB.

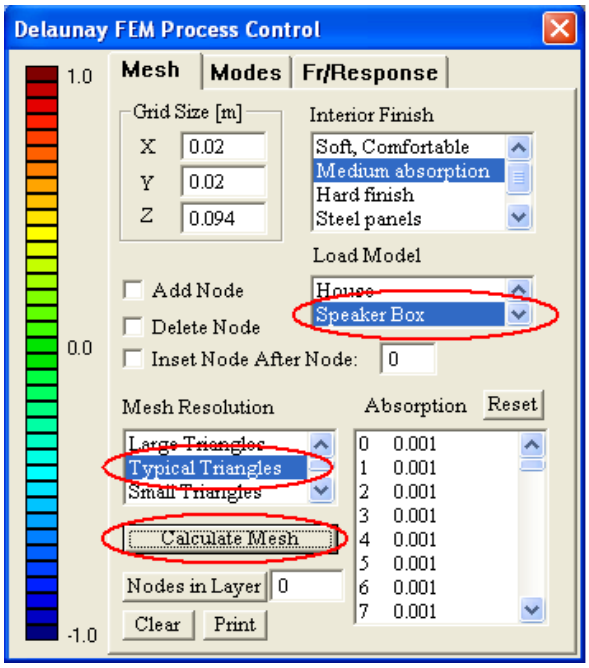


Fig 4.62 "Wedge" element control box – Mesh TAB with marked controls

Select: "**Speaker Box**" – to use built-in model
"**Typical Triangles**" – to obtain about 360 Nodes in FEM mesh.
"**Calculate Mesh**" – button to generate mesh.

Following, you should now see something like the picture below. One can review Nodes layer-by-layer to determine which nodes should be used to mimic the loudspeaker's cone.

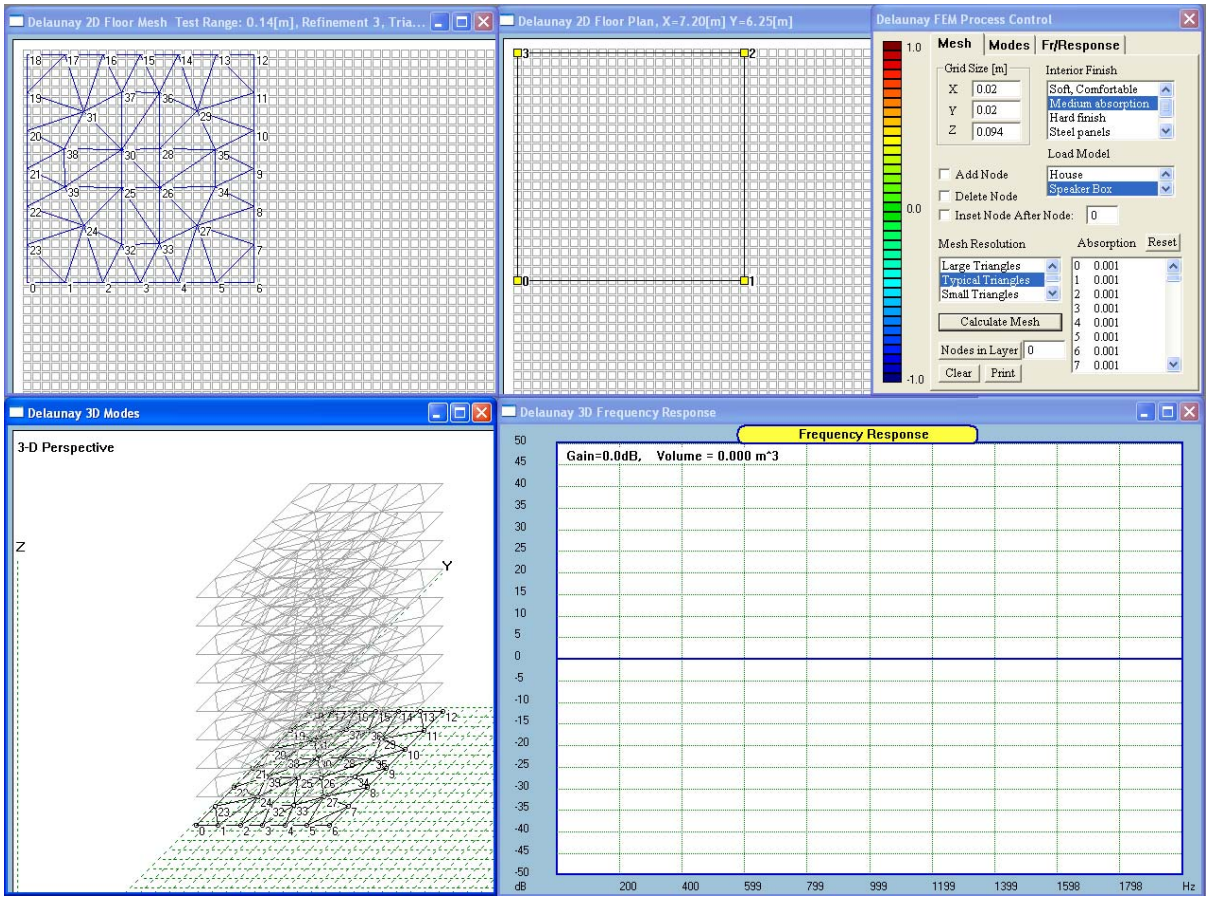


Fig 4.63 Example of "Speaker Enclosure" with node numbers shown in Layer 0.

5. Go back to "**Fr/Response**" TAB and enter node numbers you wish to use. For the built-in model, all nodes are already entered for you. Also, box volume will be calculated, based on brick element size. As you can see, the example enclosure volume is 0.173m^3 = 173Lt. This is close enough.
   If you need to change the enclosure volume or dimensions, at this stage, just go back to "**Mesh**" TAB and enter slightly different Element Size (X, Y or Z). When you go back to "**Fr/Response**" TAB, you will see the new enclosure volume re-calculated.

6. **IMPORTANT**: "**My Enclosure**" check box will enable FEM wedge Element model during enclosure frequency response calculation.

7. For now, press "**Calculate Frequency Response**" button to calculate the driving point impedance of the enclosure. Picture below was plotted for Qb=20. Simulation of "**My Enclosure**" on the FEM system predicts driver's driving point acoustical impedance, Zdr, looking into the enclosure. The Q-factor (Qb) is selected from the Enclosure Design Control box.

Blue plot = Real part of Zdr.
Green plot = Imaginary part of Zdr (reactance).
Red plot = w*Cab, reactance, if the box was pure compliance (as in the standard T/S model). It is easily observable, that Zdr deviates from the T/S model well below the first modal frequency of the enclosure.

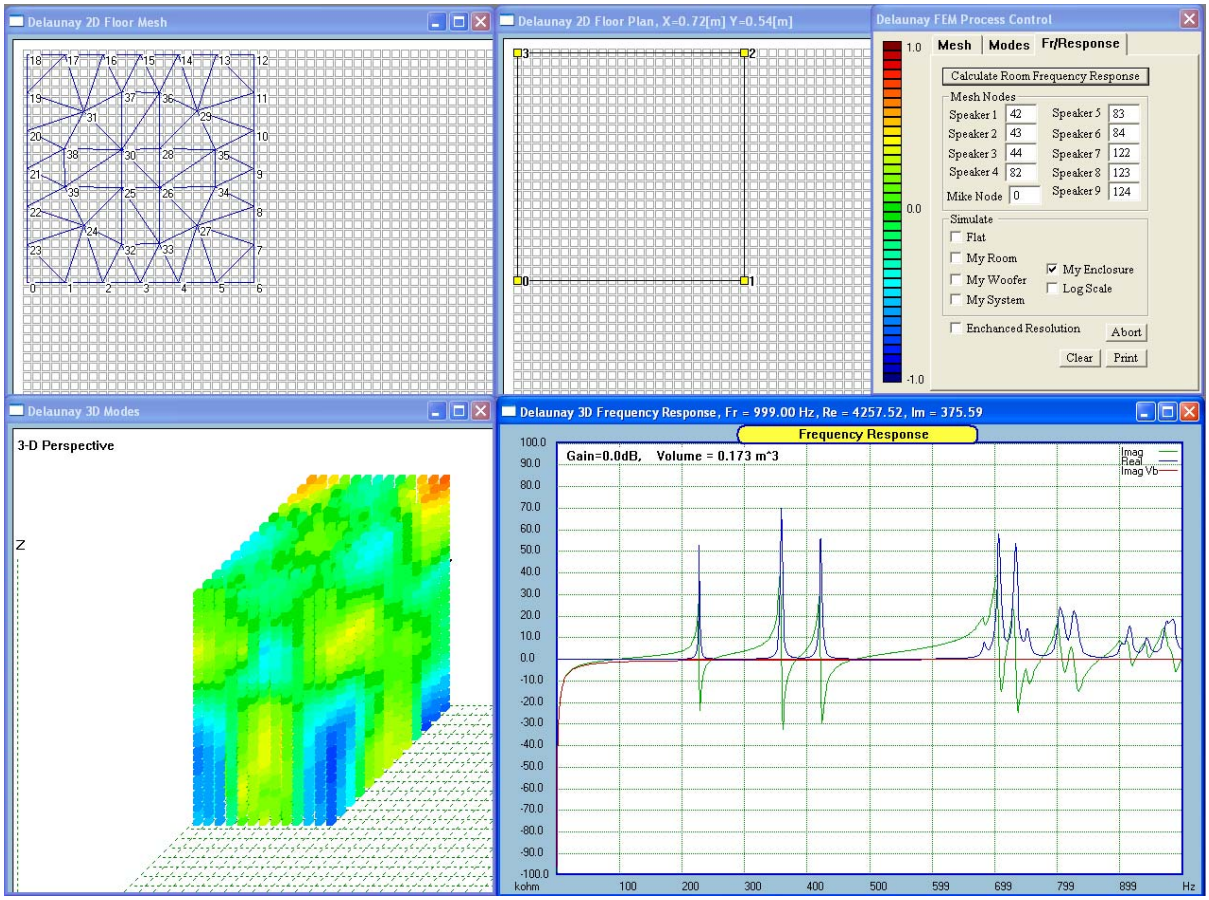


Fig 4.64 Driving point impedance of a simple rectangular enclosure

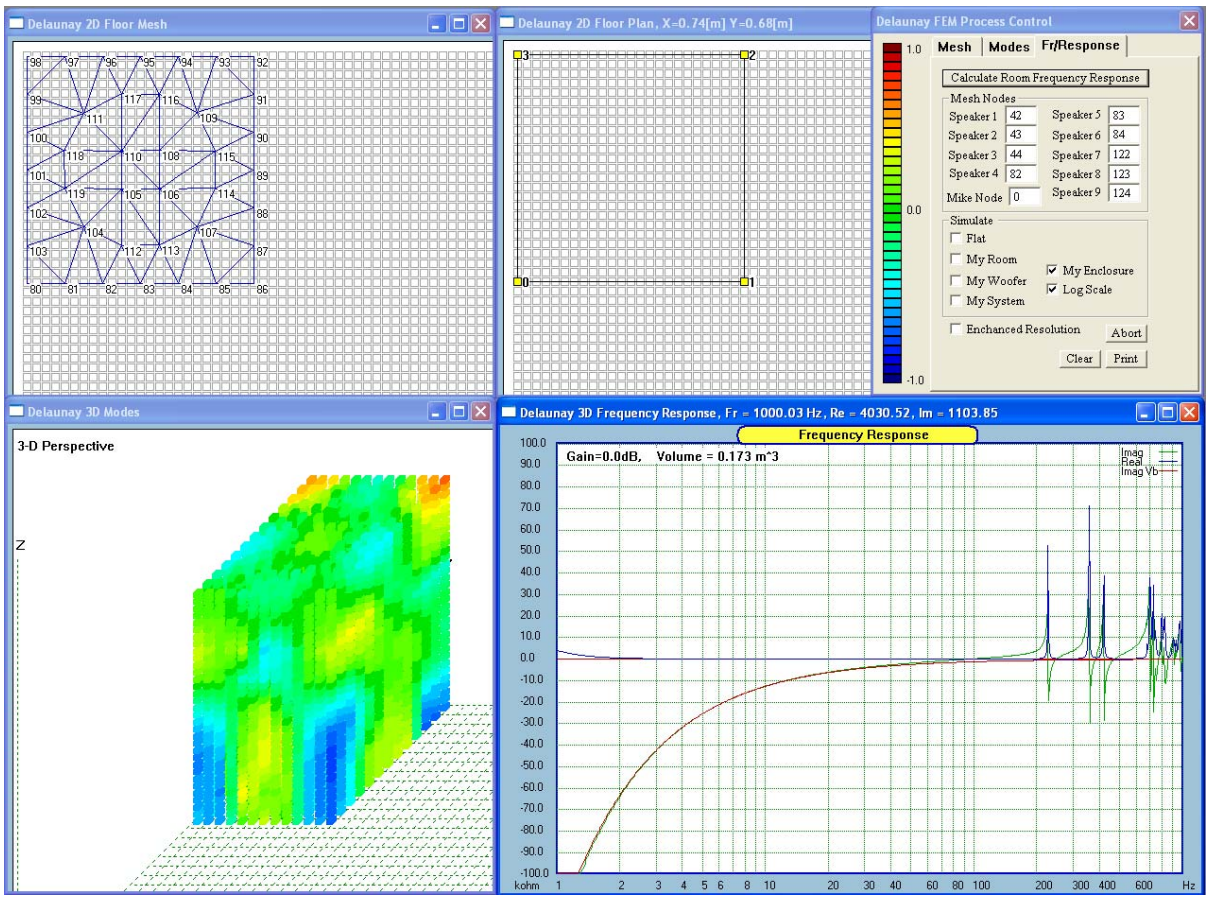


Fig 4.65 Driving point impedance of a simple rectangular enclosure – LOG scale.

Chapter 5. Enclosure Optimization Function

Background

The synthesis of the enclosure parameters using "family curves" technique was given in Chapter 4. However, it is always recommended to confirm the results of the design by applying an optimization process to the enclosure parameters found in Chapter 4. During the optimization, parameters of the enclosure are varied accordingly to the method's algorithm and the error function is calculated for every set of parameters. The error is considered to be any difference between a reference target frequency response function and the frequency response calculated for the current set of enclosure parameters. There are a number of optimization methods available for applications such as minimization of the error function. The method adopted in this program was first described by Rosenbrock in 1960. The method lends itself to a computer application. For a detailed description of the method the user is referred to [18]. The algorithm attempts to minimize the error by adjusting the parameters of the enclosure.

There are six target reference functions for a sealed enclosure:

1. Chebyshev, 2-nd order, 0.5 dB ripple,
2. Chebyshev, 2-nd order, 1.0 dB ripple,
3. Chebyshev, 2-nd order, 2.0 dB ripple,
4. Chebyshev, 2-nd order, 3.0 dB ripple,
5. Bessel, 2-nd order and
6. Butterworth, 2-nd order.

There are also 13 target reference functions for a vented enclosure:

1. Chebyshev, 3-rd order, 0.5 dB ripple,
2. Chebyshev, 3-rd order, 1.0 dB ripple,
3. Chebyshev, 3-rd order, 2.0 dB ripple,
4. Chebyshev, 3-rd order, 3.0 dB ripple,
5. Chebyshev, 4-th order, 0.5 dB ripple,
6. Chebyshev, 4-th order, 1.0 dB ripple,
7. Chebyshev, 4-th order, 2.0 dB ripple,
8. Chebyshev, 4-th order, 3.0 dB ripple,
9. Bessel, 3-rd order,
10. Bessel, 4-th order,
11. Butterworth, 3-rd order.
12. Butterworth, 4-th order and
13. Quasi-Butterworth (QB3) 3-th order.

And finally, there are 3 types of target alignments for Band-Pass Type 2 enclosure:

1. Narrow (Chebyshev type)
2. Medium (Chebyshev type)
3. Wide (Butterworth type)

The target response is selected from a dialog box invoked by pressing "Optimize Sealed", "Optimize Vented" or “Optimize BP2” buttons. The dialog box will also offer some pre-calculated parameters of the enclosure being optimized (or default values, if no driver file has been loaded). The proposed values for the 3-dB cut-off frequency, enclosure tuning (vented) and enclosure volume, were calculated for a given driver, but user may edit any of them. The "Reference" button on the dialog box should be used for plotting the reference curve only.

Optimization Strategy

Generally, optimization is a "curve fitting" process, which attempts to obtain minimum error between the amplitude response of the selected enclosure and required target function. Should the 3-dB cut-off frequency of the target response be selected badly, the global error at the end of the optimization process will be large. The proposed values presented to the user in the dialog should be considered as a good starting point for optimization. However, the reference curve should still be moved around on the frequency axis to check if a better fit can be achieved for a given driver. This is exactly what the Optimization tool offers.

The algorithm will perform 30 optimizations using 30 target responses with different 3-dB cut-off frequencies. The cut-off frequencies will be selected as 80 to 180% of the "Proposed F[3dB]" parameter editable from the dialog box eg: the first target reference curve will have its 3-dB cut-off frequency calculated as ("Proposed F[3dB]) * 0.8, the second curve: ("Proposed F[3dB]) * 0.9, and so on till the 10-th curve, having the 3-dB frequency equal to ("Proposed F[3dB]) * 1.8. The emphasis on the higher cut-off frequencies are advocated, as it generally leads to smaller enclosures. Selecting the "Proposed F[3dB]" too low, may become a source of significant errors. When the optimization process is completed, the **lowest Global Error** is highlighted by the program and the enclosure parameters (Qb, Vb and Fb, in case of a vented enclosures) are transferred to the driver's data file space.

Enclosure Optimization

The optimization process uses the data provided by the enclosure design screen. The "Enclosure Optimization" screen allows for several parameters to be edited - depending on the enclosure type. Generally, drivers with a Qt-factor greater then 0.5, should be used in sealed boxes or for Chebyshev alignments. Drivers with the Qt as high as 0.9-1.0 may also be tested with Chebyshev 4th order, 3dB ripple alignments. The optimization routines will find it difficult to produce the "optimally flat alignment" (Butterworth 4th order) with this type of driver in a vented enclosure. You may find, that the final Q-factor of the enclosure (Qb) has to be as low as 2.0 in order to reduce the humps in the frequency response. The efficiency of the system will be reduced as well. To start the optimization process, you needs to select the desired tab (click on the "Sealed", "Vented" or "BP2 TABs), depending on the designed enclosure and then select the desired action by pressing one of the three dialog box buttons. Optimization process is invoked by pressing the "**Optimize"** button and after a few moments, a message window will indicate the process is completed. **"Abort**" button simply aborts the process and **"Reference**" button plots the reference enclosure curve. Optimized values for the example vented enclosure are shown on Fig 5.1:

Vb = 175 liters Fb = 25.3 Hz Qb = 74.4 F3db = 25.4 Hz.

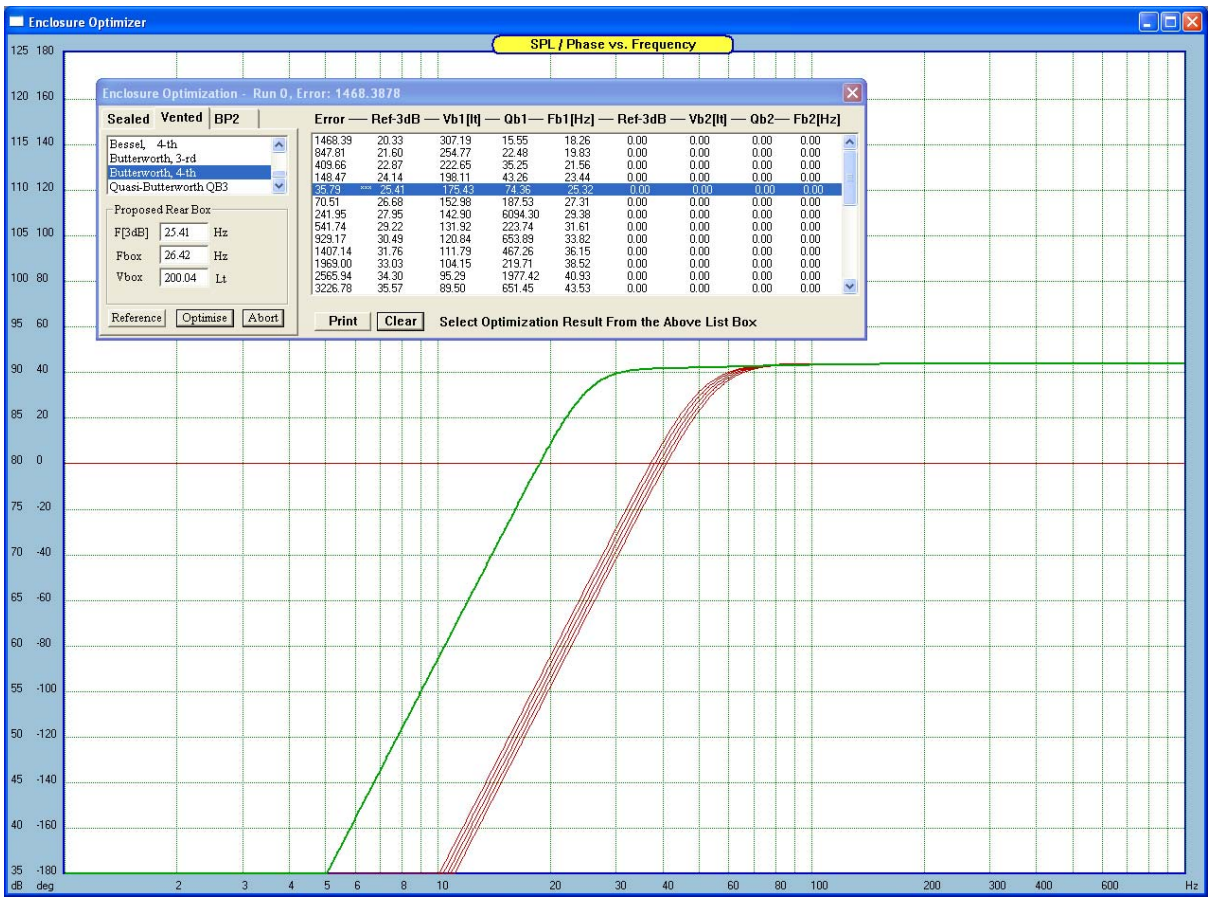


Fig 5.1 Optimized vented enclosure parameters

You can monitor the progress of the optimization process. Two parameters, relating to the progress are displayed: Current Global Error and Number of Trials. Examining the results one can observe, that the enclosure designed using the "family curve" approach has very similar parameters to the optimized box. The user should also be aware that the error function the program is trying to minimize, may have one or more local minimum. It may happen, that the algorithm will never find a global minimum, so it is recommended to monitor the value of the global error displayed on the screen. Should the error increase steadily, the process should be aborted by clicking on the "Abort" button.

The discussion here was limited to a woofer installed in a vented enclosure. However, the user would follow the same steps designing either sealed or vented boxes for a midrange or upper bass driver. The "Erase" button clears the plotting area. The complete discussion on the numerical optimization methods is beyond the scope of this manual. For the more difficult drivers, (driver with a Qt> 0.5 installed in vented enclosure) the user should try a few different starting set of parameters. The algorithm may be able to find a lower F3db cut-off frequency and flatter frequency response for a higher allowed Qb.

The user is referred to [12], [18], [19], [20] and [21]. Fig 5.2 shows a woofer driver (test301.wfr) optimized for a sealed enclosure with Qb=10.0. Driver's low Qt forced the resulting small enclosure. Fig 5.3 shows the same woofer optimized for a vented enclosure, but for Chebyshev 4-th, 0.5 dB ripple. Please note quite large box and NO losses proposed by the algorithm. Fig 5.4 shows three examples of the reference curves for BP2 enclosure.

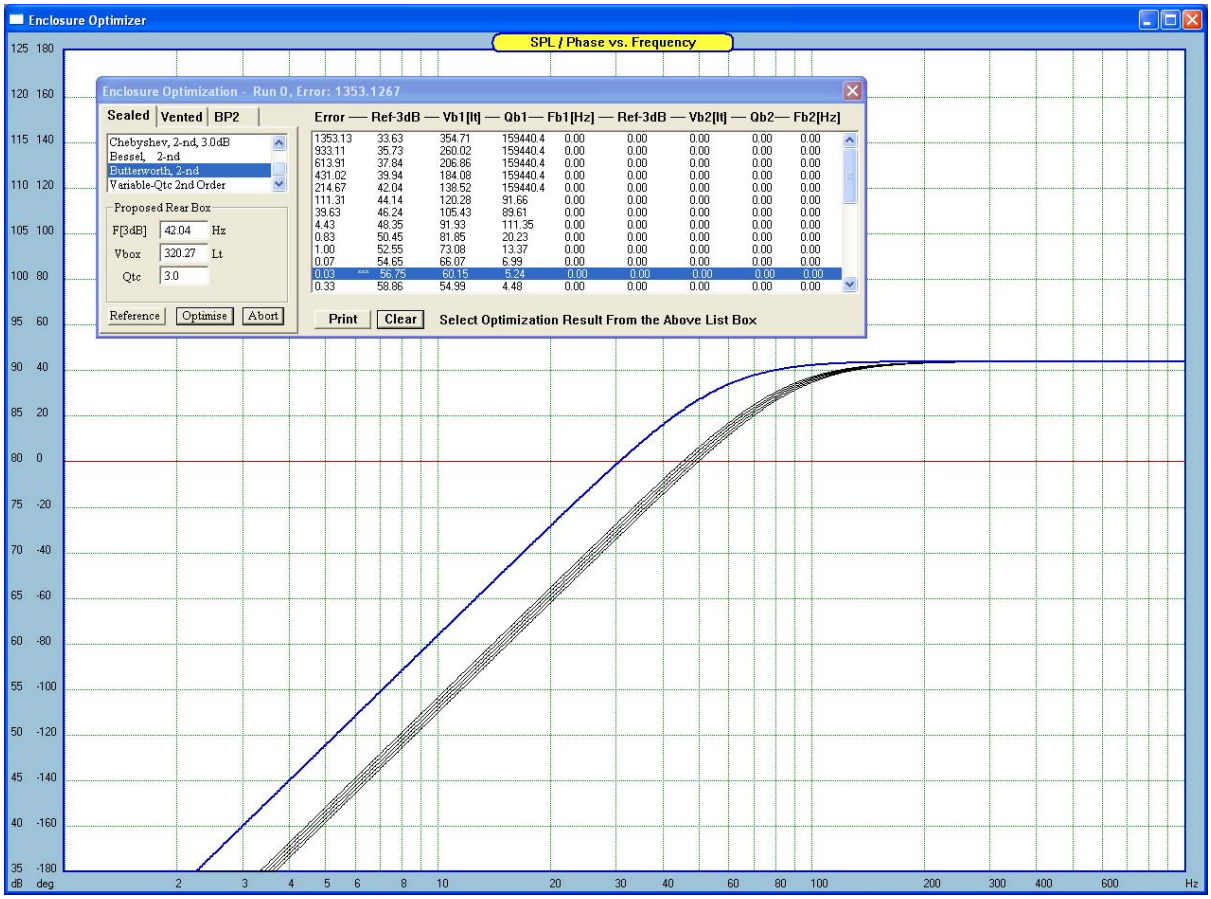


Fig 5.2 Optimized sealed enclosure parameters

As the optimization process steps-through the alignments, the list box on the right-hand side of the box gets filled with optimization results. For a sealed enclosure the results would show:

1. Error – This is a measure of how close the current SPL curve matches the target curve. The smaller the error the better match was accomplished.
2. Ref 3dB – This is the current 3dB cut-off frequency of the target curve.
3. Vb1 – Current rear enclosure volume.
4. Qb1 – Current rear enclosure Q-factor.

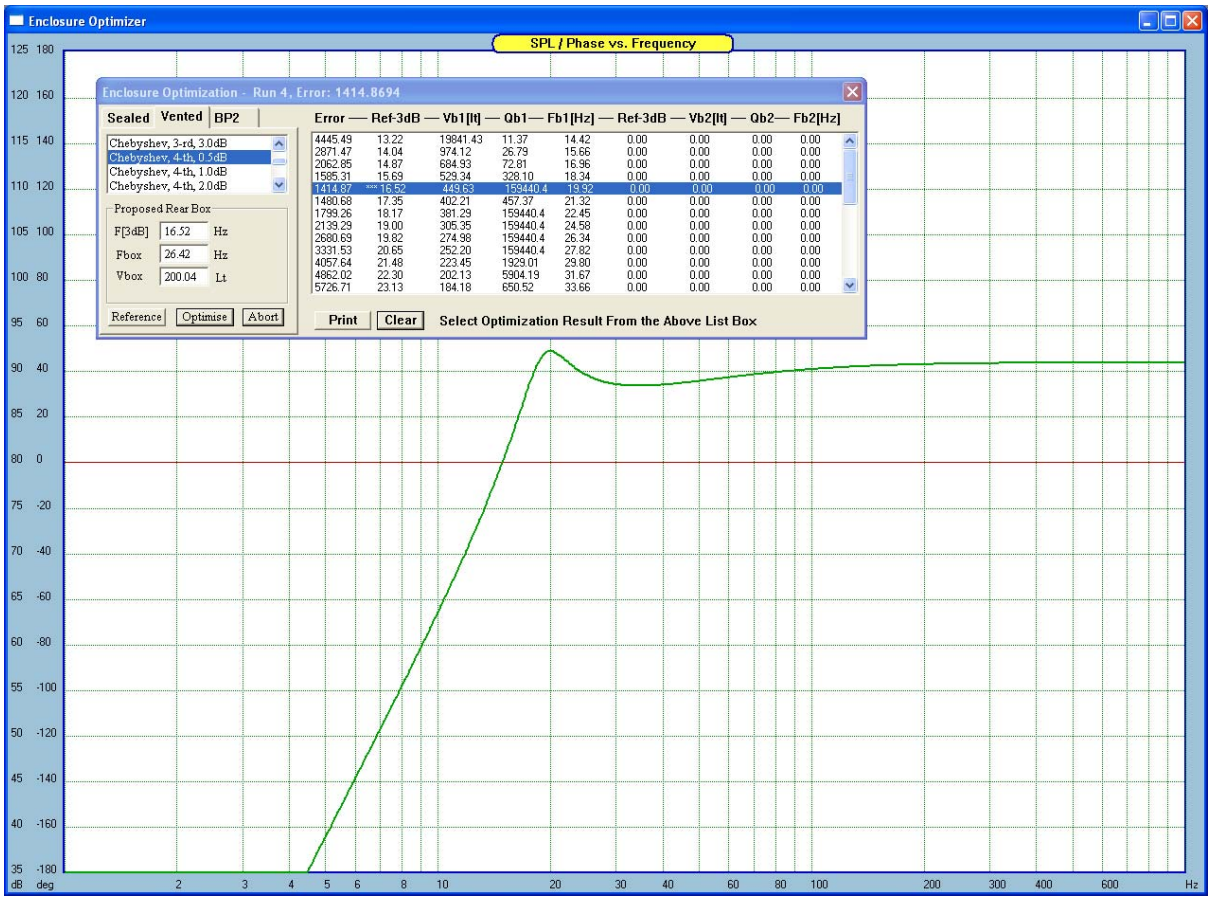


Fig 5.3 Optimized vented enclosure parameters for Chebyshev, 0.5dB ripple

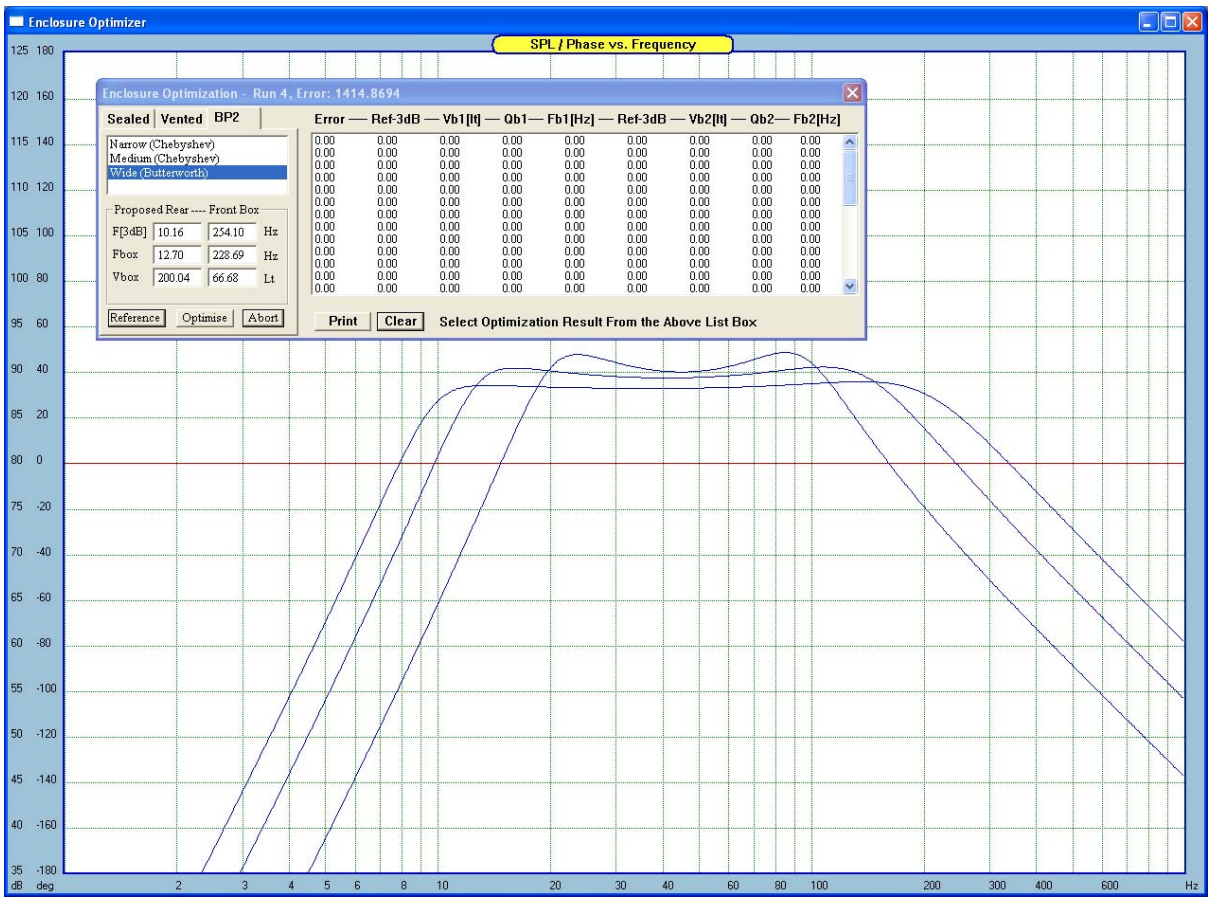


Fig 5.4 Example of the target curves for BP2 enclosure

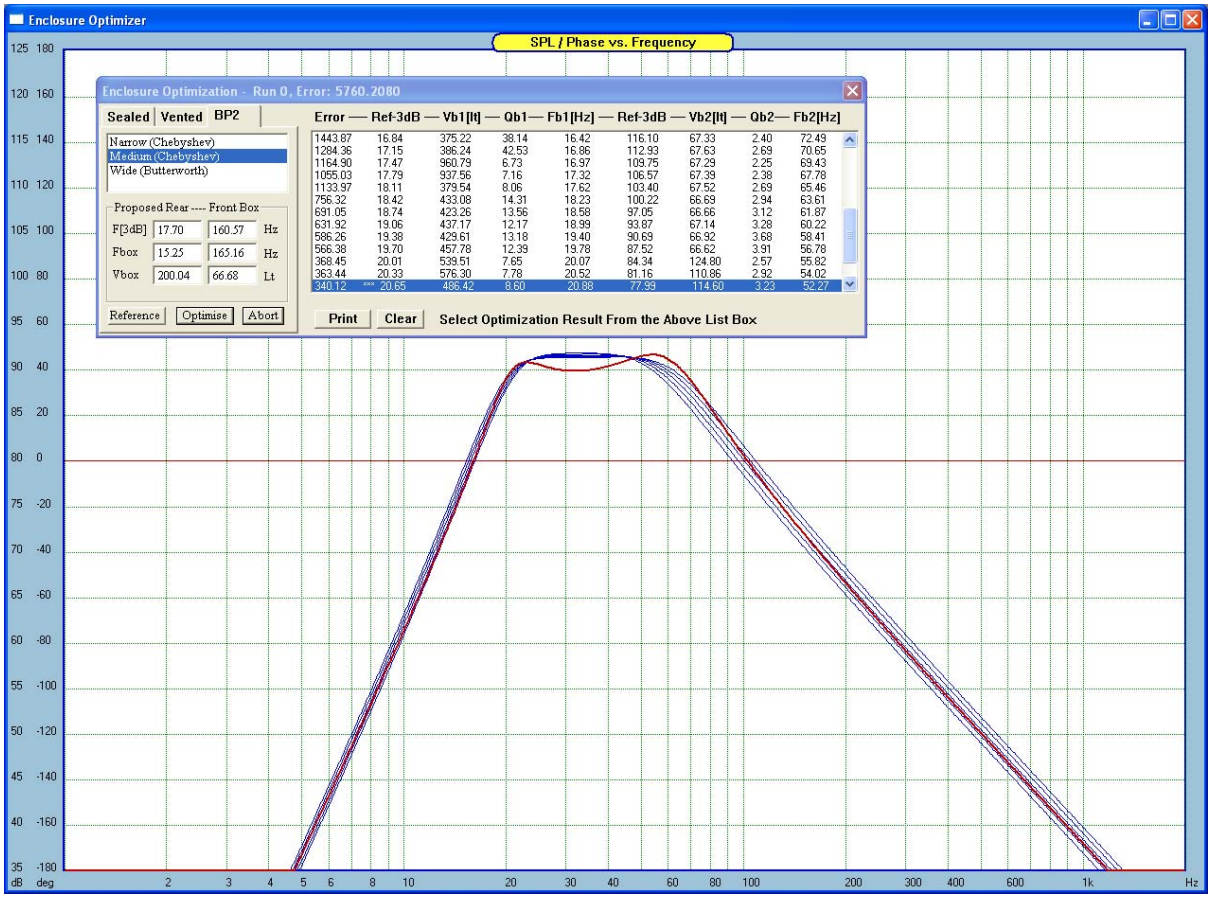


Fig 5.5 Example of BP2 optimization

Sealed enclosure can also be defined in terms of the total, or system Q-factor, Qtc. Should you decide to optimize your sealed enclosure for a particular Q-factor, you need to enter its value in the Qtc data field on the “Sealed” TAB.

The list box can handle maximum 30 entries, so there is that many alignments for the algorithm to try. Consecutive optimization results are listed in the box for your review and resulting parameters are transferred to the driver’s data space when you click on the line showing the desired parameters. Please use scrolling tab to move other results into view area. When you click on any of the lines, you will also see the resulting SPL displayed on the screen.

Enclosures can be optimized for several parameters. Even the simplest, sealed box, has at least two “degrees of freedom”:

1. **System Q-factor, Qtc**. This would be a good choice if you were interested in a particular Qtc parameter and associated time-domain response of this second-order circuit. Different type of music is often said to sound better for a particular Qtc factor. For instance, orchestral or symphonic music may sound better with a sealed enclosure with Qtc = 0.7. On the other hand, rock or disco music may prefer Qtc = 1.5 at a particular frequency.

2. **Lowest F3dB.** Here you may be interested in getting out the most of your loudspeaker. It is assumed here, that the flattest SPL is also desired. This optimization would provide you with the most extended low-frequency end.

Vented enclosure optimization would typically revolve around the lowest F3dB. However QB3 target may also be desired, as it provides sensible compromise between the good impulse response of the sealed box and better bass extension of the vented enclosure.

When optimizing for the lowest 3dB parameter, the optimization algorithm moves the target's 3dB frequency starting from the low end and progressing to the high-end of the bass range. Typical result of the optimization process for a sealed and vented enclosure will result in a bunch of results that start with a high Error, then exhibit a minimum Error value then continue into high Error area again.

The low-end high Error is the result of the driver not being able to reproduce bass frequencies well below its resonant frequency. Then the Error begins to drop and reaches minimum. This is where the size of the enclosure becomes proportional to Vas of the driver. The high-end Error is the result of the driver being too compliant (Vas too high) for the requested target F3dB in this frequency range. Here, the algorithm will try to smooth the ever increasing humps on the SPL curve by increasing box damping (reducing Qb1). However a limit is reached sooner rather than later, and the Error begins to creep up again. Somewhere in the middle is you optimal alignment for the given driver. Remember Thiele's rule-of-thumb for maximally flat alignment for a vented box: Vb=Vas, Qtc=0.38, F3dB=Fs.

Chapter 6. Enclosure Response to Rectangular Function.

Background

One of the most frequently cited criticisms of vented enclosures was their tendency to add mid-bass coloration to the reproduced sound. Early enclosures, particularly those which were misaligned, gained reputation for being "boom boxes". Even today, opinions concerning the subjective transient response of the reflex systems may be influenced by the boomy character of the many poorly designed examples still in production, of which some have Qtc values greater than 1.5. The fourth-order, 24 dB/oct final roll off for a reflex system endows it with a fundamentally poorer transient response that a sealed box. This particular characteristics inspired researchers to look for intermediate alignments and resulted in less-then-optimum efficiency alignments such as QB3. Generally speaking, the specifications of a loudspeaker system do not contain any requirements concerning the transient response. It nevertheless has to be mastered in certain conditions. A great majority of the systems will respond to a step function (rapid change in the supplied input voltage) with some ringing, evident in the transient response. The transient response here is defined as acoustic pressure in front of the box versus time after the step function was applied. So, how much ringing are you prepared to tolerate ?. Literature often advocates, that speaker systems should be matched to the type of music being reproduced. The rationale follows, that for classical music Qtc should be between 0.7 and 0.9 for the sealed system. This alignment is said to provide a refined and clean bass. Contemporary music lover would tolerate much higher levels of Qtc, perhaps as high as 1-1.5 (for sealed box). Some rock music fans like higher Qtc for sealed boxes, because it accentuates the bass (Qtc = 1.6-2.0). Different alignments will respond with different amplitude and duration of ringing. SoundEasy provides the designer with a time domain tool for plotting box response to a square-wave excitation. The frequency of the square-wave can be varied from 1.0 to 500.0 Hz and in the lower range of frequencies (1-10 Hz), the box response is identical to the transient response.

How It Is Done.

Passive loudspeaker system falls into a Linear Time-Invariant (LTI) category of systems. The nature of these systems could be studied using Difference Equations or, more appropriate for computer application, Discrete Fourier Analysis. SoundEasy takes advantage of the fact, that the frequency response function of the selected enclosure has already been formulated and can be used for process called 'Convolution'.

$$Vout(j\omega) = Vin(j\omega)*H(j\omega)$$

Convolution in time is reduced to a simple multiplication of complex functions; V() and H(). The first one, Vin() is the input square-wave transferred into the frequency domain using Fast Fourier Transform (1024-point Radix-2 FFT) algorithm. The second, H() is the frequency response of the simulated enclosure. The resulting function, Vout() is then acted upon using Inverse Fast Fourier Transform (1024-point IFFT) to obtain time plot of the system response. The block diagram of the process is shown in Fig 6.1 below.
1

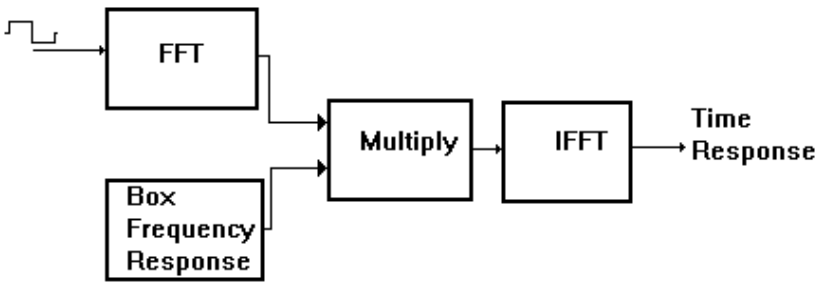


Fig 6.1

The input square-wave contains infinite number of odd harmonics. Amplitude of each harmonics is given below by the Fourier coefficient for odd and even values of 'n'.

Amplitude = 0 for n even, Amplitude = $\frac{4}{n*\pi}$ for n odd.

eg: Amplitude of F1 = 1.2732395 F2 = 0.0
F3 = 0.4244131 F4 = 0.0
F5 = 0.2546479 and so on.....

It is easy to observe, that only when all the harmonics are added with correct amplitudes and phases, the input waveform has the required rectangular shape. When the wave passes through the LTI system, the amplitude and phase of individual harmonics are changed. One can think of this process as generating 512 sinusoidal waves relating to each other with correct amplitude and phase (odd harmonics of the 1024 point FFT) and passing them all at once through the LTI system. In general, frequency response of a driver in a enclosure could be approximated as high-pass or band-pass filter. This implies amplitude and phase variations over the operating frequency range. As a result of this, some of the 512 sinusoidal waves passing through the LTI system will be affected to various degree. Some will be almost completely suppressed, others will be passed through, but shifted in phase and so on. The resulting waveform, Vout() will be again a linear combination (sum) of the 512 waves and may be vastly different from the square-wave entering the system. Transient response is much more difficult test for the loudspeaker system than a sine-wave sweep. The sine-wave sweep does not take into account phase shift of the system. Similarly difficult test, advocated by many audio reviewers is White Noise test, which contains infinite number of frequencies simultaneously present in the input signal, but their amplitudes and phases are random. Value of the Fourier coefficients indicates, that most of the wave's energy is present in the fundamental frequency F1 and several higher odd harmonics. For the low frequency transducer, it is vital to correctly recreate these frequencies. This way a solid and powerful sound can be generated.

Transient Analysis.

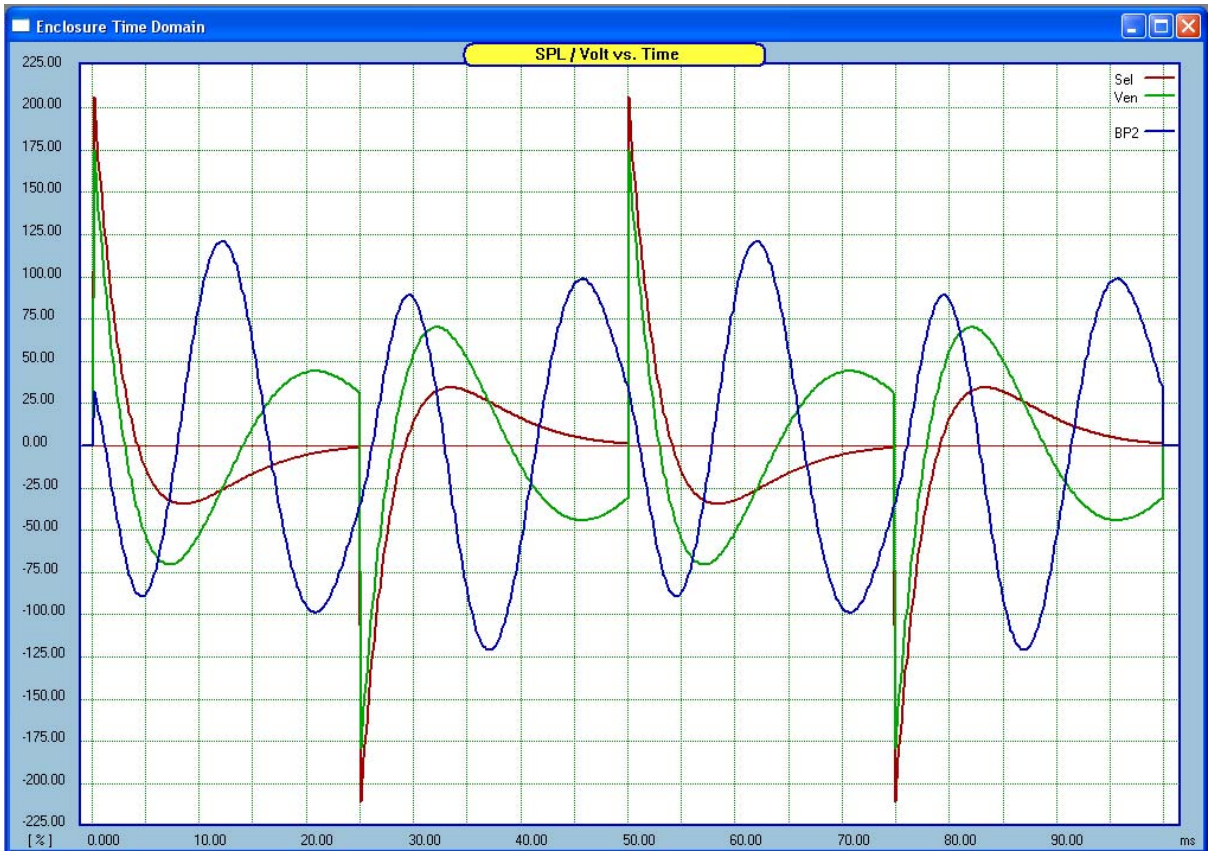


Fig 6.2 Transient response of sealed, vented and BP2 boxes

An interesting case is shown on Fig 6.2. The top section of shows frequency responses of the sealed, vented and band-pass2 enclosures. At the frequency of 50 Hz, the sine-wave sweep indicates 4dB difference between the responses, with the sealed box producing the least output, followed by the vented box and the BP2 producing the strongest output. However, when the sine-wave is replaced by the square-wave, the difference between the BP2 and other boxes becomes even more pronounced. The sealed and vented boxes clearly provide "differentiation" of the square-wave, while the BP2 box provides band-pass filtering. With the 50Hz frequency being the fundamental F1, the next odd harmonic in the square wave will be F3, at 150 Hz. This and all higher harmonics will be heavily suppressed by the BP2 enclosure and most of the wave's energy will be carried by the F1 fundamental frequency. The BP2 box offers the most energy despite passing only one (fundamental) harmonic.

The square-wave passed through the sealed and vented systems at the frequencies below F3dB system's cut-off frequency, will resemble a train of "spikes", with an initial overshoot and then gradual roll-off. If the frequency of the input square wave is still reduced, the system time response becomes transient response - see Fig 6.3. Here a 5 Hz square-wave is applied to the sealed and vented systems and the "classical" ringing can be observed. The amplitude and duration of the ringing can be now compared for all the designs. Fig 6.4 shows a sealed system with Qtc=1.5. The ringing is far too excessive for most kind of music. However, it would offer an impression of "more bass" to an untrained ear. Male voice would sound really unpleasant when listened to through this system. Fig 6.5 shows a misaligned vented system. Please note again the excessive ringing. Duration of the ripple can be read directly from the plot. Also, period of the ripple, converted to frequency (see Fig 6.5) should correspond to some peak on the frequency response. The top part of the Fig 6.5 shows a misaligned vented enclosure, which results in a pronounced peak on the frequency response around 50 Hz. To convert from period of the ripple to frequency, use the following formula:

1/(period in seconds) = (frequency of the peak in Hz)

Figure 6.3 Transient response of sealed and vented boxes at 5Hz

The time domain and frequency domain windows allow the user to **correlate enclosure responses in two domains**: **frequency and time.**

The example above illustrates well this dependency. Quite often, the response may be acceptable in the frequency domain, but will not be acceptable in the time domain. Eg; Enclosure may accentuate bass quite well (to the designer satisfaction) but excessive ringing, very disturbing on male voice, will cause the design to be modified. In order to perform time-domain analysis, please select “Time Domain” option from the “Tools”. Before you open the TD window, it suggested, that you reduce the vertical size of the main window. Simply shorten the main window from the bottom so that the 'Message' line is cut out of the visible area of the window. The reason for this is, that once you open the TD window and then click-on the main window to change some data, the main window will cover completely the TD window making it difficult to bring it back to front. If you shorten the main window before, the TD window is opened, you will have a small area of the TD window at the bottom of the screen to click-on and bring the TD window back to front. Plotting the time responses of the enclosure is achieved by (1) selecting the desired frequency of the square wave from the small editable window on the bottom of the TD screen, (2) activating the 'Plot' button to open a floating menu and (3) selecting the desired plot. Transient analysis becomes a very important factor when attempting to design loudspeaker system with active equalization.

In case of sealed boxes, a popular solution is to employ a peaked high-pass filter. However, some writers (V. Staggs, "Transient-Response Equalization of Sealed-Box Loudspeakers", JAES, December 1982) propose circuits with biquadratic transfer function. This solution is claimed to provide the desired equalization and improved transient response over the peaked high-pass filter. The peaked high-pass filter (2-nd order), together with the transfer function of the sealed box (also 2-nd order) would push the total order of the equalized system to 4. The solution proposed by Staggs, keeps the minimum order of the equalized system at the level of 3.

Modal Influence on Loudspeaker in Time Domain

It is often claimed, that for accurate sub-bass reproduction, you need a sealed enclosure. The word "accurate" is frequently used in reference to sealed enclosure time domain behavior. This excludes vented, passive radiator and acoustic bandpass woofers, because they all rely on resonant energy storage to increase efficiency and to reduce size. Indeed, the sealed enclosure only modifies already existing compliance of the driver, while vented enclosure adds air mass in the vent. This air mass, when combined with the air trapped inside the box, creates additional resonating circuit, or energy storage element, eventually affecting the sound quality produced by the loudspeaker system.

It is true, that sealed boxes do not add any additional "energy storage" elements over and above of what is already in the system. But have you ever stopped to consider, if there are perhaps any other "energy storage" elements involved in the reproduction of sound in your listening room?.

This question is not a trivial one, particularly, that your loudspeaker and you are placed in the same listening room. At the low-end of the audio spectrum (frequency less than 300Hz) the sound generated by your loudspeaker undergoes considerable changes in pressure and phase before it reaches your ears. The degree of the changes, commonly known as low-frequency room transfer function, can vary the sound in amplitude by more than +/-20dB. This variation of the room transfer function is due to room modes, or self-resonant frequencies. Please note, that we have just mentioned additional resonant frequencies. This automatically implies additional "energy storage" elements. Each room mode has associated with it its own "energy storage" element, having its own resonant frequency and Q-factor.

To establish a reference point for a simple comparison, let us consider sealed and vented enclosures with reasonably well behaved frequency responses. The sealed box has usable volume of 70lt and the vented box is larger – 130lt. Both enclosures produce in free-field the "textbook" frequency domain responses, appropriate to their design – see Figure 6.4. The sealed box exhibits –12dB/oct roll-off and the vented enclosure has –24dB/oct roll-off at low end of the spectrum. Another thing immediately obvious, is the –3dB cut-off frequency, which would be around 30Hz for the vented enclosure and about 56Hz for the sealed box.

The analysis should not stop here. The loudspeakers will be eventually placed in the listening room so the effect of the listening environment must be taken into account when considering sound reproduction. In this article, we would like to focus on the transient response or time domain characteristics of the loudspeaker generating sound waves in an enclosed space. **We are very quick to take advantage of "room gain" effect on a subwoofer, but we are much less inclined to include room modes' influence on the final sound produced by the subwoofer**. This attempt to highlight the issue is rather simple, but we hope to offer you one more angle to consider when attempting to decide upon the type of enclosure that would meet your complete set of expectations.

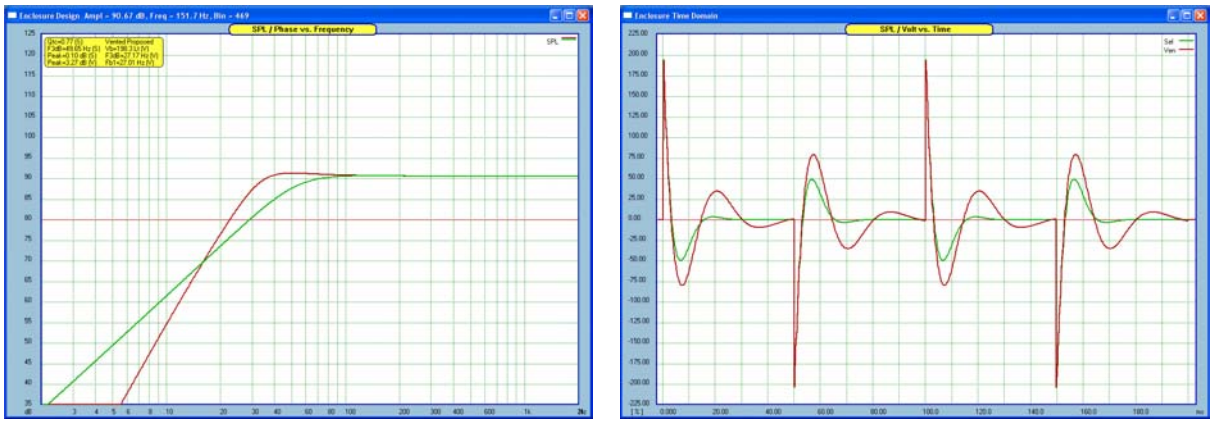


Figure 6.4 (a) SPL of sealed and vented enclosures. (b)Transient response of sealed and vented enclosures

Without any doubts, the decision is not a simple one. Just about every designer has slightly different set of preferences, so more often than not, the final design is a result of many compromises being made.

Time domain response to a rectangular pulse is also well defined for the above two enclosures. Figure 6.4b shows typical, free-field, time-domain response of vented (brown curve) and sealed (green curve) enclosures excited with 10Hz rectangular pulse. It is easy to notice, that sealed enclosure response has settling time of 15ms, while vented enclosure continues to ring till 50ms, which is more than 3-times as long. Also, the amplitude of the vented enclosure ripple around 8ms mark, is nearly twice (100%) as big as the sealed enclosure. Finally, the vented response crosses zero-line 4 times before it settles. We suppose, all the above would justify claims of superior performance of the sealed enclosure in time domain – known as having better transient response. The contribution of the "extra energy storage" element in the vented enclosure is quite evident from Figure 6.4. Those elements always tend to increase the amplitude of ripples and extend ringing time in the transient response of the enclosure.

Behavior of the sound waves in a small enclosure is a quite complex problem. However, the wave property of the sound energy makes it possible to introduce significantly simplified model of the problem, restricted to low frequencies, where only the room dimensions are of significance. In a rectangular room we are going to consider, the sound waves will be reflected and re-reflected between all 6 surfaces. When the reflections occur between opposite surfaces, the standing wave pattern generated this way is called "axial". Reflections involving four surfaces are called "tangential" and finally, reflections involving all six surfaces are called "oblique". The general, the room will exhibit an infinite number of modes.

The modes exhibit resonant character. They have natural resonant frequencies, bandwidths dependent on their damping factors, and amplification Q-factors also dependent on their individual damping. Modal density varies in frequency. Modes tend to be widely separated at lowest audio frequencies and then come closer together as the frequency of interest is increased. In addition, the effects of low frequency modes on the source frequency response are heavily dependent on the position of the source and receiver. For example: if a receiver is located in a pressure node, it will record no sound at all for a case of zero dumping factor for this particular node. In a typical listening room, exhibiting many modes, the frequency response of the source will be the vector summation of contributions of all modes, as well as the function of the source and receiver positions.

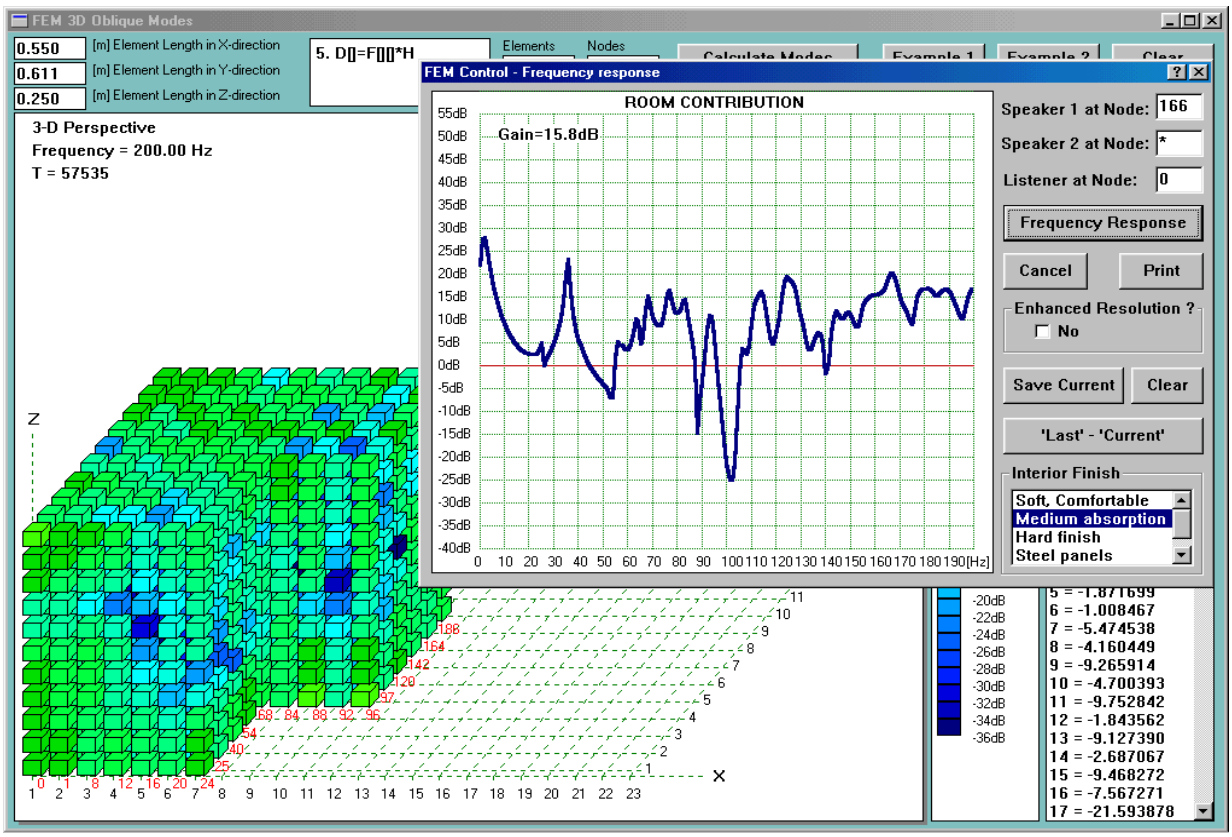


Figure 6.5. Example of a Room Transfer Function of L-Shaped Room

An example of an L-shaped room transfer function is shown on Figure 6.5 [2], where it is evident, that variation in SPL is large and there are many peaks and dips. The SPL variations are within +/-25dB and some modes have quite high Q-factor (eg: 36Hz mode).

In the following analysis of the two enclosures, we are trying to compare when placed in the room would be somewhat simpler, if we could limit the room transfer function to exhibit only 2-3 peaks. In order to create such room, we have devised an "electronic room", with only 3 peaks, using electronic circuit with three operational amplifiers. All we need to re-create room listening environment is the room transfer function, regardless of the way it was generated. we can therefore create some well defined electronic circuit and use it's transfer function as the simplified room transfer function. This approach allows me to control number of modes and their Q-factor quite easily.

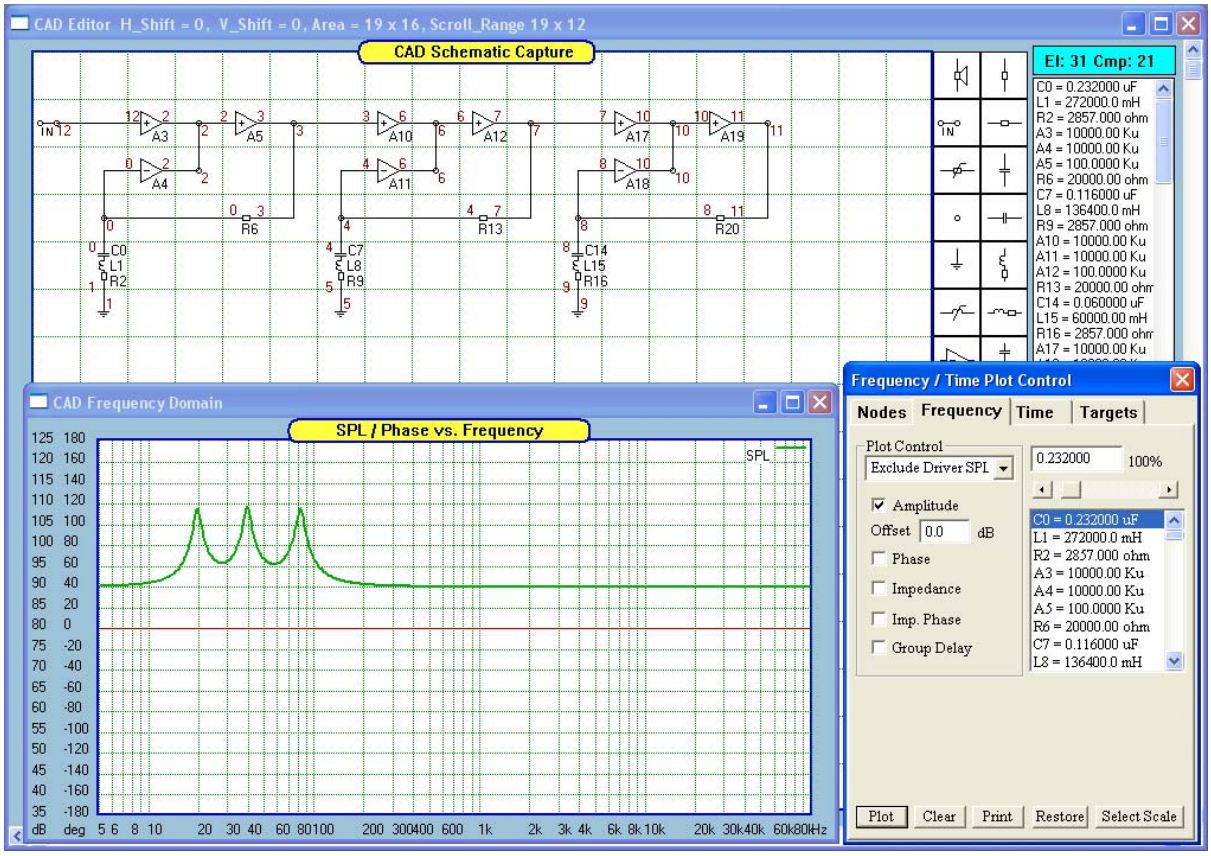


Figure 6.6 Simplified "room transfer function".

Our simplified room has only three modes at 20Hz, 40Hz and 80Hz. Each mode has Q-factor of 12 and corresponding gain of 8, therefore the humps created by the room will not add more than 18dB to the loudspeaker response. Also, we introduced only peaks, no dips, therefore we assumed no frequency band cancellation would occur. As you can see, our "electronic room" is much less severely affecting the sound than your typical listening room would.

Figure 6.6 shows the details of the electronic circuit we used to simulate our "electronic room". Basically, we have three differential amplifier stages, each having a series resonant circuit in the negative feedback loop. The loop is arranged such a way, that gain of each stage will be maximum at the resonant circuit center frequency. **The bottom section of the Figure 6.6 window shows the resulting frequency response – or room contribution.** The three modes are evident as the resonant peaks. It would be quite simple to add more peaks, or room modes to our "electronic room". All we would have to do, is to cascade more stages peaking at different frequencies. However, for the purpose of my simplified analysis, it is preferred to keep the picture as simple as possible, yet the ability to demonstrate the issues at hand must be clearly evident.

In the next step, we need to combine the "room contribution" with the two enclosures we selected for comparison. Mathematically, it is a simple multiplication of transfer functions, and the resulting combined speaker + room transfer function is shown on Figure 6.7. The drop in SPL towards the higher frequency end due the voice coil inductance, Le, in the model is removed.

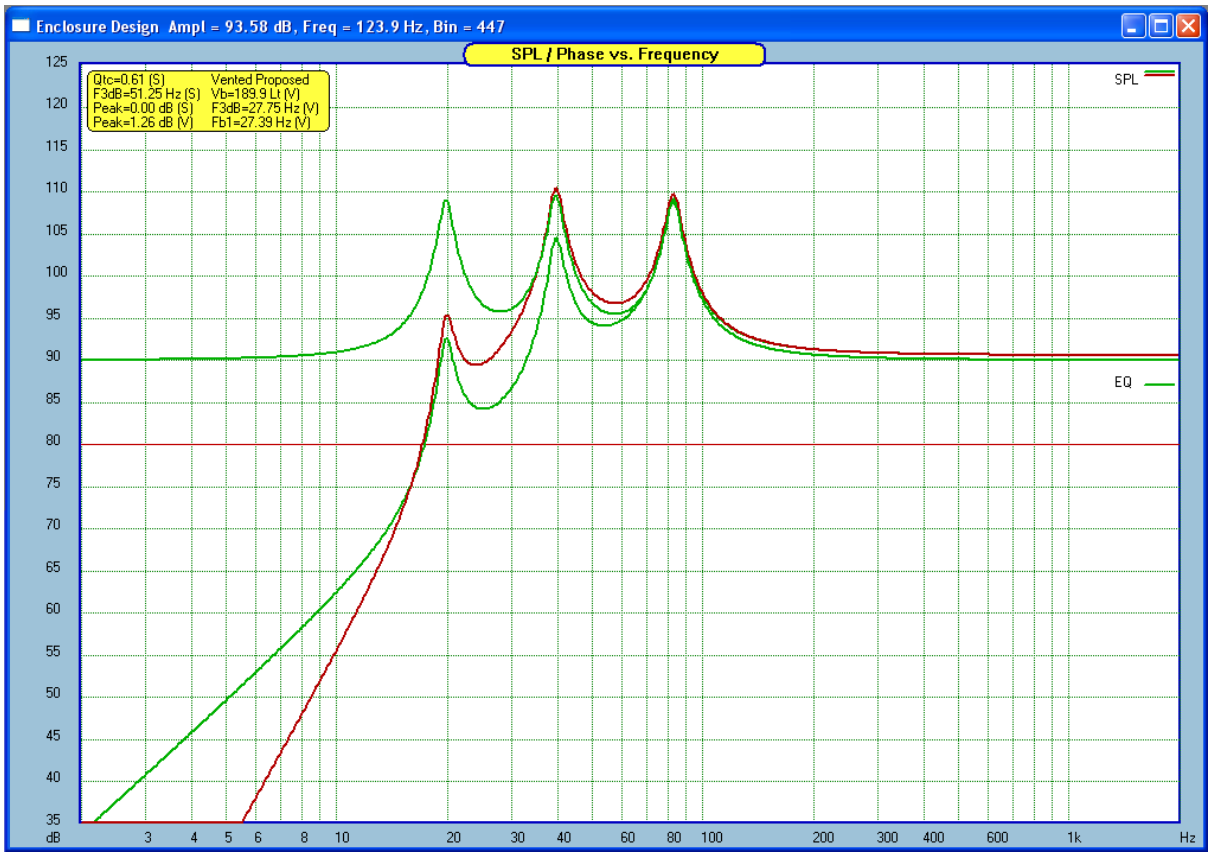


Figure 6.7. Room Transfer Function combined with both enclosures.

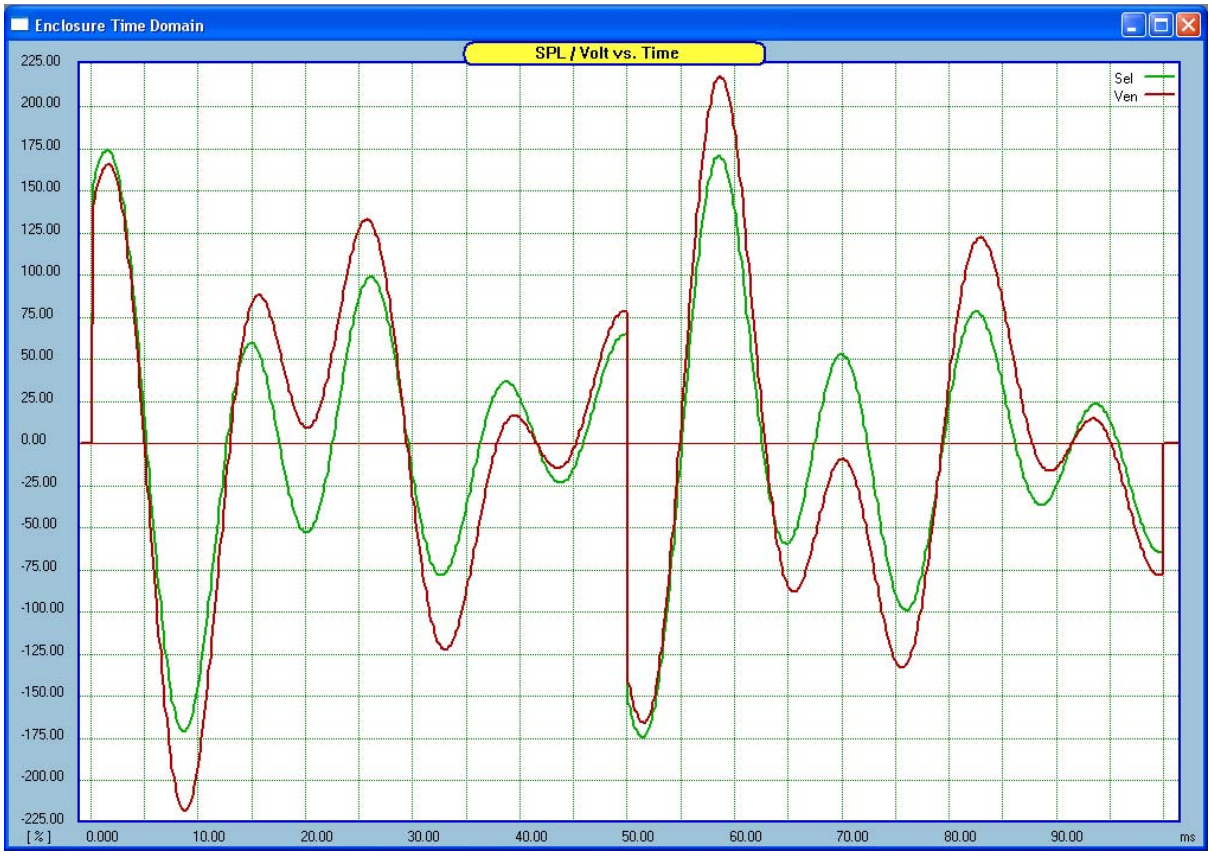


Figure 6.8. Transient response of sealed and vented boxes at 10Hz

Now, for the resulting time domain responses please review Figures 6.8 and 6.9. On Figure 6.8, sealed enclosure curve crosses zero line 8 times. Vented enclosure curve crosses zero line 4 times. Now, the vented enclosure ripple is only 30% higher than sealed enclosure. This is down from 100% for the anechoic chamber response. On Figure 6.9, vented enclosure has slightly larger ripple, but sealed enclosure has more pronounced 2-nd harmonic component. Total duration of ripples is comparable for the two enclosures.

At this stage, you would be forgiven for thinking, that the sealed box begins to look and sound awfully like the vented box. The sealed enclosure "accuracy advantage" now begins to disappear quite rapidly, even with only three simulated room modes. When we add more modes and also take into account dips due to phase cancellations within the room, we may expect the room characteristics to dominate the SPL and time domain responses even more. The actual number of room modes (or resonant storage elements) in your listening room is infinite. However, at higher frequencies they are so closely spaced, that it is impossible to separate individual modes any longer. This phenomenon actually works to your advantage, smoothing room response quite significantly.

We guess, it should be clear now, that the element responsible for this problem is the listening room itself. Please note, that both loudspeaker enclosures used for this simple comparison were relatively low-Q systems in comparison with the room modes Q-factors of 12. Therefore, you may expect, that it will be the room modes that will dominate the characteristics of the sound, rather than the type of enclosure being examined. **The saying "above 300Hz, you listen to the speaker, below 300Hz you listen to your room" begins to make more sense now.**

In general, it is quite a challenge to design high-SPL, low distortion subwoofer, capable of delivering 20Hz notes. The issue boils down to moving large volume of air. In case of a driver mounted in a sealed enclosure, the maximum SPL is clearly dependant on Xmax capability of the driver and this relates to BL non-linearity and suspension non-linearity.

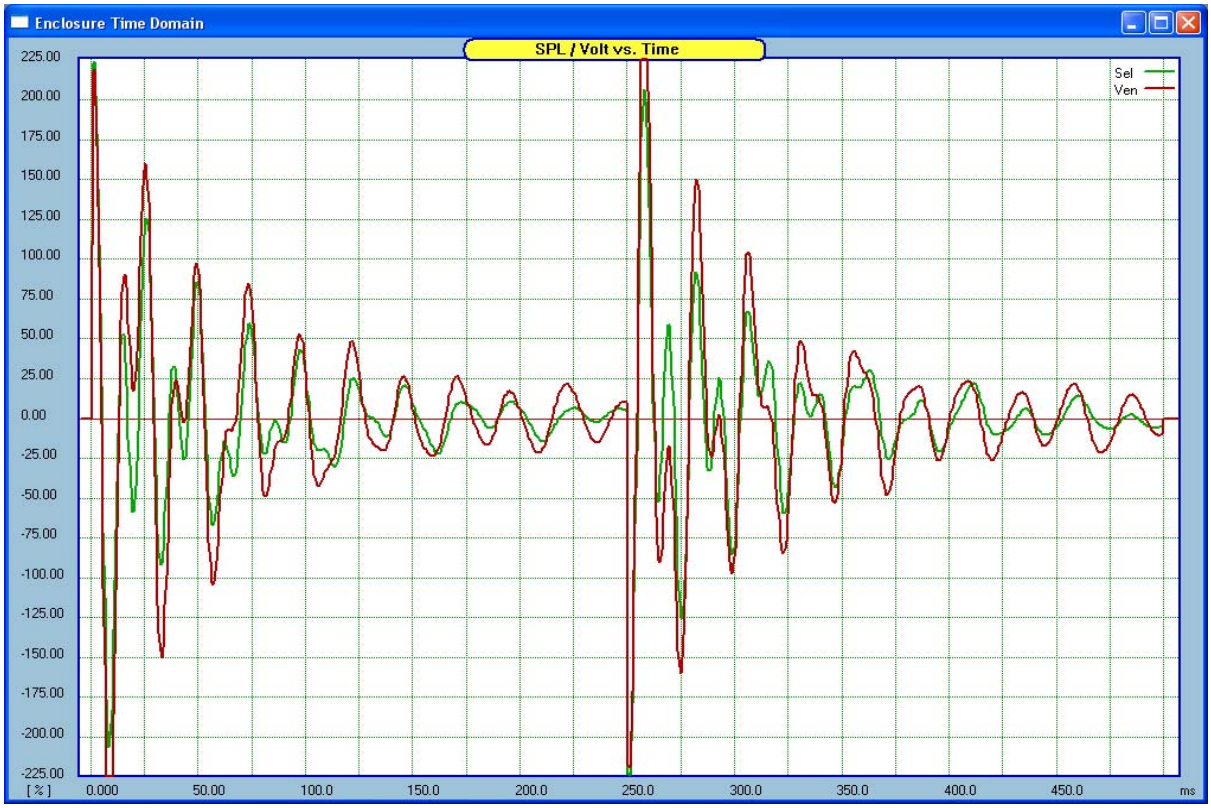


Figure 6.9. Transient response of sealed and vented boxes at 2Hz

Vented enclosure seems to have some advantage over the sealed box, because it actually reduces cone excursion around port resonance frequency. This can only work well, if the port itself is large enough and port compression and non-linearity problems are kept very low. The port takes over the job of generating the SPL, therefore reducing driver's motor system and suspension system linearity related problems. This in itself, would be worth taking into consideration when selecting appropriate enclosure for your subwoofer.

But there is more. Consider also the fact, that my vented enclosure actually produces 7.5dB more output at and around 30Hz than the sealed box. This represents more than four times (3dB +3dB) in required amplification power.

If you use electronic equalization, to raise the SPL of the sealed box at 30Hz or below, you will hit the Xmax much sooner, than you would with the vented enclosure, therefore at low end of the spectrum, you will never obtain the same SPL with the sealed enclosure as you would with the vented box.

As the final word, we would like to point you to a very interesting work of Soren Bech of Bang & Olufsen and Peter Bangs [1]. They have investigated the audibility of changes in the subwoofer lower cut-off frequency and slope. The slope was changed from 2-nd, 4-th and 6-th order and the cut-off frequency was varied between 20, 35 and 50Hz. Also, the influence of amplitude ripple and delay ripple was investigated. The results of the experiment showed, that the lower cut-off frequency has a significant influence on the perceived bass reproduction, while filter order was found not to be of significant importance. Also, amplitude ripples had a significant influence on the perceived bass reproduction, but the delay ripple had not.

If you dare to equate their filter slope with sealed and vented enclosures' slopes (sealed = 2-nd order and vented = 4-th order) and amplitude ripple with room modes gain (see Figure 5), you can draw parallel conclusions between the work of Beach and Bangs and the presentation above. In fact, the ear is quite sensitive to the amplitude ripples. Bucklien [3] found, that listeners have detected with 100% accuracy a 5dB peak (Q=10) at 85Hz. This would offer some indication as to why the room modes are so well imprinted in the sonic signature of your listening room.

References:

[1]. Bech S. and Bangs P.Quantification Of Subwoofer Requirements Part II, AES Preprint 5199, 109 Convention 2000 September, Los Angeles.
[2]. SoundEasy V4.00se manual. Chapter 15. Bodzio Software Pty. Ltd.
[3]. Bucklien R. "The audibility of frequency response irregularities" JAES 29, 1981.

Chapter 7. Car Acoustics

Background

As a listening environment, the passenger cabin of a car leaves a lot to be desired. Space is rather limited, so the loudspeakers can only be mounted where there some room left to do so, and where the sound would not be muffled or obstructed. Often, this would be in door panels, or in the rear deck under the window, so that the interior of the trunk can be utilized as a closed box. In order to model harmonic acoustic behavior of an enclosed space with a sound source, the following equation is sufficient:

$$(K[][]-k^2M[][]+j\omega C[][])p[]=j\rho_0\omega v[] \qquad k=\frac{\omega}{c} \qquad \omega=2\pi f \qquad j=\sqrt{-1}$$

Where: K[][] is called acoustic stiffness matrix, M[][] is the acoustic mass matrix, C[][] is the damping of the system, p[] is the sound pressure vector, v[] is the excitation vector in qubic meters per second, f is the test frequency and c is the speed of sound.

For the source of the sound one can assume a "point source", which is convenient, as it can be located in any of the mesh nodes. Larger sources would have to be accounted for as a part of the room boundary. Mathematically, the problem now reduces to assembly of the K[][], M[][] and C[][] matrixes and inverting the expression in the brackets. This way, vector p[] can be found for any frequency and location of the sound source represented by excitation vector v[]. Vibration of the cabin surfaces can alter the sound generated by the loudspeakers to various degree. The low quality, vibrating doors can actually resonate to the point, that the door panel behaves like a diaphragmatic absorber. This effect can alter the cabin's resonant modes and the pressure distribution quite significantly. Taken to the extreme, the cabin would be able to maintain ¼ wave resonance. Proper mathematical treatment of this type of problem would involve "fluid-structure coupled systems", where the air trapped in the interior of the cabin (fluid) influences panel resonances (structure) and vice-versa. The issue is quite complex and involves knowledge of physical properties of materials used for cabin construction. Fortunately, the quality of the interior finish in today's cars tends to be better, so that at low-to-medium sound levels, the panel vibration is less influencing the sound than in cheaper, yesterday's models with tinny doors.

In this release of the program, we would like to concentrate on somewhat simplified approach, where we assume, that vehicle interior is a well built, solid structure, with most panels exhibiting damping effect (soft but good quality finish). This damping is modeled as uniformly populated damping square matrix C[][] in the FEM analysis. In addition, you can assign absorption coefficient to each node of your mesh.

If you are professionally involved in this subject and would like to have your unique model analysed, please contact Bodzio Software with the model dimensions and geometry. We will implement it and send you updated software free of charge.

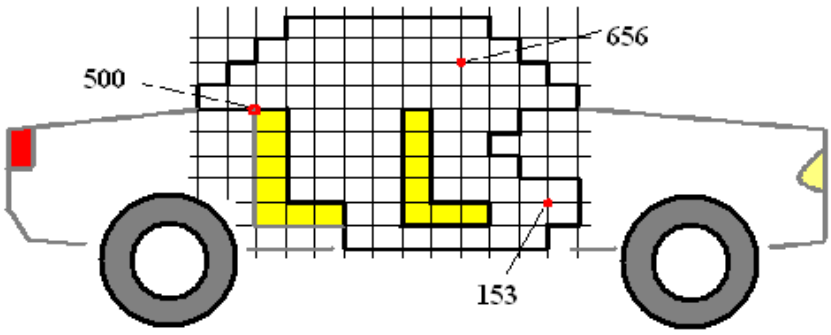


Figure 7.1 "Brick" element approximation of the passenger cabin

Furthermore, we will use simple "brick" elements to approximate the complex geometry of the passenger cabin – the white, or non-shaded area on Figure 7.1. With this in mind, we can proceed to estimate the modal influence of the cabin on the low end of the frequency response of the loudspeaker mounted in various places inside the cabin. Figure 7.1 shows the chosen locations for loudspeakers:

Speaker Location 1 – rear speaker, mounted behind the back seat (Node 500),
Speaker Location 2 – front door speaker, mounted low on the right front door (Node 153),
Microphone Location 1 – approximate position of the driver's ears (Node 656).

Finally, we can enter the above model into the computer FEM program to obtain the 3D-view like the one shown on Figure 7.2.

The Analysis

It is often beneficial to determine the resonant frequencies of the cabin first. The actual modal frequencies and the corresponding distribution of the acoustic pressure are a great help in identifying the peaks of the frequency response of the cabin itself. The FEM analysis revealed several modal frequencies below 200Hz. As you may expect, the lowest mode (65Hz) follows the longest dimensions of the car, with peaks at rear window and front pedals – see Figure 7.3. Acoustic pressure distribution for the second lowest mode is shown of Figure 7.4. Secondly, the modal analysis allows me to determine the areas to avoid for possible locations of the speaker. The speaker placed in the exact modal spot for a given frequency will produce little acoustic output on this frequency. Obviously, it goes without saying, that choice of mounting locations for loudspeakers is quite limited, as the car was designed as a means of transportation first, and as a listening environment last.

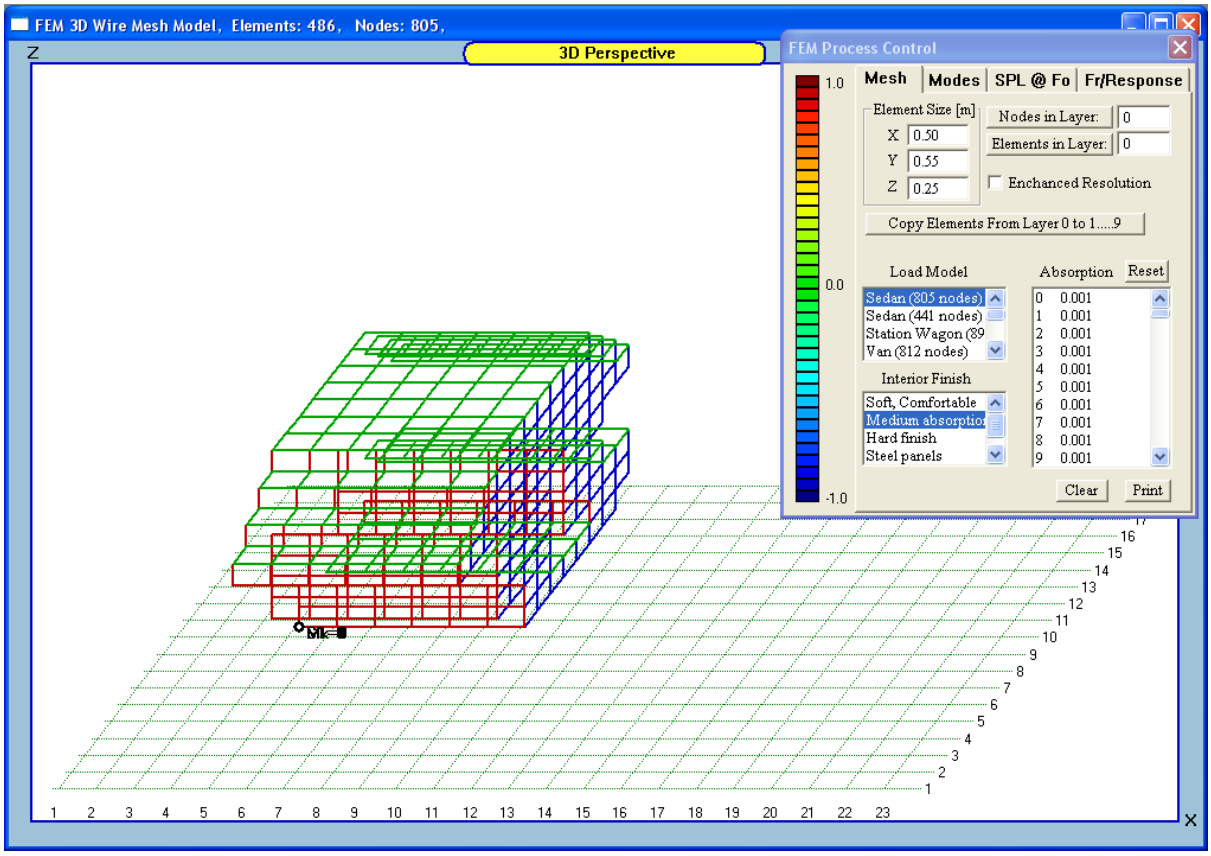


Figure 7.2. 3D-Passenger compartment model.

For the purpose of further analysis, we assume that my loudspeaker has a perfectly flat frequency response. Using such a source is quite legitimate and helpful in understanding what influence the cabin finish and geometry has on the frequency response of a typical loudspeaker. When the "perfect loudspeaker" is used, the only visible amplitude irregularities are the one's that are introduced by the environment.

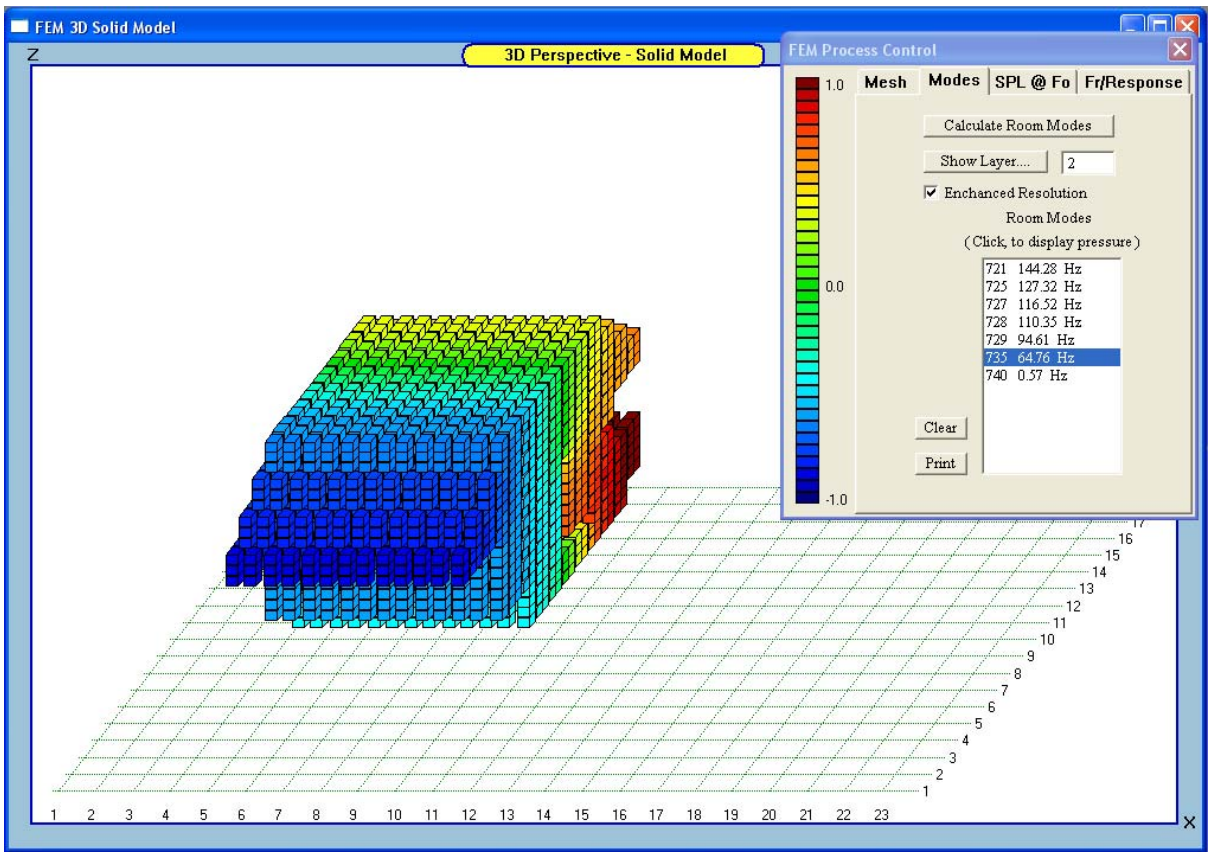


Figure 7.3. Acoustic pressure distribution for lowest mode. Red/blue color indicates high pressure and green color shows modal line.

When a "perfect loudspeaker" is mounted at the back of the cabin, under the rear window, and the listening position is assumed to be where the driver's head is, the modeled frequency response of such "perfect loudspeaker" is shown on Figure 7.5. Closer inspection of the plot would indicate: (1) there is overall gain of about 5-6dB averaged across 10-200Hz frequency range, (2) the plot exhibits a +12dB slope below 50Hz – this is due to the cavity effect, and (3) the frequency response is quite irregular between 70-95Hz and then improves as modal density increases towards 200Hz. The 65Hz mode determined from my earlier FEM analysis appears to be quite pronounced for the amount of dumping assumed earlier. Incidently, a closed box with corner frequency at around 50Hz, mounted in this car, theoretically can produce flat acoustic frequency response to DC. The 12dB/octave slope of the closed box, would ideally be compensated by the 12dB/octave rise from the cavity effect.

When the loudspeaker is mounted in the lower section of the front door, the modeled frequency response of my "perfect loudspeaker" is shown on Figure 7.6. Once again, closer inspection of the plot would indicate: (1) there is overall gain of about 7-10dB averaged across 10-200Hz frequency range, (2) the plot exhibits a +12dB slope below 40Hz – this is due to the cavity effect, and (3) the frequency response is quite irregular between 50-120Hz and then improves only slightly as modal density increases towards 200Hz. The 65Hz and 95Hz modes determined from my earlier FEM analysis appears to be quite pronounced for the amount of dumping assumed earlier. There is significant energy concentration between 55Hz to 115Hz. Also, the sharp dip at 52Hz is likely to audiable. Loudspeaker frequency response will be strongly influenced by the windows position. We have restricted my analysis to windows up position, which seals the vehicle interior and causes the cabin to exhibit resonances. Secondly, vehicle in these conditions exhibit the cavity loading which will introduce a 12 dB per octave rise in the bass at low end of the audio spectrum. The corner frequency can be estimated from the major dimension of the passenger cabin and varies between 45Hz to 90Hz with the higher frequency assigned to small cars.

If it was possible to designed vehicles with perfectly sealed passenger compartments, the loudspeaker cone moving forward would pressurize the cabin and the pressure would be maintained as long as the cone stays in the "forward" position. This is really equivalent to a DC (0 Hz) acoustic response.

The plots shown above for two different loudspeaker locations captured both effects: cabin resonances and +12dB/oct rise towards the low end of the audio spectrum. For the vehicle of the size chosen for my model, the corner frequency would be between 45Hz and 50Hz.

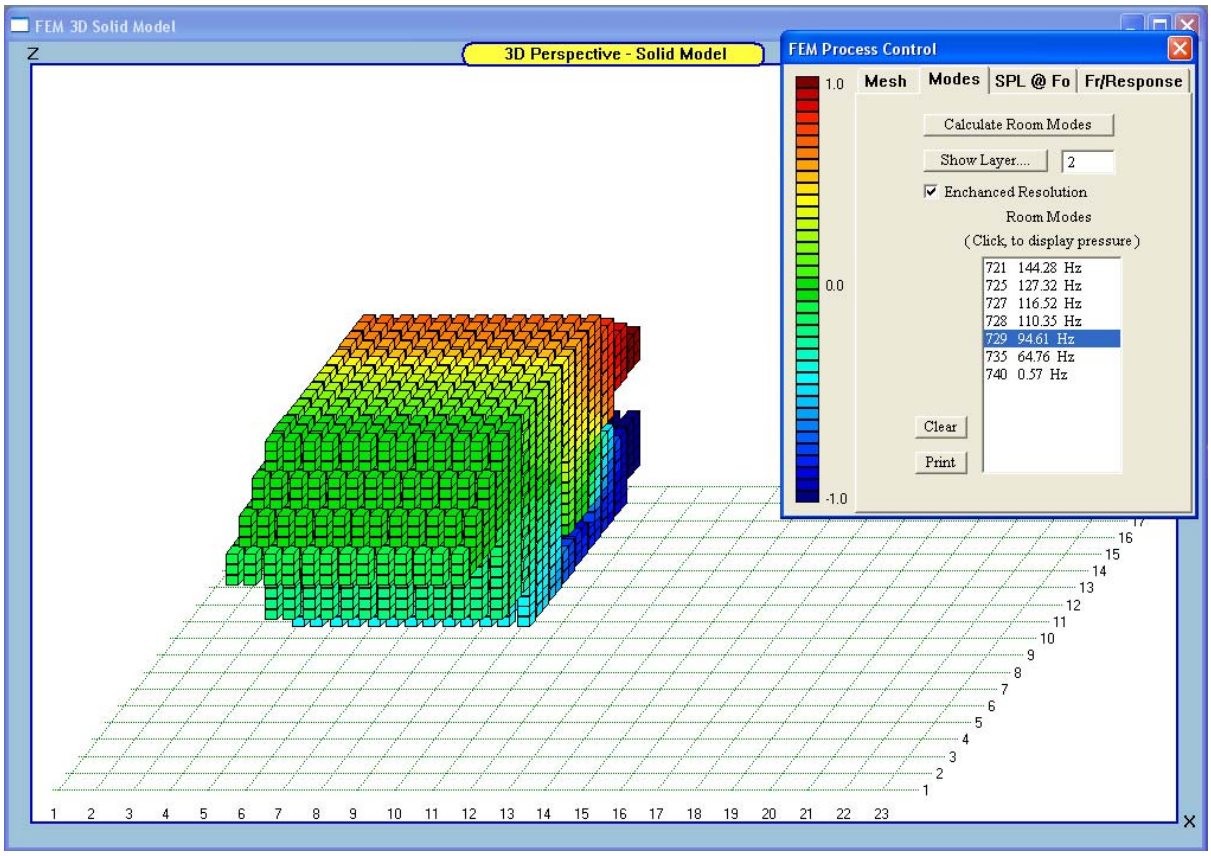


Figure 7.4. Acoustic pressure distribution for the second lowest mode.

Our earlier assumption, that the source of the sound is a "point source" implies, that it radiates the sound uniformly, in all directions. This assumption agrees well with restricting the analysis to about 200Hz, below which all sound sources appear to be omnidirectional. When both speakers are operating at the same time, the combined frequency response would be similar to the one shown on Figure 7.7. Despite its "bumpy" character, the combined frequency response would be preferred over each of the individual plots obtained for single driver operation. The peaks and valleys can not be avoided in the closed compartment, but their influence can be significantly reduced by making the interior "softer" (read: more absorbing). The remaining irregularities can be further reduced by electronic equalization, thus making the total frequency response more acceptable.

In the above short discussion, we have presented one possible approach to modeling car acoustics. The discussion was based on the steady state cabin response, where the compartment modes carry the acoustic energy. With the additional assumptions of: (1) "de-coupling" the volume of cabin air resonances from the car body resonances and (2) interior panel dumping was represented by uniformly populated dumping matrix C[][], it was possible to obtain frequency response plots for a couple of loudspeaker-microphone locations. The plots show clearly the modal (resonating) behavior of the passenger cabin together with the pressurization of the cabin at lowest audio frequencies (sometimes called the cavity effect).

The set of plots (Figure 7.7) present quite realistic situation of using contributions from two loudspeakers, with microphone placed in the driver's head horizontal plane. The accuracy of the analysis in the upper range of frequencies was limited by the size of the "brick" elements we have used. The dimensions were: Y=22cm, X=26cm and Z=14cm. Smaller elements would increase accuracy around 200Hz, but the penalty would be significantly increased computational time. The final frequency response shown on Figure 7.7 seems quite acceptable taking into account where we started (Figures 7.5 & 7.6).

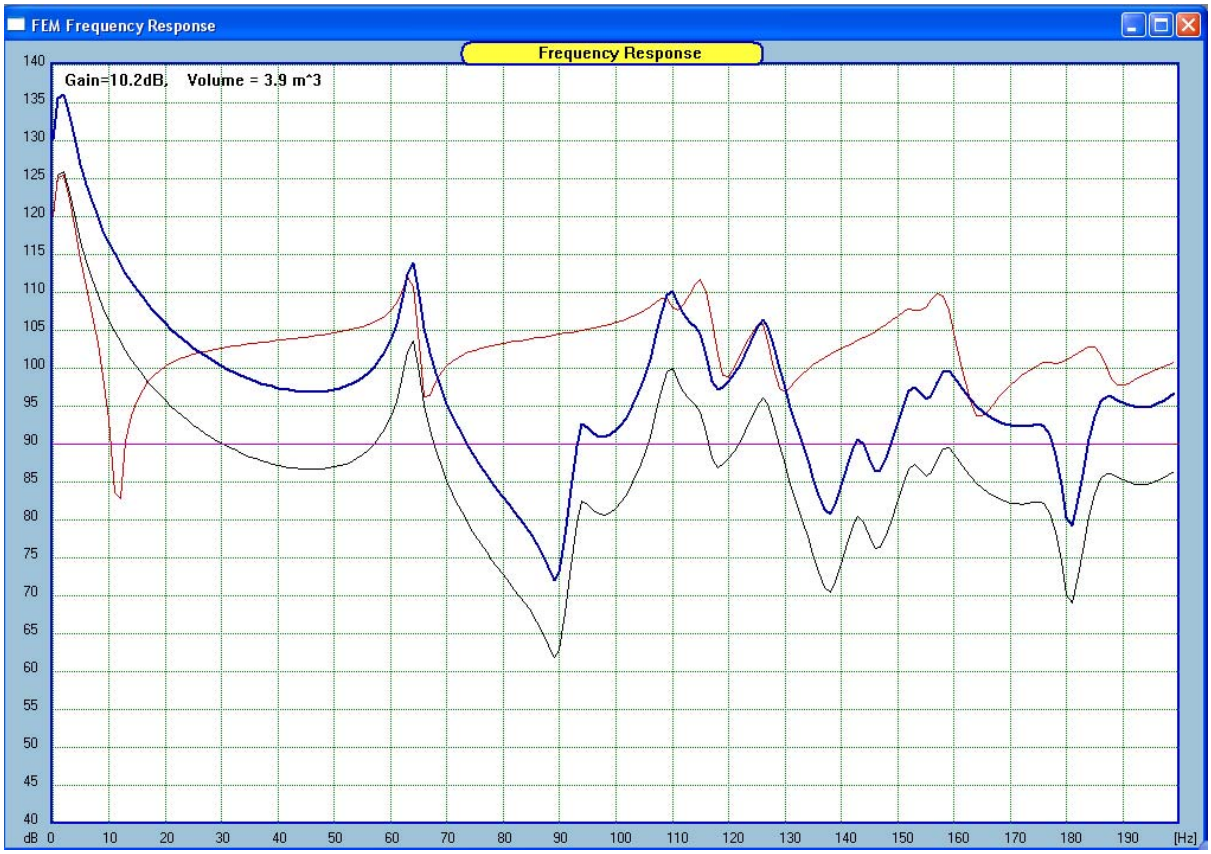


Figure 7.5. Loudspeaker at the rear of the passenger cabin (blue).

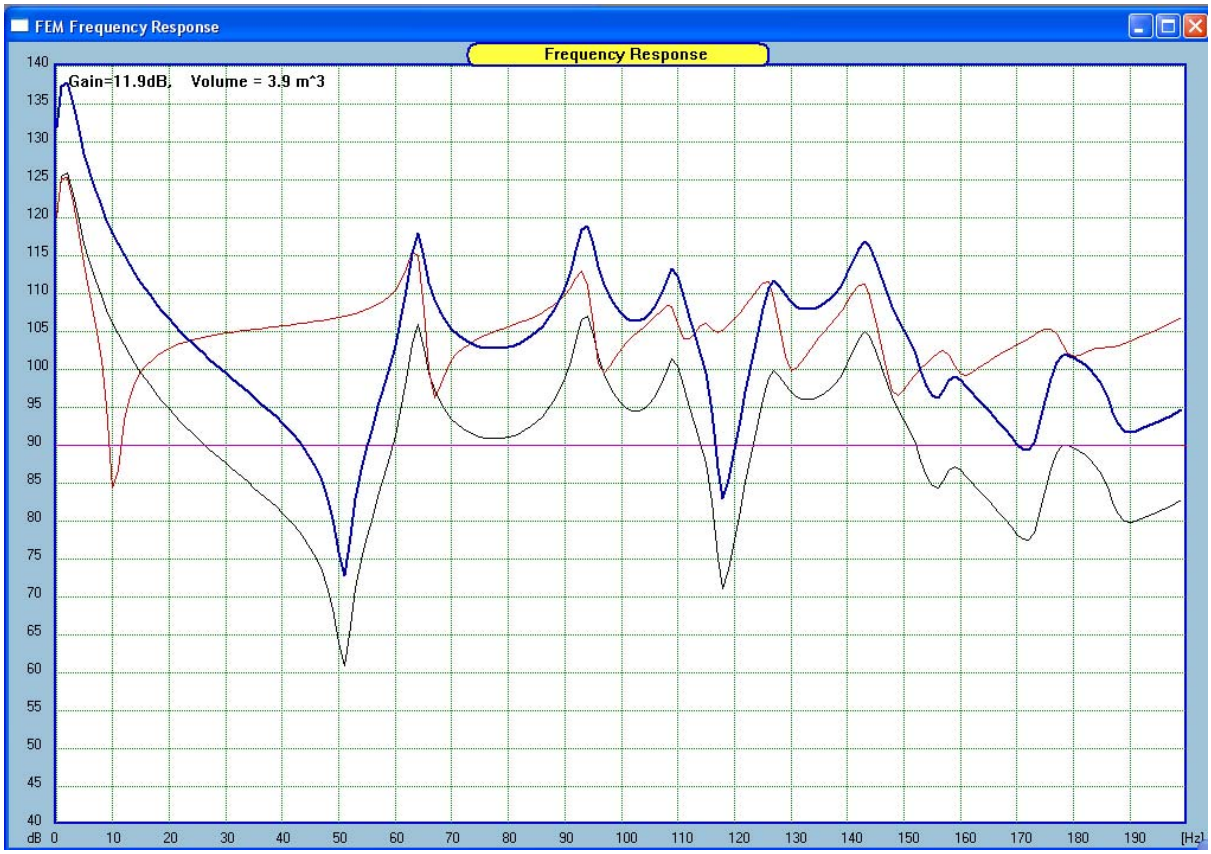


Figure 7.6. Loudspeaker mounted in the front door (blue).

Operation of the 3D FEM Car Audio Functions is explained in details in Chapter 15.

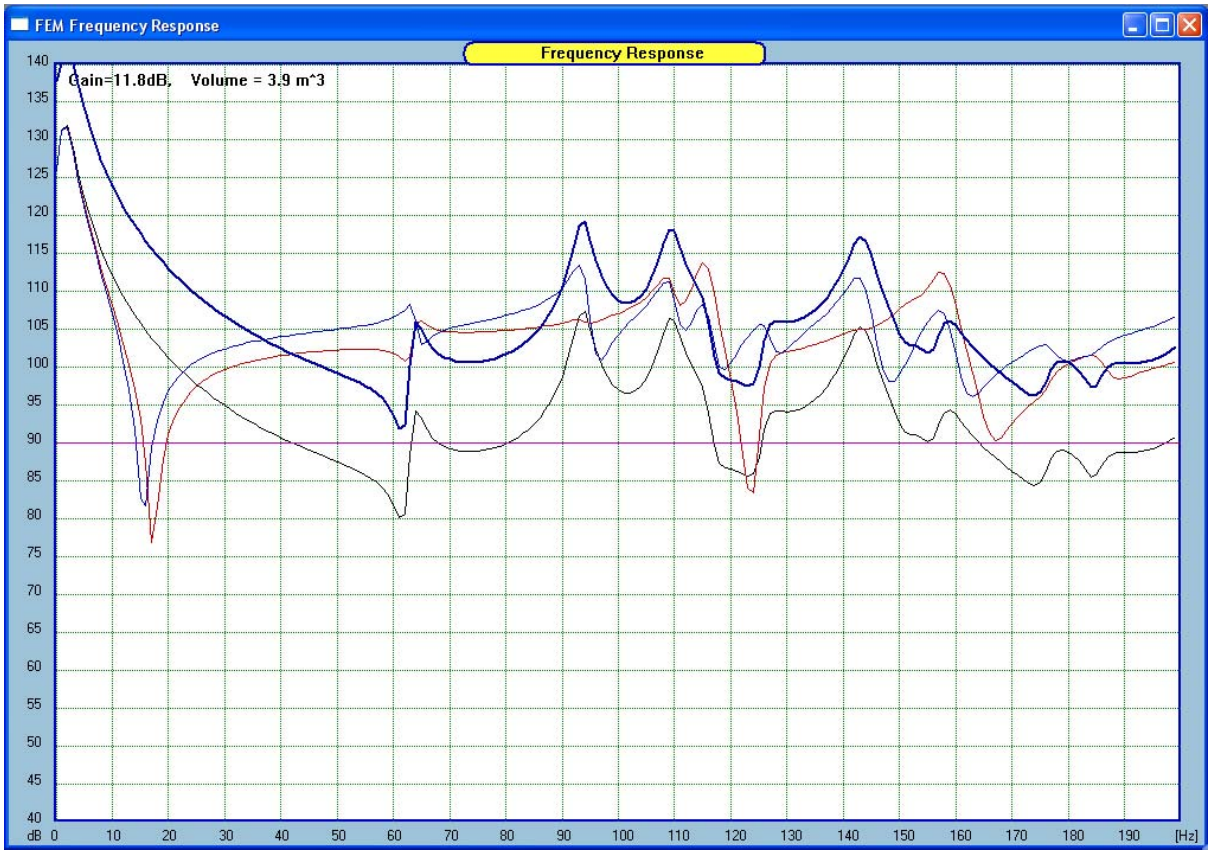


Figure 7.7. Frequency response due to SPL of two speakers combined (blue).

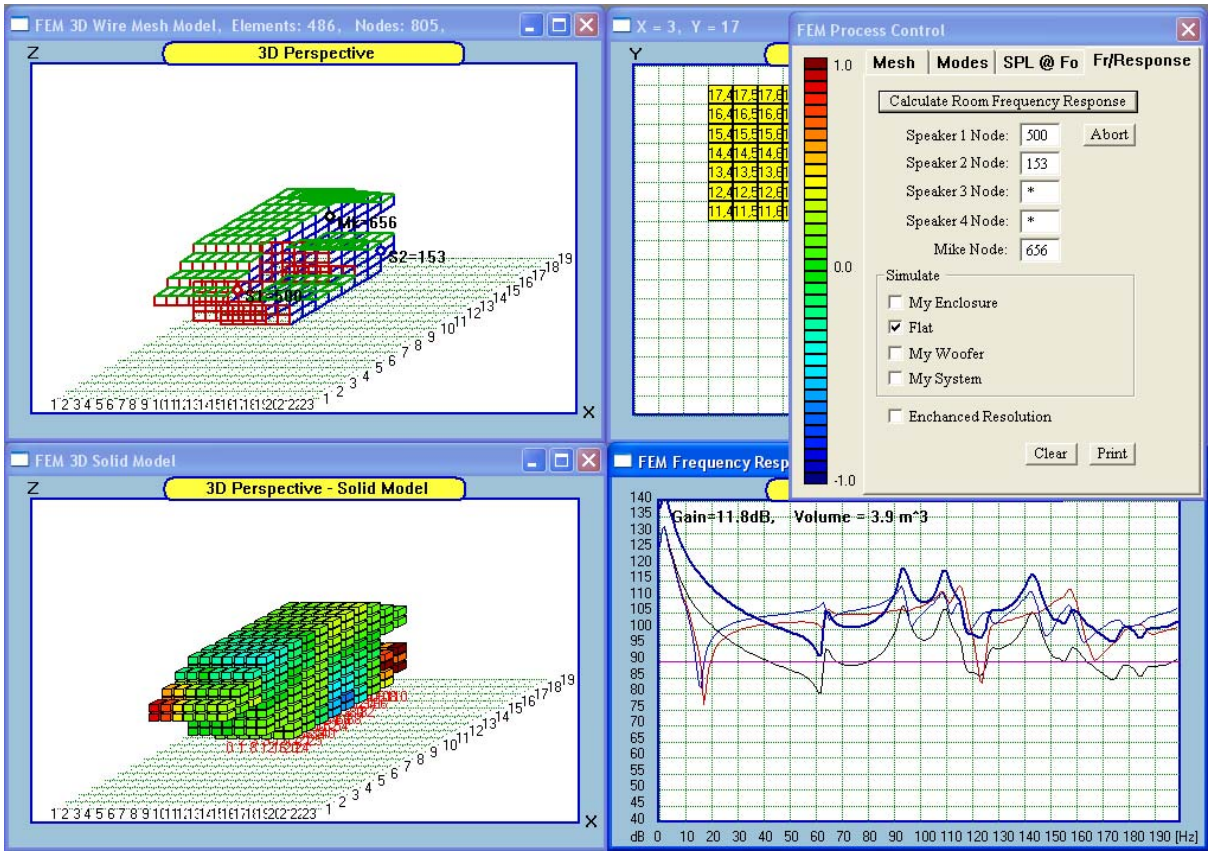


Fig 7.8 Selecting nodes for our car model.

Part 3

Working With Crossover Functions

Chapter 8. Using CAD System

Background

The two main tasks performed by the Crossover section are:

1. Design and optimization of filters and crossovers.
2. Editing and modelling of the amplitude/impedance/time compensation circuitry.

Frequency and time domain analysis is available on passive and active circuits. Fig 8.1 shows an example of a network created and analysed with this module's functionality.

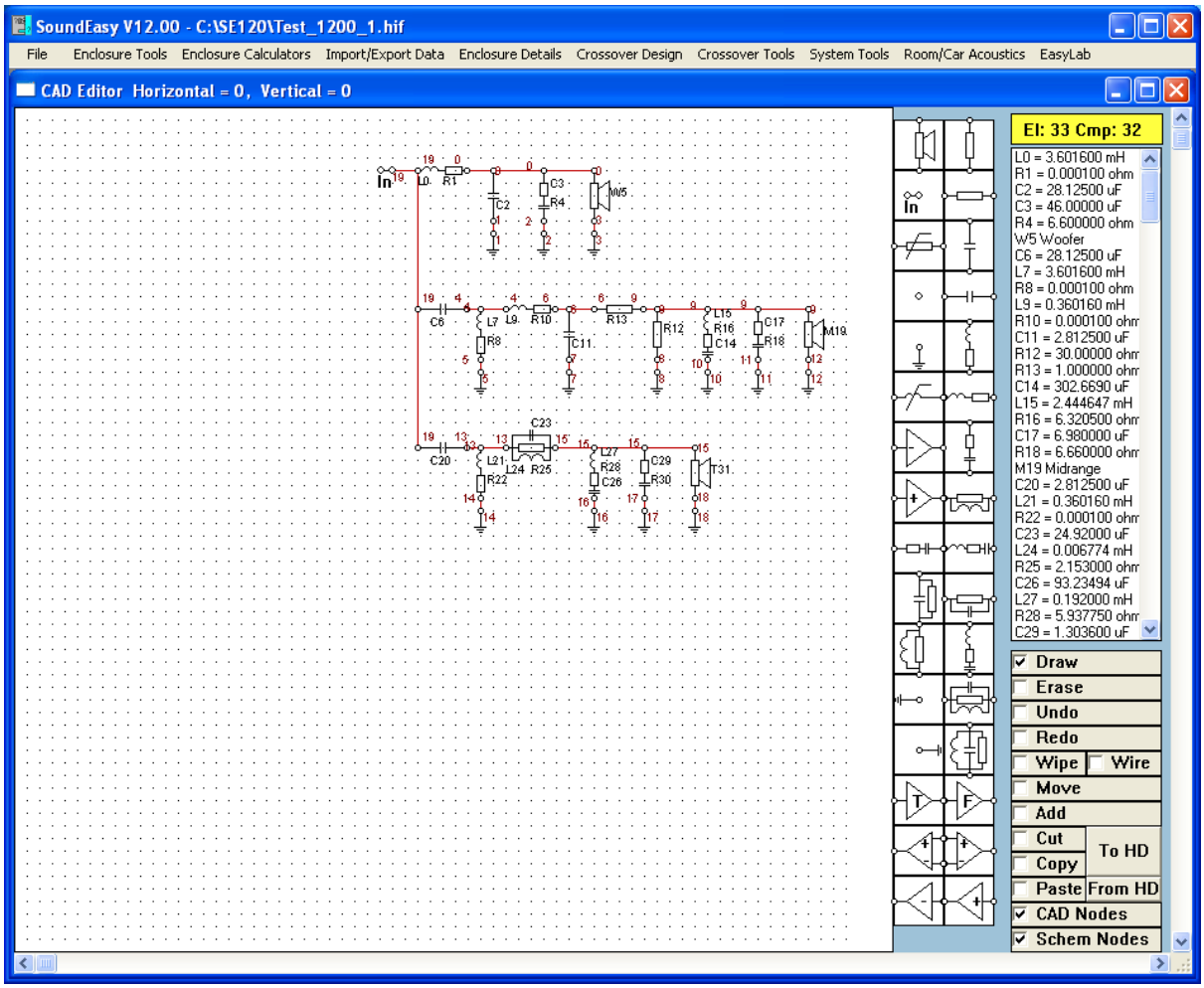


Figure 8.1. Example of fully edited schematic for a 3-way crossover

Pick And Place Component

The middle portion of the screen is occupied by the drawing area. Components located on the right hand side can be "picked-and-placed" on the drawing area using the LEFT mouse button. There are 19 passive components to choose from. Some of the components consist of 3 elements, like LRC tank circuits. This approach helps speed up the calculation processes within the program. The list box rectangle on the far right hand side is a display field and it will display automatically the component values (in the correct units). In order to pick a component from the provided list the user is required to:

1. Position the pointer arrow above the required component .
2. Press and keep pressing **the LEFT mouse button**. The selected component will be highlighted in **blue.**
3. Drag the arrow pointer (and the component with it) to the chosen drawing area, still pressing the LEFT mouse button.
4. Once the component is displayed where it is wanted, release the mouse button.
5. Go to (1) and pick the next component.

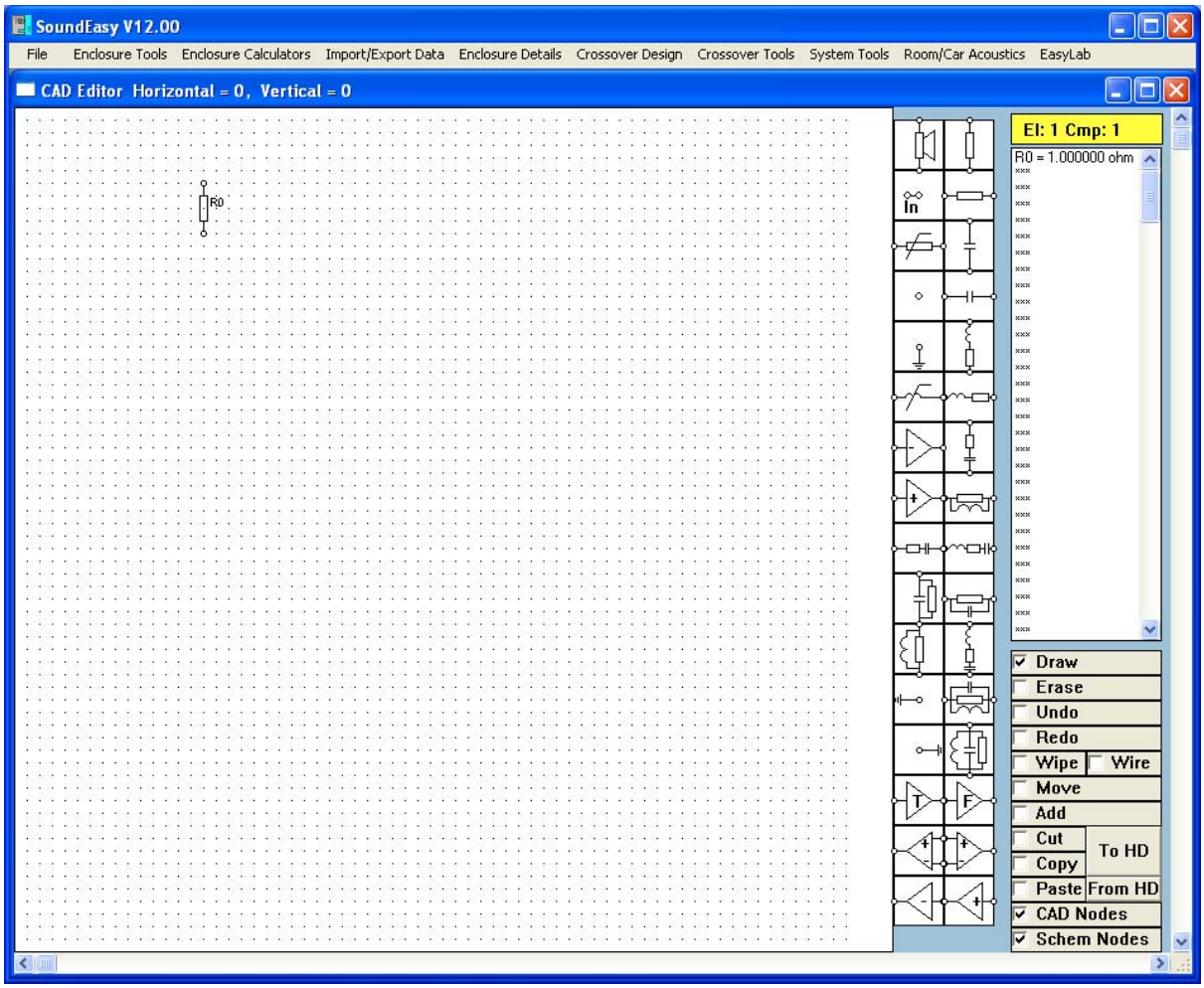


Fig 8.2 Resistor Ro placed on the CAD drawing area

To understand this simple process better still, refer to Fig 8.2 - 8.7. The process of building a small low pass filter is presented in a few steps. First, a load resistor R0 is placed on the drawing area (Fig 8.2). The resistor was "picked and placed" in the manner described above. Then, the C1 capacitor was positioned on the left hand side of the resistor. (It is assumed, the input to the network being built on the screen is on the left hand side of the screen and the output on the right hand side). Signal flow will be from the left to right as well.

You should now follow Fig 8.3 - Fig 8.6 and place all the remaining components.

Connecting Components

Various connection types are shown on Fig 8.5. Listed below is a short description of each type.

1. In order to connect resistors R0-R1, position the pointer arrow on resistor R0 first, then press and keep pressing the **LEFT** mouse button. Next, drag the pointer across until it is placed above resistor R1, then release the mouse button. The connecting line drawn initially at an angle will disappear and a neatly drawn, right angle connection will be made for you. The important thing to remember is that R0 was the start and R1 was the end of this right angle line.

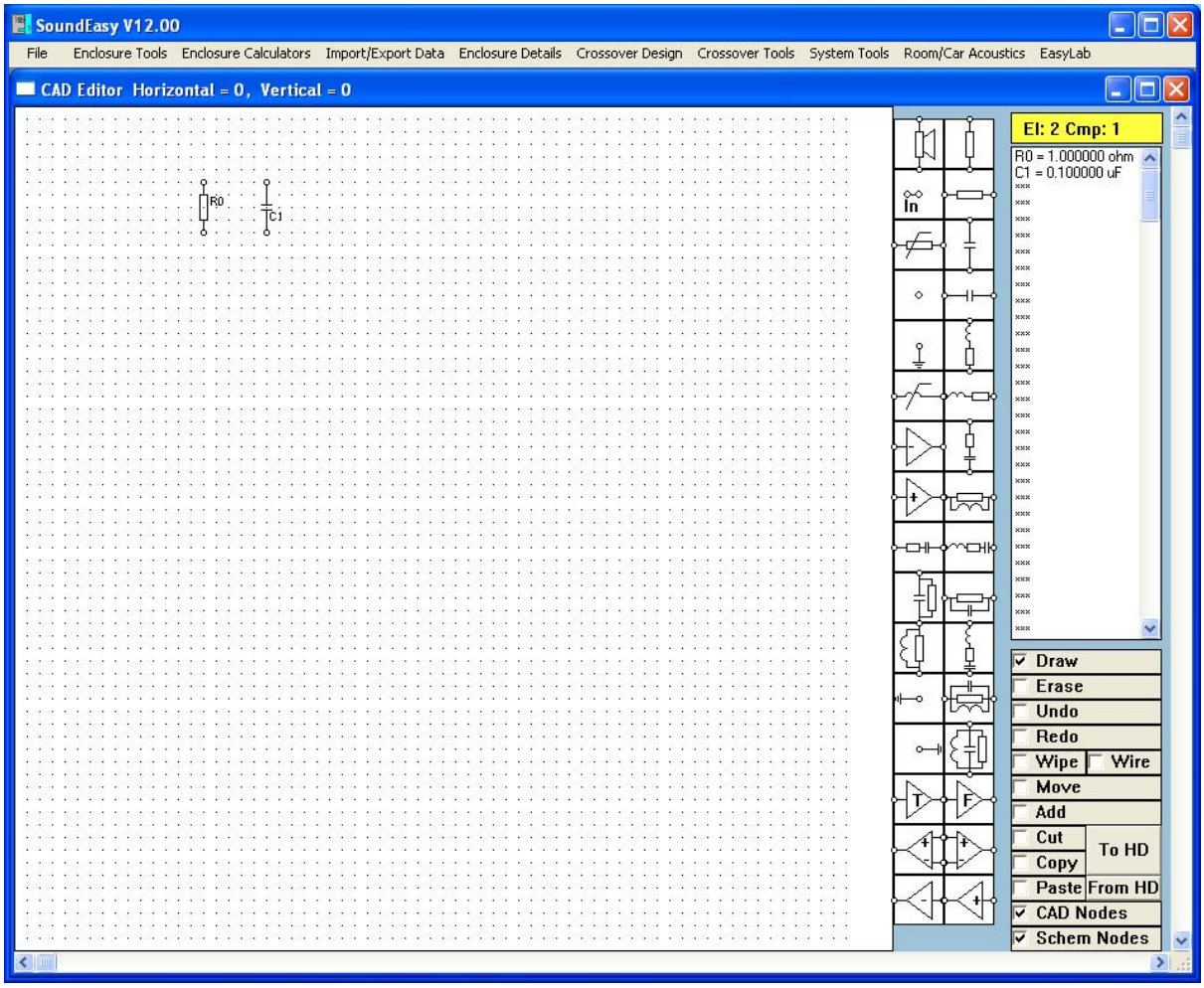


Fig 8.3 Capacitor C1 placed on the CAD drawing area

2. In order to connect resistors R2-R3, position the pointer arrow on resistor R2 first, then press and keep pressing the **LEFT** mouse button. Next, drag the pointer across until it is placed above resistor R3, and release the mouse button. The connecting line drawn initially at an angle will disappear and a neatly drawn, right angle connection will be made for you. The important thing to remember is that R2 was the start and R3 was the end of this right angle line.

3. In order to connect resistors R4-R7, position pointer arrow on resistor R7 first, then press and keep pressing the **LEFT** mouse button. Next, drag the pointer across, until it is placed above resistor R4, then release the mouse button. Again, the connecting line drawn initially at an angle will disappear and a neatly drawn, right angle connection will be made for you. Again it is important to remember that R7 was the start and R4 was the end of this right angle line.

4. In order to connect resistors R5-R6, position pointer arrow on resistor R6 first, then press and keep pressing the **LEFT** mouse button. Next, drag the pointer across until it is placed above resistor R5, release the mouse button. The line drawn initially will disappear and a neatly drawn, right angle connection will be made for you. It is important to remember is that R6 was the start and R5 was the end of this right angle line.

5. In order to connect resistors R8-R9, position pointer arrow on resistor R8, then press and keep pressing the **LEFT** mouse button. Next, drag the pointer to the right until it is placed above resistor R9, release the mouse button. The line drawn initially will be replaced by a neatly drawn, connection.

6. In order to connect resistors R10-R11 position pointer arrow on resistor R10, then press and keep pressing the **LEFT** mouse button. Next, drag the pointer down until it is placed above resistor R11, release the mouse button. The line drawn initially will be replaced by a neatly drawn, connection.

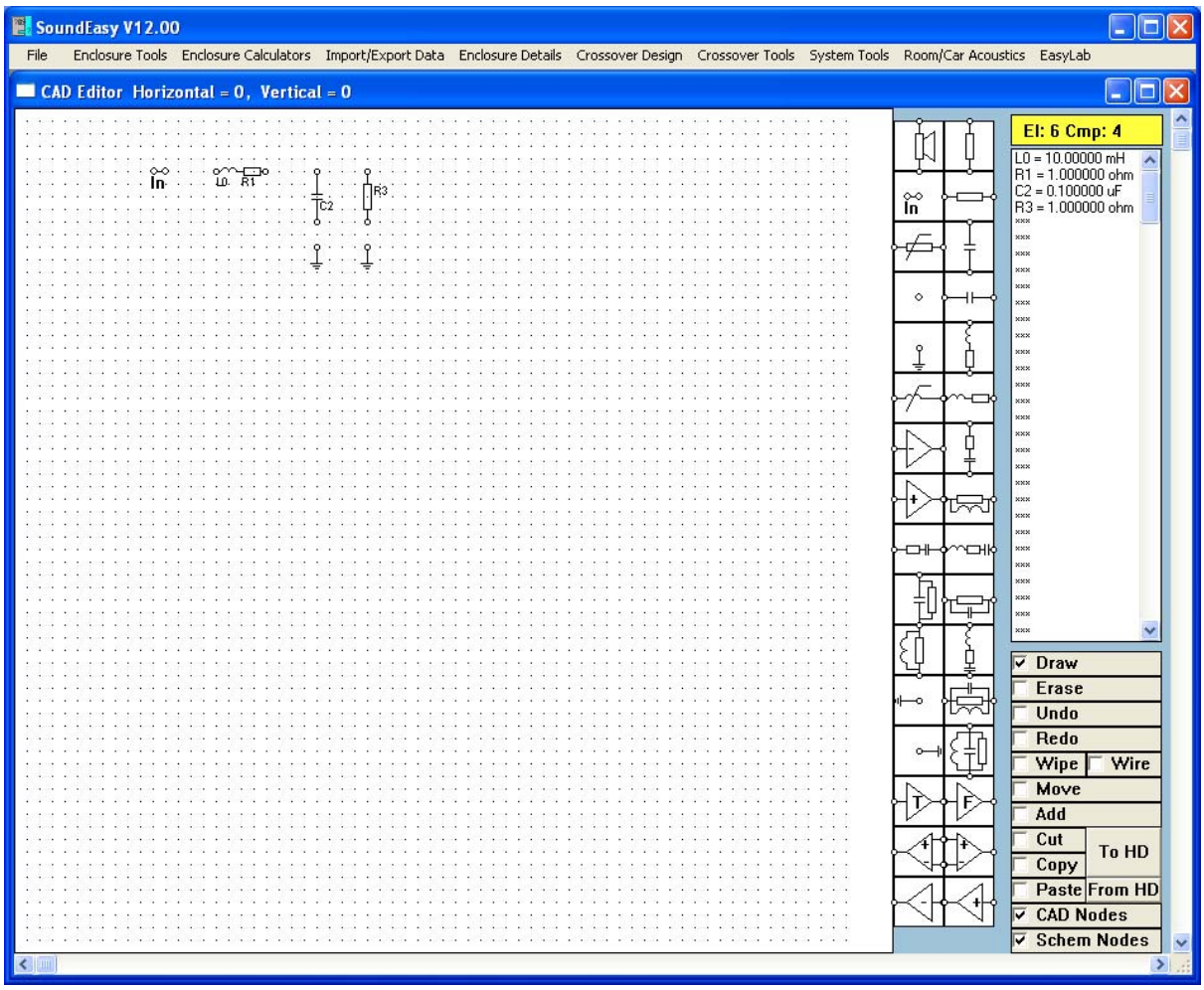


Fig 8.4 All LP filter components placed

7. In order to connect resistors R12-R13-R14, pick the "soldering point" node from the list displayed on the right hand side (the little circle) and place the node in the middle between R12 and R13 resistors. Next, position the pointer arrow on resistor R12, then press and keep pressing the **LEFT** mouse button. Drag the pointer to the right until it is placed above the node, release the mouse button. Place the pointer above the node, press and keep pressing the **LEFT** mouse button while moving the pointer right to the resistor R13. Release the mouse button. Place the pointer above the node, press and keep pressing the **LEFT** mouse button while moving the pointer down to the resistor R14. Release the mouse button. The line drawn initially will be replaced by a neatly drawn, connection. The node or "soldering point" must be placed before the connecting line was drawn. Having mastered the technique of connecting components, the user is now ready to continue with the example schematic in Fig 8.6. Place the arrow pointer on resistor R0 and press the **LEFT** mouse button, then drag the pointer to the node and release the mouse button. A left angle line is drawn connecting R0 and node - Node “0” was created. Continue with the rest of the connections following Figs 8.7 to 8.9. There is no need to exercise great accuracy when positioning the pointer arrow. The 'Crossover' tool cleans the drawing and makes all connections the correct way. Certain connections are not allowed. For example, R1-R8 cannot be connected or R6-R12 can not be connected without a node. If R0 was not there, one could place a node at the square 45 deg up and right from R6 and then connect R6 and R12 via the node. A fully developed schematic is shown in Fig 8.8. “Crossing the wires” is allowed and two nodes are assigned by the program - see Fig 8.5, R15,R16,R17,R18.

Note 1: Every network must have an Input node and some ground connections. Test signal is always injected between the Input node and the ground and measured or monitored at the nominated node and the ground (or two nominated nodes for floating loads).
Note 2: Data file for each driver can hold up to 256 elements (components + solder joints + ground connections + input node) – DO NOT exceed this number.

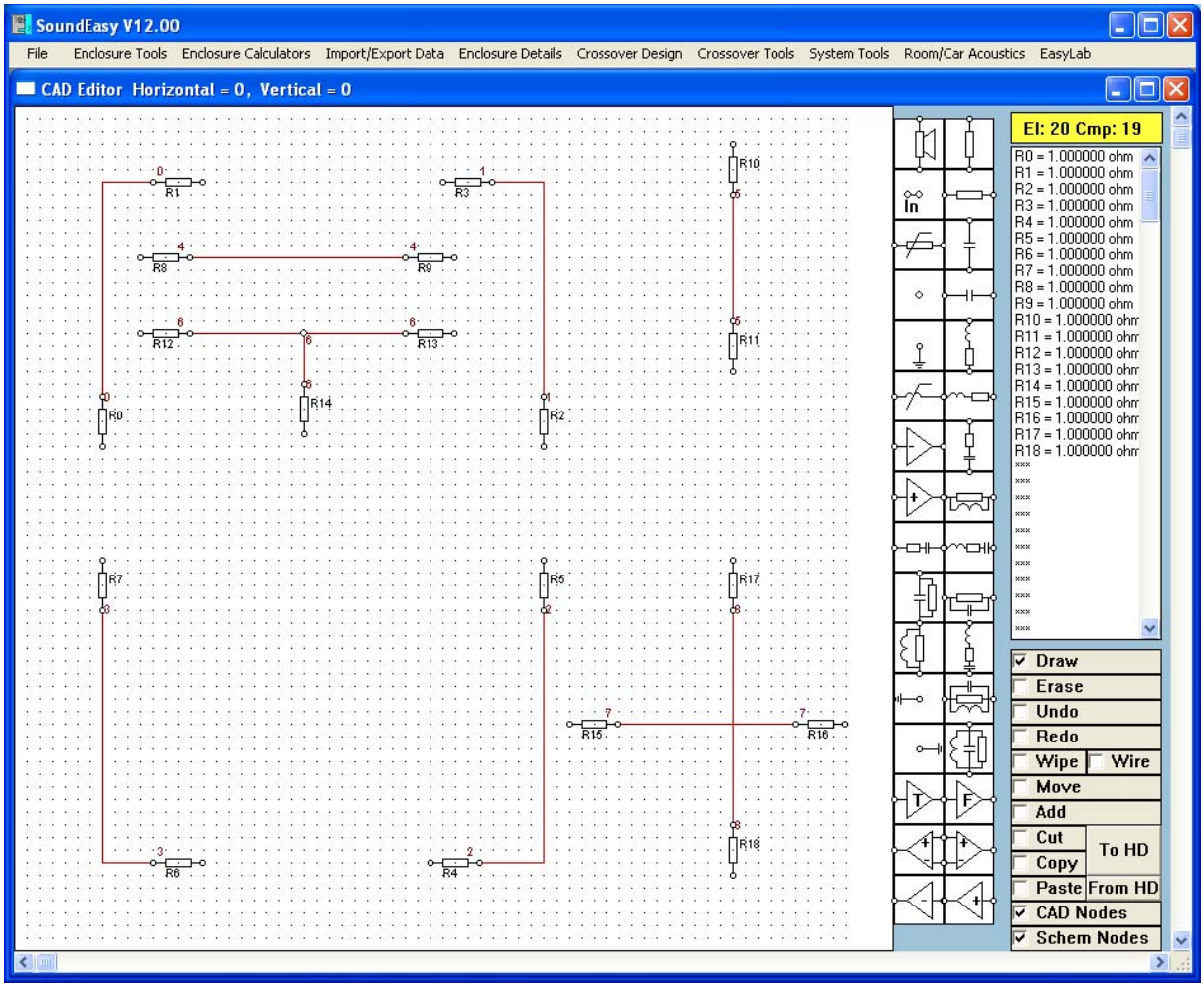
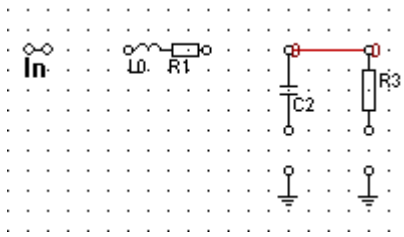


Fig 8.5 Various connection types



Please note, that link connects two nearest component terminals.

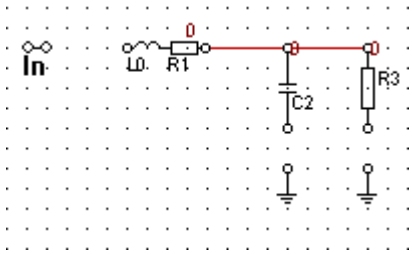


Fig 8.6 Making progressive connections between components

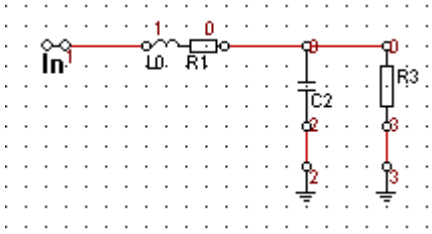


Fig 8.7 All connections for low-pass filter have been made.

Entering Component Values

Please note, that starting from V12, component values are edited using the RIGHT MOUSE BUTTON.

The final step in creating the network is assigning values to the components. In order to assign a value to a component, position the arrow pointer above the required component and **CLICK** the **RIGHT** mouse button. As a result of this, a window will be opened with 1, 2 or 3 editable fields for entering the required value. The entered data must be a positive, real number. When finished, click on the "Done" button. Enter the value of the each component in the same way. The entered data is displayed under the "Parts List" heading as soon as the data entry window is closed. When all the values are assigned, the network is now ready for analysis (see Fig 8.7).

Driver CAD symbol must be associated with appropriate file data. For instance, woofer data must be associated with woofer CAD symbol - **CLICK** the **RIGHT** mouse button over the driver symbol.

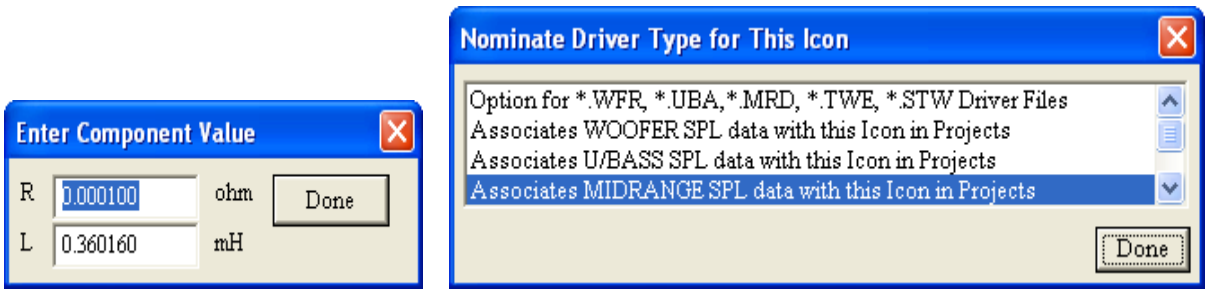


Fig 8.8 Typical Component Value Entry Box + Associating driver type with **driver icon** on the schematic

Correcting Simple Error

The schematic capture technique which was implemented in the 'Crossover' tool allows the user to perform some simple corrections on the schematic. Most of the potential errors when connecting the components will be intercepted by the module before they are allowed to occur. If an unacceptable connection is made, the program erases the wrong line and flashes an error message window. However, the program will not detect a problem if the wrong type of component was placed on the drawing area. The program allows the user to replace one component with another one dragged and placed on the top of the first (wrong) one, if the new component is of the same orientation as the old one. It can be noticed, that some components listed on the right hand side of the "Edit" window lend themselves more easily to "series" connection with the signal flow (left and right terminals) and some components are of the "parallel" type, which is easier to connect as a shunt element to the ground (top and bottom terminals).

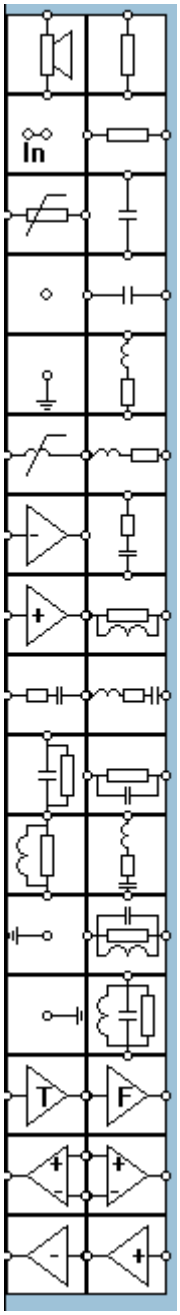
Reversed OPAMPS in CAD system

Component list now includes the following elements:

1. Resistor – 2 orientations
2. Capacitor – 2 orientations
3. Serial L-R circuit – 2 orientations
4. Serial R-C circuit – 2 orientations
5. Parallel L-R circuit – 2 orientations
6. Parallel R-C circuit – 2 orientations
7. Serial R-L-C circuit – 2 orientations
8. Parallel R-L-C circuit – 2 orientations
9. Loudspeaker element – you can select woofer / upper bass / midrange / tweeter / upper tweeter driver files to be included in the analysis for SPL/ phase and time domain.
10. Input Node
11. Variable resistance element (for Zin model)
12. Solder Joint
13. GND Node – 3 orientations.
14. Inverting OPAMP – 2 orientations
15. Non-Inverting OPAMP – 2 orientations
16. Differential OPAMP – 2 orientations
17. "T-Component" – Super Component. Assumes identity of a large number of filters, circuits and importer transfer functions.
18. "F-Component" – Implements filter transfer functions and Mimics 41 functions of Behringer 2496.

All components have been provided with two orientations. This allows you to see how the component will look in the schematic before it's placed.

Please note, that in addition to these components, you have over 100 built-in filters, crossovers and other circuits to choose from, when designing your schematic.



Reversed OPAMPS →

Fig 8.9 Menu "Pick" for CAD system.

CAD Editing Functions

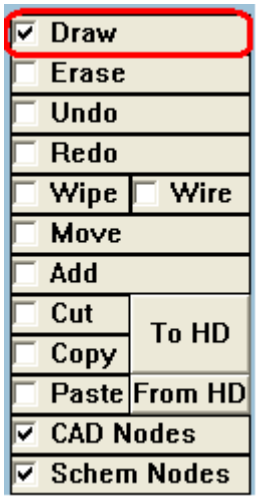
SoundEasy now includes 17 CAD processing related functions:

1. **Draw** – For "pick-and-place" components and drawing connections
2. **Erase** – For erasing single connection or component
3. **Overlap** – Changing components without disconnecting or moving them
4. **Undo / Redo -** For "undoing" the last 4 steps, and cancelling the last Undo operation
5. **Wipe** – For disconnecting and erasing the component
6. **Wipe Wire** – For erasing wires
7. **Move** – For disconnecting and moving component around
8. **Add** – For adding a complete, pre-calculated filter or even a complete crossover to your schematic
9. **Cut** – For cutting a section of the schematic
10. **Cut To Hard Disk** – For cutting a section of the schematic and saving it to HD
11. **Copy** – For duplicating a section of the schematic
12. **Copy To Hrd Disk** – For duplicating a section of the schematic and saving it to HD
13. **Paste** – For pasting the Cut section
14. **Paste From Hard Disk** – For pasting the Cut section from HD.
15. **Cad Nodes** – For switching ON/OFF dots (nodes) on CAD area.
16. **Schematic Nodes -** – For switching ON/OFF node numbering on CAD area.
17. In addition, there are over 100 **in-built filters and crossovers** to choose from. This approach allows you to use a calculator for deriving component values, so when the selected circuit is placed on the CAD screen, it will come fully functional. Possibly the simplest way of creating a filter is to load one of the in-built designs and modify it for your specific purpose. The process will be described in the next chapter.

When using functions like: loading in-built circuits, Add, Cut/Paste, Copy/Paste, Paste From HD, the selected circuit is placed in a buffer, so you can move the circuit around the CAD screen (while pressing the left mouse button) and place the circuit where you want it by releasing the left mouse button. It is likely, that you will find yourself using this functionality quite often.

1. Draw Function

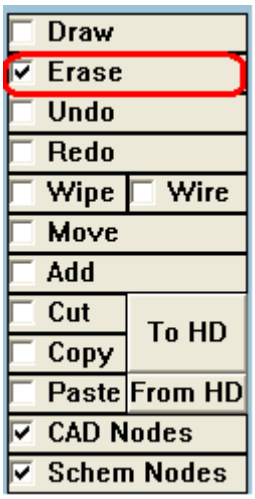
The Draw function is the possible the most fundamental function of all, as it allows you to create your schematics. You can pick-and-place selected component and also link the components already placed on the CAD screen. The function is activated by placing the mouse pointer on the "Draw" check box and pressing the left mouse button. You should now see the "checkmark" as shown below. Now you can move the mouse pointer onto the component's list item and press the left mouse button to select your choice. Keep the left mouse button pressed down, so you can pick the component and move it to where you want it on the CAD screen. Once the component is positioned, please release the left mouse button – the component you picked is now drawn on the schematic.



If you have two or more components placed on the CAD screen, you can connect them together with a link. The link is created by placing the mouse pointer tip on one of the component's nodes, and while the left mouse button is pressed down, you drag the mouse pointer to the other component's node to be connected. Then release the left mouse button.

2. Erase Function

The Erase function is your primary tool for removing components and links from the schematic. The function is activated by placing the mouse pointer on the "Erase" check box and pressing the left mouse button. You should now see the "checkmark" as shown below.



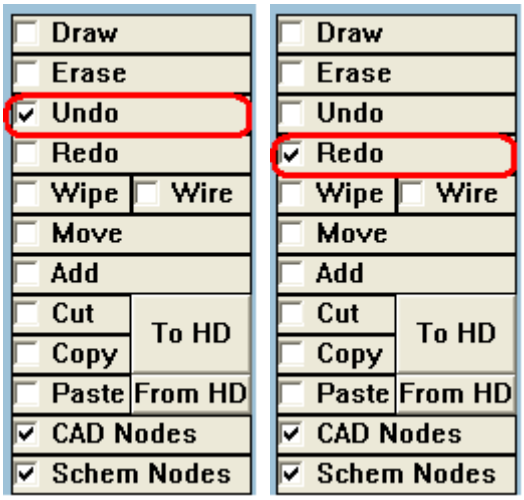
To erase a link between two components, perform the exact reverse of linking process. Place the mouse pointer tip on the start of the link, and while the left mouse button is pressed down, you drag the mouse pointer to the other component's node to be disconnected. Then release the left mouse button. Component can only be "erased" if all of it's links have been erased first. Then place the mouse pointer on the component to be erased and double-click the left mouse button. This will remove the component from the CAD screen.

3. Overlap Function

SoundEasy will accept one serial component placed on the top of another serial component and one parallel component placed on the top of another parallel component. Use the previously described "pick and place" technique to replace components. The parts list will be updated automatically. There is no button for the "Overlap Mode", just pick another component of compatible orientation and place it on the top the existing one.

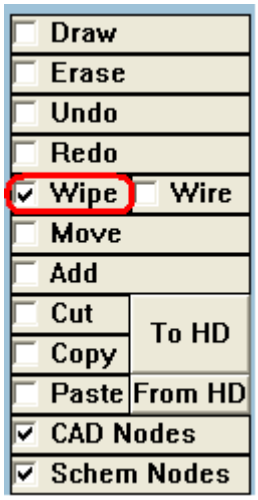
4. Undo / Redo Functions

The Undo function comes very handy if you decide to reverse the last four steps in creating your schematic. Single press on the "Undo" check box will take back one step. The program remembers the last four steps, so you can reverse all of them. If you happen to realise, that you have "undone" one change unnecessarily, you can use the Redo function to reverse the Undo action.



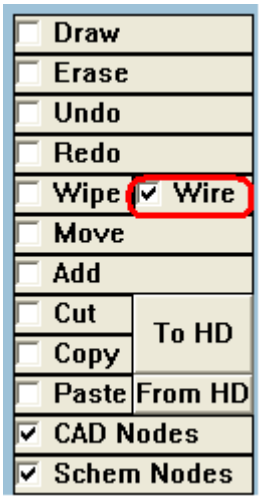
5. Wipe Function

Wipe function is a productivity-type function. In some circumstances, you may decide to remove a component from the schematic completely and permanently. You could use the Erase function to remove links one-by-one and then erase the component. Or, you can use the Wipe function to erase a component and all links connected to it. The function is activated by placing the mouse pointer on the "Wipe" check box and pressing the left mouse button. You should now see the "checkmark" as shown below. Then place the mouse pointer on the component to be removed and double-click the left mouse button. This will remove the component + all it's links from the CAD screen.



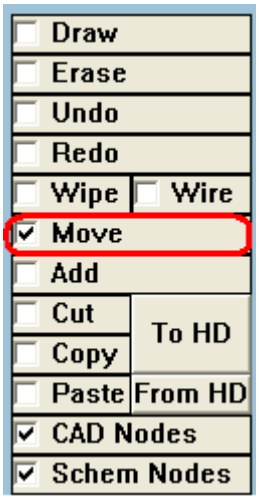
6. Wipe Wire Function

Wipe Wire function is a productivity-type function as well. If you find yourself in a situation, where you need to erase many links quickly, the Wipe Wire function will set the program in "wire removing mode", and then you can double-click on any wire to have it removed. The function is activated by placing the mouse pointer on the "Wire" check box and pressing the left mouse button. You should now see the "checkmark" as shown below. Then place the mouse pointer on the link to be removed and double-click the left mouse button. This will remove the link from the CAD screen.



7. Move Function

Components can be moved around the CAD screen freely. The component will stay connected and the links, that are affected by the move are highlighted in green. The function is activated by placing the mouse pointer on the "Move" check box and pressing the left mouse button. You should now see the "checkmark" as shown below. Then place the mouse pointer on the component to be removed and press the left mouse button. Keep the button pressed down as you move the component. Once you move the component where you want it, release the left mouse button. This will draw the component in the new location on the CAD screen.



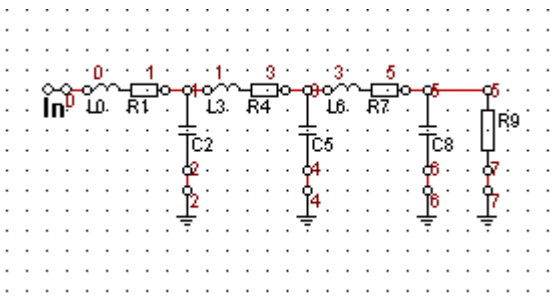


Figure 8.10 Original circuit

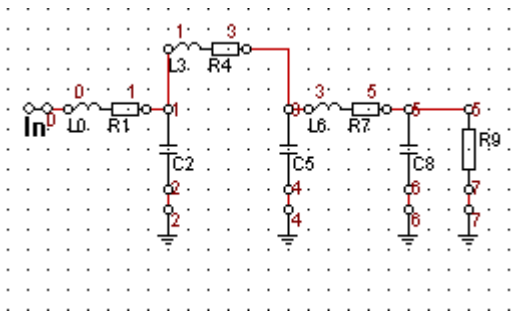


Figure 8.11 L3+R4 (lossy inductor) components have been moved.

8. Add Function

The "Add" function allows you to put the CAD system in "network add" mode. This is very powerful editing mode, as it allows you to add complete filter or crossover with all components pre-calculated.

Figure 8.12 shows the CAD screen with already uploaded 4-nd order Butterworth low-pass filter. Our goal is to create a crude band-pass filter by adding a 2-th order high-pass Linkwitz filter to it.

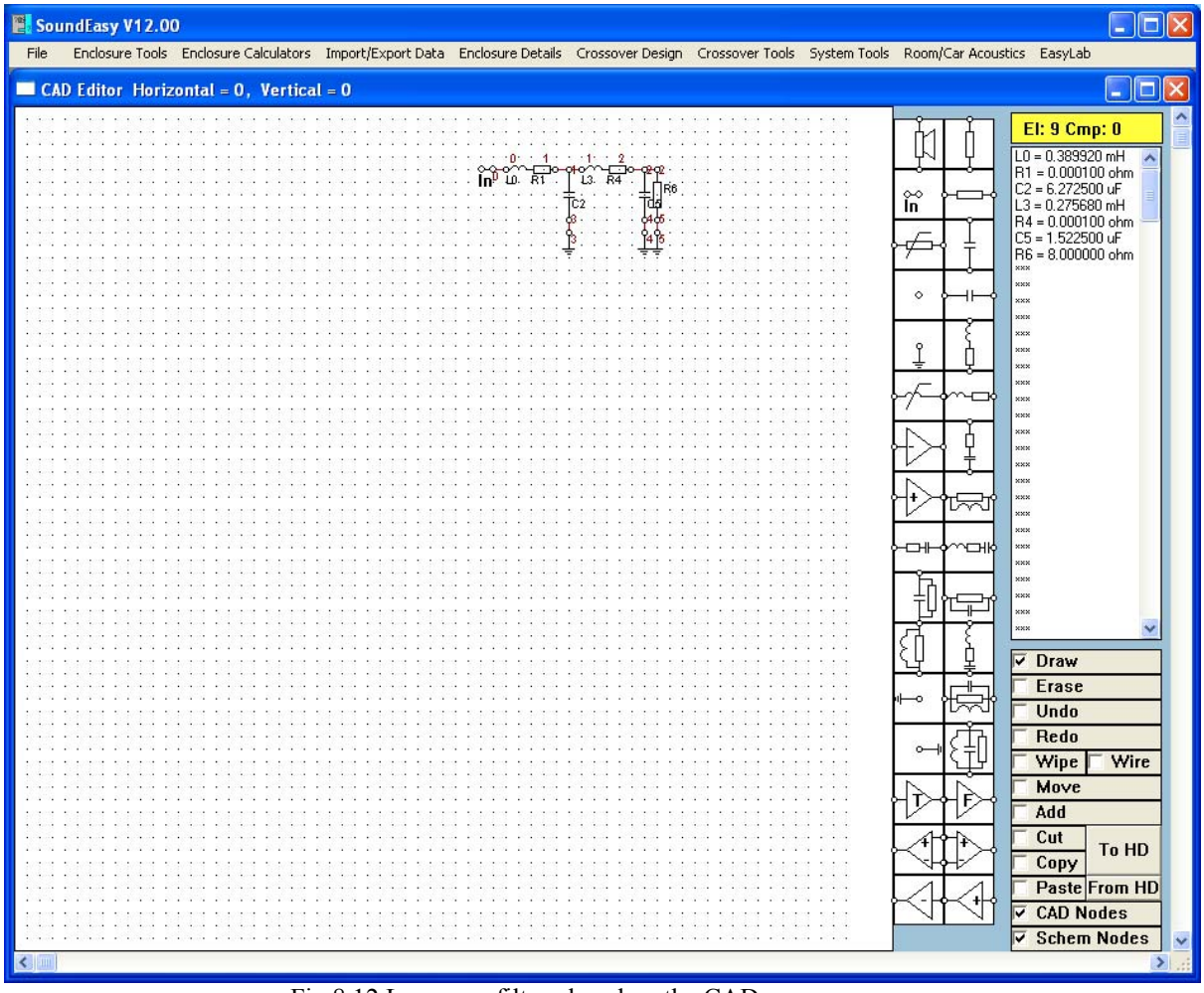


Fig 8.12 Low-pass filter placed on the CAD screen.

Next, place the CAD screen in the “Add Network” mode.

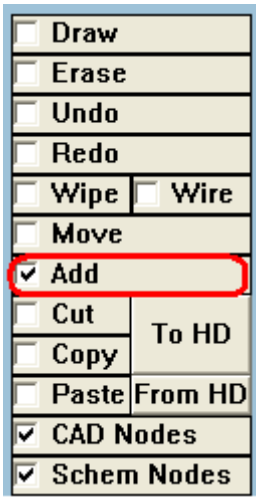


Fig. 8.13 CAD screen in “Add” mode.

Select the built-in filter you wish to use from the filters dialogue box as before. The filter will be placed at your specified location with all components calculated – see Figure 8.14

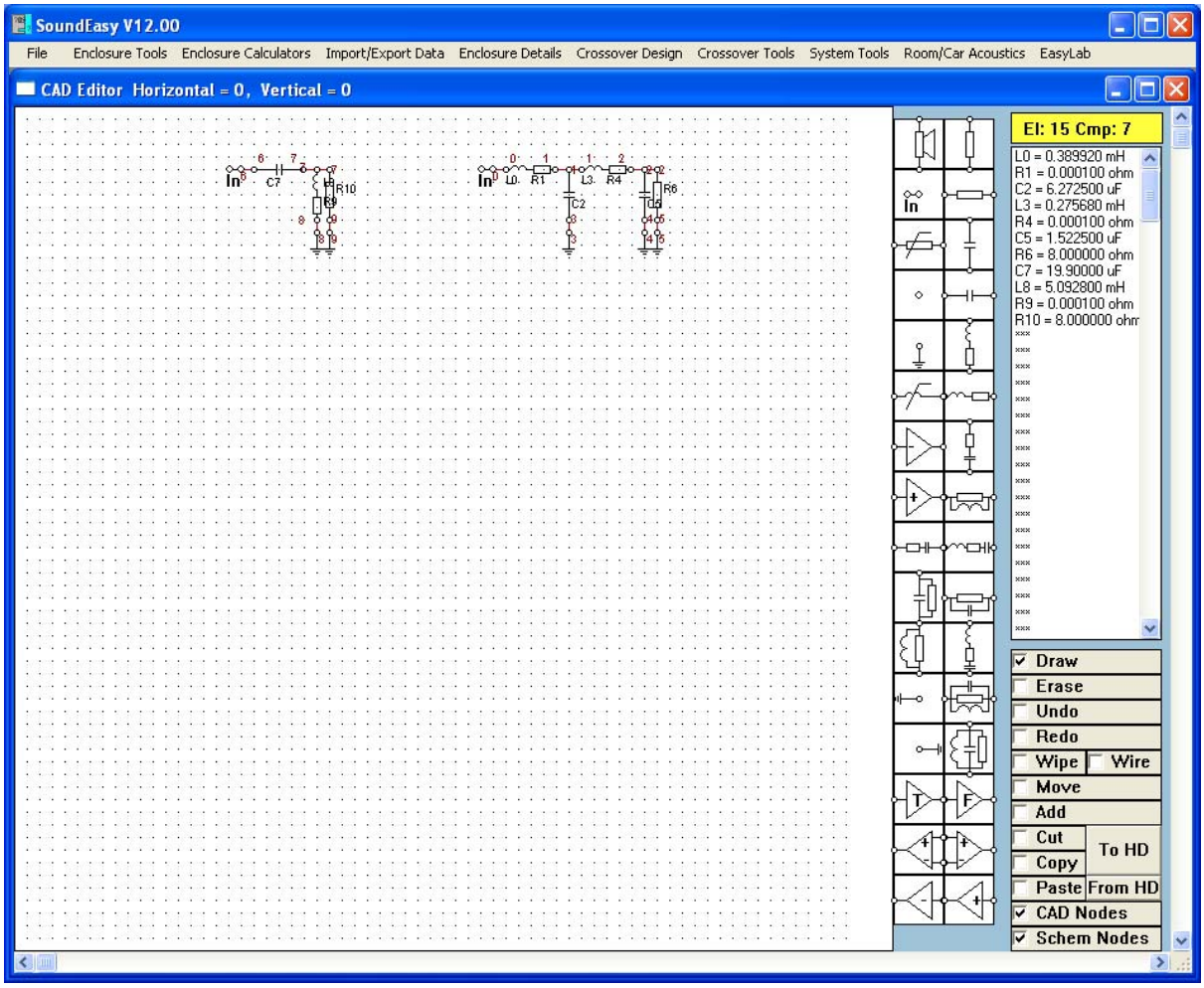


Figure 8.14 Low-pass and High-pass filters on the same screen

Please note, that you now have TWO inputs and all node numbering is duplicated – this is due to the fact, that you now have two independent circuits on one screen. Also, the first circuit has the load resistor connected to its output (R10) – and this component is now redundant.

Click on “Erase” button to erase the R10 (load for the first filter) and “IN0” (input to the second filter). Finally, click on “Draw” button and link the filters – as shown on Fig 8.15.

We have now created an asymmetric band-pass filter (uploaded 4-nd order Butterworth low-pass and 2-th order high-pass Linkwitz filters). You an continue adding more and more circuits to create quite complex asymmetrical crossover very quickly. The “Add network” function allows you to freely mix-and-match filter + new filters and crossover + new filters. You can upload a crossover and then ADD to it more filters with all components calculated. Inevitably, some more circuit manipulation will be needed to produce your final filter or crossover.

SoundEasy performs basic checks for duplicate input nodes, wrongly anchored filters. Still, we recommend, that you are sensible about selecting schematics that are too large to fit on the available CAD screen.

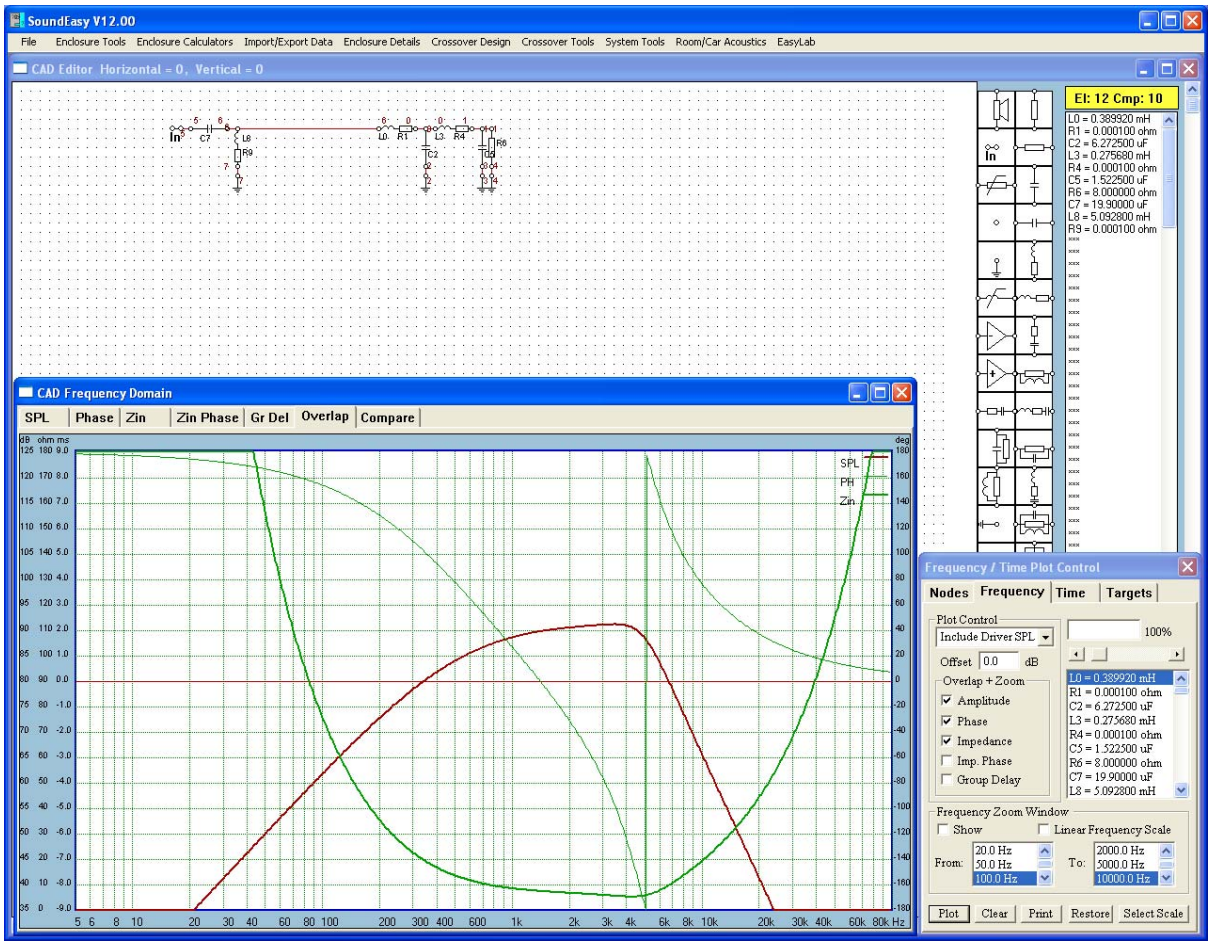
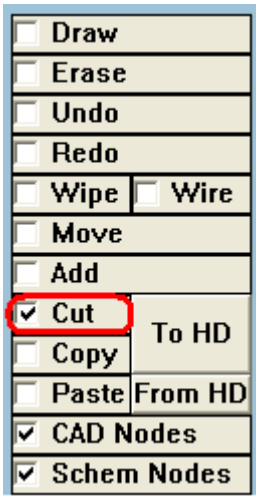


Fig 8.15 Two filters linked properly and their response as the band-pass filter

9. Cut Function

The “Cut” function allows you to quickly remove large portion of the schematic. The function is activated by placing the mouse pointer on the “Cut” check box and pressing the left mouse button. You should now see the “checkmark” as shown below. Please note, that the removed circuit is placed in a buffer, so it is still available for “Paste” operation.



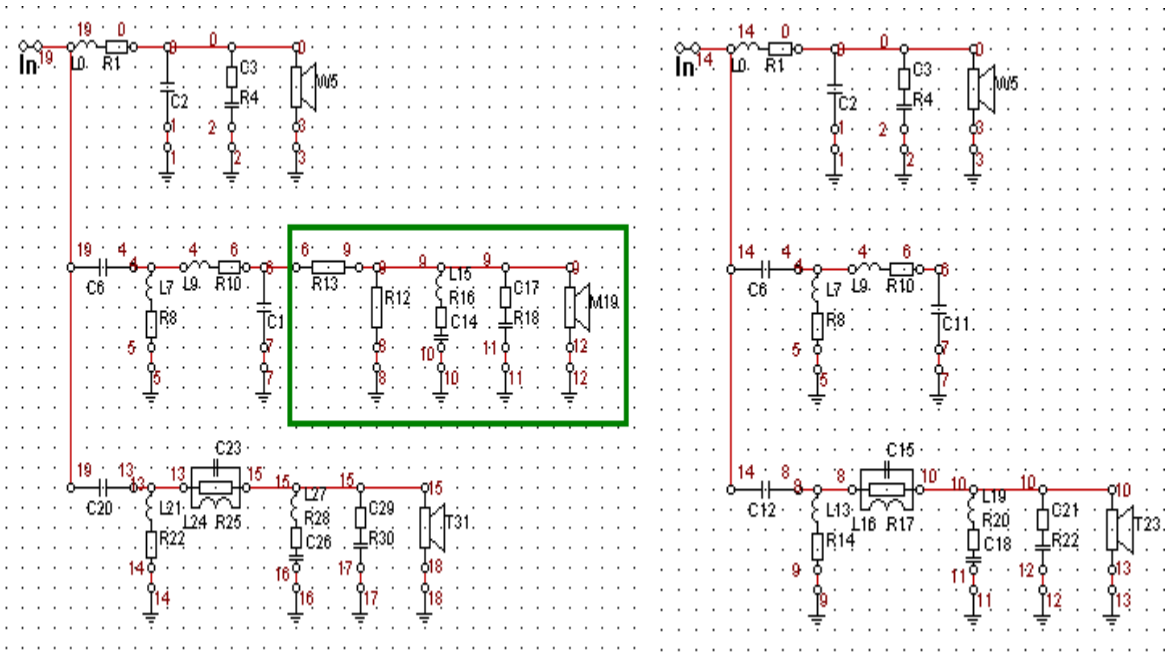


Figure 8.16. Simple schematic example “before” and “After” Cut operation

For the effect of complimentary “Paste” function see the following paragraps.

Here is how it works:

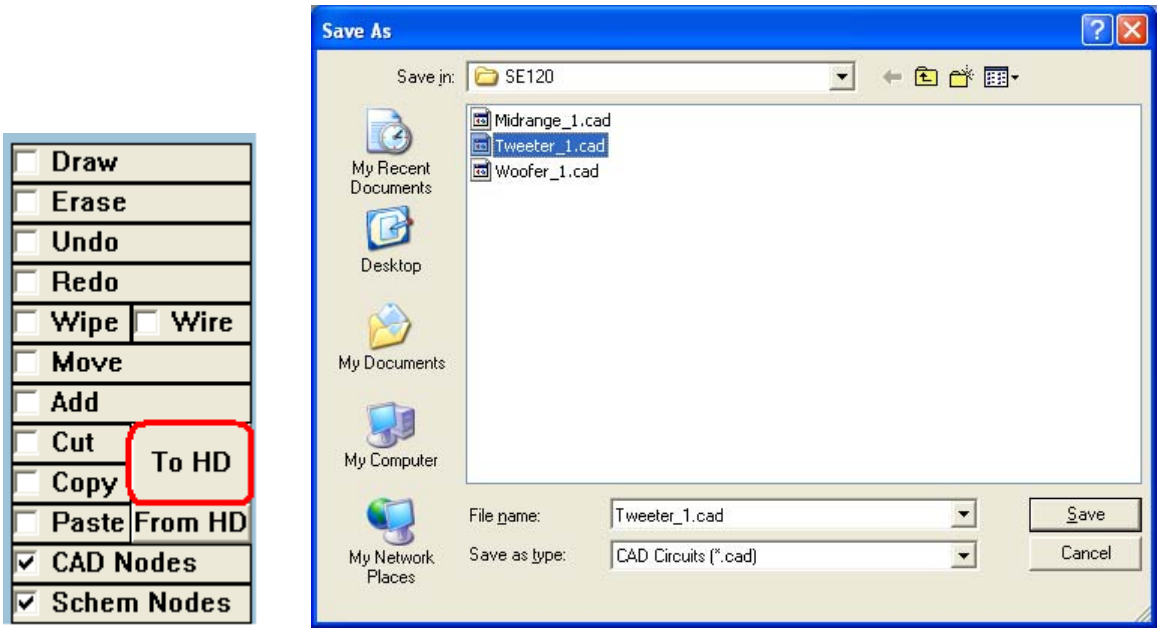
The sub network should be able to be bounded by a rectangle.

