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.

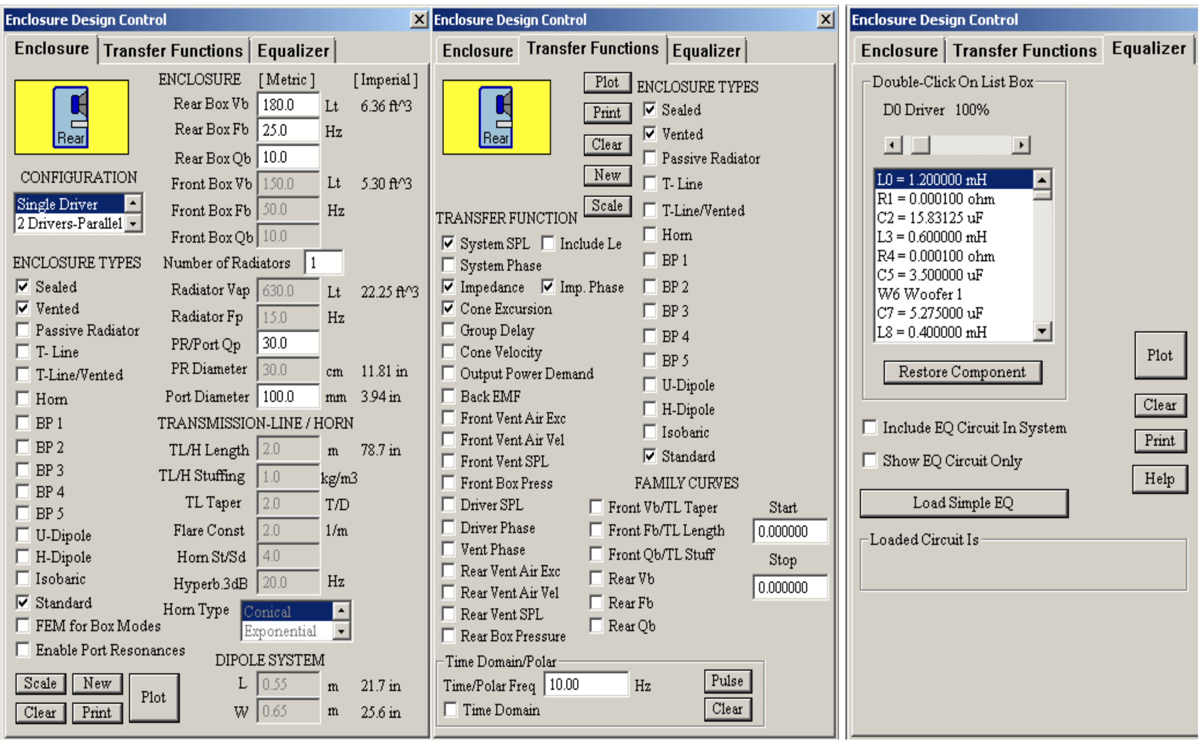


Figure 4.1. Enclosure Modeling Dialogue Box with 2 TABs.

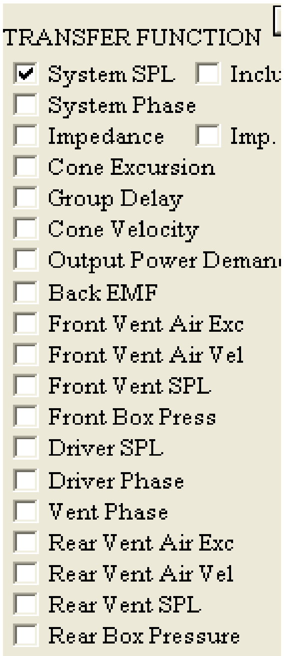
Enclosure modelling is controlled by three TABs: “**Enclosure Parameters**”, “**Transfer Function**” and “**Equalizer**”. Enclosure parameters are keyed-in to the first TAB and the parametric curves selected in the second TAB control is what gets plotted. Labelling of all functions and data fields is explained in Chapter 2, therefore, will not be expanded in here.

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.

Plotting TABs

All curves generated during design process are grouped on designated TABs. There are “System SPL”, “System Phase”, “Group delay” and many more TABs to diplay design curves. Selecting Tabs is very straightforward – just click left mouse button on any of the TABs.

Curves are nominated for plotting by checking appropriate box on the “Transfer Function” control box TAB – see below. Here, the “**System SPL”** curve was selected for plotting.



Finally, each TAB will accommodate 5 curves with different colours. This feature is activated from “Preferences” screen by checking the “Cycle Colour” box shown below.

Attaching Cursor to Curves

Calculated, discrete value of each of the plotted curves can be viewed more accurately by associating a cursor with the selected curve. To nominate a curve for parametric display, please click left mouse button on the curve of interest. The selected curve will be momentarily displayed in pink, and then you will be able to readout curve values as you move the mouse cursor across the plotting screen.

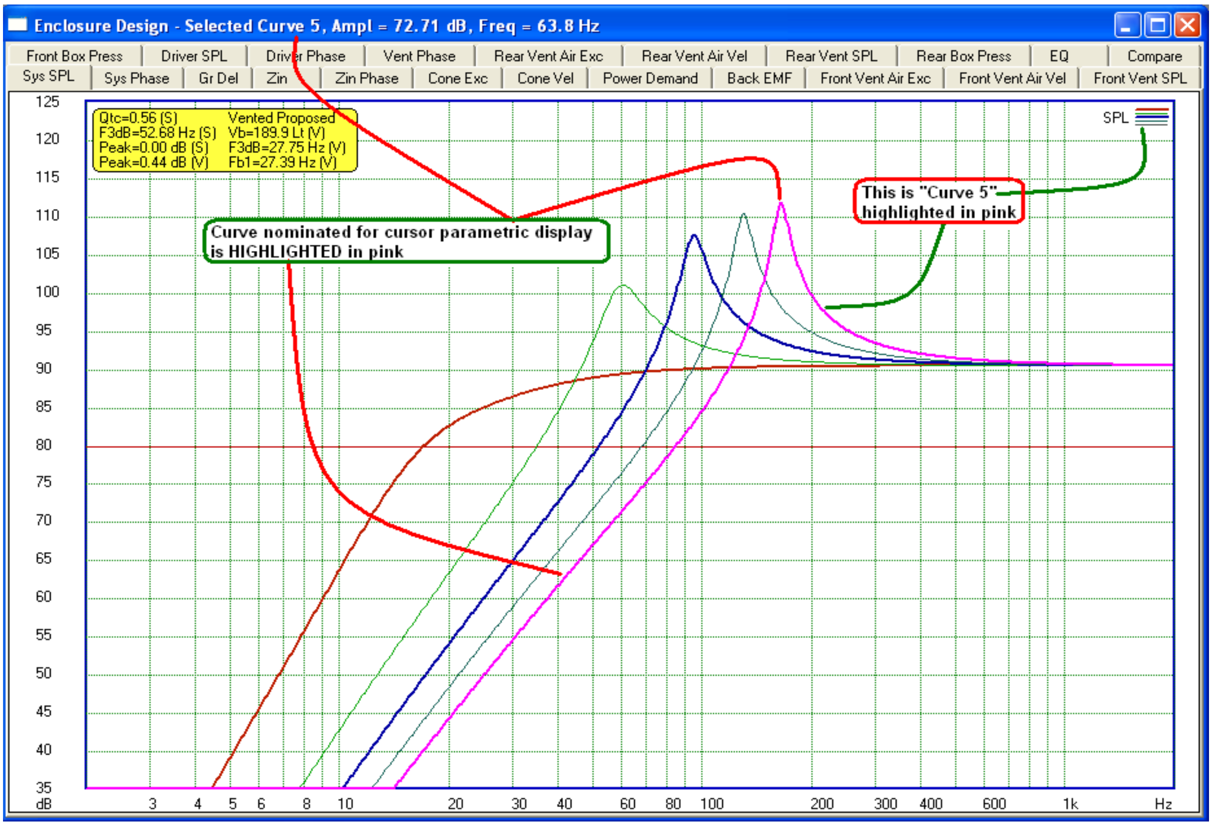


Figure 4.2. Mouse click upon curve 5 causes the cursor attachment to this curve.

Here is an example of a cursor nominated to display parameters on "Curve 5". This curve was originally plotted in gray colour, which is indicated by curve label, in the left-top corner of the plottimg screen. The mouse cursor at the time of readout was placed at Freq = 256.5Hz, and corresponding SPL value was 95.08dB – see below. These values are displaued in the top frame of the plotting window.

At the same time when you nominate the curve for cursor readout, the selected curve is automatically copied to the **Compare** screen.

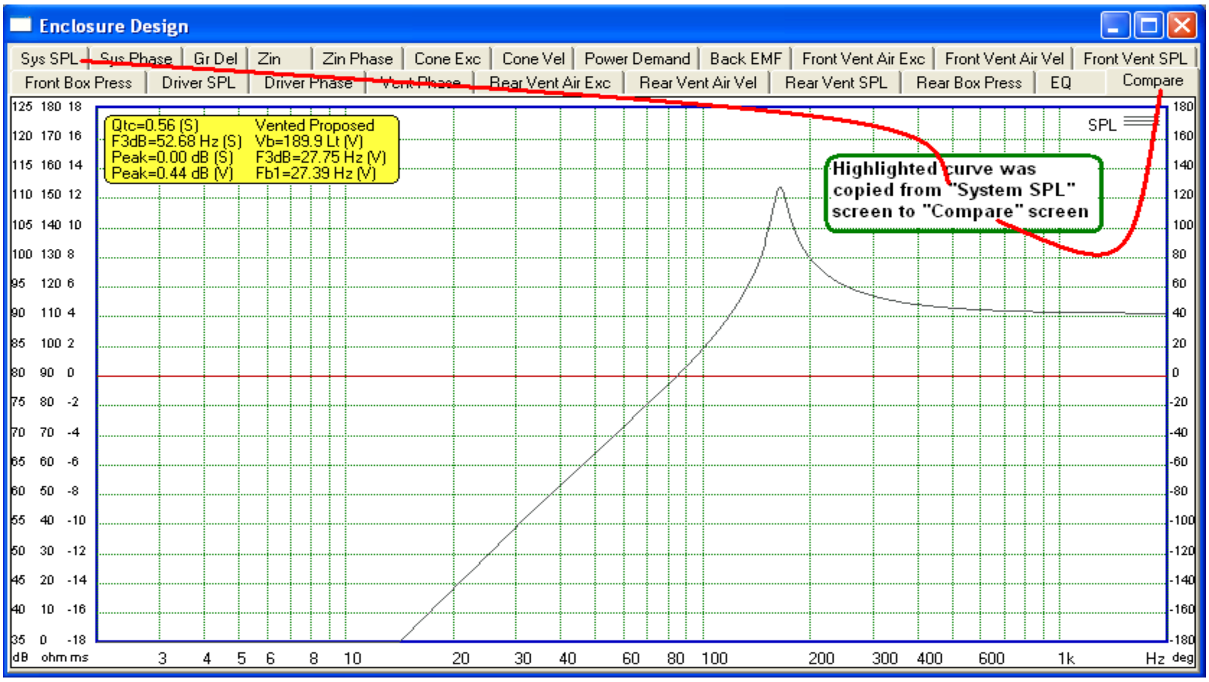


Figure 4.3. Curve 5 copied to "Compare" screen.

Enclosure Modeling/Review

To gain an insight into this tool please load the "******.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 Figure 4.4.

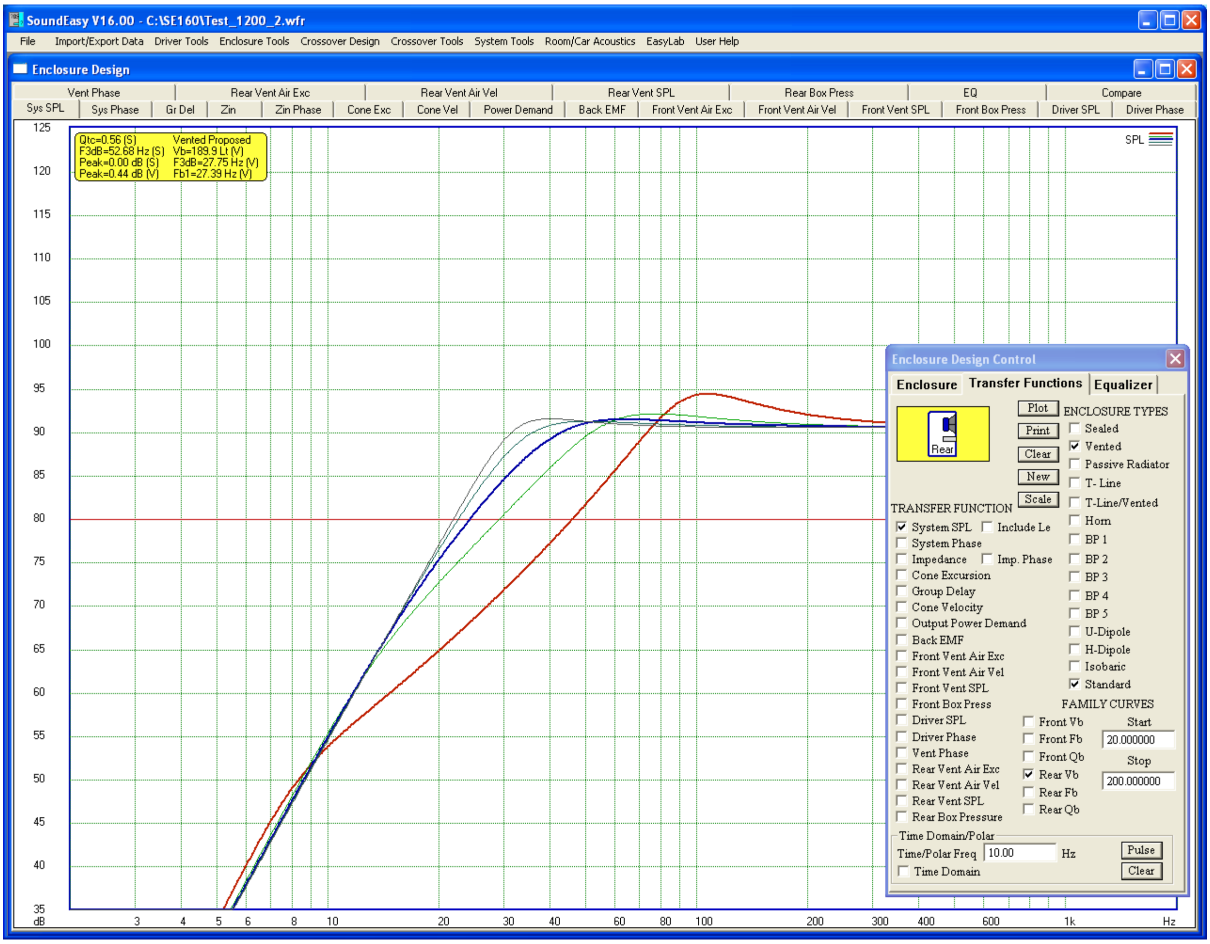


Fig 4.4 Family curves for Different Vb values.

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 4.5. Examination of these curves will lead to the conclusion, that curves corresponding to 28-32 Hz box tuning, offer the flattest frequency response.

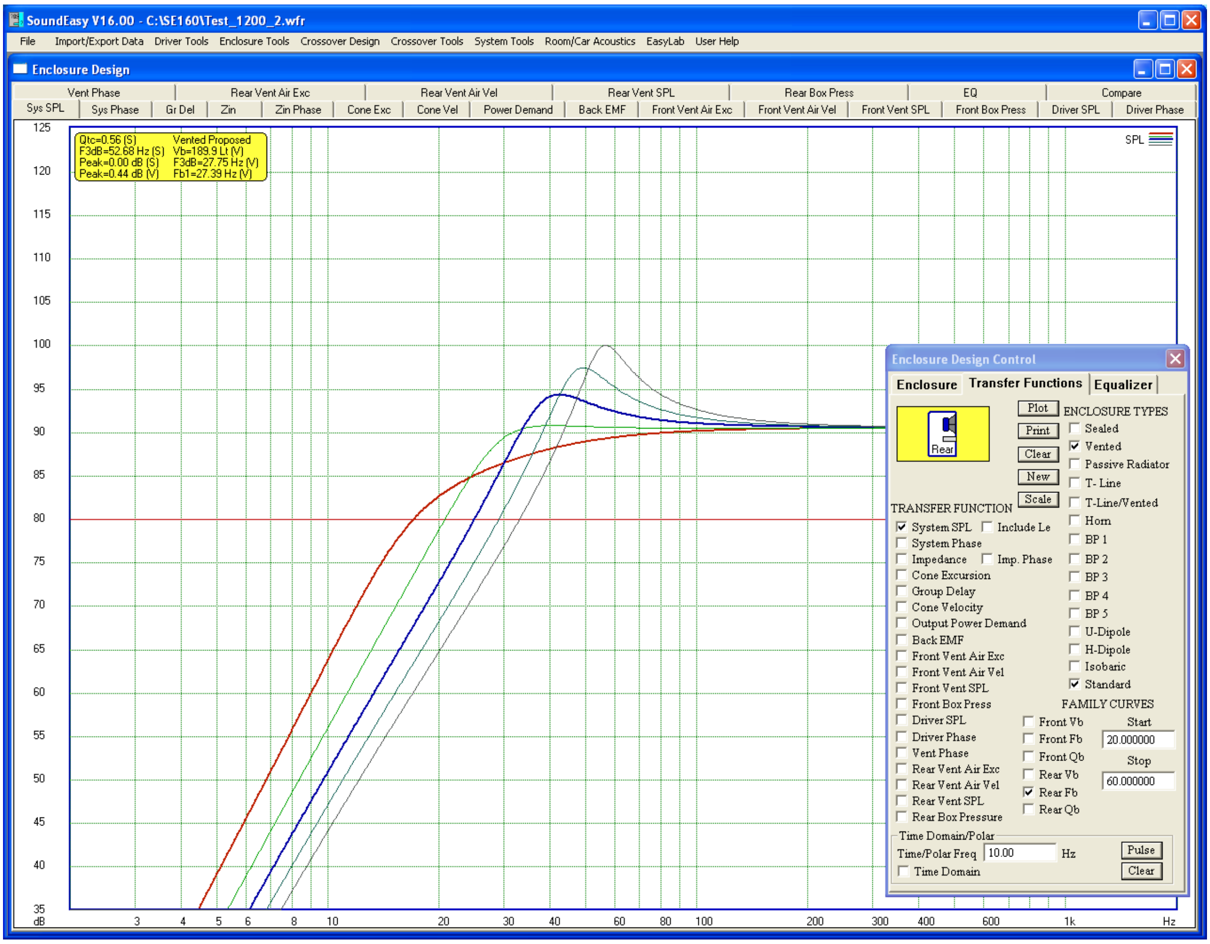


Fig 4.5 Family curves for Different Fb values

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 4.6. 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. 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".

