Chapter 18. Recommended Design Sequence – typical workflow.

As discussed in Chapter 1, SoundEasy modules/functions can (but do not have to) be used as a sequential set of tools that process incoming information using associated calculators, and pass the results to the next tool. Here is a brief look (a high-level overview) at the recommended design sequence, which describes some of the most important functions of SoundEasy. The time spent with each activity may vary, depending on the design accuracy you wish to achieve.

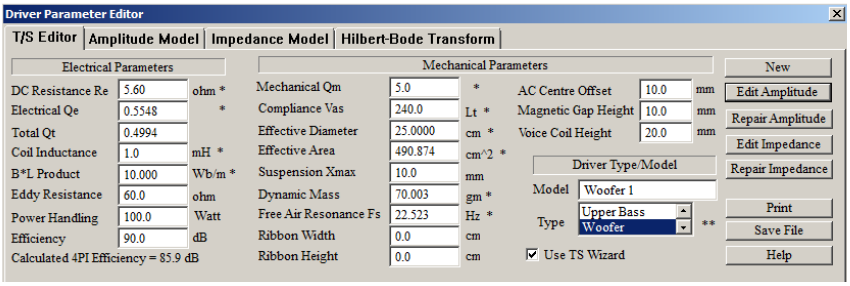
Because of the versatile way the program was designed, you may use it in as many ways as your personal preferences would dictate. But in the simplest way, the top-level overview of one of the possible design workflows may look like this:

1. Enter T/S parameters into Driver Editor.
2. Design enclosure (Fb, Vb) based on T/S parameters from (1) using Enclosure Design.
3. Design vent (length, diameter) for given Vb, Fb from (2) using Vent Calculator.
4. Plot SPL/Phse and Zin/Phase transfer functions from (2) using Enclosure Design.
5. Design front baffle (width, height) using Front Baffle Design/Diffraction.
6. Calculate diffraction for this baffle based on (5).
7. Calculate all major mechanical dimensions of the Rectangular Box, including bracing based on (5).
8. Design/Calculate filter for this driver based on transfer functions plotted in (4) using CAD Schematic Editor.
9. Evaluate the complete design for off-axis SPL/Phase, time domain performance and polar performance using Front Panel Layout functions..
10. Document the design by saving driver file.

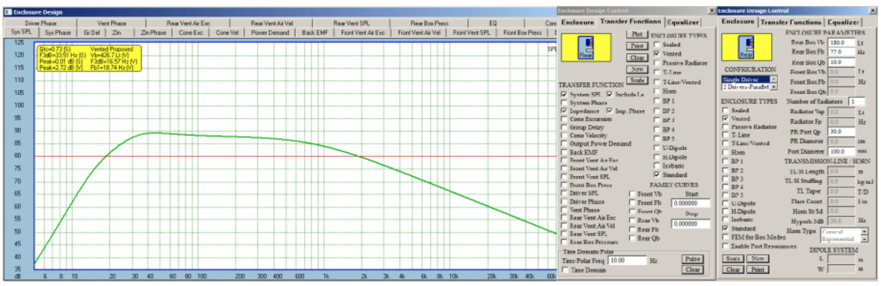
Here is an expanded description of the above workflow.

Single-Driver Design

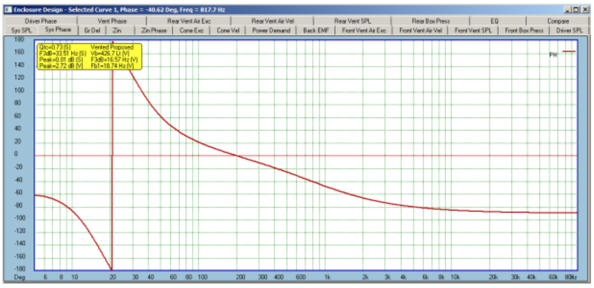
The process starts with **entering all known T/S parameters**. “Driver Tools” -> “Driver Editor” -> “T/S Editor TAB”. Here, you can also use provided test file: **Test_21_W7_1.wfr or Test_21_W7_2.wfr**

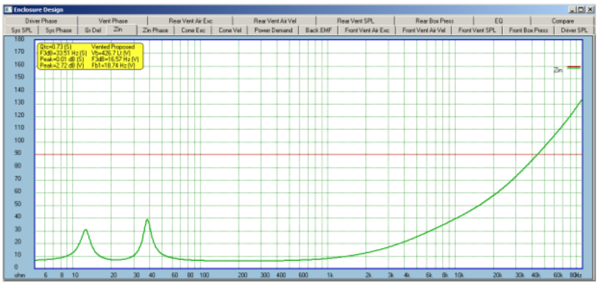


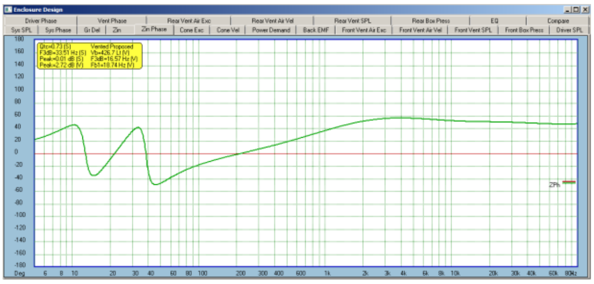
Design your enclosure using **Enclosure Design** system. This process will allow you to determine optimal enclosure volume (Vb) and enclosure tuning (Fb) for each of the open-back drivers.



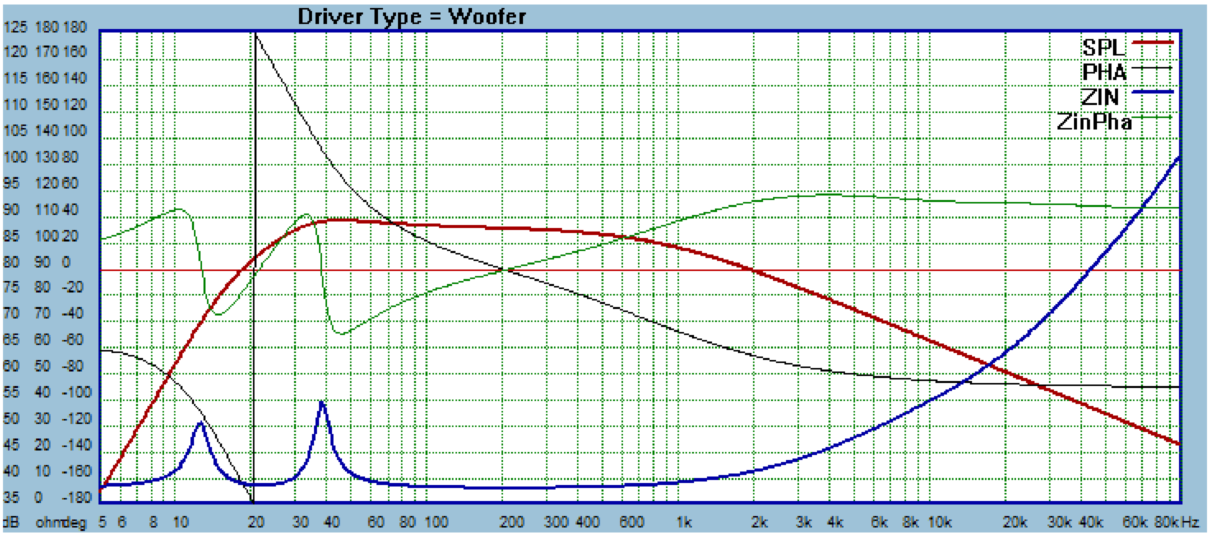
If you select “System SPL”, “System Phase”, “Impedance” and “Impedance Phase” as a selection of plotted transfer functions, and all these will be plotted over the same “Box Frequency Range” as the “System Frequency Range” (selectable from Preferences screen), you can then use these modeled curves as the amplitude and impedance transfer function of your modeled driver.







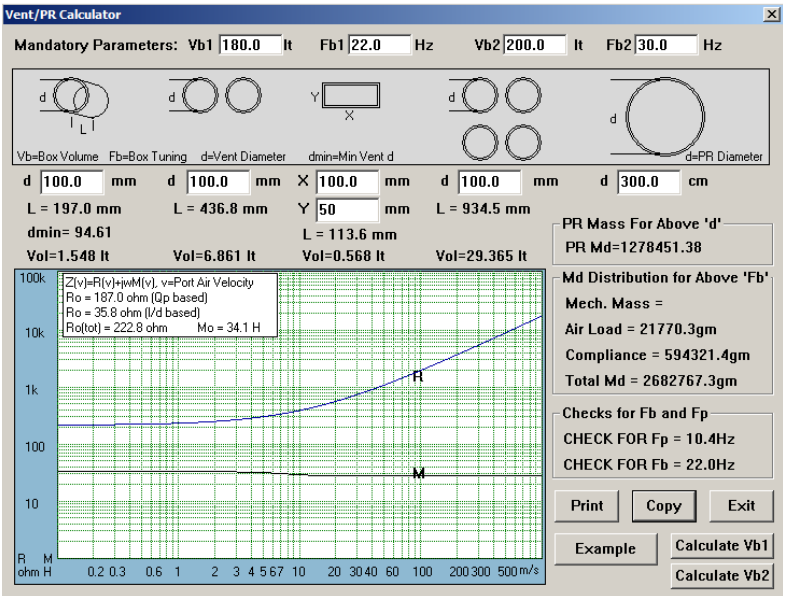
This can be accomplished by going to “Driver Tools” -> “Driver Editor” -> “Amplitude Model” TAB and pressing “Get SPL + Zin” button. You can then review if everything is correct by confirming, that the newly plotted curves have been transferred to driver file. Go to “Driver Tools” -> “Show Default Driver Data”, and you should see the curves as they are shown below.



Please note, that driver transfer function (SPL/Phase) and input impedance (Zin/Phase) can be obtained from the following sources:

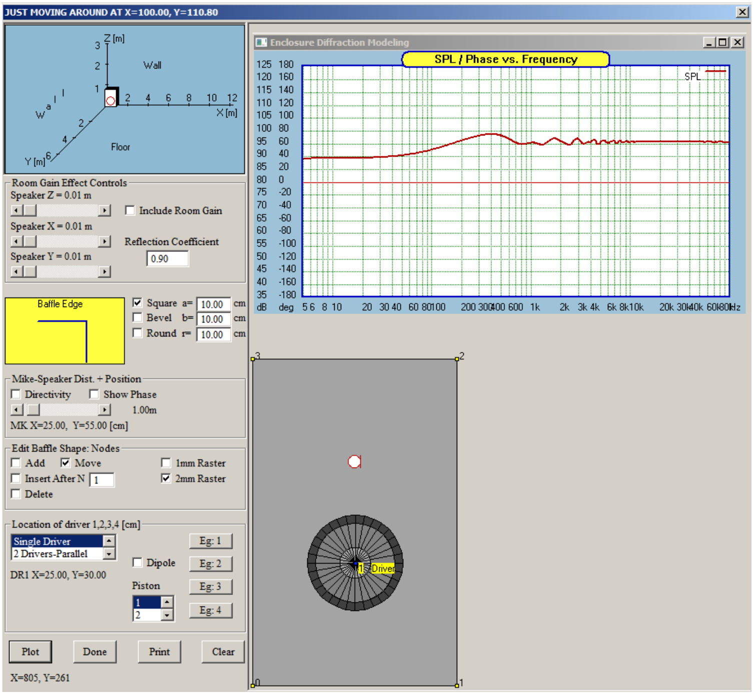
1. Direct MLS measurement
2. Import from other measurement systems
3. Modeled in Enclosure Design system
4. Via HBT method using (1) and (2).

With the Vb and Fb two parameters known, you can now complete port design, length and diameter, using provided **Vent Calculator**.



In the next logical step, you may choose to design front baffle of your loudspeaker, using **Front Baffle Design** dialogue box. Once you are happy with the size and shape of the front baffle, you can quickly **determine diffraction of this baffle**. Diffraction will be used by other sections of the program as well.