- Click on "Cut" check box.
- Place the mouse pointer on the "top-left" rectangle of the sub-network. Press the left mouse button and
- keep it pressed.
- Move the mouse pointer to the "bottom-right" rectangle of the sub-network. Release the left mouse button.

Move the mouse pointer where you want to place the sub-network. Click on the left mouse button (and keep it pressed), you can move the circuit around. This is your chance to make sure, that the sub-network is placed conveniently and will not overlap any other components. Release the left mouse button. You should be able to make **single "cut" and multiple "paste" operations**. All component values should be replicated.

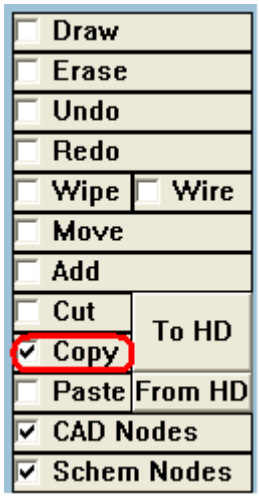
10. Cut To Hard Disk Function

The "Cut To HD" function allows you to quickly remove large portion of the schematic and save it to your Hard Disk. The function is activated by pressing the "To HD" button. Please note, that you should "Cut" the circuit first, before selecting "To HD" button. Once the button is pressed, the standard "Save As" dialogue box will be invoked to allow you to enter a file name for the cut circuit. The circuits are saved with ".CAD" extension – see below.



11. Copy Function

The "Copy" function allows you to quickly duplicate large portion of the schematic. The function is activated by placing the mouse pointer on the "Copy" check box and pressing the left mouse button. You should now see the "checkmark" as shown below. Please note, that the removed circuit is placed in a buffer, so it is available for complimentary "Paste" operation.



Here is how it works:

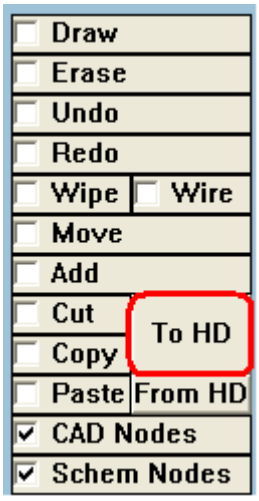
The sub network should be able to be bounded by a rectangle.

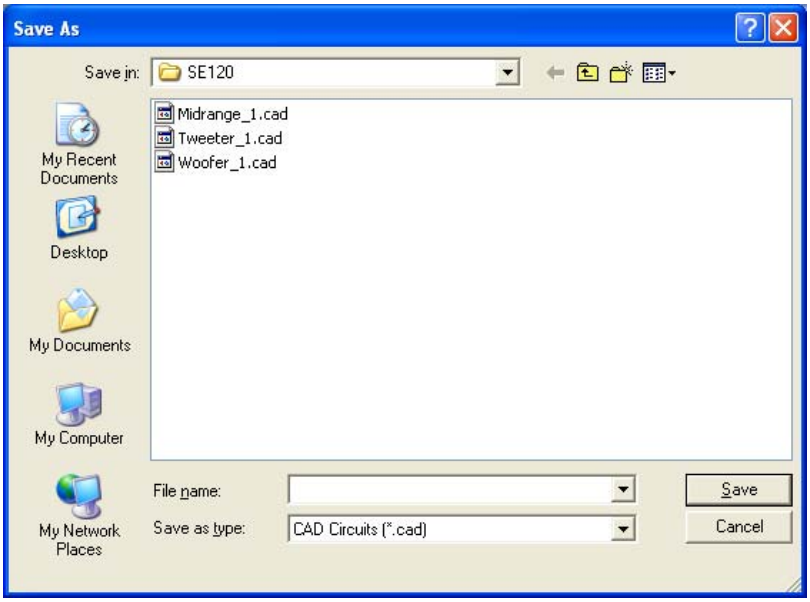
- Click on "Copy" check box.
- Place the mouse pointer on the "top-left" rectangle of the sub-network. Press the left mouse button and
- keep it pressed.
- Move the mouse pointer to the "bottom-right" rectangle of the sub-network. Release the left mouse button.

Move the mouse pointer where you want to place the sub-network. Click on the left mouse button (and keep it pressed), you can move the circuit around. This is your chance to make sure, that the sub-network is placed conveniently and will not overlap any other components. Release the left mouse button. You should be able to make **single "cut" and multiple "paste" operations**. All component values should be replicated.

12. Copy to Hard Disk Function

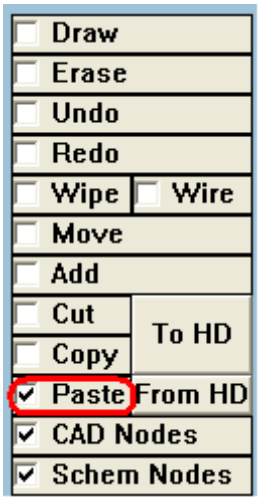
The “Copy To HD” function allows you to quickly duplicate large portion of the schematic and save it to your Hard Disk. The function is activated by pressing the “To HD” button. Please note, that you should “Copy” the circuit first, before selecting “To HD” button. Once the button is pressed, the standard “Save As” dialogue box will be invoked to allow you to enter a file name for the cut circuit. The circuits are saved with “.CAD” extension – see below.





13. Paste Function

The “Paste” function allows you to transfer the circuit from a temporary buffer onto the schematic. Obviousle, the circuit to be pasted, mush have been placed into the buffer by Copy or Cut functions already. The function is activated by placing the mouse pointer on the “Paste” check box and pressing the left mouse button. You should now see the “checkmark” as shown below.



Move the mouse pointer where you want to place the sub-network. Click on the left mouse button (and keep it pressed), so you can move the circuit around. This is your chance to make sure, that the sub-network is placed conveniently and will not overlap any other components. Release the left mouse button. You should be able to make **single "cut" and multiple "paste" operations**. All component values should be replicated.

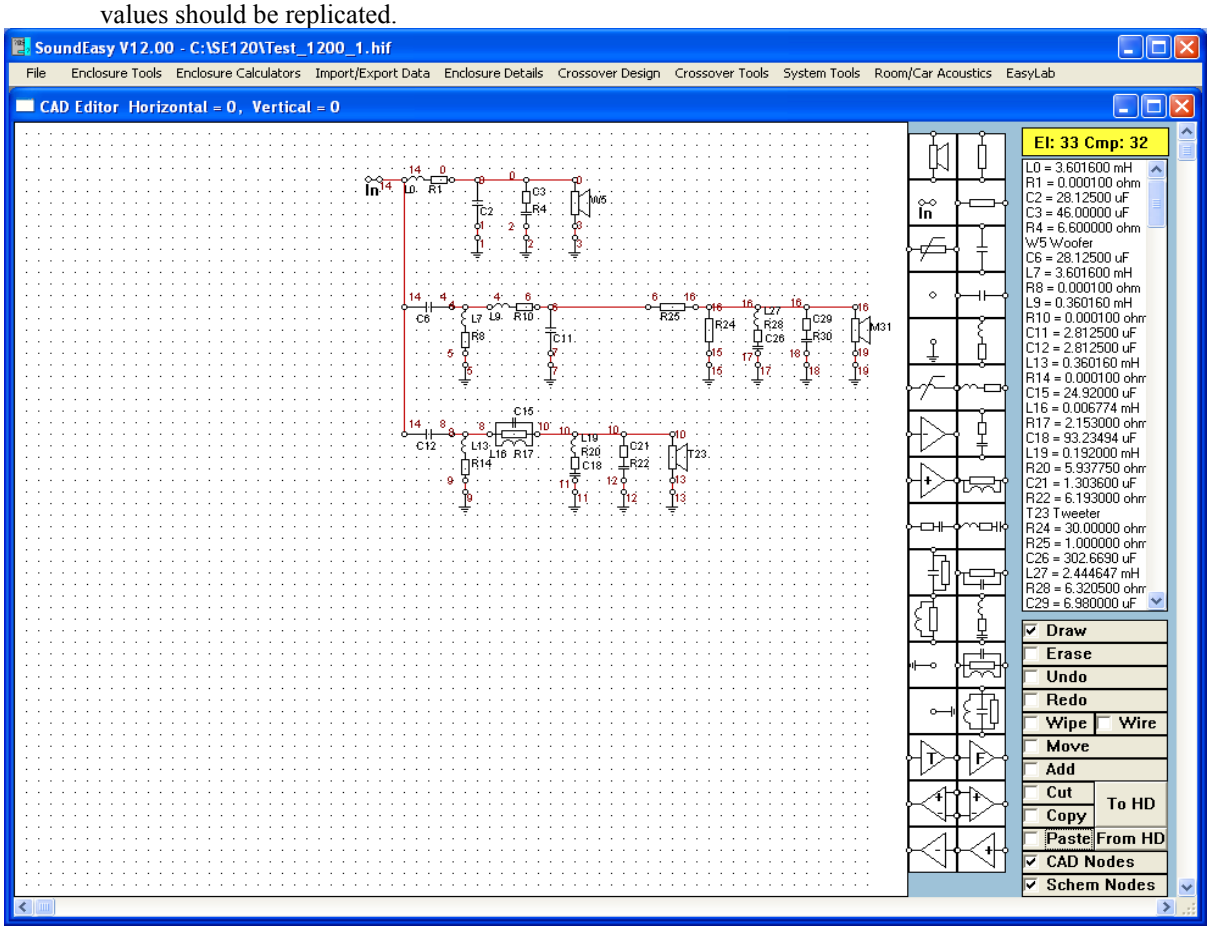
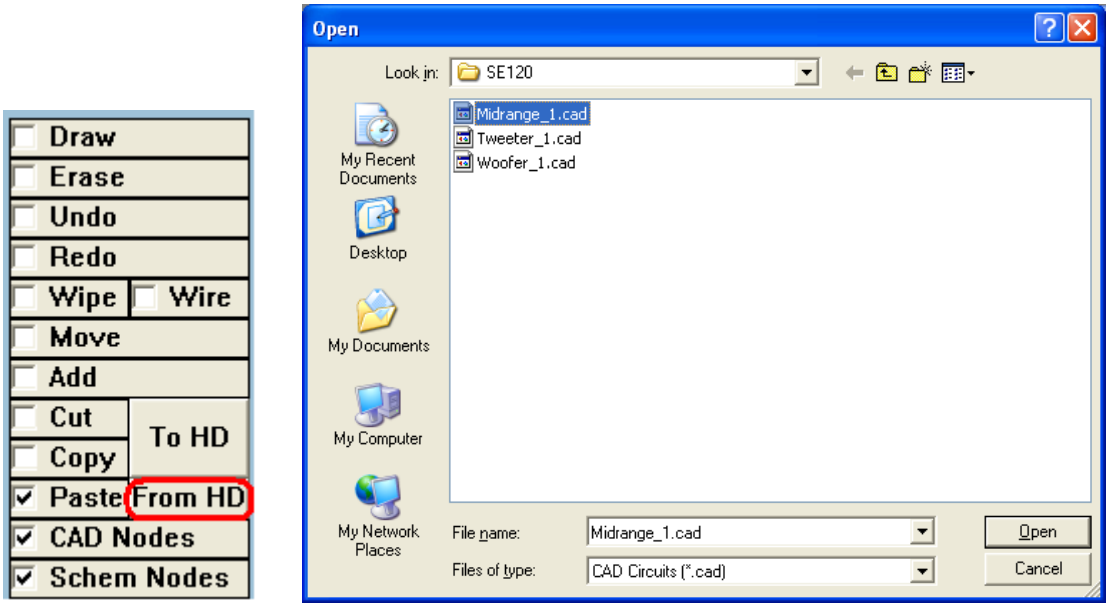


Figure 8.17. Simple schematic example “after” Cut-and-Paste.

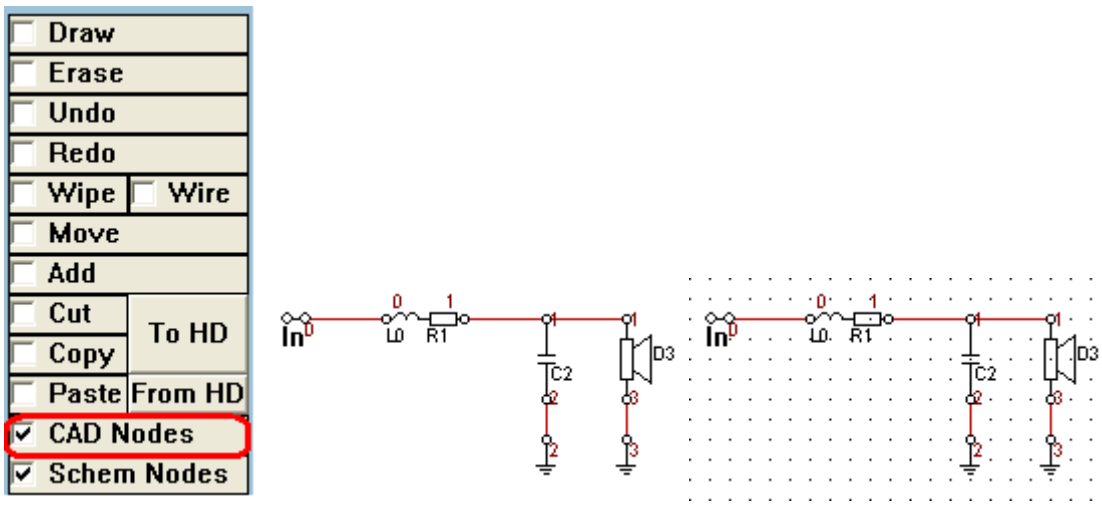
14. Paste From Hard Disk Function

The “Paste” function allows you to transfer the circuit from a temporary buffer onto the schematic. Obviousle, the circuit to be pasted, mush have been saved into the HD by Copy or Cut functions already. The function is activated by bressing the “From HD” button. Once the button is pressed, the standard “Open” dialogue box will be invoked to allow you to enter a file name for the cut circuit. The circuits are saved with “.CAD” extension – see below.



Once you select and load the circuit, move the mouse pointer where you want to place the sub-network. Click on the left mouse button (and keep it pressed), so you can move the circuit around. This is your chance to make sure, that the sub-network is placed conveniently and will not overlap any other components. Release the left mouse button. You should be able to make **single "cut" and multiple "paste" operations**. All component values should be replicated.

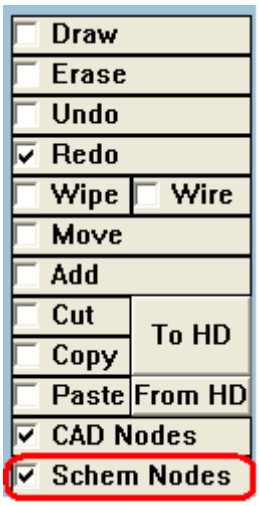
15. Cad Nodes Function

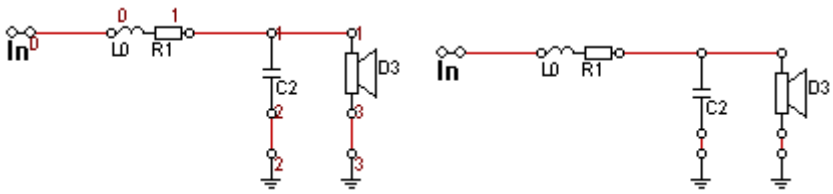


The “CAD Nodes” function is just a convenient switch to select/deselect location nodes on the CAD screen.

When the components are moved around the CAD screen, they are automatically placed on a "nodal raster". This process manifests itself as slight "jumps" when the components are moved on the screen because the program attempts to place the components onto the nodes CLOSEST to the current position. Seeing the location of CAD nodes allows you to place the components more precisely. However, this is not strictly necessarey, as you can move the components freely after they were originally placed.

16. Schematic Nodes Function





The "Schematic Nodes" function is just a convenient switch to select/deselect the display of electrical nodes on the CAD screen. You need the node numbers to tell the program where it should monitor the voltages for modelling network behaviour.

Modelling Network Behaviour

The next step in the network design is the analysis of the network's performance. This accomplished by plotting amplitude, phase group delay and dinput impedance of the network.

Nodes Tab

Schematic shown below, has the woofer connected between **"From"** node "1" and ground "4", midrange driver connected between node "9" and ground "13" and finally the tweeter driver connected between node "16" and ground "19". The "**To**" column of nodes would be used, if the driver was connected between two nodes AND none of them would be the "ground" node. For example, "series" crossovers have at least one driver connected to two floating nodes. When you assign component values to the schematic elements, you also need to assign driver type to each driver icon on the schematic. Simply double-click the right-mouse-button on the icon and you will be prompted to enter a number for identifying the driver. Driver type will be displayed in the "**Driver Type**" column. Finally, you can switch ON/OFF any driver from the analysis by checking/un-checking the boxes in the "**ON?"** column – see Figure 8.18.

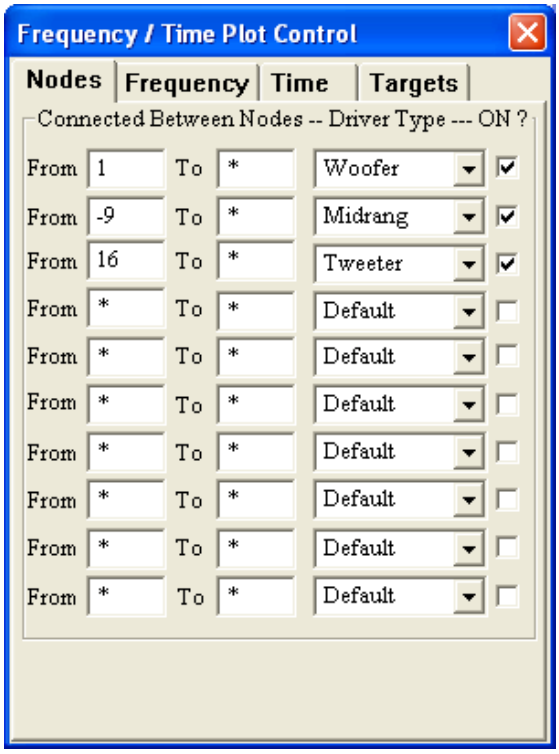
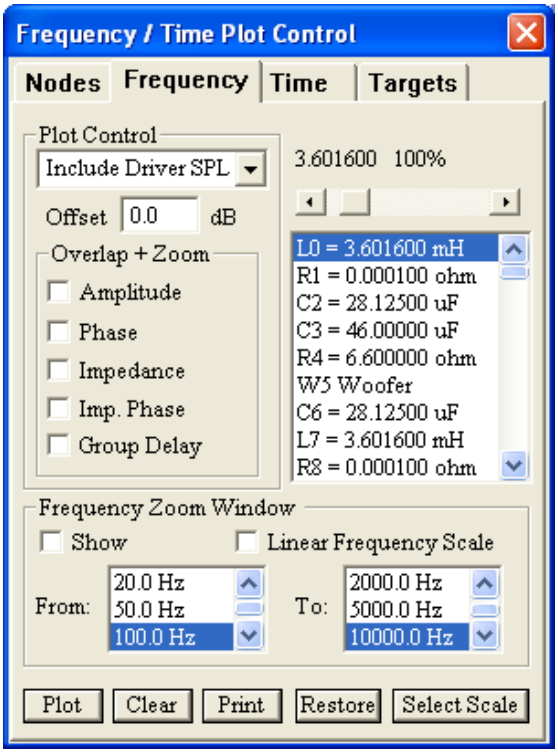


Fig 8.18 Tab for entering circuit nodes



Tab for selecting plots and component adjusting

You must enter one (for single ended loads, second Nx enter as "*") or both node numbers (for floating loads) and **numbers must be different from any ground node number**. Driver polarity is reversed simply by **reversing the sign in front of the node number(s)** – see "-" in front of midrange driver on Fig 8.18.

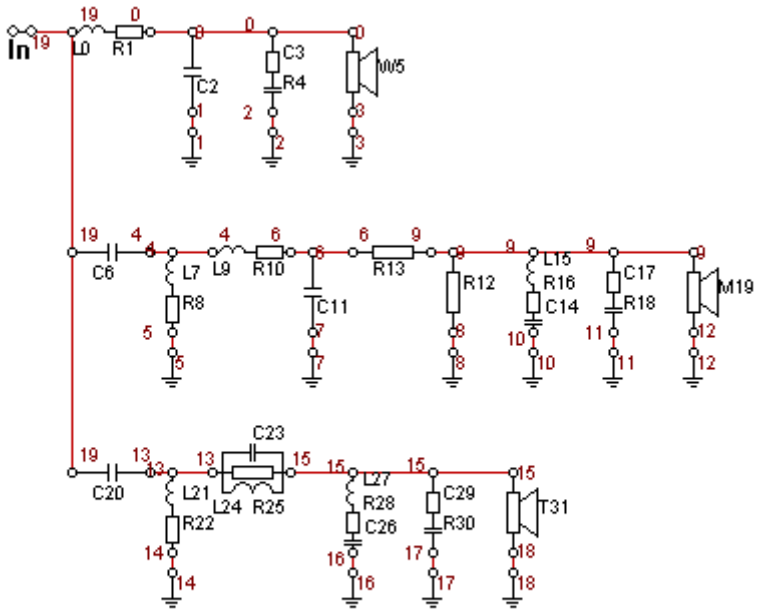


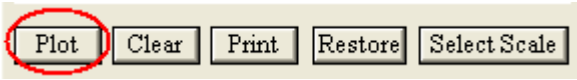
Figure 8.19. Example on nodes (red) on the schematic.

Frequency Tab

1. **Plot** – Button activates plotting for all selected functions.
2. **Clear** – Button clears the plotting window.
3. **Print** – Button opens printer dialogue box.
4. **Restore** – Button used to restore original component value (after it's value was changed by the slider).

5. **Select Scales** – Button opens dialogue box for selecting vertical scale resolution.
6. **Offset** – Editable field for offsetting SPL plots in dBs.
7. **Slider and List-box** - Double-click on the list-box component to select it into the display field above. Then, you can use the slider to adjust component's value.
8. **Plot Control** – Drop-down list box for selecting additional plotting parameters. See "**Aditional Operations"** section.

Pressing **Plot** button the **Frequency TAB** will automatically plot all 5 transfer functions on individual TABs: SPL, Phase, Zin, Zin Phase and Group Delay.



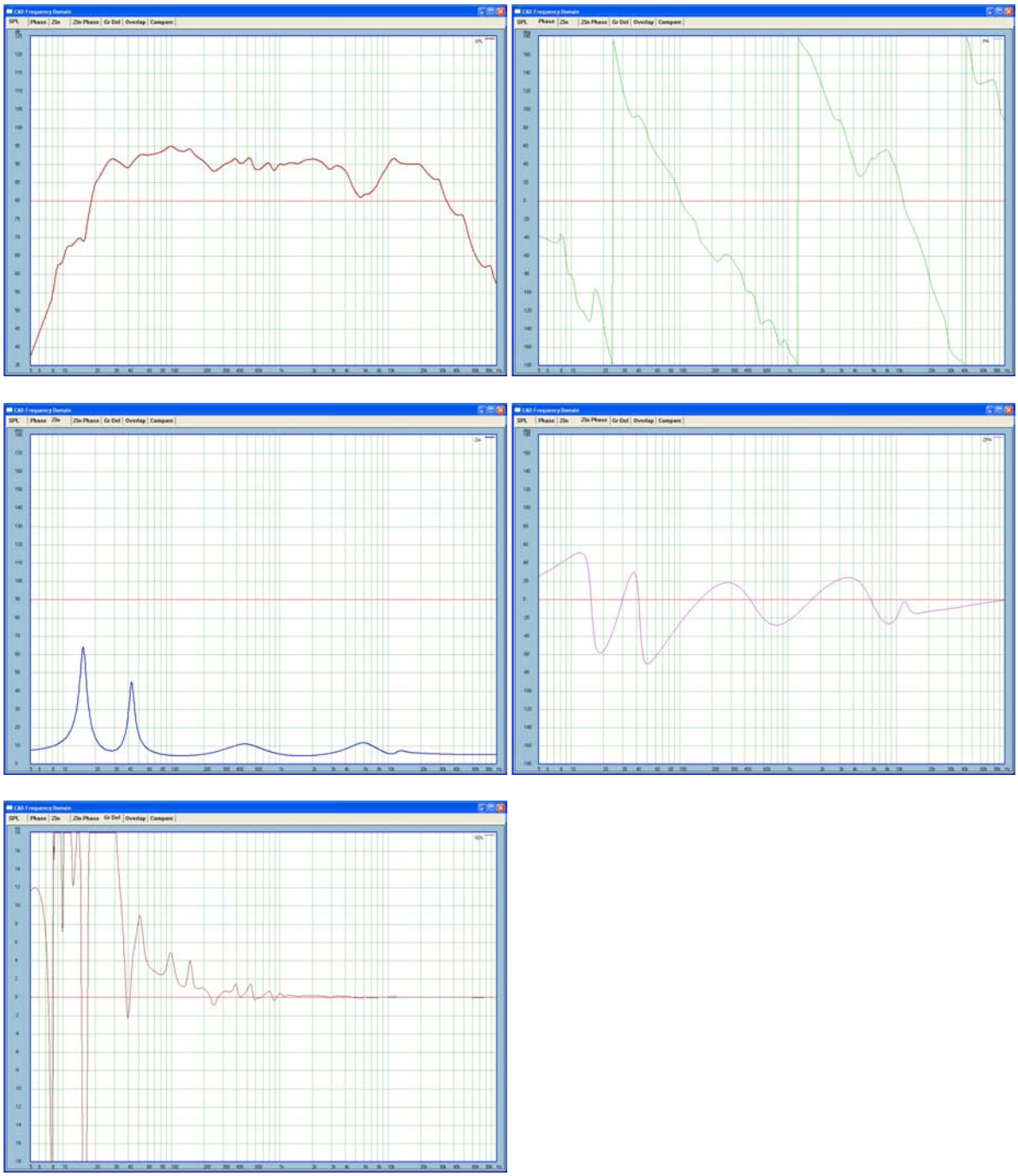


Figure 8.20 Example of 5 individual TABs plots + Overlay plot for our 3-way network.

Selecting any (or all) checkboxes on the "**Overlay + Zoom**" group **Frequency TAB** will automatically plot all 5 transfer functions on individual TABs: SPL, Phase, Zin, Zin Phase and Group Delay on the **Overlay TAB.**

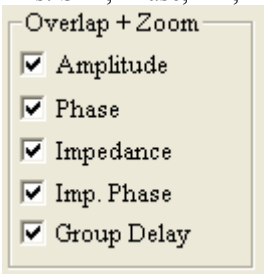
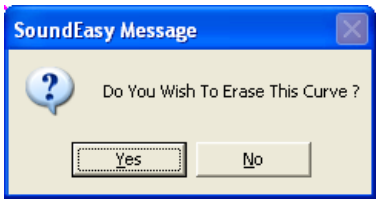


Figure 8.21 Example of 5 individual TABs plots on the Overlay TAB for our 3-way network.

Curve Compare Function

The "Compare" function allows you to cope any curve from the 5 TAB'ed plots: SPL, Phase, Zin, Zin Phase and Group Delay on the "**Compare**" TAB for easy comparason the the curves. You can select the desired curve by positioning the mouse pointer on the desired curve and pressing the left mouse button. The selected curve will be highlighted in pink. When you release the button, the curve will be transferred to the "Compare" TAB. On the "Compare" TAB you will can also erase individual curves. Once again, position the mouse pointer over the desired curve and click left mouse button. The curve will be highlighted in pink, and a message dialog will confirm, that you wish to delete this particular curve.



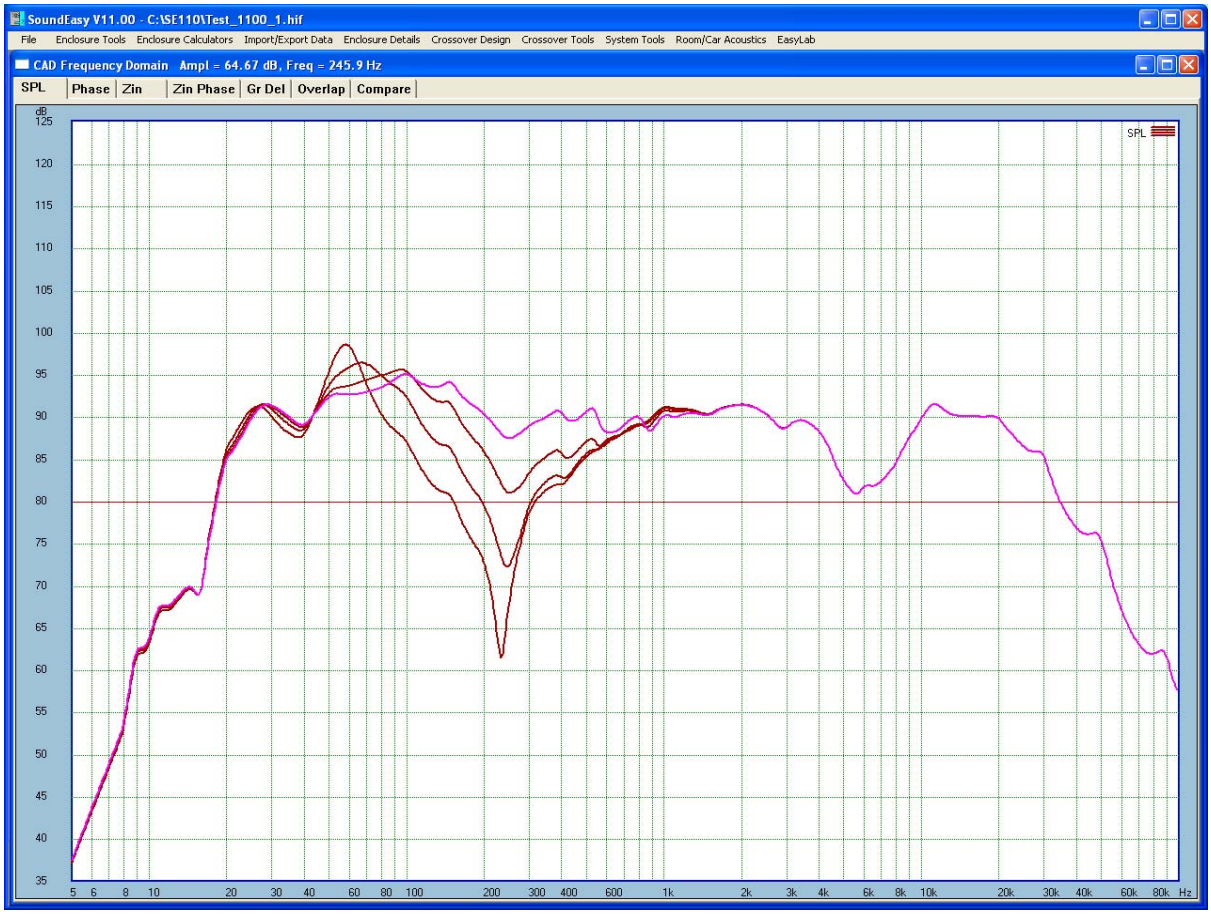


Figure 8.22 Example of selecting an SPL curve for transfer to “Compare” TAB.

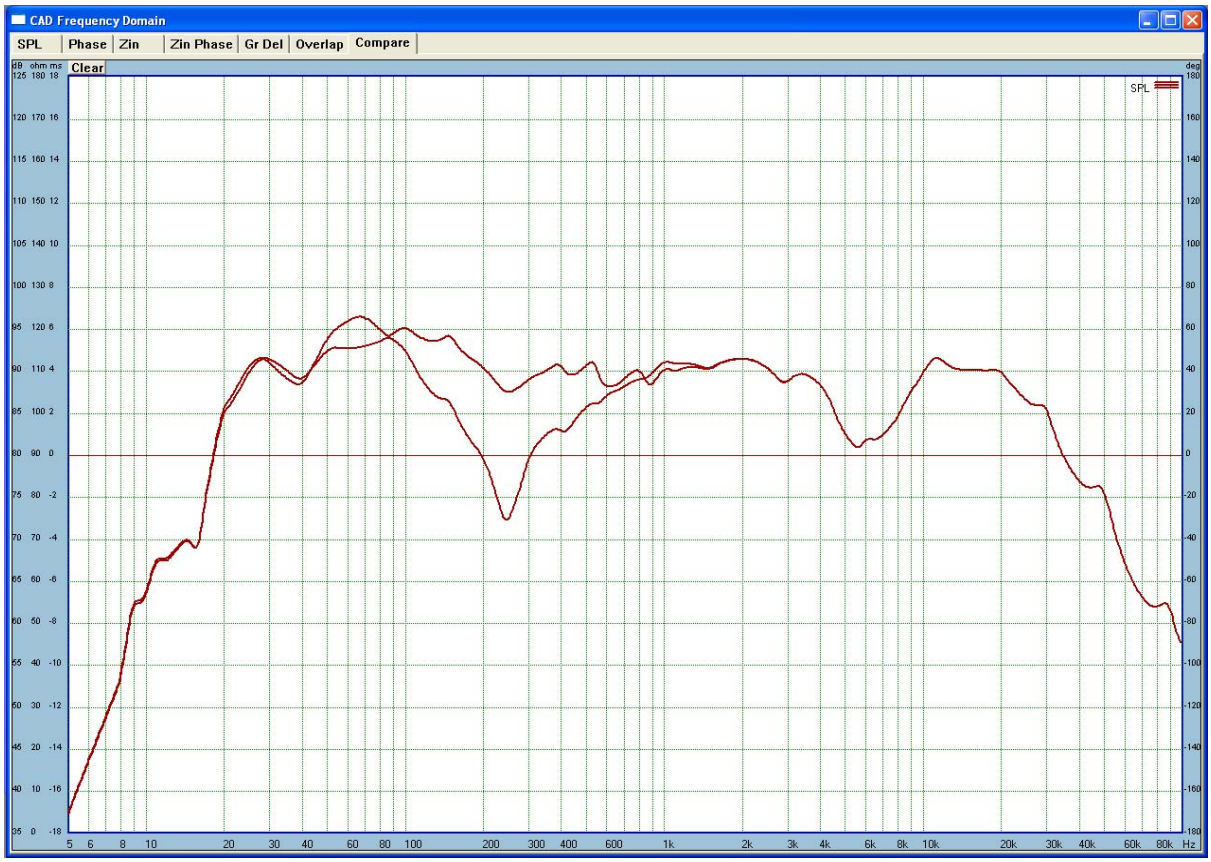


Fig 8.23 Example of “Compare” TAB with 2 SPL curves transferred form the SPL TAB.

The "Compare" TAB plotting window retains it's plots until you exit the program. It's primary function is to allow you to store curves, and compare it to other plots, as you progress through your "what-if" analysis. The "Clear" button simple clears the plots.

Time Tab

1. **FFT Method combobox –**
   Phasing ? check-box column – "In" phase indicates no phase reversal. "Out" of phase indicates phase reversed by 180 degrees.
2. **Plot ?** check-box column – "No" indicates do no plot this curve. "Yes" indicates inclusion of the curve in plots.
3. **Sum FFT button –** Plots summed response using FFT method
4. **Nodal Method combobox –**
   Phasing ? check-box column – "In" phase indicates no phase reversal. "Out" of phase indicates phase reversed by 180 degrees.
5. **Plot ?** check-box column – "No" indicates do no plot this curve. "Yes" indicates inclusion of the curve in plots.
6. **Sum Nodal button** – Plots summed response using MNM method
7. **FFT button** - Plots summed response using FFT method. You **can include driver icon in the CAD** schematic when plotting time domain response using this method.
8. **FFT Full button** - Plots summed response using FFT method. You **must not include driver icon in the CAD** schematic when plotting time domain response using this method.

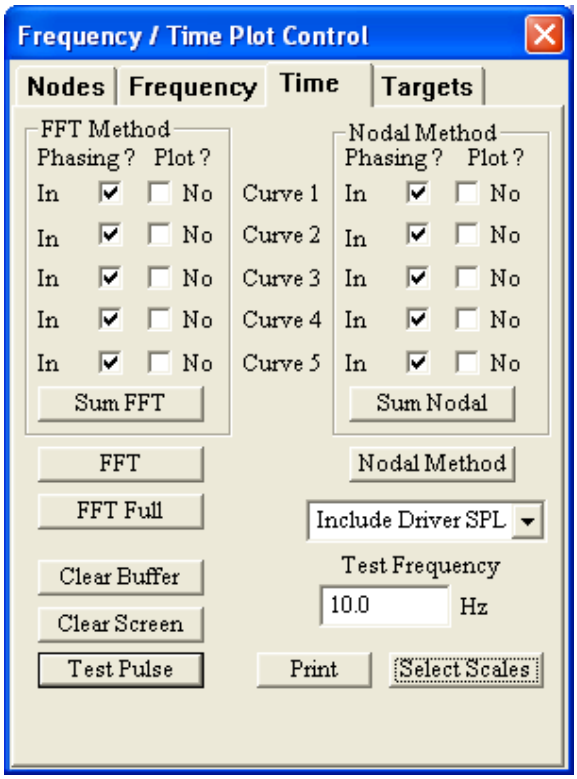
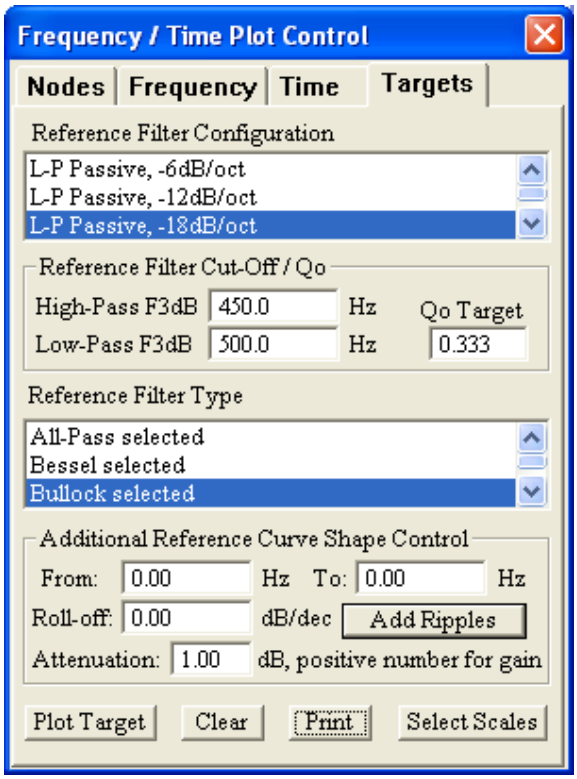


Fig 8.24. Tab for controlling Time-Domain plots



Tab for selecting target curves in CAD

9. **Clear Buffer button** – Clears memory storage allocated for 5 curves.
10. **Clear Screen button** – Simply clears the screen.
11. **Test Pulse button** – Plots the test pulse.
12. **Test Frequency editable field** – Enter your test frequency here.
13. **Drop-down list** box for selecting additional plotting parameters.
14. **Print button** – Invokes printer dialogue box.
15. **Select Scales** – Button opens dialogue box for selecting vertical scale resolution.

Targets Tab

Allows you to plot target SPL/Phase curves in the frequency domain CAD plotting window.

1. **Reference Filter Configuration** – this list box allows you to select slope order and attenuation type.
2. **Reference Filter Type** – this list box allows you to select filter type.
3. **Reference Filter Cut-Off Frequencies/Qo"** - three editable fields to accomplish just that. Enter frequencies in Hz.
4. **Add Ripples button** - Opens another box to enable you to add 3 ripples.
5. **Additional Reference Curve Shape Control**" - group has four editable fields. The **"From"** field selects the frequency (in Hz) from which the response is to roll-off. . The **"To"** field selects the frequency (in Hz) at which the response is to flatten again The third field - **"Roll-off**"- is used to enter the required tilt in dB/decade. Usually, it is a small amount: 1-3dB/decade. The **"Attenuation" parameter indicates, how much we would have to shift the 90dB reference line to match the system efficiency. For example: system efficiency is 86dB – enter "Attenuation" as –4.**
6. **Plot Target button** - Activates plotting for all selected targets.
7. **Clear button** - Clears the plotting window.
8. **Print button** - Opens printer dialogue box.
9. **Select Scales button** - Opens dialogue box for selecting vertical scale resolution.

What Is Compensation?

Measurement of the input impedance of a typical moving-coil, direct radiator loudspeaker, reveals that there are two main factors which affect the results: (1) the inductance of the voice coil and (2) the free air resonance of the vibrating system.

Voice coil inductance will result in a rising input impedance as the frequency of the signal is increased. At higher frequencies, the input impedance levels up because of the Eddy current shunt resistance across the voice coil inductor. At the lower end of the frequency range, a pronounced peak occurs. This is the free air resonance of the driver's vibrating system. For a typical driver, the DC resistance of the voice coil may vary from 3.5 - 14 ohms, but the factors mentioned can cause the modulus of the input impedance to reach 100-150 ohms. The input impedance of the driver is a complex function of frequency.

Whatever crossover network is designed, it is always assumed, that the crossover is loaded by a single resistor and this assumption is valid across the whole operating range of the filter. Now, the problem becomes more clear. In order to achieve the desired frequency response of the filter, a small and well designed passive network is inserted between the driver and crossover filter. The total impedance now presented to the crossover filter, consisting of the driver input impedance plus the compensation network impedance, is far less reactive. It might well be possible to cancel the reactive parts of the driver input impedance to almost negligible values. When the impedance of such a combination is examined, it should be an almost flat line. This is the designers ultimate goal when designing an impedance compensation network.

The second type of compensation is employed to improve the frequency response of the driver. Some small humps in the frequency response can be smoothed out or the excessive efficiency of the midrange or tweeter driver reduced to match the woofer efficiency. Once again, a small and properly designed network is inserted between the driver and crossover filter, but this time the amplitude of the driver and the compensation network is of primary interest. The third type of compensation involves time delay. In order to facilitate mounting of all drivers on a flat front panel, their acoustic centres need to be aligned. Usually, the higher frequency drivers are delayed to bring their acoustic centres in line with the woofer. SoundEasy offers a simple-to-use tool to design and evaluate many types of frequency/time compensation networks.

Using The Calculators

The 'Crossover' tool has four handy built-in calculators for quick evaluation of four simple elements of the compensation networks shown in Fig 8.25 - 8.28.

1. L-Pad Calculator (see Fig 8.25)

In order to reduce excessive sensitivity (efficiency) of the driver you need to insert a properly designed attenuator, consisting of two resistors. The L-Pad attenuator is best inserted between the driver (or driver and compensation network) and the crossover. The resistors must be able to dissipate the required power levels. To calculate the L-pad resistors, select "Calculator" option from the main menu and then highlight the "L-Pad" option. A small window will be opened with two editable data entry fields. The user is required to provide the drivers DC resistance for Re and the required attenuation for Att then to click on the "Calculate" button. The resulting values for the parallel resistor Rp and series resistor Rs will be displayed below. The input resistance of the L-Pad network will be the same as Re.

2. Zobel Network (see Fig 8.26)

Zobel Network is used to cancel the effects of the voice coil inductance. In order to calculate the RC Zobel Network, select the "Calculator" option from the main menu and then highlight the "Zobel Network" option. A small window will be opened with three editable data entry fields. The user is required to provide all parameters and then click on the "Calculate" button. The resulting values for Zobel resistor Rz and Zobel capacitor Cz will be displayed below. Driver's amplitude response and total input impedance are plotted below. Please note the flat input impedance curve.

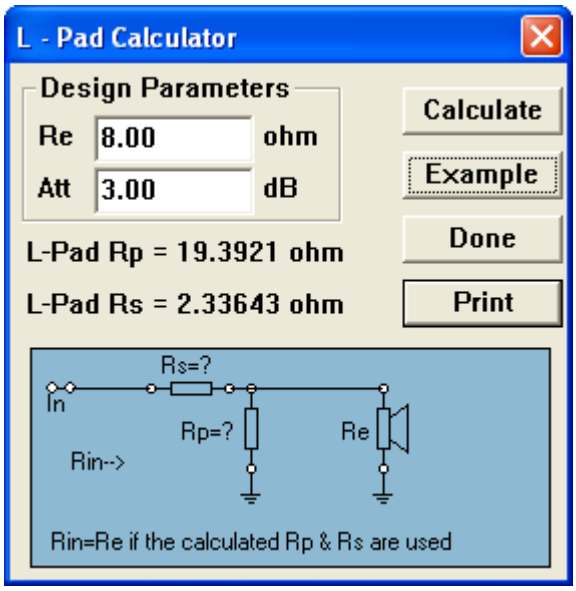


Figure 8.25 L-pad network.

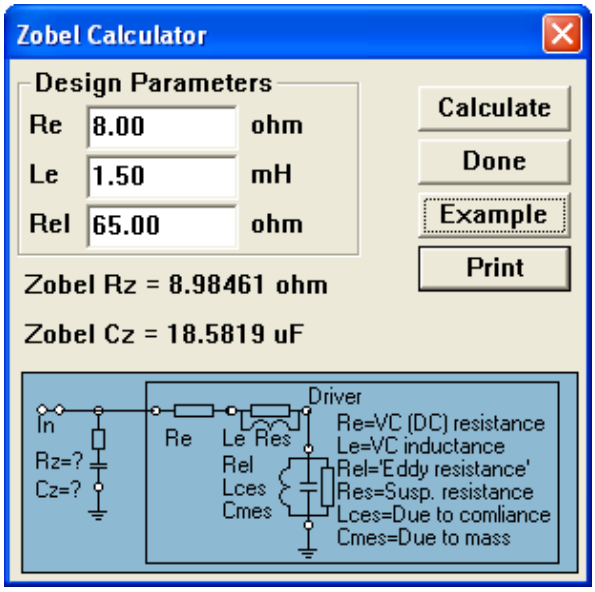


Figure 8.26. Zobel network.

3. Impedance Peak Equaliser (see Fig 8.27)

In order to calculate the components for the resonance circuit used to compensate for the impedance peak, select the "Calculator" option from the main menu and then highlight the "Parallel Notch" option. A small window will be opened with seven editable data entry fields. The user is required to provide all the data and then to click on the "Calculate" button. The resulting values for series components will be displayed below. The "before" and "after" impedance plots are shown below.

4. Amplitude Peak Equaliser (see Fig 8.28)

In order to calculate the components for the parallel resonance circuit used to compensate for the amplitude peak, select the "Calculator" option from the main menu and then highlight the "Bump Compensation" option. A small window will be opened with five editable data entry fields. The user is required to provide driver's DC resistance for Re, the amount of attenuation required in the centre of the hump for Ao, lower for Fl and higher for Fh frequencies at which the hump rises by Al,h decibels. You would typically enter Fl and Fh as the frequencies at which Al,h equals 1db and then click on "Calculate" button. The resulting values for parallel components will be displayed below.

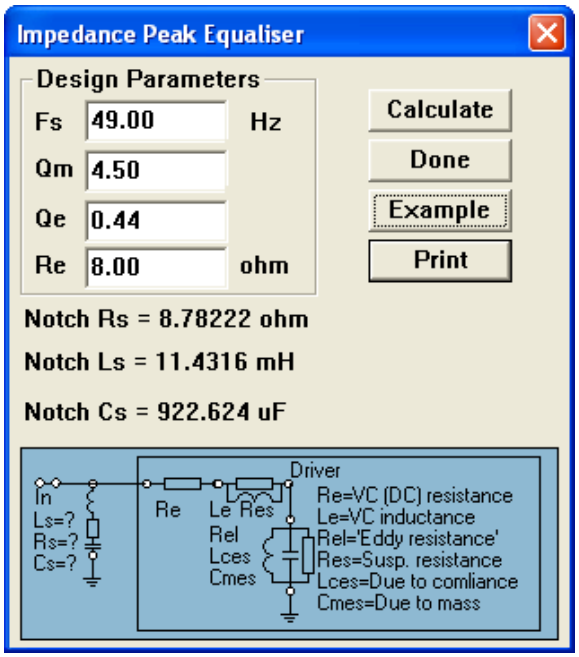


Figure 8.27. Impedance Peak Equalizer

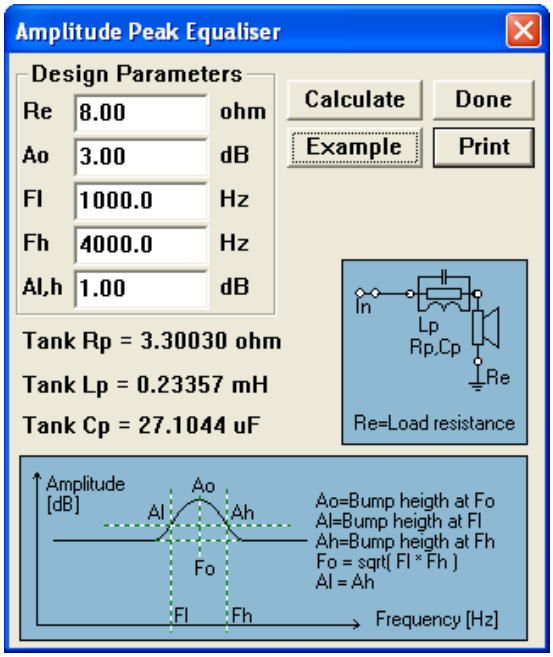


Figure 8.28 Amplitude Peak Equalizer.

Lattice Networks

Compensation networks considered in this section have constant amplitude response and variable phase and group delay responses. The are commonly termed "all-pass" filters or Lattice Filters. One method of passively realizing all-pass function uses symmetrical lattices. A very useful representative of this family of filters is shown on Fig 8.29. It depicts a general topology of the symmetrical lattice filter. Please note, that both load terminals are floating - they are not connected to the ground terminal. A practical realization of the first order lattice network is given on Fig 8.30 and second order lattice network is shown on Fig 8.31.

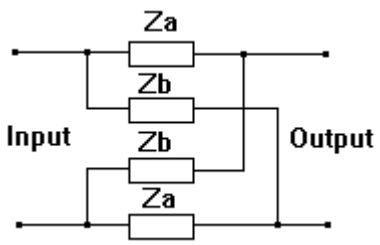


Fig 8.29 Lattice Network (Time Delay Filter).

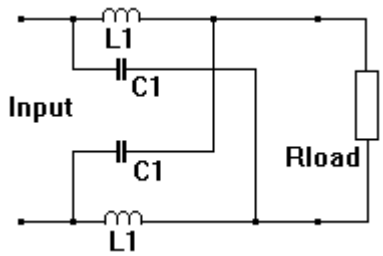


Fig 8.30 First order Lattice Network.

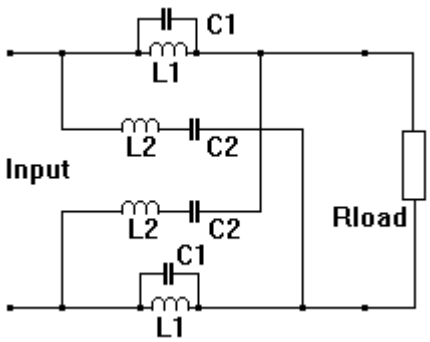


Fig 8.31 Second order Lattice Network

The larger number of elements in the symmetrical lattice arises because the network is not only symmetric with respect to the ports, but also balanced with respect to ground. For satisfactory performance, the impedance in two series arms must track at all frequencies, as must those in the two cross arms. These properties call for low-tolerance components, making the lattice network expensive to build and tune. We are mostly interested in the following characteristics of the filter:

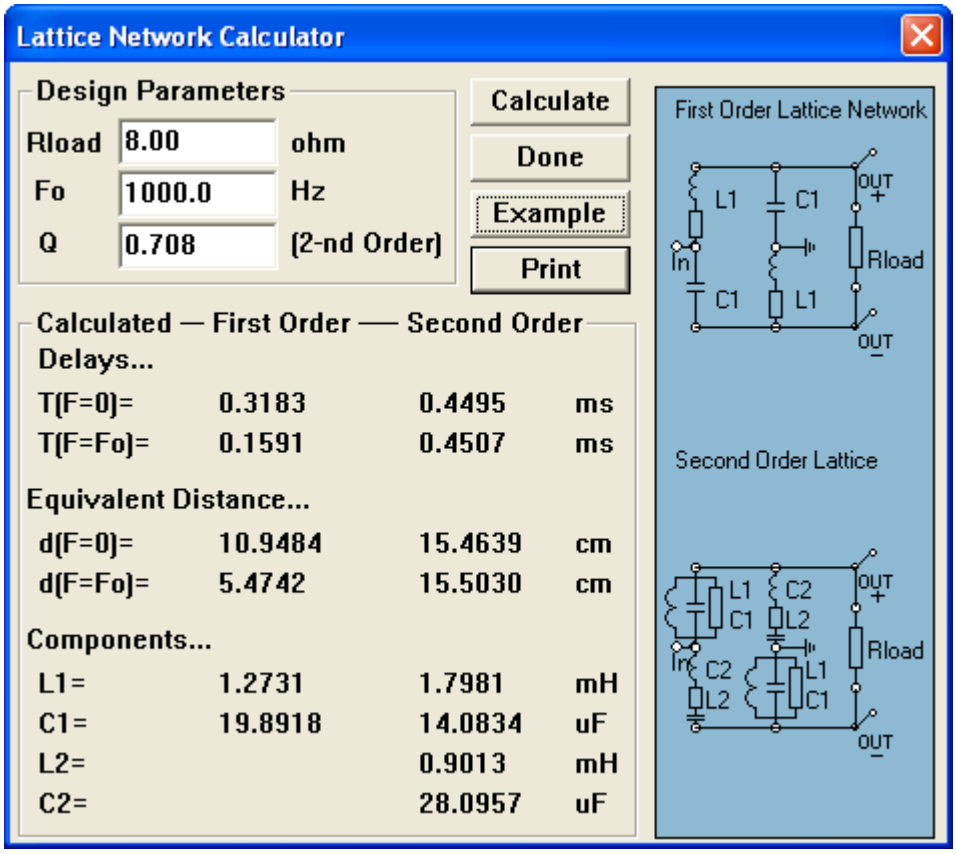


Figure 8.32. Lattice network calculator.

1. Ability to delay a signal in time without introducing frequency distortion.
2. Ability to reflect load resistance to the input terminals, the "constant resistance" property. This property will greatly assist in cascading the networks.

For the first order lattice, the above criteria will be met when: **Za = -Zb and |Za| = |Zb| = Rload**

SoundEasy offers a built-in Lattice Network Calculator (LNC) - see Fig 8.32. This simple LNC allows the user to estimate attainable time delay offered by the first and second order lattices at a desired frequency (Fo), for a given load resistance (Rload). The LNC also calculates the L and C components of the first and second order lattice networks, such as the one shown on Fig 8.30 and Fig 8.31. The resulting components will be given for a constant resistance lattice. Larger delays can be obtained by cascading first or second order networks, so that their individual time delays can be summed. Time delay introduced by the equalizer (Lattice Network) will be added to the existing (if any) delay in the circuit. The first order all-pass network introduces maximum delay at Fo = 0.0 Hz and the delay rolls-off with frequency. For the second order network, there are two parameters, that can be manipulated by the designer in order to obtain the required total delay: (1) Q-factor and (2) Fo. For Q-factors less than 0.6, the delay, introduced by the second order all-pass network tapers off with frequency. For Q-factors greater than 1.0, the second order lattice network allows the designer to create concentration of delay around a frequency, at which there is a 180 deg phase shift (Fo). For Q-factors greater than 2, the delay exhibits a pronounced peak around the Fo, with slopes of -12dB/oct. The LNC is activated from the main menu by selecting (1) "Crossover Tools" and then (2) "Lattice Network" options. The first order lattice network requires load resistance (Rload) and "cut-off" frequency to be entered on the calculator's data entry fields. The second order network requires additional Q-factor to be keyed-in.
Caution !. The LNC assumes that load resistance of the network (Rload) is the DC resistance of the driver (Re). Remember, that lattice network should only introduce the required time delay, and reflect the load resistance to its input. Modifying Rload AND storing the data file will cause the value of Re to be changed to Rload.

The total area under the delay curve is constant for lattice networks. As a consequence of this, higher Q-factors would cause concentration of the delay around the Fo at the expense of the delay at other frequencies.

Additional Operations.

The introduction to the 'Crossover' tool was based on the woofer driver. The midrange and tweeter drivers have their own dedicated menus, but the operation of the menus is exactly the same as for the woofer driver. For detailed description on loading, saving and printing the user should consult Chapter 2.It is often beneficial to determine how much the "non-ideal" load disrupts the transfer functions of the designed network. It is easier to make this observation without the driver's acoustic response - in the "Exclude Driver" mode. The "Include Driver" and "Exclude Driver" options in the list box combine or exclude the previously edited driver's transfer function with the compensation network. The Impedance curve will not be modified by this option. The same option is also implemented in the 'Crossover' tool.

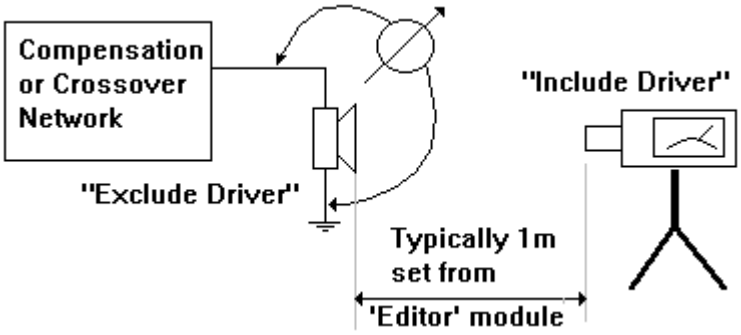


Fig 8.33 "Include Driver" and "Exclude Driver" options.

"Crossover Tools" Menu Options

There are additional menu options available at the bottom of the "Crossover Tools" main menu bar. The bottom three are system-wide options and are active, while other (non-crossover CAD) screens are opened. You may for instance require to change a component value while working in system optimizer.

1. **"Use E-12 Series"** - this menu option allows you to approximate component values to the nearest E-12 series value.
2. **"Use E-24 Series"** - this menu option allows you to approximate component values to the nearest E-24 series value. These values are more accurate than E-12 series.

"Preview" built-in circuit window

The program has a large number of built-in filter and crossover circuits to choose from. All these circuits can be loaded to the clean CAD screen with pre-calculated values. They can also be added to the existing circuit by using "Add" function. This way, crossover development is quite fast and efficient. All in-built circuits identify themselves to the user by their names, such as **"BP Active +/-18dB/oct"** circuit. This can be sometimes rather cryptic, so the circuit **"Preview"** window will come very handy in a situation, where you are uncertain, as to what the circuit name actually hides.Filter and crossover types and configurations are listed in two list boxes. Combination of user selections from both list boxes is needed for final selection of the circuit.

The **"Preview"** window is activated automatically when you open the built-in filter or crossover dialogue boxes and select any of the circuits from the provided list boxes. The **"Preview"** window will close automatically when you close the filter or crossover dialogue box. The built-in schematics are displayed automatically when you select any item from the list box, OR you can use cursor keys UP/DOWN to scroll through the schematics.

However, you may want to position and re-size the "Preview" window on the screen, so it will display full view of all built-in circuits. You will be able to keep the "Preview" window still opened, when you close the filter or crossover dialogue box using the "X" gadget in the top-right corner of the dialogue box. Now, the "Preview" window is active and can be adjusted as required.

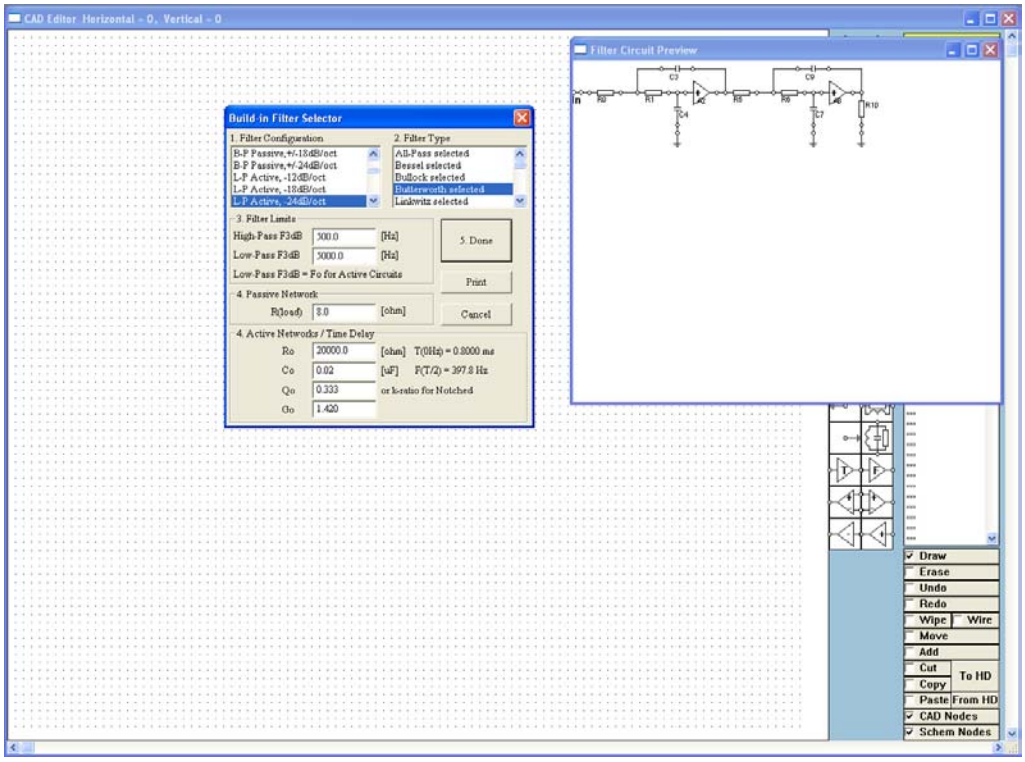


Figure 8.34 The "Preview" window.

Fast CAD Component Tweaking

In order to facilitate a shortcut to component value entry, we have implemented "Adjust" function. The function is accessed from a list box located on the "**Frequency/Time Domain Plot Control**" Tab – see Figure 8.35. The dialogue box allows you to manually adjust the value of the selected component **within 1% - 1000%** of its original value. All five curves: (1) Amplitude, (2) Phase, (3) Impedance, (4) Impedance Phase and (5) Group delay curves can be nominated to be displayed while the selected component is being tweaked. Finally, a "Clear" function is also provided to clear congested screen.

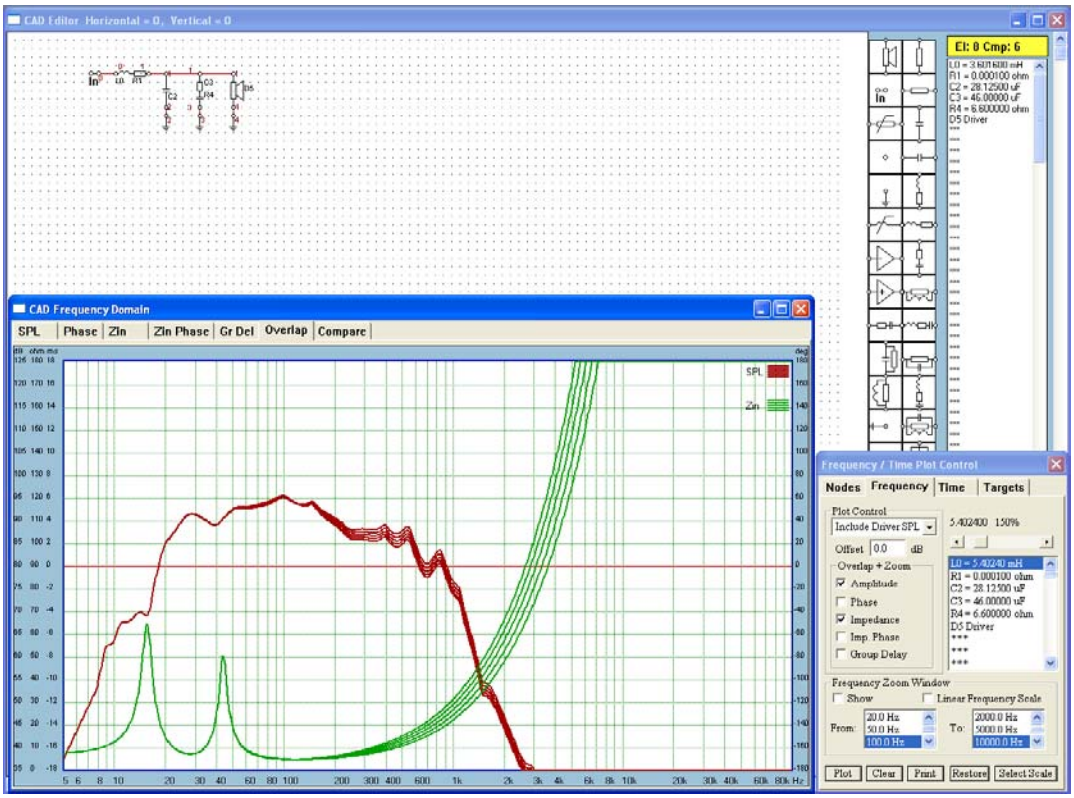


Figure 8.35. Example of "Tweaking" components using "Adjust" function.

Frequency Domain ZOOM

Frequency Domain Zoom window allows you to zoom-in the frequency domain into the selected frequency range for detailed view of the plots in Crossover CAD and System plots. The Zoom window will be automatically opened if the **"Show"** box is "checked" – see Figure 8.36.

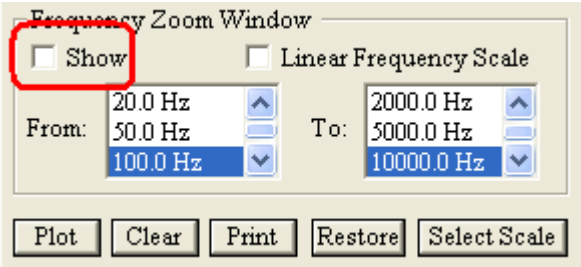


Figure 8.36 Zoom Window controls

The Zoom window has a separate settings for high/low ends of the frequency range of it's screen. These settings are fully independent of the main frequency domain screen in the CAD and System sections of the program. Vertical scale of the zoom window is linked to the main frequency domain window of the crossover section.

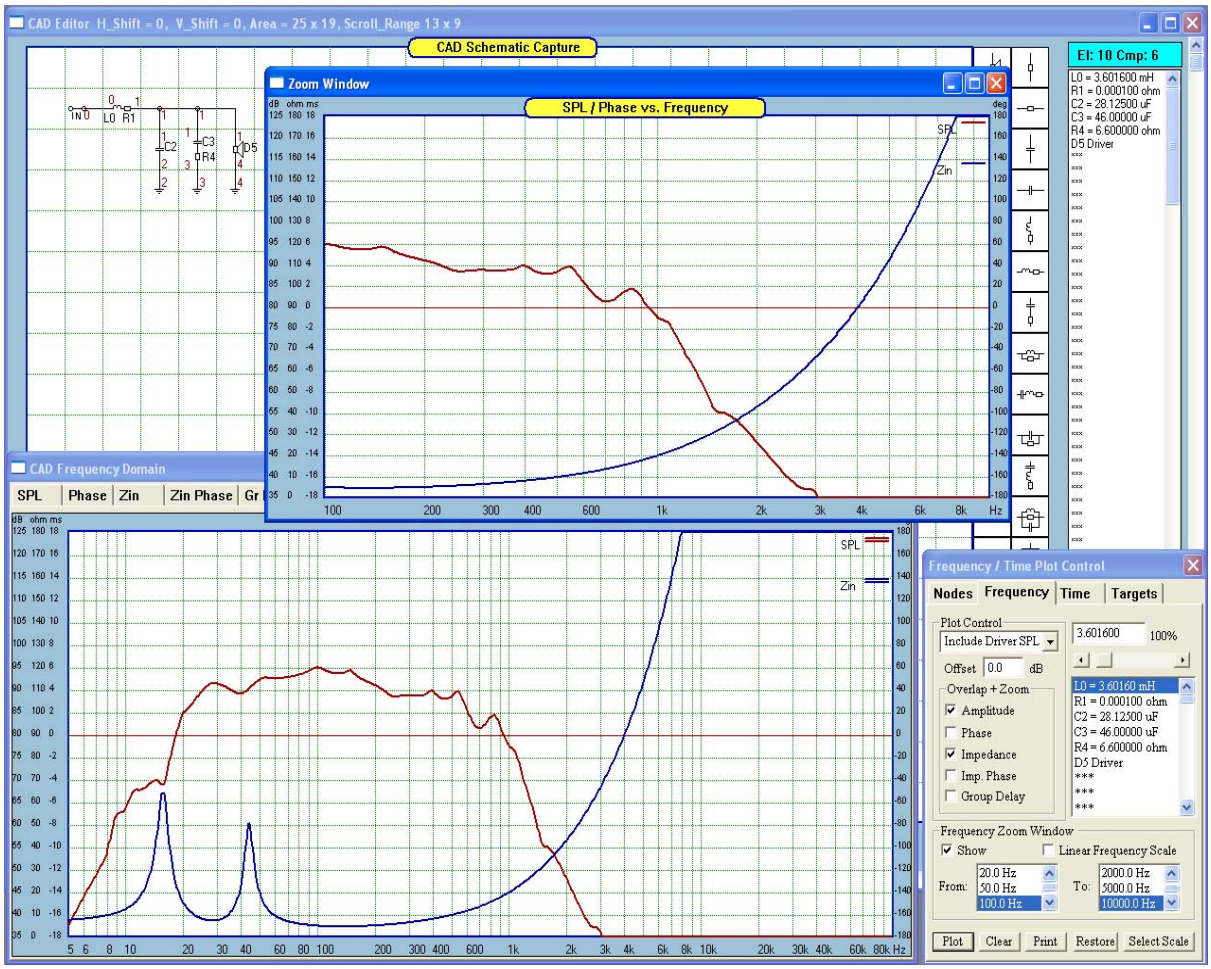


Figure 8.37 Zoom window used to focus on 100-10000Hz frequency range.

There are following practical issues are observable on the zoom window.

1. If you analyze a circuit, that has no driver-dependant data included in the analysis – like SPL or Zin from the driver file, than the circuit transfer functions are calculated directly for every frequency point of the screen. This is an excellent way of increasing screen resolution – a true zoom-in process.

2. If you analyze a circuit, that includes driver-dependant data in the analysis, then this data will be mapped onto the frequency scale of the zoom window.

Linear Frequency Scale

Another very useful feature of the ZOOM window is the ability to switch to **linear frequency scale**. Frequently, you may need to determine if the circuit under analysis is actually a "liner-phase" type or how linear the phase really is. When operating the zoom window in linear frequency scale mode, this particular property of the circuit is clearly visible.

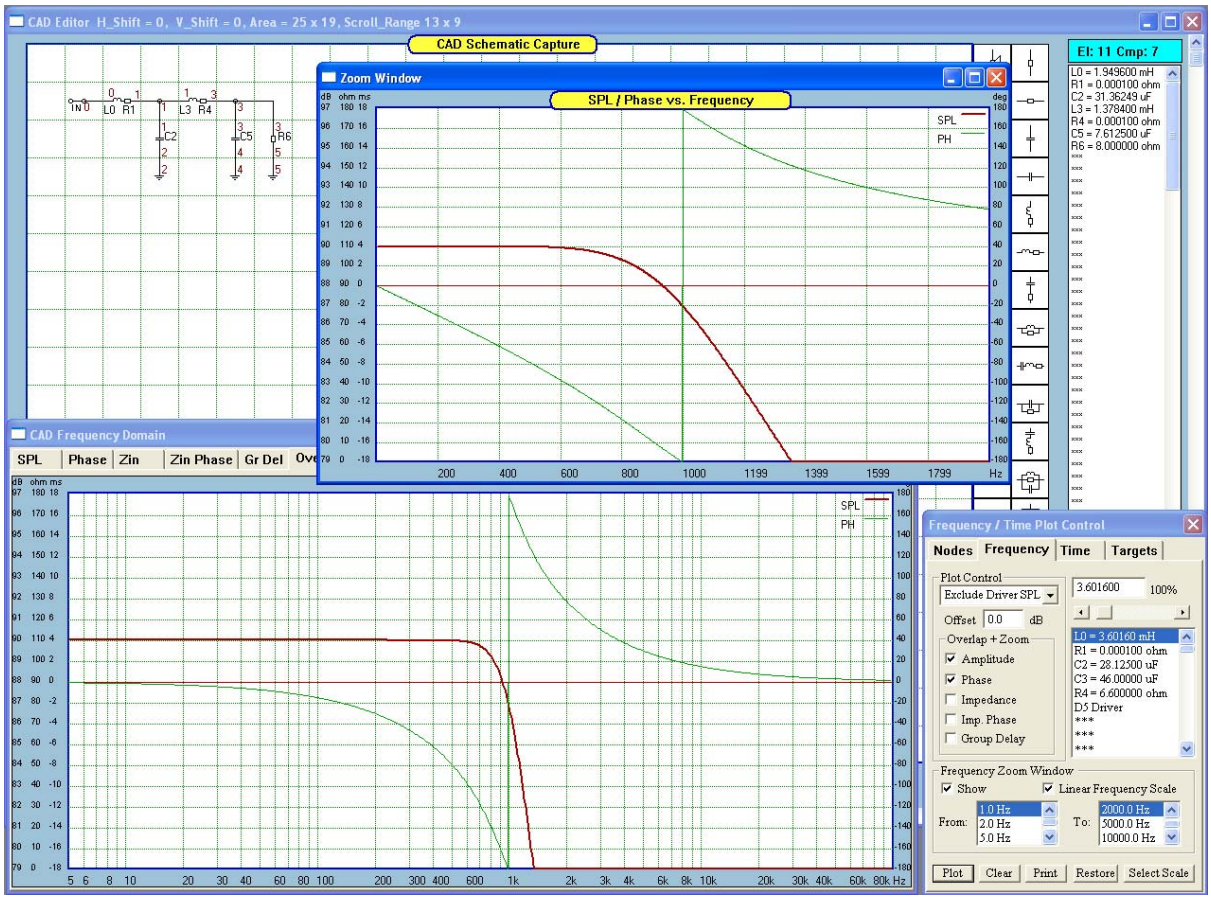


Figure 8.38. A 4-th order LP Butterworth filter shown in main window and ZOOM window with Linear frequency scale 1Hz – 2000Hz. Please note phase **non-linearity** between 400-2000Hz.

A filter will have linear phase if its impulse response is symmetric about its midpoint. As a result, systems with a linear phase response delay each frequency component of the input by the same amount, while nonlinear phase systems delay each frequency by a different amount, leading to a form of distortion known as phase distortion.

A good example of passive, linear phase filters are Bessel filters. These are characterized by good phase response within the pass-band – see figure below. The advantage of using linear frequency scale to evaluate phase response is highlighted on Figure 8.39, where the bottom (standard) window displays 4-th order LP Butterworth and Bessel filters using logarithmic frequency scale. Phase response of both filters is different, but one could not tell from this picture, which phase response is linear. The zoom window set to linear frequency scale makes it very easy.

Please also note the excellent zoom on the SPL and phase curves within 100Hz – 2000Hz shown on Figure 8.38 and Figure 8.39, while the standard screen runs in 5Hz – 100000Hz mode.

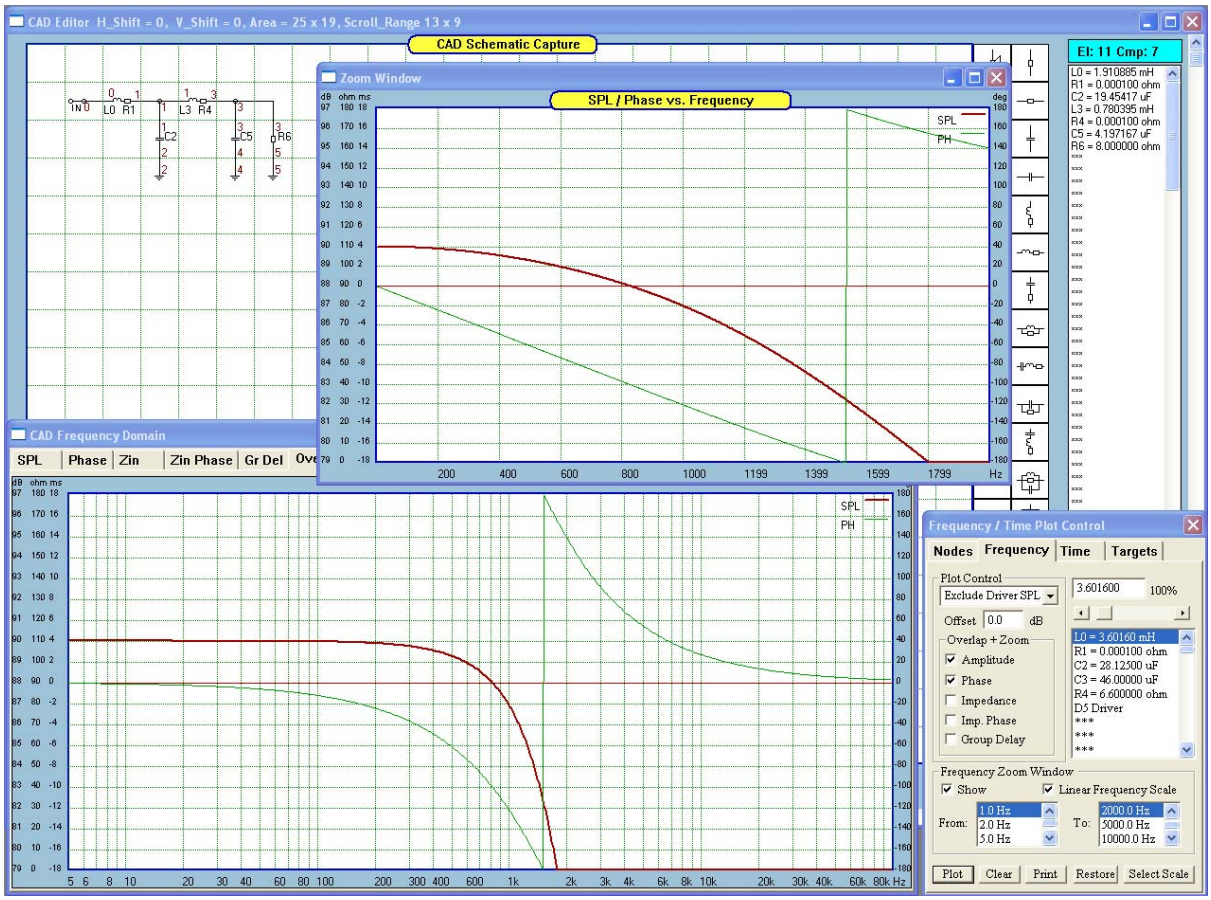


Figure 8.39 4-th order LP Bessel filter shown in main window and ZOOM window with Linear frequency scale 1Hz – 2000Hz. Please note phase **linearity** between 400-2000Hz.

The zoom window is highly automated. All requested curves will be re-plotted automatically when a new frequency range is selected. If the window is accidentally covered by another window, you can click on "Show" button twice to close and the re-open this window on the top of your screen.

Chapter 9. Filter and Crossover Design

Background

Multi-way speaker systems can provide the best possible acoustic performance with each loudspeaker specially designed to cover a limited section of the frequency spectrum. To obtain this performance one must ensure that the electrical signal input to the different loudspeakers is divided in such a way that each speaker receives only the signals it is required to reproduce.

The 'Crossover' tool has been provided to enable a thorough examination and design of the **passive and active parallel, series and cascaded filters/crossovers**. The parallel filter design enables the designer to treat each driver independently. This is possible because the driving amplifier is usually a voltage source. Only one driver is taken into account when each optimisation is performed. Crossover design is a difficult task and this manual is not intended to substitute a text book on crossover design. The user is well advised to seek additional information on the subject and the list of texts in the Appendix would be a good starting point. Please load now the test301.wfr data file. Two approaches could be taken when working with the 'Crossover' tool. First would assume, that you design "less then typical" crossover. It could be a band-pass section which has +18/-6 dB slopes or 5-th order high-pass filter. You would then be required to create schematic using the technique described in Chapter 8 and perform your own calculations for the component values. After you entered the components' values, SoundEasy will then help you with the analysis of your customised crossover.

Channel Filters or Crossovers

The ‘Crossover’ tool enables you to design and analyse:

1. Complete crossover network OR

First of all, design of a complete crossover on one screen is convenient. You can see the complete design and understand better the interaction between the individual electronic components and channels. Secondly, if your intention is to use ‘System’ screen for further analysis of the system, you need to use this approach. This is because **the ‘System’ screen is designed to analyse complete crossovers rather than individual filters**. You may stop at this moment to think about how to plot input impedance of individual channel when the complete crossover is drawn on the screen ?. We suggest, that in many cases, you will be able to change a component value in your crossover to make it act as a “switch” allowing you to keep the schematic configuration intact, but a channel disconnected. For example: two-way, second order crossover has the HP filter capacitor (C4) reduced to a VERY small value, so it practically cuts-off the HP section from analysis. Input impedance of the crossover now only includes the LP section - see Fig 9.1. If your crossover topology prohibits you from using this approach, you may need to include extra “switching” components in your crossover.

Please refer to section “First Cut Design” for more information about complete crossover design approach.

2. Individual channel (woofer, midrange, tweeter…) filter

Individual filters are useful for two reasons: (1) allow you to easily analyse input impedance of a parallel crossover and (2) driver files with the filter or EQ circuit can be loaded back into the ‘Box’ module and used in conjunction with the enclosure frequency responses. This is the approach you would take to design **an active subwoofer with EQ circuit**. Both ways will include driver frequency response and input impedance in modeling all transfer functions of your filter or crossover. Also, both approaches can be used with filter/crossover optimizer.

Using Built-In Filters

Five types of filters are built-in for faster network implementation. These are: Butterworth, Bullock, Bessel, Linkwitz and All-Pass filters. The filters are edited and selected by “Built-in Crossover” under “Edit” main menu option. This will open a dialog box from which the user may set-up the required configuration. Most of the filters are available in 1,2,3,4,5 and 6-pole versions. Low, band (symmetrical) and high-pass versions are available as passive and active networks (see Fig 9.2).

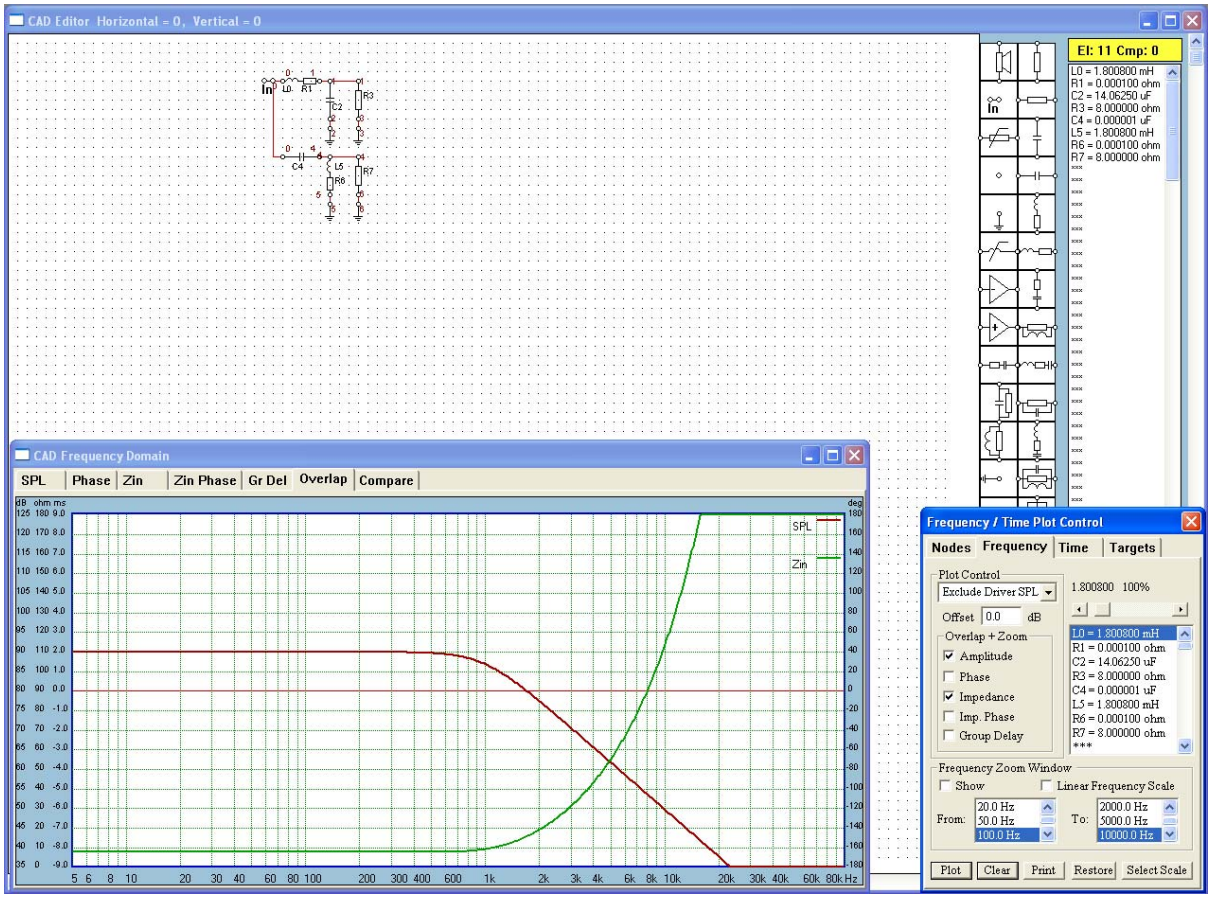


Fig 9.1 the HP filter capacitor (C4) reduced to a VERY small value

Example: In order to select the Butterworth band-pass crossover, Select “Built-in Filter” from the “Edit” menu and when a dialog box appears, follow the steps described bellow.

1. Select the required Filter Configuration from the “1. Filter Configuration” list box.
2. Select the filter (eg: Butterworth) type from the “2. Filter Type” list box.
3. Edit the high-pass or low-pass cut-off frequencies, the "-3dB" frequencies in “3. Filter Limits” data entry fields.
4. Enter required load resistance R(load) under “4. Passive Network” or arbitrary Ro or Co for “4. Active Network”.
5. Press “5. Done” button

You can now obtain a hard copy of the selected values by pressing the 'Print' button. Otherwise, click-on "Done" button to continue with the design process. SoundEasy will remove the dialog box and create the required filter. When the process is finished, a complete crossover is drawn in the screen together with the parts list displayed in the list box on the right. This filter is loaded by a single resistor. Normally, the user would want to evaluate the circuit's performance when the filter is loaded by a loudspeaker (and possibly a compensation network). To simulate this configuration for real driver, you must **load the data file first** (refer to Chapter 2), then enter required filter frequencies, select slope orders and type using previously described dialog box. There is no need to edit the load resistance. The 'Crossover' tool is now informed that the driver data file is loaded and the DC resistance of the driver is automatically selected for the calculations as the load resistance. Click on the 'Done' button when finished. When the network appears on the screen, replace the load resistor with the loudspeaker symbol - see Fig 9.4. In this way you tell the program that you wish to use data calculated by the previous tool, the 'Box' module.

Replacing the resistor is performed in the same way as used when replacing a wrong component. If in doubt, please refer to Chapter 8. Fig 9.3 shows an example data file loaded and some graphs plotted using the "Frequency Domain" window**. “Clear CAD”** menu option clears schematic only and **“Clear Data”** menu option clears all data and the CAD screen.

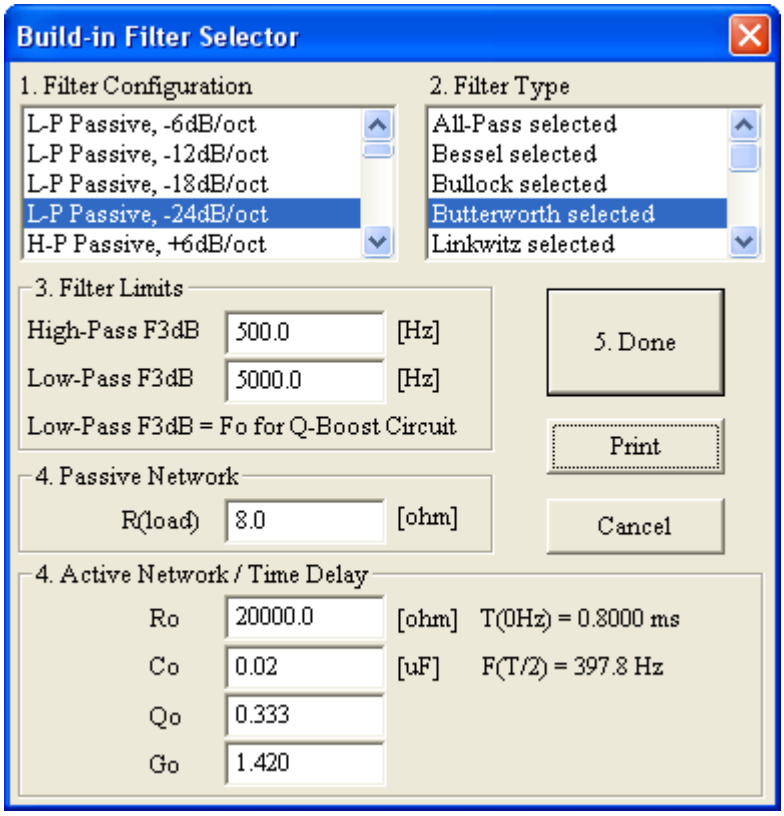


Fig 9.2 Built-in Filter Selector dialogue box

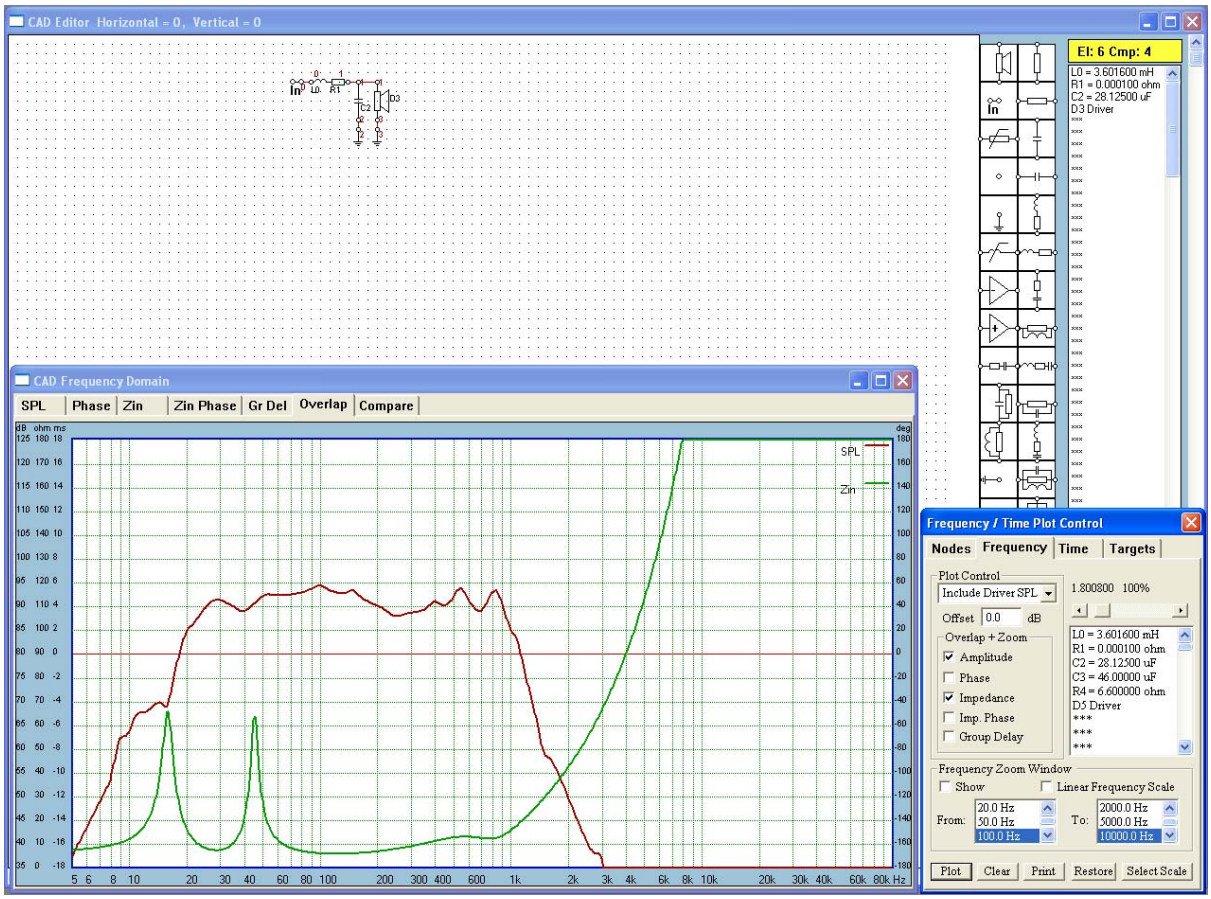


Fig 9.3 Driver file was loaded first and then built-in filter was selected

Using Built-In Crossovers

Three types of crossovers are built-in for faster network implementation. These are: Butterworth, Bessel and Linkwitz crossovers. The crossovers are edited and selected by “Built-in Crossover” under “Edit” main menu option. This will open a dialog box from which the user may set-up the required configuration. Most of the crossovers are available in 1,2,3 and 4-pole versions, 2-way to 5-way configurations. Low, band (symmetrical) and high-pass versions are available as passive and active networks (see Fig 9.4).
Example: In order to select the Butterworth band-pass crossover, Select “Built-in Crossover” from the “Edit” menu and when a dialog box appears, follow the steps described bellow.

1. Select the required Filter Configuration from the “1. Crossover Configuration” list box.
2. Select the filter (eg: Butterworth) type from the “2. Crossover Type” list box.
3. Edit the high-pass or low-pass cut-off frequencies, the "-3dB" frequencies in “3. Crossover Limits” data entry fields. NOTE: The first crossover frequency is considered to be the one between the lowest frequency driver (say, woofer) and the immediate next higher frequency driver (say, upper bass) and so on….**You must enter the frequencies in increasing order, even if you do not use all data fields**. For example, if you only intend to design a two-way system with a crossover frequency of 1000Hz, enter 1000Hz for the “First F3dB, then enter a dummy data of 2000Hz for the Second F3dB, then again a dummy data of 3000Hz for the third F3dB and so on..
4. Enter required load resistance R(load) under “4. Passive Network” or arbitrary Ro or Co for “4. Active Network”.
5. Press “5. Done” button

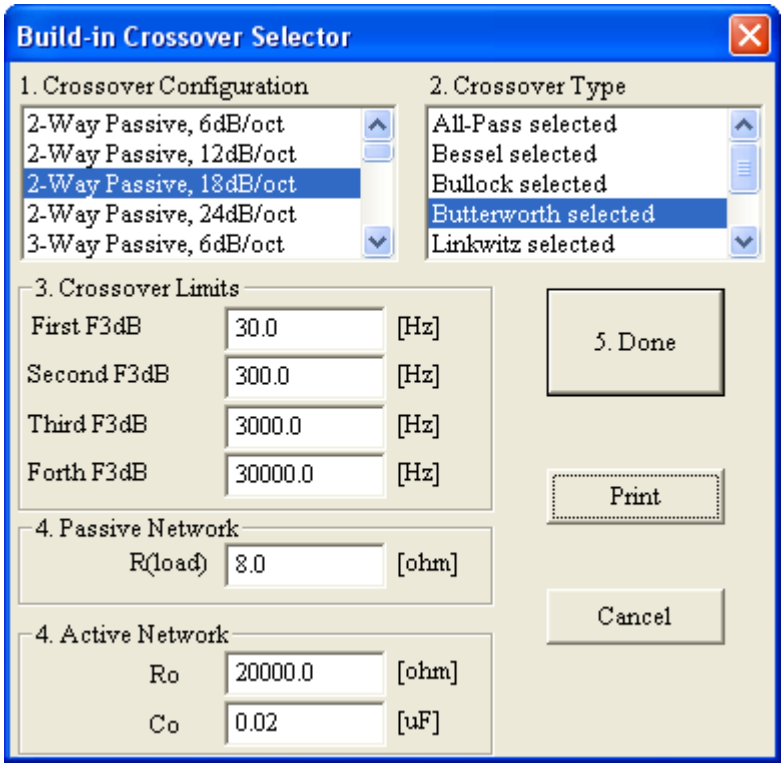


Fig 9.4 Built-in Crossover Selector dialogue box

Active Crossover

Note: Successful application of active networks must consider hardware issues like biasing, supply voltage, driving point impedances, buffering, clipping, feedback, distortion and much more. This program is only concerned with transfer functions of the circuits being modelled.

Filtering circuits employing operational amplifiers have been extensively debated and described in details in professional literature. The task of this module is therefore, to provide the user with an easy to use and flexible tool, capable of analysing active network behaviour. CAD system operation is exactly the same as for passive networks however, model used for inverting and non-inverting amplifiers requires more detailed presentation.

Operational Amplifier

In order to accommodate practical operational amplifiers, the program uses VCVS (Voltage Controlled Voltage Source) to model the operational amplifier. Practical amplifiers have finite gain, finite input impedance and greater than zero output impedance. All those elements were incorporated in the VCVS model so it become a non-ideal VCVS. Default values for the low frequency model of :

1. Inverting amplifier are: Rin=10Mohm, Rout=100ohm and Ku=100000. These parameters are taken directly from data sheets of the product.

2. Non-inverting amplifier are: Rin=1000Mohm, Rout=0.001ohm and Ku=1.0. These parameters are pre-calculated as shown in the example below. Please note, that ALL parameters are entered as positive numbers.

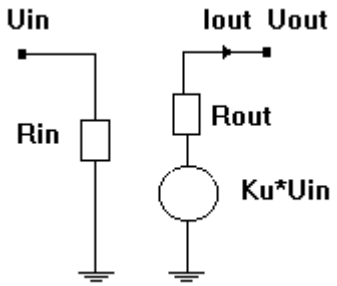


Fig 9.5 OPAMP model

All components of the model can be edited from dialogue boxes. Fig 9.5 shows model used for representing the two types of amplifiers. When the inverting amplifier is used, Ku denotes **open loop** gain. The "in circuit", actual input and output impedances can be calculated observing Miller's theorem. When the non-inverting amplifier is used, the default values have been set to represent a good quality voltage follower and Ku represents **closed loop** gain. In general case, the closed loop gain of a non-inverting amplifier is set by two external resistors, (feedback resistor - connected between output of the amplifier and the inverting input and another resistor - connected between the inverting input and the ground) so that, for example, a non-inverting amplifier with a gain of 100 can be constructed. You can simulate this case here by setting 'Gain' to 100 in the dialogue box and when you actually build the circuit, you will use two additional resistors to set the gain, as described above. Other parameters should be calculated as in the following example:

Example: Operational amplifier you selected for non-inverting application has the following data as provided by manufacturer: Zin=1Mohm, Zout=100ohm and open loop gain of Aol=10000. Your application calls for an amplification in order of 20. Set "Gain" from the dialogue box to 20 - remember, it is the closed loop gain that is required here, and when you finally build your circuit, you should use external resistive feedback to set the gain.

Rin(non-inverting OPAMP) = (1+Aol/Gain)*Zin = (1+10000/20)*1 = 51Mohm
Rout(non-inverting OPAMP) = Zout/(1+Aol/Gain) = 100/(1+10000/20) = 2ohm.

The following data should be entered into the non-inverting operational amplifier dialogue box: Rin=51Mohm, Rout=2ohm and Gain=20. In case you would use the same hardware as an inverting amplifier, you should enter the original data: Rin=1Mohm, Rout=100ohm and Gain=10000. We recommend, that you use load resistor from output of the amplifier to ground to simulate real life loads. Modification of the default values, provides the user with a powerful mechanism to customise behaviour of both amplifiers. Should your application require an implementation of a differential amplifier, use the following configuration (use the following data):

1. A1: Rin1=10Mohm, Rout1=100ohm, Gain1=1.
2. A2: Rin2=10Mohm, Rout2=100ohm, Gain2=1.
3. A3: Rin3=10Mohm, Rout3=100ohm, Gain3=100000.
4. A1 input is the inverting input and A2 input is the non-inverting input.

This configuration is used in active delay network of the first order shown in previous chapter. It is observable, that A1 acts as a simple voltage inverter and A2 is a unity gain buffer. Existence of greater than zero output impedances in A1 and A2 (100ohm) allows us to connect their two outputs in parallel - see Fig 9.6. As with the passive delay networks, one can observe flat amplitude response.

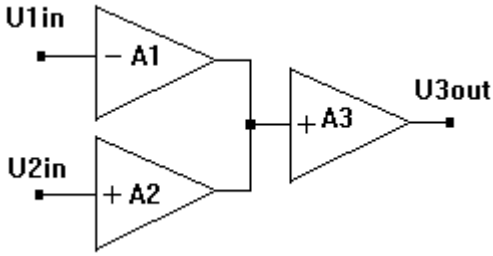


Fig 9.6 Differential OPAMP

When analysing active circuits, three plots will be of importance to the designer: (1) amplitude response, (2) phase response and (3) group delay. Input impedance of the circuit was of prime importance for passive compensation/crossover networks, as it determined quality of the crossover load. In case of active networks, loudspeakers are isolated from the active crossovers by the power amplifiers. Also, buffer preceding the active crossover will always provide near zero output impedance eliminating the need to examine the loads presented to it by the active crossover.

Network Example

Frequently recommended active crossover is a third order Butterworth filter. It will exhibit maximally flat magnitude response, it has sharp cutoff characteristics of -18dB/oct and it has flat voltage and power frequency responses with a gradual change in phase across the band. Please refer to Fig 9.8 and 9.9 Several tropologies have been proposed for this filter, out of these, multiple feedback approach offers good tradeoffs in circuit complexity, component spread and sensitivities.

A common design approach would be to design the active filter with unity gain and adjust system gains by tweaking power amplifiers' gains. We also strongly **recommend that the final gain of the active crossover filters is equal to 1, as it enables proper functioning of the optimizer modules**. Since loudspeakers are connected directly to power amplifiers, there is also no need to provide impedance compensation for the drivers. Power amplifiers will provide very good level of dumping. Separating the spectrum before the final power amplifiers makes it easier to control undesirable effects of clipping high frequency signals in the presence of high level low frequency signals.

When modelling active crossovers using SoundEasy, the loudspeaker can be connected directly to the output of the operational amplifier. Theoretically it is possible, (remember, we are not concerned with the practical hardware limitations) since the negative feedback of this particular configuration reduces output impedance of the amplifier to a small fraction of an ohm, and the amplifier starts to behave like an almost ideal voltage source.

Additionally, you can reduce output resistance of the amplifier (Rout) to a small value. Most of the "feedback" type of arrangements would fell into this category. Because of the above, you can still use the same techniques of "including or excluding" driver as it was explained for the passive networks, and the same substitute of loudspeaker symbol for the load resistor to include frequency/phase/group delay and impedance responses of the driver in the active network analysis.

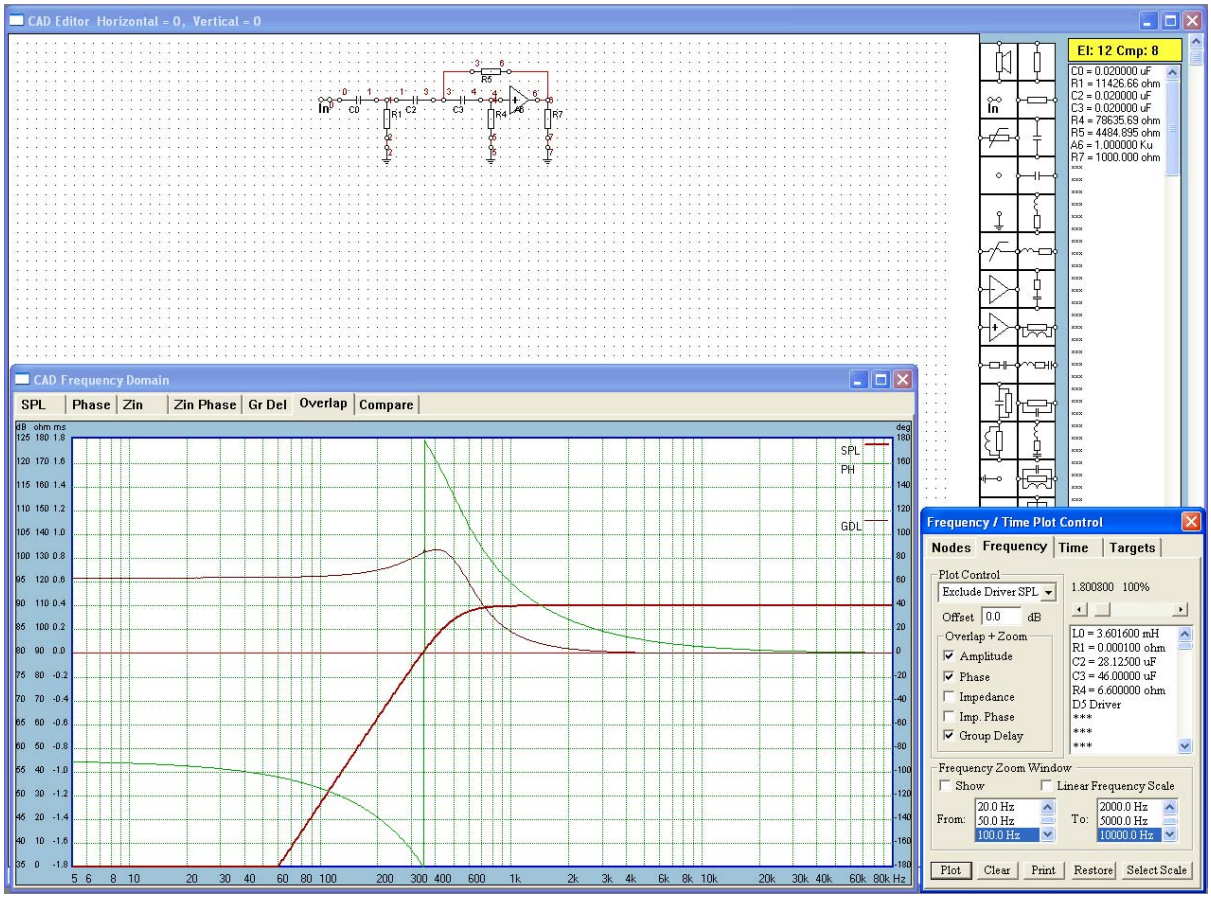


Fig 9.8 Butterworth +18dB/oct HP filter.

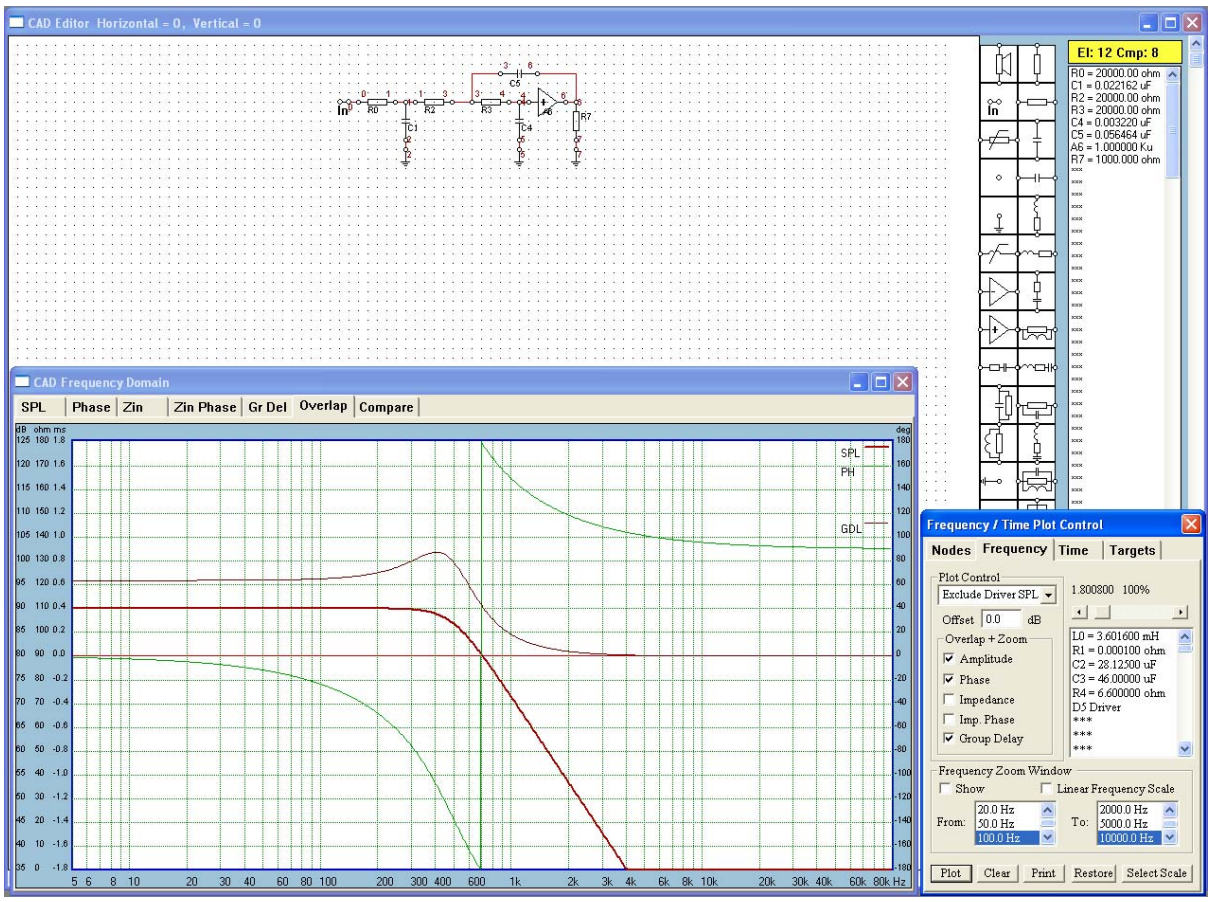


Fig 9.9 Butterworth +18dB/oct LP filter

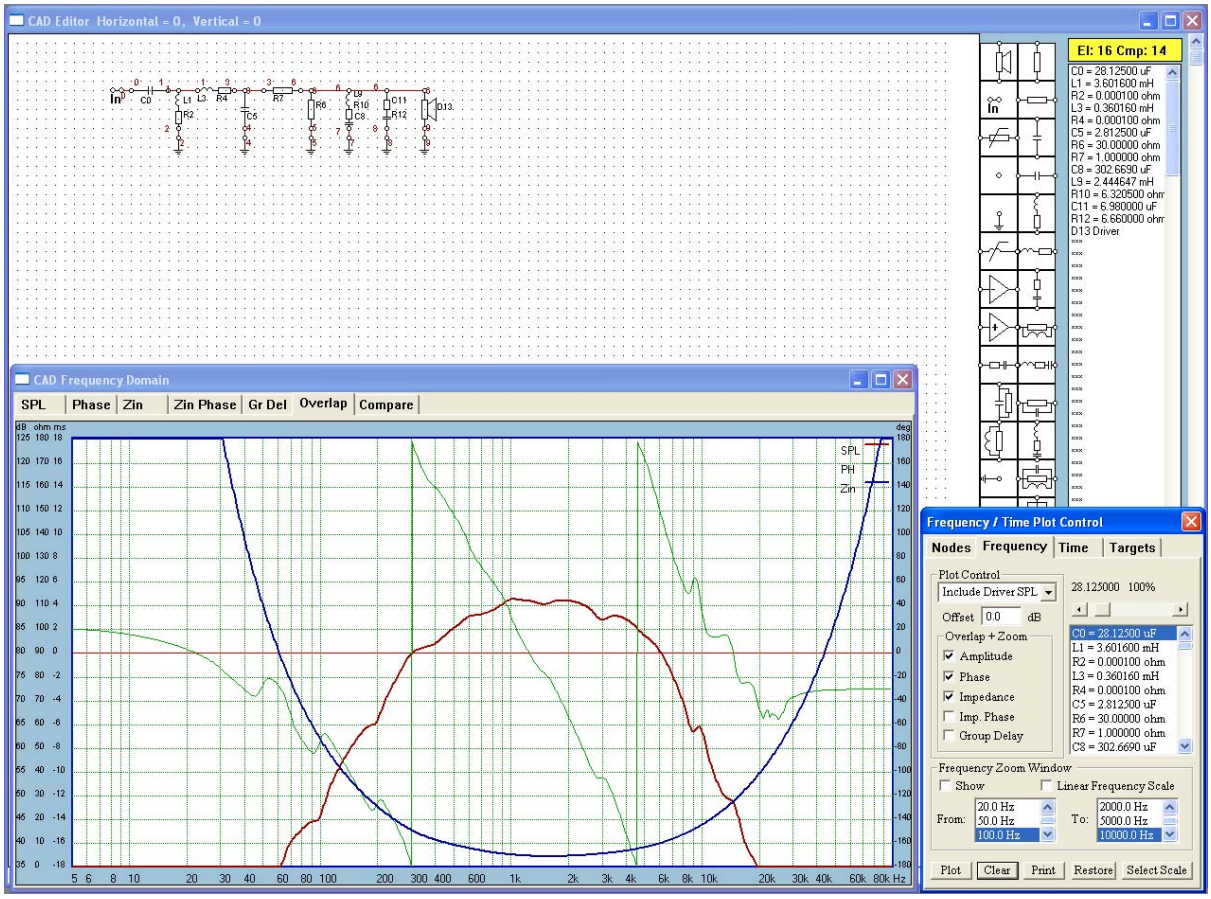


Fig 9.10 Example midrange driver.

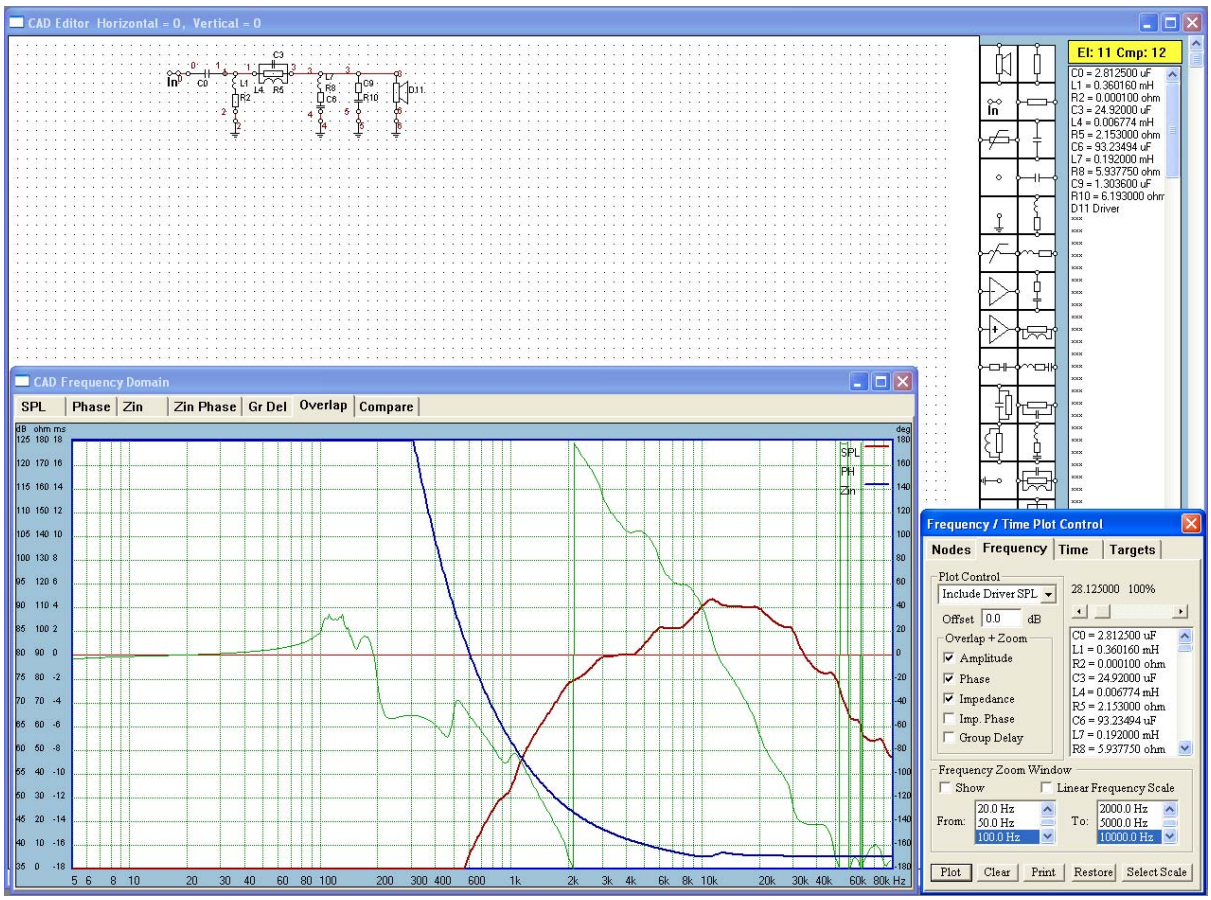


Fig 9.11 Example tweeter driver

First Cut Design (Multiway & D'Appolito)

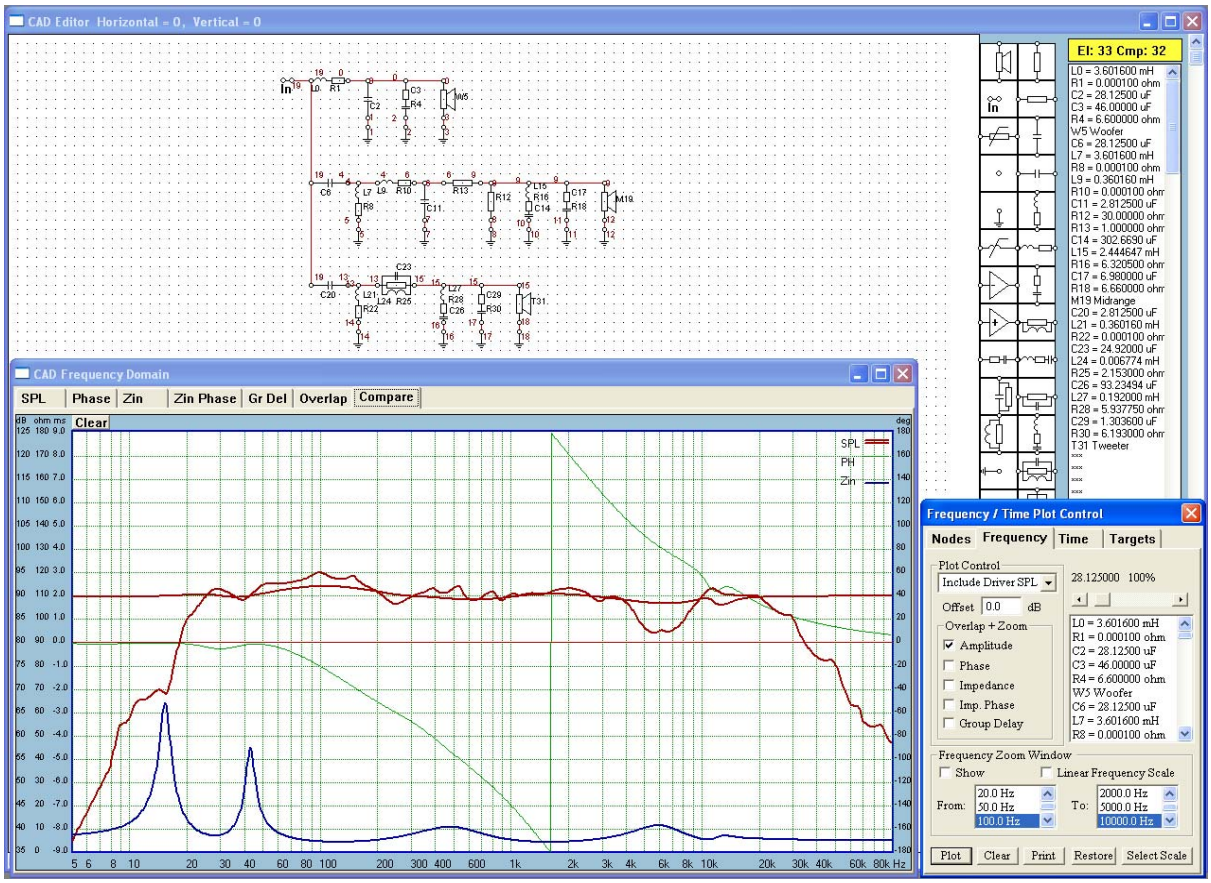


Fig 9.12 Complete 3-way crossover

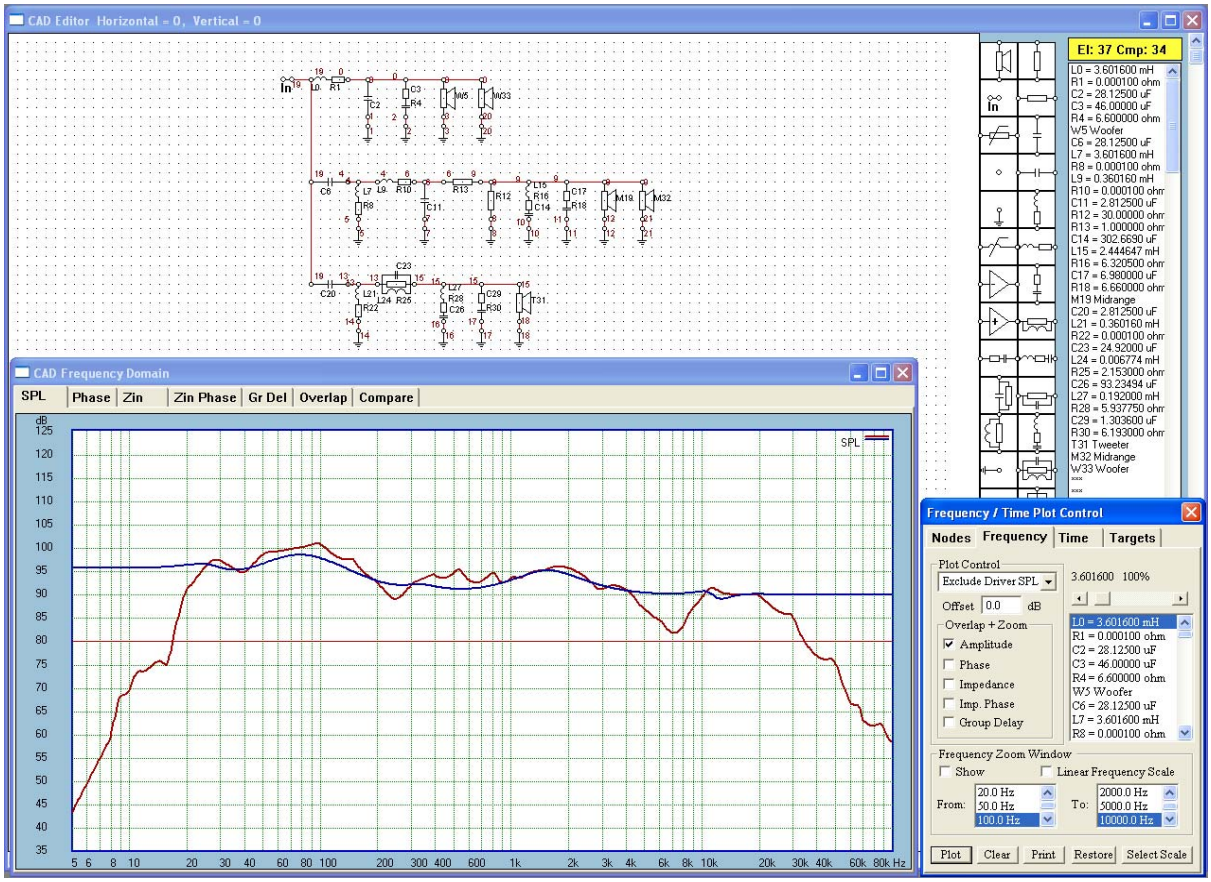


Fig 9.13 Complete WMTMW crossover

At the beginning of the design process it is often acceptable to create a “first cut” design, to check if basic ideas are feasible. To facilitate this concept, we have incorporated in the Crossover module a functionality, that allows you to create medium complexity, 5-way crossover and compensation network on one screen. The process is really no different from a single channel filter and you would proceed with the schematic as described before. An example of 3-way crossover is shown on Figure 9.12. You can plot frequency response of each individual channel or a summed response of up to 5 channels. This process is controlled from the dialogue box invoked just prior plotting. Additionally, you can load up to 5 driver files and include their frequency responses and input impedances for evaluating the complete system.

Here you must be aware of the fact, that no spatial (XYZ positioning) information is available at this moment to the program and all drivers are treated as co-located, radiating points. Full XYZ positioning is incorporated in the “System” module and you still need to use it for accurate modeling of the combined frequency response.

Also, you will able to optimize single channel or the complete crossover created this way.

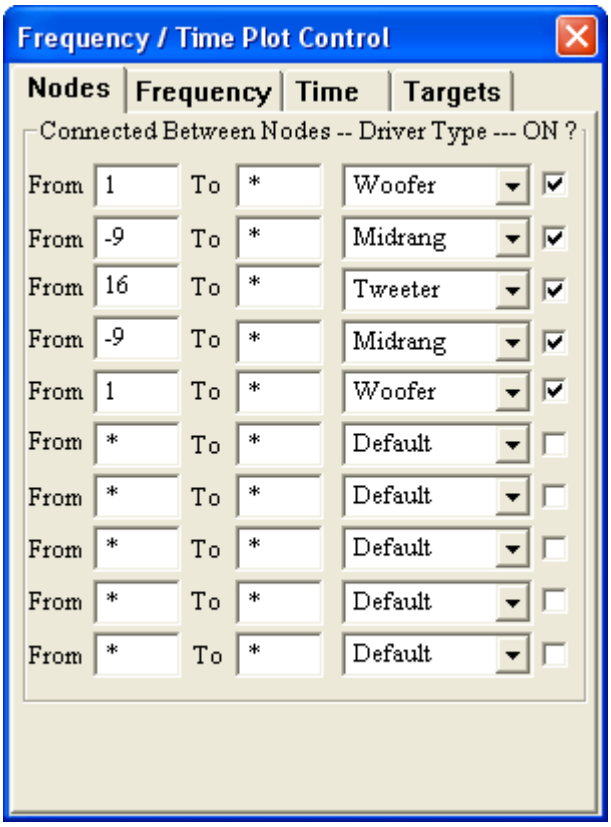


Figure 9.14. Example of node assignment for WMTMW configuration.

Working With Project Files

Project files are manipulated using "New Project", "Open Project" and “Save Project“ options. These three options are designed to accelerate the loading of the three driver's files, keep them together as one project and facilitate tracking of your system design. This file is designed to hold the following:

1. Five paths and file names for the selected drivers.
2. XYZ mounting/tilting/rotating coordinates of each driver.
3. CAD schematic of the complete crossover.
4. System optimisation parameters and more…..

The file names are then used to automatically load the drivers' data files. The names of the drivers' data files can be entered into the project using the "New Project" option from the window menu. In summary, you would follow the steps below:

To create and save Project file:
1. Activate “New Project” menu under "File" main menu option. A dialogue box with 5 editable fields for path names will be opened. Please fill appropriate field for woofer, upper bass and press on “Load” button. The selected files will be loaded into the program’s memory.
2. Save project file using "Save Project" option.

To load Project file:
1. Activate "Open Project" option.
2. Select and load required project.
3. You will be given an option to review the individual drivers' filenames.
4. Press "Load" button to load ALL the files into the program data space. The program will also re-open the default screen and will display all loaded drivers on the front baffle.

In summary:
1. Load all required driver files using “Open Project” option from the “Files” menu. This will ensure, that the frequency response and input impedance data is now available in program’s memory and will be included in the plots. Please refer to further sections of this manual for information on using project files.
2. Create complete crossover and compensation network on one screen using CAD techniques described before. Your schematic should include loudspeaker symbols. We suggest, you save the schematic now – see Fig 9.12 or 9.13.
3. Assign driver type to each loudspeaker symbol. To accomplish this, double-click LEFT mouse button on each loudspeaker symbol. A selection box will be opened for you to choose on of the 5 available driver types. You can now select (enter a number 1 for woofer, ...5 for super tweeter) which driver is assigned to this symbol. Repeat the above for all drivers. This process assigns driver’s input impedance to this particular schematic element (loudspeaker symbol) – see Fig 9.12 or 9.13.
4. Plotting control dialogue box (Fig 9.14) also contains additional fields for assigning drivers to nodes. Now, you need to **assign driver’s frequency response to a particular node and this is accomplished by selecting driver's name from the associated list box.**

An example of this process is shown Figure 9.12, where a 3-way crossover is developed.

1. Woofer (W31) is connected to node 1 (N1),
2. Midrange (M19) is connected in reverse phase to node -9 (N9) and
3. Tweeter is connected to node 16 (N16).

Figure 9.14 shows correctly filled data fields in the plotting control dialogue box. You can observe, that system frequency response will be plotted using the following formula (midrange driver is connected “out of phase”):

Frequency response = Node 1 - Node 9 + Node 16,

Also, Node 1 has woofer driver assigned to it (woofer selected in the associated data field). Node 9 has midrange driver assigned to it (midrange selected entered in the associated data field) and Node 16 has tweeter driver assigned to it (tweeter selected entered in the associated data field). Figure 9.15 shows frequency response of each individual driver + filter and also total system frequency response. Switching ON/OFF of the individual channels is accomplished by inserting * (asterisk) instead of node number in the dialogue box shown on Fig 9.14.

If you just need to disable curve plotting of specific driver(s), you can also use “Do Not Plot” list box option from the same dialogue box.

Built-in Filters (Asymmetrical Band-Pass)

Filter Type	**+6/-XdB/oct**	**+12/-XdB/oct**	**+18/-XdB/oct**	**+24/-XdB/oct**
Butterworth Passive	+6/-6dB/oct	+12/-6dB/oct	+18/-6dB/oct	+24/-6dB/oct
	+6/-12dB/oct	+12/-12dB/oct	+18/-12dB/oct	+24/-12dB/oct
	+6/-18dB/oct	+12/-18dB/oct	+18/-18dB/oct	+24/-18dB/oct
	+6/-24dB/oct	+12/-24dB/oct	+18/-24dB/oct	+24/-24dB/oct

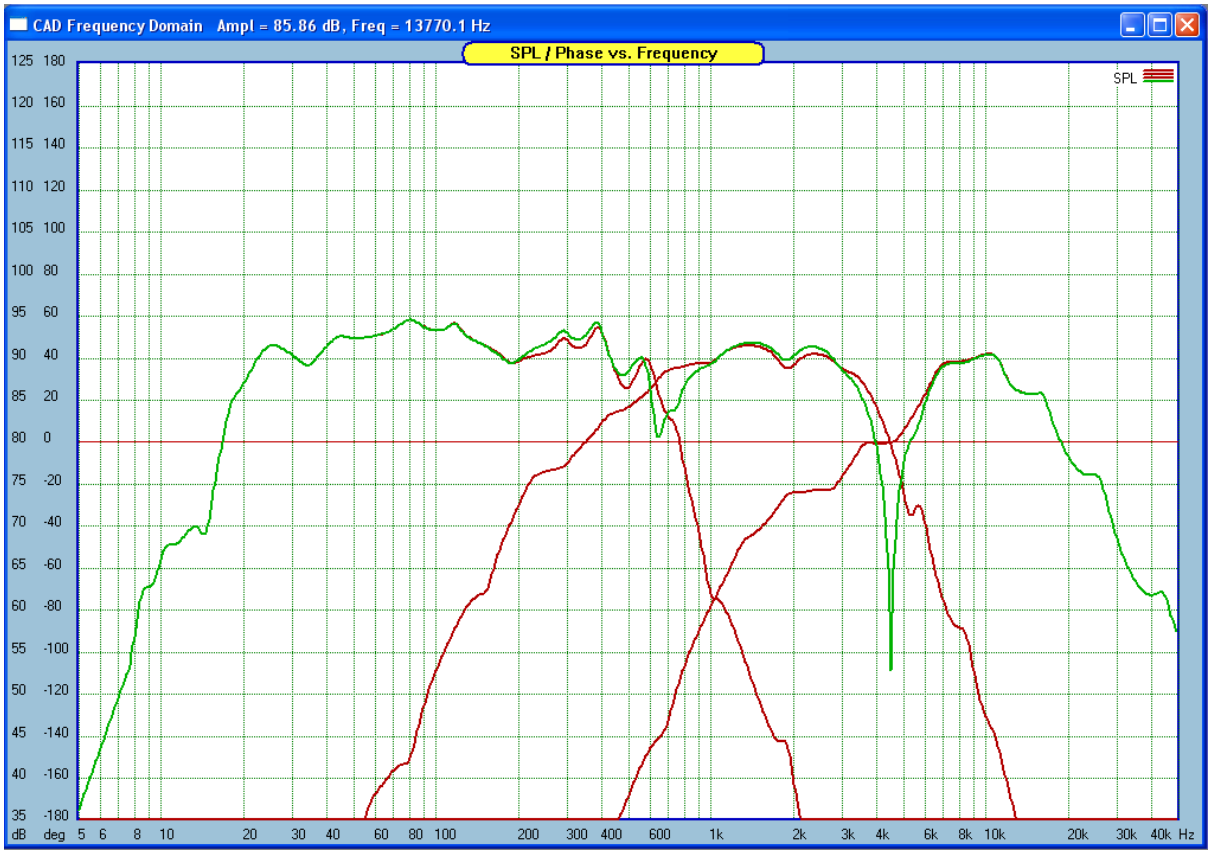


Figure 9.15 shows frequency response of each individual driver + filter and also total system frequency response of the system on Figure 9.12.

Filter Type	Low-Pass	High-Pass	Symmetrical Band-Pass
Bessel Passive	-12dB/oct	+12dB/oct	+/-12dB/oct
	-18dB/oct	+18dB/oct	+/-18dB/oct
	-24dB/oct	+24dB/oct	+/-24dB/oct
	-30dB/oct	+30dB/oct	+/-30dB/oct
	-36dB/oct	+36dB/oct	+/-36dB/oct
Bessel Active	-12dB/oct	+12dB/oct	+/-12dB/oct
	-18dB/oct	+18dB/oct	+/-18dB/oct
	-24dB/oct	+24dB/oct	+/-24dB/oct
Butterworth Passive	-12dB/oct	+12dB/oct	+/-12dB/oct
	-18dB/oct	+18dB/oct	+/-18dB/oct
	-24dB/oct	+24dB/oct	+/-24dB/oct
	-30dB/oct	+30dB/oct	+/-30dB/oct
	-36dB/oct	+36dB/oct	+/-36dB/oct
Butterworth Active	-12dB/oct	+12dB/oct	+/-12dB/oct
	-18dB/oct	+18dB/oct	+/-18dB/oct
	-24dB/oct	+24dB/oct	+/-24dB/oct
Bullock Passive	-6dB/oct	+6dB/oct	+/-6dB/oct
	-12dB/oct	+12dB/oct	+/-12dB/oct
	-18dB/oct	+18dB/oct	+/-18dB/oct
	-24dB/oct	+24dB/oct	+/-24dB/oct
Linkwitz Passive	-12dB/oct	+12dB/oct	+/-12dB/oct
	-24dB/oct	+24dB/oct	+/-24dB/oct
	-36dB/oct	+36dB/oct	+/-36dB/oct
Linkwitz Active	-12dB/oct	+12dB/oct	+/-12dB/oct
	-24dB/oct	+24dB/oct	+/-24dB/oct

All-Pass Passive	-12dB/oct	+12dB/oct	+/-12dB/oct
	-24dB/oct	+24dB/oct	+/-24dB/oct
			+/-6dB/oct
			+/-18dB/oct
Notched	-24dB/oct	+24dB/oct	

Built-in Crossovers

Filter Type	**2-Way**	**3-Way**	**4-Way**	**5-Way**
Bessel Passive	-6dB/oct	+6dB/oct	+/-6dB/oct	+/-6dB/oct
	-12dB/oct	+12dB/oct	+/-12dB/oct	+/-12dB/oct
	-18dB/oct	+18dB/oct	+/-18dB/oct	+/-18dB/oct
	-24dB/oct	+24dB/oct	+/-24dB/oct	+/-24dB/oct
Bessel Active	-12dB/oct	+12dB/oct	+/-12dB/oct	+/-12dB/oct
Butterworth Passive	-6dB/oct	+6dB/oct	+/-6dB/oct	+/-6dB/oct
	-12dB/oct	+12dB/oct	+/-12dB/oct	+/-12dB/oct
	-18dB/oct	+18dB/oct	+/-18dB/oct	+/-18dB/oct
	-24dB/oct	+24dB/oct	+/-24dB/oct	+/-24dB/oct
Butterworth Active	-12dB/oct	+12dB/oct	+/-12dB/oct	+/-12dB/oct
Linkwitz Passive	-12dB/oct	+12dB/oct	+/-12dB/oct	+/-12dB/oct
	-24dB/oct	+24dB/oct	+/-24dB/oct	+/-24dB/oct
Linkwitz Active	-12dB/oct	+12dB/oct	+/-12dB/oct	+/-12dB/oct
Butterworth type 1 Passive	-12dB/oct	+12dB/oct	+/-12dB/oct	+/-12dB/oct
	-24dB/oct	+24dB/oct	+/-24dB/oct	+/-24dB/oct
Butterworth type 1 Active	-12dB/oct	+12dB/oct	+/-12dB/oct	+/-12dB/oct
Notched	-24dB/oct			

Built-in Networks

Network Type
Passive Lattice Time Delay Networks
Linkwitz-Riley Transformer
Time Delay – First Order Circuit
Transient Perfect (TP) second order
Butterworth Q/Fo Type
Active Q-Boost
3 types of Bi-Quad circuits (HP, LP, BP)
Lintwitz & Butterworth asymmetrical 12/24dB filters.

Compensation of Diffraction Distortion

As we discussed before, diffraction distortion is best observed in the anechoic chamber. Here, the loudspeaker radiates into full surrounding space (4π radiation). However, portion of the sound is reflected from the cabinet edges and combines with the direct sound at the observer location. Resulting SPL depends on the frequency and total geometry of design. An example of this process is shown on Figure 9.16, where the front panel dimensions are X=40cm and Y=60cm and the driver is located towards the bottom of the panel. It is observable, that SPL starts rising quite early, at 50-60Hz region for this design, reaches +6dB "gain" at 300Hz and continues at this level developing small bumps.

Compensating for this frequency response distortion revolves basically around three options: (1) acoustical attenuation of the sound travelling towards the edges of the cabinet. The aim here is to prevent the diffraction phenomenon at the first place.

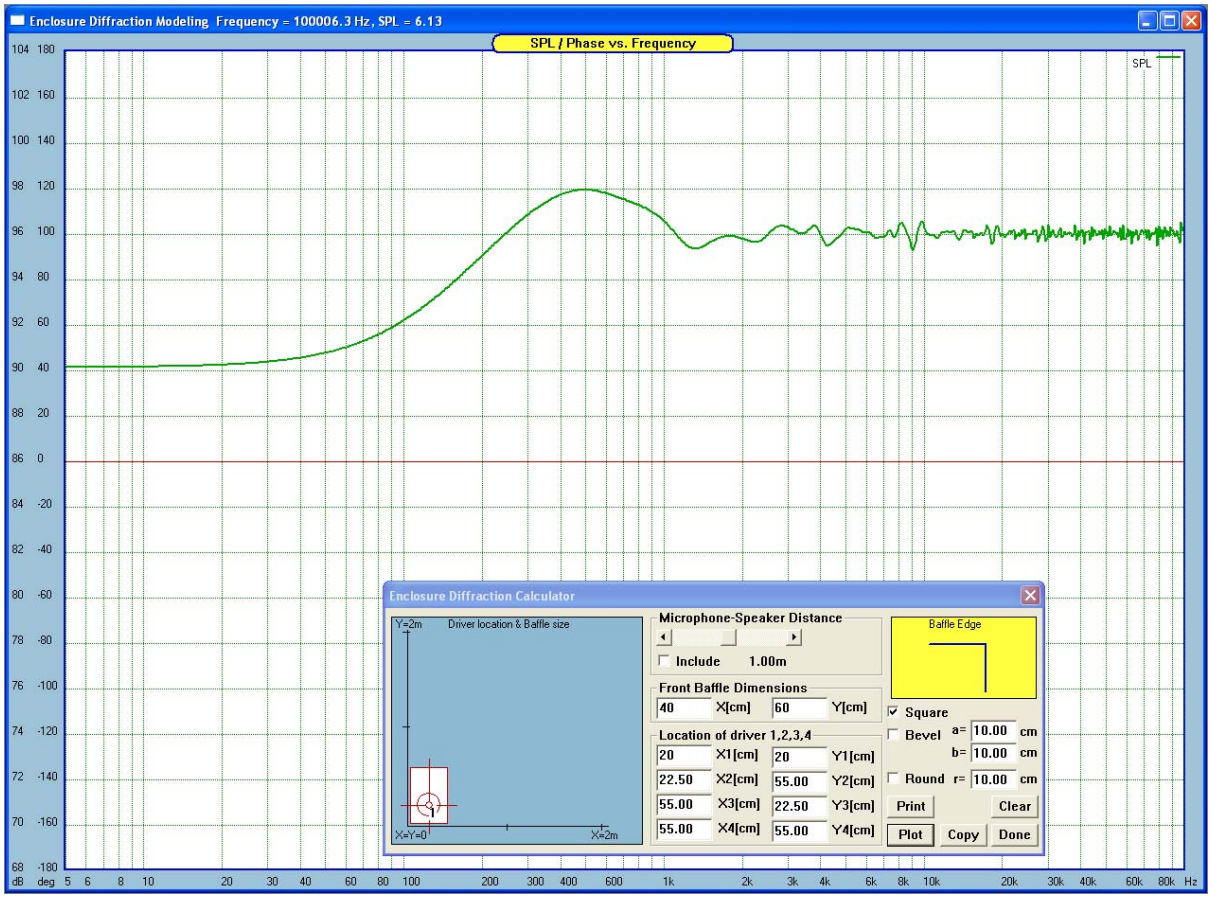
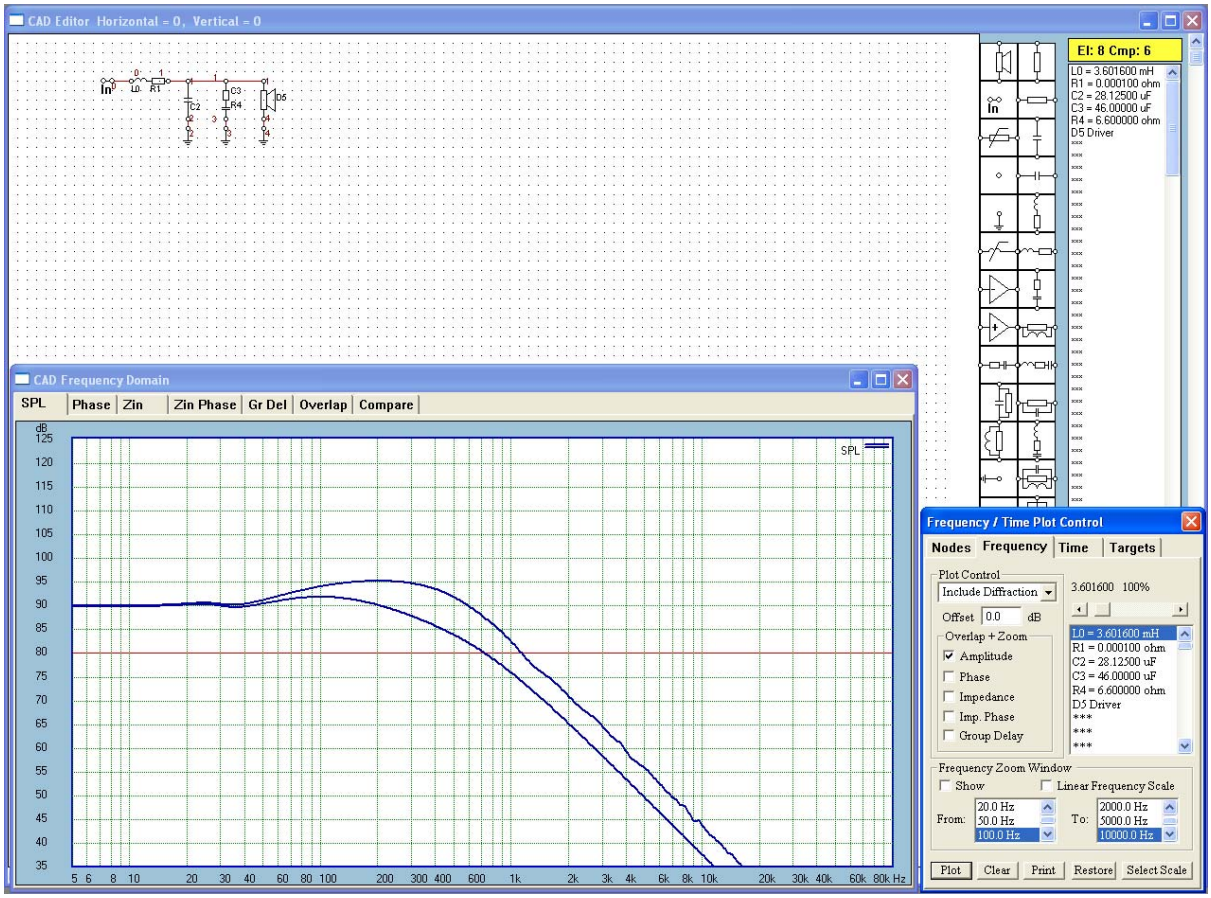


Fig 9.16 Diffraction distortion



Good results have been reported by placing sound absorbing material on the front panel around drivers and making the cabinet edges rounded. (2) D'Appolito configuration. By adding second woofer on the front panel, the low frequency SPL gain can be as high as +6dB. Once again, by careful selection of all contributing system parameters, one can reduce the +6dB diffraction step. Finally, (3) electronic compensation. Figure 9.17 shows two curves (1) crossover frequency response when connected to a real driver as a load impedance - Network Amplitude and (2) as before + diffraction loss. The extra SPL gain due to the diffraction needs to be compensated.

Electronic compensation could be arranged as passive or active circuit. However, if the passive option is chosen, you need to be aware of possible problems associated with this option. Our problem could be stated as finding a way to reduce system gain within certain frequency range. Therefore, our compensation circuit needs to contain reactances (capacitors and inductors). These additional reactances will combine with the crossover's reactances and most importantly, with already existing loudspeaker input impedance. Therefore, the resulting frequency response will be a compromise between reducing the diffraction distortion and introducing additional ripple into the frequency response. This problem is illustrated on Figure 9.18, where the +6dB diffraction step has been equalized, at the expense of +2/-1.5dB ripple below 100Hz. The circuit used was a parallel tank circuit (R7||L6||C8) and also low-pass crossover component (L0) value was modified.

Somewhat better result was obtained when active compensation circuit was implemented. Figure 9.19 shows two curves: the frequency response of the compensation circuit itself (measured at Node 3 - Excluding Driver) and the total system frequency response (measured at Node 5 - Include Enclosure Diffraction).

In order to include the enclosure diffraction effect in the graphs, please select "Include Encl. Diffr." option from the list box of the plotting screen. The plots will not contain driver's acoustic frequency response.

Also, the diffraction distortion gain is NOT included in the crossover optimization process.

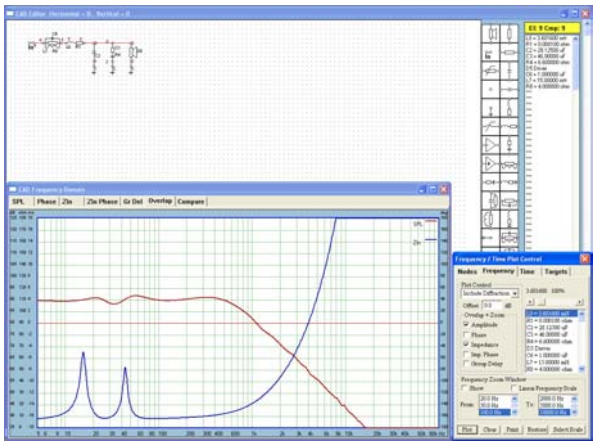


Fig 9.18 Passive equalization of diffraction

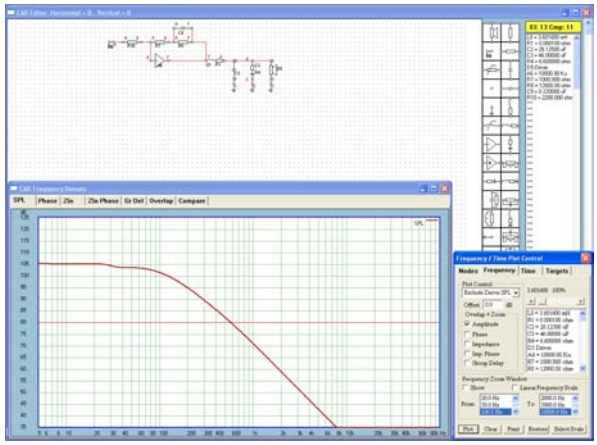


Fig 9.19 Active equalization of diffraction.

Passive Time Delay Circuits

Current SoundEasy release also incorporates two passive time delay circuits: First Order Lattice and Second Order Lattice. Both networks are accessible from the “Built-in” filters dialogue box.

1. Open the **Edit -> “Built-in Filter”** dialogue box.
2. Select **“Passive Lattice 1-st” or “Passive Lattice 2-nd”** from the “Filter Type” list box.
3. This will invoke **“Lattice Network Calculator”** dialogue box.
4. Enter the required **“Design Parameters”** and click on **“Exit”.**
5. Once you press the Left Mouse Button above the CAD screen, the Lattice Network will loaded and displayed on the screen.

You can easily cascade the networks to obtain larger delay. An example of three cascaded time delay, second order networks are shown on Figure 9.20. Please remember to keep only one Input Node in the circuit. As you can see, with the exception of terminating resistor, R36, load resistors in the preceding networks were replaced by the consecutive lattice network. Plot “0” – Nodes: 1 and -2, Plot “1” – Nodes: 4 and –5, Plot “2” – Nodes: 6 and –7.

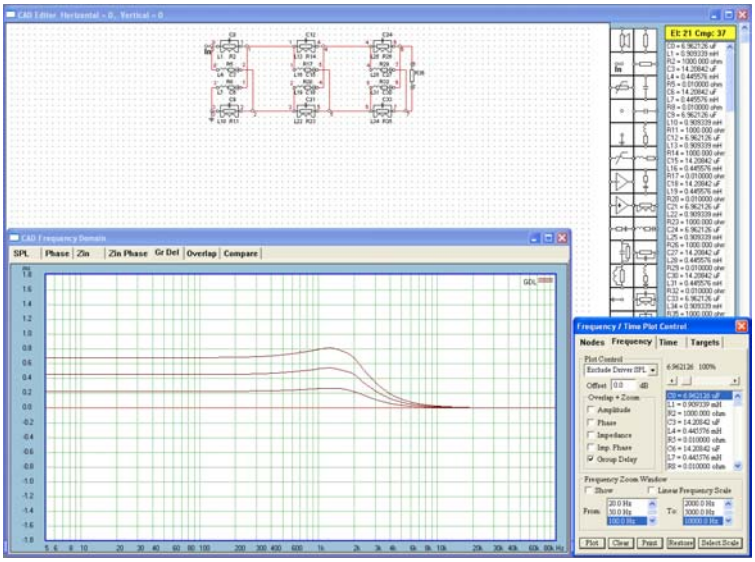


Figure 9.20. Example of cascaded Second Order Lattice and their Group Delay plots.

Multi-way Transient Perfect (TP) Crossovers

Current release of the program provides you with a full support for design and optimization of passive and active, multi-way Transient-Perfect (TP) crossovers. The TP crossovers are a class of pre-distorted filters with suitable modified frequency response, so that a square wave run through such a design, will maintain its perfect shape after the summation. Here is a short example:

Woofer:	Low-Pass,	Fc=500Hz,	Node 7
Midrange:	Band-Pass,	500-5000Hz,	Node 18
Tweeter:	High-Pass,	Fc=5000Hz,	Node 21

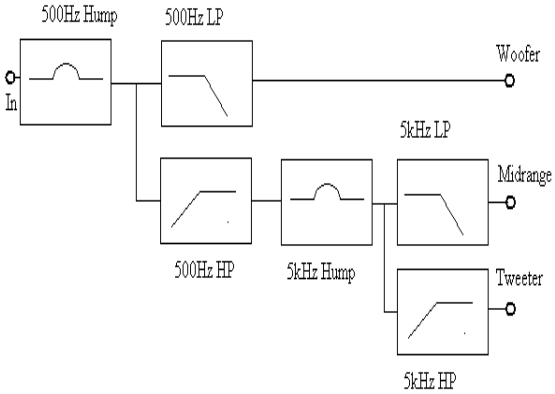


Figure 9.21 3-Way TP Crossover block diagram

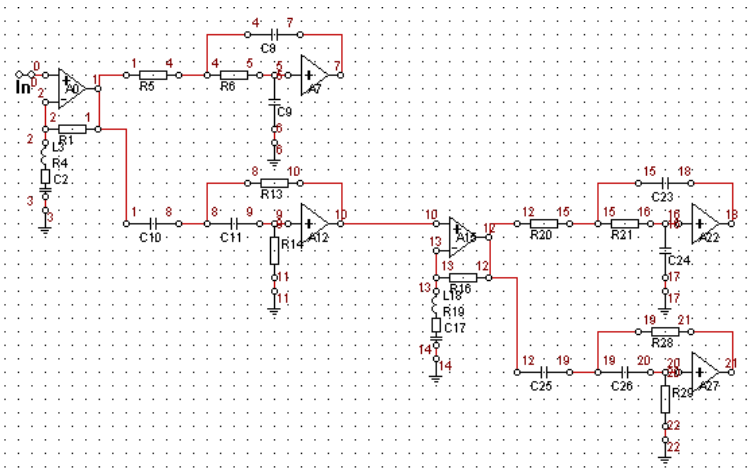


Figure 9.22. 3-Way TP Crossover

Principle of creating multi-way TP networks:

Cascade another 2-way TP network (Hump+LP+HP) from tweeter port. Therefore, 4-way, 5-way… crossovers can be assembled the same way.

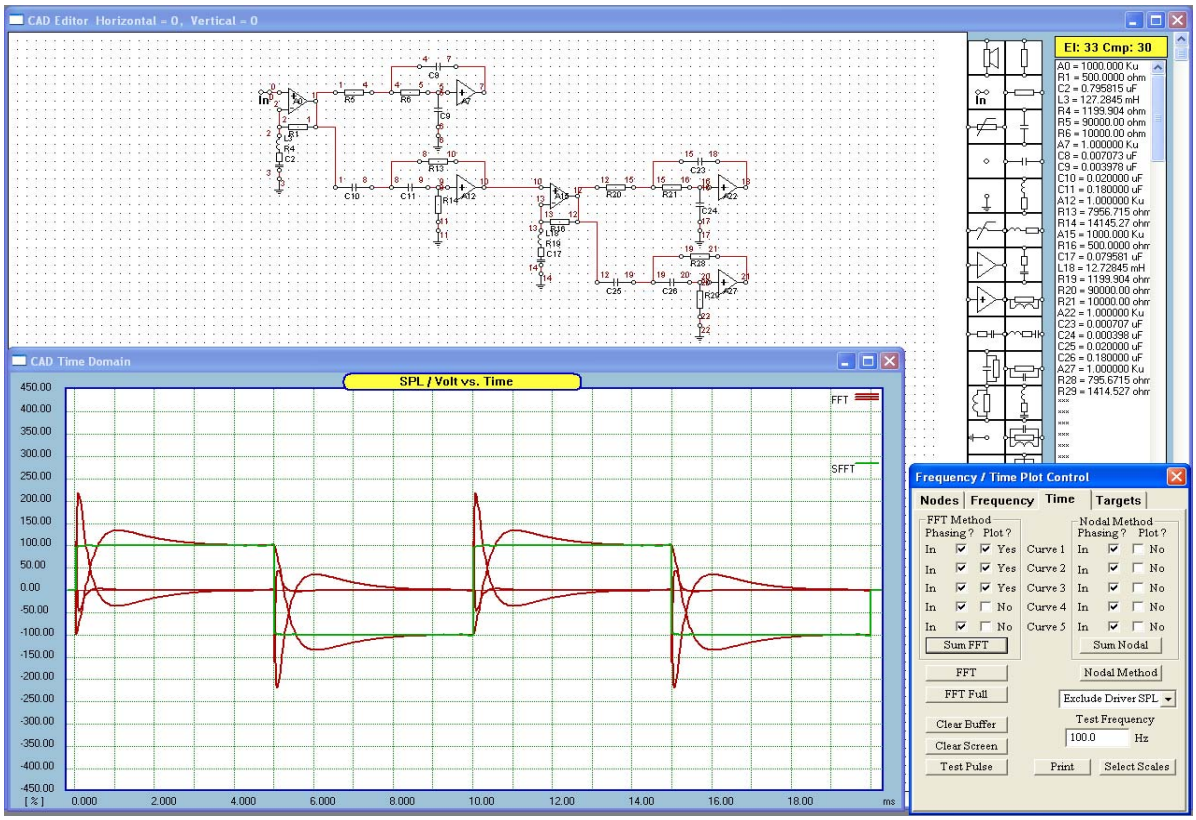


Figure 9.23 Green color is the summed time response of 3 channels – perfect square wave

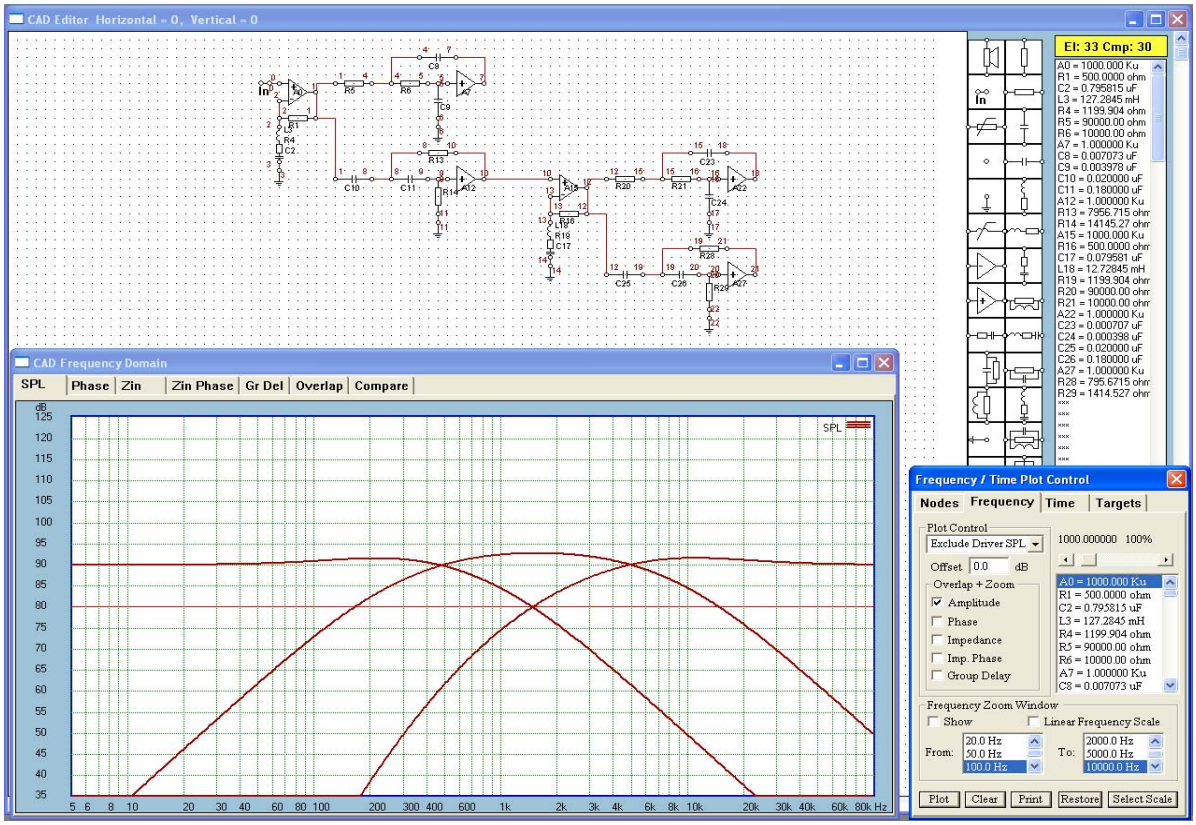


Figure 9.24. Red – SPL – individual channels.

Additional requirements:

1. The active implementation of the 3-way TP network requires three (3) power amplifiers to be connected to each of the crossover outputs. Obviously, 4-way TP network requires 4 amplifiers and so on….
2. Gain of the each channel (power amplifier + speaker's SPL) MUST be set exactly the same.
3. 2-way TP HP/LP filters can be implemented as passive networks, therefore you can get away with single power amplifier.

SoundEasy 2-nd order TP Calculator is shown below. The 2-way, 2-nd order active crossover with EQ correction is your building block. All you need to enter is crossover frequency and overlap parameters. Filter Section parameters and Equalizer Section parameters are calculated automatically from the two mentioned above. However, you can still edit Filter and Equalizer parameters to force the program into "what-if" analysis. Targets for HP and LP sections of the TP crossover are also built-in for optimizations of the full acoustic response of the crossover – see figure below.

Acknowledgement:
The 2-way TP Crossover and Calculator concepts are due to the excellent papers from John Kreskovsky.
(www.geocities.com/kreskovsky/John1/html)

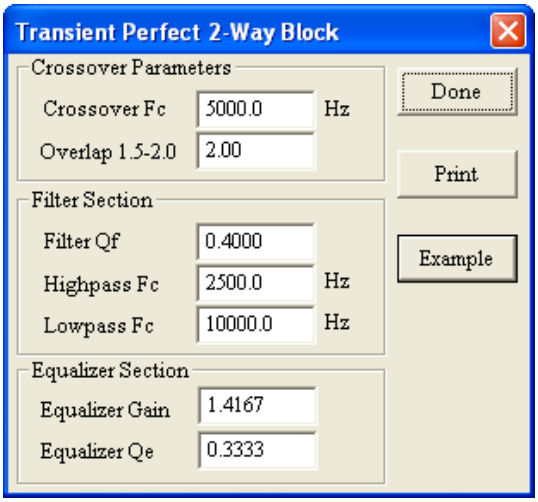


Figure 9.25
2-nd order TP Calculator – your "building block"

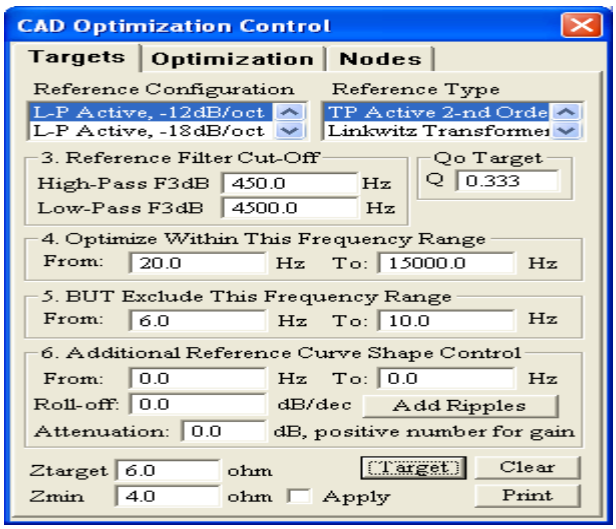


Figure 9.26
HP and LP Optimizer templates for TP, 2-nd order.

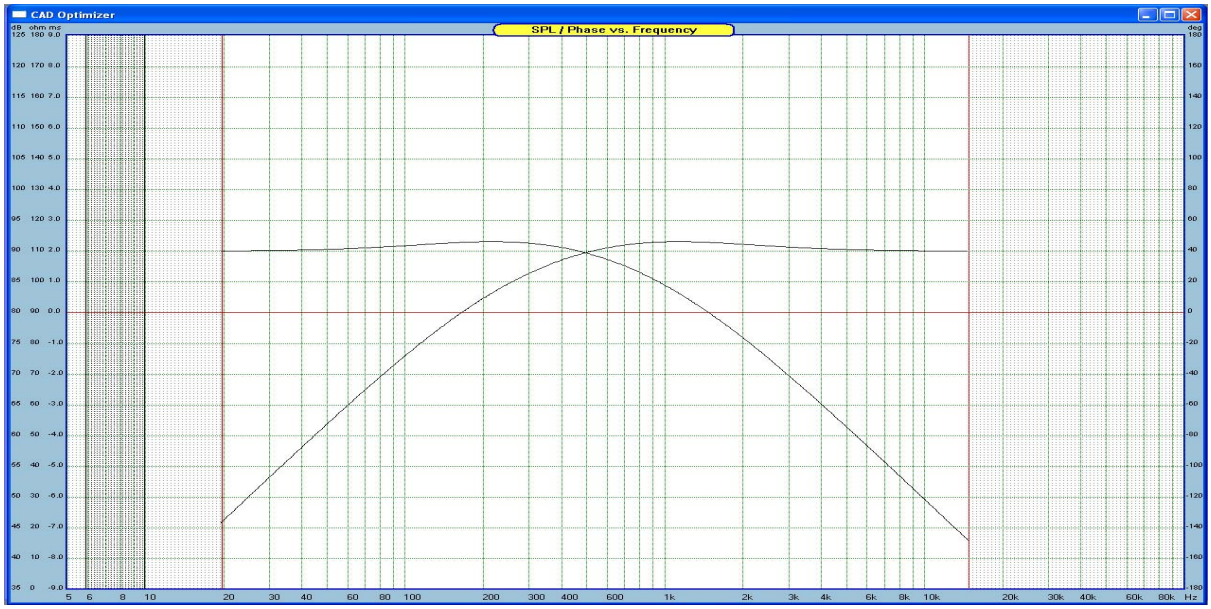


Figure 9.27. Optimizer TP reference curves example.

TP 2-nd order crossover version with GYRATOR replacing inductor in the EQ circuit.

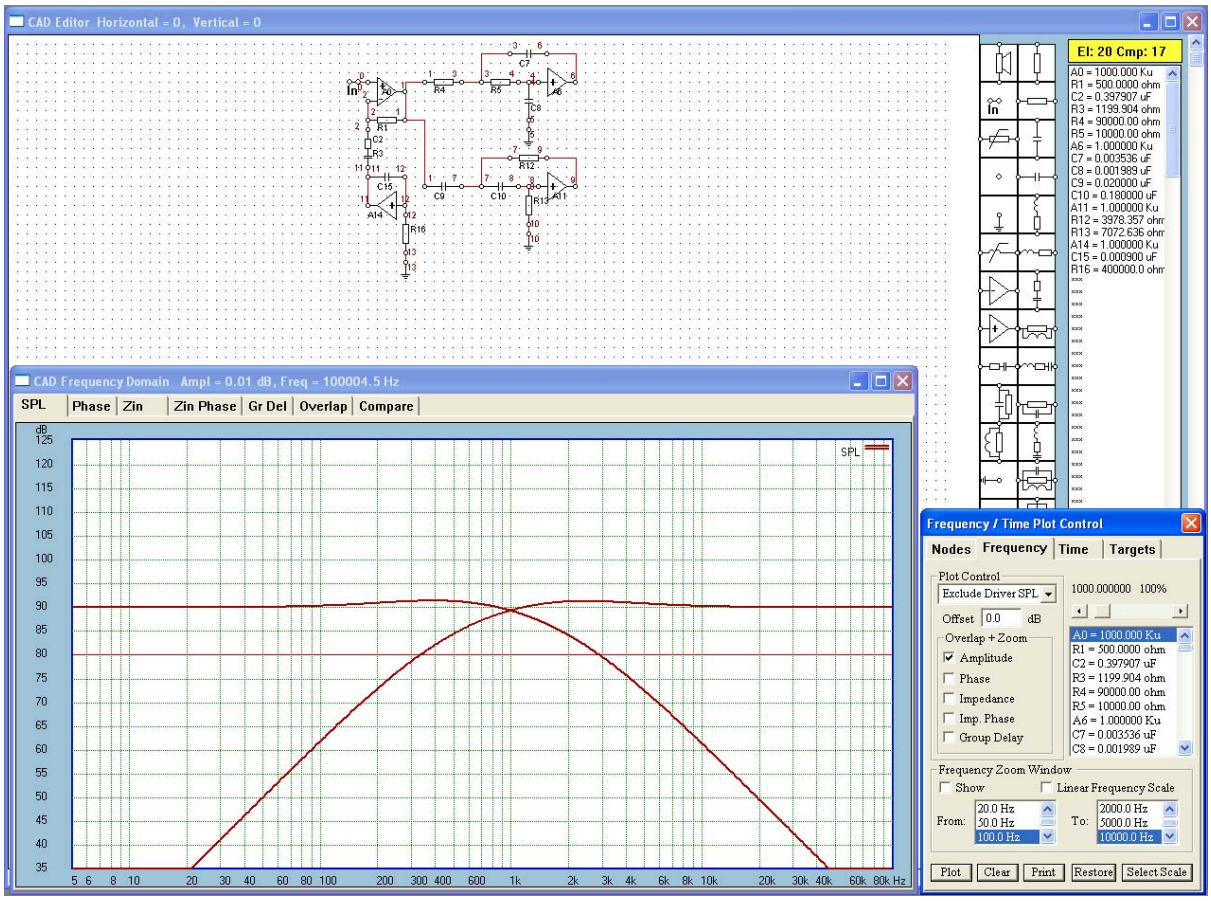


Figure 9.28. TP 2-nd order crossover version with GYRATOR replacing inductor in the EQ circuit

Note: A16 output impedance is the used as one of the GYRATOR's components. Outputs are Node 5 (LP) and Node 9 (HP).

Approximated value of the inductor, L, created with the gyrator:

L = (R17 – Rout)*Rout*C15
Rout = 200 ohm, output impedance of the A16. Rout should be selected from 200-470 ohm.
R17 = 400 kohm
C15 = 0.009 uF
Hence: L = 70mH

Approximated value of the inductor's, Q, created with the gyrator:

Q = XL/(Rout+R4)
Rout = 200 ohm
R4 = 1100 ohm
L = 0.070 H
Hence: Q = 0.33

Approximated value of the gain, G, created with the gyrator:

G = 1 + R2/(Rout+R4)
Rout = 200 ohm
R4 = 1100 ohm
R2 = 500 ohm
Hence: G = 0.385

Digital Filter

There are several methods of implementing a digital filter on your PC. One that looks attractive and is simple to use would possibly involve:

1. Connecting your CD-player to the sound card input,
2. Connecting several identical power amplifiers + loudspeaker drivers to the sound card outputs,
3. Generating required filter transfer functions – using CAD screen and
4. Playing music through the sound card – now your "digital crossover".

Set up described above, requires your to purchase and measure drivers and actually built the enclosures, therefore the procedure significantly increases costs of your design process and really goes well beyond computer modelling.

There are two basic areas where experimenting with digital filters may be of advantage:

1. Quick comparison of filter topologies. You may be interested in actually listening to a sub-woofer channel with different crossover orders or type or EQ-circuit.
2. Equalizing driver channel response. For instance, your goal is to produce **acoustical response of the driver channel that is identical to 4-th order LR filter**. What is needed in this case, is the digital equalizer, that will correct the imperfect driver's response (ALL small SPL irregularities) to the ideal 4-th order LR filter.

In any case, you must exercise extreme care with amplifiers in order to avoid potential damage to drivers from incidental signals that are outside their intended frequency range.

To accomplish the above, the program implements important DSP technique, **the overlap-add method, and FFT convolution**. The overlap-add method is used to break long time domain signals into smaller segments for easier processing. FFT convolution uses the overlap-add method together with the Fast Fourier Transform, allowing signals to be convolved by multiplying their frequency spectra. For longer time domain samples, the FFT convolution tends to be faster than standard convolution, while producing exactly the same result.

FFT convolution uses the principle that multiplication in the frequency domain is equivalent to convolution in the time domain. The sampled 8192-sample blocks of input signal is transformed into the frequency domain using the FFT, multiplied by the frequency response of each filter, and then transformed back into the time domain using the Inverse FFT. By using the FFT algorithm to calculate the Discrete Fourier Transform, convolution via the frequency domain can be **faster** than directly convolving the time domain signals. The final result is the same; only the number of calculations has been changed by a more efficient algorithm. For this reason, FFT convolution is also called **fast convolution**.