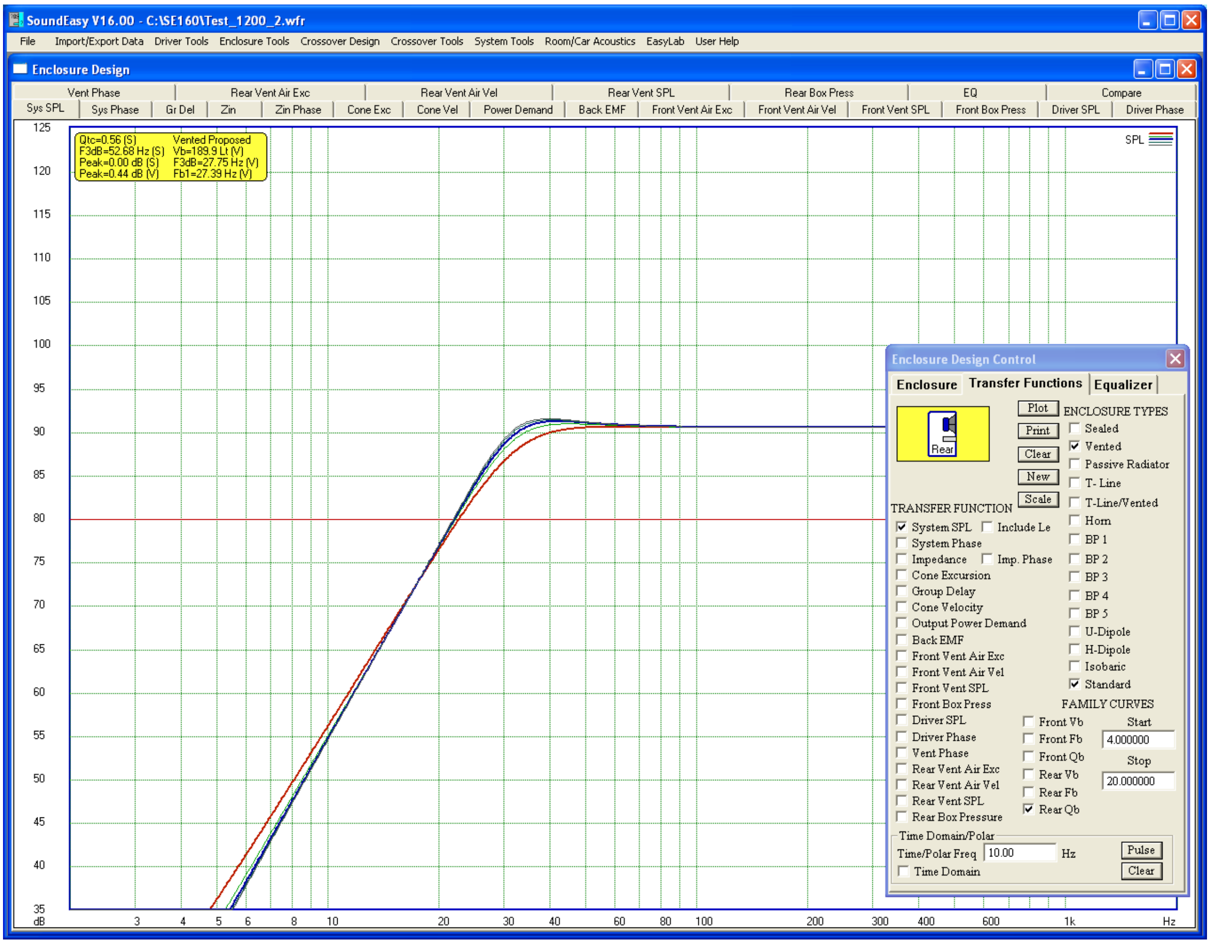


Fig 4.6 Family curves for different Qb values.

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 4.7. 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.

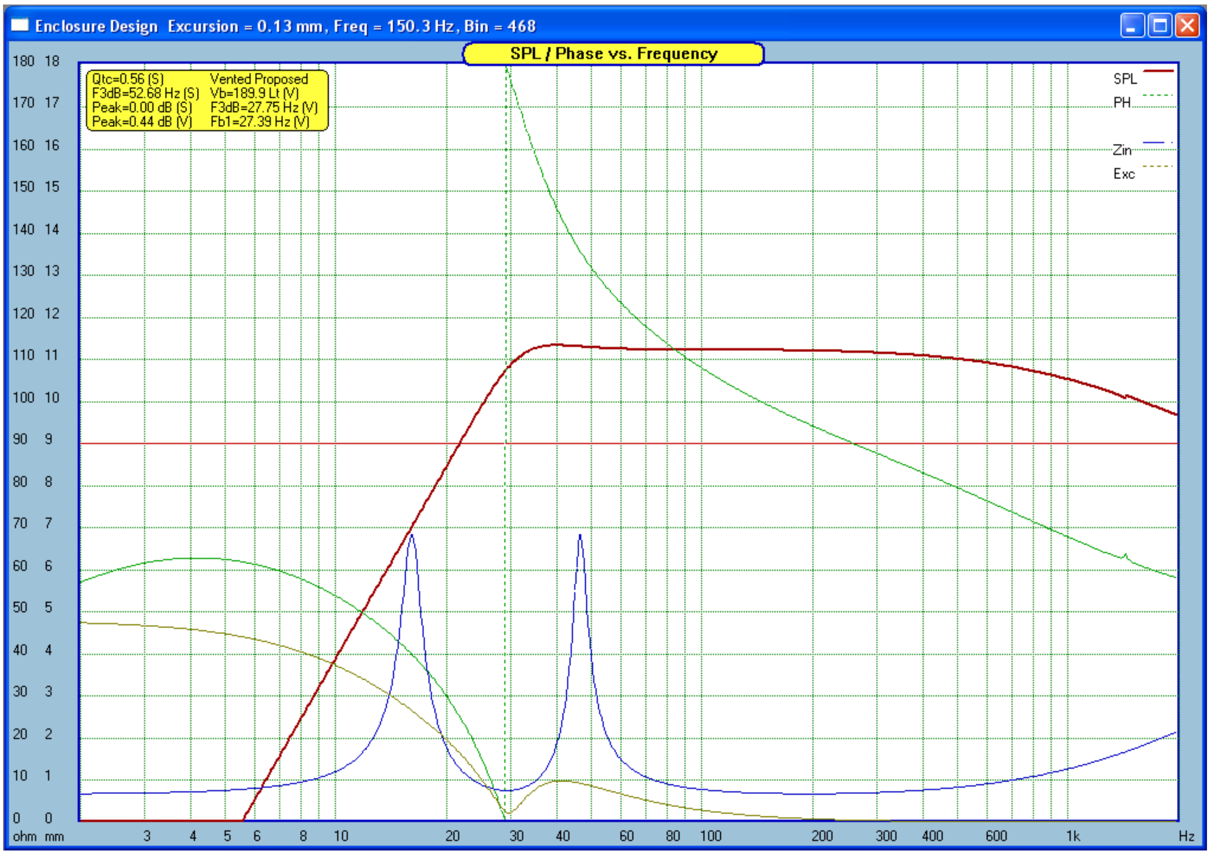


Fig 4.7 Various vented box plots

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 4.8) 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.

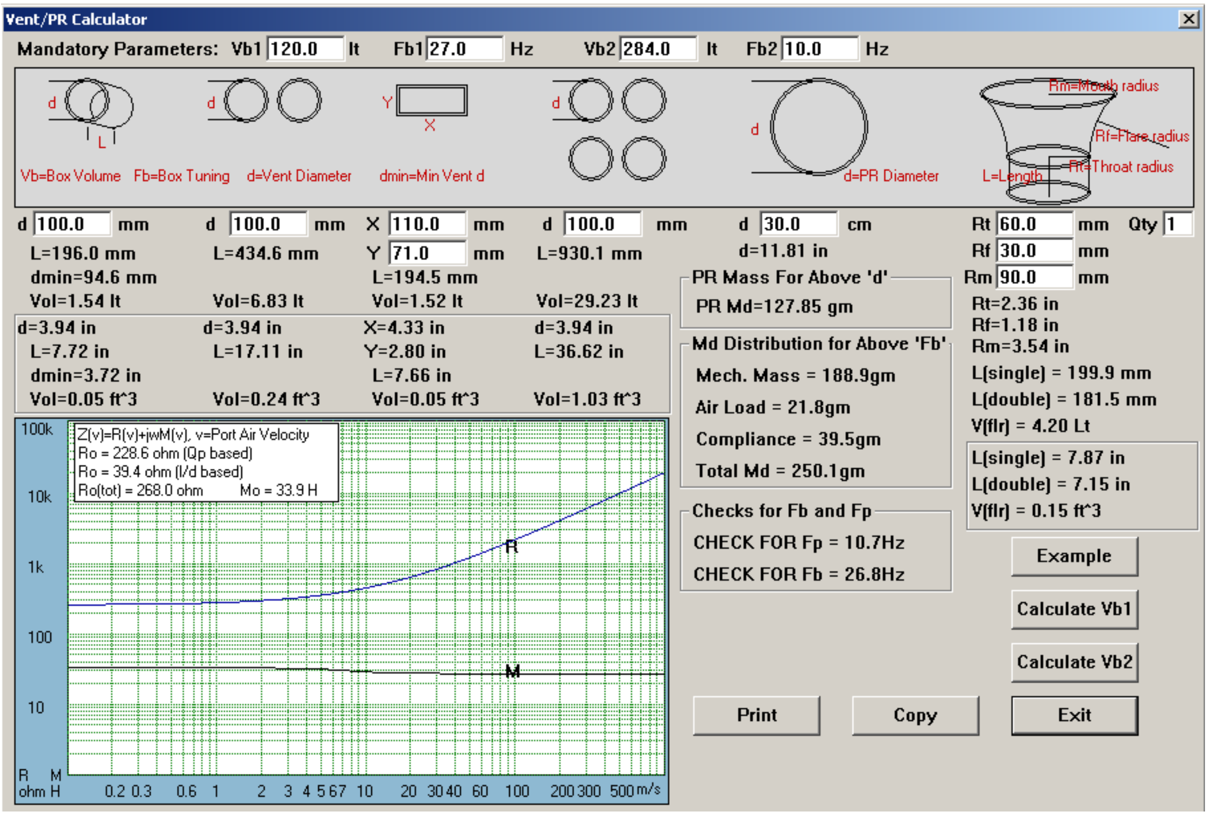


Figure 4.8 Vent Calculator

Flare radius for the flared vent is calculated from Mouth radius Rm, and Throat radius Rt data.

Enclosure Shapes

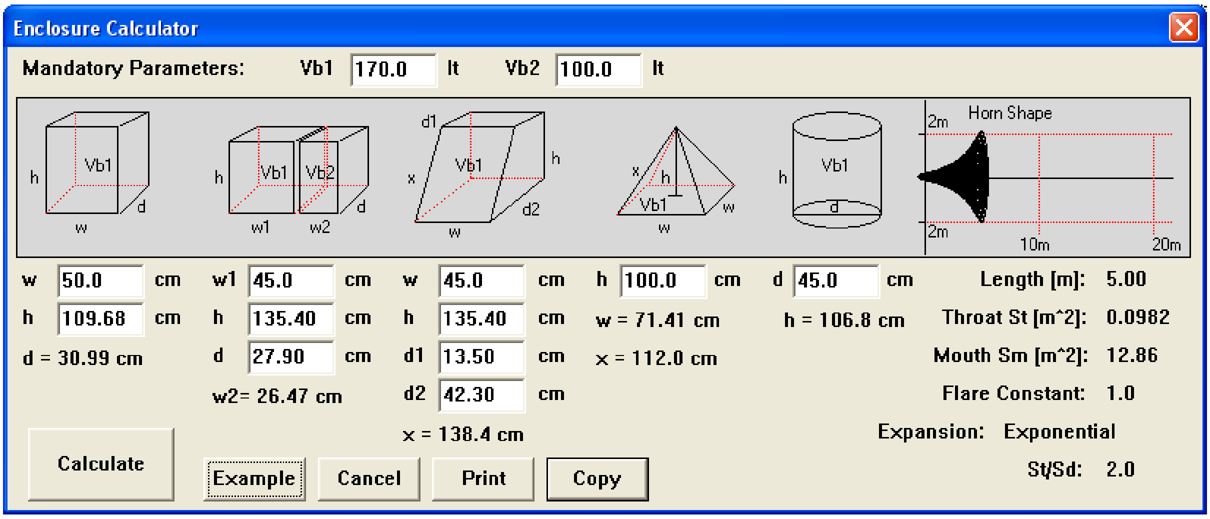


Figure 4.9. Enclosure schapes calculator

The Enclosure Shapes Calculator (see Fig 4.9) 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 4.10

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.

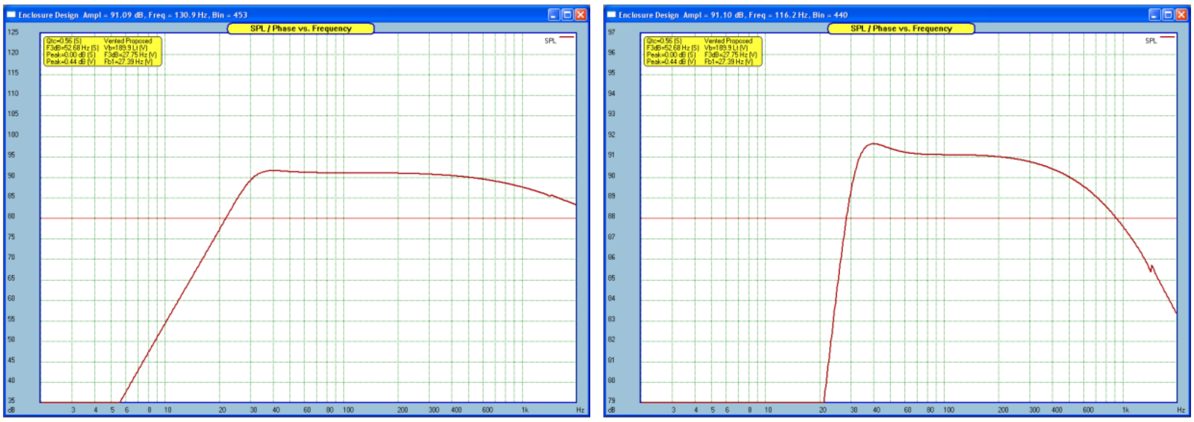


Figure 4.10. Adjusting vertical resolution.

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. Figures presented later show, 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].

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.

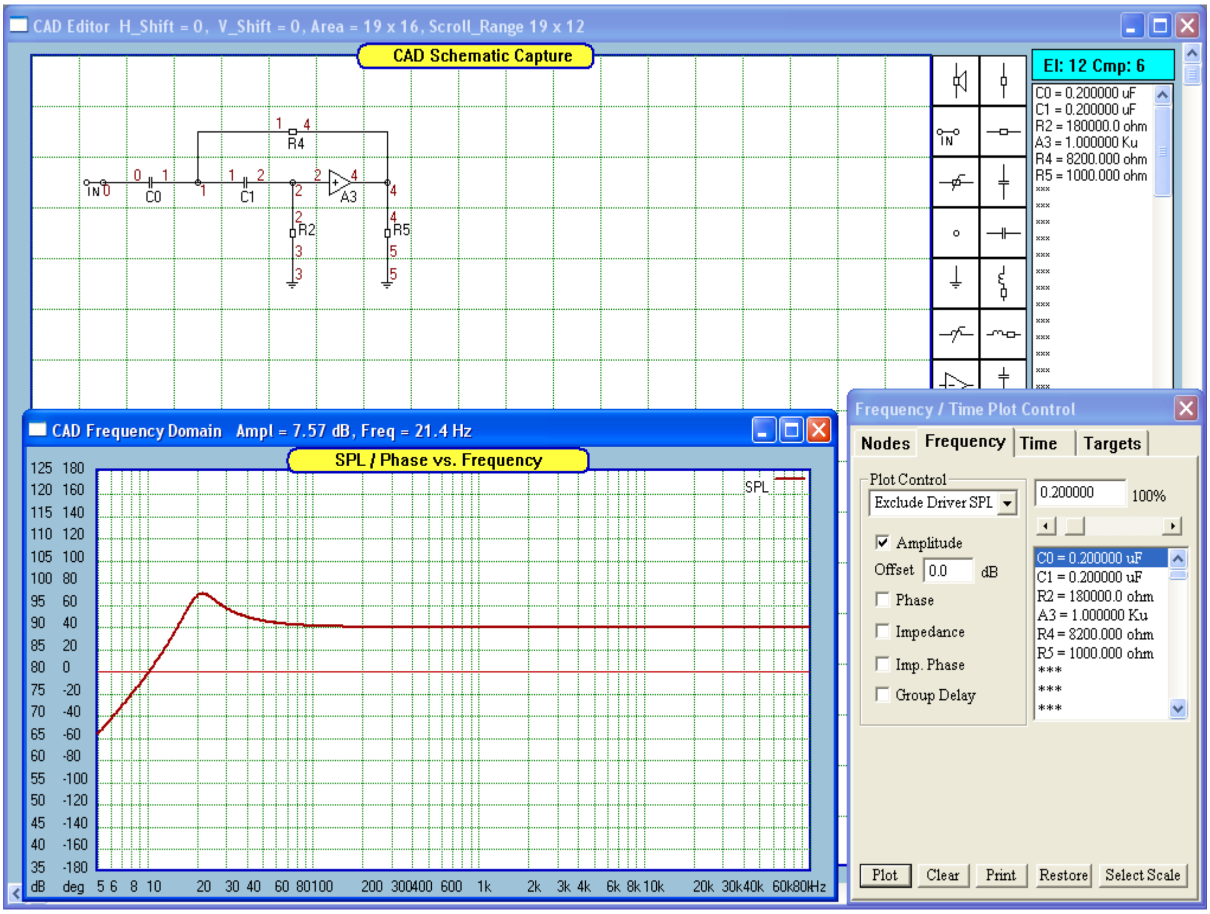


Figure 4.11 Active EQ circuit.

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.