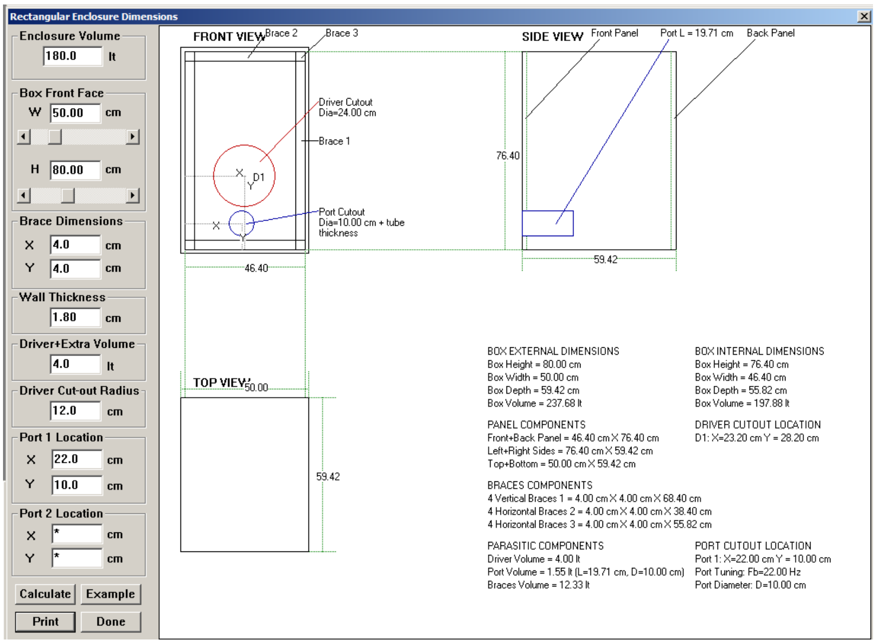
Go to “Enclosure Design” and select “Front Baffle Design/Diffraction”.



Here, you would design the front baffle and place the driver and microphone at the desired locations for calculating diffraction. In the next step, given the front baffle dimensions and required box volume, you can design the actual box and bracing.

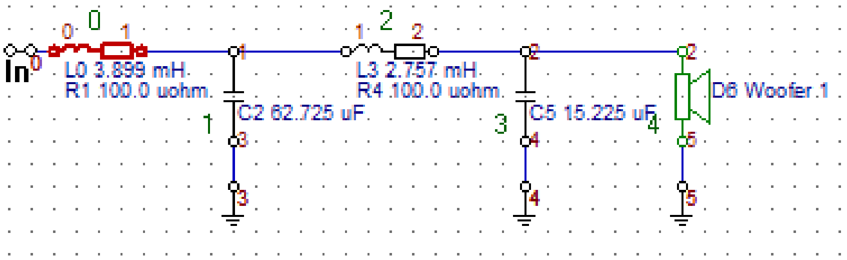
If the enclosure is of a standard type, with a typical rectangular baffle, you can then proceed to **Rectangular Box** menu. Select “Enclosure Tools” -> “Rectangular Box”.

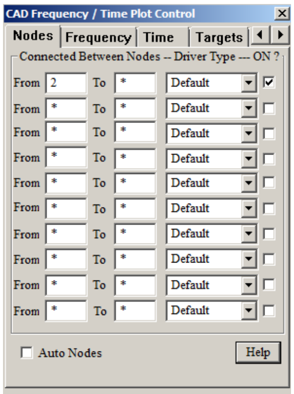
There, you will be able to determine bracing size and dimensions, depth of the enclosure, with all data taking into account driver’s volume, bracing volume and port volume. You are ready to build the prototype of your enclosure now.

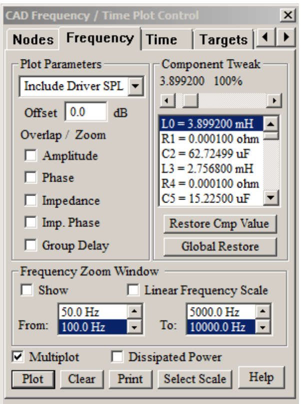


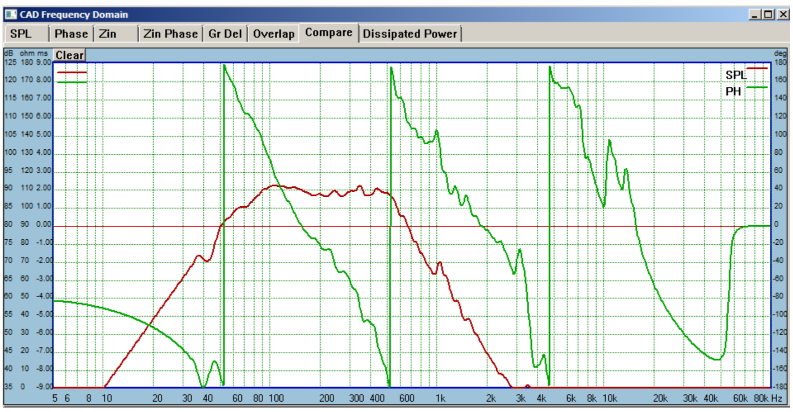
Each driver file may also include individual filter for this driver. This comes handy if you develop any “single driver” loudspeaker - like a subwoofer or a “wideband” loudspeaker. There is simply no need to go through all aspects of setting up full project. Single driver file will do the job just fine. To develop a filter, go to “Crossover Design” -> “CAD Schematic Editor” and “Frequency/Time Domain”.

This includes trying various crossover topologies and the **optimizing** for the best performance of the loudspeaker.







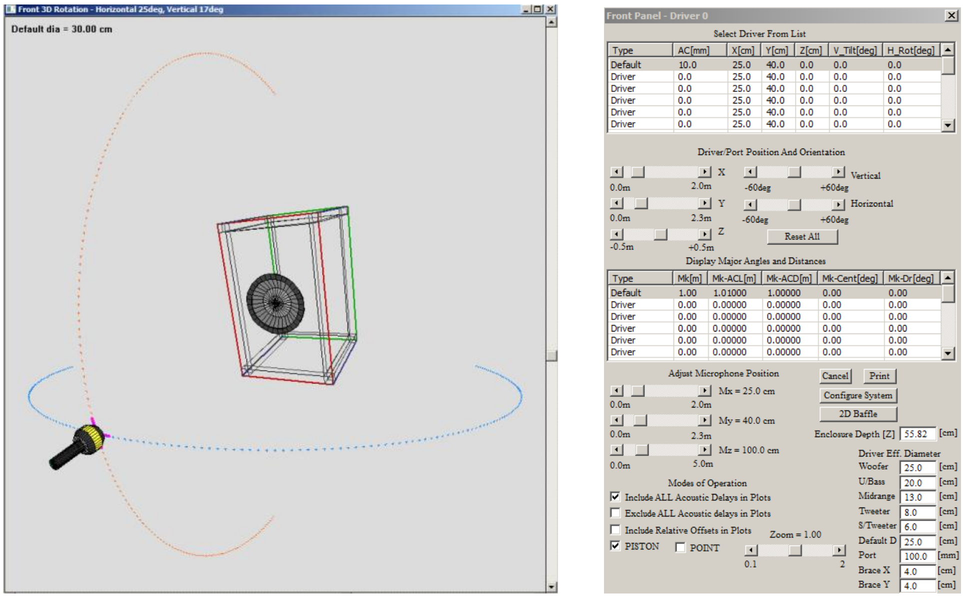


Filter designed for each driver and stored in the driver file, **will be replaced by the complete crossover developed in the project file**. This way, you can freely use driver files in the project file, without concerns about duplicating filters.

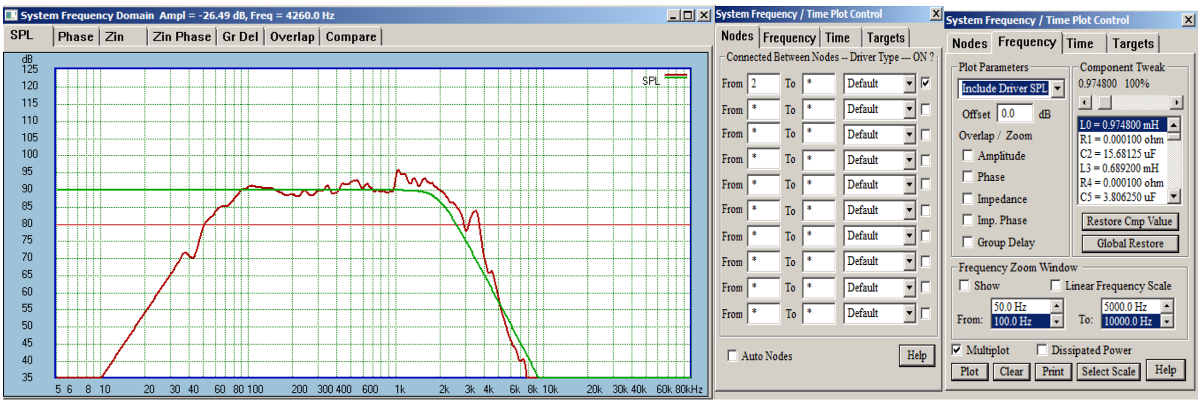
In the next step (on the mechanical path) the dimensions of the front baffle (from **Front Baffle Design**) and bracing size parameters (from **Rectangular Box**) are passed into the **3D System Design**, where you can set-up your test system (your box and measurement microphone) and model on/off axis performance of the whole system.

Select “System Tools” -> “Front Panel Layout”.

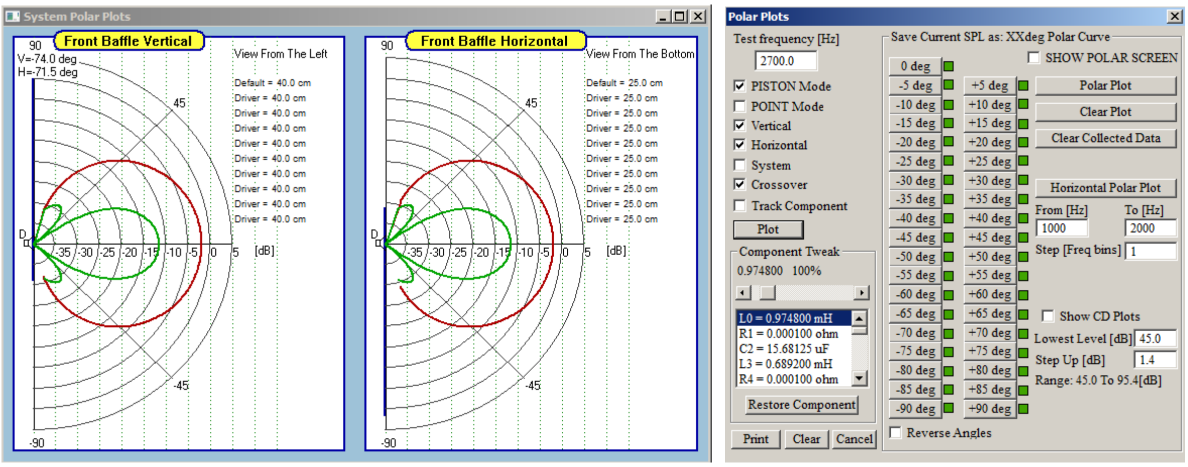
When the screen is first opened, the default viewing reference is right behind the microphone. Therefore, you may need to use the vertical and horizontal screen sliders to re-adjust your viewing reference slightly to the left/right or up/down – see picture below. Driver icon is activated by selecting “Front Panel” -> “Configure System” and checking “Driver Type ON” checkbox.