The overlap-add method is based on the fundamental technique in DSP: (1) decompose the input signal into smaller blocks, (2) process each of the blocks using fast convolution, and (3) recombine the processed blocks into the output, time-domain signal. Fourier Transform treats our 8192-sample block of input data, X(t) and the filter impulse response H(t) as though they are periodic. Convolution treats them as thought they are not. In order for convolution theorem to be of any use in our case, we need to either perform the convolution in a "circular" manner (which often causes problems at the edge of the processed blocks), or pad the blocks with zeros, so that "circular convolution" becomes "linear convolution".

We anticipate to use the FFT routine to build our "block filter" because it requires far fewer calculations than a conventional FIR filter for long impulse responses. For example: For an impulse response of length N, an FIR filter does N^2 calculations for each block of N input samples. A block FFT filter has to do 2 FFTs and N multiplications for each block of N input samples. This is $2(N*\log_2 N)+N$ calculations. If N=1024, then the FIR filter does 1024^2 = **1 million calculations** for each input block, while the FFT filter does 2(1024*10)+1024 = **21 thousand**. However, the problem with our "Block filter" right now is the circular convolution operation does not exactly match the linear convolution operation. In practical application, you would hear quite loud "clicks" from the loudspeaker when using circular convolution-based block filter.

The method described here is a popular way to overcome the problems with circular convolution. There are actually two popular methods: the overlap-and-add method and the overlap-and-save method. Presented below is only the first one. In our case, the basic difference between linear and circular convolution is that the block of the 8192-sample input signal is appended with a 8192-long string of zeros. The total length of the input block of data is now 16384-samples. These zeros allowed the last impulse in the input signal to generate it's impulse response without wrapping around to the start of the signal. The block FFT filter we are trying to design, should do the same thing.

Another thing worth noticing, is that we can save some computational time for each block of input data, because the filter transfer function, H(f) does not change for each block. We can calculate it once, store and re-use for all input blocks of data. Please also notice, that the program does not need to start calculating the filter transfer function as the FFT of its impulse response padded with N-zeros. We already have the filter's transfer function represented in frequency domain. Listed below is a step-by-step algorithm for an impulse response that is N samples long:

1. Calculate H(f) – this is actually done by the CAD screen functions – zero padding "equivalent" is performed in frequency domain. The H(f) is now 2N long.
2. Read-in N samples from the input waveform device of the soundcard.
3. Pad the N samples with N zeros – now, the sample length is 2N long.
4. FFT the input block of samples, and designate it as X(f) – FFT is now 2N long.
5. Multiply X(f) and H(f) to obtain Y(f) – there are 2N multiplications.
6. IFFT Y(f) to get the time domain signal, y(t) - the IFFT is also 2N long.
7. Divide y(t) into 2 blocks, the N1 and N2 samples, N1=N2.
8. The N1 samples are added to the N2 samples from the previous block, then written out to the soundcard.
9. The N2 samples of current y(t) are stored for re-use in the next block.
10. Go back to step 2 for next block of input data.

For "linear convolution" the formula: $2(N*\log_2 N)+N$ needs to be modified. Firstly, after zero-padding, the FFTs are performed over 2N blocks of data. Secondly, the number of multiplications is also doubled to 2N. Therefore, the final formula is:

$$2(2N*\log_2(2N)) +2N = 2(2N*(\log_2(2)+ \log_2(N))+2N = 4N(1+ \log_2(N))+2N = \mathbf{6N + 4N* \log_2(N)}$$

Taking into account the figure (N=1024) from the previous example: 6*1024 + 4*1024*10 = **46 thousand**. This is still much less than **1 million calculations** for each input block processed with FIR filter. The method described above was implemented in the program.

Operation of the Digital Filter Dialogue Box

Before you can use the Digital Filter function, you need to have the filter or crossover completely designed on the CAD screen. This step is no different from your normal CAD activity. You should also plot the amplitude response of the filter to make sure, that design works correctly. Also, if your filter has too much gain, you will drive the sound card output DAC into saturation. **We strongly recommend, you should start slowly and familiarize yourself with the methodology and the set-up quite carefully**. A good starting point is a simple RC low-pass filter with –6dB/oct slope.

Now, its time to connect the amplifiers to the sound card outputs and your CD-player to the input of the sound card. Please note, that there is no volume control on the input of the sound card, therefore, you need to make sure, that the sound card will not be driven into clipping. The digital filter uses "Line-in" input of the sound card, so chances are, that CD-player output voltage matches the input sensitivity of the sound card "Line-in" input. The volume control on the "Digital Filter" dialogue box is only used to control output level from the sound card. Again, please be careful not to overdrive the output DAC of the sound card, as sever clipping will occur. To make things worse, your loudspeakers are now connected **directly** to amplifiers outputs', so any clipping distortion may cost you dearly.

Before you can play music through the sound card, you need to have the crossover transfer function calculated. To accomplish this, press the **"Calculate Filters" button**.

To start the filtering process, simply press **"Start Filter"** button and to end the process, press the **"Stop Filter"** button. If your sound card is suitable for this function, there will be no warning messages and the sound will appear at the output of the sound card. Here is the functionality of all controls:

1. **Drivers Group**
   - **Dr1** – Check box to turn ON driver connected to the first pair of nodes in the schematic.
   - **Dr2** – Check box to turn ON driver connected to the second pair of nodes in the schematic.
   - **Dr3** – Check box to turn ON driver connected to the third pair of nodes in the schematic.
   - **Dr4** – Check box to turn ON driver connected to the fourth pair of nodes in the schematic.
   - **Dr5** – Check box to turn ON driver connected to the fifth pair of nodes in the schematic.
   - **Delay [ms]** – Enter required delay for each driver.

2. **Sample Rate** – Select 48kHz OR 44.1kHz ONLY.
3. **Calculate Filters** – Press this button to calculate and plot SPL /Phase of your individual filters (channels) in the crossover. You must have your filter transfer function calculated first, before you can start filtering the music.
4. **Start Filter** - Press this button to start Digital Filter. You should now hear the sound in each channel, filtered by your crossover.
5. **Stop Filter** – Press this button to suspend the Digital Filter function.
6. **Plot System** – Press this button to plot total crossover response.
7. **Clear** – Press this button to clear plots.
8. **Done** – Press this button to close the dialogue box and exit.
9. **Select Curve For Plotting Group**
   - Phase – Check this box for plotting filter and system phase.
   - Amplitude – Check this box for plotting filter and system SPL.
10. **Volume** – Use this slider to adjust sound card output.
11. **Mono Group**
    - **1x2 way to 1x4 way** – Check one of these boxes to set the sound card to required mono resolution.
12. **Stereo Group**
    - **2x2 way to 2x4 way** - Check one of these boxes to set the sound card to required stereo resolution.
13. **Component's List Box** – Doubleclick on the component you need to adjust.
14. **Component Adjust Slider** – Use this slider to change component's value from 1% - 1000%.
15. **Restore Component** – You can restore the original component's value by pressing this button.
16. **Project 1** – Press this button to make Project 1 current and active.
17. **Project 2** – Press this button to make Project 2 current and active.
18. **Project 3** – Press this button to make Project 3 current and active.

You can add **Delay to the signal path** for each driver be entering delay in milliseconds in the provided data entry fields. Corresponding physical offsets will be displayed in the "Offset" column. You can also change the component value using the slider, while the DF is active. Switches Dr1 – Dr5 will turn ON/OFF drivers and the processed signal.

When you start the Digital Filter, it is most preferred, that you limit yourself to changing component values only. This is to minimize sound breaks introduced by forcing your computer to deal with additional CPU tasks.

You may also notice, that Digital Filter plotting window can not be customized – for the same reason – to minimize CPU usage. If you need to change the size or location of the plotting window, please Stop the filter first, close the control box and select "Digital Filter Plotting" main menu option. Now, you can change the parameters of the window, and they will be "remembered" when you use Digital Filter next time.

Please note, that output signal from the soundcard will appear on the lines corresponding to order of nodes in the schematic. That is: Line 1 – will output signal corresponding to the first pair of nodes. Line 2 will output signal corresponding to the second pair of nodes and so on….

This has very important implications when changing projects. YOU MUST ASSURE, that same drivers in each project, are connected to nodes in the same order. Otherwise, tweeter in one project will end up connected to woofer channel in another project.

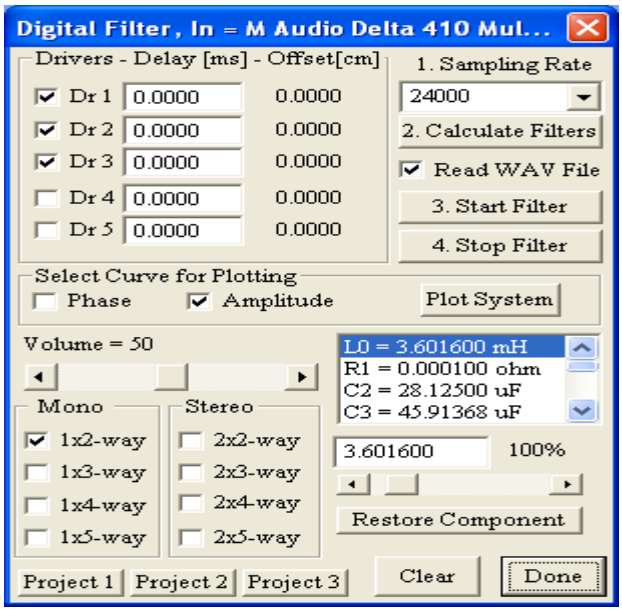


Figure 9.29. Digital Filter Control Box

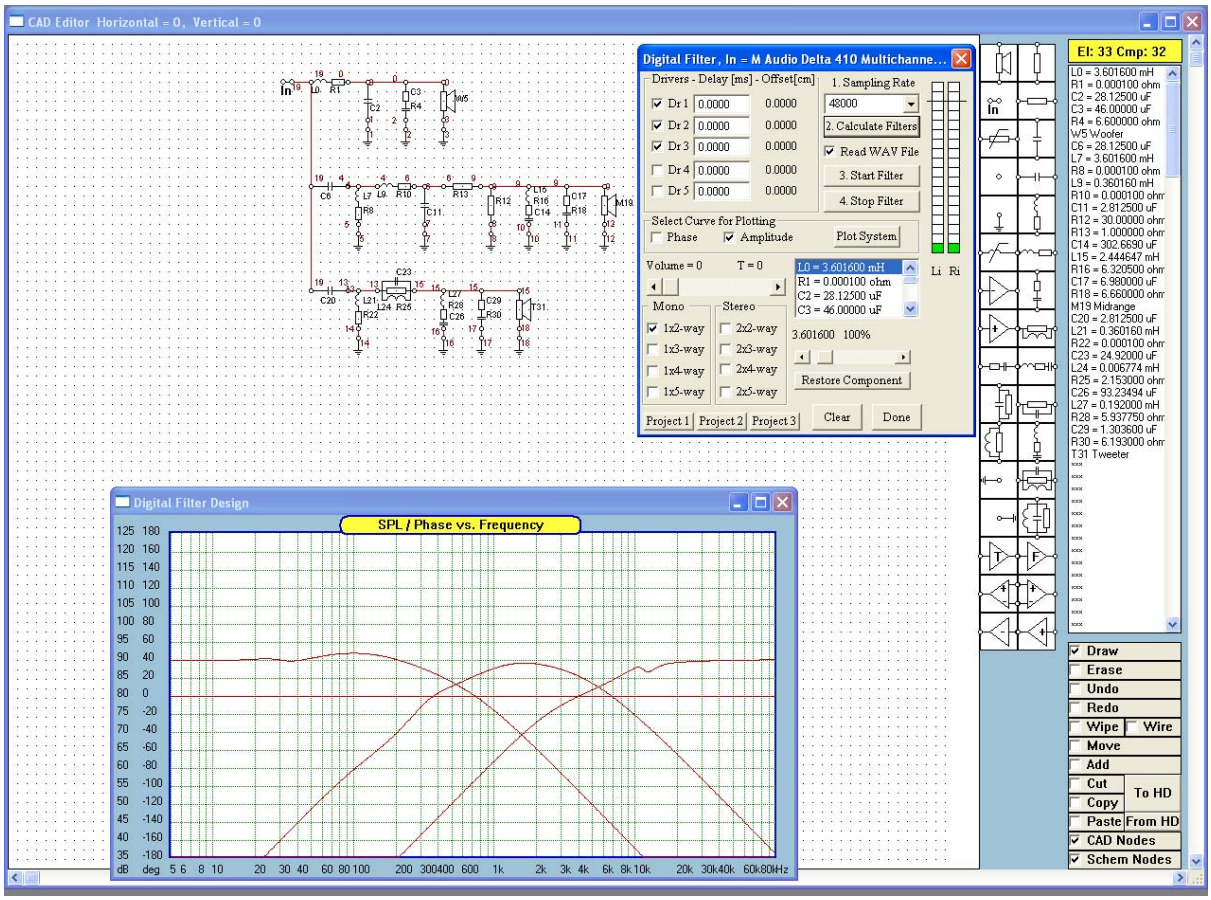


Figure 9.30. Example of a 3-way crossover implemented as digital filters using your sound card.

You can add **delay to the signal path** for each driver be entering delay in milliseconds in the provided data entry fields. Corresponding physical offsets will be displayed in the "Offset" column.

The technique implemented in this release of the program **does require Microsoft ™ Multi-Media Extensions (MME) to be implemented in your sound card**. This specific approach was taken, because it allows you to use quite inexpensive sound card to its full potential. You can have up to 8 independent DSP filtered outputs, that allow you to create and audition 4-way, stereo speaker system. Certainly, there are many 5-way systems available on the market, however, the vast majority of the designs revolve around 3 or 4-way implementations. One modelling option opened for you is to use Digital Filter function to mimic an active crossover. In this case, in the final implementation, all the drivers would be connected to their individual amplifiers, therefore, the input impedance of the driver would not enter the design equation at all. There is no need to use loudspeaker symbols anywhere on the schematic for this option. In another option – as shown in the passive crossover example shown above - the magnitude response of each filter is influenced by the actual input impedance of the driver connected at the output of the filter. Here, we are trying to design a passive crossover, that will eventually be used with real drivers connected across designated outputs. Filter **DSP transfer function automatically excludes driver's SPL**, regardless of the selection in the list box on the frequency response screen. However, you need to connect loudspeakers symbols across your crossover output to get the drivers' input impedances included in the Digital Filter transfer function. This will mimic the behaviour of the passive crossover loaded with real drivers, even though for the purpose of the DSP simulation, your drivers are now connected to the soundcards+amplifiers with near-zero output impedances.

When using the Digital Filter option to audition your crossover design, it is unlikely, that your ears will outperform the results of rigorously performed SPL measurements. Particularly, when your crossover has already been optimised for a specific design goal. Please remember, that one of the most critical tasks of the crossover is to protect your drivers from signals they were not designed to handle. Excessive modifications performed on the crossover "by ear", may affect critical system parameters such as: polar response and system power handling. It is therefore strongly recommended, that you proceed very slowly, with full understanding of "pros" and "cons" of what you are doing. It is impossible for Bodzio Software Pty. Ltd. to predict the exact level of compatibility of all hardware installed in your computer. With our best intentions, we suggest to you to take EVERY precaution when operating the digital Filter option.

BEFORE using the Digital Filter option, please consult Chapter 18 – "Computer Related Issues" section for setting your mixer device and general comments on compatibility.

Disclaimer

Bodzio Software Pty. Ltd. assumes no responsibility for any damage to the hardware running the SoundEasy program or any equipment connected to this hardware. It is entirely User's responsibility to check the signals generated from the sound card before connecting to any external equipment.

Digital Equalizer

The idea of Digital Filter can be extended into **much more advanced function of Digital Equalization**. Our Digital Equalizer implements dual functionality:

1. Filtering signals intended for a given driver – thus operating as normal crossover section,
2. Equalizing driver's irregular SPL within the filter's bandpass frequency range.

Mathematically, the equalizer performs the following operation:

$$EQ(f) = \frac{a + jb}{c + jd}$$

Where (a+jb) is the driver's transfer function and (c+jd) is the target filter transfer function. The driver's transfer function can be imported into the program from an external measurement equipment or measured by built-in MLS or analogue measurement systems.

The issue of Tails

Please refer to Figure 9.31. In every frequency range where the target SPL curve is ABOVE the driver's SPL curve the equalizer (or the EQ(f) function) will attempt to produce gain to compensate for the driver's faster roll-off. This issue is particularly evident for woofers below the –3dB cut-off frequency and tweeters above the –3dB cut-off frequency. An example of a **severe** "tail issue" (EQ(f) - green curve) is shown below.

Figure 9.31. Unmodified driver's SPL response causes severe tailsbelow 26Hz and above 4kHz.

Here, the EQ(f) will produce rapidly rising frequency response, inversely proportional to driver's roll-off – this is often called "tails issue". **Please note, that wherever the target (blue) curve is ABOVE the driver's SPL (black) curve, the <u>EQ(f) (thick green) curve shows gain above 90.0dB (which is the reference efficiency)</u>.** There are three ways to compensate for the tails:

1. Manipulate the nominator the EQ(f) function - here you would employ the Hilbert-Bode Transform to flatten the lowest end of the SPL curve for the woofer.
2. Manipulate the denominator of the EQ(f) function – here, you would modify the frequency response of the target filter.
3. Use (1) and (2) together.

For example, woofer driver can have it's SPL curve modified at the low-end of the audio spectrum. The tool to accomplish this is the Hilbert-Bode Transform. The HB transform will generate mathematically correct transfer function (a + jb) necessary for the rational function EQ(f). It needs to be emphasized, that changing the amplitude alone will not accomplish the task. You need both: real AND imaginary component of the driver's transfer function – this can be easily accomplish using the HB Transform.

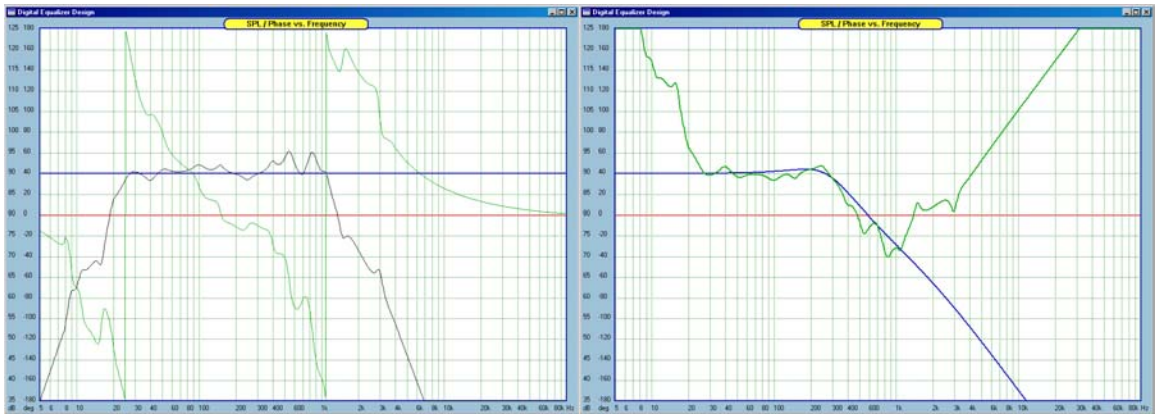


Fig. 9.32. Unmodified driver's response (left) causes severe tails (right) below 26Hz and above 4kHz

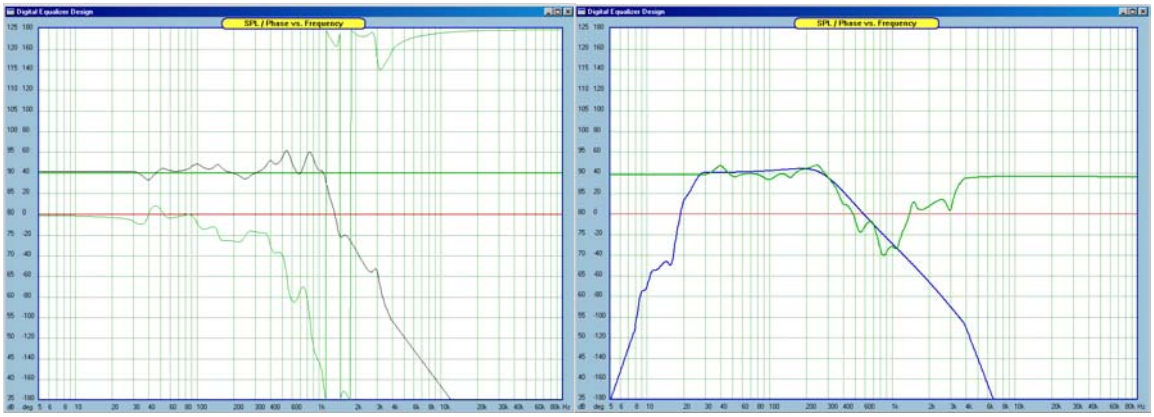


Fig 9.33. Modified driver (left) does not create tails (right)

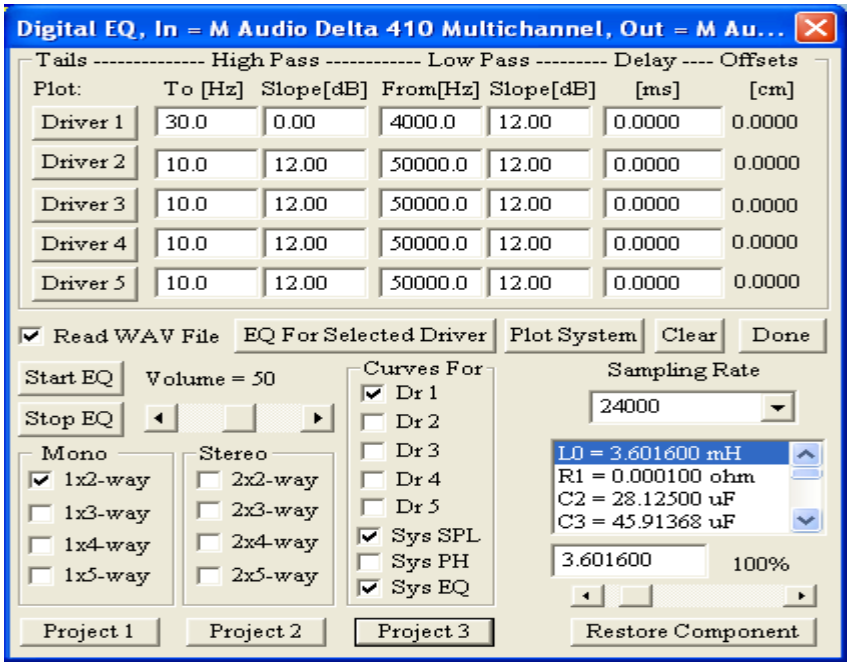


Fig 9.34. Driver 1 HBT settings for the "tail removal"

Recommended Use of the Equalizer

It is assumed, that the equalizer will be used with several audio power amplifiers connected to the output of the sound card. Therefore, input impedance of the drivers will not enter the design process. Also, there is no need for any impedance compensation networks. The Digital Equalizer is invoked from the main menu by selecting the "Digital Equalizer" option.

1. Select your drivers, paying particular attention to acoustic roll-off on low-end and high-end of the operating frequency range.
2. Choose crossover configuration. Here, you should consider the acoustic roll-off speed of the driver vs. the chosen filter. If the target filter rolls-off faster than driver, the tails will not be a problem, otherwise, you will need to modify driver's transfer function (HBT) to get the tails right.
3. Prepare drivers for EQ(f) function using HB transform. All HB Transform parameters are saved into the data file. Ultimately, you will be using project files for the equalizer function and your crossover will be saved as the project file. Note, that you can also use driver file with the equalizer function.
4. Consider how to approach midrange drivers as explained in (2).
5. Open the CAD screen and choose crossover from the built-in selection. It is recommended that the crossover alone be displayed on the CAD screen – no driver impedance modification components should be included in the schematic. However, the crossover can be of any type and include any SPL modification elements. This way you can define quite "esoteric" target functions for all drivers using their crossover sections. **Terminate the crossover with appropriate driver icons**.
6. Inspect the plots and if you are happy with the tails, simply select "Start EQ" button on the "Digital Equalizer" dialogue box. This will start the filtering process and produce the sound at the output of the sound card.

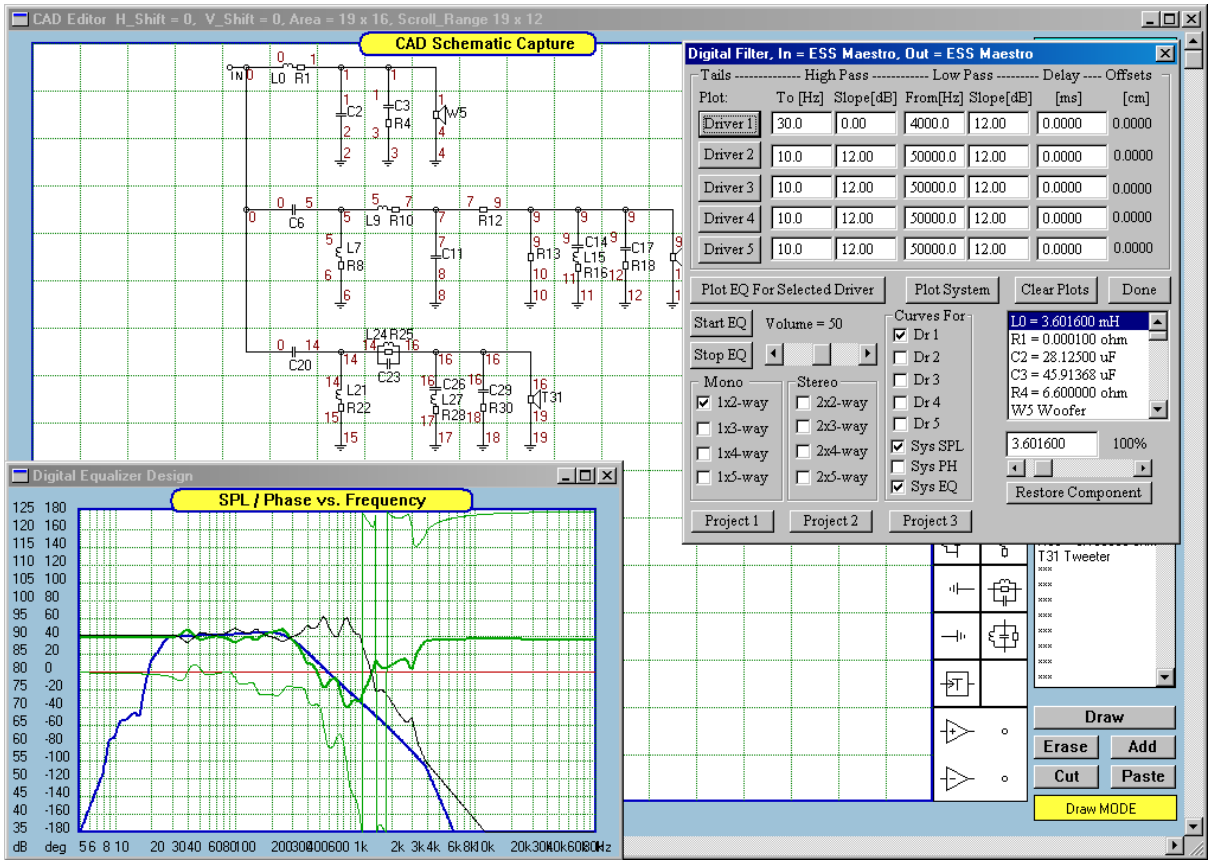


Fig 9.35. Example of all tools used for Digital Equalization

Please note, that output signal from the soundcard will appear on the lines corresponding to order of nodes in the schematic. That is: Line 1 – will output signal corresponding to the first pair of nodes. Line 2 will output signal corresponding to the second pair of nodes and so on….
This has very important implications when changing projects. YOU MUST ASSURE, that same drivers in each project, are connected to nodes in the same order. Otherwise, tweeter in one project will end up connected to woofer channel in another project.

As for the Digital Filter, before you can play music through the sound card, you need to have the crossover transfer function calculated. To start equalization process, simply press **"Start EQ"** button and to end the process, press the **"Stop EQ"** button. If your sound card is suitable for this function, there will be no warning messages and the sound will appear at the output of the sound card. Here is the functionality of all controls:

1. **Tails Group**
   - **Driver 1 - To / Slope / From / To / Delay** – Use these data fields to engage Hilbert-Bode Transform to modify Driver 1 SPL curve for the purpose of Tail Correction.
   - **Driver 2 - To / Slope / From / To / Delay** – Use these data fields to engage Hilbert-Bode Transform to modify Driver 2 SPL curve for the purpose of Tail Correction.
   - **Driver 3 - To / Slope / From / To / Delay** – Use these data fields to engage Hilbert-Bode Transform to modify Driver 3 SPL curve for the purpose of Tail Correction.
   - **Driver 4 - To / Slope / From / To / Delay** – Use these data fields to engage Hilbert-Bode Transform to modify Driver 4 SPL curve for the purpose of Tail Correction.
   - **Driver 5 - To / Slope / From / To / Delay** – Use these data fields to engage Hilbert-Bode Transform to modify Driver 5 SPL curve for the purpose of Tail Correction.
1. **Curves For Group**
   - **Dr1** – Check box to turn ON driver connected to the first pair of nodes in the schematic.
   - **Dr2** – Check box to turn ON driver connected to the second pair of nodes in the schematic.
   - **Dr3** – Check box to turn ON driver connected to the third pair of nodes in the schematic.
   - **Dr4** – Check box to turn ON driver connected to the fourth pair of nodes in the schematic.
   - **Dr5** – Check box to turn ON driver connected to the fifth pair of nodes in the schematic.
   - **Sys SPL** – Check box to enable plotting of SPL curves.
   - **Sys PH** – Check box to enable plotting of phase curves.
   - **Sys EQ** – Check box to enable plotting of EQ curves.
2. **Start EQ -** Press this button to start Digital Equalizer. You should now hear the sound in each channel, filtered by your crossover.
3. **Stop EQ** – Press this button to suspend the Digital Equalizer function.
4. **Plot EQ For Selected Drivers** – Press this button to plot equalizer's SPL for selected driver.
5. **Plot System** – Press this button to plot total crossover response.
6. **Clear Plots** – Press this button to clear plots.
7. **Done** – Press this button to close the dialogue box and exit.
8. **Volume** – Use this slider to adjust sound card output.
9. **Mono Group - 1x2 way to 1x4 way** – Check to set the sound card to required mono resolution.
10. **Stereo Group - 2x2 way to 2x4 way** - Check to set the sound card to required stereo resolution.
11. **Sampling Rate** – Select 48kHz OR 44.1kHz ONLY.
12. **Component's List Box** – Doubleclick on the component you need to adjust.
13. **Component Adjust Slider** – Use this slider to change component's value from 1% - 1000%.
14. **Restore Component** – You can restore the original component's value by pressing this button.
15. **Project 1** – Press this button to make Project 1 current and active.
16. **Project 2** – Press this button to make Project 2 current and active.
17. **Project 3** – Press this button to make Project 3 current and active.

You can add **Delay to the signal path** for each driver be entering delay in milliseconds in the provided data entry fields. Corresponding physical offsets will be displayed in the "Offset" column. You can also change the component value using the slider, while the DF is active. Switches Dr1 – Dr5 will turn ON/OFF drivers and the processed signal. When you start the Digital Equalizer, it is most preferred, that you limit yourself to changing component values only. This is to minimize sound breaks introduced by forcing your computer to deal with additional CPU tasks.

You may also notice, that Digital Filter plotting window can not be customized – for the same reason. If you need to change the size or location of the plotting window, please Stop the filter first, close the control box and select "Digital Filter Plotting" main menu option. Now, you can change the parameters of the window, and they will be "remembered" when you use Digital Filter next time.

If you need to attenuate a driver by certain amount of dBs, the way to implement this is to attenuate the target function by the required amount OR reduce the gain in the power amplifier by the required amount of dBs. If you run a comparison between several equalizers, you may prefer to use the method shown on Figure 9.36, as this will automate the whole process significantly and you will not have to change the power amplifier gain every time. In the example shown on Figure 9.36, the L-Pad circuit resistors are Rs = 3.5ohm and Rp = 10.2ohm. These values are calculated by the built-in L-Pad calculator for the required 5dB attenuation and load resistance of 8ohm.

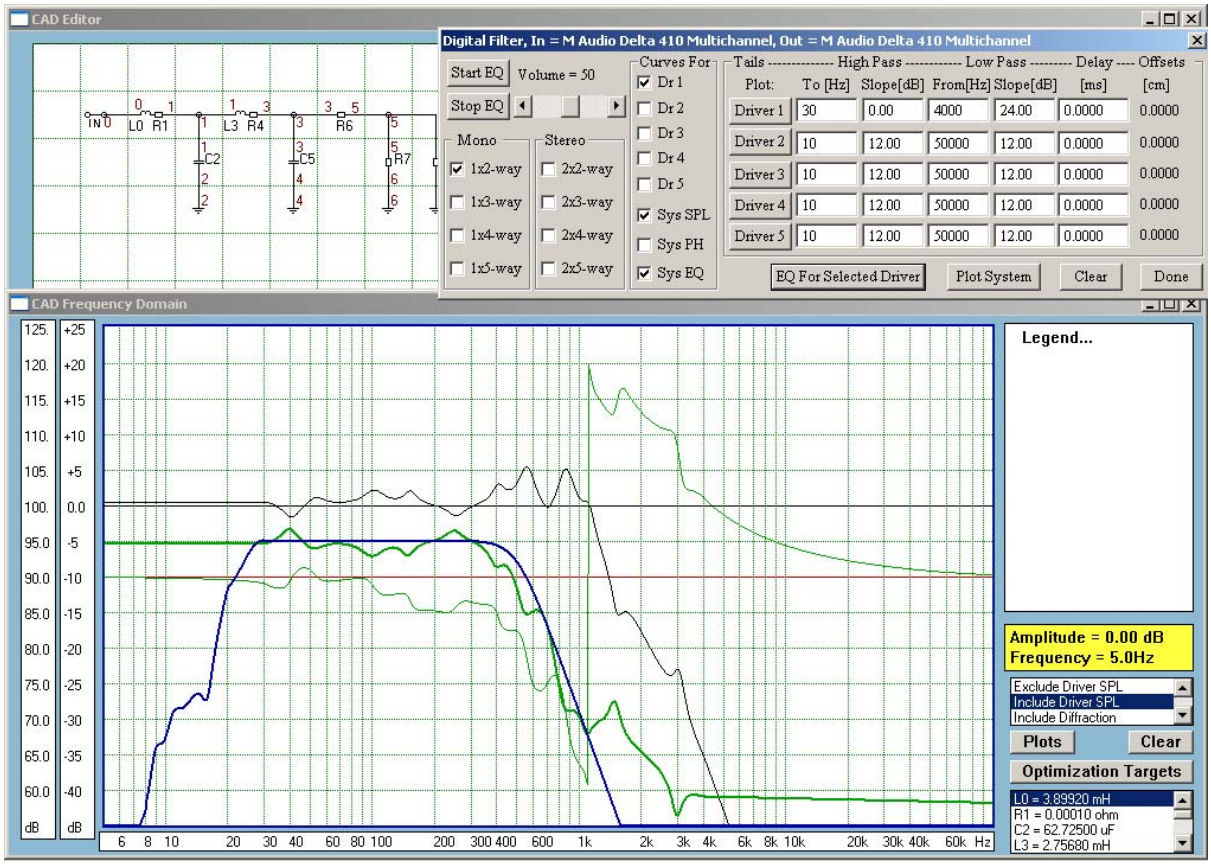


Fig 9.36. Reducing Target level by-5dB – inserting L-Pad 3.5/10.2ohm.

Please remember, that loudspeakers are treated by the equalizer as purely resistive element having a value of R(load) as entered from the Built-in Filter or Crossover dialogue boxes**. All crossover elements (your target curve) are calculated using the same R(load) for terminating resistor. This is necessary for the target functions (ideal filters) to maintain their "perfect" shape, rather than being distorted by the loudspeaker input impedance.** In fact, target function shape can be modified in various ways to produce the desired curve. You can insert any circuit for creating peaks, dips and shelves on the ideal filter frequency response. You can also use active components to generate the target function – however, you MUST be very careful with the gain of the active circuits. Introducing gain in the equalizer may drive the output DAC of the sound card into saturation and clipping

In summary, the equalizer creates a mirror-image of the driver's frequency response and modifies it such a way, that the combined transfer function (driver's SPL + EQ(f)) sums to the requested target curve. The target curve may be the "ideal" LR 4th-order filter or quite "esoteric" curve created specifically for R&D purposes.

It is impossible for Bodzio Software Pty. Ltd. to predict the exact level of compatibility of all hardware installed in your computer. With our best intentions, we suggest to you to take EVERY precaution when operating the digital Filter option. BEFORE using the Digital Equalizer option, please consult Chapter 18 – "Computer Related Issues" section for setting your mixer device and general comments on compatibility.

Disclaimer

Bodzio Software Pty. Ltd. assumes no responsibility for any damage to the hardware running the SoundEasy program or any equipment connected to this hardware. It is entirely User's responsibility to check the signals generated from the sound card before connecting to any external equipment.

See system example below. A 3-way, 24dB/oct Butterworth stock crossover was used with cut-off frequencies of 400Hz and 4000Hz.

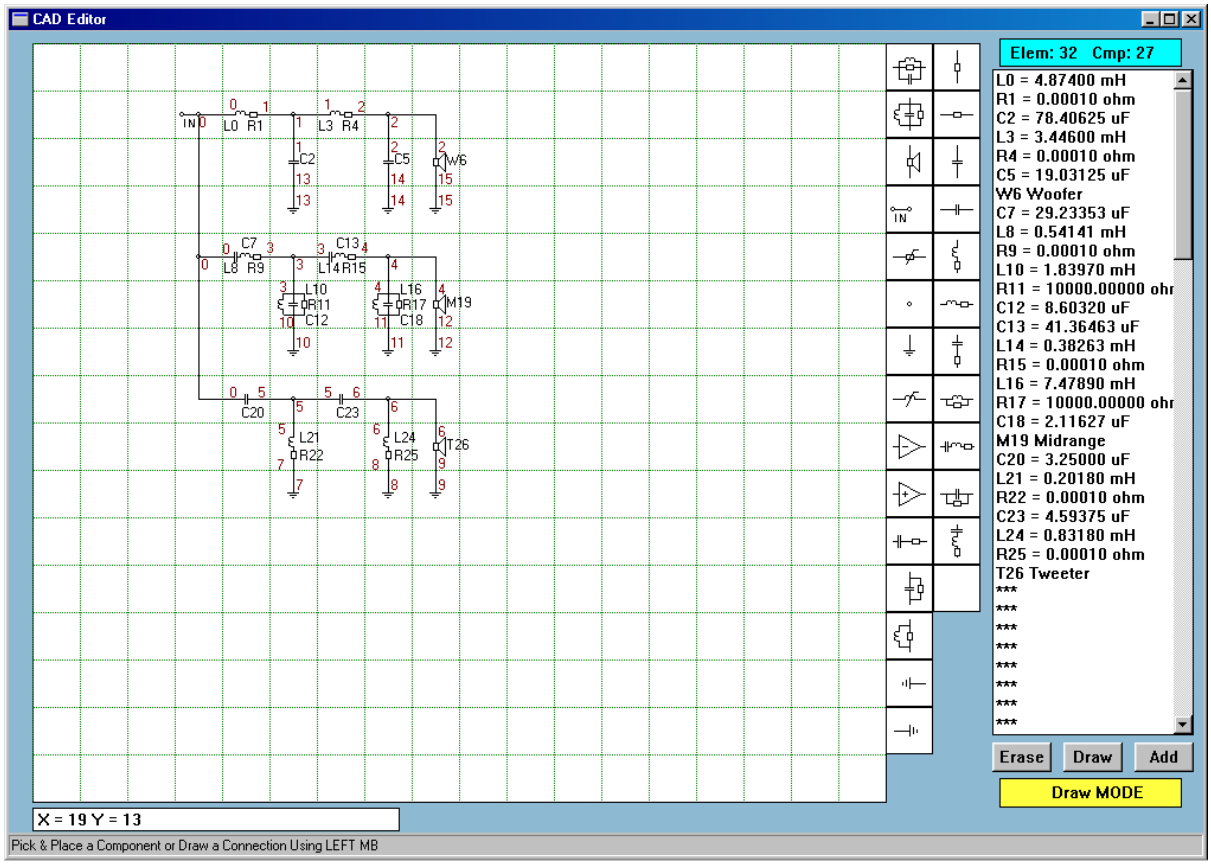


Figure 9.37. 3-Way system ready for Digital EQ function

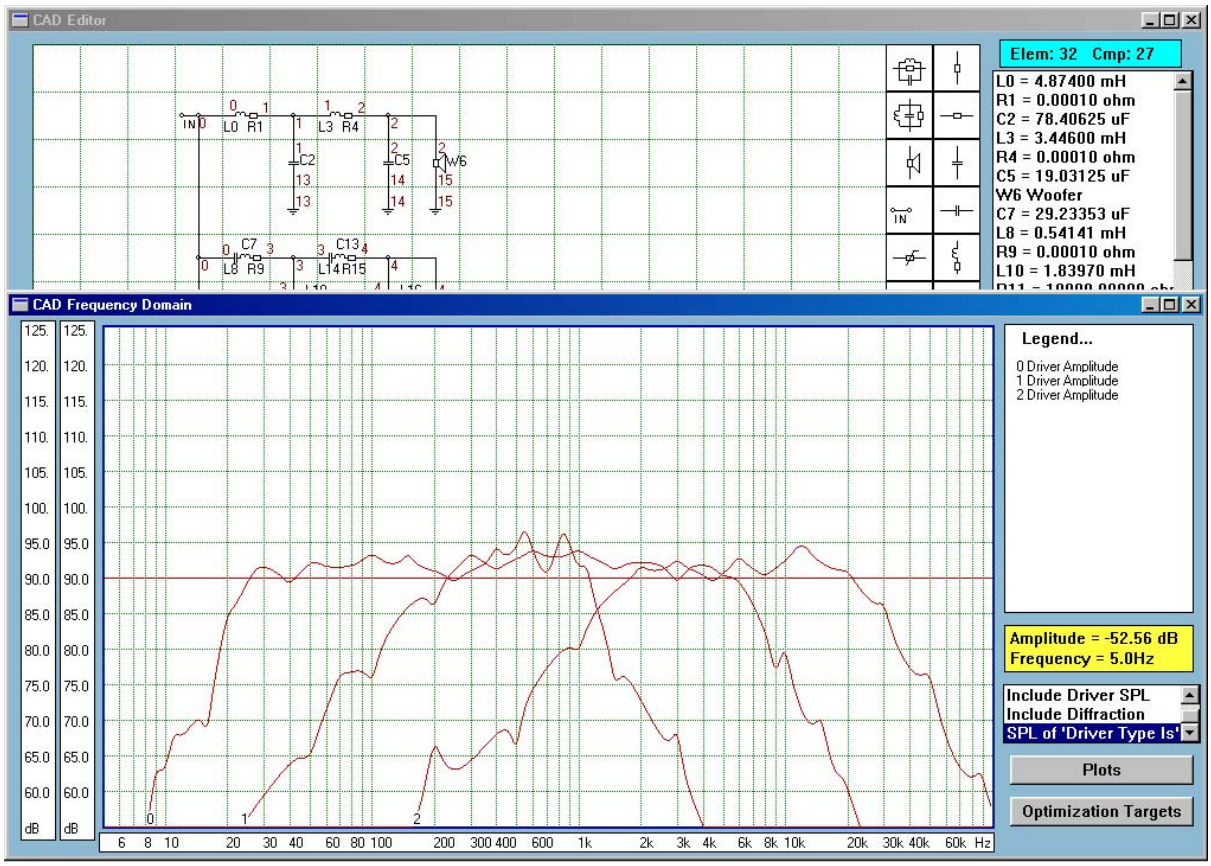


Figure 9.38. Woofer, midrange and tweeter drivers frequency responses.

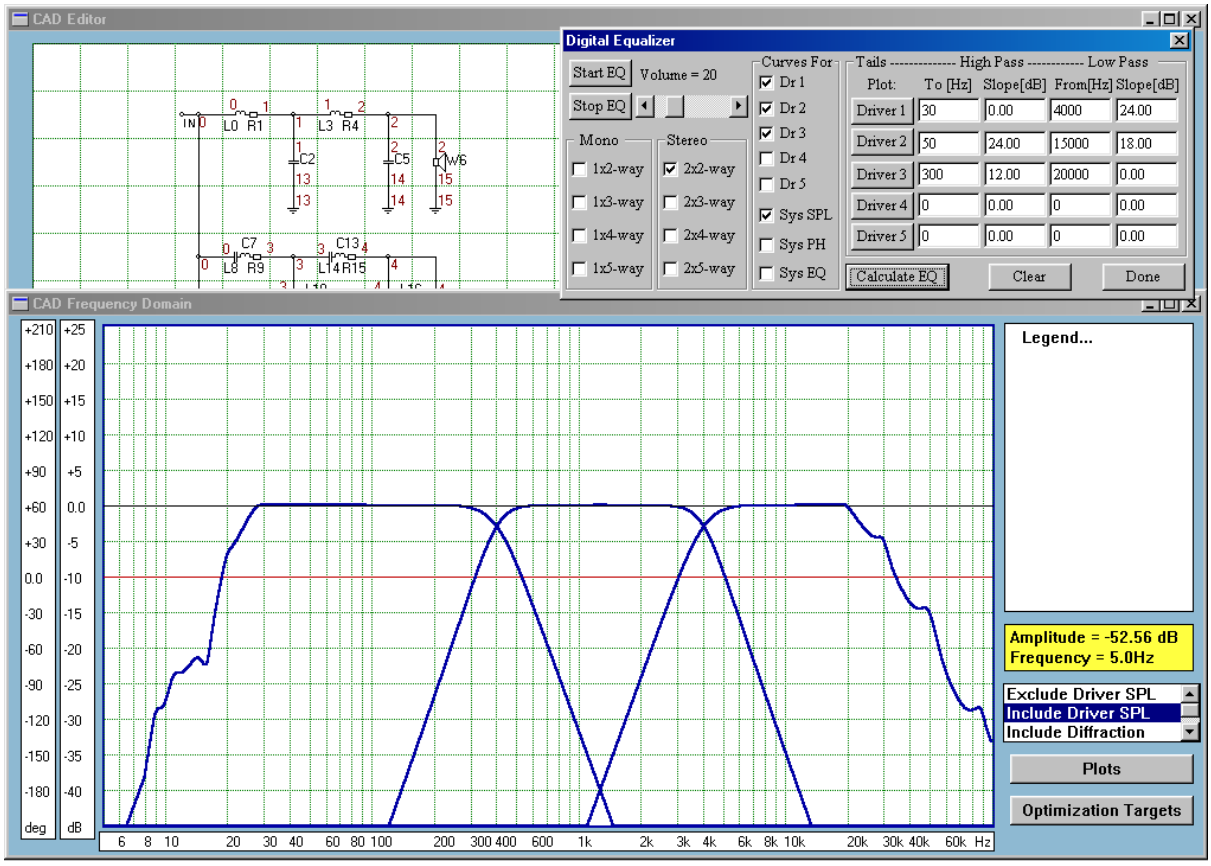


Figure 9.39. All 3 drivers were equalised – these are their system (driver+EQ) responses

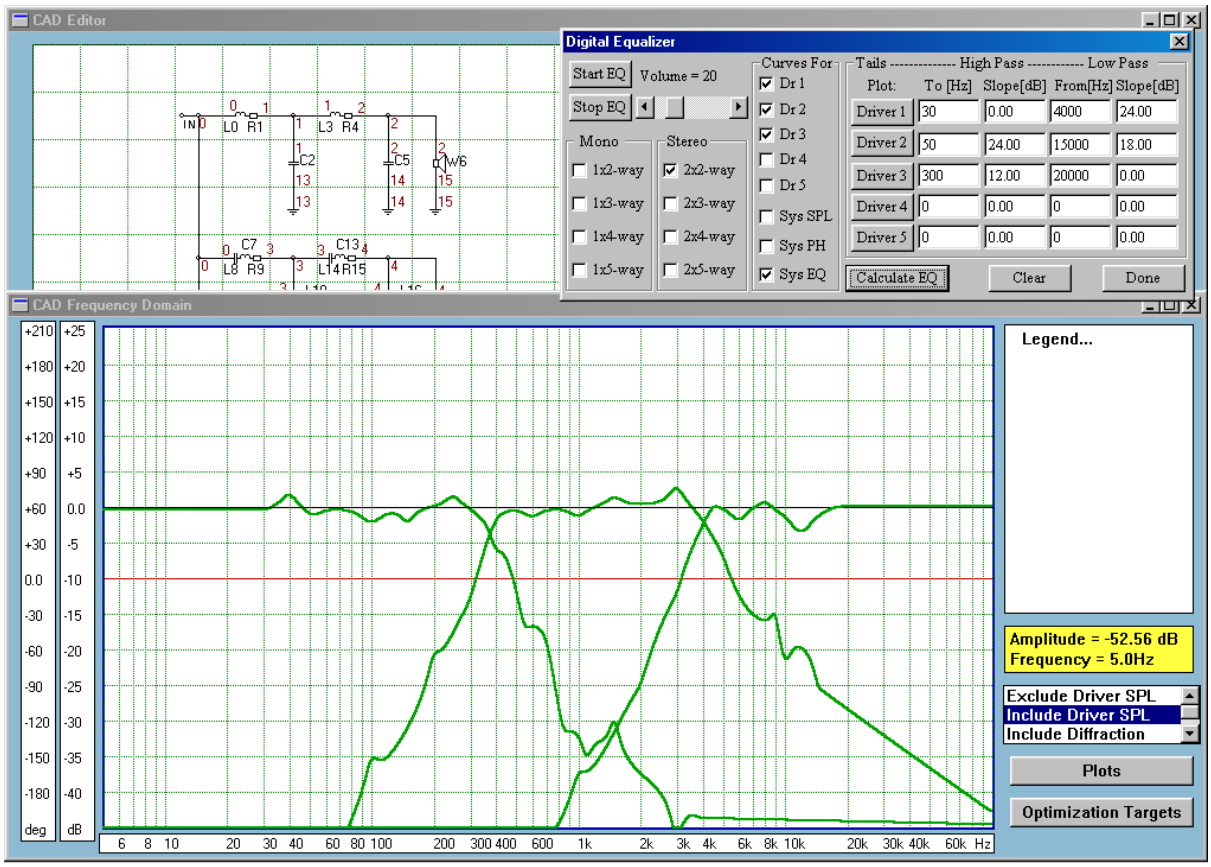


Figure 9.40. All three EQ curves used to obtain responses as on Figure 9.39

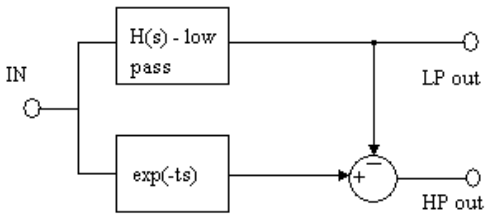
Time Delayed Filters

Time delayed filters are based on the requirement, that low-pass and high-pass channel responses must sum to a unity. $\mathbf{H_L(s) + H_H(s) = 1}$

This requirement alone did not unfortunately produced good polar responses and J. Vanderkooy came with a simple modification to the above formula, that showed much greater promise – a time delay element was introduced.

$$\mathbf{H_L(s) + H_H(s) = \exp(-\tau s) \quad OR \quad H_H(s) = \exp(-\tau s) - H_L(s)}$$

At the time when the modification was proposed, those filters could not be realized. There was no readily available time delay elements. Now, with the advent of DSP techniques, those limitations have been lifted and you can implement the whole family of time-delayed filters quite easily. Auditioning of the filters can be facilitated via Digital Filter option.



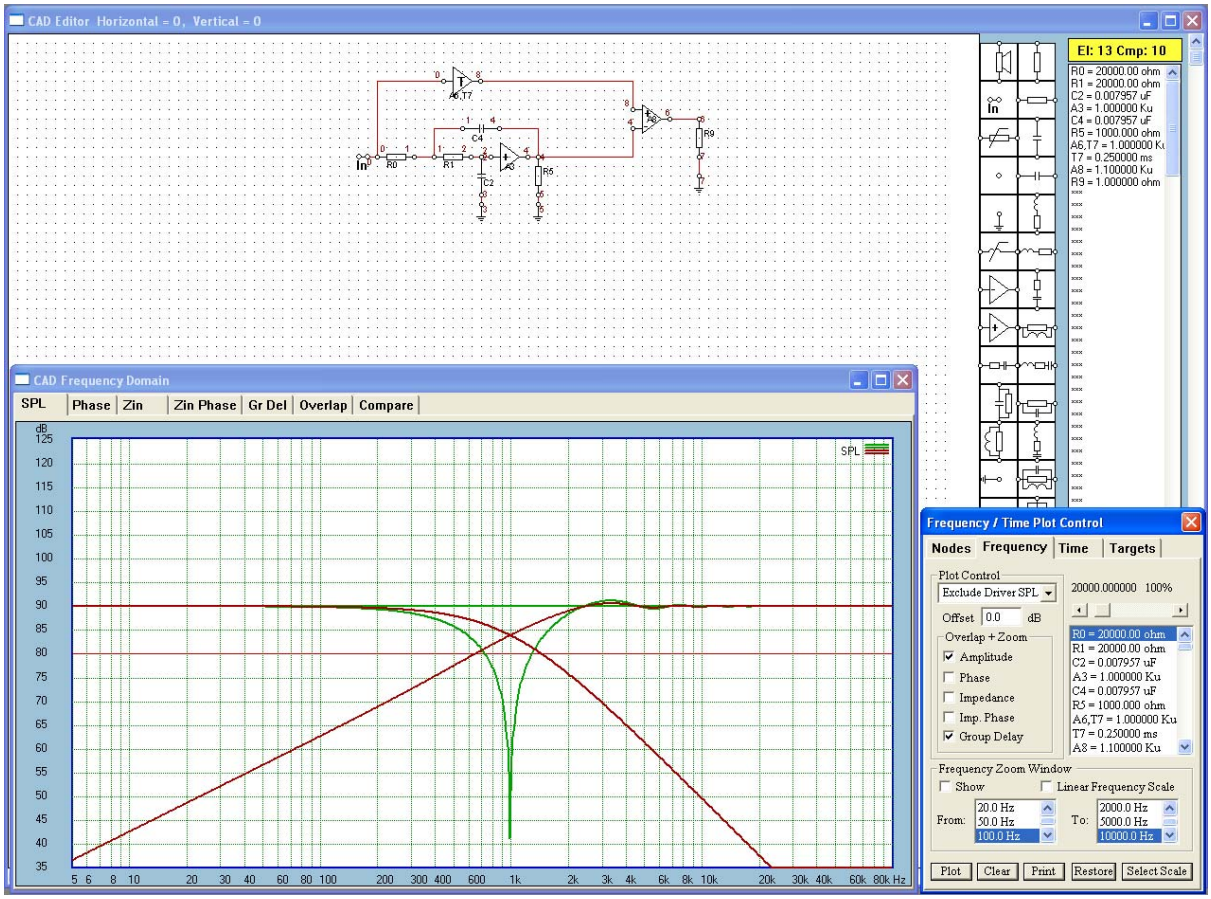


Figure 9.41. 2-way crossover built using time-delayed filters. LP=Node4, HP=Node6. (Note inverted response)

Low-Pass Linkwitz-Riley filter with cut-off frequency of 1000Hz. Delay element introduces 0.250msec delay and is connected to the non-inverting input of the differential amplifier with gain = 1.10. Summed response is "perfectly flat" line at 0.0dB level.

Please note the R9 load resistor for A8 – you must NOT leave the output unconnected.

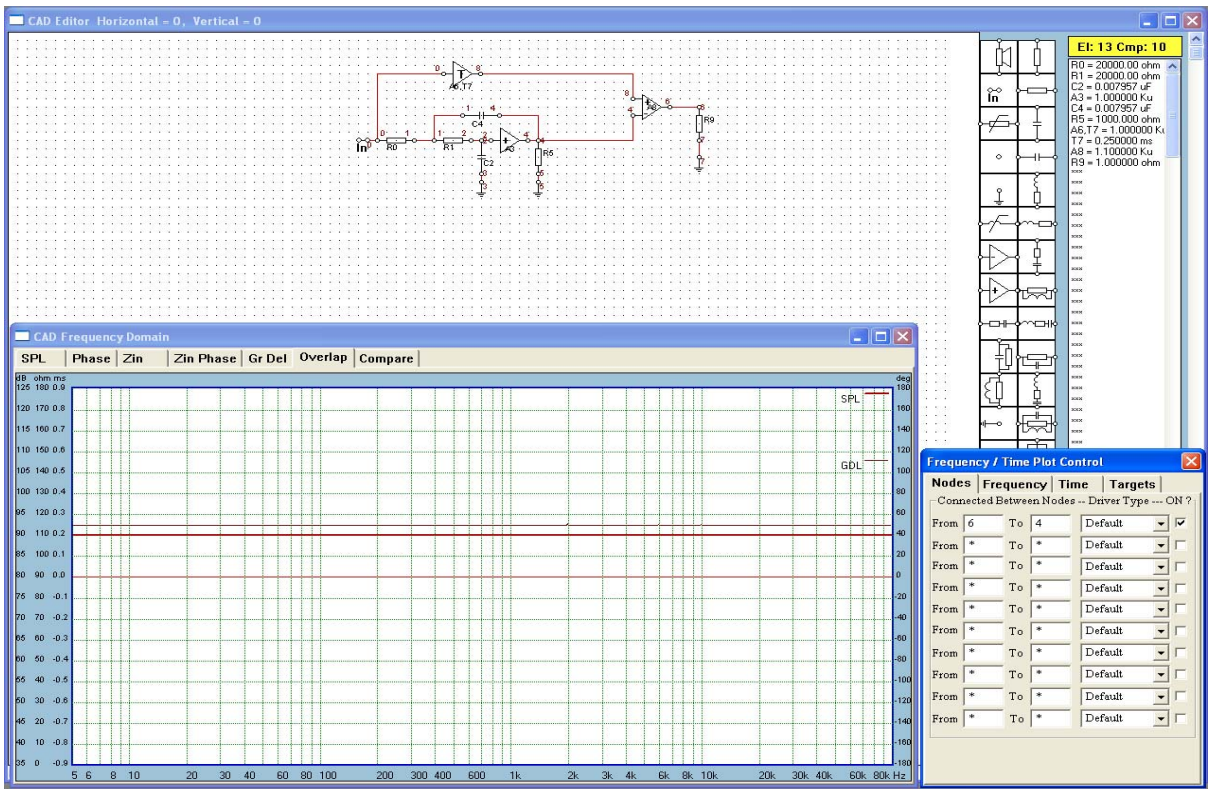


Figure 9.42 Group delay of the time-delayed crossover - CONSTANT

The delay set in the delay element is recommended to equal the low-pass section group delay at 0Hz, that is 0.250ms. Group delay is implemented as the first frequency derivative G(ϖ) of the phase response P(ω).

$$G(\varpi) = \frac{\partial P(\varpi)}{d\varpi}$$

Since the group delay is constant, this indicates, that the phase response is linear.

Driver Delay Offset using Digital Filter

When implementing Time Delayed filters or crossovers, you may still desire to provide additional delay for some drivers to compensate for the difference on acoustic centre offsets. This delay can be conveniently implemented using the fields provided in the "Digital Filter" dialogue box. The delay must be entered in **milliseconds** as positive numbers.

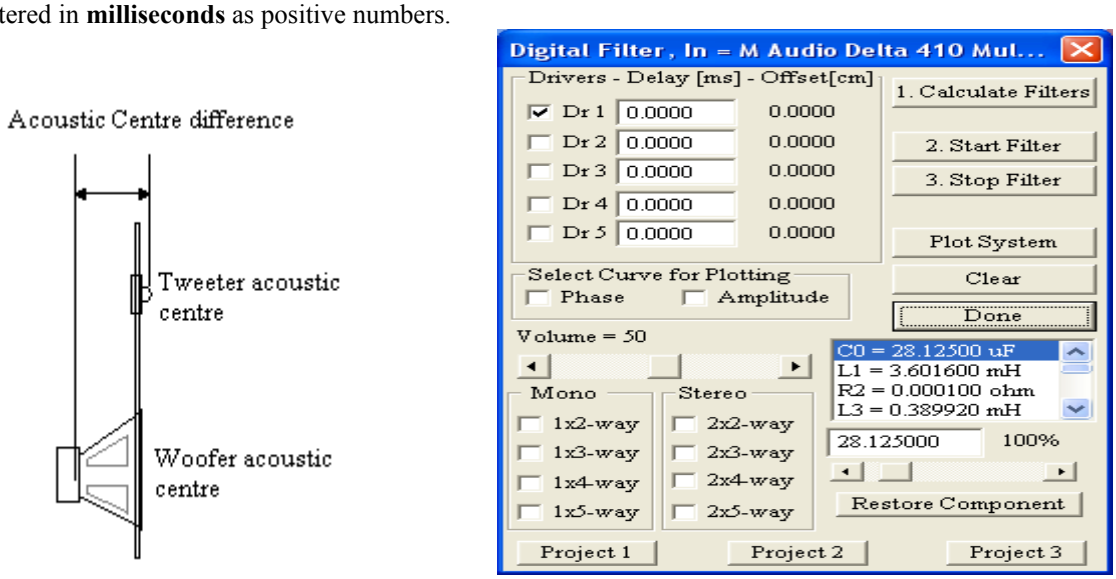


Figure 9.43. Illustration of the acoustic centres difference.

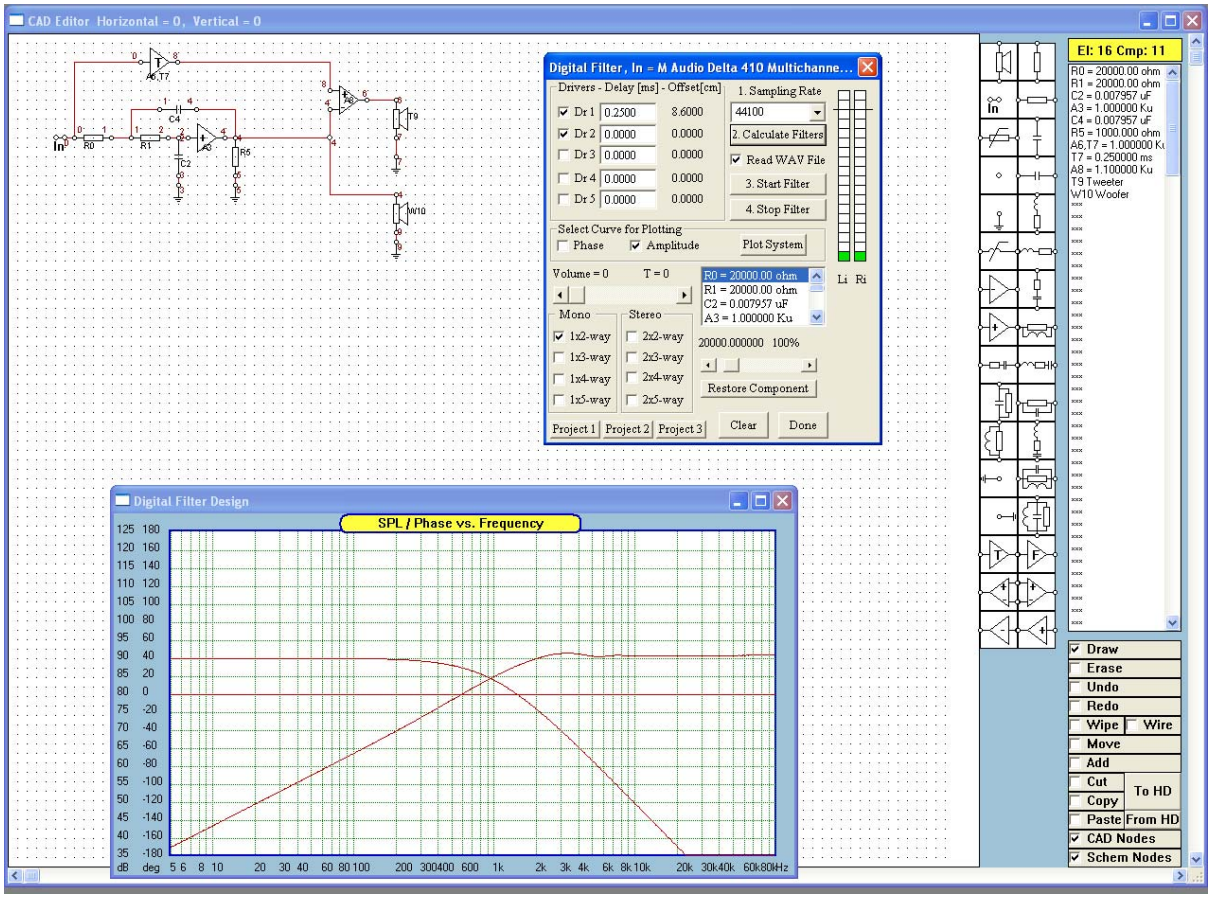


Figure 9.44. Digital Filter with additional delay of 0.25ms for driver 1.

Super-Component

The concept of **Super-Component** introduces a very powerful function into the program. The Super-Component (SC) can be described as follows:

1. Electrical implementation is identical to Voltage-Controlled-Voltage-Source (VCVS), with editable gain, input resistance and output resistance.

2. Frequency response of the SC can be any of the built-in filters or networks. **In addition, the SC can assume user edited frequency response loaded from previously saved target file.**

3. The SC can also introduce a pure time delay into the circuit, and all parameters from (1) are still available.

Practical implementation of the SC would be limited to built-in filters or networks. You would simply substitute the SC with the corresponding electrical network. The network could be actually quite complex, as the SC allows you to use any built-in filter/network configuration. Using the SC as a time delay element or assigning user edited frequency response to it, can be implemented in practice via Digital Filter or Digital Equalizer. Selecting and assigning various frequency responses from the "Edit Super-Component Data" dialogue box is very simple, as the dialogue box works the same as previously described filter/crossover selection dialogue boxes.

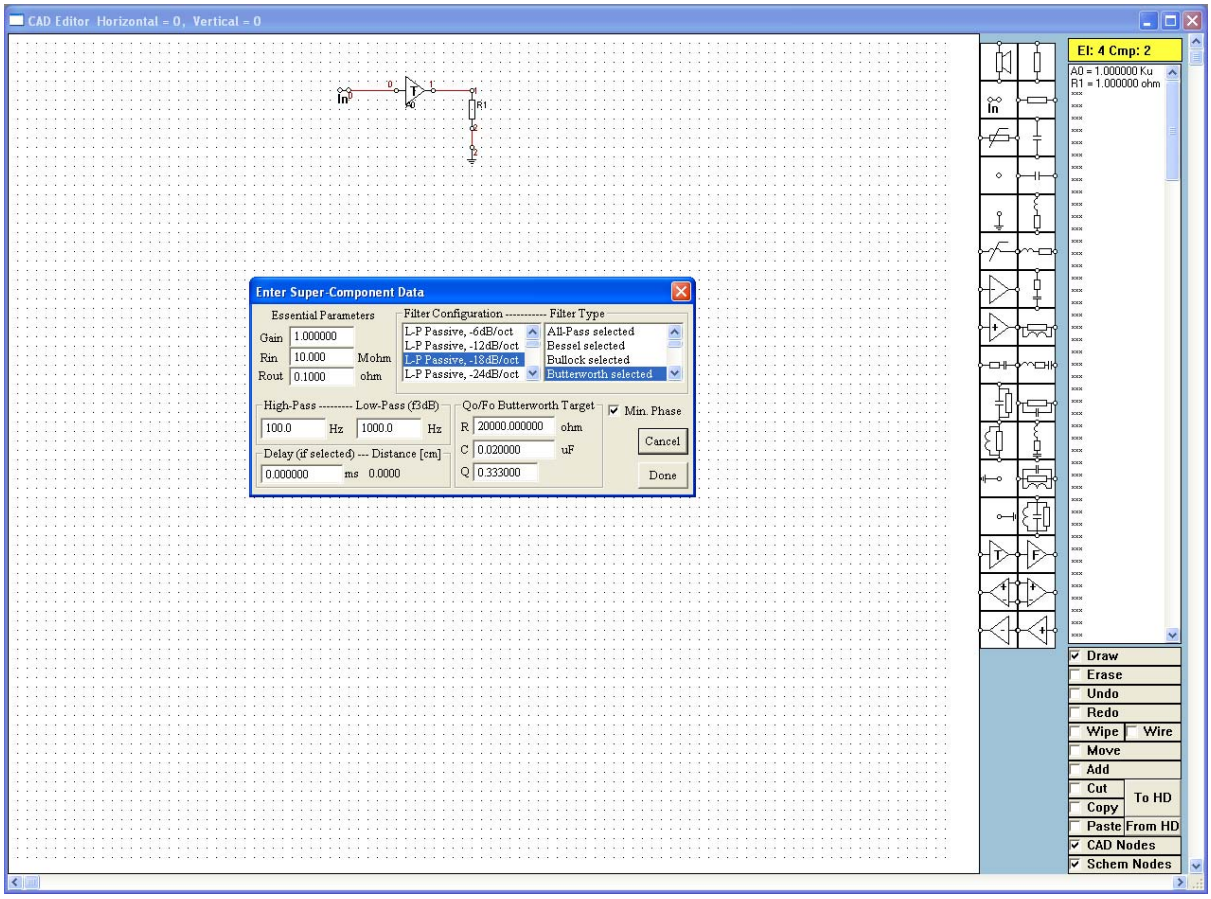


Fig 9.45. Super-Component Editor dialogue box

1. **Essential Parameters Group**
   a. **Gain** – Use this field to enter Super-Component's gain.
   b. **Rin** – Enter input resistance of SC.
   c. **Rout** – Enter output resistance of SC.

2. **Filter Configuration** – Select order of the filter from this list box.

3. **Filter Type** – Select filter's type from this list box

4. **High-Pass** – Enter high-pass 3dB cut-off frequency.

5. **Low-pass** – Enter low-pass 3dB cut-off frequency.

6. **Delay** – Enter required time delay. You will see the equivalent acoustical distance calculated.

7. **Qo/Fo Butterworth Target** – Enter additional filter parameters for certain Butterworth filters.

8. **Cancel / Done** – Use as described on the buttons.

9. **Min. Phase** – See below.

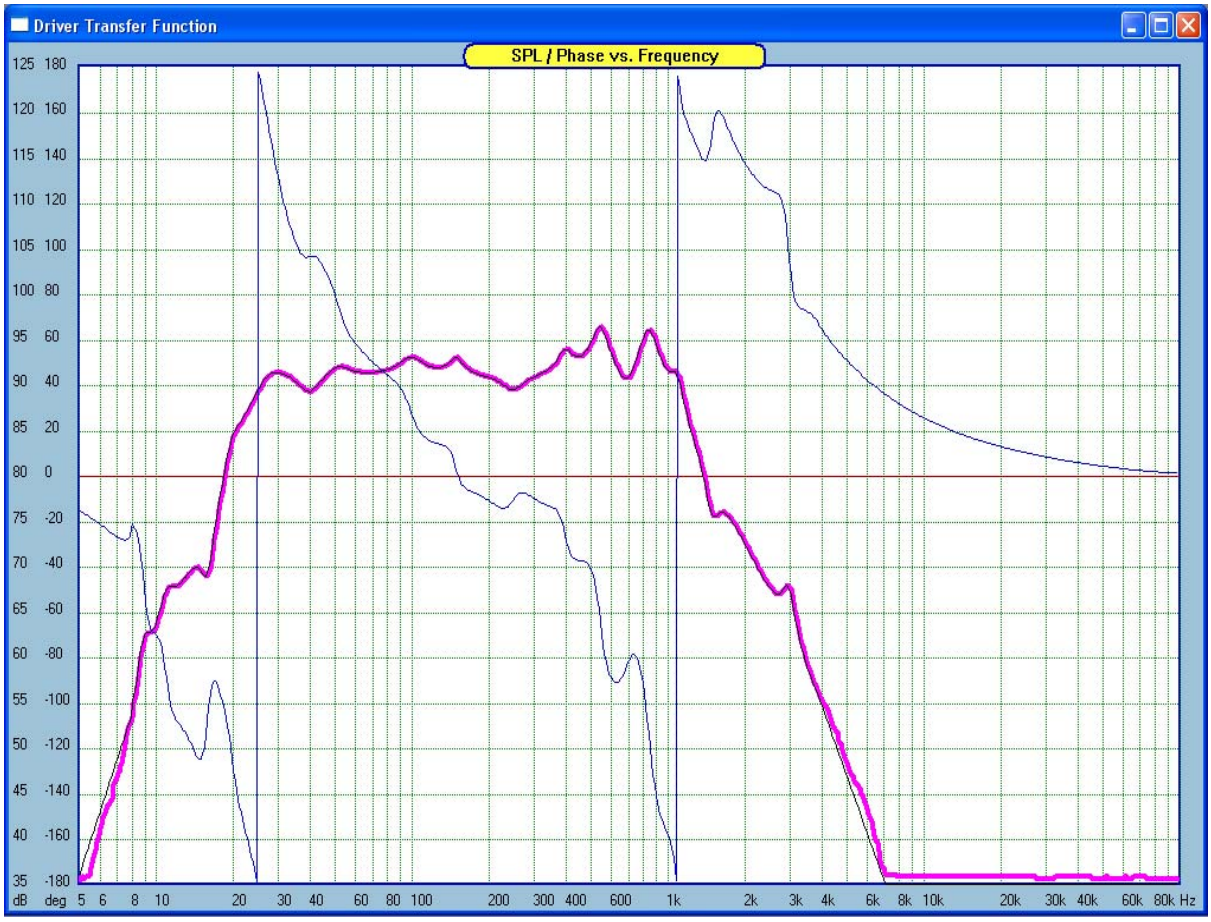


Figure 9.46. **Step 1** – Editing frequency response.
Step 2 – Running HBT on the curve
Step 3 - Saving the results of HBT as a target file.

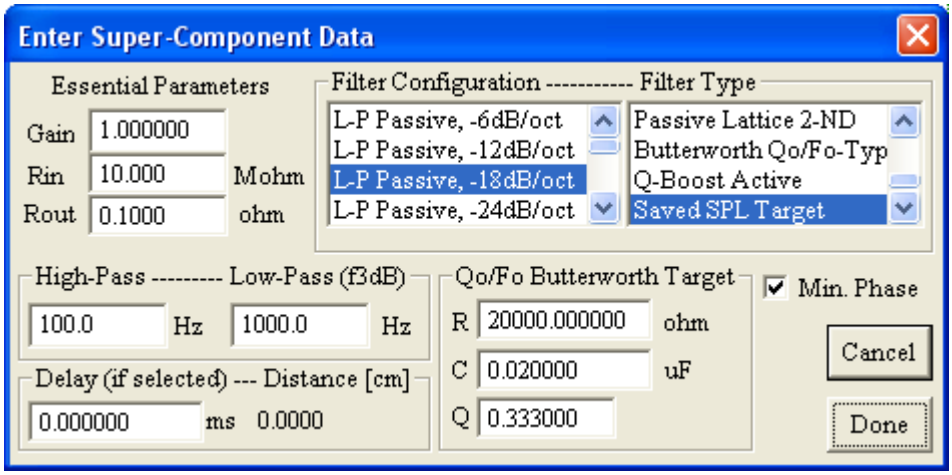


Figure 9.47. **Step 4** - Selecting user edited frequency response into Super-Element

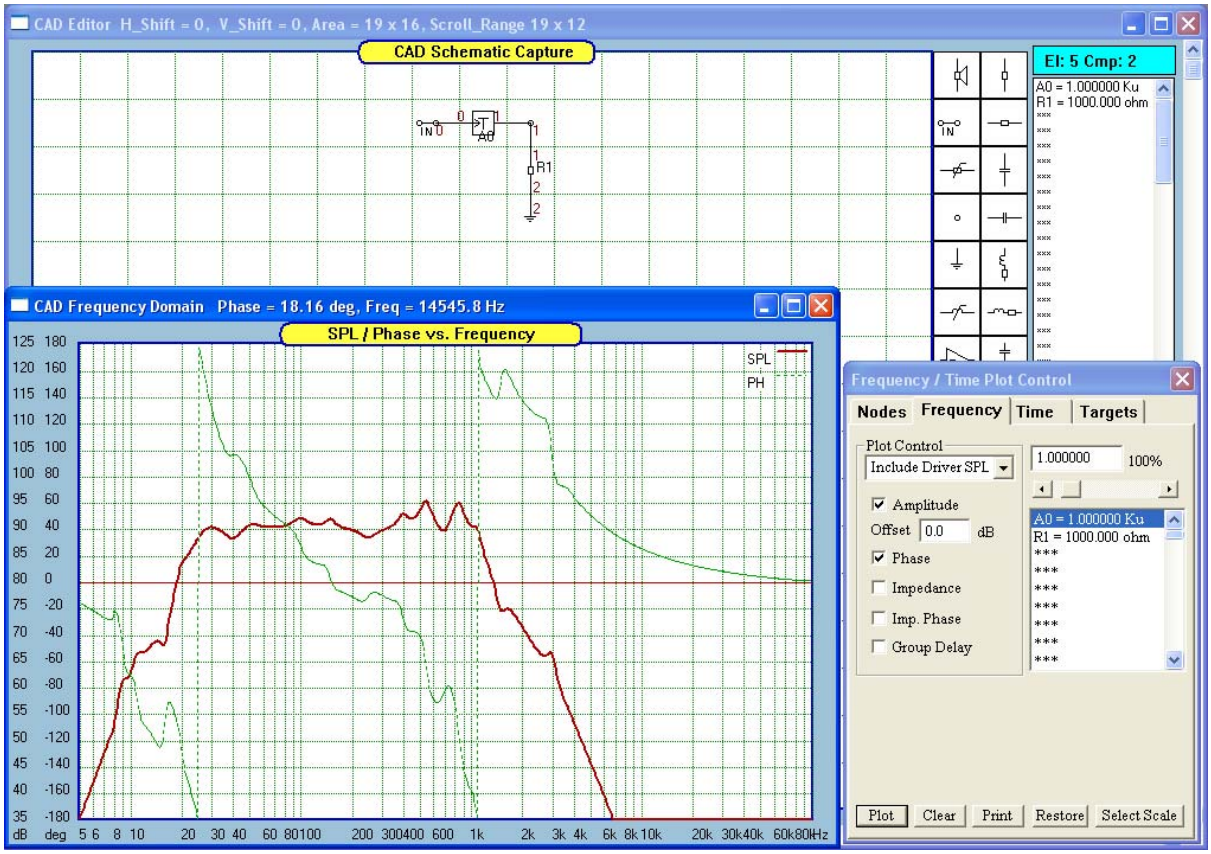


Figure 9.48. Frequency response of the Super-Component is the same as edited on Figure 9.46.

Mega-Components

The frequency response plotted on the "Frequency Domain" CAD plotting screen can be saved into the target file as well – you can only do this when the plotting screen is **opened and active**. This immediately opens another opportunity for the application of the Super-Component. The idea is explained using Figures 9.49 and 9.50.

Figure 9.49 shows an example of 5 Super-Components with assigned very different frequency responses: 1 low-pass filter and 4 band-pass filters. Output impedance of each Super-Component is set to 1ohm and outputs are connected in-parallel, so there will be some mutual loading effect reducing the gain (by 4times) of the whole system in this particular configuration. This is never a problem, as the mutual loading can be compensated by increasing the gain of each Super-Component anyway.

After the frequency response was plotted, it was then saved into the target file. In the next step, the screen was cleared and a simple Super-Component network was created as shown on Figure 9.50. This time, the Super-Component was assigned the frequency response from the previously saved file and gain set to 10x (or 20dB). Now, the single Super-Component exhibits the combined frequency response of the previous 5 Super-Components. This turns it effectively into a **Mega-Component**.

It is easy to observe that the process describe above can be carried out recursively infinite number of times. The resulting frequency response, created this way, can be extremely complex.

In addition, the **Super-Components and Mega-Components can save you drawing space** significantly. As you can see, a circuit, that can be as big as the whole screen can now be substituted in many cases by a single component. As the **Super-Components and Mega-Components** are implemented as VCVS, it makes them particularly suitable for active circuit implementation. However, since the input and output impedances are editable, you can easily use these elements in passive filters and crossovers. Please note, that depending on their complexity, it may not be possible to implement the **Super-Components and Mega-Components** in any other way, except for Digital Filter and Digital Equalizer. This is where they are mostly intended to be used.

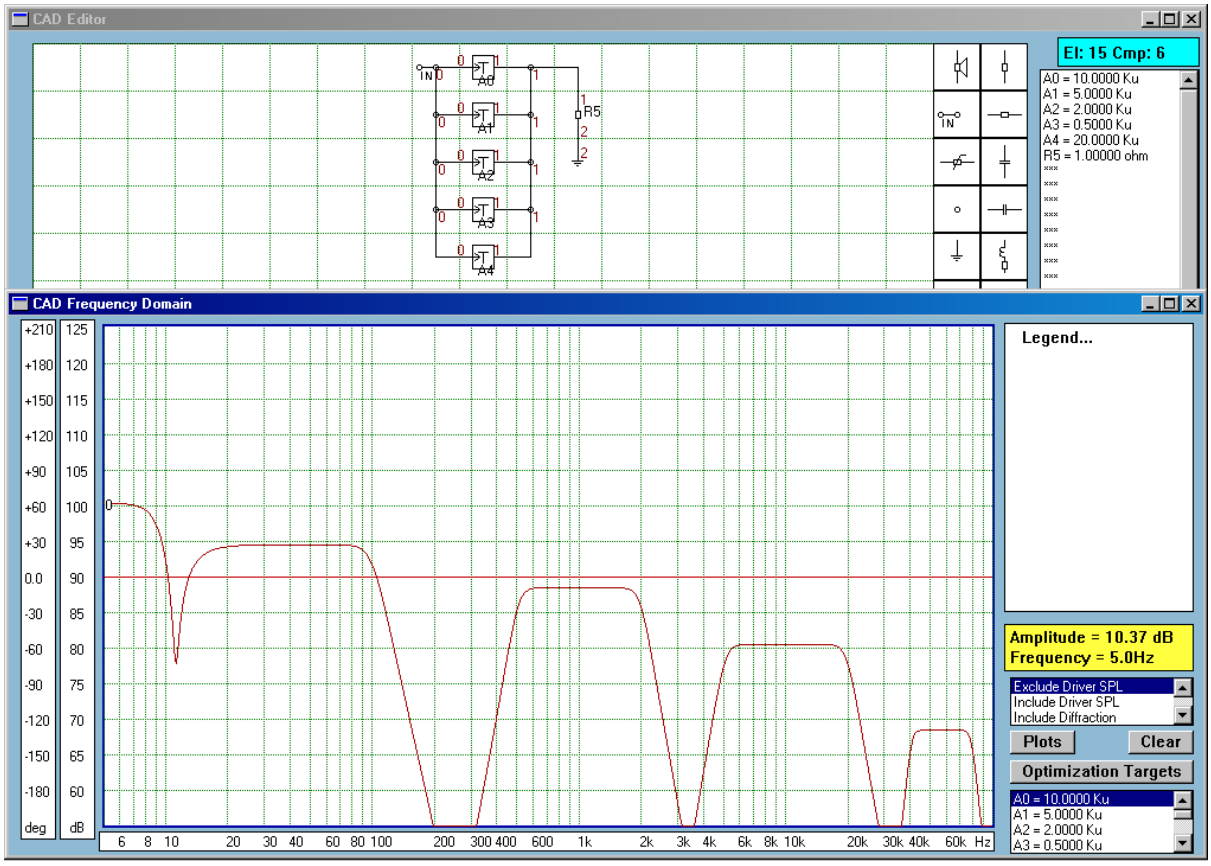


Figure 9.49. SPL of 5 **Super-Components** connected together.

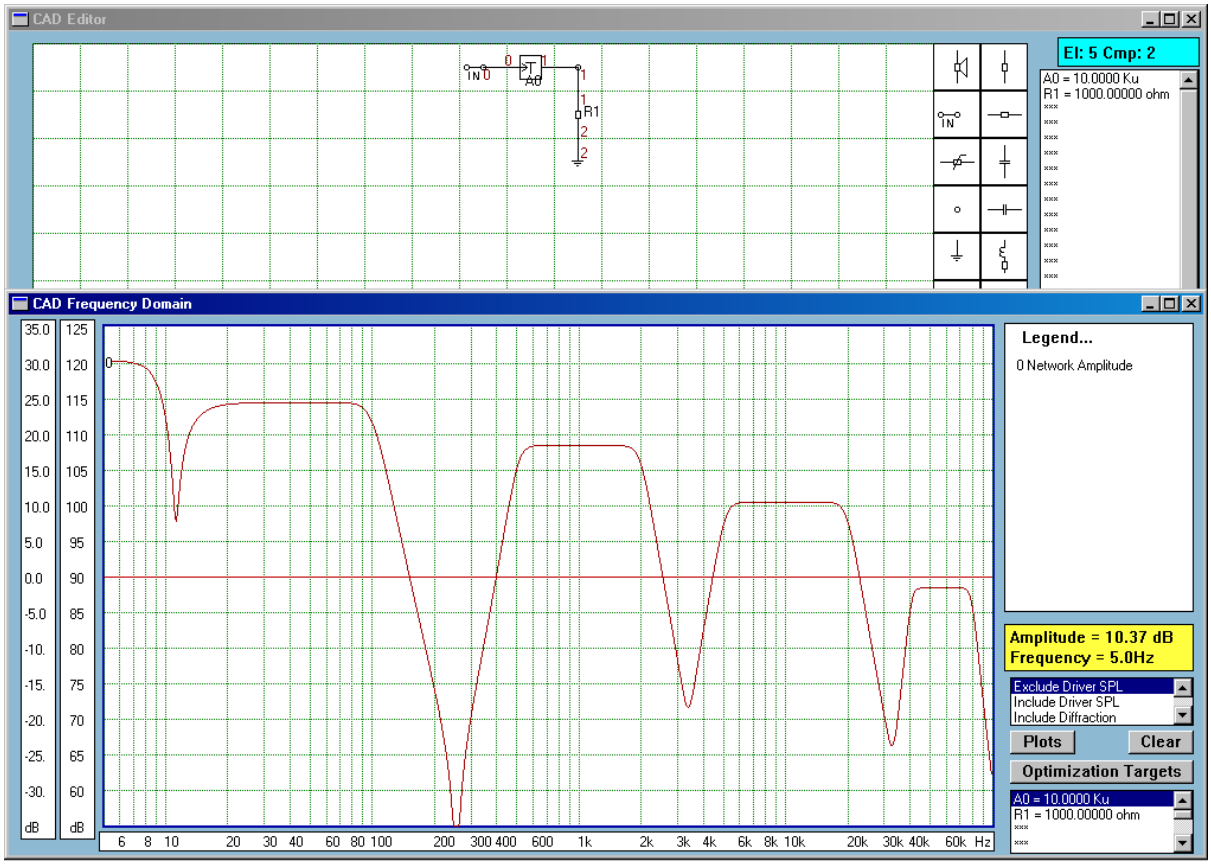


Figure 9.50. Frequency response from Figure 9.49 was saved and assigned to single **Mega-Component**.

Linear Phase Filters

Linear Phase Filters can be created using Super-Component's feature called **"Min.Phase".** When this Option is checked, the phase response of the circuits created via Super-Component is the normal, "minimum-phase" response. However, un-checking the box causes the phase to become linear, irrespectively of the magnitude response. Interestingly, the magnitude-phase responses no longer hold Hilbert-Bode relationship. Phase is controlled via delay and can be set to zero or any constant, positive value by entering data into the **"Delay"** field.

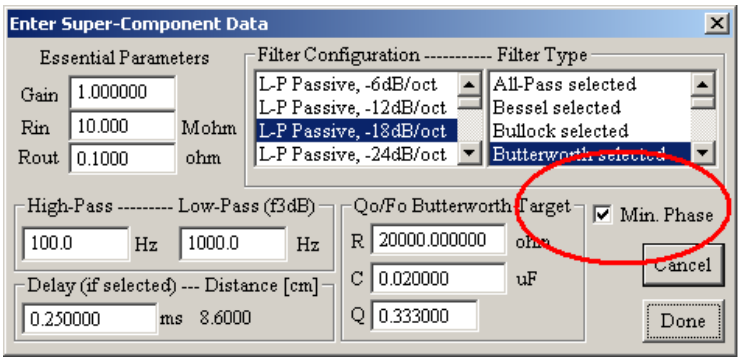


Figure 9.51. Enabling Minimum-Phase response.

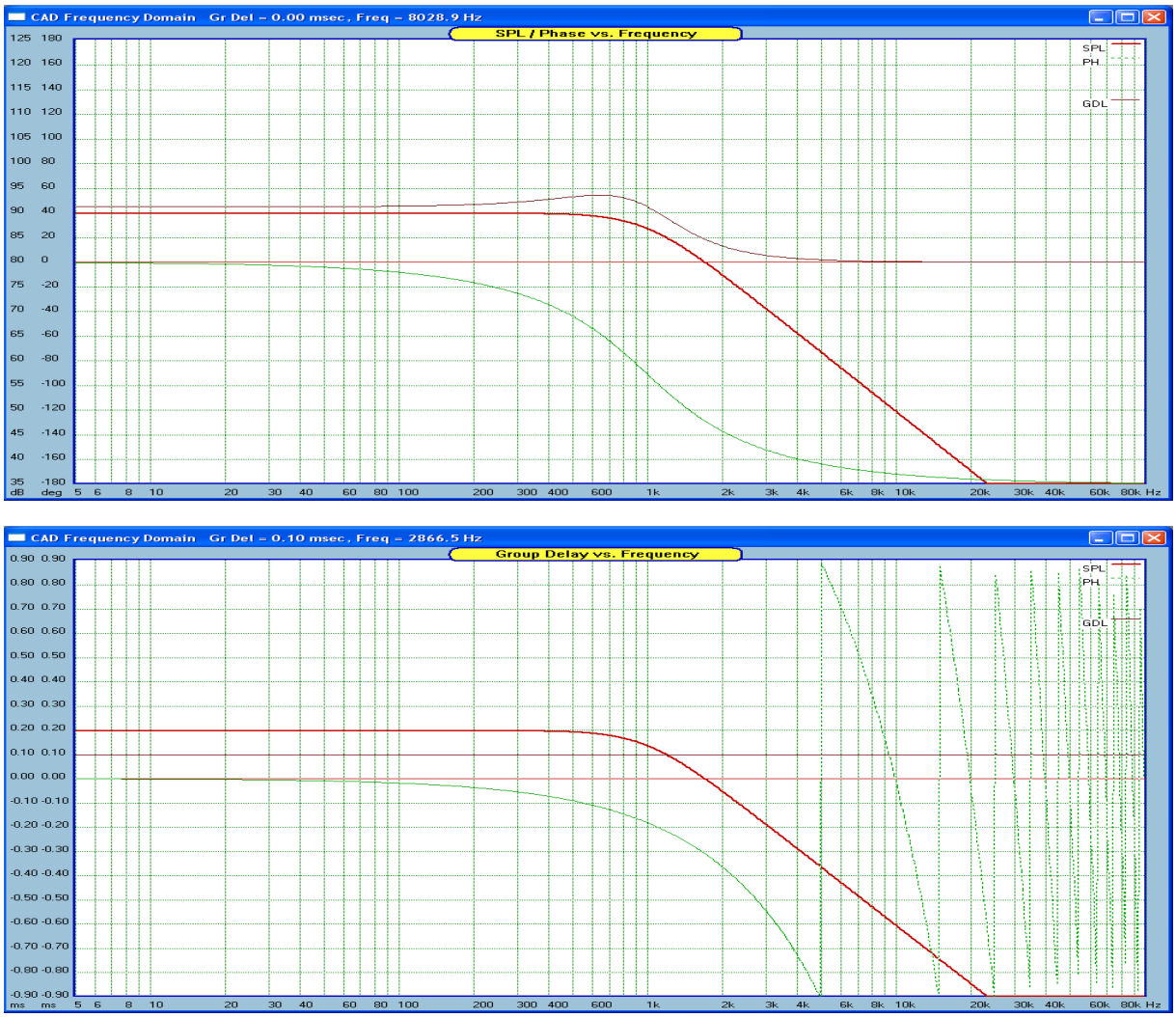


Fig 9.52. Here is an example of low-pass filter with minimum-phase (upper) and linear phase (lower).

Please note the typical group delay in the upper plot and constant group delay (0.10 ms) on the lower plot.

Modular Crossover

The concept of Modular Crossover (MC) design and optimization is intended to compliment the traditional "discrete component" crossover approach. It is anticipated, that modular crossover will find its use in emulating and selecting the parameters and optimization of DSP Crossover hardware already available on the market. A good example of such device is Behringer DXC2496 device. The DCX2496 is capable of emulating several popular filter configurations such as low-pass, high-pass, shelving filters and variable-Q type filters. The DCX2496 is a very fast, real time filtering device, but lacks SPL optimization functions or SPL importing functions for proper crossover design. The MC approach would allow you to emulate the behaviour of DCX4296 and at the same time, design and optimize the crossover, including desirable driver's AC offsets (delays).

One can hope and anticipate, that future DSP hardware filters will allow for significant increase in emulation capabilities in terms of greater number of Impulse Response coefficients to be processed. This would eventually facilitate implementation of the Digital Equalizer functionality already provided in SoundEasy.

Using the MC approach, crossover filter's **Transfer Function is expressed in terms of cut-off frequency, slope roll-off (or Q-factor) and design type**. This is in contrast to a list of component values for traditional, discrete component-type crossovers. For instance, using MC methodology, we would be talking about a second-order, Butterworth low-pass filter, with a cut-off frequency of 1000Hz.

All MC filters are implemented as building-blocks. Current software implementation offers you a number of filter types and configurations to choose from. You can find the MC icon tagged as **"F" (for Frequency-dependant component)** on the CAD screen component pickup section, together with the other components. When you pick-and-place the F-component on the CAD screen, you can double-click the left-mouse button to open a control dialogue box. This box allows you to select the desired parameters for this particular F-component. The box is shown below. All rules and recommendations for drawing schematic apply to the F-component as well, so they will not be repeated here. Operation of the dialogue box is really self-explanatory.

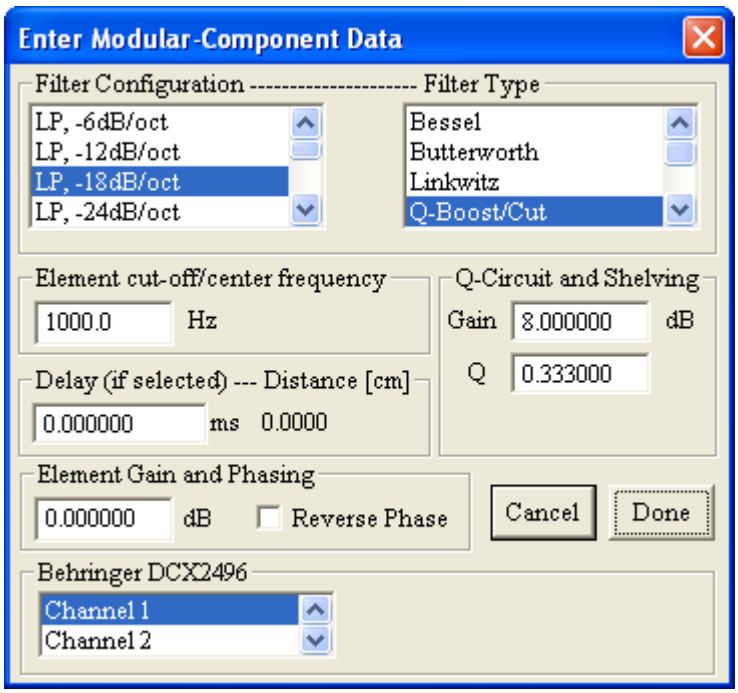


Figure 9.53. F-component control box

You will notice, that each F-component is displayed on the CAD screen with a short list of it's most characteristic parameters. The list occupies some space on the drawing area, therefore it is advisable to space the F-components more sparsely on the screen, to enable all information to be displayed without overlapping.

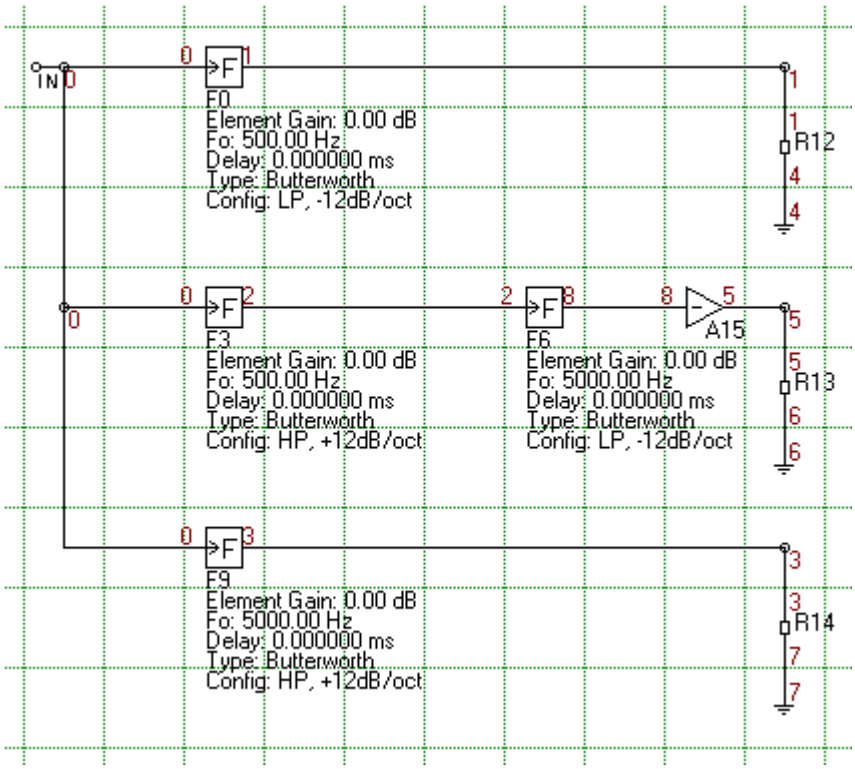


Fig 9.54. 3-way crossover with 500/5000Hz cut-offs. Note A15 to invert midrange channel.

Current release of the program implements the following F-components:

1. +/-6dB/oct Low-pass shelving filter,
2. +/-6dB/oct High-pass shelving filter,
3. +/-12dB/oct Low-pass shelving filter,
4. +/-12dB/oct High-pass shelving filter,
5. +6, +12, +18, +24, +48dB/oct Butterworth High-pass filters,
6. -6, -12, -18, -24, -48dB/oct Butterworth Low-pass filters,
7. +6, +12, +18, +24, +48 Bessel High-pass filters,
8. -6, -12, -18, -24, -48dB/oct Bessel Low-pass filters,
9. +12,+24 Linkwitz High-pass filters,
10. -12, -24dB/oct Linkwitz Low-pass filters,
11. Q-Peaking/Notch filter (or Parametric)
12. Phase inversion. **The A15 is really redundant, as you can invert phase by checking the "Reverse Phase" box on Fig 9.53.**

All F-components allow for individual gain adjustment. The gain is expressed in dB's, and you can enter negative gain as well for attenuation purposes. **In addition, all F-components can have a time delay added to their Transfer Function.**

The F-components can be mixed with all standard passive (RLCs) and active (OPAMPs) components as you wish. The F-component has a very high input impedance and very low output impedance, so effectively it behaves in the circuit as a VCVS (Voltage Controlled Voltage Source, or an OPAMP if you prefer) with **frequency dependant** Transfer Function – hence it's name the "F-component".

Frequently, the MC approach may speed-up your design. For instance, imagine, that you have your 3-way crossover designed already with discrete components on the CAD screen, but you want to change the cut-off frequency of a low-pass filter. Depending on the complexity of the filter, a number of components would have to be re-calculated and changed. To accomplish this, you could load another low-pass filter onto the CAD screen, with components calculated for the new frequency and re-link the output to the corresponding driver. Then check the system SPL for the desired outcome. If OK, you could erase the old filter (CUT / PASTE function) and use the new one. You could do this again and again until the circuit worked as desired. However, the MC approach would be a better option to start with.

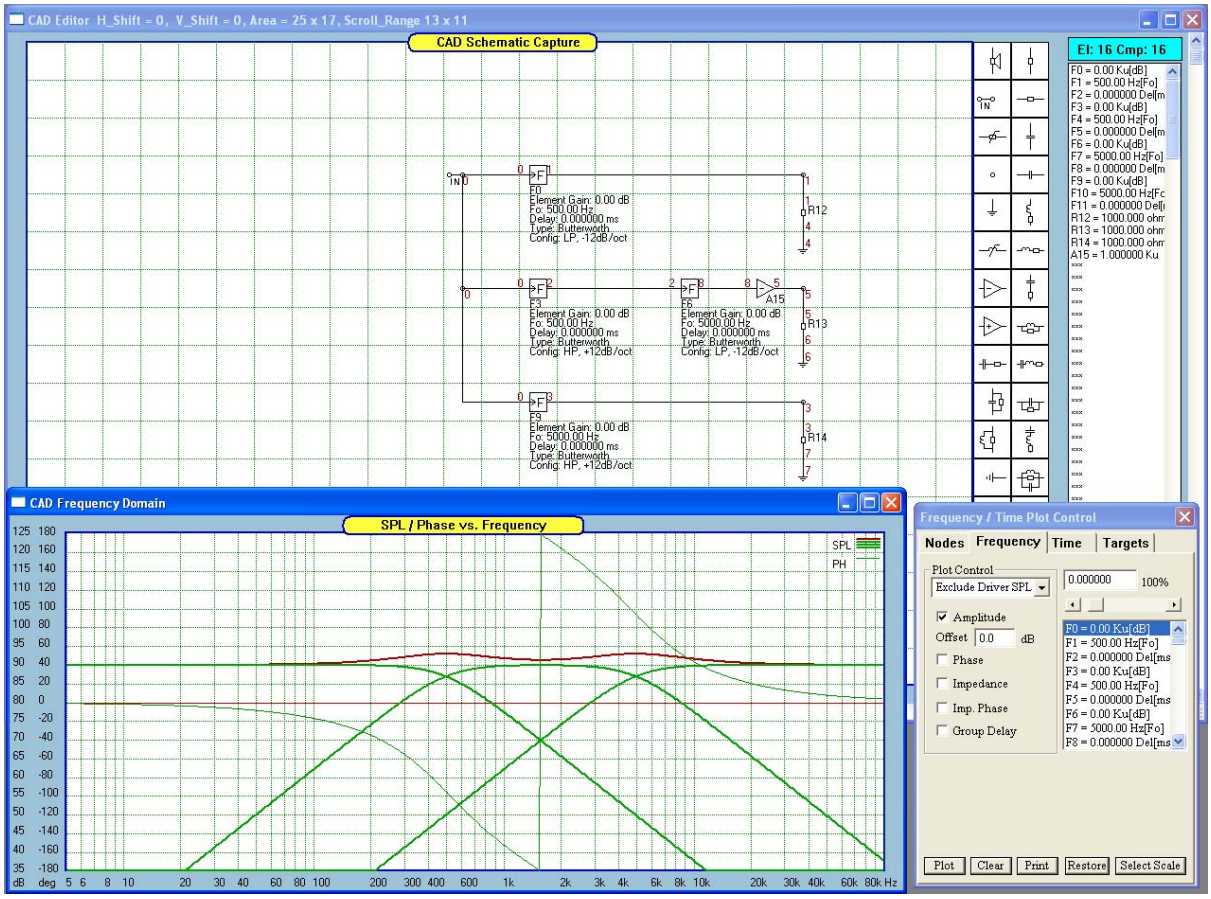


Fig 9.55. An example of a 3-way system crossover with SLP and phase plots.

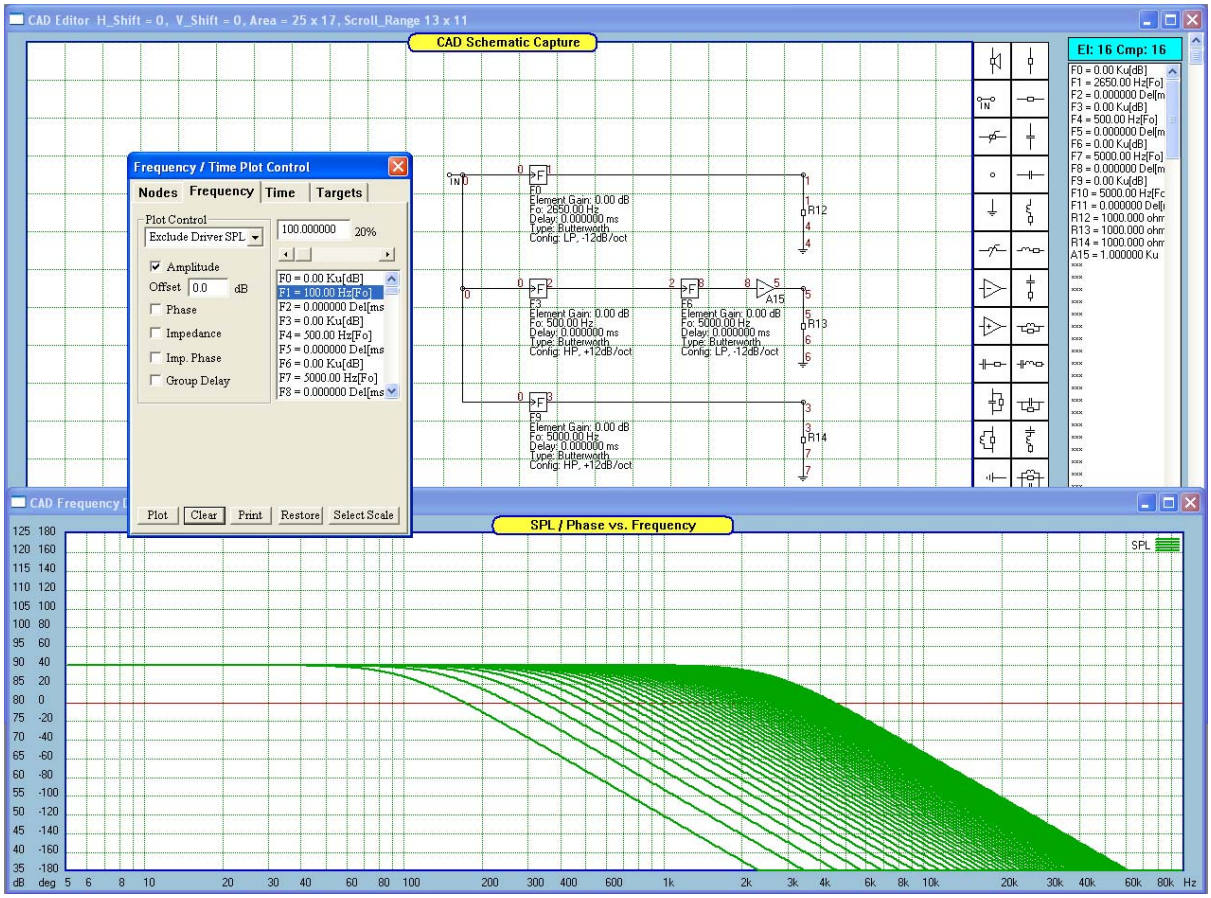


Fig 9.56. Adjusting cut-off frequency with a slider.

Here is how it would work:

1. Built your 3-way crossover using F-components.
2. You can now **adjust cut-off frequencies, gains and Q-factors with the slider**, after highlighting this parameter in the component list box – see Fig 9.56
3. You can also perform initial optimization of the MC crossover.
4. Once you are happy with the result, you can easily built your final crossover with discrete components by selecting corresponding filter parameters when populating the schematic with built-in filters.

You MC can also be optimized. The F-component parameters can optimized for gain, cut-off frequency and Q-factor.Finally, the F-component can be used with Digital Filter and Digital Equalizer functions.

Using The DCX2496 Digital Crossover.

The MC concept within the program has been developed to enable an alternative transition path from the traditional discrete/passive components of the crossover to digital-type crossovers. You know already about SoundEasy Digital Filter and the powerful Digital Equalizer functions, utilizing your PC's soundcard. The DCX2496 option, will enable you to switch off your PC and use only the DCX as the crossover. The digital crossover technology is currently a lot more expensive than simple passive crossover. However, it offers some features, that the traditional passive crossovers are lacking. One of those features is ability to add a broadband delay to any channel.

Currently available DCX2496 device can be used with MC concept, but it is unable to function as the SoundEasy Digital Filter. An external digital crossover would have to be able to accept the concept of an acoustic channel defined by it's impulse response, IR, rather than the more traditional, mathematically defined filter parameters for a DSP engine. The situation is even more complicated for the Digital Equalizer function, because the IR of the Digital Equalizer is defined by 64000 coefficients (in SE V9.00). The other problem is with Linear Phase filters, which also can not be emulated by the DCX2496.

Typical usage of the MC concept and components would be to create a crossover with EQ and shelving circuits as you see fit, then proceed to optimize the modules and parameters, and finally audition your design using either your PC's soundcard or the DCX2496. This process is similar to the design and audition of a standard, passive, discrete component crossover. To audition the crossover, you can simply use your multi-channel sound card, or, if you have the Behringer DCX2496 Digital Crossover, you can use it for the same purpose.

Almost all F-components (or settings) of Modular Crossover can be transferred to Behringer Digital Crossover box – DCX2496. Please note, that MC has more options than currently available DCX2496 (with software version 1.14). Therefore some of the Bessel filter settings can not be transferred to the DCX2496 crossover. Additionally, the DCX2496 uses fixed frequency grid of 320 points between 20Hz and 20kHz. This may place some limitations on the accuracy of the cut-off frequency settings of your selected crossover frequencies and EQ centre frequencies. Please also note, that the finest Delay resolution setting on DCX2496 is 2mm, and the gain can be changed within +/-15dB.

You can transfer the settings manually, by simply keying them into the DCX2496 or electronically, by selecting menu option "Sych DCX2496 to MC" under the "Crossover Tools" main menu.

Before you can remotely flash the DCX2496 please make sure, your DCX2496 is connected to COM1 serial port and powered ON. Failure to do so, may hang up your PC with the program waiting for a response from a disconnected or inactive DCX2496.

The DCX2496 has a rich setting array. However, please bear in mind, that this device is originally intended as a PA/stage audio processor, with functionality more suitable for PA work than home Hi-Fi system. We will be adapting a sub-set of DCX2496 functionality, keeping in mind limitations described above. Here is a typical configuration of the DCX2496, that you will be working with.

1. Channel Configuration is set to STEREO, 3WAY system: LMH LMH (CH1,2,3 CH4,5,6)
2. Channels are linked. Therefore, changes applied to channels 1,2,3 will automatically be transferred to channels 4,5,6.
3. Input section of DCX2496 has flat frequency response and gain set to 0dB.
4. All frequency response processing functions are implemented in the output section of DCX2496.
5. Each DCX2496 output section has 1 HP filter and 1 LP filter.
6. Each DCX2496 output section has up to 9 EQ circuits.
7. Cut-off frequencies of DCX2496 crossover filters are set to "FREE" mode (not linked) to allow you to set them individually.
8. During flashing process, all outputs are muted first, then new settings are applied, and finally all outputs are unmated.
9. Channels with no filters selected in, will have flat frequency response.

The freedom of CAD pick-and-place method allows you to built your MC schematic in any order/way you like it. On the other hand, the DCX2496 allows you to built each of your processing channels by selecting ON/OFF fixed channel components. Therefore, some mapping process must take place, in order to associate MC schematic components with the available DCX2496 audio processing components.

When you double-click on the MC component, the program pops-up a dialogue box with all available settings for this component. At the bottom of the dialogue box you will find a short list box designed for nominating the DCX2496 channel, that this component belongs to. There are only three settings needed to choose from:

1. Channel 1
2. Channel 2
3. Channel 3

Here, you must nominate the channel, that this component belongs to. This way, the program can correctly transfer the settings AND configuration of your crossover to the DCX2496. Currently, you can use DCX2496 to emulate STEREO 2-way and STEREO 3-way crossovers.

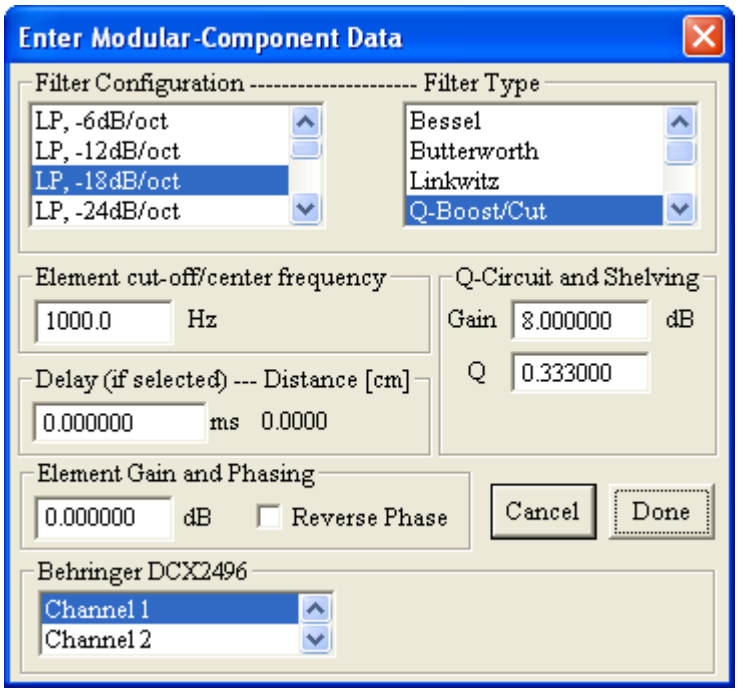


Figure 9.57. Behringer DCX2496 Channel assignment.

For instance, for a simple 3-way MC , you would allocate LP filter to Channel 1 and connect your woofer to it, BP filter to Channel 2 and HP filter to channel 3. Please remember, that DCX2496 does not implement Band Pass filters. You must construct such filter from a High-Pass + Low-Pass combination of the F-components.

Please make sure, that you are comfortable with the process of allocating the F-components to a specific DCX2496 channel. The process is very simple, but very important for setting the DCX2496 correctly. For instance, if you make an error and allocate a HP filter to a channel other than your tweeter, you will leave the tweeter totally unprotected from low frequencies – the tweeter will receive full audio spectrum via channel with FLAT low-end frequency response.

When you select menu option "DCX2496 Default" under "Crossover Tools". The DCX device will be set to the following values:

1. Channel Configuration is set to STEREO, 3WAY system: LMH LMH (CH1,2,3 CH4,5,6)
2. Channels are linked. Therefore, changes applied to channels 1,2,3 will automatically be transferred to channels 4,5,6.
3. Input section of the DCX2496 has flat frequency response and gain set to 0.0dB.
4. All output sections of the DCX2496 have gains set to o.0dB.
5. All frequency response processing functions are implemented in the output section of the DCX2496.
6. Output 1,4 has HP = 30Hz and LP = 300Hz filters.
7. Output 2,5 has HP = 300Hz and LP = 3000Hz filters.
8. Output 3,6 has HP = 3000Hz and LP = 20000Hz filters
9. All EQs, Dynamic EQs and Delays are OFF.
10. Polarity of all outputs is NORMAL.
11. Cut-off frequencies of the DCX2496 crossover filters are set to "LINKED" mode.
12. During flashing process, all outputs are muted first, then new settings are applied, and finally all outputs are unmated.

Using the "DCX2496 Default" menu option is a quick way of testing the RS232 link between your PC and the DCX device. When you select the "DCX2496 Default" menu option, you should see all red "Mute" LEDs on the DCX2496 front panel lighting up for a short time, and then going off, so the correctly filtered signals would appear on the outputs of the DCX2496 device. You can easily perform this checks with SoundEasy. The RS232 link is only active when the actual data is transferred from the PC to the DCX2496. When this process is completed, the DCX2496 becomes virtually a "stand alone" device. You can then use Spectrum Analyser screen to perform the above tests with the built-in White Noise signal generator. An alternative method for testing the DCX2496 would be of course the Digital or Analogue measurement screens. An example of DCX2496 SPL measured with SoundEasy MLS screen is shown below. Sampling frequency for channel H was set to 96kHz.

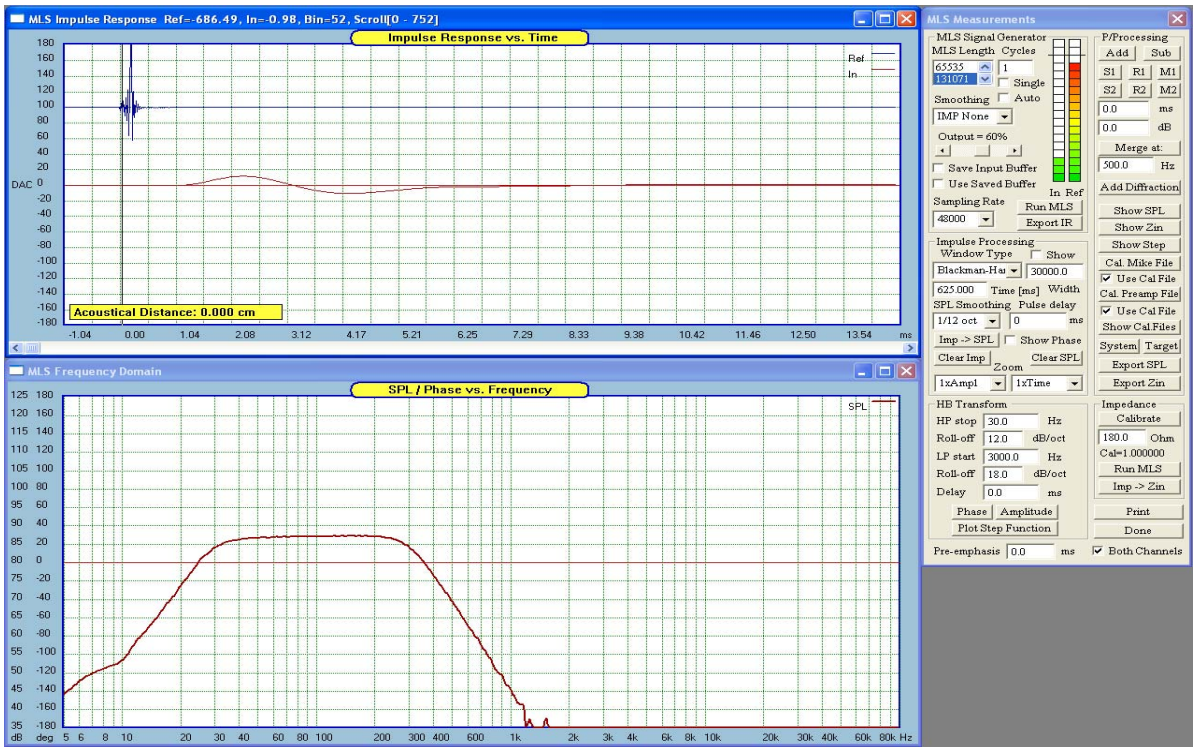


Fig 9.58. DCX2496 Default setting for L channel (IR + SPL).

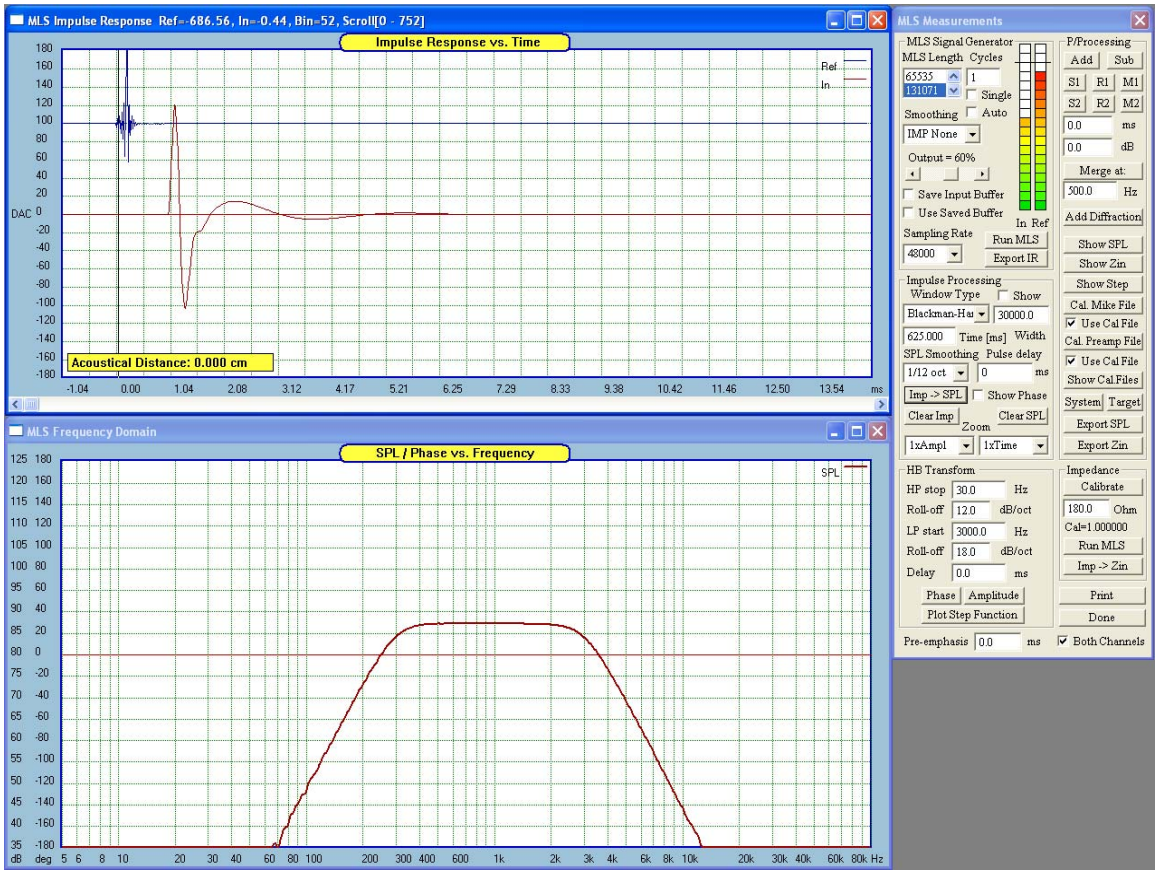


Fig 9.59. DCX2496 Default setting for M channel (IR + SPL).

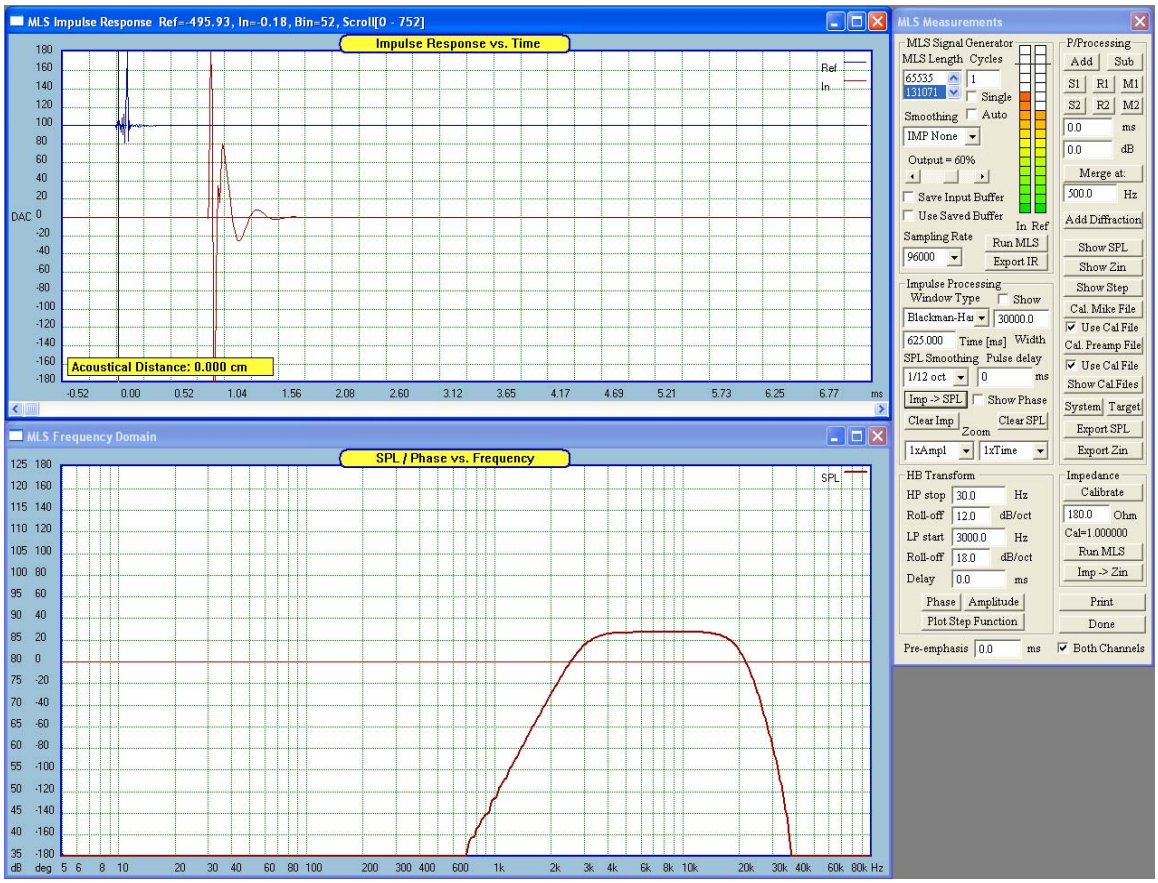


Fig 9. 60. DCX2496 Default setting for H channel (IR + SPL).

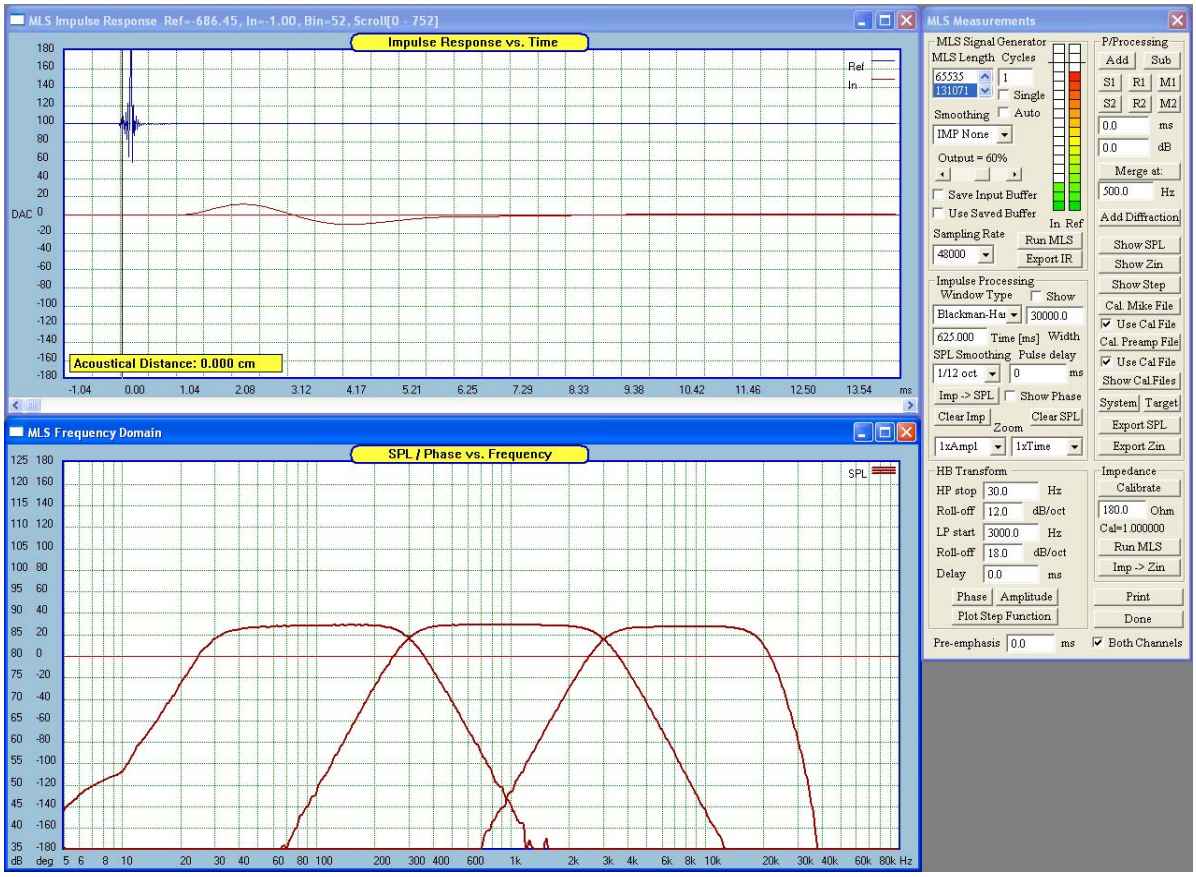


Fig 9. 61. DCX2496 Default setting for L+M+H channel (SPL).

Please note, that SoundEasy V9.00 transfers MC "Gain" as the combined DCX2496 output section gain – see DCX block diagram.

Disclaimer:

SoundEasy V9.00 is compatible with Behringer DCX2496, software version 1.14.
Please note, that, depending on the available technical information, Bodzio Software Pty. Ltd. makes every effort to maintain compatibility with DCX2496 software versions. However, we can not guarantee, that all Modular Crossover settings, will be successfully transferred to DCX2496. This is due to variations in DXC2496 software implementations.

Bi-Quad Filters

$$H(s) = K\,(s + z1)(S + z2) / [(s + p1)(s + p2)]$$

This equation for transfer function H(s), is called a biquadratic equation, or a biquad for short. The **zn** terms in the numerator represent zeros and the **pn** terms in the denominator represent poles. Sallen-Key filters, state-variable variable filters, multiple feedback filters and many other types are all biquads. There also is a "biquad" topology, and three of such filters have been implemented in this release of the program.

Bi-Quad filters are selectable from the built-in "Filter Selector" dialogue box, using **"Filter Type"** list box. There are three types of Bi-Quads available. Implementation is based on OPAMPS and as the filter includes capacitors and resistors, you will need to nominate Ro, Co, Qo (Q-factor), Go (gain at resonance) and high-pass, low-pass frequencies. Please note, that low-pass F3dB is used as a band-pass filter center frequency.

For a Bi-Quad, as Fc changes, the bandwidth stays constant, but the Q value changes. Thus, if you change Fc in the frequency domain, as Fc increases, the Q value increases and as Fc decreases, the Q value also decreases. It allows very high Q values, it can be configured in a 3 or 4 amplifier configuration, and it too is less sensitive to external component variations.

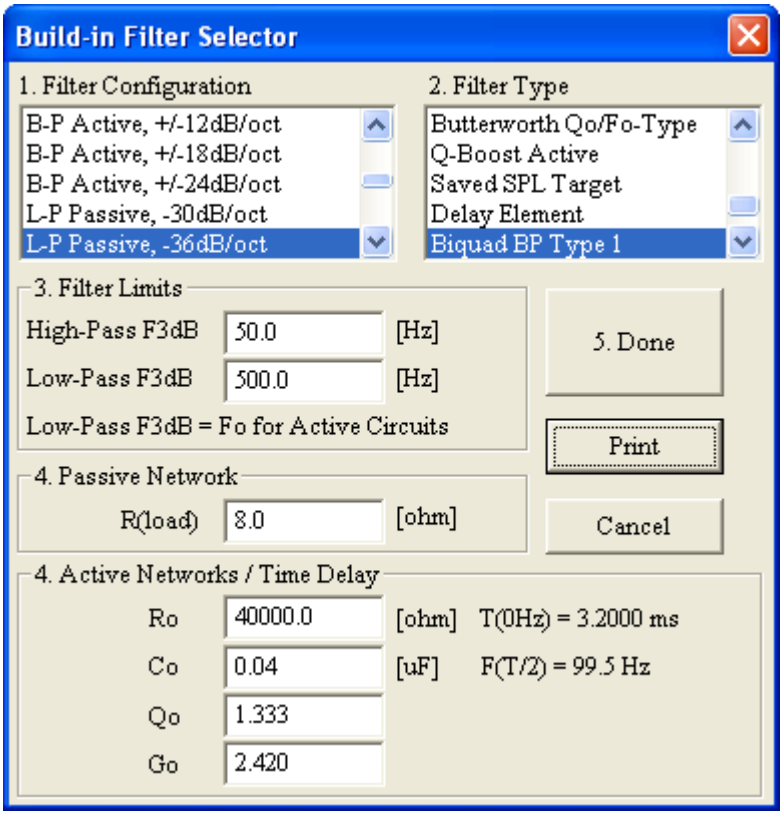


Fig. 9.62. Nominate Ro, Co, Qo (Q-factor), Go (gain at resonance)

Bi-Quad Type 1

Bi-Quad Type 1 is a band-pass filter. Selectable parameters are center frequency - Fo, quality factor - Qo and gain - Go.

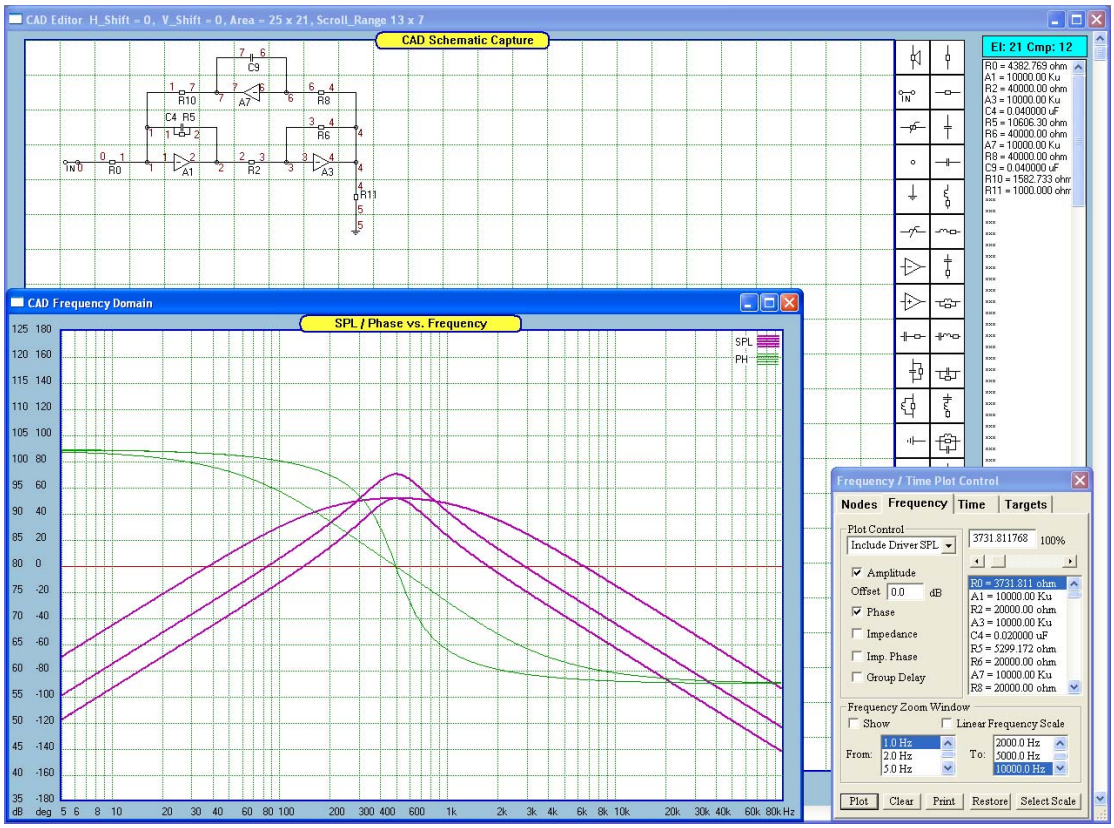


Figure 9.63 Examples of transfer function of Bi-Quad Type 1.

Bi-Quad Type 2

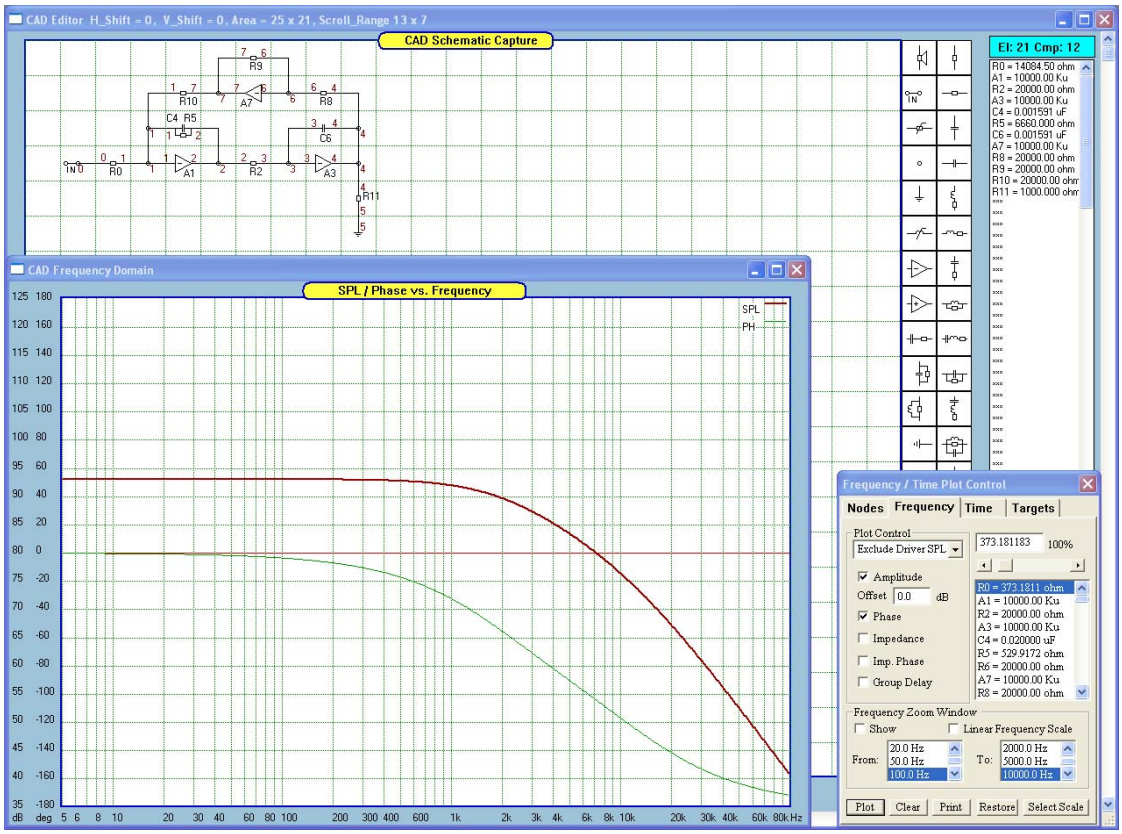


Figure 9.64 Examples of transfer function of Bi-Quad Type 2.

Bi-Quad Type 3

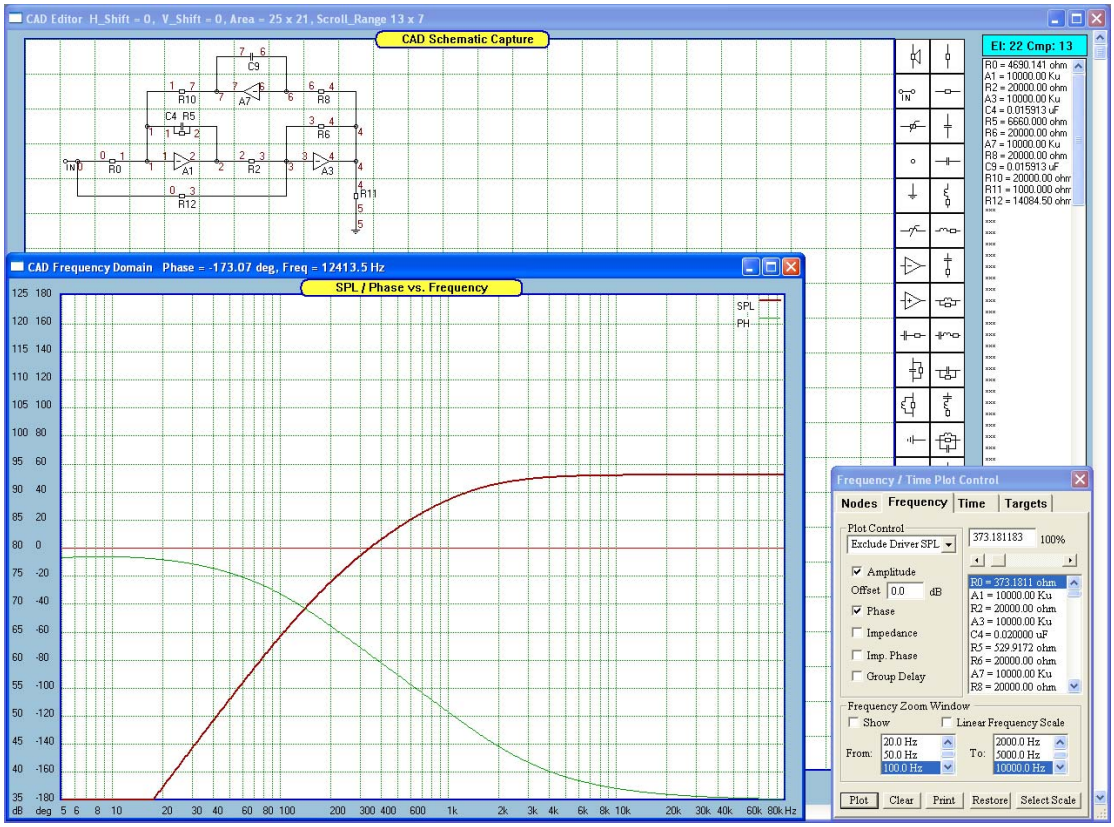


Figure 9.65 Examples of transfer function of Bi-Quad Type 3.

Linkwitz, Butterworth Asymmetrical Filters

At some occasions, you may need to use asymmetrical filters directly or, as templates/targets for your circuit to be optimized to. Asymmetrical filters can be created very easily in the program, but two such configurations are built-in for your convenience. These are: +12/-24dB/oct and –24/+12dB/oct active filters and targets.

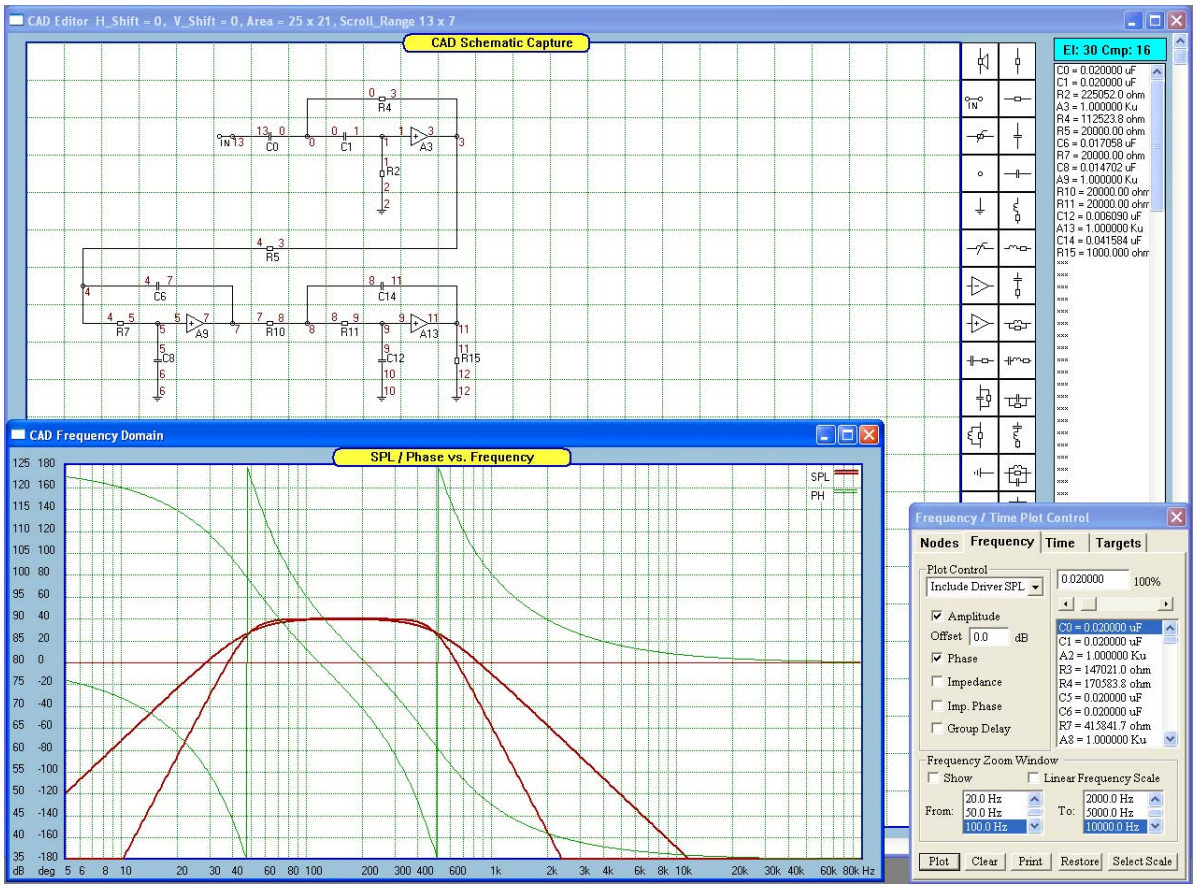


Figure 9.66. Linkwitz, Butterworth Asymmetrical Filters.

Linkwitz-Riley "Transformer"

When the subwoofer system is designed with the size in mind, it may happen, that the required enclosure volume causes a pronounced peak in response. This is an obvious sign, that the box is too small for the selected driver (its Vas). However, the peak can still be equalized and the low-end boost provided at the same time. The circuit recommended for this job is the Linkwitz-Riley filter and its example is shown on Fig 8.25. This circuit provides bass boost and correction for the peak. Design formulas are as follows:

1. Choose $f_{0,}Q_0, f_p, Q_p$,

2. $k = \dfrac{\dfrac{f_0}{f_p} - \dfrac{Q_0}{Q_p}}{\dfrac{Q_0}{Q_p} - \dfrac{f_p}{f_0}}$,

3. Choose C_2,

4. $R_1 = \dfrac{1}{2 * \pi * f * C_2 * [2 * Q_0(1+k)]}$,

5. $R_2 = 2 * k * R_1$,

6. $C_1 = C_2 * [2 * Q_0(1+k)]^2$,

7. $C_3 = C1 * \left(\dfrac{f_p}{f_0}\right)^2$,

8. $R_3 = R_1 * \left(\dfrac{f_0}{f_p}\right)^2$

Example presented on Fig 8.24 below, shows the Linkwitz-Riley filter calculated for Qo=2, fo=70, Qp=2, fp=25 and C1=10nF. Press “Load Filter” to have the circuit displayed.

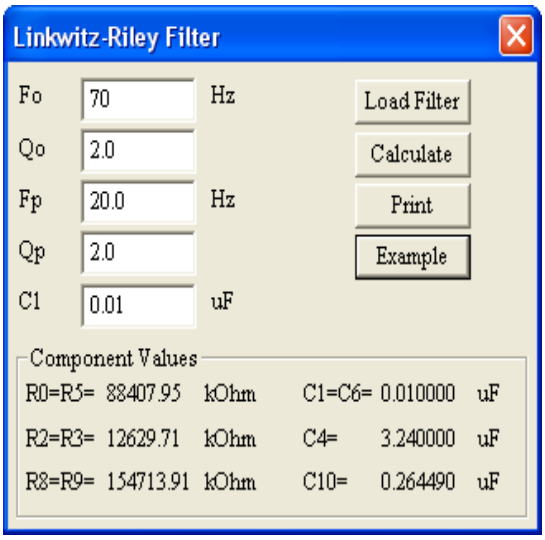


Figure 9.67. Linkwitz-Riley ‘Transformer”.

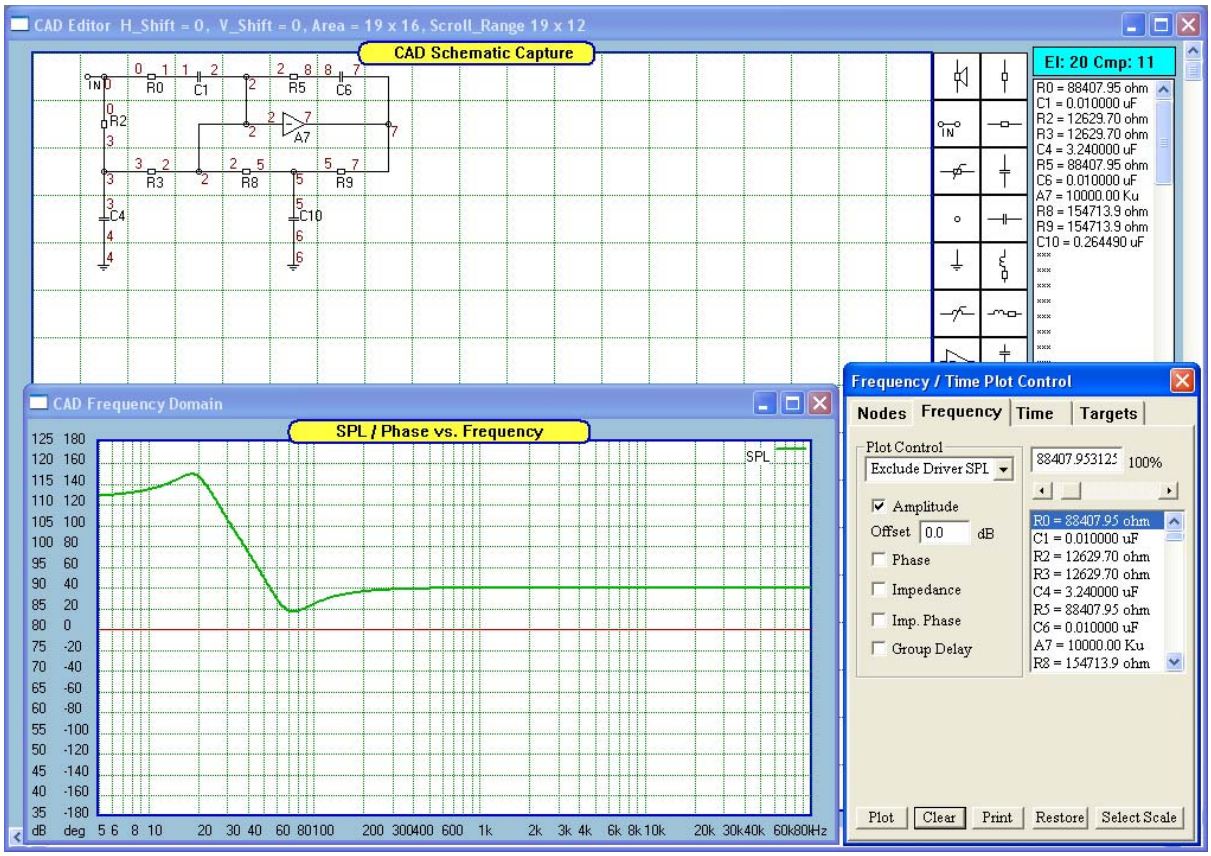


Figure 9.68. Frequency response of the L-R Transformer

Time Delay Circuit

As simple, first-order time delay circuit, is also available from the built-in networks. The delay is controlled by the Ro and Co components selected from “Filter Selector” dialogue box. The circuits can be easily cascaded, to obtain larger delays. The performance of the time delay circuit shown on Figure 8.27 can be estimated from the Filter Selector dialogue box. Entering values for Ro and Co will modify the time delay at To=0.0sec and also the frequency, F(T/2), at which the time delay falls to half of To value.

Both values: To and F(T/2) are displayed on the Filter Selector dialogue box. This way you can estimate how many sections on the active delay circuit you will need to obtain the required delay at the desired frequency.

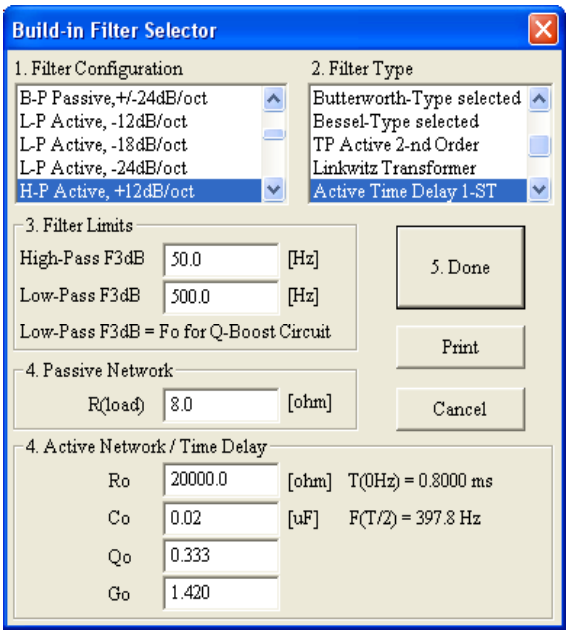


Figure 9.69. Time Delay circuit

In the example on Figure 9.70, the goal was to obtain 0.8ms delay at 100Hz.

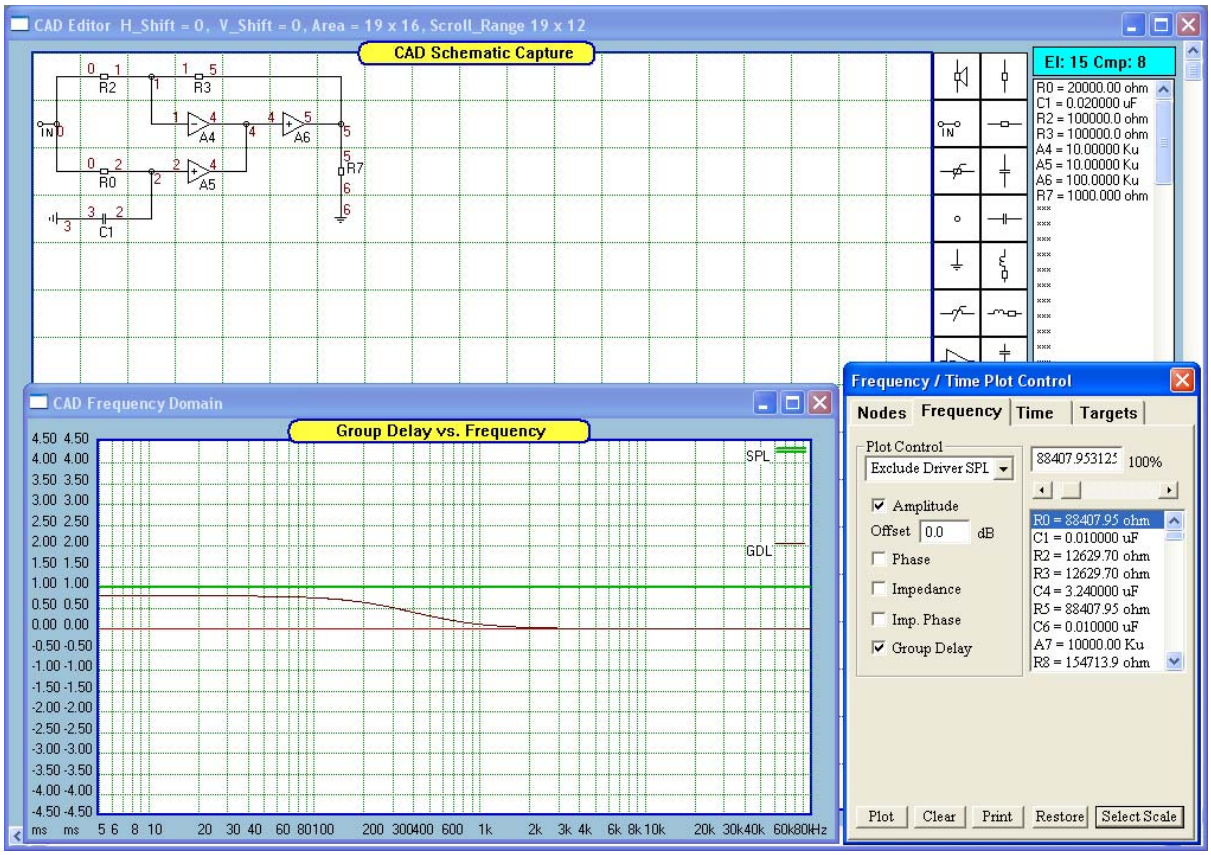


Figure 9.70. Built-in time delay circuit.

Notched Crossover / Filter

Notched filters are characterised by a null in the frequency response close to the transition frequency. Consequently, the SPL roll-off quite rapidly around the transition frequency. Additionally, the sum of the LP+HP filters sum to the flat response.

"k" is the ratio of lower notch frequency in the high-pass response to the crossover cut-off frequency, and is equal to the ratio of crossover cut-off frequency to lower notch frequency in the low-pass response.

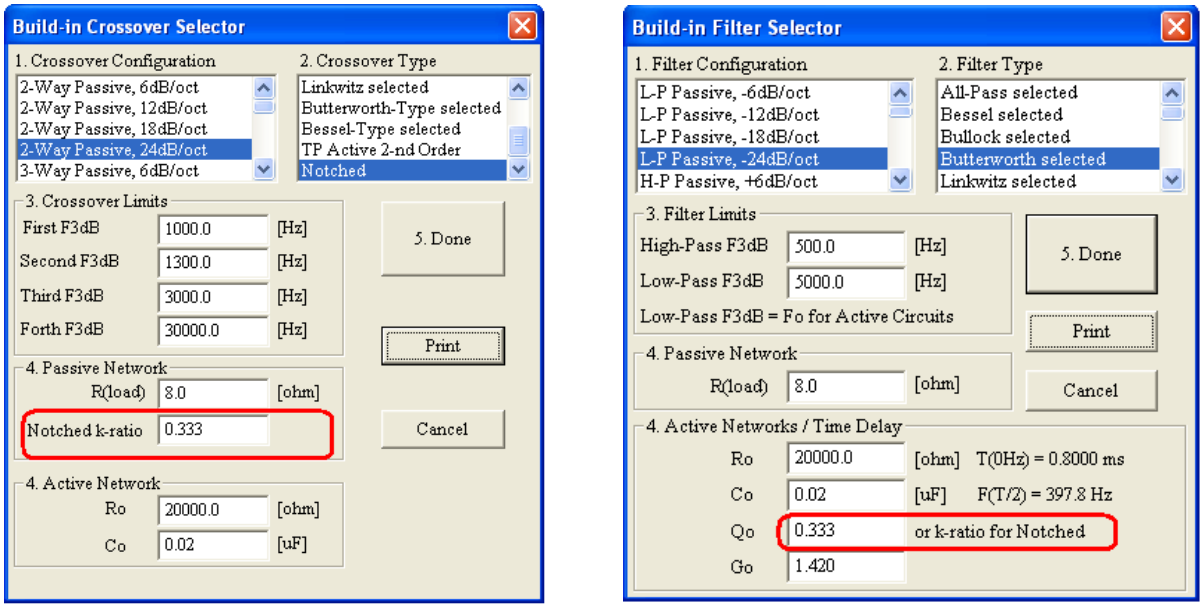


Figure 9.71. Notched k-ratio in the dialogue box.

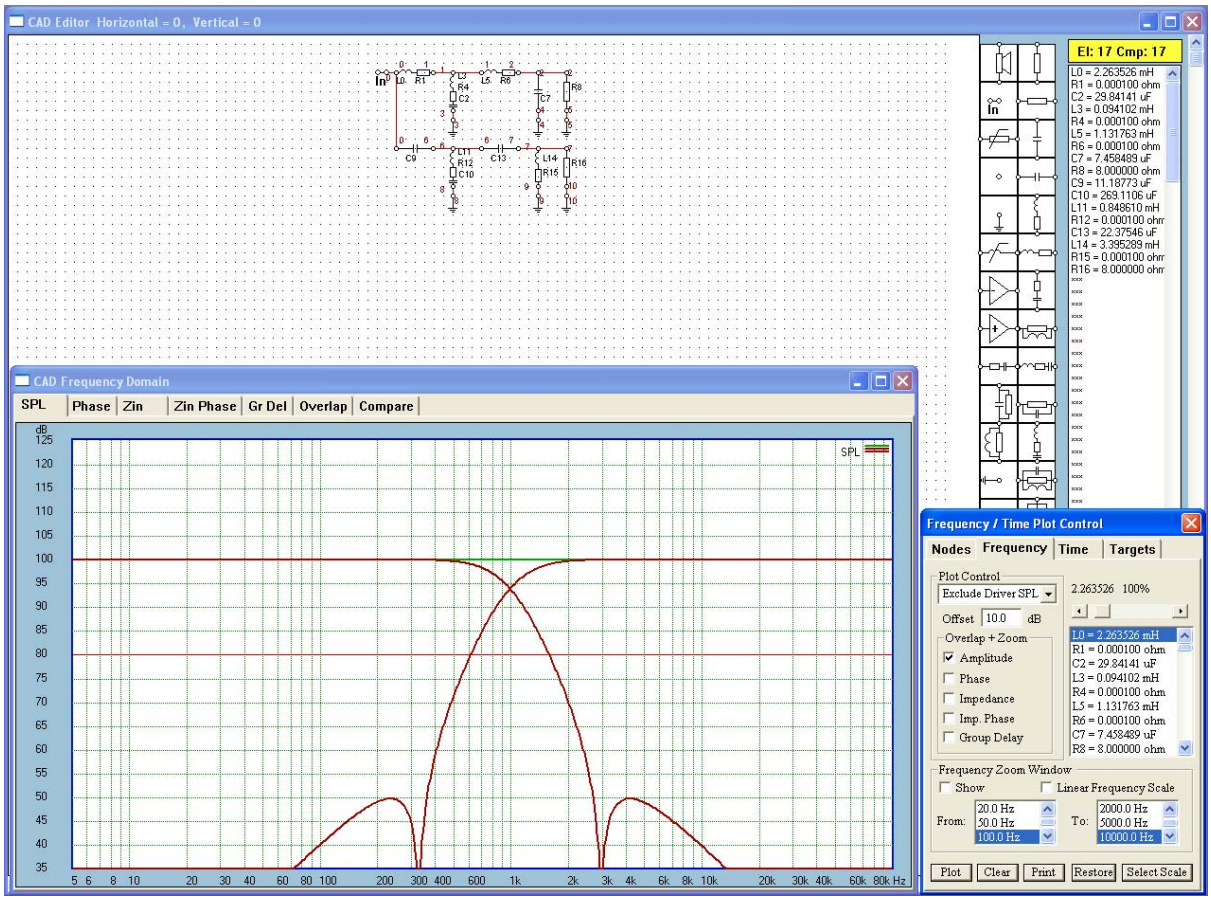


Figure 9.72 Example of a notched crossover.

Chapter 10. Filter/Crossover Optimization

Background

Classical approach to crossover filter design is based on electrical filter theory. The models used are based on the assumption that:

1. The input impedance of the driver is purely resistive thus constant vs. frequency.
2. The frequency characteristics of the drivers are independent of frequency.

In reference to assumption 1, the program offers the CAD tool which helps the designer to neutralize the reactive parts of the input impedance of the driver. However, assumption 2, if left unresolved, may totally undermine the performance of the designed system. The “Enclosure Design” tool, introduced in Chapter 1, is the first step on the way to obtain a correct frequency response of the low-, band- or high-pass channels of the system. The "channel" is understood to be a summed response of the following components:
1. Driver - complex transfer function,
2. Enclosure - modifies input impedance of the driver,
3. Compensation network - affects the input impedance and frequency response,
4. Crossover - affects frequency response.

Therefore, the filter transfer functions and component values calculated from the filter theory should only be used to construct the first approximation of the crossover filter. The final values of the components should be only decided upon after the optimization of the filter has been performed and it produced satisfactory improvement in the filter performance. This task has been assigned to the optimizer screen. The default window of the tool with the main control box is shown on Fig 10.1, where an example data file has been loaded and displayed on the screen.

Optimization

You may decide to optimize only the crossover network component values. It is very likely, that majority of the designs would follow this option. However, we will also show an optimization example, where the amplitude bump will be eliminated by optimizing previously calculated components. However, since all the passive/active components entered from the CAD screen may be actively involved in the optimization, the user may wish for example, to split the compensation network into L-pad attenuator and impedance compensation network for the midrange driver (and lock them) AND crossover components - as the second group of components. The second group will not be locked - the optimizer screen will optimize only the crossover (unlocked) components. Please upload now “Test_900_2.mid” data file. When the process is completed, select **“Filter/Crossover Optimizer**” under **“Crossover Design**” menu option. The screen that will be opened next will look similar to the one shown on Fig 10.1.

Your first step in the optimization process is to select 'From' and 'To' frequency range of the optimization. It is easier to make the decision while looking at the actual frequency response of the selected filter. The frequency response can be plotted using **“Old Values**” button on the “**Optimization**” tab. Upon examination of the responses, we decided to set the 'From' frequency to 200.0Hz and 'To' frequency to 12000.0 Hz. You must now set-up a target response for the optimizer. Operation of this dialogue box has been described on page 10.12. Let's assume, that the target curve shape was selected as a second order (+12/-12dB per octave) Butterworth response and cut-off frequencies have been set to 500Hz and 5000Hz for high-pass and low-pass respectively.

If you now depress 'Optimize' button, the optimization process will commence at the end of which the curves as on the Fig 10.2 will be plotted (you can go ahead and press the 'Optimize' button for the sake of experiment. If you do so, you must re-load the test_902.mid file, when the optimization process is completed). This is not exactly what we expected. The brown line represents the filter response and is badly distorted at higher frequencies. The total channel response is however very good. It resembles very well the targeted, 2-nd order Butterworth band-pass filter. To explain the problem please refer back to Fig 10.1. Here we can observe, that initial channel response exhibits already faster roll-off at higher frequencies then the 2-nd order target function. This is due to the fact, that channel response is a combination of the driver and filter responses. Driver's frequency response starts to roll-off around 5kHz and this combined with the selected filter response will produce much faster roll-off than the targeted 2-nd order curve. The program has managed to modify the component values such a way, that the overall response resembles the targeted function, but this result is clearly unacceptable.

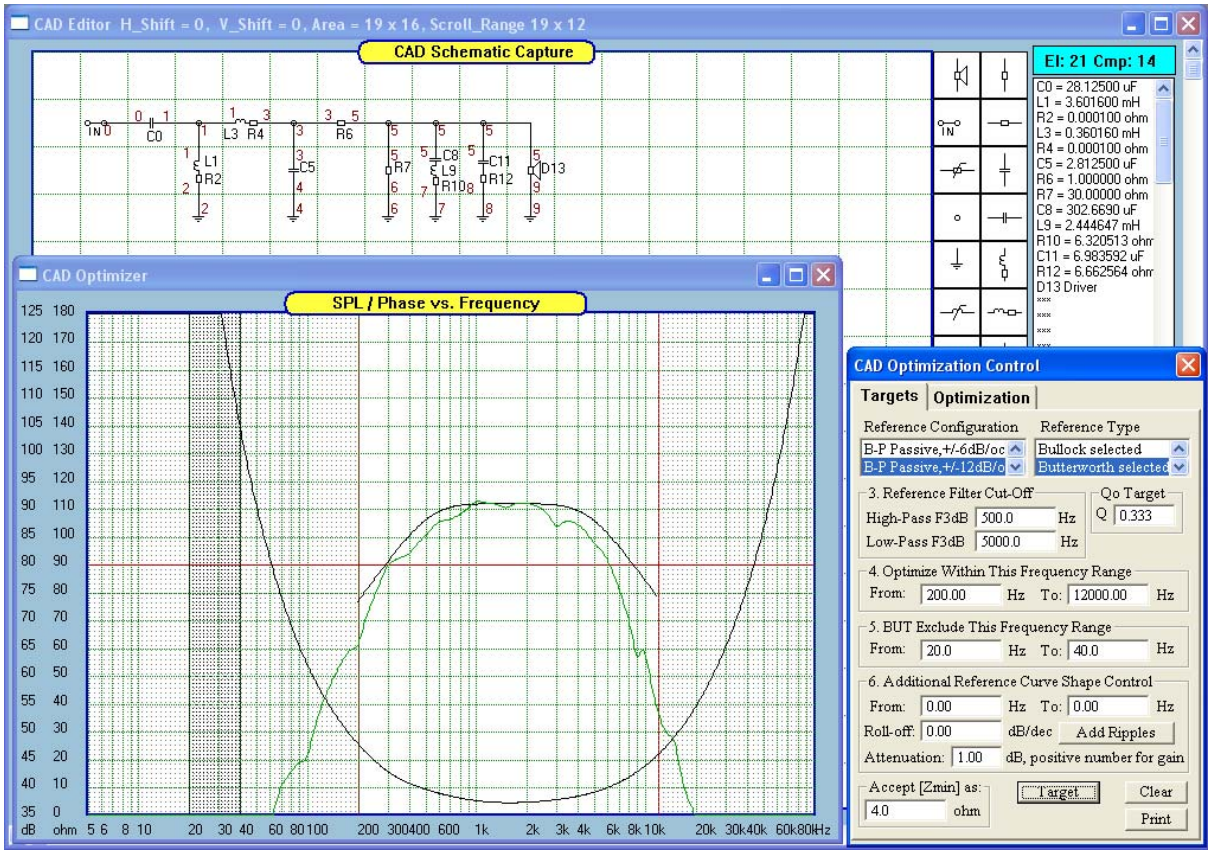


Fig 10.1 Optimization control dialogue box

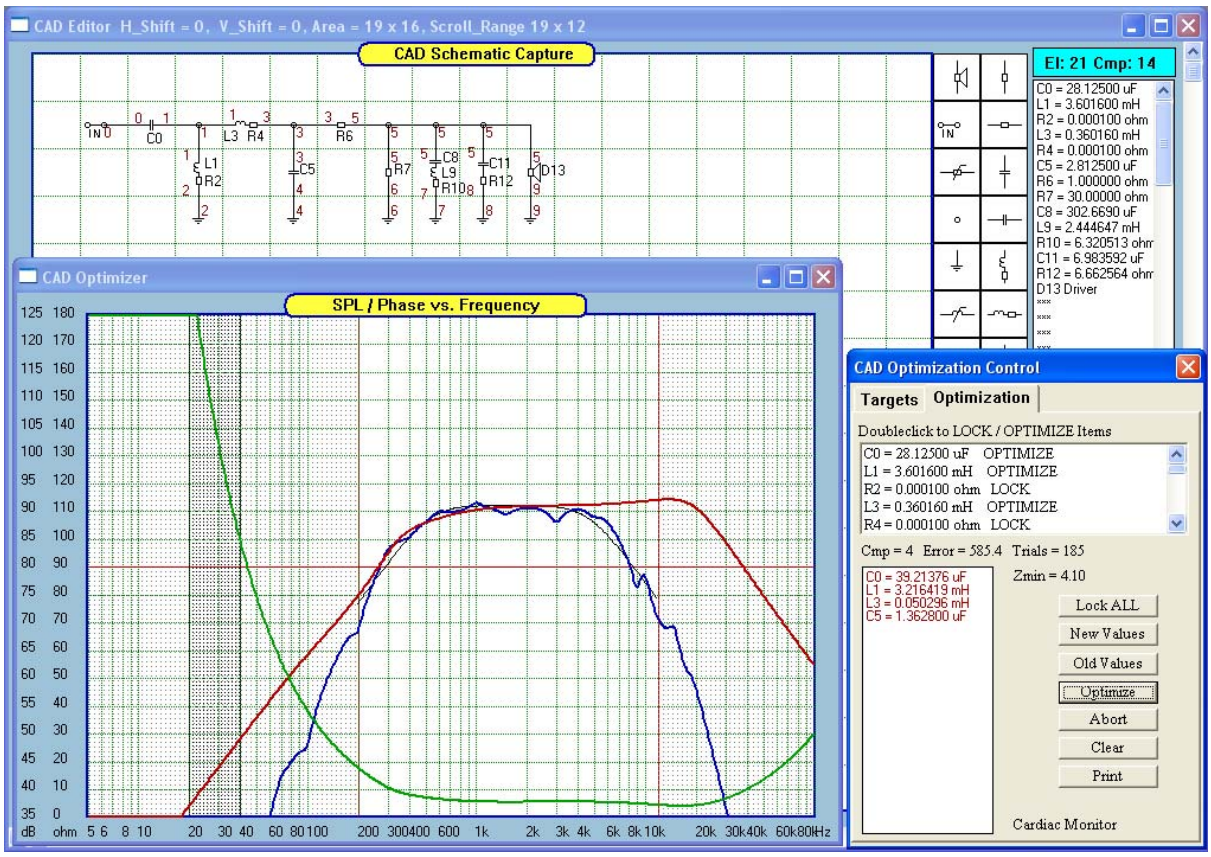


Fig 10.2 The results of wrongly selected optimization parameters

A better solution here would be to target 4-th order band-pass filter as the final response and then perform the optimization. Fig 10.3 and 10.4 show this scenario. Please note, that frequency range of optimization was narrowed because a filter with faster roll-off was selected. Should the old frequency range be kept, the algorithm would be optimizing the response down to -35dB level which is not recommended. When the optimization process is completed, you should obtain plots shown on Fig 10.4.

