Loudspeaker Diffraction Distortion and Radiation Impedance

By Bohdan Raczynski

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 I 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 [2] for example, develops several useful rules for improving listening pleasure in typical room, with the emphasis on reducing 1-st reflections. Therefore, typical listening environment is nowhere near the "reverberant room", which reflects all the sound produced inside it and makes the diffraction distortion difficult to separate. Finally, there is the issue of consistency in the design approach. Here is one point of view. It is known, that room modal response at low frequencies (standing waves) causes 30dB variations in the low end of the frequency response of the loudspeaker, but this has NOT deterred anybody from using technically sound approach (Small/Thiele parameters) for proper design of the enclosure.

Why should this approach be limited only to the low end of the audio spectrum ?. In the mid to high frequency range room reflections will also distort the frequency response of the loudspeakers, so should the design methodology be relaxed and the diffraction distortion neglected in this frequency range ?. We would advocate careful consideration of the "whole picture". At least as long as loudspeakers are tested and compared in anechoic chambers and placed away from the walls in well designed listening rooms. You see, at some point of the design process, you need to gain the confidence, that you are creating the best source of sound you can come up with. And if it still does not sound right to you, it must be the listening room environment, which needs corrections - NOT the speakers. And if you ever even think about having your speakers evaluated in an anechoic chamber, diffraction distortion should become your daily bread.

Situation becomes even more complex if several identical drivers are placed on the front baffle. Mutual radiation impedance effect becomes evident strongly at lower frequencies with the cut-off frequency determined by the geometry of the design.

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

Diffraction distortion

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

Plainly speaking, when testing in the anechoic chamber, at high frequencies the speaker is radiating into "half space" (**2π)** i.e. it is only radiating into the forward hemisphere. At low frequencies the speaker radiates into "full space", (4**π)** exhibiting a loss of bass when implemented in typical speaker enclosures. The difference in the SPL is 6dB and is referred to as the "6 dB baffle step" or the enclosure's "diffraction loss". The location of the "baffle step" on the frequency scale is dependent on the dimensions of the baffle. The smaller the baffle the higher the transition frequency. Diffraction Loss modeling performed for this article is based on the GTD (Geometric Theory of Diffraction). Figure 1 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.

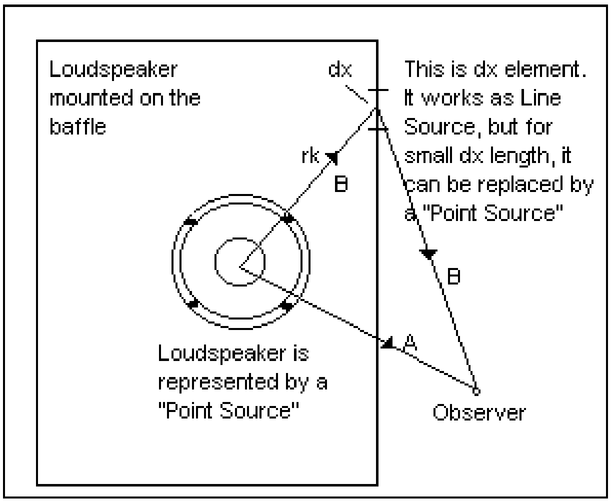

Figure 1. GTD concept.

Loudspeaker enclosure or baffle and the placement of the driver contribute up to +10dB to the frequency response of the system. The GTD using ray model is applied to determine the exact amount of SPL deviation due to the diffraction.

In this model, sound rays, B, propagate along the surface of the baffle and are scattered when encounter the edge of the baffle. This secondary sources combine with the direct rays, A, produced by the loudspeaker and the resulting frequency response is far from being flat. To calculate the contribution of baffle edge, total length of the baffle edge is quantized into a number of sections of length ***dx***. The average distance from each section to the "point source" representing the loudspeaker is ***rk***. If ***dx*** is made sufficiently small, it can be replaced with a "diffraction point source".