Frequency-domain performance can be inspected in various way for several settings of the microphone locations and so on – see below…

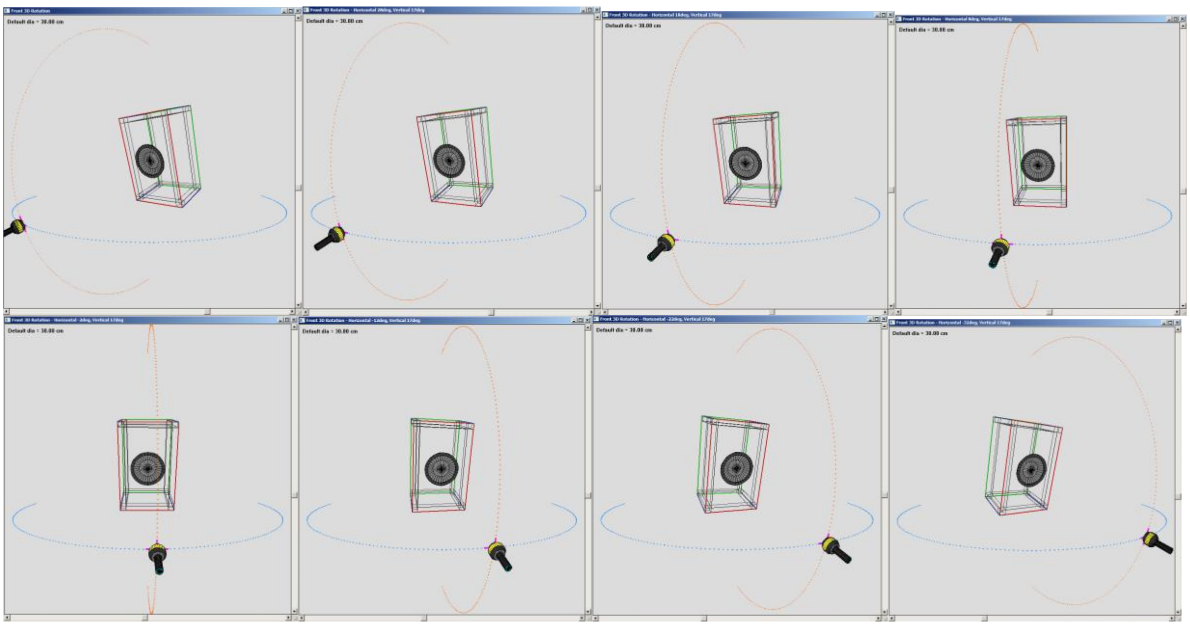


In order to obtain some insight into **polar behavior of the loudspeaker**, go to “System Tools” -> “System Polar Plots”. The “Polar Plots” control dialogue box and a plotting screen will open and allow you to examine various aspects polar performance of the loudspeaker – see below.



For **inspecting time-domain performance**, go to “System Tools” -> “Front Panel Layout” and “System Tools” -> “Frequency/Time Domain” and select “Time” TAB in the control dialogue box – see below..

All of the above performance verifications can be viewed from +/-90deg horizontal and +/-90deg vertical angles – as exemplified below.



Save the driver file now.

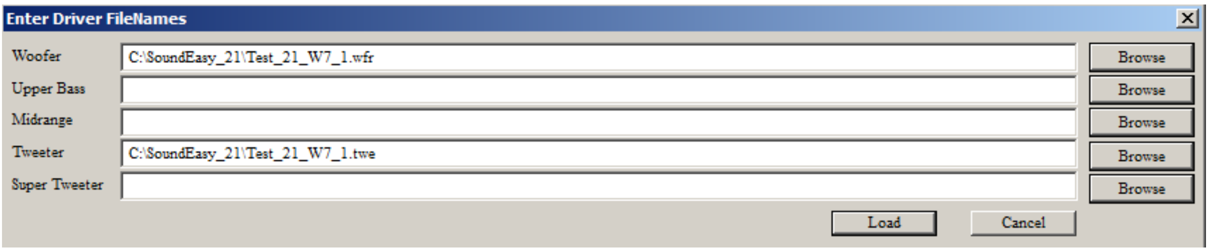
Multi-Way System Design

System Design typically refers to complete system with crossover and starts with obtaining frequency response (amplitude and phase) and input impedance of each driver. This is accomplished as described before - via MLS measurements (recommended), importing this data from a number of other measurement systems or **Enclosure Design** system.

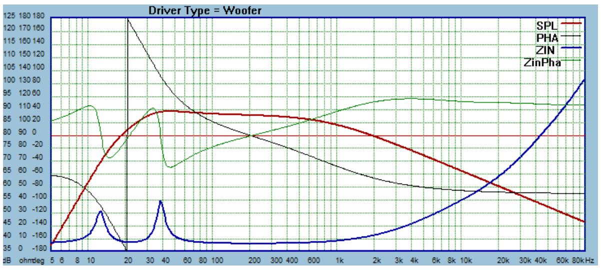
If you choose to model the loudspeaker responses in the **Enclosure Design** system, please plot also the input impedance and phase of the driver in the proposed box. These plots must be executed with the **Box Frequency Range** set to the same range as the **System Frequency Range** (both set in Preferences screen). Save each driver file now.

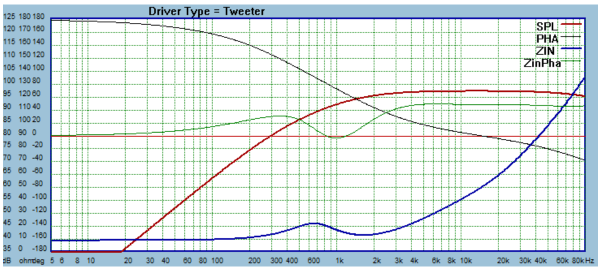
At this stage, you should have all driver files for your loudspeaker system saved on your hard drive. Now, you can create **Project File**, and this will allow you to pull together all previously edited driver files.

You can also load project files: **Test_21_W7_System_2.hif** or **Test_21_W7_System_1.hif**



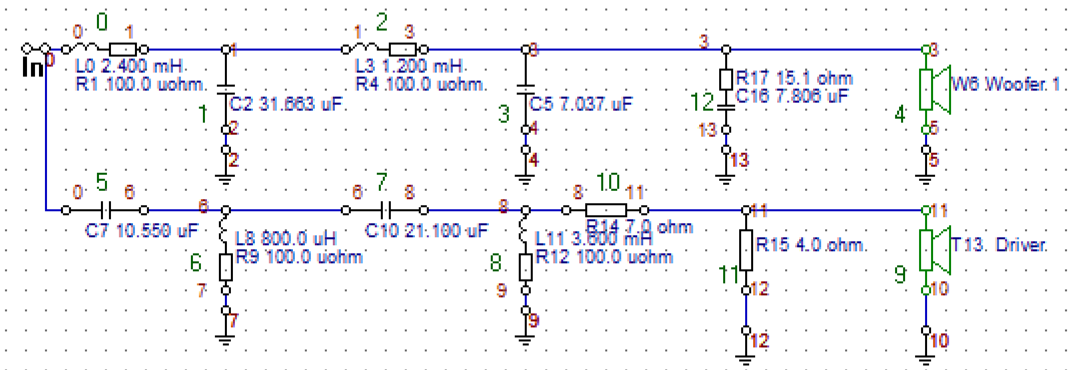
Woofer and tweeter transfer functions **edited in Enclosure Designer** are shown below. When you load the project file, you can review transfer functions (SPL, Phase, Zin, Zin Phase) of all loaded drivers all drivers using “Driver Tools” -> “Show Woofer Data” and so on… This is quite handy feature – see below.

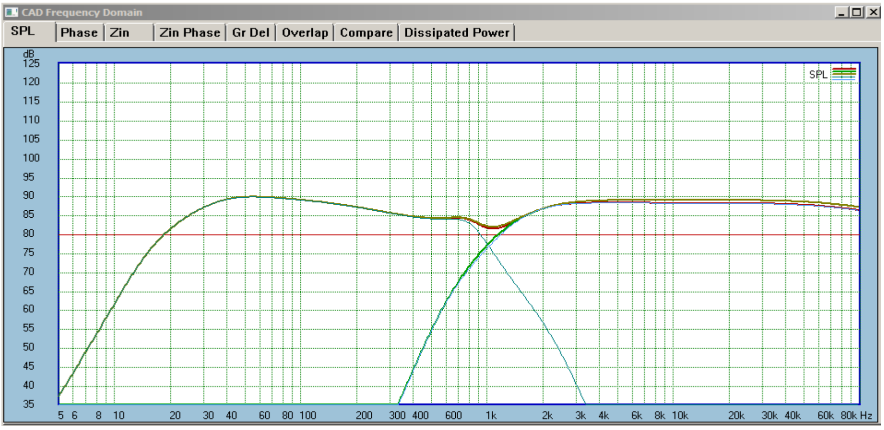




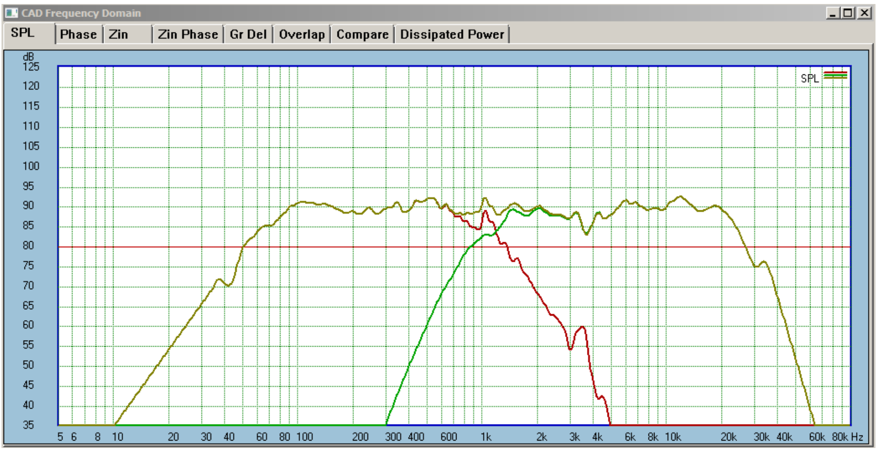
Frequency response and input impedance of the drivers are used by the **Crossover CAD System**, where you will spent a lot of time tweaking the crossover design. There are also many calculators to assist you with this task. To get there, go to “Crossover Design” -> “CAD Schematic Editor” and “Frequency/Time Domain”.

This includes trying various crossover topologies and the **optimizing** for the best performance of the whole system.



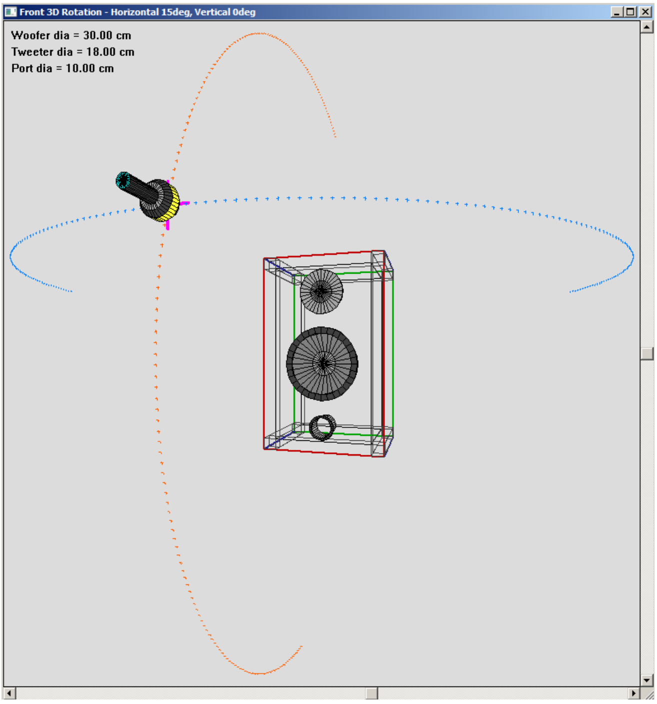


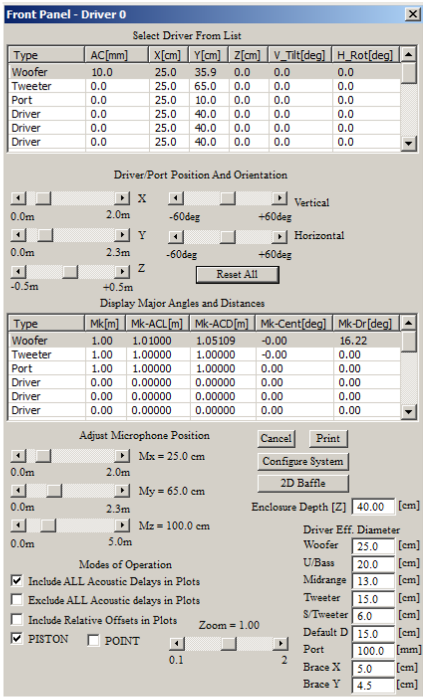
Example of driver files SPL/Zin data created in Enclosure Design screen



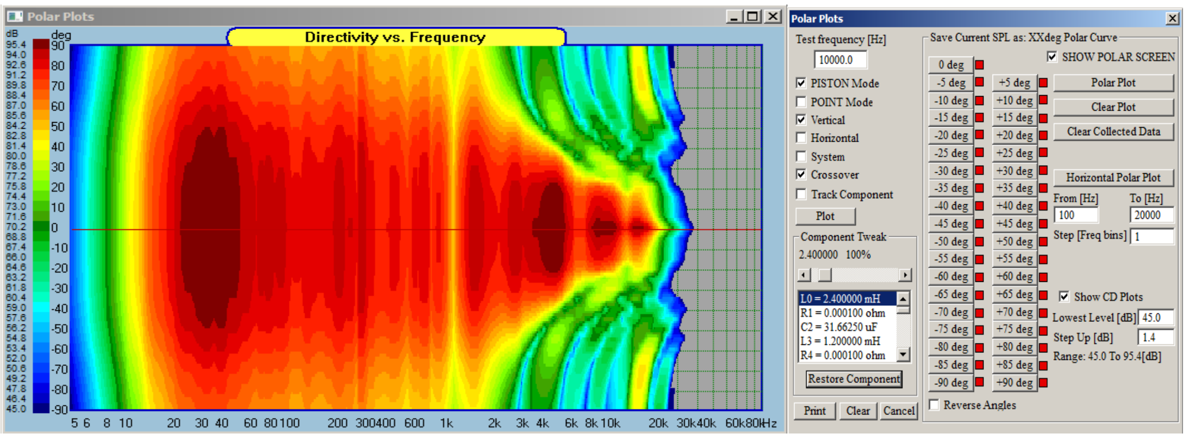
Example of driver files SPL/Zin data measured on real drivers.