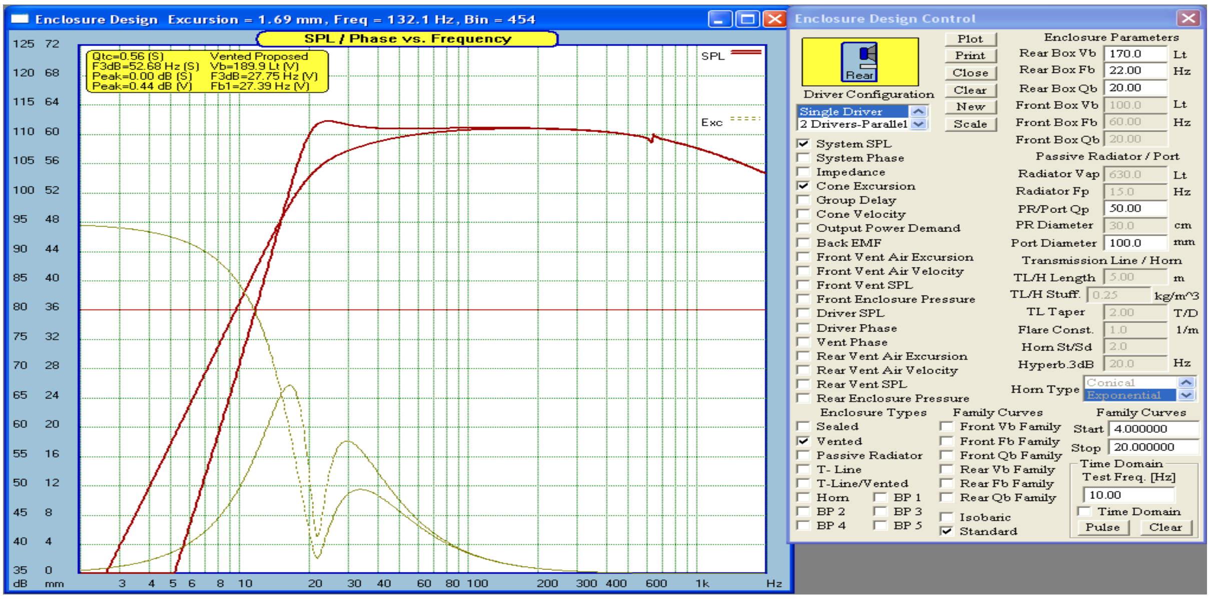


Fig 4.12 Vented enclosure “Unassisted” and with EQ Circuit.

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 4.11. 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 4.12 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 4.13 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 4.13 Vented enclosure (upper curve), Sealed enclosure (lower curve) + EQ circuit

File Selection for Enclosure Equalization

Enclosure equalization can be performed by loading one of the pre-designed equalization circuits, therefore a variety of equalization options for the enclosure can be explored.

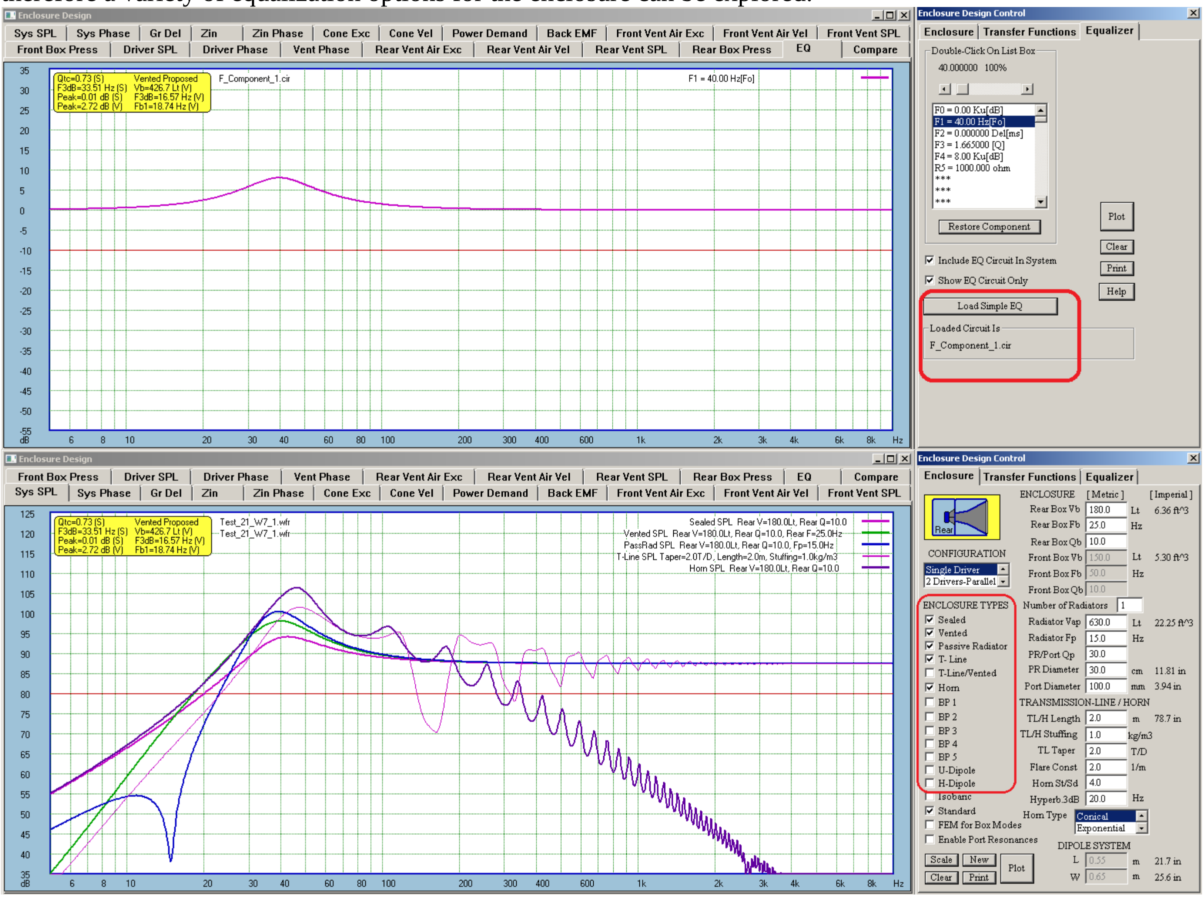


Figure 4.14. Selecting an EQ file for various enclosures.

Front Baffle Design and Diffraction Distortion

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", (**4π**) 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 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.

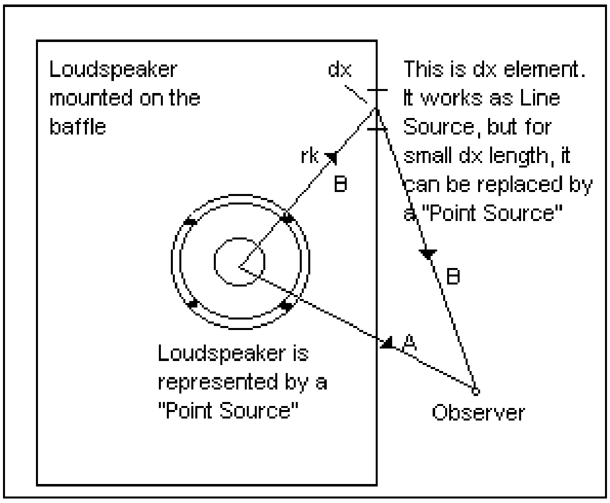


Fig 4.15. GTD concept.

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.

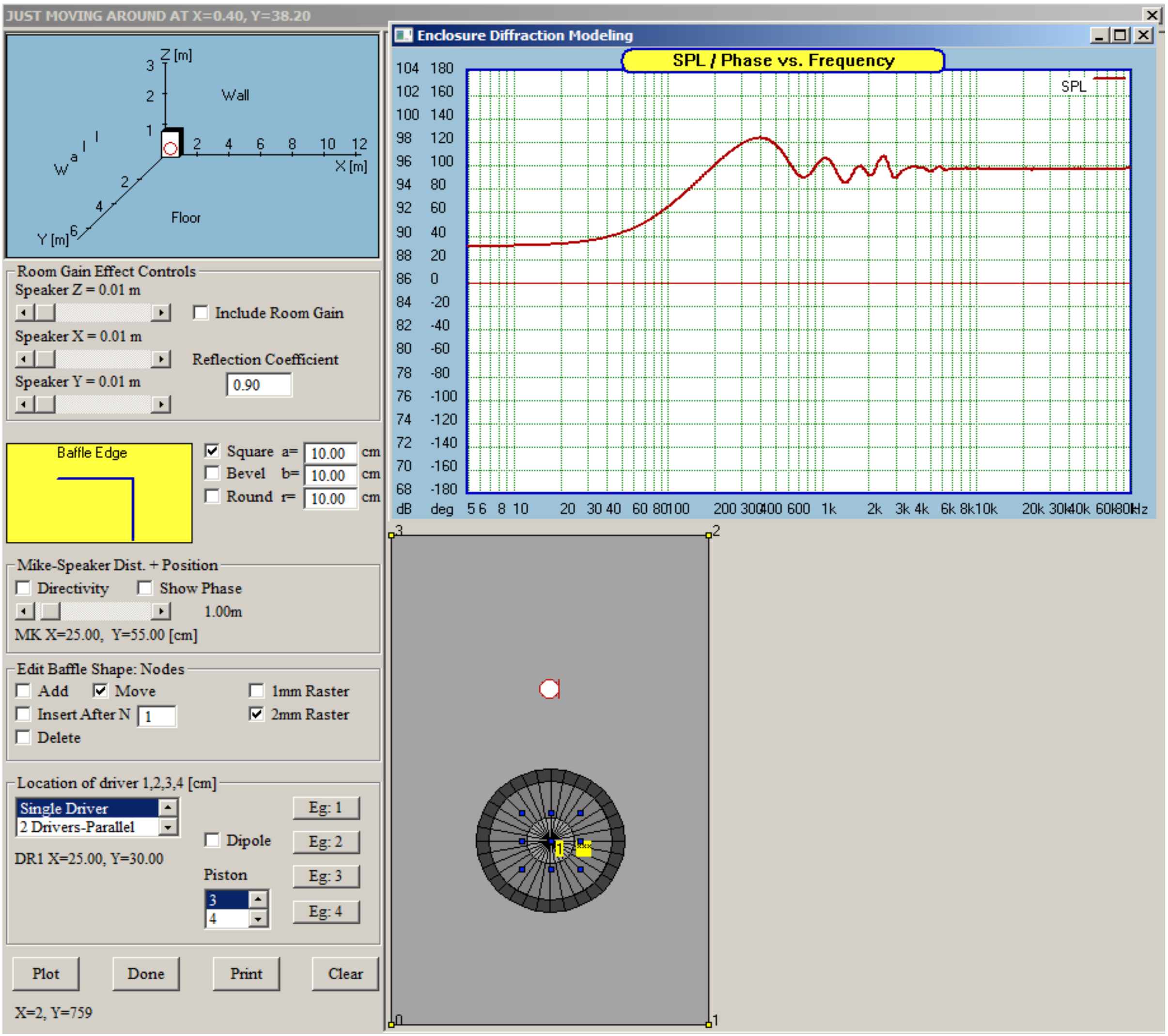


Figure 4.16. Single driver diffraction – 2dB/div, 90dB Ref.

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".

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. The program automatically divides each baffle side into a number of ***dx*** line elements. Loudspeaker's directivity is modeled as large number of "point sources" contributing to the direct SPL. For the purpose of modeling standard diffraction loss, we also assume, the reference distance to the observer to be the standard 1 meter. This situation is depicted on Figure 4.15. **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 below. 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.

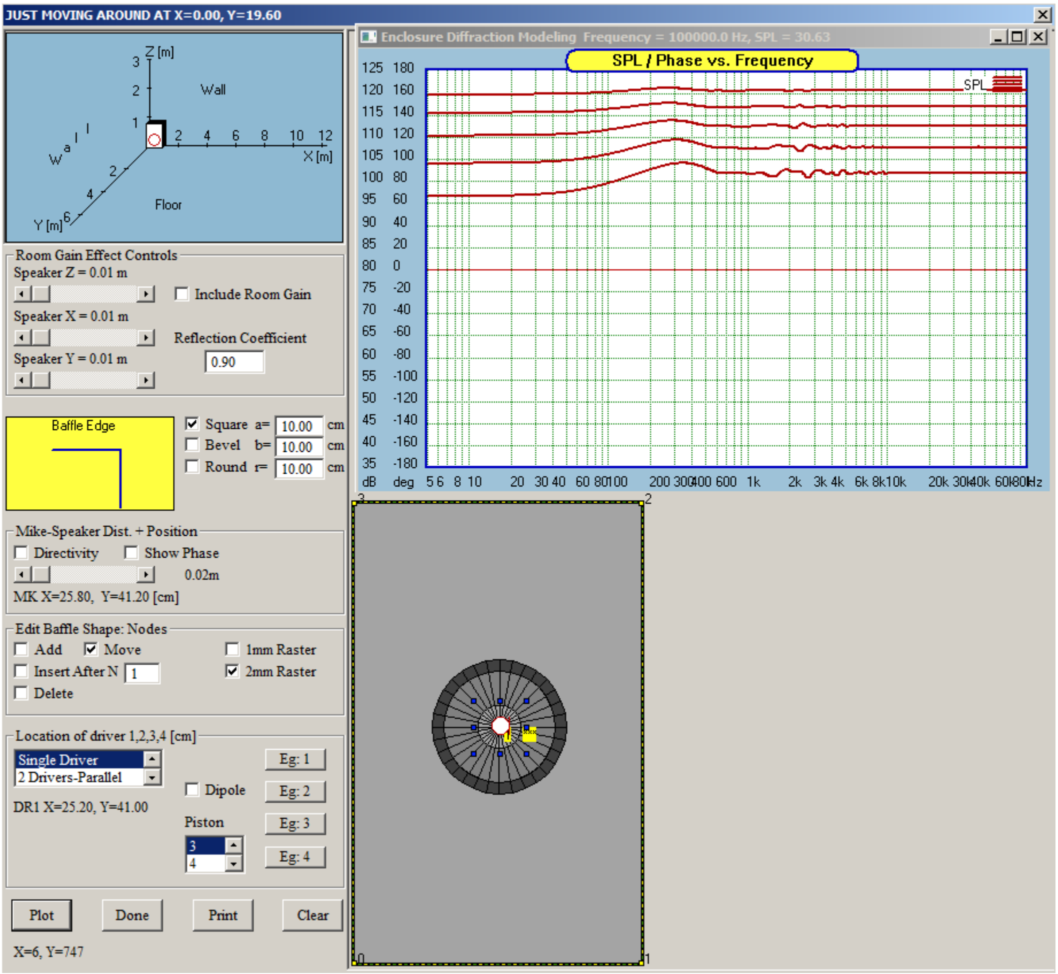


Figure 4.17. Near-field diffraction – 5dB/div, 90dB Ref.

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.

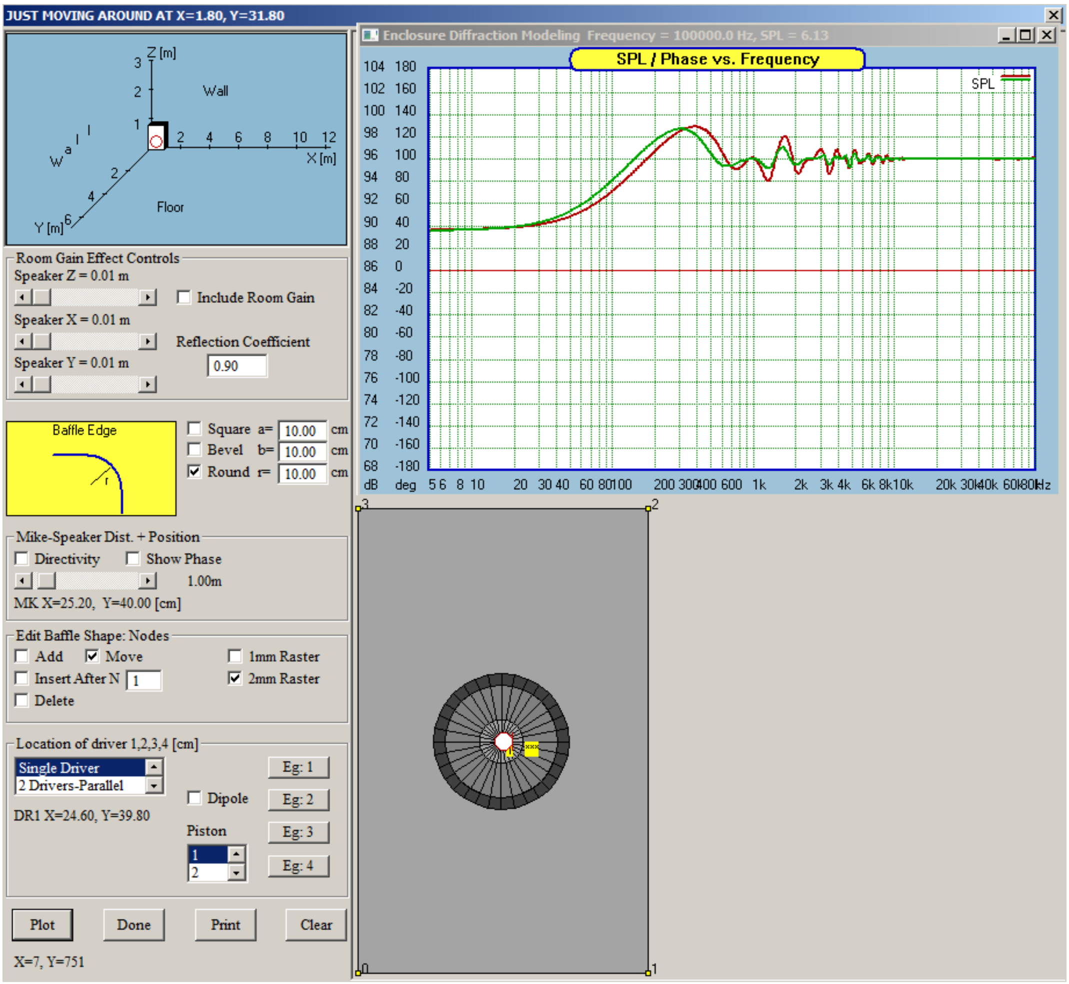


Figure 4.18. Trying different edge finish Square (Brown) and Round (Green).

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

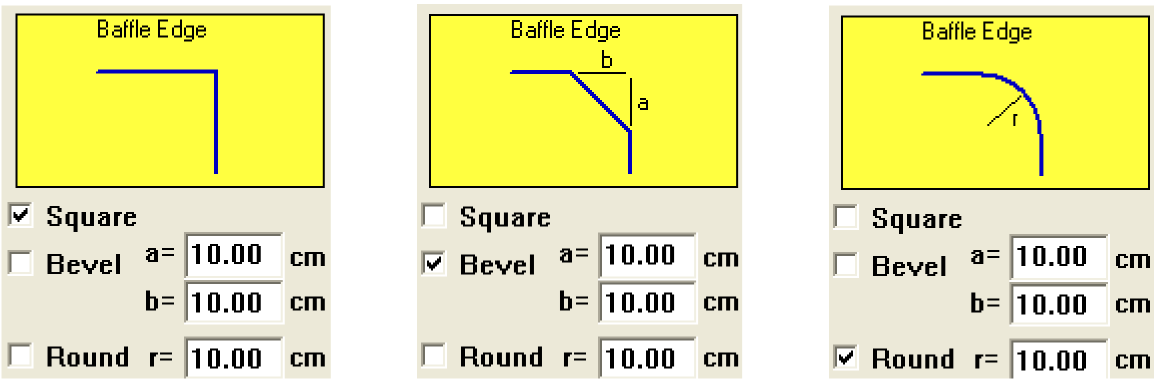


Figure 4.19. Three enclosure edges: square, bevel and round.

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 4.19
3. Round-type – there is only the radius, r, to be determined – see Figure 4.19

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 4.20) 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.

