Auri: A Double Chamber Reflex Speaker

"Sometimes They Come Back. . . Again." This is the title of a 1996 horror movie [sequel]. In this article there's nothing scary, just an old reflex variation that reappears.

By Claudio Negro

Auri is a floor-standing two-way speaker that uses the unusual double chamber reflex (DCR) enclosure. This kind of system was proposed in December 1961 in *Electronics World* by G.L. Augspurger and later on resumed first by R. N. Marsh (*Speaker Builder* 3/80) and then by David B. Weems (*Speaker Builder* 4/85 and 1/92 and in several of his books). To avoid confusion, **Fig. 1** shows the bass reflex and four of its variations, including the one of Augspurger-Weems. All measurements were done using the freeware program *Speaker Workshop*[1], with the exception of the room response, and all the SPL values are not absolute.

INSIDE THE DCR

The speaker is made of two chambers, one of which is twice the size of the other, with the woofer placed in the bigger chamber (**Fig. 2**). In the first chamber, V_1, there are two ports: the first, P_1, tunes V_1 with the outside while the second port, P_2, tunes the two chambers, $V_1 - V_2$. Finally, the third port, P_3, tunes V_2 with the outside. Therefore, three different resonance frequencies will be produced: F_1 related to $V_1 - P_1$; F_2 related to $V_1/V_2 - P_2$; F_3 related to $V_2 - P_3$. Keeping $V_1 = 2 \times V_2$ and the three ports equal to each other, in length and diameter, results in $F_2 = 1.126 \times F_1$, while $F_3 = 2.17 \times F_1 = 1.93 \times F_2$.

In the impedance response of a DCR system, there are three peaks instead of the typical two of the bass reflex. You can also see two F_B frequencies: in the lower one (F_2) the woofer sees just one chamber ($V_1 + V_2$), as if there isn't any partition between the two chambers; the other one (F_3) is present at a little bit less than one octave of F_2, with $F_3 = 1.93 \times F_2$. In the DCR impedance chart (**Fig.**

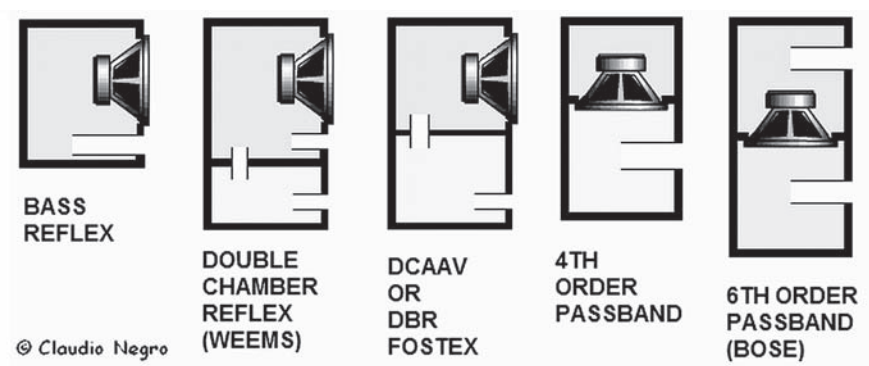

FIGURE 1: The bass reflex and some of its variations.

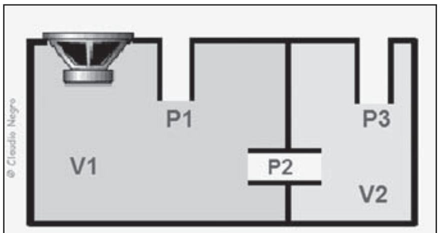

FIGURE 2: DCR chambers and ports.

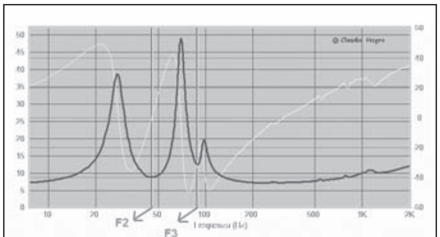

FIGURE 3: DCR impedance response.

3), I found that F_2 = 46.1Hz and F_3 = 89Hz, measured looking at the two dips where the impedance module assumes the relative minimum. The measured values of F_2 and F_3 are confirmed by applying said formulas.