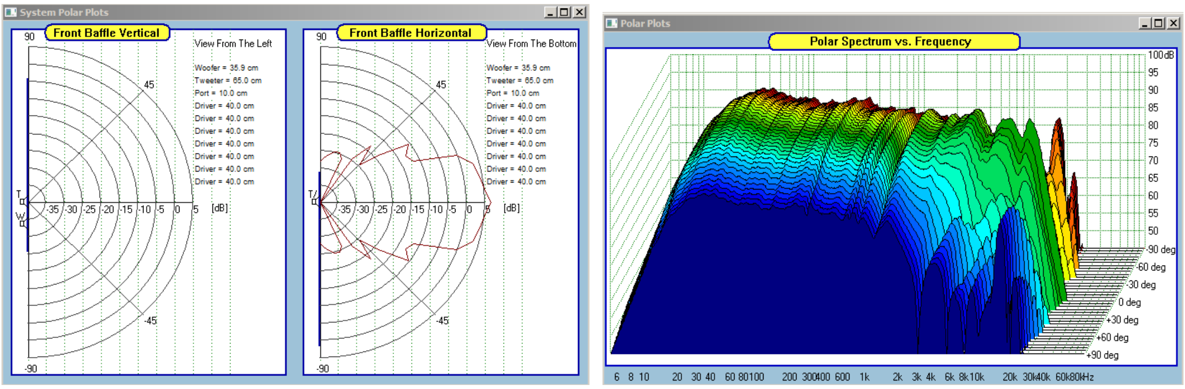
Once crossover is determined, you can move to the final stage of the modeling process – the **3D System Design.** Here, you will be testing and reviewing **on/off axis performance of your system**. The 3D approach allows you to better visualize the measurement process, which may involve multiple microphone locations.





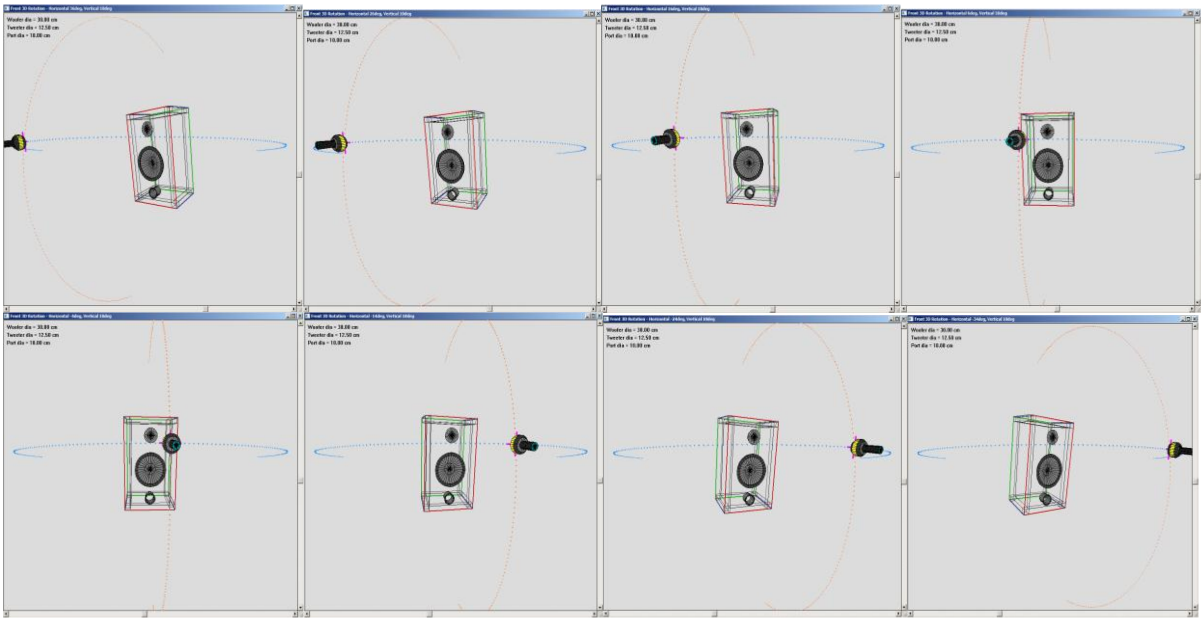
There is a number of other functions you can execute to better analyze your system being modeled/measured.





For **inspecting time-domain performance**, go to “System Tools” -> “Front Panel Layout” and “System Tools” -> “Frequency/Time Domain” and select “Time” TAB in the control dialogue box – see below..

All of the above performance verifications can be viewed from +/-90deg horizontal and +/-90deg vertical angles – as exemplified below.



There is also a very **powerful MLS measurement system** with full post-processing and a number of analysers used to disseminate collected impulse response – but this is a separate subject.

Finally, preferably after gaining some experience with the program, room and car acoustics can also be modeled. Save the project file now.

More Advanced Approach

More advanced users or loudspeaker designers will find significant freedom in using the program, and are able to tune it to their individual needs. This "other way" of using the program has been best described by John Kreskovsky of Music and Design. Quote from diyaudio website:

"…. SoundEasy is a very complete tool set…... True, it won't design a speaker for you or lead you by the hand through the process. What it will do is allow you to look at difference aspects of design w/o first having to do something else. It is very powerful and very flexible, whether you want to design a complete speaker or just an active circuit to address some particular equalization function. Think of it like a carpenters tool box. You may not need every tool in the box for a specific job, but it's likely if you go looking for a tool to do a job will find it in there. In a sense, it's up to the user to organize the tools in a manner most useful to him….."

Here is a more detailed description of some of the functionality built into the program.

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.

System Tools 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. 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.

2. 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

3. 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

4. Cumulative Spectrum Decay Screen / STFT Analysis / Wavelet Analysis

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

5**. 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

6. 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.

7. 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.

8. 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 !!.

Recommended reading.

[1] L.L. Beranek, ACOUSTICS, McGraw-Hill, New York.
[2] P.M. Morse, VIBRATION AND SOUND, McGraw-Hill, New York.
[3] E.R. Geddes, AN INTRODUCTION TO BAND-PASS LOUDSPEAKER SYSTEMS, JAES Vol 37.
[4] LOUDSPEAKERS PART 1, An Anthology, JAES Vol 1 to 25.
[5] LOUDSPEAKERS PART 2, An Anthology, JAES Vol 26 to 31.
[6] A.N. Thiele, MEASUREMENT OF TWEETER PARAMETERS, 22 nd International Convention and Exhibition, Australia 1989.
[7] M. Abramowitz, HANDBOOK OF MATHEMATICAL FUNCTIONS, Dover Publications, New York.
[8] A.H. Davis, MODERN ACOUSTICS, G. Bell & Sons, London.
[9] R.H. Small, CONSTANT-VOLTAGE CROSSOVER DESIGN, JAES Vol 19.
[10] R.M. Bullock, PASSIVE THREE-WAY ALL-PASS CROSSOVER NETWORKS, JAES Vol 32.
[11] P. Grade, ALL-PASS CROSSOVER SYSTEMS, JAES Vol 28.
[12] G.J. Adams, COMPUTER-AIDED LOUDSPEAKER SYSTEM DESIGN PART 1 AND 2, JAES Vol 26.
[13] P.L. Schuck, DESIGN OF OPTIMISED LOUDSPEAKER CROSSOVER NETWORKS USING A PERSONAL COMPUTER, JAES Vol 34.
[14] R.P.De Wit, A.J. Kaizer, J,F Op De Beek, NUMERICAL OPTIMISATION OF THE CROSSOVER FILTERS IN A MULTIWAY LOUDSPEAKER SYSTEM, JAES Vol 34.
[15] S.H. Linkwitz, ACTIVE CROSSOVER NETWORKS FOR NONCOINCIDENT DRIVERS, JAES Vol 24.
[16] W. Waldman, SIMULATION AND OPTIMISATION OF MULIWAY LOUDSPEAKER SYSTEMS USING A PERSONAL COMPUTER, JAES Vol 36.
[17] R.F. Allison, THE INFLUENCE OF ROOM BOUNDARIES ON LOUDSPEAKER POWER OUTPUT, JAES Vol 22.
[18] H.H Rosenbrock, AN AUTOMATIC METHOD FOR FINDING THE GREATEST OR LEAST VALUE OF A FUNCTION, Computer J, Vol 3.
[19] P.R Adby, M.A.Dempster, INTRODUCTION TO OPTIMISATION METHODS, Chapman & Hall, London.
[20] D.J. Wilde, OPTIMUM SEEKING METHODS, Parentice-Hall, NJ.
[21] D.M. Burley, STUDIES IN OPTIMISATION, Intertext Books,
[22] Philips Data Book C3, LOUDSPEAKERS, 1985.
[23] Lorantz Audio Services Data Book, LOUDSPEAKERS 1992, Australia.
[24] H. Flanders, J.J. Price, CALCULUS WITH ANALYTIC GEOMETRY Academic Press, New York.
[25] M. Rossi, ACOUSTICS AND ELECTROACOUSTIC, Artech House, 988.
[26] V. Dickason, LOUDSPEAKER DESIGN COOKBOOK, The Marshall Jones Co. 1987.
[27] K.C Ng, ELECTRICAL NETWORK THEORY, Pitman, London 1977.
[28] J. Borwick, LOUDSPEAKER AND HEADPHONE HANDBOOK, Butterworth & Co.
[29] C.B. Rorabaugh, CIRCUIT DESIGN AND ANALYSIS FEATURNIC C ROUTINES, Tab Books 1992.
[30] M. Colloms, HIGH PERFORMANCE LOUDSPEAKERS, Pentech Press, 1991.
[31] D.E. Johnson, ELECTRICAL CIRCUIT ANALYSIS, Prentice-Hall, 1989.
[32] L. Beranek, ACOUSTICAL MEASUREMENTS, Acoustical Society of America, 1988.
[33] J.D. Turner, ACOUSTICS FOR ENGINEERS, MacMillan, 1991.
[34] J. Vlach, COMPUTER METHODS FOR CIRCUIT ANALYSIS AND DESIGN, Van Nostrand, 1992.
[35] A. Ioinovici, COMPUTER-AIDED ANALYSIS OF ACTIVE CIRCUITS, Marcel Dekker, 1990.
[36] L. Chua, COMPUTER-AIDED ANALYSIS OF ELECTRONIC CIRCUITS, Prentice-Hall, 1990.
[37] H. Press, NUMERICAL RECIPES IN C, Cambridge Univarsity Press, 1992.
[38] R. D. Strum, DISCRETE SYSTEMS AND DIGITAL SIGNAL PROCESSING, Addison-Wesley, 1989.
[39] G. James, ADVANCED MODERN ENGINEERING MATHEMATICS, Addison-Wesley, 1993.
[40] P. Embree, C LANGUAGE ALGORITHMS FOR DIGITAL SIGNAL PROCESSING, Prentice-Hall,1991.
[41] C. P. Bernardin, C/MATH TOOLCHEST FOR ENGINEERING AND SCIENTIFIC APPLICATIONS, Prentice-Hall,1993.
[42] S. Nakamura, APPLIED NUMERICAL METHODS IN C, Prentice-Hall,1993.
[43] R. Kielkowski, INSIDE SPICE, McGraw-Hill, 1994.
[44] M. E. Van Valkenburg, ANALOG FILTER DESIGN, Holt, Rinehart & Winston, 1989.
[45] J. Helszjan, SYNTHESIS OF LUMPED ELEMENT, DISTRIBUTED AND PLANAR FILTERS, McGraw-Hill 1989.
[46] W. W. Seto, ACOUSTICS, McGraw-Hill, INC. 1970.
[47] J. Middlehurst, PRACTICAL FILTER DESIGN, Prentice-Hall, 1993.
[48] S. D. Stearns, R.A. David, SIGNAL PROCESSING ALGORITHMS USING FORTRAN AND C. Prentice-Hall, 1993.