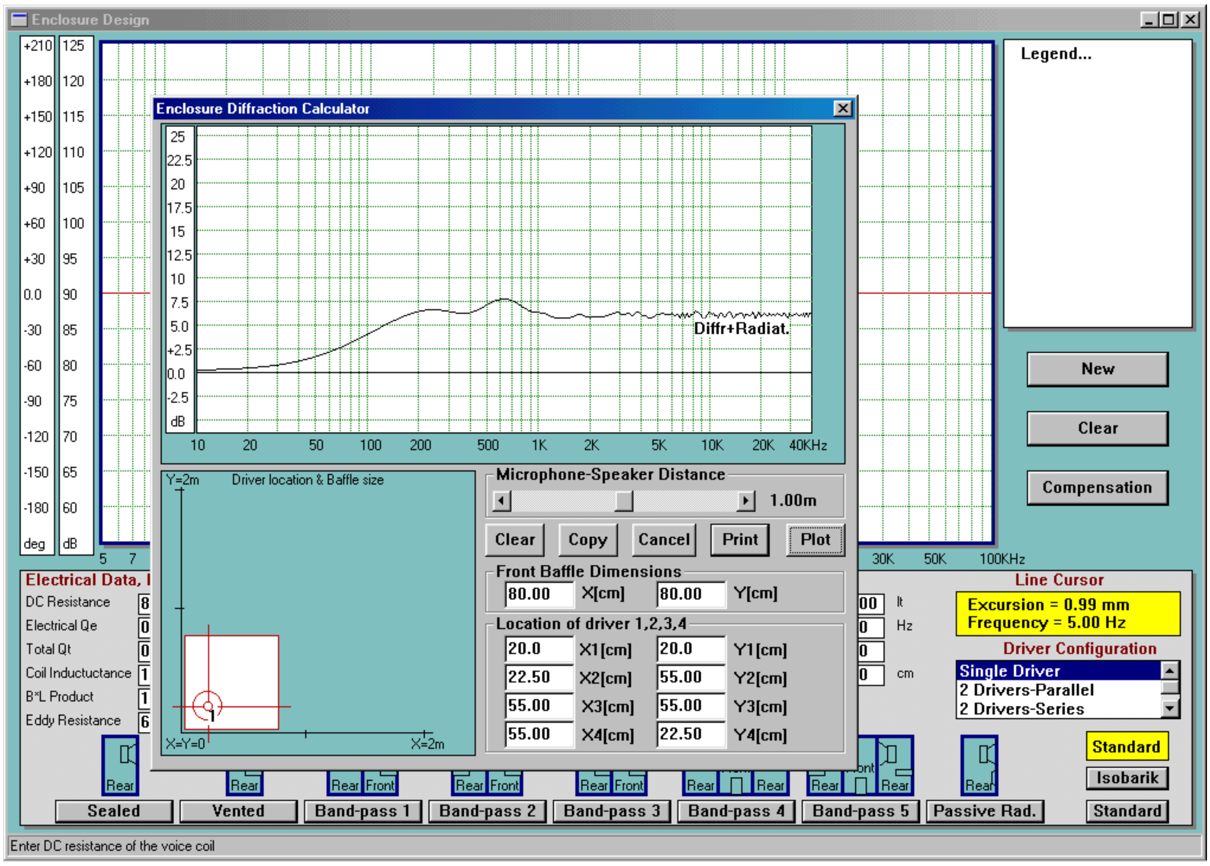

Figure 2. Single driver diffraction

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. I have selected 48 ***dx*** line elements to represent top and bottom edge of the baffle and 96 ***dx*** line elements to represent the left and right side of the baffle. Loudspeaker driver is modeled as 8 "point sources" contributing to the direct SPL. For the purpose of modeling the diffraction loss, I also assume, the reference distance to the observer to be the standard 1 meter. This situation is depicted on Figure 2.

You can also change loudspeaker location on the front baffle, the dimensions of the baffle and microphone distance. **Asymmetrical mounting of the loudspeaker will result in "smoother" frequency response**. The enclosure diffraction effect is much less pronounced when the test microphone is placed much closer to the loudspeaker. The distance that can be preset the "direct" sound (ray A) has to travel is only 1cm, but the "diffracted" (ray B) wave has to travel to the edge of the enclosure and back, therefore it will arrive at the test microphone at much lower level. This situation is depicted on Figure 3. 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.