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

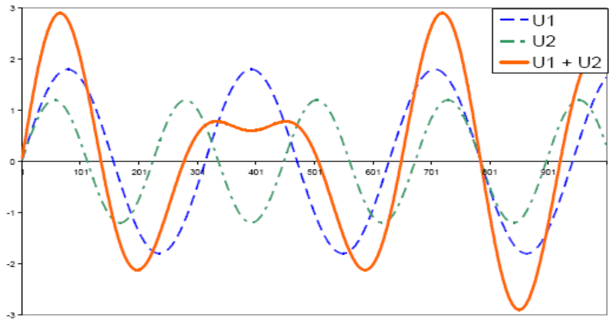


Figure 4.20. Example of addition of uncorrelated voltages.

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 (correlated voltages addition) 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.

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

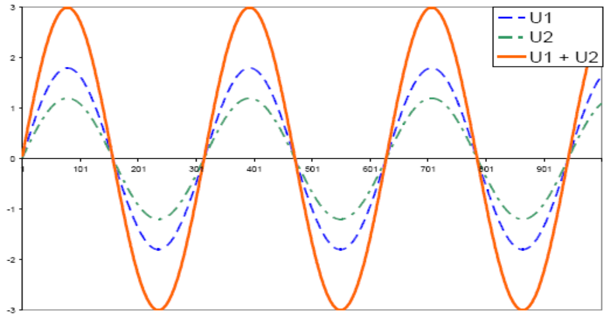


Figure 4.21. Example of addition of correlated voltages.

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.

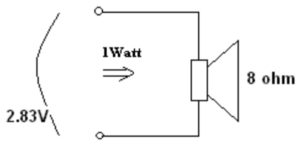
Combining All the Above

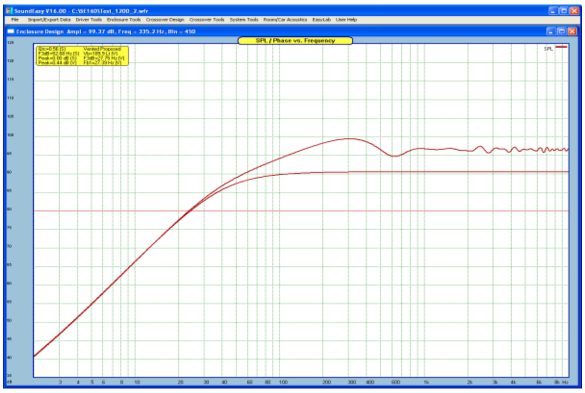
We are now in good position to review the SPL gains due to:
1. enclosure diffraction,
2. mutual radiation impedance and
3. multiple drivers in the same enclosure.

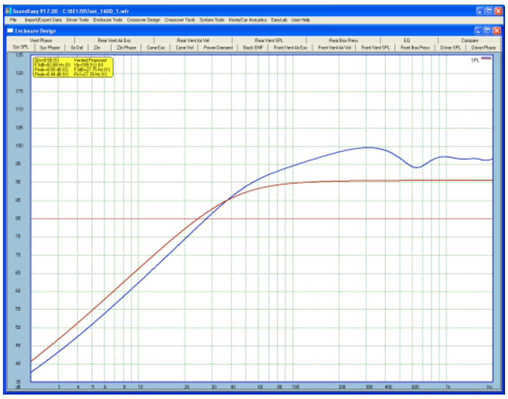
We assume that: (1) our amplifier is an ideal voltage source - this is the case of most currently available amplifiers. The amplifier will deliver 2.83VRMS to the load. (2) all drivers are identical and have real impedance of 8ohm. For the input voltage of 2.83VRMS, the loudspeaker will deliver **90dB SPL into 4PI space**. 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 below.

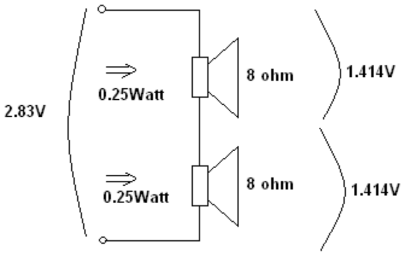



Single driver


Two Drivers Connected in-series

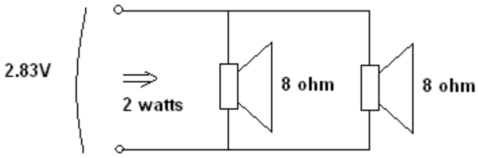
Two Drivers Connected in-series

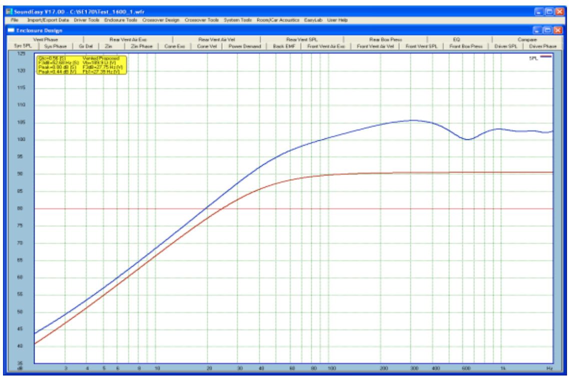
Being the ideal voltage source, the amplifier will also cope well with the 16ohm load impedance being now presented to it (two 8-ohm loudspeakers connected in-series). Each driver will now receive only half of the 2.83VRMS voltage generated by the amplifier. With this in mind, each speaker will receive only 0.25W of electrical power (U*U/R = 1.415*1.415/8 = 0.25 W) Each driver will now generate only 84dB SPL, so that total SPL of the system is now 90dB. The diffraction effect will add +6dB in the upper end of the operating frequency range. The electrical power delivered to the system is now only a half (0.5W) of the single driver configuration.

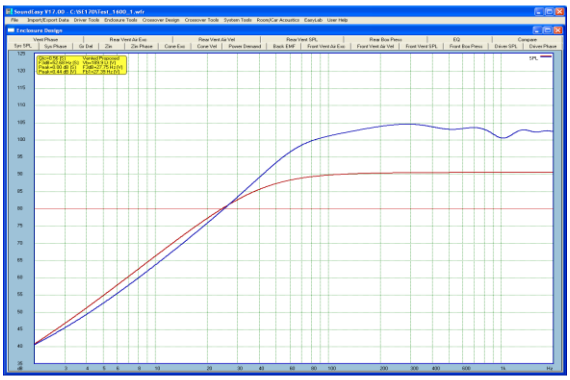


Two drivers connected in-parallel

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 96dB. The diffraction effect will add on the top of it +6dB in the upper end of the operating frequency range. The final SPL gains curve is shown on figure below. It is worth noticing, that electrical power delivered to the system is now twice (2.0W) of the single driver configuration.

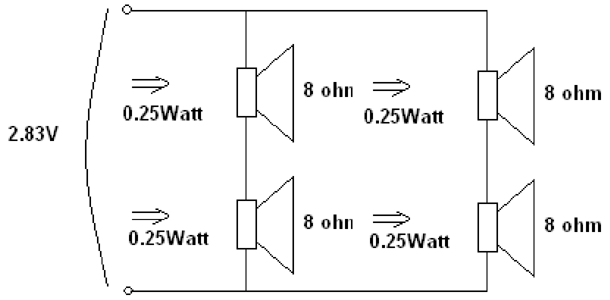



Two driver connected in parallel


Four drivers connected in-series and in-parallel

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 +6dB. The final SPL gains curve is shown on figure above. 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", as compared to the single driver configuration. The main advantage of this configuration is 4 x power handling of the single driver.



Operation of the Diffraction Distortion Dialogue Box

The "Front Baffle Design/Diffraction" dialogue box can be opened from the main menu, under the "Enclosure Tools" option. It is now tha main front baffle design tool. 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) 4 Drivers Parallel/Series and (4) Single Driver in smaller box. Activated "Enclosure Diffraction" box is shown on above. There are several parameters editable by the user:

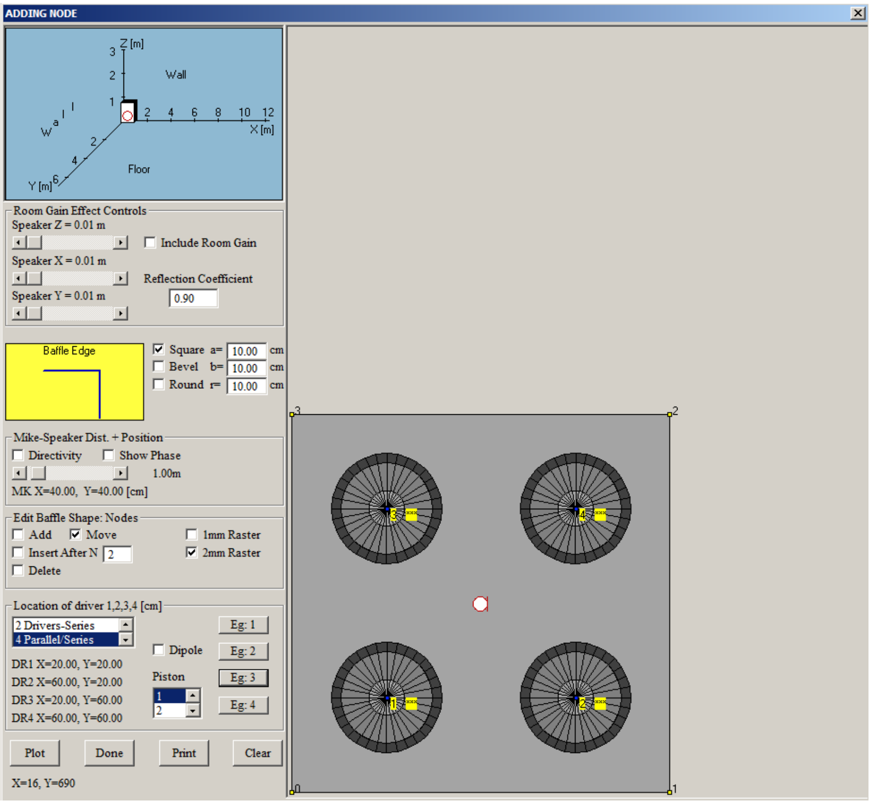


Fig 4.22 Front Baffle Design / Baffle Diffraction control dialogue

Microphone-Speaker Distance. The distance can be adjusted between 1mm and 20000mm (20meters) using horizontal sliding bar. Dimensions of the front baffle (x-coordinate and y-coordinate) are shown in centimeters. Location of all four drivers - this parameter is shown 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. There is also a check-box to include the Speaker-Microphone distance and driver positioning in diffraction modeling of multiple drivers. The four buttons activate the following functions:

1. **Clear** - clears the plotting area.
2. **Cancel** - closes the dialogue box.
3. **Print** - prints the screen for immediate hard copy.
4. **Plot** - calculates diffraction, low frequency coupling and plots the results.
5. **Listbox** - for selecting multiple driver's configuration is activated by doubleclick.

The "Front Baffle Design" system and "Front Panel Layout" system share the same coordinates system. The relationship between those two is explained on thediagram below. Maximum size of the front baffle is **603cm x 806cm**.

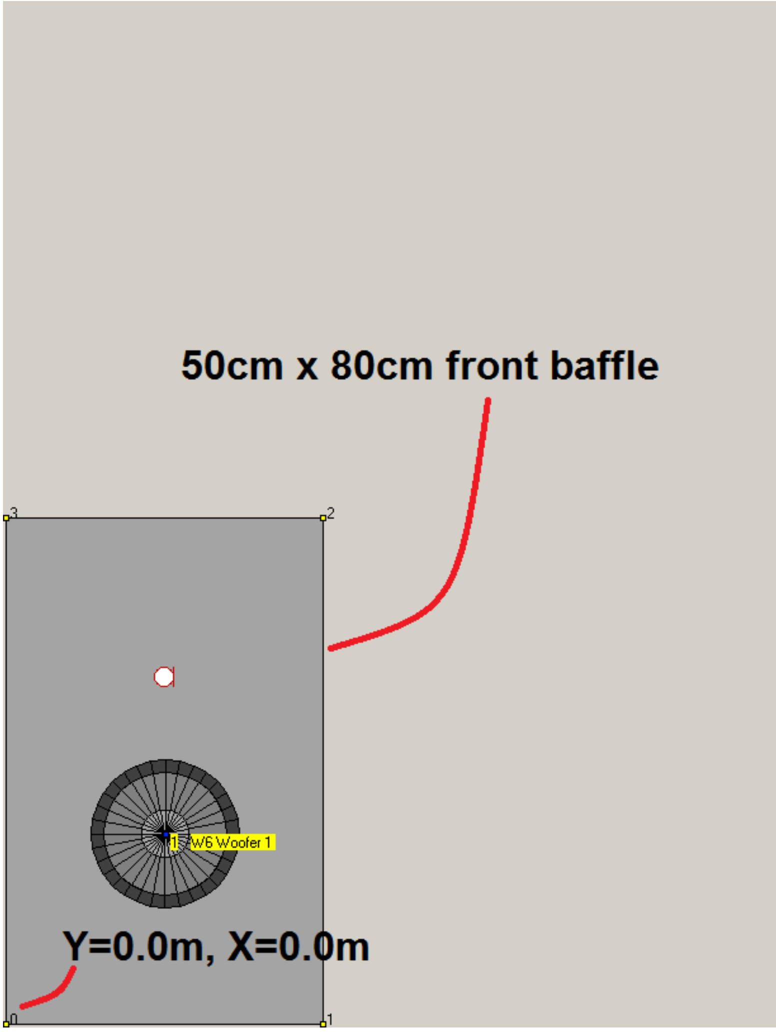


Fig 4.23 "Front Baffle Design" system and "Front Panel Layout" system share the same coordinates system.

Diffraction Analyser Functions

Diffraction Analyser function provides additional insight into performance of a loudspeaker in a box and is intended to compliment Enclosure Design module. Currently, the following functions are implemented into the Diffraction Analyser:

1. Diffraction from enclosure edges. There are 3 types of edge approximations available.
2. Inclusion of directivity of the driver. Here, the loudspeaker cone is modeled as 41 discrete point sources, and upon summation, provide additional information about driver's directivity.

3. Mutual radiation resistance loading. Drivers mounted on the same baffle generate additional air-load pressure for each other. This effect increases load resistance and consequently improves efficiency at low frequencies by up to 3dB.

4. Room corner loading. When a loudspeaker is placed near an intersection of 3 planes (room corner), a significant augmentation of low frequencies occur. This analysis is rather academic, as it refers to “concert” or stage listening conditions. Typically, you will be listening to your speakers in an enclosed space.

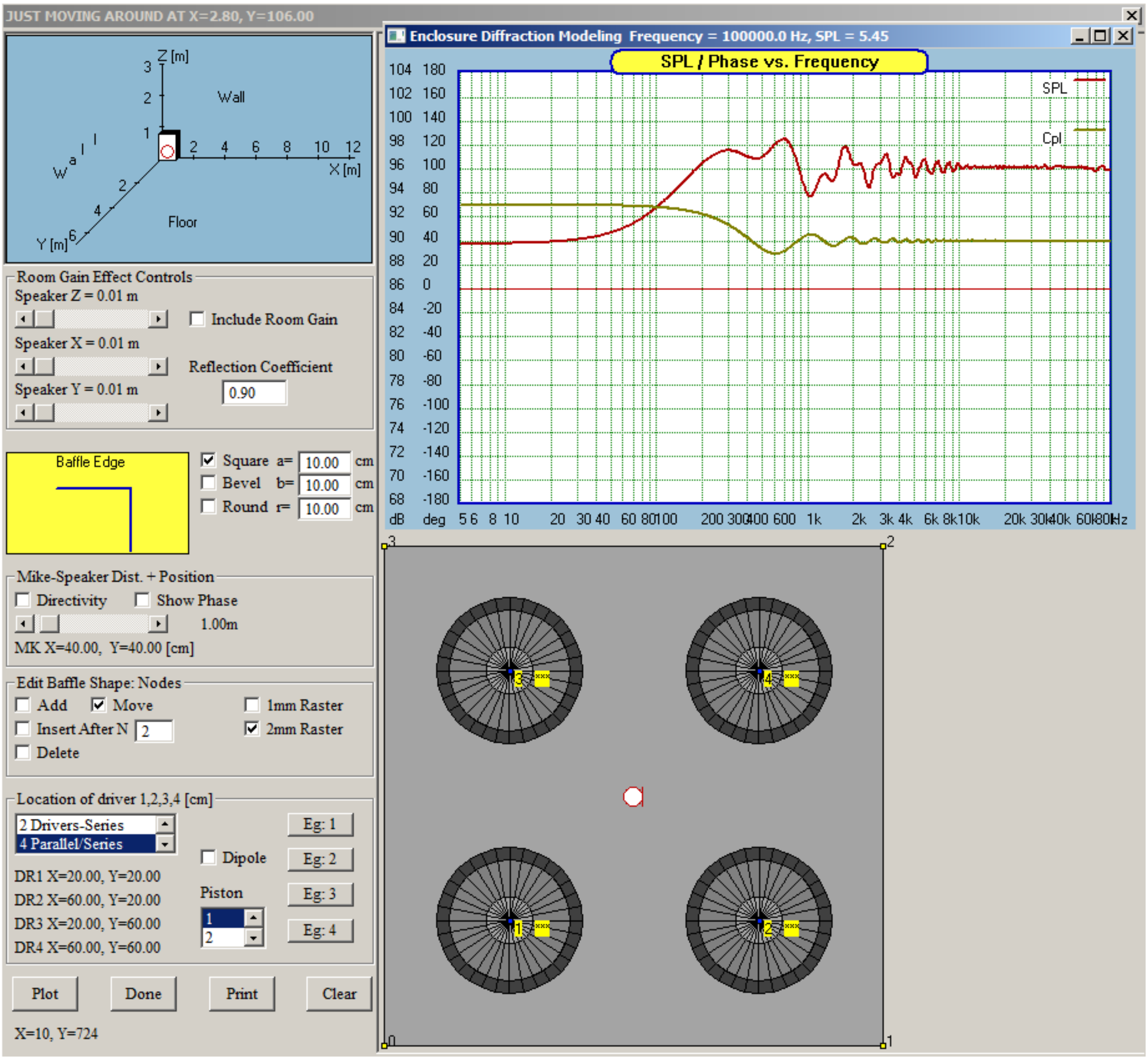
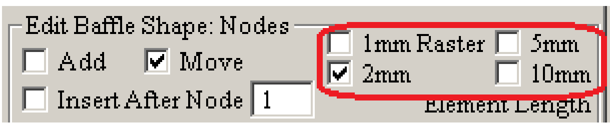


Fig 4.24 Four drivers connected in-series and in-parallel

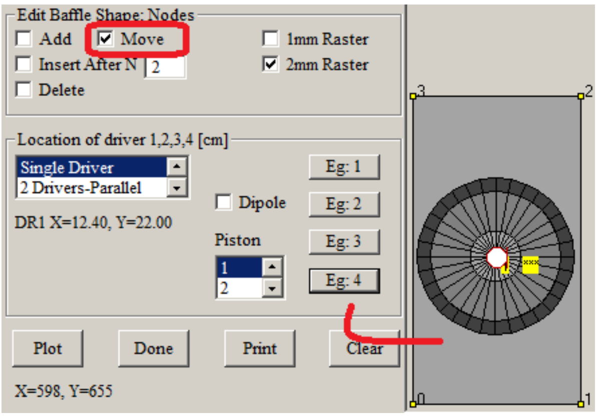
GUI of Diffraction Analyser

In order to perform diffraction analysis, you need to define the size and shape of the front baffle of the loudspeaker system. Current implementation allows you to “draw” the baffle shape on the provided raster. The raster offers **600 points along X-axis and 800 points along Y-axis**. The distance between raster points equals to 1 screen pixel, and has been chosen to facilitate “hand drawing” using mouse. The **size of the raster** is editable.