[49] L.E.Kinsler, A.R. Frey, A.B.Coppens & J.V. Sanders, FUNDAMENTALS OF ACOUSTICS, John Wiley & Sons, 1982.
[50] O.C. Zienkiewicz, R.L. Taylor, THE FINITE ELEMENT METHOD, Vol 1, Vol 2 McGraw-Hill, 1991
[51] W.B. Bickford, A FIRST COURSE IN THE FINITE ELEMENT METOD, IRWIN, Boston 1990.
[52] R. D. Cook, D. S. Malkus & M.E. Plesha, CONCEPTS AND APPLICATIONS OF FINITE ELEMENT METHOD, John Wilet & Sons, Canada, 1989.
[53] M. Petyt, INTRODUCTION TO FINITE ELEMENT VIBRATION ANALYSIS, The Bath Press, Avon, 1990.
[54] G. R. Buchanan, FINITE ELEMENT ANALYSIS, McGraw-Hill, 1995.
[55] R. J. Bernhard, A FINITE ELEMENT PROCEDURE FOR DESIGN CAVITY ACOUSTICAL TREATMENT, J. Acoustic Society of America, 83, June 1988.
[56] S. Sakai, Y. Kagawa and T. Yamabuchi, ACOUSTIC FIELD IN AN ENCLOSURE AND ITS EFFECT ON SOUND-PRESSURE RESPONSES OF A LOUDSPEAKER, Loudspeakers Vol 3. Journal of the Audio Engineering Society Vol 32-Vol 39.
[57] M. Petyt, J. Lea and G.H. Koopman, A FINITE ELEMENT METHOD FOR DETERMINING THE ACOUSTIC MODES OF IRREGULAR SHAPED CAVITIES, Journal of Sound and Vibration, Vol 45, 1976.
[58]. J. Vanderkooy and S.P. Lipshitz, "Power Response of Loudspeakers with Noncoincident Drivers - The Influence of Crossover Design", 74th Convention of the AES, New York, 1983.
[59]. M. E. Engebretson, "Low-Frequency Sound Reproduction", JAES, Vol 32, May 1984.
[60]. M.R. Gander and J.M.Eargle, "Measurement and Estimation of Large Loudspeaker Array Performance", JAES, Vol 38, April 1990.
[61] D.B. Keele, "Effective Performance of Bessel Arrays", JAES, Vol 38, October 1990.
[62]. Harry F. Olson, "Direct Radiator Loudspeaker Enclosures", Audio Engineering, November, 1951.
[63]. Ole Lund Christensen, "A practical guide to acoustical design of control rooms and placement of loudspeakers", Copenhagen AES Convention 1989.
[64]. J. BACKMAN, The Nonlinear Behaviour of Reflex Ports. AES preprint 3999.
[65]. J. VANDERKOOY, Nonlinearities in Loudspeaker Ports. AES preprint 4748.
[66]. N.B ROOZEN, J.E.M. VEAL, J.A.M. NIEUWENDIJK, Reduction of Bass-Reflex Port Nonlinearities by Optimizing the Port Geometry, AES preprint 4661.
[67]. A. SALVATTI, D. BUTTON, A. DEVANTIER, Maximizing Performance from Loudspeaker Ports, AES preprint 4855.
[68]. G. TRURSTON and others, Nonlinear Properties of Circular Orifices, JASA, Volume 29, Number 9, 1957.
[69]. A. NOLLE, Small-Signal Impedance of Short Tubes, JASA, Volume 25, Number 1, 1953.
[70]. G. THURSTON, C.MARTIN, Periodic Flow through Circular Orifices, JASA, Volume 25, No 1, 1953.
[71]. D. BIES, O. WILSON, Acoustic Impedance of a Helmholtz Resonator at Very High Amplitude, JASA, Volume 29, Number 6, 1957
[72]. U. INGARD, H.ISING, Acoustic Nonlinearity of an Orifice, JASA, Volume 42, Number 1, 1967.
[73]. U. INGARD, On the Theory and Design of Acoustic Resonators, , JASA, Volume 25, Number 6, 1953.
[74]. G. THURSTON, Periodic Fluid Flow through Circular Tubes, JASA, Volume 24, Number 6, 1952.
[75]. R. SMALL, Direct Radiator Loudspeaker Analysis, An Anthology of Articles on Loudspeakers from JAES Vol. 1 -Vol. 25 (1953-1977)
[76]. V. DICKASON, Loudspeaker Design Cookbook, 5th Edition, Old Colony Sound Lab. 1997.
[77]. JOSEPH D'APPOLITO, Ph.D., Testing Loudspeakers, Audio Amateurs Press, Old Colony Lab.1998.
[78]. J.BORISH and J.B ANGELL, An Efficient Algorithm for Mesuring the Impulse Response Using Pseudorandom Noise, JASA Vol 31, No7, August 1983.
[79]. W.T. CHU, Impulse-Response and Reverberation-Decay Mesurements Made by Using a Periodic Pseudorandom Sequence. Applied Acoustics, 29, 1990.
[80]. L.R.FINCHAM, Refinements in the Impulse Testing of Loudspeakers, JASA, Vol 33, No 3, March 1985.
[81]. D.D. RIFE and J. VANDERKOOY, Transfer -Function Measurement with Maximum-Length Sequences. JASA, Vol 37, No 6, June 1989.

Software Security.

SoundEasy has an in-built hardware Security Key. It comes as a module, which must be plugged into the USB port of your computer. 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 using appropriate driver for your system. 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 the USB port of an IBM (c) (or true compatible) computer.

Using other hardware keys.

Not Allowed.

.

Specific hardware problems.

If the program does not start with the USB Key plugged-in, or a dialogue box pops-up during start-up of the program informing you of an error - please contact Bodzio Software.

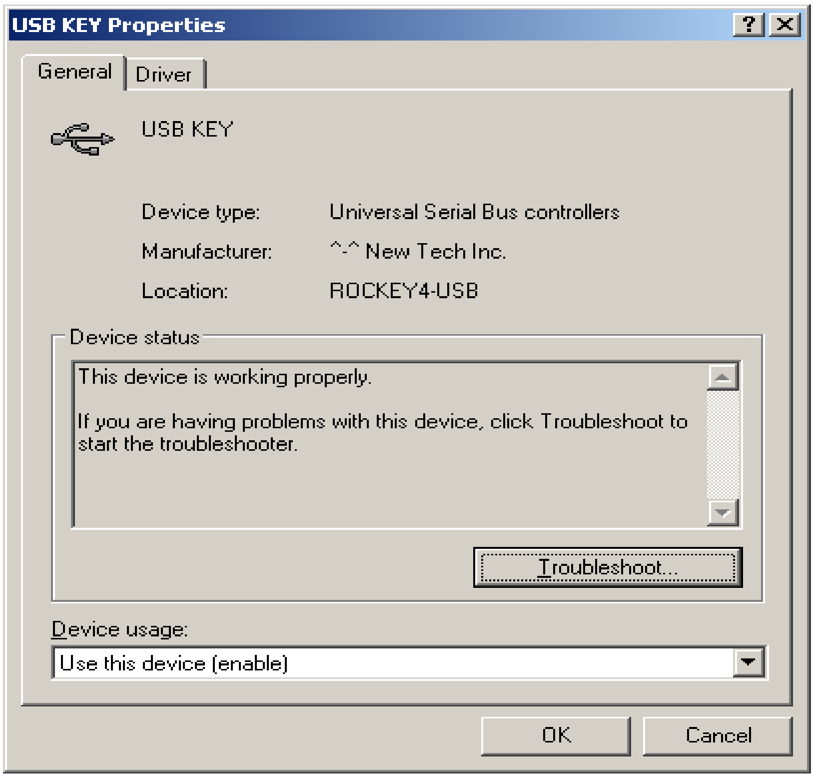
USB Dongle.

Your CD-ROM package includes a dongle installation programs. You need to select the dongle installation program, that is compatible with your operating system. You can run it from your CD-ROM and all you need to do is to follow a couple of simple steps.

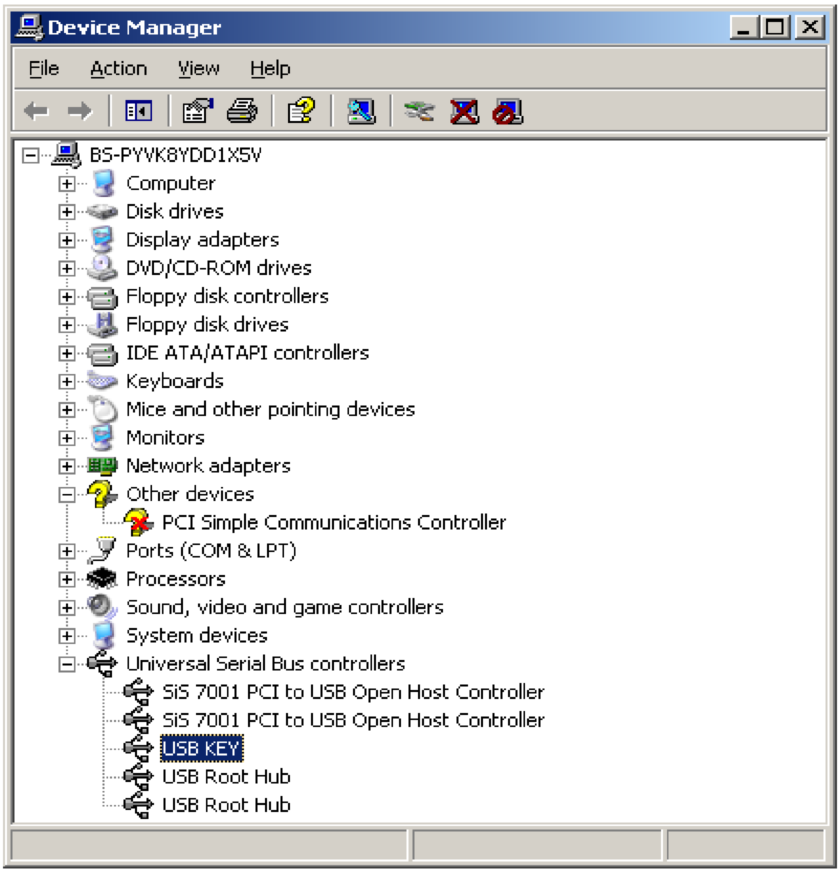
When the program has finished, you should find the USB driver: **RockUsb.sys** placed in your **C:\WINDOWS\system32\drivers** directory. We suggest you use Windows Explorer and check for the presence of this driver.

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.