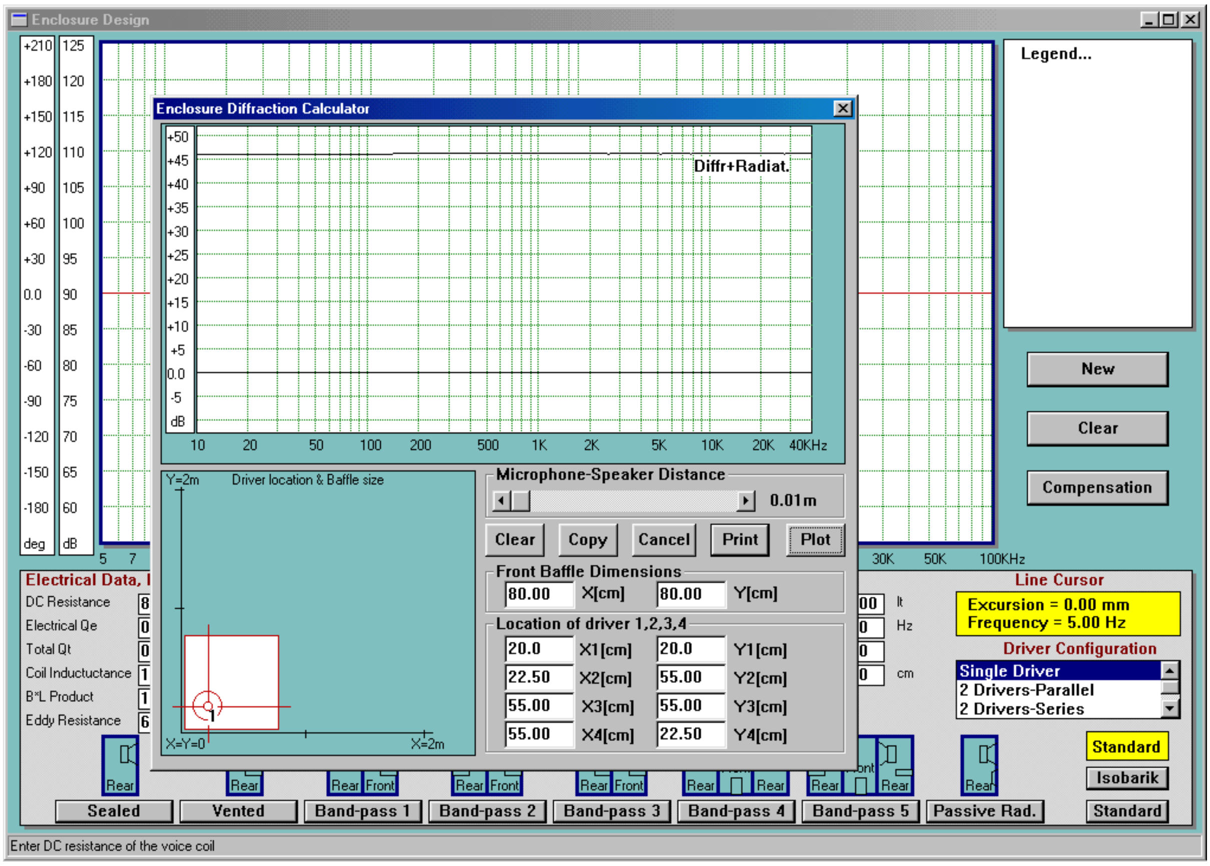

Figure 3. Near-field diffraction

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.

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 [3] 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) 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 [4] indicated, that this additional increase in effciency will hold to a frequency above which the diaphragms no longer "couple". This phenomenon has been experimentally verified by Gander and Eargle in [5]. 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 [6], investigating the performance of Bessel Arrays concluded that an array of two loudspeakers simply connected in-parallel, exhibits maximum SPL increase of 6dB, but only up to a frequency where the sources are about 1/4 wavelength apart.

Signal Summation Strategies

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

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

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

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

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

The Phasor Summation technique was used in calculating diffraction from the enclosure edges. As you may recall, we specifically looked at each signal's path length and added all arrivals as vectors.

It is fair to point out, that Phasor Summation method may produce results that are mathematically quite correct, but would appear worse than listening tests would indicate. This disparity is associated with the ear's critical bandwidth and suggests, that reproduced sound should not be evaluated based on single frequencies, but rather on 1/3 octave bands. This is rather interesting, as the optimum modeling concept seems to be a combination of Power Summation and Phasor Summation techniques. In any case, for a truly representative modeling the sound pressure magnitude and phase needs to be included in the prediction process.

Combining All the Above

We are now in good position to review the SPL gains due to:

1. enclosure diffraction,
2. mutual radiation impedance and
3. multiple drivers in the same enclosure.

We assume that: (1) our amplifier is an ideal voltage source - this is the case of most currently available amplifiers. The amplifier will deliver 10VRMS to the load. (2) all drivers are identical and have real impedance of 4ohm. For the input voltage of 10VRMS, the loudspeaker will deliver 100dB SPL. Finally, (3) for the purpose of evaluating SPL levels for different combination of drivers, we will keep the amplifier output constant (no change in volume level**). The 0.0dB reference level shown on following figures corresponds to 100dB SPL.**

Single driver.

This is our reference case. The amplifier will deliver 25W electrical power to the speaker (U*U/R = 10*10/4 = 25 W) and the loudspeaker will now generate 100dB 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 2. Being the ideal voltage source, the amplifier will also cope well with the 8ohm load impedance being now presented to it (two 4ohm loudspeakers connected in-series). Each driver will now receive only half of the 10VRMS voltage generated by the amplifier.