In case of the test_902.mid driver, its efficiency was 1.5dB higher (92.5dB) then that of the selected woofer (91dB) so that the value of 1.5dB was used for calculating L-Pad (if this is your design goal). Now, the midrange channel efficiency is (92.5-1.5)dB=91dB. This is 1dB over 90dB SoundEasy reference and the same 1.0 dB is now entered in the 'Attenuation' field of the optimizer module. The "Attenuation" parameter indicates, how much we would have to shift the 90dB reference line to match the system efficiency.

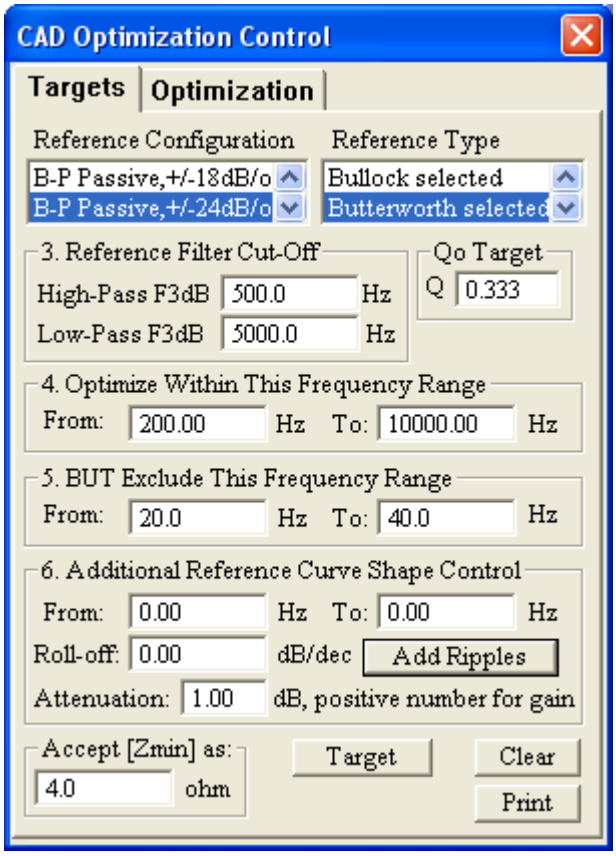


Fig 10.3 Better selected optimization parameters

Selecting Optimization Range

You can edit the boundaries of the optimization range in form of "From" and "To" frequencies. The optimization range will have a great effect on the speed and accuracy of the optimization process. Fig 10.5 shows the woofer driver channel (Test_900_2.wfr data file) after the optimization. Only the crossover was optimized and Butterworth low-pass filter was used for reference. The optimization process was performed over 50 - 2000 Hz frequency range. In order to estimate the frequency range, click on "Use Original Values" button on the main optimization screen. The complex transfer function of the driver will be plotted on the screen. Next, select the targeted frequency response. Having these two curves on the same screen, you can observe, that above 2000Hz the curves differ significantly. The driver's transfer function rolls-off more rapidly and in order to bring it closer to the reference frequency response, some gain would actually be needed. The driver's output is only 20 dB below the nominal level and the driver would contribute very little to the overall system frequency response.

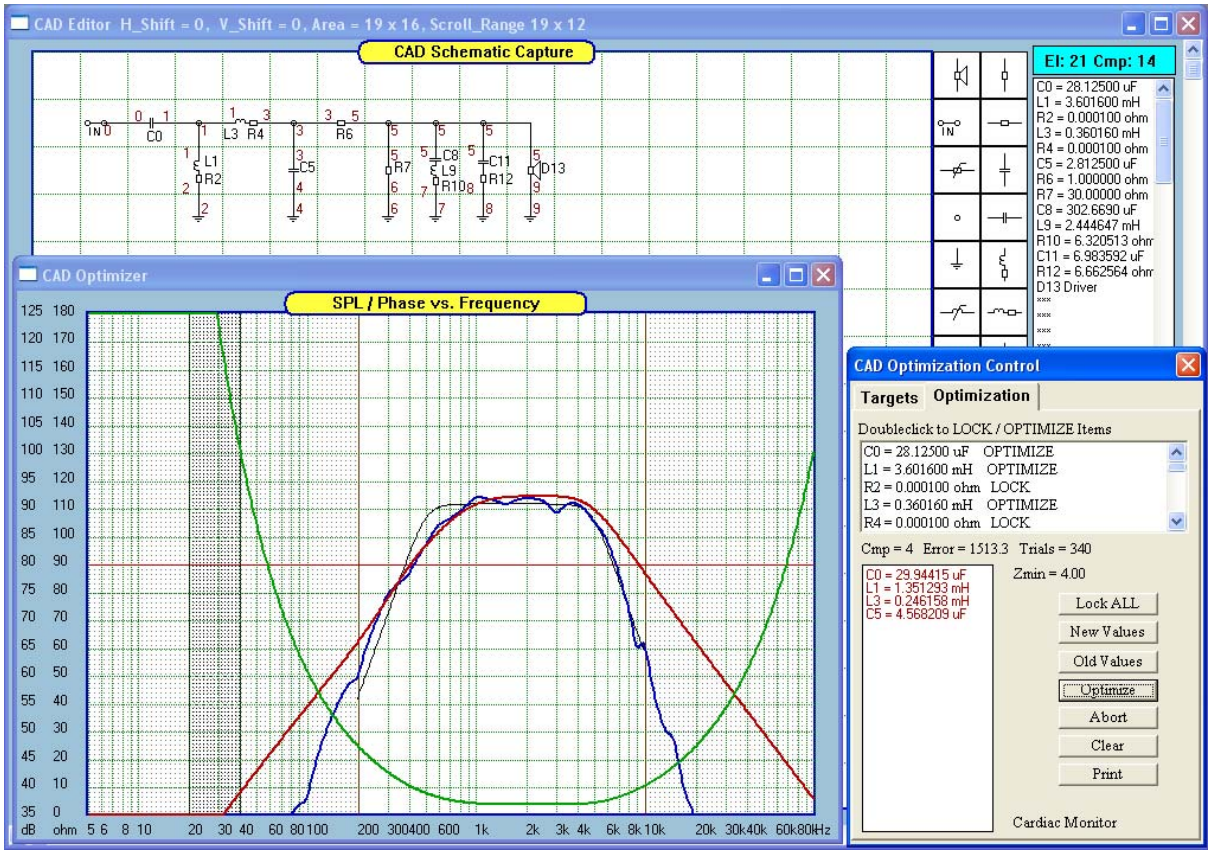


Fig 10.4 The results of better selected optimization parameters

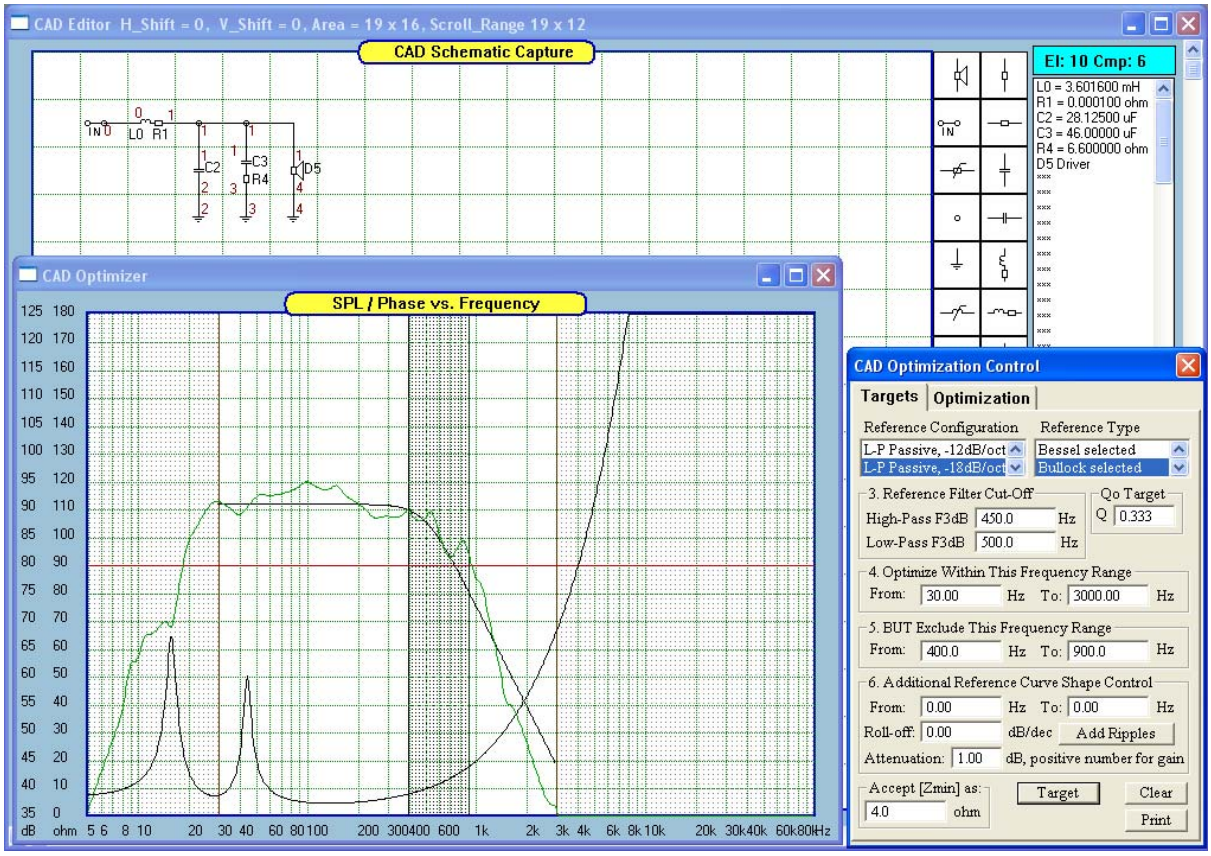


Fig 10.5 Dialogue box showing proposed optimization parameters

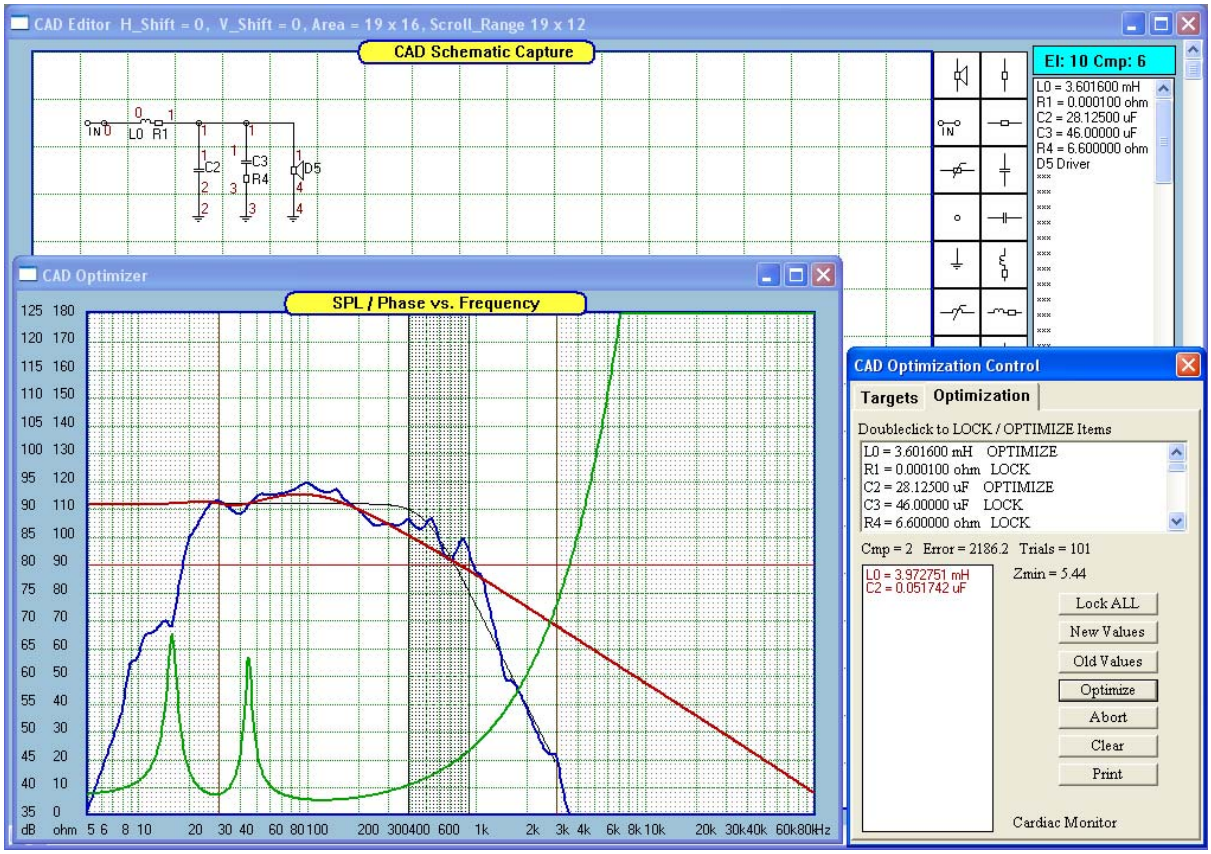


Fig 10.6 Results of the optimization + new filter response (brown).

If the frequency range above the 2000 Hz mark is included in the optimization process, the algorithm used would attempt to compromise frequency response within the useful frequency range, that is 50 - 2000Hz, in order to minimize error between 2000 -3000Hz. This is a highly undesirable situation and can be a source of significant errors in the optimization process. It would therefore be more sensible to select even narrower frequency range eg: 20 - 1200 Hz for the optimization range. In this example, a woofer driver was taken into consideration, but the described method of selecting the optimization range can be applied to other drivers.

It is always recommended to vary the optimization boundaries and perform two or three optimizations and inspect, at least visually, the degree of fit between the target response and the total channel frequency response. If you wish the restart the optimization process always from the same starting conditions (component values), which is recommended you must not save the previous values and re-load the data file each time prior optimization. Save only the results you want to be used by the next module. You can also EXCLUDE a frequency range from optimization. Simply specify the "From:" and "To:" frequencies in the "Exclude" frequency range box – see Fig 10.3.

Additional Shape Control

All reference target curves can be pre-distorted by adding a slope or plateau to the basic shape of the filter. Fig 10.7 shows an example of pre-distorted, -24dB/oct Butterworth filter. The distortion was generated as follows:

1. "Attenuation" was entered as 10dB, therefore, the whole curve is shifted up by 10dB.
2. From 20Hz to 100Hz the frequency response rolls-off by 15dB/dec.

One (but not only) possible application of the pre-distortion is to deal with diffraction distortion. Diffraction distortion typically causes the SPL curve to rise from around 200Hz and flattens around 5kHz (depending on the geometry of the enclosure). Therefore, you may aim to have your crossover designed and optimized around the diffraction distortion. The **"Roll-off"** field accepts positive number for creating rising edge and negative number for creating slopes. Also, the **"Attenuation"** field can be used to add gain – enter positive number or loss – enter negative number.

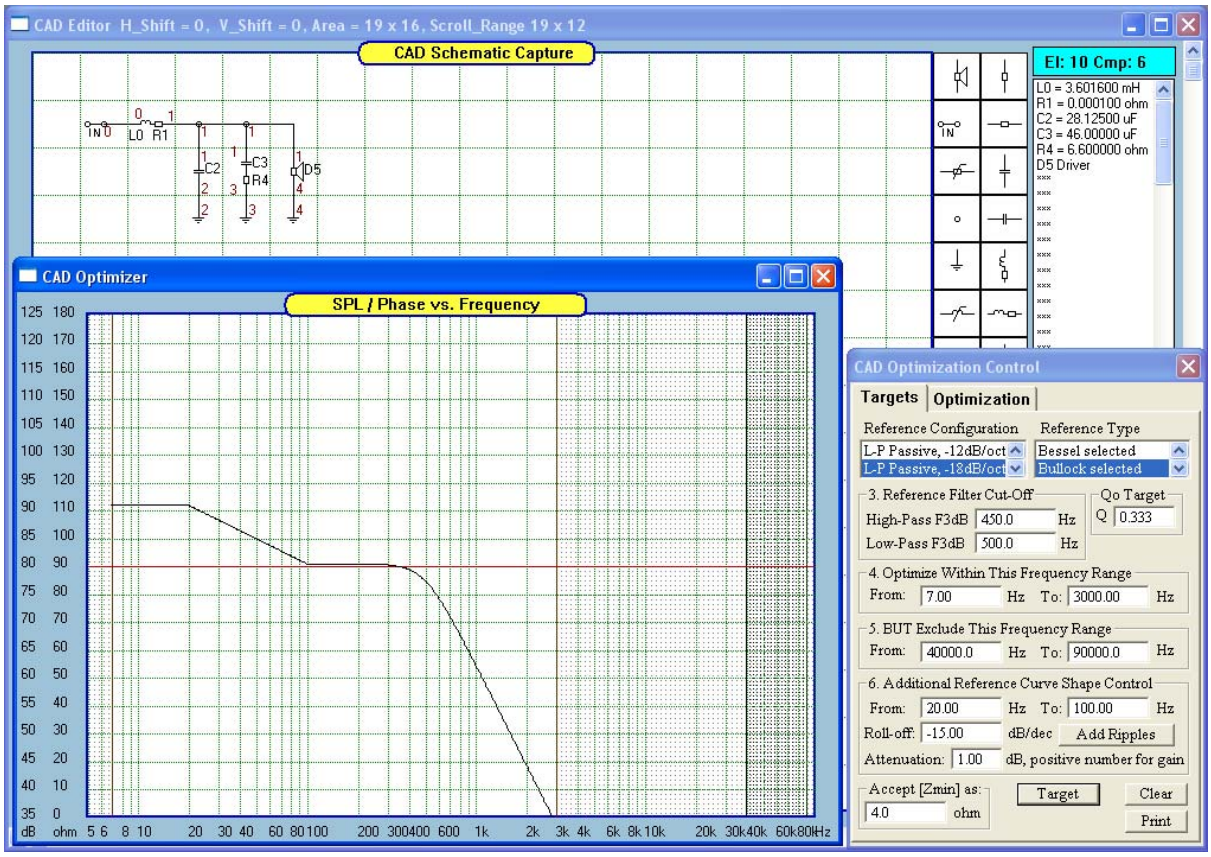


Fig 10.7 Additional shape control – pre-distorted filter

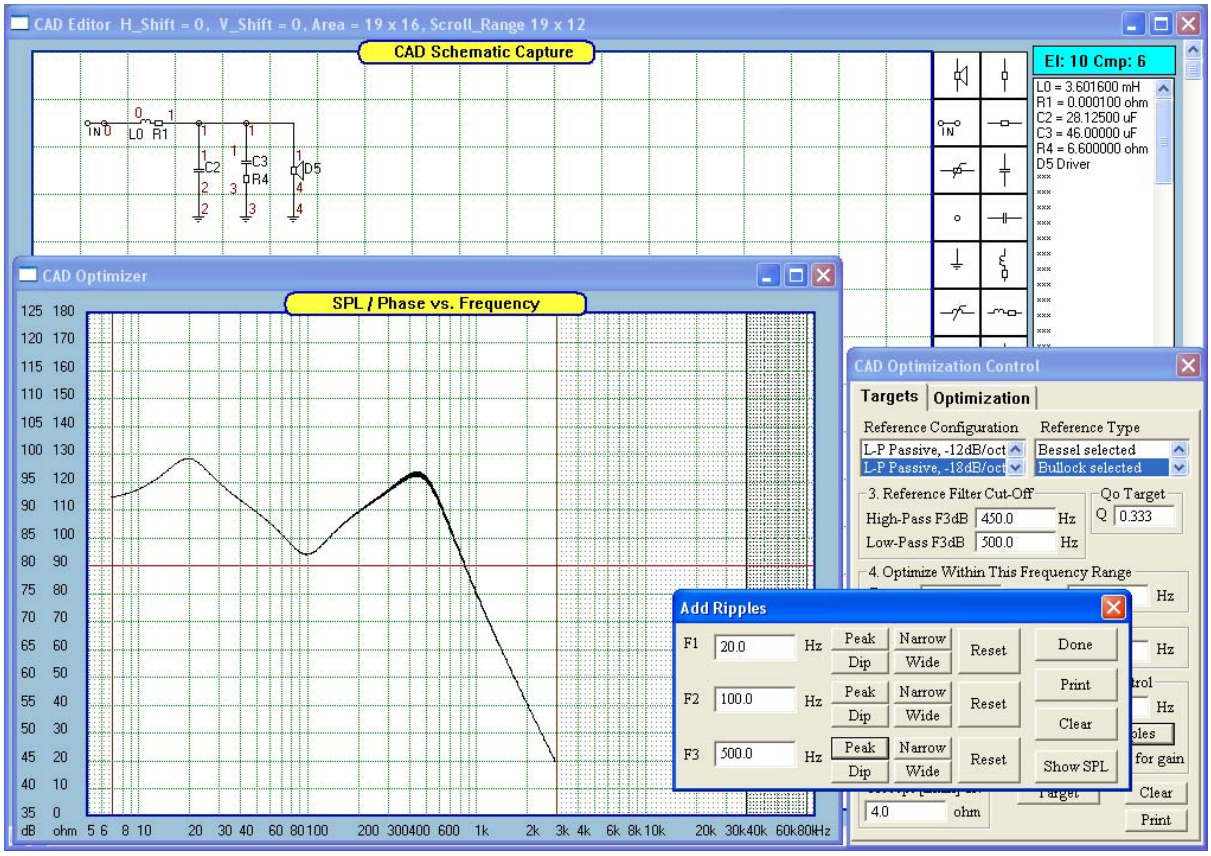


Fig 10.8 Additional shape control – adding ripples

Adding Ripples

In addition to the above pre-distortion, three "ripples" can be added to the target curve. Each ripple can be a peak or a dip in the frequency response. In order to incorporate the ripples into the target curve, please select "Add Ripples" button on the "Optimization Control" dialogue box. This dialogue will be replaced with smaller control box "Add Ripples" allowing you to control all parameters of the newly introduced ripples:

1. **Width** of the ripple – use "Narrow" or "Wide" buttons.
2. Location on the **frequency** scale – enter frequency in the "F1", "F2" or "F3" data field.
3. **Height or depth** of the ripple – use "Peak" or "Dip" buttons.
4. **Cancellation** of the ripple – press "Reset" button.
5. **"Show SPL" – use this button to display the plot SPL with optimized values.**

The "Clear" button will clear and re-plot the target curve on the main optimization screen. Finally, when you are happy with the result, press "Done" button to close the "Add Ripples" dialogue box. All parameters of the ripples are saved to the data file.

Locking components.

If it is required, that not all components are to be optimized, you may resort to locking the selected items. In order to tell the program, which components are to be excluded from the optimization process, you must **DOUBLECLICK** on the required component, listed in the **"Optimization**" tab list box. The selected item becomes highlighted and also corresponding items under **"Optimization**" tab and "**Optimize/Lock**" columns become highlighted automatically. This process is particularly important when larger number of components is involved. And not all of them fit into the display area of the list box. You may need to scroll the list box in order to bring the required component into the box display area

In summary, **DOUBLECLICK** in the list box on the component you wish to exclude from optimization process. Corresponding items become highlighted automatically. Also, the "**Optimize/Lock**" column **TOGGLES** from OPTIMIZE to LOCK. The lock/unlock status table of each component **is saved in the data file** and the default value is "Optimize" for all components. Optimization routine will terminate if two consecutive trials do not reduce global error by arbitrarily small amount. Since active crossovers employing negative feedback are quite immune to variations in gain, it is highly advisable to lock the gain of the amplifier as it would cause the optimizer to terminate.

Filter Optimization Example

We assume to following:

1. Woofer efficiency, η woofer = 91dB
2. Midrange efficiency, η midrange = 92dB
3. Efficiency difference, η difference = 92 – 91 = 1dB, **higher for midrang**e.

Example:

To provide maximally flat midrange filter frequency response, so that:
1 Final midrange channel efficiency is the same as woofer, that is 91dB.
2 Final acoustic frequency response of midrange channel follows +/-18dB/oct Butterworth filter from 500-5000Hz for –3dB drop at the corner frequencies.

Solution:

1. Efficiency difference is nullified by inserting an L-Pad (R6+R7) attenuating signal by 1dB. Those values were calculated assuming driver's Re for the load.
2. Driver's impedance resonant peak is equalized by series notch (C10+L11+R12).
3. Driver's voice coil inductance if equalized by Zobel Network (C8+R9).
4. Amplitude response peak around 1500Hz is equalized with Series Tank (L14+R15+C16).

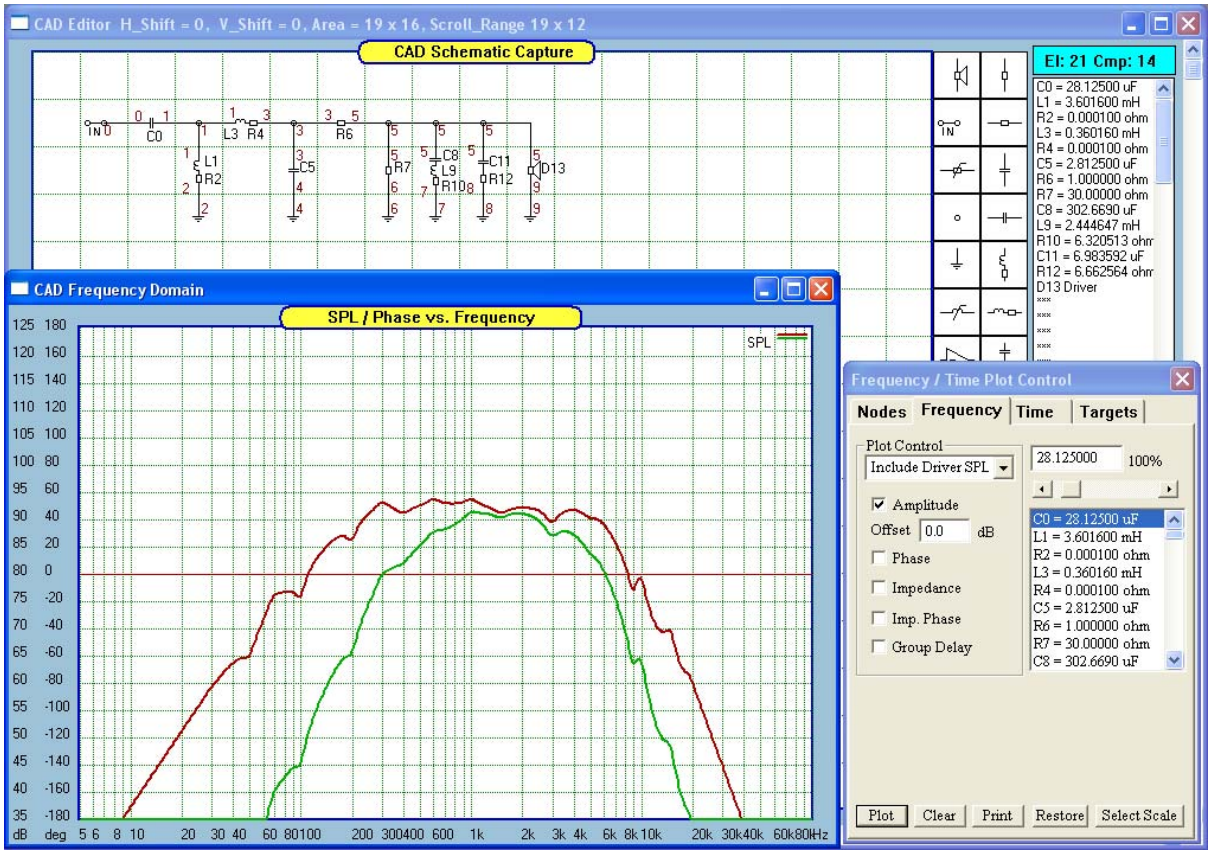


Fig 10.9 “Driver Only” – brown curve, and complete midrange filter – green curve.

Having dealt with the impedance and amplitude irregularities, we can now experiment with the Optimizer to determine if we can improve of the midrange channel frequency response. At this moment, channel frequency response is far from being optimal with the following problems:

1. Attenuation at 5000Hz is excessive, -10dB.
2. Attenuation at 500Hz is also excessive, -8dB.

The overall shape of the midrange channel acoustic response (green curve) on Fig 10.9 is far from being optimal for the selected Butterworth filter.

Excessive attenuation at 5000Hz is caused by the combination of driver’s natural roll-off, which starts at 4500Hz and filter corner frequency of 5000Hz. Clearly, the combination is poorly chosen and the driver should not be used for this frequency range. However, our goal is to illustrate the design process, therefore let us see, how the optimizer would deal with this problem. We can immediately observe, that standard circuit components for Butterworth 2-nd order filter do not meet these requirements. However, driver’s natural frequency roll-off will add to the steepness of the slopes below and above corner frequencies. To start the process and illustrate the procedure, we select +18/-24dB/oct reference target filter. Our initial evaluation of the problem indicated, that midrange driver’s SPL was 1dB too high. The L-Pad attenuator was therefore inserted in the driver’s path to reduce the level. Now, the midrange driver SPL is plotted on the screen 1dB lower (around 91dB level). However our target reference curve is still sitting on 90dB level. If nothing is done about it, the optimizer will try to compensate for this 1dB difference in levels and it is almost guaranteed, that optimized component values will be wrong.

Fig 10.10, shows the result of optimization for Attenuation = 0.0dB. Please note the Error = 602.3 Fig 10.11 shows the same optimization process for Attenuation = 1.0dB. The results are significantly better as Error is equal to 490.5. Final optimization was performed for Attenuation = 2.0dB (Fig 10.12) and the resulting Error is 558.

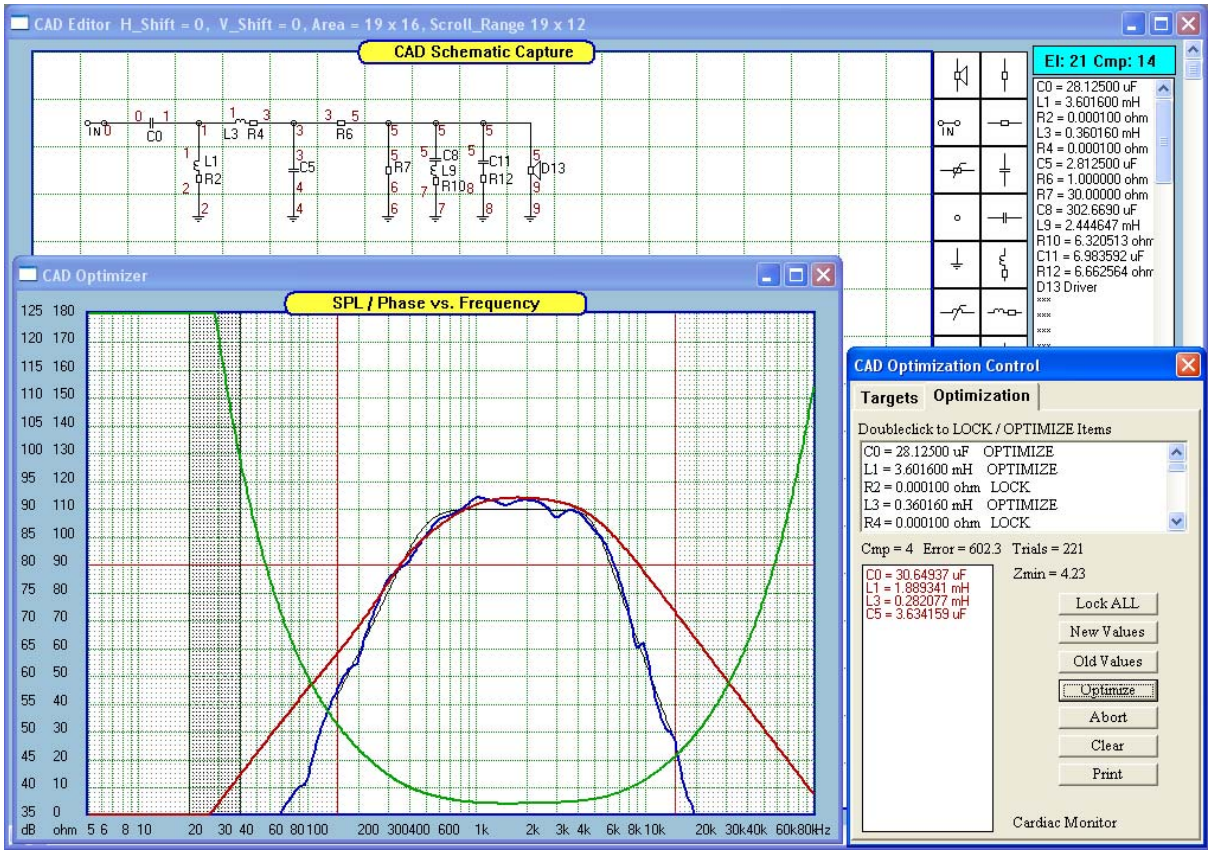


Fig 10.10. Target curve: +18/-24dB/oct, and 0.0dB attenuation

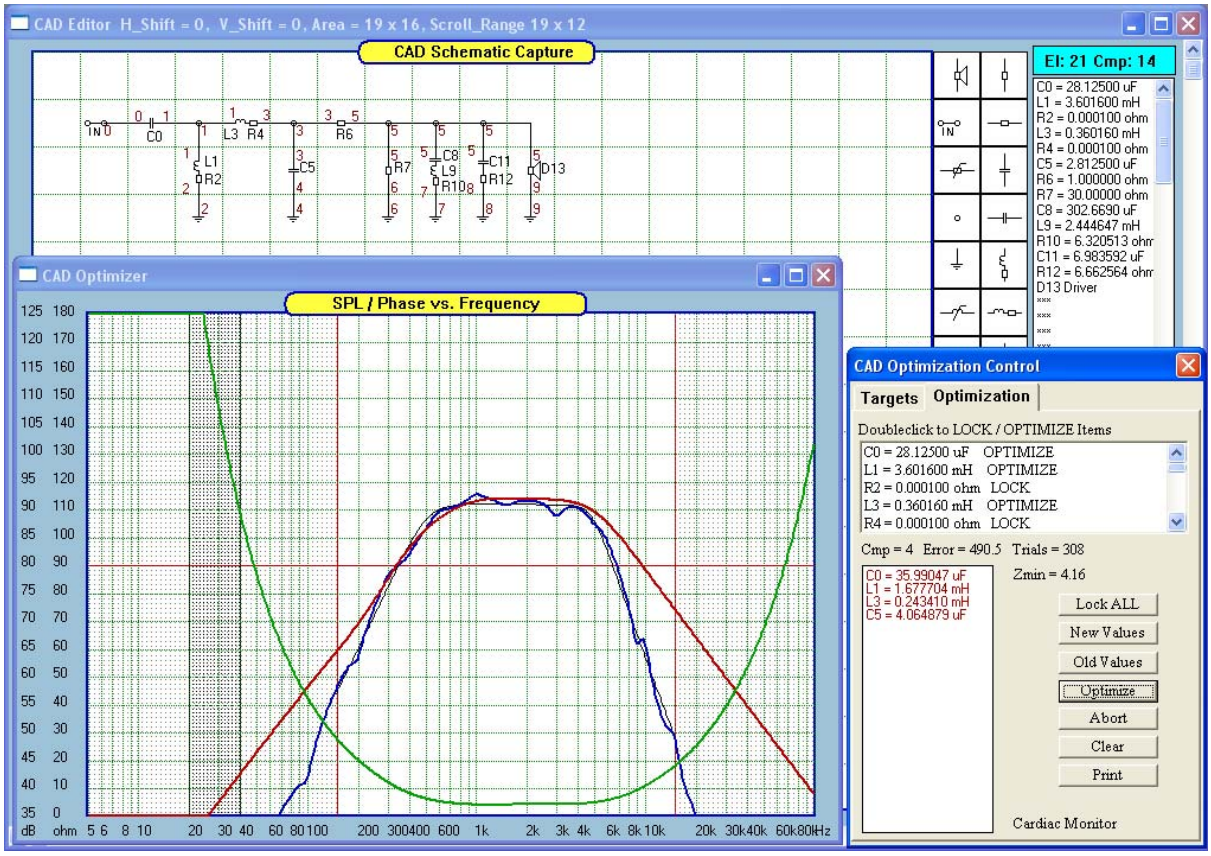


Fig 10.11. Target curve: +18/-24dB/oct, and 1.0dB attenuation

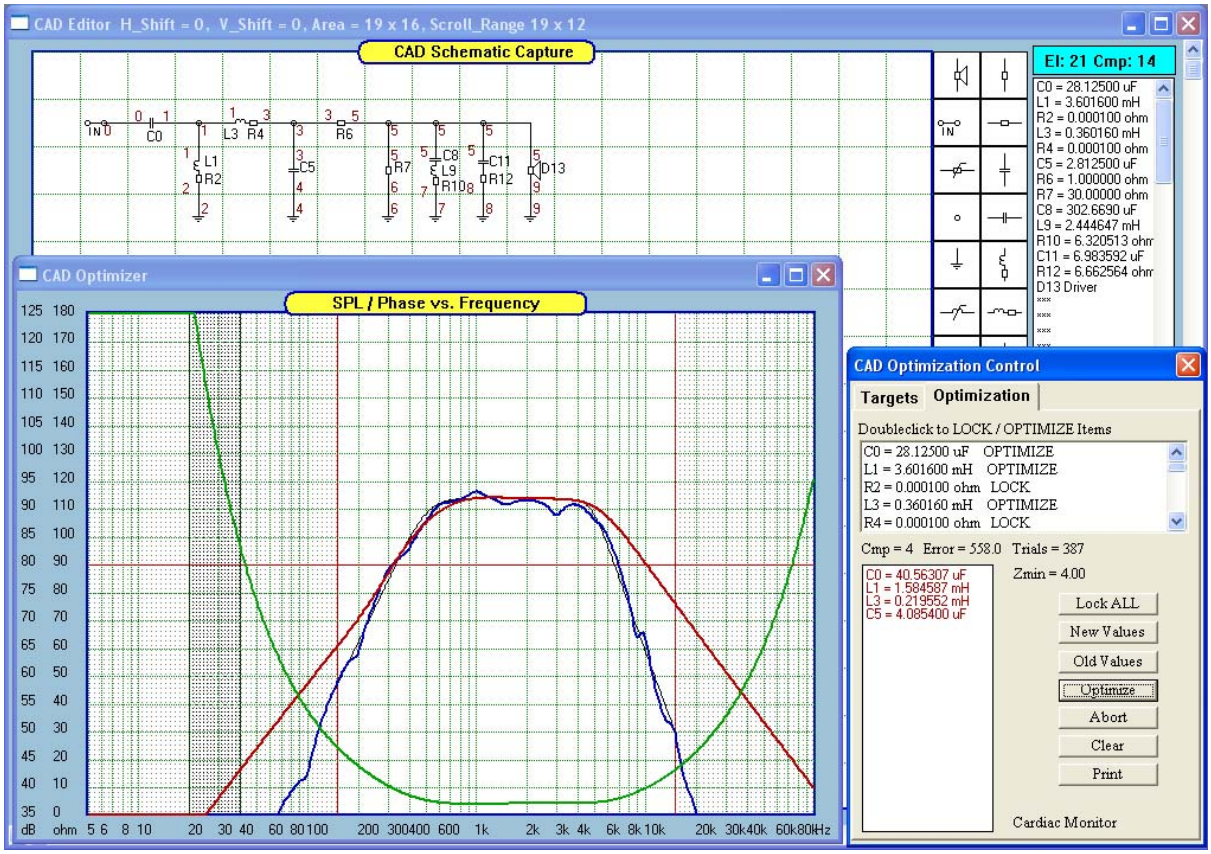


Fig 10.12. Target curve: +18/-24dB/oct, and 2.0dB attenuation

The lowest Error is now 490.5 and it is clearly the lowest of all three optimizations. Our choice of Attenuation parameter was based **on inserted L-Pad attenuation, SPL of the driver (92.5dB) and the 90dB reference the program defaults to.** In general, the whole optimization process is mathematically intensive and depends on accuracy of entered data:

1. Correctly entered (or imported) SPL curves.
2. Matching SPL curves data and SPL value. For example, if you entered 95dB for SPL of the driver, but its frequency response curve points are nowhere near this level, you have a problem.
3. Properly selected L-Pads.

It needs to be stated here, that it is quite acceptable to correct diffraction problems by reducing SPL of the midrange and tweeter drivers. In this case, you may elect to increase L-Pad attenuation, so the whole SPL curve will be shifted down by several dBs. In this case, you would need to shift accordingly the target curve for optimization, and the Attenuation field is just the way to accomplish this. As it is observable from Fig 10.10-10.12, the Optimizer also decided to shift the upper slope of the filter to about –12dB/oct. This is due to the unfortunate choice of the driver/filter combination, where driver alone rolls-off too early and quite steeply. Other component values were adjusted for the "best fit" to the targeted +18/-24dB/oct target response.

To visualize the results of the optimization, the original midrange response and the optimized midrange response were plotted on Fig 10.13 and 10.14 respectively. It can be easily observed, that the optimized curve is significantly better match to the required Butterworth, 3-rd/4-th order bandpass filter with 500 – 5000Hz corner frequencies and efficiency of 91dB.

Interestingly, the improvement in performance was accomplished without changing the topology of the filter. The asymmetrical filter was selected as an example only, but please observe, that the original frequency response of the midrange driver (Fig 10.9) also exhibits asymmetrical character.

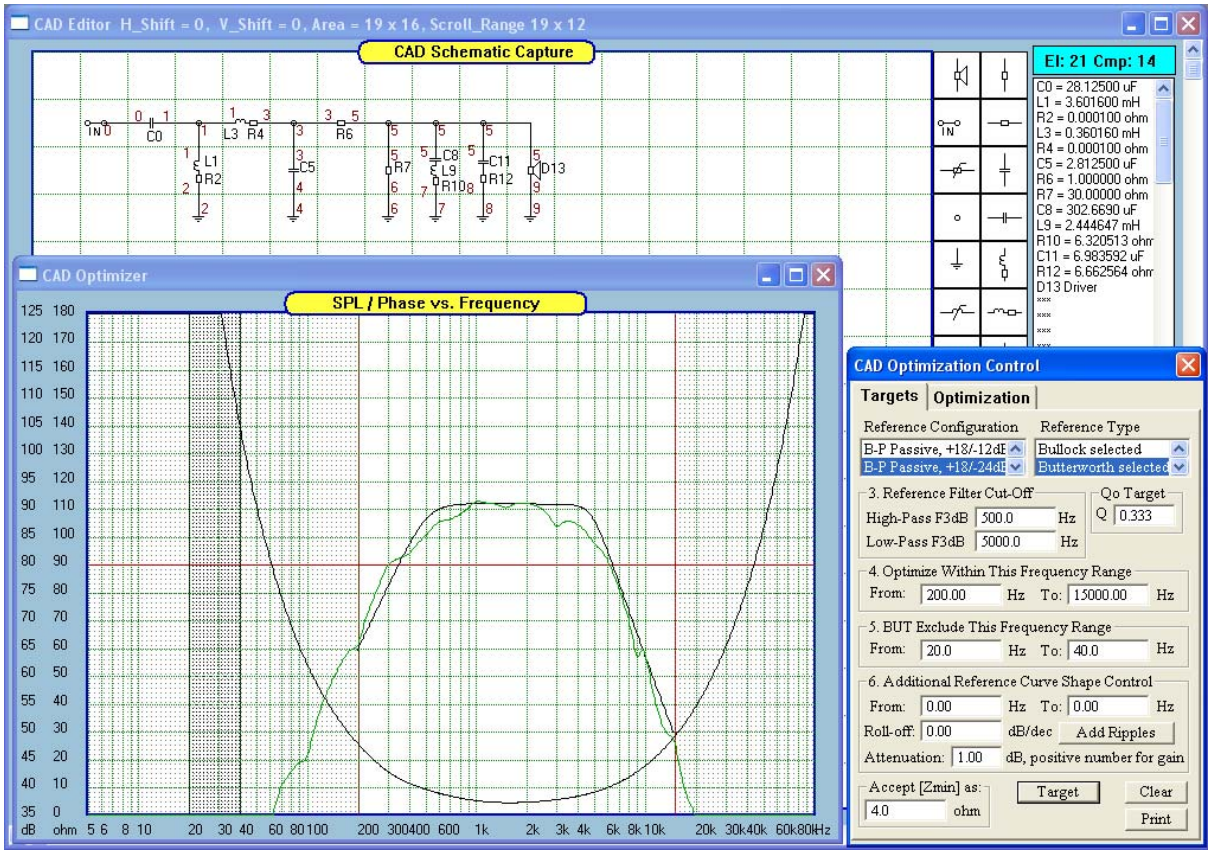


Fig 10.13 Original midrange filter – curve 1, +18/-24dB/oct target – green curve.

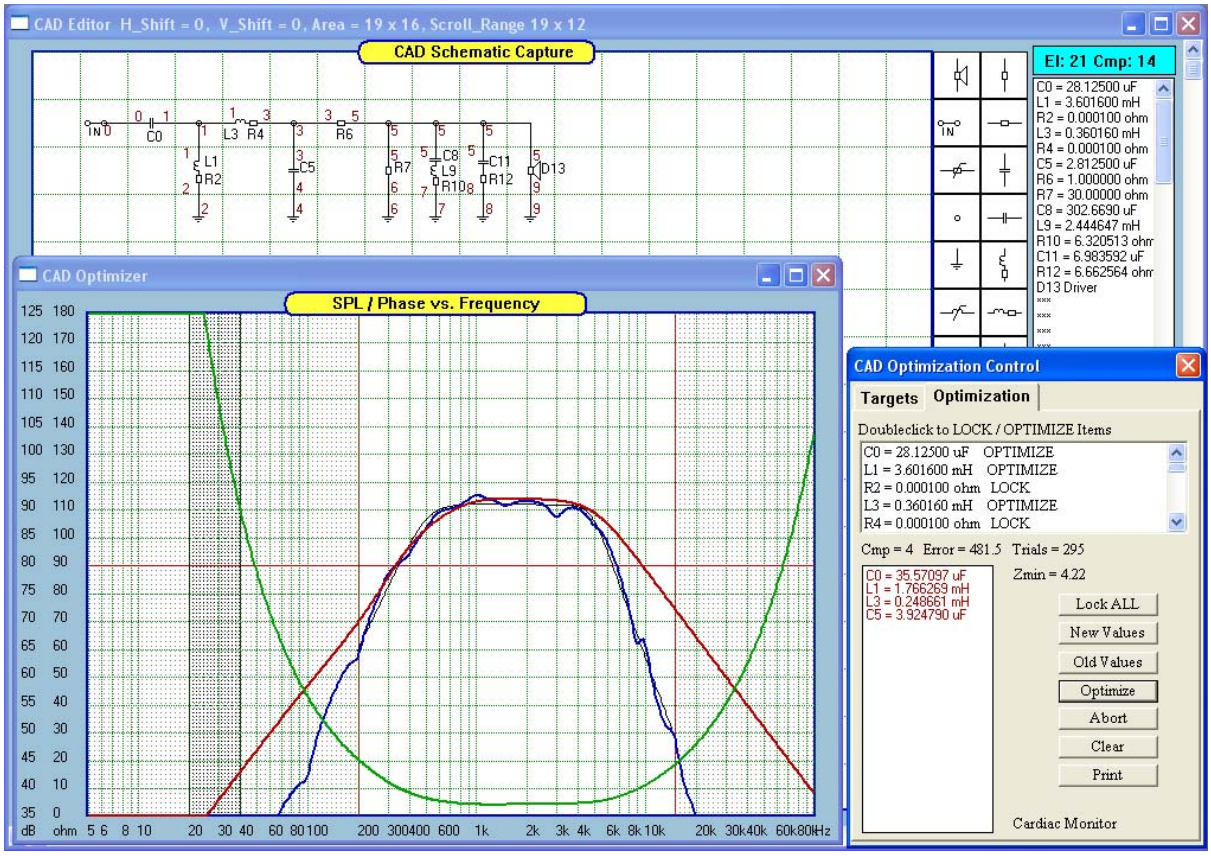


Fig 10.14 Optimized midrange filter – curve 1, +18/-24dB/oct target – blue curve.

In summary – please refer to Figure 10.15

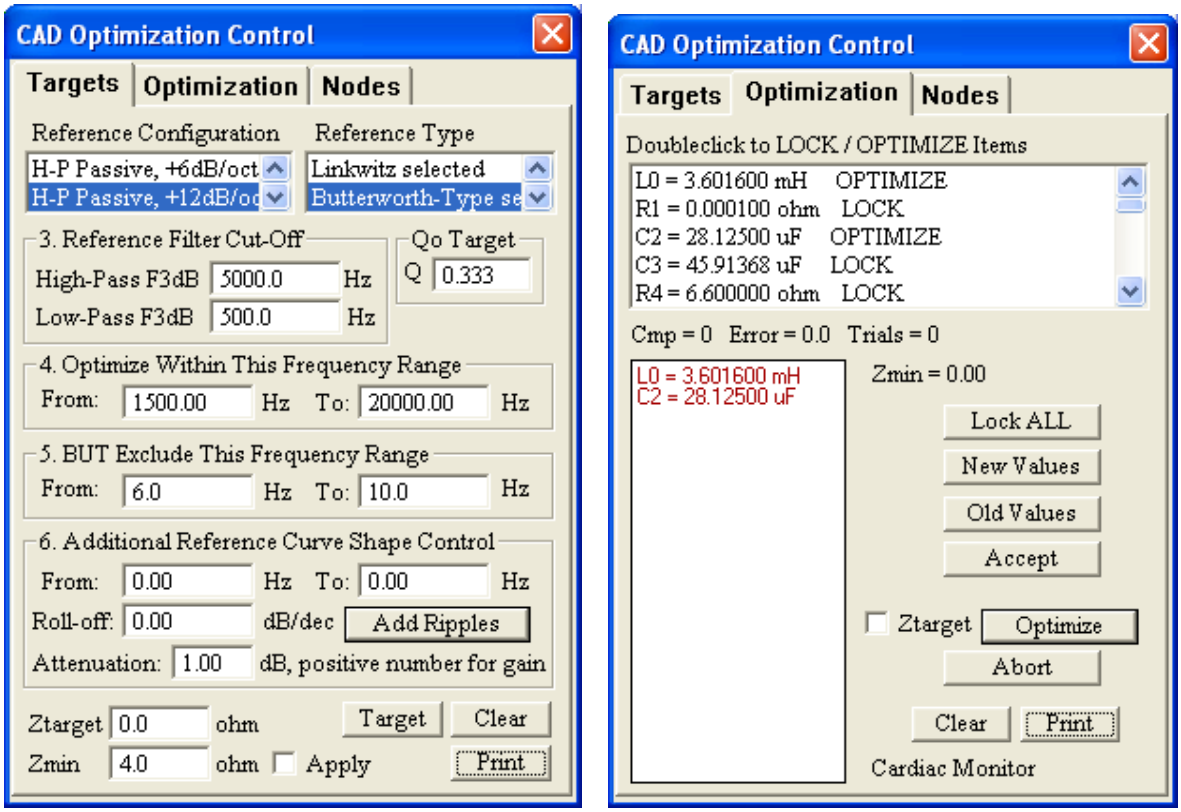


Figure 10.15. Optimization Controls.

Optimization process is intended to adjust crossover components, so that the overall acoustical response of the channel (woofer, midrange and so on…) is close to the ideal electrical filter response. **Optimization Control Dialogue Box** has the following controls:

Targets Tab

1. Select the required Filter Configuration from the **"Reference Configuration**" list box.
2. Select the filter (eg: Butterworth) type from the "**Reference Type**" list box.
3. **"Reference Filter Cut-Off**" – High-pass and Low-pass frequencies of the filter.
4. **"Optimize Within This Frequency Range"** - two editable fields to accomplish just that. Enter frequencies in Hz.
5. **"BUT Exclude This Frequency Range"** – again, two editable fields to accomplish just that. Enter frequencies in Hz.
6. **"Additional Reference Curve Shape Control**" - group has four editable fields. The **"From"** field selects the frequency (in Hz) from which the response is to roll-off. . The **"To"** field selects the frequency (in Hz) at which the response is to flatten again The third field - **"Roll-off**"- is used to enter the required tilt in dB/decade. Usually, it is a small amount: 1-3dB/decade.
7. The **"Attenuation" parameter indicates, how much we would have to shift the 90dB reference line to match the system efficiency.** For example: system efficiency is 86dB – enter "Attenuation" as –4.
8. It is recommended, to use **"Target"** button on the dialogue box to review system target reference line before commencing optimization process.
9. **"Clear"** button – simply clears the screen.
10. **"Add Ripples"** – this button closes the "Optimization Control" box and opens another box to enable you to add 3 ripples.
11. **"Accept Zmin as:"** - editable data field to enter the lowest acceptable input impedance your amplifier can tolerate. **MUST be greater than 0.1ohm**
12. **"Print"** button – prints the screen.

Optimizer Tab

1. **"Lock All"** button – press to lock out all components from optimization.
2. **"New Values"** button – press this button if you are happy with the optimized values and you want to use them across the module.
3. **"Old Values"** button – press this button if you want to return to the original component values.
4. **"Accept"** – press this button to accept newly optimized values.
5. **"Clear"** button – simply clears the screen.
6. **"Abort"** button – press this button to STOP optimization process.
7. **"Optimize" –** pressing this button invokes the "Optimization Control" dialogue box.
8. **"Print"** button – prints the screen.

Nodes TAB is the same as described in CAD section of this manual.

Constrained Optimization.

At times, you may need to assure, that the best optimization process does not lead to the resulting input impedance dropping dangerously low – that may cause potential problems for the power amplifier. If you feel, you need to impose a limit on the minimum input impedance of the system or crossover, please enter this value in the provided **"Accept [Zmin] as** data field.

The results of the optimization will typically be not as good as for "unconstrained optimization" , that is setting the Zmin to a low value of say: 1.0 ohm.

An example of two types of optimization are given on Figure 10.16 and Figure 10.17. Firstly, constrained optimization was performed on the tweeter filter and the requested minimum input impedance was set to 4.50 ohm. Next, practically unconstrained optimization was performed by setting the Zmin = 1.0 ohm. The result was the Error was much lower for the unconstrained optimization. However, both results were a significant improvement over the original SPL curve.

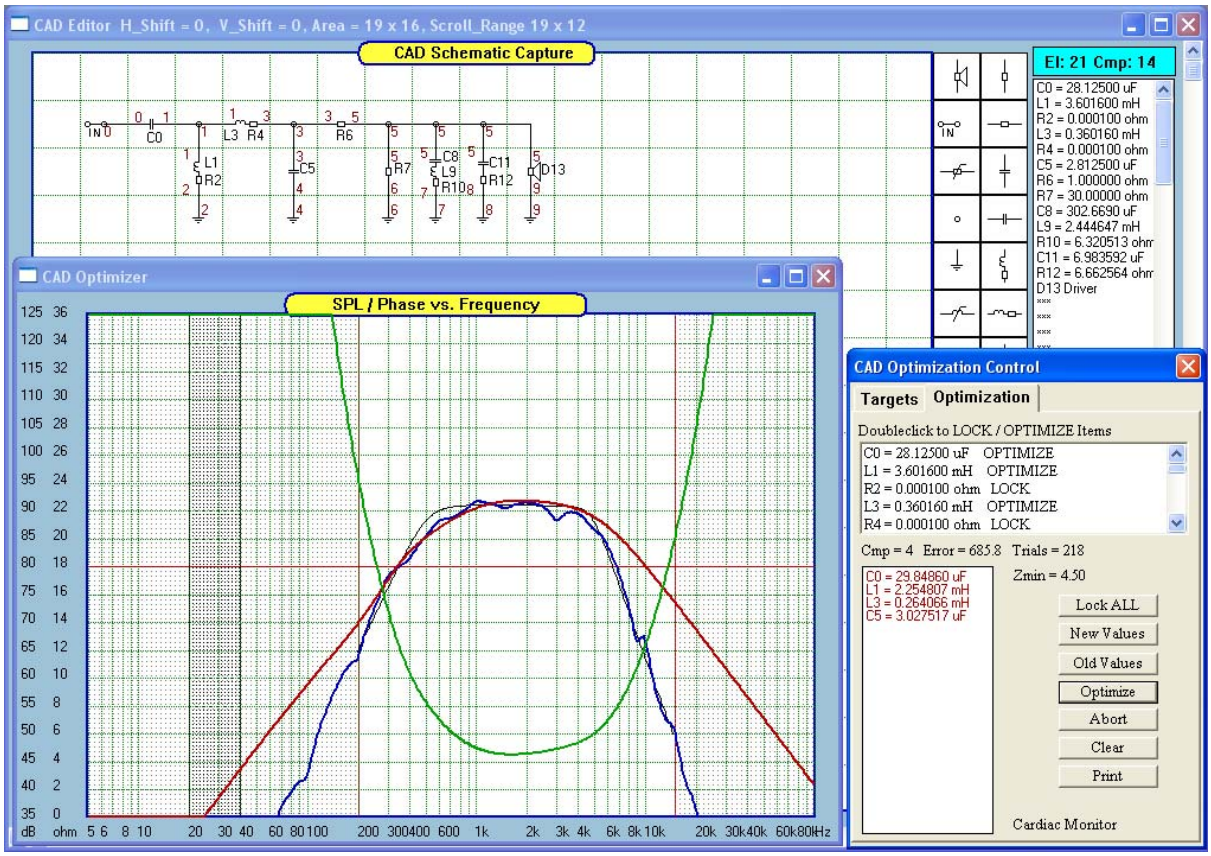


Figure 10.16. Results of constrained optimization – Error = 685.8.

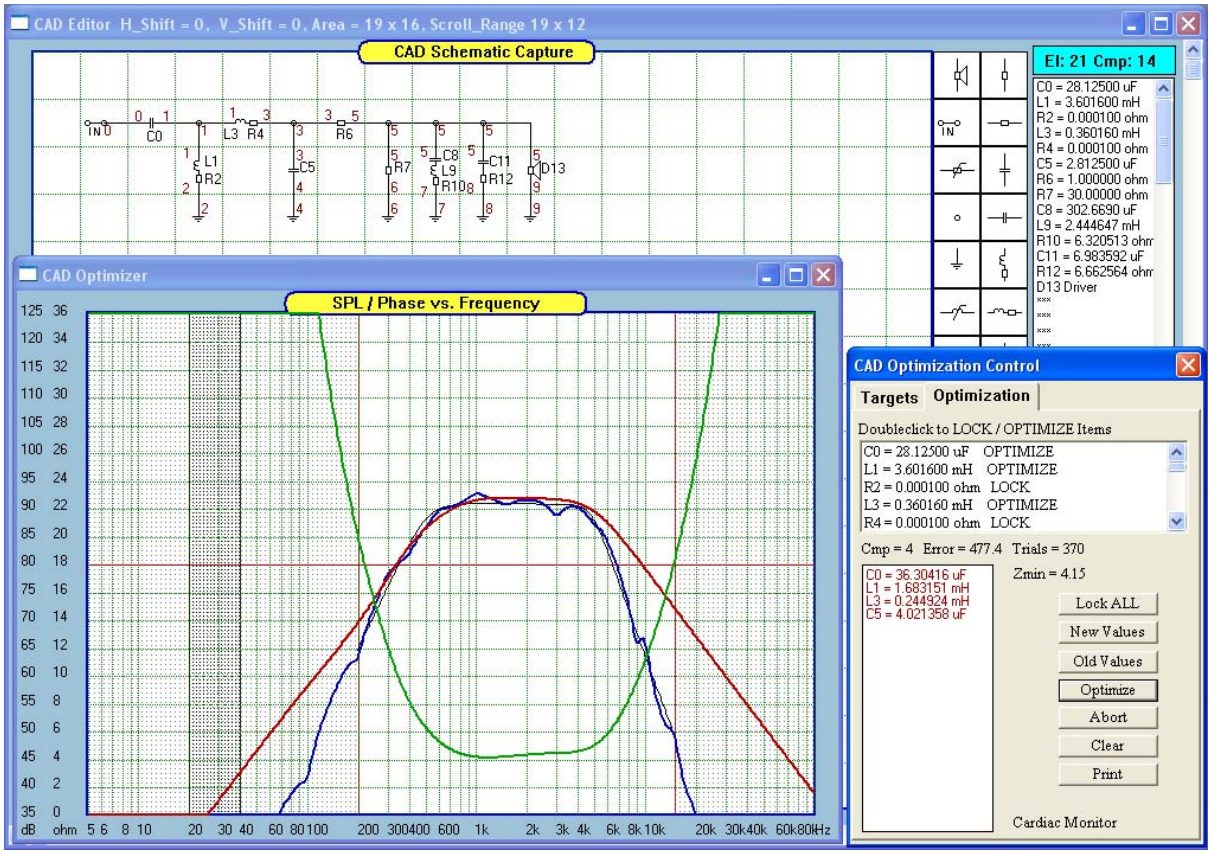


Figure 10.17. Results of unconstrained optimization – Error = 477.4.

User Defined Target Curves

When running the Optimizer process, you may at times desire to use pre-defined target SPL curves. SoundEasy provides you with a means to generate practically any target SPL curve using the Editor Screen. Any curve plotted on the Editor Screen can be used as the SPL target curve in the Optimizer process. At the end of the editing process, you will be able to save the newly created curve as ASCII file, containing all 750 data SPL points. This file can be further edited, if desired.

You may decide to create the entire target SPL curve using one of the available text editor programs as well, and then simply import it into SoundEasy Optimizer screen. In this case, we suggest you make your life easier by making an ASCII "template" of the target file. Simply start SoundEasy and select "Import/Export Data" -> "Export SPL/Phase at 0.0dB" while the program is in the default stage. This will create complete ".trg" file with 750 data points, showing the frequency column data and all SPL entries equal to 0. The frequency data points are already there for you, so this file can now be quickly edited for the desired SPL data using the external text editor.

In order to create the target SPL curve, we recommend you should start with the "Driver Parameter Editor" -> "Amplitude Model" tab, to start editing process for the target SPL. The process is identical to editing the SPL curve of a driver and has been described before. Figure 10.18 below shows the target SPL curve created using "Amplitude Model" tab (to get the initial shape of the curve). You can hand-draw (add) distortions after that.manually on the Editor Screen using the mouse pointer. This illustrates the possibility, that you can edit every single pixel on the Editor Screen.

The process of editing the target SPL curve is entirely graphical. For "fine tuning" the curve, you should "click" the left mouse button over the thick, pink SPL curve – this will activate small gray-colored screen cursor and then you can use cursor keys to move the screen cursor , at the same time re-drawing the SPL curve.

Finally, you must create Transfer Function from the edited template using HBT TAB – see Figure 10.19. This process will create Frequency – SPL – Phase columns in your target file. Now, the target file is ready to be used in the Optimizer process.

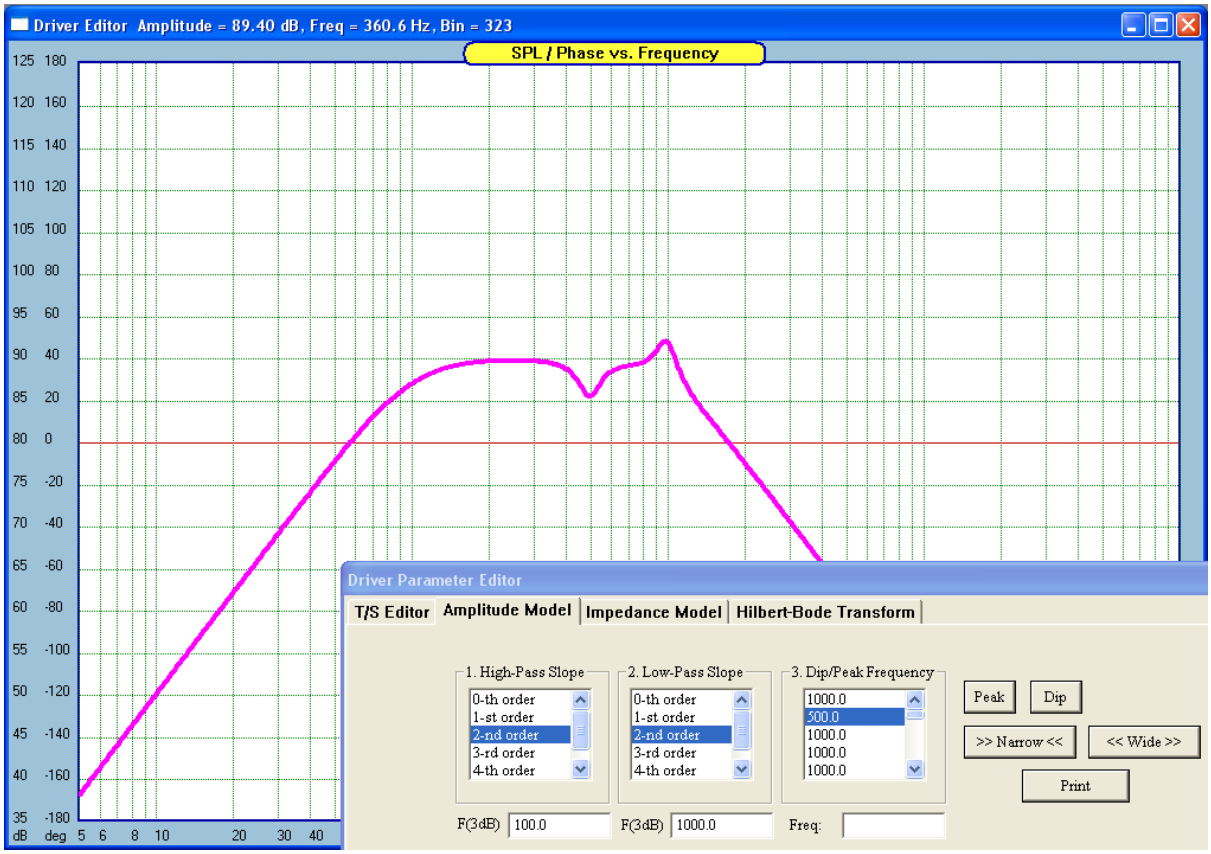


Figure 10.18. Editing SPL target curve using Editor Screen.

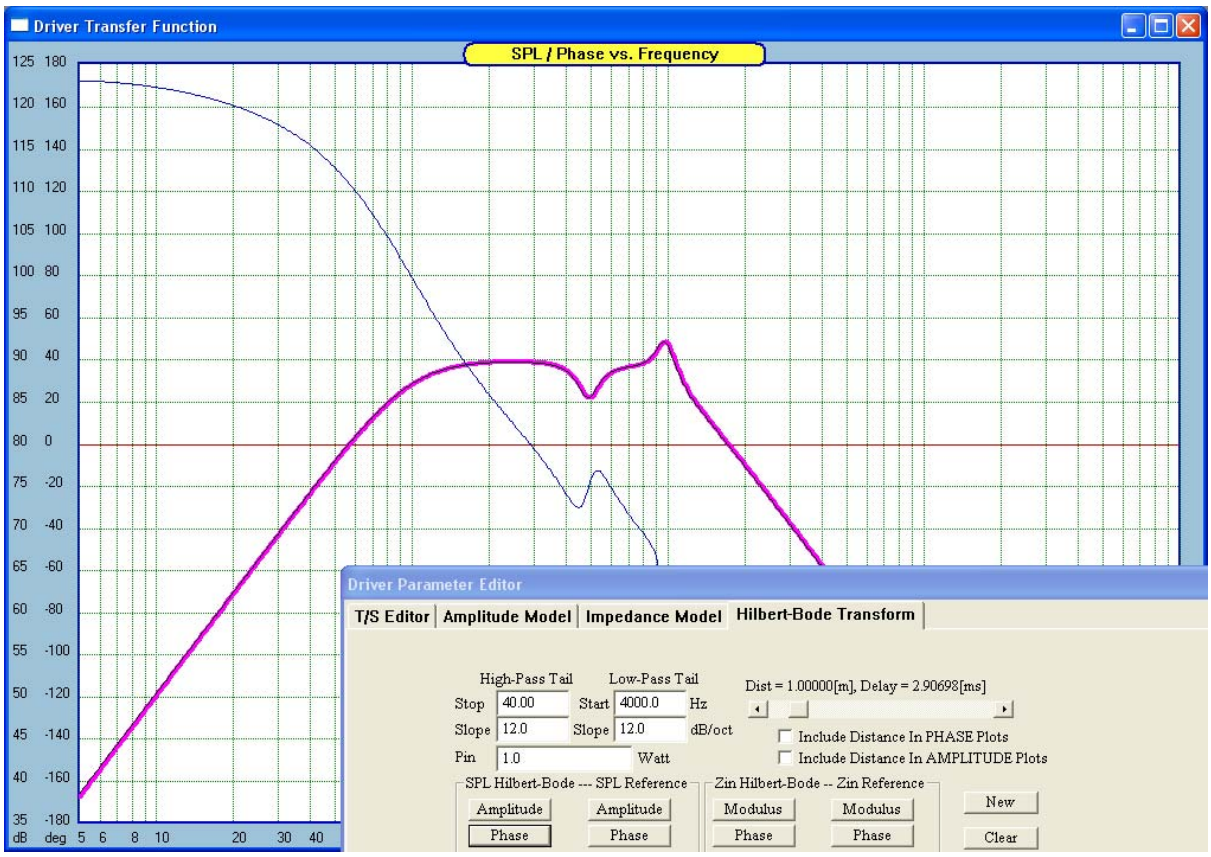


Figure 10.19. Converting edited SPL target curve to Transfer Function, using HBT function.

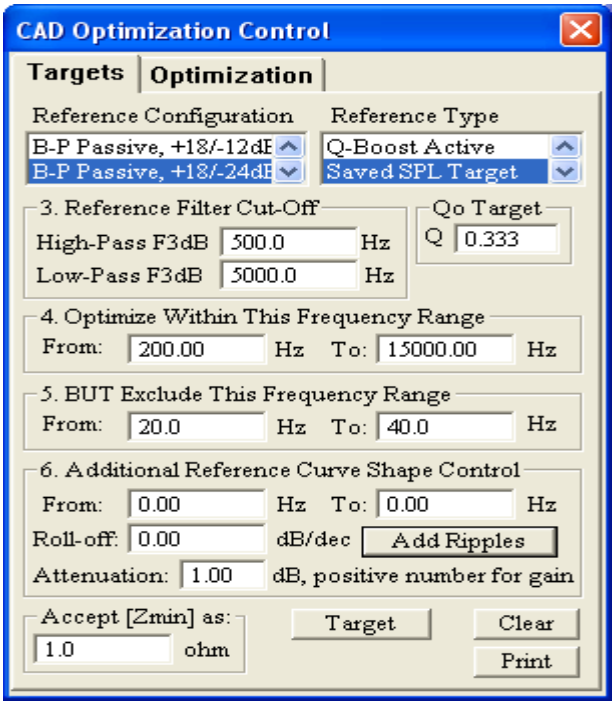


Figure 10.17. Selecting ".trg" file for target

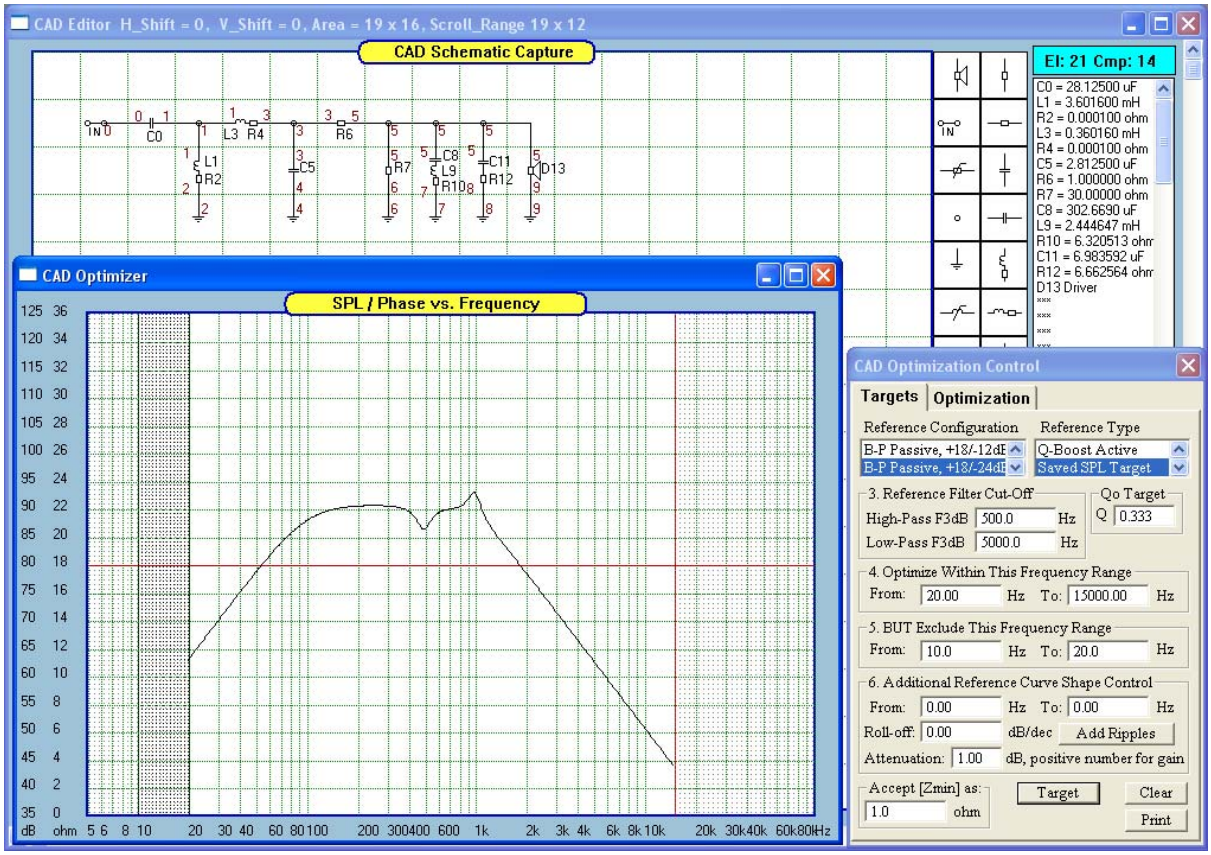


Figure 10.18. Target SPL curve

Conjugate Impedance Optimizer

Compensating driver's input impedance due to the semi-inductance of the voice coil, can be assisted by using Conjugate Impedance Optimizer. The Conjugate Impedance Optimizer function is primarily designed to work for Zobel networks, but you may be able to use it for other circuits as well. **The Conjugate Impedance Optimizer always attempts to adjust selected circuit elements, so that the input impedance into this circuit is a flat line (constant resistance), centered around Ztarget**.

Optimization results can be generally sensitive to selecting initial values of the components, optimization parameters and optimization frequency range.

The frequency range of optimization was selected to start above minimum impedance point for a "raw" impedance curve. In our example, it is deemed to be 300Hz. Frequencies below this point, should be EXCLUDED from optimization. The upper bound of the optimization frequency range should be the screen limit (100kHz in our example).

As for typical optimization requirements, you need to LOCK/OPTIMIZE appropriate components. In our example, both, C0, R1 are being optimized – see Figure xx.xx. To start Conjugate Impedance Optimizer function, please check the **Ztarget** box and press **"Optimize"** on Optimization TAB.

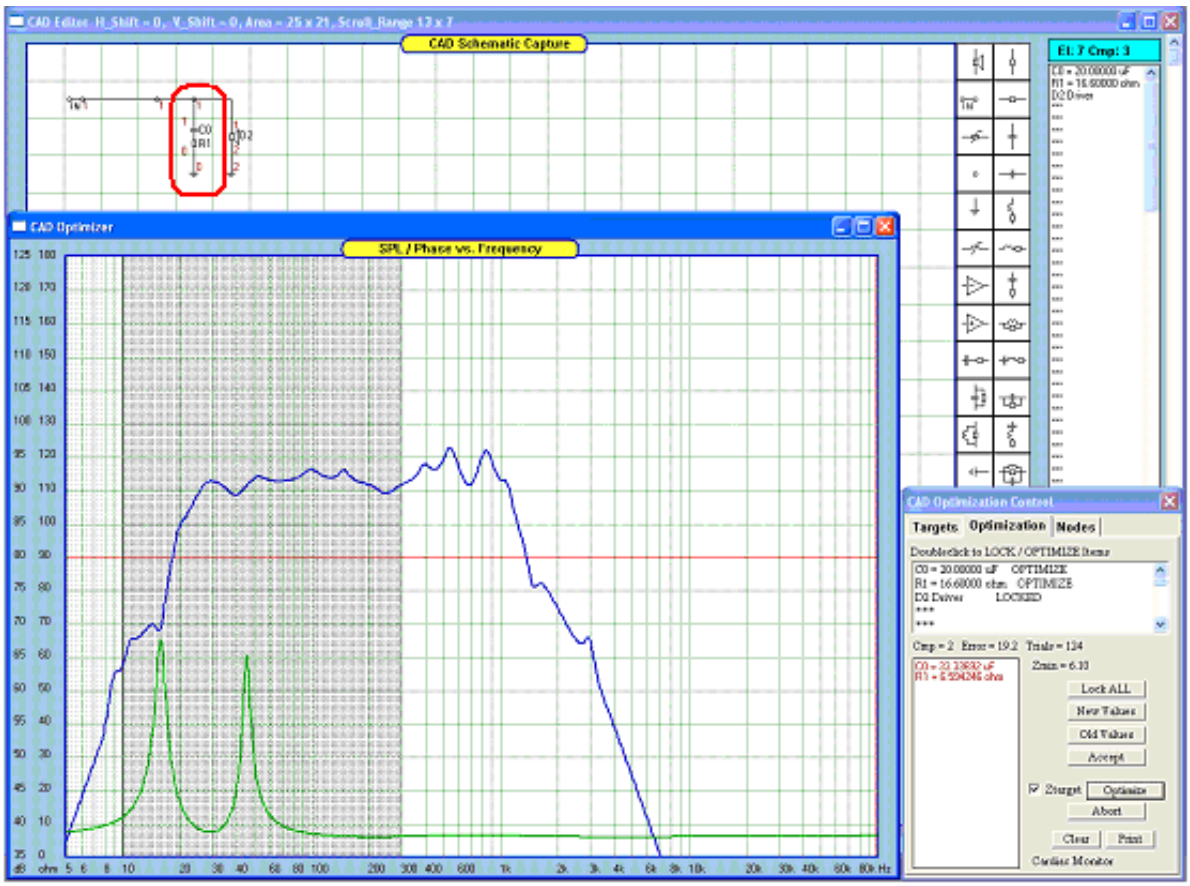


Figure 10.19 Basic schematic and highlighted Zobel network for input impedance compensation.

It is suggested, that for typical 12" woofer, with Re=6.0ohm, you may stat with initial Zobel values of R1 = 16ohm, C0 = 20uF – see Figure 10.19 above. **Zmin** (minimum allowed impedance) should be set just above the Re value, so it was set to 6.1ohm, and **"Apply"** box checked to activate it. **Ztarget** should be set above **Zmin**, so it was selected as 6.2ohm.

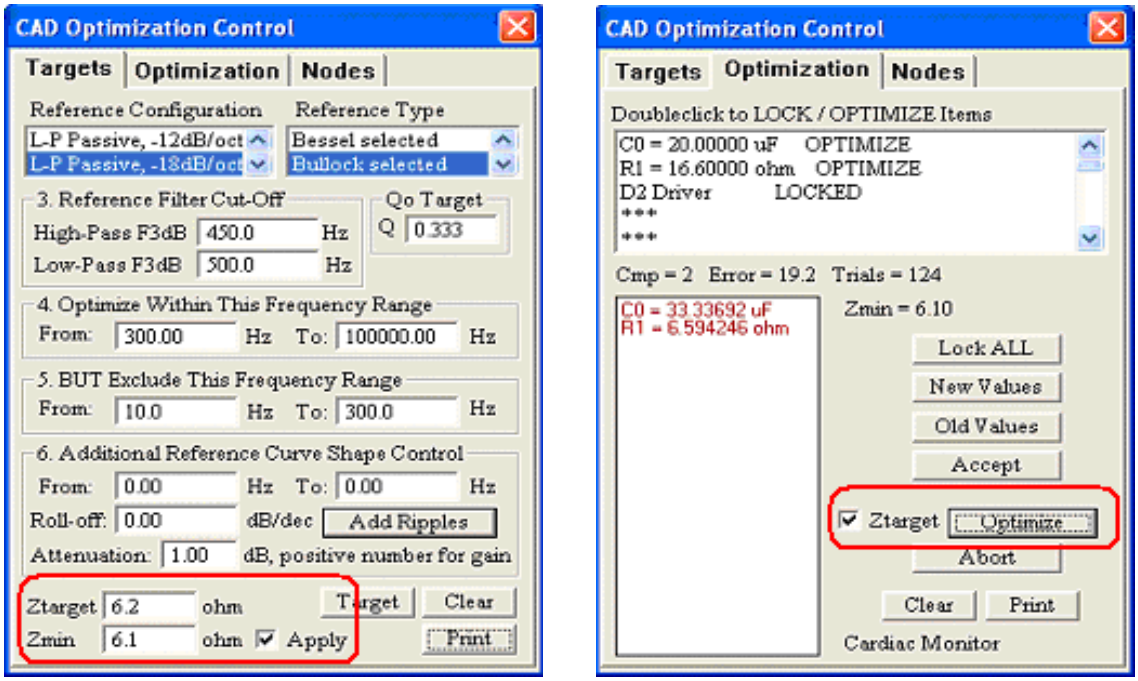


Figure 10.20 Location of controls for the Conjugate Impedance Optimizer.

Chapter 11. Crossover in Time Domain

Background

Over the past decade there has been growing recognition of the importance of loudspeaker transient performance. Accurate transient reproduction ability has been associated with phase response of the system. Voltage transients, when transformed into the frequency domain, are characterised by the presence of a rich array of frequency components simultaneously, which are related to each other with correct phase. When the phase relationship between these components is distorted during the reproduction, the resulting transients are also affected. Amplitude response alone is no longer sufficient to evaluate quality of the design and assure the user, that all forms of audible distortion have been compensated.

Rather than displaying phase response and drawing conclusions from it, a better test is being proposed. A step function, or square wave is being fed into the system and the time response to this excitation recorded. You may recall from Chapter 6, that square wave is composed from many individual sine waves, having specific phase relationship. It is obvious, that if the system under test, disturbs the existing phase relationship in the test signal, then the sine wave components will not add correctly after passing through the system and the resulting output will no longer resemble a square wave. A good point to start evaluating the transient response is the crossover. This is an area, where the designer has some degree of freedom in selecting the configuration and component values of the filters. The 'Time response' function module operates on the crossover section of the data file. However, it can be used as a "free standing" module for testing new ideas or circuit configurations - without saving the results of your work. Another area of concern is the difference in the acoustic path length from the acoustic centre of each driver to the listener. This problem can be studied in details using the "System" tools.

There are several methods being used in analysing electrical circuits in time. The 'Crossover' tool offers two methods: (1) Fast Fourier Transform (FFT) and (2) Matrix Nodal Method. For the first method, the basic idea of "How it is done" was explained in the previous chapter, so it is sufficient to state, that "Crossover" tool also uses Convolution in Time method to obtain time response of the analysed circuit. The Nodal Method will be explained in more details in the remainder of this chapter. Evaluating channel response in frequency domain only, offers the designer only a limited insight into the quality of the design. As it will be shown later, crossover filters, which display only a small ripple in the summing region, do not add voltages correctly in the time domain. As a matter, of fact, most passive crossovers don't. Those which do, are called "Constant Voltage" crossovers (Small, JAES, January, 1971, Vol 19, Number 1). The analysis can be performed in two modes explained below. Extensive menu offers the user several options for plotting and saving the curves in the provided buffers. Time Domain analysis window can be opened from the main menu.

Time Response Menu

There are several options available to the user from the floating menu of the FFT screen. The options are explained below:

1. **Standard Mode -** to be used with the files or circuits, that include drivers and their frequency responses.

   * Calculates the network transfer function for FFT fundamental frequency (selected from the editable window - Test Frequency) and harmonics up to 40kHz.
   * Calculates FFT of the square wave.
   * Calculates the product of the above (Convolution).
   * Calculates IFFT and plots the result on the screen (time response).
   * The curve is saved in the buffer.

2. **Extended Mode -** to be used with the files or circuits, that DO NOT include drivers and their frequency responses.
   * Calculates the network transfer function for ALL FFT harmonics.
   * Calculates FFT of the square wave.
   * Calculates the product of the above (Convolution).
   * Calculates IFFT and plots the result on the screen (time response).
   * Saves the curve in the buffer in the first available location.

Time Tab

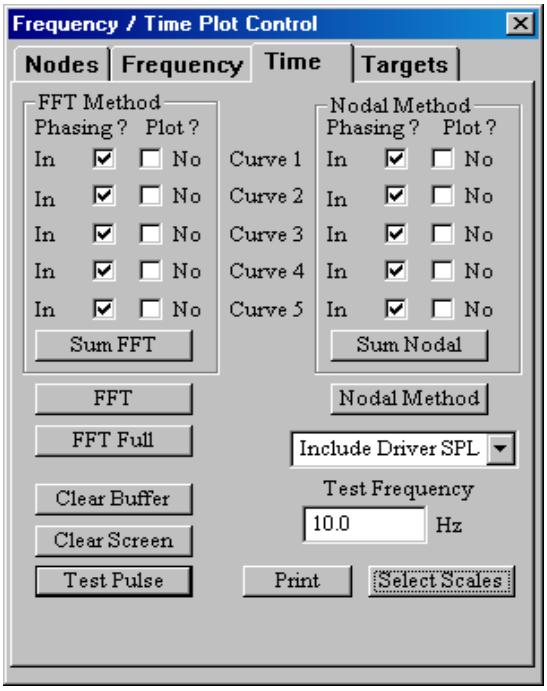


Figure 11.1. Tab for controlling Time-Domain plots

1. **FFT Method combobox –**
2. **Phasing ?** check-box column – "In" phase indicates no phase reversal. "Out" of phase indicates
3. phase reversed by 180 degrees.
4. **Plot ?** check-box column – "No" indicates do no plot this curve. "Yes" indicates inclusion of the curve in plots.
5. **Sum FFT button –** Plots summed response using FFT method
6. **Nodal Method combobox –**
7. **Phasing ?** check-box column – "In" phase indicates no phase reversal. "Out" of phase indicates phase reversed by 180 degrees.
8. **Plot ?** check-box column – "No" indicates do no plot this curve. "Yes" indicates inclusion of the curve in plots.
9. **Sum Nodal button** – Plots summed response using MNM method
10. **FFT button** - Plots summed response using FFT method. You **can include driver icon in the CAD** schematic when plotting time domain response using this method.
11. **FFT Full button** - Plots summed response using FFT method. You **<u>must not</u> include driver icon in the CAD** schematic when plotting time domain response using this method.
12. **Clear Buffer button** – Clears memory storage allocated for 5 curves.
13. **Clear Screen button** – Simply clears the screen.
14. **Test Pulse button** – Plots the test pulse.
15. **Test Frequency editable field** – Enter your test frequency here.
16. **Drop-down list** box for selecting additional plotting parameters.
17. **Print button** – Invokes printer dialogue box.
18. **Select Scales** – Button opens dialogue box for selecting vertical scale resolution.

Standard Mode vs. Extended Frequency Mode

Individual time response of every frequency channel may reveal additional useful information. Area of particular importance however, is the "crossover" range of frequencies. This module adds up to five time responses, in- or out- of phase, stored previously in Standard or Extended Frequency Mode. The Standard Mode should be used when plotting time response of any circuit developed with the help of other SoundEasy modules. As it was explained in Chapter 1, the program operates within 1 - 100000Hz frequency range. You may still decide to use Extended Frequency Mode, but the plotted time responses will have reduced accuracy for higher sampling frequencies (frequency of the square wave) - see Fig 11.3. For example: let's assume, that sampling frequency was selected as 20 kHz.

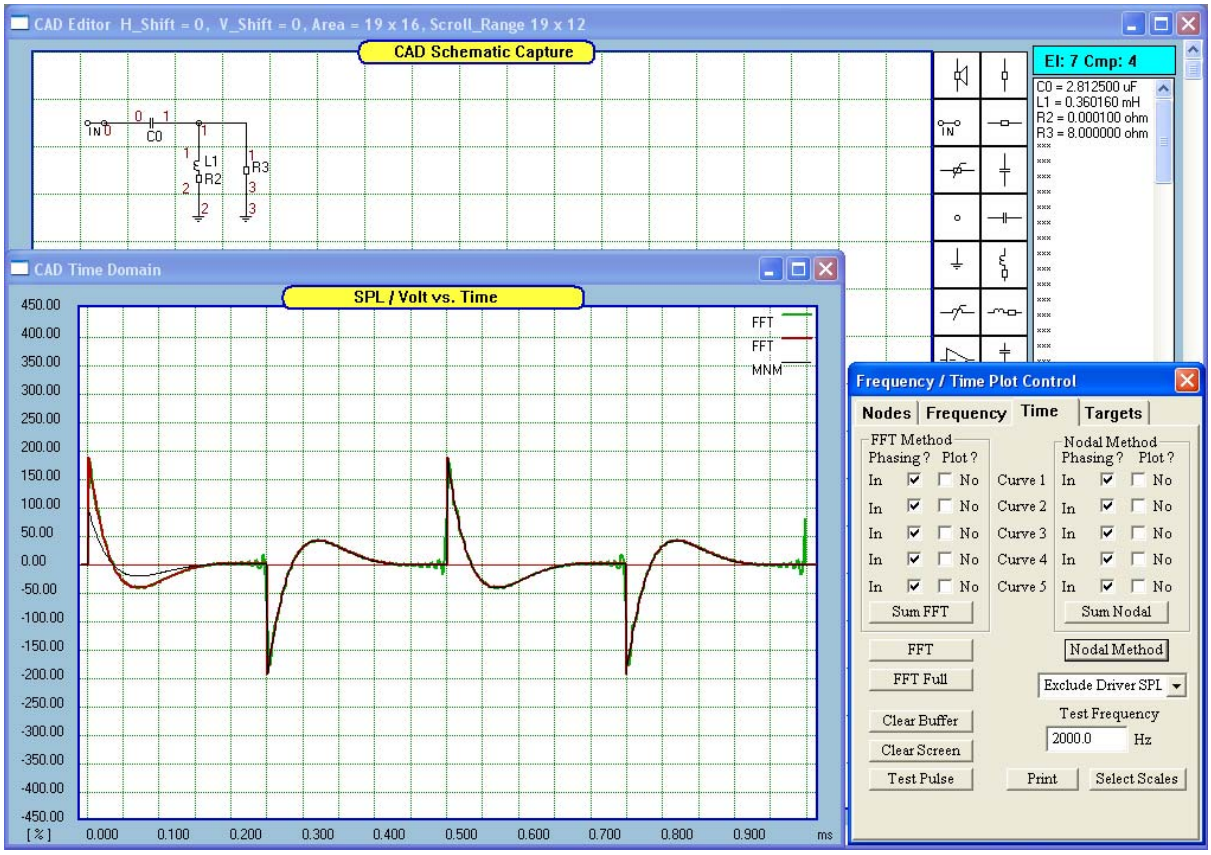


Fig 11.2 Three options for obtaining time plots: FFT – green, Full FFT – brown, MNM black.

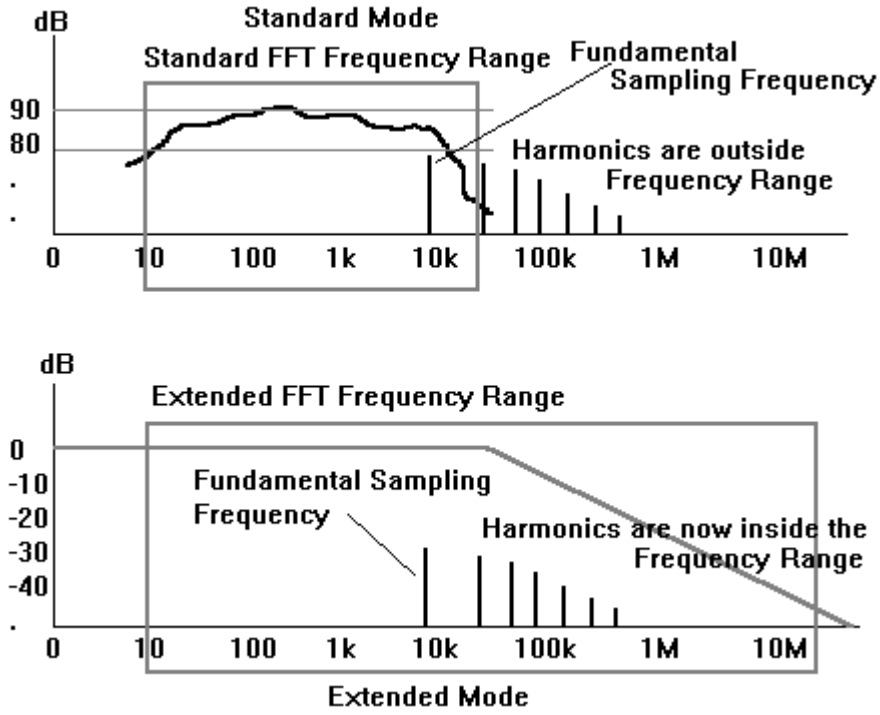


Fig 11.3

The first odd, non-zero harmonic (F2) of the square wave will be 60 kHz. The next non-zero harmonic (F3) will be 100kHz and so on. All the harmonics, F2, F3 and higher fall outside the 10 - 40000Hz frequency range where the frequency response of the proposed circuit is not calculated (it is actually assumed to be zero). As a result of this, the higher harmonics of the FFT process can not contribute to the final result of the Convolution in Frequency and the error is produced. The size of the error can be estimated from the Fig 11.2. The figure shows two time response plots of the same circuit. The curve labelled "0" was plotted in Standard Mode and it can be observed, that the curve has a few "waves" on it. The result of missing harmonics was explained in the previous chapter, when the process of building a perfect square wave was explored. The second curve, labelled "1" was obtained in the Extended Mode. The Standard Mode will yield good results below 1 kHz sampling frequency and will offer reasonable approximation up to 5 - 6 kHz. This should be sufficient for most of the midrange-tweeter crossover points used in three-way systems. In the Extended Mode, SoundEasy is able to calculate and add all 1024 (or rather 512 odd) harmonics and produce accurate results. This mode is recommended for all study and analysis of CAD circuits (but remember, the circuit must not contain loudspeaker symbol or "Include Driver" mode).

Nodal Method - How it is done ?.

The circuit you have designed is analysed by Nodal Method using complex number matrixes. In the frequency or time domain the process broadly follows the steps described below.

1. The designer creates the circuit using CAD methods described in the previous chapters.
2. All necessary information about the circuit is stored in a large matrix, which will later be used as a circuit reference.
3. Based on (2), incidence matrix Aa[][], relating to passive elements and Ab[][], relating to active elements are created.
4. Datum node (rows corresponding to ground connections) are then removed from the Aa[][] matrix.
5. The Aa[][] matrix is then transposed to $Aa^T[][]$ and Ab[][] is transposed to $Ab^T[][]$.
6. Complex branch conductance matrix Ga[][] for passive components is formulated next.
7. Complex branch conductance matrix Gb[][] and branch resistance matrix Rb[][] for active components are formulated next.
8. Independent complex branch current vector J[] is formulated.
9. Matrix Rb[][] is inverted to $Rb^{-1}[][]$.
10. Complex nodal conductance matrix Gn[][] = Ga[][] - Ab[][]* $Rb^{-1}[][]$*Gb[][]* $Ab^T[][]$.
11. Now, algorithm enters the main loop and calculates circuit response for individual frequencies or time steps.
11.1 Nodal current source vector Jn[]=A[][]*J[].
11.2 Nodal voltage vector Vn[]=-Jn[]/Gn[][]. At this stage, inversion of matrix Gn[][] is performed.
11.3 Branch voltage vector V[]= $A^T[][]$*Vn[].
11.4 Passive branch current vector Ia[]=Ga[][]* $Aa^T[][]$*Vn[].
11.5 In the time domain, the algorithm now solves first order differential equations for simple components and integro-differential equations for compound (LRC) components. Second-order Runge-Kutta method was implemented.

In the time domain, the algorithm advances a small fraction of a second at a time. This time length is calculated from the formula: **Time_step = 1/(Test Frequency * 5840)**

The "Test Frequency" can be entered from the provided data field on the plotting screen. When testing circuit with very small time constants (RC, RL elements), it is important to use small time steps (or high Test Frequency) otherwise, the circuit response will decay before the next time increment.
Time constant for RC circuits: t = R*C, Time constant for RL circuits: t = L/R.

Fig 11.4 and 11.5 show a couple of simple circuits and their time responses plotted using the FFT method and Nodal Method. The user will notice a good agreement between both methods. Occasionally, the two methods will produce different results, particularly, when the frequency response has sharp, high amplitude peaks. This would require much smaller time steps used by Nodal Method. User editable time steps will be provided in future releases of SoundEasy. The user should be aware, that there is no single best method of analysing circuits in time domain.

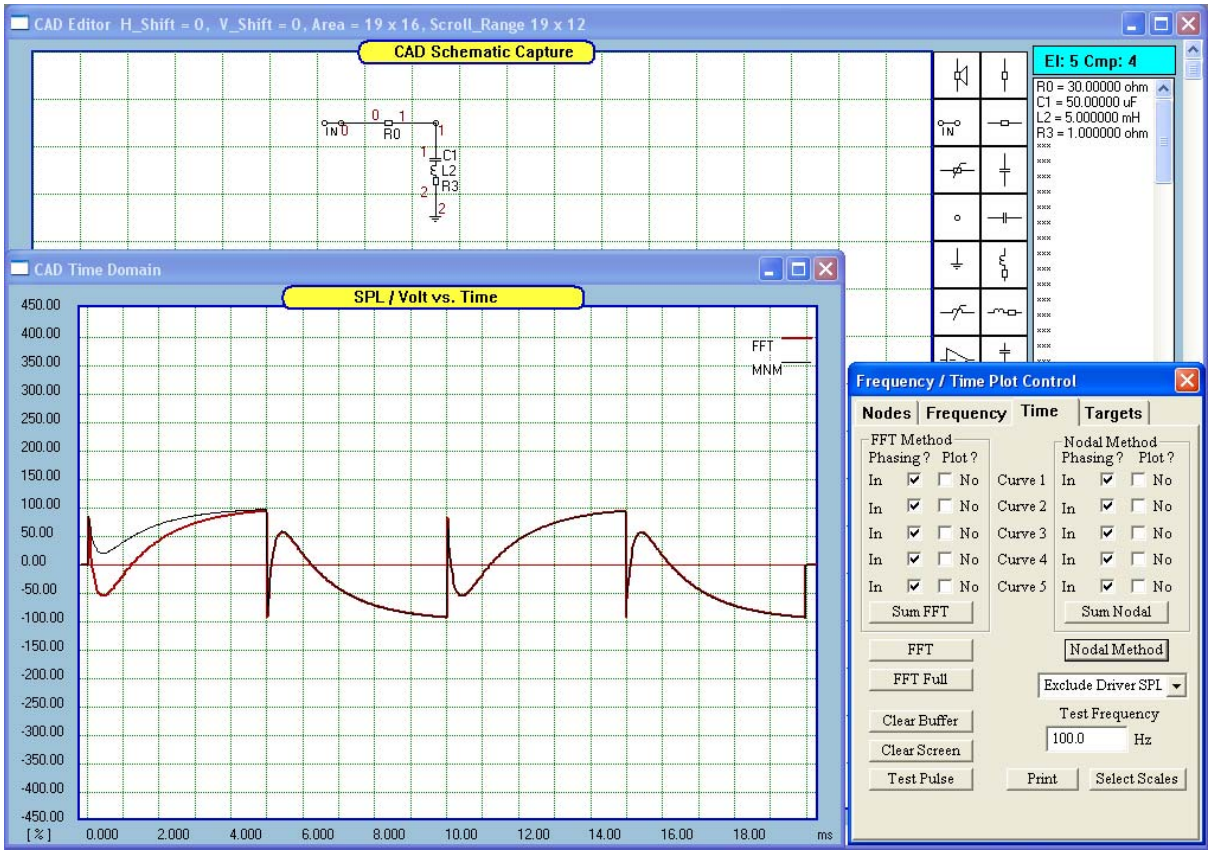


Fig 11.4 Example 1 comparing FFT (brown) and MNM (black).

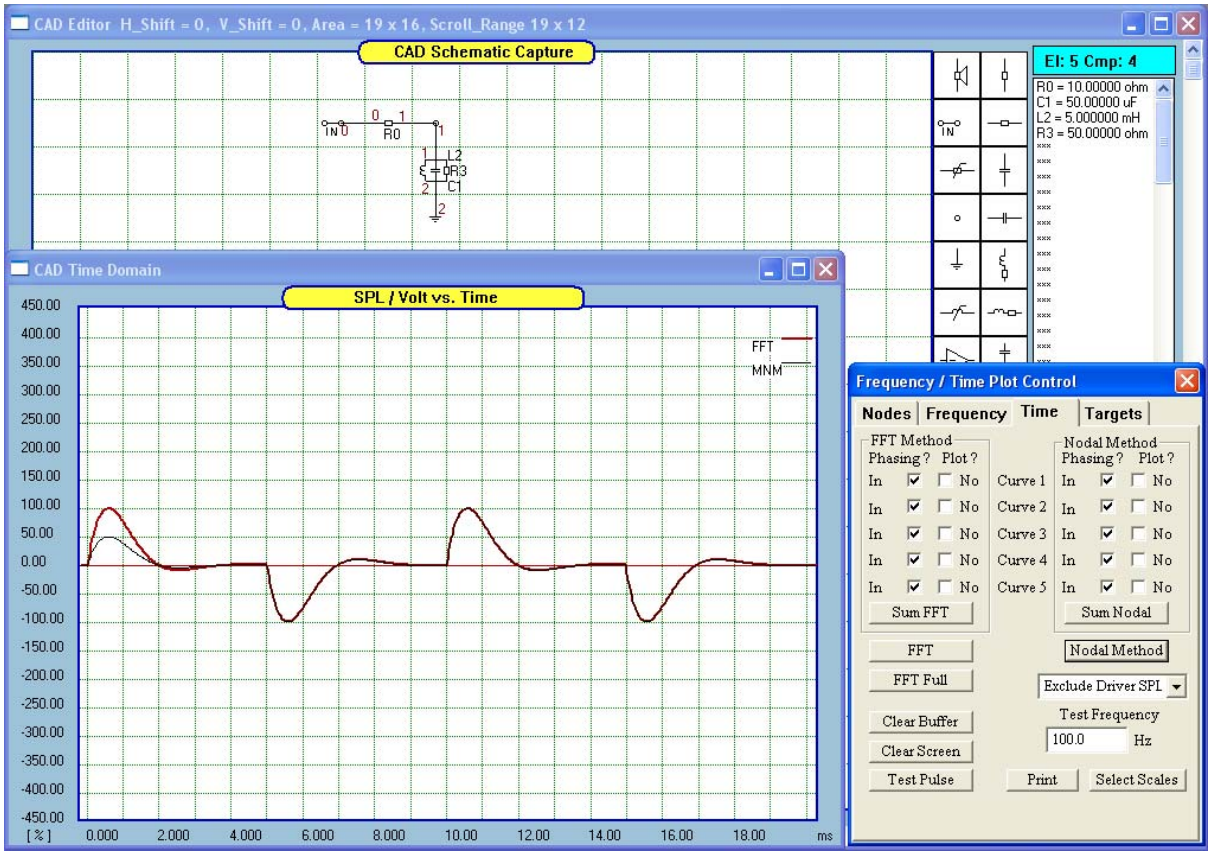


Fig 11.5 Example 2 comparing FFT and MNM

Adding several time responses

The time response function contains a buffer for five time response curves. The user may elect to plot the time response with or without the save option. The saved plots are labelled [SAVED] on the legend. In order to add the curves correctly, all saved plots must have been plotted using the same sampling frequency. Summation process may be performed with the plots being added in phase or out-of-phase. The user will be prompted to select the required configuration from a dialogue box prior the actual summed response being plotted - see Fig 11.1. When the buffer is full, eg: five curves have been saved, it will not accept any more data. To empty (clear) the buffer the user is required to select the "Clear Buffer" option from the floating menu, activated by the "Plot" button.

Concept of "fill-in" driver

Analysis in the time domain usually reveals additional limitations of passive crossovers. In order to overcome these limitations the concept of "filler driver" has been advocated by Bekgaard of Bang & Olufsen.

Fig 11.6 shows frequency responses of o tree-way Butterworth crossover and Fig 11.7 shows time response of the same crossover. It is observable, that the summed time response does not even resemble the expected square wave. This crossover is not a "constant voltage" crossover. The user is encouraged to review several other types of crossovers for the benefit of learning more about their advantages and disadvantages.

One interesting case is depicted on Fig 11.8. The figure shows time responses of a first-order high-pass and low-pass crossovers and their summed response. It is observable, that this crossover is a constant voltage crossover and the resulting summed time response is a perfect square wave. The first order crossover is advocated by many designers as "the best there is" for this reason alone. However, loudspeaker drivers used with the 1-st order crossover must be of highest quality and reliability. They must have high power handling capabilities and be tolerant to out-of-band signals. A selection of drivers meeting these requirements is manufactured by Dynaudio. The "fill-in" driver concept revolves around inserting in an extra driver together with its crossover network - the filler driver. The new combined crossover becomes "constant voltage" crossover.

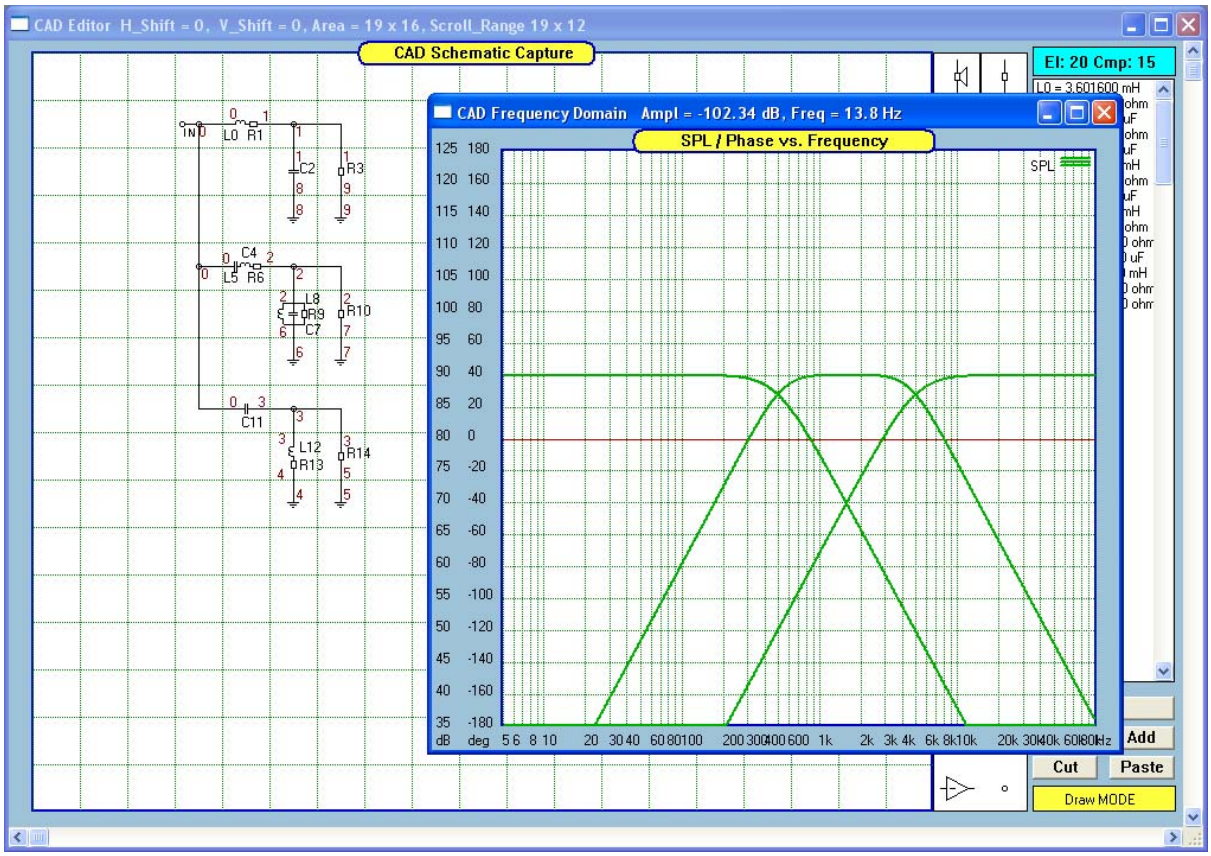


Fig 11.6 Shows frequency responses of o tree-way Butterworth crossover

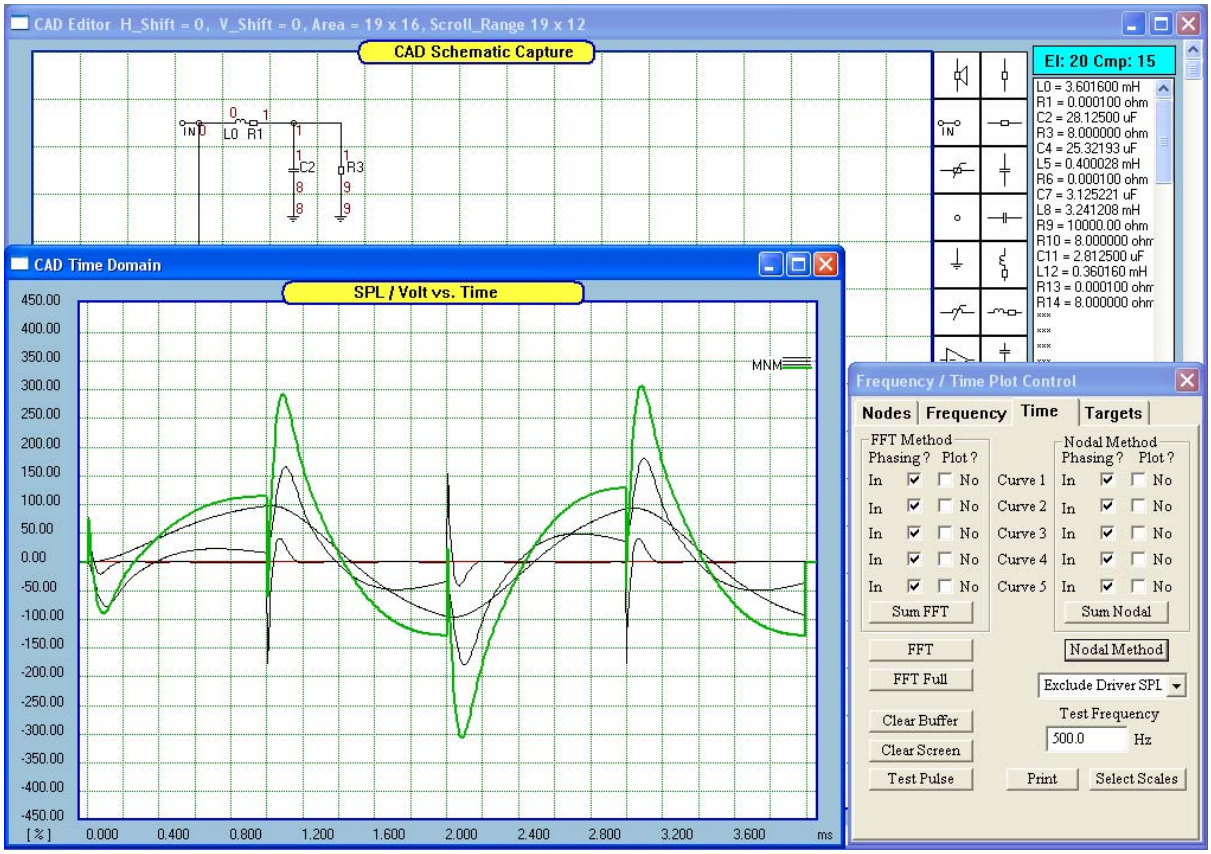


Fig 11.7 Time response of o tree-way Butterworth crossover (Figure 11.6) at 500Hz

Please review the following figures:

1. Fig 11.8
   * Curve 0 - Woofer **second order** low-pass filter frequency response.
   * Curve 1 - Tweeter **second order** high-pass filter frequency response.
   * Curve 2 - Filler driver - **first order** band-pass filter frequency response.

2. Fig 11.9
   * Curve 0 - Woofer time response @ 1000 Hz.
   * Curve 1 - Tweeter time response @ 1000 Hz.
   * Curve 2 - Filler driver time response @ 1000 Hz.
   * Curve 3 - Summed response - a perfect square wave.
     All time plots were added in-phase.

For exact formulas and more technical information about this concept, the user is referred to the original work of Bakegaard (JAS, May 1977, Volume 25, No 5).

You may wish to continue experiments in the following areas:

1. Filler driver connected via +12/-12 dB/oct and higher order band pass filters in 2-way crossover. This research may be beneficial if the filler driver is required to work in higher power systems.

2. Applicability of the filler driver in 3-way and 4-way crossovers of various orders.

3. Time delay filters (Lattice Networks).

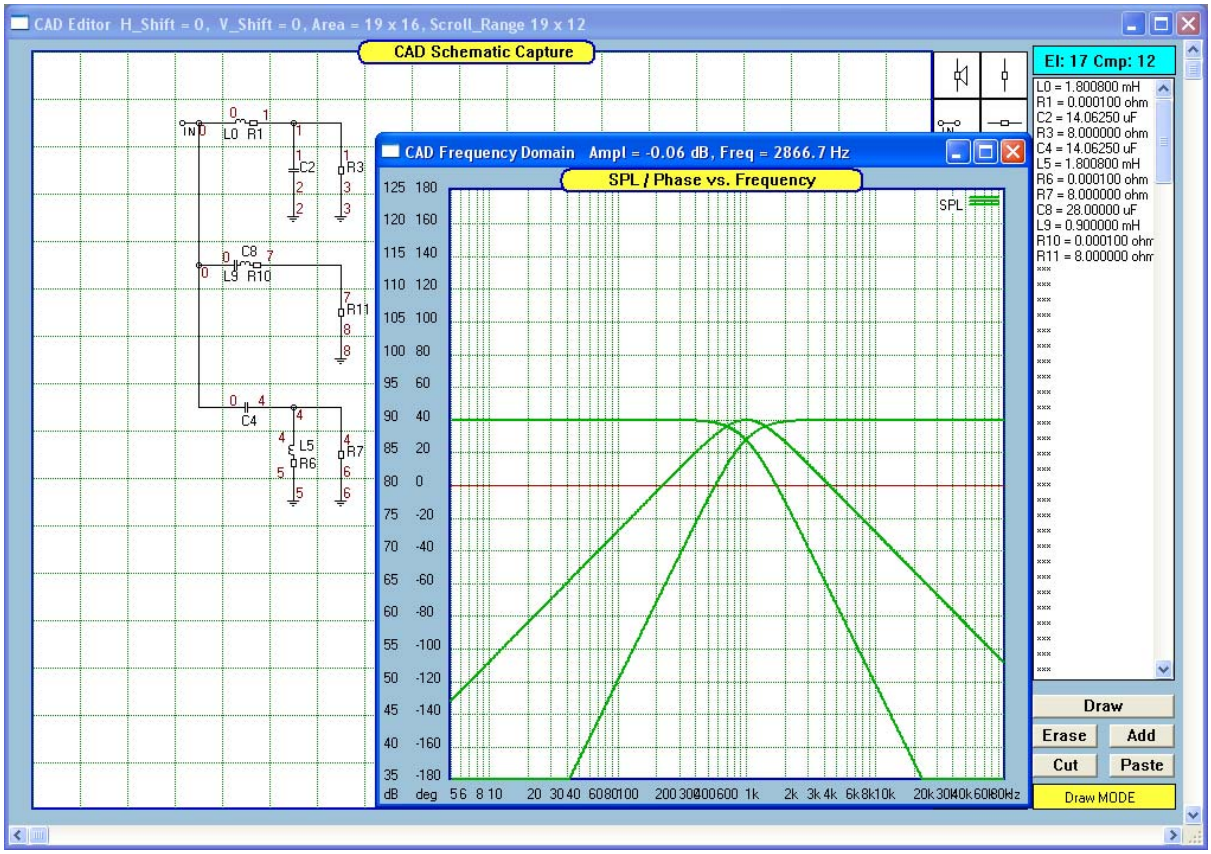


Fig 11.8 Three-way crossover with filler driver

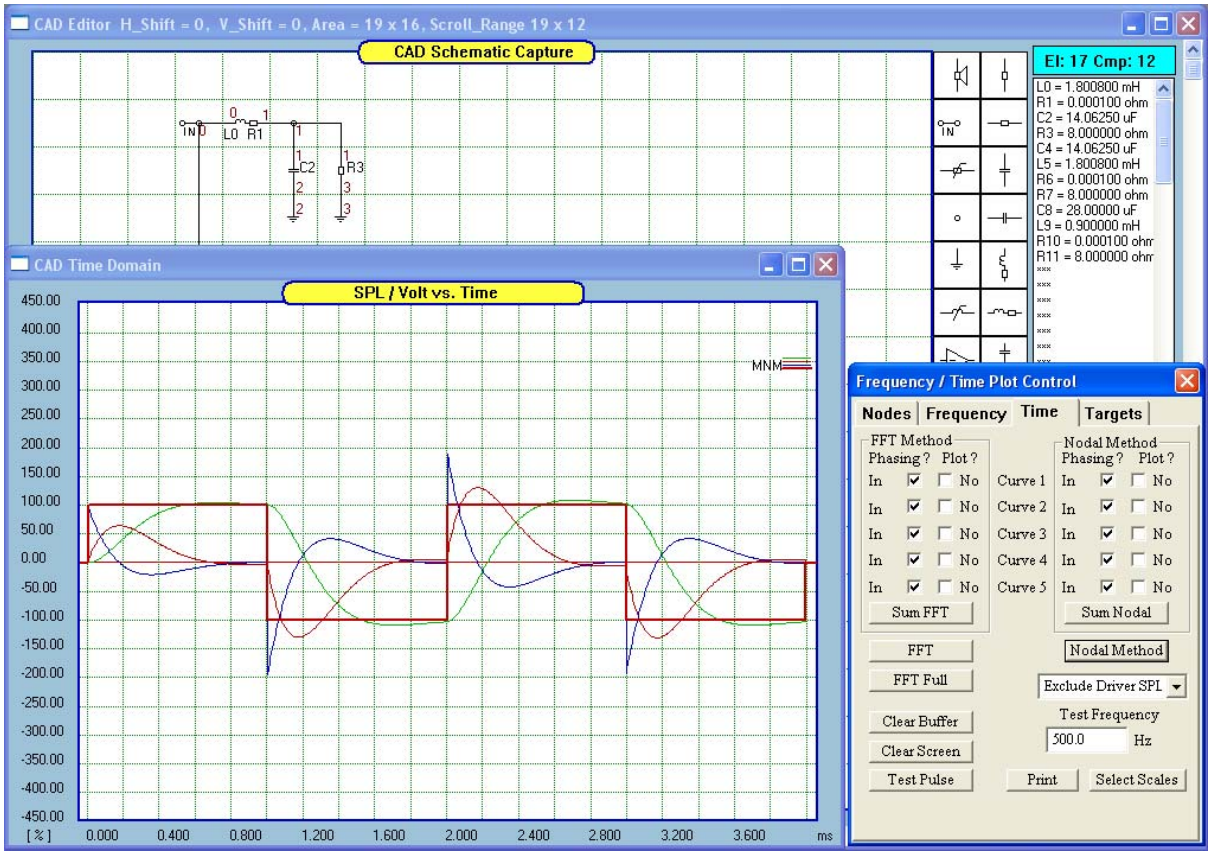


Fig 11.9 Curve 3 - Summed response - a perfect square wave (all drivers in-phase).

Part 4

Working With System Screen

Chapter 12. System Response

Putting It All Together

So far, the designer using SoundEasy tools, has been focused on the performance of the individual components of the system under development. The following issues have been covered:

1. A library of fully edited drivers has been created on a mass storage media.
2. Individual drivers have been selected for the system.
3. Enclosures have been designed.
4. The input impedances of the drivers have been compensated.
5. The efficiency of the higher frequency drivers have been compensated (where necessary).
6. The frequency response of the drivers have been analysed and equalised (where necessary).
7. All other circuitry (eg. time delay) have been added.
8. The individual crossover filters have been designed.

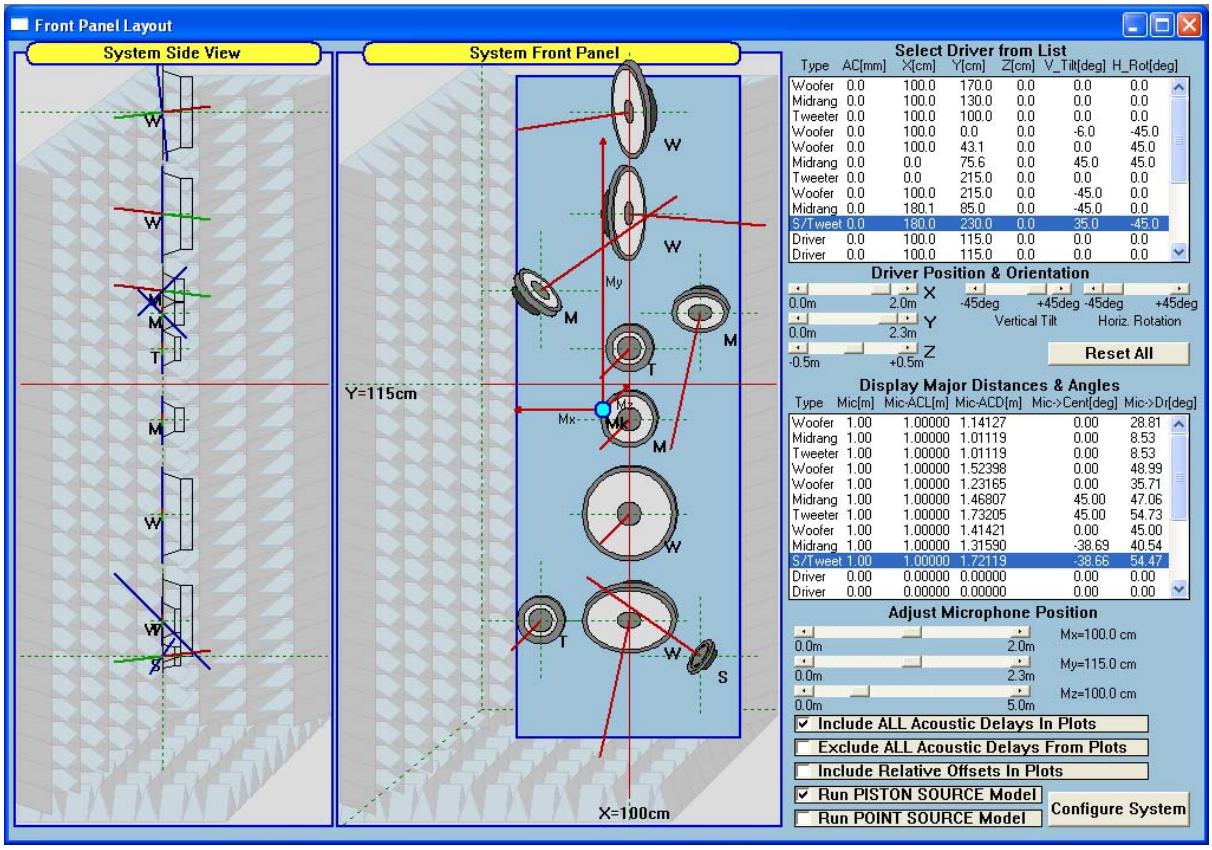


Fig 12.1 System Module

Main "Front Panel" Screen

The "Front Panel Layout" screen enables you to place (in 3D), tilt and rotate loudspeakers within 2m x 2.3m "imaginary" baffle. Diffraction effects are not modeled on this baffle, only driver's SPL curve, with the exception of off-axis simulations on piston model. Also, this screen allows you to position your test microphone. Finally, you may decide to select one of the two built-in models for the driver, or check the response without acoustic delays. Driver diameter is necessary for the models to work correctly and it needs to be entered from the "**Driver Parameter Editor**" tab. The get the driver displayed on this screen, you must turn it ON first, by selecting "**Configure System**" button. This will open the tabbed box with "Node" tab for turning ON/OFF any driver in the system.

There is a good chance, that you have developed the crossover first, and the drivers will be switched ON already. Perhaps the easiest way to work with this screen is to simply follow the controls from top-to-bottom.

1. **Select Driver From List** – click on the driver listed in this box to gain access to its location parameters: X, Y, Z, position and Vertical Tilt and Horizontal Rotation.
2. **X slider** – position the driver horizontally (0 – 200cm).
3. **Y slider** – position the driver vertically (0 – 230cm).
4. **Z slider** – position the driver baffle offset (0 – 500cm).
5. **Vertical Tilt** – tilts the driver UP/DOWN
6. **Horiz Rotation** – rotates the driver around.
7. **Reset All** – resets all driver locations the the default: X=100cm, Y=115cm, Z=0.0cm, Vertical tilt=0.0deg and horizontal rotation = 0.0deg.

The next list box located just below the controls is for display purposes only. The columns show the following information:

1. Mic – the Z-distance from test microphone to the baffle in meters.
2. Mic ACL –Total distance from the test microphone to the baffle in meters.
3. Mic ACD – Total distance from the test microphone to the driver's acoustic center in meters.
4. Mic->Cent - Vertical angle between the microphone and the center of the baffle in degrees.
5. Mic-> Dr – Vertical angle between the microphone and the driver in degrees.

Microphone location is adjustable by moving the **Mx, My, Mz sliders**. The "**Include ALL acoustic Delays", "Exclude ALL acoustic Delays" and "Include AC + Z-offsets in Plots**" check-boxes are really self explanatory.

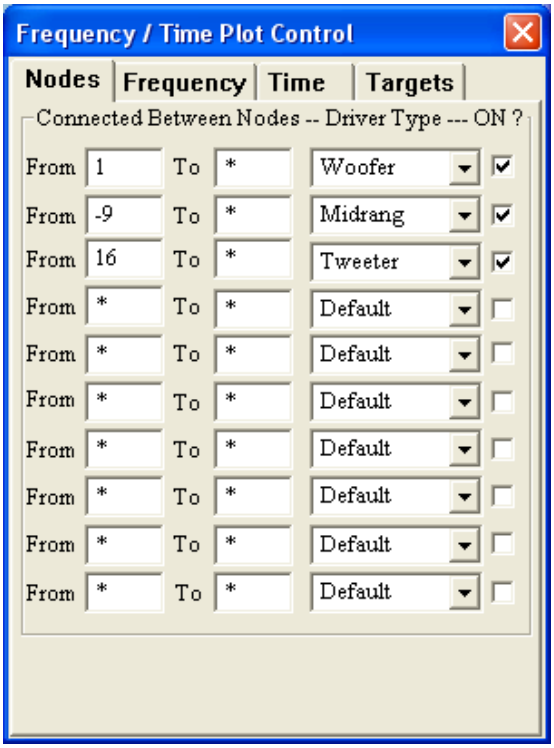


Figure 12.2 "Configure System" box

The "**To**" column of nodes would be used, if the driver was connected between two nodes AND none of them would be the "ground" node. For example, "series" crossovers have at least one driver connected to two floating nodes. Driver type will be displayed in the "**Driver Type**" column. Finally, you can switch ON/OFF any driver from the analysis by checking/un-checking the boxes in the "**ON?"** column – see Figure 12.2.

You must enter one (for single ended loads, second Nx enter as "*") or both node numbers (for floating loads) and **numbers must be different from any ground node number**.

"Configure System" – button which opens a control box allowing you to switch ON/OFF drivers in the system. SoundEasy software provides you with a high degree of flexibility in configuring the system. All 10 drivers can be switched ON or OFF the system. Also, when the driver is switched ON, it may be connected in-phase or out-of-phase with other drivers. The above process is accomplished via a dialogue box, invoked using the "Configure System" button. The phasing of the system is usually referenced to the woofer and it is preferred to leave the woofer in the "In-phase" state. You can then connect all other drivers (their crossovers) accordingly to your system configuration. The drivers switched ON must be loaded into the program data space. Do not attempt to switch ON a driver that was not loaded before. You may however switch OFF any driver from your system for the purpose of "what-if" analysis.

Tilting and Rotating Drivers

Tilting or rotating drivers can be used to simulate enclosures with tilted or non-flat front baffles. Drivers mounted on "sloppy" baffles are really operating "off-axis" when tested with microphone suspended on the same height as the driver. Typically, the driver is facing slightly "upwards". You can observe the resulting SPL on Figure 12.4 below. Even though the microphone AND the driver are located at the same height (Y-distance) the resulting SPL has all the characteristics of "off-axis" measurement – notches and SPL reduction. Now, you have two options to go back to the "on-axis" measurement conditions:

1. Tilt the driver back to 0.0deg tilt position.
2. Move the microphone up, in the direction of the main axis of the driver. This is shown on Figure 12.5, and as you can see, the frequency response is fully restored, except for a slight reduction in overall level due to somewhat increased mean distance between the microphone and the driver.

In this example, we are showing only one driver for clarity reasons. You would use the same reasoning for a complete system.

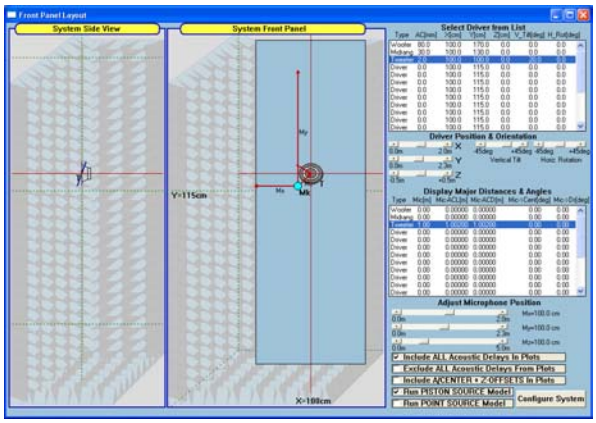


Figure 12.3 Tweeter driver tilted up by 20 deg.

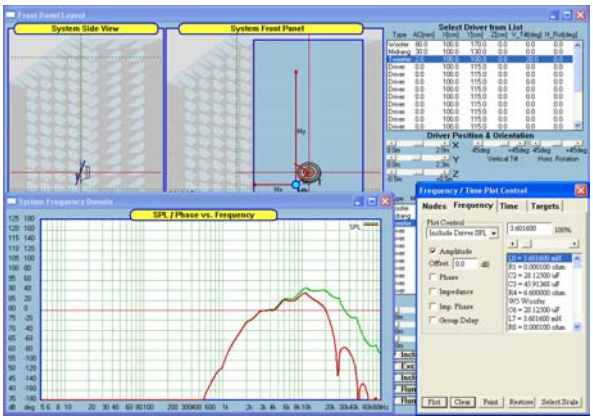


Figure 12.4 Tweeter tilted up 20deg and microphone re-positioned back on-axis.

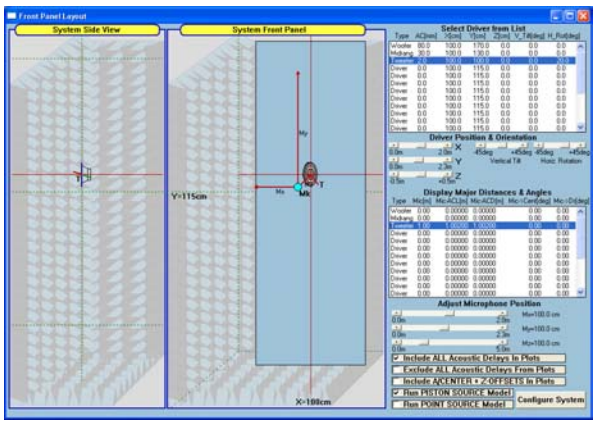


Figure 12.5a Tweeter driver rotated right by 20 deg.

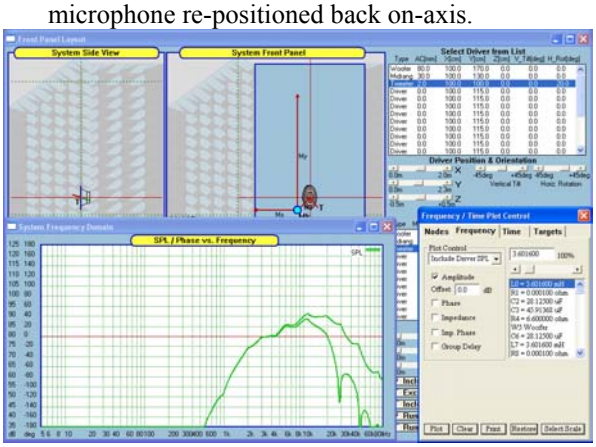


Figure 12.5b Tweeter rotated right 20deg and microphone re-positioned back on-axis.

Off-axis Response

The off-axis response are better simulated in PISTON MODE. Remember, that point sources radiate in all directions, therefore, the "off-axis" response will be identical to the "on-axis" response for single driver.

In the PISTON MODE, the program recognizes, that microphone is being moved off-center and is actually closer to one edge of the driver. If you can imagine dividing the piston area of the driver into many smaller radiating areas, you will realize, that moving the test microphone off-center, creates a difference in path lengths and thus the arrival times of pressures generated by each small radiating area. Vectoral summation (eg; taking into account individual phases) of all contributing pressures, creates peaks and nulls in the resulting frequency response. Peaks – when all contributions add constructively and nulls – when all contributions happen to cancel each other.

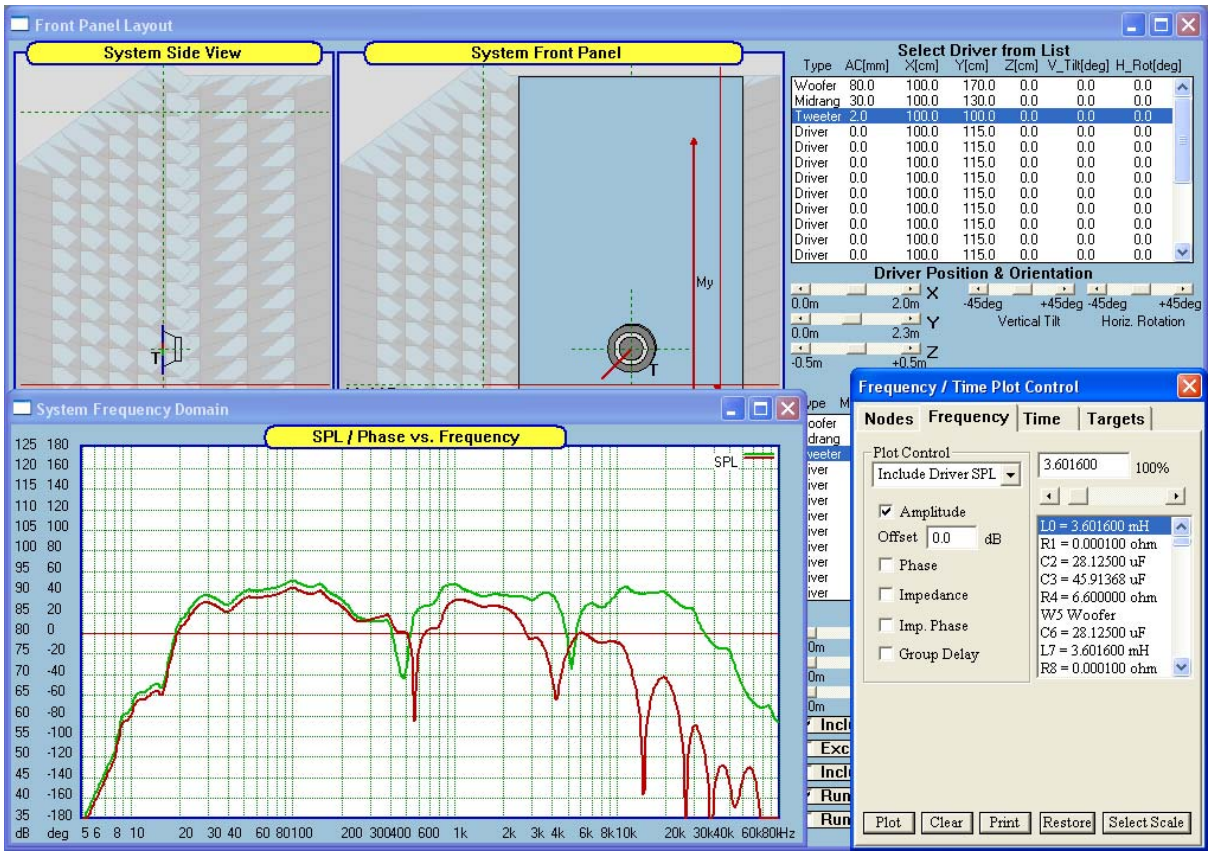


Figure 12.6 System SPL off-axis – brown curve.

SoundEasy uses a model, that divides the driver piston area into 81 smaller sections and then calculates contributions of each individual section. A more accurate model would be accomplished with many more even smaller radiating areas combining into the total acoustic output of the driver. However, this approach results in increase of computational time, therefore, current SoundEasy model uses 81 areas. Model implemented in the program enables you to predict the "off-axis" response of a tilted driver as well. This will come very handy when simulating the "off-axis" response of a tilted front baffle system with 5 drivers.

Note: Drivers' and microphone configuration set-up on the "Front Panel Layout" screen is remembered by all other screens in this module. This enables you to perform off-axis optimizations, time domain plots, vertical/horizontal "polar plots" and vertical/horizontal "straight-line plots".

System Vertical and Horizontal Response Plots

Polar plots (vertical or horizontal) modeled on your computer are quite easy to obtain. Assuming correctly implemented software algorithms, typically, a few clicks of the mouse is all that you require. However, practical verification measurements are nothing short of a problem. Ideally, you would set-up your test jig in an anechoic chamber and use computer-controlled, sophisticated turntables and microphone movement equipment to obtain the required degree of accuracy. This is because typical polar plots require transducer movement in TWO dimensions – on the X-Y plane.

In your "home laboratory", it would be unlikely, that you would go to these lengths to verify the theoretical predictions of the polar plots of your design. But what if you could MODEL and MEASURE your system responses using only one axis to move the test microphone ?. Moving the test mike along a straight line simplifies the measurements significantly. In most cases, a long, stretched tape-measure is all that is required to position the test microphone accurately in small increments. You can easily obtain excellent "off-axis" responses, and this is exactly what the polar plots are representing. Now, if your software is also capable of modeling microphone movements in straight lines, you can easily correlate and verify model predictions with the actual "straight line" measurements, correcting for the variance in distance.

System vertical and horizontal polar response can be evaluated using **"Polar Plots"** dialog box. The box opens when you select "System Polar Plots" item from the main menu. The associated plotting screen will actually generate two plots**.** There will be a change in SPL due to the distance, as the microphone is moved further up or down or left and right. Please note, that SPL plots generated on the "Frequency Domain" plotting screen are therefore affected by the microphone movements. As a good hint to remember is, that frequency response plots are generated for a "stationary" microphone setup over the whole range of frequencies. Polar plots are generated for "stationary" frequency over the whole range of microphone positions. All these plots **must** correlate with each other. For example, if you see a notch in the frequency response at some frequency and particular microphone location, the same notch must be present in polar plot performed at this frequency.

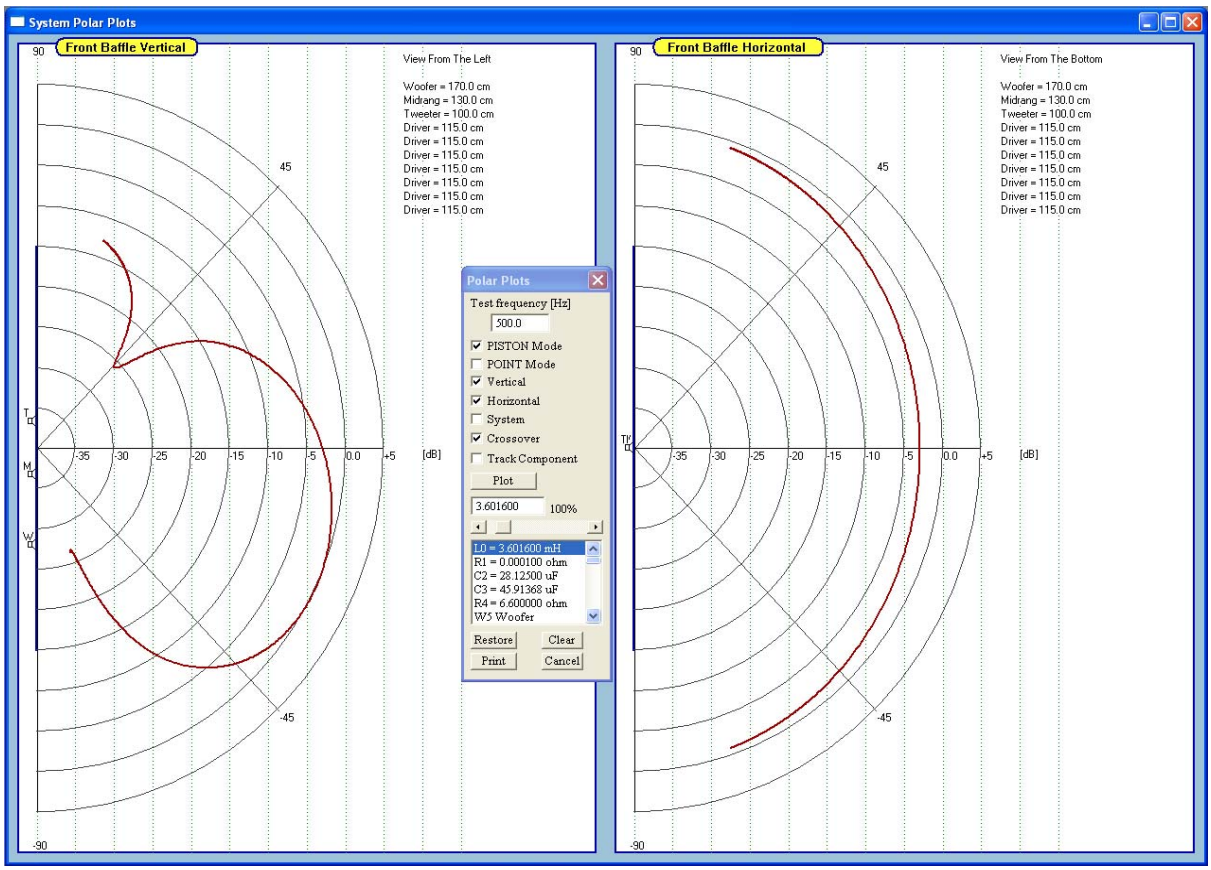


Figure 12.7. System Vertical and Horizontal Polar Plots box.

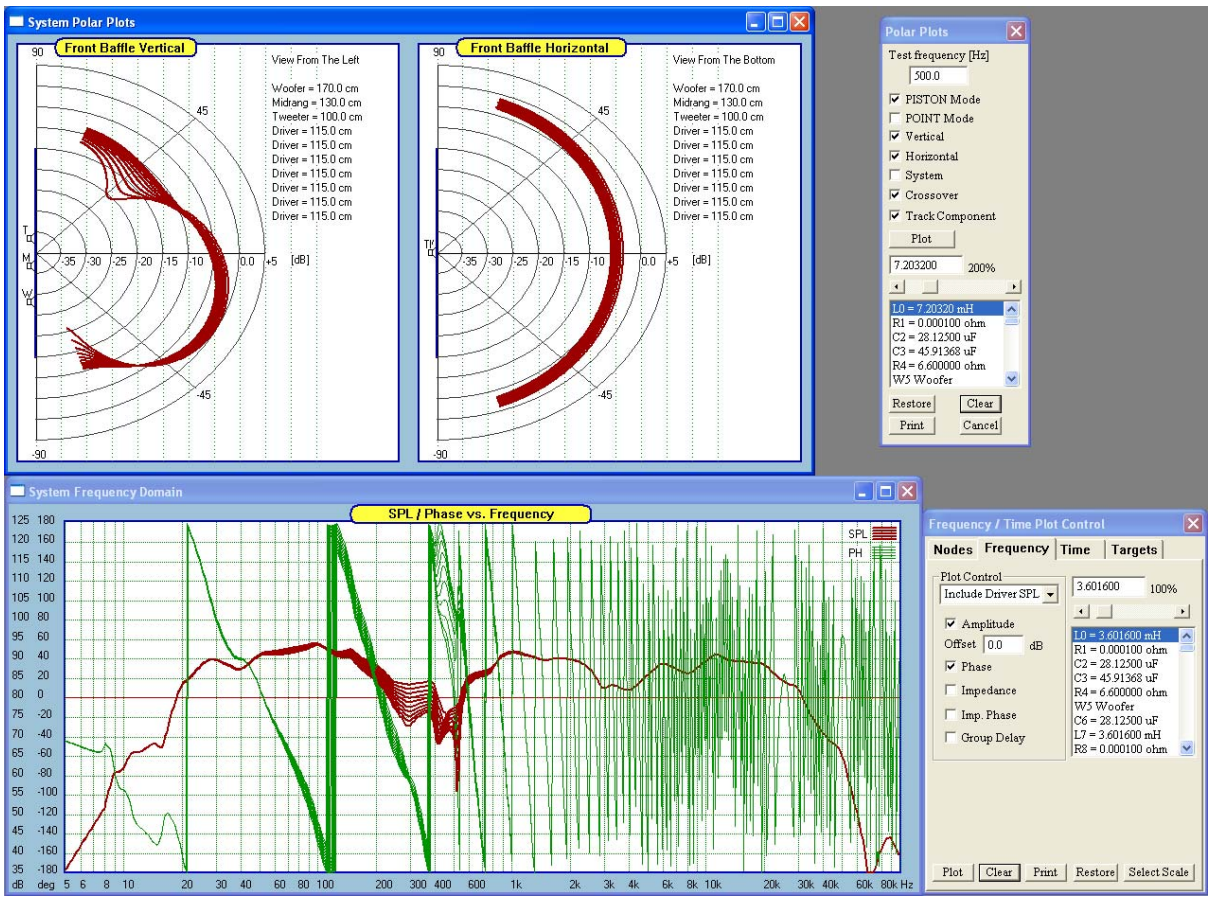


Figure 12.8. Example of "Polar Plots".

The dialogue box is rather self-explanatory and contains the following controls:

1. **"Test frequency"** – this is an editable data field for you to enter the frequency of interest in Hertz.
2. **"Plot"** - use this button to generate vertical/horizontal polar plots of the crossover.
3. **"Vertical"** – use this check box to select Vertical plots.
4. **"Horizontal" -** use this check box to select Horizontal plots.
5. **"List Box"** – Doubleclick on any component to select it for slider adjustments.
6. **"Restore"** – button to RESTORE component value to the 100% prior adjustement.
7. **"Clear"** - button to clear the plotting area and re-prints the scale.
8. **"Print"** button – simply prints the screen.
9. **"Cancel"** button – closes the box.
10. **"PISTON Mode" –** check box to select PISTON mode.
11. **"POINT Mode"** – check box to select POINT mode.
12. **"Slider"** – adjust component values 1%-1000%
13. **"Track Component"** – select this check box if you have System frequency domain screen opened and you want the SPL plots to be updated automatically.

If the microphone on the main system setup screen is moved, the center of the polar response will follow the ovement.

Is often desirable to visualize changes Frequency Response while checking Polar Response. The Polar Plots dialogue box has built-in functionality allowing you to change a selected component and then adjust it with the slider within 1-1000% of the original value. All changes are automatically reflected in Polar Plots curves. However, you can also observe SPL/Phase changes at the same time. Please make sure, that **the System Frequency Domain plotting screen OR CAD Frequency Domain plotting screen are opened** and the "Track Component" check box is "checked" on the "Polar Plots" dialogue box. This siftuation is depicted on Figure 12.8

Working With Project Files

Two new window menu options have been provided with the 'Sum-plot' tool. These are "New Project" and "Load Project" options. These two options are designed to accelerate the loading of the three driver's files, keep them together as one project and facilitate tracking of your system design. This file is designed to hold the following:

1. Five paths and file names for the selected drivers.
2. XYZ mounting/tilting/rotating coordinates of each driver.
3. CAD schematic of the complete crossover.
4. System optimisation parameters and more…..

The file names are then used to automatically load the drivers' data files. The names of the drivers' data files can be entered into the project using the "New Project" option from the window menu. In summary, you would follow the steps below:

To create and save Project file:

1. Activate “New Project” menu under "File" main menu option. A dialogue box with 5 editable fields for path names will be opened. Please fill appropriate field for woofer, upper bass and press on “Load” button. The selected files will be loaded into the program’s memory.
2. Save project file using "Save Project" option.

To load Project file:

1. Activate "Open Project" option.
2. Select and load required project.
3. You will be given an option to review the individual drivers' filenames.
4. Press "Load" button to load ALL the files into the program data space. The program will also re-open the default screen and will display all loaded drivers on the front baffle.

Phantom Project Files.

SoundEasy does a lot of background work checking presence and validity of data. when performing the design tasks with fully edited driver files, using methods described in this manual, the subsequent module checks consistency of data provided by the previous module and, usually issues no warnings. Processing of the data follows broadly the steps described below:

1. 'Box' module provides:
   * driver's complex transfer function, driver's type and
   * electro-mechanical parameters – **remember to include driver’s diameter for Piston Model**
   * enclosure type and enclosure parameters, calculates driver's impedance vs. frequency.
2. ‘Crossover’ module provides:
   * provides complex transfer function of the compensation circuit.
   * provides total impedance vs. frequency function of the driver mounted in the box, with the impedance compensating components attached.
   * includes compensation complex transfer function in the final crossover transfer function.
3. 'System' module adds vectorally all individual transfer functions in XYZ space.

It is clear, that subsequent functions relies heavily on the data provided by the previous module. Phantom files are an attempt to "fool" the program. They provide a short-cut and force the algorithm into using default values to much greater extend. Phantom files are useful when performing crossover or compensation circuit study. These files do not contain drivers' transfer functions or electro-mechanical parameters. However, there are still some data, that must be entered in order to run the program.

Working With Several Projects

SoundEasy allows you to open up to **3 projects at the same time** and store them in RAM memory. This approach requires you to have somewhat more advanced design skills, however, it offers you an excellent

way of making quick comparison between projects, **cutting-and-pasting from one project to another** and so on. Projects are controlled from the main menu and Digital Filter and Digital Equalizer dialogue boxes.

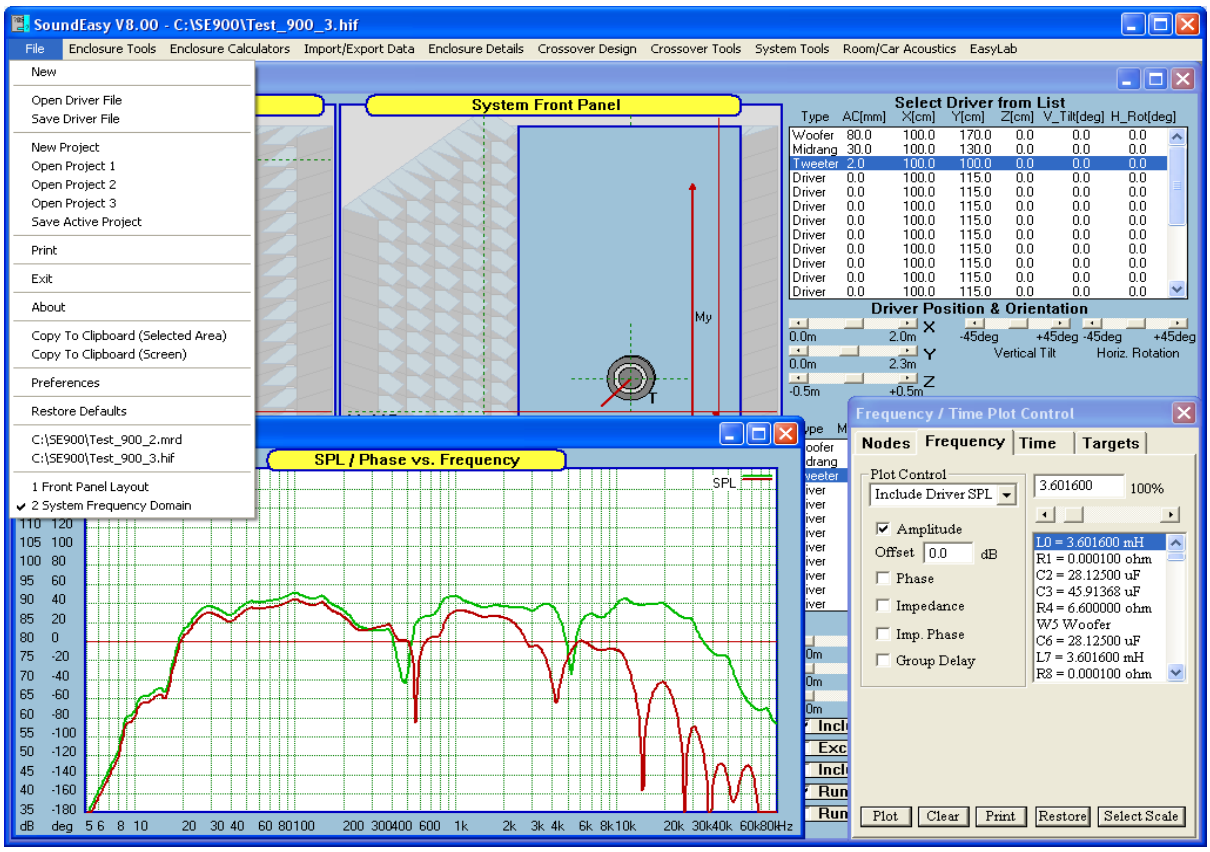


Figure 12.9. Loading all 3 projects into the program data space.

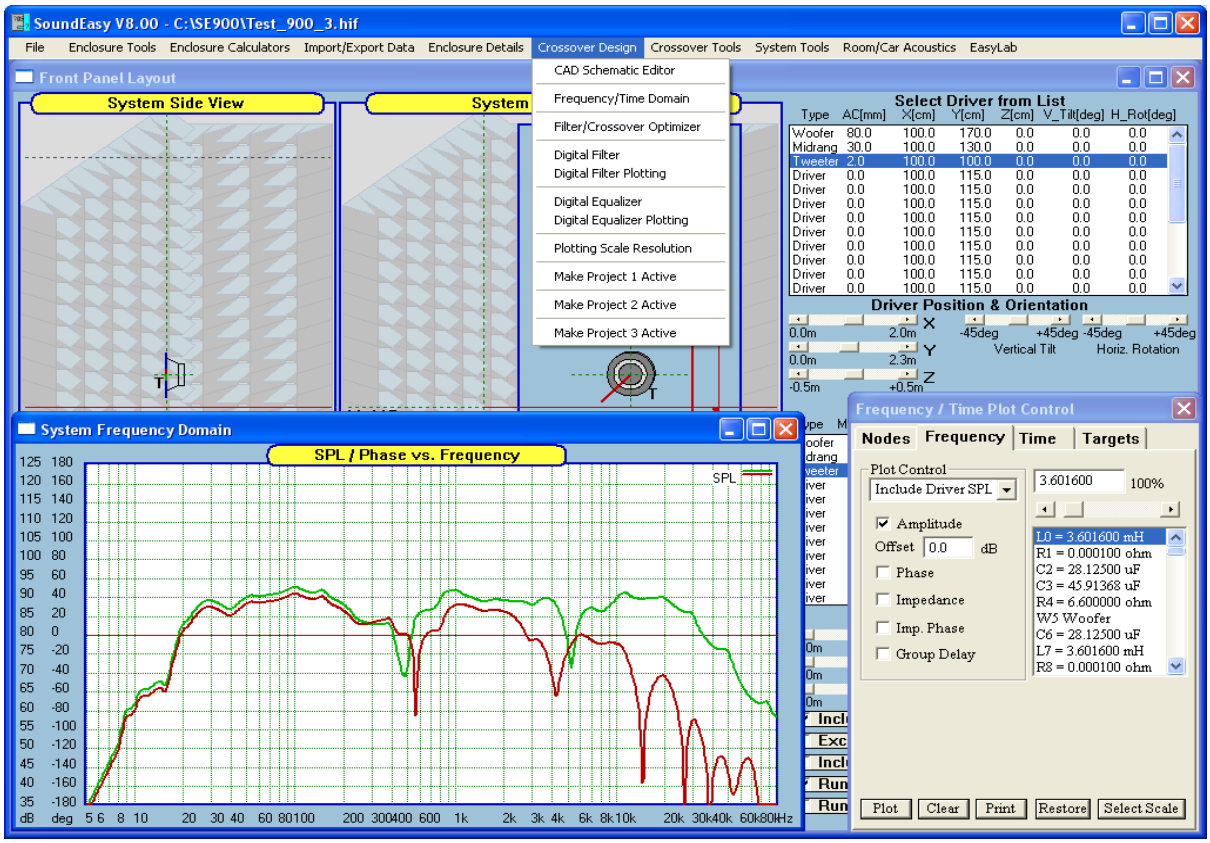


Figure 12.10. Activating Projects from “Crossover Design” menu.

System Relative AC offsets

When all drivers in a multi-way speaker system are placed on the front baffle, their "acoustic center – to – microphone" distances, ***Di***, will be affected by the distance between the microphone and the baffle itself. By subtracting this distance from the total distance ***Di,*** we can view the "relative acoustic distances" (or offsets) between the drivers. The relative offsets, primarily determine interaction between the drivers at the crossover points.

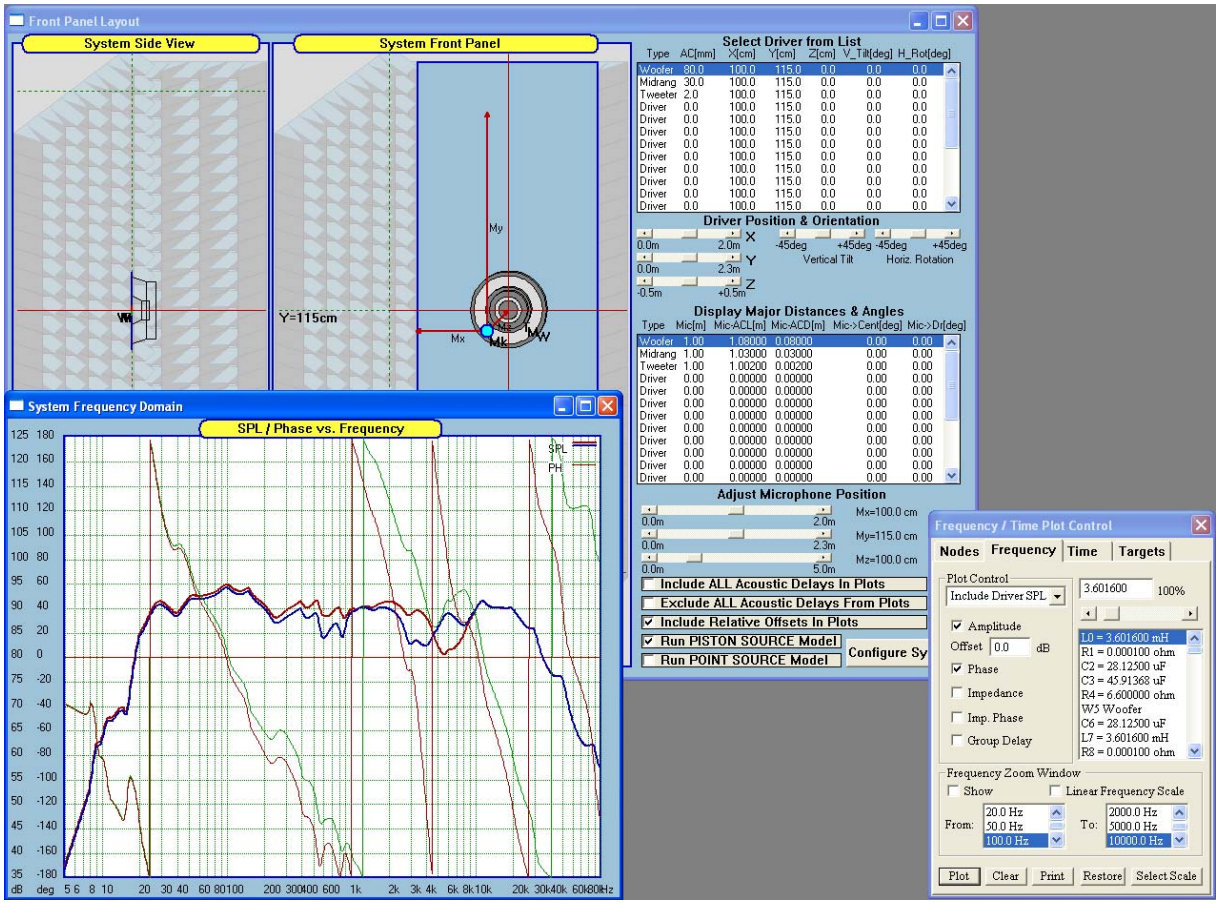


Figure 12.11. Relative acoustic distances (red) phase response.

Chapter 13. System Optimization

Background

Crossover optimization is considered to be the final stage of fine tuning the design. After the individual channels have been designed and optimized, The optimizer tool should be employed to determine if any further improvement in the frequency response of the system can be obtained. There are a number of factors which must be considered when determining the crossover cut-off frequencies and the speed of the roll-off. Some of the loudspeaker drivers on the market must be used with crossovers having a high roll-off speed because of their power handling characteristics. Should this be the case some caution is advised when using the optimizer tool. The algorithm employed within the module is not bound by the power restrictions of the driver. The prevailing factor assumed by the algorithm is the minimization of the error between the target and system frequency response.

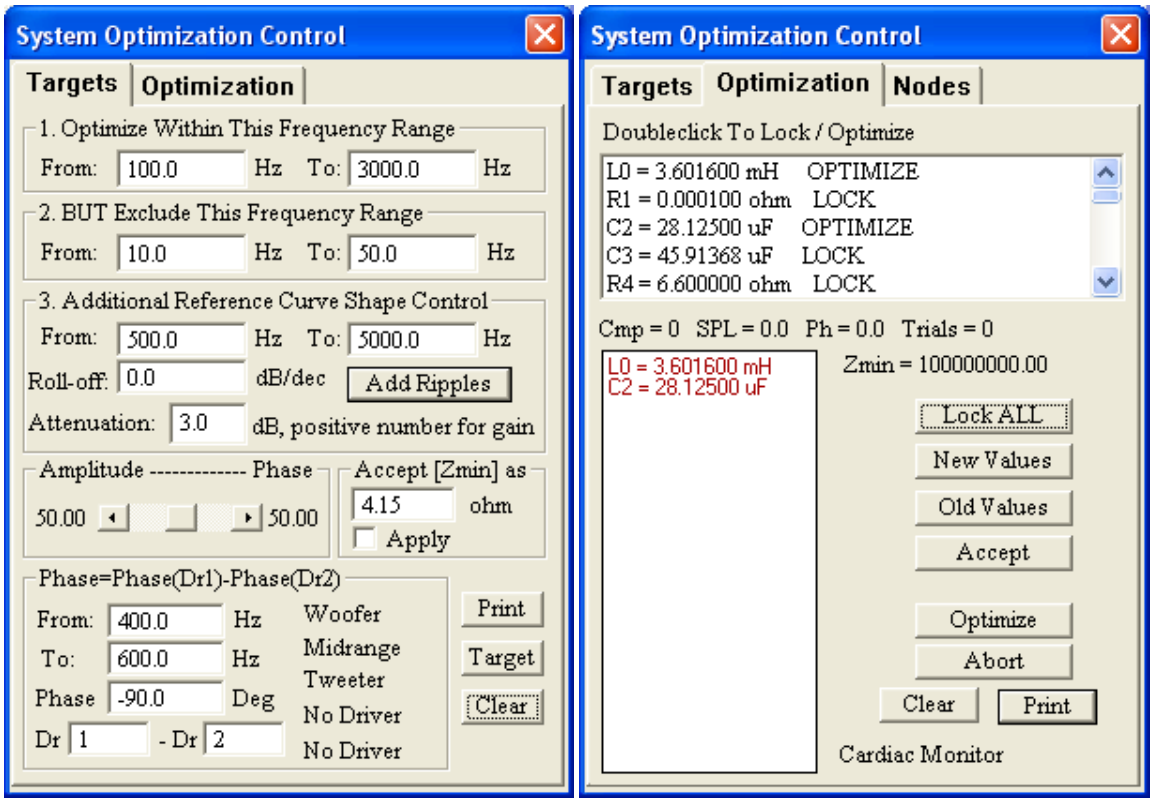


Figure 13.1.Optimization Control Dialogue Tabs

System Optimization Control Tab

1. **"Optimize Within This Frequency Range"** - two editable fields to accomplish just that. Enter frequencies in Hz.
2. **"BUT Exclude This Frequency Range"** – again, two editable fields to accomplish just that. Enter frequencies in Hz.
3. **"Additional Reference Curve Shape Control**" - group has four editable fields. The **"From"** field selects the frequency (in Hz) from which the response is to roll-off. . The **"To"** field selects the frequency (in Hz) at which the response is to flatten again The third field - **"Roll-off**"- is used to enter the required tilt in dB/decade. Usually, it is a small amount: 1-3dB/decade.
4. The **"Attenuation" parameter indicates, how much we would have to shift the 90dB reference line to match the system efficiency. For example: system efficiency is 86dB, enter "Attenuation" as –4.**

5. It is recommended, to use **"Target"** button on the dialogue box to review system target reference line before commencing optimization process. The "**Target**" button allows you to plot the reference curve on the screen without commencing optimization and is particularly useful when using "tilted" mode.
6. **"Optimize"** – use this button to start optimization process.
7. **"Add Ripples"** – this button closes the "Optimization Control" box and opens another box to enable you to add 3 ripples.
8. **"Accept [Zin] as**: - editable data field to enter the lowest acceptable input impedance your amplifier can tolerate. **MUST be greater than 0.1ohm**
9. **"Apply"** – checkbox to enable/disable Zmin restriction.
10. **"Phase=Phase(Dr1)-Phase(Dr2) Group"**
    "From" – enter required starting frequency of the band.
    "To" – enter required end frequency of the band.
    Phase response is typically changing much more rapidly with frequency, than the amplitude response. Therefore, the selected frequency range should be significantly narrower than typical SPL optimization frequency range.
    "Phase" – enter desired reference phase value.
    "Driver #" – enter the first driver number as displayed in "Drivers Used" list.
    "Driver #" – enter the second driver number as displayed in "Drivers Used" list.
    Phase response to be optimized it the phase difference between those two nominated drivers.
11. **"Amplitude ----- Phase Group"** - use this slider to apportion mutual importance level to amplitude and phase.

Optimization Tab

1. **"Lock All"** – Button to lock all values from optimization.
2. **"Clear"** button will erase the plotting area.
3. **"Abort"** button will cause the program to exit from the optimization routine.
4. **When the "New Values"** button is clicked upon, the original component values are substituted for the newly optimized.
5. When you click on the **"Old Values"** button, the process is reversed. Once again, plotting the reference curve and the driver's frequency response leads to determination of the optimization boundaries.
6. **"Accept"** – press this button to accept newly optimized component values
7. Click on **"Optimize"** button to commence the optimization process or to edit the target response parameters.

Nodes Tab – the same as previously described for Cad system.

Researchers recommend, that only one channel should be optimized at a time. Hence, the optimizer tool allows the user to select woofer, upper bass, midrange, tweeter or super tweeter channels, together with the frequency range of the optimization. The frequency range must match the selected driver.

This program has been developed for use with reactances including their "real life" losses. Thus inductors are modeled with their associated resistances and the complete model is the used by the optimization modules. As a result of this, the optimization algorithm may recommend increasing the inductor's resistance, should you decide not to LOCK it. Please remember, that it will have some effect on the total Q-factor of the driver connected to this crossover branch. It is also recommended to experiment with inductors having lower resistance losses, if only to avoid power loss.

Setting-up For System Optimization

We are now going to perform possibly the simplest optimization on a 3-way system. A step-by-step approach will be presented, that can be subsequently expanded for more complicated optimization cases.

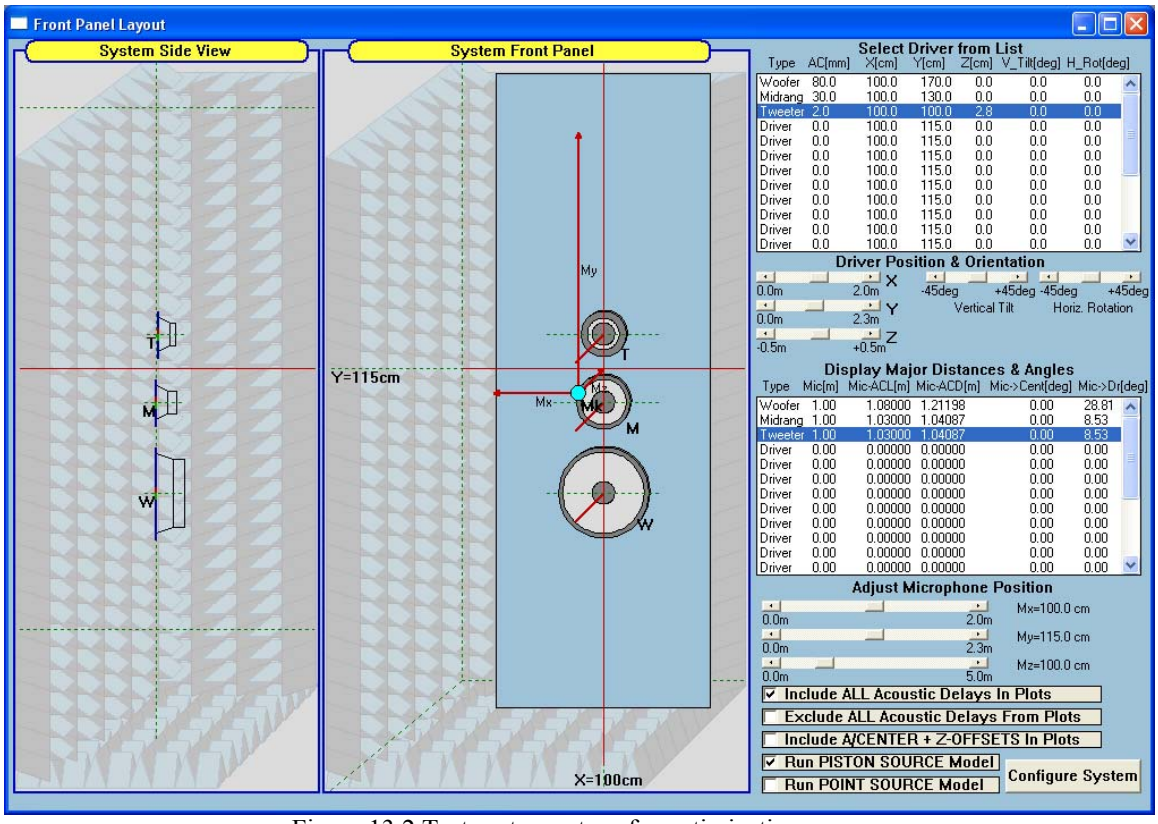


Figure 13.2 Test system set-up for optimization.