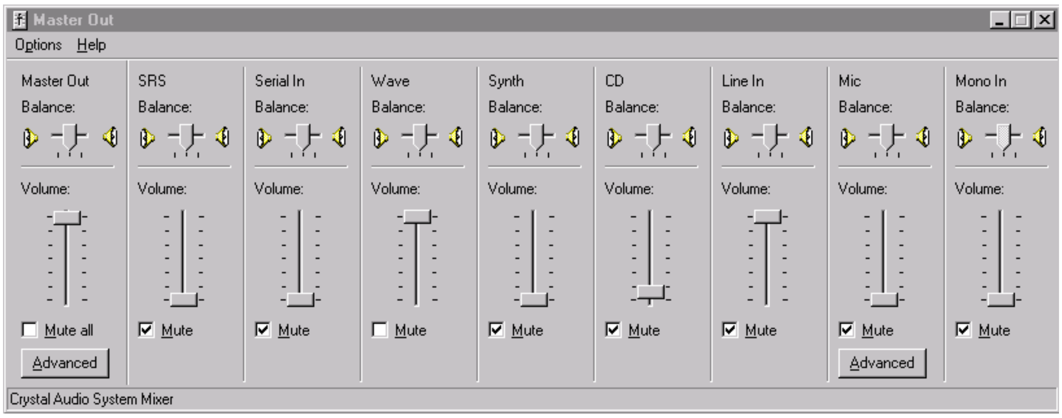
Latest generation SoundBlaster cards (e.g. SBLive) have a very good frequency response but can be used with SoundEasy **only at the sampling frequency of 24 kHz and 48 kHz on some computers** because they work internally always at 48 kHz. Other sampling frequencies are obtained by an internal frequency 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:

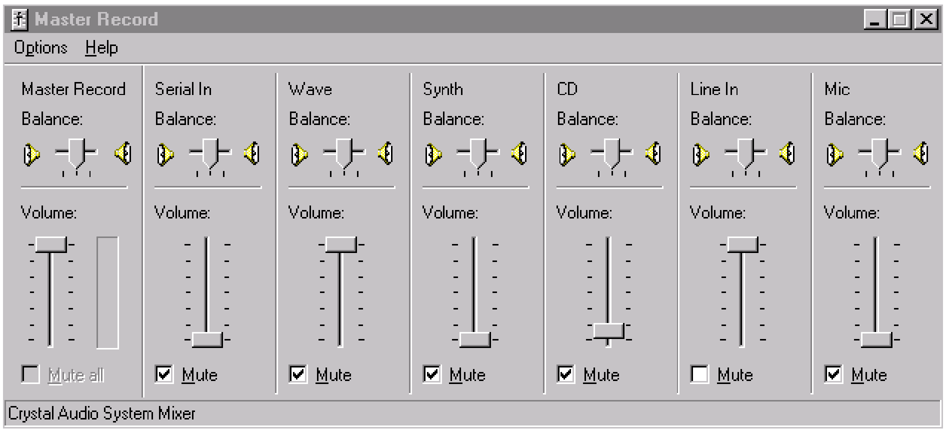
1. Windows2000 may not "inform" SoundEasy software of the correct maximal sampling frequency allowable for a specific sound card. As a result, a sampling frequency not allowable may be selected, causing the system to hang up. See your sound card manual for the maximal allowable sampling frequency.
2. 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.

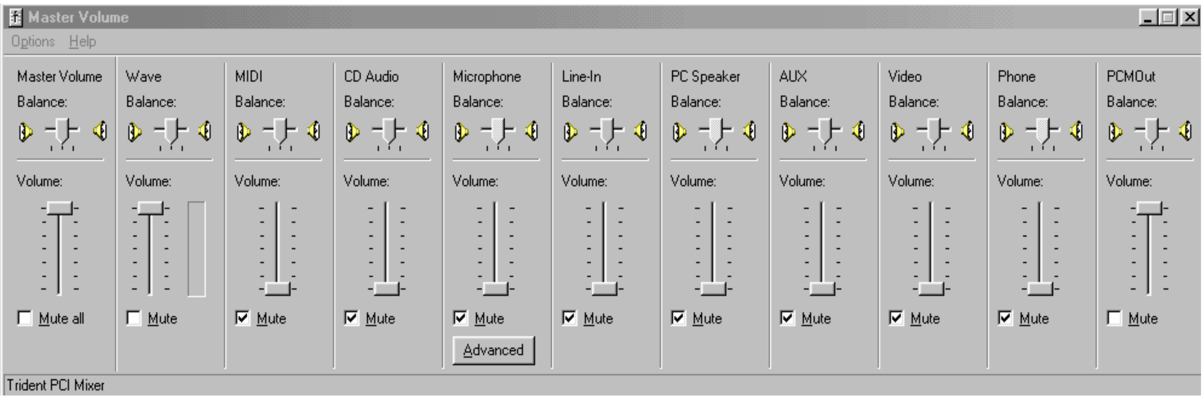
VERY IMPORTANT NOTE



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.

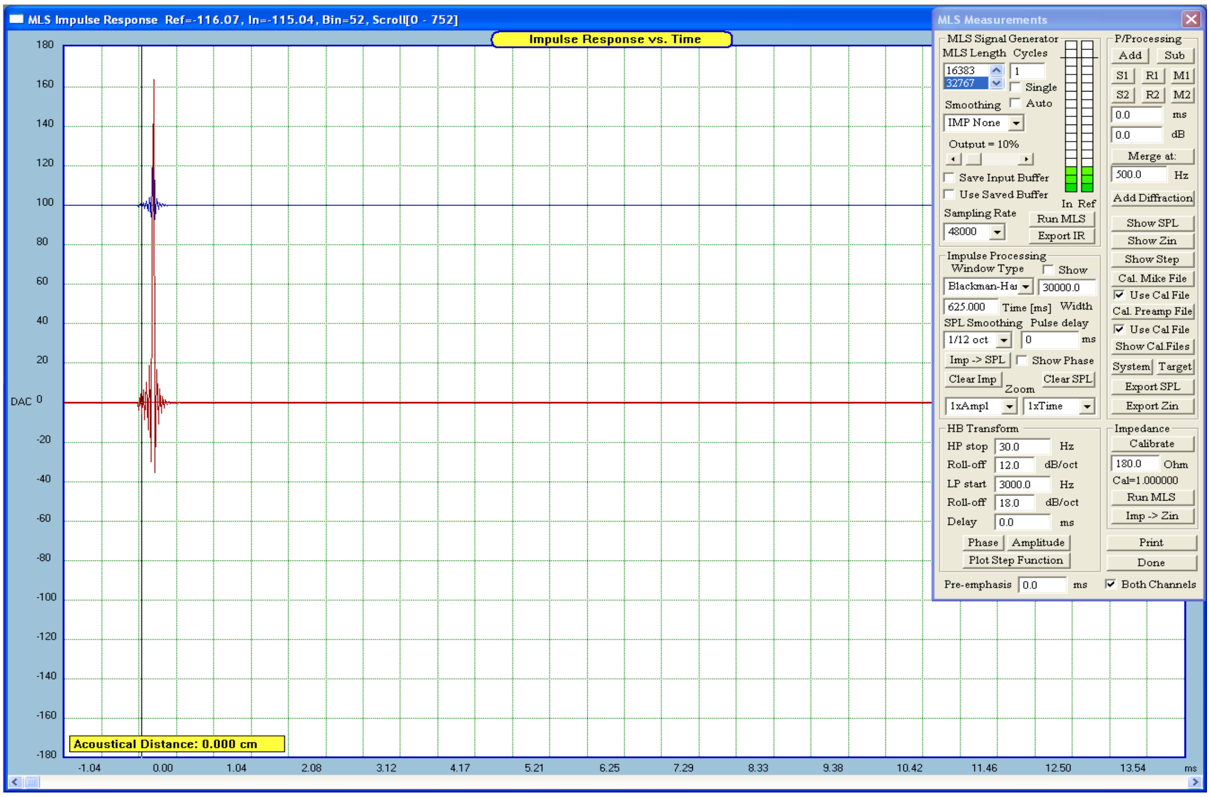
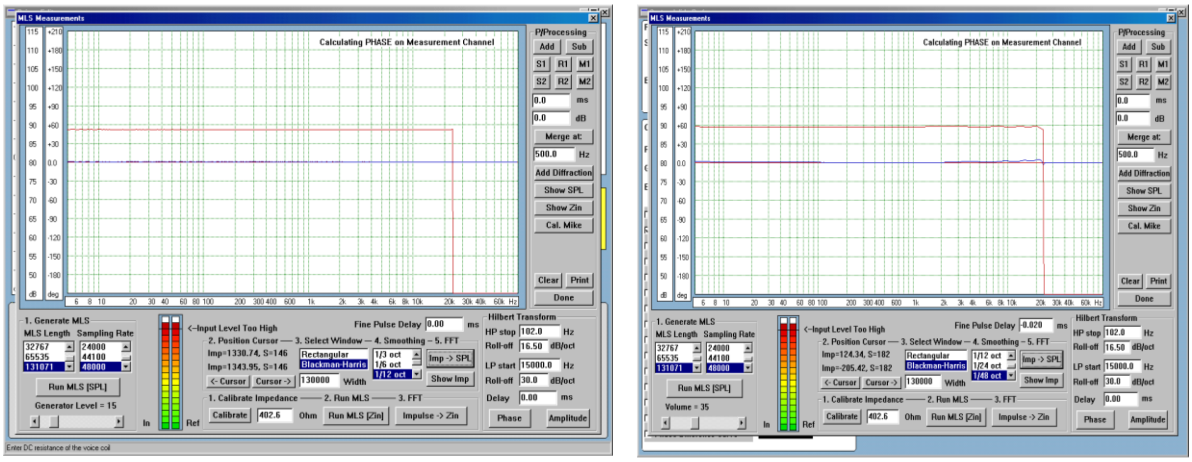


Figure 18.1 Example of impulse response when both probes are connected to power amplifier output.

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



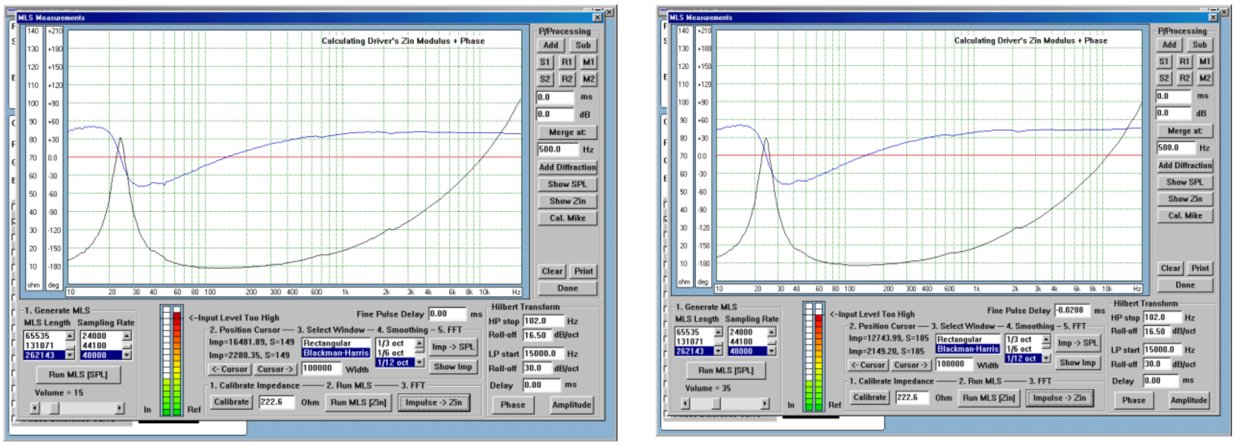
NON-MULTIPLEXED TRIDENT 4D Wave DX   MULTIPLEXED CARD

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.