With this in mind, each speaker will receive only 6.25W of electrical power (U*U/R = 5*5/4 = 6.25 W) Each driver will now generate only 94dB SPL, so that total SPL of the system is now 97dB. The diffraction effect will add +6dB in the upper end of the operating frequency range and the mutual radiation impedance effect will add +3dB in the lower end of the operating frequency range. The final SPL gains curve is shown on Figure 2. The electrical power delivered to the system is now only a quarter (12.5W) of the single driver configuration.

Two drivers connected in-parallel.

Being the ideal voltage source, the amplifier will cope well with the 2ohm load impedance being now presented to it (two 4ohm loudspeakers connected in-parallel). Each driver will generate 100dB SPL, so that total SPL of the system is now 103dB. The diffraction effect will add on the top of it +6dB in the upper end of the operating frequency range and the mutual radiation impedance effect will add +3dB in the lower end of the operating frequency range. The final SPL gains curve is shown on Figure 4. It is worth noticing, that electrical power delivered to the system is now twice (50W) of the single driver configuration.

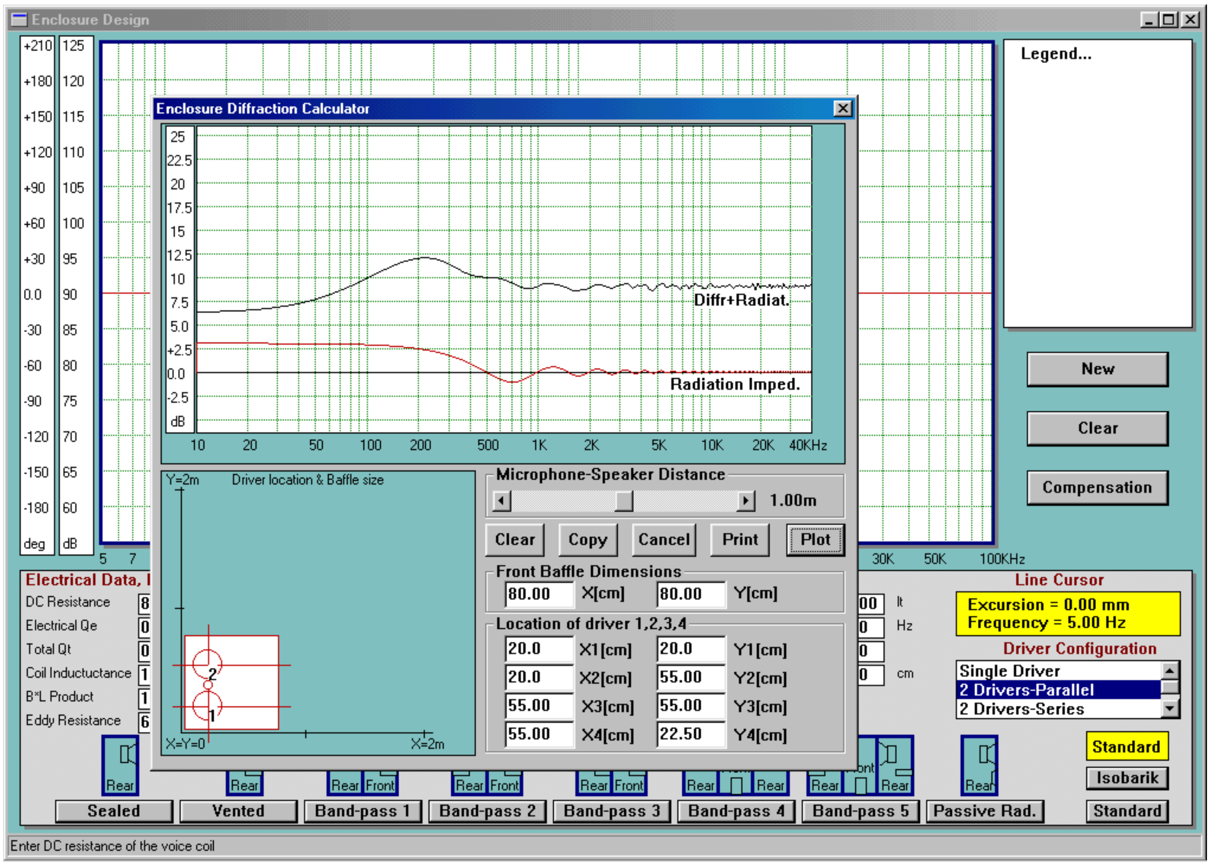

Figure 4. Two drivers connected in-parallel

Two drivers connected in-series.