Loudspeakers that we decide to use in this example have the following characteristics:

1. Woofer: Efficiency = 91.0dB, acoustic offset = 80.0mm
2. Midrange: Efficiency = 92.5dB, acoustic offset = 30.0mm, L-Pad = -1.5dB
3. Tweeter: Efficiency = 91.0dB, acoustic offset = 2.0mm.

It is clear, that we are aiming at **system efficiency of 91.00 dB**. Additionally, only the midrange-tweeter crossover point will be optimized, however, you may apply the same principles to any other driver optimization. Finally, for the purpose of illustrating the optimization process, we are interested in **maximally flat frequency response above 1.5kHz**.

Topology of our design is depicted on Figure 13.2. Here we have the tweeter driver mounted on the top (Y=100.0cm) followed by midrange driver (Y=130.0cm) and the woofer is mounted at the bottom of the baffle (Y=170.0cm). As you can see, this would be a typical arrangement for a 3-way loudspeaker system.

Where do you place the microphone ?. As we have decided to focus on optimization of the midrange-tweeter crossover section, therefore, we will place the test microphone right between those two drivers (Y=115.0cm). This will be confirmed by equal distances from both AC centers of drivers to the test microphone (1.04087m) and equal angles from both drivers to the test microphone (8.53deg) – see Figure 13.1. The selected microphone location is actually quite common location for the test microphone.

Right now, we have a situation, where the acoustic center of the midrange driver is further away from the test microphone by 28.0mm (30.0 – 2.0 = 28.0mm). We have two options to compensate for this: (1) introduce electronic delay – just as described in previous chapters, or (2) physically offset the tweeter by the 28.0mm to bring it "in-line" with the midrange acoustic center. In this example, we decided on the second option. **This final set-up is clearly depicted on Figure 13.1**.

Estimating the Required "Attenuation" Parameter and the Frequency Range

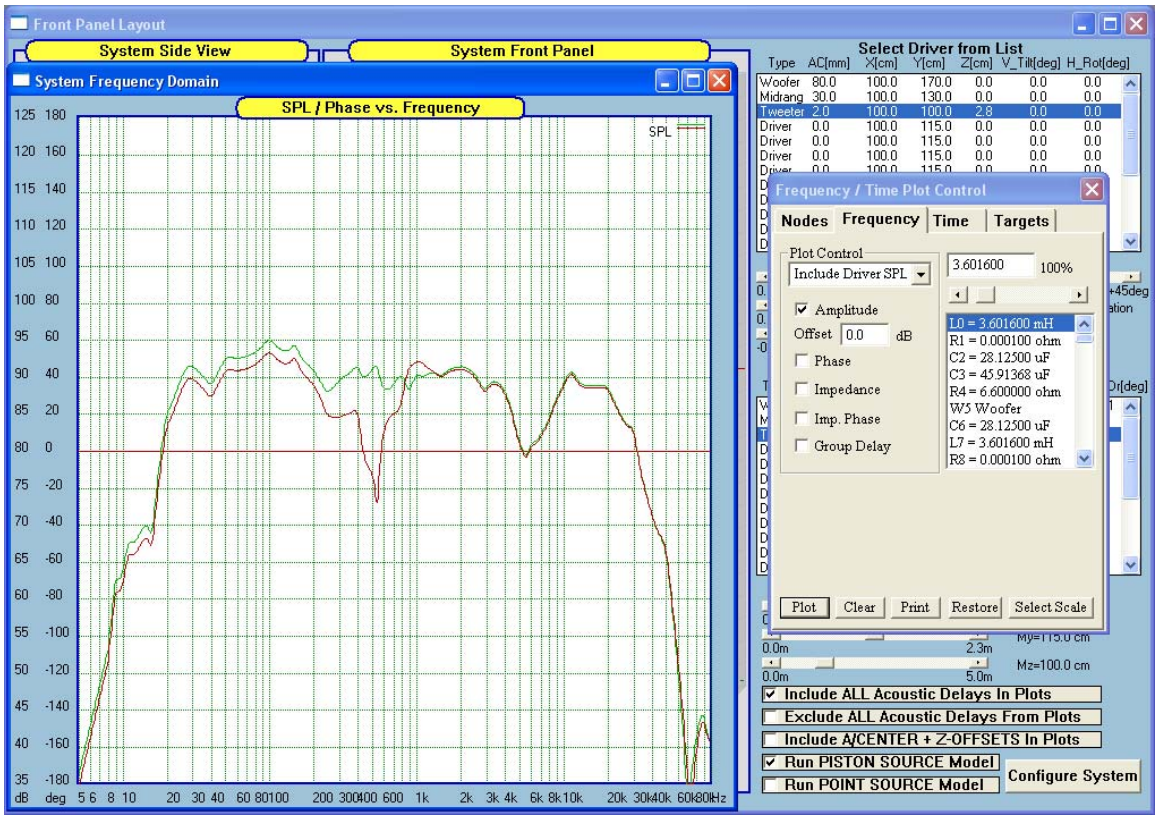


Figure 13.3 Lower plot (red) is about 0.2dB within 1.5kHz-20kHz frequency range.

Assuming, that all your drivers were measured at 1.00 meter distance (not including acoustic offsets) we are now trying to estimate, **how much additional attenuation we have introduced by placing the test microphone at 1.0409meter away from midrange and tweeter**. This is about 4.09cm further away.

Let us focus on the frequency range of interest: 2kHz – 10kHz, which is where the intended optimization will take place. Make sure, the you set the check box at the bottom of the "Front Panel Layout" to "INCLUDE Acoustic Delay In Performance Plots" position, and plot the system frequency response.

Next, set the check box at the bottom of the "Front Panel Layout" to "EXCLUDE Acoustic Delay From Performance Plots" position, and re-plot the system frequency response on the same screen. You should obtain a plot that (in this example) is a fraction of a decibel above the previous plot within 2kHz – 10kHz frequency range. Our estimation would be appr. 0.2dB above the first plot – see Figure 13.3.

Therefore, the **system efficiency, as measured at our test microphone location will be**

91.0dB – 0.2dB = 90.80 dB.

Now, the **"Attenuation" parameter indicates, how much we would have to shift the 90dB reference line to match the system efficiency** (as above, 90.80dB in this case). Therefore, the "Attenuation" parameter is **set to 0.80dB**.

Please note, that midrange crosses over to tweeter at 5kHz. Therefore, we set the optimization frequency range 1.5kHz to 20kHz – which is approximately 2 octave below and above the 5kHz cut-off.

Deciding Which Components to Optimize

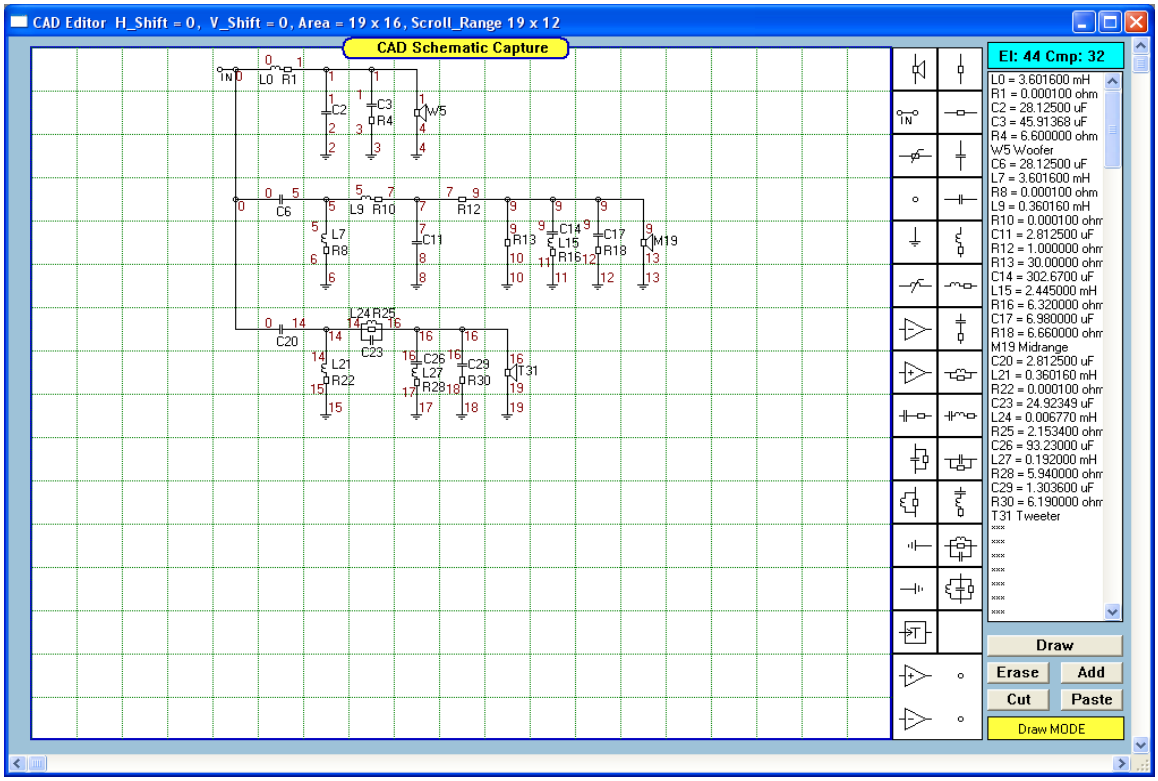


Figure 13.4 Here is our crossover to be optimized.

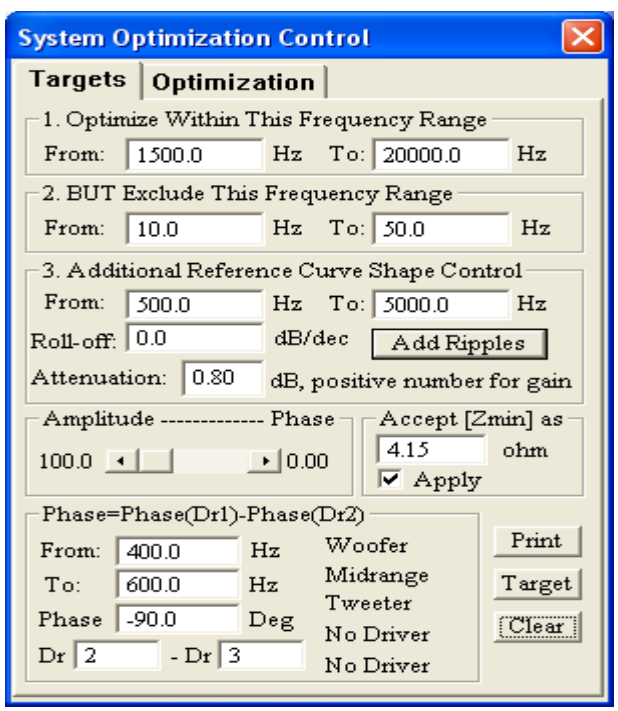


Figure 13.5. Fully preset optimization parameters for the midrange driver

We strongly advocate, that only one driver channel should be optimized at a time. Therefore, we will perform two-step optimization:

1. In the first step of optimization, we will concentrate on the low-pass section of the midrange band-pass filter: L9 = 0.36016 mH and C11 = 2.8125 uF - please refer to Figure 13.4
2. In the second stage of optimization we will concentrate on the high-pass section of the tweeter filter: C20 = 2.8125uF and L21 = 0.36016 mH - please refer to Figure 13.4.

Optimization Results

Figure 13.5 shows fully preset optimization parameters for the midrange driver and Figure 13.6 shows the results of optimization. Please note, that **SPL Error = 1168.2 after midrange optimization.**

If you were to finish the optimization now, you would press "New Values" button, so the SoundEasy can use these new values in other screens.

Figure 13.7 shows fully preset optimization parameters for the tweeter driver and Figure 13.8 shows the results of optimization. Please note, that **Global Error = 720.7 after tweeter optimization.**

Again, **PLEASE PRESS the "New Values" button now, so the SoundEasy can use these new values in other screens and later save them into the project file.**

In order to visualize the improvement, please consider Figure 13.9. The lower plot is the "Before" and upper plot is the "After" optimization result. Clearly, the frequency response is significantly flatter. It is always recommended to check the frequency response of individual filter AFTER OPTIMIZATION to check if the basic filtering characteristics of the crossover still hold. Also, the input impedance (Figure 13.8) is quite stable, therefore at this point of time, we would conclude that **optimization process met our initial requirements of maximally flat response above 1.5 kHz**.

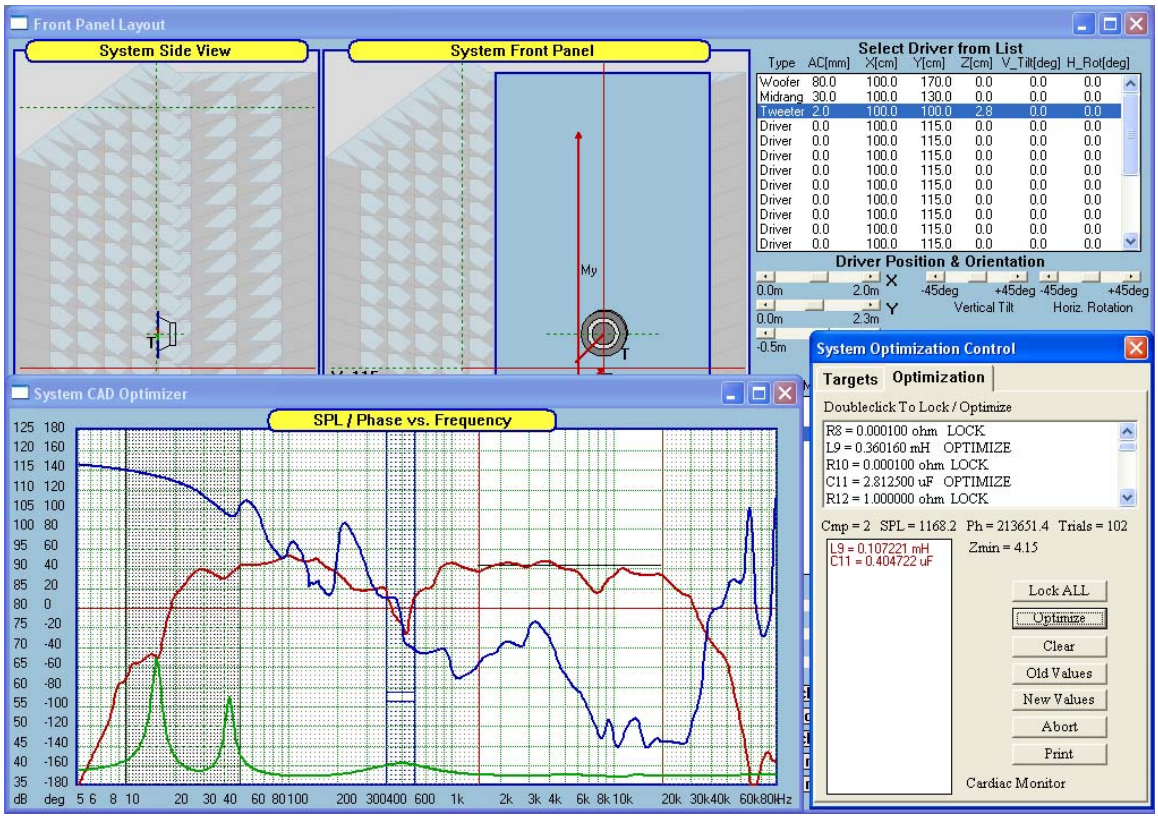


Figure 13.6 Results of the midrange driver optimization.

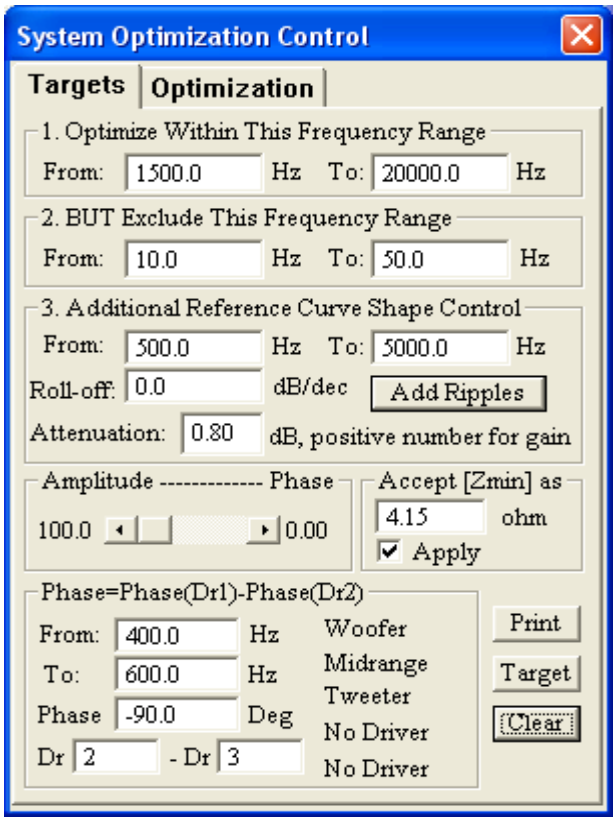


Figure 13.7. Fully preset optimization parameters for the tweeter driver

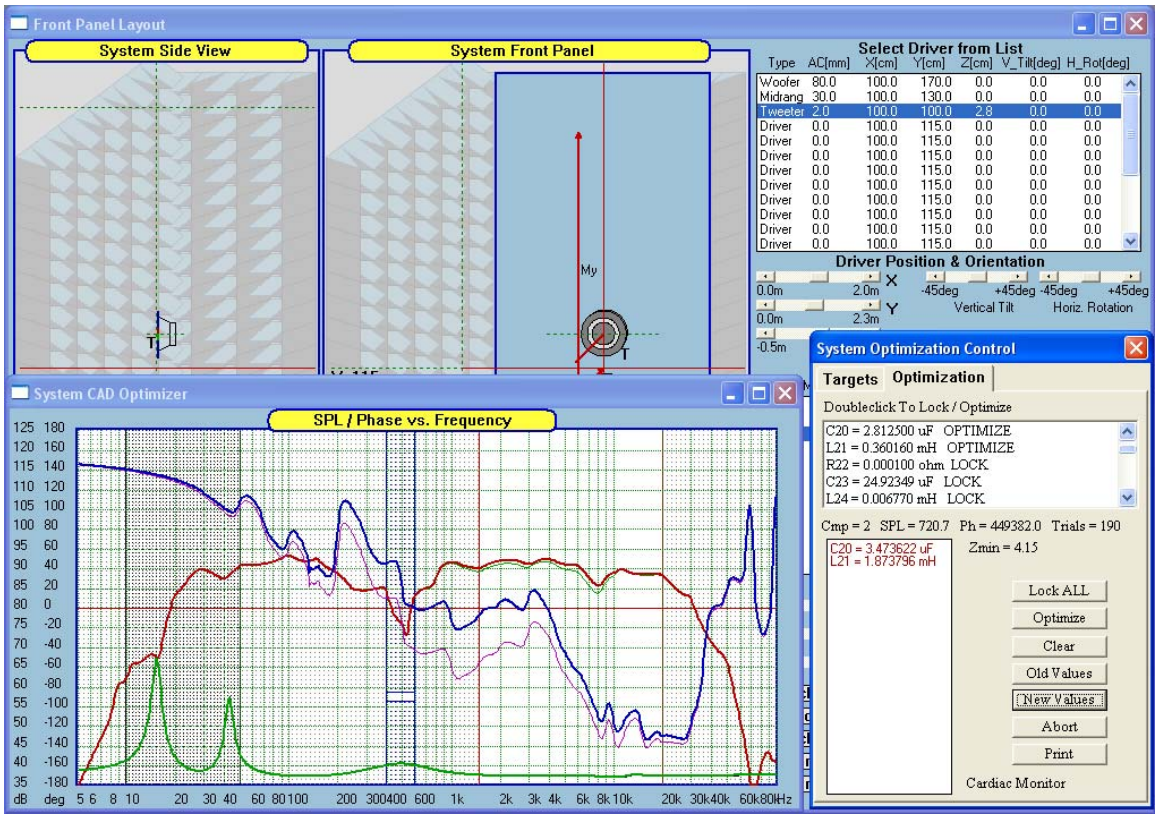


Figure 13.8 Results of the tweeter driver optimization

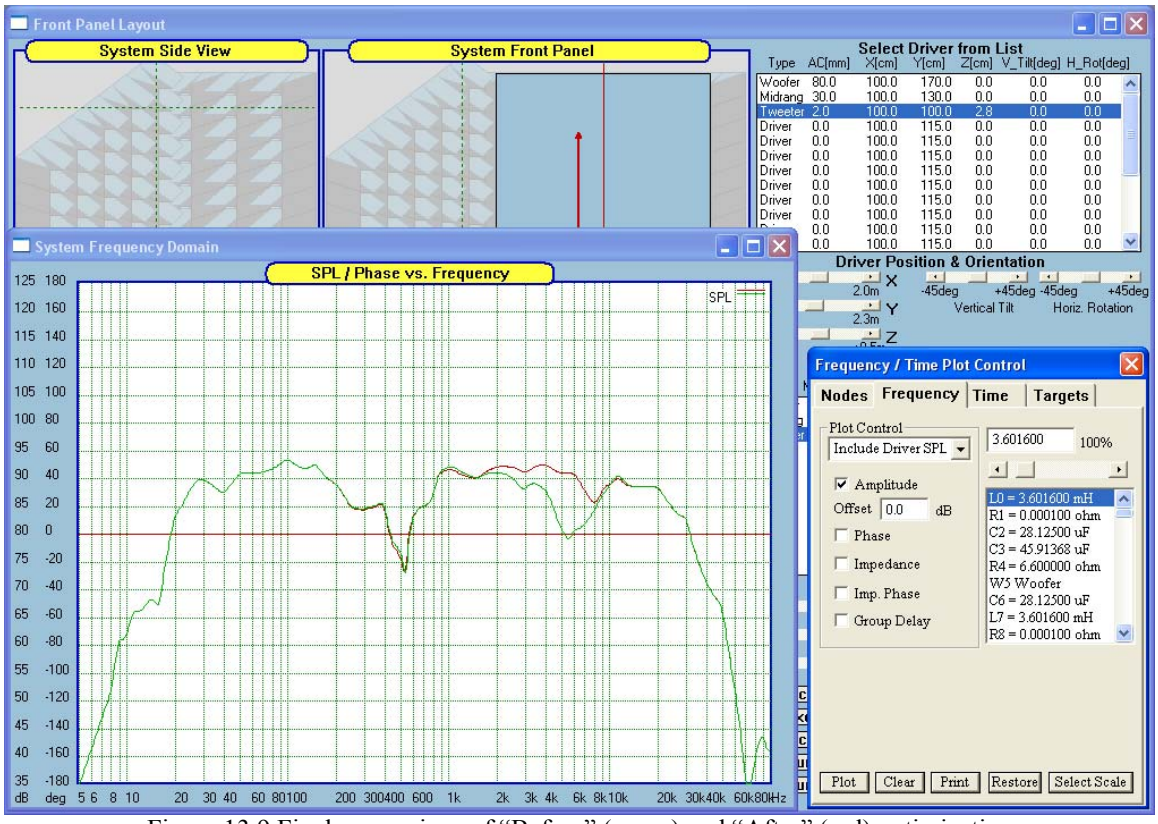


Figure 13.9 Final comparison of "Before" (green) and "After" (red) optimization.

Final Comments On Optimization

Next step is to critically examine the final component values of all the filters and their final frequency responses. Please remember, that crossover filter must also protect the driver from out-of-band frequencies and drastic deviation from the required shapes may not be in the best interest of the system power handling point of view. You are strongly encouraged to experiment with the optimization tools. A fast computer may be a definite advantage as it will shorten the optimization time. Two examples were given, one in Chapter 10 and one in Chapter 13 to illustrate the selection of the optimization frequency range. The designer needs to take into account the following factors:

1. The crossover point.
The selected frequency range needs to overlap the adjacent filter in order to perform the optimization properly, but excessive overlap is not recommended.

If the overlap is excessive, it will increase the optimization time and may cause some problems since the algorithm will try to compromise the degree of fit between the useful frequency range and the excessive overlap. Also, if the channel output falls below -15 or -20 dB below the nominal level, the driver contributes very little to the system and there is no need to Optimize beyond this level.

2. The optimization time.
Generally, the wider the range, the longer it will take to complete optimization.

3. The required flattens of the system frequency response.

As mentioned in Chapter 1, all optimization tools modify the data and the results need to be saved back to the data file.

Optimization algorithm.

Rosenbrock's "Rotating Coordinates Method" has been around since 1960. It is one of the most robust methods available for optimization when the derivatives are not available. This program implements the algorithm for the purpose of finding a minimum value of an error function of n variables; f(n1,n1,n3.....nn). The error is understood as positive or negative deviation from a reference function, which in our case is user selected reference frequency response of a given filter. The total error is calculated as follows:

$$f(n) = \sum_{k=1}^{520} (f_{resp}(k,n) - f_{ref}(k))^2$$

where: $f_{freq}(k,n)$ is the currently calculated frequency response,

$f_{ref}(k)$ is the user selected reference filter frequency response.

Summation is performed for all discrete frequency steps k=1...520. There is 'n' components (variables) in the crossover filter. Practically, the algorithm, operating in n-dimensional space, is attempting to find the shortest way from current location, defined by component values, to a new location, where the error function would ideally assume global minimum. From a current location, the algorithm is making 'n' trial moves in specifically calculated directions, with specifically calculated steps. The error function is calculated in every direction and care is taken to make sure, that every direction has been checked with a successful step (reduction in error) followed by failure (increase in error). The direction of the smallest error is then rewarded by increasing the step in this direction by 3 times. Finally, the whole coordinate system is aligned with the direction of the smallest error such a way, that the first coordinate is aligned with the direction of the smallest error. As one can observe, calculation of directions, rotating the system and calculation of step size becomes the focal point of the algorithm.

Gram-Schmidt ortogonalization process for finding directions is employed. In order to construct 'n' mutually perpendicular vectors (directions) **ξ1**, **ξ2.....ξn**, supporting vectors **A1, A2.....An** are constructed first:

A1 = (a1, a2.....an)
A2 = (0, a2.....an)
.....
An = (0, 0, 0,an)

The first direction ξ1, is found by normalizing **A1.**

$$\xi_1 = \frac{A1}{\left[\sum_{i=1}^{n} a_i^2\right]^{1/2}} = (\xi_{11}, \xi_{12}, \ldots\ldots \xi_{1n})$$

In the next step, **A2** is used to construct a vector **B2** normal to ξ1,

$$B2 = A2 - \xi_1 \left[\sum_{i=1}^{n} \xi_{1i} a_i\right] = (b_{21}, b_{22}, \ldots\ldots b_{2n})$$

B2 is then normalized to obtain the second direction **ξ2**,

$$\xi_2 = \frac{B2}{\left[\sum_{i=1}^{n} b_{2i}^2\right]^{1/2}} = (\xi_{21}, \xi_{22}, \ldots\ldots\ldots \xi_{2n})$$

B3, ξ3, **B4, ξ4**, are calculated in the same manner. Generally, direction **ξk**, is found as:

$$Bk = Ak - \xi_{k-1}\left[\sum_{i=1}^{n} \xi_{k-1} a_i\right] = (b_{k1}, b_{k2}, \ldots\ldots b_{kn})$$

$$\xi_k = \frac{Bk}{\left[\sum_{i=1}^{n} b_{ki}^2\right]^{1/2}} = (\xi_{k1}, \xi_{k2}, \ldots\ldots\ldots \xi_{kn})$$

Pre-Distorted / Tilted Frequency Response

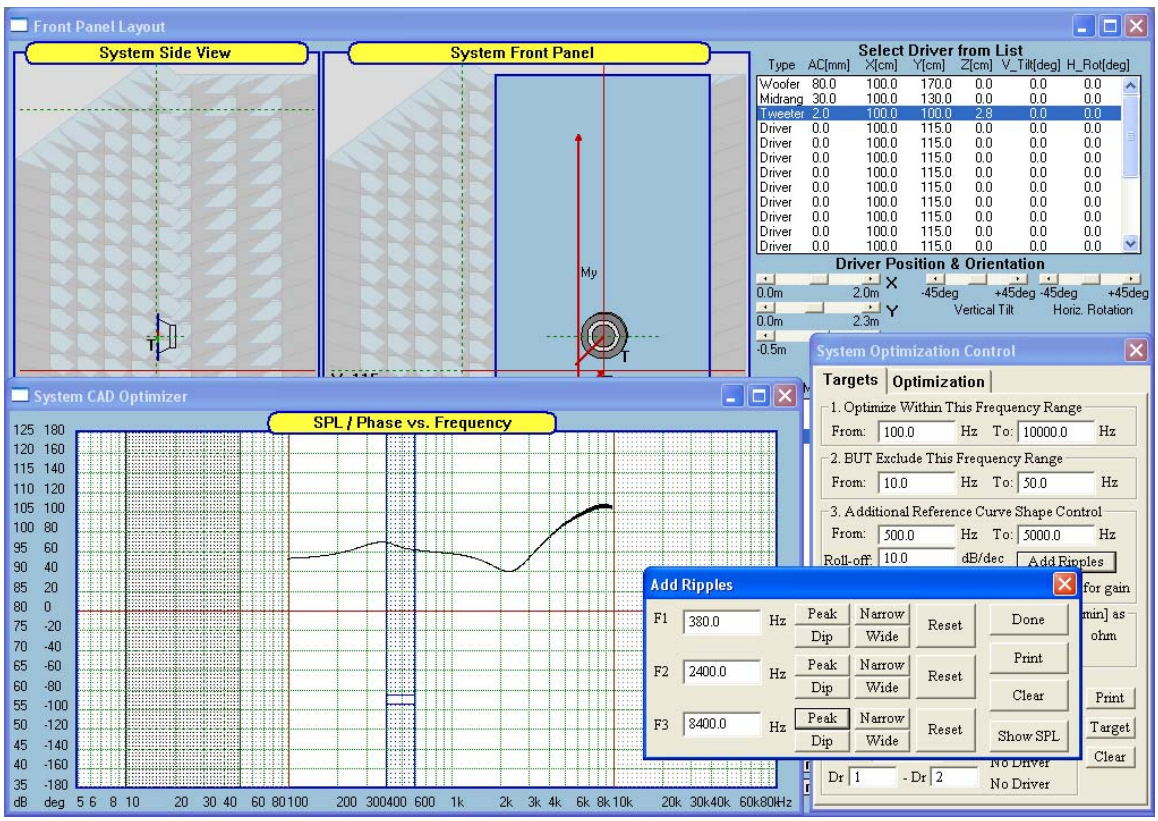


Figure 13.10. Various options for pre-distorting the target frequency response.

At times, you may wish to experiment with system frequency response, that is dropping toward higher end. Some systems may be particularly prone to these characteristics, such as WMTMW arrays. Here, the two woofers could add as much as 6dB to the efficiency of the system, while the midrange drivers contribute only 3dB. This may be the exact advantage you are looking for in the system. Obviously, selecting less efficient woofer/midrange drivers restores the balance. In a station like this, a smooth transition between the drivers can be achieved by tilting the frequency response by some fixed amount of dB/decade (NOT dB/oct).

Amplitude optimization alone will produce flat frequency response at the expense of phase relationship between adjacent drivers. This phase relationship, determines vertical radiation pattern around the crossover frequency. You could easily end-up with drastically different properties of your system, and really lose control over this important property of your crossover if the phase is allowed to be modified too much by the SPL optimization process alone. However, including phase in the optimization parameters, will prevent it from being excessively modified.

Even better, you can assign more importance to the phase (by moving the slider to the right) and actually have the phase relationship improved around the frequency band of interest. Changes in the vertical radiation pattern are the best observed using “System Polar/Line Plots” dialogue box. System optimization process can be now quite involving and often you may find yourself running the optimizer several times in order to explore various settings for phase, Zin and SPL. **Finally, we suggest, that you will select "Include A/Center + Z-Offsets in Plots" option on the system "Front Panel Layout" screen to reduce the number of phase reversals on the phase difference plot.**

Adding Ripples

In addition to the above pre-distortion, three “ripples” can be added to the target curve. Each ripple can be a peak or a dip in the frequency response. In order to incorporate the ripples into the target curve, please select “Add Ripples” button on the “Optimization Control” dialogue box. This dialogue will be replaced with smaller control box “Add Ripples” allowing you to control all parameters of the newly introduced ripples:

1. **Width** of the ripple – use “Narrow” or “Wide” buttons.
2. Location on the **frequency** scale – enter frequency in the “F1”, “F2” or “F3” data field.
3. **Height or depth** of the ripple – use “Peak” or “Dip” buttons.
4. **Cancellation** of the ripple – press “Reset” button.
5. **“Show SPL” – use this button to plot SPL with current optimized values.**
6. The “Clear” button will clear and re-plot the target curve on the main optimization screen.
7. Finally, when you are happy with the result, press “**Done”** button to close the “Add Ripples” dialogue box. All parameters of the ripples are saved to the data file.

Chapter 14. Room Acoustics

Background

Thus far, many conditions which affect the performance of a loudspeaker system have been examined. The drivers and the enclosure also interact with each other and the surroundings. This can affect the frequency response of the designed system. Listening rooms display vastly characteristics to anechoic rooms hence there is a need to evaluate the influence of the room boundaries and modes on the frequency response. The 'Room Acoustics' option with its three screens helps to understand these issues. Assumptions are as follows [5]: (1) The loudspeaker behaves like a monopole sound source, (2) The average absorption of the room surfaces is more than 3% and (3) The three closest boundaries are almost perfectly rigid reflectors. Most normal rooms will easily fit into these descriptions. The positions in which the loudspeaker systems are placed within the room are often dictated by the convenient spaces which happen to be available rather then by any scientific rules. However, if the best possible quality of sound reproduction is desired from the system, it needs to be positioned subject to a better understanding of the influencing factors. If a loudspeaker is moved about in the corner of the listening room, the user will notice, that the increase in the power output at lower frequencies can be as high as 9 db. This increase in the power output of the source can be explained with the help of the radiation resistance.

The efficiency of a typical moving-coil direct radiator loudspeaker system is in the order of 1-5%. This poor efficiency is due to the relatively small mechanical load which the air presents to the diaphragm and it's associated enclosure. Hence, most of the input power is dissipated in energy losses by heating the voice coil. A small percentage only is consumed by working against the resistive part of the air load (radiation resistance) to generate acoustic radiation. When the loudspeaker system is placed near one or more room boundaries, a part of the sound wave radiated by the loudspeaker diaphragm is reflected by boundaries back on to the diaphragm. If the wavelength of the sound radiation is much larger then the distance between the loudspeaker diaphragm and the boundary, then the combination of the outgoing and the reflected waves results in an increase in the radiation resistance.

Because the radiation resistance is only a very small part of the total mechanical impedance of the system, the amplitude of the diaphragm velocity will be virtually unchanged. This constant diaphragm velocity and the arrival of the reflected sound waves in-phase with the radiated sound increases the radiation resistance and then the power output of the system particularly in corner position. As the frequency is increased, radiation resistance falls until the reflected sound waves arrive in exact opposite phase to the radiated sound wave. This creates a severe dip in the power output of the system, as the air load becomes very small.

Enclosure Placement

The "**Room Image Method**" - enclosure positioning, function is incorporated under the **"Room/Car Acoustics**" menu and invoked from the main menu of this module (see Fig 14.1).

The first button - "Clear" is used to clear the plotting area. The next seven buttons provide plots for the indicated enclosure type. "Your Woofer" button recalls/generates plot for the woofer which has been created as the amplitude response (Transfer Function) with the "Editor" tool and "Your System" curve must have been created by the user as a summed response of the configured system. Box distance from the three boundaries (corner of the room) is selected using three scroll bars, one for each direction: X, Y and Z. Every time one of these bars is activated, the corresponding position of the enclosure is changed by a small increment. The minimum distance allowed is 5 cm, and the maximum distance allowed is displayed on the graph at the end of each X,Y,Z coordinate. The graphs are labeled automatically. You can display the woofer frequency response in various modes (enclosures) or your system frequency response. Figure14.1 shows an example of enclosure placement and the resulting responses. If a loudspeaker measured in a true free field, with no reflecting surfaces, such as a very good anechoic chamber, or outdoors far away from any large object, including the ground, the sound radiates into a full sphere solid angle, identified as 4π steradians. Placing the loudspeaker on a floor reduces the solid angle by half, and the sound pressure at low frequencies will be increased by approximately 6 dB because sound that would have propagated away will be reflected back. Leaving it on the floor and backing it up against a wall reduces the solid angle by two factors of two, to π steradians and the sound pressure is elevated by about 12 dB. Sliding the loudspeaker into a corner reduces the solid angle into which the sound radiates by a further factor of two, to $\pi/2$ steradians, and the pressure goes up by another 6 dB for a total low-frequency gain of about 18 dB! With respect to amplifier power and stress and strain on the loudspeaker, these acoustic gains are absolutely free, so wise people take advantage of them. An unfortunate side effect is that some irregularity is introduced at upper bass and lower mid frequencies.

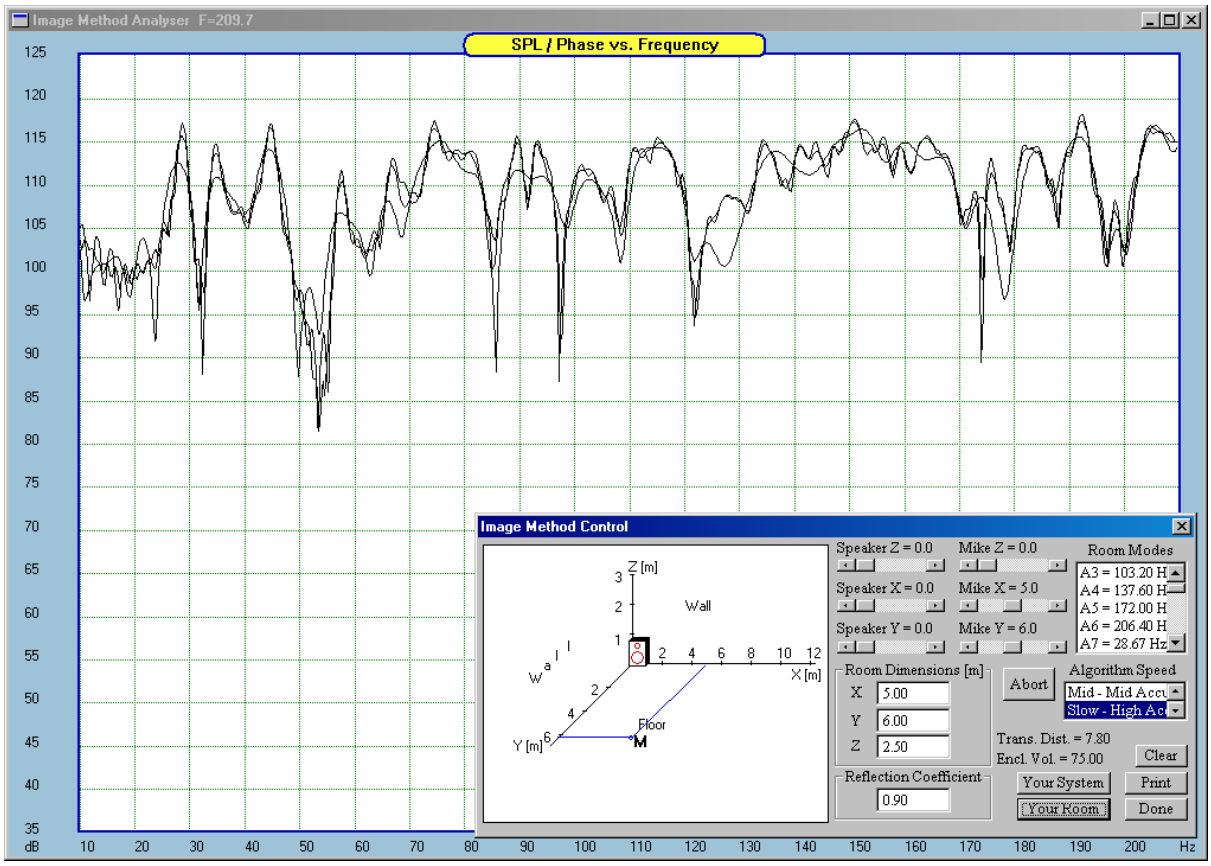


Fig 14.1 Image Method Screen

Image Method

Behavior of the sound waves in a small enclosure is a quite complex problem. However, the wave property of the sound energy makes it possible to introduce significantly simplified model of the problem, restricted to low frequencies, where only the room dimensions are of significance. In a rectangular room we are going to consider, the sound waves will be reflected and re-reflected between all 6 surfaces. When the reflections occur between opposite surfaces, the standing wave pattern generated this way is called "axial". Reflections involving four surfaces are called "tangential" and finally, reflections involving all six surfaces are called "oblique". The general expression for modal frequencies in a rectangular room is a well-known formula:

$$\omega_i = \pi c \sqrt{\left(\frac{n_x}{X}\right)^2 + \left(\frac{n_y}{Y}\right)^2 + \left(\frac{n_z}{Z}\right)^2}$$

Where: c is the speed of sound, c=344m/s, X,Y and Z are the principal room dimensions and n_x, n_y and n_z are positive integers (including zero).

The modes exhibit resonant character. They have natural resonant frequencies, bandwidths dependent on their damping factors and amplification Q-factors also dependent on their individual damping. Modal density varies in frequency. Modes tend to be widely separated at lowest audio frequencies and then come closer together as the frequency of interest is increased. In addition, the effects of low frequency modes on the source frequency response are heavily dependent on the position of the source and receiver.

For example: if a receiver is located in a pressure node, it will record no sound at all for a case of zero dumping factor for this particular node. In a typical listening room, exhibiting many modes, the frequency response of the source will be the vector summation of contributions of all modes, as well as the function of the source and receiver positions.

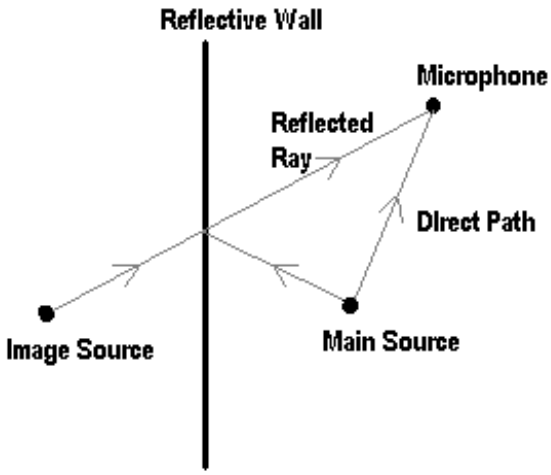


Fig 14.2 Image Model for single reflection

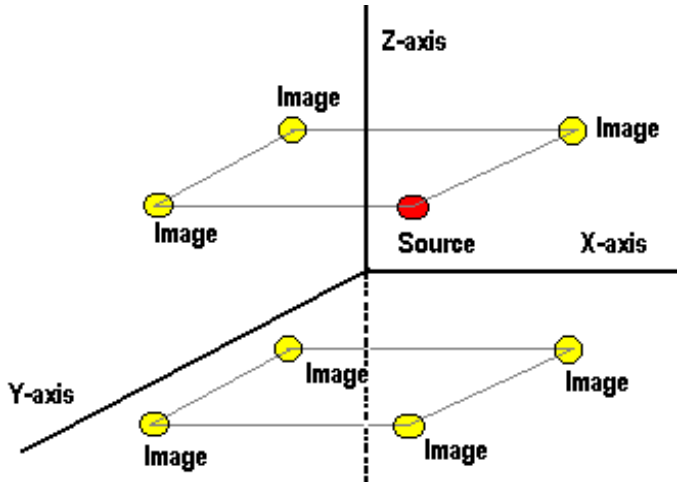


Fig 14.3 Image cluster (1 source, 7images) formed at a corner.

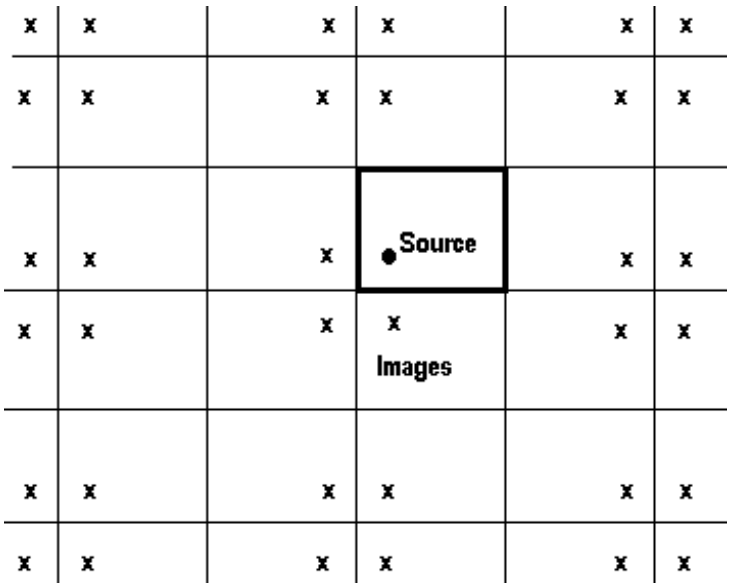


Fig 14.4 Positions of virtual sound sources on XY plane.

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Using the Room Modes Screen

Please upload project file "test301.hif" into the program and select "**Room Image Method**" screen from the main menu - see Fig 14.1. The control dialogue box contains 3 scroll bars to position the speaker up to several meters from the corner and correspondingly, 3 scroll bars to move the "microphone" around.

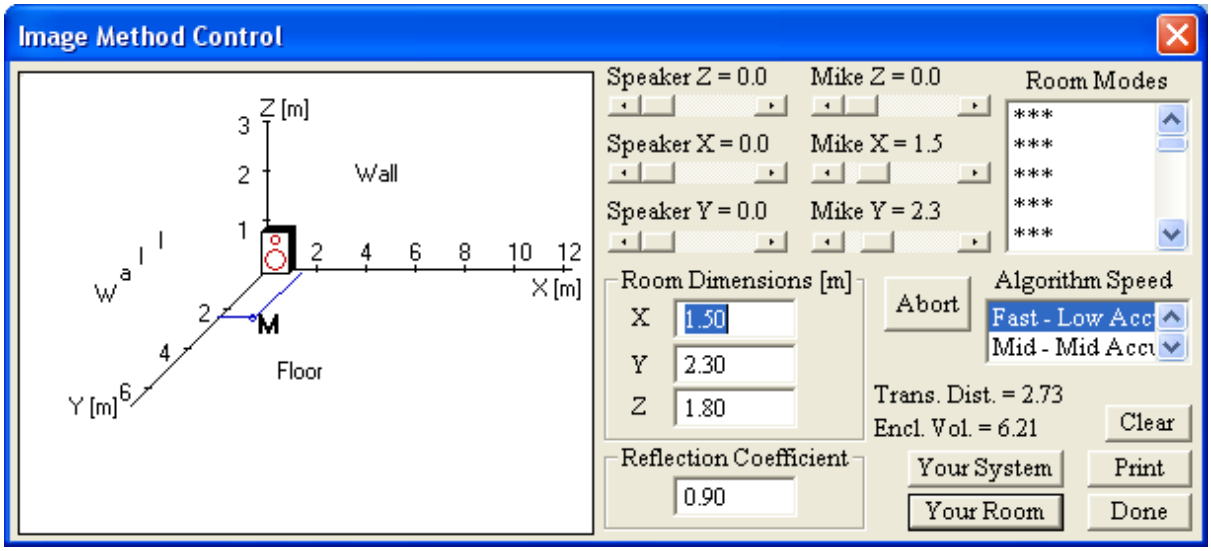


Figure 14.5. Image Method dialogue box.

Controls available in the Image Method Control box are quite simple:

1. **"Speaker Z"** – Controls location of the speaker along Z-axis (vertical).
2. **"Speaker X"** – Controls location of the speaker along X-axis (horizontal).
3. **"Speaker Y"** – Controls location of the speaker along Y-axis (horizontal).
4. **"Mike Z"** – Controls location of the microphone along Z-axis.
5. **"Mike X"** – Controls location of the microphone along X-axis.
6. **"Mike Y"** – Controls location of the microphone along Y-axis.
7. **"Room Dimensions X,Y,Z" –** Enter desired room dimensions in meters.
8. **"Reflection Coefficient" –** Accepts average reflection coefficient of the wall.
9. **"Algorithm Speed/Accuracy" –** Slow speed=high accuracy.

The dialogue box will display transducer distance and total enclosure volume. Pressing **"Your Room"** button will plot room contribution to the transfer function between transducers. Pressing **"Your System"** will add your system response onto the room transfer function. The **"Clear", Print" and "Done"** buttons do what the label indicates.

The loudspeaker can be position within X = 5m, Y=4m and Z = 3m distances from the room corner. The microphone can be located within X = 10m, Y = 12m and Z = 4m from the room corner. The horizontal frequency response is of 10-200Hz linear resolution. Room dimensions are editable from the three fields shown on the right-hand side. There is no built-in limit on the room size, but the screen is intended analyze small rooms. The room dimensions can be greater than graphical representation of the set-up shown on the bottom-left hand of the screen, however, the smallest room size is 1m x 1m x 1m. Pressing the "Plots" button invokes a floating menu, from which you can select various enclosures to be examined and also imported woofer and system frequency responses. The "Clear" button clears the plotting area. It is clearly observable, that the SPL frequency response of a loudspeaker placed in a confined space is far from being flat. The fluctuations in level are most pronounced at low end of the audio spectrum, where the modes are few and separate. The frequency response of loudspeaker will be further influenced by the actual location of the loudspeaker and the listener. The so called "corner location" allows the loudspeaker to excite all room modes - see Fig 14.1. On the other hand, locating the listener in the exact center of the room, further distorts the frequency response - see Fig 14.6.

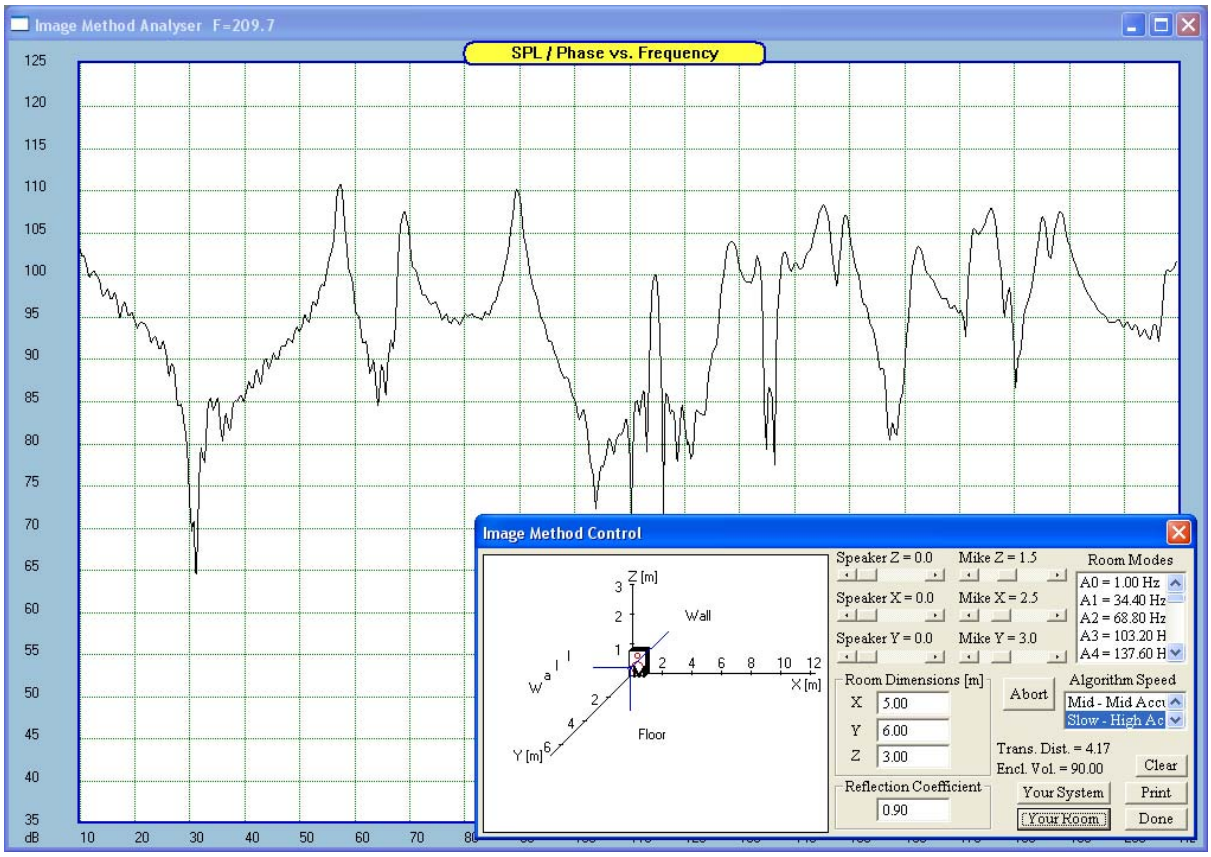


Fig 14.6 Central location of the microphone

Frequency independent, average room Reflection Coefficient (RC) can be entered in the corresponding data field on the bottom-right hand section of the screen. Please note, that entering the RC=0.01 (negligible reflection) corresponds to testing your design in the anechoic chamber. Indeed, when you use this low RC, the corresponding frequency response plot should look exactly as the one representing anechoic chamber. The RC=0.90 will produce highly distorted frequency response by contribution of the undamped room modes - see Fig 14.1.

The predicted "low end" frequency response is based on modal summation method. Calculations assume an empty rectangular room and the same absorption coefficient on all surfaces. The "real world" music room will rarely (if ever) comply with the above conditions. The room will contain furniture, the absorption coefficient of the floor and walls will be different and varied with frequency. The sum of all these factors will result is some differences between predicted and measured frequency response.

There is still quite a lot of sense in evaluating the predicted responses. The model is quite realistic in predicting frequency response deviations due to a particular location of the loudspeaker and listener. You will be able to select the best location for your speakers and plan the location of the listening area. For example: you will notice, that modal theory predicts large "dents" in the already highly irregular frequency response, when you place the listener or speaker in the center of the room.

Having worked with the screen for a while, you will find, that it is generally not desirable to place the loudspeaker or listener in the exact node of any given frequency, as this would create undesirable cancellations of the sound. The knowledge of modal frequencies and nodes' locations is essential in understanding your acoustic environment. This is why the FEM approach for predicting room acoustics in "non-rectangular spaces" is so successful. Even if the basic FEM implementation does not produce the frequency response, it will generate all modal frequencies and associated pressure patterns. All that is needed next is to review the pressure patterns and avoid placing speaker and listener in the nodes.

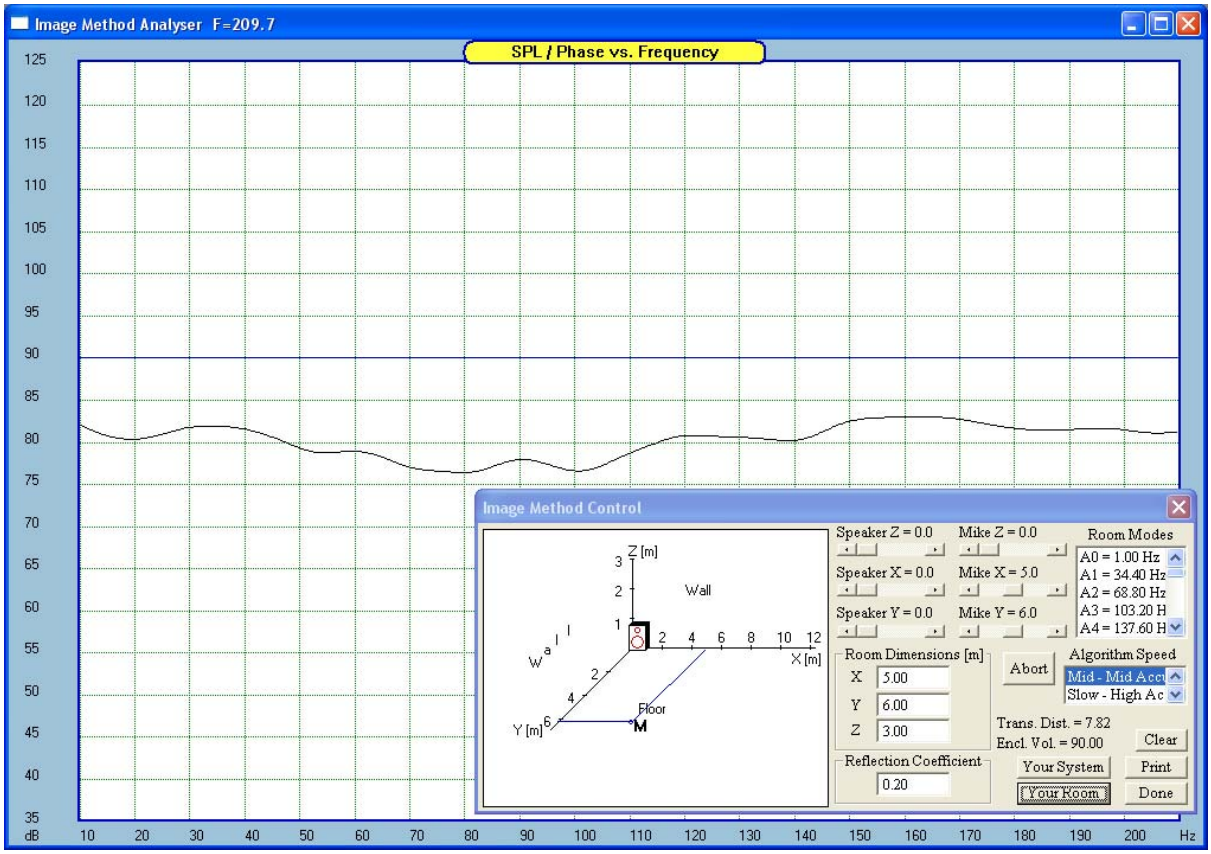


Figure 14.7. Low reflection coefficient.

Room Acoustics Background

At some stage you will arrive at the end of the loudspeaker system design process. It is probably fair to assume, that considerable effort went into designing and building the "perfect system", possibly including anechoic flat frequency response from 20Hz to 20kHz and many other desired and unique features. There is one more challenge to consider on the way to actually enjoying the results: the listening room acoustics. Room acoustics issues are complex and for many of us the "learning factor" will be significant. Absorption, reflection and diffusion are the issues to be considered, together with ray-tracing and modal analysis. In the worst case, misunderstanding in this area may obliterate most of your design effort and a well designed system will loose much of its strengths. The room characteristics can never be removed from the listening experience and they must be considered if you are serious about what you hear. A crossover network, even if not perfectly optimized, will not introduce irregularities in the system frequency response of more than 3-4dB. Room acoustic may add as much as 15-18dB due to "room gain" factor and also cancel the acoustic output completely (well, almost) causing variations in the sound pressure up to 30-40dB. It is therefore highly desirable to understand tradeoffs and risk factors involved in evaluating the last component in the audio chain - the listening room. There are "rules of thumb" and common knowledge about placing the loudspeakers, but when it comes to showing the actual pressure distribution within the room and predicting room modes (resonances) for complex shape rooms, the situation is quite different. Among many "risk" factors, the following three seem to stand out: (1) wrong positioning of the loudspeakers, (2) wrong location of the listening position and (3) poor room acoustics. As a minimum solution to the above problems one would consider avoiding (1) and (2) and optimizing (3). The remainder of this chapter will proceed along this idea, keeping complexity to the introductory level.

Mid-frequencies Problem

From the theoretical point of view, below 300 Hz, the average listening room must be considered as a resonant cavity. In this frequency range, reflections result in standing waves and the room becomes a resonating chamber. Above 300Hz the Ray-Tracing Model becomes more useful. Here, we assume that the ray's incident angle is the same as the reflection angle. In the home situation, the shape and size of the listening room are already fixed. For the purpose of reviewing the issues associated with room acoustics in mid-frequency range, we would like to start with a simple case depicted in Fig 14.8. Let's assume that there are only two audio paths arriving at the test microphone.

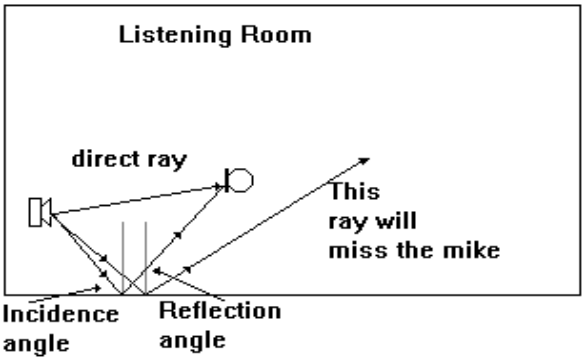


Figure 14.8 Incident and reflection angles.

If the reflecting surface is hard, there is no phase change between incident and reflected rays. Please note that the two angles are equal, so not every ray will reach the microphone. Figure 14.8 depicts the case, where two rays are combining at the microphone: (1) direct ray and (2) single ray reflected from the floor. The difference in length between these two paths is 10 cm. The 10 cm distance happens to be half of the wavelength (180 deg shift) of 1700Hz sinewave. If we add two sinewaves of the same frequency and shifted 180 deg, the result should be total cancellation. Indeed, this is clearly depicted in Fig 14.9. The first null occurs at 1700Hz followed by a +6dB peak at 3400Hz. At 3.4kHz the 10cm distance is equivalent to 360 deg phase shift, so both waves combine constructively. Next, another null occurs at 5100Hz, as the phase shift over 10cm distance equals 360+180 deg (phase reversal) and so on.

Adding More Rays

Adding more paths, each one delayed by a different amount, produces the effect depicted in Fig 14.10. In this example we have 8 rays added together - one direct and seven delayed. All delayed paths are 100 % reflected, there is simply no absorption in the reflecting surfaces. The resulting frequency response is heavily distorted. Similar analysis can be performed for any location inside the listening room. Are all the reflections necessarily bad?. Imagine that you have some perfectly absorbing material placed on all room surfaces and there are no reflections whatsoever. This is basically the situation in the anechoic chamber - the room is acoustically "dead". Most people would discard such a room for listening pleasure. A solution seems to be somewhere in-between those two extremes.

Most distortions of the curve depicted in Fig 14.10 are attributed to early reflections, which would still have sufficient amplitude to compete with the direct ray. These are the first bounce reflections from the loudspeakers off the nearest room surfaces to the listening positions. Therefore these early reflections should be looked at first. You could determine the paths of the reflected rays using the graphical method discussed before. The reflection points are then covered with an absorbing rugs or acoustical tails - depending on the required absorption.

The next step is to look at the lateral reflections (reflections from the side walls). It is recognized that lateral reflections contribute to the image of the sound stage and spaciousness. It is therefore desirable to keep the lateral reflections only partially absorbed by the reflecting surfaces. This situation is depicted in Fig. 14.10, where three rays representing early reflections are 90% absorbed, two lateral reflections are 50% absorbed and the back wall reflections are 75% absorbed.

Professional literature lists absorption coefficients for a number of common building and decorating materials, so we advise you to consult the books on this matter. Absorption capability of the materials are heavily frequency dependent and are best in the mid-to-high frequency range. Porous materials, like foam and fiberglass, are very effective at mid and high frequencies, but loose efficiency at low frequencies. Therefore the above model and discussion is only valid for frequencies above 300Hz.

Please consult Chapter 15 for more information on Absorption Coefficients.

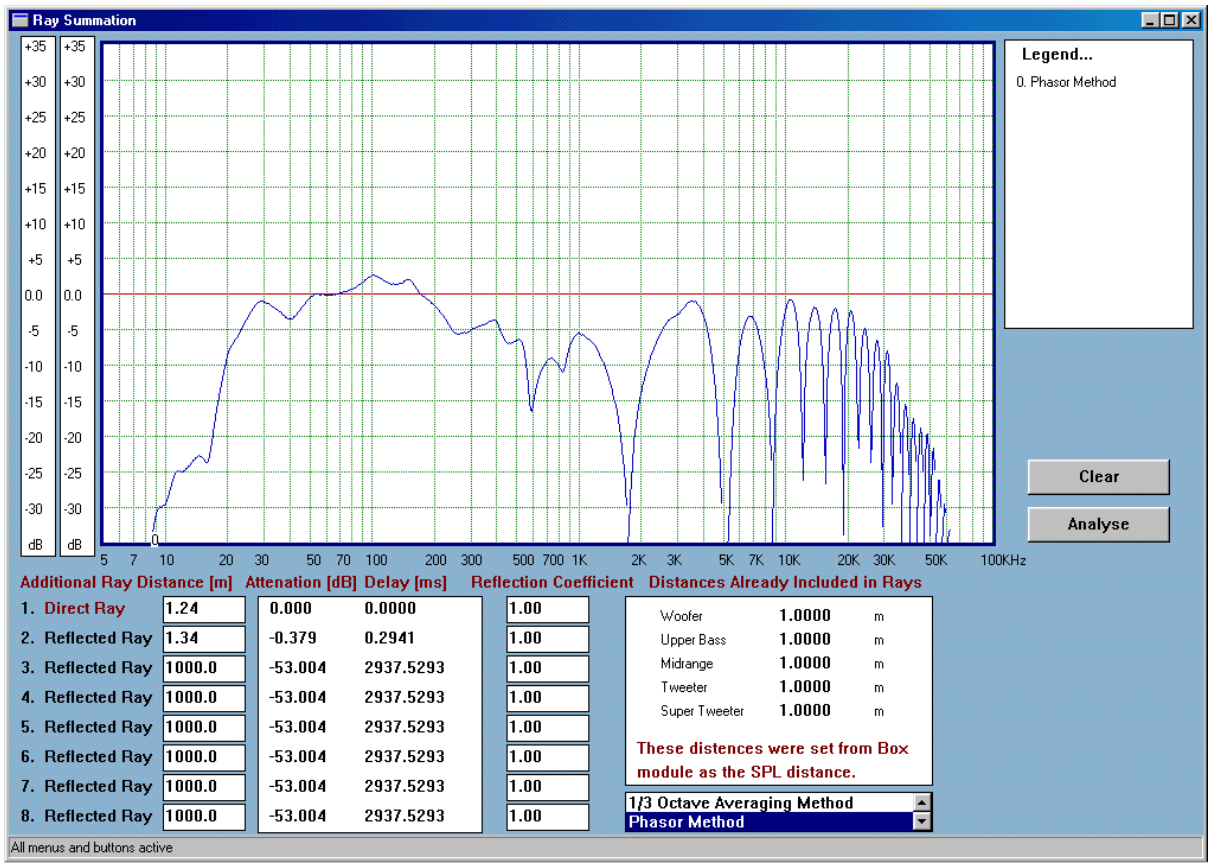


Fig 14.9 Two rays added – path difference 10cm

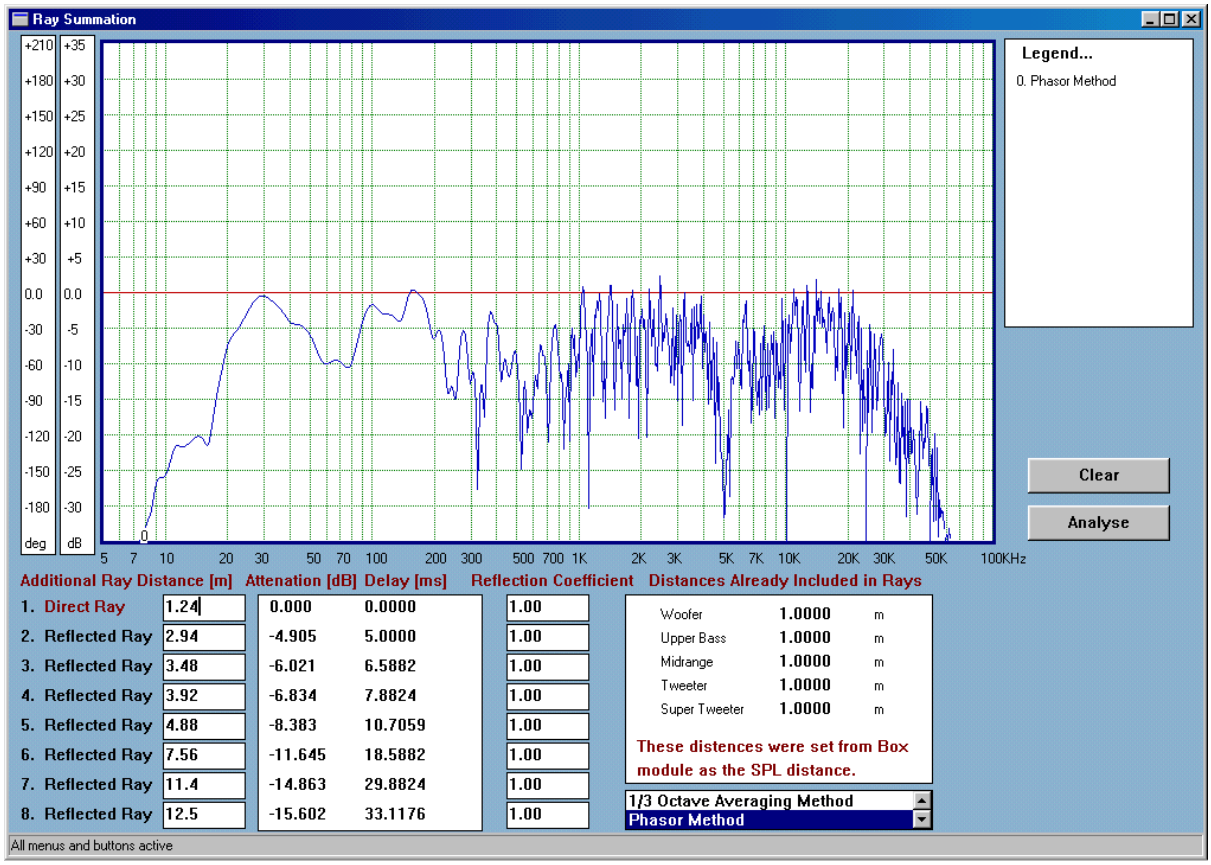


Fig 14.10 Eight rays added with various delays and 100% reflection

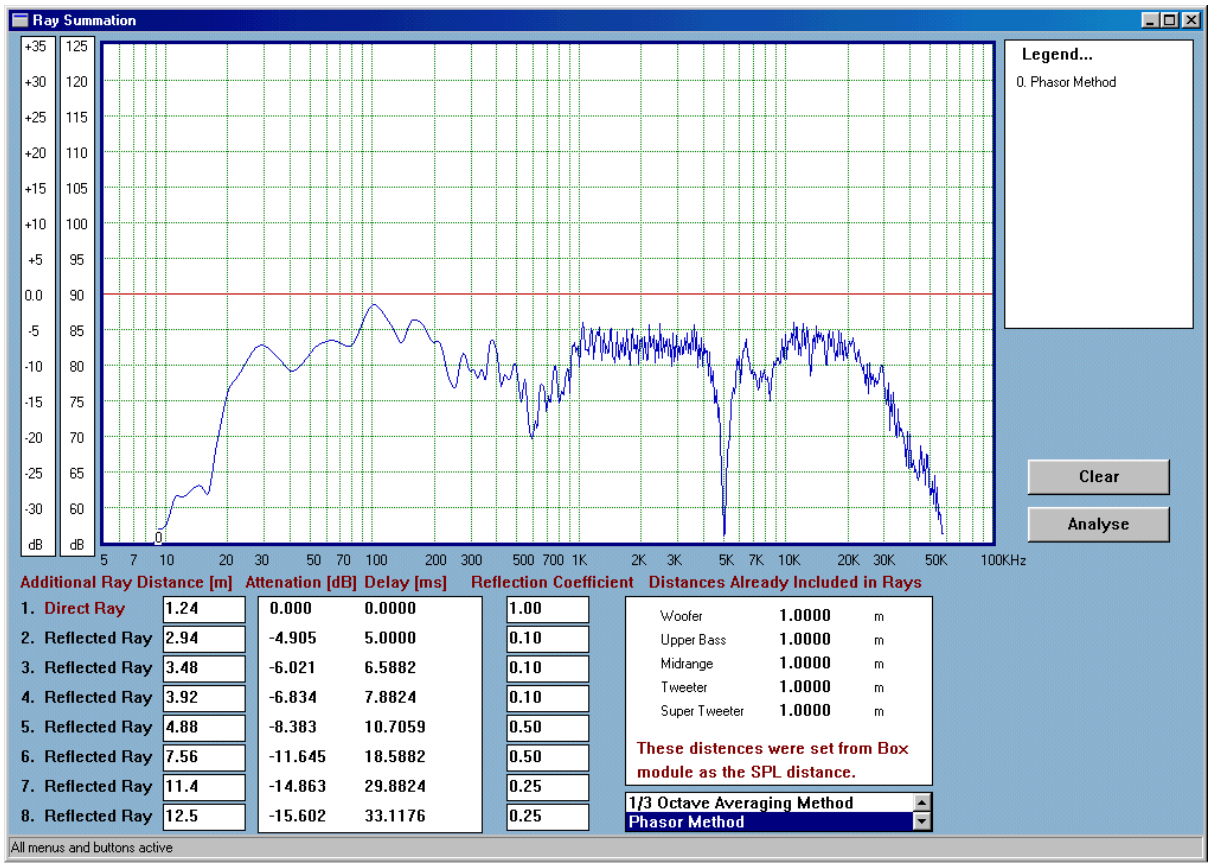


Fig 14.11 Eight rays added with reflection coefficients

The main purpose of the exercise above was to model summation of up to 8 audio rays on frequencies of 300Hz and above. Each ray may have different propagation path, with two parameters associated with it : (1) length, translated into attenuation expressed in dBs and phase change and (2) reflection coefficient, as the ray may be bounced off partially absorbing surface. The summed response of 8 rays is then plotted on the screen.

Room modes.

Listening room acoustics at the low end of the spectrum tends to be one big compromise, especially in small rooms. Standing waves are a fact of life in acoustically untreated rooms, and this will of course degrade the performance of a well designed loudspeaker system. Alternatively, installation of a loudspeaker system capable of energizing lowest room modes may well reveal that the room acoustics are less than perfect.

Room modes (natural resonant frequencies) occur across the entire frequency range. Professional treatment of the modes involves “bass traps” - typically a large absorbing element located in corners of the room. This idea may be readily applicable to a listening room in a recording studio, but for the rest of us it may be difficult to get all members of the family to agree to it. If this is your situation, there are a few things you may need to consider to avoid basic mistakes when analyzing low frequency issues of the home listening room.

Every room mode has associated acoustic pressure patterns with it. The pressure varies from 0 (null, no sound) to maximum (peak of the standing wave). When music is played in the room, the modes are excited accordingly and the pressure pattern changes and shifts dramatically within the boundaries of the room.

All modes, ie: axial, tangential and oblique, can be characterized by the following: (1) Bandwidth B - inversely proportional to the reverberation time: B = 2.2/RT60. Reverberation time depends on absorption, so the more absorption, the shorter the RT60 time and wider the mode bandwidth. (2) Decay - again, mode decay depends on the distribution of the absorbing material in the room and (3). Density - increases with frequency. Above 300 Hz room response smoothes markedly with frequency.

Let us now define the problem: you are the proud creator of a system capable of reproducing sound down to 20Hz, but when you sit down and listen to the sound, the low end around 20Hz is definitely missing. Assuming the problem is non-trivial, you are most likely sitting in the null of the pressure pattern for the 20Hz mode. The importance of knowing your room modes and associated pressure distribution is now becoming clear.

Mathematically, if the room has a simple rectangular shape an elegant formula can be used to determine room natural frequencies and some more work is required to determine pressure distribution. If the room has complex shape, the problem involves solving Helmholtz equation in 3-dimensions, looking for eigenvalues and eigenvectors. This process is handled very well by Finite Element Method (FEM) approach. If necessary, the FEM can be quite accurate. The accuracy depends on many factors and some of these are: the size of the element used, and order of approximating functions. The compromise on the other side involves computer processing speed and memory requirements. Therefore, in this example, we decided to accept errors of less than 5%, understanding that we deal here with an “approximate” method.

Digital Room Equalizer

Digital Equalizer provided with the CAD section of the program is capable of "non-minimum phase" equalization of the loudspeaker system, and therefore, correcting amplitude and time domain delays of 5-way loudspeaker system. An extension of this concept is equalization of the "speaker-to-ear" room transfer function – the Digital Room Equalizer (DRE). The DRE is a "minimum-phase" equalizer, which in addition, includes a delay function as well.

At low frequencies, a typical listening room transfer function is governed by the room modes (resonances). Room resonances, behave as minimum-phase phenomena, and can be dealt with using a minimum-phase equalizer devices.

This new feature has been implemented to allow you to "deconvolve" or "equalize" areas of your listening room. The issue of room equalization has been described in many publications, and seven of them are listed below to start your creative thinking process flowing. Possibly start with item 1, as it is mostly descriptive, and then item 4.

1.**The Science of Audio** - a series of lectures by Floyd E. Toole, Ph.D. Vice President Acoustical Engineering Harman International Industries, Inc. (**easy, excellent reading**).

2. **Digital equalization methods for audio systems** (Preprint 2598), J. Mourjopoulos University of Patras, Patras**,** Greece.

3. **Loudspeaker and room correction using digital signal processing** (Preprint 2756), Colin Bean, Peter Craven, B&W Loudspeakers, Worthing, England.

4. **Practical Adaptive Room and Loudspeaker Equaliser for Hi-Fi Use** (Preprint 3336) CRAVEN, Peter G. Consultant to B&W Loudspeakers, Steyning, United Kingdom, GERZON, Michael A. Technical Consultant, Oxford, United Kingdom.

5. **Digital room equalization.** Reproduced from Studio Sound, unknown issue 1991. From M. GERZON archive. http://www.audiosignal.co.uk/Gerzon%20archive.html

6. **Automated In-situ Frequency Response Optimisation of Active Loudspeakers.** Andrew Goldberg, and Aki Mäkivirta, Genelec Oy, Olvitie 5, 74100 Iisalmi, Finland.

7. Listening Test Results from a new Loudspeaker/Room Correction System. Lars G. Johansen (1), Per Rubak (2) (1) Aalborg University, Institute of Electronic Systems, (2) Engineering College of Aarhus, Denmark. Audio Engineering Society, Convention Paper 5323.

Authors of reference (7) summarized at the end of their presentation: *" Our conclusions are quite optimistic. Given we are operating with an acoustically fine and well damped listening room, superb loudspeakers, and a nearly optimal positioning of loudspeakers and listener, it must be expected to be a hard task further to improve reproduction quality by electronic means. Yet, our results show that practically oriented correction techniques do pay off even under these conditions. Measurements and correction design using less optimal listening rooms and cheaper loudspeakers show that our design system works equally well as in the near ideal scenario – revealing even greater objective improvements."*.

Room equalization involves knowledge of psychoacoustic and general/traditional room acoustics expertise. Room equalization cannot perform miracles, and is not a panacea for badly reverberating rooms with long decay times. Possibly the best results are obtained by employing some common sense acoustical treatment in the room together with DRE techniques. The improvements obtained are generally quite noticeable.

The goal of room equalization is to improve the perceived quality of sound reproduction in a listening area. The room transfer function is position dependent, therefore perfect equalization within a reasonably large listening area is unlikely, and even an acceptable equalization is typically a far from a "flat line" SPL curve.

It is possible that equalization even degrades the sound in larger listening area if it is based only on a single point measurement and without trough's correction. The frequency response in nearby positions can actually become worse (more irregular) after the equalization designed using only a single point measurement is applied. A well known method to prevent this undesirable outcome is to use an average of SPL responses measured within the listening area.

The actual DRE process involves using MLS measurement system to capture averaged room IR and obviously the DRE screen itself. It is therefore a non-trivial process and you are well advised to take small steps on your learning curve.

Practical tips on Room Equalization

1. We want the equalizer to flatten the "in-room" frequency response, but not at the expense of boosting dips or notches in the frequency response to the point where the boost causes amplifier and speaker overload or massive amounts of the boosted frequency at other listening positions. Fortunately, the ears are much less sensitive to the odd dip in a frequency response than to peaks of similar amplitude. We can take advantage of this phenomenon and be sensible about the boost associated with the troughs. Therefore, DON NOT attempt to fully equalize the troughs or dips in the original room transfer function. The program runs the RDE screen in **"Pixel Editor"** mode, so you can actually "draw" on the screen and correct the measured frequency response by smoothing the dips. By doing so, the "inverse" filter will not have peaks in it's frequency response.

2. Room equalization needs to be of minimum phase type to avoid problems with audible pre-responses when cancellation of the room response is imperfect. One of the remarkable things about minimum phase filters is that the overall frequency and phase response of the filter can be computed, by the HBT, just from a knowledge of the amplitude frequency response alone. In other words, if one knows what frequency response one wishes to correct, the phase response aspects of the filter follow automatically. As you can see, the whole equalization process is performed in the frequency domain rather than in the time domain. This allows you to correct the measured room frequency response to partially eliminate dips in the response.

3. Frequency response in higher frequency range (the statistical region) should be smoothed out by applying smoothing at capturing room IR stage. Typically, the SPL in the statistical region just resembles "noise", and needs to be smoothed out before application of the DRE. Standard acoustical treatment of the room is also very helpful in this frequency range. Initially, you can simply "switch off" this region from DRE action. This is accomplished by applying a flat LP tail, starting from say 500Hz (or lower) to the measured room SPL. Since the measured SPL is now flat above 500Hz, the DRE filter will also be flat – no room correction above 500Hz. Gradual experiments into the statistical region would allow you to determine how much benefit you can gain in mid-high frequencies range using the DRE. The benefits are likely to be tied to individual room's acoustics.

4. Another advantage of being able to derive minimum phase equalizers just from a measurement of the amplitude of the frequency response is that you can measure the SPL response at several points across a listening area, and equalize for the averaged frequency response at all these points, rather than for the frequency response measured at just one point. This gives better results across the whole listening area, at the expense of less good results very near one single point. In typical domestic listening rooms, averaging for six points across, say a sofa, can give useful room equalization across the listening area.
   One advantage of averaging the measured response at several room points is that this tends to "fill in" measured dips in the frequency response, so you have less headache, trying to decide how much of the dip you need to leave untreated.

5. It is left entirely to your discretion to decide where to, on the frequency scale, apply tails for the HBT. Just a few points to remember:

   - Application of tails should not result in frequency response with sharp "corners". This type of frequency response is mathematically correct, but is not natural to realizable audio components.
   - Do not attempt to create bass boost or treble boost in the overall DRE filter frequency response, especially at the low/high ends of the frequency range.
   - Neutral gain (around 0dB) will be accomplished if the measured room SPL is around 90dB. Please monitor the "In" and "Out" LED level indicators during the DRE operation.

If you are a “seasoned acoustician” you should be able to run DRE with filtering function averaged over 5-10 locations and show the results of the DRE operation as a Cumulative Spectral Decay plots.

If you are new to this technology, you may want to initially concentrate on single measurement approach to the design of the inverse filter DRE and verifications measurement. This way, it may be easier for you to grasp all issues involved.

Description of the DRE Process

The complete DRE process is broadly described below. Some of the steps are quite involving and will be detailed in the paragraphs that follow.

1. Select what DRE channel (Left / Right / Both) is being measured by “checking” the box in DRE control dialogue.
2. Make spatially averaged SPL measurements in your listening area, for Left / Right or Both channels. Use MLS to perform the measurements and averaging.
3. Transfer the averaged SPL into DRE using the Curve Arithmetic Buffer.
4. Remove excessive troughs / dips from the averaged SPL using DRE Pixel Editor screen.
5. Select HBT tail frequencies for low- and high-end of the frequency range. This may be different for each channel.
6. Select the desired slope of HBT tails, so that the final DRE filter has flat or rolling slopes at low- and high frequency extremes. Again, this may be different for each channel.
7. Plot all relevant DRE curves for visual inspection.
8. Run DRE.

Extensions to this process involve modifications to the target filter transfer function by means of tilting the whole filter or adding ripples to the transfer function. It needs to be emphasized here, that the DRE process is a “challenge”.

We recommend, that you first practice without any amplifiers/speakers connected to the DRE, just to make sure, that you are comfortable with the process.

Making average SPL measurements.

You can use the MLS screen to make all room IR measurements. However, the process commences by selecting **Left / Right OR Both channels,** for which room transfer function is being currently measured. This selection is performed from the DRE control box, under Channel column. Please note, that you can move the results of Curve Arithmetic from the **Buffer** to the **Left / Right / Both channels of the Digital Room Equalizer**. First, please select where you want the results to go, by selecting appropriate check box on the **DRE** control dialogue box. Then select **“Save Buffer”** to **Curve 0**, on the MLS “Post-Processing” tab. This actions will save the Buffer content into driver data space (Curve 0) and also into the DRE data space, as requested from DRE dialogue box.

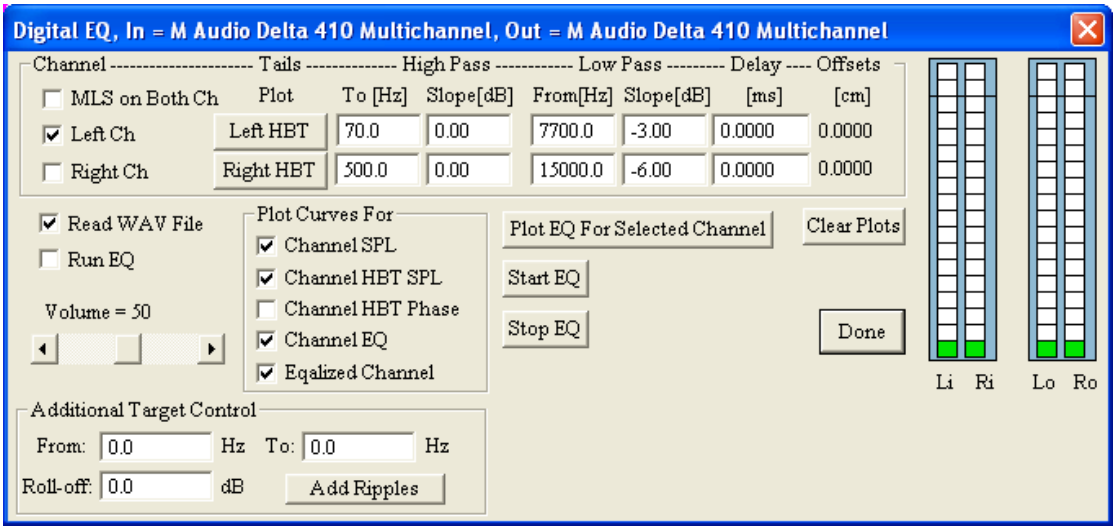


Figure 14.12 DRE control box.

1. Measure SPL at several (5-10) nominated locations. It would be recommended to save each individual measurement as IR-file (check **"Save Input Buffer"** box in the **"Acquire IR"** TAB) for future use. It is also useful, to label each measurement as a file name when saving the IR to a file. This will allow you to "mix-and-match" locations at will for different average locations at later stage.

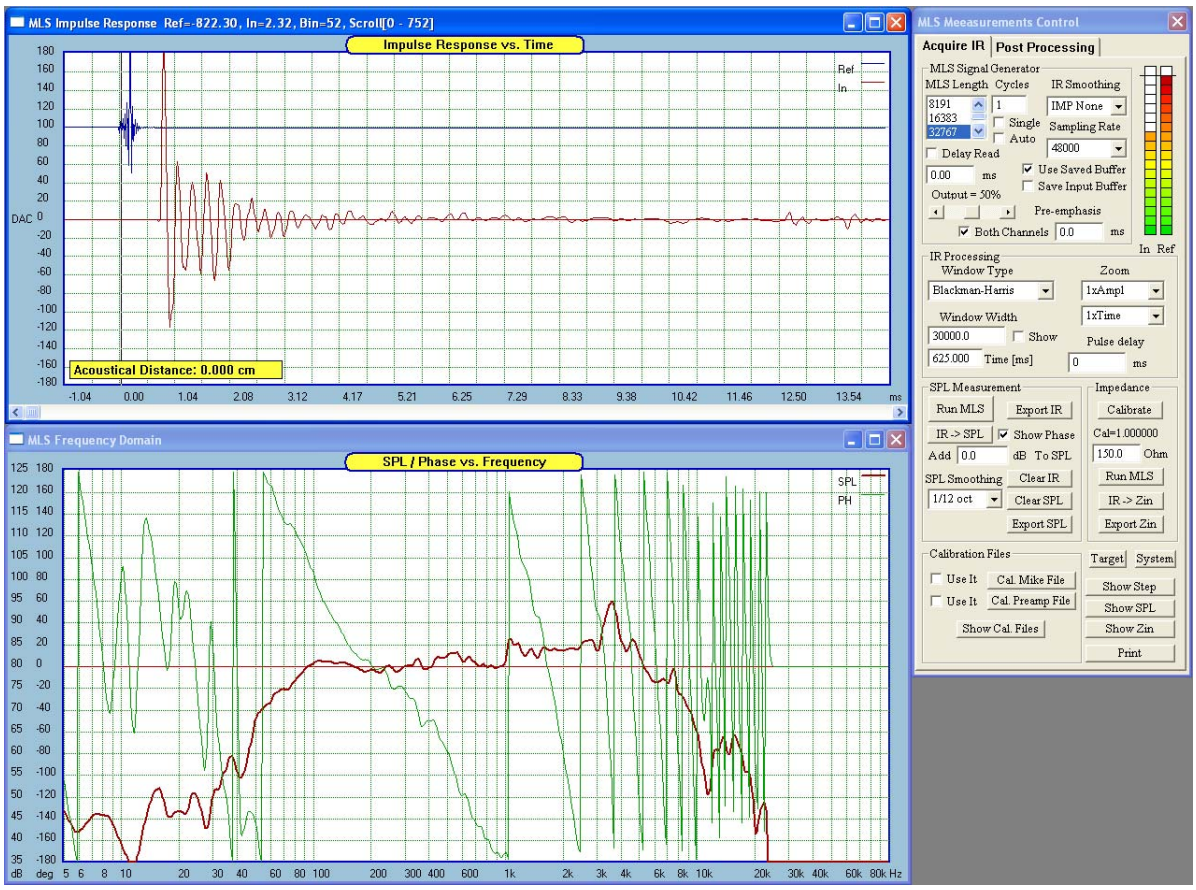


Fig 14.13. MLS is used to measure room transfer function

2. Load each IR file into MLS screen and perform **"IR->SPL"** transformation. Store each resulting SPL curve into the five **"Post-processing"** locations. When all five curves have been processed, perform "Add" operation. If the resulting "averaged" curve ended up to high on the SPL scale, "Subtract" the required number of dB's from it. Up to five files can be included in the averaging process at one time, so the averaged curve may end-up being 13-14dB higher. You can average more locations by performing the above process sequentially.

   <u>You should aim at having the spatially averaged SPL curve around 90dB level. This is a 0dB gain for the DRE.</u>

3. The results need to be moved into the "driver" data space (from Buffer to curve 0) in order to show up in the DRE screen. The rest of the operations can be performed from the DRE dialogue box.

Removing troughs/dips.

When applied to the DRE, the concept of the "inverse filter" simply means "mirror image" of the measured SPL. The measured SPL dips become filter's peaks and vice-versa. The obvious problem here is, that the 30-40dB cancellation troughs end-up as massive peaks and will damage the components of your audio chain. Clearly, the dips need to be reduced.

When you open the DRE window, the screen is automatically set to **"Pixel Editor"**mode. You can actually "draw" on the screen and correct the measured frequency response by smoothing the dips. By doing so, the "inverse" filter will not have peaks in it's frequency response. As a first approximation, you may aim at having the deep peaks reduced to 5-6dB shallow, rounded dents in the measured SPL curve.

The process of “drawing” the SPL using the Left Mouse Button and cursor keys has been described in early chapters of this manual. An example of an undesired dip in the averaged room transfer function is shown below. Here, a large dip at 200Hz (pink curve) will result in corresponding peak in the DRE filter response (green curve).

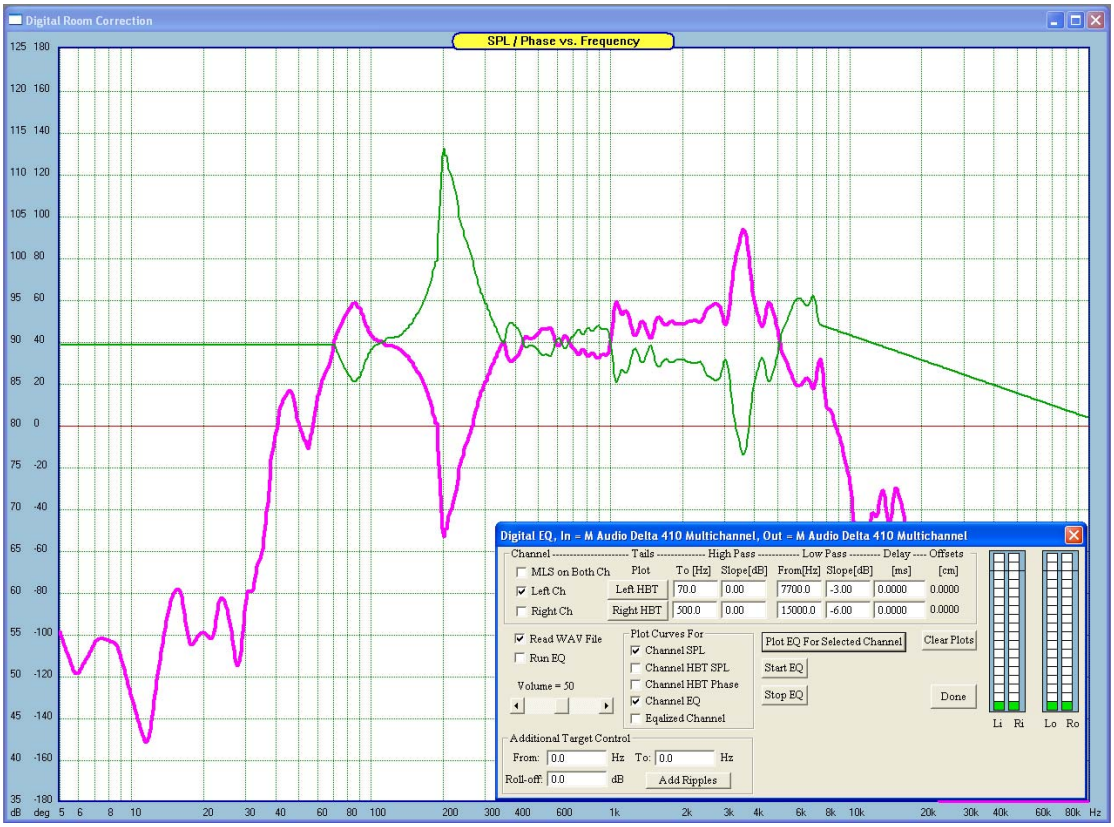


Fig 14.14. Undesired dip in the averaged room transfer function

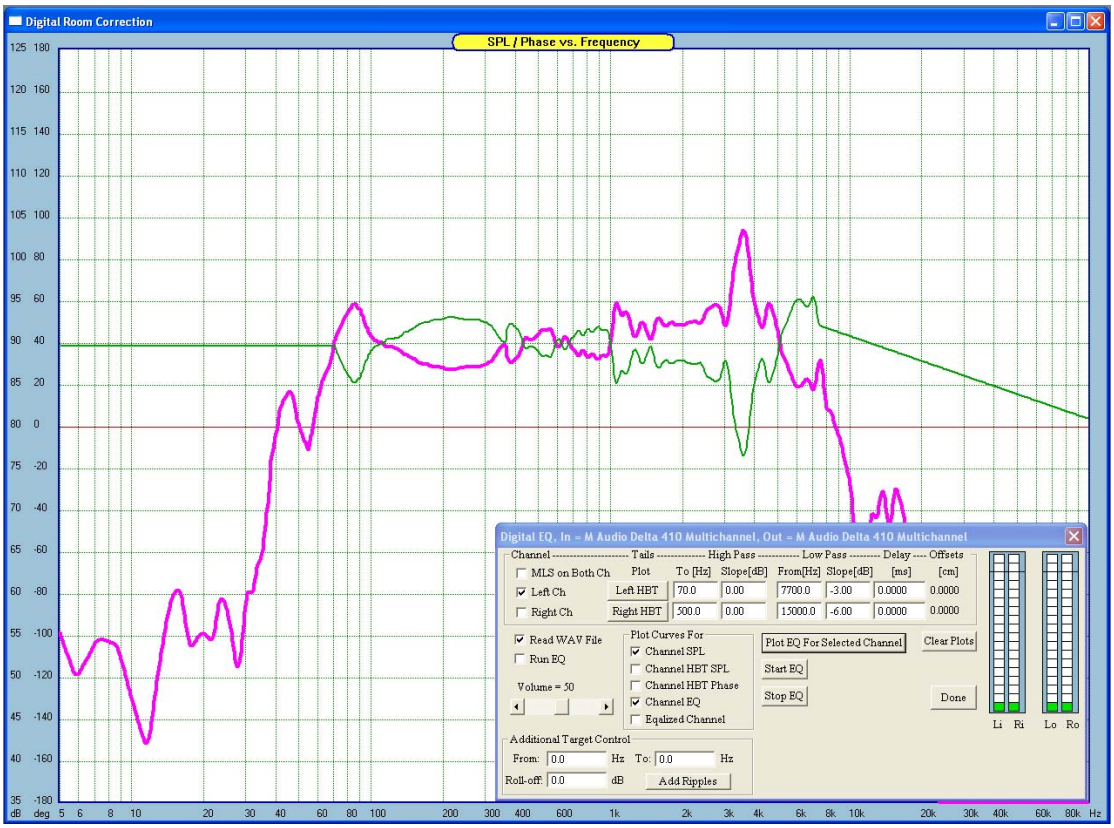


Fig 14.15. The dip is removed from the averaged room transfer function

However, if the dip is removed from the averaged room transfer function (pink curve), the peak on DRE filter response also disappears – see Figure 14.15 above.

Selecting HBT parameters

Selecting the HBT parameters is quite an important issue. Let's assume, that the spatially averaged room transfer function looks like the pink curve on Figure 14.16 below.

We know already, that anything above 90dB line will be attenuated and anything below 90dB line will be amplified by the DRE.

Upon visual examination of the pink curve, we can determine, that we can gain a bit of bass EQ boost at the low-end (about 6dB), if the HBT high-pass filter (the "To" frequency) is selected at the short leveled section of the pink curve at around 70Hz. This is because this section is about 6dB below 90dB.

The slope for high-pass filter is selected to be flat (0dB). This way, all frequencies below 70Hz will be amplified by the constant 6dB factor. Please do not attempt to select positive slopes (positive value in this data field) here. This would result in DRE trying to unnecessarily boost the lowest end of the frequency spectrum. It would be advisable to enter negative value here to allow the DRE filter to roll-off towards the frequency extremes.

On the high-side of the frequency range, we would like the DRE to cancel the high peak at 3.5kHz. Therefore, the HBT low-pass (the "From" frequency) is selected as 7.7kHz. This is where a small peak is residing.

The slope for low-pass filter is selected to be rising -3dB (yes, negative value). This way, all frequencies above 7700Hz will be attenuated by the constant -3dB factor in the final DRE filter response. Please do not attempt to select positive slopes (positive value in the data field) here. This would result in DRE trying to unnecessarily boost the high end of the frequency spectrum. It would be advisable to enter negative value here to allow the DRE filter to roll-off towards the frequency extremes.

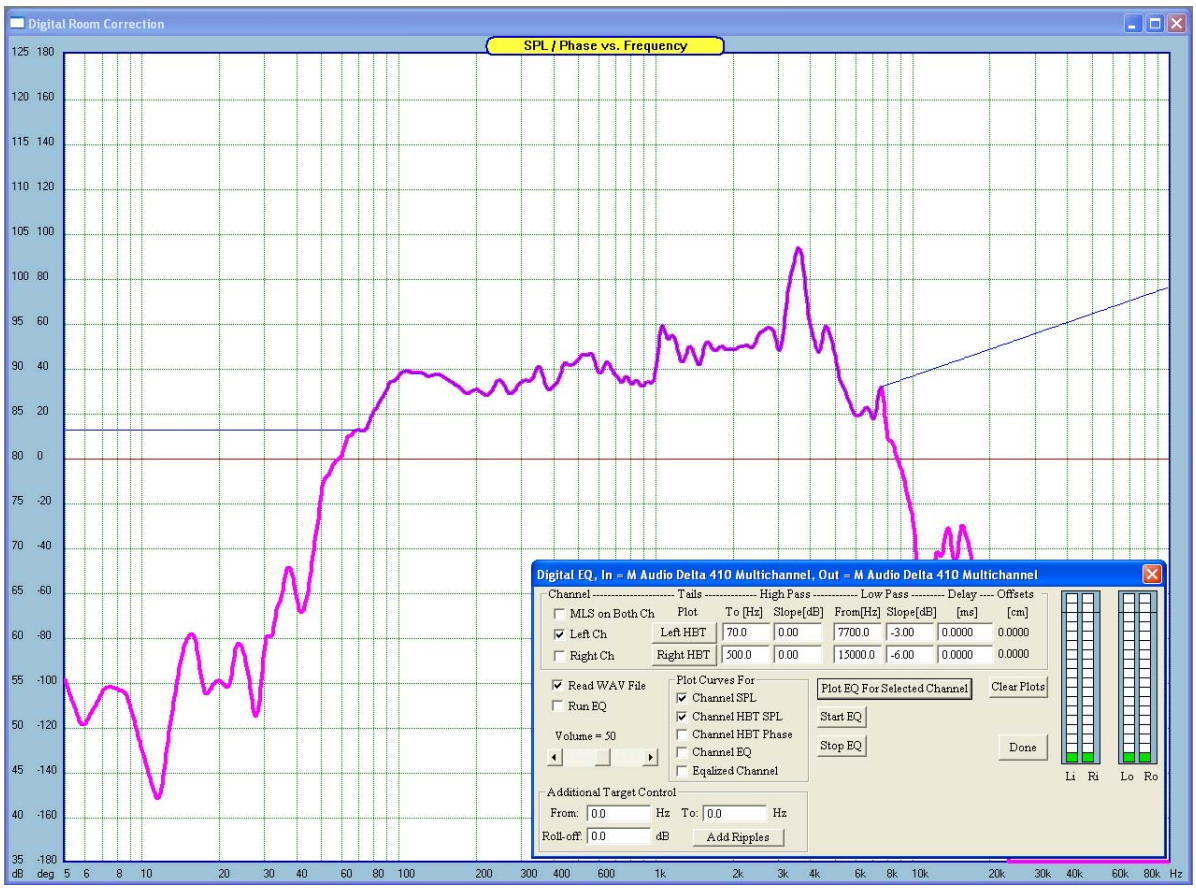


Fig 14.16. Example of HBT (blue plot).

Now, we are ready to confirm the above assumptions and plot the plot the HBT(blue curve) + DRE (green curve) filter responses – see Figure xx.xx below. It is observable, that the DRE filter is a mirror image of the HBT response, centered around 90dB level line.

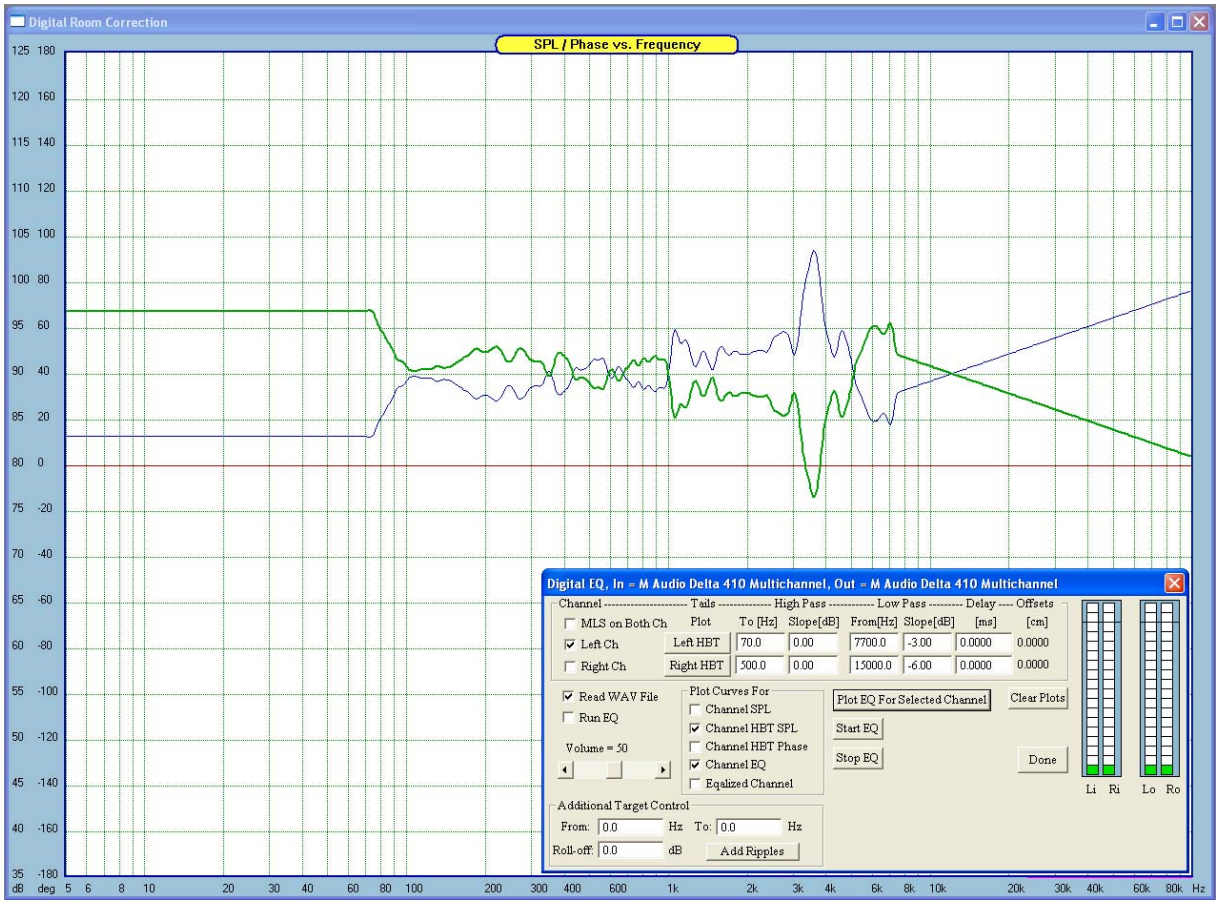


Fig 14.17. DRE (green curve) centered around 90dB.

Finally, we can review the TOTAL frequency response of the DRE + room after equalization. This situation is depicted on Figure 14.18 below.

Please view Figure 14.18. The pink curve is the "un-equalized" room transfer function. The thin black curve is the corresponding, equalized room transfer function.

The example presented in this paragraph showed fairly smoother low-end room transfer function. There was no significant dips, therefore we did not have to use pixel editor to account for the excessive dips. Therefore, the final, simulated, equalized room transfer function looks very smooth and even within the pass-band. Please also note, that we have gained bass extension from 100Hz down to 65Hz. Real-life results will be compromised by the need to account for the dips, so the final equalized room transfer function will not be a flat line.

Fig 14.18 Equalized (black curve) room transfer function

DRE Controls

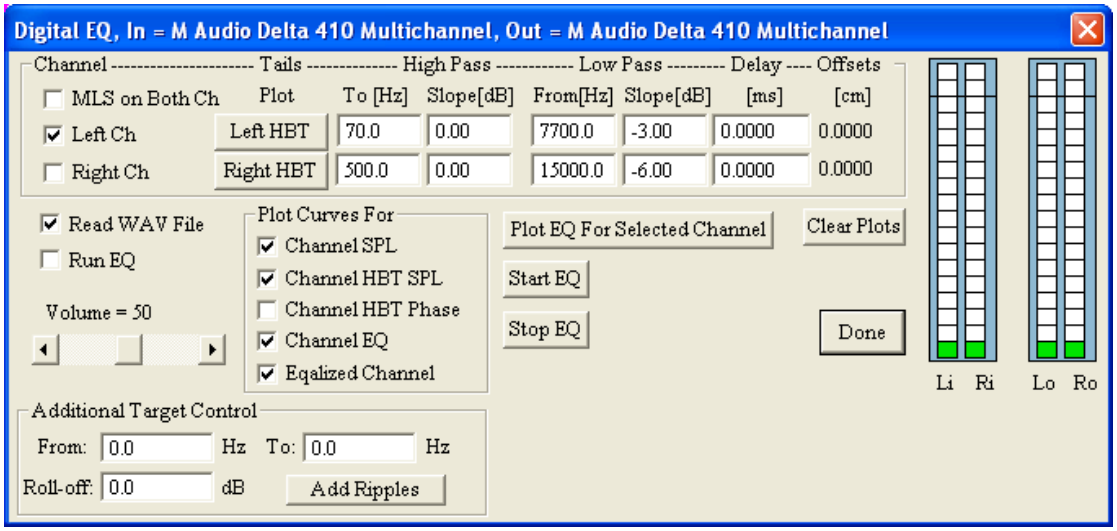


Fig 14. 19 DRE controls.

Channel group

1. **"MLS on Both Ch"** – check box to select the averaged SPL room transfer function measurement from MLS to be stored in both channels.
2. **"Left Ch"** - check box to select the averaged SPL room transfer function measurement from MLS to be stored in the Left Channel.
3. **"Right Ch"** - check box to select the averaged SPL room transfer function measurement from MLS to be stored in the Right Channel.

Tails group

1. **"Plot Left HBT"** – button to plot left channel HBT.
2. **"Plot Right HBT"** – button to plot right channel HBT
3. **"To"** – data field to enter cut-off frequency for high-pass section of HBT for left / right channels.
4. **"Slope"** – data field to enter slope speed in dBs, for high-pass section of HBT for left / right channels.
5. **"Delay"** – data fields to enter delays for left / right channels.
6. **"Offsets"** – displays delay data converted to centimeters.

Additional Target Control group

1. **"From"** – data field to enter starting frequency for the roll-off.
2. **"To"** – data field to enter ending frequency for the roll-off.
3. **"Roll-off"** – data field to enter steepness of the roll-off.
4. **"Add Ripples"** – button to activate control box for entering 3 ripples onto the target DRE curve.

Plot Curves group

1. **"Channel SPL"** – check to display averaged SPL room transfer function
2. **"Channel HBT SPL"** – check to display HBT SPL of the room transfer function
3. **"Channel HBT Phase**" – check to display HBT Phase of the room transfer function
4. **"Channel EQ"** – check to display DRE transfer function
5. **"Equalized Channel"** – check to display DRE transfer function + averaged SPL room transfer function.

"Read Wav File" – check box to select signal source as WAV file directly from the HD.
"Run EQ"- check box to ENABLE / DISABLE the DRE.
"Volume" – slider to adjust output volume.
"Plot EQ For Selected Channel" – Button to plot any of the selected curves.
"Start EQ" – Button to start DRE play.
"Stop EQ" – Button to stop DRE play.
"Clear Plots" – clears the display.
"Done" – closes the DRE.
"Li / Ri" – Input level indicators for Left / Right channels.
"Lo / Ro" – Output level indicators for Left / Right channels.

Chapter 15. Finite Element Analysis

The FEM.

Before we proceed further, a few words of explanation on the FEM are needed. This concept has been around for several decades and lends itself particularly well to solving Helmholtz equation for complex volume shapes. Within a volume V, enclosed by a surface S, the pressure p must satisfy wave equation:

$$\nabla^2 p + (\frac{\omega^2}{c^2})p = 0 \qquad \text{c = speed of sound, } \omega \text{=frequency of vibrations}$$

On the hard surface, S, the normal velocity $\partial p / \partial n = 0$. The solution to the above problem, expressed by an equivalent variational principle is:

$$\delta \int_V \frac{1}{2}[(\nabla p)^2 - (\omega^2 / c^2) p^2] dV = 0$$

The volume in question is divided into a large number of smaller elements, each having several nodes. We concentrate on an 8-node (8-corner) "brick" element with linear shape functions approximating pressure distribution within the element. The bricks are placed together to approximate the required volume and shape of the room. The solution of the problem is expected to assign each node a pressure value for every room mode (eigenvalue). When shape functions of the element are introduced into the variational principle, the equation is reduced to matrix eigenvalue problem:

$$[K - \frac{\omega^2}{c^2} M]\{p\} = 0$$

K is the "stiffness" matrix, M is the "inertia" or mass matrix. Now the matrix equation can be solved by standard eigenproblem methods.

Generally, the user of an FEM program is dependent on the detailed knowledge about theories, algorithms and assumptions behind the program for the proper selection of models and algorithms. The FEM knowledge base is huge and readily available, but the immediate question is: do you need to become an FEM expert if you only need to analyse your room acoustics once?. Please read on.

Analyzing FEM output

Note: SoundEasy offers two FEM screens for more complete analysis of your room acoustics and enclosure driving point impedance. In order to save memory, the screens share same of the data space. We therefore recommend, that you perform analysis on one screen and SAVE results of your work to disk before opening the other FEM screen. If you start new analysis, please ALWAYS use "Clear" button on each screen to ensure consistent starting point for the FEM process.

In general, the loudspeakers tend to activate those modal frequencies, which have partial or full maximum at this particular location. Conversely, room modes that have null at the loudspeaker location can not be energized. Because some absorption and transmission losses are always present in the room, the modes can not develop infinite pressure patterns and will always decay when the source of excitation is removed. However, as long as there are reflections in the closed volume of the room, the modes will be present. A solution that is always recommended is to install bass traps. If the economical and aesthetic factors are on your side the problem can be solved, or at least partially eradicated.

Bass traps will greatly reduce reflections, thus preventing the standing waves phenomenon taking full effect and you will gain much more freedom in positioning the speakers. You simply kill the problem right where it starts. We have browsed the Internet for some help in this area and we can testify that there are several companies offering good solutions to this problem. Room modes that need to be looked at are typically located below 100 Hz, so you need to make sure that bass traps you wish to implement are really efficient at such low frequencies. If, for whatever reasons, the bass traps are not an option for you we suggest careful review of the modal pressure plots. From this moment onward, we are not trying to solve the problem, but rather select the "lesser evil". Good FEM simulation software usually calculates all room modes and associated pressure patterns all at once, so when this lengthy process is completed you only need to flip through the list of modal frequencies to get the pressure pattern displayed on the screen. What you looking for are the location of pressure peaks and nulls for each mode (frequency). Larger loudspeaker systems are typically made as floor standing and for purely aesthetic reasons you will not place them in the middle of the room, but rather against the wall or even in the corner. From the pressure plots, you will notice that pressure peaks also like walls and corners.

Therefore, if you end up energizing room modes, you may as well energize as many of them as possible. If the room has many modes spaced evenly, the sound field tends to be smoother than in a room with only a few well-separated modes. The latter room sounds boomy at those frequencies. Next, the pressure plot analysis should reveal the locations of pressure nulls. This is essential, as one of the issues of prime importance here is to avoid cancellation of sound. Having located the nulls, you know at least, which areas of the room to avoid for quality listening. Many of us are would rather accept some amplification of sound than cancellation of sound (and our effort). Moreover, if this particular mode is too loud you can always use electronic equalization to trim it down. The downside of living with the room modes is that the acoustic frequency response (loudspeaker + room) WILL be irregular.

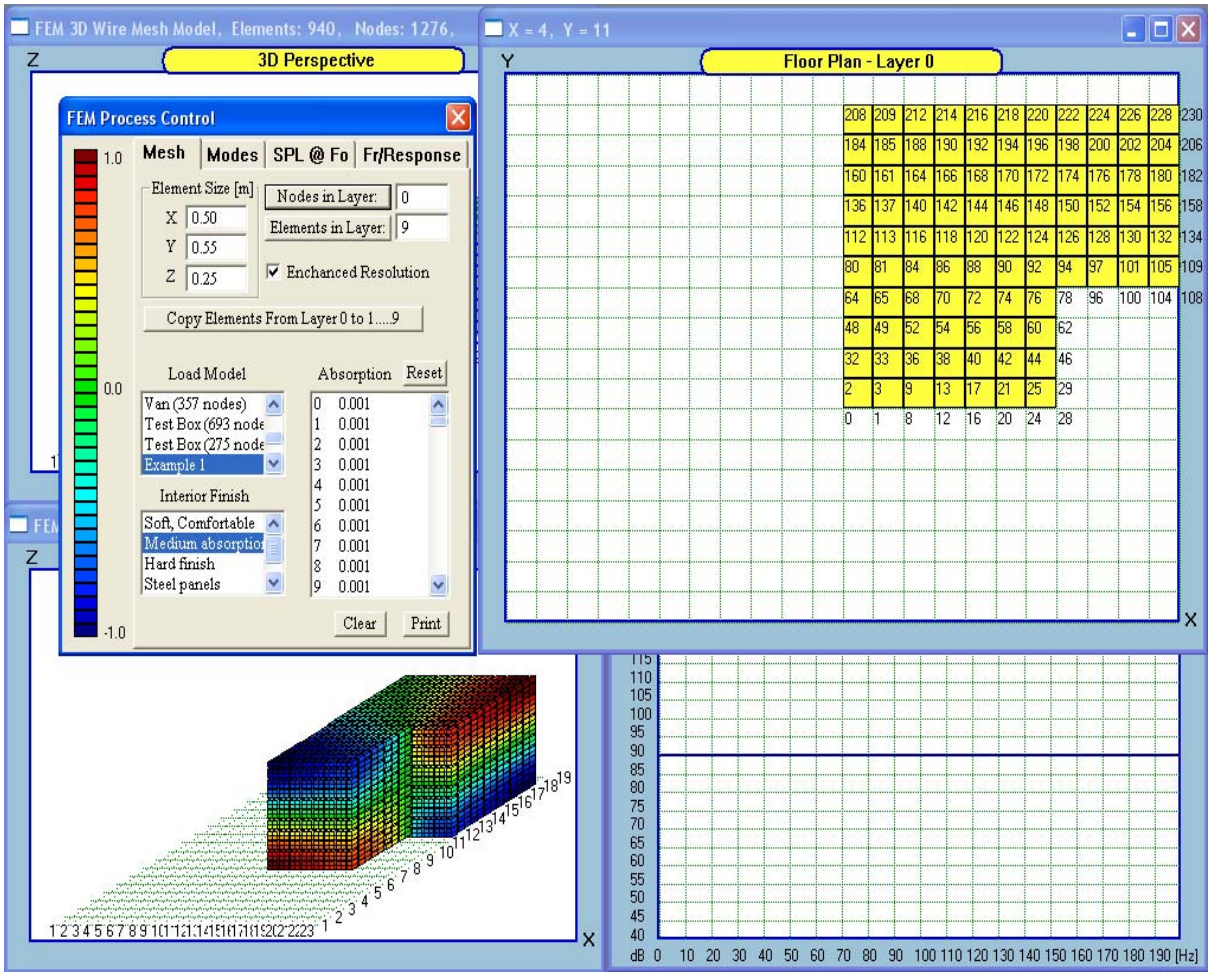


Figure 15.1 Example of a floor plan.

That is, if you avoid a pressure null at one frequency, you are likely to sit in a pressure null of another frequency. The problem here could be therefore stated in terms of what frequency range is my favorite one, so I can preserve it and which one I can sacrifice. That is, if you avoid a pressure null at one frequency, you are likely to sit in a pressure null of another frequency. The problem here could be therefore stated in terms of what frequency range is my favorite one, so I can preserve it and which one I can sacrifice.

As it is implemented, the Finite Element Method is a sophisticated tool for solving room acoustics problems. The analysis includes:

1. Creating geometrical mesh, representing the interior of the listening room. The FEM is ideally suited for complex shapes of the room.
2. Finding room natural frequencies – modes.
3. Plotting SPL distribution within the room for modes. The room can be “sliced” horizontally, so you can view the SPL of each node.
4. Finding SPL distribution for any frequency.
5. Plotting Speaker – Microphone room transfer function. Four speakers are allowed, connected “in-phase” or “out-of-phase”. This effectively gives you the analysis of two dipole woofers in room.

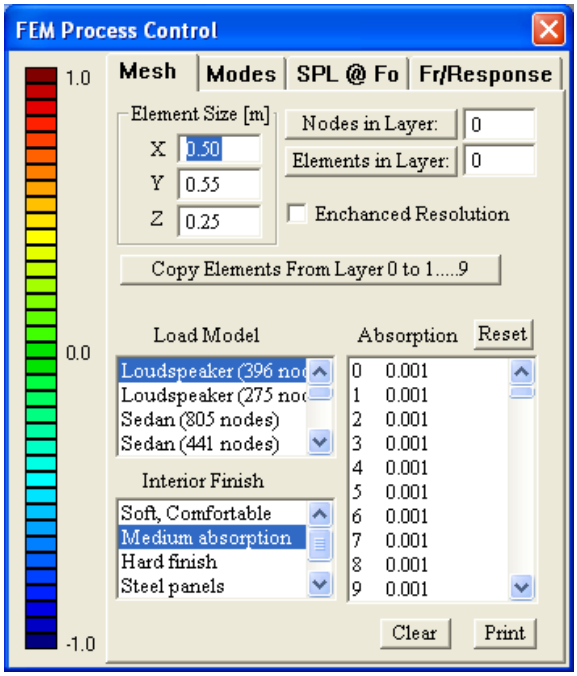


Figure 15.2. Mesh Control Tab

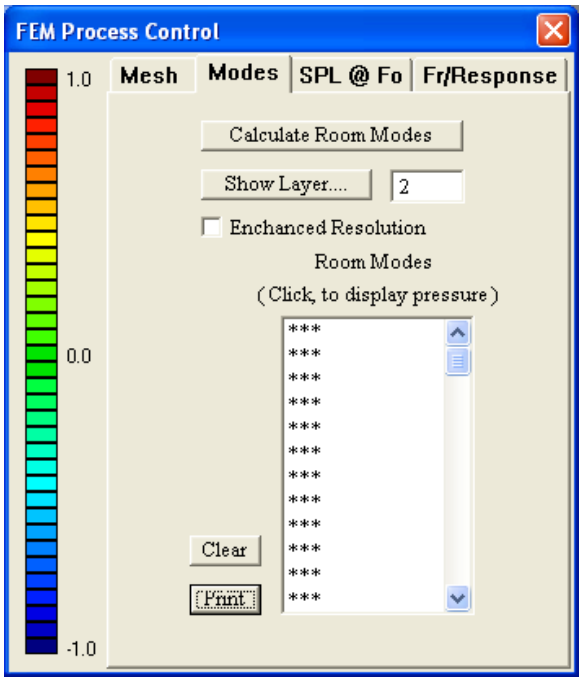


Figure 15.3. Modal Calculator Tab

Mesh Tab

You should start the FEM process by measuring your listening room and making a simple sketch of the floor plan. This will be helpful in determining the "brick element" dimensions, that you will use to "fill-in" the interior of the listening room – process called "meshing". Smaller brick element allows you to more accurately approximate the interior and model SPL at higher frequencies more accurately, however, there is a limit of 1350 element imposed, so you can not exceed this number of element in your analysis. Also, please be aware, that calculation time increases very dramatically, as you increase number of elements. You should start with the Test Box of 275 elements and see how long the algorithm takes on your computer to complete the calculations. It is often beneficial to start with larger-size elements and perform crude analysis. Then, progressively reduce the size of elements, thus increasing the accuracy of the room mesh model. You will clearly see how the number of elements affect the calculation time. The brick element dimensions are entered from **"Element Size X/Y/Z"** data fields. You can start entering mesh parameters by loading any of the built-in models (**Load Model** list box) and expanding from there, OR you can start with elements in "Layer 0" , then elements in "Layer 1", and so on, being entered into the **"Floor Plan**" window. You will see immediately the 3D structure being built on the "3D Perspective" screen. There are 10 layers of elements allowed (therefore, a total of 11 nodal layers), stacked on the top of each other to approximate your listening space. Simply enter layer number and click on **"Element in Layer**" button to edit this particular layer of nodes. You can review node numbers in each layer by entering nodal layer number and pressing the "Nodes in Layer" button. If your room happens to be rectangular, you can simply copy the bottom element layer to all other layers by pressing "**Copy Elements from Layer 0 to 1…9**" button. Average absorption within the room can be controlled by selecting "Interior Finish" option from the provided list box.

Individual node absorption can be edited from the "Absorption" list box. Simply DOUBLECLICK left mouse button on the desired node and edit the coefficient. With the help of Absorption Coefficient, you will be able to approximate opened windows, carpets, heavy drapes and many other objects made from materials of known absorption. The absorbing area is determined by the location and the number of adjacent nodes with nominated Absorption Coefficients. As a refresher, in the next paragraph, we discuss some basic concepts of Absorption Coefficient.

The **"Enhanced resolution**" button causes the SPL variations being displayed in finer mesh.

Erasing elements from the mesh is accomplished by (1) selecting the element layer and (2) pressing the Right Mouse Button above the element you wish to erase.

Modes Tab

You can calculate room modes by pressing the **"Calculate Room Modes**" button. When the process is complete, the modes below 150Hz will be listed in the provided list box. Click on any of the modal frequencies in the list box to have the corresponding SPL distribution displayed in the "3D Perspective – Solid Model" window. You can also review the SPL distribution layer-by-layer. Simply enter layer number and press **"Show Layer**" button

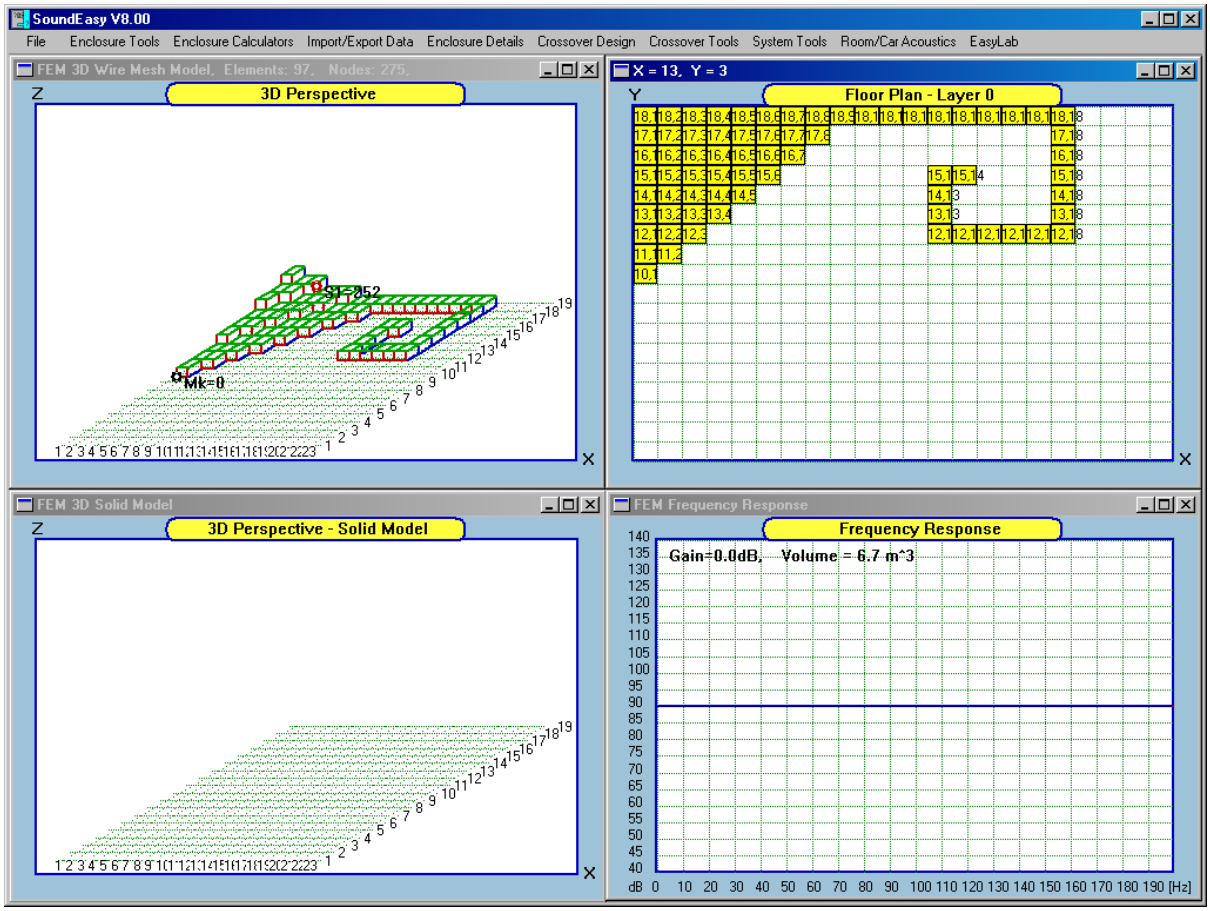


Figure 15.4. Example of building the FEM mesh model.

SPL @ Fo Tab

Sometimes, it may be beneficial to calculate SPL at frequencies other than modal frequencies. You can accomplish this using the "SPL @ Fo" tab. There are only two editable fields: frequency of interest (in Hz) and layer to be displayed after the SPL distribution was calculated.

Frequency Response Tab

The Tab has the following controls:

1. **Speaker 1 Node** - This is an editable field and is used to enter nodal location of Speaker 1. You can review node numbering by going to the "Mesh" tab. The editable field next to the "Nodes in Layer" button allows you to specify layer number you wish to inspect for nodes. Negative sign in front of the number indicates reversed phase.
2. **Speaker 2/3/4/5/6/7/8/9 Node** - Same as above. If you have only one speaker, you must enter '*' (asterisk) in this field. Negative sign in front of the number indicates reversed phase.
3. **Mike Node** - This is an editable field and is used to enter nodal location of the Listener. Again, you should review numbering of nodes as described in bullet 1.

4. **Calculate Frequency Response** - Pressing this button will activate the algorithm. The process can be quite lengthy (2-3hours), particularly for larger number of nodes. Plotted frequency response represents acoustic pressure developed within the room as generated by up to 9 speakers.
5. **Enhanced Resolution** - This check box allows you to double the resolution of the mesh approximating the listening room.
6. **Simulate** group of options:
   Flat = No other components are included
   My Room = SPL in the room with last calculated T/S-type enclosure SPL superimposed.
   My Woofer = SPL in the room with woofer transfer function superimposed.
   My System = SPL in the room with complete system transfer function superimposed.
   My Enclosure = Select to calculate Driving Point Impedance of your speaker enclosure. <u>This option MUST be selected for FEM modal analysis of your loudspeaker enclosure.</u>
   Log Scale = Switch to logarithmic scale.
7. **Clear** - Simple clears the display.
8. **Print** - Button invokes printing dialogue box.
9. **Abort** – Simply aborts plotting.

<u>Note: Frequency response range for FEM analysis is selected from Preferences screen same as Box range.</u>

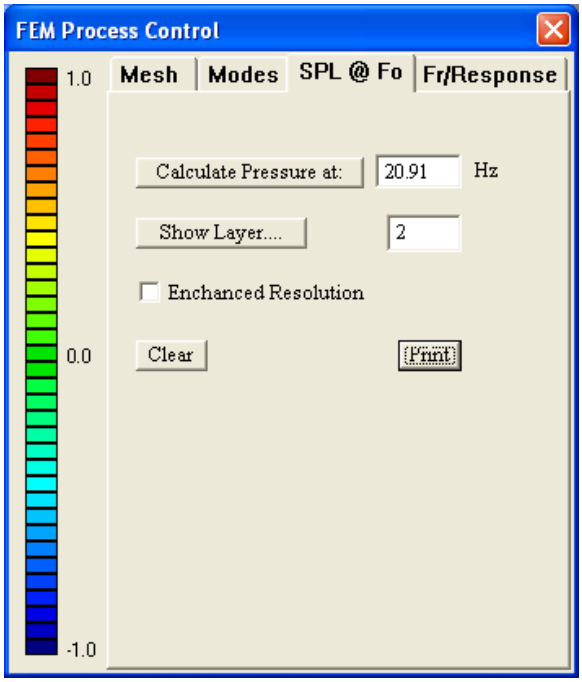


Figure 15.5. SPL @ frequency Fo Calculator

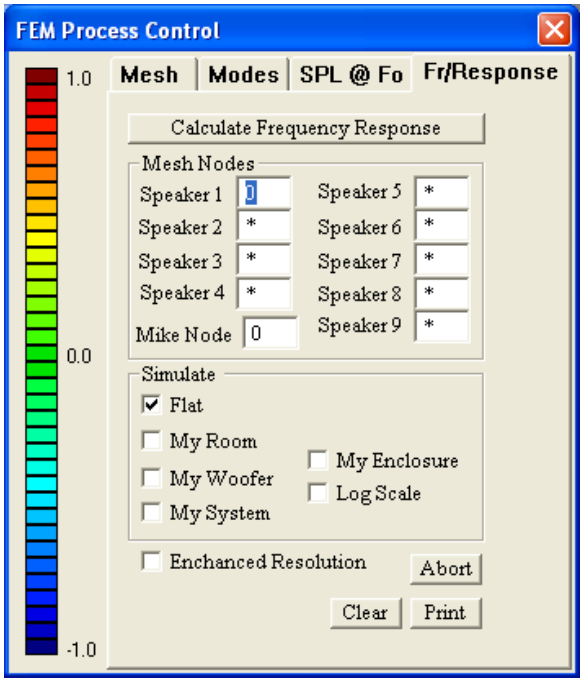


Figure 15.6 Frequency response from Nodes to Mike

The object being analyzed by the FEM screen can be sliced vertically into 10 layers: Layer 0 to Layer 9. Each layer can be entered separately, so that overall vertical shape of the object can be very complex indeed. Selection of the currently edited layer is accomplished from the control box shown on Figure 15.2. It is worth noting the numbers associated with each group of elements. The numbers suggest the flow of the activities needed to be performed sequentially.

1. **Assemble Elements in Each Layer**. There is a data field associated with the "Elements in Layer" button. This field controls which layer is currently being edited. Please enter numbers from 0 (for Element Layer 0) to 9 (for editing Element Layer 9) into this field. After entering the number press on the "Show Elements in Layer" button. If nothing was entered into this layer yet, you will see clean floor plan layer. In case, of the room under analysis being rectangular, that it the shape of the floor plan extends right up to the ceiling, please enter only Element Layer 0 shape and then use "Copy Layer 0 to 1…9" button to fill all 9 remaining element layers with the same shape as the bottom one.

2. **Review Nodes in Each Layer.** The associated button, "Nodes in Layer" will display node numbers of each of the 11 Node Layers **(please note, that there is 10 Element Layers and 11 Node Layers)**. The Layer number is entered from the associated data field.

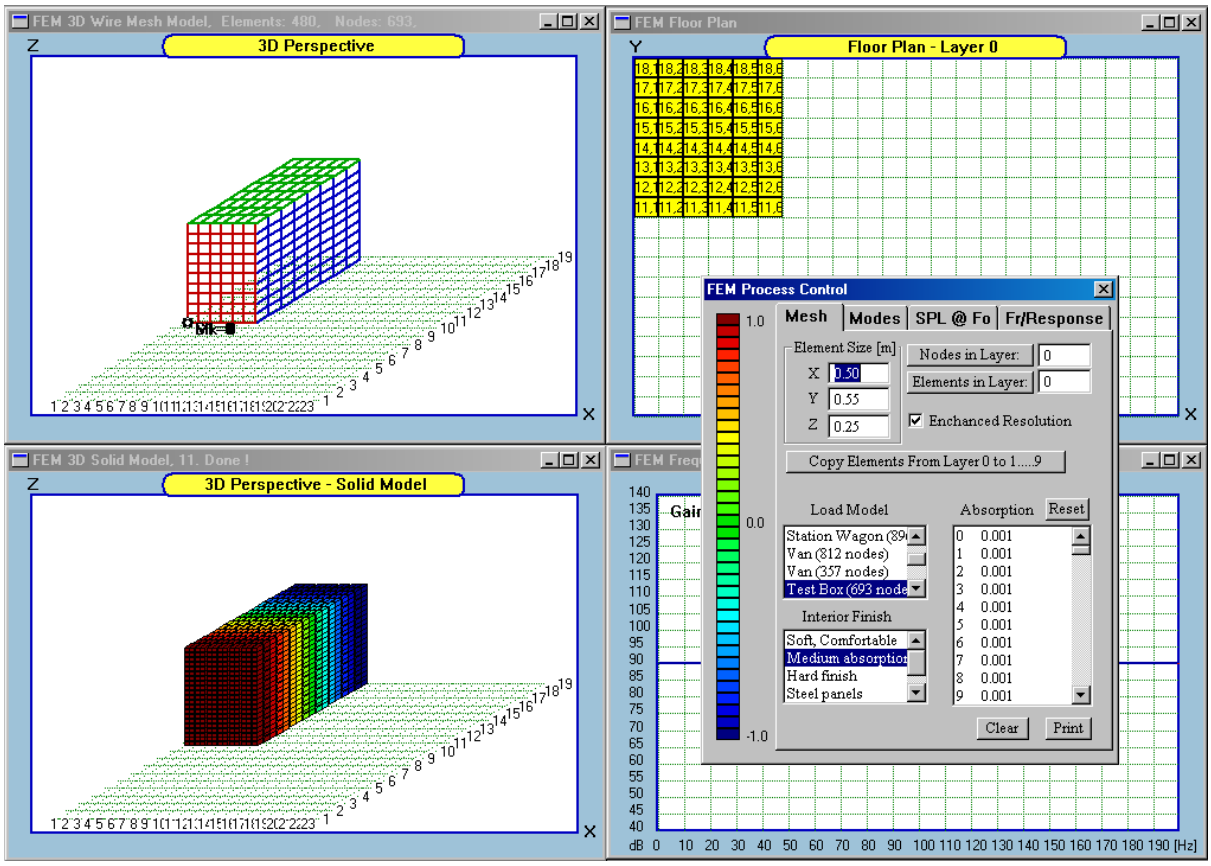


Figure 15.7. Example of modal calculation.

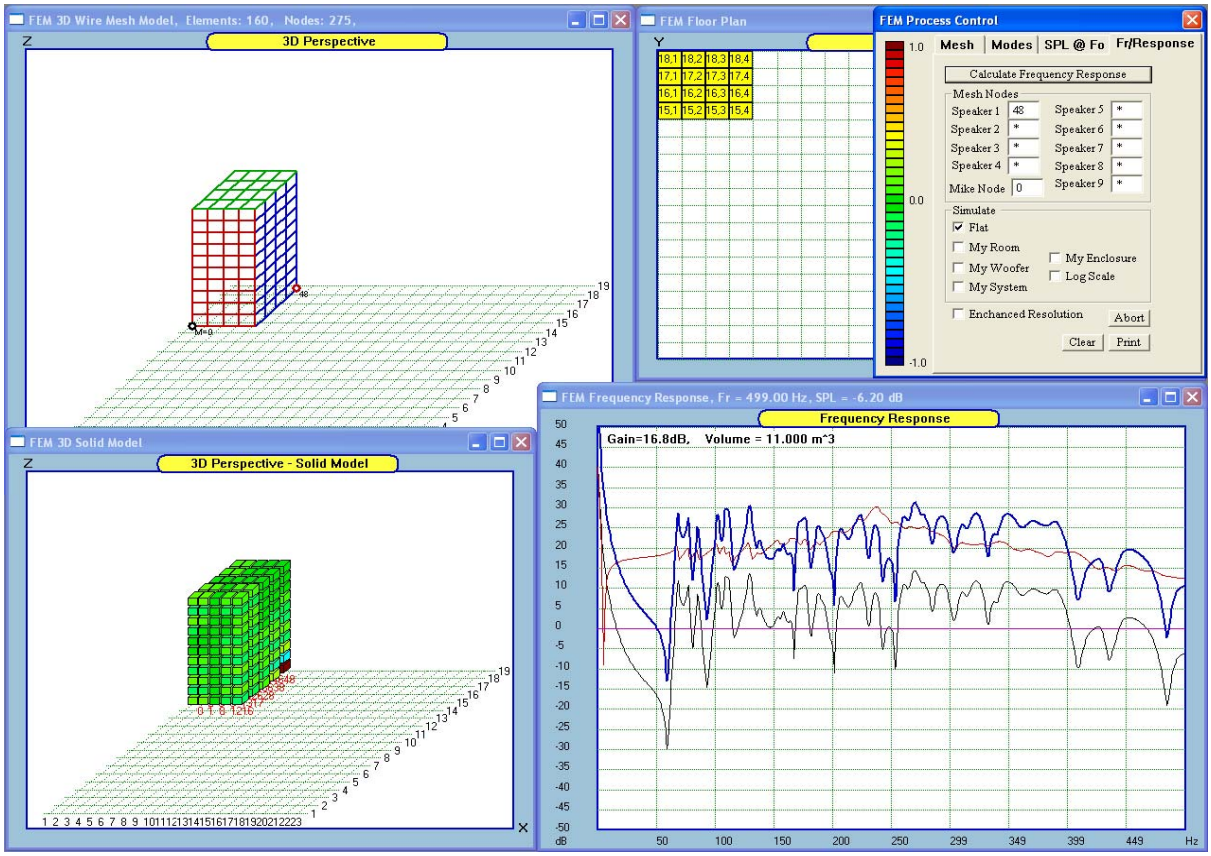


Figure 15.8. Example of frequency response plots

3. **Resonance Modes**. Calculate Room Modes is the next step in the process of analyzing the room acoustics. The FEM process is often quite long if you deal with large number of nodes and slower computer. When it has finished, you can review the whole object layer-by-layer using "**Show Layer**" button and entering Element Layer number into the associated field.

4. **Pressure From Active Sources**. Knowing the room modes enables you to make decision about placing one or two sound sources and selecting the test frequency. Then press the "**Calculate Pressure at**:" button. When the process is completed, you can review individual Element Layers by pressing the "**Show Layer**" button.

5. Finally, if the finer mesh is required, please use "Enhanced Resolution" option.

Listening Room example

For the purpose of further analysis, we selected a listening room having the shape indicated in Fig. 15.1. The floor plan is bounded by the Nodes: 0,28,78,108,230 and 208 corners. The "brick element" we used to model the internal volume of the room has the following dimensions: X=0.5 meter, Y=0.55 meter and Z=0.25meter. The room is acoustically untreated and walls are 100% reflective - this would be a good approximation of a great majority of home listening environments at frequencies below 70Hz.

1. Fig. 15.9 shows pressure distribution of the 27Hz (lowest) mode.

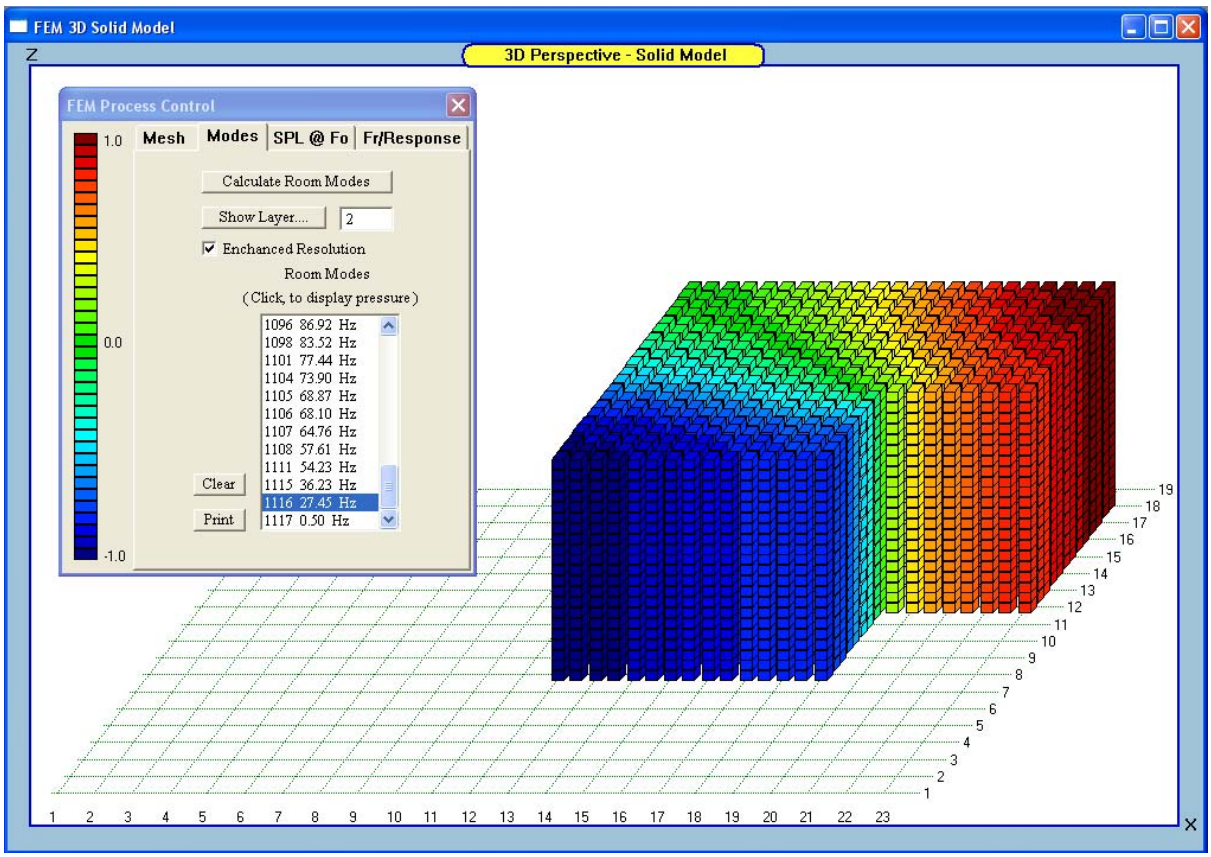


Fig. 15.9 shows pressure distribution of the 27Hz (lowest) mode

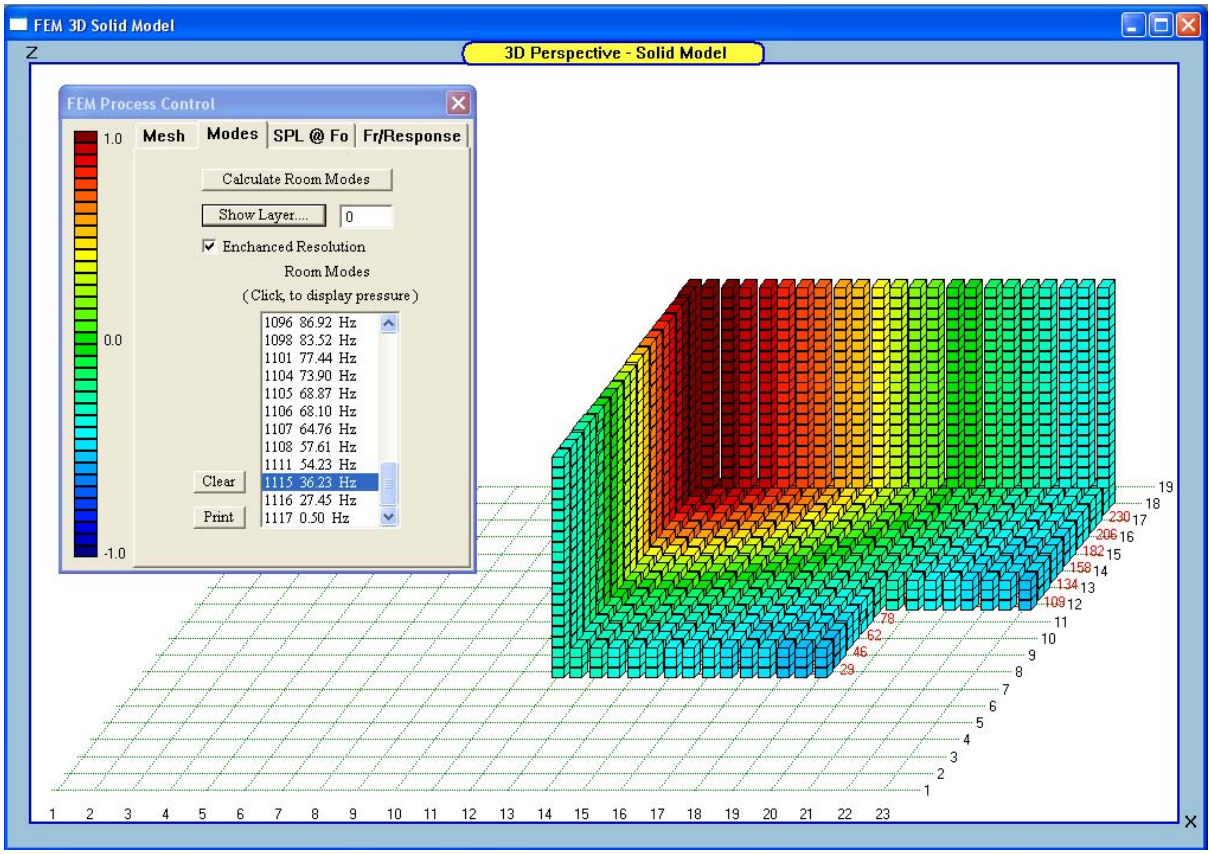


Fig. 15.10 shows pressure distribution of the 36Hz mode, showing Layer 0.

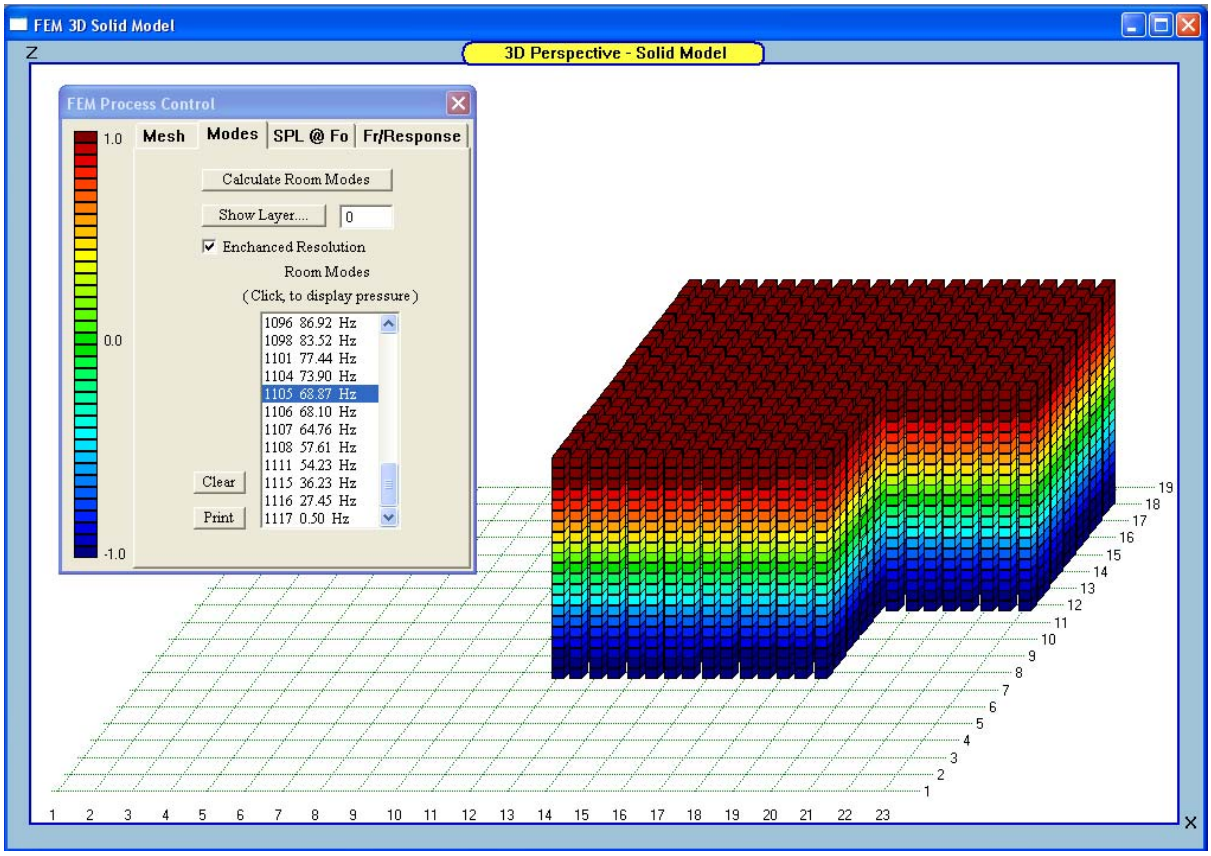


Fig. 15.11 shows 68.6Hz mode developed between the floor and ceiling

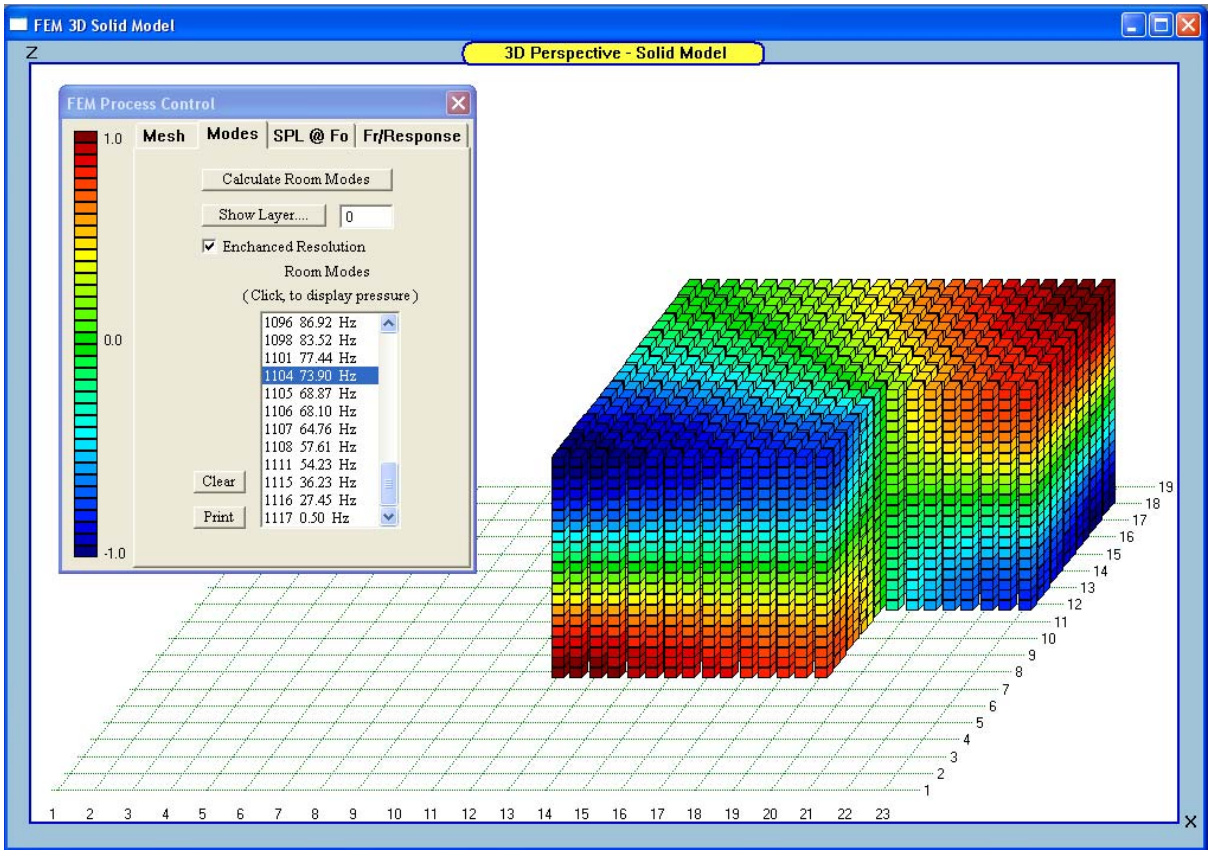


Fig. 15.12 shows tangential mode, 73.6Hz, developed between corners Nodes 78-208 and the ceiling

2. Fig. 15.10 shows pressure distribution of the 36Hz mode, which appears as a gently curved line running from X=12,Y=11 to X=18,Y=18 coordinates.
3. Fig. 15.11 shows 68.6Hz mode developed between the floor and ceiling.
4. Fig. 15.12 shows tangential mode, 73.6Hz, developed between corners Nodes 78-208 and the ceiling.

As mentioned before, higher modes, for this particular type and size of modeling elements, tend to attract a small percentage error. You can reduce the error and increase overall accuracy by selecting elements of smaller size. For the given room dimensions, the lowest mode is 27Hz and it can only develop if you place the loudspeaker in corner Node 230 and select your listening position in corner Node 0 (see Fig. 15.1). The area to avoid for listening position is the straight line between corners Node 78-208. The second lowest mode is 36Hz (see Fig. 15.10) and this one is not readily predicted without the FEM method. To energize this mode, one would place the second loudspeaker right in the Node 208 corner and avoid listening positions half-way along the walls Nodes 208+230 and Nodes 0+208 and also middle of the room. Two things need to be emphasized strongly here: (1) by placing your loudspeakers in the corners of the room, you take advantage of what is known as “room gain”. It can add as much as 10dB at low end frequencies to your loudspeaker output, as compared with the “free space” response, (2) room modes, being based on standing waves phenomenon, also amplify the sound at specific frequencies. Therefore, you do not need to locate your listening position at nodes with 100% pressure. We would strongly suggest aiming for 40-60% pressure nodes locations but avoid nodes of 0% pressure at all costs. With the above in mind, you will find that placing the subwoofers in corners Node 208 and 230 and selecting you listening area framed by [X13,Y8], [X17,Y8], [X13,Y10], [X17,Y10] points secures reasonable (not perfect) reproduction of the lowest two modes. As you can see, one would recommend a satellite system with two subwoofers and two satellites, one located in corner Node 208 and the other half-way along the Nodes 208+230 wall. The X,Y and Z nodes can be easily translated into physical space coordinates of the room with the help of the element XYZ dimensions. Higher modes, shown in Figures 15.10, 15.11 and 15.12 are useful in understanding what is the cost of this particular arrangement and we suggest you draw your own conclusions here. Having reviewed the modes, it should become easier to see, why the acoustically untreated room is a compromise. On the positive side however, you can now make an intelligent choice and understand tradeoffs between choosing this particular positioning of loudspeakers and listening area. Some of us will be tempted to take advantage of the room modes.

Say that you are interested, above all in reproducing 20Hz wavelengths - because you purchased these special woofers and designed this unique enclosure and so on. In our opinion, this is a quite legitimate point of view. Today's electric bass guitars have 5 strings and easily generate fundamental frequency of 20-30Hz. Special sound effects recorded on digital video discs are also rich in low frequencies. If your thinking follows along these lines, you need to perform Modal Analysis of your listening room, determine pressure distribution patterns and only then make an educated decision about your listening area.

Introducing a source of sound

In order to model harmonic acoustic behavior of an enclosed space with a sound source, the following equation is sufficient:

$$(K[][]-k^2M[][]+j\omega C[][])p[]=j\rho_0\omega v[] \qquad k=\frac{\omega}{c} \qquad \omega=2\pi f \qquad j=\sqrt{-1}$$

Where: K[][] is called acoustic stiffness matrix, M[][] is the acoustic mass matrix, C[][] is the damping of the system, p[] is the sound pressure vector, v[] is the excitation vector in qubic meters per second, f is the test frequency and c is the speed of sound.

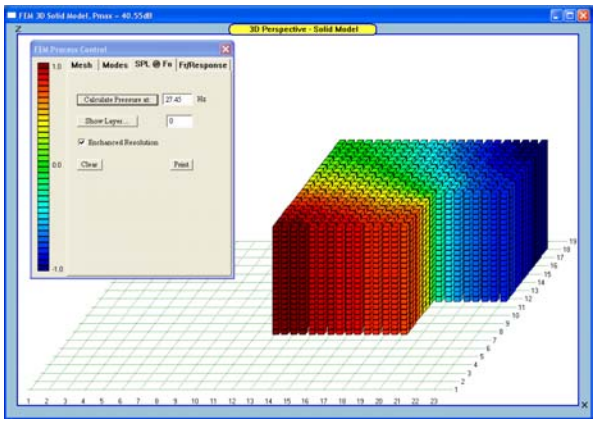
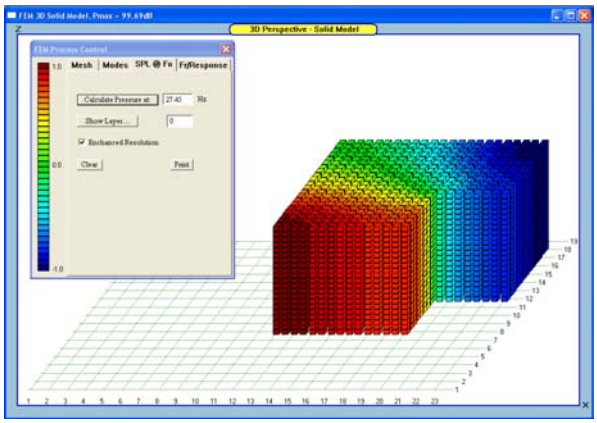


Fig 15.13 Source of sound at Node 184**, Pmax=40dB**



Source of sound at Node 0, **Pmax=99dB**

Assuming C[][]=0, that is no damping in the system, the above equation simplifies to:

$$(K[][]-k^2M[][])p[]=j\omega v[]$$

For the source of the sound I assume a "point source", which is convenient, as it can be located in any of the mesh nodes. Larger sources would have to be accounted for as a part of the room boundary. Mathematically, the problem now reduces to assembly of the K[][] and M[][] matrixes and inverting the expression in the brackets. This way, vector p[] can be found for any frequency and location of the sound source represented by excitation vector v[]. For room modes, the expression in the brackets:

$$(K[][]-k^2M[][])=0$$

SoundEasy uses "brick" (8-node) elements to approximate the volume of the listening room and the functions describing the pressure distribution between the nodes are linear. This is important, as knowing the pressure at the nodes, one can calculate pressure at any distance between them. One of the limitations of the above approach is that FEM can not model the "close field" sound pressure very accurately. It would take much more dense (finer) mesh to do this job properly. We are not however concerned with this limitation, because the job at hand is to look at the whole room. For the purpose of illustrating the application of the FEM method we selected a listening room shape indicated on Figure 15.1. This is an L-shaped room with wall Nodes 0-28 slightly longer than Nodes 108-230 wall. This deliberate lack of symmetry will make the analysis perhaps more difficult, but will emphasize the usefulness of the FEM.

Source at room nodal line

We can now keep the test frequency at the first room mode and move the source to the pressure null line of the first mode. This situation is depicted on Figure 15.13 and it is easily observable, that despite radiating the exact room modal frequency, the source fails to **fully energize** the room mode

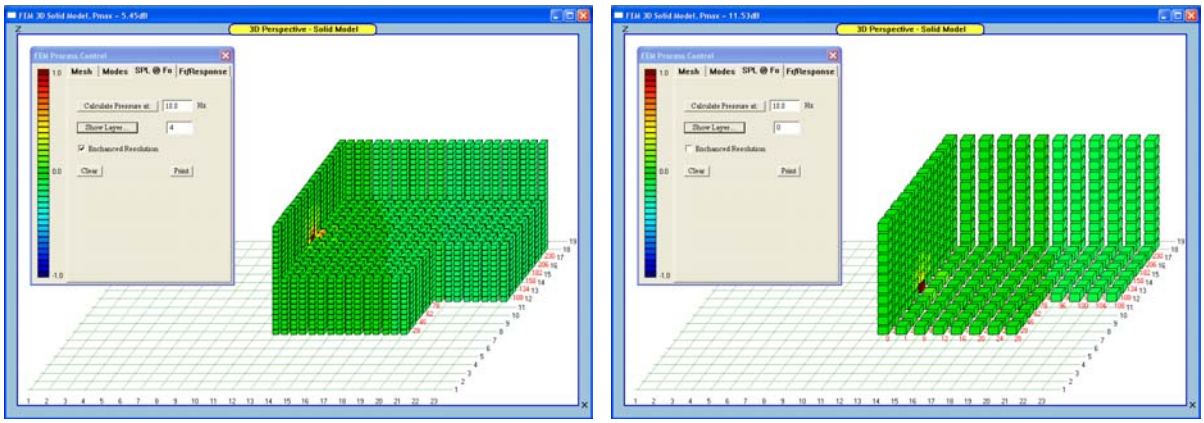


Fig 15.14 Source of sound at Node 620 and 18.8Hz, Pmax=5.45dB, Node 80, Pmax=11.45dB

Consider the following sequence: (1) If the "point source" radiates in free space, it would produce some sound pressure, which I mark as "unity". (2) When this source is placed in front of a hard wall (half-space) the sound pressure is reinforced by the reflections from the wall. (3) If in addition, the source is moved to the wall-floor junction, the resulting sound pressure is further reinforced and finally, (4) if the source of the sound is moved into the corner of the room, the sound pressure is up by some 16-18dB. Now, (5) if the frequency of the sound source happens to be one of the room resonant modes, the sound pressure at the source is dramatically magnified by the resonant effect and lack of dumping. (We assumed matrix C[][]=0). The display is arranged such a way, that nodal pressure is normalized to the maximum pressure within the room. This maximum pressure is marked "0.0dB", so all other nodes will exhibit negative pressure values, as shown on the color coded "Pressure Map" on each screen dump. **The actual level of the "Pmax=" is displayed in windows caption bar, relative to sound pressure radiated from a "point source" in free space**. Generally, if the Pmax were above 80dB mark, you would be approaching a resonant mode. When the Pmax exceeds 80dB you hit the mode. Level of 40dB indicate, that the source fails to fully energize the mode.

Levels below 18dB relate to various locations of the source at non-modal frequencies. In general, sound field of the complex-shape enclosure can be modeled for any location of the source or sources of the sound. For the non-modal frequencies, the resulting sound field typically exhibits variations near the source and a gradual increase in intensity toward opposite wall. Please note, that computer rounding errors also contribute to the value of Pmax at modal frequency and finally, the modal frequencies, as given in the previous article, are calculated and rounded to the second decimal point, as in "real life", there is no need for greater accuracy.

At a flat wall location, the Pmax= parameter is close to a typical wall-floor value of 6dB. The source was located at Node 620. At Node 80, (wall/floor joint) Pmax equals approximately 12dB. Finally, at the corber location, the Pmax=17.84dB, again, approximately the theoretical 18 dB for corner location.

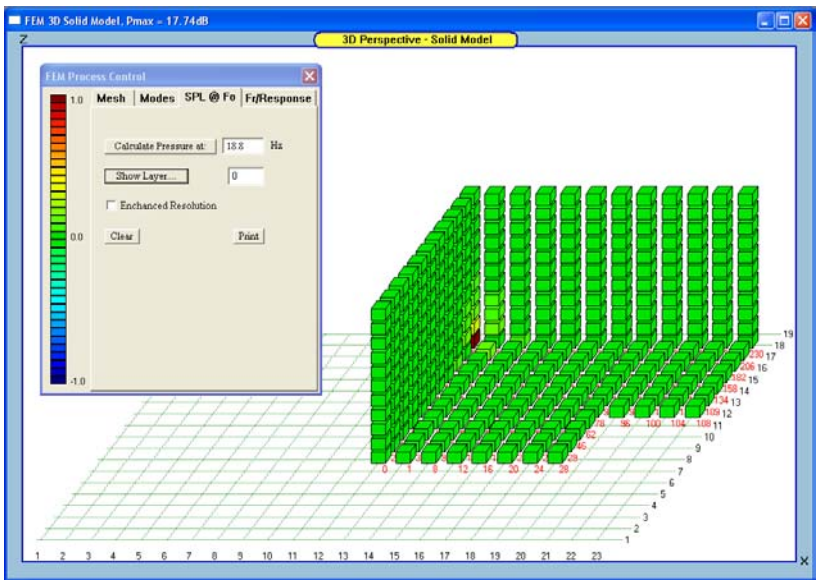


Figure 15.15. Corner location and non-modal frequency, Pmax=18dB.

3D Delaunay FEM Screen

Method of Analysis

Architectural style of today reflects our relaxed lifestyle. As a result of this, open layout of several interconnecting rooms or areas is very common. This type of dwelling allows the air and sound to freely flow from one room to another (no doors) and as such, one large, complex shape cavity is created. An example of such complex cavity is shown above on Figure 15.16. Here we have a floor plan of a house comprising of an entry hall, sitting room to the left, formal dining room to the right, and kitchen and living room further down the hall. **Our task is to obtain lowest (<100Hz) modal frequencies of the dwelling and pressure pattern distribution for those frequencies**. For the purpose of analysis, it was assumed that walls absorb negligible amount of acoustic energy. The longest distance between the most distant walls is typically much larger than the distance between floor and ceiling.

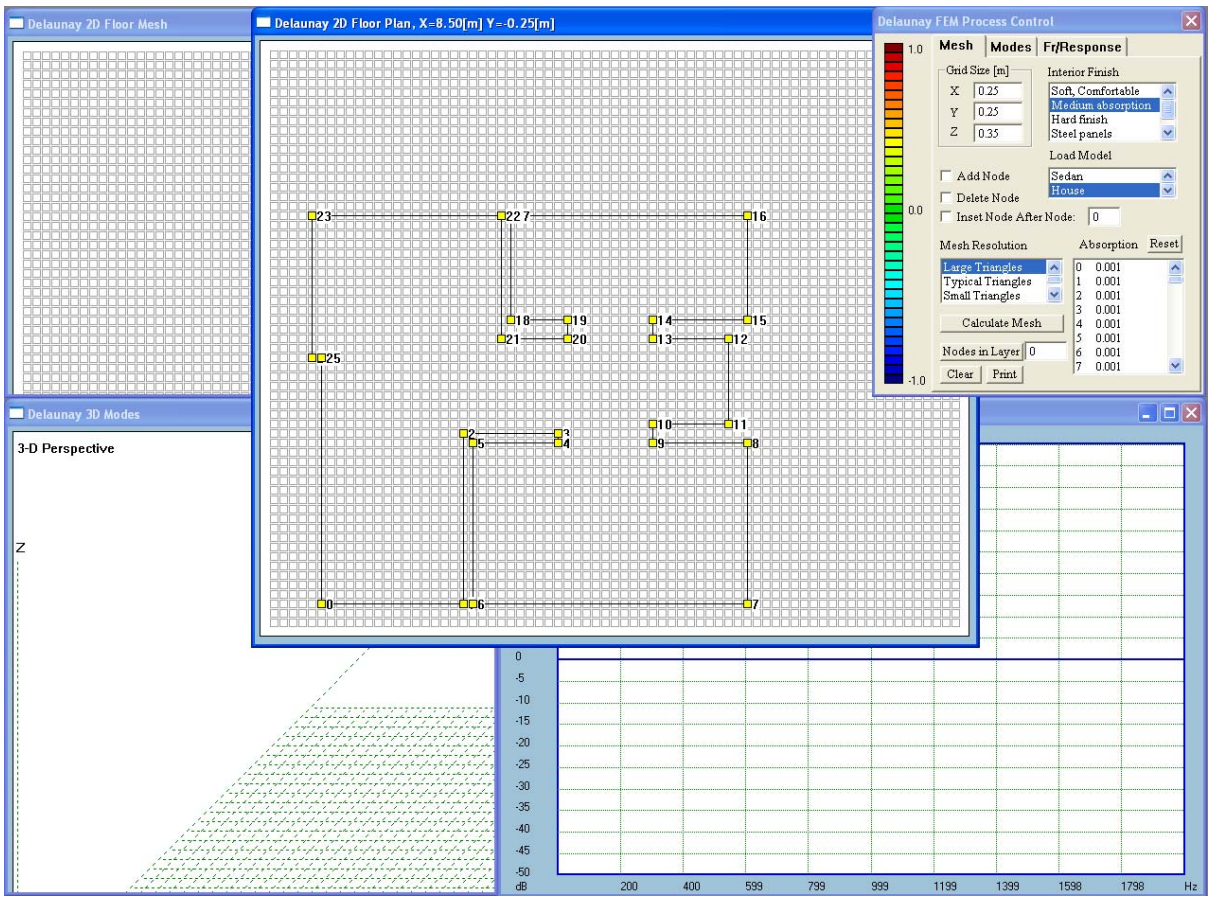


Fig 15.16 . An example of such complex cavity is shown above

Therefore the lowest modes will be tangential modes - these are modes NOT involving floor or ceiling reflections. In order to calculate the height of the element, you need to **measure the floor-wall distance and divide it by 6**. For example: distance between floor and wall is 3.30 meter. Therefore the height of the "wedge" element is 3.30/6 = 0.55 meter. The smallest increments (resolution) along the X and Y axes is also determined by the user.