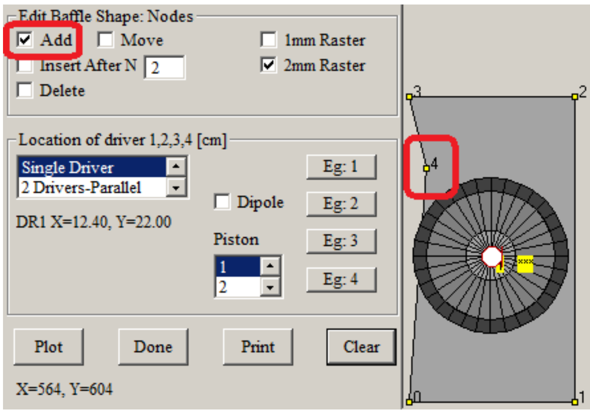
Therefore, for smaller enclosures, you would select smaller raster size, and for larger boxes you can select larger raster size. To facilitate baffle drawing process, the analyzer provides you with several editing functions, that are selectable via provided checkboxes. **Maximum Baffle Size = Raster Points x Raster Size**.

However, since all boxes have at least three vertices (like triangular front baffle), the quickest way to draw the baffle is to use one of the built-in “Examples” and then simply modify the baffle by adding more nodes and moving them around.



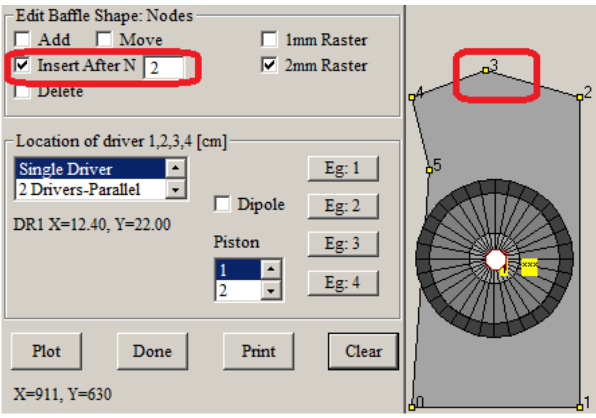
Additional benefit of this approach, is that the “Node 0” is correctly placed in the origin of the drawing area.

1. **Add Node** – Select “Add Node” and then move the mouse pointer above the baffle. After you moved the mouse to the correct raster point, click the LEFT mouse button. A small yellow rectangle will appear, marking the selected node. Adding nodes places them AFTER the existing node with the largest number.

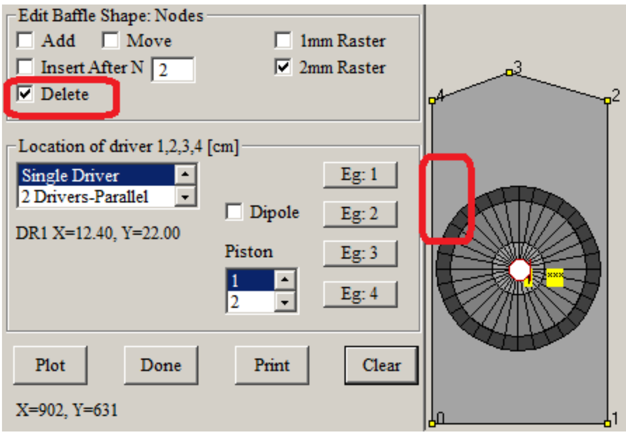


   One point, that is very important to remember here is, that your graph representing the front baffle MUST be closed. So the last node is always placed on the top of the very first node in the graph. You will see a message on the screen when this condition is detected.

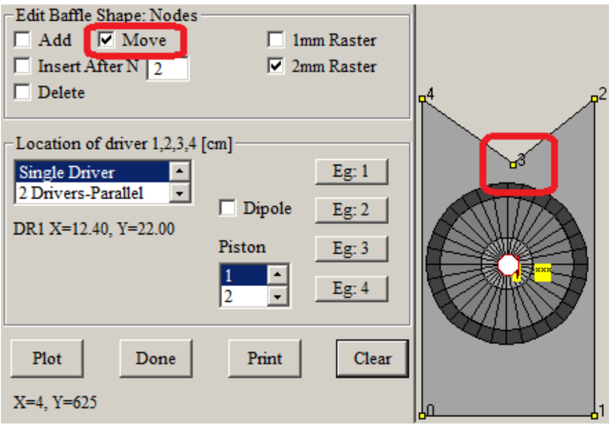
2. **Insert Node** after N-th node – Typically, you would use this function to make your baffle more elaborate in shape. After you selected this option, move the mouse to the correct raster point, click the LEFT mouse button. A small yellow rectangle will appear, marking the newly inserted node.



3. **Delete Node** – If you placed a node in a wrong location, or placed too many nodes, you can erase a node by switching to "Delete" function. So, select this mode, and place the mouse pointer close to the unwanted node. Then press the LEFT mouse button. The "Auto Guide" function will automatically place the mouse poiner on this node, and when you release the button, the node will be deleted. Example below shows Node 5 deleted.



4. **Move Node** – Sometimes, you may decide, that pushing nodes around will simplify the drawing. Should this be the case, please select "Move Node" and place the mouse pointer close to the desired node. Then click the LEFT mouse button to lock the node, and move it to the new location. Example belo shows Node 3 moved.



You can move around nodes associated with the baffle, nodes associated with the loudspeakers and a node associated with the test microphone.

Diffraction analyzer can also be used to review directional properties of the loudspeaker setup. To turn this function ON, please "check" the **"Directivity" checkbox**. Subsequent plots will include: (1) driver "piston model" consisting of 41 radiating points on the cone, (2) microphone X-Y-Z positional information, and (3) loudspeaker X-Y positional information. You will observe, that when turning "Directivity" OFF, the diffraction is modeled as a "minimum phase" phenomenon. Phase information on the SPL plotting screen can be displayed by toggling the **"Phase" checkbox**. You will observe, that phase response corresponds to "Directivity" checkbox settings. Since the diffraction modeling function compliments the enclosure design activities, this dialogue box includes for your convenience, the same **loudspeaker configuration setting control (1, 2, or 4 drivers**) as the enclosure design module. This is again implemented as a list-box with the above settings.

There are also 4 built-in examples of baffles with different driver configurations. You can access the examples by **pressing "Eg 1", "Eg: 2", "Eg: 3" or "Eg: 3" buttons**. You could possibly use those as a starting configurations for your designs. The **"Eg: 4"** is a single driver configuration with microphone placed right in front of the driver. There is no baffle defined for this case. This could be used as your initial configuration. Finally, the **"Piston**" listbox allows you to select piston approximation using a number of point sources. Larger number of point sources will increase alculation time for diffraction.

Diffraction Segments Length

The length of the diffraction segments used to calculate SPL contributions from the edges, can be selected as 2cm, 4cm and 8cm.

Typically, you would select shorter segmants when designing smaller enclosure.

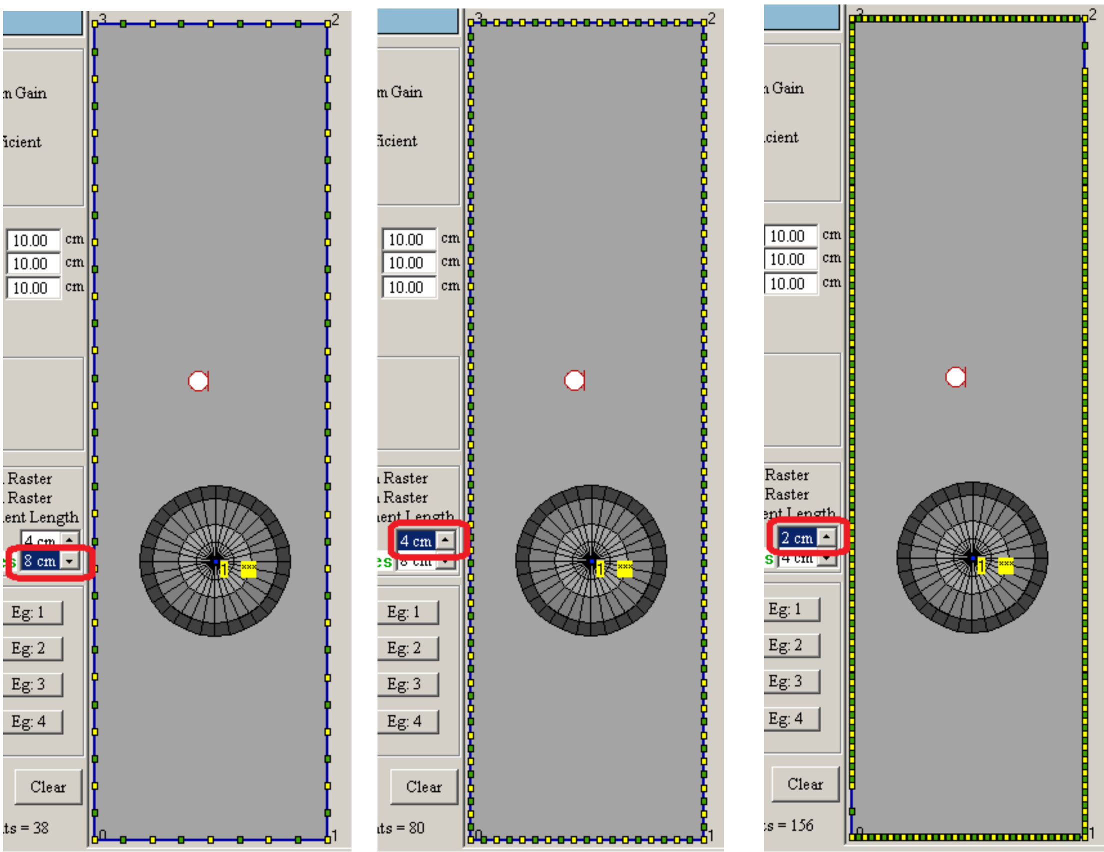


Figure 4.25. Selecting diffraction segments length.

Manual Nodal Coordinates

Nodal coordinates can be edited manually as well. This is accomplished by invoking a small dialogue box when Right Mouse Button is clicked upon the baffle drawing area. Once the box in visible, you can use Left Mouse Button to click upon the row with the Node Number you wish to edit. The node number and X/Y coordinates are transferred for editing and when you finished, please click on “Update This Node” button. When all editing is done, you can close the dialogue box by pressing on “Done” button.

This feature may come handy when you encounter difficulties using your mouse for entering vertices very close to the left/bottom edge of the drawing area. Should this be the case, the you can use the mouse pointer to position the vertices (nodes) in approximate locations, and then enter the exact coordinates using the list box provided.

Press on “Done” button when finished entering coordinates – this will close the box.

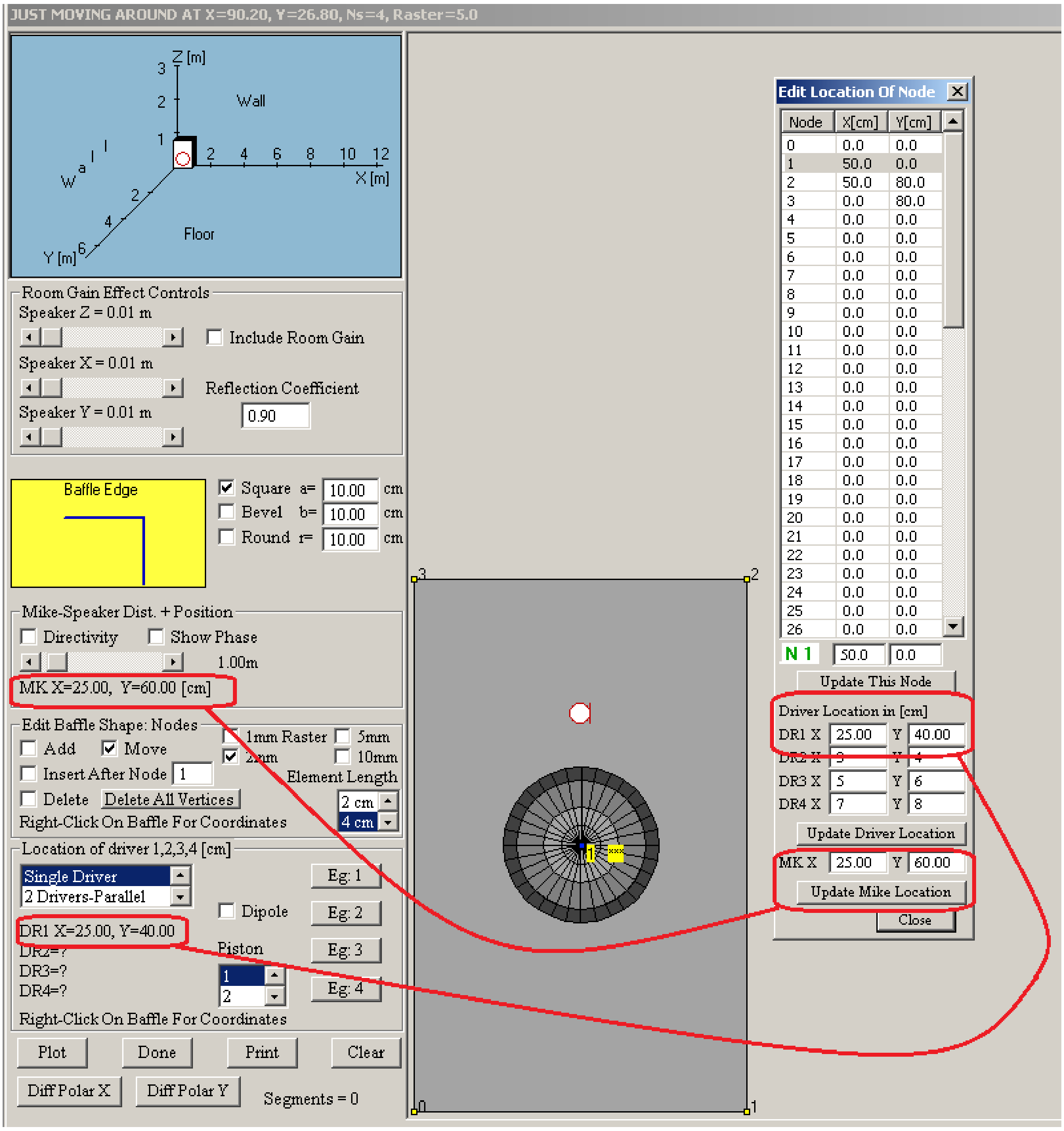


Figure 4.26. Entering nodal coordinates manually.

The “**Delete All Vertices**” button comes handy when you need to start fresh, without any nodes inserted. The button does not delete loudspeakers or microphone – just the nodes.

Drivers coordinate location on the front baffle can be edited from a floating dialogue box, activate by a mouse right-click above the baffle. After editing the coordinates, please click on “Update Driver Location” button and then close the dialogue box.

Finally, microphone location can be edited the same way.

Dipole Configuration

Diffraction effect can also be observed in dipole loudspeaker systems – see figure below. The built-in algorithm assumes that baffle thickness in zero.

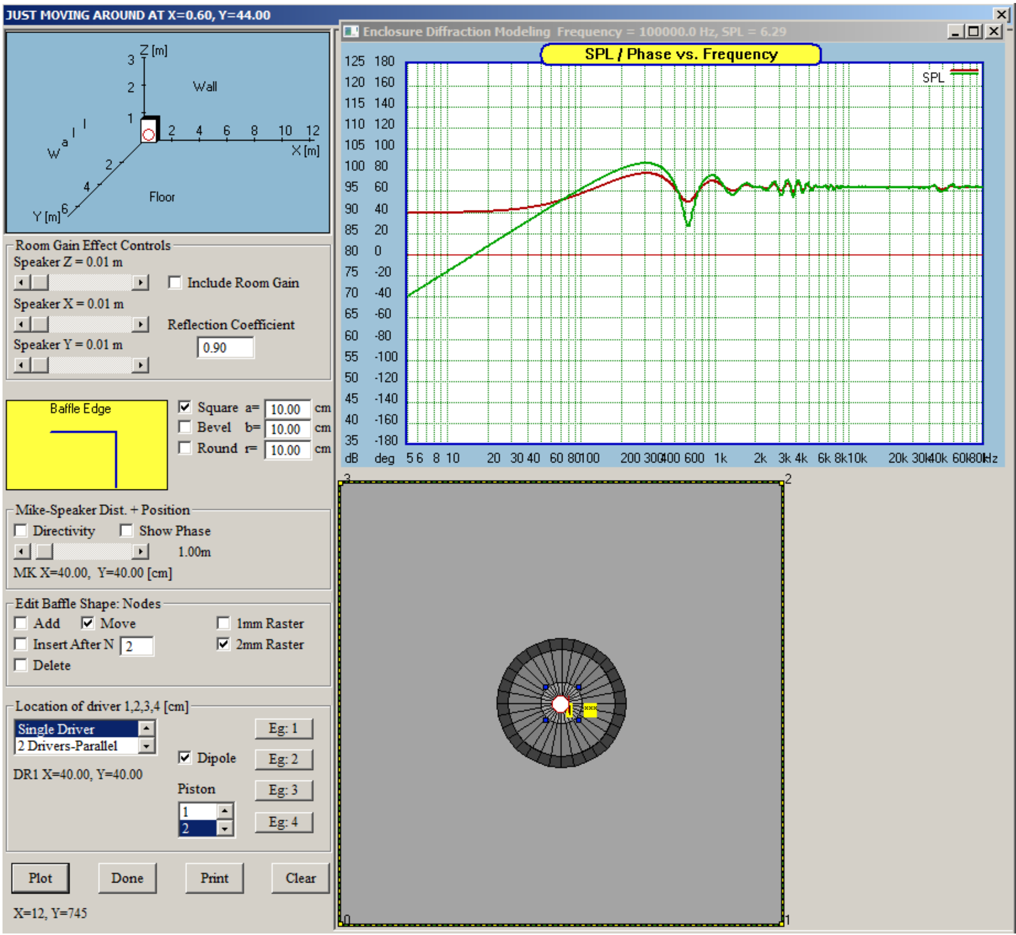


Figure 4.27 Example of conventional (brown) and dipole baffle diffraction

Room Gain Effect (Corner Placement)

Roy Allison has shown in his paper "The Influence of Room Boundaries on Loudspeaker Power Output", that loudspeaker frequency response in the low end of the frequency range is heavily influenced by it's placement near reflecting boundaries.