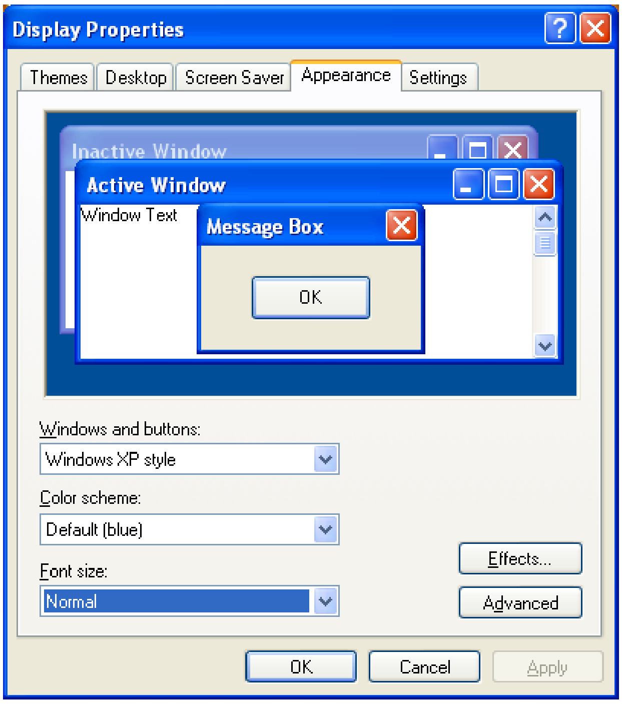


Figure 18.4. Setting Windows XP “Font Size” to **Normal.**

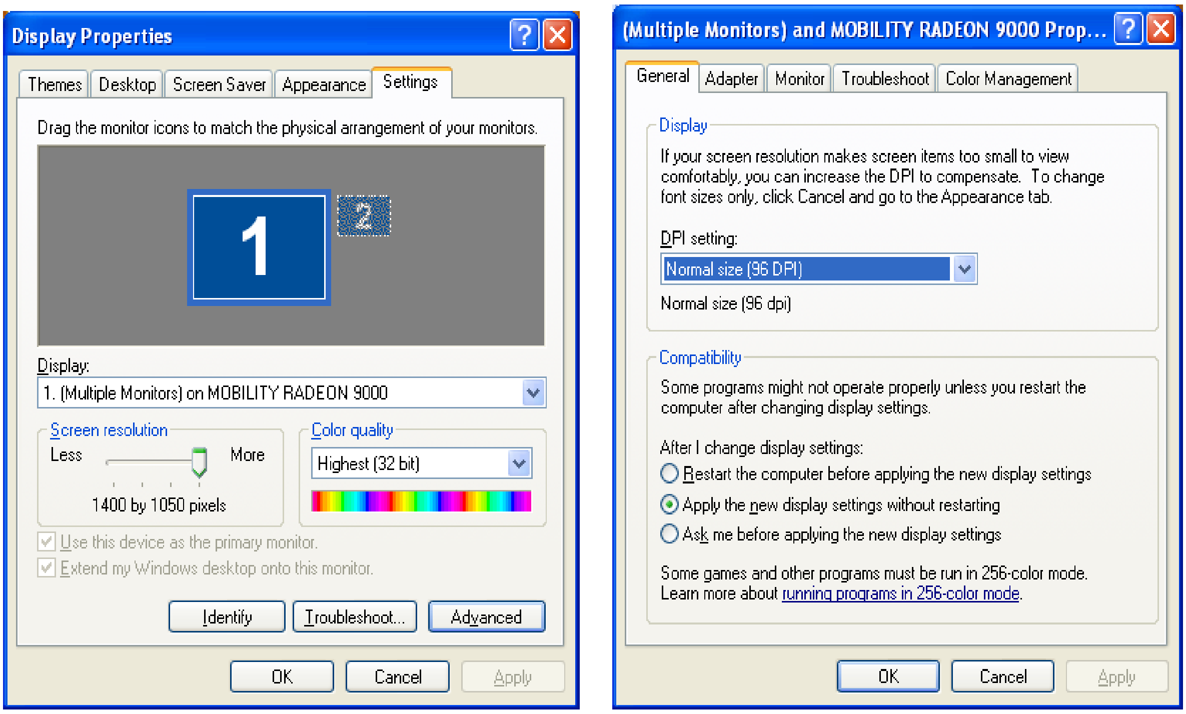


Figure 18.5. Select “Settings” TAB and “Advanced” to go to -----> Select **“Normal size 96 DPI”**.

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.

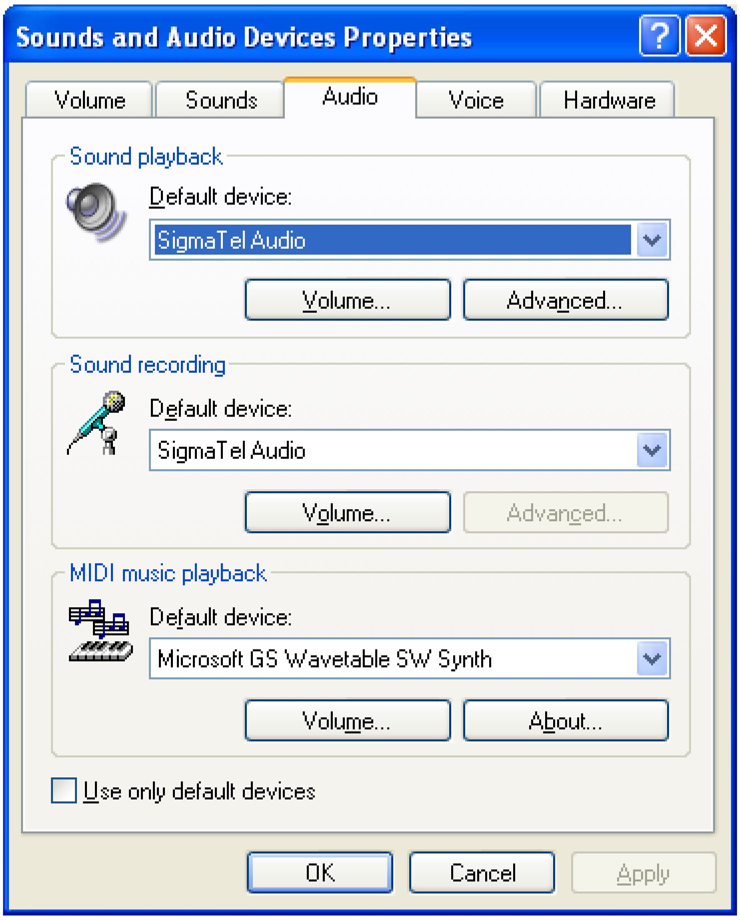


Figure 18.6. Selecting Default Audio Devices.

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.

4. **Digital Filter and Digital Equalizer default to MMC/WDM 8-channel output settings for the sound cards.**

Motherboard Audio Codec Issues

Contemporary PC measurement system should be able to deliver 24bit sampling resolution. This is however dependant on sound card quality or motherboard audio codec – if you decide to use motherboard audio.

Older sound cards (but still available on eBay) like Delta1010LT will provide 24bit sampling resolution on both: recording and playback side. Also good quality motherboard audio codec like ALC1150 Audio Codec supports 24 bit sampling for both: “Line In” input and outputs – see below. As an example, the ALC1150 codec is implemented in GIGABYTE Z97X-UD5H motherboard.

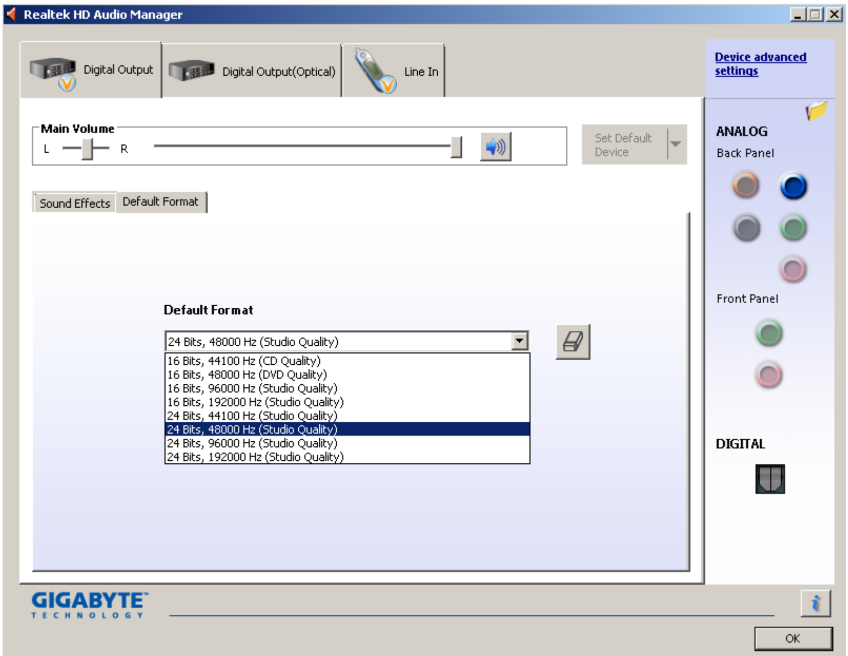


Figure 18.7. Realtek Audio Codec ALC1150 Line-In options.

Somewhat less fortunate implementation of audio codecs has been delivered in motherboards like ASUS P6X58D-E and GIGABYTE H170-H03. There is only 16bit sampling depth implemented on recording side, however, the full 24bits are available on playback side.

In order to help maximizing the overall performance of the MLS measurement system, the Preferences screen allows you to select 16bit/24bit option for sound cards**. The selection is applicable to both: the recording AND playback of the measurement process.** If your PC has only 16bit sampling depth available on the recording side, you MUST select “16 bit” option for the playback on your PC as well, and select “16-Bit Sound Card” from the Preferences screen.

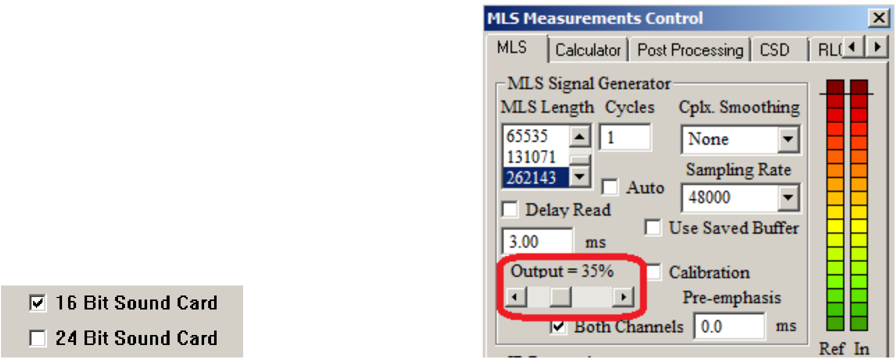
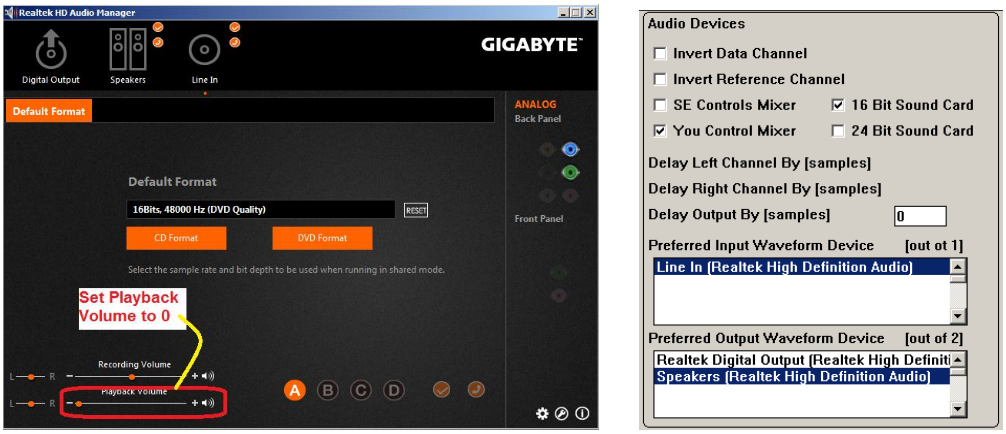


Figure 18.8. Preferences 16bit settings and MLS system audio level adjustment to 35%.

Please note, that due to MLS signal “crest factor”, the output level should be less that 50% lower than maximum available output. **It is recommended to set it to 34-35%.** Shown below are two examples of motherboard audio codecs with 16bit sampling on the recording side and 24bit sampling on the playback side.

IMPORTANT - In the “Line Input” section of the sound card Manager, please set “Playback Volume” to 0 to prevent direct feed from the output to the input of the sound card.

GIGABYTE H170-H03 Recording Side – please note, Recording Volume set to 50%, Playback Volume = 0.