This way, a large and **complex floor plan may include thin walls** in the model - this feature is essential in modeling complex floor plan of the house. The Delaunay Triangulation process is fully automated and only requires that the you provide the floor plan as the input. This is accomplished by drawing the perimeter of the area being analyzed as a **Planar Straight Line Graphic (PSLG)**. The graph must be entered in an **anticlockwise** direction. Please select the "3D Delaunay Screen" from the main menu.

The steps are described below:

1. Position mouse arrow pointer on the drawing right above the first or "start" node to be marked and CLICK the LEFT mouse button. The selected "start" node will be marked with a number when the button is clicked.

2. Move the mouse arrow pointer to the **right of the start node**, where you wish the "end" node of the line should be and CLICK again the LEFT mouse button. The line will be drawn during mouse movements and when the button is released, the "end" node will be marked, and numbered. **It is important to "orient" the SLPG anticlockwise right at the beginning, by positioning the second node to the right of the first node.** At this moment, you have one straight line with two nodes drawn on the plotting area. This represents a wall in your floor plan.

3. Position the mouse arrow pointer where you want to place the next node of the graph and CLICK the LEFT mouse button again. The node will be marked with a number and you are have the second "wall".

4. Continue to move the mouse arrow pointer to where you wish the consecutive nodes to be placed and CLICK on every node. Lines will be drawn automatically and nodes will be marked, and numbered.

5. Continue drawing walls until the closed, bounded floor plan is completed.

When drawing SLPG, you must remember the following: The graph MUST be oriented anticlockwise and MUST be closed - that is the "end" node of the last wall MUST be "start" node of the whole SLPG.

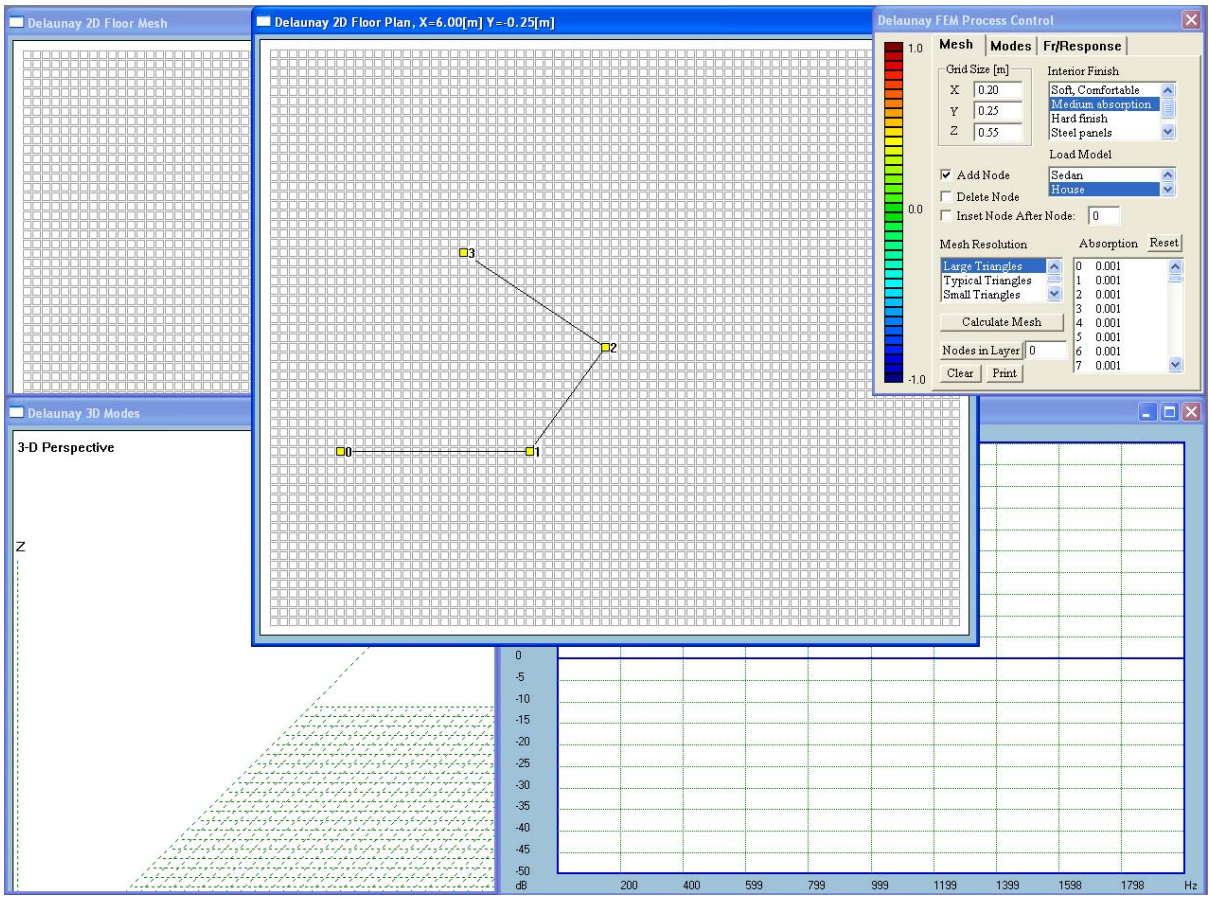


Fig 15.17 shows the first three walls entered.

How it is done - Vibration basics

One of the types most common types of analysis for structures are: (1) the natural frequency analysis or eigenvalue analysis and (2) steady state response of the structure to a harmonic force at a selected frequency. We are going to apply both method to analyze a "listening room" acoustics. The general equation of motion describes the balance required for all inertial and damping forces in the equation.

$$M[][]*\frac{\partial^2 D[]}{\partial t^2}+C[][]*\frac{\partial D[]}{\partial t}+K[][]*D[]=F[]$$

Where M[][] represents the structure mass matrix,
C[][] is the structure damping matrix,
K[][] is the structure stiffness matrix,
F[] is time varying nodal load vector and
D[] is the nodal displacement vector with its acceleration and velocity.

Eigenvalue Analysis

Assuming, that all nodes vibrate in a sinusoidal manner with a peak displacement P, we can write:

$$D[]=P*\sin(\omega t)$$
$$\frac{\partial D[]}{\partial t}=P\omega\cos(\omega wt)$$
$$\frac{\partial^2 D[]}{\partial t^2}=-P\omega^2\sin(\omega t) \qquad \omega=2\pi f \qquad j=\sqrt{-1}$$

Also, we shall assume no damping of the structure, which implies C[][] = 0 and no external forces, so that F[] = 0 as well. The general equation of motion simplifies now to:

The above equation can now be solved as a standard eigenvelue and eigenvector problem.

$$(K[][]-k^2M[][])P[]=0 \qquad k=\frac{\omega}{c}$$

Theoretically, we imply that if the structure is deformed into a mode and then released, it would continue to vibrate indefinitely due to the absence of damping. However, practical structures always have some damping, so that the vibrations will eventually decay.

Frequency Response Analysis

In order to model harmonic acoustic behavior of an enclosed space with a sound source, the following equation is sufficient:

$$(K[][]-k^2M[][]+j\omega C[][])p[]=j\rho_0\omega v[]$$

Where: K[][] is called acoustic stiffness matrix,
M[][] is the acoustic mass matrix,
C[][] is the damping of the system,
p[] is the sound pressure vector,
v[] is the excitation vector in qubic meters per second,
f is the test frequency and
c is the speed of sound.

For the source of the sound one could a "point source", which is convenient, as it can be located in any of the mesh nodes. Larger sources would have to be accounted for as a part of the room boundary. Mathematically, the problem now reduces to assembly of the K[][] and M[][] matrixes and inverting the expression in the brackets. This way, vector p[] can be found for any frequency and location of the sound source represented by excitation vector v[]. Solving this equation for different discrete input frequencies determines the frequency response of the structure. When the test frequencies are equal to the natural modes, the solution algorithm fails due to division by zero.

Stiffness and Inertia Matrixes

Derivation of stiffness and inertia matrixes is beyond the scope of this article, however, I would like to describe briefly the elements involved. Our interest will focus on a volume V enclosed by a surface S. The surface is acoustically hard, so that normal velocity on the surface is zero. Under those conditions the pressure p, must satisfy the wave equation

$$\nabla^2 p + \left(\frac{\omega^2}{c^2}\right) p = 0 \qquad\qquad \frac{\partial p}{\partial n} = 0$$

Variational principle, equivalent to the above equations can be prescribed as follows:

$$\delta \int_V \frac{1}{2}\left[(\nabla p)^2 - \left(\omega^2 / c^2\right) p^2\right] dV = 0$$

In order to maintain continues pressure between the nodes, an 8-node "brick" elements can be used to fill the volume V. Elements k[][] of the stiffness matrix K[][] can be expressed as follows:

$$k_i[][] = \int_{Vi} B_i^T[][] B_i[][] dV$$

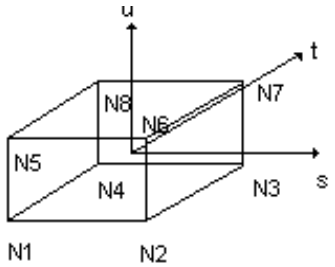
Where:

$$B_i = \begin{bmatrix} \dfrac{\partial N_i}{\partial x} \\ \dfrac{\partial N_i}{\partial y} \\ \dfrac{\partial N_i}{\partial z} \end{bmatrix}$$

And the elements m[][] of the inertia matrix M[][] is expressed as follows:

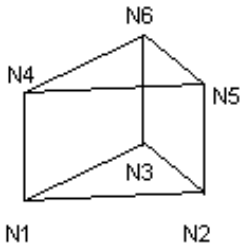
$$m_i[][] = \int_{Vi} N_i^T[] N_i[] dV$$

Both elemental matrixes for 8-node element are 8x8 in size. Vector N[] represents eight linear shape functions defined as below:



N1 = (1 - s)(1 - t)(1 - u)/8 N2 = (1 + s)(1 – t)(1 – u)/8
N3 = (1 + s)(1 + t)(1 – u)/8 N4 = (1 – s)(1 + t)(1 – u)/8
N5 = (1 – s)(1 – t)(1 + u)/8 N6 = (1 + s)(1 – t)(1+ u)/8
N7 = (1 + s)(1 + t)(1 + u)/8 N8 = (1 – s)(1 + t)(1 + u)/8

For the 6-node "wedge" element, vector N[] represents six linear shape functions defined as below:



N1 = s(1 - u)/2 N2 = t(1 – u)/2
N3 = (1 - s - t)(1 – u)/2 N4 = s(1 + u)/2
N5 = t(1 + u)/2 N6 = (1 - s - t)(1+ u)/2

With the above in mind, the variational principle can be expressed in matrix form as:

$$(K[][]-k^2M[][])P[]=0 \qquad k=\frac{\omega}{c}$$

Due to complexity of the resulting expressions for element k[][] and m[][] they are best evaluated by using numerical integration techniques. Integration can be facilitated by Gauss integration scheme, requiring integration limits of +1 and –1. We can change integration from x,y,z space to s,t,u space with the help of Jacobian J[][] of the transformation resulting immediately in the volume element dV being expressed as follows:

$$dV=|J|dsdtdu$$

Where |J| is the determinant of Jacobian matrix J[][].

Jacobian of the Transformation

$$\begin{bmatrix}\frac{\partial N_i}{\partial s}\\ \frac{\partial N_i}{\partial t}\\ \frac{\partial N_i}{\partial u}\end{bmatrix}=[J]\begin{bmatrix}\frac{\partial N_i}{\partial x}\\ \frac{\partial N_i}{\partial y}\\ \frac{\partial N_i}{\partial z}\end{bmatrix}$$

Where J[] is a ***Jacobian*** of the transformation expressed as follows:

$$[J]=\begin{bmatrix}\frac{\partial N_1}{\partial s} & \frac{\partial N_2}{\partial s} & \frac{\partial N_3}{\partial s} & \frac{\partial N_4}{\partial s} & \frac{\partial N_5}{\partial s} & \frac{\partial N_6}{\partial s} & \frac{\partial N_7}{\partial s} & \frac{\partial N_8}{\partial s}\\ \frac{\partial N_1}{\partial t} & \frac{\partial N_2}{\partial t} & \frac{\partial N_3}{\partial t} & \frac{\partial N_4}{\partial t} & \frac{\partial N_5}{\partial t} & \frac{\partial N_6}{\partial t} & \frac{\partial N_7}{\partial t} & \frac{\partial N_8}{\partial t}\\ \frac{\partial N_1}{\partial u} & \frac{\partial N_2}{\partial u} & \frac{\partial N_3}{\partial u} & \frac{\partial N_4}{\partial u} & \frac{\partial N_5}{\partial u} & \frac{\partial N_6}{\partial u} & \frac{\partial N_7}{\partial u} & \frac{\partial N_8}{\partial u}\end{bmatrix}\begin{bmatrix}x1 & y1 & z1\\ x2 & y2 & z2\\ x3 & y3 & z3\\ x4 & y4 & z4\\ x5 & y5 & z5\\ x6 & y6 & z6\\ x7 & y7 & z7\\ x8 & y8 & z8\end{bmatrix}=$$

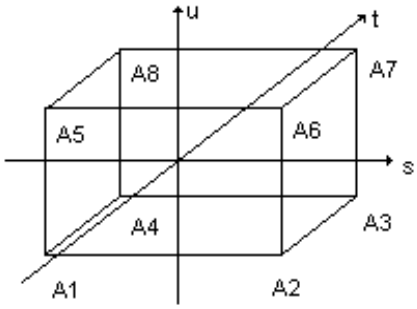
$$\begin{bmatrix}\sum_{i=1}^{8}\frac{\partial N_i}{\partial s}x_i & \sum_{i=1}^{8}\frac{\partial N_i}{\partial s}y_i & \sum_{i=1}^{8}\frac{\partial N_i}{\partial s}z_i\\ \sum_{i=1}^{8}\frac{\partial N_i}{\partial t}x_i & \sum_{i=1}^{8}\frac{\partial N_i}{\partial t}y_i & \sum_{i=1}^{8}\frac{\partial N_i}{\partial t}z_i\\ \sum_{i=1}^{8}\frac{\partial N_i}{\partial u}x_i & \sum_{i=1}^{8}\frac{\partial N_i}{\partial u}y_i & \sum_{i=1}^{8}\frac{\partial N_i}{\partial u}z_i\end{bmatrix}$$

Now, the derivatives of the shape functions in respect to x,y and z coordinates can be calculated by inverting the Jacobian matrix J[][].

$$\begin{bmatrix}\frac{\partial N_i}{\partial x}\\ \frac{\partial N_i}{\partial y}\\ \frac{\partial N_i}{\partial z}\end{bmatrix}=[J]^{-1}\begin{bmatrix}\frac{\partial N_i}{\partial s}\\ \frac{\partial N_i}{\partial t}\\ \frac{\partial N_i}{\partial u}\end{bmatrix}$$

Gaussian Integration

Example of Gaussian integration scheme for the 8-node "brick" element and N=2 is being used and the location of the points together with the associated weights is shown below.



	Weight	s	t	u
A1	1.0	-0.57735	-0.57735	-0.57735
A2	1.0	0.57735	-0.57735	-0.57735
A3	1.0	0.57735	0.57735	-0.57735
A4	1.0	-0.57735	0.57735	-0.57735
A5	1.0	-0.57735	-0.57735	0.57735
A6	1.0	0.57735	-0.57735	0.57735
A7	1.0	0.57735	0.57735	0.57735
A8	1.0	-0.57735	0.57735	0.57735

Assembly of System Matrixes

Local nodes of individual elements are always numbered as 1 to 8 for "brick" elements and 1 to 6 for "wedge" elements. However, when the complete mesh has been generated, **global** coordinates of all nodes and global numbering scheme for all components become available, and at this point of time we know which elements share the same nodes. It is therefore possible to assign the global node numbering and the global element coordinates to the **local** nodes of each element. The process is called mapping and can sometimes be "tricky". Assembly of the stiffness K[][] and inertia M[][] system matrixes involves calculation of derivatives of the shape functions ***Ni*** and can be quite complex process.

Solution of Eigenvalue problem

Eigenvalue problem can be solved using Hauseholder Algorithm and Tridiagonal QLI Algorithm. However, this process requires symmetrical matrixes for its input. Assuming that matrixes K[][] and M[][] are symmetric of order n, the eigenvalue problem can be stated as:

$$(K[][] - \lambda M[][])X[] = 0$$

The X[] is a vector called eigenvector and λ is a scalar called eigenvalue. The above can be converted to standard eigenvalue problem by pre-multiplying both sides by inverted matrix M[][].

$$A[][] = M^{-1}[][] * K[][]$$

However, matrix A[][] is often non-symmetrical despite M[][] and K[][] being both symmetrical. If the method adopted for solving the eigenvalue problem requires A][] to be symmetrical, the following procedure may be used:

1. Using Choleski method, decompose M[][], so that only upper triangular U[][], matrix will be used:
2. Formulate A[][] as:

$$M[][] = U^T[][] * U[][] \qquad A[][] = (U^T[][])^{-1} * K[][] * (U^{-1}[][])$$

3. Matrix A[][] is now symmetrical and suitable for Hauseholder Algorithm and Tri-diagonal QLI Algorithms.

Generating Mesh

Generating mesh is a large subject in its own right. Mesh is the way to translate information about the physical parameters of the object being analyzed into a set of numbers understood by the program. Vastly simplified version is presented here, however, good results can be achieved for a number of applications.

Delaunay Triangulation Algorithm

The process commences with the **Planar Straight Line Graphic** (PSLG) description. In simple terms, the PSLG it is the closed contour (perimeter) encompassing the area to be triangulated. The coordinates (nodes) are entered as a graph oriented counterclockwise, and using mouse pointer. The next step is the Delaunay Triangulation process. It needs to be noted, that final results of the triangulation is expected to be a uniform mesh with a predetermined maximum triangle size, suitable for Finite Element Method mesh. Delaunay triangulation of a point set ***P***, is a triangulation with the property that no point in the point set ***P*** falls in the interior of the circumcircle (a circle that passes through all three vertices) of any triangle in the triangulation. Initially, the point set ***P*** is the set of nodes of the PSLG, as determined by the user.

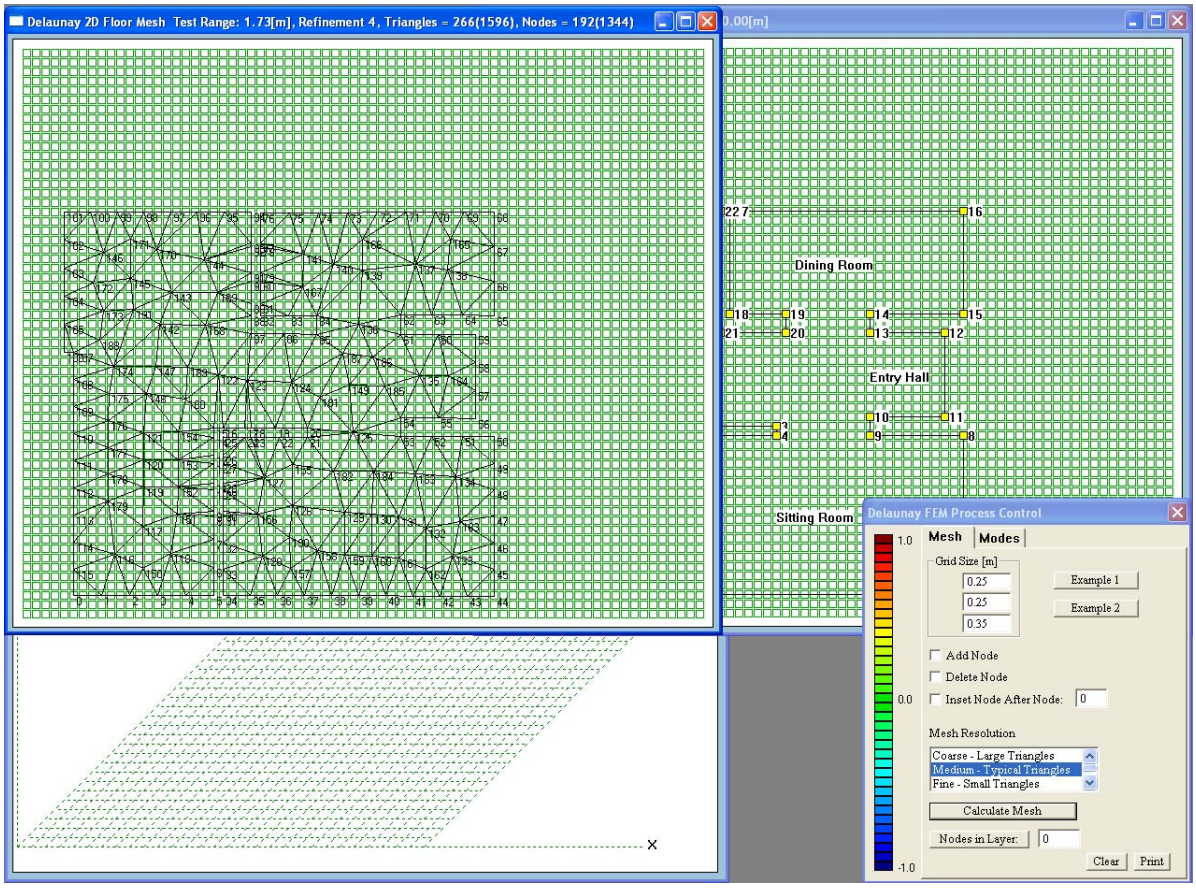


Fig 15.18 Delaunay Trangulation completed.

The point set P is then immediately expanded by splitting each edge of the PSLG into smaller sections whose length is determined by the maximum triangle size, as specified by the user. The resulting point set ***P*** is then triangulated. At this moment, there is no points inside the PSLG, therefore the first pass of the triangulation produces very long and "skinny" triangles. The record of the longest edge ***L***, is maintained for future reference. In the next loop, the longest edge is compared against the required triangle size and if a triangle with a longer side is found, the triangle is destined to be split by inserting an extra point into the offending triangle. Once all triangles are checked and all extra points are inserted, the whole point set ***Pnew*** is then re-triangulated. The process continues until there is no more extra points to be inserted. The following can be noted:

1. Each triangulation pass produces "denser" set of points from which Delaunay triangles are build. However, it is wasteful to check the whole point set ***Pnew*** for the purpose of building Delaunay triangles. Only the points located no further from the current "seed" edge than longest edge ***L***, need to be considered for potential new Delaunay triangle.
2. The algorithm maintains a list of all edges that were already used as "seeds" for constructing Delaunay triangles. These edges are never used again in a current pass.
3. The Delaunay triangulation uses **Orientation Test** - Check whether or not a point lies to the left of a vector. If point "C" lies to the left of the vector AB, the cross product of the vectors AB *AC will be larger than 0. If points "A", "B", and "C" are colinear, the cross product of these two vectors will be equal to 0. If point "C" lies to the right of the vector AB, the cross product of the vectors AB * AC will be smaller than 0.

4. Also, the Delaunay triangulation uses **In Circle Test** - Check whether or not a point lies inside the circumcircle of a triangle. The algorithm uses the InCircle Test to make sure that there is no point that is located inside the circle whichpasses through the three points it found. These three points can be now connected to form a Delaunay triangle.
5. Finally, the algorithm checks for "crossed edges" of triangles.

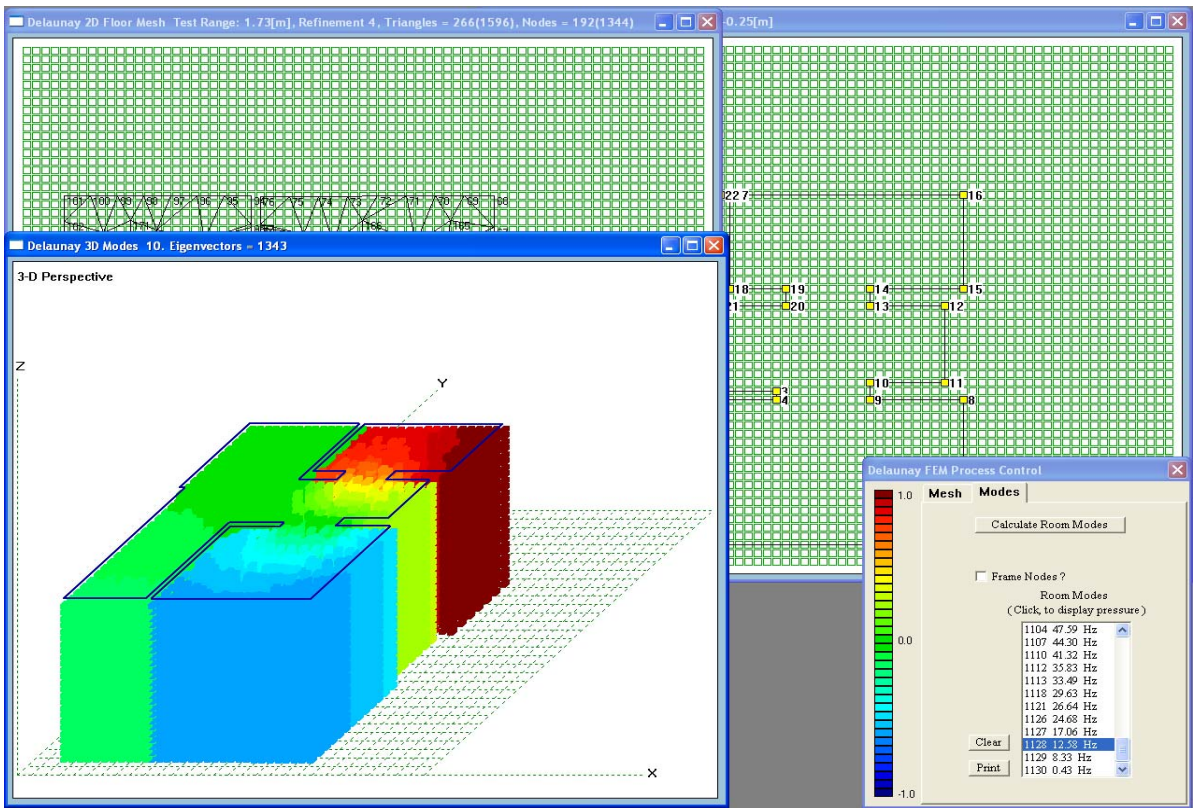


Fig 15.19 Sound field pressure distribution in the house.

Finite Element Method has been successfully implemented to accomplish the task of determining complex-shape room modes, because the method lends itself particularly well to complex geometry of the space being analyzed. All essential mathematical components of the whole analytical process have been nominated and briefly explained. Attempt to solve a problem of this nature without a computer would prove to be too formidable task to undertake, if only because of the shape complexity of the volume being analyzed. The FEM suite of tools are well suited to help you out in a situation like this. Data generated by the computer is intended to assist you in making educated decisions about the loudspeaker-listener interaction in your **individual** listening environment.

Control Dialog Boxes

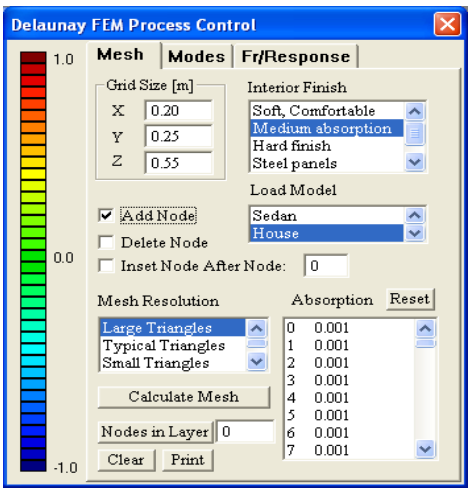


Fig 15.20 Mesh control TAB

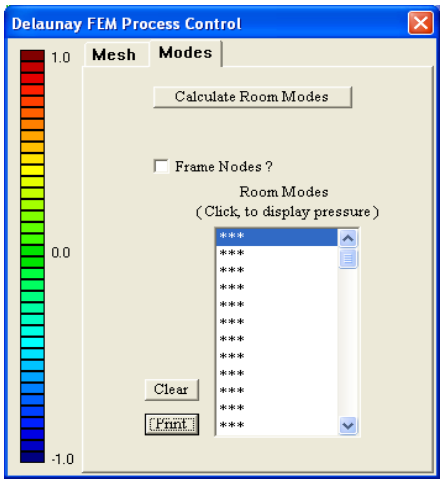


Fig 15.21 Delaunay Modes TAB.

Delaunay Mesh TAB

1. **Grid Size** [in meters] – X, Y, Z dimensions. Enter the required dimensions into these data fields.
2. **Load Model –** A built-in set of examples.
3. **Interior Finish –** Select the closest fit of the walls of the interior.
4. **"Add Node"** – if checked, you can add a node to the SLPG graph.
5. **"Delete Node"** – if checked, you can delete a node to the SLPG graph. Example below shows most of the internal nodes deleted from the "open plan house" example.
6. **"Insert Node After Node"** – if checked, you can insert a node to the SLPG graph. The new node will be inserted after the nominated node.
7. **Absorption** – Select absorption coefficient of individual nodes. **Reset** button included.
8. **"Mesh Resolution"** – List box to select 4 sizes used for Delaunay triangles.
9. **"Calculate Mesh"** – Press this button to generate triangular mesh of the floor plan.
10. **"Nodes in Layer"** – You can review node numbering in each of the 7 layers (0…..6).
11. **"Clear"** – Clears the mesh and SLPG graph.
12. **"Print"** – Opens printer dialogue box.

Delaunay Modes TAB

1. **"Calculate Room Modes"** – Once you are happy with the mesh, you can calculate room modes for this particular floor plan.
2. **"Frame Nodes"** – The resulting color-coded nodal pressure is displayed with the rectangles framed.
3. **"Room Modes"** – List box which allows you to select and display nodal pressure patterns for each calculated mode. Just click on the desired modal frequency.
3. **"Clear"** – Clears the mesh and SLPG graph.
4. **"Print"** – Opens printer dialogue box.

Frequency Response Tab

1. **Speaker 1 Node** - This is an editable field and is used to enter nodal location of Speaker 1. You can review node numbering by going to the "Mesh" tab. The editable field next to the "Nodes in Layer" button allows you to specify layer number you wish to inspect for nodes. Negative sign in front of the number indicates reversed phase.
2. **Speaker 2/3/4/5/6/7/8/9 Node** - Same as above. If you have only one speaker, you must enter '*' (asterisk) in this field. Negative sign in front of the number indicates reversed phase.
3. **Mike Node** - This is an editable field and is used to enter nodal location of the Listener. Again, you should review numbering of nodes as described in bullet 1.

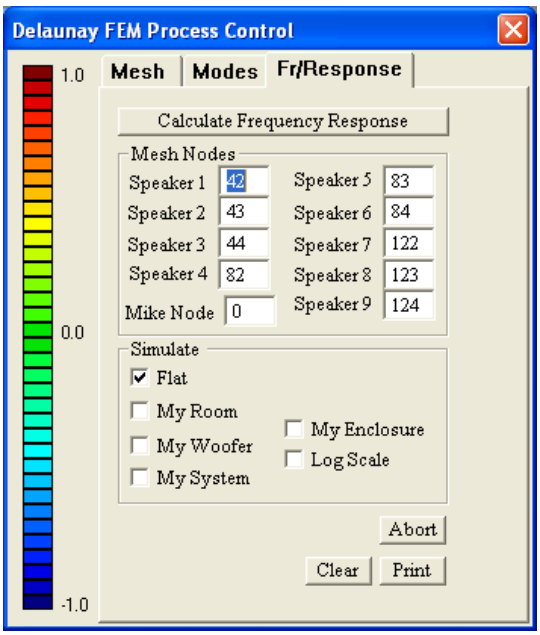


Fig 15.22 Delaunay Frequency response TAB

4. **Calculate Frequency Response** - Pressing this button will activate the algorithm. The process can be quite lengthy (2-3hours), particularly for larger number of nodes. Plotted frequency response represents acoustic pressure developed within the room as generated by up to 9 speakers.
5. **Simulate** group of options:
   Flat = No other components are included
   My Room = SPL in the room with last calculated T/S-type enclosure SPL superimposed.
   My Woofer = SPL in the room with woofer transfer function superimposed.
   My System = SPL in the room with complete system transfer function superimposed.
   My Enclosure = Select to calculate Driving Point Impedance of your speaker enclosure. <u>This option MUST be selected for FEM modal analysis of your loudspeaker enclosure.</u>
   Log Scale = Switch to logarithmic scale.
6. **Clear** - Simple clears the display.
7. **Print** - Button invokes printing dialogue box.
8. **Abort** – Simply aborts plotting.

<u>Note: Frequency response range for FEM analysis is selected from Preferences screen same as Box range.</u>

The colored scale on the left-side of the dialogues box indicated normalized pressure at a node. Please note, that the display windows can be moved around and re-sized, however, the display is not scaled accordingly to the size of the window. Scaling the graphics will be incorporated in later releases of the program.

The process of creating a mesh, starting with the SPLG graph, has been described in some details already. You have a choice of using several mesh densities, by selecting the triangle size from the provided list box. You will notice, that triangle size is selected independently of the meshing grid size (the X / Y / Z element size). The grid resolution should reflect the size of the object being analysed. For instance, when analysing the acoustics of an open floor house, you may want to select X=0.25meter, Y=0.25 meter and so on. This will allow you to draw plastered walls between rooms with correct scale. After that, you can decide on the actual FEM mesh density, by selecting one of the available triangle sizes.

On the other hand, when you work with loudspeaker enclosure modal analysis, a better choice of grid would be X=0.01m or X=0.02m and so on.

Generally, large triangles will result in less dense mesh, and this will result in lower usable frequency of analysis. However, it will also give you much shorter calculation time. So we would recommend, that you start with less dense mesh and then increase the density, if you need the analysis progressing accurately into higher frequencies (above 200-1000Hz).

After the mesh has been created, you can review the location of individual layers / nodes by editing the layer number and pressing the **"Show Layer**" button. For this software release, there are 9 layers of nodes, numbered 0-9. Evidently, there are 8 layers of wedge elements.

The Delaunay triangulation (wedge element) FEM scheme is equipped with very similar functionality to the brick element FEM scheme. File Upload and Save functions are available via the main menu and the saving and loading process is a standard, Windows-type of control dialog. Once you are happy with the mesh density, it is always recommend to save the mesh into a file, for future reference.

One of the features of the **"Fr/Response"** TAB, allows you to automatically view the location of a node in the mesh, when you edit the node number in one of the nine "loudspeaker" data fields provided on this TAB. The layer with this particular node will be highlighted in the **"3D Mesh Window"**, so please keep it opened at the same time.

Changing the type (LOG – LIN) frequency scale in the SPL plotting window can be performed when the **"My Enclosure" checkbox in NOT selected**. Generally, it is recommended to make the scaling decision before plotting, and set the type of the scale then.

For instance, after you calculate the mesh (after the Calculate Mesh button was pressed), you can save the resulting mesh into a file and later load it back into the program – see main menu options above. This ensures, that exact same mesh will be used for all subsequent calculations. Figure 15.23 was created with re-loaded mesh.

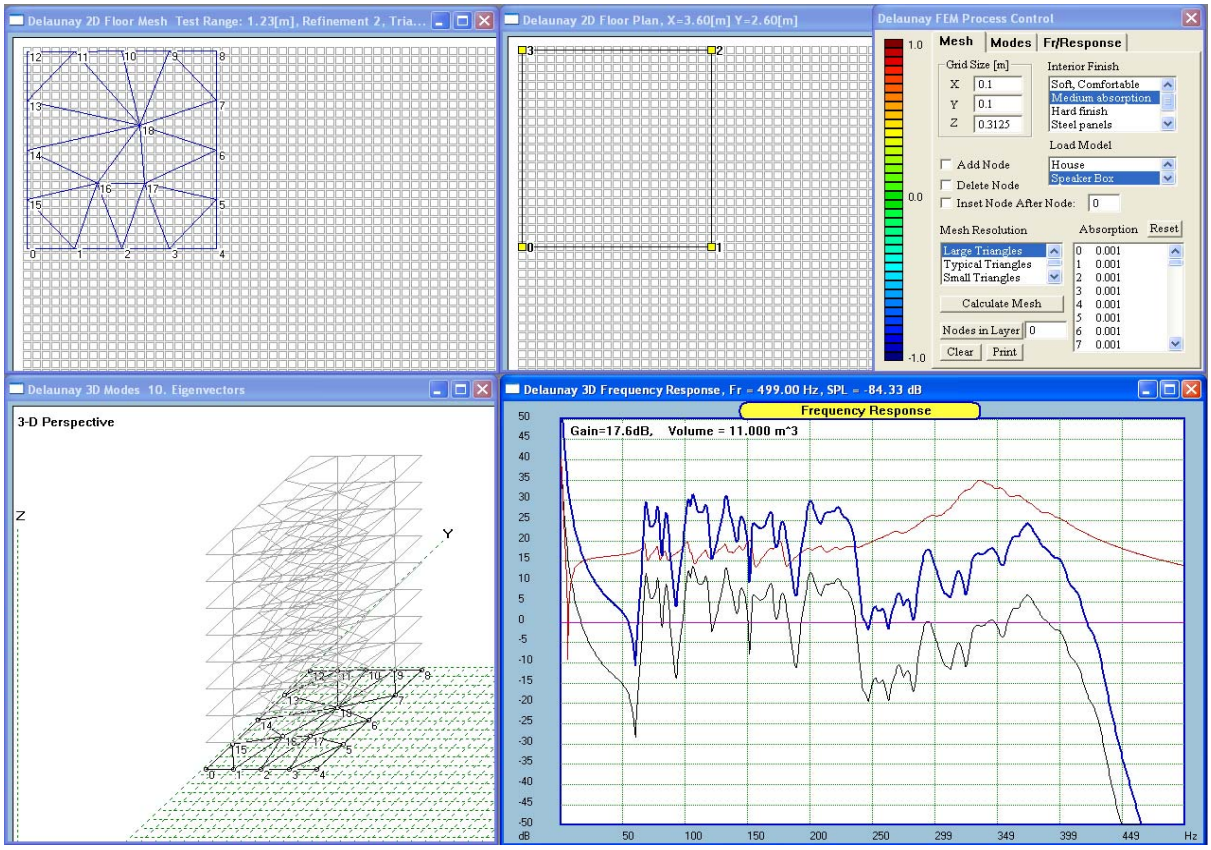


Fig 15.23 FEM SPL potting function.

It is easy to observe, that the SPL plot above is valid within ~200Hz range. This is due to the size of triangles used for mesh creation.

Some aspects of subwoofer placement

As we know, all rooms (a cavity volume enclosed by walls) have resonant frequencies at which the SPL generated by a source can be quite large. The frequencies at which resonances occur (called **modes**), depend on the geometry of the room. At a resonance frequency the pressure pattern inside the room will consist of **antinodes**, where the pressure is maximum, and **nodes**, where the pressure is zero. In a room with hard (eg. reflective) walls, the pressure will always be maximum at the wall or in a corner when the room is excited at any of its modal frequencies. If the source is located at an anti-node for a given mode, the room response will be greatest. If the source is located at a pressure node for a given mode, the room will not respond regardless how powerful the source is. Some of us (myself including) take advantage of our listening room's acoustic behavior to improve the performance of our systems. Placing a loudspeaker or subwoofer against a wall or in the corner of a room allows the low frequency modes of the room to enhance the low frequency performance of the loudspeakers. When the source is placed at a node for that frequency standing wave pattern, the maximum room response drops to zero at that frequency. Moving the speaker some distance away from the node restores back (at least partially) the room's response. Modal patterns only occur when the room is being driven at a modal frequency. At any other frequency, the pressure waves radiating outwards from the source reflect from the walls, but do not combine to produce a modal pressure pattern. As a result, there is no nodes and antinodes and the pressure can actually fall to zero at a wall. You will see these irregular patterns on Figure 15.17 – Figure 15.21.

Eliminating Room Modes Using Multiple Subwoofers

If a woofer is placed against a wall, it energizes all of the modes along the length of the room. If the woofer is placed in the corner (of rectangular room, as we saw before, in non-rectangular rooms this will not be the case in general) , it will energize all mores in this room. On one hand, we may want to have more low frequency output, but on the other, room modes have very definitely resonant character and will add excessive SPL at this frequency and will add ringing in time, as we find out in the next chapter. Understanding what happens in listening rooms at low frequencies requires determination of the room modes. In addition, we need to understand what happens if we start moving the source of the sound (loudspeaker) and the listener around the

room. Finally, we need to take another look at the sound pressure level distribution when several loudspeakers are placed in the room. The floor plan of our example room is shown below.

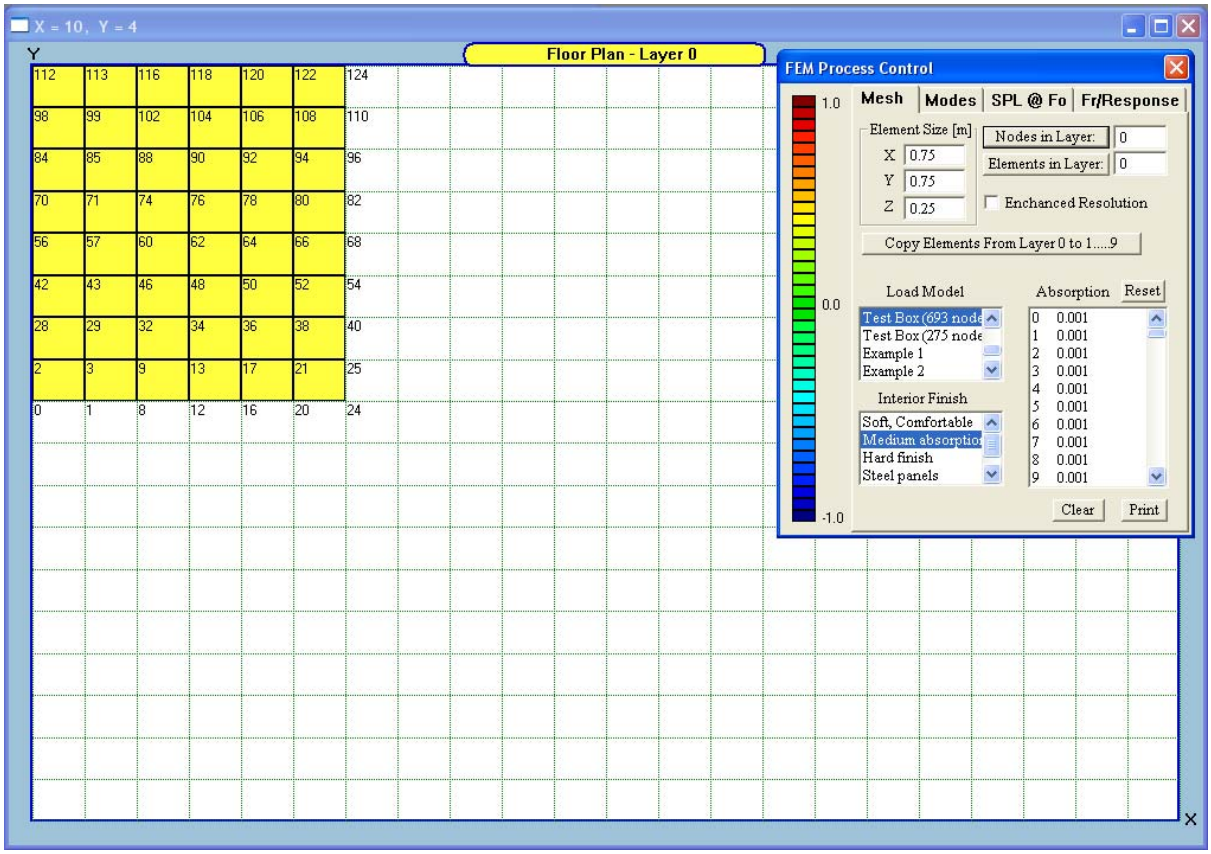


Figure 15.24 Floor plan of our test room

Calculating and displaying modal pressure pattern certainly helps the understand where the pressure peaks are located. In addition, a color scheme is used to indicate weather the pressure peak is maximum in positive polarity (red color on the chart) or maximum in the negative polarity (blue color on the chart). It is easy to observe, that the sound pressure on opposite side of a null in standing wave pattern are in opposite polarity. This gives us an excellent opportunity to place second subwoofer in the negative polarity area to cancel the positive maximum pattern generated by the first subwoofer.

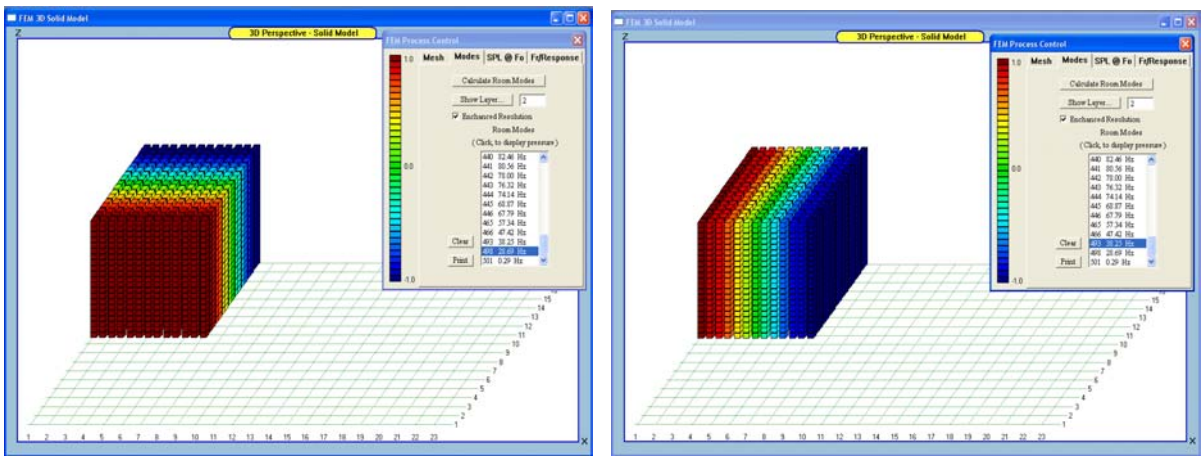


Figure 15.25. Modes: 28.7Hz and 38.3Hz

To visualize the modal problem in terms of the loudspeaker frequency response (SPL), we can start examining the SPL plot as taken with a simulated sealed enclosure with F3dB = 30Hz and placed in one of the corners – this would be corner 124 as marked on our floor plan. Nodal frequencies of 28.7Hz, 38.2Hz, 47.4Hz 57.3Hz 67.7Hz and so on, are clearly visible on the SPL plot as a series of sharp peaks – see Figure 15.21.

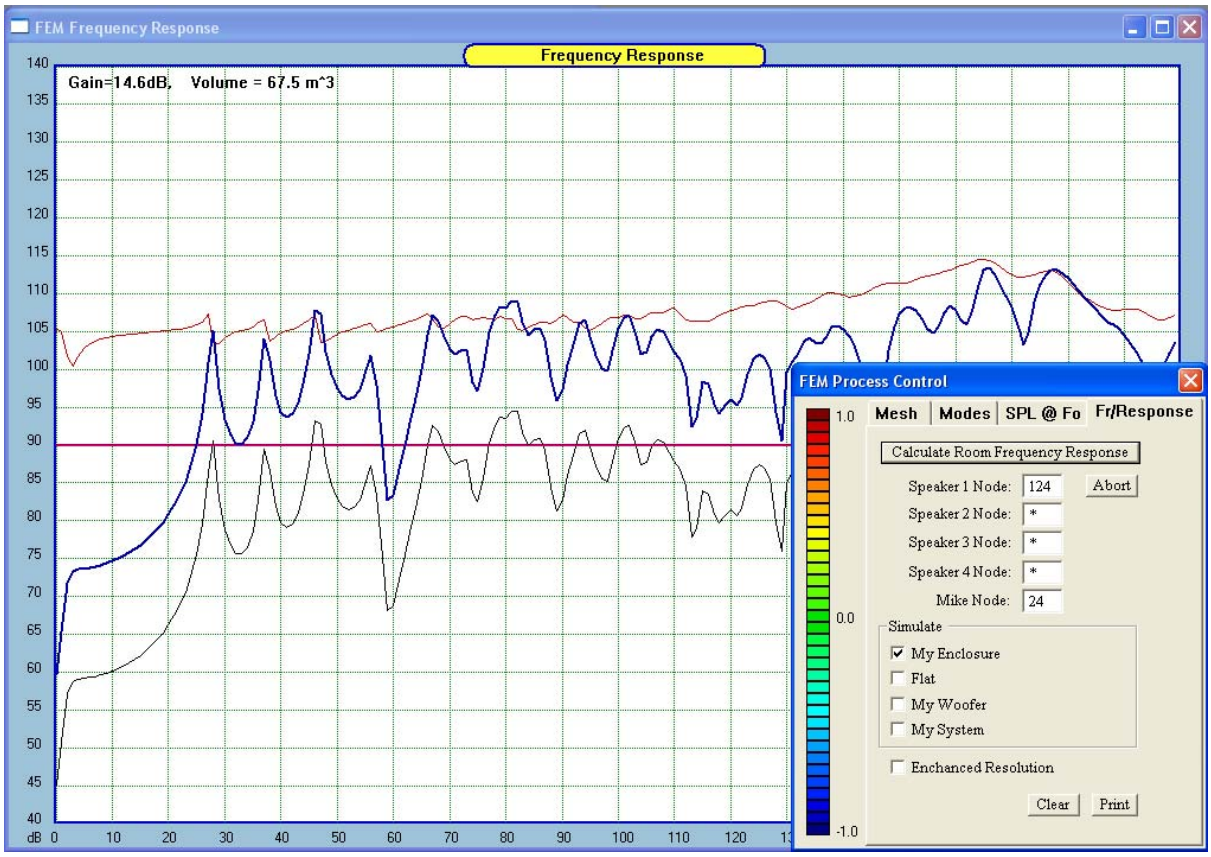


Figure 15.26. The room's modal influence is clearly seen on SPL curve – peaks at low frequencies.

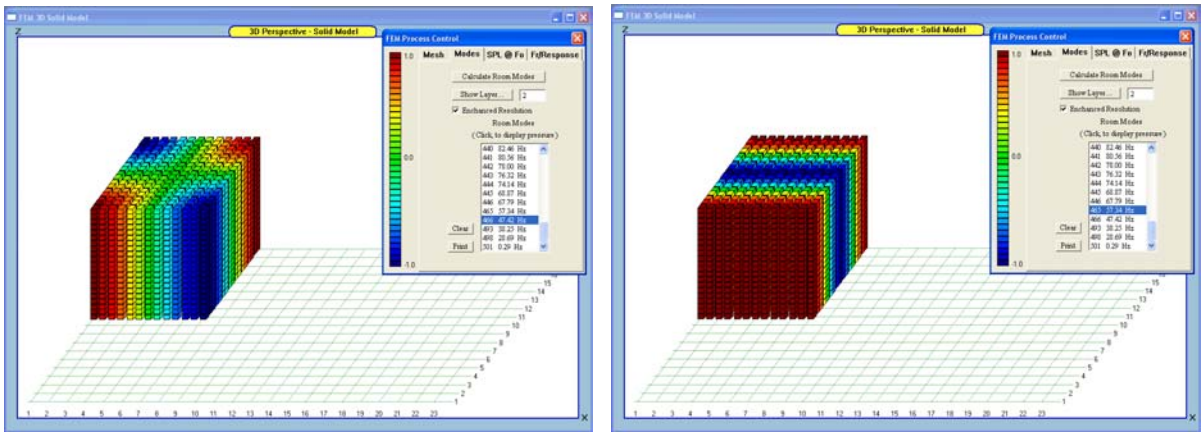


Figure 15.27. Modes: 47.4Hz and 57.3Hz

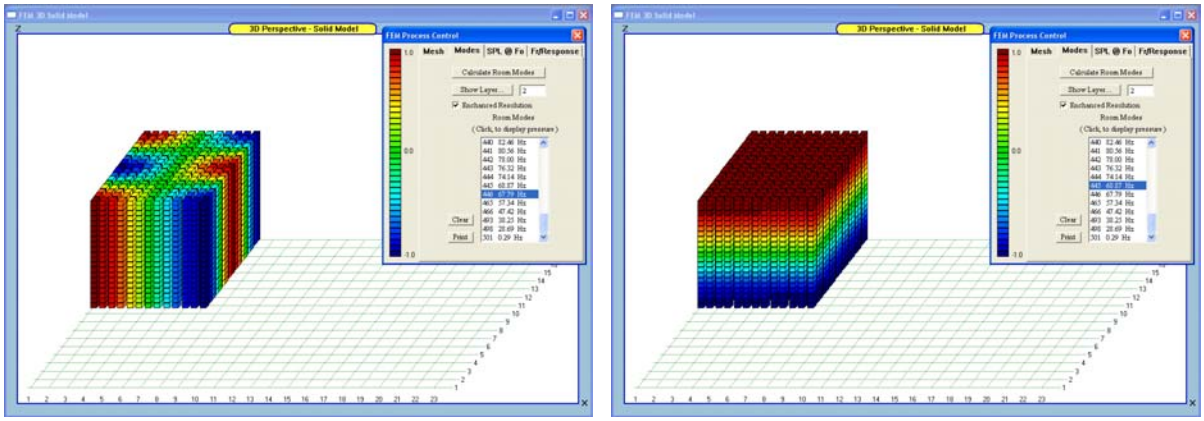


Figure 15.28. Modes: 67.8Hz and 68.7Hz

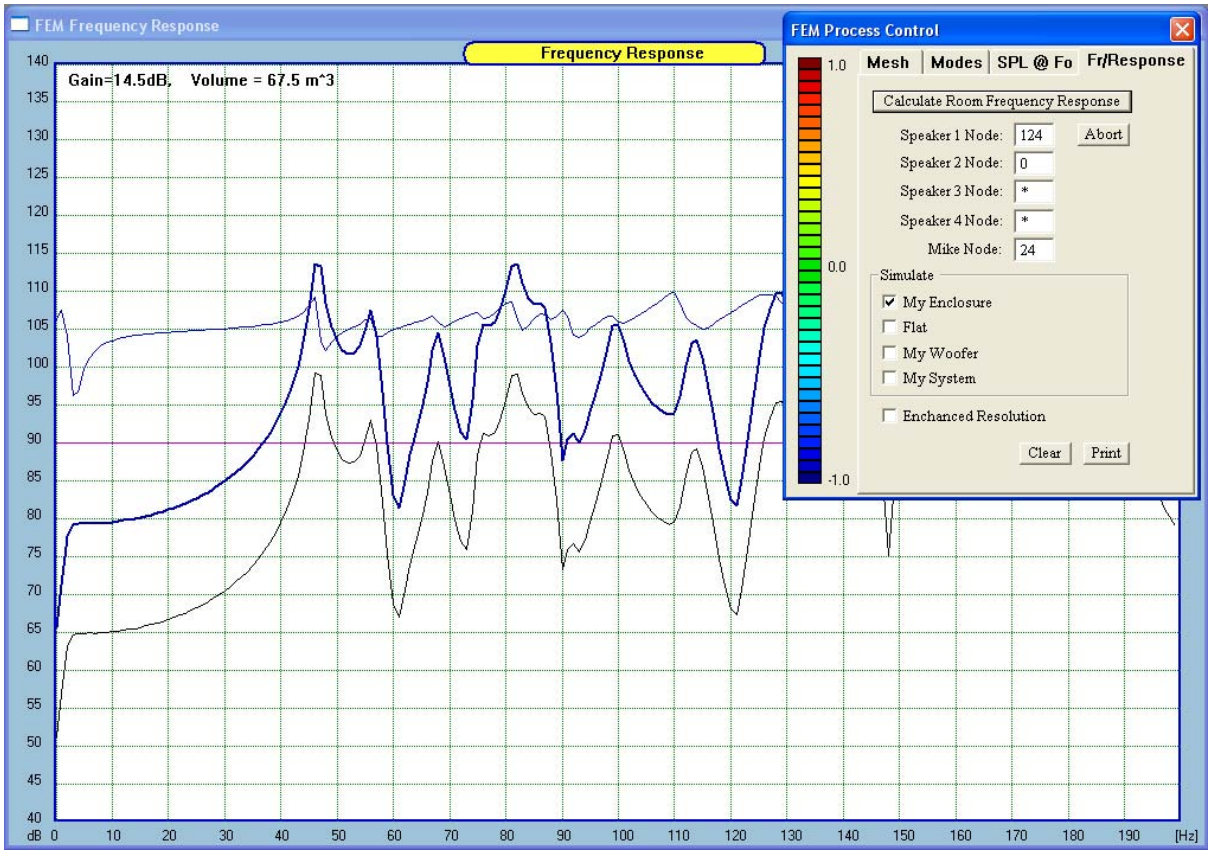


Figure 15.29 Two woofers at opposite corners

As we place more subwoofers in the room, we are able to cancel other low-frequency modes at 28.7Hz and 38.3Hz – see Figures 15.22 and 15.23. There is one **tangential** mode of 47Hz left. This mode generates the same polarity pressure at the opposite corners of the room and as such, our corner subwoofer placement would not eliminate this mode. You may need to use electronic equalization for this mode.

There is another very positive thing happening in our room. If you examine the SPL below 25Hz, you will find, that it has been steadily increasing as we were canceling more nodes using additional woofers. This may be of interest to many readers, as it will be much easier to conceal 4 smaller subwoofers in the room instead of one really large device. At 10Hz (movie special effects) the gain is about 12dB for 4 subwoofers – see Figure 15.30.

Secondly, by distributing subwoofers, we have achieved a 12dB improvement in frequency response smoothness below 50Hz. Please compare Figure 15.26 (107 – 73dB = 34dB variations) and Figure 15.30 (107 – 85dB = 22dB variations). The SPL variation reduction can be further improved by adding more subwoofers.

Anything In-Between – Modal Equalization.

Electronic filtering or equalization for the purpose of improving the perceived quality of sound reproduction via loudspeakers in a listening room has been in widespread use for a number of years now. Intelligent equalization can work, and is useful in improving the overall perceived quality in sound reproduction.

With the advent of currently available powerful capabilities of digital signal processing (DSP) technology and fast desk-top computers, some researchers have become interested in more complex methods of equalization. These methods include “room de-reverberation” or inverting the room transfer function for the purpose of using it as a filter. It is probably rather intuitive to realize, that equalization is **position-dependant**. Audio waves that add in one location will most certainly subtract in another location, or at least interact with differing amplitudes and phases. As the wavelengths are becoming shorter and shorter at higher frequencies, the listener position accuracy is becoming more critical. If taken to absurd level, this would require you to remain very still during your listening experience.

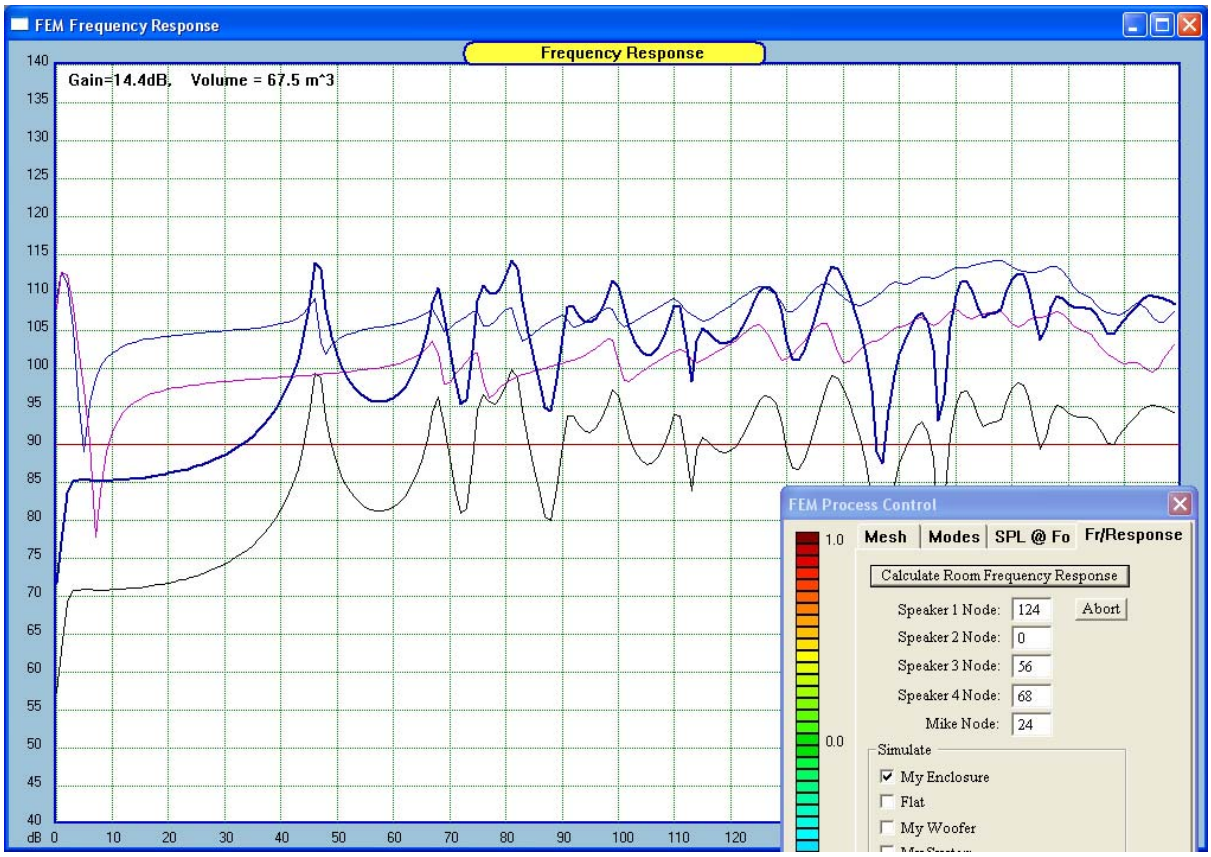


Figure 15.30. Four woofers: two at opposite corners and two half-way along the longest wall

Secondly, it should also be fairly obvious, that any attempt to equalize a dip, would require extra power. And a lots of it too. A 10 dB dip compensation requires 10 times more power. A 20 dB boost uses 100 times more power. Because "dip compensation" is no different than adding a sharp resonance to the frequency response, we have just added a 10 or 20 dB resonance that will be very disturbing at most locations other than the one we are attempting to equalize.

It can not be stressed enough, that the key to successful room equalization is in understanding what can and cannot be corrected with the help of filters or DSPs.

Room resonances, add artifacts to the sound. Some people refer to this as "room playing along with the musicians". In the frequency domain, this effect would manifest itself as peaks and dips in the frequency response. In the time domain, room resonances manifest themselves as amplified and extended ringing of the impulse response. Acoustically, room resonances are those things that are referred to as "boomy bass", "one-note-bass", "muddy bass", "ringing", "coloration". They make it difficult to follow bass notes.

Room resonances behave mostly as minimum-phase phenomena, therefore, they can be equalized with the following in mind.

1. Carefully measure and analyze what the problem is. Attempt to determine where the room modes are and what is their order. Computer modeling and predictions will offer many useful hints here.
2. Is it a peak, that you attempt to deal with or a dip ?.
3. Dips caused by sound-wave cancellations are impossible to equalize. Dips are very position sensitive.
4. Peaks are possible to deal with. Typical room will exhibit modal Q-factors of finite value, possibly in the vicinity of 5-20. This would indicate, that selective SPL output reduction at the offending room modes may actually flatten the frequency response to the acceptable level. This issue can be addressed with carefully matched parametric equalizers.

You may expect, that after peak equalization, the subsequent room measurements would return significantly shorter ringing of the impulse response and significantly smoother frequency response. Electronic, selective modal equalization (Type I) will be discussed at the end of this chapter.

Absorption Coefficient At The Node

It is often important to control absorption of some selected area of your listening room. You may have absorbers fixed on the walls or heavy carpet laid on some area of the room. This information needs to be available to the program for more accurate SPL predictions. The FEM Screen allows you to **assign the absorption coefficient to a single node**, anywhere within your model. The node number you wish to assign the absorption value must be known, so the right time to enter the coefficients would be after you created the model. You can use any of the built-in FEM models or edit your own and save it. Entering the coefficients is quite straightforward.

1. Examine node number you wish the Absorption Coefficient to be associated with. Node numbers for each layer can be displayed as explained before.
2. Knowing the node numbers, you can now DOUBLECLICK on the coefficient's list box to open a small data-entry dialogue for entering **Absorption Coefficients between 0.001 – 1.00 (0.1% - 100%).**

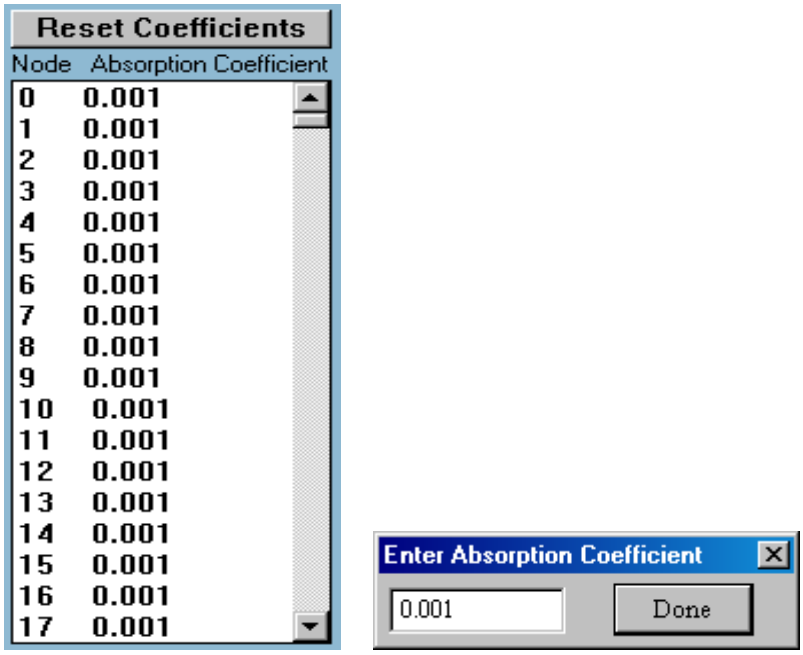


Figure 31. Entering Absorption coefficients.

Large areas with high absorption can dramatically change the pressure distribution within the room. Example on Figure 15.32 shows pressure distribution for the built-in "Test Box 1" at around 110Hz, where all nodes have uniformly identical absorption of 0.001. Figure 15.33 shows dramatic change after the front wall of the room was made totally almost absorbent (AC = 0.99). Please note on the associated frequency response plot, that the "0Hz" pressure mode (always present in enclo0sed spaces) is no longer there for this case.

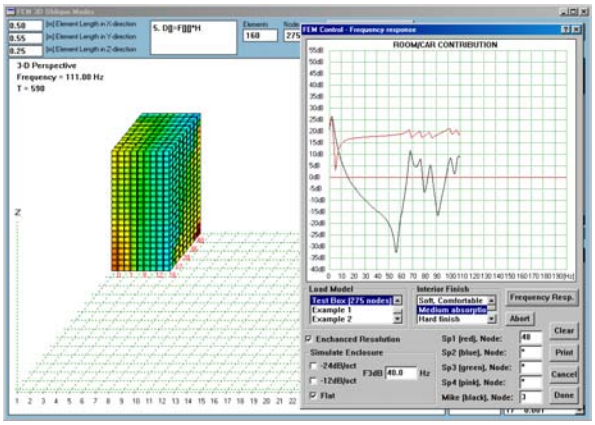


Figure 15.32. AC = 0.001 at all nodes

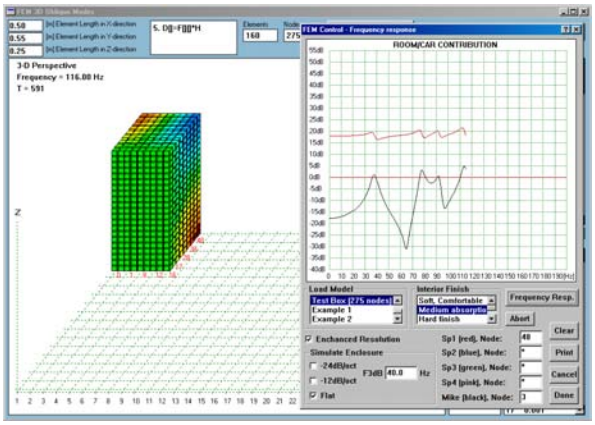


Figure 15.33 AC = 0.99 at the front wall.

It is observable on Figure 15.33, that the nodal pressure pattern for 79Hz node (blue and red colour) is confined to rear corners of the room. It does not extend beyond half –way towards the front of the room. If the room was closed, the blue and red pattern would continue to right to the front wall. The way we defined this example, practically "removes" the front wall from the model. The microphone is placed and Node 3, which is near the left-bottom corner and is right in front of the "missing" wall.

Whit the help of Absorption Coefficient, you will be able to approximate opened windows, carpets, heavy drapes and many other objects made from materials of known absorption. The absorbing area is determined by the location and the number of adjacent nodes with nominated Absorption Coefficients. As a refresher, in the next paragraph, we discuss some basic concepts of Absorption Coefficient.

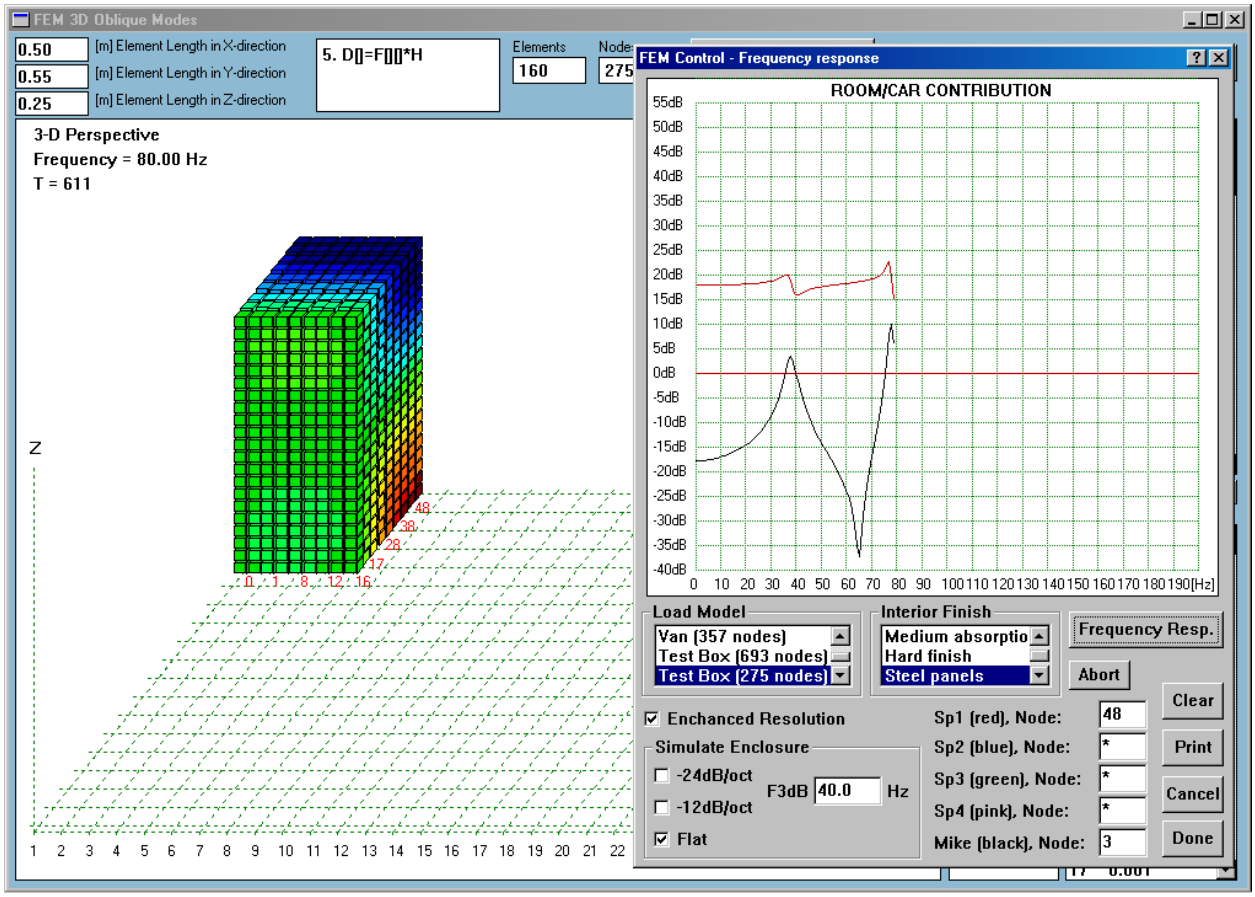


Figure 15.34. Front wall of the room is 100% absorbent (opened to the outside world).

Absorption

Acoustic wave incident on a surface can be absorbed or reflected back. The amount of absorbed energy depends on physical properties of the surface material. We would normally define the fraction of the wave energy absorbed by the material as the **absorption coefficient "a"**.

Typical values of a are between 0 and 1. If our surface is composed of a material exhibiting absorption coefficient of 1 (for instance an open window), then all of the incident energy would be absorbed into the surface. An absorption coefficient of 0.4 means, that 40% of the incident energy would be absorbed into the surface and 60% is reflected back into the room.

The ability to absorb sound will vary with frequency for any given absorbing material. Typically, higher frequencies are better absorbed than lower frequencies because sound power of a typical audio spectrum diminishes with frequency, therefore the absorbing material does not need to works as hard.

There are some interesting materials listed in the absorption table. For instance, heavy folded curtains absorb nearly all sound energy at 4kHz and above. This would be a good choice for the back wall of the listening room if you need to suppress the rear reflections at those frequencies. Another material good for mid-range frequencies would be thick fiberglass, followed by acoustic paneling.

Placement and absorption characteristics of the material will help you modify the acoustical properties of your listening room. You can (1) control early reflection so some degree and (2) affect total reverberation time of the room.

The FEM screen models the SPL response for frequencies below 200Hz. We would recommend using the Absorption Coefficients from the first column (125Hz) from the table below.

Tables of absorption coefficients are published, usually stating the figure at six standard frequencies. Shown below are the coefficients of some common substances at these frequencies:

Material	**125 Hz**	**250 Hz**	**500 Hz**	**1kHz**	**2 kHz**	**4 kHz**
	a	a	a	a	a	a
Concrete	0.01	0.01	0.015	0.02	0.02	0.02
Plasterboard	0.29	0.1	0.05	0.04	0.07	0.09
Acoustic panelling	0.15	0.3	0.75	0.85	0.75	0.4
Ac. Tiles	0.1	0.35	0.7	0.75	0.65	0.5
Brick	0.024	0.025	0.03	0.04	0.05	0.07
Carpet, thin	0.05	0.1	0.2	0.25	0.3	0.35
Carpet, thick,u/lay	0.15	0.27	0.35	0.42	0.5	0.6
Curtains	0.05	0.12	0.15	0.27	0.37	0.5
Curtains,heavy,folded	0.2	0.35	0.5	0.65	0.8	0.95
Fibreglass, 1 in	0.07	0.23	0.42	0.77	0.73	0.7
Fibreglass, 2 in	0.19	0.59	0.79	0.92	0.82	0.78
Fibreglass, 4 in	0.38	0.89	0.96	0.98	0.81	0.87
Floor,wood on joists	0.15	0.2	0.1	0.1	0.1	0.1
Floor,wood blocks	0.05	0.05	0.05	0.07	0.1	0.1
Glass	0.03	0.03	0.03	0.03	0.02	0.02
Plaster on lathe	0.3	0.2	0.1	0.07	0.04	0.03
Plaster on brick	0.024	0.027	0.03	0.037	0.039	0.034
Plywood, 3/8 inch	0.11	0.11	0.12	0.11	0.1	0.09
Ply, 3/16 on 2 in batten	0.35	0.25	0.2	0.15	0.05	0.05
Wood, 3/4 in solid	0.1	0.11	0.1	0.08	0.08	0.11

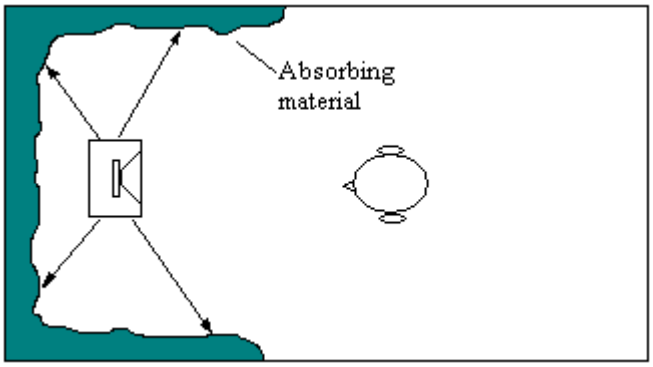


Figure 15.35 Absorbing material eliminates most of the early reflections from behind the speaker

In order to measure how much sound a particular surface material and area absorbs, we multiply the area (in square meters) by the absorption coefficient, to get the result in **Square Meter Absorption Units** or **smau** (the unit of sound absorption based on the square meter).

Average Absorption Coefficient

$$\bar{a} = \frac{a_1 s_1 + a_2 s_2 + ...a_n s_n}{S}$$

Where $a_1, a_2 \ldots a_n$ are the individual absorption coefficients of surfaces $s_1, s_2, \ldots s_n$.

Modeling Real Sub-Woofer Placed in Room

Evaluating real loudspeaker performance in-room can be accomplished by measuring or importing the "free-air" or anechoic frequency response of the loudspeaker and then using the FEM screen to create room transfer function. The "free-air" loudspeaker frequency response is automatically incorporated in the room transfer function if you check the **"My Woofer"** checkbox in the **"FEM Control – Frequency response"** dialogue box. Sub-woofers are modeled as a point sources located at the assigned nodes (Node 48 in the example below). You can place up to 4 sub-woofers in the room.

Please remember, that the FEM screen uses linear frequency scale 1 – 200Hz.

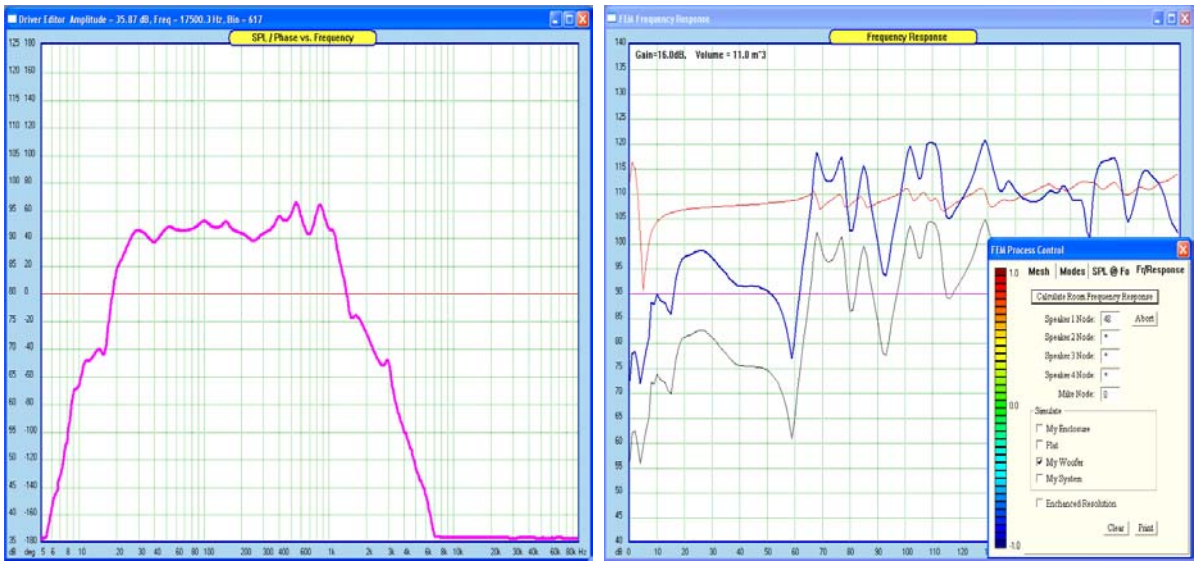


Figure 15.36. "Free air" and "in-room" frequency response of the same driver.

Electronic Selective Modal Elimination

As we mentioned before, loudspeaker-room system is dominated by the room properties for the rate of energy decay. At higher frequencies, typically above a few hundred Hertz, the energy decay is controlled to a required level selectively placing necessary amount of absorbing material in the room. This is generally adequate, as long as the wavelength of sound is small compared to dimensions of the room. At low frequencies passive means of controlling reverberant decay time become more difficult because the physical size of necessary absorbers increases and may become prohibitively large compared to the volume of the space, or absorbers have to be made narrow-band. Related to this, the cost of passive control of reverberant decay greatly increases at low frequencies. One option left is Electronic Modal Elimination using notch filtering approach. Example circuit is shown on Figure 15.29.

A6 output impedance (see Figure 15.29) is the used as one of the GYRATOR's components. Output is Node 4 and input is Node 0. Approximated value of the inductor, L, created with the gyrator:

L = (R7 – Rout)*Rout*C6

Rout = 100 ohm, output impedance of the A6. Rout should be selected from 100-470 ohm. Approximated value of the inductor's, Q, created with the gyrator:

$$Q = XL/(Rout+R4)$$

Methods for optimizing the response at a listening position by finding suitable locations for loudspeakers have been proposed but cannot fully solve the problem. Because of these reasons there has been an increasing interest in active methods of sound field control at low frequencies, where active control becomes feasible as the wavelengths become long and the sound field develops less diffuse.

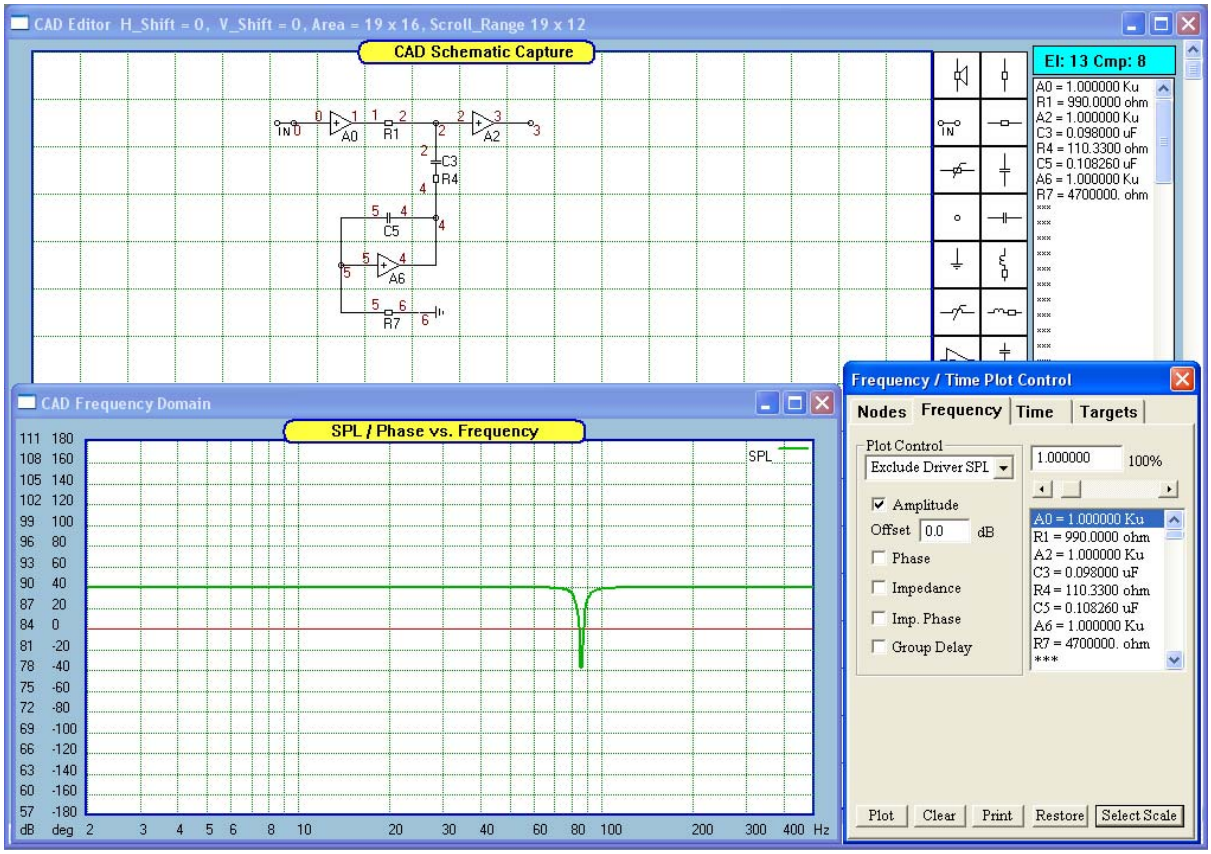


Figure 15.37. Electronic equalizer notch set to Fo = 87Hz.

Modal resonances in a room can be audible because they modify the magnitude response of the primary sound or, when the primary sound ends, because they are no longer masked by the primary sound. Detection of a modal resonance appears to be very dependent on the signal content. Olive et al. reports that low-Q resonances are more readily audible with continuous signals containing a broad frequency spectrum while high-Q resonances become more audible with transient discontinuous signals.

Traditional magnitude equalization attempts to achieve a flat frequency response at the listening location either for the steady state or early arriving sound. Both approaches achieve an improvement in audio quality for poor loudspeaker-room systems, but colorations of the reverberant sound field cannot be handled with traditional magnitude equalization. Colorations in the reverberant sound field produced by room modes deteriorate sound clarity and definition. Modal equalization is a novel approach that can specifically address problematic modal resonances, decreasing their Q-value and bringing the decay rate in line with other frequencies.

Modal equalization decreases the gain of modal resonances thereby affecting an amount of magnitude equalization. It is important to note that traditional magnitude equalization does not achieve modal equalization as a by-product. There is no guarantee that zeros in a traditional equalizer transfer function are placed correctly to achieve control of modal resonance decay time. In fact, this is rather improbable. A sensible aim for modal equalization is not to achieve either zero decay time or flat magnitude response. Modal equalization can be a good companion of traditional magnitude equalization.

A modal equalizer can take care of differences in the reverberation time while a traditional equalizer can then decrease frequency response deviations to achieve acceptable flatness of response. We have presented two different types of modal equalization approaches, Type I modifying the sound input into the room using the primary speakers. Type I systems are typically minimum phase. There are several reasons why modal equalization is particularly interesting at low frequencies. At low frequencies passive means to control decay rate by room absorption may become prohibitively expensive or fail because of constructional faults. Also, modal equalization becomes technically feasible at low frequencies where the wavelength of sound becomes large relative to room size and to objects in the room, and the sound field is no longer diffuse. Local control of the sound field at the main listening position becomes progressively easier under these conditions.

Type I system implements modal equalization by a filter in series with the main sound source, i.e. by modifying the sound input into the room. Modal equalization is a method to control reverberation in a room when conventional passive means are not possible, do not exist or would present a prohibitively high cost. Modal equalization is an interesting design option particularly for low-frequency room reverberation control.

Having designed suitable notch filter using CAD screen, you **must plot its frequency response** to place the SPL data in the location accessible to FEM screen. To include the plot from CAD screen in FEM calculations, simply check the "My System" check box on the "FEM Control – Frequency response" dialogue box.

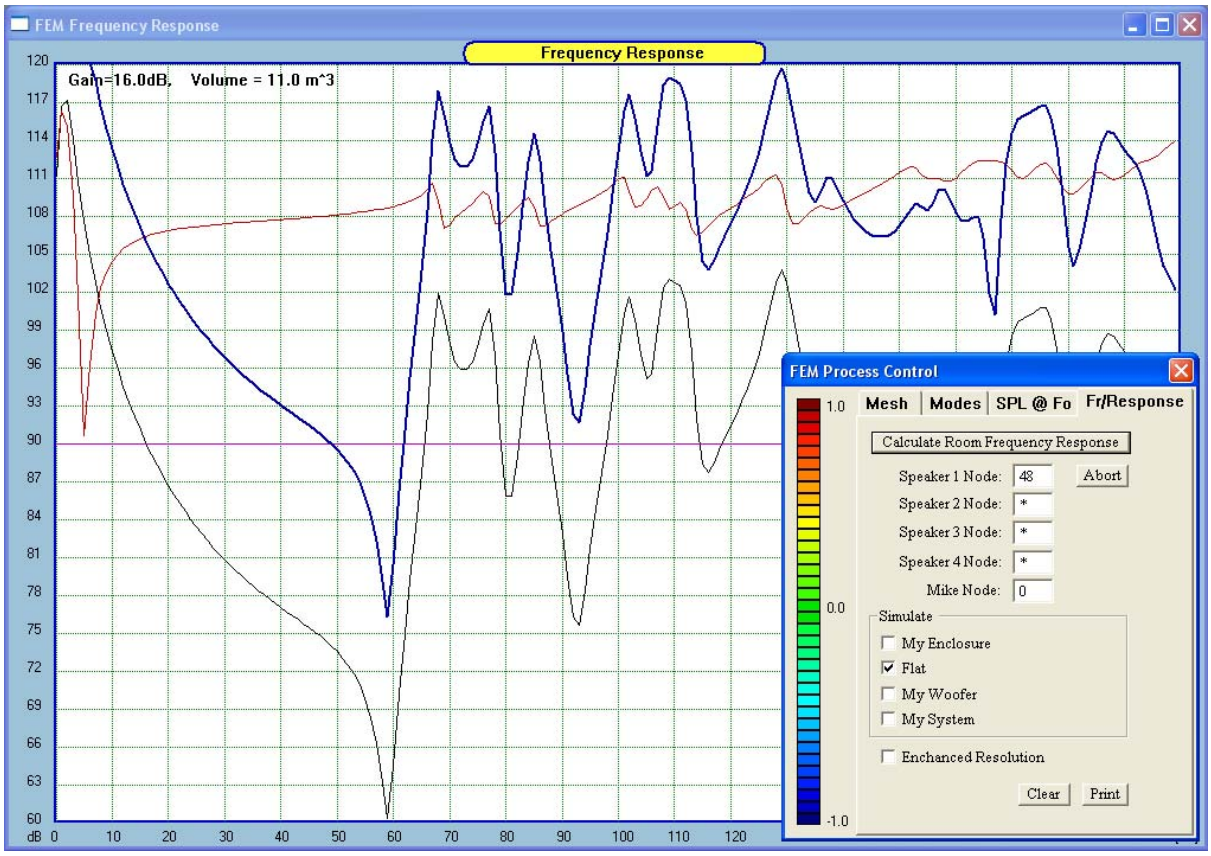


Figure 15.38. Room transfer function **before** EQ applied @ 87Hz.

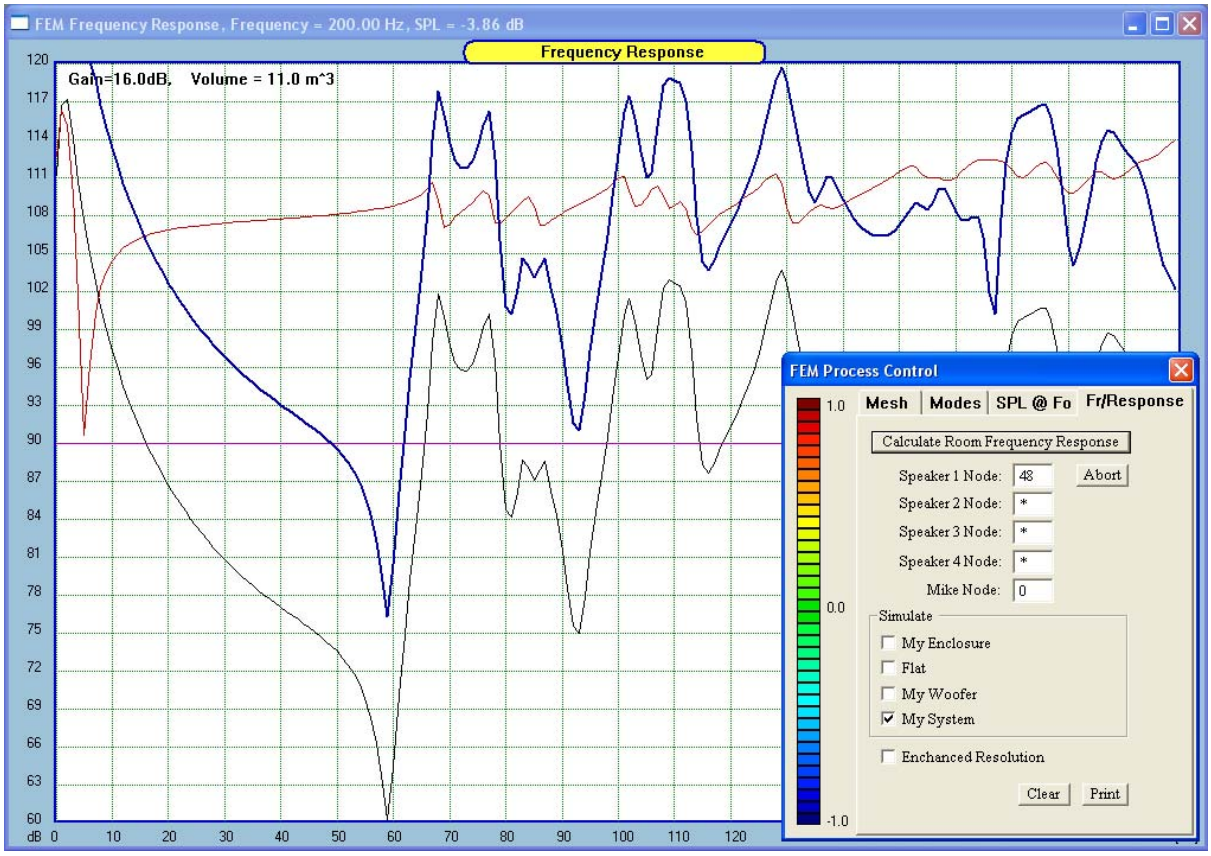


Figure 15.39. Room transfer function **after** EQ applied @ 87Hz.

EasyLab Advanced

Chapter 16. Linear Analog And Digital Measurements

Short Review of the Measurement Methods for SPL/Phase

When measuring the SPL of your loudspeaker, the utmost attention must be applied **to keeping the environmental effects out of the measurement**. The typical goal of an SPL measurement is to measure the transducer or system, without measuring the environment. To achieve this goal several methods of SPL measurement have been proposed. To measure SPL without environmental reflections, two methods can be employed:

1. Setup the environment so as not to produce any significant reflections, or
2. Use gating to remove reflections.

When making continuous-tone SPL measurements, any walls or other reflecting surfaces will produce multi-path signal reflections, which will be summed together into the response measurement. These reflections will result in significant irregularities (narrow peaks and dips) in the response curve that are not actually there in the transducer itself. As such, these measurements are environment dependant if the goal of the measurement is to analyze only the loudspeaker and not the room plus the loudspeaker. There are several types of environments which can be employed to eliminate, at least to a large degree, any multi-path reflections caused by walls or boundaries. In fact, these methods are recommended as the ones which can be used when measuring full range frequency response curves down to low frequencies. Gating method tends to be effectively used at mid and high frequencies, but still does not solve the problem of low frequency measurement. Valid results are basically obtained from the frequency where the period of the sinusoidal test signal is smaller than the gate "ON" duration. Continuous-tone SPL measurements can be successfully carried out in several types of environments:

1. Anechoic chamber,
2. Ground plane,
3. Half-space and
4. Near-field.

Anechoic Chamber Type Measurements

Anechoic measurements require either a specially built anechoic chamber, or suspending the speaker in mid-air, far from any boundaries. This method includes the enclosure diffraction effects. The purpose of an anechoic chamber is to absorb all sound striking the walls so that no reflection occurs. This is typically accomplished by covering the walls of a room with sound absorbing materials (pyramid-shape). However, even rigorously designed chambers still do not provide the ideal absorbing properties at low frequencies, where the total absorption is extremely difficult to achieve. For this reason most chambers exhibit a constant pressure behavior as frequency is decreased. At low frequencies the pressure field will cause a rise in level with decreasing frequency typically 12dB/Octave. In the mid and high frequencies the chamber will usually have good anechoic characteristics. True anechoic measurements are beyond the reach for most of us.

Ground Plane Type Measurements

Ground plane measurements are quite similar to anechoic measurements. Because the measurement includes both the direct sound from the enclosure, plus the reflection from the ground plane, the SPL for the same microphone placement distance will be 6dB higher than the anechoic SPL level. The only requirement for a ground-plane measurement is a large reflective surface, with no obstructions for a radius of 8-10 meters or more.

The procedure is to place the loudspeaker on its side on the ground, and then place the microphone on the ground in front of the speaker. This method includes the enclosure diffraction effects. The increase in SPL is about 6dB, the frequency being determined by the total area of the baffle. The ground plane method is a very good (inexpensive) testing solution and all that is required is a flat surface with some open area around it.

Half-Space (Pit) Measurements

Half-space or pit measurements are made by placing the loudspeaker in the small pit ground with the drivers facing up, and the microphone placed right above the enclosure. Again, the microphone should be directly in line above the high frequency driver. This type of measurement does not include the enclosure diffraction effects. The radiation field is half of a full sphere so the level is 6dB higher than a free field measurement.

Near Field Measurements

The measurement methods discussed in the sections above are examples of far field measurements. The microphone is far enough away from the source that the sound radiation at the microphone is spherical. If the microphone is moved to a very close distance near the piston surface of the source, the radiation becomes a pressure field. This is known as the near field. By placing the microphone closer to the source, (5 mm) boundary reflections are eliminated since the reflection amplitude relative to the direct amplitude will be much smaller.

This type of measurement does not include the enclosure diffraction effects.

The recommended distance applicable to this method is dependent on the frequency range being measured. For measuring woofers, placing the microphone within 5mm from the cone will produce adequate near field curves. For ports, the test microphone should be placed in the port mouth. For midrange drivers placing the microphone as close as possible without touching the cone is necessary. For tweeters near field measurements may not be practical.

A good choice would be to take near field measurements on the woofer and port, and a gated measurement on the full enclosure to capture the mid and high frequency ranges. These curves could then be merged together to form a complete full range curve.

In far field measurements, the SPL output from the woofer and port are summed together acoustically. When taking near field measurements on the woofer, the port output does not appear in the pressure at the woofer. For this reason, ported enclosures require separate measurements to be taken at the woofer cone, and at the port. These must then be summed together mathematically.

The real goal here is to be able to measure the near field SPL, and then to predict what the SPL would be at a far field distance such as 1 Meter. This can be done if the piston area, Sd, for the source is known. It is simply a matter of scaling. The following formula gives the ratio of far field pressure to near field pressure at 1 Meter. Scaling Factor for Near Field Measurements (half-space):

$$\mathbf{P(far) = P(near)*20\log[\ 0.2821*\sqrt{(Sd)}\ ] \qquad [Sd \text{ in } m^2]}$$

For a full-space environment, you would divide this ratio by 2, (or subtract 6dB).

Near-field loudspeaker measurements correctly predict the true full-space, far-field response only in the driver's piston range of operation below roughly f = 10897/d Hz where d is largest dimension of the largest diaphragm or port in centimeters.

Maximum-length Sequences

The program measures a linear, time invariant (LTI) system's impulse response using a pseudo-random excitation called a maximum-length sequence (MLS). ML sequences have some unique theoretical and practical features separating them from other test signals commonly used to measure linear systems such as narrow pulses, white noise and swept sine signals

Mathematical properties of the MLS excitation allow efficient computation of very long impulse responses. When a large FFT is applied to an extended wide-band impulse response the result is a wide-band transfer function with very fine frequency resolution. Therefore, there is no need for zoom processing with the MLS approach. SoundEasy uses FFT of the same length as MLS (well, plus 1 of course). Therefore, there is no FFT length selection.

When using ML sequences for LTI system's measurements you should **keep in mind, that the MLS method is accurate only for LTI systems** meaning their transfer function does not change over the measurement interval. Therefore, you should not attempt to make room measurements while the room acoustics are changing. This might occur if enough people are moving around a room during a measurement session. The opposite is also true – if your sound card (or PC) is unable to keep the In/Out timing accurate, the MLS measurement system will not work.

Basic MLS/FFT specification

The program will generate 6 different MLS sequences and use associated FFTs as shown below:

MLS order	MLS length	FFT length
13	8191 points	8192 bins
14	16383	16384
15	32767	32768
16	65535	65536
17	131071	131072
18	262143	262144

Smoothing: 1/3, 1/6, 1/12, 1/24 and 1/48 octave smoothing
Sampling: 11025, 12000, 22050, 24000, 44100, 48000, (96000 where applicable).

The MLS signal is periodic with period 2^N -1 where N is the order of the sequence. The MLS generator is clocked by the sample rate clock. Therefore, the period of a MLS, in seconds, is equal to the sampling interval times the sequence period in points. For example, for a 48 kHz sampling rate (20.8 microseconds sampling interval) a 262144-point MLS will have a period of 5.452 seconds.

The MLS can be regarded as a train of pulses of equal amplitude but appearing in positive or negative direction in pseudo-random manner. Such signal is fed into the loudspeaker, so that he microphone will pick up the response to these pulses. The application of Hadamard transform, causes the individual responses at different times to be re-arranged, so they appear simultaneous and in one direction. It is now easy to understand, why the LTI requirement for the system – it must not change during the measurement.

True Frequency Resolution

The system transfer function is computed by applying an FFT to the impulse response or a portion of it as in the case of windowed, anechoic loudspeaker measurements. The displayed frequency resolution is given as the sample rate divided by the FFT size (i.e. 48000/262144=0.183Hz). Displayed frequency resolution is just the FFT bin separation. However, the true frequency resolution of a measurement is **at best equal to 1/T** where T is the length of the impulse response segment used in the FFT calculation (i.e. the width of a rectangular time window). **Thus, it is possible to have a very fine displayed resolution but a rather coarse true resolution.** For example, you could apply a 32768-point FFT to a 512-point segment of impulse response. In that case, the large FFT only acts to interpolate the true frequency resolution to form a much finer displayed resolution.

Data-tapering window shapes would yield even worse resolution than a rectangular window. In the frequency domain, SoundEasy indicates the true frequency resolution by **a gray-shading the invalid section of the graphics display**. The data falling in the grayed frequency range should be regarded as invalid.

For example, if the time window T = 4 milliseconds the lowest accurate frequency is 250 Hz. The true frequency resolution at all higher frequencies will, in this case, also be 250 Hz. Therefore, the next correctly measured and calculated data point will be at 500Hz, and the next at 750Hz, and so on…..
Please note, that between 250Hz and 750Hz there is only ONE valid data point !.

To obtain better true resolution you must measure an extended impulse response and use an equally long time window and FFT size to compute the transfer function – go outdoor, close-mike or anechoic.

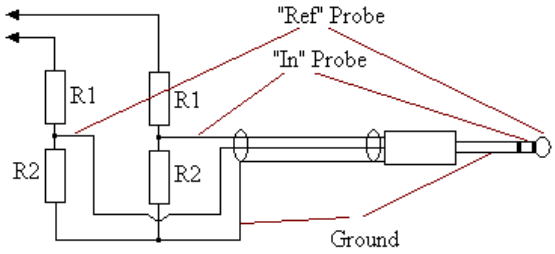
Noise In Presentation of MLS Measurement Results

This is significant issue. Virtually all MLS measurement systems include some kind of **averaging** of the results in time/frequency domains, necessary to produce better quality graphs. Typical figures are: average up to 16 MLS periods for a 12 dB SNR improvement. Further improvement can be obtained with 50-100 MLS runs. Please be aware, that if you use 262144-long MLS, clocked out at 48000Hz, you will generate 5.5 seconds of noise. The 100 MLS runs will result in 550 seconds (9.2 minutes) of loud noise. This timing becomes quite comparable with analog sweep measurement time. Averaging in frequency domain is typically performed for 1/3, 1/6, 1/12, 1/24 and 1/48 octave smoothing. Several other innovative, noise reduction schemes have been adopted in MLS software available on the market. This release of SoundEasy is designed to be a fast, high-resolution, single-shot, MLS tester, providing you with good quality presentation graphs. Therefore, we have implemented our propriety averaging scheme coupled with the Hilbert-Bode Transform to provide you with tools and choices capable of generating data over any desired frequency range**. The B-H Transform, on the top of it's usual task, essentially acts as a filter, generating and perfectly smoothing data outside the selected band – below "HP stop" and above "LP start".**

Additional Hardware Needed

No hardware devices are supplied with the program. You will need additional hardware as listed below:

1. **Audio power amplifier** capable of about 50W of power. The amplifier does not need to be of a "test equipment" grade. Any device with -80dB noise floor and the –3dB frequency range of 5Hz-30000Hz will do. The frequency/phase response of the amplifier will be automatically compensated for by the software. Input impedance of the amplifier need to be at least 10 times higher than output impedance of your sound card.
2. **Test microphone**. We have used CLIO "pencil type" electret microphone with a very simple, single-OPAMP, pre-amplifier. Bias voltage for the microphone was provided by the pre-amplifier. We live the choice of the microphone and pre-amplifier entirely to you. Just bear in mind, that the maximum non-clipped input voltage to your sound card is in the vicinity of 2V, so your test microphone + pre-amplifier should be able to provide this much voltage.



3. **Resistive probes**. For impedance measurements at 2.83V level, a simple voltage dividers of R1=47kΩ, R2=22kΩ may be sufficient. If you need to measure the impedance curve for different power levels, you may need to select different divider ratio. For SPL measurements, you may need to use different dividers again, depending of your particular voltage of interest.

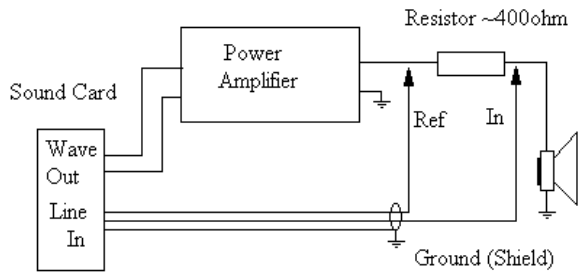
4. Additional audio cables as required.

As you can see, the above list is short and simple. You may already have your power amplifier, resistors and cables ready for use and the only piece of hardware you will need is good quality test microphone.

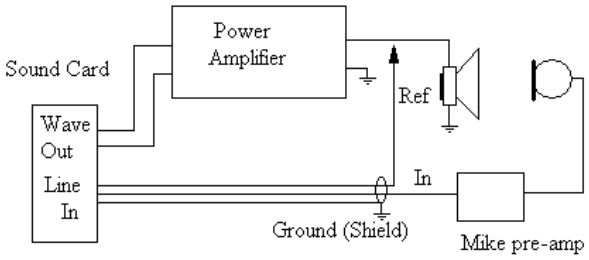
Operation of the MLS (Digital) Screen

The screen uses **two-port measurement technique**. One port ("Ref" probe) provides reference signal for amplitude and phase, and the other port ("In" probe) is actually used to measure the characteristics of the device under test – your loudspeaker. This simple arrangement allows the program to automatically calibrate out the amplitude/phase response of the power amplifier needed for the tests and also the amplitude/phase of the output circuitry of the sound card. Microphone calibration SPL/phase data needs to be imported into the screen.

When making impedance measurements, Zin, the "Ref" probe, sensing V1, is connected to the output of the power amplifier and the "In" probe, sensing V2, is connected right across the loudspeaker. The measured impedance is calculated from the formula: Zin = R*(V2/V1)/(1 – V2/V1). It is easy to observe, that V1 is a reference signal for voltage and phase. Phase of the input impedance is simply the difference between the phase of voltage V1 and the phase of voltage V2.



When making SPL measurements, the "Ref" probe, sensing V1, is connected to the output of the amplifier and the "In" probe, sensing V2, is connected to the microphone pre-amp output. The measured SPL (V2) is corrected in amplitude and phase accordingly to amplitude and phase measured by V1.



The **SPL measurements are relative**. They do not show the actual, calibrated results in dBspl. However, you can actually obtain very good results if you correlate MLS SPL screen curves with the indication of an inexpensive (but calibrated) SPL meter.

There is a range of SPL meters available on the market and you should get one for your home lab. For the best accuracy, correlation could be performed using "half-space" technique to avoid multi-paths or echoes picked up by the SPL meter.

Measurement Method for SPL

The MLS measurement technique involves three basic steps:
A. Exciting the loudspeaker under tests with the MLS signal.
B. Correlating the captured response using Hadamard Transform - impulse response of the loudspeaker.
C. Taking FFT of the impulse response to produce the frequency response curve.

Other parameters relate to FFT length, MLS length, sampling frequency and smoothing. Window type and length depends what type of measurement your are performing. You can easily get the job done selecting the highest sampling frequency, longest MLS and FFT and 1/12 octave smoothing. Follow the steps below to perform MLS measurements of the sound pressure (SPL/phase) of your loudspeaker.

1. **Connect the test setup**: power amplifier, test probes, microphone and the "loudspeaker under test".
2. **Generate MLS sequence using "Acquire IR" TAB**
   - Select "MLS Length" from the provided list box – on SB Live! Sound card set to 131071.
   - Select sampling rate from the provided list box– on SB Live! Sound card set to 48000.
   - Select Generator level from 0% and rise it slowly to the required level.

- Press “Run MLS [SPL]” button. You should now hear the noise coming out from the measured speaker. The microphone will pick-up this signal and your “Input Level LED” indicator should lit-up for “In” bars and “Ref” bars. **The “In” indicator corresponds to the MEASURED signal from your microphone and the “Ref” indicator corresponds to the REFERENCE signal probe (connected to the output of the power amplifier).**

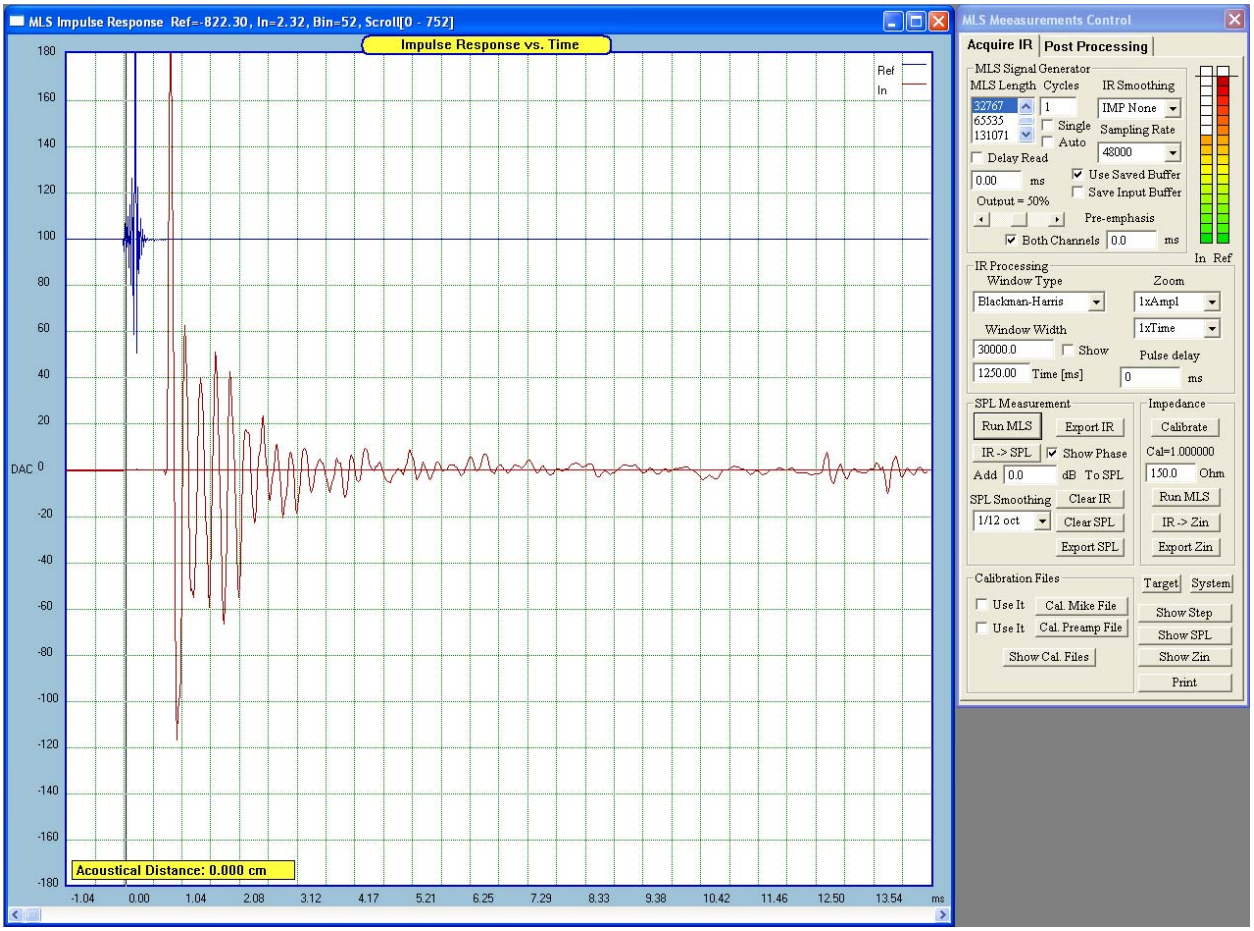


Figure 16.1. Impulse responses of "Ref" signal - TOP TRACE and **"In" signal - LOWER TRACE**

3. **Select windowing parameters**.
   When the MLS sequence has finished, you should see the impulse response of the measured signal (from the microphone) on the screen. The program will automatically place the time cursor 8 samples to the left of the peak value.
   - Position the time cursor on the screen now. This is accomplished by simply moving the mouse pointer onto the time domain screen and clicking the LEFT mouse button.
   - In order to remove the “time-of-flight” sound delay, position the time cursor right before the impulse starts to rise. You can do the fine cursor adjustments in 1 step increments/decrements by pressing the **“<-” and “->” Cursor buttons on your keyboard**.
   - Select the type and the width of the time window. For “close microphone” techniques you can select wider windows and for echoic environment, you should select narrow windows, aiming at cutting out the first echo.
   - Please DO NOT create wider windows than the length of the MLS.
   - **“Show”** – Check-box enabling the window to be displayed.
   - “**Type”** – Select window type using this list box
   - **“Width**” – Enter window’s width in **“samples”** into this data field.
   - **“Time”** – Enter window’s width in **“miliseconds”** into this data field.
4. **Select smoothing parameter**.
   This is a simple selection of one of the provided smoothing ratios: 1/3, 1/6, 1/12, 1/24 and 1/48 octave.

5. **Select additional delay.**
   - Adding a pure delay (enter **positive** number in the "Pulse Delay" field) causes the phase curve to tilt downward from left to right when displayed on a frequency scale. Conversely, removing a delay (enter **negative** number in the "Pulse Delay" field) causes the corresponding phase curve to tilt upwards.
   - A basic problem in loudspeaker phase measurements is that there is a large acoustical delay associated with the space between the loudspeaker and microphone. Placing the marker just before the Impulse response first arrival removes most of this delay in the time domain. Any residual delay can be removed through the "Pulse Delay" in the frequency domain. This field allows you to manually enter any desired phase delay.
6. **Generate SPL curve**.
   Simple press the **"IR->SPL"** button and wait a short while for the calculations to finish. Please note, that wider smoothing band will result in increased computation time.
   - "**Show Phase**" – check this button if you prefer phase response to be plotted as well.
   - "**Zoom**" – You can zoom in on the Impulse Response amplitude and time domain by selecting desired magnification from the provided list boxes.

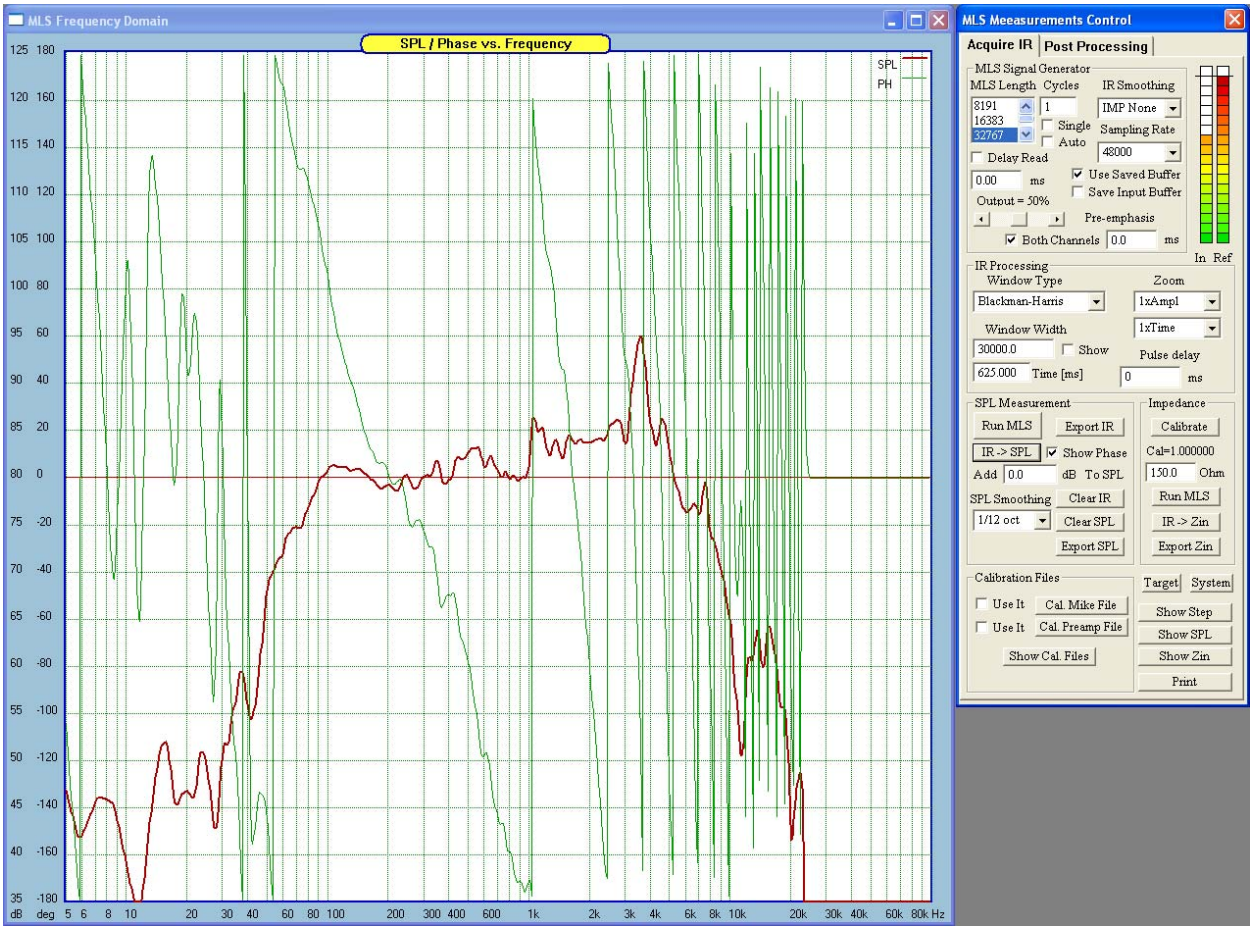


Figure 16.2 Example of SPL/phase response (1/12 oct smoothing) with echoes due to window of 30000 samples.

7. **If you need to extend the measured SPL/phase curves beyond the measurement frequency range**.
   This may happen if you selected screen frequency range wider then the sound card measurement frequency range. For example: reliable (with automatic equalization via Ref probe) frequency range of SB Live! card is approximately 10Hz – 20kHz. You have selected SoundEasy screens 1Hz 100kHz wide for your project work.
   The tool to use in this case, is the Hilbert-Bode Transform group-box provided at the right-bottom corner of the measurement screen. The transform is driven by 5 editable parameters and they should be selected to obtain the **best match for phase and amplitude between measured signal and calculated transform over the widest frequency range**. Typically, good match can be obtained between 20Hz and 20kHz. Generally, you would select the parameters to provide initial match for amplitude response and then fine tune the parameters to provide the match for phase.

Please note, that "Roll-off" parameters can be as high as +/- 100dB to +/- 120dB if the measured frequency response is highly irregular (resonant peaks) at the extremes of the measurement range. The program calculates phase using a Hilbert-Bode Transform from any magnitude curve. This is known as a Minimum Phase response and it does not contain any delay or positional information. It is a phase response produced solely from the magnitude data curve. This is very convenient for SPL measurements since the distance from the source to the microphone is completely unimportant. The phase curve will have no excess delay. This eliminates the usual problem of removing the arrival delay from the path of the source to microphone. "A minimum phase system is one which is able to transfer input energy to its output in the least amount of time for a given frequency response". Mathematically, you would calculate the phase response of a minimum phase system by taking the Hilbert-Bode Transform of its log magnitude response (i.e. magnitude expressed in decibels). This relationship is also reciprocal: the log magnitude response of a minimum phase system equals the Hilbert-Bode transform of its phase response. Because the Hilbert-Bode Transform requires an infinite integral over all frequencies, however, any numerical computation of minimum phase is necessarily an approximation. Understanding the characteristics of the phase transform and how it works can be very important if maximum accuracy is to be obtained. **HBT is available from "Post Processing" TAB**.

For some data curves, the transform can be run directly and will provide excellent results with little attention to the original magnitude data. In other cases, a far better result can be obtained by correcting certain areas of the magnitude curve before running the minimum phase transform. Typically, the regions which can cause problems are **at the ends** of the measurement bandwidth. The transform must project the slope of the magnitude curve beyond the measured bandwidth. It does this by taking the last slope found as the curve reaches the low and high ends of the frequency range. It then assumes that: (1) the slope found at the low end will remain constant down to DC, and (2) the slope found at the high frequency limit will remain constant towards infinity. For this reason it is of fundamental importance that the slope of the magnitude curve at the high and low ends of the frequency range be approaching or already at their final asymptotic values.

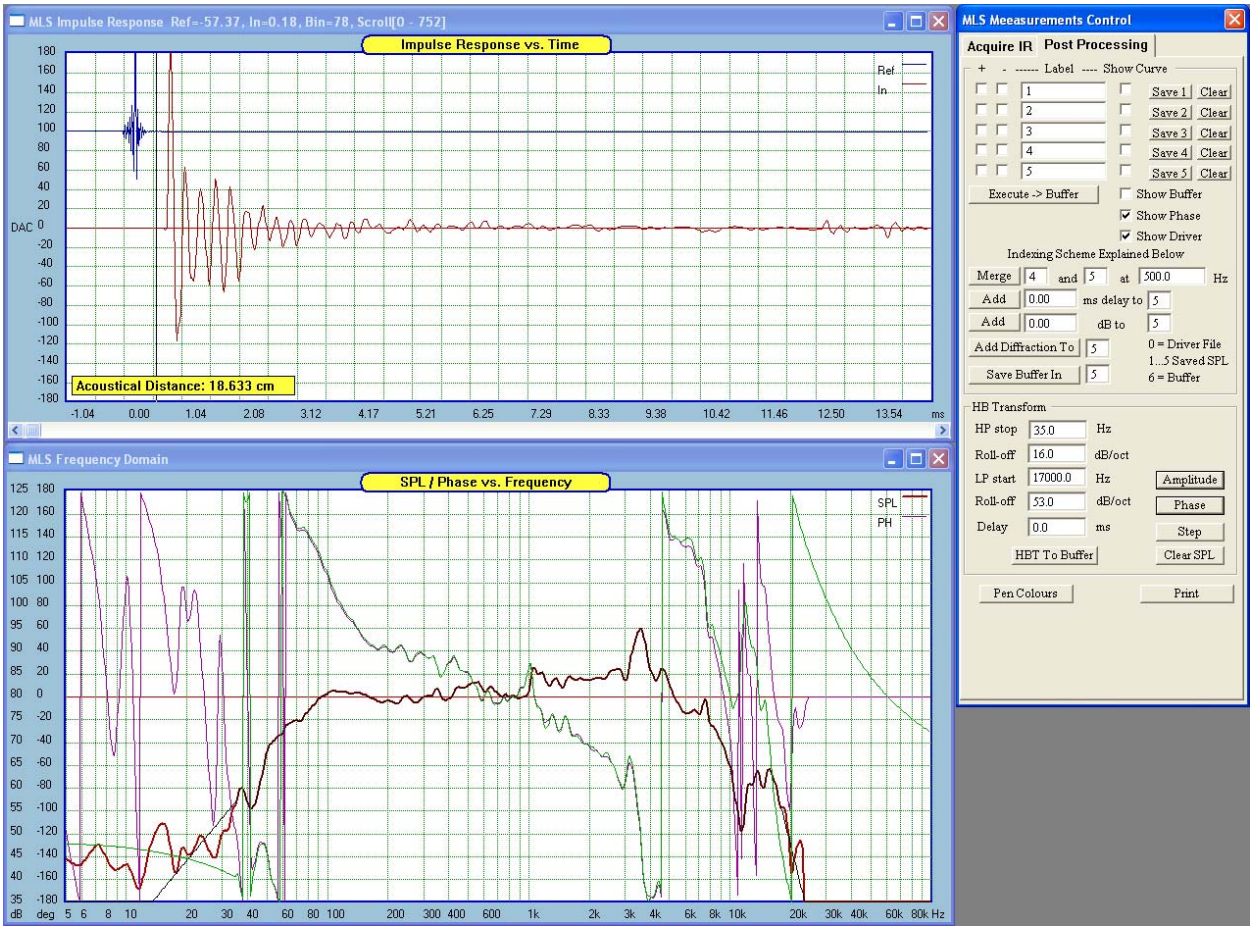


Figure 16.3 Hilbert-Bode Transform phase matching well with the measured phase from appr. 30Hz – 10kHz.

While most loudspeaker drivers usually exhibit nearly minimum phase behavior, this is not necessarily the case for complete loudspeaker systems due to possible non-minimum phase crossovers or, if the acoustic centers of the drivers do not lie in the same plane. Truly minimum phase loudspeaker system is a indeed an uncommon occurrence.

8. **Measured data (modulus of the amplitude and phase) is saved into the program data space as imported data. Therefore SoundEasy makes no distinction between imported measurements and its own measurement system data. However, if you decide to use Hilbert-Bode Transform on the measurement screen, the output of the transform will be saved directly into the Transfer Function data space.**

Measurement Method for Impedance

When making impedance measurements, Zin, the "Ref" probe, sensing V1, is connected to the output of the power amplifier and the "In" probe, sensing V2, is connected right across the loudspeaker. The measured impedance is calculated from the formula:

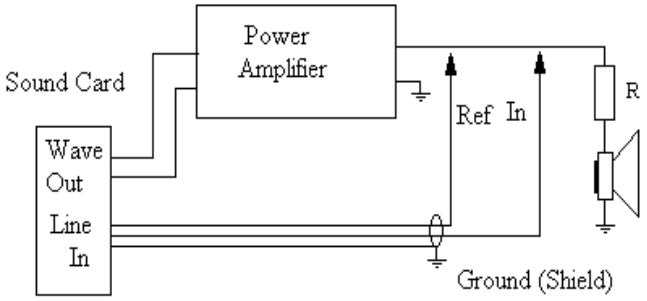
Zin = R*(V2/V1)/(1 – V2/V1), R > 400 Ω is recommended for constant current method

The method used for direct impedance measurements is close to constant current. This is because the source impedance (resistor R = 400 Ω, or higher inserted between the power amplifier output and the loudspeaker) is high so that the current following through the voice coil remains relatively constant**. Using lower resistance R is perfectly acceptable**. The formula above indicates clearly, that input impedance is heavily dependant on the voltage ratio, as measured by left and right channel of the sound card. There is a potential for significant error, as the voltage gains in both channels may not be exactly equal. More importantly, if you use probe resistive dividers, the voltage divider ratios may not be equal in both probes due to resistor tolerances. In order to ensure accuracy of the impedance measurements, you must CALIBRATE the system before a valid impedance measurement can be taken. During the calibration process, you will feed the same signal to both probes and the program will measure and memorize the difference in actual readings. This information will be use next for scaling the measured voltages.

CALIBRATION FOR IMPEDANCE MEASUREMENT

Follow the steps below to perform calibration for impedance .

1. **Connect the test setup**: power amplifier, test probes. Connect the test probes TOGETHER, at the output of the power amplifier – as shown below. Power-up the power amplifier.



2. **Select "Digital MLS**" from the main menu to invoke MLS measurement screen.
3. **Perform calibration**
   - Select Generator level from 0-100%.
   - Press "Calibrate" button in the Impedance group box. The setup will now generate 1000Hz tone needed for measurement of both channels. **The "In" indicator corresponds to the MEASURED signal from your loudspeaker terminal and the "Ref" indicator corresponds to the REFERENCE signal probe (connected to the output of the power amplifier).**

Having calibrated the system you can immediately follow the steps below to perform MLS measurements of the input impedance and phase of your loudspeaker.

1. **Connect the test setup**: power amplifier, test probes, test resistor and the "loudspeaker under test". Power-up the power amplifier.

2. **Select "Digital MLS**" from the main menu to invoke MLS measurement screen.

3. **Generate MLS sequence**
   - Select "MLS Length" to 262141. Select sampling rate to 48000. Select Generator level from 0-100%.
   - Press **"Run MLS**" button in the "Impedance" group of controls. You should now hear the noise coming out from the measured speaker. The microphone will pick-up this signal and your "Input Level LED" indicator should lit-up for "In" bars and "Ref" bars. **The "In" indicator corresponds to the MEASURED signal from your loudspeaker terminal and the "Ref" indicator corresponds to the REFERENCE signal probe (connected to the output of the power amplifier).**

4. **Select windowing parameters – this is done automatically**.
   - When the MLS sequence has finished, you should see the impulse response of the measured signal (from the microphone) on the screen. The program will automatically place the time cursor 10 samples to the left of the peak value. Select the type and the width of the time window. **Please DO NOT create wider windows than the length of the MLS.**

5. **Select smoothing parameter**.
   This is a simple selection of one of the provided smoothing ratios: 1/3, 1/6, 1/12, 1/24 and 1/48 octave.

6. **Select additional Zin scaling.**
   - The measured Zin curve can be shifted up/down by entering suitable data in the "Correction" data field.

7. **Generate Impedance curve**.
   Simple press the "Impulse->Zin" button and wait a short while for the calculations to finish. Please note, that wider smoothing band will result in increased computation time. If you need to see the impulse response again (to re-position the time marker), simply press "Show Impulse" button.

8. **If you need to extend the measured SPL/phase curves beyond the measurement frequency range**.
   This may happen if you selected screen frequency range wider then the sound card measurement frequency range. For example: reliable (with automatic equalization via Ref probe) frequency range of SB Live! card is approximately 10Hz – 20kHz. You have selected SoundEasy screens 1Hz 100kHz wide for your project work.

   1. The SPL/phase and impedance/phase data curves measured by the MLS screen are saved as "imported impedance " data. Therefore, you can view them in the "Editor" and "Transfer Function" screens the same way as you would deal with the imported measurements. For instance, you can use the "Transfer Function" screen to re-create Zin transfer function, if you wish. This step is however NOT NECESSARY. The MLS measurement screen will calculate transfer functions for SPL and impedance exactly the same way as the "Transfer Function" screen does. Please note the **correct behavior of the phase response**. The phase response decays to zero at low-end of the frequency response. This is because the modulus of the input impedance becomes purely resistive at low end of the audio frequency range. Also, the phase response approaches asymptotically 45^0, at the high-end of the audio frequency spectrum. This is due to "semi-inductance" element, rather than normal electrical inductance of the voice coil dominating at these high frequencies. This behavior can be clearly seen on full-range frequency screens, particularly, after HB Transform has been applied. This a very practical "sanity check" for your impedance measurements. If you can not see this pattern over an extended frequency range – something is wrong with the measurement. Finally, please note somewhat unusual parameters needed for Hilbert-Bode Transform to track the impedance curve correctly. The low-end slope has 0.0 dB/oct slope and the high-end slope of the impedance modulus is rising, therefore -3.4dB/oct (negative) value was entered. HBT is a unique and extremely useful tool in acoustics.

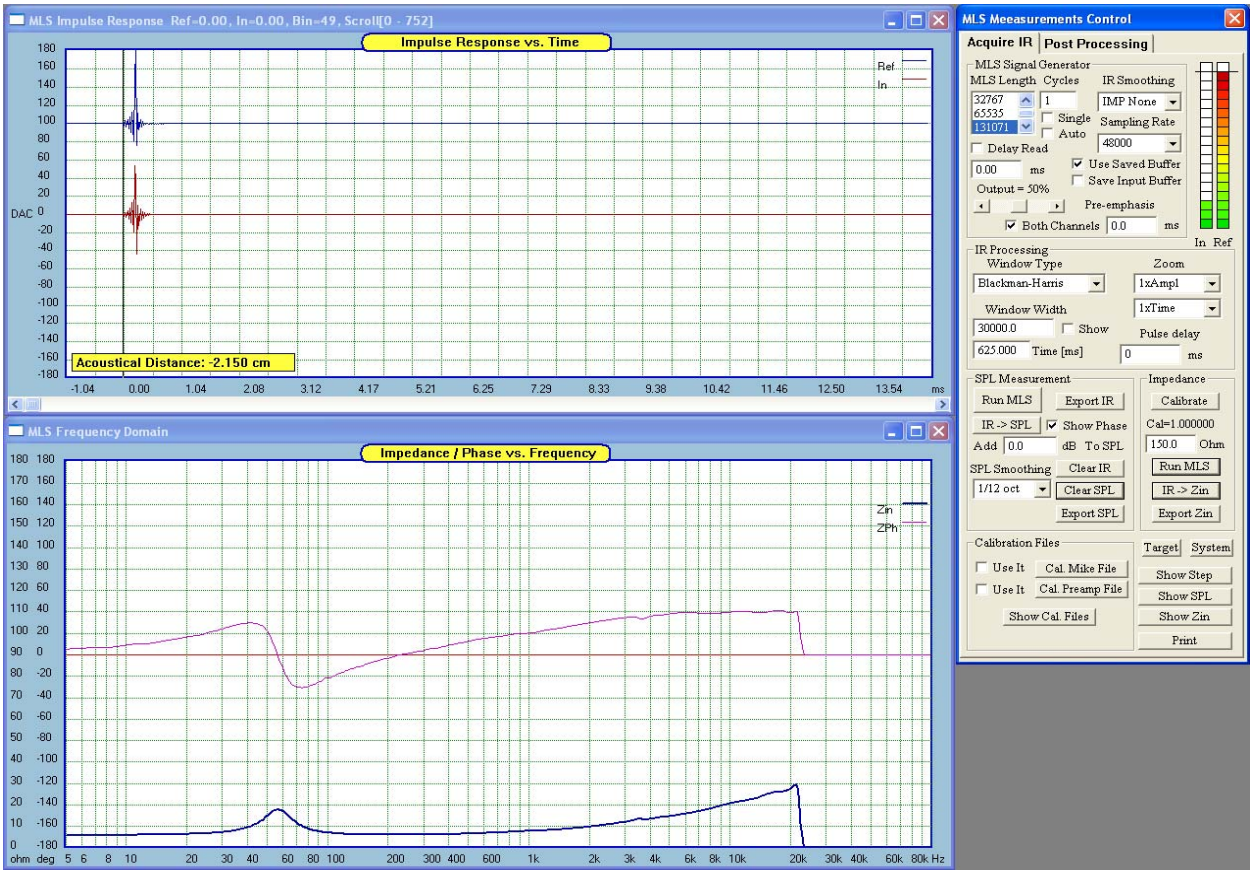


Figure 16.4 Input impedance Zin/phase over narrower frequency range of 5Hz – 20kHz.

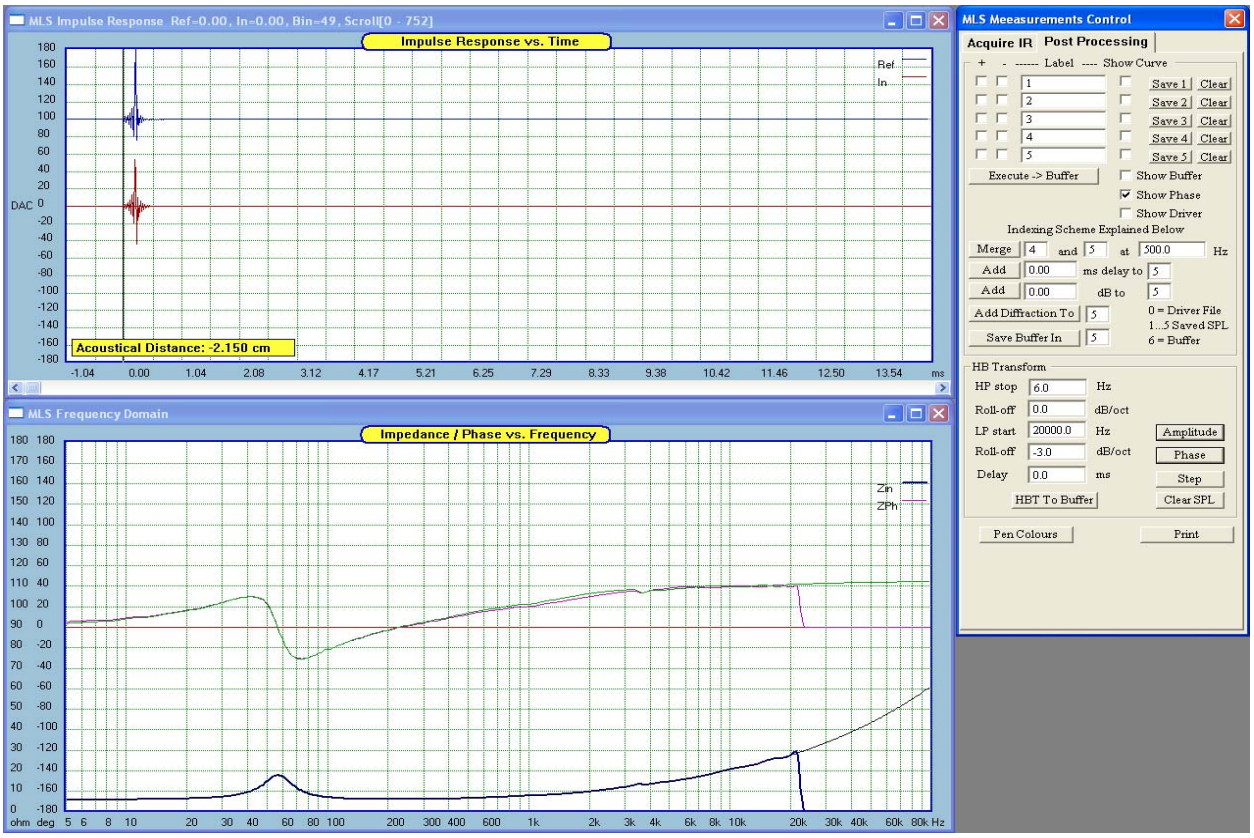


Figure 16.5 HB Transform extends input impedance Zin/phase over full frequency range

Operation of the Sine-wave Sweeps (Analog) Screen

Sine-wave sweep techniques are possibly the oldest measurement method for testing SPL and input impedance. The method is using continuous tones for impedance measurements or anechoic type of SPL measurements, or involve gated (short bursts of sine-waves) tones for eliminating possible echoes in SPL measurements. The analog screen uses also two probes, however, the reference probe is used exclusively for amplitude level type of reference.

The phase response for SPL and Zin measurements is mathematically generated via Hilbert-Bode Transform. Therefore, the phase response will be the minimum-phase response. The Hilbert-Bode Transform group box allows you to add any delay to the minimum-phase response, so that you can generate phase response including the "time-of-flight" of sound delays between the test microphone and the loudspeaker.

The **SPL measurements are relative**. They do not show the actual, calibrated results in dBspl. However, you can actually obtain very good results if you correlate analog SPL screen curves with the indication of an inexpensive (but calibrated) SPL meter. There is a range of SPL meters available on the market and you should get one for your home lab. For the best accuracy, correlation could be performed using "half-space" technique to avoid multi-paths or echoes picked up by the SPL meter.

The Analog Screen has built-in all required functions needed for both type of measurements. Here is the description of the screen functions:

1. **Generate Tone.**
   - The data entry field can be used to enter the desired frequency and the "ON" and "Stop" buttons control the generator.

2. **Sweep Generator.**
   - The data entry fields can be used to enter the desired frequency range and the "RUN" and "Stop" buttons control the generator.
3. **Output.**
   - Use this slider to adjust output voltage from the sound card.

4. **"In Ref".**
   - LED indicators of the input level. Do not allow the indicator to enter the last "square".

5. **Sampling Rate**
   - Select desired sampling rate for the sound card. We recommend you use the highest available sampling rate.
6. **Smoothing**
   - When you finished with the frequency response plot, you can apply smoothing over 1/48 to 1/3 oct.

7. **Gated Sweep**
   - The data entry fields can be used to enter the desired timing parameters - see **Gated Sine-wave Sweep Measurements** section, and the "**RUN**" and "**Stop**" buttons control the generator.

8. **Hilbert-Bode Transform**.
   - The tool to use for generating phase response matching the measured amplitude/impedance curves. The Hilbert-Bode Transform group-box provided at the right-bottom corner of the measurement screen. The transform is driven by 5 editable parameters and they should be selected as described in many other parts of this manual.
     Screen above, shows you an excellent example of BHT usage to generate "textbook perfect" Zin/Phase curves.
9. **Impedance measurement – Resistive Divider Method + Calibration**.
   - The data entry fields can be used to enter the desired resistor value range and the "RUN" and "Stop" buttons control the generator – see **Measurement Method for Impedance** section for more details.

10. Utilities – see section **MLS and Analog Measurement Screen Utilities – Post-Processing**

Measurement Method for Impedance

When making impedance measurements, Zin, the "Ref" probe, sensing V1, is connected to the output of the power amplifier and the "In" probe, sensing V2, is connected right across the loudspeaker. The measured impedance is calculated from the formula:

$$\mathbf{Zin = R*(V2/V1)/(1 - V2/V1)}$$

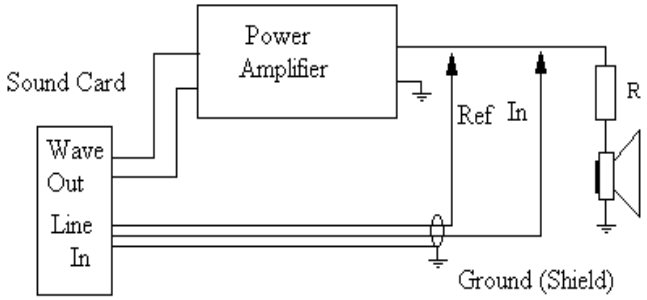
The method used for direct impedance measurements is close to constant current. This is because the source impedance (resistor R = 400 Ω, or higher inserted between the power amplifier output and the loudspeaker) is high so that the current following through the voice coil remains relatively constant. **Other resistor values can be used**. The formula above indicates clearly, that input impedance is heavily dependant on the voltage ratio, as measured by left and right channel of the sound card. There is a potential for significant error, as the voltage gains in both channels may not be exactly equal. More importantly, if you use probe resistive dividers, the voltage divider ratios may not be equal in both probes due to resistor tolerances. In order to ensure accuracy of the impedance measurements, you must CALIBRATE the system before a valid impedance measurement can be taken.

During the calibration process, you will feed the same signal to both probes and the program will measure and memorize the difference in actual readings. This information will be use next for scaling the measured voltages.

CALIBRATION FOR IMPEDANCE MEASUREMENT

Follow the steps below to perform calibration for impedance .

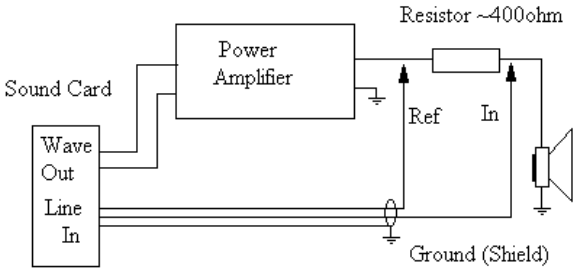
1. **Connect the test setup**: power amplifier, test probes. Connect the test probes TOGETHER, at the output of the power amplifier. Power-up the power amplifier.



2. **Select "Analog Measurements**" from the main menu to invoke MLS measurement screen.
3. **Perform calibration**
   - Select Generator level from 0-100%.
   - Press "Calibrate" button in the Impedance group box. The setup will now generate 1000Hz tone needed for measurement of both channels. **The "In" indicator corresponds to the MEASURED signal from your loudspeaker terminal and the "Ref" indicator corresponds to the REFERENCE signal probe (connected to the output of the power amplifier).**
     The input voltage applied to the on SB Live! Sound card can be as high as 2.2Volts to get the LED bars indicator right to the maximum non-distorted level. In order to obtain good Signal-To-Noise characteristics of the measured signal, you should always try **to maximize the "In" signal level, but without crossing the "Input Level Too High" level**.

Having calibrated the system you can immediately follow the steps below to perform analog sweep measurements of the input impedance and phase of your loudspeaker.

Connect the test setup: power amplifier, test probes, test resistor and the "loudspeaker under test".
Power-up the power amplifier



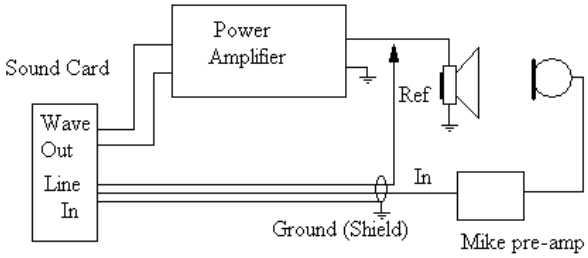
1. **Select "Analog Sweeps**" from the main menu to invoke sine-wave sweeps measurement screen.
2. **Run the test**
   - Select sampling rate from the provided list box– on SB Live! Sound card set to 48000.
   - Select Generator level from 0-100%.
   - Press "Run" button in the Impedance group box. You should now hear the tines coming out from the measured speaker. The probes will pick-up this signal and your "Input Level LED" indicator should lit-up for "In" bars and "Ref" bars. **The "In" indicator corresponds to the MEASURED signal from your loudspeaker terminal and the "Ref" indicator corresponds to the REFERENCE signal probe (connected to the output of the power amplifier).**
3. **Select additional Zin scaling.**
   - The measured Zin curve can be shifted up/down by entering suitable data in the "Zin +/- ohm" data field.
4. **Generate Impedance Phase curve**.
   - Please note somewhat unusual parameters needed for Hilbert-Bode Transform to track the impedance curve correctly. The low-end slope has 0 dB/oct slope and the high-end slope is rising, therefore -3.4dB/oct (negative) value was entered. Phase delay was set to 0.0.

Figure 16.6 HB Transform extends input impedance Zin/phase over full frequency range 5Hz-100kHz

Sine-wave Sweep Measurements

Sine-wave sweep measurements should be used for **anechoic type** of measurements. The technique is very simple and involves generating continuos tones of 1second duration stepping "From" and "To" selected frequencies.

1. **Connect the test setup**: power amplifier, test probes, microphone and the "loudspeaker under test". Power-up the power amplifier and the microphone pre-amplifier.



Select "Analog Sweeps" from the main menu to invoke analog measurement screen.

2. **Generate Sweep sequence**
   - Select sampling rate from the provided list box– on SB Live! Sound card set to 48000.
   - Select Generator level from 0-100%.
   - Press "Run" button under "Generate Sweep" column. You should now hear the tones coming out from the measured speaker. The microphone will pick-up this signal and your "Input Level LED" indicator should lit-up for "In" bars and "Ref" bars. **The "In" indicator corresponds to the MEASURED signal from your microphone and the "Ref" indicator corresponds to the REFERENCE signal probe (connected to the output of the power amplifier).**
3. **Select smoothing parameter**.
   This is a simple selection of one of the provided smoothing ratios: 1/3, 1/6, 1/12, 1/24 and 1/48 octave.

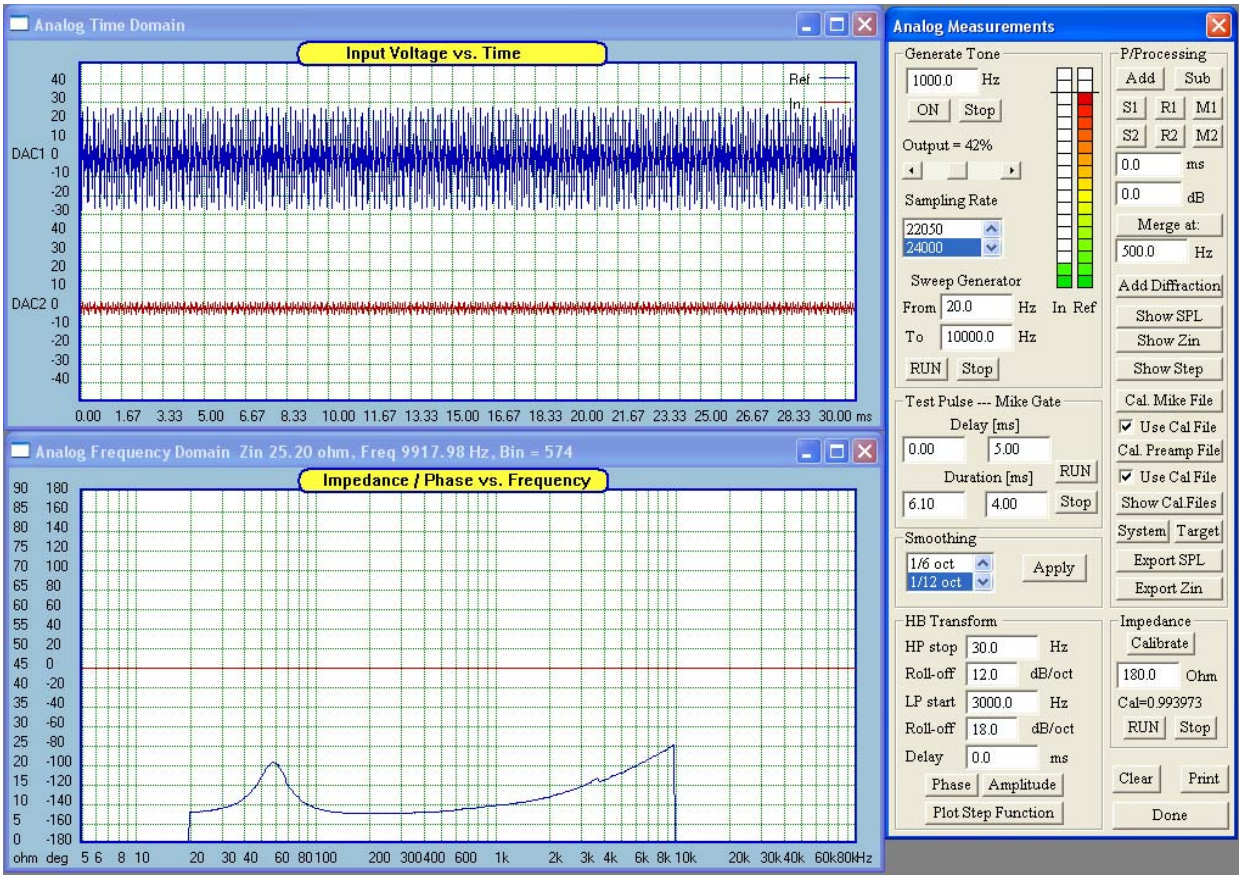


Figure 16.7 Example of Zin sweep measurement from 20Hz to 10000Hz - small woofer out of box.

4. **Select additional delay/amplitude scaling.**
   - Adding a pure delay (enter **positive** number in the "Phase" field of the HB Transform) causes the phase curve to tilt downward from left to right when displayed on a frequency scale. Conversely, removing a delay (enter **negative** number in the "Phase" field) causes the corresponding phase curve to tilt upwards.

5. **Generate smoothed SPL curve**.
   Simple press the "Amplitude" button and wait a short while for the calculations to finish. Please note, that wider smoothing band will result in increased computation time.
   - If you need to see the impulse response again (to re-position the time marker), simply press "Show SPL" button.

Gated Sine-wave Sweep Measurements

Gated sine-wave sweep measurements can be considered equivalent to "windowed" measurements using MLS technique. The purpose is the same – to find some means of cutting out the echoes of early reflections. The analog screen provides you with the means to control the timing of generating the test tone burst and the gating of the microphone probe. **Please DO NOT modify parameters of the sweep during its run – STOP the sweep first**.

You should be able to see both pulses: the reference pulse picked by the "Ref" probe from the output of the power amplifier and the measured pulse, as received by the microphone and picked up by the "In" probe. The position of both bursts on the time domain screen is a function of several variables: (1) microphone distance (time-of-flight delay), (2) the speed of message passing within your computer, and (3) latency of the sound card hardware. It is therefore difficult to specify exact timing, common for all possible combinations of PC+sound cards+acoustic delays. We recommend the following approach to resolve timing issues.

1. Review your test setup and determine critical distances. Basically, there are two main distances you need to know. The distance between the microphone and loudspeaker, from which you will be able to determine the time-of-flight delay – T_m. Secondly, the earliest reflection distance, from which you will be able to determine the when the first reflection will hit the test microphone - T_r. Your microphone gate duration, T_g, is simple the difference between those two.

$$\mathbf{T_g = T_r - T_m \ [ms]}$$

2. Assuming that you can only gate-in tones of Tg duration, it makes little sense to generate tone bursts much longer than Tg. Therefore, you could simply calculate the burst duration, Tb, by adding a small margin (2-3ms) to the Tg.

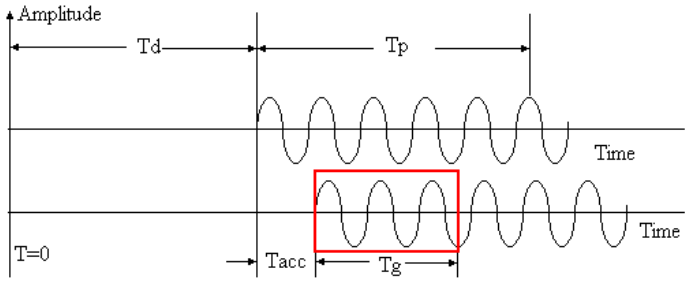
$$\mathbf{T_p = T_g + 3.0 \ [ms]}$$

3. Since the microphone distance delay, **Tacc**, card latency and messaging system, **Td**, add only delay to the signal reception, it makes sense not to increase this delay any further, therefore you may want to enter 0.0 into "Test Pulse delay" data field.

4. Final parameter needed for execution is the microphone gate delay, **Tdel**. Since this parameter depends so much on what is happening in your system, we suggest, that you determine it dynamically. Simply set the screen to "Time Domain" and start Gated Sweep by pressing "Run" button under the "Gated Sweep" column. The goal is to adjust the "Delay" and "Duration" parameters of the Microphone Gate, so that the whole **Tg** length of the test burst can be seen on the screen. You are looking at the bottom trace in time domain – this is your "In" pulse.

   If you adjust the start of the microphone gate exactly to the beginning of the received burst, you will be able to extend your microphone gate ON period to the full duration = Tg time.

$$\mathbf{Tdel = Td + Tacc \ [ms]}$$



Timing diagram for Gated Measurements

This is exactly what you want to accomplish, because the length of the microphone ON gate determines the lowest frequency limit of your test setup. We strongly recommend, you should really optimize the test setup and timing, so you can push the low-end cut-off frequency of the setup as low as possible.

F(cut-off) = 1/Tg [Hz]

It easy to observe, that as with the MLS/FFT measurement systems, the 1/Tg is the low frequency limit for the test setup. However, the 1/Tg is also the frequency RESOLUTION for MLS/FFT based systems above the 1/Tg limit. This is a serious limitation, which does not apply to the analog gated sweep measurement. Frequency RESOLUTION above the 1/Tg is as fine as you wish, and is determined by the screen's frequency range.

5. Having adjusted gating parameters you are now ready to actually plot the frequency response. Even though the low-end frequency limit is 1/Tg, you can start with a frequency twice as low.

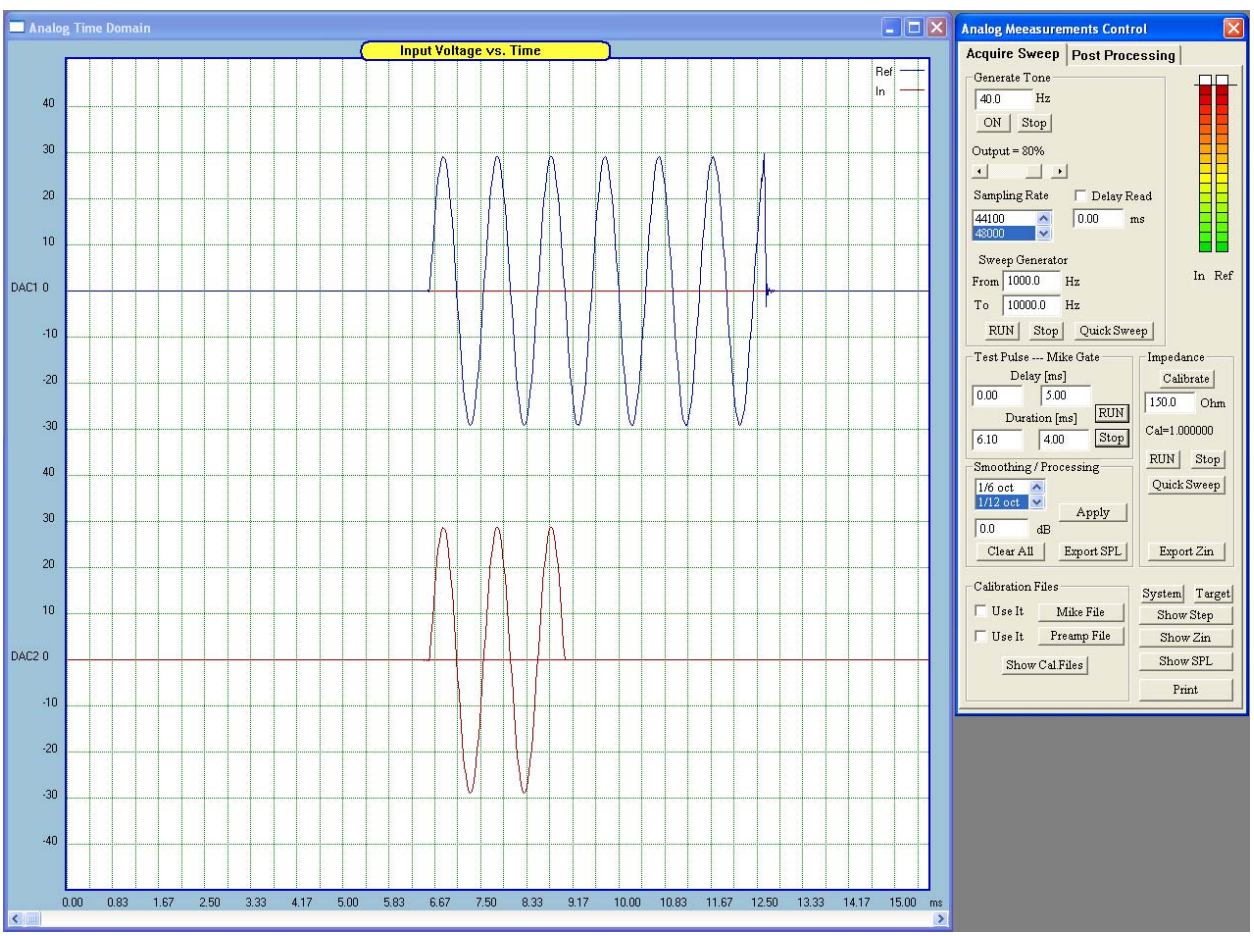


Figure 16.8. Time domain screen using analog gated sweep method .

MLS and Analog Measurement Screen Utilities – Post-Processing

MLS Measurement System Curve Arithmetics.

As the measurement and processing capabilities of the MLS system continue to grow, so is the demand for more controls on the dialogue box. As the result of this, the MLS dialogue box has been re-designed and now incorporates two tabs: **"Acquire IR"** tab and **"Post Processing"** tab. Various curve arithmetic can be performed on 5 curves. This includes:

- Adding / subtracting up to five curves at a time.
- Adding time delay to a curve.
- Adding / subtracting X dBs to a curve. This moves the SPL curve up / down.
- Performing Hilbert-Bode Transform on a curve.
- Erasing individual curve.
- Merging two curves at a specific frequency.
- Saving the results of each operation in a buffer. Buffer content can be moved to any of the 5 curves data space.
- Add box diffraction to the curve.

Theoretically, you can process unlimited number of curves. For instance, the results of the processing of the first five curves are saved into the buffer and 4 new curves can be processed. The fifth curve is the old result from the buffer. You can repeat this process indefinitely, and use the MLS screen to make all IR measurements.

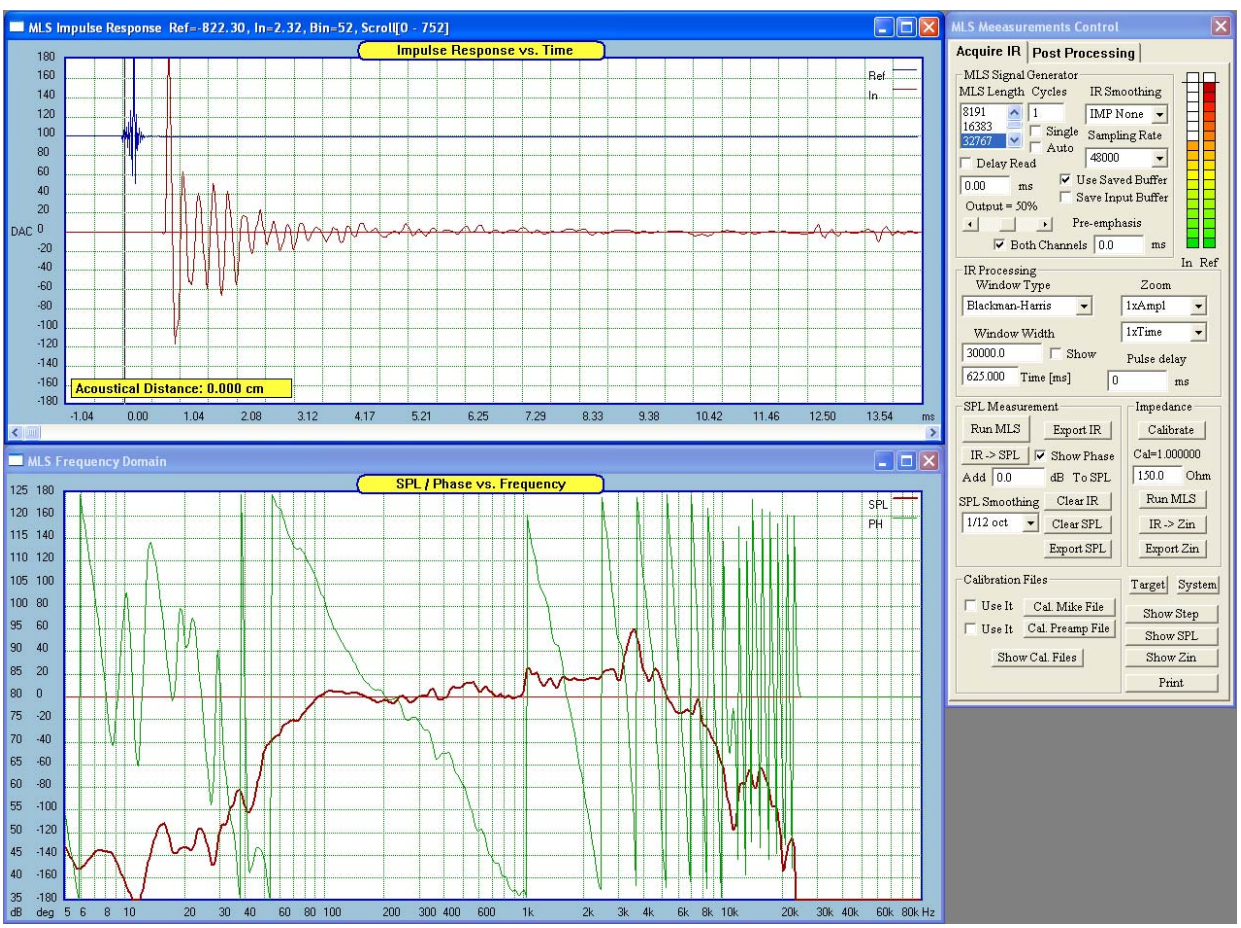


Figure 16.9 Collected IR (top plot) and IR transformed to SPL (bottom plot)

1. Measure SPL of the system under test using **"Acquire IR"** tab. It is not necessary to do this, but it would be recommended to save each individual measurement as an MLS buffer for future use. It is also useful, to label each measurement as a file name when saving the IR to a file. This will allow you to "mix-and-match" measurement results as you wish.

2. Load each MLS buffer file into MLS screen and perform **"IR->SPL"** transformation. Store each resulting SPL curve into the five "Post-processing" locations, using **"Post Processing"** tab. When all required curves have been stored, perform required arithmetic. Up to five files can be included in the process at one time. You can engage more curves by performing the above process sequentially.

2.1 Perform MLS measurement (or load saved MLS buffer into the program).
2.2 Press **"IR -> SPL"** button to perform Inverse Fourier Transform on the Impulse response.
2.3 Go to **"Post Processing"** tab.
2.4 Press "Save 1" button. This will save SPL and phase data into curve 1 data space.
2.5 Go to Step 2.1 and repeat the IFFT process, but this time press "Save 2" button on **"Post Processing"** tab.
2.6 Up to 5 curves can be stored this way.

Each curve can be labeled using a short name entered under the "Label" column on the **"Post Processing"** tab. You will notice, that the color of the "Label" field matches the color of the curve. The curve can be displayed with different colors. You can select the color of each of the processed curves by pressing the **"Pen Color"** button on the **"Post Processing"** tab BEFORE you press "Save X" button. A dialogue box will be opened, where you can select the desired curve color.

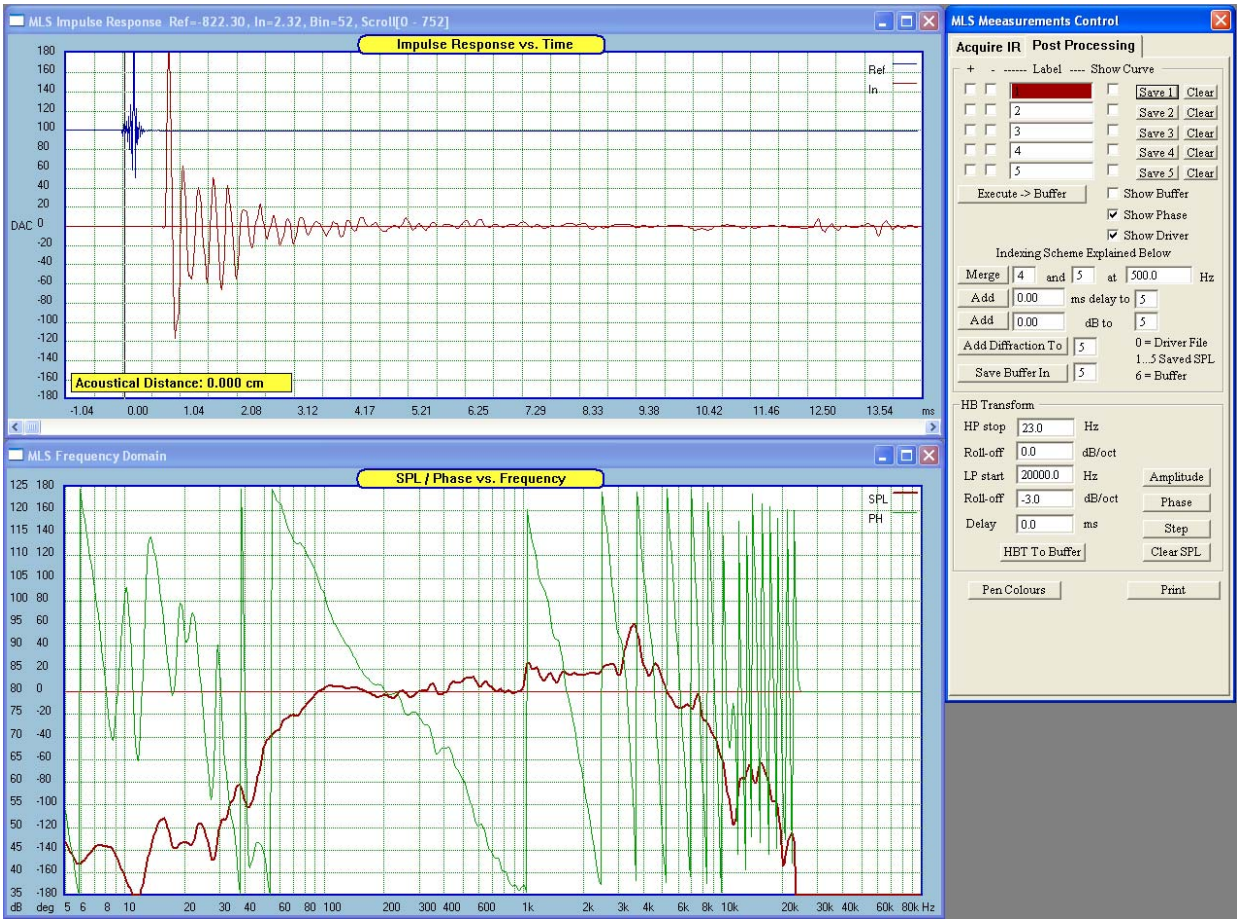


Figure 16.10 SPL (bottom plot) has the same color as the label in Post Processing TAB.

The "Post Processing" tab has several check boxes for you to select, to indicate what function is to be performed. There are **"+"** and **"-"** check boxes to indicate that the selected curves are to be added or subtracted.

There are also **"Show Curve"** check boxes to indicate which curve is to be displayed, and additional options for showing phase together with SPL for all curves.

All the above mathematical and graphical operations are executed when **"Execute -> Buffer"** button is pressed. Please note, that the results are always stored in the buffer data space. You can display the result of the operation, if the **"Show Buffer"** check box is selected.

<u>Processing of the curves is based on a very simple indexing scheme:</u>

0 - is the index of the original SPL/phase measurement result, which has also automatically been copied to the driver's data space, when pressing **"IR->SPL"** button on the "IR Acquire" tab. This way, it will appear in the Driver Editor screen and DRE screen.
1….5 - are the indexes of saved curves.
6 – is the index of data in the buffer.

Example 1: Adding driver and vent SPL measurement results.

- Measure driver's SPL using close-mike technique and save the result in Curve 1
- Measure vent's SPL using close-mike technique and save the result in Curve 2.
- Check "+" box and "Show Curve" box against Curve 1.
- Check "-" box and "Show Curve" box against Curve 2.
- Check "Show Buffer" box to show the result.
- Press **"Execute -> Buffer"** button.
- Enter '0' and Select **"Save Buffer In"** button to move the results to driver's file.

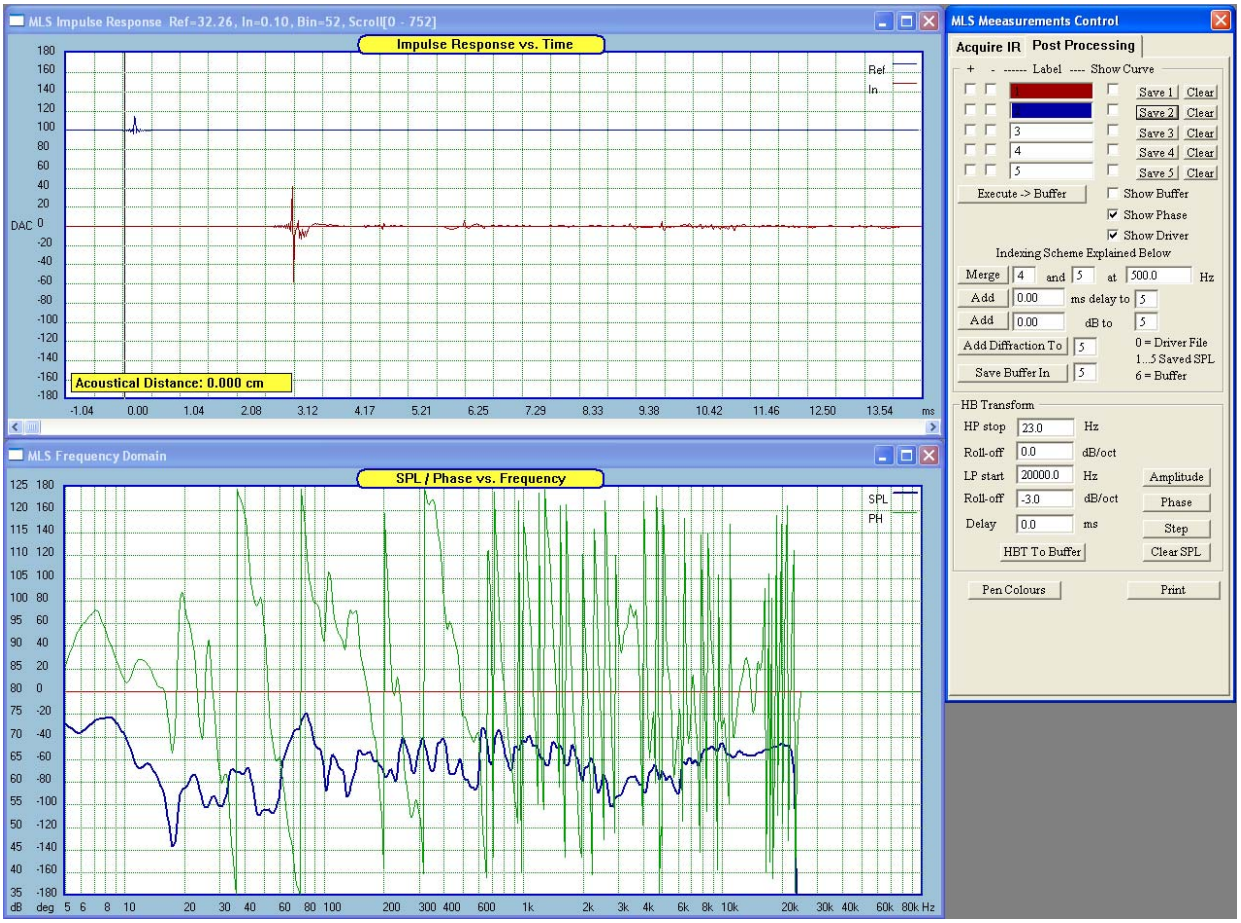


Figure 16.11 Second SPL plot collected (bottom plot).