4PI – radiation into full space. This would be testing of loudspeakers in an anechoic chamber.
2PI – radiation into half-space. In this scenario, the loudspeaker would be placed on infinite baffle.
PI – radiation into quarter space. The loudspeaker is placed at the intersection of two planes (walls).
PI/2 – radiation into 1/8space. Here the loudspeaker is placed in a corner, formed by a three mutually perpendicular boundaries.

Subsequent works of R.V. Waterhouse in "Output of a Sound Source in a Reverberation Chamber and Other Reflecting Environments" offered a formula for an acoustic power radiating from source located near reflecting boundaries. Implementation of this formula is based on radiation from coherent sources approach, therefore, each boundary add +6dB to the total SPL.

Controls are very simple and include **X / Y / Z sliders** for positioning of the loudspeaker in the corner. Since typical domestic walls/floors are not 100% reflective, you can nominate a **reflection coefficient,** averaged for all reflecting boundaries. This number may be entered into "**Reflection Coeff**" data field. The **"Include Room gain**" checkbox activates this function.

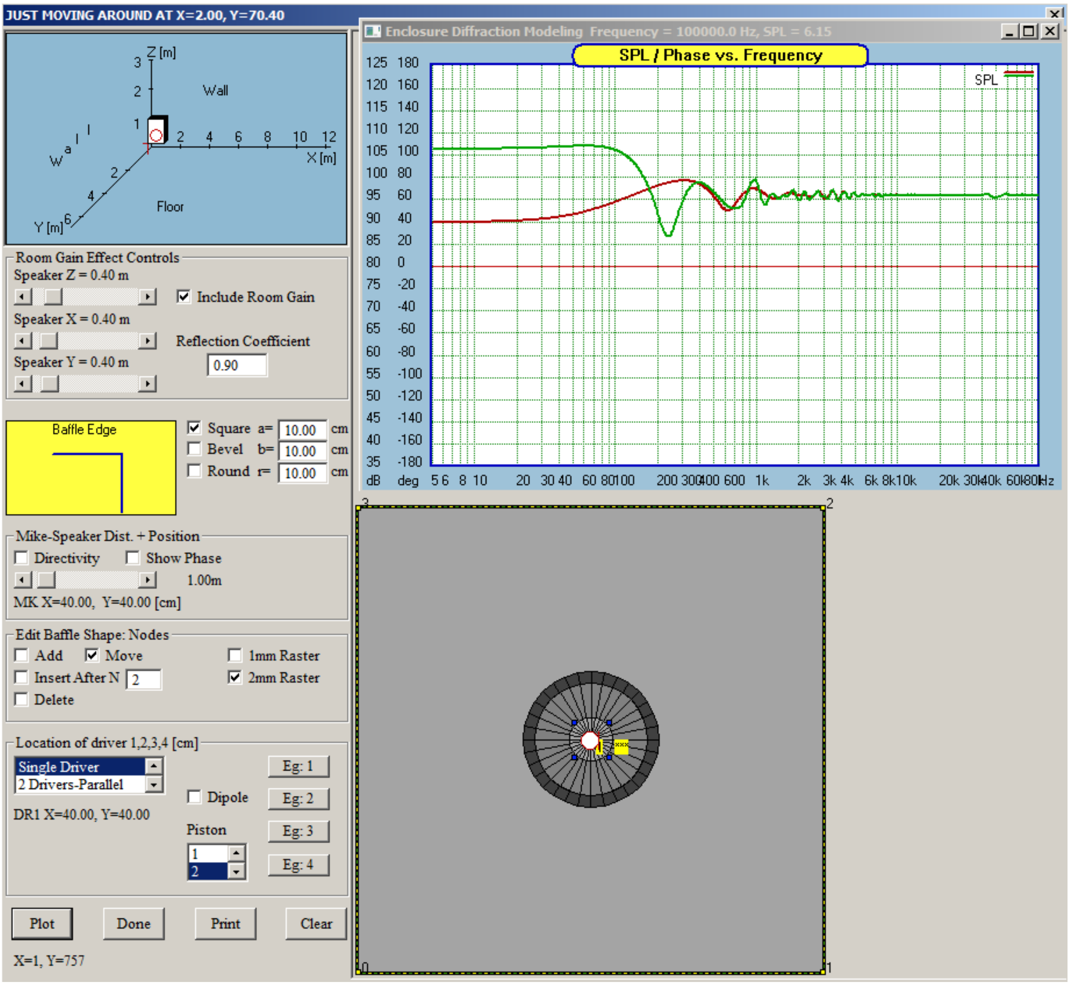


Fig 4.28 Effects of corner placement (green curve) on the SPL. Brown curve is just diffraction effect alone.

It needs to be noted, that the above functionality is mostly academic. Typical listening room is an enclosed space, and such, it will exhibit self-resonant modes. The resulting SPL will be heavily influenced by the modes and will exhibit a number of peaks and dips below Schroeder frequency. It is recommended to use FEM tools for modeling SPL in enclosed spaces.

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 4.24. 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 4.25 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 4.29 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

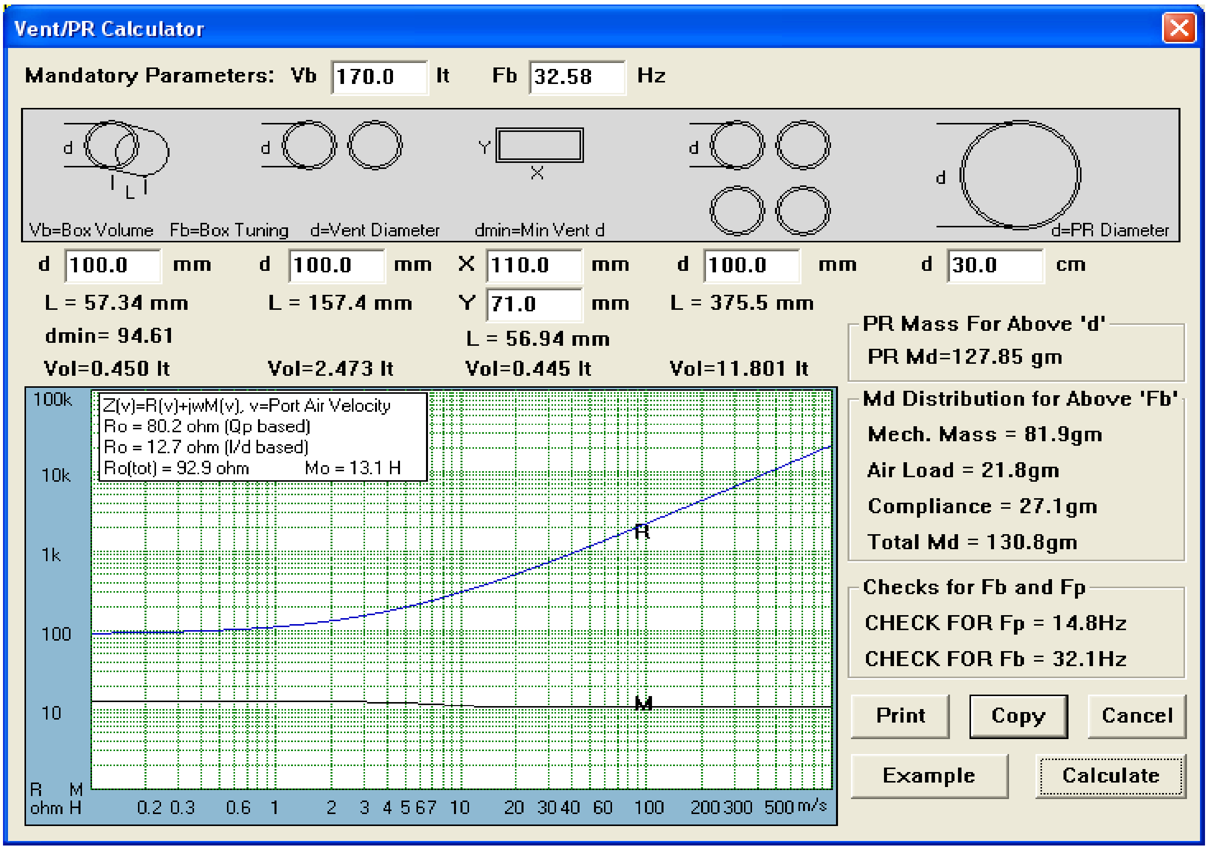


Fig 4.30 Tuning PR

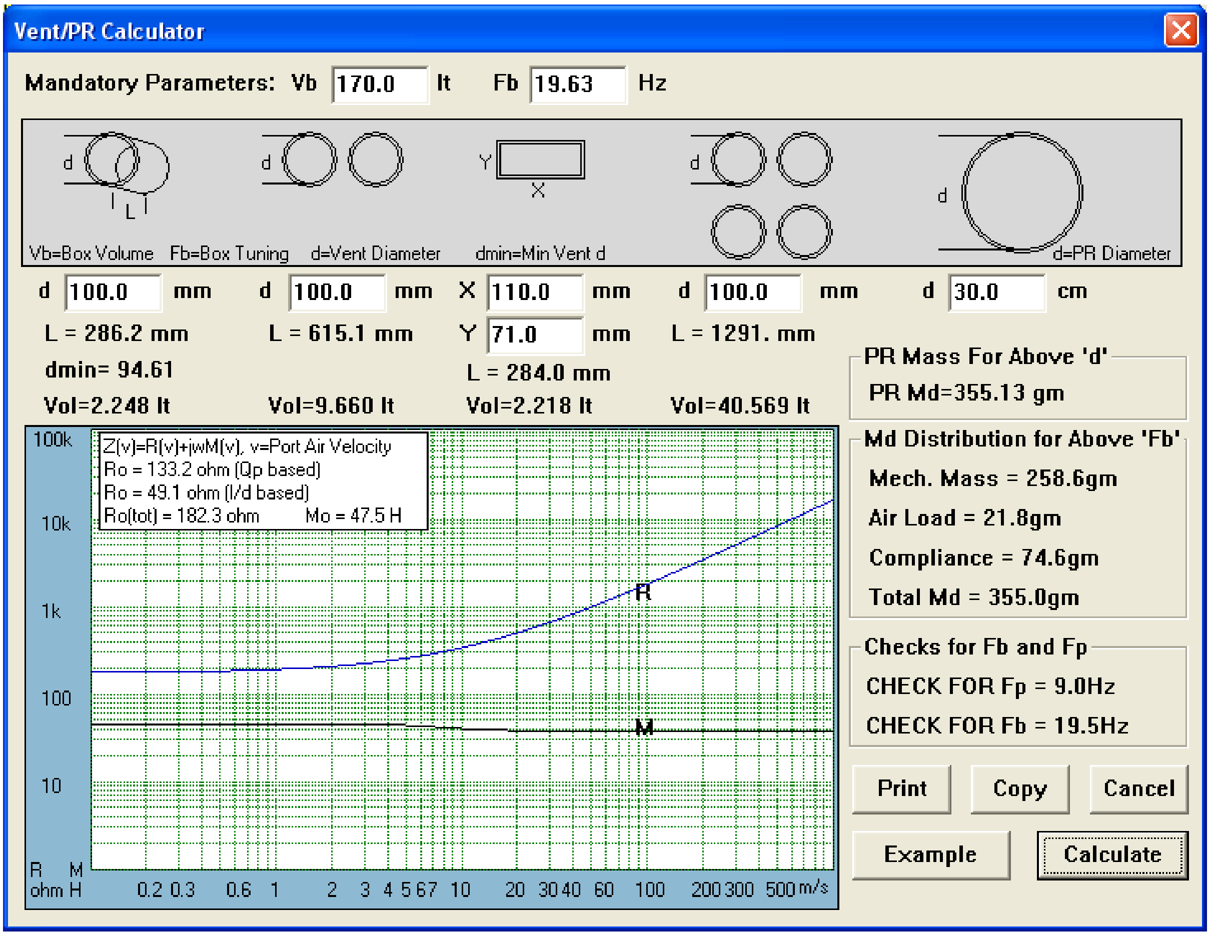


Fig 4.31 Re-Tuning the PR - continued

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 4.26. 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.

Please note, that you can enter the number of passive radiators from the Enclosure design Control box.

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.

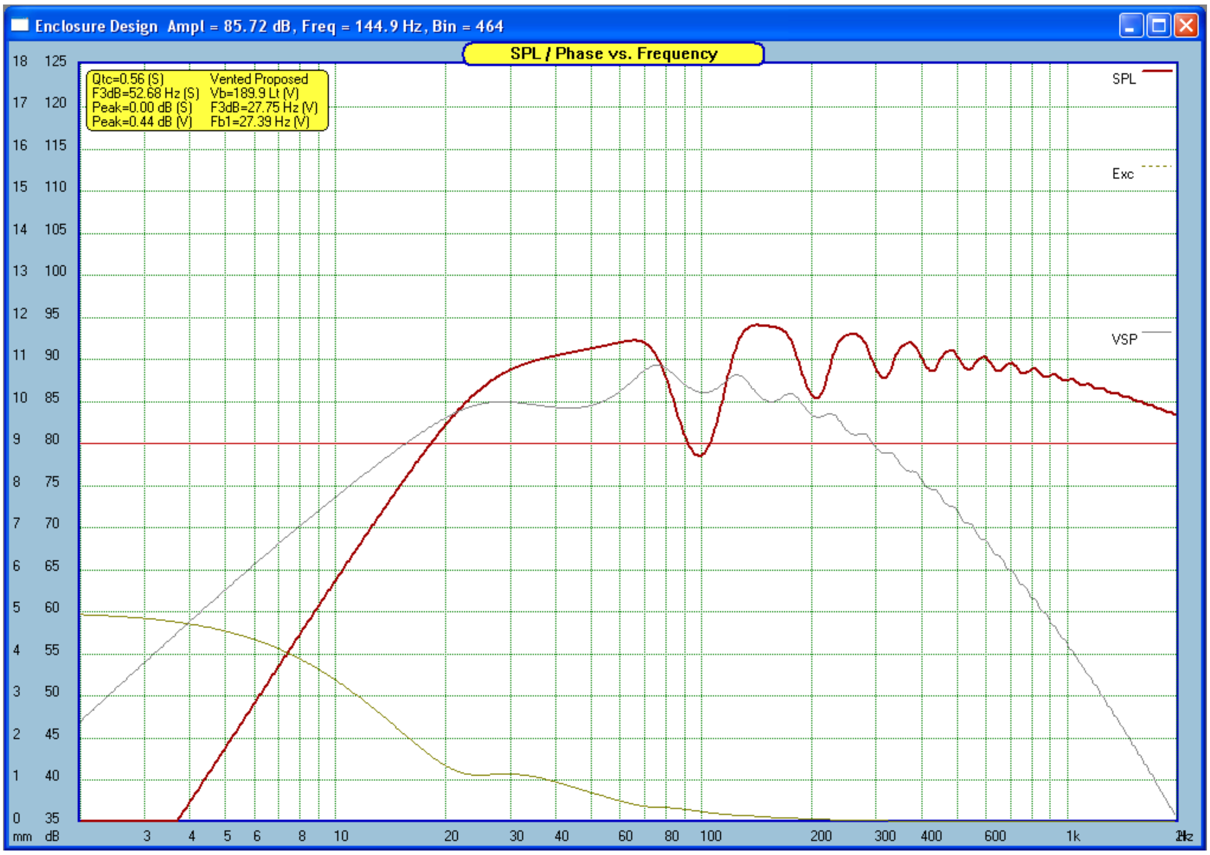


Fig 4.32 Transmission Line basic characteristics

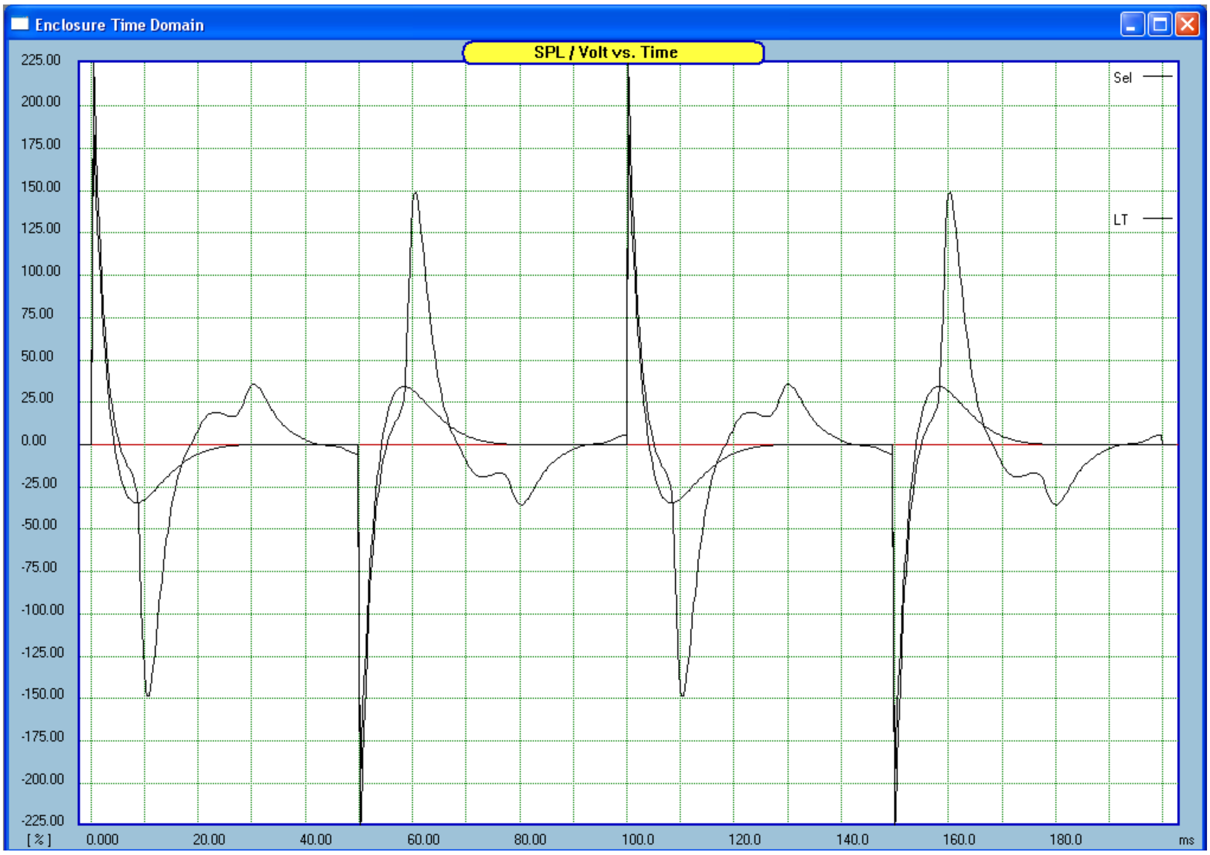


Fig 4.33. TL and Sealed box system transient response

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 4.28 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 4.29. 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.