To prove that the DCR F_2 is equal to the F_B of the corresponding reflex system, thus taking off the separation between the two chambers, I compared their impedances. Neither system used an absorbing mat, and the reflex total volume was 0.8 ltr bigger than the DCR one, because of the absence of the chamber's separation panel. Notice in **Fig. 4** how similar the dips between the peaks are, just to confirm that the F_2 of the DCR is equal to the reflex F_B; that is, at F_2 the DCR sees just one chamber ($V_1 + V_2$), simplifying its design.

The typical frequency response of a DCR speaker has a dip just above F_3. But, according to Augspurger and Weems, you needn't worry about this because of its modest amplitude and also because dips are less audible than peaks. I agree with this statement, because you need a well-trained ear to notice what is missing. In case the dip is very deep, Weems suggests you place some absorbing mat inside P_2; however, doing so will increase the port losses altering the known parameters in a not easily predictable way.

The chart in **Fig. 5** is the near field response of just the woofer, thus without having measured and added the ports responses, and it's valid from 20Hz. F_2 is recognized by the big notch, because in a reflex system the woofer less energetic emission happens at its F_B. In this chart you notice that F_2 is equal to 44.5Hz, while in the impedance chart it is equal to 46.1Hz. This difference is normal because usually acoustic data and electric (impedance) data do not agree exactly.

However, I suggest you use the woofer near field notch as the more reliable

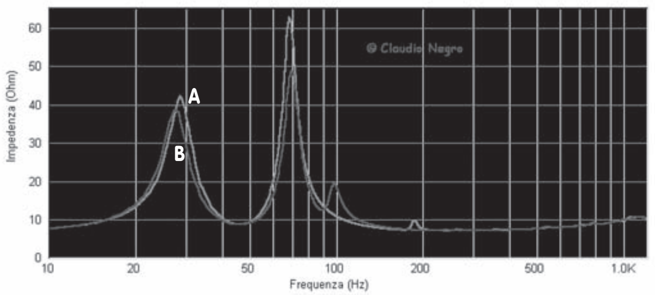

FIGURE 4: Impedance response comparison: REFLEX (A), DCR (B).

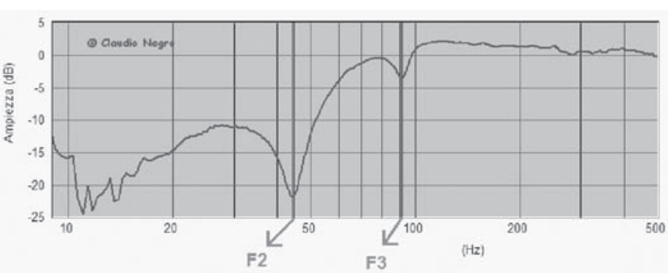

FIGURE 5: DCR near-field response without the ports.

measurement of the reflex tuning frequency. Notice the typical double chamber reflex dip near F_3. But what causes it? To understand why it happens, I studied the two external port acoustic emissions, P_1– P_3, looking at the near field phase chart of P_1– P_3 and of the woofer (**Fig. 6**). Below F_2 the ports are in phase with each other but out of phase with the woofer; between F_2– F_3 the ports are in phase with each other and with the woofer as well; over F_3 the ports are out of phase with each other, which is what causes the DCR typical dip.

The advantage of the DCR compared to a conventional reflex system—as reported by D. B. Keele—is a reduction of the cone excursion around F_3, with a consequent reduction of the distortion. In fact, in a reflex speaker the cone excursion is minimal at F_B (while in a closed box the cone excursion is maximal near the F_C); therefore, the diaphragm nonlinearity distortion, as well as the intermodulation one, is less than in a closed box at the system tuned frequency.

In a DCR the reduced cone excursion happens at two frequencies, F_2 and F_3, and in a wider range compared to a reflex system. Moreover, the use of two chambers in a tower speaker, where it is easier to produce standing waves, helps to reduce the phenomenon of the resonances. In fact, in **Fig. 4**, where I compared the impedance of the DCR with a reflex box obtained by eliminating the chamber's division panel, the reflex peak at 180Hz, due to the box height resonance, disappears.