GIGABYTE H170-H03 Playback Side

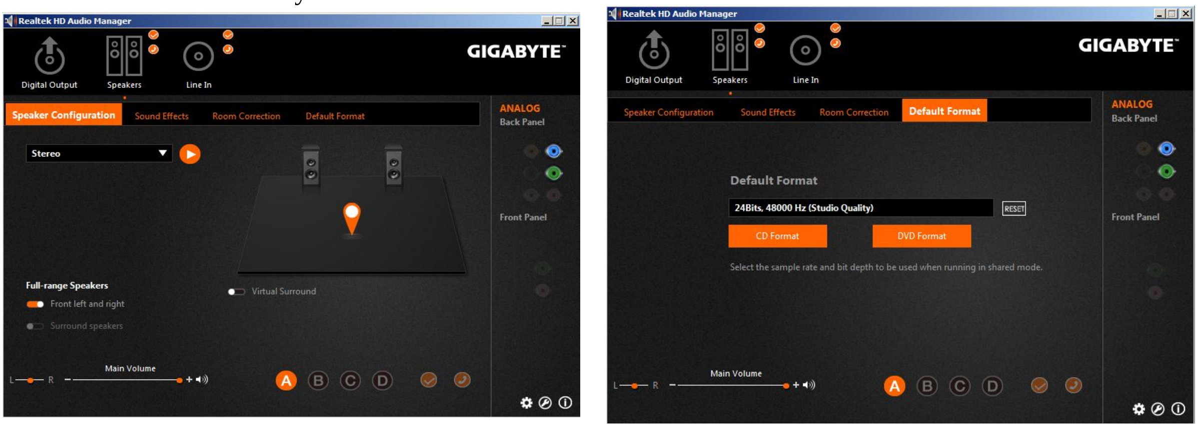


Figure 18.9. Audio Codec GIGABYTE H170-H03

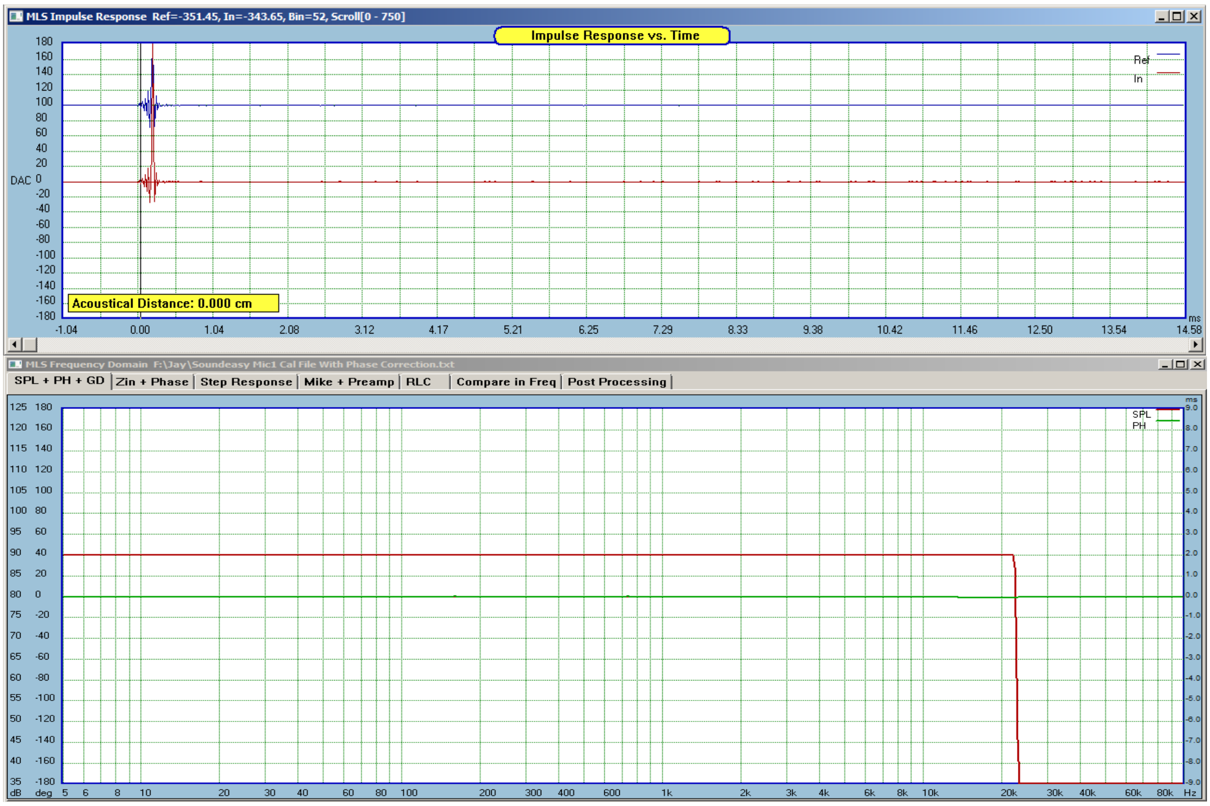
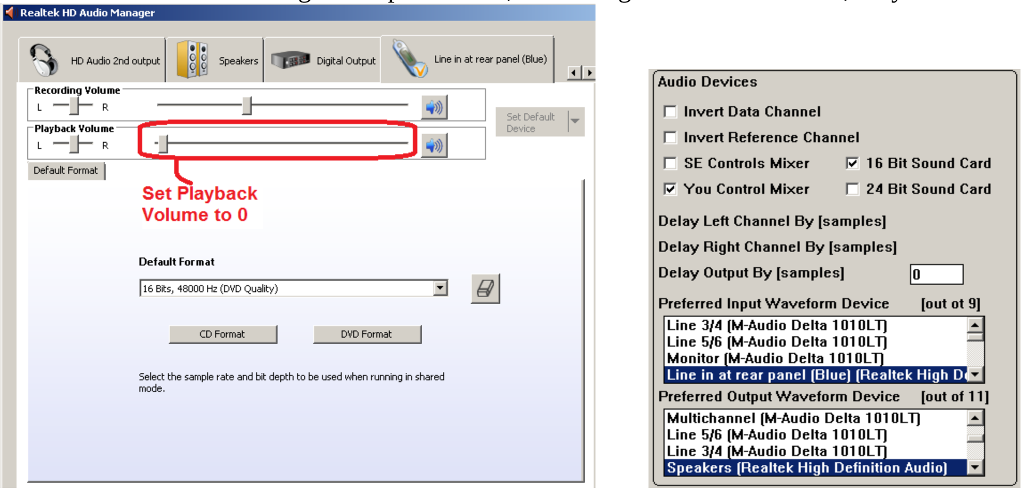


Figure 18.10. Resulting Loop-Test performance

ASUS P6X58D-E Recording Side - please note, Recording Volume set to 35%, Playback Volume = 0.



ASUS P6X58D-E Playback Side - please note, Main Volume set to 70%

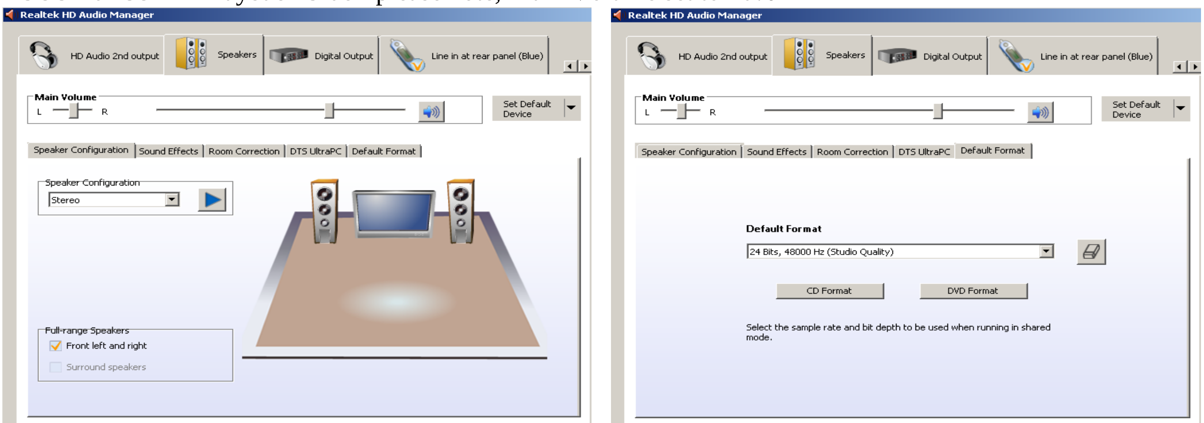


Figure 18.11. Audio Codec GIGABYTE H170-H03

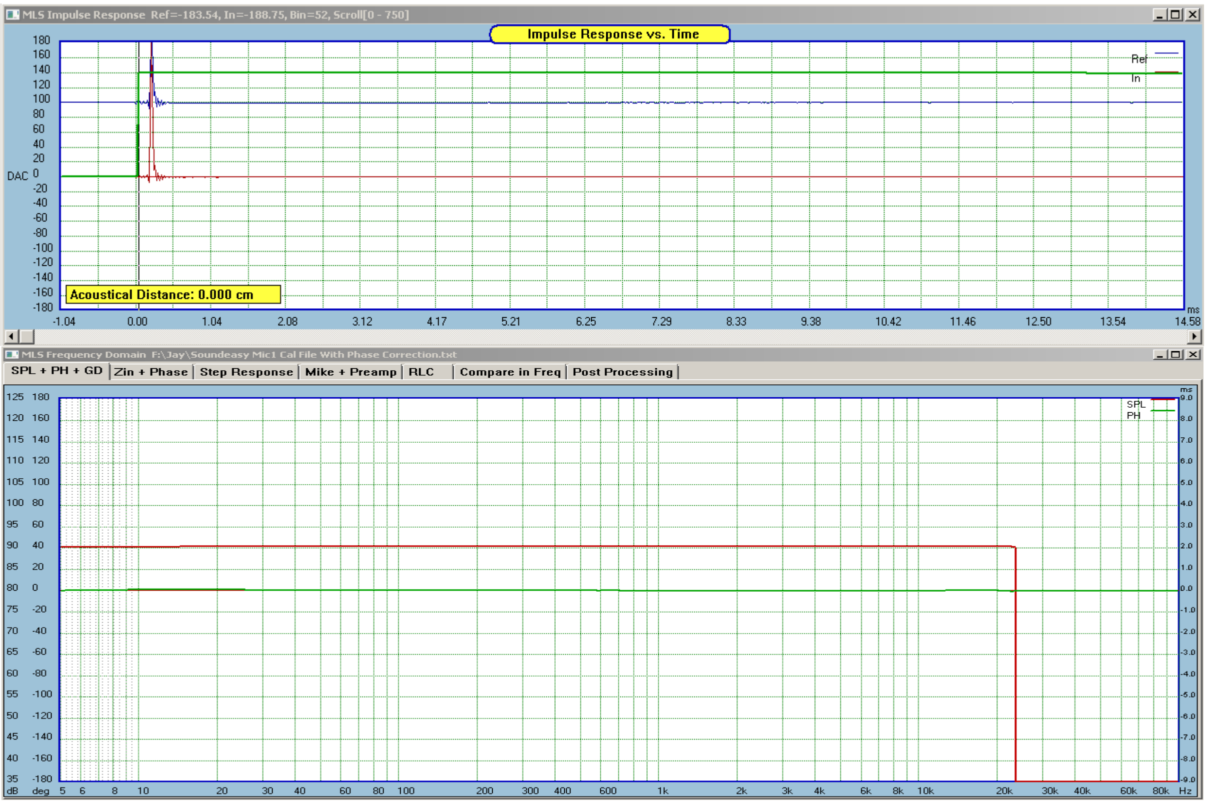


Figure 18.12. Resulting Loop-Test performance

<u>It is strongly recommended, that prior to using SoundEasy software, you confirm the sampling rate and bit depth resolution settings with the sound card driver software or motherboard driver audio software for BOTH: playback and recording and set them to the same values.</u>

Interestingly, a website with a compilation of all available motherboards with ALC1150 coded lists **284 motherboards available**. The ALC1150 codec is capable of outperforming typical sound card or CD-player for signal to noise ratio.

http://us.hardware.info/products/19/9372-realtek-alc1150-motherboards?page=2#allproducts