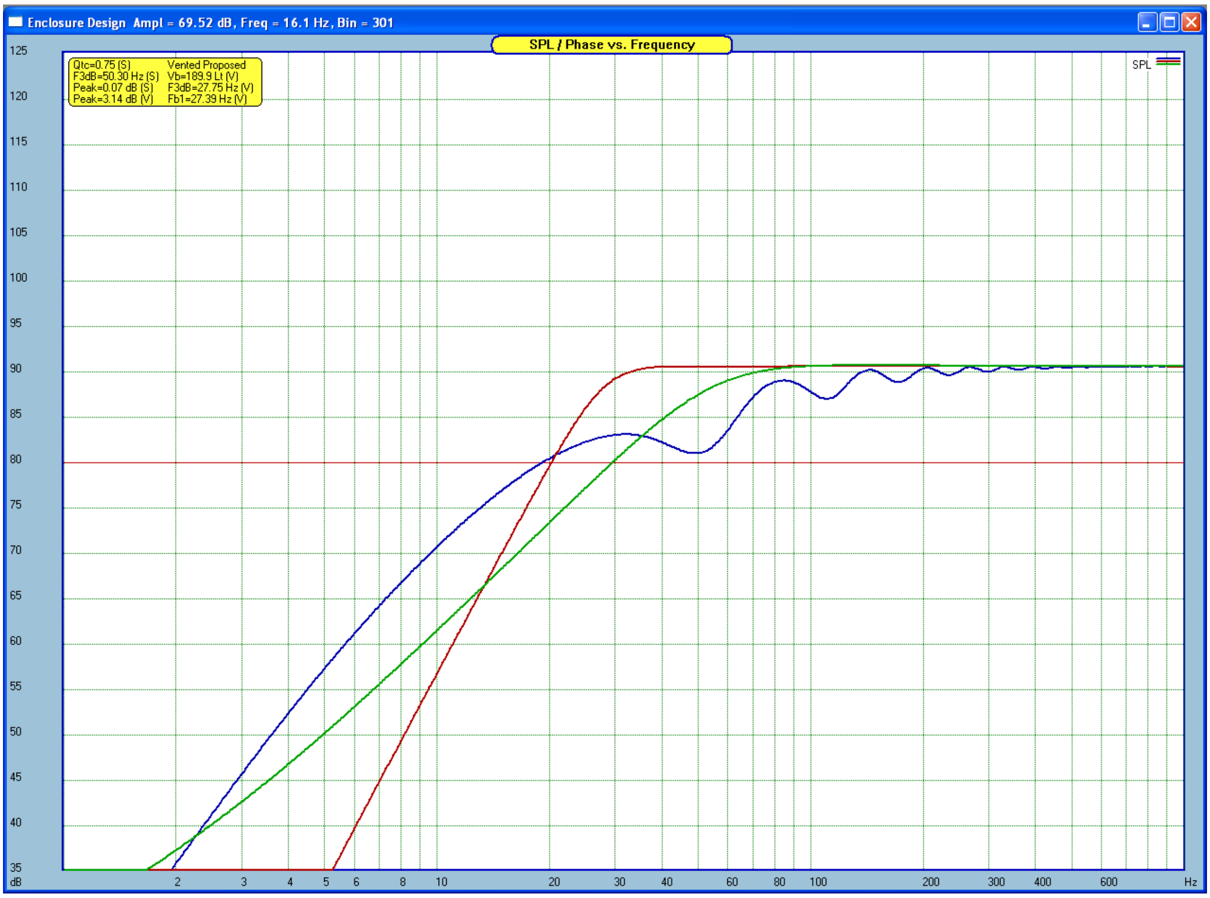


Figure 4.34 Sealed (green), Vented (brown) and TL (blue) system SPL

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.

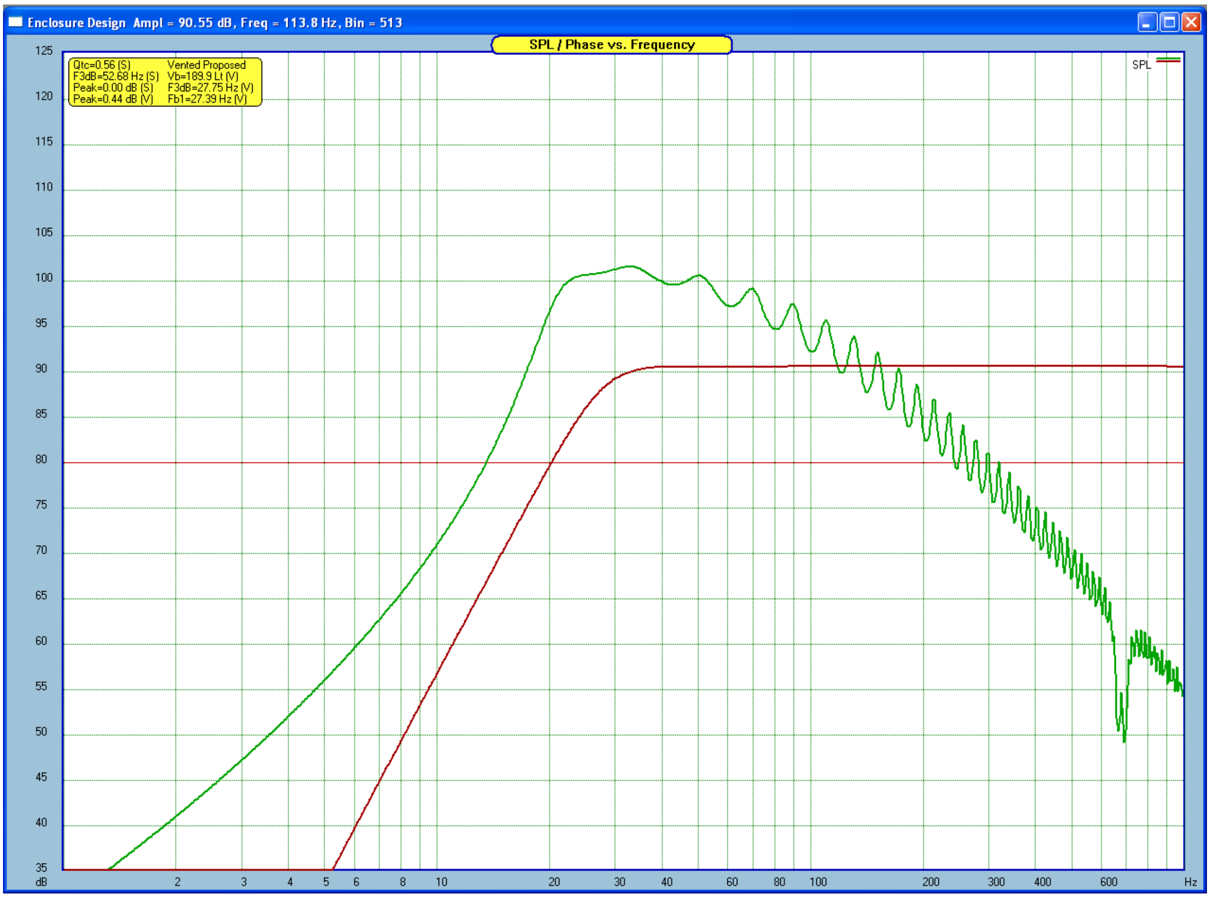


Fig 4.35. Horn (green) and vented box (brown) loudspeaker system comparison

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 4.30. 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.

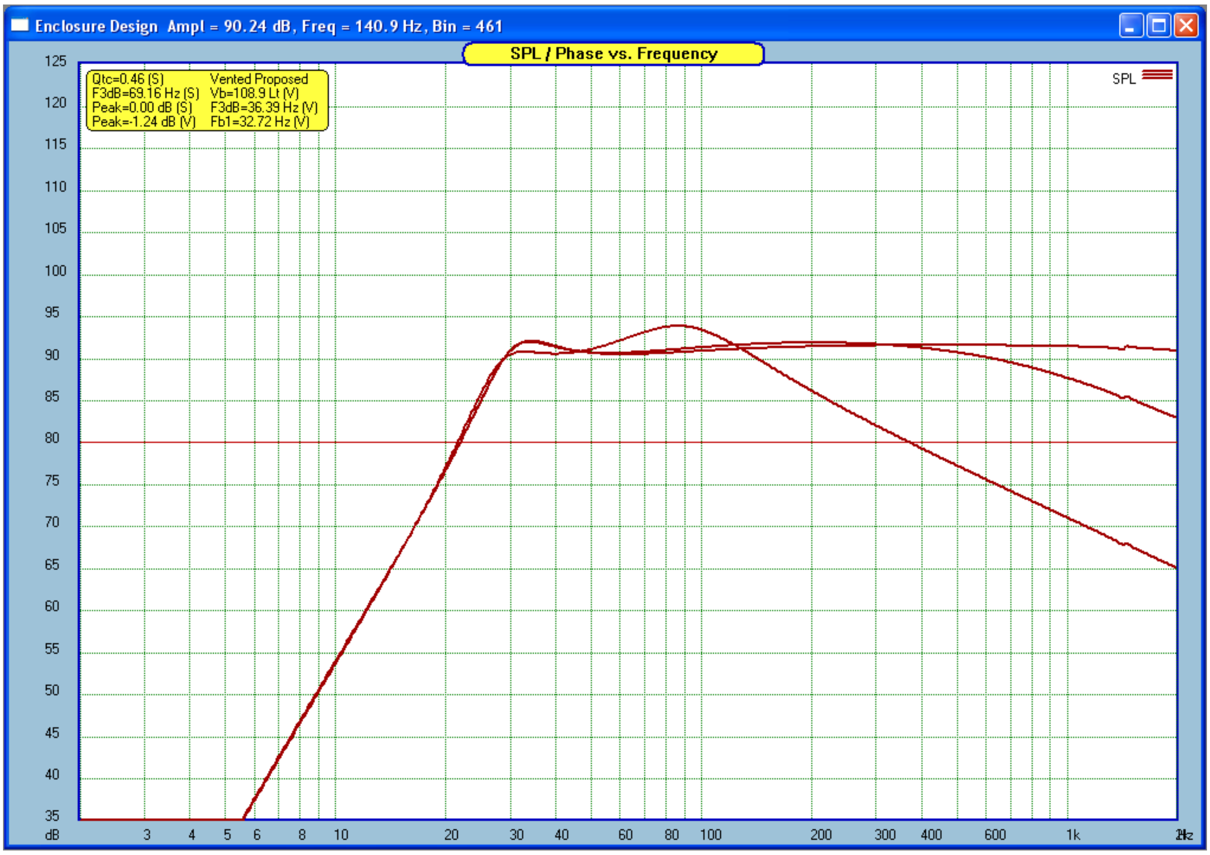


Figure 4.36. SPL comparison foe VC=0.2mH, 1.2mH and VC = 10.0 mH.

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 4.32. Polar plot at 20Hz clearly indicates directional characteristics of this enclosure.

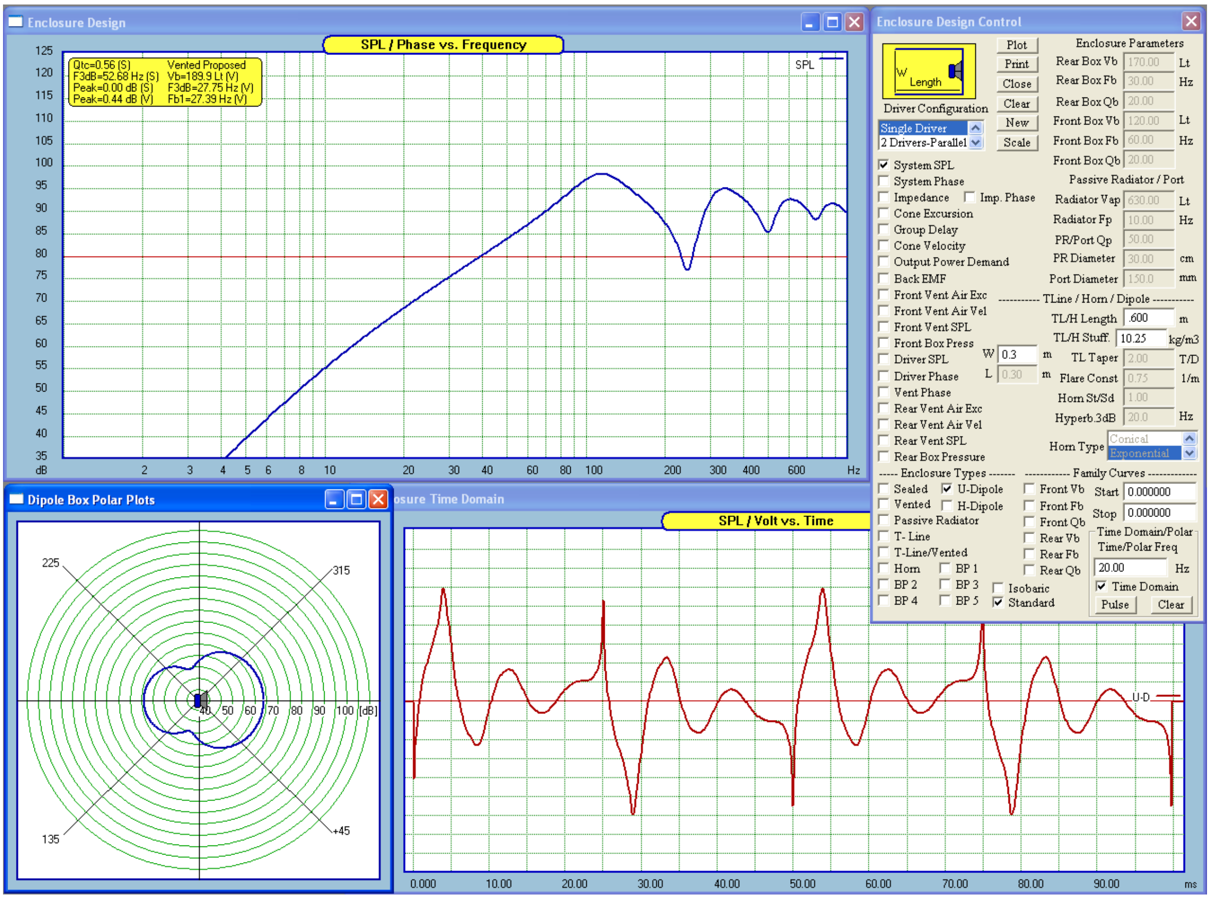


Fig 4.37. U-Dipole SPL, time domain response and polar plot.

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.

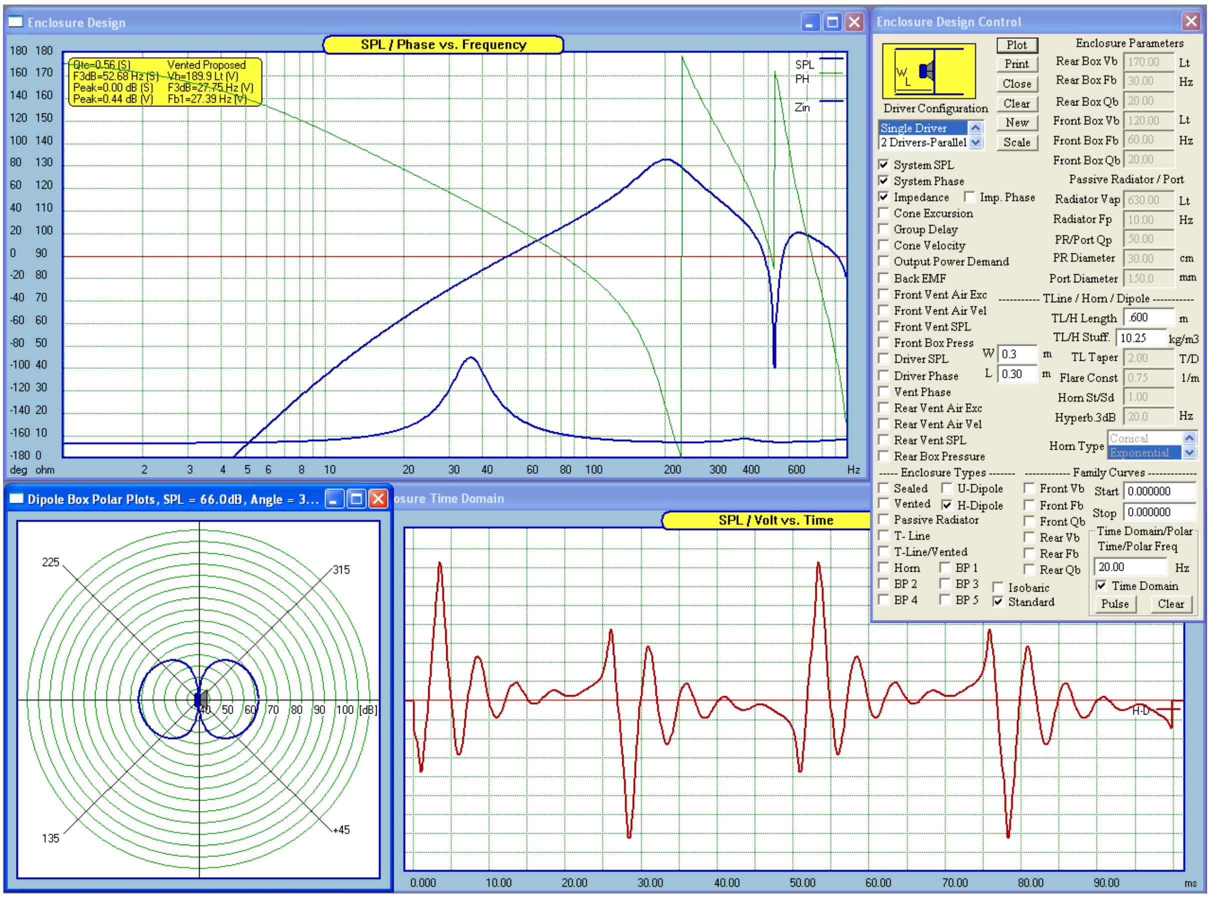


Fig 4.38. H-Dipole – note very characteristic polar plot.

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 4.34.

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.