DCR DESIGN

To start, select a reflex alignment for the driver in use, taking into account all the known procedures to obtain a good-sounding reflex system. To help you decide, you can use one of the numerous software programs, such as the freeware *Unibox*[2]. Then divide the total volume of the reflex (V_B) to obtain a partition that divides the box into two unequal chambers, with one twice as big as the other. You mount the woofer in the bigger chamber. For example, if your reflex V_B is 60 ltr, divide it by 3 to get the smaller chamber volume (V_2 = 20 ltr), which you multiply by 2 to get the bigger chamber volume ($V_1 = 20 \times 2 = 40$ ltr). As you have seen before in the impedance comparison chart, you can assume that $\mathbf{F_B = F_2}$; therefore, you can go on calculating F_1 and F_3 using the known formulas.

But what happens when the chamber volumes are not kept in the 2:1 proportion? I investigated this by leaving the V_2 volume the same while increasing V_1 by one liter first (A) and then decreasing V_1 by two liters (C), in comparison to the ideal 2:1 volumes ratio (B). The DCR total volume ($V_1 + V_2$) was 35 ltr. In the impedance chart in **Fig. 7** you can see the result: While the F_2 is almost the same for the three curves, the F_3 shows a notable difference in the smaller-than-ideal curve.

To continue the chamber ratio change study, I compared the near field response of the three systems in the impedance comparison chart. I measured without adding the port responses because I wanted to see the system tuning frequency ($F_B = F_2$) given by the typical response notch. The near field chart of **Fig. 8** confirms what you already saw in the impedance comparison: The three curves show a F_2 difference inside 0.8Hz; after 70Hz A (one ltr bigger V_1 chamber) and B (ideal 2:1 ratio) curves are similar in shape, while C (two ltr smaller V_1) starts falling down at 83Hz, drawing a deeper and wider dip, for sure more audible. My conclusion is that you should never keep the volume of V_1 less

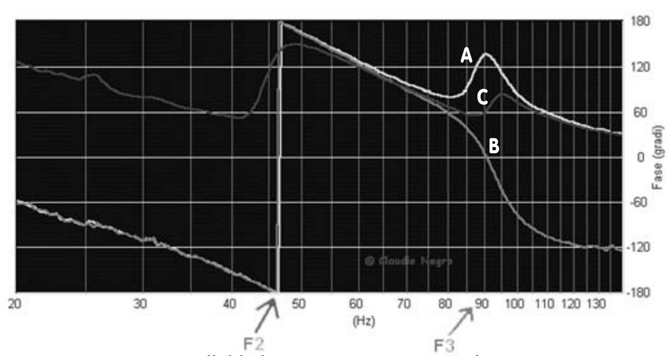

FIGURE 6: DCR near-field phase: P1 (A), P3 (B), woofer (C).

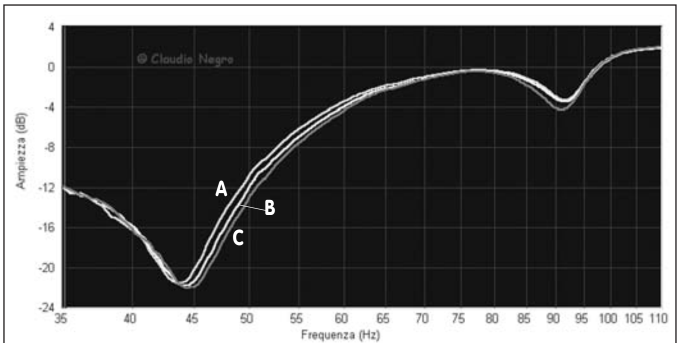

FIGURE 8: DCR near-field comparison with V1: bigger (A), ideal (B), smaller (C).