Example 2: Merging curves.

- Measure SPL 1 and save the result in Curve 1
- Measure SPL 2 and save the result in Curve 2.
- Check “Show Curve” box against Curve 1.
- Check “Show Curve” box against Curve 2.
- Check “Show Buffer“ box to show the result.
- Enter “1” and “2” and desired merge frequency.
- Press **“Merge”** button.
- Enter ‘0’ and Select **“Save Buffer In”** button to move the results to driver’s file.

In the following simple graphical example, **red SPL curve** from Figure XXX was added to **blue SPL curve** from Figure YYY and this produced the **pink SPL curve** on Figure ZZZ. The pen color was changed from red -> blue -> pink.

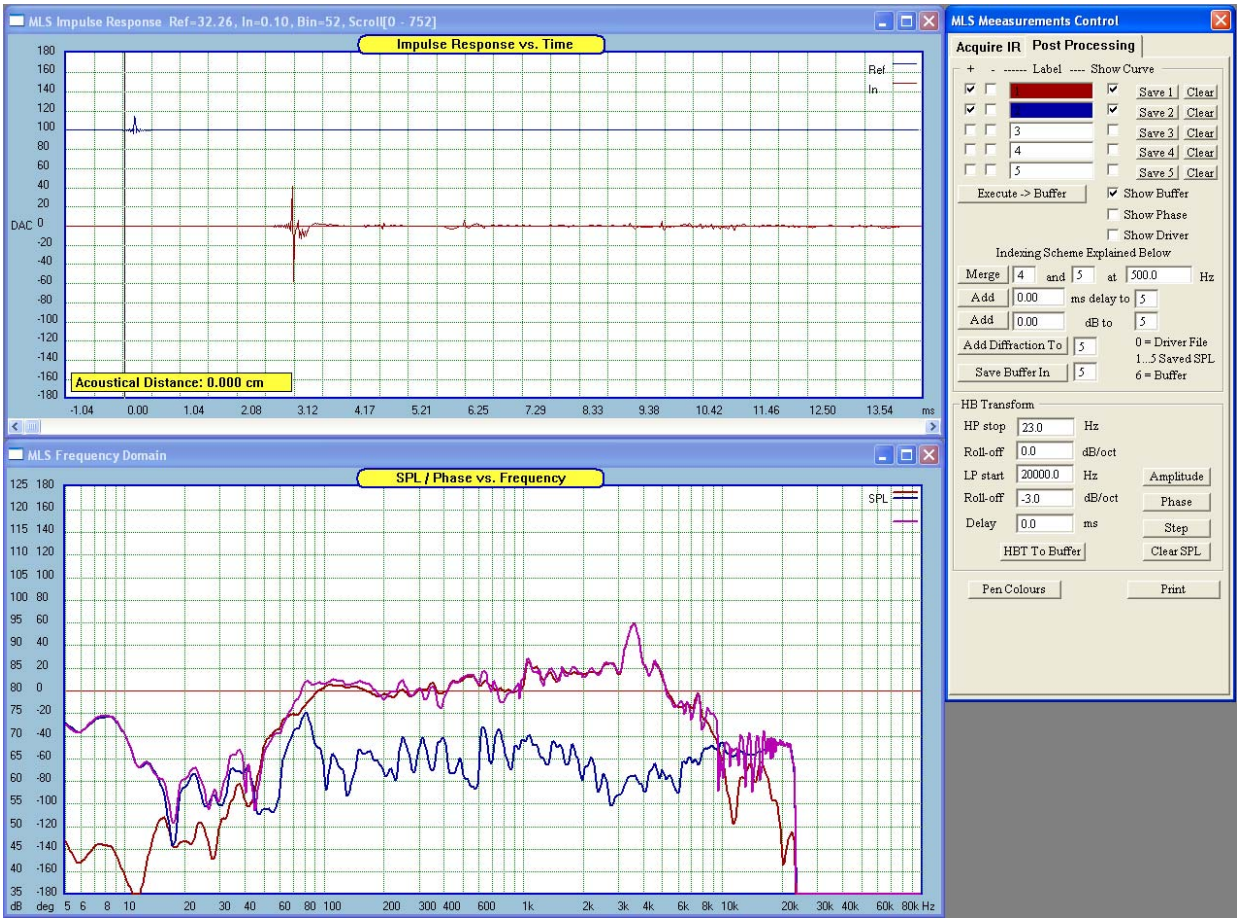


Figure 16.12. SPL (bottom plot) from Fig 10 + 11 added together (blue plot).

Please note, that you can move the results of Curve Arithmetic from the **Buffer to the Left / Right / Both channels of the Digital Room Equalizer**. First, please select where you want the results to go, by selecting appropriate check box on the **DRE** control dialogue box. Then select **“Save Buffer”** to **Curve 0**, on the MLS “Post-Processing” tab. This actions will save the Buffer content into driver data space (Curve 0) and also into the DRE data space, as requested from DRE dialogue box.

Curve Arithmetic Group

1. **"+" , "-"** – Column of check boxes to select arithmetical operation on the curve. "+" means, that the curve will be added, and "-" means, that the curve will be subtracted vectorally. Both check boxes blank indicate, that the curve will be disregarded.
2. **"Label"** – Column of 5 labels to use with the curves.
3. **"Show Curve"** – Column of 5 check boxes to select which curve will be displayed on the screen.
4. **"Save X"** – Column of save buttons to store the last SPL curve, generated on "Acquire IR" TAB.
5. **"Clear"** – Press to clear previously saved curve.
6. **"Execute -> Buffer" –** This button executes the +/- operations and saves the result in the buffer.
7. **"Show Buffer" –** Check this box to display current curve in the buffer.
8. **"Show Phase" –** Check this box to display phase on ALL plots.
9. **"Show Driver" –** Check this box to display currently available driver SPL/phase.
10. **"Merge" –** Button and data fields to merge two selected curves at the desired frequency.
11. **"Add" –** Button and data fields to add X.XX ms of delay to nominated curve.
12. **"Add" –** Button and data fields to add X.XX dBs to nominated curve.
13. **"Add Diffraction" –** Button and data field to add diffraction to the nominated curve.
14. **"Save Buffer In" –** Button and data field to move buffer content to the nominated curve data space.

HBT Group

1. **"HP stop"** – Enter the corner frequency for low frequency tail.
2. **"Roll-off" -** Enter the slope of the low-end tail.
3. **"LP Start"** – Enter the start frequency for the high end tail.
4. **"Roll**-off" – Enter the slope for the high-end tail.
5. **"Delay"** – enter additional delay in ms.
6. **"Amplitude"** – Button to display SPL of the HBT.
7. **"Phase" -** Button to display phase of the HBT.
8. **"Step"** – Button to display step response calculated from HBT.
9. **"Clear SPL"** – Clears SPL/Phase screen.
10. **"Pen Colors"** – Press to invoke Pen Color dialogue box. Please remember to select the colour of the plotted curves and labels BEFORE you plot or save the curve.
11. **"Print"** – Simply prints the screen.

Analog Measurement System Curve Arithmetic.

The Analogue Measurement System benefits from the dialogue has "Acquire Sweep" tab and "Post Processing" tab. Various curve arithmetic can be performed on 5 curves.

Curve Arithmetic Group

1. **"+" , "-"** – Column of check boxes to select arithmetical operation on the curve. "+" means, that the curve will be added, and "-" means, that the curve will be subtracted vectorally. Both check boxes blank indicate, that the curve will be disregarded.
2. **"Label"** – Column of 5 labels to use with the curves.
3. **"Show Curve"** – Column of 5 check boxes to select which curve will be displayed on the screen.
4. **"Save X"** – Column of save buttons to store the last SPL curve, generated on "Acquire IR" TAB.
5. **"Clear"** – Press to clear previously saved curve.
6. **"Execute -> Buffer" –** This button executes the +/- operations and saves the result in the buffer.
7. **"Show Buffer" –** Check this box to display current curve in the buffer.
8. **"Show Phase" –** Check this box to display phase on ALL plots.
9. **"Show Driver" –** Check this box to display currently available driver SPL/phase.
10. **"Merge" –** Button and data fields to merge two selected curves at the desired frequency.
11. **"Add" –** Button and data fields to add X.XX ms of delay to nominated curve.
12. **"Add" –** Button and data fields to add X.XX dBs to nominated curve.
13. **"Add Diffraction" –** Button and data field to add diffraction to the nominated curve.
14. **"Save Buffer In" –** Button and data field to move buffer content to the nominated curve data space.

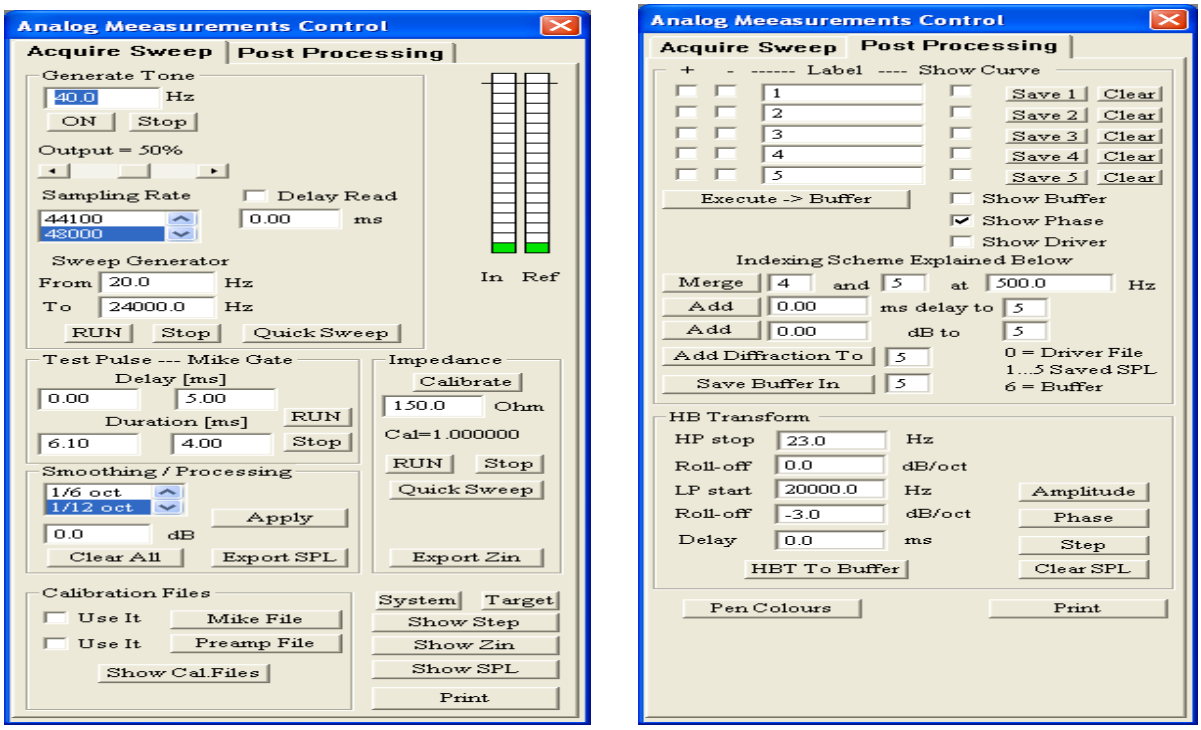


Figure 16.13. Analogue Measurement System Controls

HBT Group

1. **"HP stop"** – Enter the corner frequency for low frequency tail.
2. **"Roll-off"** - Enter the slope of the low-end tail.
3. **"LP Start"** – Enter the start frequency for the high end tail.
4. **"Roll-**off" – Enter the slope for the high-end tail.
5. **"Delay"** – enter additional delay in ms.
6. **"Amplitude"** – Button to display SPL of the HBT.
7. **"Phase"** - Button to display phase of the HBT.
8. **"Step"** – Button to display step response calculated from HBT.
9. **"Clear SPL"** – Clears SPL/Phase screen.
10. **"Pen Colors"** – Press to invoke Pen Color dialogue box. Please remember to select the colour of the plotted curves and labels BEFORE you plot or save the curve.
11. **"Print"** – Simply prints the screen.

Acquire IR Group

1. **Show SPL** – This button plots the last SPL curve.
2. **Show Zin** – This button plots the last input impedance curve.
3. **Show Step** – This button plots the last Step Function curve.
4. **Cal Mike File/Use It** – Button to plot mike calibration file and checkbox to include it in calculations.
5. **Cal Preamp File / Use It** - Button to plot pre-amplifier cal. file and checkbox to include it in calculations.
6. **Show Cal Files** – Button to plot both calibration curves.
7. **System** – Button to plot predicted system response for comparison with measured response.
8. **Target** – Button to invoke dialogue box controlling target function plotting.
9. **Export SPL** – Button to export SPL/Phase curve in ASCII file.
10. **Export Zin** – Button to export Zin/Phase curve in ASCII file.
11. **Save Input Buffer** – Check box to activate file saving option for measured impulse response.
12. **Use Saved Buffer** – Check box to load saved impulse response from file.
13. **Window Type** – Two windows are available: Rectangular and Blackman-Harris.
14. **Show** – When checked, the window will be plotted in the Impulse Response window.
15. **Width** – Enter window's width here.
16. **Imp->SPL** – Main button to calculate frequency response from Impulse rsponse.
17. **Show Phase** – Plots phase as well, when the SPL is calculated.

Microphone Calibration File

Microphone calibration file could look as the example below:

```
Lsp-LAB:Compensation File v1.10
00128.MCP
318
BEGIN.
200.0000  1.5773  0.0000
202.9091  1.5830  0.0370
…
…
19740.2988 -3.5650 -134.3056
20000.0000 -3.7538 -136.2404
END
```

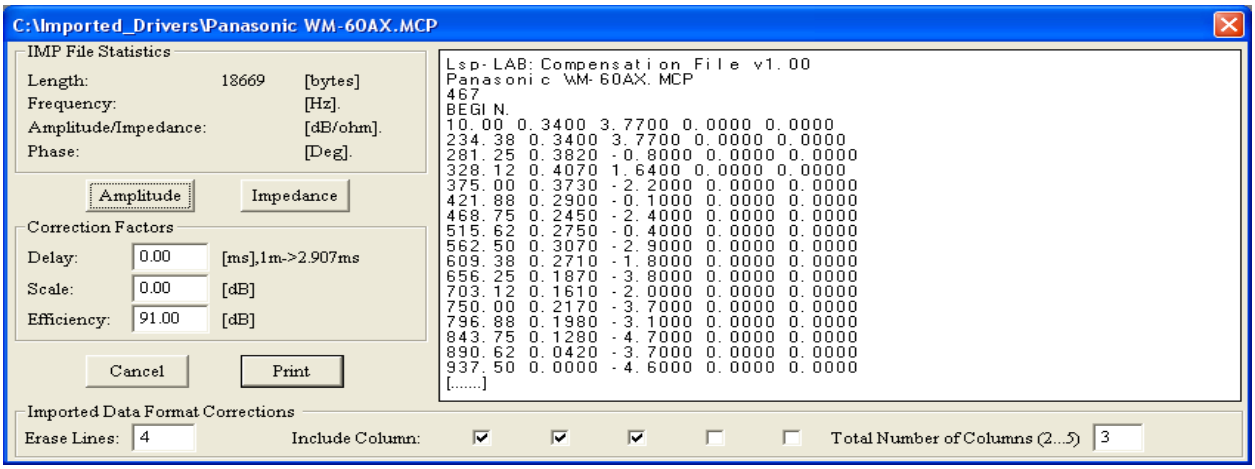


Figure 16.14 Importing microphone calibration file

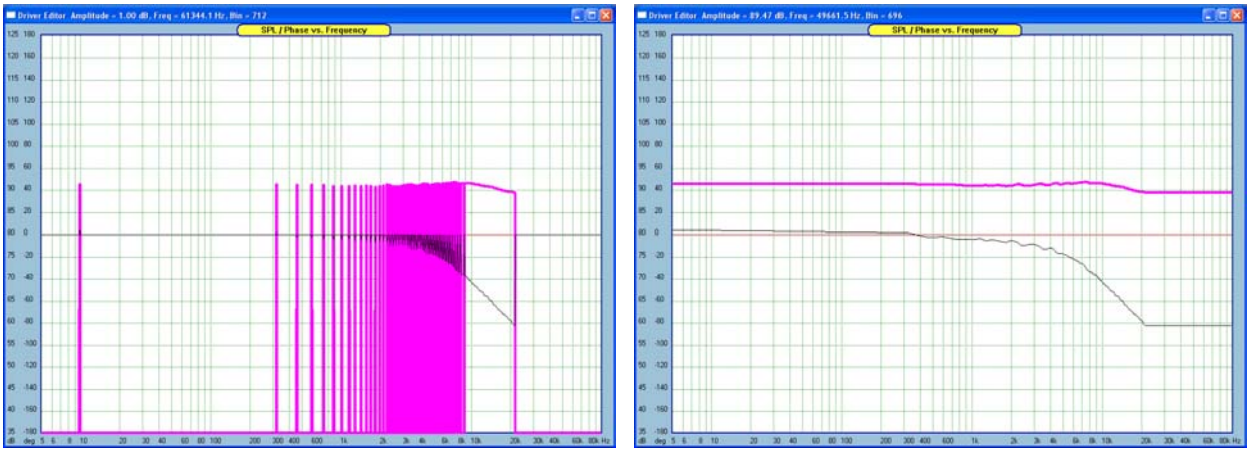


Figure 16.15 Imported and “Repaired” microphone calibration file.

Next, import the file into SoundEasy using "Import IMP" files as the option. The program will ask you to enter reference efficiency and you can enter say, 90.0dB to shift the relative SPL values in dB somewhere around 90dB display area of the screen. The 90.0 dB will be automatically subtracted when the fully edited microphone calibration file is imported into MLS or Analog measurement system screen. The file imported into SoundEasy may look like the one shown on the example below. As you can see, the file in its current state is not of much use and needs to be further processed. In the next step, you should press the **"Repair Amplitude"** button, resulting in display as shown above. This fractional data will be used by the Hilbert-Bode Transformer to re-create full SPL/phase information over the whole frequency spectrum. Please invoke the "Transfer Function" screen and select HB-Transform parameters for the best match with the "template" data. In the example below, the low-frequency slope has 0.0dB/oct roll-off and the high-frequency slope has 26.0dB/oct roll-off.

Now, Perform Hilber-Bode Transform and EXPORT this file as, say "XXXXXX_Corrected.txt" file. The SE exported file should look as below and **contain 750 lines – you can easily check for this using a text editor. It is important to use the microphone calibration file exported by SoundEasy. The MLS and Analog measurement screens are expecting this particular format and the 750 fully valid data points in the file.**

Example of Microphone calibration file exported by SoundEasy.

```
5.00     92.59     -0.04
5.07     92.59     -0.04
5.13     92.59     -0.04
5.20     92.59     -0.04
5.27     92.59     -0.04
….
….
93610.93     78.82     60.31
94855.23     78.74     52.35
96116.07     78.65     44.28
97393.66     78.57     36.11
98688.24     78.48     27.83
```

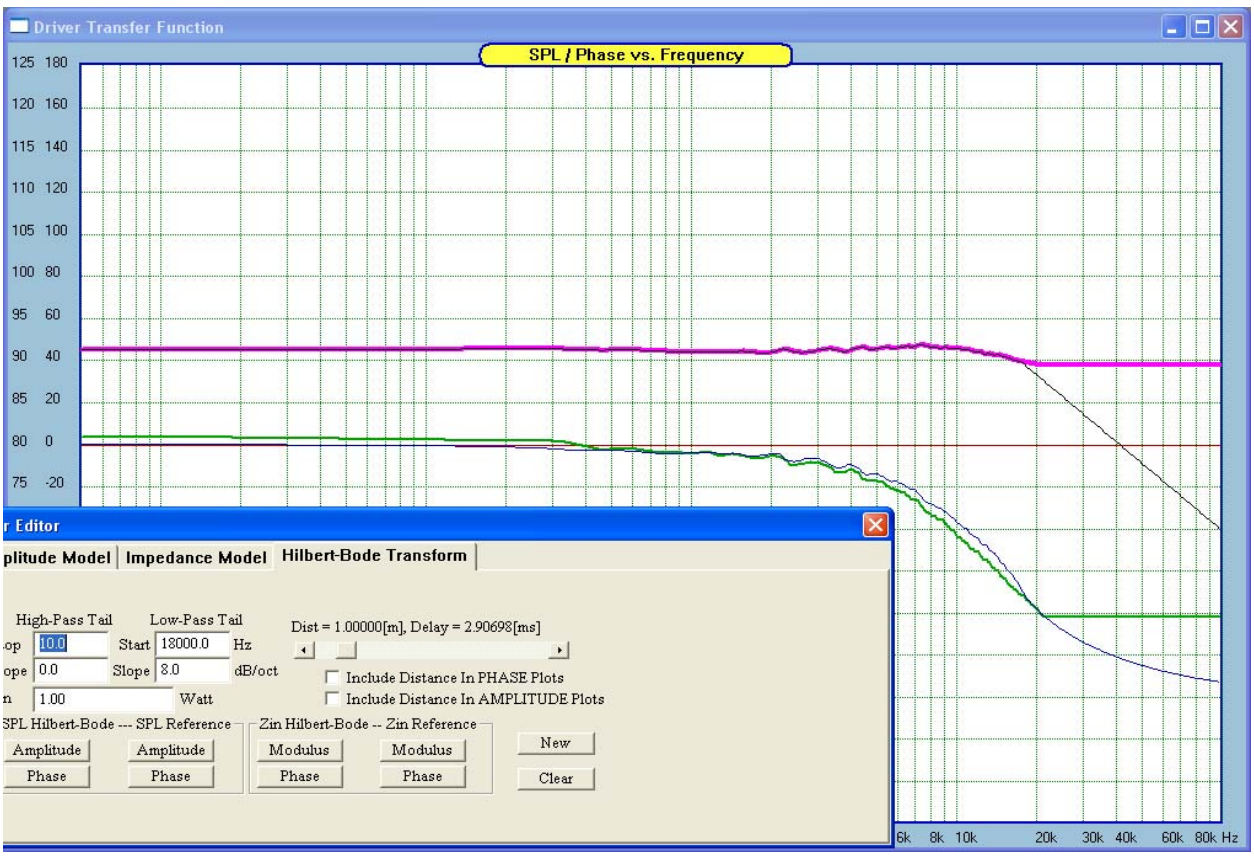


Figure 16.16 Fully edited microphone calibration file.

Select "Cal.Mike" button on the Analog or MLS screen to import the "XXXXXX_Corrected.txt" file. The actual curves will not be shown on the screen, but this amplitude response and phase response will be used for correcting ALL SPL measurements.

Exporting Data

Exporting measured SPL and impedance data from analog and digital measurement screens directly from the measurement screen can be accomplished by pressing "**Export SPL**" (creates .frd file) and "**Export Zin**" (creates .zma file) buttons. Data is exported "as you see" it on the screen.

Spectrum Analyzer/Signal Generator

Spectrum Analyzer screen is intended to show the frequency content of the audio data currently entering your sound card. Here are some of the uses for an audio spectrum analyzer:

1. Analyze the frequency characteristics of live/recorded music.
2. Analyze distortion (harmonic content) of audio equipment.
3. Analyze room acoustics using also the built-in pink noise generator.

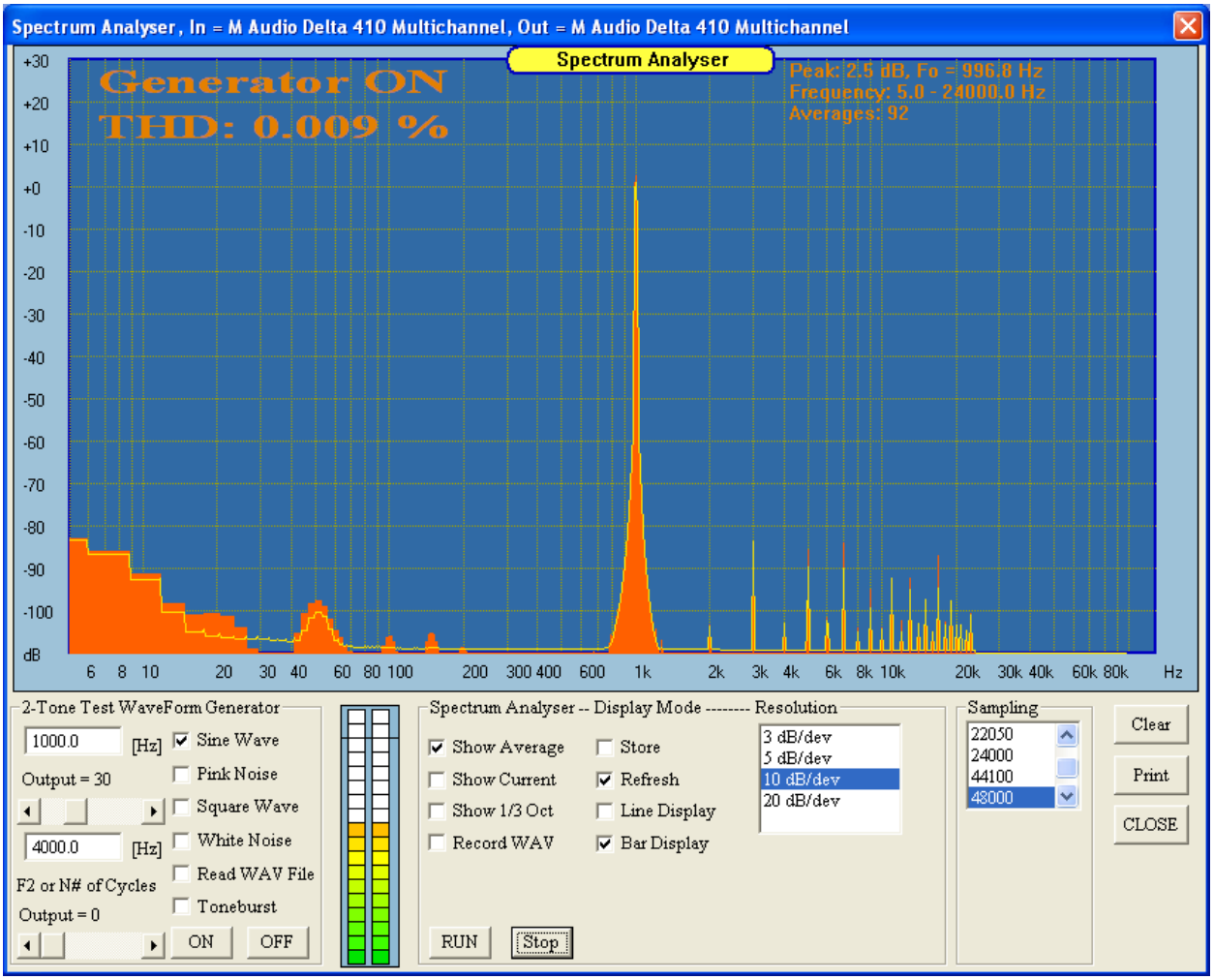


Figure 16.17 EasyLab spectrum analyzer screen.

Another very useful feature is the built-in Pink Noise generator. The characteristics of the pink noise call for a spectrum drop towards higher end of audio range with a rate of 10dB/decade. The noise is a very useful tool in room acoustic analysis, showing room resonance and room transfer function for different speaker/microphone locations. The preferred use of this technique would be to click on "Show Average" check box to allow a number of samples (sweeps) to accumulate over time. Additional generators include White Noise, Square Wave, Tone Burst and Two-Tone Generator signals.

Total Harmonic Distortion, THD parameter, is calculated in reference to the fundamental frequency as entered in the Test Waveform Generator. If you use the internal generator for this measurement, the fundamental frequency of the signal is given automatically by generator. However, if you intend to measure THD on an externally generated signal, the program will track the fundamental frequency of your external signal generator's frequency. The THD measurement is performed automatically on all input signals and the resolution is 0.01%.

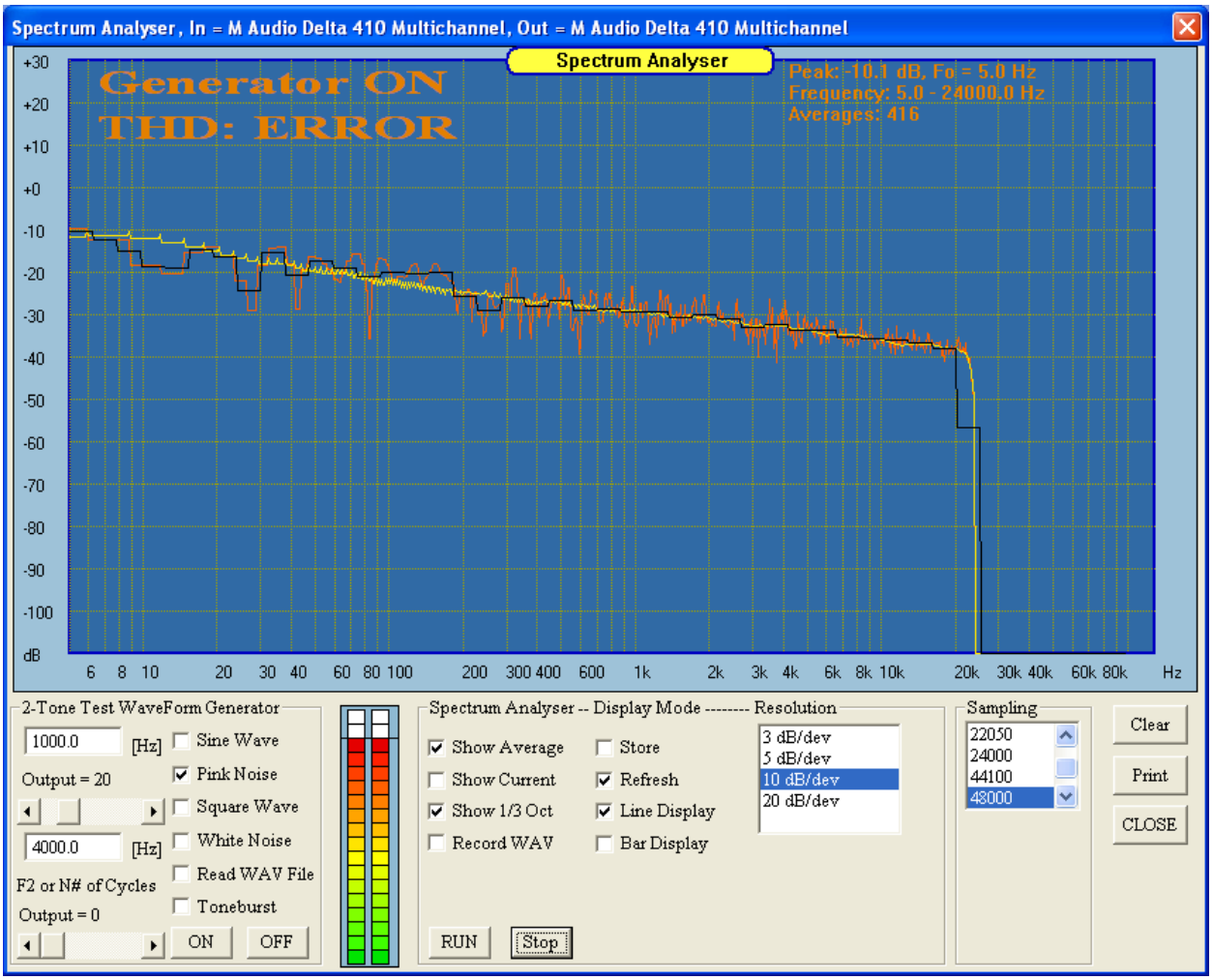


Figure 16.18. EasyLab spectrum analyzer with Pink Noise.

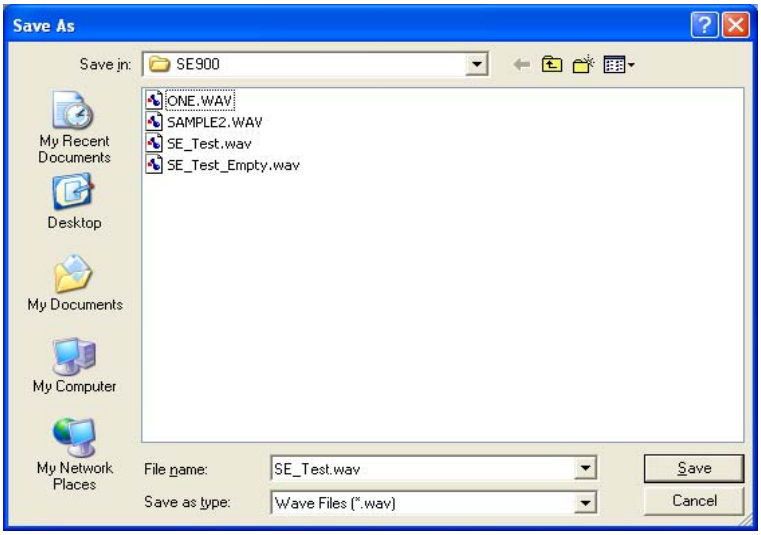
Test Waveform Generator Controls

The device has the following controls:

1. **Frequency** 1 data entry field – type-in the desired frequency of your test signal's sine wave.
2. **Output** level slider – use this control to adjust the voltage level out of your sound card.
3. **Frequency 2** data entry field **-** type-in the desired frequency of your second test signal's sine wave. This field doubles as the "Number of Cycles" in the burst for ToneBurst-type of signals.
4. **Output** level slider **-** use this control to adjust the voltage level of the second signal out of your sound card
5. **Sine Wave** – click on this check box to generate sine wave out of the sound card.
6. **Pink Noise** - click on this check box to generate pink noise out of the sound card.
7. **Square Wave** - click on this check box to generate square wave out of the sound card.
8. **White Noise** - click on this check box to generate white noise out of the sound card.
9. **Read WAV File – This check fiels takes priority over the other checkboxes.** Use it to red WAV files created during previous recording session with SoundEasy – see Record WAV.
10. **ToneBurst -** click on this check box to generate burst of tone (F1) with number of cucles (F2) out of the sound card
11. **ON** button – click on this button to start signal generator.
12. **OFF** button – click on this button to turn the generator off. You should turn the generator OFF before closing the Spectrum Generator dialogue box.
13. **Input Level Indicator** – please observe the input level into the soundcard, so that **the signal does not get clipped.**

Spectrum Analyzer controls

1. **RUN** button – press this button to start spectrum analysis.
2. **Stop** button – press this button to stop spectrum analysis.
3. **Show Average** - click on this check box to display averaged results of several sweeps.
4. **Show Current**. - click on this check box to display only current result of the sweep.
5. **Show 1/3 Octave** - click on this check box to display 1/3 oct result of the sweep (black plot).
6. **Record WAV** – You can record about 45Mb of data per song using this facility. Once you stopped recording. A "File Save" box pops up, where you can enter the name of the recorded file. All recordings are made in STEREO with 48kHz sampling to give "studio quality" sound.



7. **Store** - click on this check box to display only current result of the sweep.
8. **Refresh** - click on this check box to display only current result of the sweep.
9. **Line Display** - click on this check box to display only current result of the sweep.
10. **Bar Display** - click on this check box to display only current result of the sweep.
11. **Resolution** (vertical) - click on this check box to display only current result of the sweep.

Dual Channels Oscilloscope/Signal Generator

Oscilloscope is a very handy device for analyzing signals in time domain. There is an array of controls provided for optimizing the quality of the display of the analyzed signal. You can use single channel mode or take advantage of the dual mode capabilities. Synchronization of the input signals is necessary for stable display and includes "slope" triggering and selection of synch channels. The built-in signal generator provides you with ready-to-use test signals and all controls are on the same screen, so you do not need to browse the menu each time you need to change your test signal. **However, you do need to exercise caution when making live tests on your loudspeakers. You may accidentally use a test signal that will destroy the driver with too much power or a wrong spectrum.**

Test Waveform Generator Controls

The device has the following controls:

1. **Frequency** 1 data entry field – type-in the desired frequency of your test signal's sine wave.
2. **Output** level slider – use this control to adjust the voltage level out of your sound card.
3. **Frequency 2** data entry field **-** type-in the desired frequency of your second test signal's sine wave. This field doubles as the "Number of Cycles" in the burst for ToneBurst-type of signals.
4. **Output** level slider **-** use this control to adjust the voltage level of the second signal out of your sound card
5. **Sine Wave** – click on this check box to generate sine wave out of the sound card.
6. **Pink Noise** - click on this check box to generate pink noise out of the sound card.
7. **Square Wave** - click on this check box to generate square wave out of the sound card.

8. **White Noise** - click on this check box to generate white noise out of the sound card.
9. **Read WAV File – This check fiels takes priority over the other checkboxes.** Use it to red WAV files created during previous recording session with SoundEasy – see Record WAV.
10. **ToneBurst -** click on this check box to generate burst of tone (F1) with number of cucles (F2) out of the sound card
11. **ON** button – click on this button to start signal generator.
12. **OFF** button – click on this button to turn the generator off. You should turn the generator OFF before closing the Spectrum Generator dialogue box.
13. **Input Level Indicator** – please observe the input level into the soundcard, so that **the signal does not get clipped.**

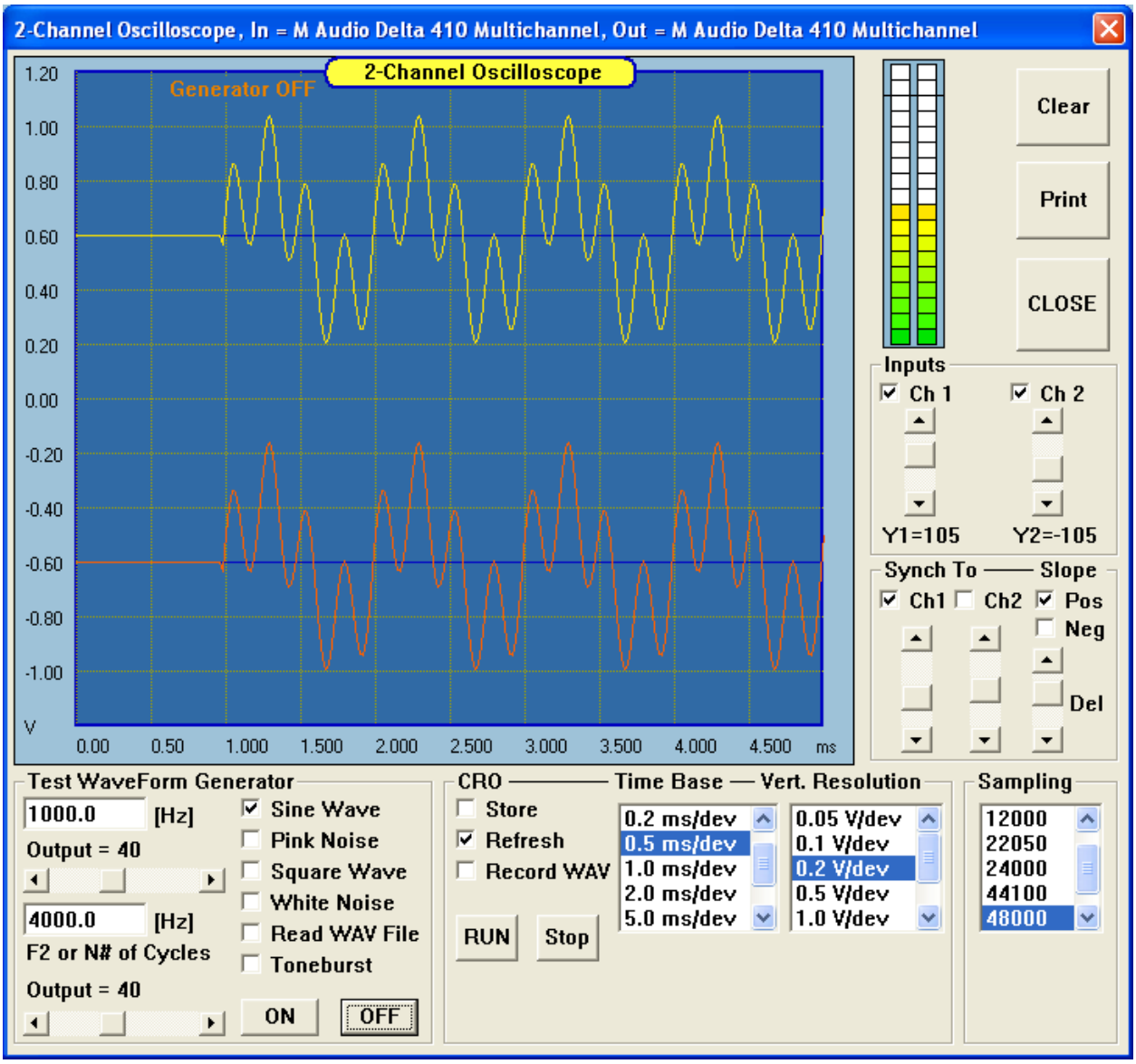


Figure 16.19. EasyLab Oscilloscope screen – Two-Tone signal 1000Hz + 40000Hz.

14. **Sampling list box –** use the selection provided to switch the sound card to the desired sampling frequency

Oscilloscope Screen Controls

1. **RUN** button – press this button to start display analysis.
2. **Stop** button – press this button to stop display analysis.
3. **Store** - click on this check box to display only current result of the sweep.
4. **Refresh** - click on this check box to display only current result of the sweep.
5. **Record WAV** – You can record about 45Mb of data per song using this facility. Once you stopped recording. A "File Save" box pops up, where you can enter the name of the recorded file. All recordings are made in STEREO with 48kHz sampling to give "studio quality" sound.
6. **Time Base** – select desired time scale.
7. **Vert. Resolution** (vertical) – Select appropriate vertical gain.

Inputs

1. **CH – 1** – click to turn ON channel 1.
2. **CH – 2 –** click to turn ON channel 2.
3. **Vertical slider for CH –** 1 – use it to position/move Ground Line for CH 1 on the display screen.
4. **Vertical slider for CH –** 2 – use it to position/move Ground Line for CH 2 on the display screen.

Synchronization

1. **Synch To CH – 1** – click to synchronize display to channel 1. Synch Level is adjustable
2. **Synch To CH – 2** – click to synchronize display to channel 2. Synch Level is adjustable

Slope

1. **Pos** – click to synchronize display to positive (going through "0" from negative to positive voltage).
2. **Neg** - click to synchronize display to negative (going through "0" from positive to negative voltage).
3. **Delay** – Moves the displayed signal right, as per set delay

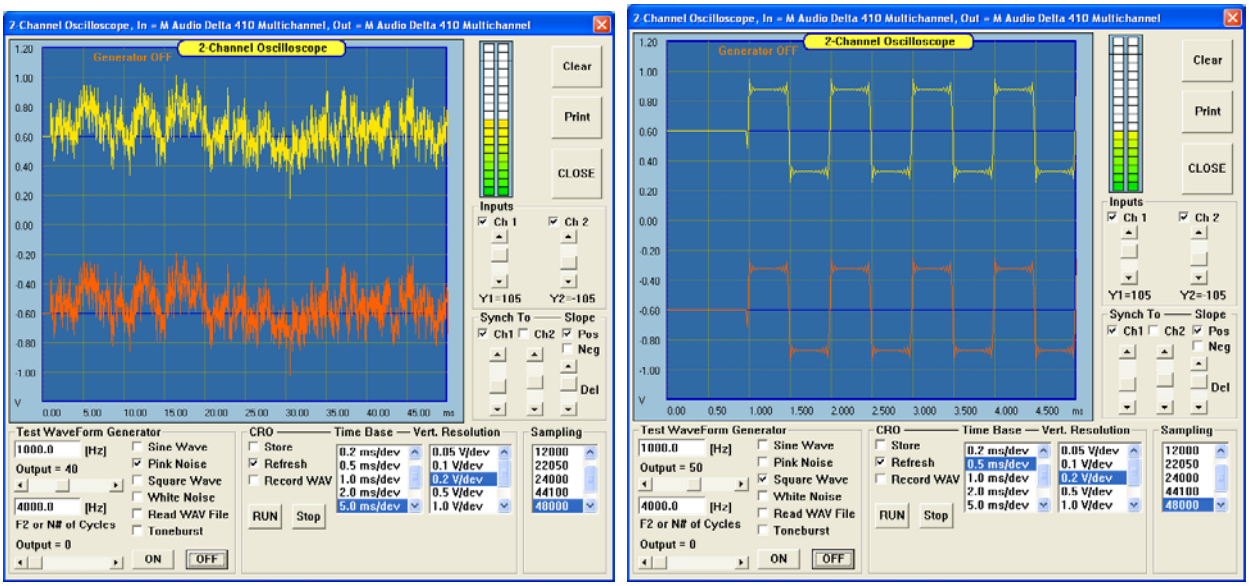


Figure 16.20. Pink Noise Generator Square Wave Generator

Additional Controls

1. **"Clear"** – clears the display.
2. **"Print"** – use this button to obtain a hard copy of the screen.
3. **"Done"** – closes the Oscilloscope screen.

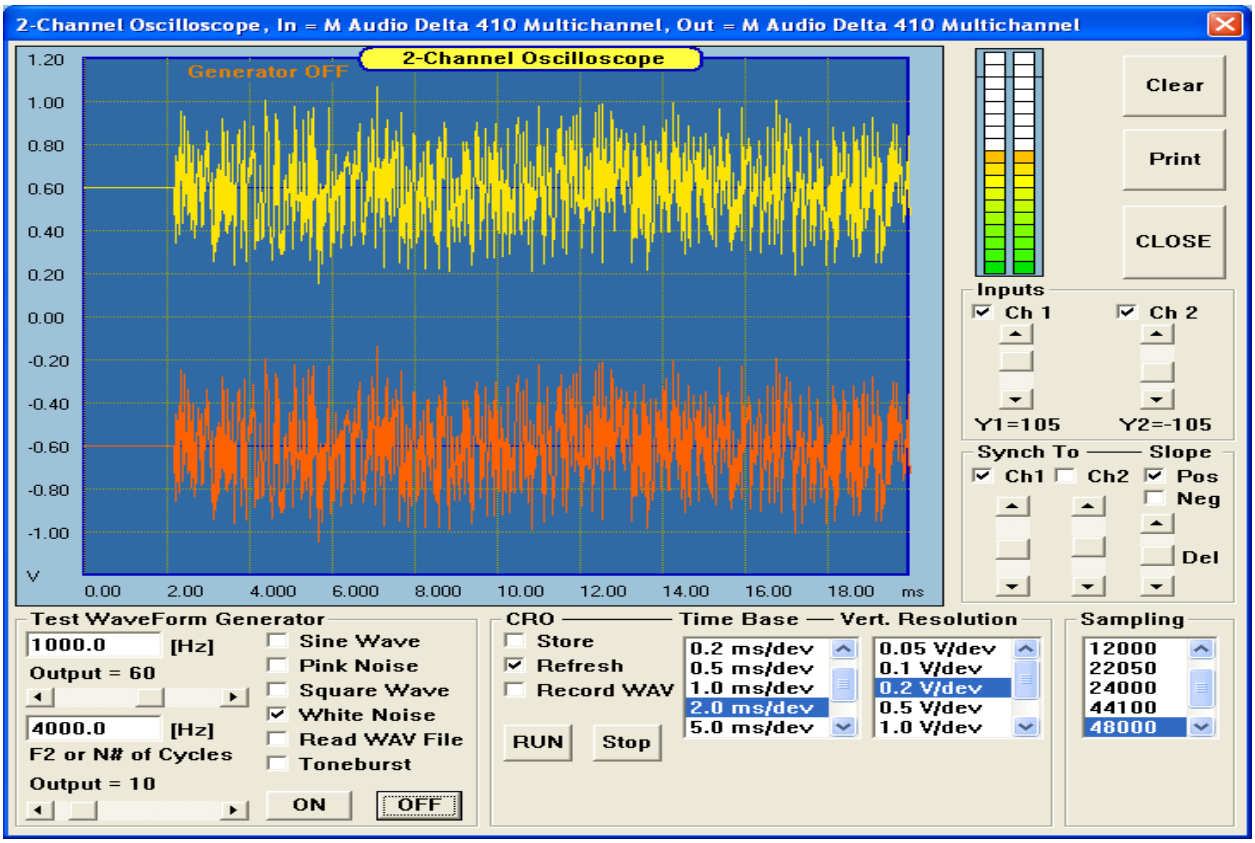


Figure 16.21 White Noise Generator

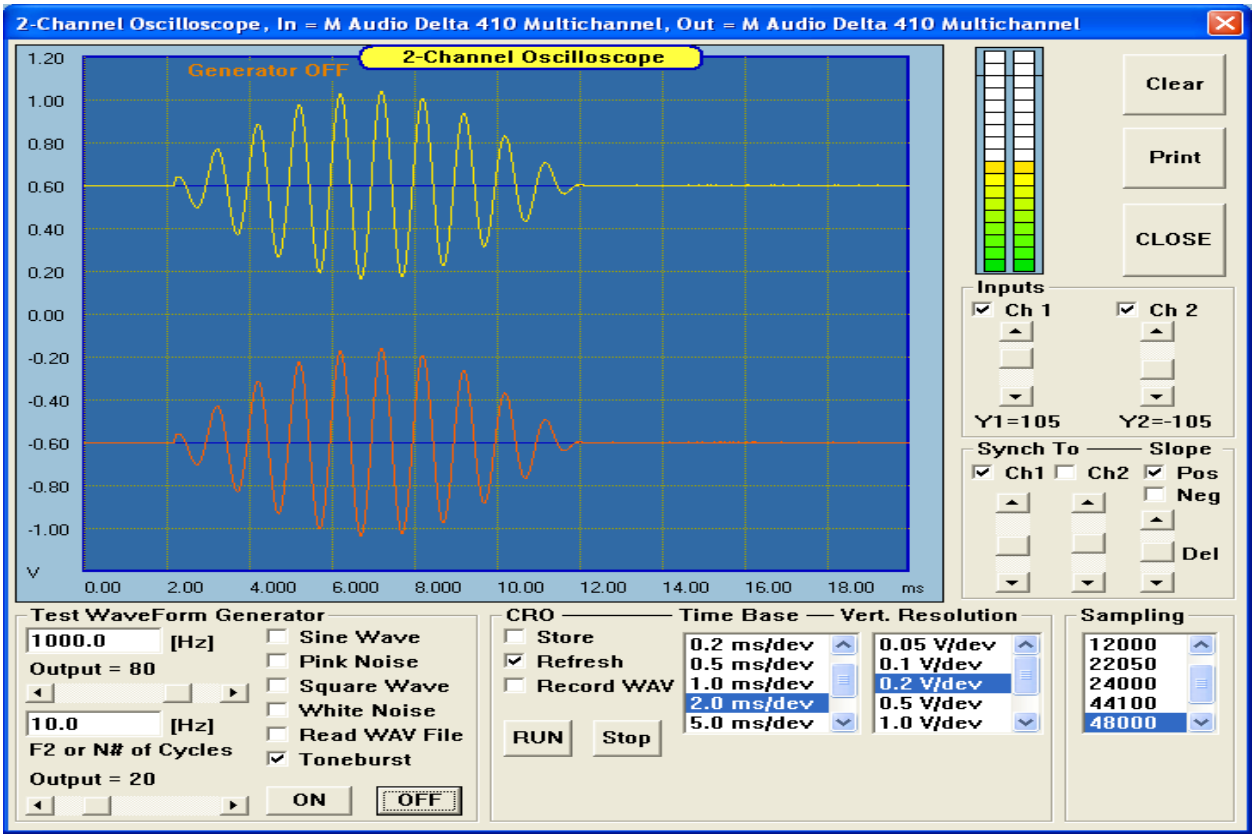


Fig 16.22. ToneBurst Generator.

Cumulative Spectral Decay and Tone Burst Waterfalls

Cumulative Spectral Decay plots are derived from impulse response. Therefore, the first step is to perform the classical MLS-type measurement on your loudspeaker and obtain the Impulse Response – see **Operation of the MLS Screen** section. Now you are ready to start decimating the impulse response to see what it is composed of. Each slice is then fed into the FFT to obtain corresponding frequency response. The first slice of the CSD plot (at zero delay) is the same as the standard frequency response. Successive slices are computed by shifting the windowed impulse response segment (as determined by the marker and cursor positions) right by a specified number of samples. This process is repeated many times generating the successive slices, which form the CSD plot. You can select the number of time points to shift between slices through the provided list box "**CSD Time Step**".

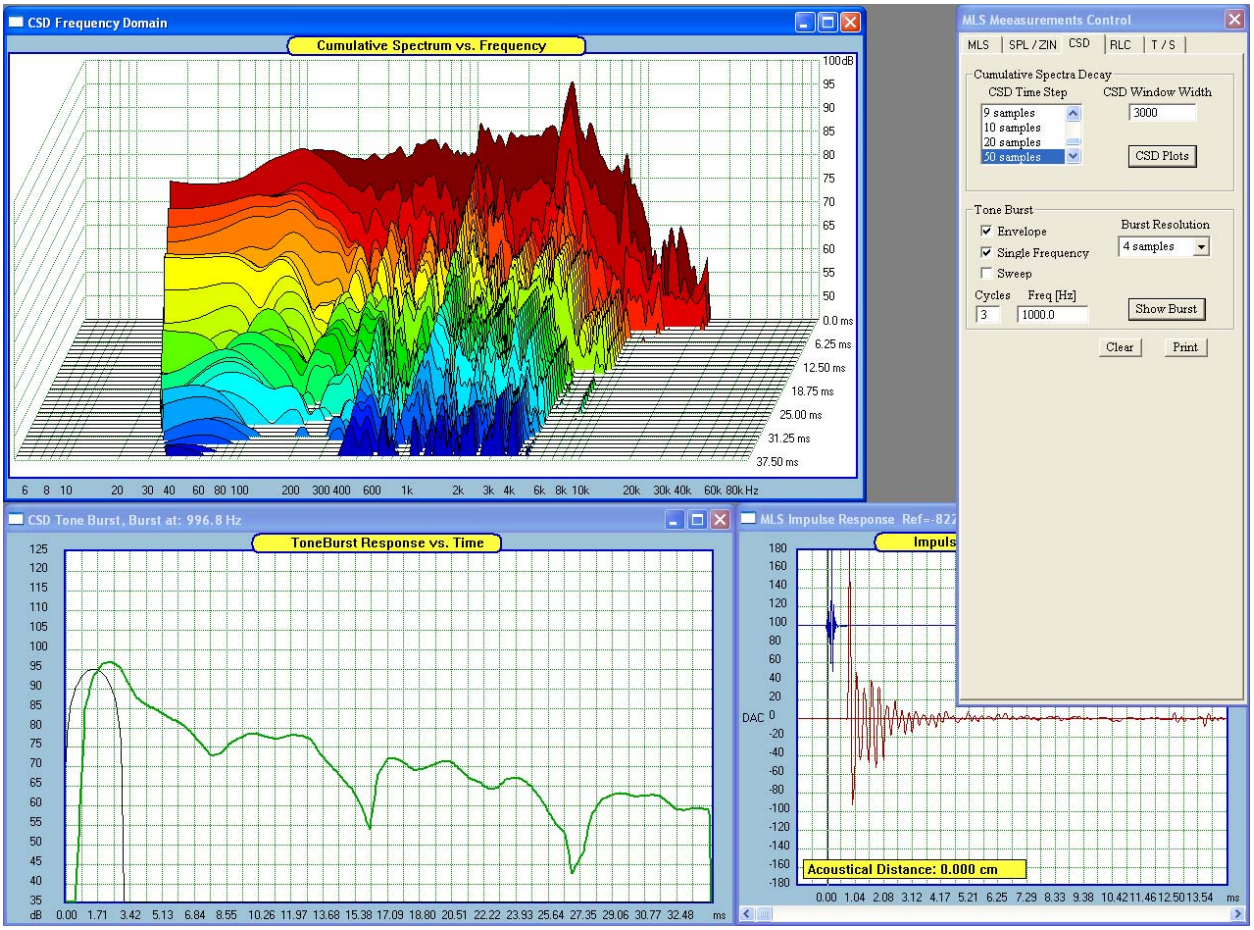


Figure 16.22. Impulse response ready to be decimated into CSD. 1000Hz ToneBurst response.

Impulse response of the loudspeaker under test is obtained the same way as you would perform MLS measurements. You need to get the hardware set-up ready and functional. Next, you would select the gating time window and smoothing option. If you need to remove the "time-of-flight" for the sound to reach the test microphone, please position the time cursor accordingly. Please also note, that SPL plot will not change when the time domain cursor is repositioned.

Finally, press the " CSD Plots" button to start the waterfall plots. CSD waterfall plots are frequently used to detect and visualize mechanical resonant characteristics in speaker cones, enclosures or horns. A resonance will manifest itself as a slowly decaying ridge along the time axis, due to its slow energy release over time. In other words, CSD plot shows you the frequency-decomposed decay. The basic idea behind the waterfalls involves using ever shorter time window to FFT process shorter and shorter section of the impulse response. This has profound consequences in the frequency resolution of each "slice".

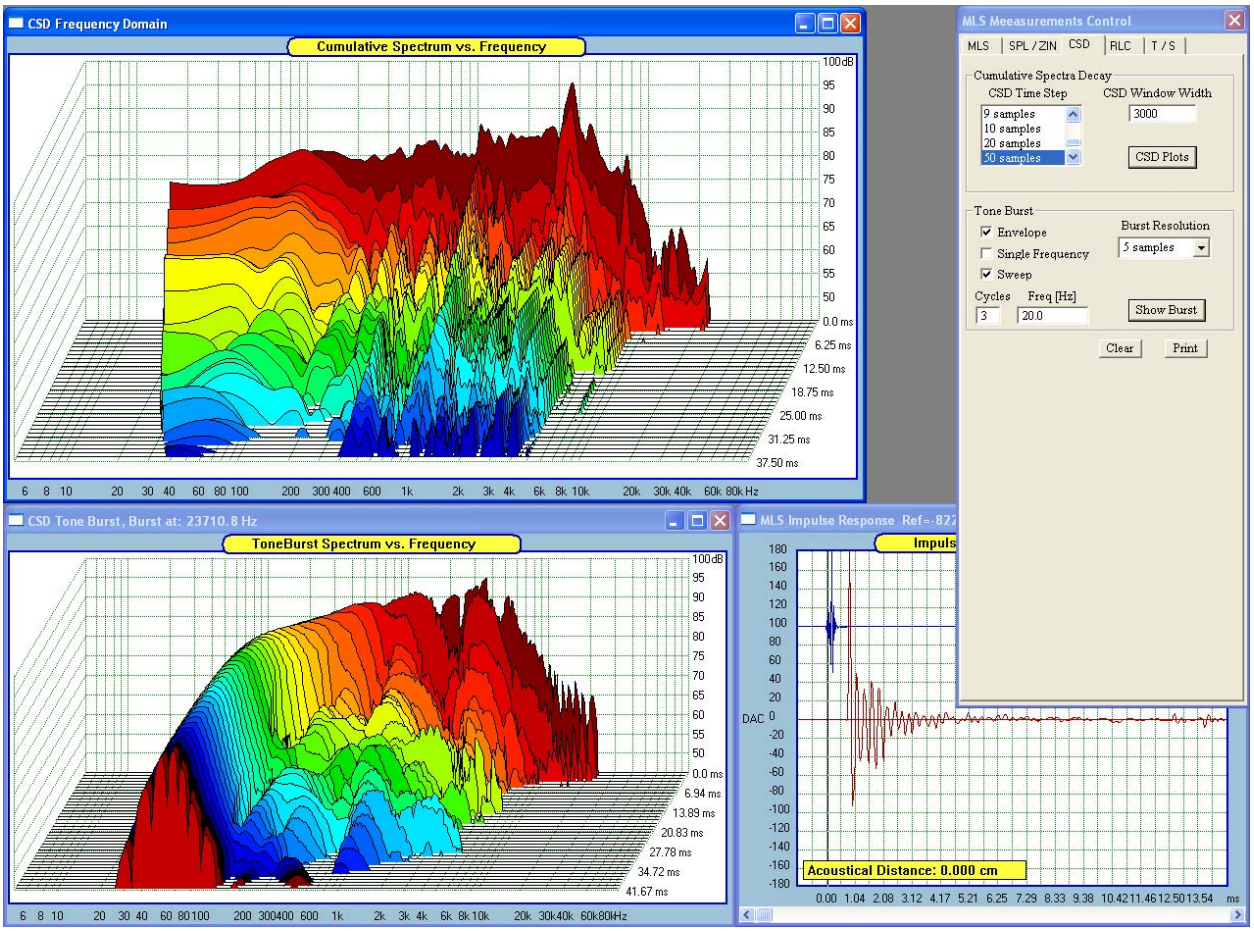


Figure 16.23. Example of CSD waterfalls AND ToneBurst waterfalls.

As the time window, T, gets shorter, the Fc = (sampling rate)/(window width) [Hz] gets larger and larger. Let's assume, that there are 25 "slices" to be computed and the "Time step" is selected as 20 samples. With the initial time window of 1000 samples and sampling rate of 48000 Hz, we have: **Fc1 = 48000/1000 = 48Hz resolution.** When the CSD process has finished, the time window has been reduced by 25 slices x 20 sample = 500 samples. **Fc2 = 48000/(1000-500) = 96Hz resolution.**

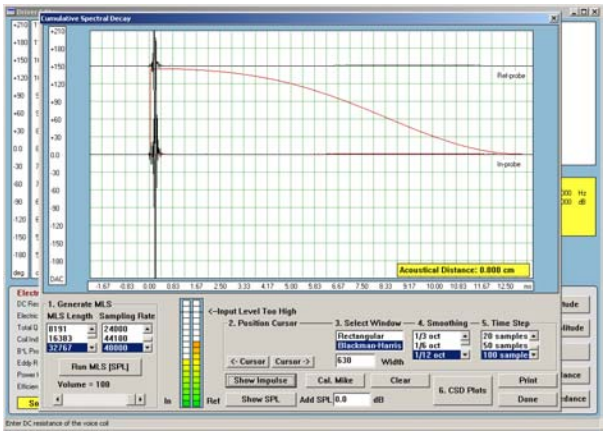
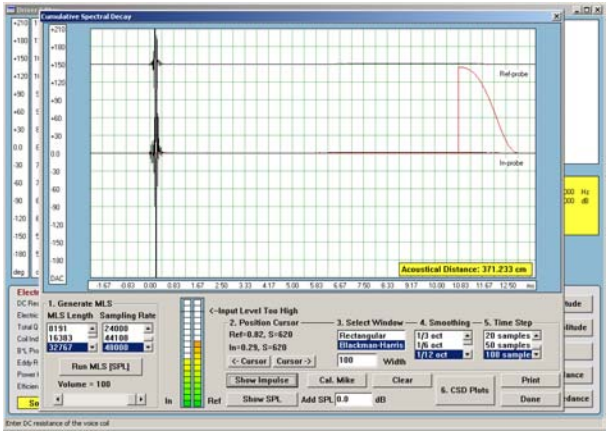


Figure 16.24 Time window at the start of the analysis



Time window at the end of the analysis

The resulting loss of resolution AND the lowest cut-off frequency is automatically accounted for in the program. In another example below, you can observe the effect of 3500 samples long initial time window progressively reduced by 36 slices and 100 samples/per step settings – the invalid portion of the CSD plots are simply not displayed.

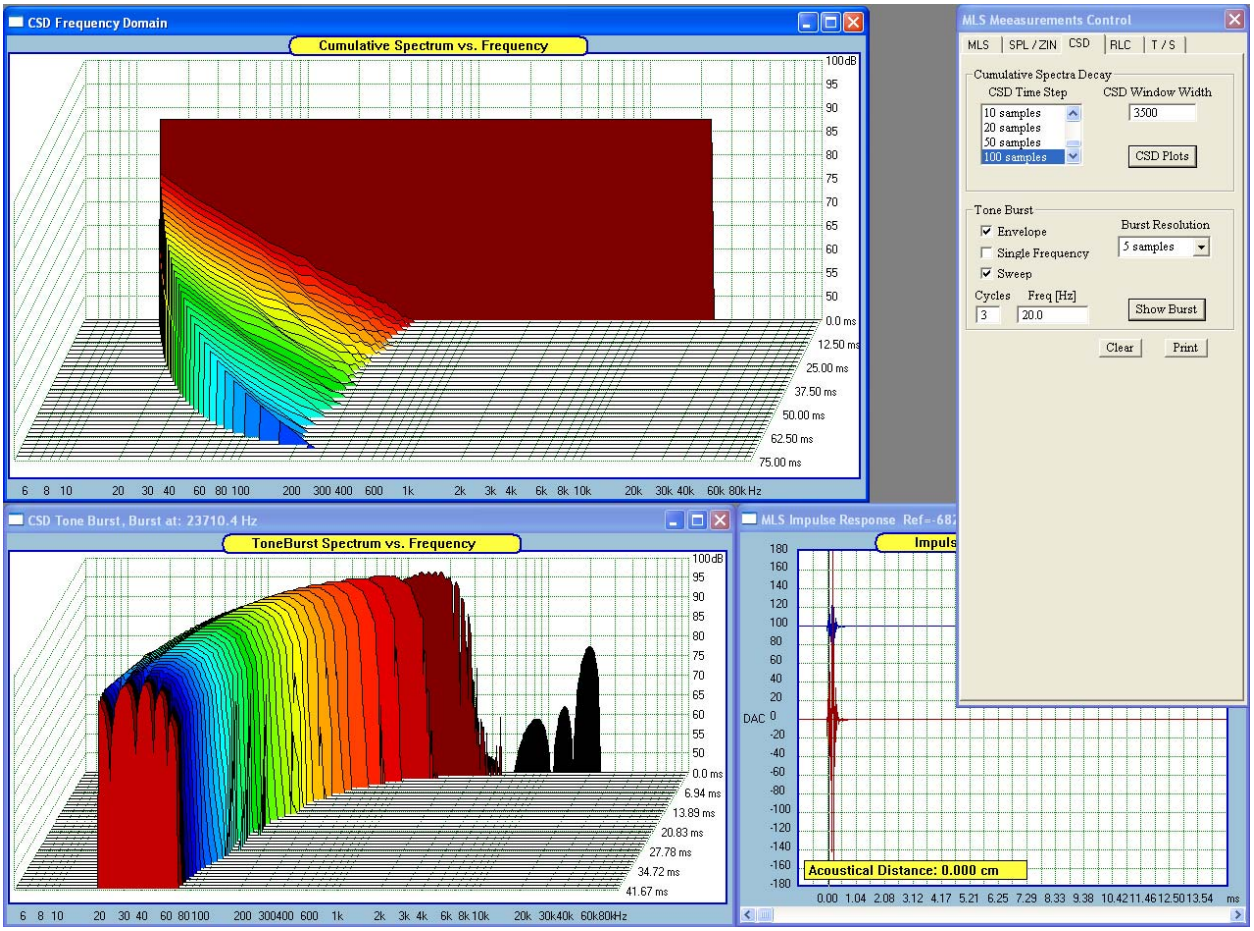


Figure 16.25. Result of "anchoring" the tail of the window and moving the starting point

Measurement results can be also saved into the nominated files and recall at the later time for post-processing. "Save Input Buffer" check box activated saving mechanism and "Use Saved Input Buffer" opens loading dialogue box.

When the "Cumulative Spectra" dialogue box is opened, you will notice a section in the middle of the box – see Figure 16.23. It's called "Tone Burst" and contains all controls necessary to perform tone-burst excitation analysis of the collected Impulse Response. You can generate tone-burst response at individual frequency or you can also generate waterfall-like family of responses, where the excitation was a tone burst of several cycles of each frequency. The responses were then assembled to resemble a waterfall-like display.

The measurement procedure is very similar to the standard CSD techniques. Firstly, you must collect the Impulse Response of your loudspeaker. The top section of the control box includes all standard controls for generating MLS signal and collecting the IR. The exact procedure was described already in this manual. If you need to use mike calibration file, please do so, using provided controls as well. Collected Impulse Response will be shown in the CSD Impulse response window.

We can now proceeded to the "Tone Burst" section:

1. **"Single Frequency"** – check this box to generate tone burst response for a selected frequency
2. **"Sweep"** – check this box to generate waterfall-like 3D display with a defined starting frequency.
3. **"Cycles"** – data field to enter number of cycles of in the tone burst.
4. **"Freq"** – enter frequency response of the tone burst, OR starting frequency for the Sweep mode.
5. **"Burst Res"** – Burst Resolution. Select the time scale using this control. For instance, if you select "5 samples", the program will display every 5-th sample of the convolution result. This way, you can zoom-in or –out on the time scale.
6. **"Burst"** – pressing this button activates calculation and display.

You will notice, that convolution calculations take much longer time at lower frequencies, therefore please select the starting frequency for the "Sweep" mode sensibly. In a way of an example, please collect the IR of your midrange driver and set the controls to:

1. **"Single Frequency"** – check this box.
2. **"Sweep"** – un-check this box
3. **"Cycles"** – enter number 3
4. **"Freq"** – enter 1000.
5. **"Burst Res"** – Select "5 samples"
6. **Press "Burst" button**. Now, see Figure 16.23 for example.

T/S (Thiele/Small) Parameters Extraction

Characterization of loudspeaker involves many parameters and can be quite involving task. There are small-signal parameters, large-signal parameters (temperature dependence, BL(x), Cas(x), Le(x), Xmax), frequency response, polar characteristics and so on.

In this section, we focus on a sub-section of this array of data that has become known as small-signal parameters or Thiele/Small parameters. Both researchers provided us with a good insight of the process of analysis and synthesis of the loudspeaker system based on those parameters.

It is perhaps worth pointing out, that Thiele's working model was a significant simplification. Consequently, following his published papers, more work went into expanding the model and making it more accurate. For instance, some research activities focused on more accurate representation of the voice coil impedance at higher frequencies. As a result, SoundEasy implements two impedance models. One model is an expanded and much more accurate, 4-stage LR version of the simple "single segment" representation of the "semi-inductor" in the voice coil. The second model represents somewhat different approach and involves "variable resistor" and variable inductor" proposal from J. Wright.

Basic small-signal parameters are:

Bl	-	Product of magnetic flux density, B and length of the voice coil wire in the magnetic gap
Cas	-	Acoustical compliance of the loudspeaker's suspension system
Cms	-	Mechanical compliance of the loudspeaker's suspension system
Fs	-	Resonant frequency of the driver in "free-air", including air load
Le	-	Inductance of the voice coil
Mms	-	Mechanical mass of the driver's vibrating system, including air load
Qms	-	Driver's Q-factor at resonance, considering only mechanical losses
Qes	-	Driver's Q-factor at resonance, considering only electrical losses
Qts	-	Driver's Q-factor at resonance, considering ALL losses
Re	-	DC resistance of the voice coil

Sd	-	Effective cone area (piston area)
Vas	-	Volume of air exhibiting the same compliance as the driver's suspension
η_0	-	Reference efficiency
SPL	-	Sound Pressure Level, usually measured for 1watt input power at 1 meter distance

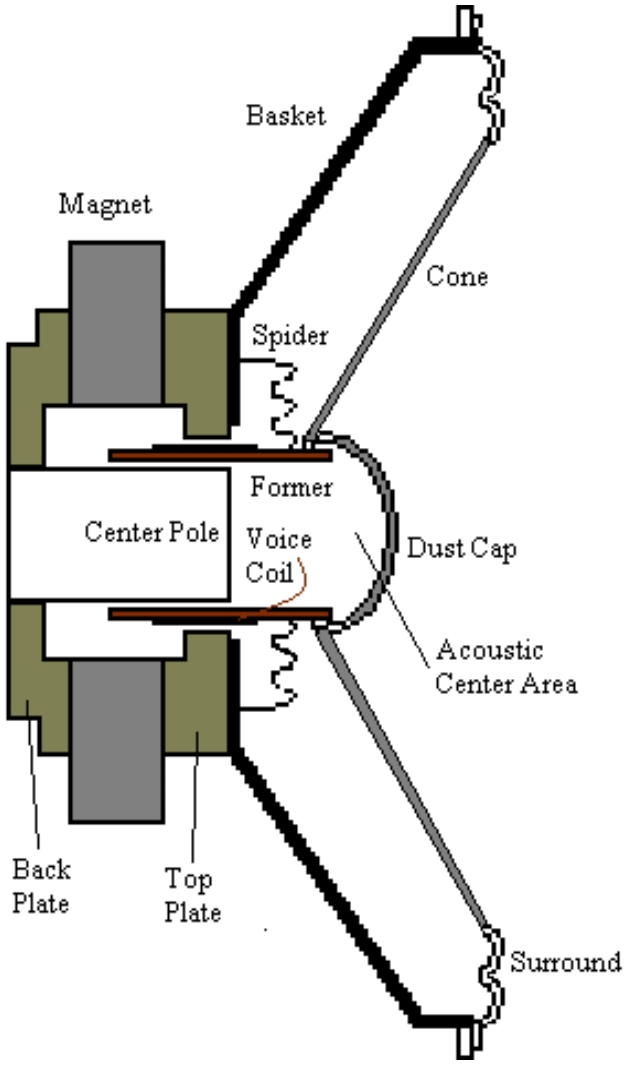


Figure 16.27. Cross-section view of the electro-dynamic loudspeaker

The surround + spider are commonly referred to as the "suspension system" and the magnet + the voice coil are referred to as the "motor system" of the loudspeaker. The "acoustic center" of the loudspeaker is the area most often located somewhere behind the dust cap. The exact acoustic center of radiation needs to be determined by measurement and is most often conveniently referenced to the mounting plane of the loudspeaker basket. In this case, you would refer to the "acoustic center offset".

The small-signal, electro-mechanical parameters described in this section are derived from input impedance of the driver. The only other data you will need to supply is the "effective" cone diameter and properly measured DC resistance, Re, of the voice coil. The Re value could be extracted from the impedance curve as well, but **better accuracy can be expected from a digital ohmmeter with 3 digit accuracy**.

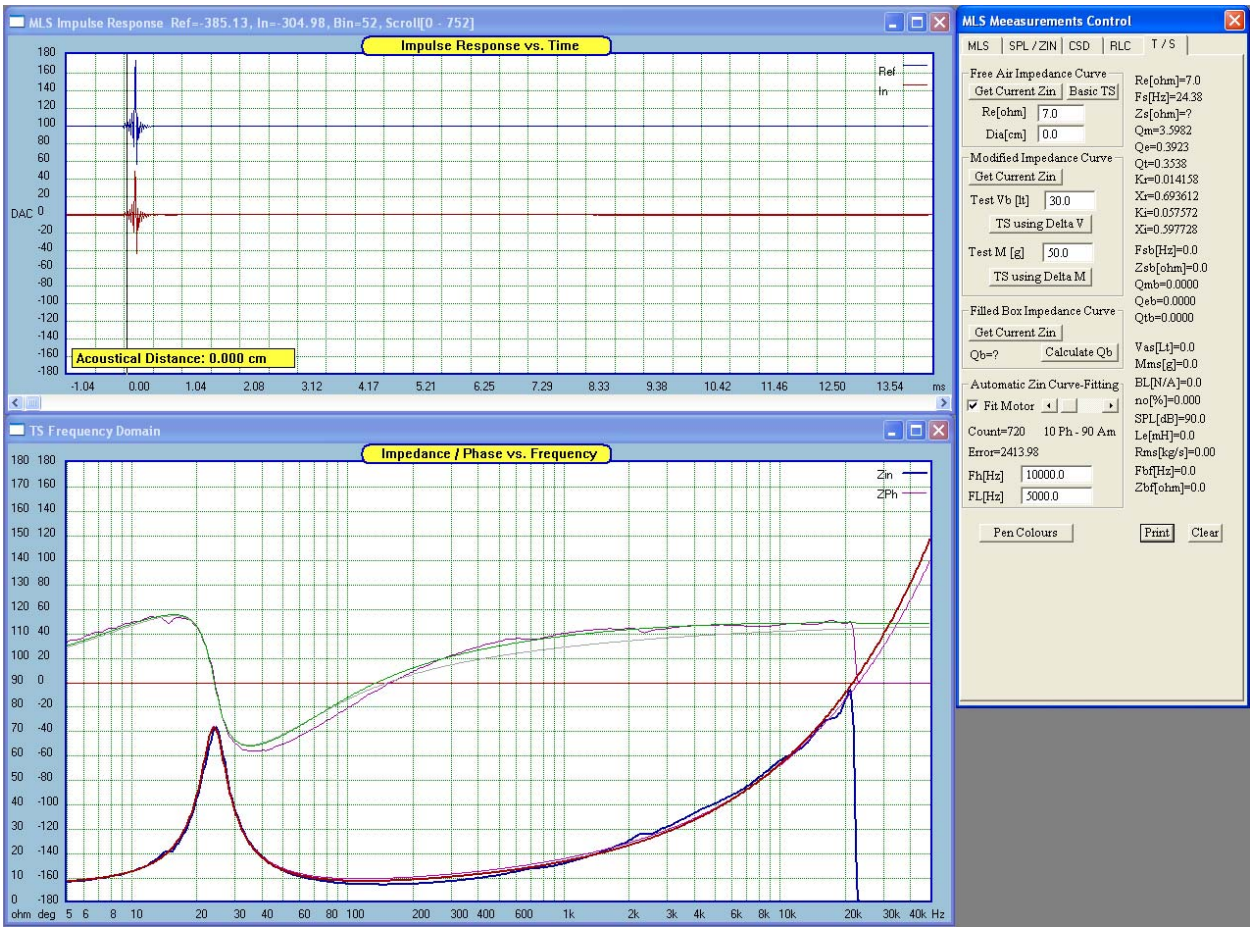


Figure 16.28 Free-air input impedance curve with basic TS parameters extracted

There are two simple methods for extracting Vas and other T/S small-signal parameters:

Added Mass method

The Added Mass technique requires additional mass, M_{Add} to be attached to the driver's cone and the new, "free-air" resonant frequency determined again. The extra mass should be large enough to cause resonant, Fs, shift by at least 25%.

$$M_{ms} = M_{Add}\left(\frac{F_s * F_s}{F_{sb} * F_{sb}} - 1\right)$$

Where:
Madd is the added test mass,
Fs is the loudspeaker free-air resonance,
Fsb is the loudspeaker resonant frequency with the test mass, Madd, attached.

With the Mms value calculated, we can proceed to establish the Vas as follows:

Cms = 0.0253/(Fs*Fs*Mms) and finally **Vas = ρ*c*c*Sd*Sd*Cms**

Where c, is the speed of sound, Sd, is the driver's piston area and ρ=1.18kg/cubic meter

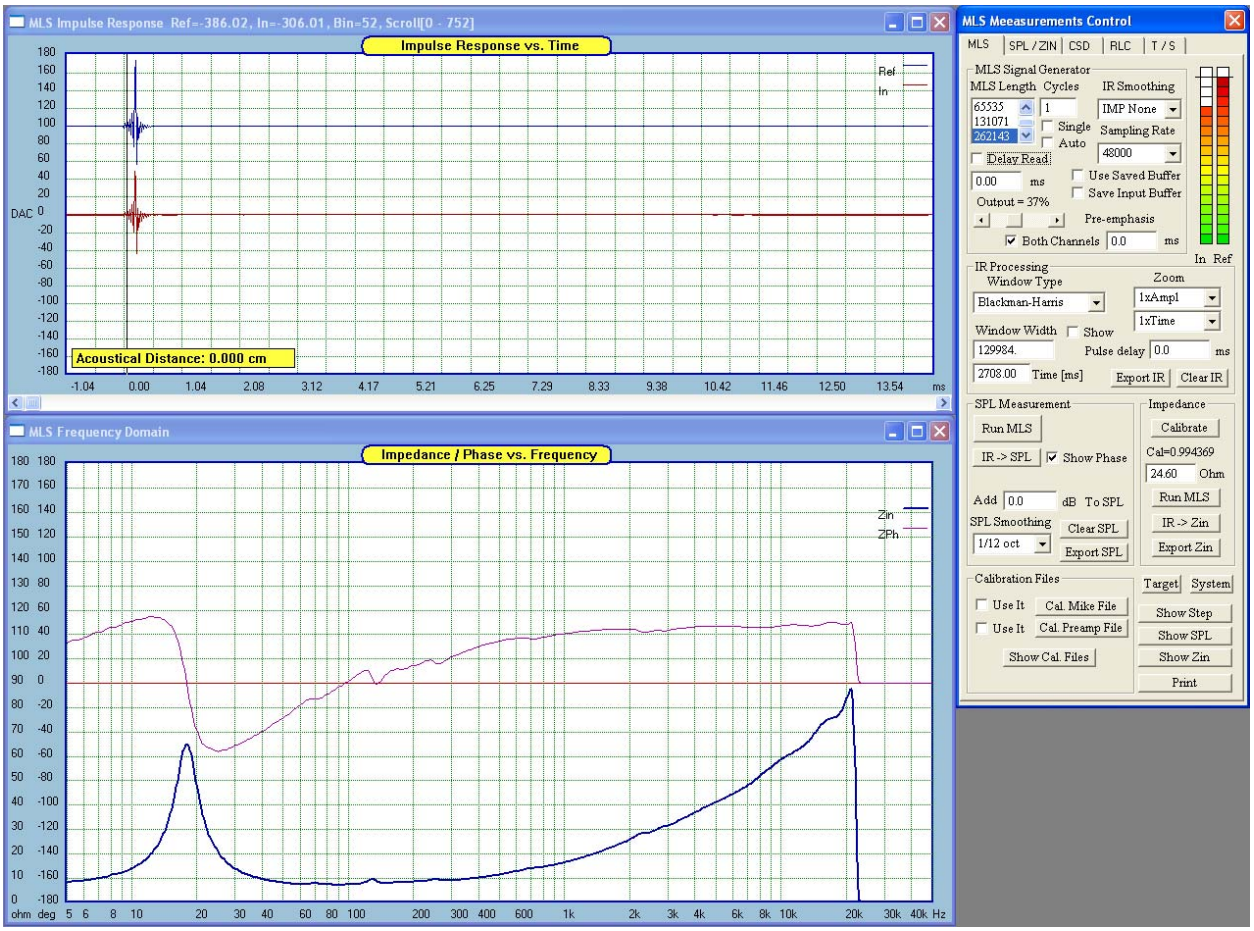


Figure 16.29. Shifted resonance due to added extra mass.

Delta Vas method

The Delta Vas method uses a well designed and well built test enclosure for the purpose of mounting the driver in it and measuring the new Fsb and new Qesb "in-the-box". This arrangement will shift the resonant frequency of the driver in the box upwards. You should aim at the enclosure volume, Vb, that would cause the resonant frequency to shift by at least 25%.

$$V_{as} = V_b \left(\frac{F_{sb} * Q_{esb}}{F_s * Q_{es}} - 1 \right)$$

Where:

Vb is the test enclosure volume,
Fs is the loudspeaker free-air resonance,
Qes is the electrical Q-factor of the driver in free-air
Fsb is the loudspeaker resonant frequency when mounted in the test enclosure with Vb volume
Qsb is the loudspeaker electrical Q-factor when mounted in the test enclosure with Vb volume

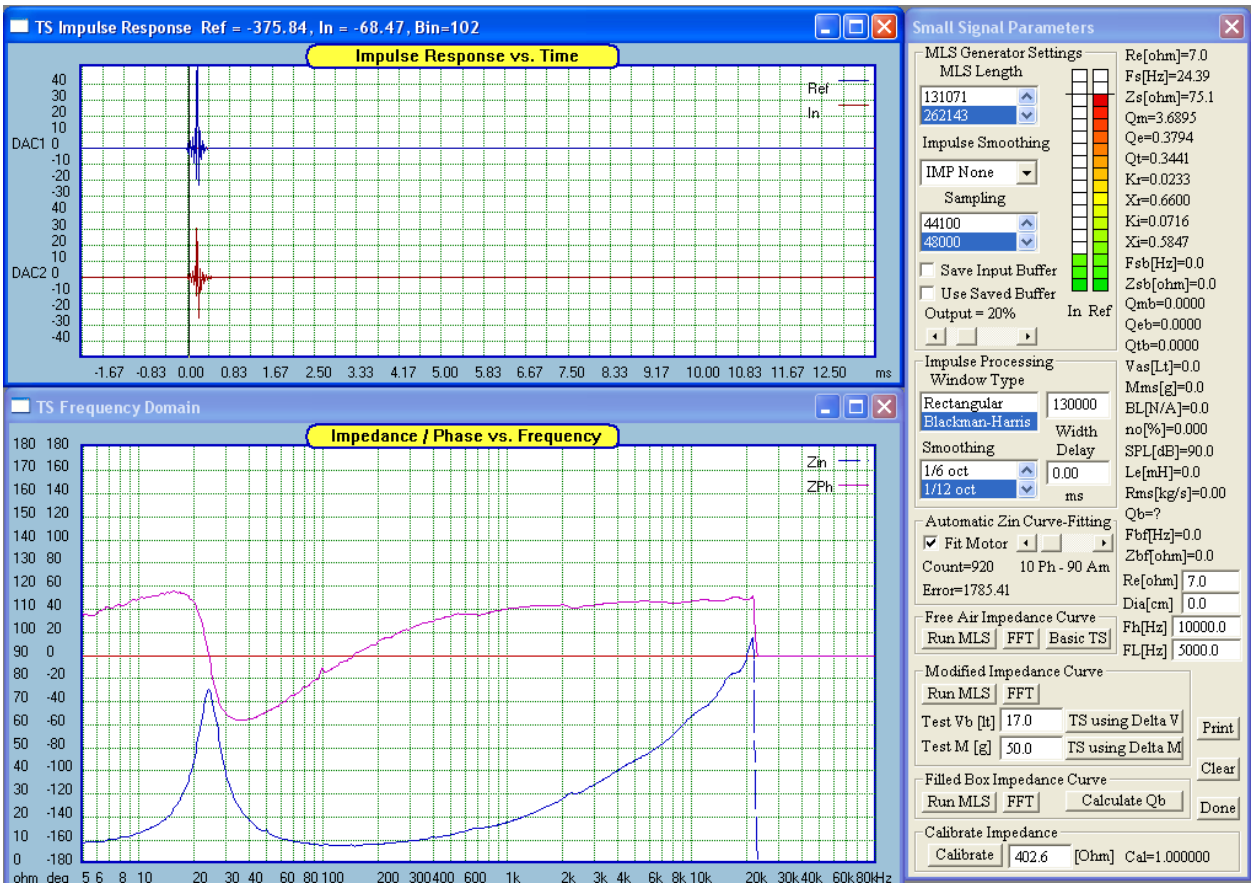


Figure 16.30. Free Air impedance curve

Extracting Voice Coil Parameters

This method of extracting parameters of the voice coil is attributed to J.Wright. Basically, the "free-air" input impedance of the driver, Zin(ω) is evaluated at two frequencies selected past the midrange frequency range. SounEasy uses f1 = 5kHz and f2 = 10kHz frequencies. Extracted parameters are used to calculate "variable inductor" and "variable resistor", which together, approximate the voice coil "semi-inductor" quite well. The method has the potential to provide much more accurate representation of the loudspeaker impedance at higher audio frequencies. Here are the formulas used by the program.

$$\omega_1 = 2\pi f_1$$

$$\omega_2 = 2\pi f_2$$

$$X_r = \frac{\log \dfrac{\mathrm{Re}(Z_{in}(\omega_1))}{\mathrm{Re}(Z_{in}(\omega_2))}}{\log \dfrac{\omega_2}{\omega_1}} \qquad K_r = \frac{\mathrm{Re}(Z_{in}(\omega_1))}{\omega_1^{X_r}}$$

$$X_i = \frac{\log \dfrac{\mathrm{Im}(Z_{in}(\omega_1))}{\mathrm{Im}(Z_{in}(\omega_2))}}{\log \dfrac{\omega_2}{\omega_1}} \qquad K_i = \frac{\mathrm{Im}(Z_{in}(\omega_1))}{\omega_1^{Xi}}$$

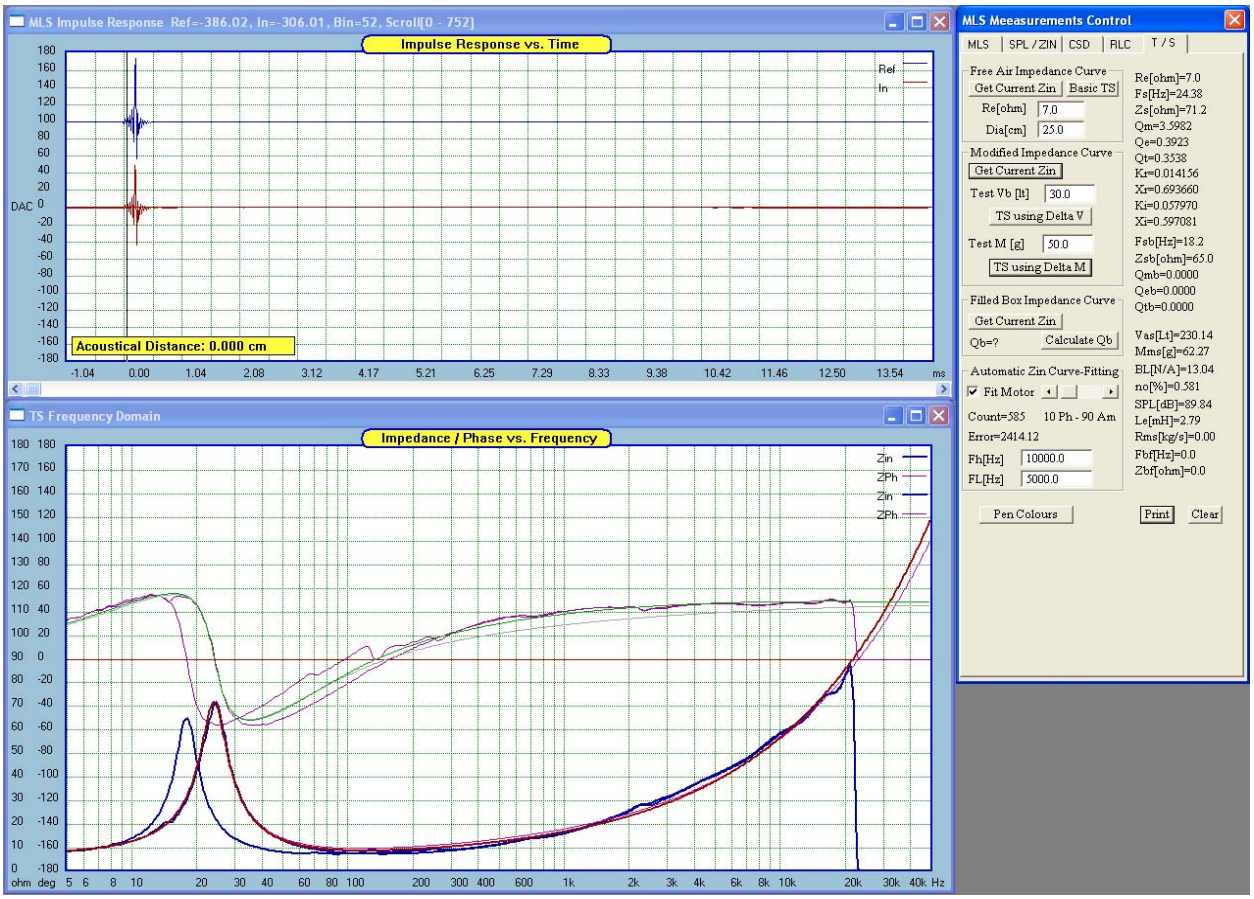


Figure 16.31. “Delta M” method.

Example of measured T/S set of parameters for a 12” woofer, using Delta Mass Method (added 50gram) are:

1. Fs = 24.38 Hz
2. Zmax = 71.2 ohm at Fs
3. Qm = 3.5982
4. Qe = 0.3923
5. Qt = 0.3538
6. Le at 152Hz = 2.79 mH
7. Mms = 62.27 gram
8. Vas = 230.13 Lt
9. BL = 13.04 N/A
10. Efficiency = 0.581%
11. SPL = 89.84 dB/1W/1m
12. Kr = 0.014156, Xr = 0.69366
13. Ki = 0.05797, Xi = 0.597081
14. Re = 7.0 ohm (measured separately with ohmmeter)
15. Diameter = 25.0 cm (measured separately)

Measurement Method for T/S Parameters Extraction

In general, you can use any MLS length and sampling frequency combination. However, more often than not, you will find, that certain sampling frequencies do not produce good results. This is to do with soundcard internal frequency conversions and clocking issues. Also, you can use test resistor of any reasonable value (10-500ohms), however we recommend, that you use the highest resistor value you can work with and also the longest MLS sequence. The long MLS sequence will assure good frequency resolution required at the low end of the frequency band (48000/262000 = 0.18Hz) and then you can use Blackman-Harris smoothing window of 131000 length, giving you smoother curves and still good resolution of 0.36Hz.

Higher resistor value will assure, that your test setup is working indeed in "small-signal" area. This is often forgotten issue, that we are trying to extract small-signal (no BL(x) or Cas(x) type of non-linearity involved).

When making impedance measurements, Zin, the "Ref" probe, sensing V1, is connected to the output of the power amplifier and the "In" probe, sensing V2, is connected right across the loudspeaker. The measured impedance is calculated from the formula:

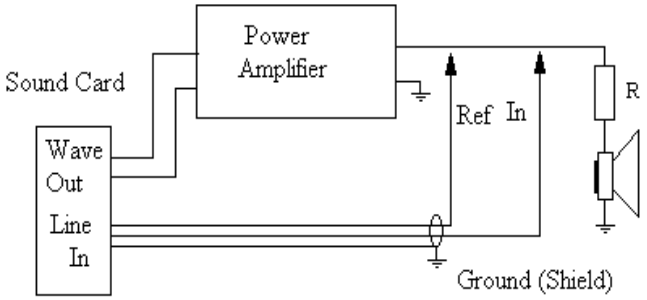
Zin = R*(V2/V1)/(1 – V2/V1), R > 400 Ω is recommended for constant current method

The method used for direct impedance measurements is close to constant current. This is because the source impedance (resistor R = 400 Ω, or higher inserted between the power amplifier output and the loudspeaker) is high so that the current following through the voice coil remains relatively constant**. Using lower resistance R is possible**. The formula above indicates clearly, that input impedance is heavily dependant on the voltage ratio, as measured by left and right channel of the sound card. There is a potential for significant error, as the voltage gains in both channels may not be exactly equal. More importantly, if you use probe resistive dividers, the voltage divider ratios may not be equal in both probes due to resistor tolerances. In order to ensure accuracy of the impedance measurements, you must CALIBRATE the system before a valid impedance measurement can be taken. During the calibration process, you will feed the same signal to both probes and the program will measure and memorize the difference in actual readings. This information will be use next for scaling the measured voltages.

CALIBRATION FOR IMPEDANCE MEASUREMENT

Follow the steps below to perform calibration for impedance measurements.

4. **Connect the test setup**: power amplifier, test probes. Connect the test probes TOGETHER, at the output of the power amplifier – as shown below. Power-up the power amplifier.

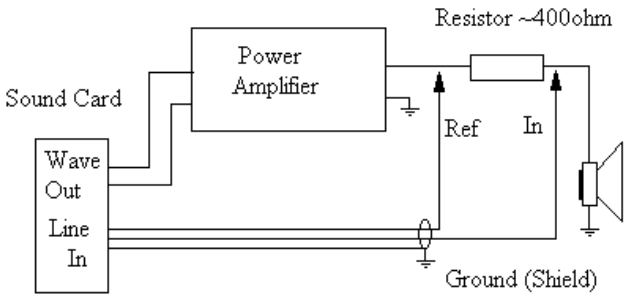


16.32 Calibration for impedance measurements.

5. **Select "T/S Parameters**" from the main menu to invoke MLS measurement screen.
6. **Perform calibration**
   - Select Generator level from 0-100%.
   - Press "Calibrate" button in the Impedance group box. The setup will now generate 1000Hz tone needed for measurement of both channels. **The "In" indicator corresponds to the MEASURED signal from your loudspeaker terminal and the "Ref" indicator corresponds to the REFERENCE signal probe (connected to the output of the power amplifier).**

Having calibrated the system you can immediately follow the steps below to perform MLS measurements of the input impedance and phase of your loudspeaker.

9. **Connect the test setup**: power amplifier, test probes, test resistor and the "loudspeaker under test". Power-up the power amplifier.



16.33 Test setup for small-signal parameters extraction. Resistor can be used from 10-500ohm.

10. **Select "T/S Parameters**" from the main menu to invoke corresponding MLS measurement screen.

11. **Set up MLS sequence**
    - Select "MLS Length" from the provided list box – on SB Live! Sound card set to 262141.
    - Select sampling rate from the provided list box– on SB Live! Sound card set to 48000.
    - Select Generator level from 0-100%.

12. **Select windowing parameters – this is done automatically**.
    - When the MLS sequence has finished, you should see the impulse response of the measured signal on the screen. The program will automatically place the time cursor 10 samples to the left of the peak value.
    - Select the type and the width of the time window. **Please DO NOT create wider windows than the length of the MLS.**

13. **Select smoothing parameter**.
    This is a simple selection of one of the provided smoothing ratios: 1/3, 1/6, 1/12, 1/24 and 1/48 octave.

14. **Select additional Impulse smoothing.**
    - The calculated impulse response will be smoothed before it is processed with FFT. However, calculation time will be longer.

15. **Enter "Re" AND "Effective Diameter" data - IMPORTANT**.

16. **Generate Impedance curve – this is two-step process**

    - 7.1. Simple press the **"Run LMS**" button in the **"Free Air Impedance Curve"** group and wait a short while for the calculations to finish. You should now hear the noise coming out from the measured speaker. Your "Input Level LED" indicator should lit-up for "In" bars and "Ref" bars. **The "In" indicator corresponds to the MEASURED signal from your loudspeaker terminal and the "Ref" indicator corresponds to the REFERENCE signal probe (connected to the output of the power amplifier).** Please note, that wider smoothing band will result in increased computation time.

      Now, press the **"FFT"** button to obtain the Zin curve for free-air loudspeaker.
      At this point, you can press "**Basic TS"** button to obtain a subset of the loudspeaker parameters.

    - 7.2 Finally press the **"Run LMS**" button in the **"Modified Impedance Curve"** group and wait a short while for the calculations to finish. You should hear again the noise coming out from the measured speaker. Your "Input Level LED" indicator should lit-up for "In" bars and "Ref" bars.

      Now, press the **"FFT"** button to obtain the Zin curve for the added mass technique OR the added Vb technique.

17. **You are now ready to extract the small-signal parameters. Depending on the chosen method, enter "Test Vb" – for the Delta V method OR enter "Test M" for Delta M method and press corresponding button to extract the T/S parameters.**

Advanced MLS And Analogue Measurement Features

Improving Accuracy of TS Parameters Extraction

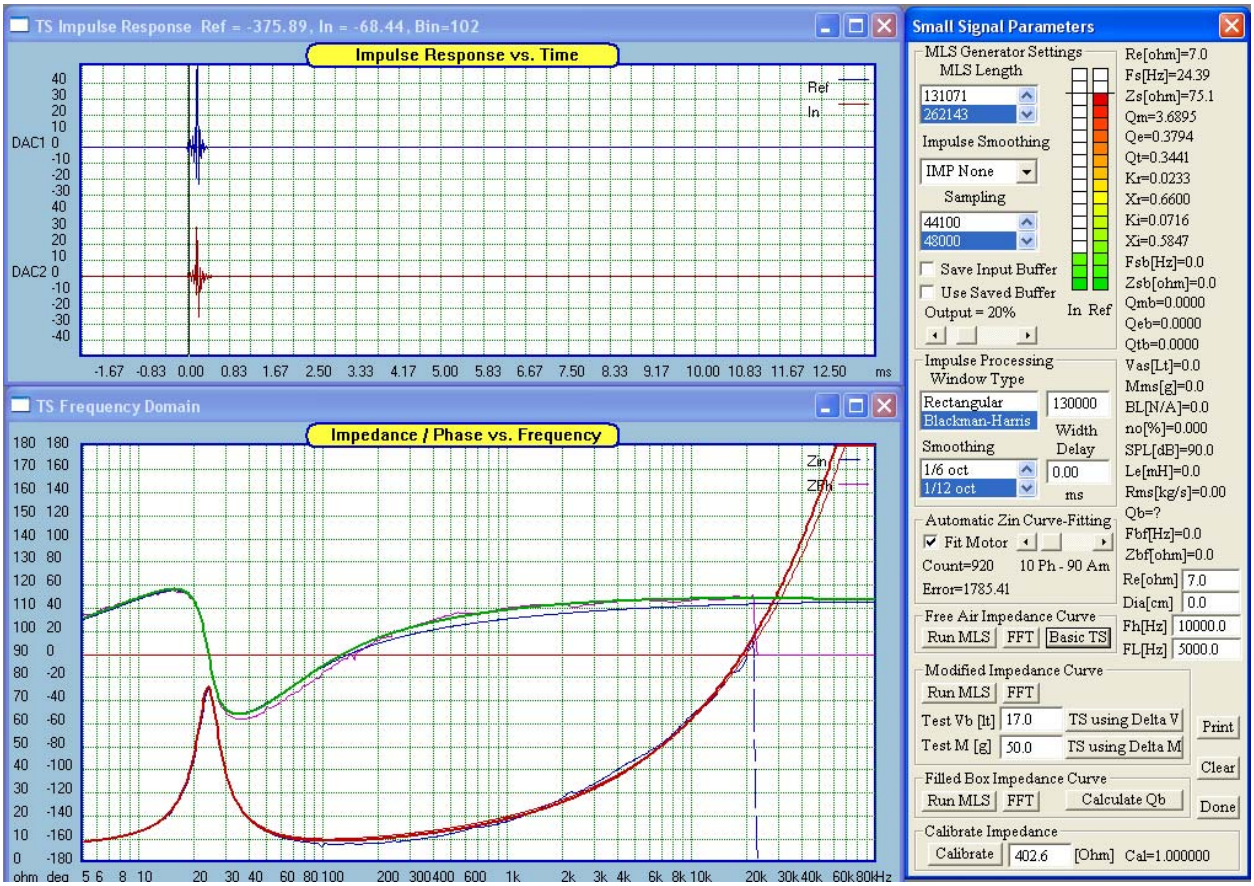
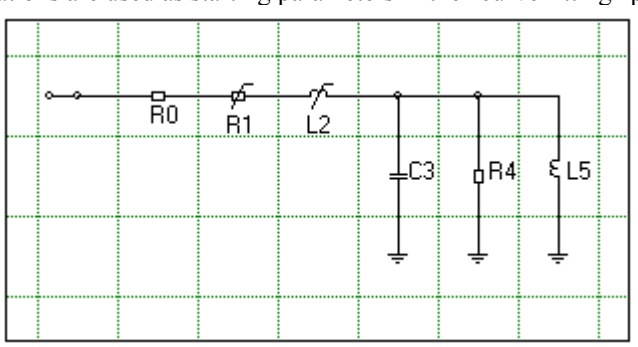


Figure 16.34. Kr, Ki, Xr, Xi. parameters calculated via "curve fitting" from the impedance curve.

Loudspeaker input impedance model developed by J.Wright is proving to be quite popular. The most obvious reason for this is the need for only 4 elements to characterize the high frequency section of the impedance curve. Taking a closer look at the results of calculated Kr, Ki, Xr, Xi reveals, that indeed a good match between calculated and measured curves can be accomplished for modulus of the input impedance. The phase of the input impedance is somewhat less accurate on some loudspeakers. To overcome this problem, we have implemented additional operation of "curve fitting" the modeled Zin onto the measured modulus and phase of the input impedance. Now, the modeled curve has to fit the measured curve across the whole measurement frequency range – not only at two selected frequencies, as calculated before. The values of Kr, Ki, Xr, Xi as calculated by previously given equations are used as starting parameters in the "curve fitting" process.



R0 = Re
Ts = 1/(2*π*Fs)
R = Re*Qm/Qe
C = (Ts*Qe)/Re
L = (Ts*Re)/Qe
R1 = Kr *ω^{Xr}
L2 = Ki*$\omega^{(Xi-1.0)}$

Simple impedance model for curve fitting of Kr,Ki, Xr,Xi.

The results of the optimization are shown on Figure 16.27. It is easily observable, that both: modulus and phase of the input impedance are now much better matched to the measured results. The new Kr, Ki, Xr, Xi are highlighted in red. The "curve fitting" mechanism will be automatically activated if you check the **"Fit Motor"** check box and is implemented as optimization (error minimization) algorithm. The algorithm accounts for both: (1) modulus and (2) phase of the input impedance assigning weighting in accordance with the position of the slider provided below. For instance, **10Ph – 90Am** indicates, that Phase contributes 10% and Amplitude (modulus) contributes 90% to the total error. This way, you can guide the optimization algorithm for the best fit between phases and amplitudes for each measured speaker individually.

Extracting Enclosure Qb Parameter

The method for extracting enclosure Qb, is based on Dr. D'Appolito's concept described in his Loudspeaker Design Cook book. You will need to perform three runs of MLS measurements on your loudspeaker under different conditions and the program will extract the Qb from collected data. Extracting Qb parameter is provided from **"T / S" TAB**. Please follow the steps outlined below:

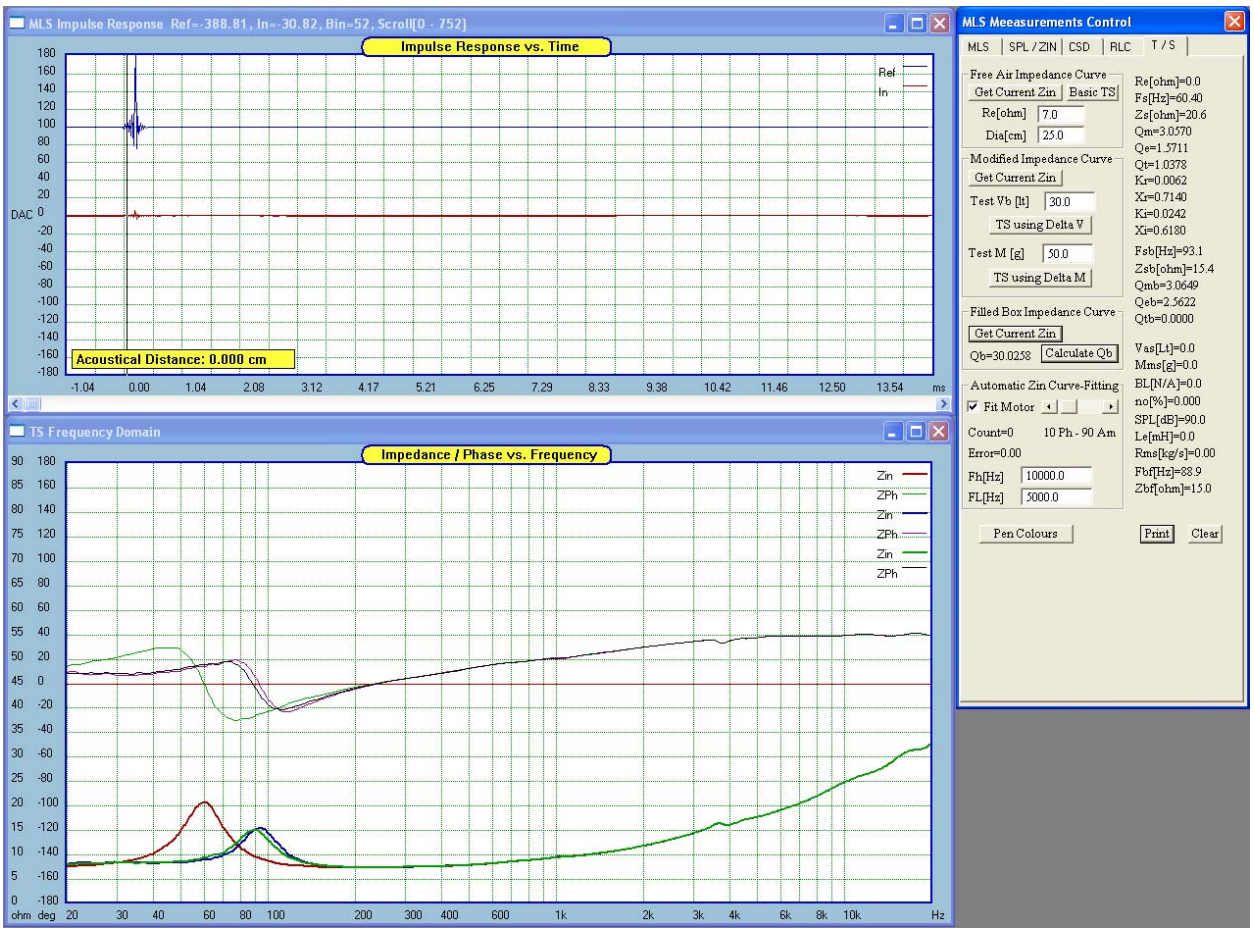


Figure 16.35 Extracting Qb process.

1. **Setup MLS** – Select MLS length (262143), Sampling Rate (48000), Volume (say 30), and your test resistor value (360ohm in our example).
2. **Calibrate system** – follow impedance calibration procedure described before.
3. **Select window parameters** – Blackman-Harris and 130000-width.
4. **Select smoothing parameters –** 1/6 oct and perhaps you may want extra smooth curves with Impulse Averaging – thus select one of the options from the "**IR Smoothing"** list box**.**
5. **Run 1 (Free Air Impedance Curve) –** Place the loudspeaker in the free-air conditions. Press **"Run MLS"** and then **"IR->Zin"** buttons on the MSL screen to capture the first impedance curve**.**

6. Go to **"T/S"** TAB and press "Get Current Zin" in the "Free Air Impedance Curve" group.
7. **Run 2 (Modified Impedance Curve) –** Mount the loudspeaker in **unfilled** enclosure you wish to measure Qb. Press **"Run MLS"** and then **"IR->Zin"** buttons on the MSL screen to capture the second impedance curve**.**
8. Go to **"T/S"** TAB and press "Get Current Zin" in the "Modified Impedance Curve" group.
9. **Run 3 (Filled Box Impedance Curve)** – Mount the loudspeaker in **filled** box you wish to measure Qb. Press **"Run MLS"** and then **"IR->Zin"** buttons on the MSL screen to capture the third impedance curve**.**
10. Go to **"T/S"** TAB and press "Get Current Zin" in the "Filled Box Impedance Curve" group.
11. Finally, enter value for Re in the T / S TAB, and **press "Calculate Qb" button** – if all previously measured parameters check out, you will se Qb calculated and displayed on the screen in the results area.

Figure 16.35 shows and example measurements taken on a small 8" woofer. Green impedance curve is the "**free-air**" plot. The red impedance curve is the "**unfilled box**" plot and the black impedance curve is the "**filled enclosure**" plot.

Zoom-In and Scroll Impulse

Having captured the Impulse Response of the loudspeaker, you may wish to review the time zone around the start of the pulse. Or perhaps you wish to review the impulse past the current window display. There are several controls allowing you to zoom-in and scroll the impulse. **Scroll is accomplished by moving the slider at the bottom of the "MLS Impulse response" window.** Zoom-in feature is implemented in time domain and amplitude domain. There are two list boxes located in the "Zoom" area. Amplitude zoom can be selected from x1 to x32 and time zoom can be selected from x1 to x8 zoom.

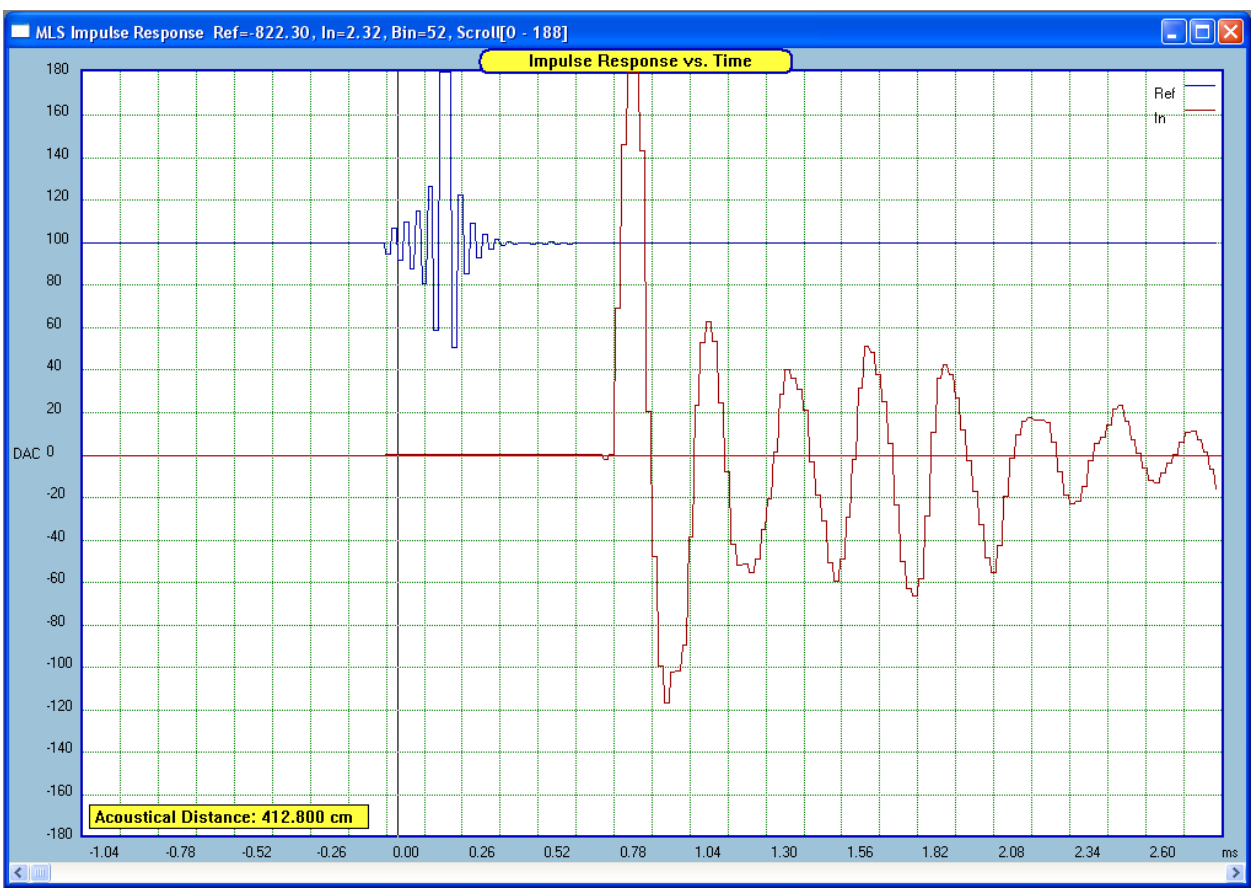


Figure 16.36. Example of "4 x Zoomed-in" impulse.

Pre-emphasized MLS signal

Standard MLS signal is basically a broadband spectrum noise. When testing higher frequency driver using such signal, you could unknowingly drive the loudspeaker into non-linear area of operation. In cases like this, it may be beneficial to use MLS signal with reduced low-frequency spectrum – the pre-emphasized MLS signal. **The pre-emphasis feature is activated automatically, by entering greater than zero value into "Pre-emphasis" data field**. Pre-emphasis is typically employed as a first order, high-pass circuit and the cut-off frequency is expressed as a time constant.

Example on Figure 16.37, shows the spectrum of pre-emphasized MLS signal with the time constant of 1.0ms. Performing normal measurements, you would use the pre-emphasized MSL signal to excite the DUT and connect the "Ref" probe to the amplifier's output and the "In" probe to the mike pre-amp output (typical SPL measurements configuration). If you perform loop-test with both probes connected to the same signal, the frequency response will show as flat.

You can also output the normal MLS signal on one sound card output and the pre-emphasized MLS signal on the other output. This will give you the opportunity to review the actual spectrum of the pre-emphasized MLS signal. Simply clear the "Both Channels" check box and connect the broadband MSL output to "Ref" probe and the pre-emphasized MLS to "In" probe – see below. Please note, that MLS length is 131071 and Blackman-Harris window is 30000.

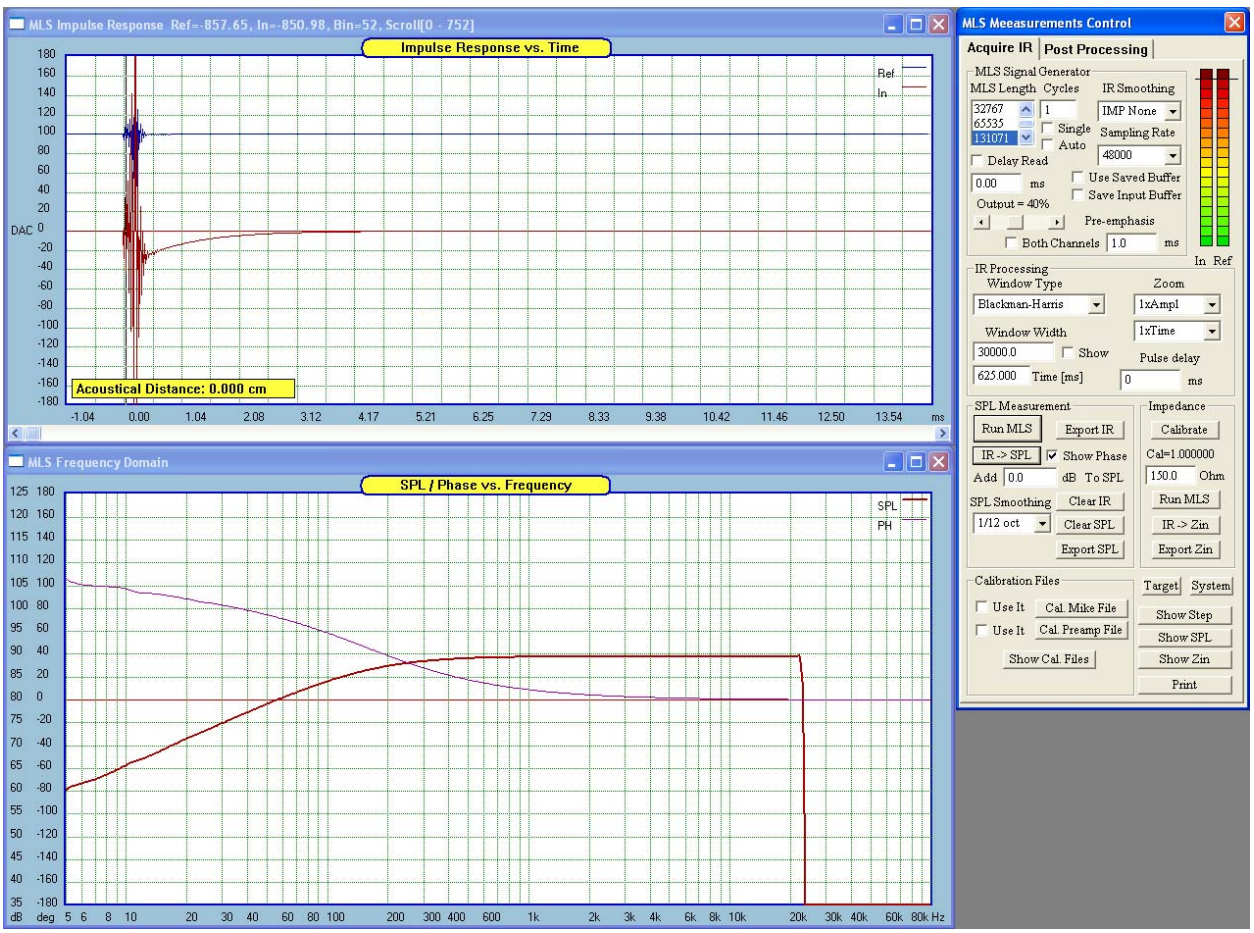


Figure 16.37. Example of pre-emphasized MLS signal

Using Input Impulse Buffer

There maybe some of the post-processing activities, that may require re-taking the measurement several times. Should you require to do this, SoundEasy now has the ability to save the captured data into a buffer and reading it from the buffer repeatedly. Because of this, you only need to make single MLS measurement and then you can switch off the test equipment. Saving the buffer is performed via a standard file saving dialogue box, that allows you to assign a file name to the saved buffer, when the **"Save Input Buffer"** check box is checked. After that, you can check the **"Use Saved Input Buffer**" box and then read the saved buffer rather then make repetitive measurements. **You can save and read unlimited number of Impulse Responses for later post-processing.**

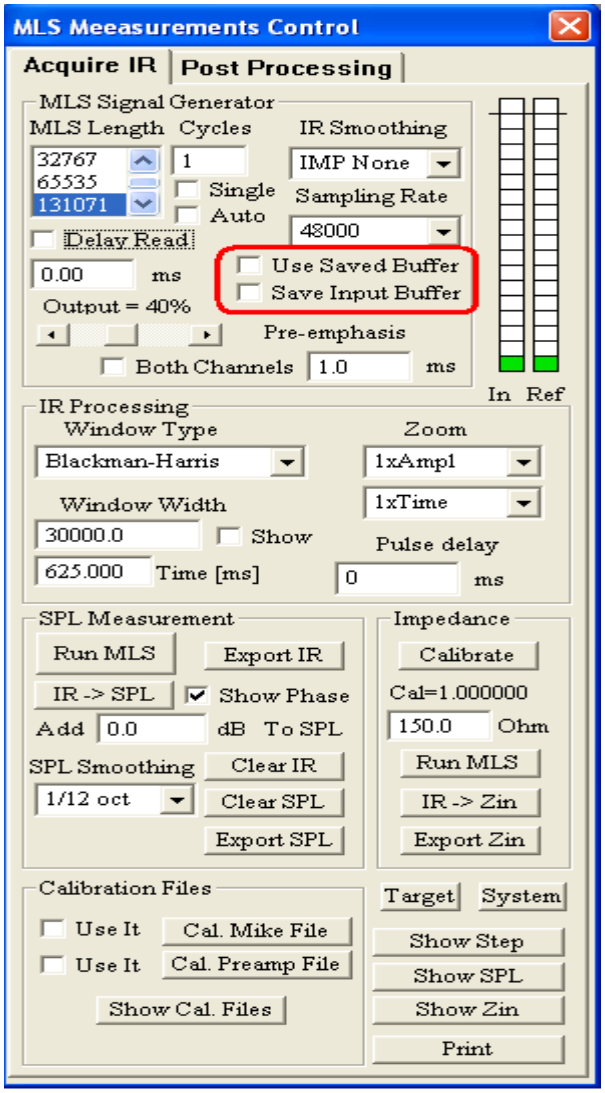


Figure 16.38. "Save Input Buffer" and "Used Saved Input Buffer" check boxes.

Impulse smoothing

Impulse smoothing may be useful when you require better phase clarity at higher frequencies. Several smoothing options are available and selectable from "pull-down" list box located in the "Generate MLS " group of controls. If you wish to switch off the impulse smoothing, please select **"IMP None"** from this list box. The **"IMP 1/32"** provides the strongest smoothing. An example of non-smoothed impulse phase response is shown on Figure 16.39. It is easy to observe, the phase response around 10kHz undergoes several jumps. When the impulse smoothing with "IMP 1/32" option was selected, the phase response around 10kHz and also HBT response are much smoother.

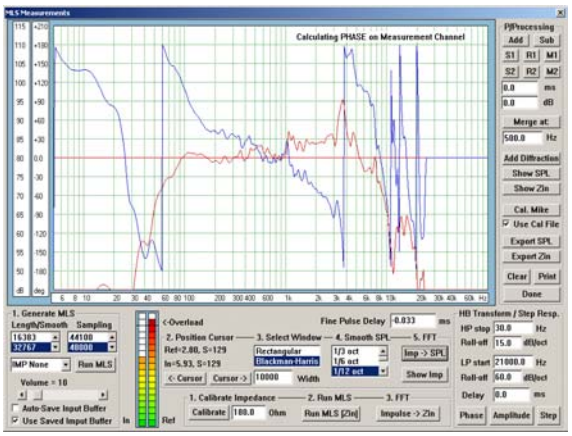


Figure 16.39 No impulse smoothing

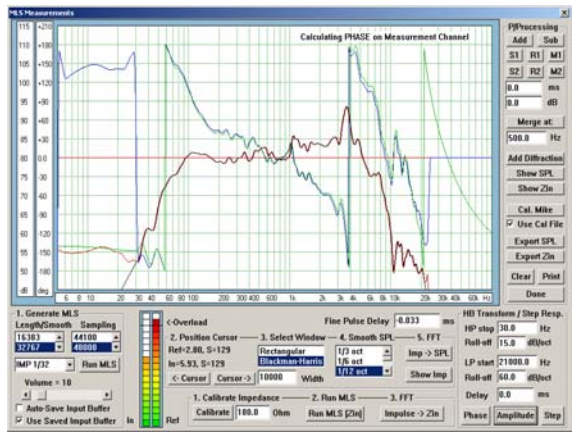


Figure 16.40. “IMP 1/32” impulse smoothing.

Step Response

<u>Step response is calculated from Hilbert-Bode Transform</u>. This approach allows you to basically use HBT to “clean” high and low-end of the spectrum before the step response is calculated.

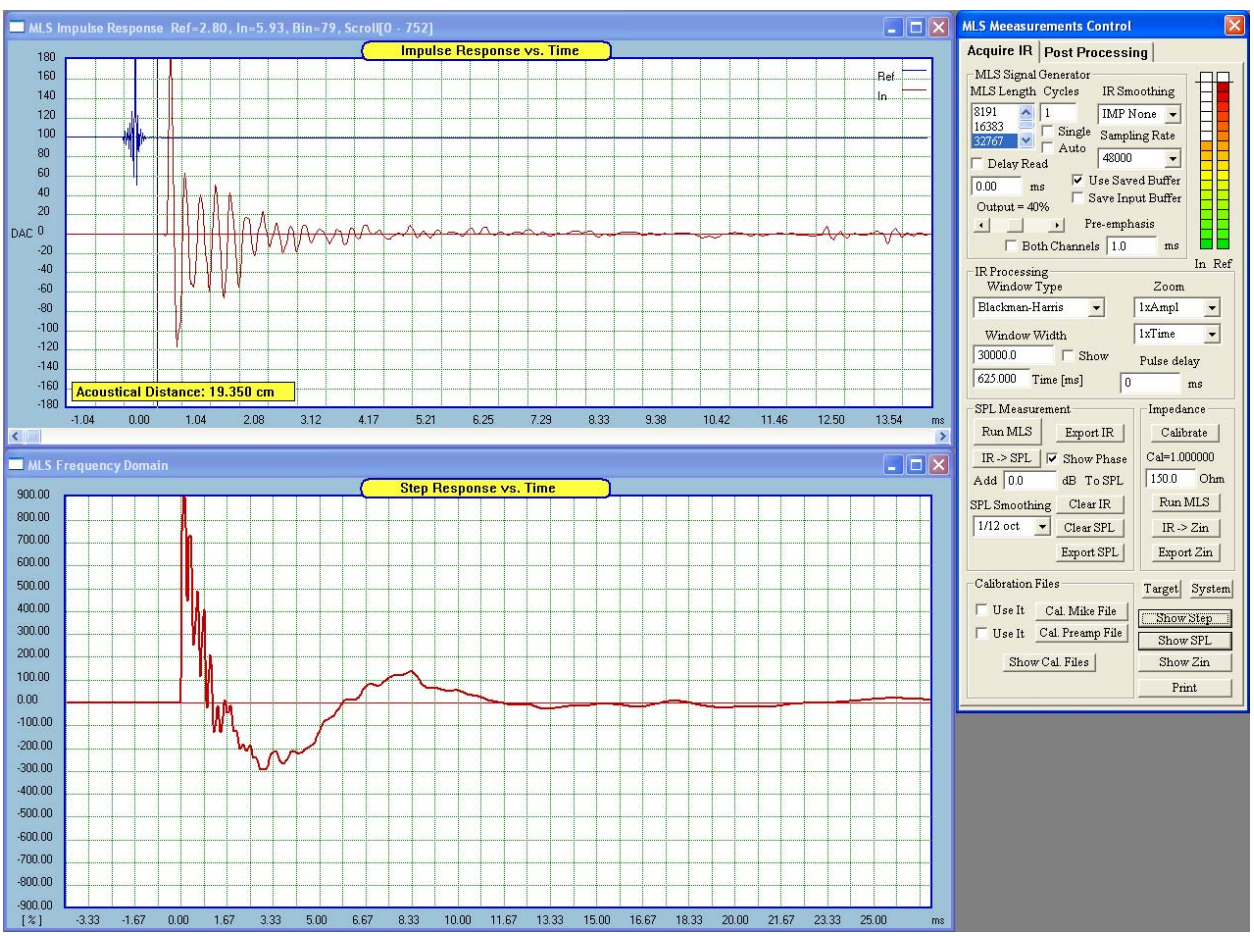


Figure 16.41 Step Response.

Step response calculated for signal captured around 80dB level (see Figure 16.39) may be a little to low in amplitude when the Step response is calculated. This my be prevented by adding, say 20dB to the amplitude just before the HBT is calculated – see Figure 16.42.You can also adjust time domain axis for Step Response plots. The scale is tied to the sampling rate, so by changing temporarily sampling rate, you can select different scales.

1. Simply enter 20dB into the [dB] **"Add"** on the "Acquire IR" TAB,
2. Press **"IR->SPL"** button to actually add the 20dB to the curve – see Figure 15.42.

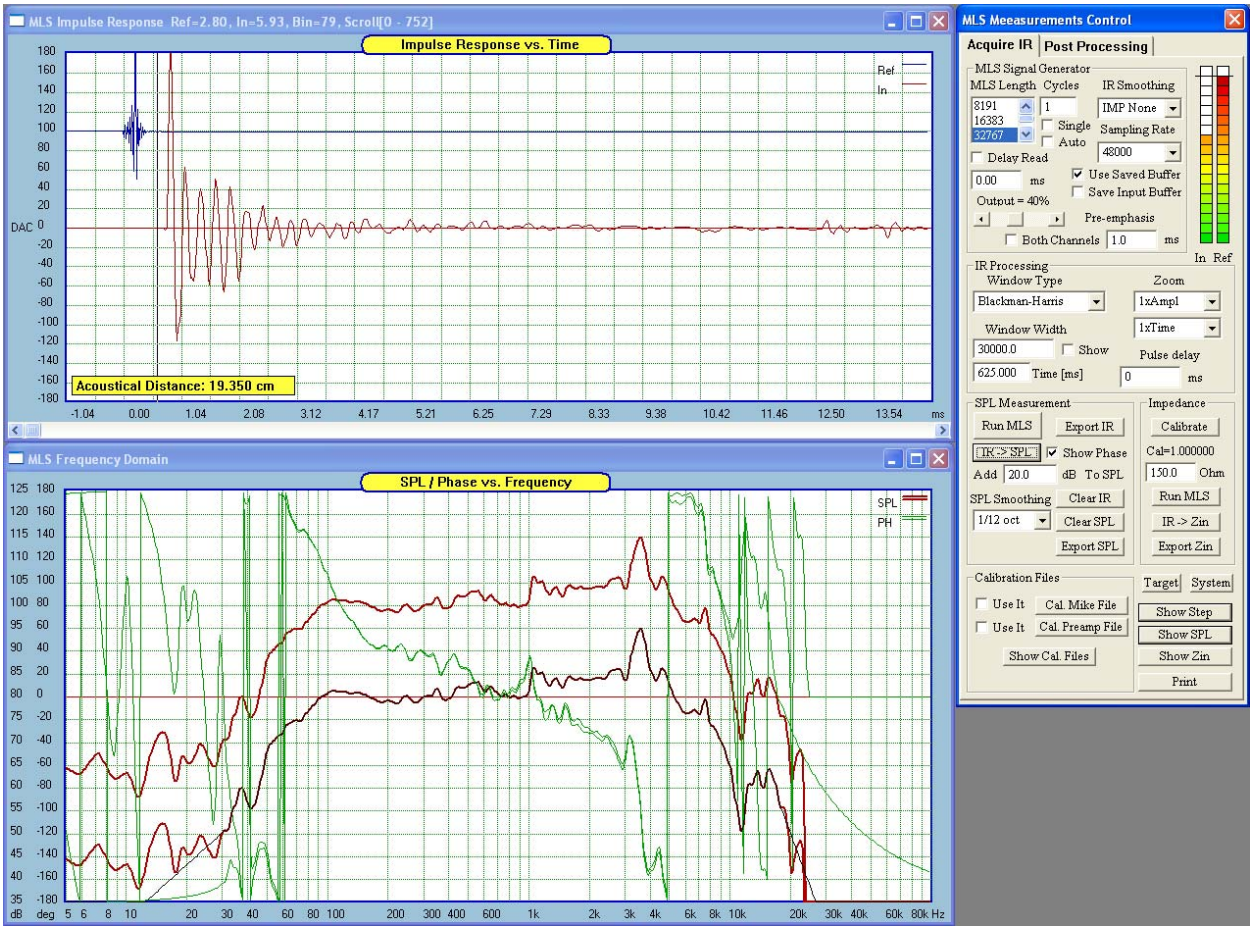


Figure 16.42. 20dB was added to the example curve.

Turning OFF Microphone And Pre-Amplifier Calibration Files

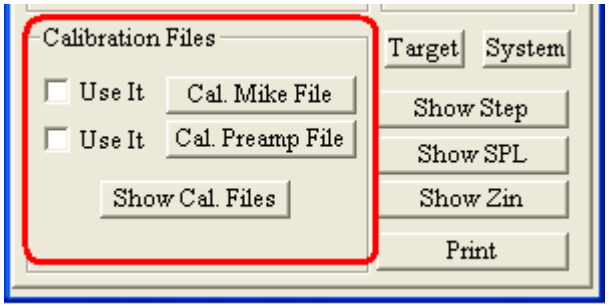


Figure 16.43. Turning OFF/ON mike and pre-amp calibration files.

The MLS and analog screens can be easily used with or without the microphone/pre-amp calibration files. Switching function is accomplished by checking/un-checking the box left to the "Cal. Mike File" and "Cal. Preamp File" buttons.

Adjusting Vertical Impedance Scale

To adjust impedance scale in MLS and Analog screens please use "**Vertical Resolution Control**" dialogue box, available from "**Enclosure Calculators**" menu.

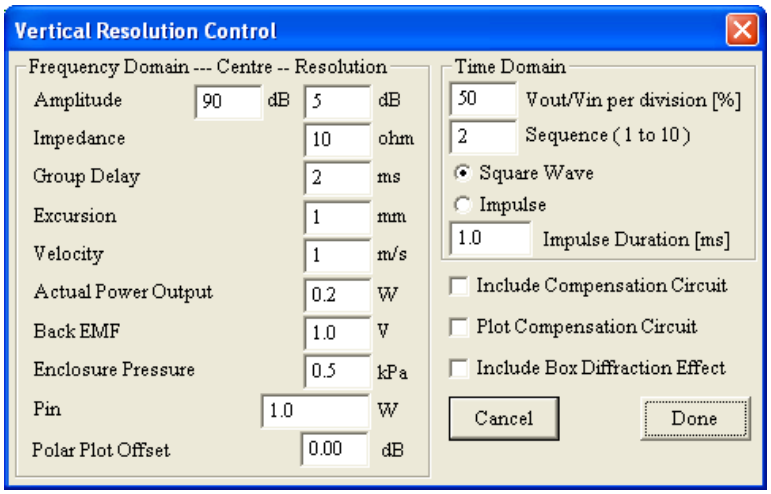


Figure 16.44 "Vertical Resolution Control" dialogue box – controls MLS impedance scale.

Measuring R, L, C Components

The RLC measurement system is intended to provide you with evaluating components, that are typically used **in Hi-If passive crossover networks**. This function is provided by the "RLC" TAB in the Easy Lab section.

Sound Easy determines the reactive components by measuring the impedance over a user-defined frequency range and then curve-fits simple component model to the measured data. Using broad frequency range rather than single frequency ensures more accurate results. Impedance measurements are performed using MLS-type of signals, as this method is quite fast. All MLS-related signal generating and capturing controls are on the **MLS TAB,** and the process of generating and capturing impedance-related measurements has been described before. Intuitively, one would select the measurement frequency range in accordance with the designated frequency range of operation of the component. For instance, 3.0mH coil could be used in low-pass filter below 1-2kHz. Therefore, you would set the screen resolution to 5Hz – 10kHz to obtain good representation of the coil's reactance. Inversely, a 0.47uF capacitor could be used in higher frequency range of filters, so more prudent choice of the frequency range would be 200Hz – 50kHz and sampling rate of 96kHz.

Component's models: $\mathbf{Xc = 1/(2\pi fC)}$ **and** $\mathbf{Xl = 2\pi fL}$

Output impedance of the soundcard is typically in the order of kilo-ohms, which does not match the need for providing higher drive for low-impedance components such as small inductors. We would therefore highly recommend, that you use your trustworthy power amplifier, as you did for other impedance measurements. The recommended test setup is shown on Figure 16.45. Please remember, that you need to perform calibration procedure as described in "CALIBRATION FOR IMPEDANCE MEASUREMENT" section before.

The value of the Reference Resistor (~400ohm on the schematic) can be varied depending on the component value. For instance, small capacitors will measure more accurately with higher resistors and vice-versa.

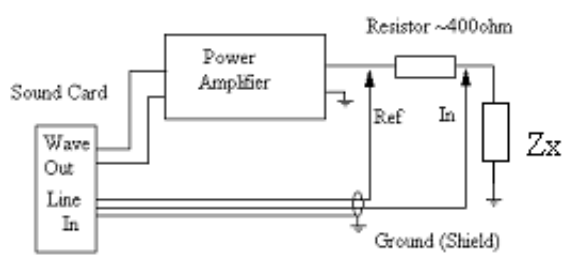


Figure 16.45. Test setup for measuring **Zx**

Typical measurement sequence looks as follows:

1. Connect the test setup as for calibration
2. Enter the Reference resistor value and perform calibration.
3. Generate Impedance Curve
   Select MLS length – 131071 recommended, sampling rate – 48K or 96K
   Set Volume to obtain full LED indication right up to the "Overload" level
   Select Window – Blackman-Harris, 130000 recommended
   Select Smoothing – 1/12 oct
   Press "**Run MLS**" in the Impedqnce MLS section – this will capture impulse response of the component under test.
   Press "**IR -> Zin**" button – this will plot the component's reactance.

Now is a good time to re-evaluate if the Reference Resistor and the frequency range were selected sensibly. You should aim at the plots looking like on 16.46 to 16.50 depending on the component type. Once you are happy with the reactance curve, representing the component value over broad frequency range, you are ready to "curve-fit" the built-in component's model to the measured data. There are three buttons provided in the **"RLC" TAB**.

1. Press the **"Find R" -** button to activate resistor fitting function
2. Press the **"Find R+L"** button to activate inductor fitting function
3. Press the **"Find C"** button to activate capacitor fitting function

Please focus now on Figure 16.49 (measurements of 3.0mH inductor). It is observable, that screen frequency range was selected to show a **flat reactance section from 5Hz to 50Hz**. This was done purposefully, as the program is able to determine the DC resistance from the flat section of the curve, located towards the left side of the screen.

Resistors

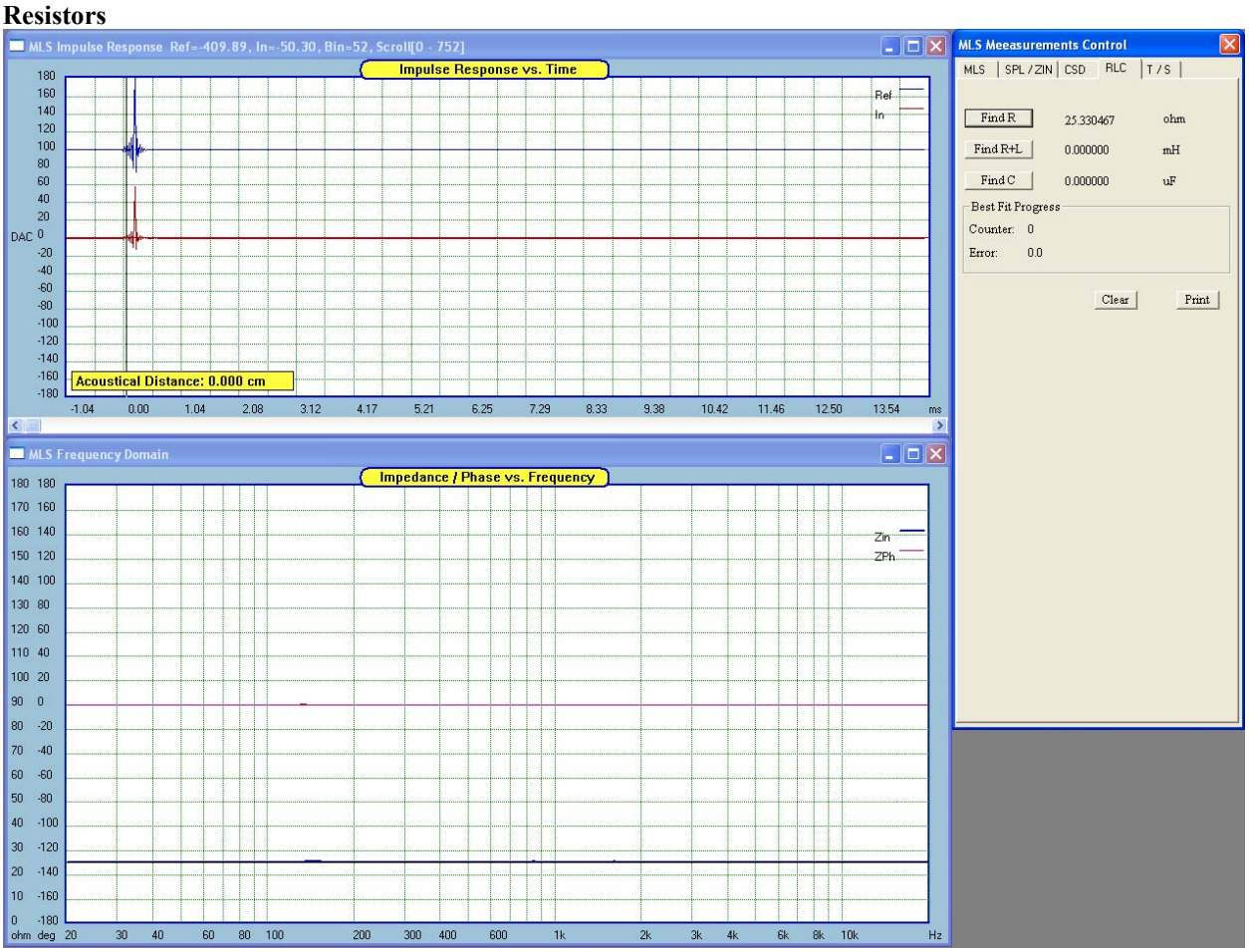


Figure 16.46. R = 25.0 ohm resistor

Capacitors

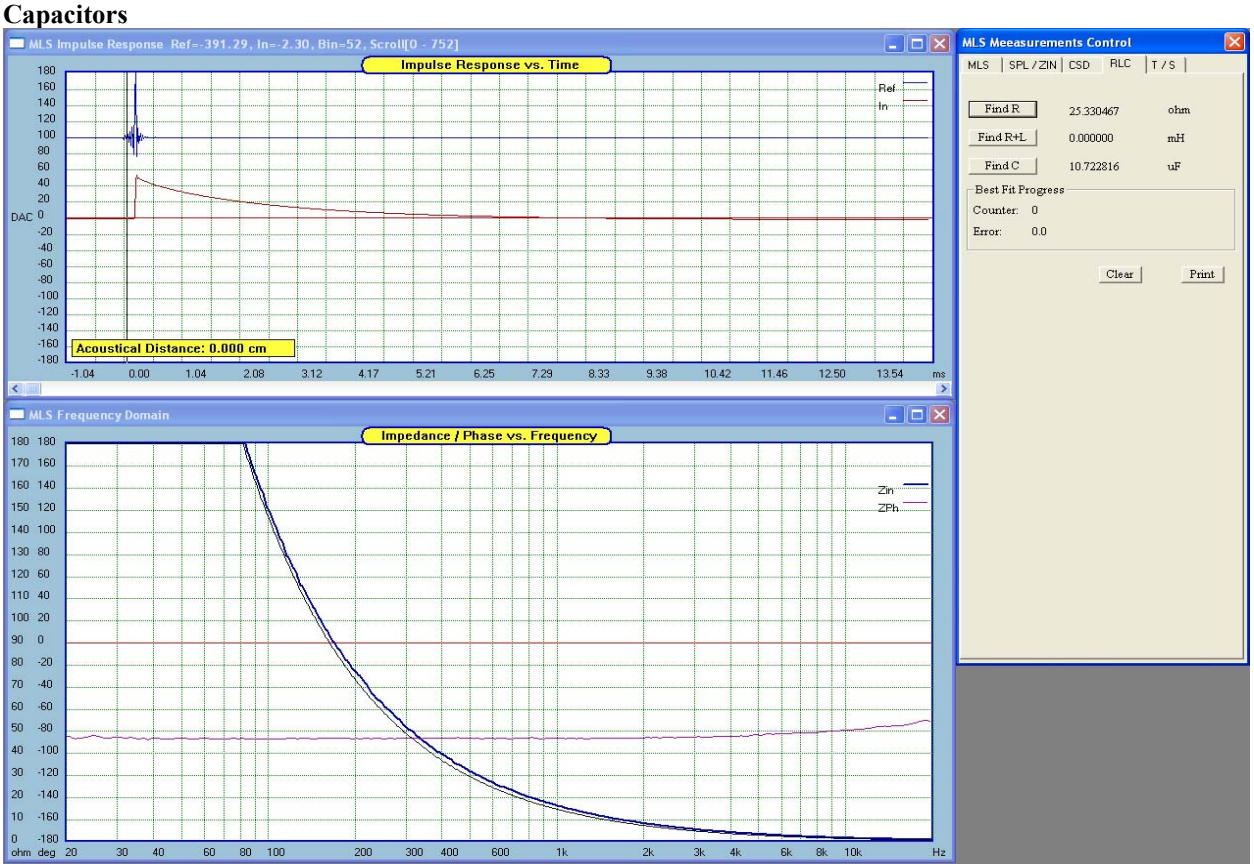


Figure 16.47. **Label: C = 10.0uF, 5%, Measured C = 10.72uF**

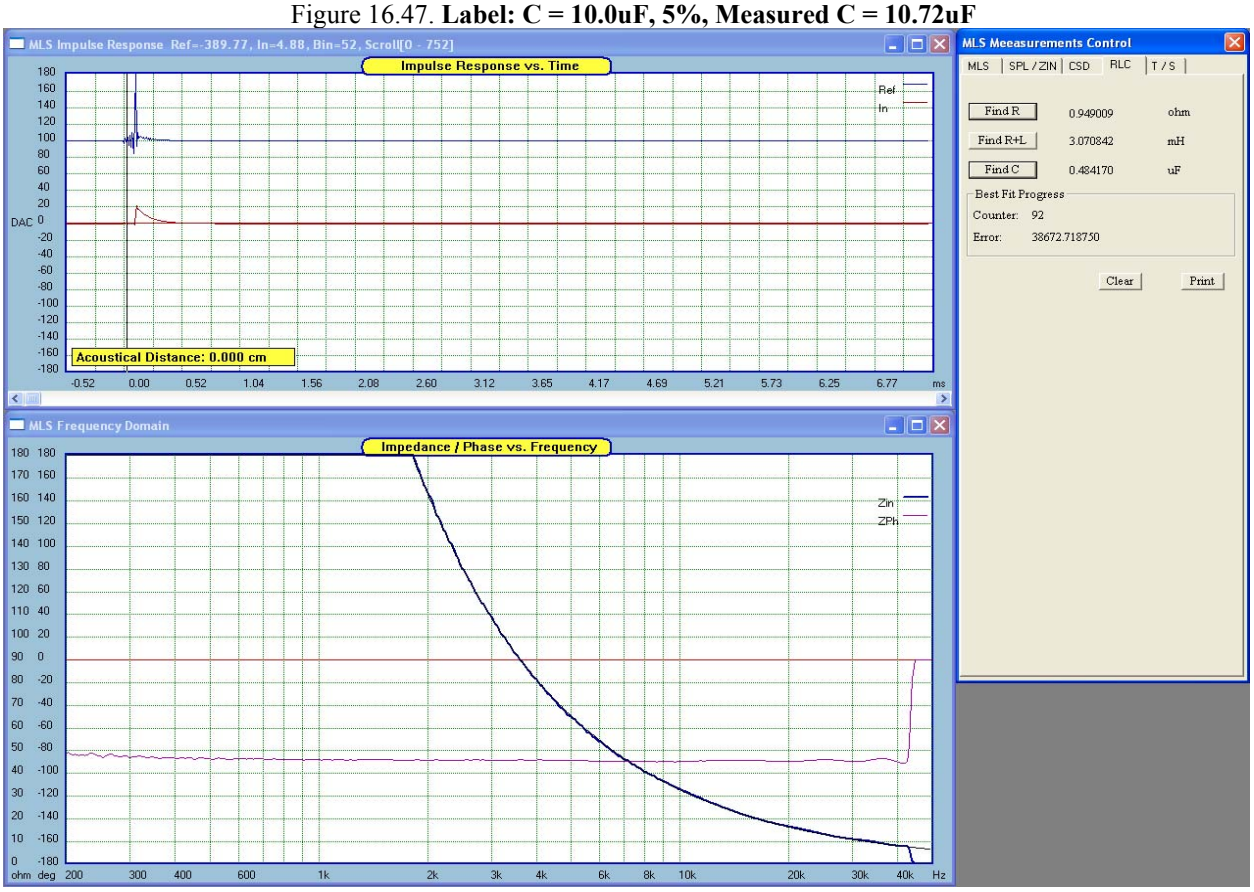


Figure 16.48. **Label: C = 0.470uF, 5%, Measured C = 0.484uF**

Inductors

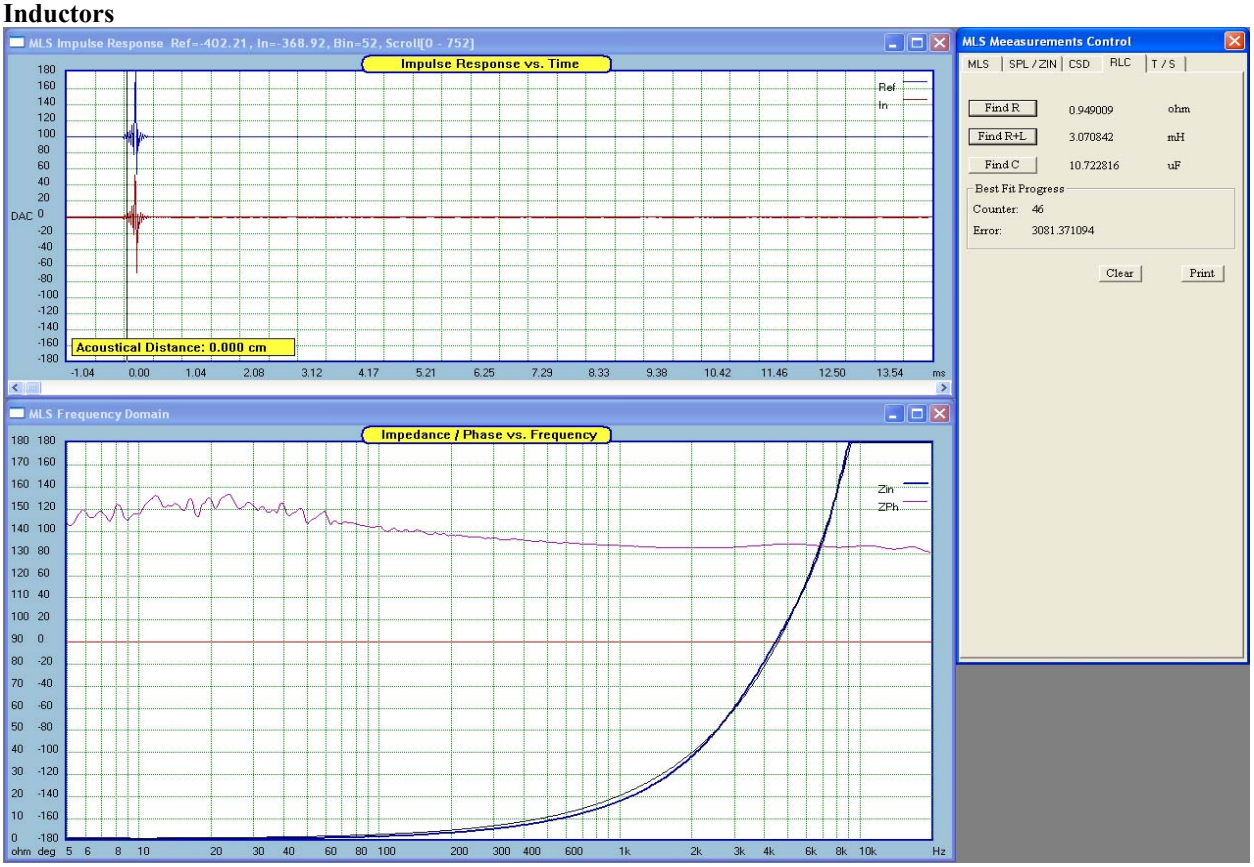


Figure 16.49. **Label: L = 3.0mH, RDC=1.0ohm, Measured L = 3.07mH, RDC= 0.94ohm**

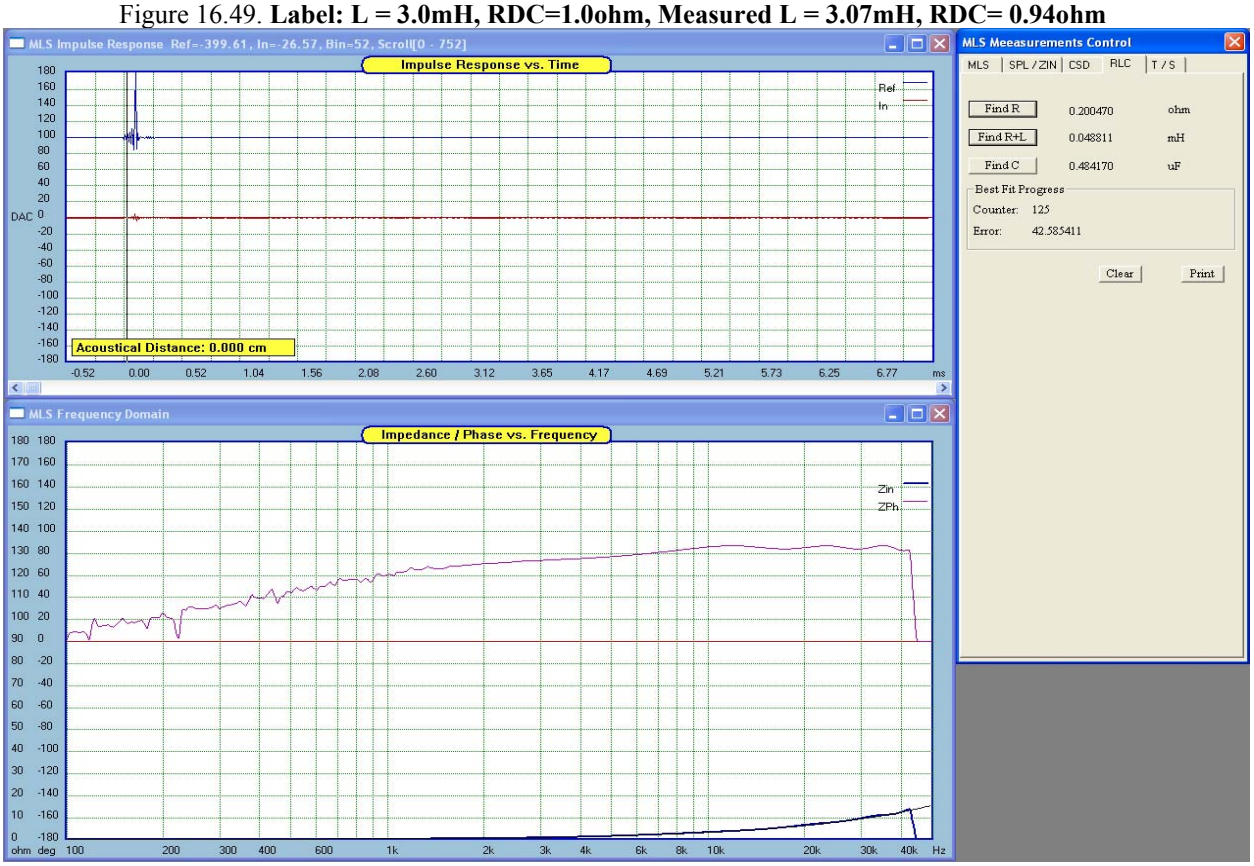


Figure 16.50. **Label: L = 0.05mH, RDC=0.10ohm, Measured L = 0.04881mH, RDC= 0.200ohm**

Single MLS Channel Mode

If you need to determine a frequency response of one of your correction circuits in your amplifier, you may be interested in the "Single MLS" mode of operation of the MLS screen. An example of the usability of this mode would be the frequency response measurements performed on parametric, shelving or other EQ circuits.

In a normal MLS measurement mode, both measurement inputs of the soundcard (Ref +In) are connected somewhere to the output of the audio amplifier. Therefore, no matter what are the internal parametric settings on your amplifier, the "Ref" probe will receive already equalized and corrected signal.

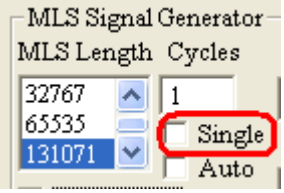


Figure 16.51 Single Mode

The "Single MLS" mode will help in this situation. The concept here is to obtain a reference SPL of your amplifier with flat EQ settings. This signal will be picked by the "Ref" probe in normal MLS mode of operation. Then you would switch the MLS system to "Single MLS" mode by checking the box. Now, the measurements are ONLY taken via the "In" probe and referenced to the previously captured "Ref" signal. As you can see, the "Ref" and In" are now decoupled. Here is the typical usage step-bu-step:

1. Run normal, dual-channel MLS measurement with "Single MLS" checked OFF. During this run, the "Ref" impulse is saved in a buffer.
2. Check the "Single MLS" box to indicate, that the next measurement will be using Impulse Response from "In" probe and the last saved "Ref" IR for reference.
3. Run FFT to convert IR -> SPL.
4. "Clear Imp" button re-sets the "Ref" input buffer to a default IR, which is close to the loop-back IR.

There one thing worth mentioning here. When you control your soundcard using a computer program, the PC will signal the current status of the soundcard using Interrupt Requests (IRQs). These are time-dependent events, and it may happen, that subsequent IRQ response from your CPU will be delayed by one sample in comparison to the previous measurements. This simply means, that your stored "Ref" IR will not peak at the same distance to subsequent "In" measurements, and as a consequence, a phase shift occurs between the stored "Ref" and current "In" IRs. If your PC happens to create this event occasionally, there is nothing much you can do about it, except for trying to re-capture the "Ref" IR and hoping, that next "In" impulses will be more stable. Obviously, dual-probe measurements are always correct, because you "In" probe is always referenced to the current "Ref" IR.

Auto Mode

Another very handy function of the MLS measurement screen is the "Auto Mode". Selecting this mode switches the MSL system into continuous measurement/FFT operation. The process runs as follows:

1. MLS system generates the measurement pulse.
2. DUT response pulse is collected in dual-channel measurement mode.
3. Collected IR if converted to SPL using FFT.
4. SPL is displayed.
5. If the "Auto" box is still checked, the process goes to Step 1 to continue the cycle.

Setting up your MLS system to the "Auto Mode" enables you to have both hands free to make your crossover adjustments. You could even position your PC screen close to where you have your crossover hardware and just let the MSL screen run in "Auto Mode". Here is how to run the mode:

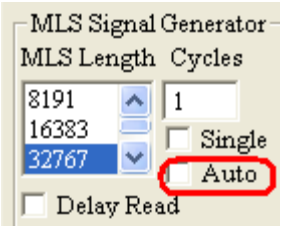


Figure 16.52 Auto Mode

1. Set ALL necessary MLS parameters first.
2. Uncheck "Single" box to indicate, that dual-channel option will be used.
3. Set number of "Cycles" to 1.
4. Check "Auto" box.
5. Press "Run MLS" button. The program will run IR/FFT cycles automatically.
6. To stop the process, **uncheck the "Auto" box**.

Cyclic Mode

In this mode, EasyLab SPL and Zin sections will generate 1 – 10 MLS impulses and will average the results for FFT calculations. This mode promises an improvement in noise immunity.

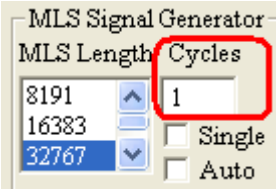


Figure 16.53 Cyclic Mode

Activating this mode is very simple:

1. Enter the desired number of MLS impulses in the "Cycles" data field.
   You must select a number from 1 to 10.
2. Press "Run MLS" button.

Export Impulse Response

If you need to make use of the information embedded in the collected Impulse response, you can export the IR into an ASCII file. The file can be read by any text editor, EXCEL and so on. The file has the following format:

Figure 16.54 Export IR button

XX.XXXXXX YY.YYYYYY

X = time step
Y = IR discrete values.

To export the IR simply press the “Export IR” button. A standard dialogue box will let you save your file.

"Delay Read" MLS Measurement mode.

Allows you to make delayed MLS measurements referenced to stored “Ref” channel.

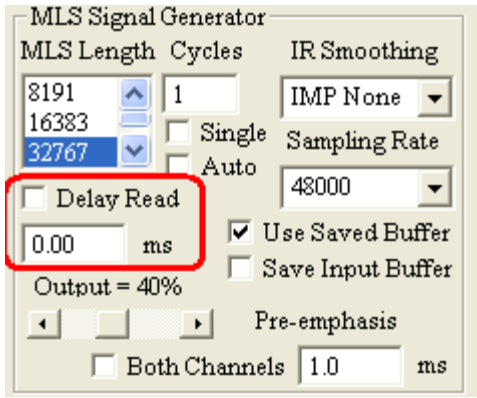


Figure 16.55 Delay Read function

In some measurements, you may experience a significant time delay introduced by the system under test. Situation like this would develop when measuring a loudspeaker at larger distances (1-5ms), when measuring room transfer function (3-20ms) or perhaps when measuring a transfer function of a DSP processor with longer processing delay (1ms -1sec).

The **“Delay Read”** function allows you to define the length of time, that the MLS system will wait before processing the incoming system under test response. The delay is defined in milliseconds and is recalculated to sampling intervals. For instance, a delay of 1.0 ms is recalculated to be 48 sampling intervals for 48kHz sampling. The Delay Read function is applicable to “single shot” MLS test signals.

Please avoid entering the delay time longer than the actual delay introduced by the system under test. The Impulse Response of the system under test still needs to arrive after (or at best, at the same time as) the reference Impulse Response.

EasyLab settingas stored to Preferences file.

Almost all EasyLab settings are saved in the Preferences file. This way, you can close and then re-open the program with the last-saved measurement settings automatically available.

“Scroll” function in Analogue Measurement System time domain window.

The **“Scroll”** function allows you to view the complete input buffer, by scrolling it’s sections into the screen view. It will be particularly useful in viewing the results of the **“Quick Sweep”**. Example below, shows the SPL curve of a low-Q notch filter, together with the section of the “In” buffer, where the notch is visible – well, the signal is actually very low at the notch point.

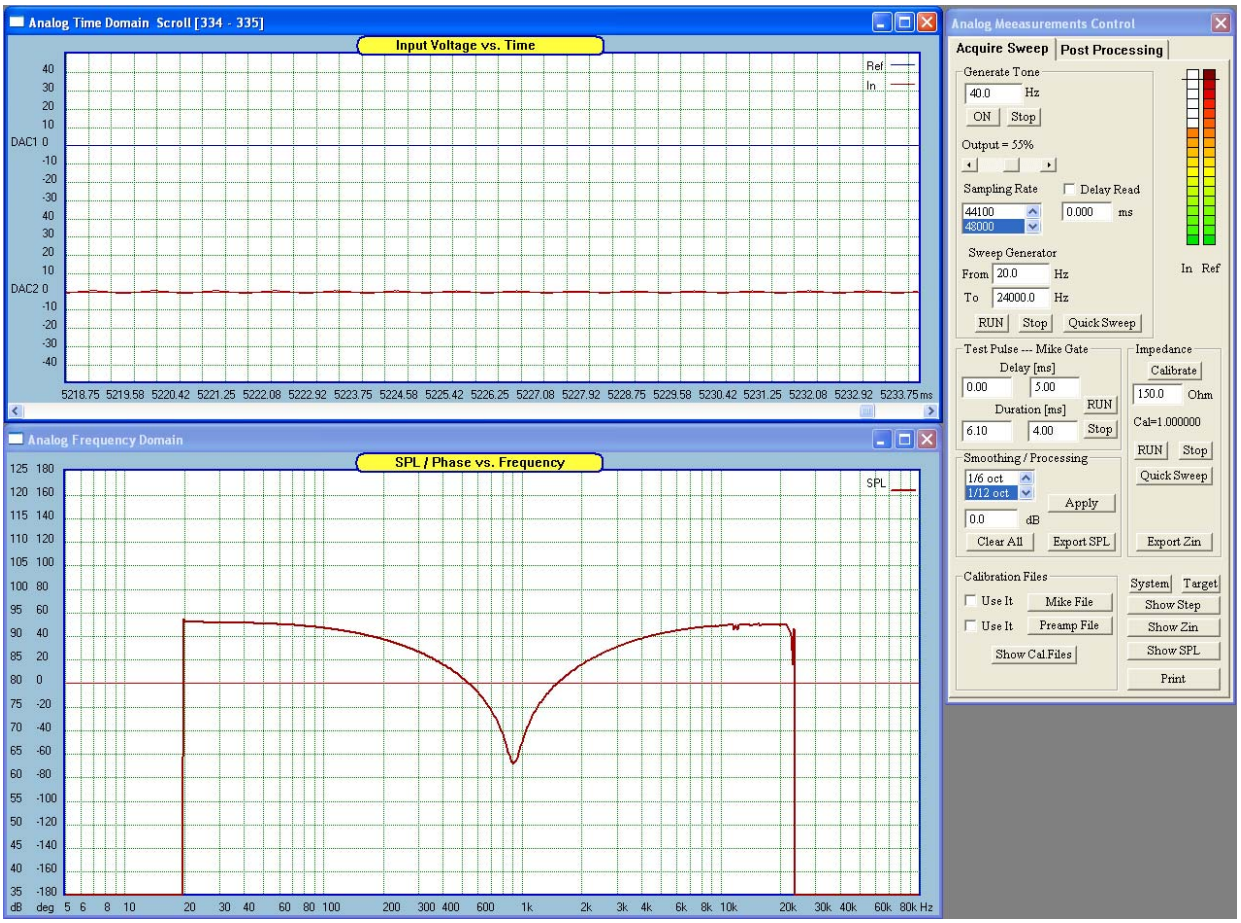


Figure 16.56. Scroll Input buffer function in Analogue Measurement System

"Quick Sweep" SPL function in Analogue Measurement System.

The **"Quick Sweep"** is a time-synchronized function. The decoding algorithm needs to know the precise moment of time, that the sweep signal commences. In order to facilitate this synchronization, one output of the sound card generates only a bi-polar reference spike. This spike is meant to be diverted directly to the "Ref" input of the sound card.

We recommend a direct connection between the output and the input of the sound card. The bi-polar spike could have been embedded into the actual sweep signal, but this would result in the spike being amplified and passed through the speaker. This could affect the speaker itself and also the frequency response of the speaker could eliminate the spike. Therefore a separate synch channel is created for the purpose of delivering the bi-polar marker undistorted to the sound card input.

As the test signal passes through the system, it may undergo a delay. For example, placing your microphone 1meter away from the AC of the driver will result in the time-of-flight delay of approximately 3ms. You will be able to see, that the sweep signal collected by the "In" probe, starts about 3ms past the bi-polar spike, displayed on the "Ref" channel. Leaving the buffer like this would result in significant SPL error towards higher frequencies , as the decoding algorithm assumes the start of the sweep to be at the "Ref" spike. The buffer needs to be manually shifted by the 3ms. This can be accomplished easily by the **"Delay Read"** function.

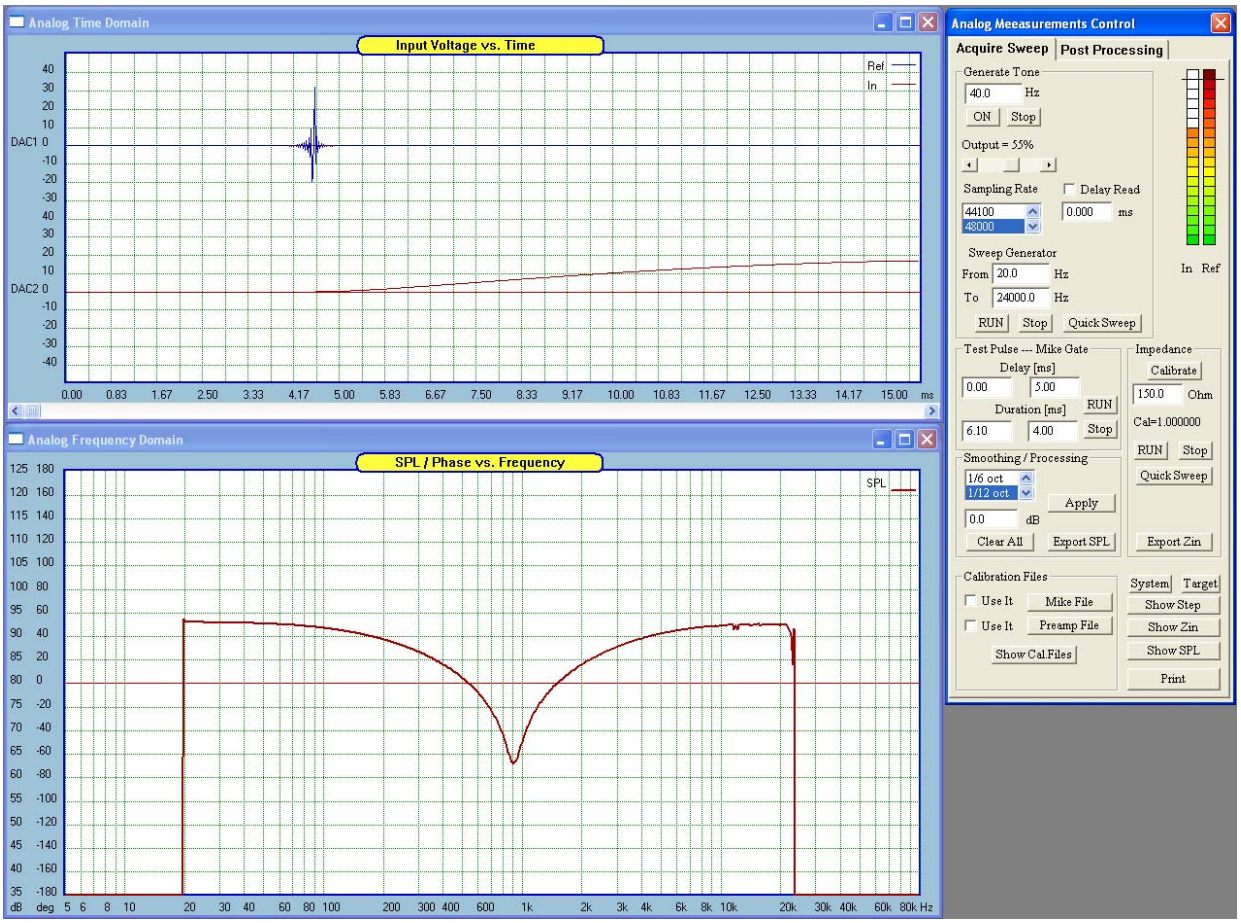


Figure 16.57. Quick Sweep employed to view a Notch Circuit.