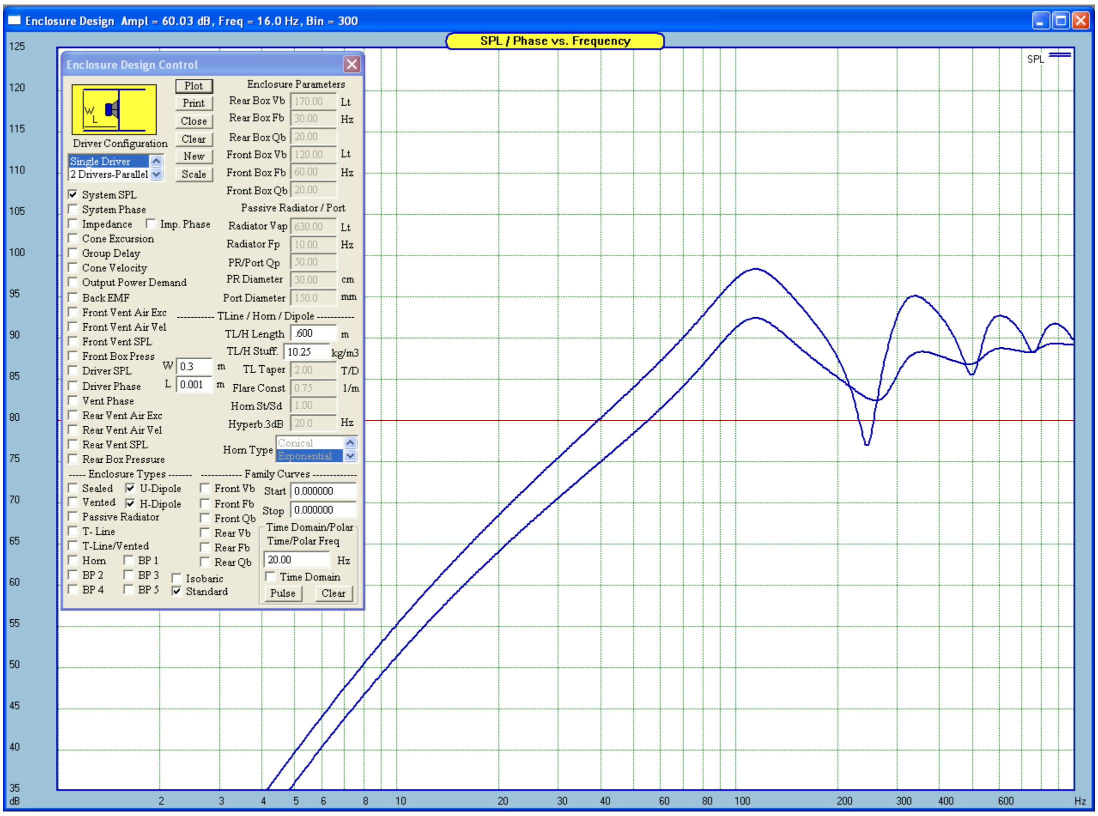


Fig 4.39. Converting H-Dipole into U-Dipole.

Polar plots vertical offset for H / U Frame dipoles

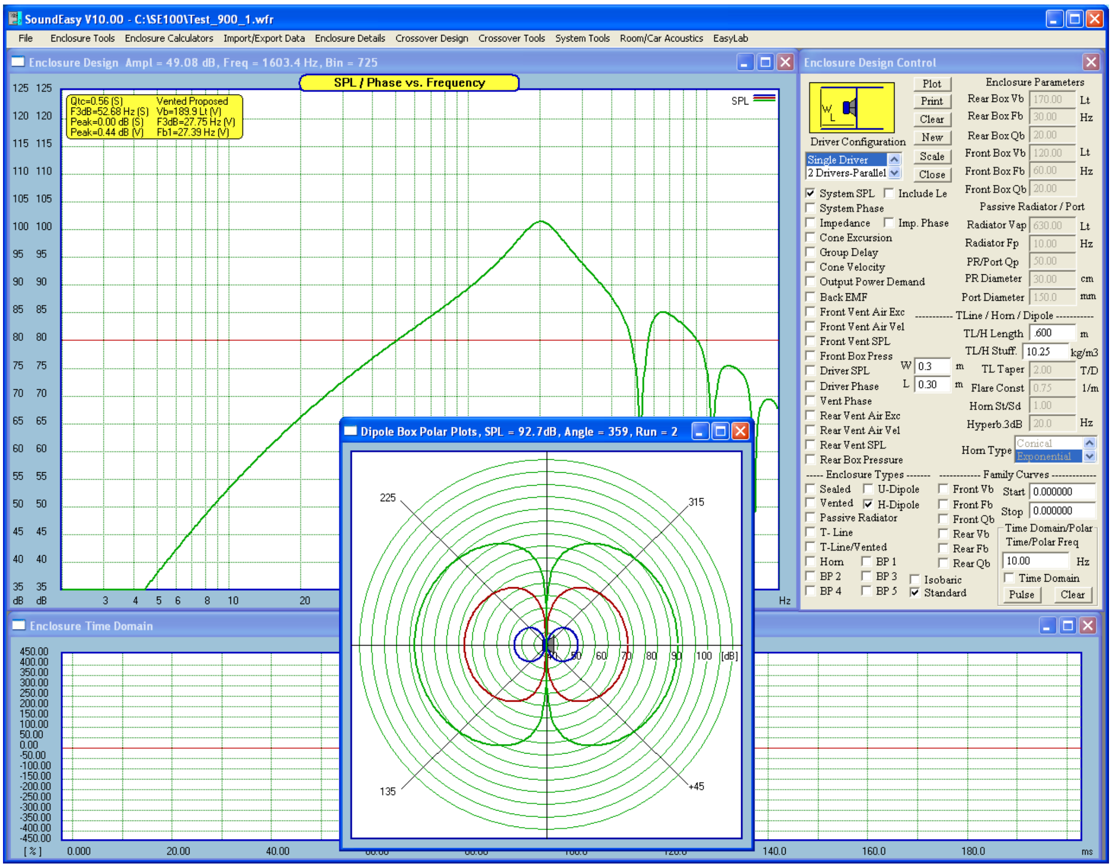


Fig 4.40. H-Frame dipole loudspeaker plotted to three offsets in 20dB steps

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 4.35 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.

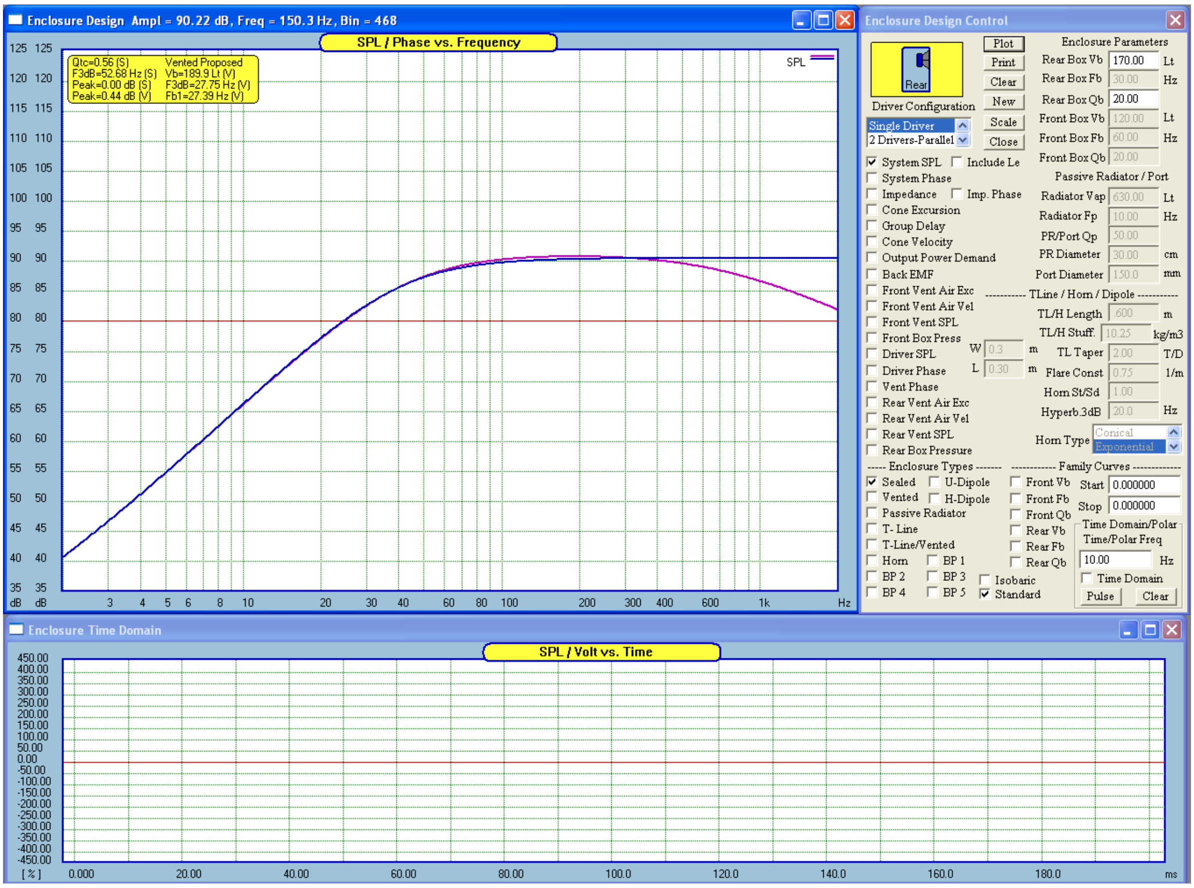


Fig 4.41 Example of plots with / without Le.

Equalizer TAB

Equalizing loudspeaker SPL curve is becoming one of the most important tools in subwoofer design. This is because equalization allows the designer to relax the size-for-bass extension restriction for purely passive implementations.

The Equalizer TAB is designed to facilitate all aspects of EQ circuit adjustments, with immediate effect visible on the SPL curve of the system. The usage of this feature can be enhanced by pre-building of your favourite EQ circuits, that you anticipate to use in your designs. The circuits could be tested and analysed using the in-built CAD system, so you are fully familiar with their limitations and advantages. Each EQ circuit should be saved in a file with a meaningful name, so you can quickly identify basic circuit features by looking at it’s file name.

If your EQ circuit is a +/-6dB, +/-12dB shelving circuit, a peaking variable-Q circuit, a low-pass filter, a high-pass filter, then the selection is even simpler. The Equalize TAB has a selection of circuits built-in for easy access. There is a good chance, that your EQ circuit is one of those available, as a built-in circuit. The usage of this feature can be described as follows:

1. Load a driver file. This will provide you with all driver/enclosure parameters for modelling.

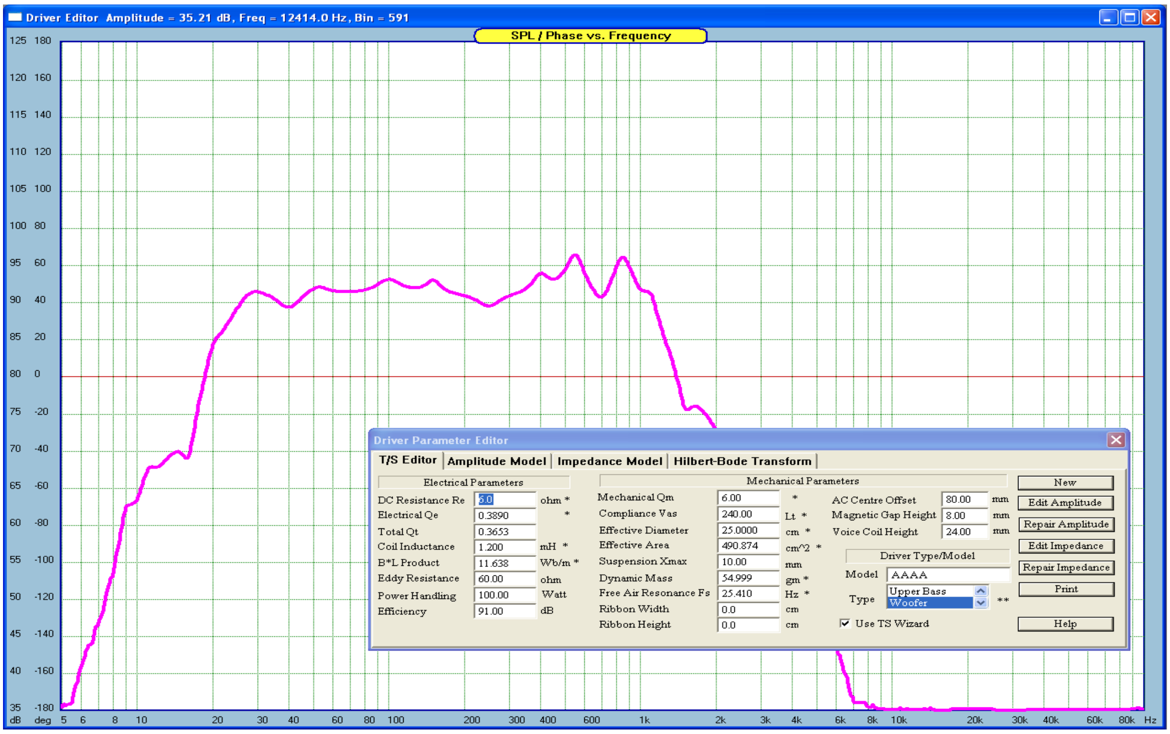


Fig 4.42 Driver file loaded into the program.

2. Go to Enclosure Design section and select SPL curve for plotting, and also all necessary parameters of the enclosure.
3. Select Equalizer TAB.
4. Select “Load Circuit” option from the main menu to load appropriate EQ circuit. Once loaded, the name of the file will be displayed in the Equalizer TAB.

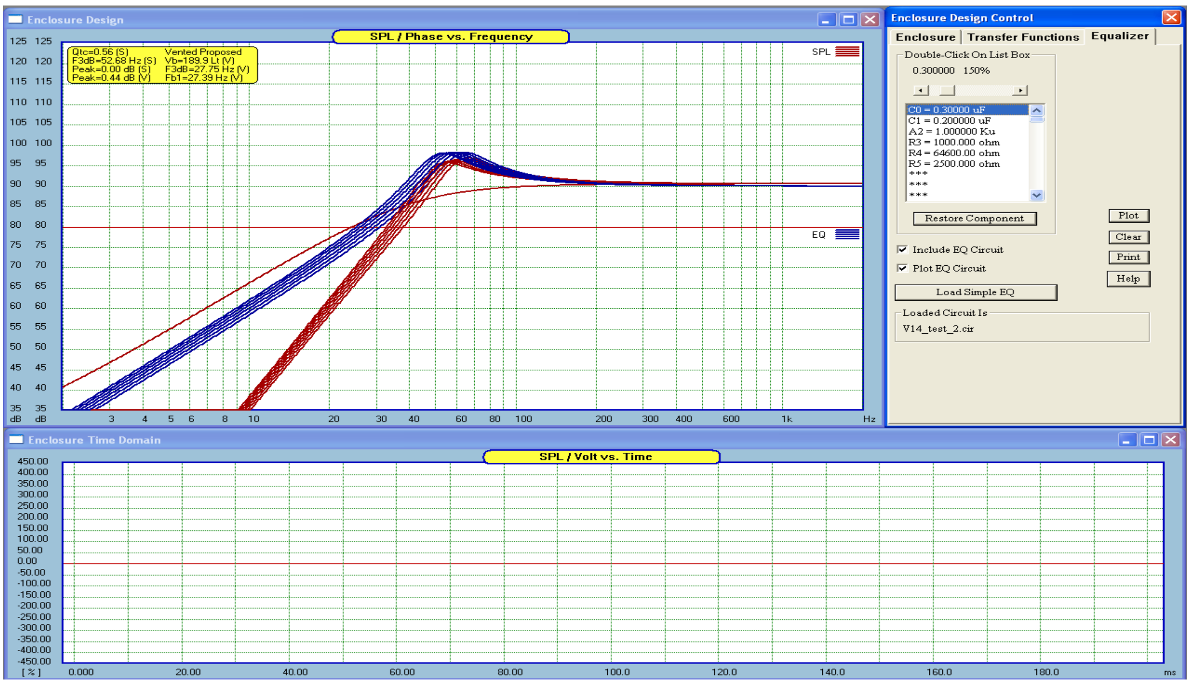


Figure 4.43. Effect of adjusting a component value on enclosure’s SPL.

5. Check the “Include Equalization Circuit” and “Plot Equalization Circuit” boxes.
6. Press “Plot” button to see the effect of the EQ circuit on the SPL curve.
7. If you need to fine-tune the EQ component values, please DOUBLECLICK on the required component listed in the list-box. This will select this particular component for slider adjustments. This process is the same as the one used in the CAD and other sections of the program.
8. Adjust the component value. Once you are happy with the results, you can save the EQ circuit into the file, and also save the driver file.

Please note, that the driver file will now also contain the newly designed EQ circuit. So the next time you need to re-edit the EQ circuit, there is no need to look for it. It will be included in the driver file. **As a final note, the EQ circuit must not include driver’s symbol in the schematic.**

Imperial Units Display

T/S Editor, Enclosure Design and Vent Calculator display data in Imperial Units as well as SI units.

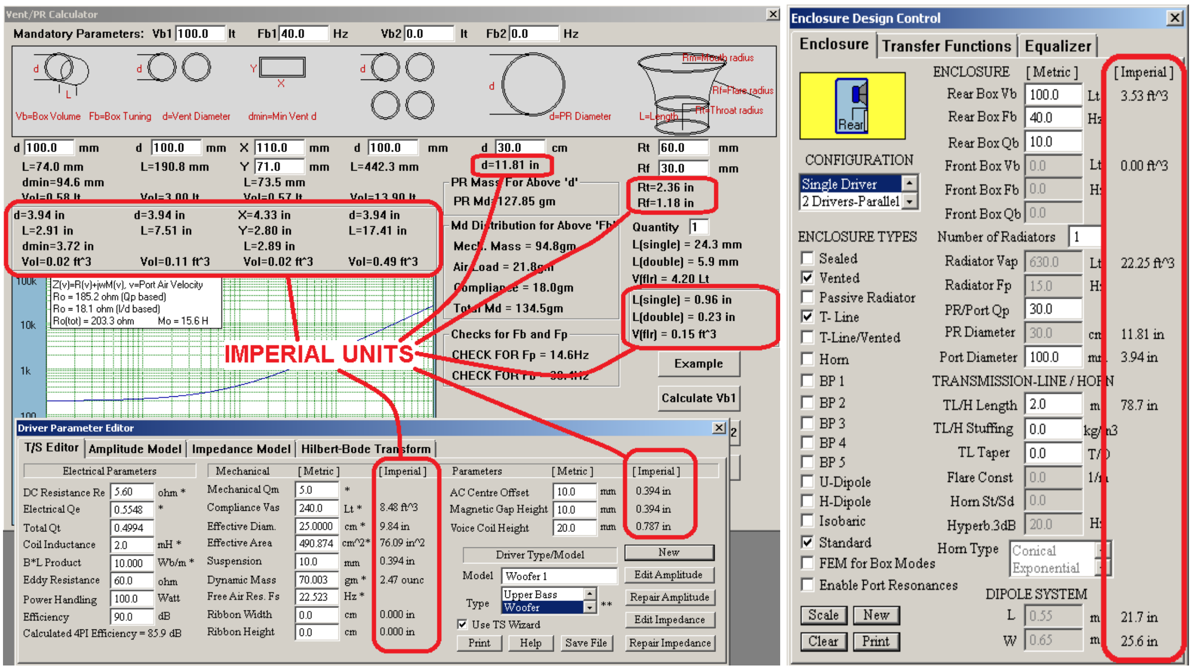


Figure 4.44. Automatic Imperial Units display in various dialogue boxes.

Enclosure Design Curve Labelling

All curves plottsd in Enclosure Design section are clearly labelled, including the Family Plots curves. An example presented below shows Family Plots for selected Rear Box Vb from 50Lt – 450Lt for Vented enclosure.

Figure 4.45 Family Plots with labels.