Each driver will now receive only half of the 10VRMS voltage generated by the amplifier. With this in mind, each speaker will receive only 6.25W of electrical power (U*U/R = 5*5/4 = 6.25 W) Each driver will now generate only 94dB SPL, so that total SPL of the system is now 97dB. The diffraction effect will add +6dB in the upper end of the operating frequency range and the mutual radiation impedance effect will add +3dB in the lower end of the operating frequency range. The electrical power delivered to the system is now only a quarter (12.5W) of the single driver configuration.

Four drivers connected in-series and in-parallel.

This type of configuration results in the system input impedance equal to that of single driver (4ohm). Therefore the electrical power delivered to the system is now 25W. The SPL gains curve can now be constructed from two SPL levels representing drivers connected in-series (the whole curve will raise by +3dB) and again, the mutual radiation impedance effect, which will add +3dB in the lower end of the operating frequency range. The final SPL gains curve is shown on Figure 5. 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 6.25W 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".

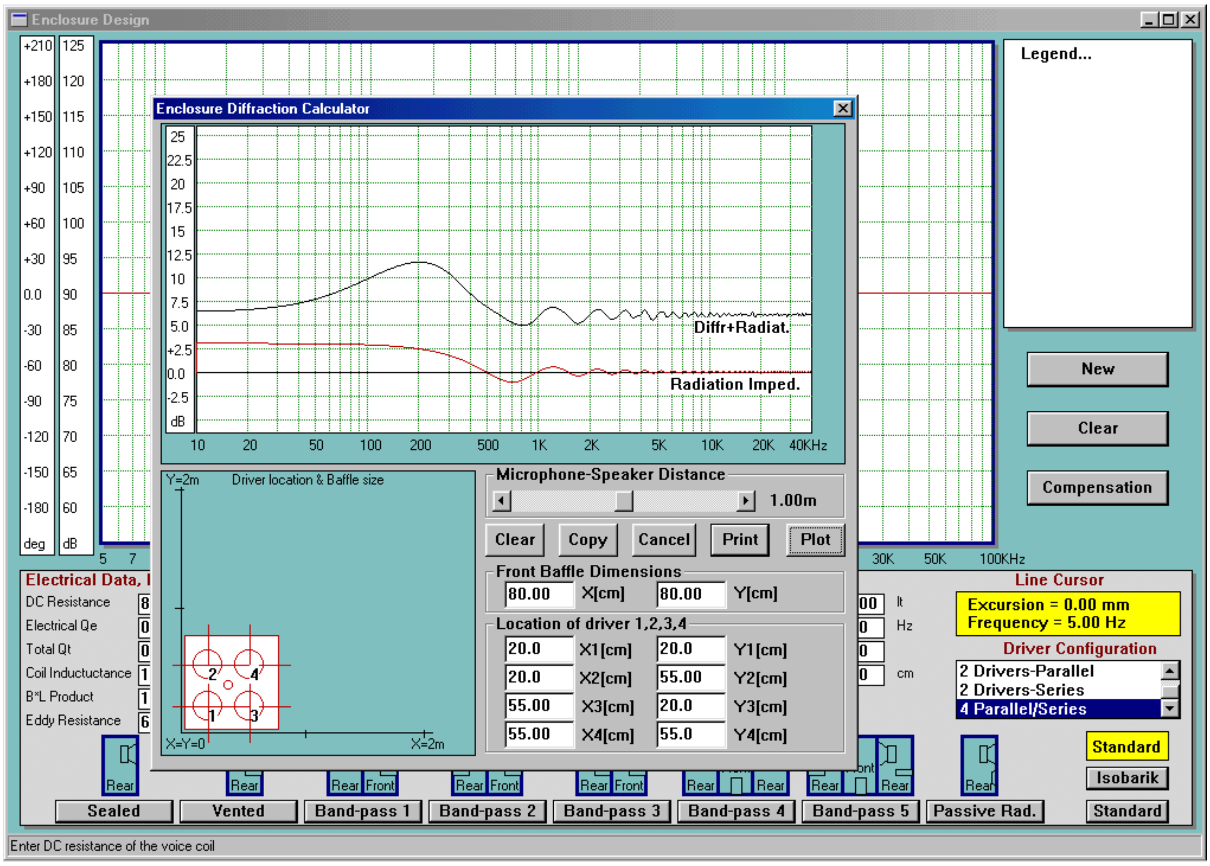

Figure 5. Four drivers connected in-series and in-parallel - Quad Box

References:

[1]. Harry F. Olson, "Direct Radiator Loudspeaker Enclosures", Audio Engineering, November, 1951.

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