The **"Quick Sweep"** SPL function can be very useful, when performing measurements in non-reflective environment and also for close-field SPL measurements, where the energy of the reflections can be negligible. The most recommended settings would be **Sampling: 48kHz, Sweep: 20Hz – 24kHz, screen frequency range 5Hz-100kHz .** The above settings would results in sweep time of 5-10seconds and allow you to assess the SPL nearly as fast as the MLS measurement does. The original SPL curve after smoothing and HBT would produce full complex driver transfer function.

It is observable, that sampling the low end of the audio spectrum would take most of the available buffer space. Therefore, setting sweep range below 20Hz is not recommended, as it would dramatically reduce the high-end of the sweep frequency range.

"Quick Sweep" Zin function in Analogue Measurement System.

The **"Quick Sweep"** function will be particularly useful when measuring input impedance, Zin. There is no time delay involved in the measurement, so it relatively simple process. However, there is one significant difference between SPL and Zin test signaling. The impedance, Zin measurement requires two inputs. Therefore, the bi-polar synch pulse is actually embedded into the test sweep signal.

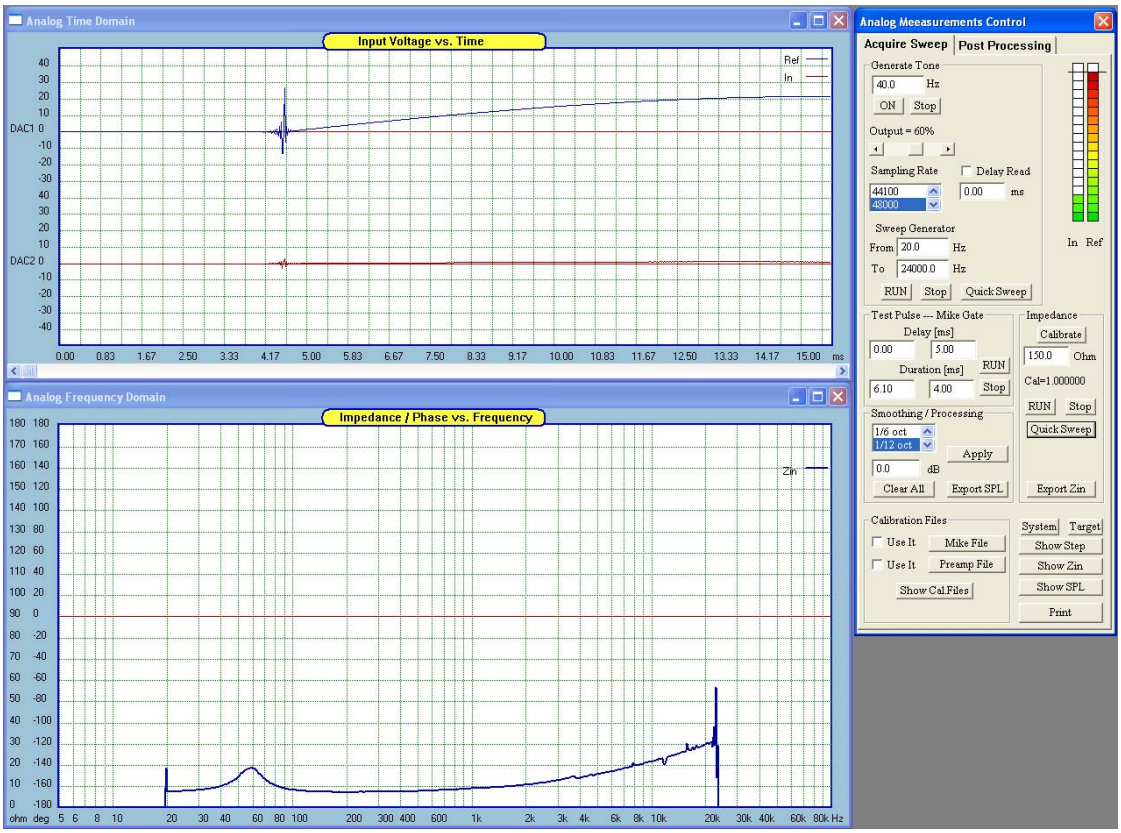


Figure 16.58. Quick Sweep employed to measure Zin of a loudspeaker.

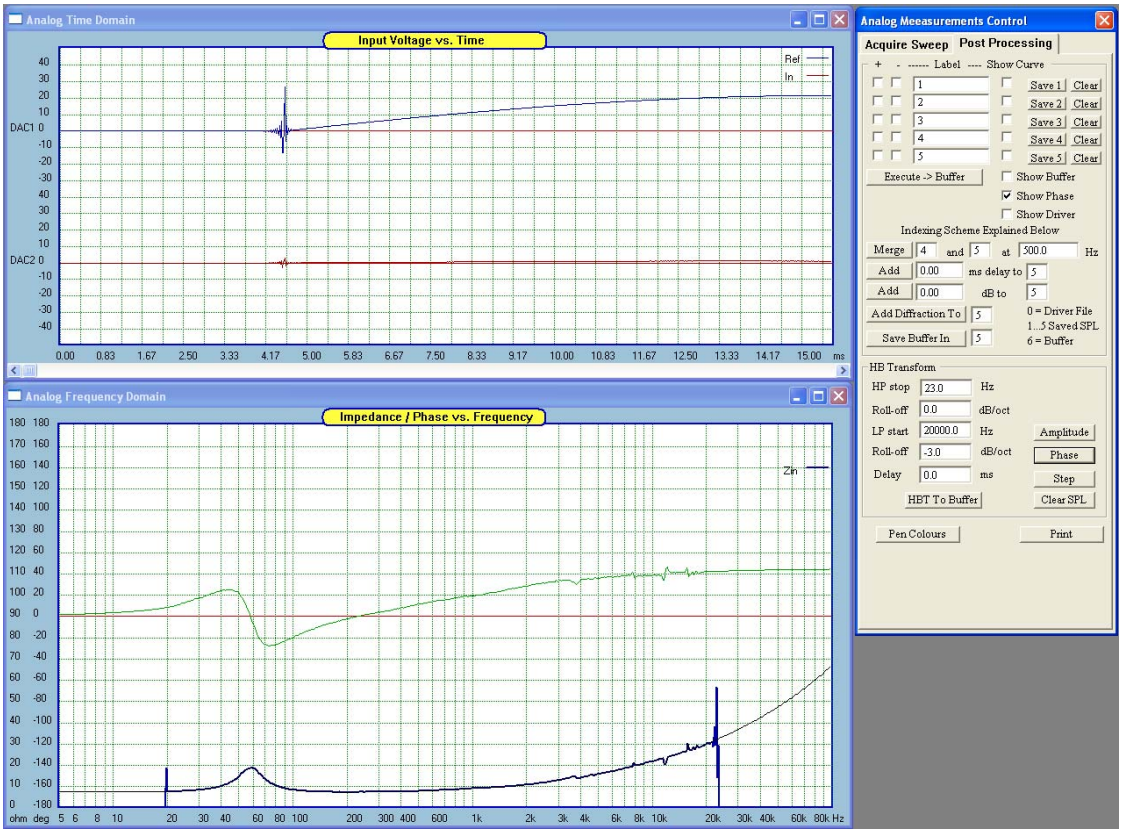


Figure 16.59. Quick Sweep combined with HBT delivers full complex input impedance Zin.

As for the SPL measurement, the original Zin curve after smoothing and HBT would produce full complex driver impedance function.

Smoothing of Impedance measurement in Analogue Measurement System.

Smoothing of the input impedance curve will be very useful before you perform the HBT on this curve to obtain complex input impedance set of data points. The resulting Zin magnitude and phase curves will be clean, smooth and presentable.

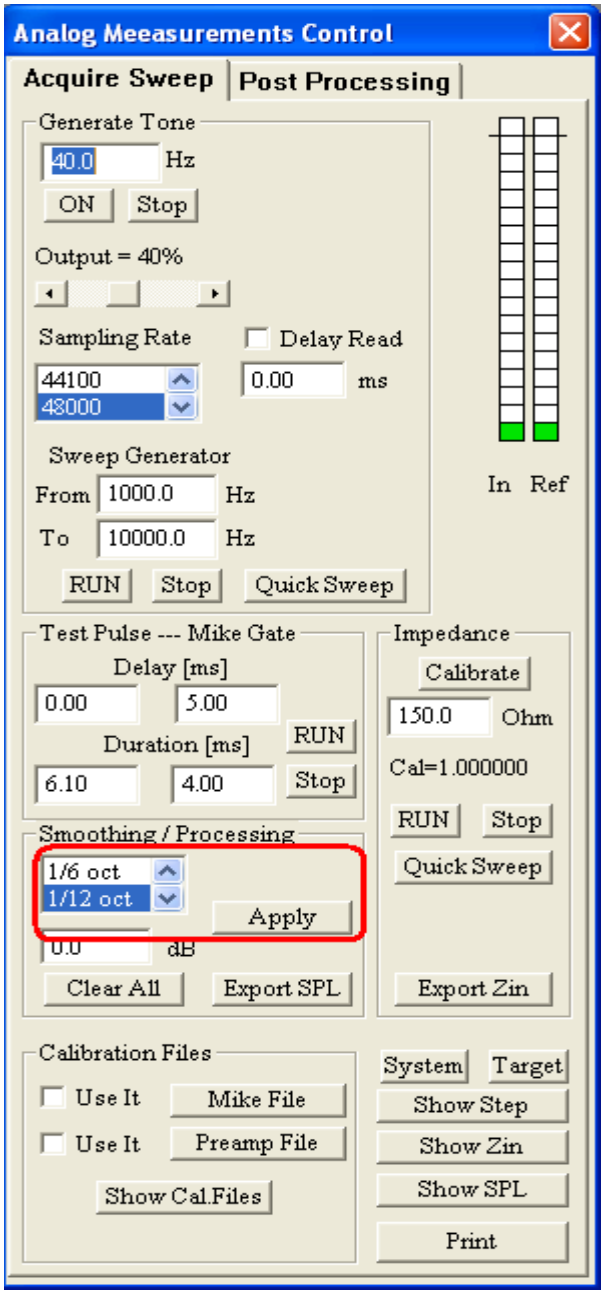


Figure 16.60. Smoothing parameters for Zin. Comes handy when using Quick Sweep.

RT60

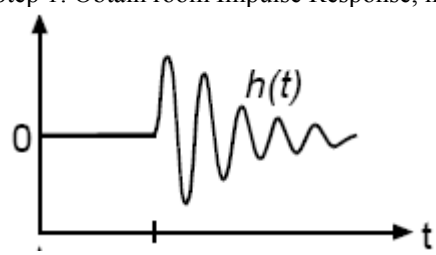
Reverberation Time (RT60) is recognized as the most important acoustical room parameter. It is defined as a time interval required for the sound energy to decay 60 dB after the excitation has stopped. The RT60 is estimated from the **Energy Decay Curve, L(t),** after the sound source is switched off. The energy decay curve is typically irregular and noisy curve, and some more mathematical processing needs to be done to enable a reliable calculations of the RT60.

The method of calculating the reverberation time (RT60) is due M. Schroeder, ("New Method of Measuring Reverberation Time". J. Acoust. Soc. Am., vol. 37 (1965) pp. 409-412). Schroeder has shown by statistical analysis that the room averaged energy decay r(t), can be obtained from the **backward integrated squared impulse response h(t)**. Here more complete formula for L(t):

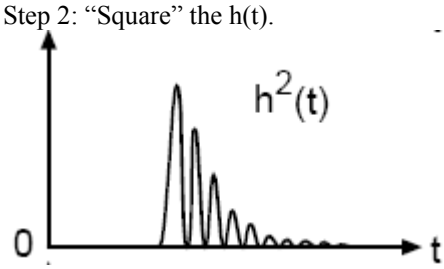
$$L(t) = 10\log(N\int_{t}^{\infty} h^2(\tau)d\tau)$$

Where: N is the noise power per unit bandwidth. The noise is considered to be a “stationary white noise”.

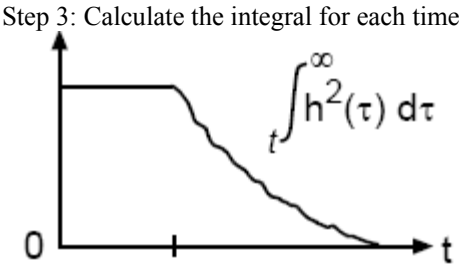
Step 1: Obtain room Impulse Response, h(t) using MLS technique.



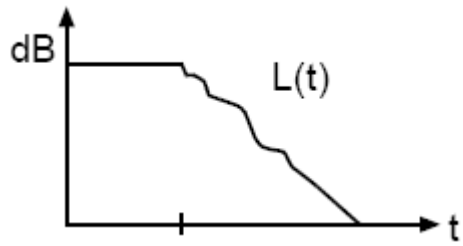
Step 2: “Square” the h(t).



Step 3: Calculate the integral for each time step from t = 0.



Step 4: Improve the RT curve by linear approximations (optional).



The **ISO3382** (ISO-3382, Acoustics – measurement of the reveberation time of rooms with reference to other acoustical parameters. 1997) also defines two additional parameters:

T20 - is the reverberation time determined from the average slope of the energy decay curve obtained from part of the decay curve between -5dB and -25dB.

T30 - is the reverberation time determined from the average slope of the energy decay curve obtained from part of the decay curve between -5dB and -35dB.

These can be useful, if the background noise level in the room makes it difficult to estimate RT60 from the L(t) curve.

Noise in RT60 measurements

Background noise is most likely to be present during your RT60 measurements and will manifest itself as a long, noisy tail after the RT60 has decayed to negligible values. In fact, many RT60 measurements do not even result in the decay curve shown over the full 60dB range. Simply because the noise level may be significantly higher than bottom end of this curve.

It would be beneficial to reduce or remove the noise from the measurements, so that a larger portion of the decay curve was available for estimation of RT60, thus making the estimations more reliable.

Two methods are being proposed:

1. **Truncation Method** – When selecting this method, the noise tail is simply chopped off after the selected truncation point. The essence of this method is to select the truncation point such a way, that the resulting RT60 curve is most linear, when measuring a single, decoupled room. The truncation point is selected by how much of the noise tail is to be removed from the calculations. It is simply the percentage of the Impulse Response (tail section) to be removed.

2. **Subtraction Method** – When selecting this method, the mean-squared value of the noise in the tail is being subtracted from the measurement before the Schroeder Backward Integration takes place.

The **Subtraction Method** was described by W.T Chu in "Comparison of reverberation measurements using Schroeder's impulse method and decay-curve averaging method" in J. Acoust. Soc. Am. 63(5), May 1978.

$$\left\langle s^2(t)\right\rangle = N\int_t^{\infty}[r(\tau)+n(\tau)]^2\,d\tau = N\int_t^{\infty}[r^2(\tau)+2r(\tau)n(\tau)+n^2(\tau)]d\tau$$

The $\mathbf{s^2(t)}$ is the ensemble average of the squared decaying sound pressure at a receiving point. The $\mathbf{r(\tau)}$ is the desired and measured signal, and $\mathbf{n(\tau)}$ is the unwanted noise. N is proportional to power spectral density.

As **n(τ)** be eider positive or negative, the second term will integrate to zero. The third term is the one, that needs attention. Mean-squared value of the noise tail after truncation point is calculated and subtracted from the **n^2(τ)**.

$$\bar{n}^2 = \frac{1}{L}\sum_{i=1}^{L} n_i(\tau) \qquad \text{therefore:}$$

$$\left\langle s^2(t)\right\rangle = N\int_t^{\infty}[r^2(\tau) + 2r(\tau)n(\tau) + (n^2(\tau) - \bar{n}^2)]d\tau$$

When using Subtraction Method for RT60 estimations, you should consider several factors affecting the process. Here is an example. Say, for instance that you have measured a medium-size room with **MLS of 262k** length and sampling rate of **48k.** This will result in **IR length = 262/48 = 5.45sec.** The Schroeder integral is calculated right to the end of the IR, so this is quite a long response to calculate. If your room has an RT60 of 300ms, for example, you will end up with rather short decay curve and a very long noise tail. The noise tail is **18 times longer than the useful recorded measurement**. The power of the noise tail will effectively compare with the power of the useful section of the recorded Impulse Response and will affect accuracy of the subtraction method. Simply speaking, too much power will be subtracted from the Schroeder curve.

In a situation like this, it is recommended, that you go back to MLS TAB and select MLS length as 32k. Do not repeat the measurement. This way, effective IR length taken into account for Schroeder integral RT60 calculations is 32k instead of 262k (or **IR length = 32/48 = 0.666sec)**. This way, the noise tail is only about **2 times longer then the useful recorded measurement**, so the balance is much better. Noise reduction methods greatly improve the estimation of RT60.

Making RT60 measurements

The process of measuring RT60 starts with collecting room Impulse Response using MLS signal, which is accomplished from the MLS TAB. All steps involved in setting up MLS generator, connecting your hardware are the same as for making standard SPL measurements, with some modifications. Here are some additional, simple recommendations for your measurement setup.

1. Microphone position must be at least 1 m from reflecting surfaces and not too close to any source position; the minimum distance from a source position can be calculated by the equation:

$$d_{\min} = 2\sqrt{\frac{V}{cT}[m]}$$

where: *V* is the room volume, *c* is the speed of sound, *T* is an estimate of the expected reverberation time.

2. The sound source should be preferably omni-directional.
3. The microphone should be omni-directional.

Also, one needs to be sensible about selecting the MLS length. This will affect the processing time for RT60 (Schroeder Backward Integration of 262k IR is a very long process) and also the averaged noise power in the tail. If you encounter this kind of issue, you can "Abort" and recover following the hints in the last paragraph.

From the definition of RT60, we understand, that "the clock starts ticking" <u>after the excitation signal stopped</u>. Therefore, you need to discard the direct signal contribution from the collected IR. To accomplish this, you need to select the MLS IR screen and move the cursor past the direct response pulse. Where exactly you place the cursor?. This will depend on your measurement setup, and obviously the distance between your measurement microphone and the speaker. Figure 16.61 below shows an example of removing the direct sound signature from the IR.

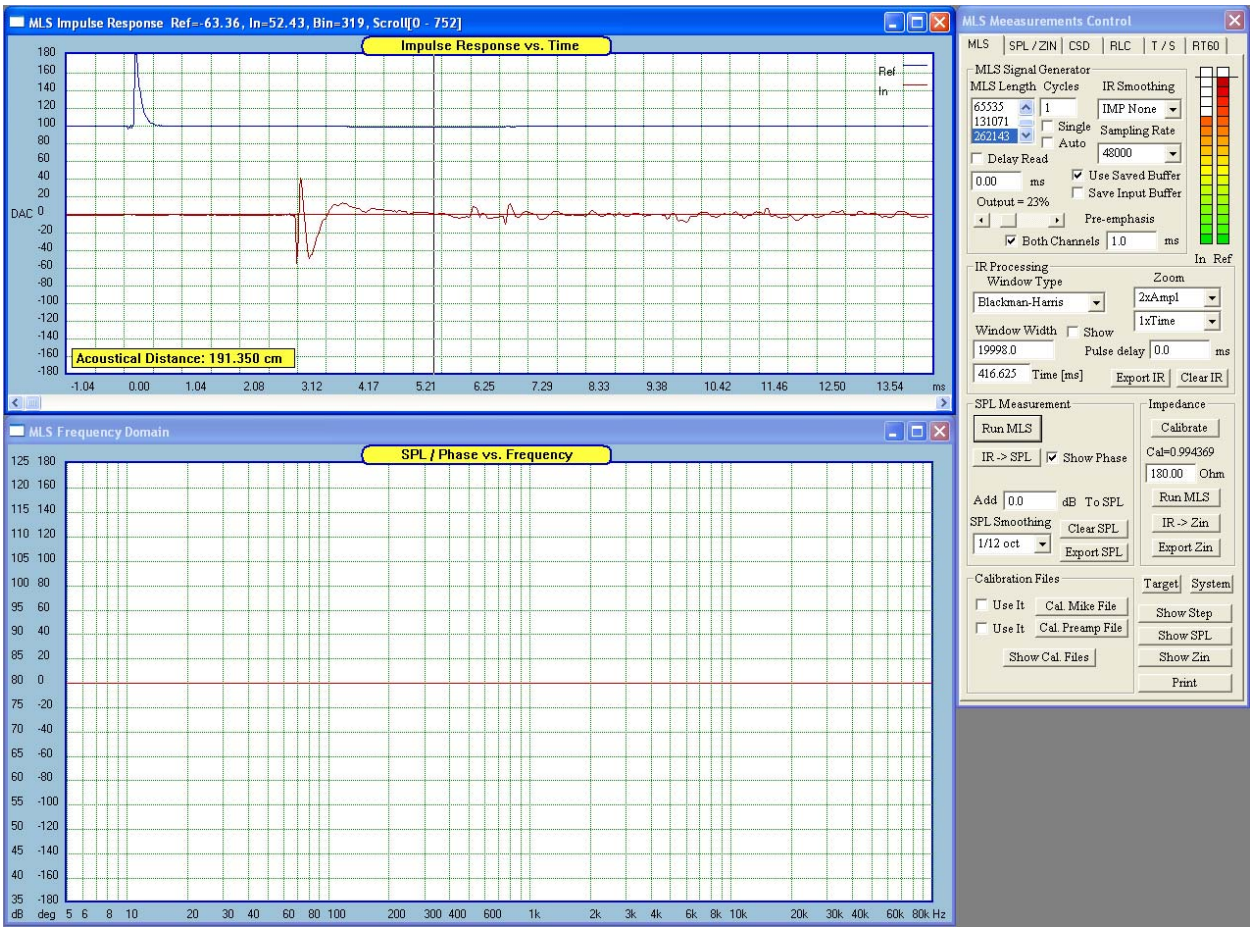
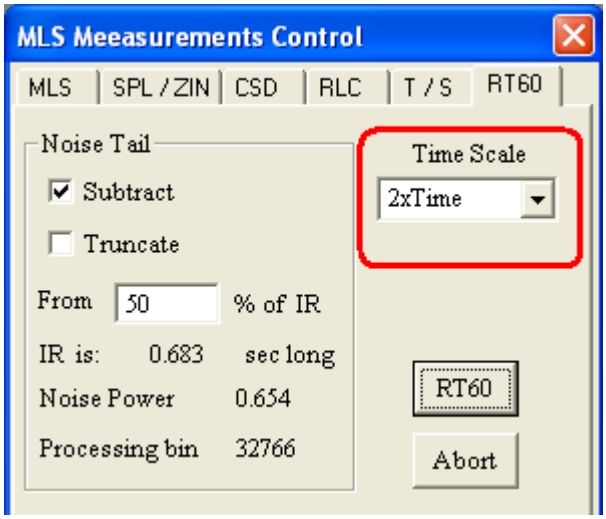


Figure 16.61 Removing the "direct sound" from collected IR.

When you open the RT60 display screen, you should consider selecting the time scale first. This is needed for proper display of long room IR, mapped onto the selected screen time scale. You can select time scale from **300ms to 16 * 300ms = 4.8 sec**. Obviously, you would select longer time scale for the expected longer RT60. Please note, that this is not a zoom function. The time scale is selected from the available list box.



Initially, you may select **Noise Tail + Truncate + 0% of IR settings**. Next, hit the **RT60** button and you should see that the program is processing measured IR bins and calculates RT60. Assuming everything went OK, you should see something similar to Figure 16.62 below.

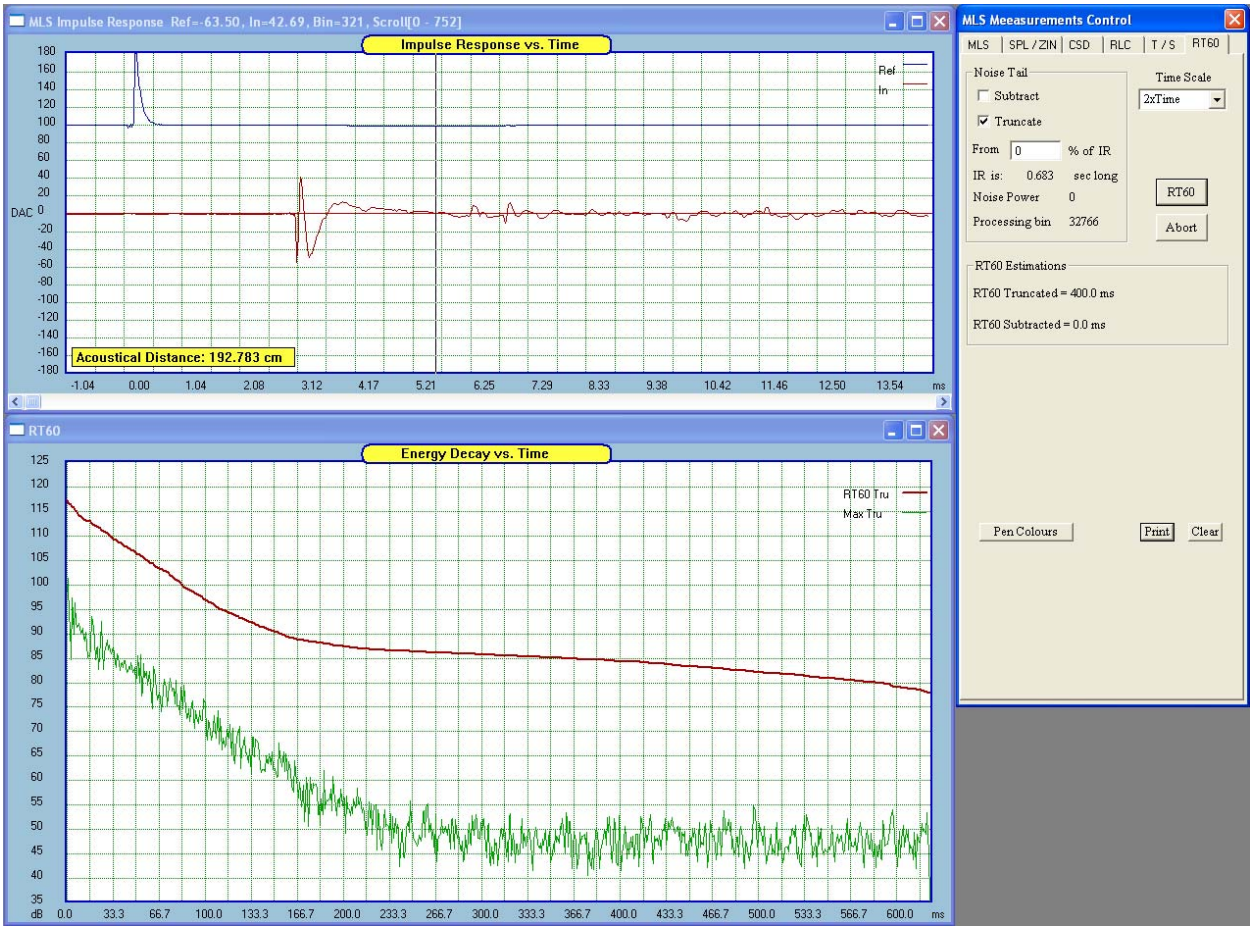


Figure 16.62. RT60 calculated without noise reduction techniques.

It is immediately observable, that the noise tail is contributing to the integration process, and rising the RT60 curve towards the right side of the plot. This prevents the RT60 curve to attain much better linearity.

In this test, the RT60 "Truncated" result is 400ms, and it is quite clear, that this result is not based on linear RT60 curve. In the next step, you can try to truncate say, 40% of the noise tail by entering "40" in the "From" data field in the "Noise Tail" group.

You can now try to calculate RT60 Truncated again, and visually inspect it's linearity. The result is presented on the Figure 16.63 below. Now, the RT60 Truncated curve looks much better. By doing this simple exercise, you can easily observe, that RT60 curve has almost lost the flatter section, therefore, you are obviously moving in the right direction.

In addition, if you try to calculate RT60 using "Subtraction" method (check the "Subtract" check box and press the RT60 button again), you will notice, that the RT60 (green) curve is quite linear up to 233ms. At this point, the curve is -45dB below the maximum value, and can be used for calculating **RT30** (time interval required for the sound energy to decay 30 dB after the excitation has stopped). This is exactly what the algorithm does. It will calculate RT30 and double it's value to get the RT60.

In the final step, RT60 Subtracted with 50% truncation (**RT60 = 305ms**) and RT60 Truncated with 65% truncation (**RT60 = 310ms**) were deemed to be the optimum. Please note, that RT60 values were 1.6% from each other – see Figure 16.64.

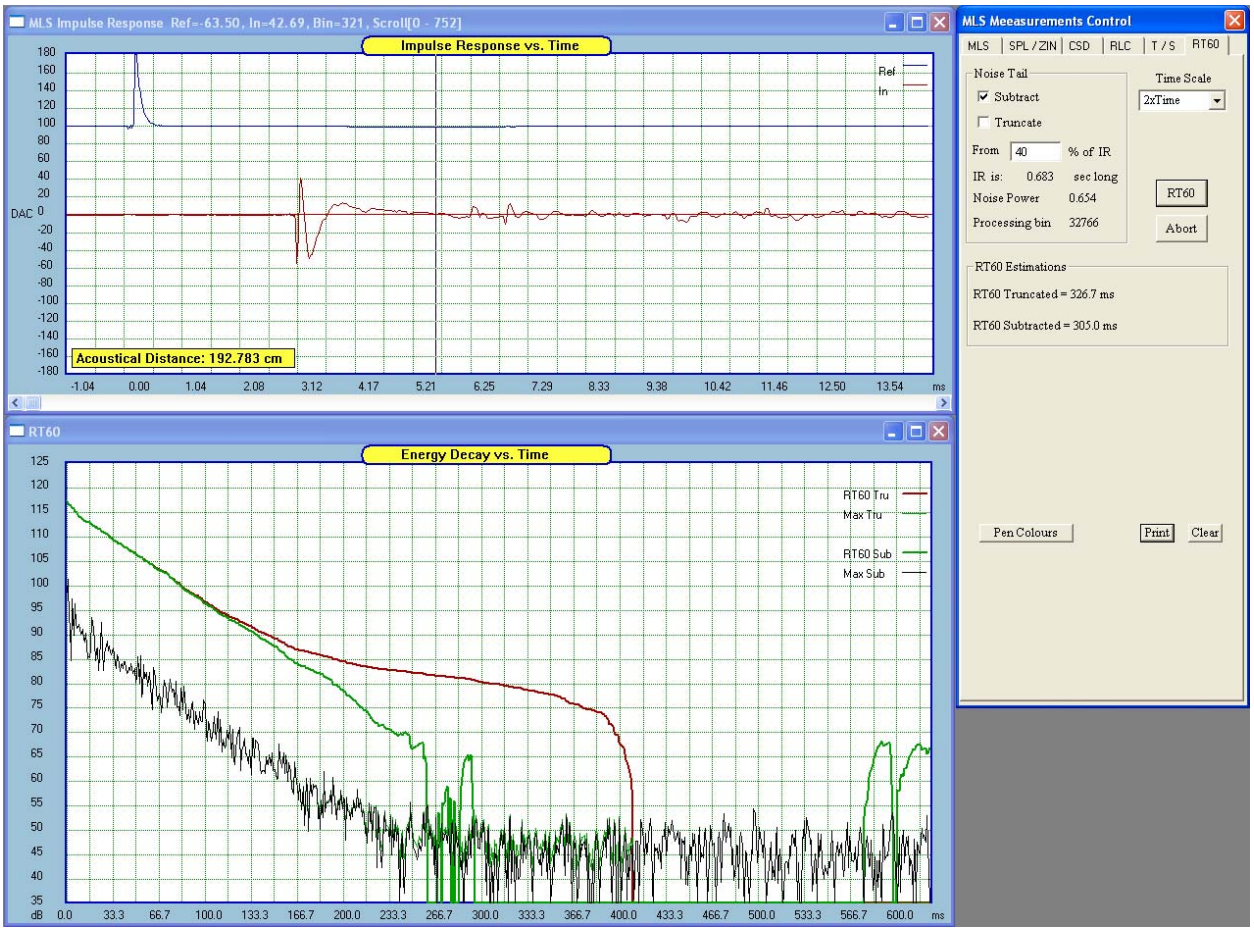


Figure 16.63. RT60 calculated with 40% of the noise tail truncated.

Description of the RT60 TAB Controls

1. **Noise Tail group**
   "Subtract" – check to calculate RT60 using subtraction method.
   "Truncate" – check to calculate RT60 using truncation method.
   "From" – enter the percentage of the tail you wish to be truncated.
   "IR is xxxxx sec long" – indication of how long is the collected room IR. Comes handy for estimating Time Scale setting.
   "Noise Power" – a figure of merit. If the noise tail is fairly flat, this figure will not change much for different truncation points.
   "Processing bin xxxxxx" – indication where you are with Schroeder integration process. You can "Abort" and change MLS length setting if the process drags forever.

2. **RT60 Estimations**
   "RT60 Truncated" = xxxxx ms – Result of Truncation method
   "RT60 Subtracted" = xxxxx ms – Result of Subtraction method

3. **Time Scale –** controls the time scale length
   "1x xTime" – Sets the time scale 0 - 300ms.
   "2x xTime" – Sets the time scale 0 - 600ms.
   "4x xTime" – Sets the time scale 0 - 1200ms.
   "8x xTime" – Sets the time scale 0 - 2400ms.
   "16 xTime" – Sets the time scale 0 – 4800ms.

4. **Abort** – Button to abort the RT60 process.
5. **Pen Colors** – Button to invoke pen colours dialogue.
6. **Print** – prints the screen.
7. **Clear** – Clears the screen

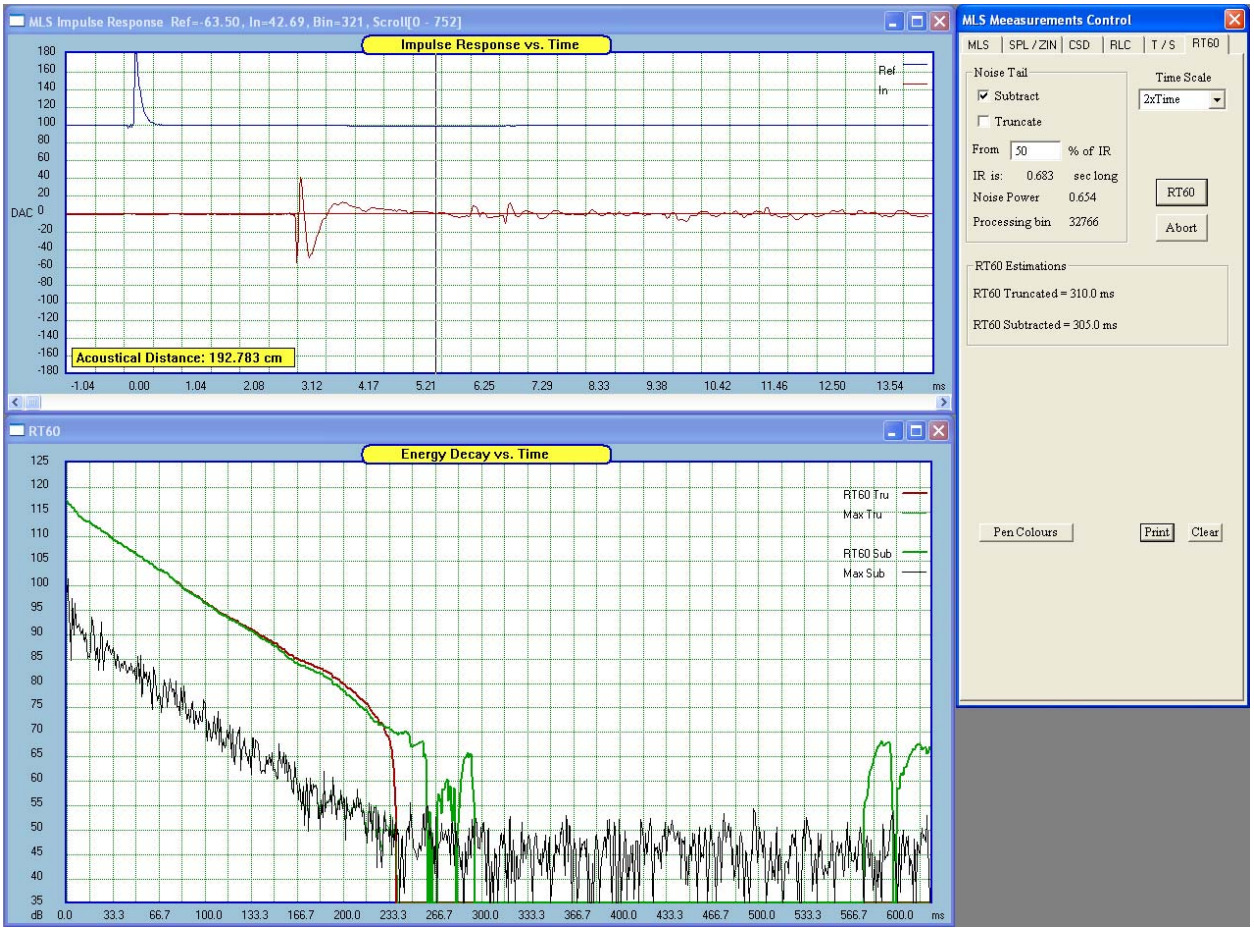


Figure 16.64 RT60 Subtracted with 50% truncation and RT60 Truncated with 65% truncation.

Chapter 17. Analog Nonlinear Measurements And Modeling

Non-linear Parameters Estimation using Volterra Series Expansion

The nonlinear parameters extraction process is quite involving. The general strategy of estimating non-linear distortion follows the works of A.J.M. Kaizer. and is based on **truncated Volterra series expansion**. In short, the strategy is as follows:

1. Measure linear (or small signal) parameters of the driver – $\mathbf{BL_0}$, $\mathbf{K_{0,\ and}}$ $\mathbf{Le_0}$.
2. Measure loudspeaker first (F1), second (F2) and third (F3) harmonics over the low frequency region, with cone excursion close to Xmax.
3. "Curve Fit" the measured harmonics to the built-in model by manipulating coefficients of the BL(x) , K(x) and Le(x) equations:

$$\mathbf{BL(x) = BL_0 + b_1{*}x + b_2{*}\ x^2}$$
$$\mathbf{K(x) \ = K_0 + k_1{*}x + k_2{*}\ x^2}$$
$$\mathbf{Le(x) \ = Le_0 + l_1{*}x + l_2{*}\ x^2}$$

The $\mathbf{BL_0}$, $\mathbf{K_{0,\ and}}$ $\mathbf{Le_0}$ are "force factor", stiffness (K = 1/Cms) and voice coil inductance known from the linear theory. The **b1,b1, k1,k2 and l1,l2** coefficients are manipulated during the curve fit process to obtain the best match with the measured harmonics over the desired frequency range. When the optimization process is finished, you should be able to obtain reasonable good fit between the measured and calculated data, indicating that calculated **b1,b1, k1,k2 and l1,l2** nonlinear coefficients represent real life distortions as close as possible – see the example Figure 17.1 below.

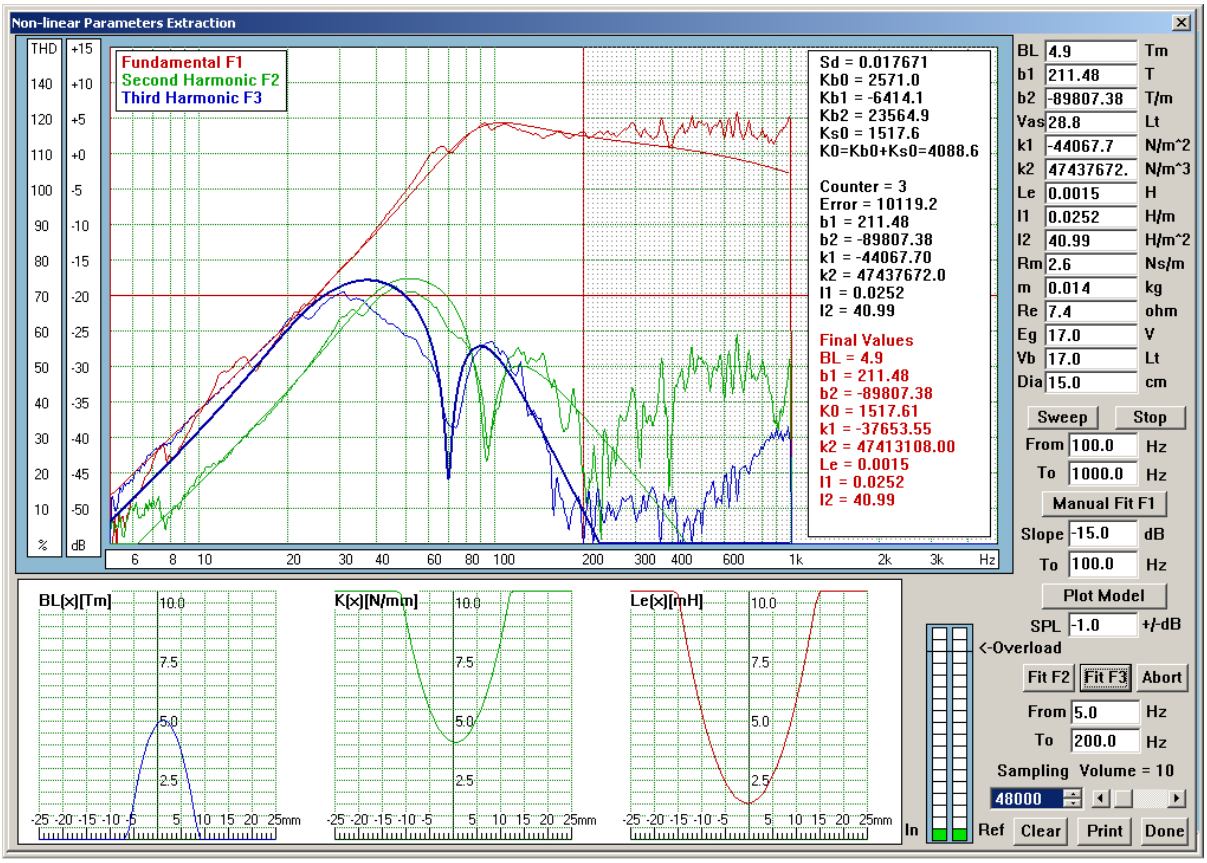


Figure 17.1 Estimation of non-linear coefficients for Volterra series expansion.

Comments On Performing The Measurements

There are many factors to consider before the actual measurements can be performed. Please remember, that the loudspeaker under test needs to be driven with a power level capable of producing high cone excursions (comparable with Xmax). The test case described in this manual uses an 8" driver, with power handling capabilities of 30W. The loudspeaker is mounted in an 17 liter enclosure and driven with approximately 20Watt of power.

1. All measurements are performed below 200Hz, so it is quite important to measure the loudspeaker correctly at such a low frequency range. The program will automatically generate, the sweep sine wave and track all three (fundamental, second harmonic and third harmonic) frequencies. However, you need to make sure, that reliable hardware and test environment for measuring frequencies down to 10Hz is provided.

   The measurement issues will be much reduced if you have an access to the anechoic chamber. The curves will be much smoother and will measure without the environment reflections and resonances. Basically, the issue to consider is the enclosure diffraction and overdriving the test microphone. The test microphone can be located at 1meter away from the loudspeaker. If the loudspeaker's efficiency is 85dB/W/m, the 20watt applied to the speaker will produce 98dB SPL. Most test microphones should be able to handle this pressure.

   If you do not have access the anechoic chamber, you may have to resort to the "close mike" techniques. The issue now is more complex. For instance, if you decide to keep the same power (20Watts) and bring the microphone closer to the speaker (say 12.5cm distance) the SPL pressure will produce 116dB at the microphone location. This is significantly more than discussed before and you need to make sure, that the microphone and the pre-amplifier can handle this conditions without introducing clipping distortions.

2. Voice coil inductance varies with frequency and the inductor is actually a "semi-inductor" type. The Volterra series expansion model assumes that the inductor is actually a "normal" inductance. Therefore, there will be a discrepancy between measured inductor value and the modeled inductor. The value of the inductor increases several times going towards low frequencies. On the example below, the inductance increase 6-fold from 20kHz down to 150Hz – see Figure 17.2 below (brown curve).

   This issue is quite critical and in our test example, the value of Le was increased from 0.8mH to 1.5mH, (2x) in order to account for the inductance variability below 200Hz.

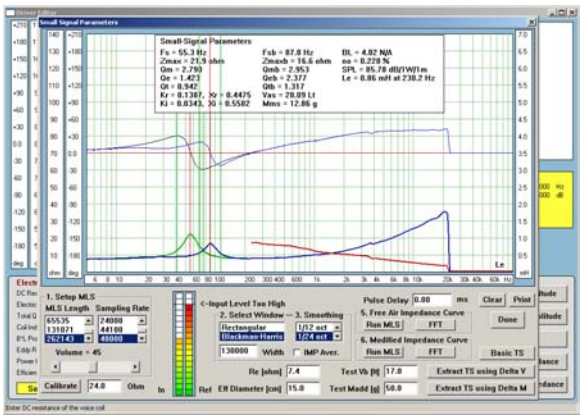


Figure 17.2 Measurement of linear parameters

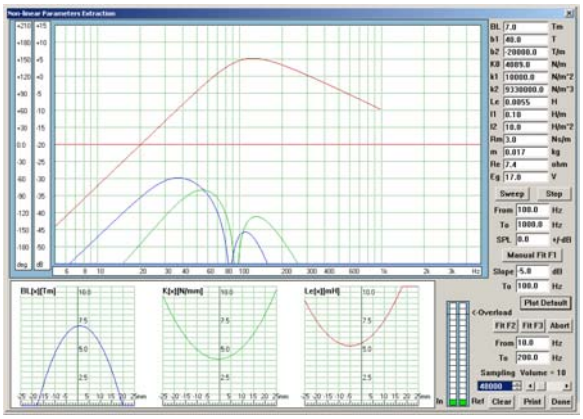


Figure 17.3 Default values for Volterra model.

3. Initial conditions for the "curve fitting" process are also important. The curve fitting mechanism is nothing more than looking for a minimum error between the measured data and the model. The mechanism will be therefore prone to typical "local-vs-global" minimum error issue. If you select unfortunate starting conditions for the optimizer, there is quite distinct possibility, that the optimizer will find "local" minimum, rather than "global" minimum. Starting conditions can be improved by inspecting the "default" curves plotted by pressing the "Plot default" button.

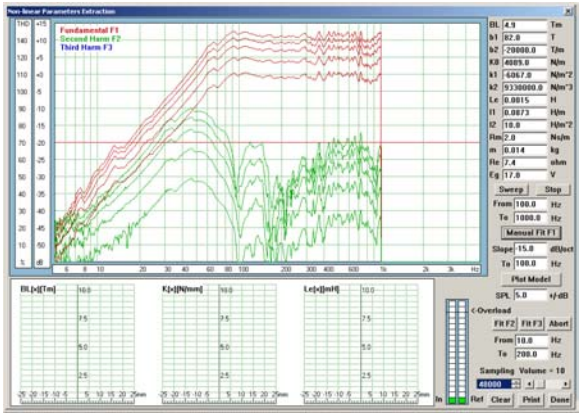


Figure 17.4 Second harmonics

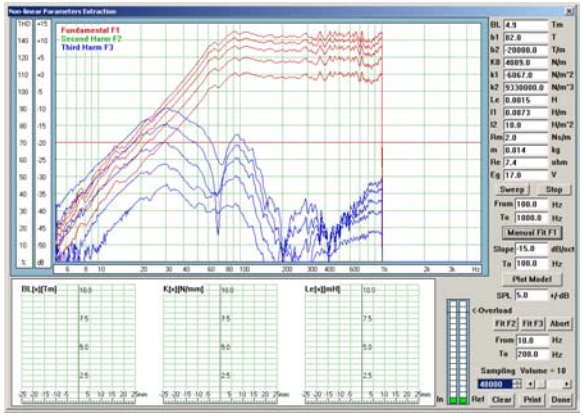


Figure 17.5 Third harmonics

4. The Volterra series expansion model is a truncated model, implemented for the purpose of evaluating the non-linear behavior of the loudspeaker. It is applicable to "weekly nonlinear" systems. In order to evaluate the required input voltage into the loudspeaker, we have plotted the **fundamental (red curve), second (green curve) and third (blue curve)** harmonics of our test loudspeaker for the following power levels top to bottom:

   34.0 Vpp……. 19 watts
   26.8 Vpp……. 11.8 watts
   20.0 Vpp……. 6.6 watts
   13.4 Vpp……. 3 watts
   7.8 Vpp……1 watt

The curves above 200Hz should be discarded, as the plots were taken in a very "bad" acoustic environment and room influence (modes) is clearly visible from the shape of the plots. However, the major characteristics of the second and third harmonics curves are clearly visible. The following can be observed:

The shape of the second harmonic curves does not change very much with the input power level. The shape of the second harmonic curve is tied up to a straight line and the second harmonic distortion alone, relates mostly to "offset centers" or "skewed" curves and these parameters do not change significantly with the cone excursion. The overall level of the second harmonic plots will move up or down, depending on the input power, however the shape of the curve remains quite similar.

The third harmonic curve behavior is somewhat different. The shape of the third harmonic curve is tied up in our model to the characteristics of a parabola. The shape of the third harmonic curve relates to how "wide" or "narrow" the parabola is, and the location where you are on the parabola curve. This ultimately changes the shape of the third harmonic curve as the cone excursion is changed.

The accuracy of our truncated model will therefore depend on input power level to the loudspeaker. We recommend, that you should use input voltages, that produce cone deflections between 60-100% of Xmax. From the examples above, it is easy to observe, that the third harmonic curve begins to stabilize its shape above 6.6 watts of input power on a small, 30 watt rated loudspeaker. In this case, a 19 watts would be recommended.

5. Loudspeaker small-signal parameters change depending on the "break-in" duration. It is recommended to run the loudspeaker with 50-60% of rated power for at least 1 hour prior making the small-signal measurements. This would loosen up and stabilize the suspension, making it ready for the much longer non-linear tests. The danger here is, that the non-linear test runs for several minutes and by itself, is a form of "breaking-in" loudspeaker. If you start with "cool" loudspeaker for small-signal tests and then proceed right away into the non-linear tests, you effectively test the same speaker under somewhat different suspension state (stiffness).

6. The measurement system/environment may introduce its own characteristics at such a low frequencies. An example is shown below, where the measured SPL of the fundamental frequency does not follow +12dB/oct curve, expected for the loudspeaker placed in a sealed box.

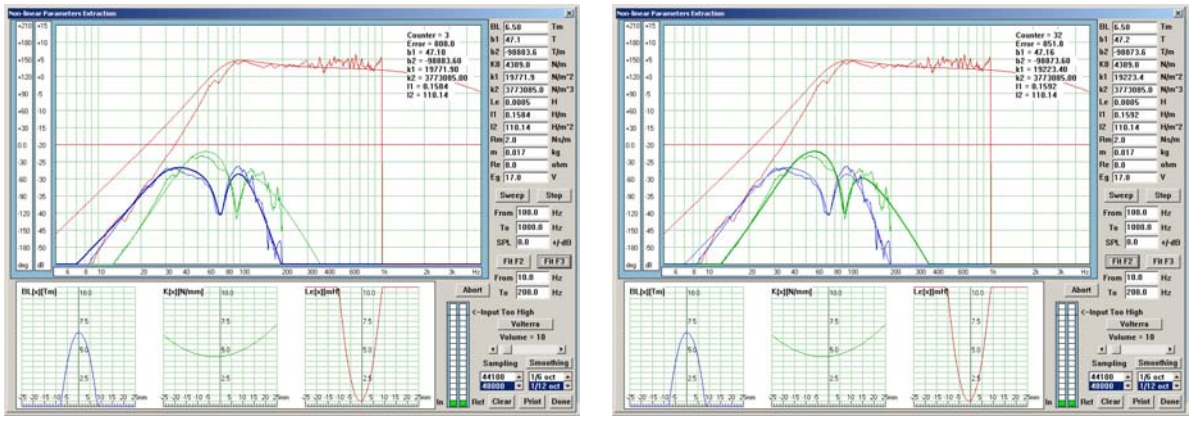


Figure 17.6 SPL mismatch on the fundamental frequency impacts curve fit for second and third harmonics.

In a drastic situation, the optimizer may produce erroneous results or can even "flip" the sign of the coefficient k2, if the test equipment does not have flat enough frequency response – see below. In this case, you need to review the total frequency response and environment of your testing process. If you are confident, that this is not the problem, you would manually correct the sign of the k2 (or b2) coefficient and attempt to re-optimize from perhaps slightly different starting conditions.

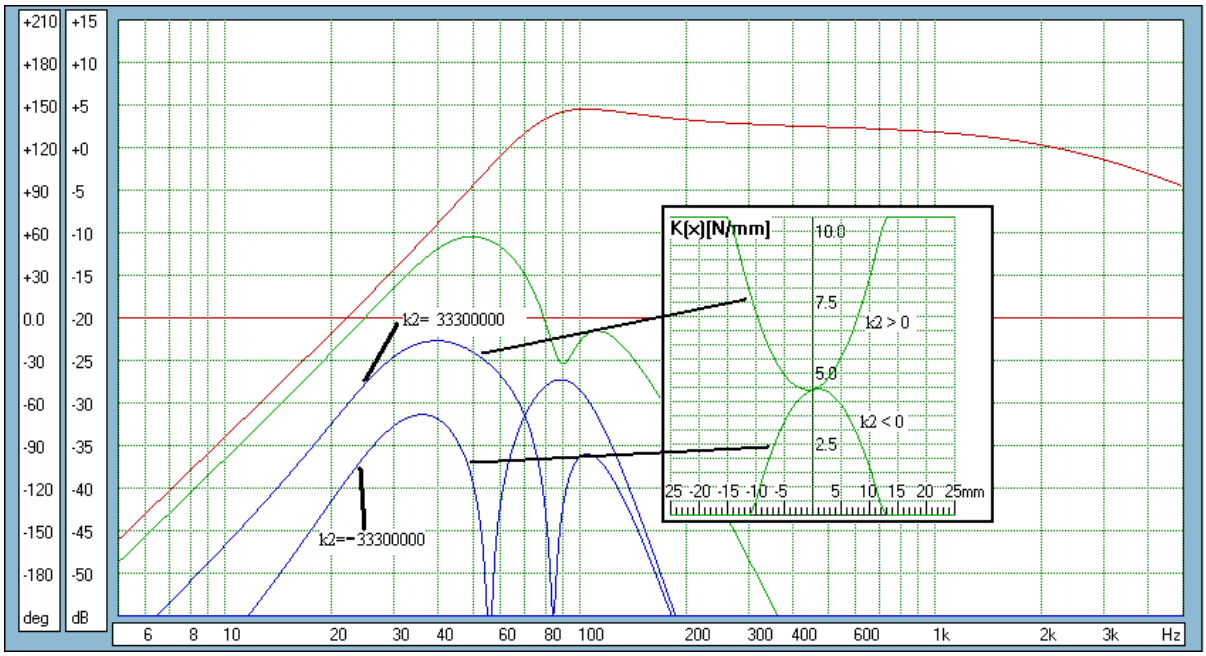


Figure 17.7 Effect of a negative k2 coefficient.

If you do have the problem with the total frequency response of the test equipment as shown above, you can remedy it to a significant degree by entering the required number of decibels you need to "lift" or "drop" the imported data, starting from the left-hand side of the screen (entered xx.x dB), and gradually tapering off and continuing to the nominated frequency entered right below the "Slope" field, where the modification no longer applies.

Making The Measurements

The block diagram below shows general idea for the test setup. The loudspeaker needs to be placed in a **sealed enclosure** of known volume. Stiffness of the enclosure will be added to the stiffness of the driver to give the total stiffness K0. The built-in non-linear model assumes that sealed box will be used in this test. Capturing all three harmonics over the required frequency range is accomplished in the following steps:

1. Set the sampling rate and output level using "**Sampling**" list box and "**Volume**" slider located at the bottom right-hand corner of the screen.

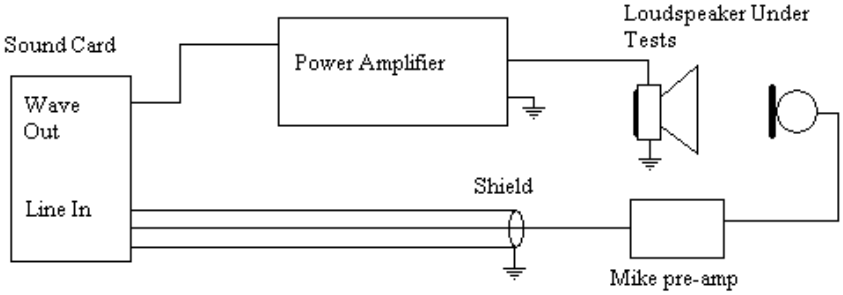


Figure 17.8 Suggested measurement setup.

2. Select sweep range by entering "**From**" and "**To**" frequencies under the "Sweep" button.
3. Commence the SPL measurements by pressing the "Sweep" button. You should see a picture similar to the one displayed below. You will notice, that the program tracks the fundamental frequency and second and third harmonics and placed a black marker line at the measurement point – see below.

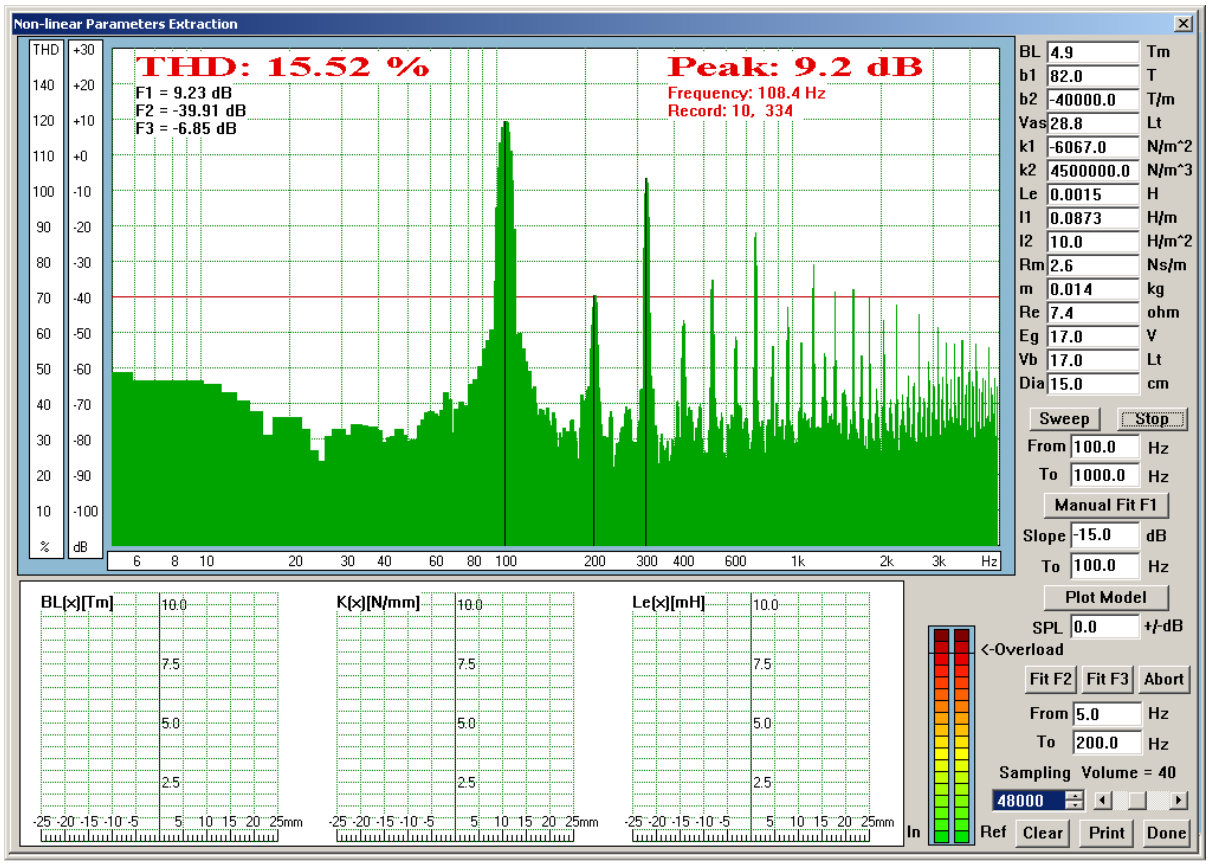


Figure 17.9 Fundamental = 108Hz, second harmonic = 216Hz and third harmonic = 324Hz (see black markers)

4. It is important **NOT** to overload the input of the sound card (or any other component in your measurement chain) as this will add to the harmonics level distorting the measurements. Figure above is just an example of overloaded soundcard input just to show the harmonics tracking process.

5. When the sweep process is completed, the program will display the captured fundamental and harmonics – see below. Please note, that the SPL scale has automatically switched to 5dB/div. All three curves are stored in the current driver data file.

Please note, that the information just collected (plots of the fundamental and the first two harmonics) is of extreme value to the designer. You could easily stop here and review the plots for the purpose of evaluating the distortion level of your design. The measurement procedure described up to this point is applicable to any type of enclosure and driver combination. You can simply use this feature as a universal distortion analyzer.

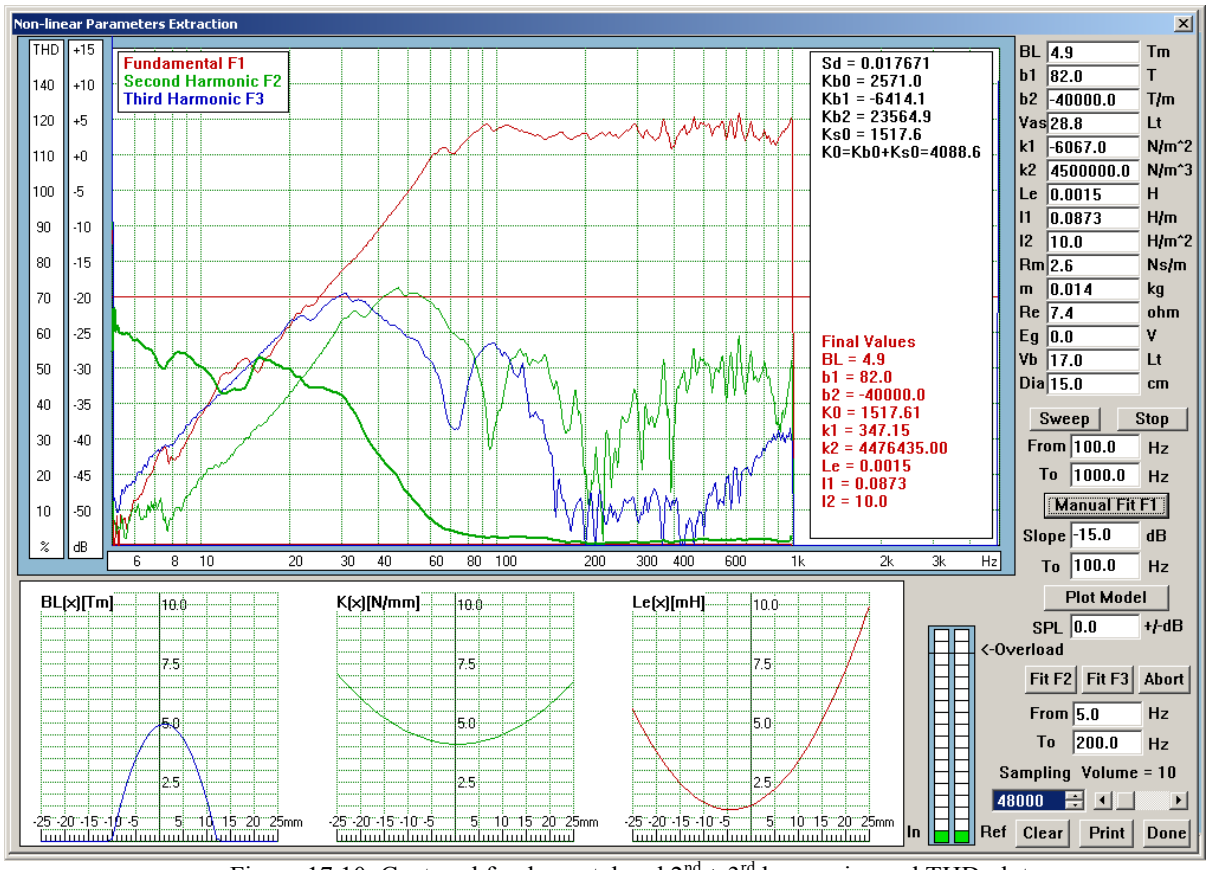


Figure 17.10. Captured fundamental and 2nd + 3rd harmonics and THD plot

6. You should now enter the parameters of the model into the fields provided:

   Eg – is Vpp/2 (half of the peak-to-peak input voltage [V].
   Re – is the DC resistance of the driver's voice coil [ohm].
   m – is the vibrating mass (including air load) [kg].
   Rm – is the mechanical resistance of the vibrating system [ohm].
   Le – is the voice coil inductance at lower end of the frequency range [H]. Typically you would enter about 150% of the VC inductance at 100Hz.
   Vas – is the loudspeaker Vas. [Lt]. Typically, the stiffness coefficient Ko = 1/(1/Cab + 1/Cms) would be used here, but because Vas is so widely used, we decided to use Vas and internally calculate Ko instead.
   BL – is the motor force, as measured in linear section[Tm].
   Vb – is the test box internal volume in liters [Lt].
   Dia – is the loudspeaker diameter in centimeters [cm].

   For now, you can use **k1,k2,b1,b2,l1 and l2** as provided default values. Enclosure non-linearity is built into the model using the following equations:

$$\mathbf{K_{box} = K_{b0} + K_{b1}*x + K_{b2}*x^2}$$

$$K_{b0} = \frac{\rho_0 c^2 Sd}{Vb}, \quad K_{b1} = K_{b0}(\gamma + 1.0)\frac{Sd}{Vb}, \quad K_{b2} = K_{b1}(\gamma + 2.0)\frac{S_d^2}{V_b^2}$$

$$K_{s0} = \frac{\rho_0 c^2 Sd}{Vas}$$

where: Sd = (Dia/2)*(Dia/2)*π, γ=1.40, Vas – is the equivalent compliance volume of the driver, Vb - is the enclosure volume, ρ_0 = 1.18kg/m^3.

The Kb0, Kb1 and Kb2 parameters are automatically calculated and displayed on the screen.

7. The above parameters should allow you to plot the SPL of the fundamental frequency. At this stage, the goal is to manually provide reasonably good match between the plots of the fundamental SPL measured on your driver (red wiggly curve) and the fundamental SPL of the built-in model (smooth red curve) over the frequency range defined under the "Fit F2" and "Fit F3" buttons. This process could be automated, but there are some advantages by doing it manually – you can simply "brute-force" the parameters as you want them. The process is rather simple. Press **"Manual Fit F1"** button and adjust data under this button for "Slope" and "To" frequency to get close second order slope matching. Secondly, you may need to move the model curves UP/DOWN using the "SPL" data (under the "Plot Model" button) depending on the amount of system attenuation and test voltage applied to the speaker.

   The **thick green curve** displayed on the screen is a **"semi-THD"** plot. It is called "semi" as the formula for calculating it only includes the second and third harmonics. It is however useful in the analysis.

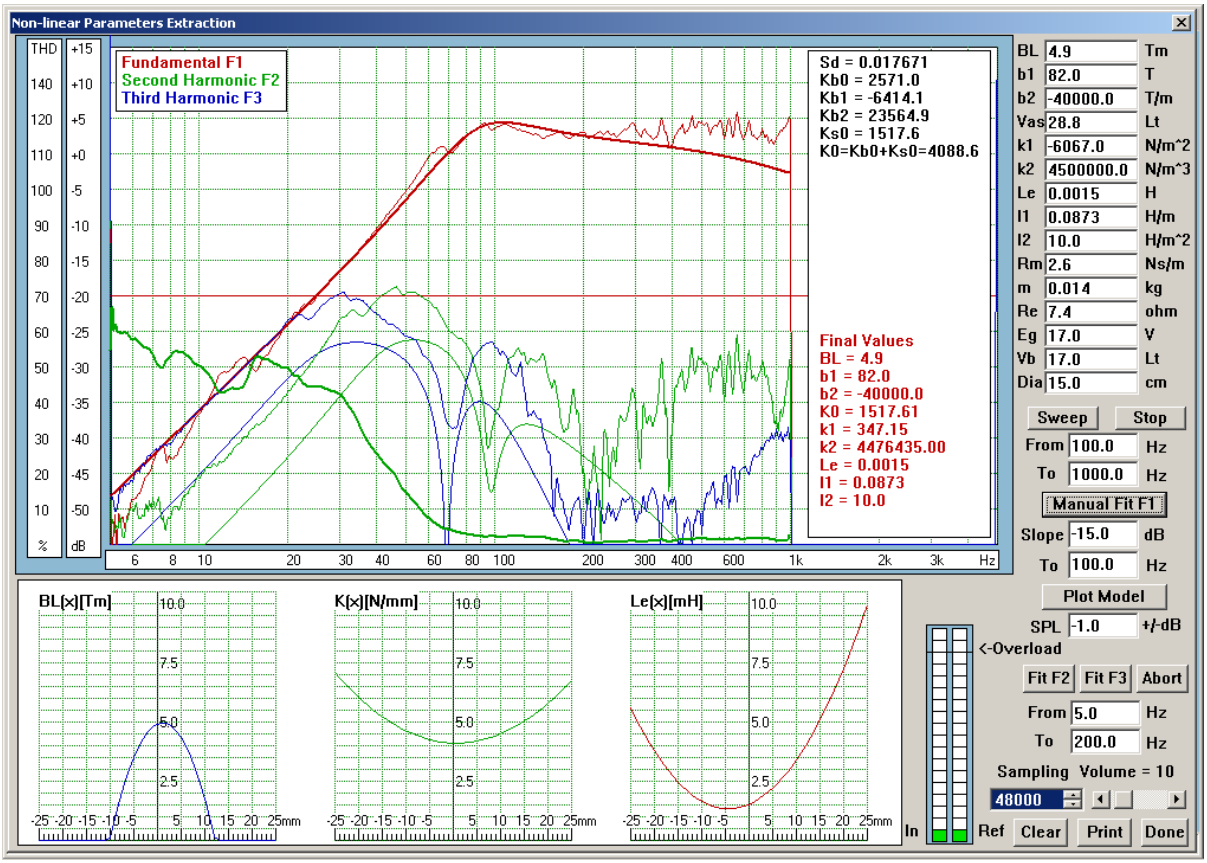


Figure 17.11 Matching SPL of the first harmonic from 5 Hz to 200 Hz – thick red curve

8. The final two steps in the process are "curve fitting" of the model into the measured second and third harmonics. At this stage, please enter the desired frequency range for the optimizer - the data fields are located under "Fit F2", "Fit F3" buttons**. It is recommended to keep the 5Hz – 200Hz default frequency range.**
9. Now press **"Fit F2"** button. The optimizer will perform two consecutive optimizations for the second harmonic. The only model parameters that are manipulated at this stage are: **k1, b1 and l1**.
10. Next, press **"Fit F3"** button to fit model into the third harmonic. The only model parameters that are manipulated at this stage are: **k2, b2 and l2**.
11. **It is strongly recommended to <u>repeat steps 9 and 10 several times in this order</u>. This is because results of the second order optimizations affect the third order optimizations. You should see it very clearly, that further attempts to fit the curves do not produce any significant reduction in error and this is the time to conclude the process. It is often beneficial to disturb Rm, m, BL, Vas, Dia parameters by 5-10% to see the reaction of the curve fitting process.**

The results of the curve fitting process on our example speaker are shown below.

BLo = 4.9 [Tm] **b1 = 211.48 [T]** **b2 = -89720.87 [T/m]**

Kso = Ko – Kbo = 4088.6 – 2571.0 = 1517.61 [N/m]
k1s = k1 – kb1 = -44067.7 - (-6415.1) = -37653.55 [N/m$^{2]}$
k2s = k2 – kb2 = 48005816.0 - 23564.9 – 47981252.0 [N/m^3]

Leo = 0.0015 [H] **l1 = 0.0252 [H/m]** **l2 = 41.09 [H/m^2]**

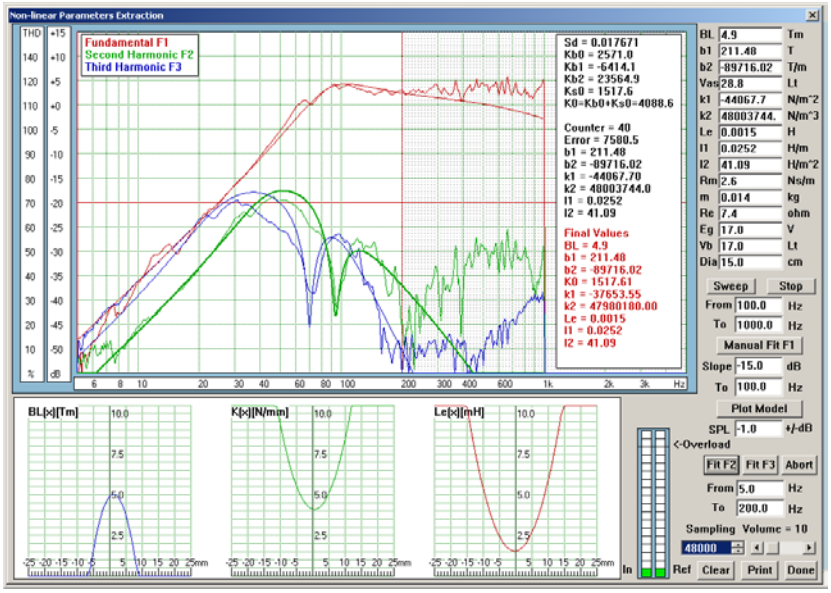


Figure 17.12 Result of "curve fitting" for F2

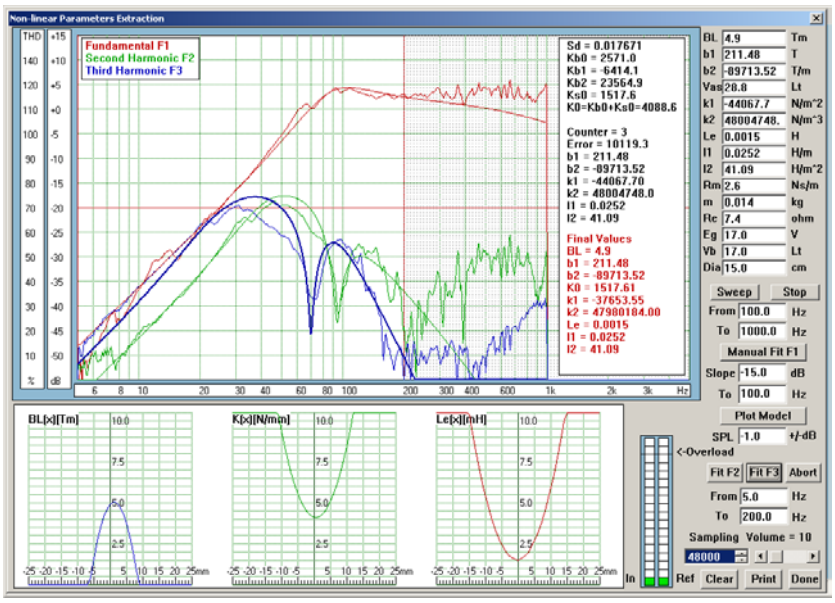


Figure 17.13 Result of "curve fitting" for F3

As mentioned in the introduction, the measurement process describe above is complex. In many instances you can split the whole operation into two steps:

1. Capturing fundamental, second and third harmonics for the purpose of evaluating the nonlinear distortion of the design. This process is quite simple, as all the necessary functions are highly automated and only require you to frequency sweep the system under test. The process of generating the sweep signal and tracking and capturing of the harmonics is fully automated.
2. Extracting coefficients of the BL(x) , Ks(x) and Le(x) for the purpose of further modeling of the system. Please be aware, the extracted coefficients are largely dependant on the accuracy of the measurements. Also, all three coefficients represent parabolas. This type of curve is a truncated approximation of the real-life characteristics of the loudspeaker non-linearity and will characterize some speakers better than others

Making Measurements Using Differential Equations Method

Preferably, the test should be conducted in anechoic chamber or at least in "half-space" acoustic environment. Loudspeaker under test is mounted on a test fixture in "free air". The fixture does not provide any enclosure, it only serves to mechanically support the driver. The loudspeaker will be driven quite hard – up to 20Vpeak, so special precautions need to be taken. For instance, if the loudspeaker's efficiency is 85dB/W/m, and 20watt applied to the speaker will produce 98dB SPL at 1meter. Most test microphones should be able to handle this pressure.

If you do not have access the anechoic chamber, you may have to resort to the "close mike" techniques. The issue now is more complex. For instance, if you decide to keep the same power (20Watts) and bring the microphone closer to the speaker (say 12.5cm distance) the SPL pressure will produce 116dB at the microphone location. This is significantly more than discussed before and you need to make sure, that the microphone and the pre-amplifier can handle this conditions without introducing clipping distortions. The basic test setup is shown below and captured loudspeaker response for linear (red) and nonlinear operation (blue) is shown on Figure 17.15.

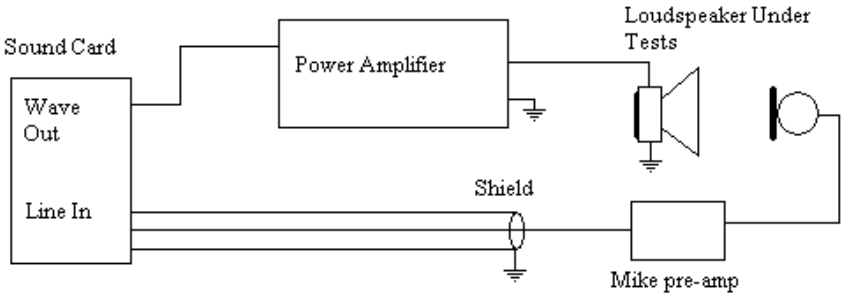


Figure 17.14. Test setup for Differential Equations Method.

Capture Issues to Consider

The essence of this measurement method is to excite the loudspeaker with sufficiently high voltage, so that it can be driven into the nonlinear region of BL(x) and Cas(x), record the cone excursion and then curve fit model parameters to the measured data in time domain.

The output of the sound card will deliver a tone burst of 10-20 periods of sine-wave at the frequency selected from the **"Generate Tone Burst"** group of controls. In response to the burst, we are attempting to capture the steady-state response of the loudspeaker, free of any transient distortions. There may be several reasons why you may want to cut out the initial part of the captured burst. For one, the loudspeaker needs to finish the initial ringing due to applications of large input voltage burst. Secondly, the sound card may produce some voltage fluctuations on the initial part of very first buffer output due to its internal workings – this was our observation when using Delta 410 card from M-Audio.

To overcome the above issues, you may need to uncheck the **"Save Linear Result"** and **"Save Nonlinear Result"** check boxes (to prevent the test results from being automatically saved to disk) and generate tone burst several times observing captured signal on the screen. The aim is to **see several periods** of the chosen frequency, with the **same amplitude in each period** – see Figure 17.15. The captured signal will be aligned to the left side of the screen, ready for curve fitting process. The alignment will be guided by the negative-to-positive voltage transitions of the captured signal. Therefore, it is important, that we cut out all "false" or undesirable transitions. On Figure 17.15, we have captured 2.5 periods of 40Hz distorted sine-wave and 8 initial "bad" transitions were skipped. It is observable, that linear response AND nonlinear response of the loudspeaker are correctly captured and automatically aligned on the screen.

The actual output voltage generated by the sound card can be adjusted using "Volume" slider in the **"Generate Tone Burst"** group of controls. We would recommend to set the card output between 75-90% and use your external power amplifier volume control to adjust the actual voltage measured across the loudspeaker. You will need to measure the **peak value** of the actual voltage across the loudspeaker for linear test and nonlinear test. These voltages will be used by the program to perform necessary internal calibrations. For entering the voltage amplitude of small-signal (or linear) operation, please use **"Linear Speaker Vin"** field – default is 2.50volts. For entering the voltage amplitude of large-signal (or non-linear) operation, please use **"Nonlinear Speaker Vin"** field – default is 20.00volts.

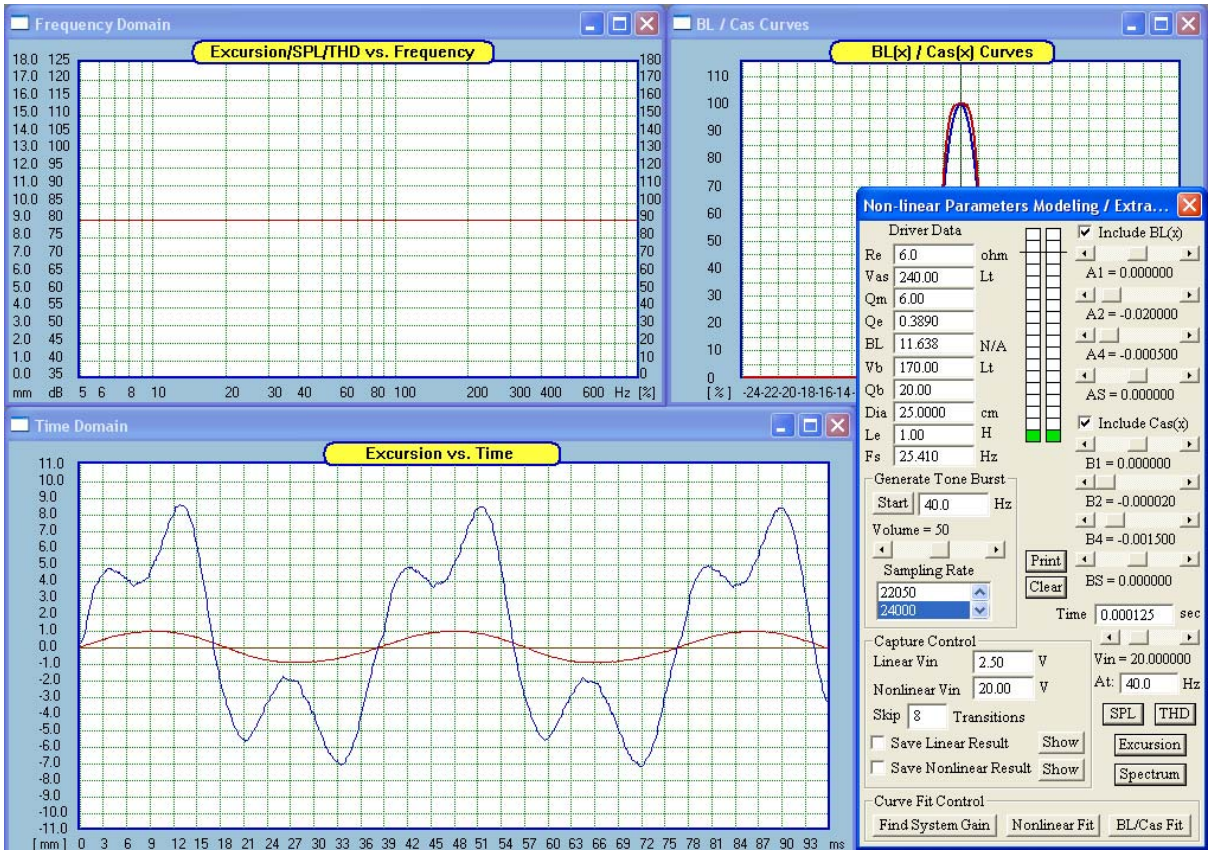


Figure 17.15. Captured loudspeaker response for linear (red) and nonlinear operation (blue).

Sampling rate for this test is fixed at 24kHz. It can not be changed from the screen list box. Loudspeaker parameters required by this screen are basic "free air" T/S data. ALL T/S fields included on this screen need to be supplied with correct data. The default starting data for BL(x) and Cas(x) exponential fit curves is shown below.

$$BL(x) = BL_0 e^{A4(x-S)^4 + A3(x-S)^3 + A2(x-S)^2 + A1(x-S)}$$

A4 = -0.0005 Wide (low) or narrow (high) of the curve as it falls to Y=0
A2 = -0.05 Flat (low) or Peaked (high) around X=0
A1 = 0.0 Asymmetry about X=0
S = 0.0 Shift of the entire curve about the X=0

$$Cms(x) = Cms_0 e^{A4(x-S)^4 + A3(x-S)^3 + A2(x-S)^2 + A1(x-S)}$$

A4 = -0.0005 Wide (low) or narrow (high) of the curve as it falls to Y=0
A2 = -0.04 Flat (low) or Peaked (high) around X=0
A1 = 0.0 Asymmetry about X=0
S = 0.0 Shift of the entire curve about the X=0

The captured data for linear and non-linear operation of the loudspeaker is automatically saved into the hard disk the way you see it on the screen. You can disable saving process for liner and non-linear data by checking or un-checking the two boxes located in "Capture Control" group. You can only "check" one of the boxes at a time, but you can uncheck both of them. The program makes sure, that you can not have them both checked at the same time. The corresponding **"Show"** buttons re-load the saved data and display it on the screen. Assuming, that you have captured the measurement results for your speaker, you may load them both onto the screen. Looking at the shape of captured example non-linear response, it is easy to observe, that the curve is asymmetrical around 0.0V. This would lead us to expect some asymmetrical shape in BL(x) , Cas(x) or both.

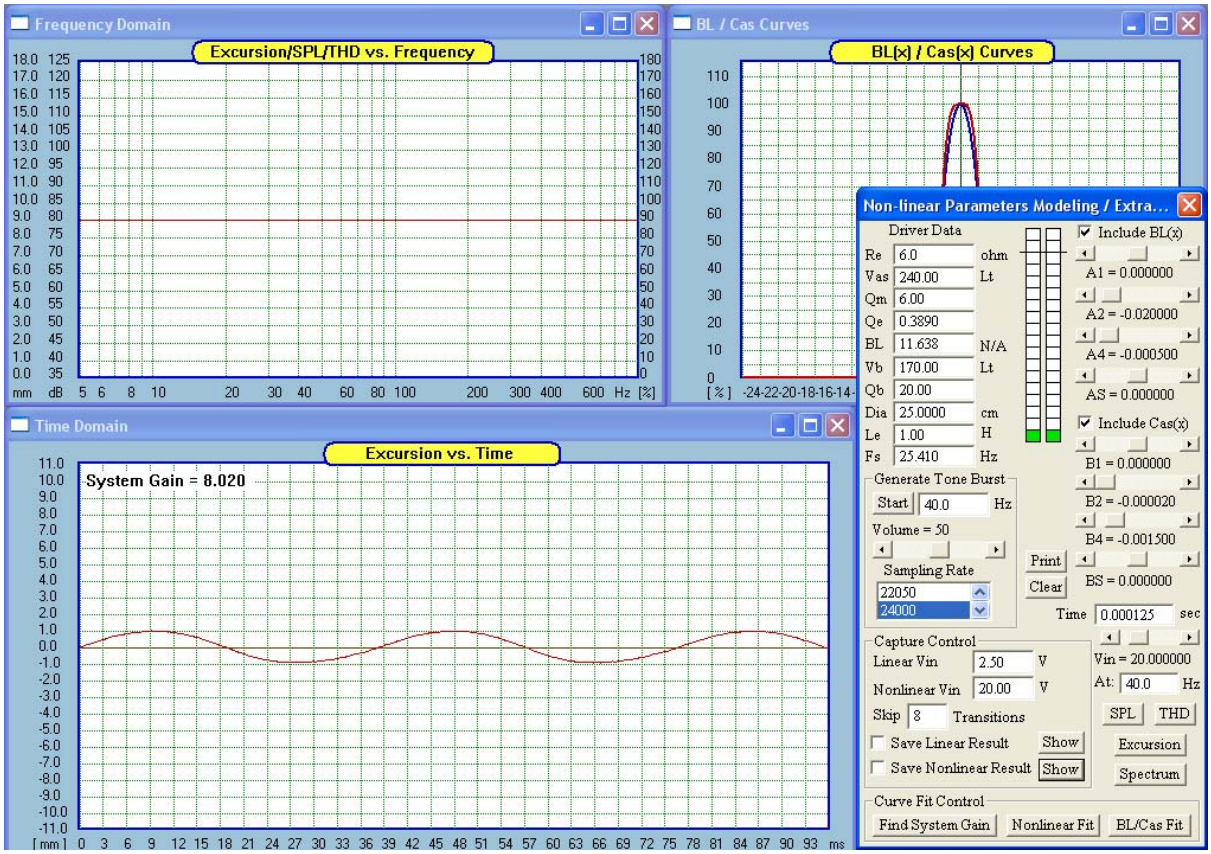


Figure 17.16. Result of "Find System Gain" operation – your system gain is determined.

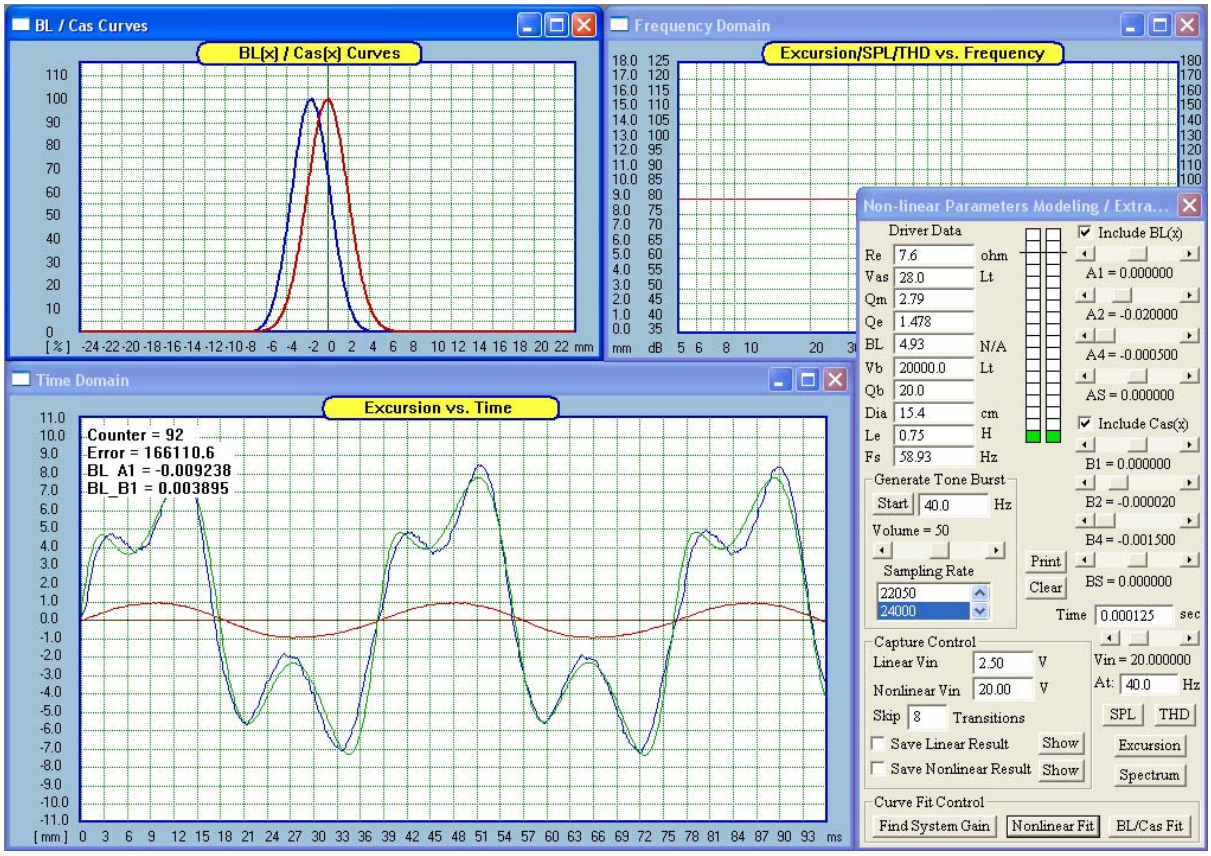


Figure 17.17. Result of "Nonlinear Fit" – driver parameters are found.

When the curve fitting process is completed, you can press **"Show BL/Cas Fit"** button and you should see both curves displayed on the screen – see Figure 17.18 as an example. Optimized valued for BL(x) curve (the A-coefficients) and Cas(x) curve (the B-coefficients) are shown on the right-side of the screen and the sliders are automatically set to those values.

BL(x) curve:
A1 = -0.011779
A2 = -0.042807
A4 = -0.000114
S = -2.818824

Cas(x) curve:
B1 = -0.004713
B2 = -0.040595
B4 = 0.000000
S = -0.090285

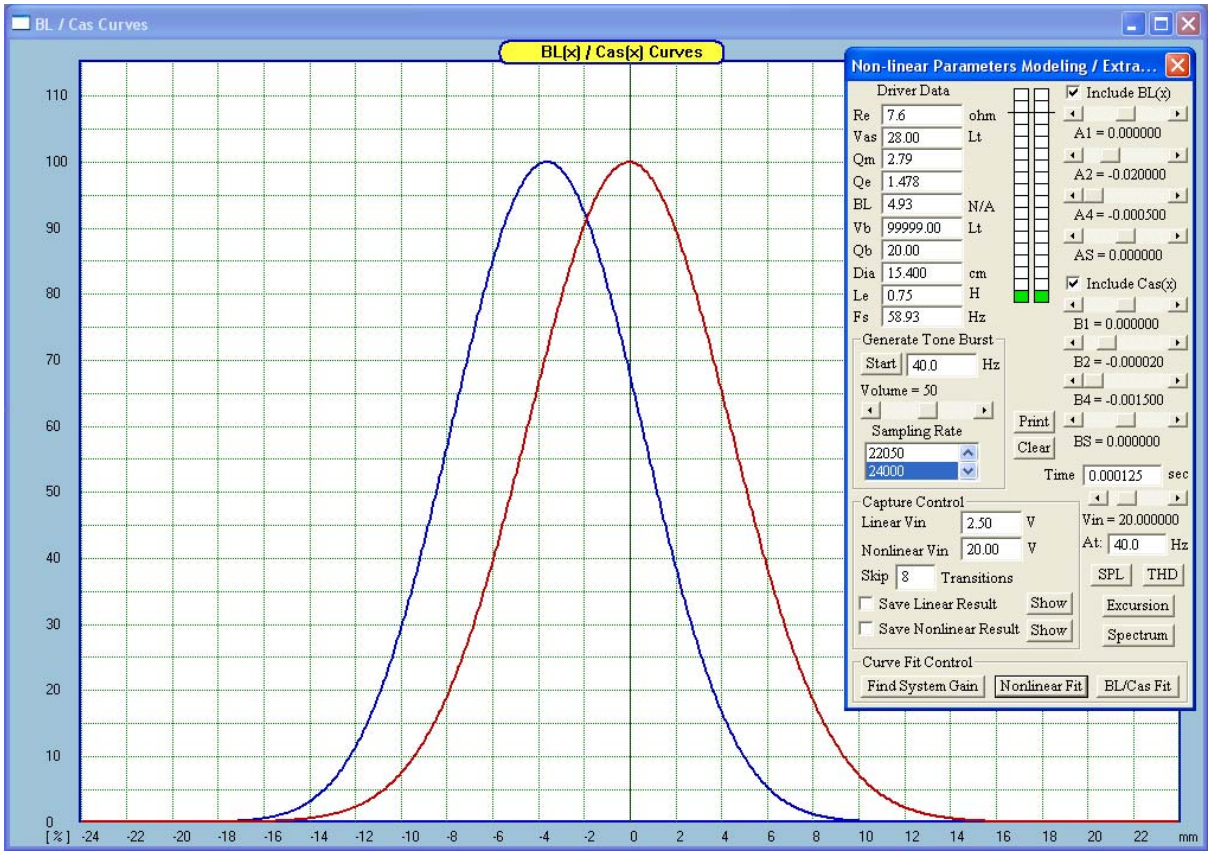


Figure 17.18. Optimized BL(x) and Cas(x) curves.

Other controls on this screen will be operational when you load all required T/S parameters first and include:

1. **"SPL" -** button used for plotting frequency response and cone excursion of the driver in sealed test enclosure.
2. **"THD" -** button used for plotting THD of the driver in sealed test enclosure.
3. **"Excursion" -** button used for plotting cone excursion of the driver in sealed test enclosure.
4. **"Spectrum" -** button used for plotting spectrum response of the driver in sealed test enclosure.
5. **"At" xxxx "Hz" -** data field used for entering frequency at which cone excursion is modeled.
6. **"Time" xxxx "sec"** - data field used for entering time step for Runge-Kutta algorithm.

7. **BL(x) Shape**
   A4 = Wide (low) or narrow (high) of the curve as it falls to Y=0
   A2 = Flat (low) or Peaked (high) around X=0
   A1 = Asymmetry about X=0
   S = Shift of the entire curve about the X=0

8. **Cas(x) Shape**
   B4 = Wide (low) or narrow (high) of the curve as it falls to Y=0
   B2 = Flat (low) or Peaked (high) around X=0
   B1 = Asymmetry about X=0
   S = Shift of the entire curve about the X=0

9. **"Include Cas(x)"** – check box to include compliance effects in model.
10. **"Include BL(x)"** – check box to include magnetic motor in model.
11. **"Vin"** – slider to adjust input voltage across models' terminals.
12. **"Clear"** – button to clear the screen.
13. **"Driver Data" –** 10 data fields for Small-Thiele parameters.

Measurement method

1. Enter Driver's T/S parameters.

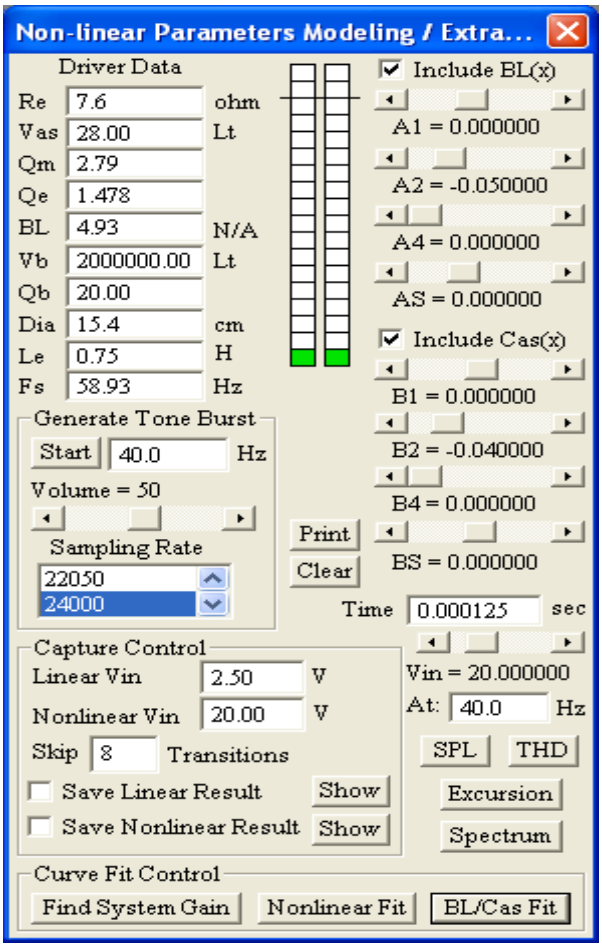
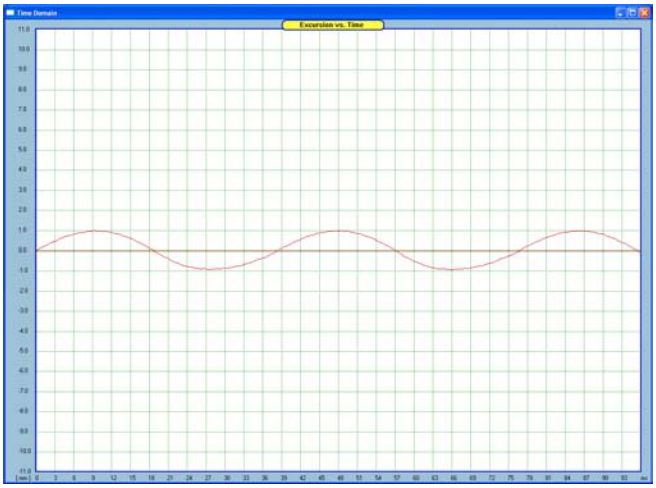


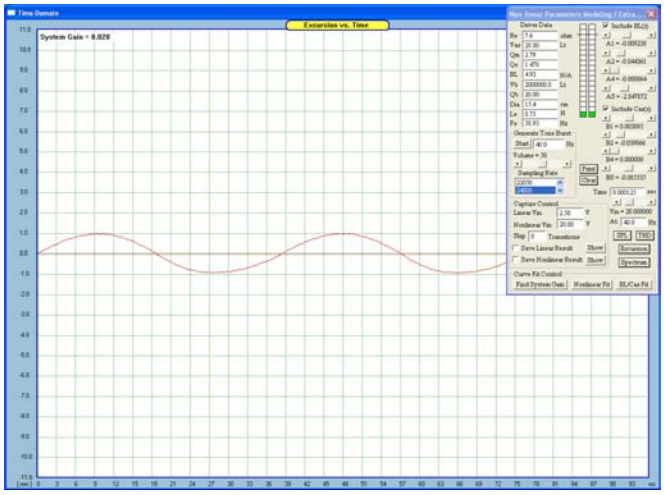
Fig 17.19. Driver T/S parameters entered.

Please note, that this driver was mounted on a baffle, and Vb was entered as very large: Vb=200000Lt.

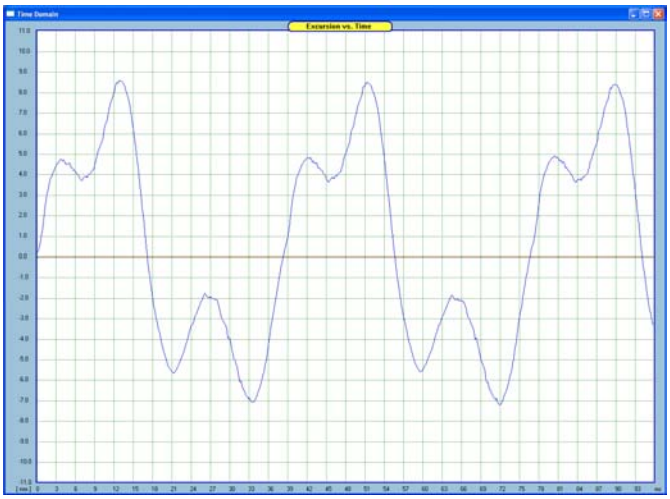
1. Select Tone Burst frequency – say 40Hz.
2. Select Volume – say 50 level.
3. Set output voltage across loudspeaker terminals to 2.500Volt PEAK.
4. Enter the same 2.500 into "Linear Vin" data field.
5. Press "Start" button and play the burst several times to see, if you can capture 2-3 periods of good, clean sinewave – see figure below:



6. Check “Save Linear Result” and press “Start” button to finally save the result.
7. Press “Find System Gain” button to calibrate internal data.

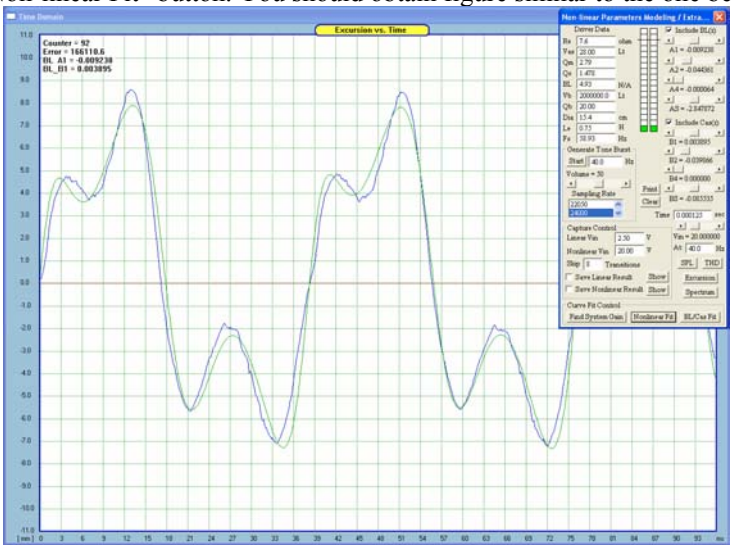


8. Increase voltage across loudspeaker terminals to driver the speaker into distortion. You should see something like figure below:

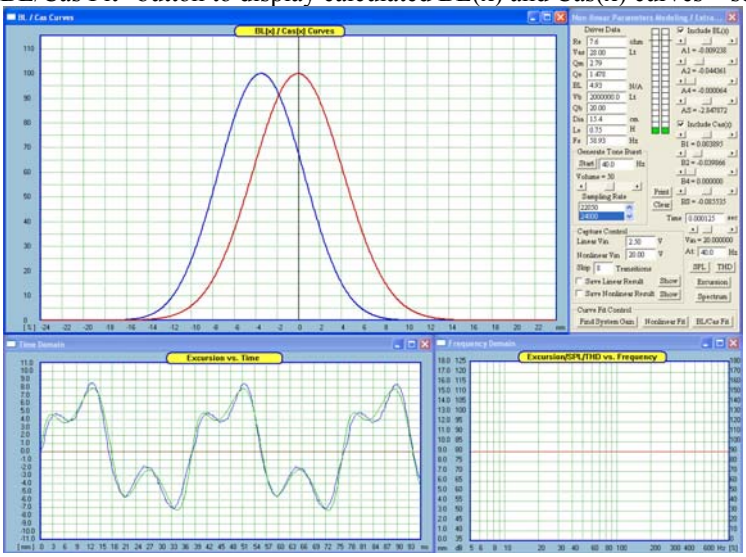


9. Enter the same increased voltage (say 20.0) into “Nonlinear Vin” data field.
10. Press “Start” button and play the burst several times to see, if you can capture 2-3 periods of good, clean sinewave – see figure above
11. Check “Save Nonlinear Result” and press “Start” button to finally save the result.

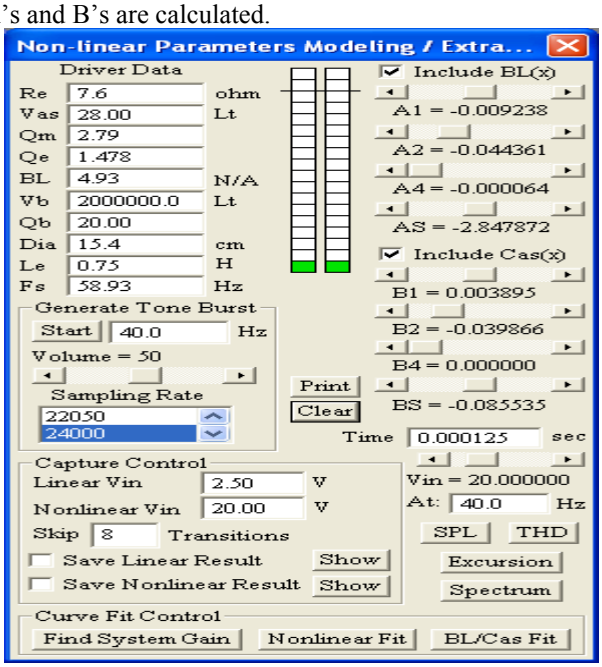
12. Press “Non-linear Fit” button. You should obtain figure similar to the one below:



13. Press “BL/Cas Fit” button to display calculated BL(x) and Cas(x) curves – see below.



14. All Non-linear A’s and B’s are calculated.



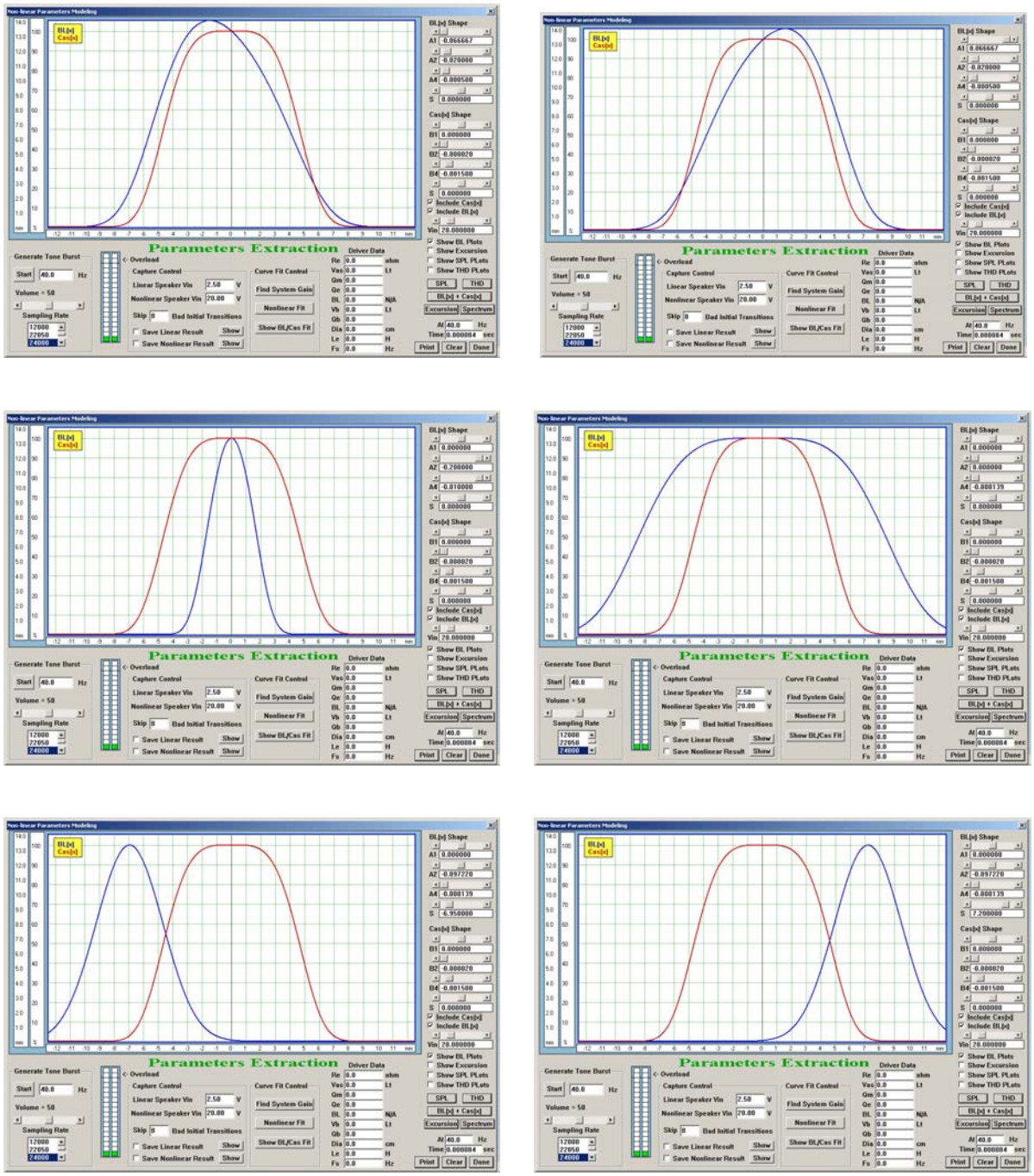


Figure 17.20. Effects of different A1, A2, A4 and S slider positions on BL(x) curve

Modeling Second Order Systems

The "**Non-Linear Parameters Modeling**" screen can also be used to predict SP/ THD/ Cone Excursion / Spectrum of a loudspeaker mounted in a sealed enclosure.

1. Enter **"Driver Data**" T/S parameters
2. Select **"Vin**" voltage.
3. Select test frequency "**At:" in Hz**
4. Press **SPL / THD / Excursion or Spectrum** buttons for appropriate plots**.**

Shown below, is an example of a loudspeaker in a 170lt sealed enclosure modeled with the given BL(x) and Cas(x) parameters.

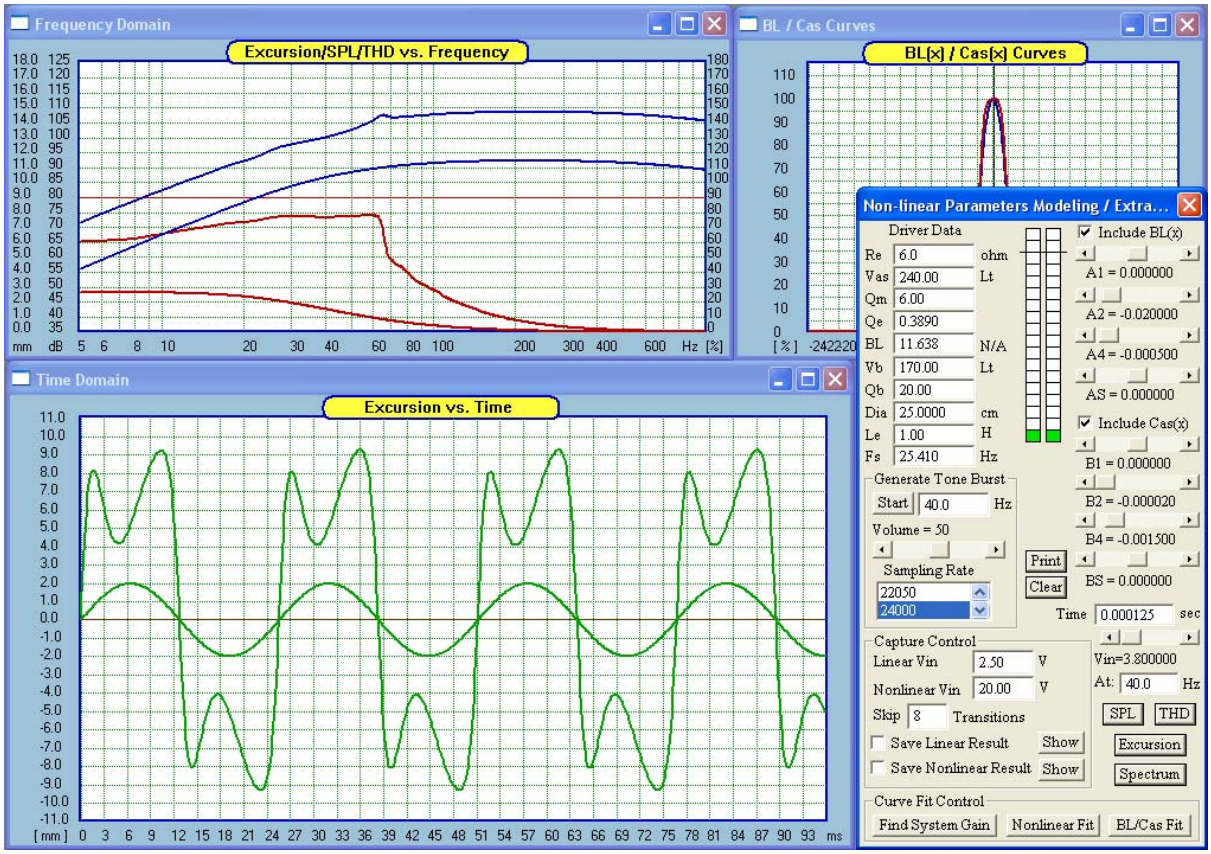


Figure 17.21. Example of modeling second order systems.

Nonlinear Parameters Extraction "Screen 1" THD extended to display 110dB.

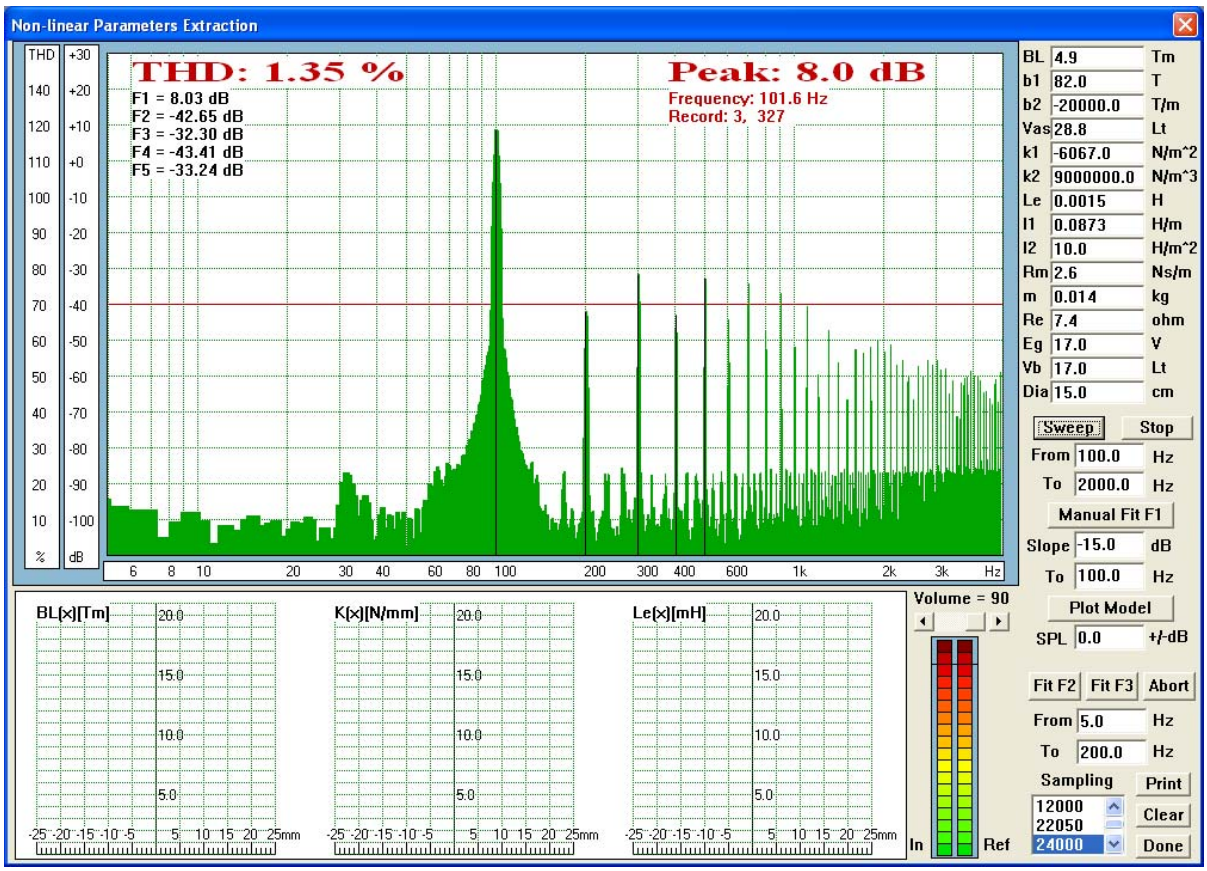


Figure 17.22. Distortion screen runs in 110dB mode during measurement.

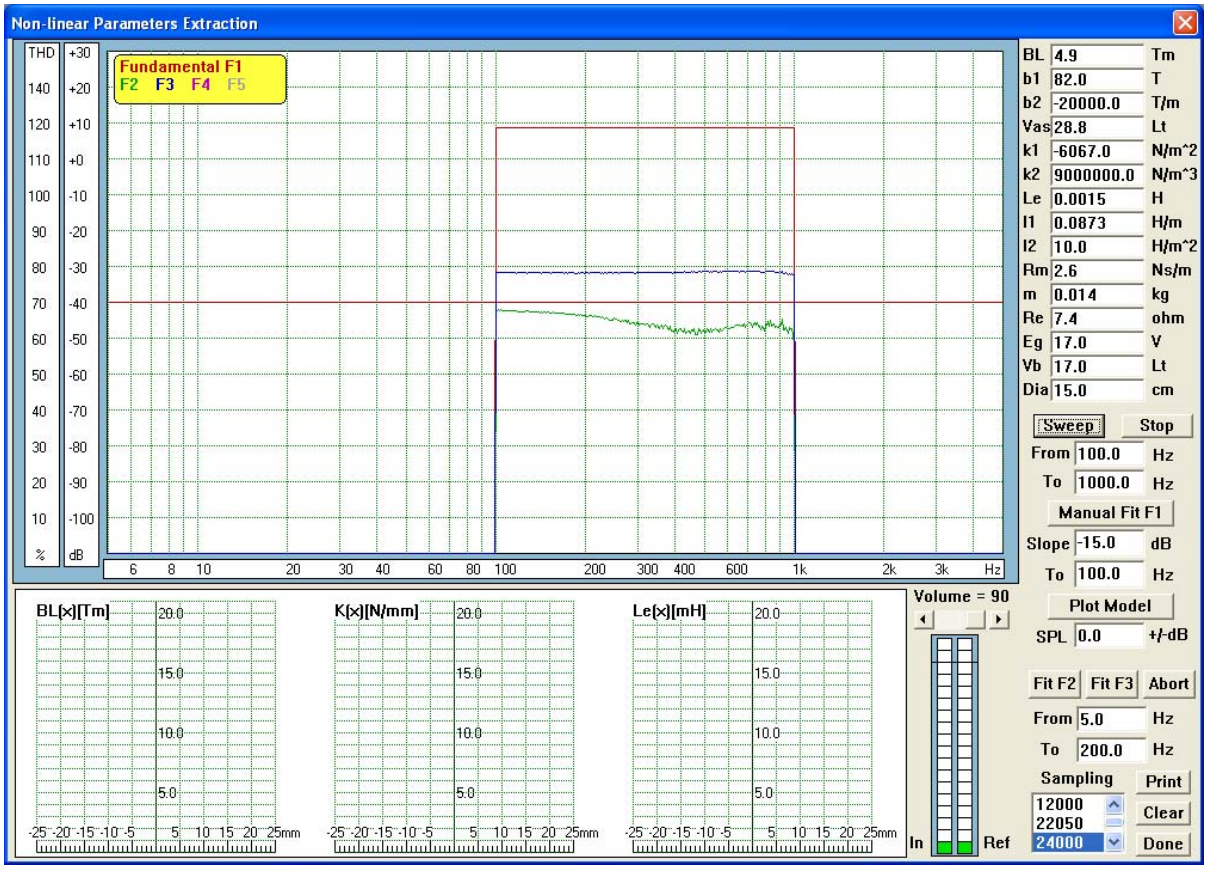


Figure 17.22. Distortion screen displays 110dB mode after measurement.

Chapter 18. Recommended Design Sequence.

As discussed in Chapter 1, SoundEasy modules/functions should be used as a sequential set of tools that process incoming information and pass the results to the next tool. Here is a brief look (a high-level overview) at the recommended design sequence, which describes the most important functions of SoundEasy. The time spent with each tool may vary, depending on the design accuracy you wish to achieve.

Box Module – Creating a Database of Fully Interchangeable Drivers & Enclosure Design

1. Enter all electromechanical parameters of the driver using the 'Driver Parameter Editor' tool. SoundEasy is a complex mathematical "number cruncher" and you need to enter all required parameters in order to fully utilize its processing power. There is number of ways you can explore, to come up with the best estimation of a missing parameter. Please contact us if you are unsure of how to deal with a particular situation. Generally, the program will attempt to calculate (or check) some of the parameters for you, while you enter the data. Particularly**: Qe, Qm, BL, Fs, Mmd exhibit heavy interdependence and will be checked repeatedly as you enter the data.** Next thing to do, is to **edit impedance model you wish to use with the enclosure design screen**. If you have entered ALL parameters on the 'T/S Editor" tab, the 'Impedance Models' tab invoked from the 'Calculators' menu will have entries pre-edited for you. At this moment, the 'Enclosure Type Selected' should be set to 'Driver' and 'Model Selection' check boxes should be checked to the model you wish to use.

You are now ready to select 'Enclosure Design' screen and enter enclosure parameters in order to model your box. For instance, for sealed box you need to enter: 'Rear Box Vb' and 'Rear Box Qb'. For vented enclosure you will need to additionally enter: 'Rear Box Fb', 'Port Qp', and 'Port Diameter'. Continue the same way with other enclosure types you wish to analyze.

2. Use one of the two methods described in Chapter 2 and 3 to enter the magnitude of the frequency response of the driver or its impedance, using the tools provided. If you decide to use 'Impedance Models' tab for generating the input impedance curve, you do not have to enter (draw) impedance curve manually using mouse on the 'Editor' screen. The dialog box enables you to select targeted enclosure and will calculate whole input impedance function of a driver mounted in this enclosure.

3. Re-create the **transfer function** and **input impedance** of the driver (magnitude/phase) using the 'Hilbert-Bode Transform' tab. **Remember, you are responsible for correctness and quality of all data points above 35dB line (bottom of the Editor screen) and SoundEasy is responsible for the rest of the curves between 1Hz and 100kHz**. Do not forget to save your file.

4. Alternatively, use importing functions provided with the program. The data importing dialogue boxes typically have three fields you can use to modify the imported data right during the importing time. You can modify amplitude and time delay of the imported data. One of the **sources of data errors** can be the noise level in your measurement system (sound cards). This problem may manifest itself as amplitude errors and particularly **phase errors at lower signal levels**. SoundEasy allows you to add small time delay to somewhat compensate for the measurement system errors. Comparison of the imported phase with no acoustic delay with the phase generated by the built-in Hilbert-Bode Transform will tell you if your measured data is correct.

A simple test (developed by one of the SoundEasy users) for your sound card is to measure the amplitude and phase response of a –40dB or –50dB resistive voltage divider (simply use 2 resistors). This test will tell you if the amplitude and phase are BOTH FLAT LINES and if your sound card based measurement system is accurate. In addition, the H-B Transform will help you to **determine the acoustic center offset** of your driver. For instance: A driver is mounted on a baffle. The test microphone is placed say, 25cm (250mm) from the front of the baffle (your reference plane). Perform the amplitude and phase measurement on this driver and import this data into SoundEasy. When you attempt to match the imported vs. H-B Transform generated phase responses, using the mike-driver distance slider, you will find, that the slider has to be set to say, 280mm for the phases to match. The difference (30mm in this case) is the acoustic center offset for this driver.

There is an excellent treatment of the subject of acoustic center offset given by D'Appolito [77, page 144, 128-133]. Please read these pages, as SoundEasy follows the general guidelines of using the acoustic center offsets in multiple driver system design. This approach is very powerful, as it allows you to swap correctly edited driver files in your projects with a "click of a button", and still maintain the overall accuracy of the design process.

Re-create the **transfer function** and **input impedance** of the driver (magnitude/phase) using 'Hilbert-Bode Transform' tab. Do not forget to save your file. This step is necessary for two reasons: (1) if everything goes well, you can rest assured, that you have properly edited data files and (2) this step flags the attempt to create amplitude and impedance transfer functions, so other tools can read the data with some confidence.

Completing the all the above, you have now completely characterized your driver. The important item to remember is, that you must run 'Hilbert-Bode Transform' tab on amplitude and impedance data to assure full compatibility of entered/imported files with requirements of all SoundEasy tools. Please refer to Fig 2.1

5. Explore/design a suitable enclosure(s) for the driver(s) using 'Enclosure' screen. At this stage, you may wish to perform "what-if" analysis for the alternative enclosures. If you attempt to design a rectangular enclosure (sealed or vented), you may find our simple **"Enclosure Details" calculator** handy. It **provides construction details in a CAD-like fasion** and it calculates panel dimensions, accounting for all extra parasitic volumes inside the enclosure. Bracing is also dimensioned and calculated. Implemented as a separate dialogue box, is a **"Diffraction Analyser".** This calculator can accommodate up to 4 drivers on 2m x 2m baffle and generate diffraction + mutual radiation impedance curves. Diffraction distortion can be added to all enclosure plots.

6. We strongly recommend, (particularly if your design POWERED subwoofer) that you **take your time and perform Large-Signal analysis of your design**. This may be an 'eye opener' even for the experienced designers and may save you future costs of replacing damaged woofers. Both, BL(x) and Cas(x) curves can be approximated to offer the best match with the driver's data. The manual contains a large chapter devoted to the Large-Signal analysis with detailed explanations of possible practical issues.

7. Use the 'Optimizer' screen to optimize the enclosure size (Vb.), tuning (Fb) and Q-factor (Qb). When you experiment with the 'Minimum Box Q-factor' setting on the Optimizer screen, the algorithm may be able to suggest an enclosure parameters, that will make the driver more suitable for yet another application. For example, by lowering allowed Qb, the Optimizer may be able to suggest a bass-midrange application for your small-size woofer. Remember to save the data file for further processing.

8. You may also wish to explore the time response of the chosen box using the 'Time Domain' screen.

9. Room/Car Acoustics is a "free-standing" module, that has built-in several models of the most common vehicles. By employing the FEM it allows you to place up to **4 speakers and a test microphone** inside the passenger cabin or a room for the low-frequency cabin-contribution evaluation.

Crossover Module – CAD electrical/electronic issues

10. Design the compensation/crossover network using the 'Crossover Design' tool. A good starting point is to create the project file to upload the drivers' files generated previously by the 'Enclosure Tools' module. If edited correctly, these files are now expected to contain SPL/impedance data of the selected drivers. Then, use one of the built-in crossovers or **create your own design on the CAD screen**. The "pick-and-place" technique was described in the previous chapters.
You will need to configure the crossover network by making sure, that drivers are connected to the correct filters (crossover outputs) and are phased the way you want them. **You would place the drivers' icons on the schematic at the correct filter terminations.**
Next, doubleclick on the icons to assign appropriate drivers there (by doing this, you tell the program **what impedance data needs to be assign to those nodes**, and how to label the driver on the schematic – just like labeling all other components). Phasing of the drivers will be done when you perform frequency response plotting. As you know, you can add and delete a component or a connection. You can also add a new network, selected from the built-in suite of networks.

11. The CAD process also involves crating the impedance, amplitude, time delay and phase compensating networks using the CAD screen provided. You would then enter the components' values and open the **"Frequency Domain"** screen and evaluate the network and driver+network performance versus frequency.

Now, just before plotting, you will be prompted to phase the drivers by placing correct (positive or negative) numbers in the provided fields for the test nodes. Also, you will need to specify the type of driver connected between those nodes (by doing this, you tell the program **what amplitude data needs to be assign to those nodes**). You will find the calculators included with the tool very handy. They are a good starting point for component values used in various equalization circuits.

12. The next possible step is to perform the "what-if" analysis and change some of the components' values. You may not be able to obtain the exactly calculated values from the component supplier. Very often, there is a scope for performance improvement from the component values provided by the calculators.

13. At this stage, **"Time Domain"** screen would normally be used for detailed analysis of time response of single filter or the whole crossover. Two concepts of achieving "Constant Voltage" crossover type of responses have been described in this manual. (1) using first order filters and (2) using "filler driver". Alternatively, you may wish to determine how far away from the "ideal" design is your project. There are several test signals provided for detailed analysis. Also, there are TWO methods, simple FFT and **the powerful Modified Nodal Method** (used by SPICE-like CAD programs), provided for you to chose from. Generally, you would use FFT if you have a driver(s) connected to the network and MNM for detailed circuit analysis without any drivers.

14. Optimization is an important stage in the network design. You should invoke the **'Optimizer'** screen to perform single filter or complete crossover optimization over the critical frequency range. The target frequency response can be heavily modified to suit your requirements using the provided tools. The program accepts **Zmin and Phase, as constraints** and will display minimum system input impedance on the optimizer screen. If you are happy with, you should save the results of the optimization process back to disk (and perhaps to a new file for reference). The new crossover component values will replace the old ones.

If you happened to design a subwoofer and wish to check the box performance with the active equalizer, you need to **save your network as the "driver file" (.wfr, .uba….)**, so you can later re-use it with the 'Box' module for enclosure+EQ performance evaluation**. This should be followed by a full Large-Signal (temperature, input power, cone excursion, THD) analysis**.

A **Digital Filter function** enables you to listen to your crossover design implemented as FIR filter . The crossover is mimicked by your sound card. **Digital Equalizer** is a unique feature allowing for equalization of irregular SPL of the driver to a "ruller-flat" response of an electrial filter.

Sum Plot Module – System Assembly, System Placement & Room Acoustics Issues

15. Invoke the 'System' tool for final review of the system responses in **3D configuration**. If edited correctly, all your drivers' files should contain **SPL measurements performed at the same microphone distance for each driver (recommended 1m, and on-axis) and correctly edited acoustical offsets** from the reference plane – usually the front baffle mounting surface. Use the project file created by 'Crossover' module to upload the drivers' files and crossover schematic. Plot and examine the system performance in frequency and time domain screens. The frequency response may still be improved by optimizing individual crossover filters for the total flatness of the frequency response. Invoke the **'System Optimizer"** screen for working on individual drivers. It is recommended, to perform the filter optimization first and then the total crossover optimization.

As in the Crossover module, shaping the **target frequency response is extremely flexible** and optimization can **be Zin-constrained**. You must save the results of the optimization process back to the project file in order to update stored component values.

17. Explore **'off-axis'** performance and **tilt** drivers at will, if you design a box with complex front baffle shape. Also, design 5-way system or WMTMW array in separate projects and compare their performance on **"Frequency Domain"** and **"Time Domain"** screens. These screens are specifically not refreshed to make the comparison possible. There are also interesting plots of **vertical and horizontal polar and "straight-line" (easier to measure at home lab) plots** available.
Geometric Acoustics - if one assumes that the dimensions of a room are large compared to the wavelength, then sound waves may be considered in much the same way as light rays are treated in optics. This situation frequently occurs in architectural acoustics, especially in **large auditoria**. A limitation of the geometrical approach is that usually only primary and possible secondary reflections can be studied, it is restricted to frequencies of 300 Hz and above. You would NOT attempt to evaluate your subwoofer placement using Ray-Tracing techniques in smaller rooms. When employing wave theory (like FEM), a room is considered as a complex resonator possessing many normal modes of vibration which are excited when a sound source is introduced to the room. The acoustic energy generated by the source acts to excite these room modes with the resulting sound energy residing in the standing waves established in the room. The characteristic frequencies of these vibrations depend on the room size and shape whereas the damping (or absorption) of the resulting waves depends upon the boundary conditions. Thus, every room imposes its own characteristics on to the sound source present

18. Evaluate speaker placement and room acoustics using provided **two FEM** screens and **'Image Method'** screen. **As a rule of thumb, you listen to your speakers above 300Hz and you listen to your room (room modes) below 300Hz.** It is absolutely essential, that you evaluate placement of your loudspeakers in your listening room. The FEM provides you with the most accurate modeling for the low-frequency characteristics of your listening room and seems to be the only practical method for those difficult, non-rectangular rooms. **The Sum-Plot module provides you with a unique "room contribution" analysis function, that sets the standard for low-cost, FEM room acoustics modeling software**.

19. EasyLab – Advanced. Versitile and powerful measurement system and post-processing

1. Analog Measurement Screen

1. Sine-wave tone generator
2. Sine-wave sweep generator
3. Gated sine-wave generator
4. Volume control
5. Sampling rate and smoothing selector
6. SPL and Zin capture and plot amplitude and phase
7. Built-in Hilbert-Bode Transform
8. Curve arithmetic – add, subtract, merge, add diffraction, add mike calibration, export.

2. MLS Measurement Screen

1. MLS generator
2. Time cursor positioning
3. Time-gate window selector
4. Volume control
5. Sampling rate and smoothing selector
6. SPL and Zin capture and plot amplitude and phase
7. Built-in Hilbert-Bode Transform
8. Curve arithmetic – add, subtract, merge, add diffraction, add mike calibration, export.
9. Modified measurement of Zin for low value calibration resistors.

3. Spectrum Analyser Screen

1. Sine-wave, square-wave, pink-noise and white-noise tone generators
2. Average/instant spectrum display
3. Store/Refresh screen modes
4. Line/Bar display modes
5. Sampling rate selector
6. Vertical resolution
7. Volume control

4. Dual Channel Oscilloscope Screen

1. Sine-wave, square-wave, pink-noise and white-noise tone generators
2. Volume control
3. Store/Refresh screen modes
4. Time base resolution
5. Sampling rate selector
6. Vertical resolution and display position
7. Channel selector display
8. Slope Trigger
9. Synch to channel 1 or 2

5. Cumulative Spectrum Decay Screen

1. MLS generator
2. Time cursor positioning
3. Time-gate window selector
4. Volume control
5. Sampling rate and smoothing selector
6. SPL and capture/adjust and plot
7. Time step selector
8. Microphone calibration file

6. **TS Parameters Extraction Screen**
    1. MLS generator
    2. Sampling rate, volume, time window, smoothing selector
    3. Calibration for the reference resistor
    4. Fine Pulse delay correction
    5. Delta Mass Method
    6. Delta Compliance Method
    7. Basis TS parameters estimated without enclosure

7. **Non-linear Parameters Estimation Screen (Volterra Expansion Method)**
    1. Free-standing Distortion Analyzer – sweeps and plots F1, F2, F3,F4 and F5 over designated frequency range
    2. Adjustable sampling rate and volume
    3. Curve-fit process for extracting non-linear coefficients for Volterra series expansion model.

8. **Non-linear Parameters Estimation Screen (Differential Equations Method) + Modeler**
    1. Implements "Exponential Fit" rather than simple quadratic approximation of the BL and Cms curves.
    2. Includes adjustment of 4 parameters for BL(x) and Cms(x).
    3. Simpler and quicker to use.

9. **RLC Meter**
    1. MLS Generator + controls.
    2. "Curve-fitting" techniques for extracting component's value.

We hope, that the above description is simple to follow, yet practical, as it is intended to keep you **focused on the design issues, that SoundEasy can solve for you**. More experienced users usually find, that because of its flexibility, the program can be pushed much further in solving your "non-standard" problems.

Bodzio Software wishes you many successful designs and hopes that you will find SoundEasy a useful design tool. Bodzio Software also wishes to thank all SoundEasy users, who provided valuable feedback on the scope and functionality of the program. As a result of your input, the program is more powerful, more automated and hopefully, simpler to use. Please keep the feedback line busy !!.

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Software Security.

SoundEasy has an in-built hardware Security Key. It comes as a module, which must be plugged into the parallel port of your computer. The module operates on IBM (c) compatible parallel port built to Centronics standards, with DB-25 connector. The key is completely transparent, allowing normal computer communication. The program will not run (dialog box will inform you first) unless the security key is installed. The key should be installed before attempting to run SoundEasy program. Installation procedure does not require the key to be plugged-in.

How to install the key.

To install the Security Key, attach it to a parallel printer port of an IBM (c) (or true compatible) computer. The side to be attached to the port is labelled "Computer". Screws are provided to connect the key securely to the port. The printer cable is then attached to the outside connector of the Security Key.

Using other hardware keys.

Some hardware keys are designed to "sit between" the computer and the printer. Such keys do not pass all signals through to the printer. Only minimum number of lines are passed, which enables proper printer functioning. If SoundEasy Security Key is installed behind the other hardware keys, it may not be able to communicate properly. For this reason, always attach SoundEasy Security Key directly to the printer port, and attach any other security devices to the program's Security Key.

Specific hardware problems.

SoundEasy Security Key has been modelled on the IBM (c) printer adapter, therefore is compatible with the wide range of hardware adhering to the IBM (c) standard for parallel interfaces. However, not all computers properly implement this standard. For example, some computer printer ports do not provide pull-up resistors to power the Security Key. This is common problem with laptop computers. Generally, **if a printer is plugged in and powered ON, it will provide power.** Problems in this area may be corrected by replacing the parallel port with a truly IBM-compatible parallel port, or by installing a second printer adapter card. If a printer is not available, power can usually be provided by an external AC/DC adaptor plug. If the program does not run when the Security Key is installed, check your printer first. If the printer is plugged-in, it may have to be turned ON. This is a known problem with some of the IBM (c) compatibles. If the printer is not plugged-in into the parallel port, but the Security Key is, you should not experience any problems. If the problem still persists, contact Bodzio Software.

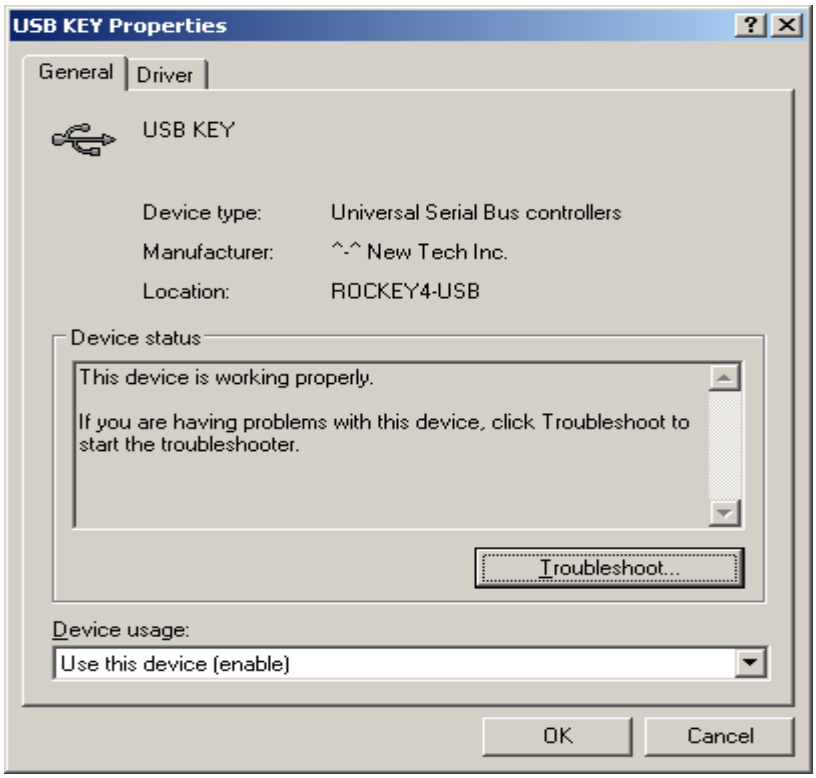
USB Dongle.

You CD-ROM package includes a dongle installation program that automatically detects your operating system and installs appropriate driver. The program is called "**InstDrv.exe**". You can run it from your CD-ROM and all you need to do is to follow a couple of simple steps.

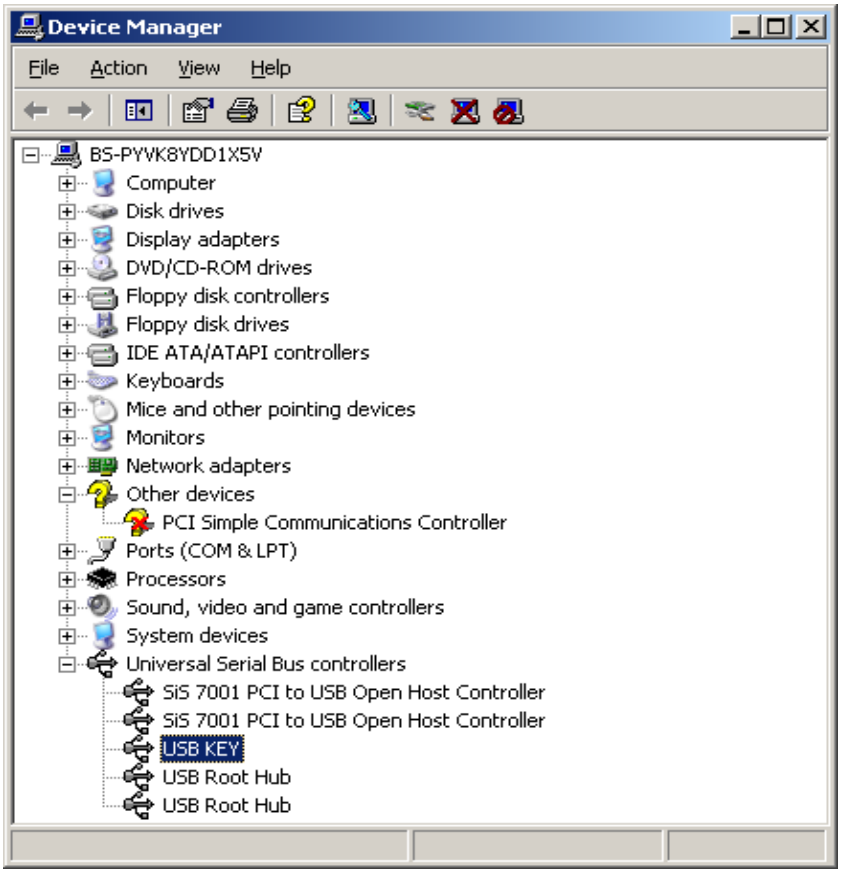
When the program has finished, you should find the USB driver: **RockUsb.sys** placed in your **C:\WINDOWS\system32\drivers** directory. We suggest you use Windows Explorer and check for the presence of this driver.

If you experience error messages related to the USB dongle ("ERR_NO_ROCKEY", indicating, that driver is not recognized by the operating system) when starting SoundEasy, please make sure, that the driver was installed properly and is now recognized by your computer's Device Manager. On WinXP™ system the following may be helpful:

1. Go to "Control Panel" -> "System" -> "Hardware" -> "Device Manager"
2. Select "Universal Serial Bus Controllers" ->"USB Key"
3. Double click on "USB Key", this will open "USB Key Properties" dialogue box.
4. You should see message "This Device is working properly" displayed in the tab "General" – see below.



5. If not, you will be given an option to re-install the dongle driver.
6. Please select the **C:\WINDOWS\system32\drivers** directory for the location of the driver.
7. Re-install the driver. When finished, you should see no question marks (?) next to the USB key – see below.



Computer Related Issues

SoundEasy is using your computer hardware resources as test equipment, and needs timely access to it. The following guidelines should be observed during use of the program:

1. Close all other applications accessing the soundcard that SoundEasy is using. Better yet, close ALL other sound programs.
2. Operation of SoundEasy while the computer is connected to a network may cause acquisition - close the network connections.
3. Don't browse the internet (or even have a browser open).
4. Do not attempt to burn CDROMs during measurements.
5. You may need to disable screensavers (please check Windows documentation for instructions).
6. Make sure, that all sound processing devices and effects, such as 3D sound effects, 4-channel output, equalizers, echo and reverb effects are **DISABLED**.

Possibly the best approach is to exit all other programs when attempting to use measurement system of SoundEasy.

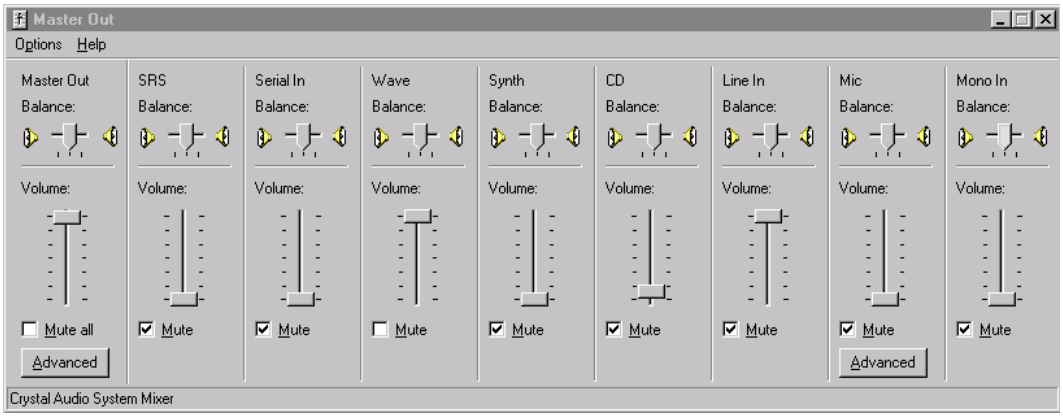
Latest generation SoundBlaster cards (e.g. SBLive) have a very good frequency response but can be used with SoundEasy **only at the sampling frequency of 24 kHz and 48 kHz on some computers** because they work internally always at 48 kHz. Other sampling frequencies are obtained by an internal frequency conversion that can give poor results with MLS measurements - only 48 and 24 kHz sampling frequencies must be used.

Windows operating system is generally quite stable and adequate for audio measurement software. However, some issues may still surface:

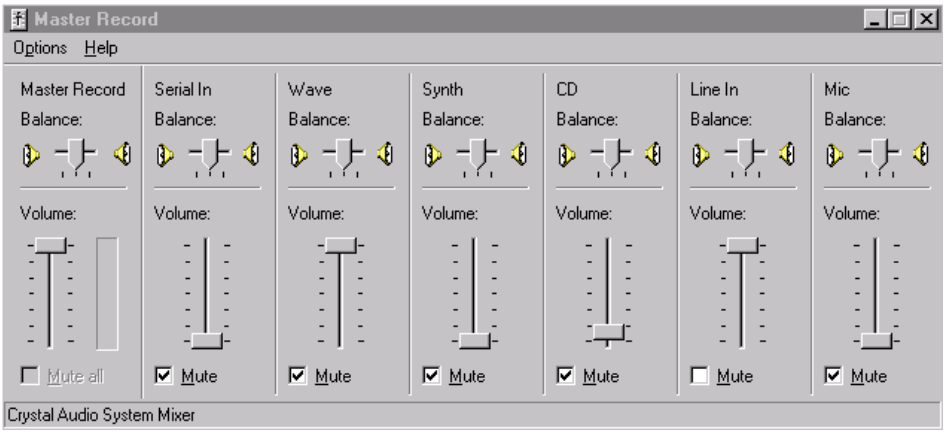
1. Windows2000 may not "inform" SoundEasy software of the correct maximal sampling frequency allowable for a specific sound card. As a result, a sampling frequency not allowable may be selected, causing the system to hang up. See your sound card manual for the maximal allowable sampling frequency.
2. The audio mixer of Windows2000 may set some audio devices, the "Enable"-"Disable" checkboxes to inverted state. SoundEasy attempts to control mixer settings when the measurement system is invoked. This can be disabled via SoundEasy "Preferences" screen. If you suspect, that mixer settings are incorrect, please disallow SoundEasy mixer control and set the mixer manually.
3. Must NOT have hardwired record-to-play or play-to-record audio paths. If these features are included, must be disabled via the system mixer applet.
4. Have good, reliable, and fully-debugged Windows 32 bit drivers.
5. **Because of Windows drivers problems (inability to access the hardware), the Fiji and Pinnacle sound cards are not generally suitable for use with SoundEasy.**
6. **Impulse Response Is NOT Still.** When the measured Impulse Response has a correct shape for a loop-back measure, but does not remain still during the cycle measures, the reasons could be different:
   - The sound card works internally at 48 kHz and emulates other sampling frequencies (this is the case, for example, of the SB Live). In this case, for MLS measurements, only 48 and 24 kHz sampling frequencies **must** be used. Other frequencies will produce an unclean and not still Impulse Response.
   - The sound card shares an IRQ with other devices of your system. **NOTE**: this affects every audio software working in your system, so it could be worth to avoid it, even if you will not use SoundEasy. The worst situation concerns IRQ sharing between audio and video devices because during sampling graphic information is often shown and this can cause problems. Please visit http://dmzweb4.europe.creative.com for more complete information on conflict resource on your PC.

The input voltage applied to the on SB Live! Sound card can be as high as 2.2Volts to get the LED bars indicator right to the maximum non-distorted level. In order to obtain good Signal-To-Noise characteristics of the measured signal, you should always try **to maximize the "In" signal level, but without crossing the "Input Level Too High" level**. If the SoundEasy "Preferences" screen is set to control your sound card mixer, SoundEasy will to read all input and output sources and set them accordingly via sound card driver. Some hardware is not completely compatible with the SoundEasy mixer. Should this be your case, **please DISABLE SoundEasy mixer control from the "Preferences" screen** and use the standard Windows™ mixer or the mixer application provided by the audio card manufacturer.

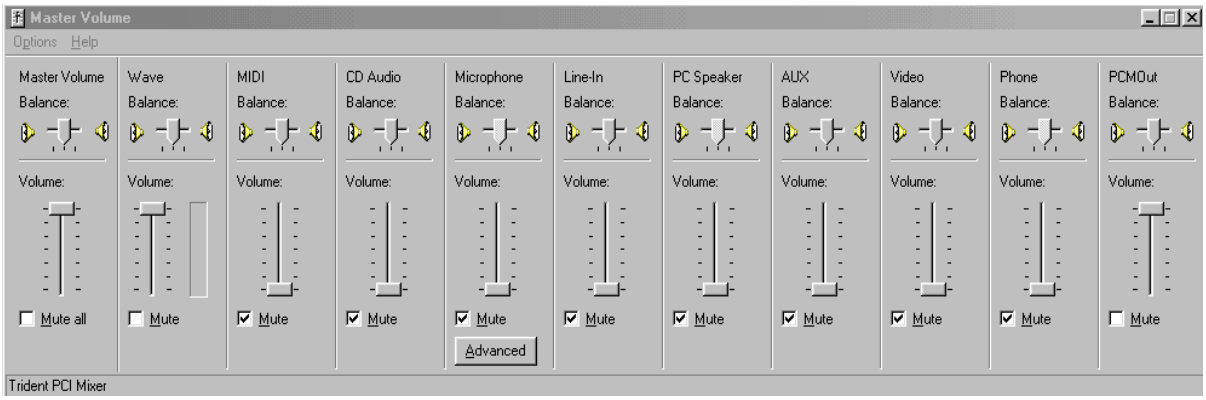
VERY IMPORTANT NOTE



SoundEasy uses "Wave Device" for outputting all signals. Therefore, this device MUST be enabled and volume set to maximum. Also, the "Master Out" volume control MUST be enabled and set to maximum. ALL other inputs or outputs should be disabled and volume control set to minimum – see example above above.



SoundEasy uses "Line-in" for input inputting microphone+pre-amplifier signals. Therefore, this device MUST be enabled and volume set to maximum. Also, the "Master Record" volume control MUST be enabled and set to maximum. The "Mute ALL" box MUST be disabled. ALL other inputs should be disabled and volume control set to minimum – see example above.



Example of TRIDENT sound card mixer settings for Master Volume - Notice PCM Out must be UMNUTED And volume set to full.

In order to perform basic checks shown below, please connect both probes to the output of the sound card. Select the appropriate signal level – close to maximum – and press “Run MLS” button. You should see picture similar to Figure 18.1.

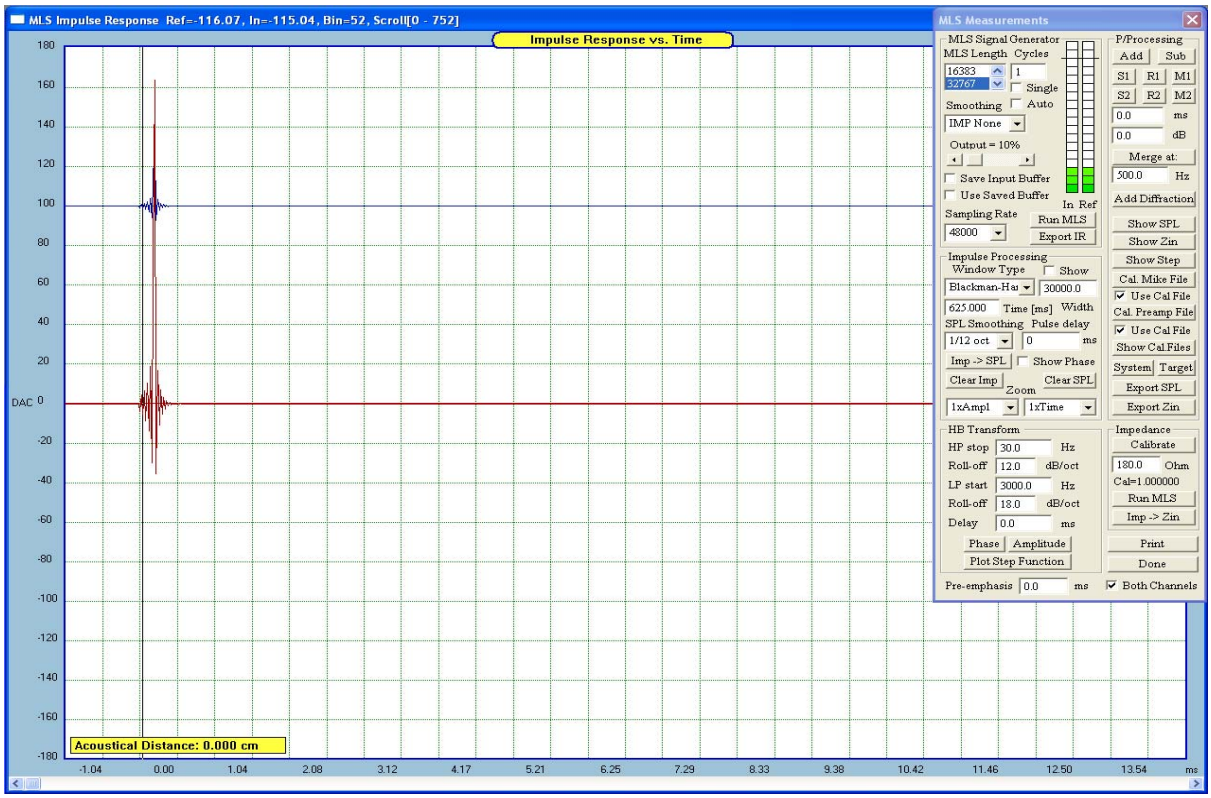
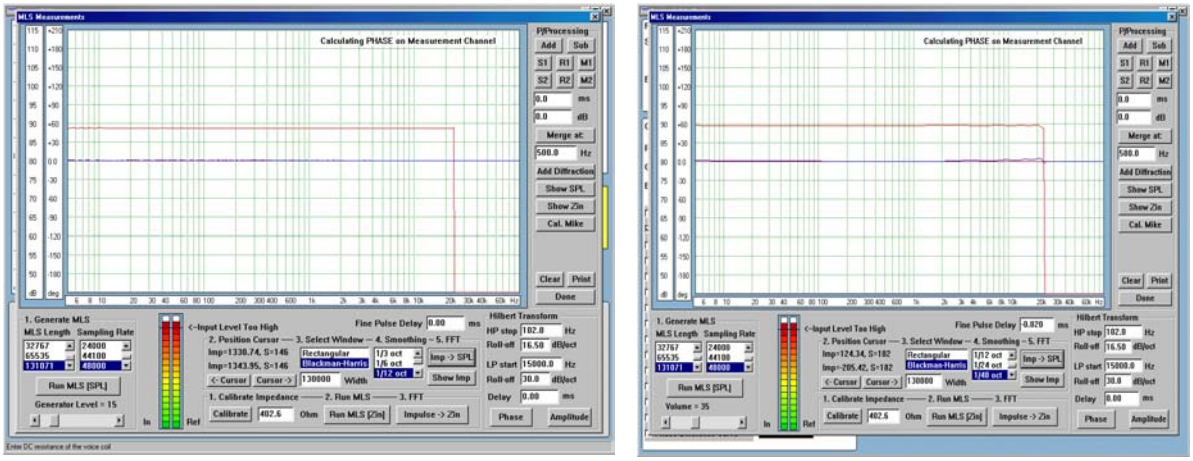


Figure 18.1 Example of impulse response when both probes are connected to power amplifier output.

Please note: BOTH impulses are pointing in the same direction. Now, please press “Imp->SPL” button and you should see pictures similar to Figure 18.2 below



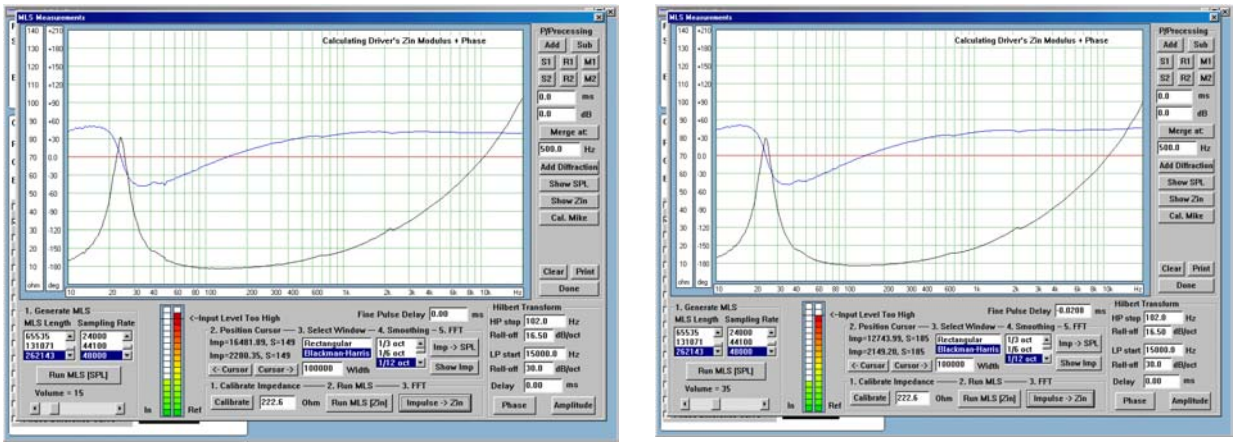
NON-MULTIPLEXED TRIDENT 4D Wave DX MULTIPLEXED CARD

Figure 18.2 Example of amplitude (red) and phase (blue) responses of the multiplexed and non multiplexed sound card.

Please note the negative delay equal to one sampling period (-0.0208 ms for 48kHz sampling frequency) entered into the "Fine Pulse Delay" data field on the multiplexed card. Also, the SB Live! 5.1 inputs need to be inverted to obtain impulse response going up - in the positive direction. Both impulses must be pointing in the same direction – either up or down – for the phase relationship to be correctly computed by the FFT.

It is essential, that you are able to obtain **stable MLS results** from your sound card as indicated on Figures 18.1 and 18.2. Please note, that "Ref" impulse is scaled down about 5 times to leave more screen room for the all important "In" signal. **There is no point conducting further measurements, until you are able to confirm that your hardware works correctly.** Please observe the following points:

1. Both impulse responses must be pointing in the same direction. Otherwise phase relationship will be incorrect.
2. Ideally, both impulses should have similar level, close to the indicated limit. This may not be possible for impedance measurements.
3. For sound cards that **DO NOT** use input multiplexing (eg: Trident Wave), the "Fine Pulse Delay" data field should be set to 0.0 - there is no need for correcting multiplexer-induced delay.
4. For sound cards that **DO** use input multiplexing the "Fine Pulse Delay" data field should be set to **one sampling frequency period** - you must correct for multiplexer-induced delay.
   You need to calibrate for this delay each time you change the sampling frequency.
5. We strongly recommend, that you start checking your sound card by setting the sampling frequency to 48kHz. We had good success ratio with this sampling rate on the sound cards we tested.
6. If you use analog screen for making measurements, the sampling frequency will be a lot less critical and restricted. In fact, we were able to use all sampling frequencies applicable to the given sound card with analog measurement screen. You computer timing and IRQ signaling is practically unimportant when using the analog screen, but you should still be aware of the mixer control issues.
7. Please be aware of mixer incompatibility problems with some sound cards. The recommended way is to disable SoundEasy mixer control and set mixer input-output configuration manually using Windows™ provided mixer control program. Examples were given above. Once you are able to obtain plots as indicated on Figures 16.12 and 16.13, you will know, that your sound card can perform the actual MLS measurements.
8. You will need good quality, shielded leads for the probs. Please note, that if you use resistive dividers with your probes (for instance when making Zin measurements), you will end up with high impedance on both ends of the leads: on one side, there will be the high input impedance of the sound card, on the other side, there will be the high resistance of the resistive divider. Unshielded leads will be prone to pick AC hum and thus distort the measurements.



NON-MULTIPLEXED TRIDENT4D Wave DX MULTIPLEXED CARD

Figure 18.3. Impedance measurements with two different sound cards.

9. When making impedance measurements, we **recommend,** that you set your sound card to a very long MLS (eg: 262k) and the time window to about half of the MLS length - see above. Zin measurements are most often required for a wide frequency range and good frequency resolution.. Also, multiplexed input sound cards need additional compensation entered into "Fine Pulse delay" data entry. **The delay equals (Negative) ONE SAMPLING PERIOD (eg: -0.0208 ms for 48kHz sampling) . The card on the left is Trident 4D Wave DX and on the right – multiplexed card.**
10. Some sound card will not keep it's own time alignment (necessary for MLS measurements) when volume controls are reduced. You need to keep the relevant input and output volume controls at maximum and use SoundEasy volume control and resistive probe dividers to obtain appropriate signal levels.

Missing Components From Control Boxes

If you PC does not display dialogue boxes correctly, chances are, that you set your **Fonts** to "Large" or **DPI settings** for your fonts to "Large size 120 DPI". Shown below are Win XP "Display Properties" control boxes.

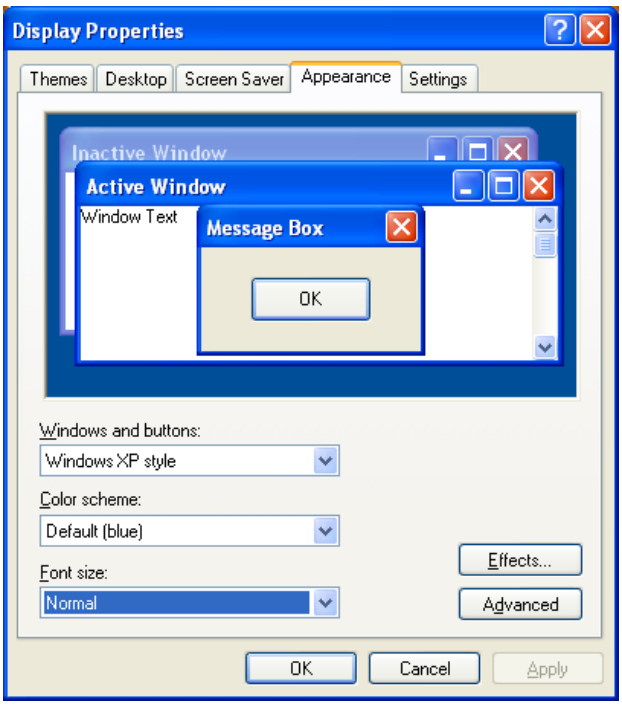
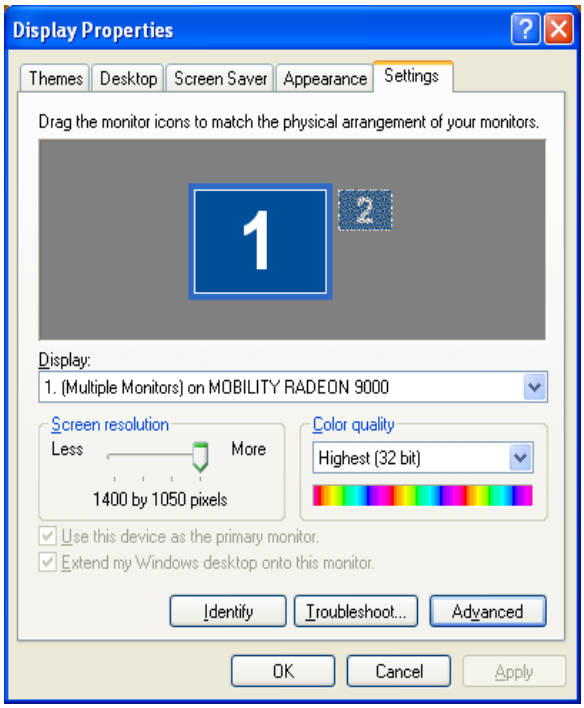


Figure 18.4. Setting Windows XP "Font Size" to **Normal.**



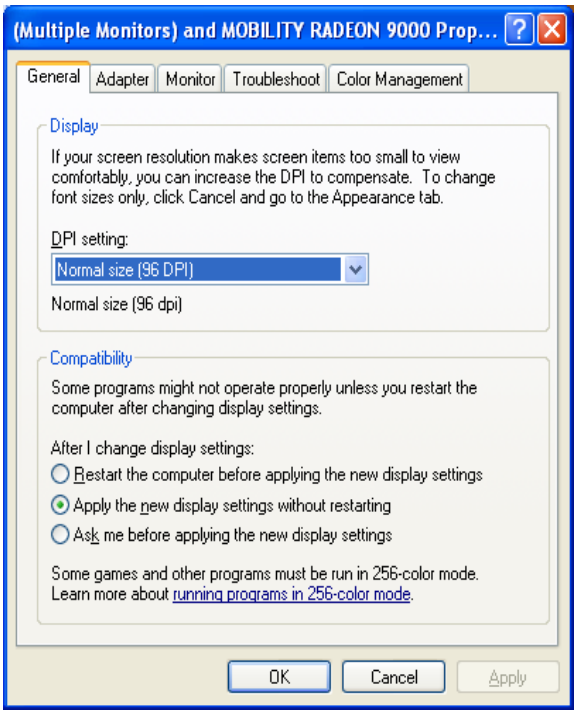


Figure 18.5. Select "Settings" TAB and "Advanced" to go to -----> Select **"Normal size 96 DPI"**.

Sound Card Selection

SoundEasy V10 and later releases, can control the selection of the soundcard from the program's "Preferences" screen. However, should you experience any SoundEasy messages related to "soundcard device handles", as a first step, you may try in addition, to select correct **Default Devices** for Sound playback and Sound recording you wish to use. See settings below for an example of the card you want WinXP to make available to SoundEasy to use. In this case, it is "Sigma Tel Audio" card built-in a DELL Latitude D600 LapTop. The WinXP control box shown below is accessed via Control Panel -> Sound and Audio Devices.

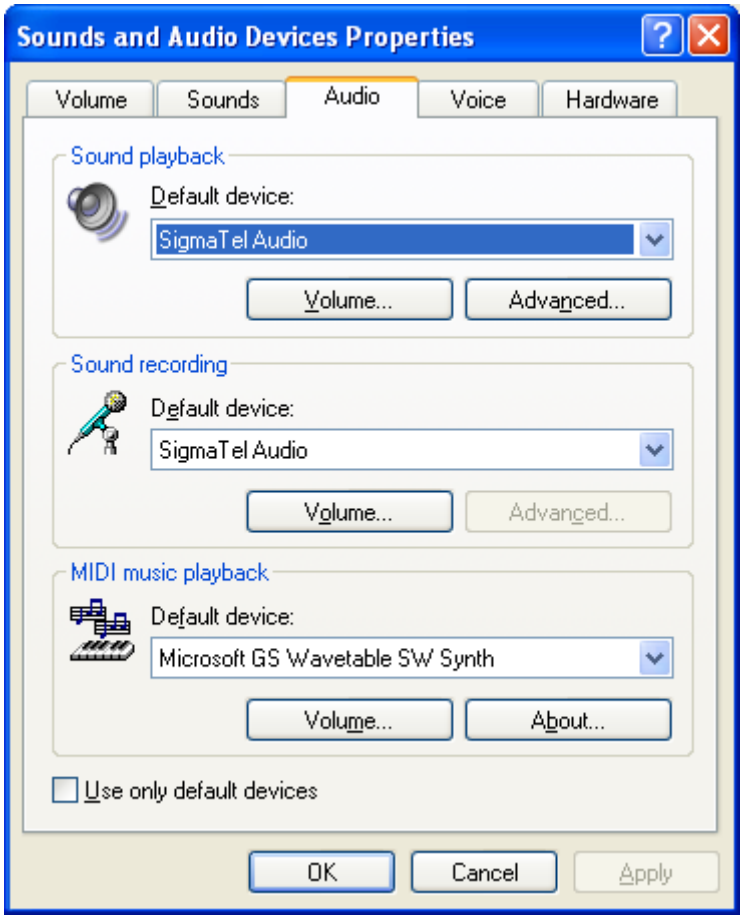


Figure 18.6. Selecting Default Audio Devices.

1. EasyLab Measurement System is expected to work with all standard, 2-channel sound cards. This could be PCI-type cards found in desk-top PC or sound cards built-into the motherboard in case of laptop devices.

2. Digital Filter and Digital Equalizer should also run on standard sound cards for 1x2-way system. configuration. You can therefore emulate simple system and then decide if this type of functionality would interest you. If yes, you can progress into more advanced, multi-channel sound cards. There are basically three sound cards compatible with DE/DF function: Delta-410, Firewire-410 and Echo sound cards.

3. Digital Crossover, the Behringer DCX2496 is programmed via serial port, RS232. You can only use this device as the digital crossover, compatible with Modular Component concept of SoundEasy. Current settings allow for 2x2-way configuration or 2x3-way configuration. Please note, that the DCX2496 is NOT a WAVE-type sound card device, therefore, it can not be used for EasyLab measurements.