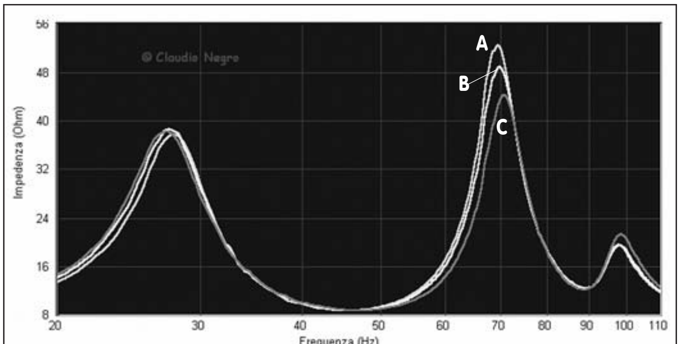

FIGURE 7: DCR impedance comparison with V1: bigger (A), ideal (B), smaller (C).

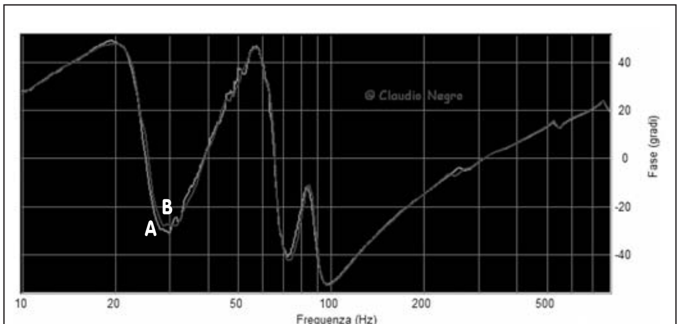

FIGURE 9: DCR impedance phase: P2 tube inside V1 (A), in the middle (B).

than $2 \times V_2$, as it can cause a more evident typical DCR dip; even using $V_1 > 2xV_2$ (inside a 5% increase) doesn't produce problems, just a lower F_2 in comparison to the ideal 2:1 response.

To calculate the port dimensions, knowing the chamber's volume and the desired F_2, just use the volume of the larger chamber (V_1) that tunes at F_1 using the known port calculation formula or a software program. Remember that $F_2 = 1.126 \times F_1$, therefore $F_1 = F_2 / 1.126$. Then use the resulting port dimension for all three ports, so that they are all identical.

Another issue I studied is the position of P_2. Should the tube be inside V_1 or V_2 or in the middle of the two? I measured the impedance phase of the DCR first with the P_2 tube inside V_1 (A) and after with it in the middle of the two chambers (B). As you can see in **Fig. 9**, F_2 is the same for both curves, meaning that the tube position is irrelevant. However, the P_2 tube inside V_1 phase (A) shows more spikes, certainly caused by its closer distance to the P_1 tube. Therefore, in this case, it's better to place P_2 in the middle of the two chambers.

AURI DRIVERS

I intended to use a ribbon tweeter because I like to experiment with uncommon drivers, knowing their sound is rich with details but with a limited vertical dispersion. For some people (mainly ribbon driver resellers), a limited vertical dispersion is a positive feature because it makes the driver less dependent on the listening room. On the other hand, some people maintain that using ribbons produces only one narrow listening point, making group listening difficult.

My thought is that room reflections and reverberations have a big role, explaining why the same speaker might sound good in a room but better in another. One of the advantages of building a speaker is that you can fine-tune it, through the crossover, for your listening room.

But back to my tweeter choice, I ended up with the ATD DDRT120, which is identical to the Silver Flute YAG20-1. Unfortunately for my wallet, the price paid for the ATD in Italy is very different from the Silver Flute sold in the US! It often happens that in my country cheap drivers become hi-end (in price) ones. Anyway, in a two-way system it is important to use a tweeter with a low resonance frequency because its crossover cutoff frequency should be at least one octave above its Fs: the ATD Fs is at 1100Hz, making it a good choice. The woofer, however, needs an extended off-axis frequency response.

Ideally you should use each driver until it works like a rigid piston that can be easily calculated through the formula $F = 345/2d$, where d is the driver effective diameter expressed in meters. However, you must consider the cone material, which can perform miracles in increasing the driver rigidity, and thus its rigid piston frequency range. In fact, the Focal 7K 4412 (my woofer choice), whose cone is made of polykevlar, showed a very good off-axis response allowing a high cutoff frequency; moreover, it has the right Qts for a vented enclosure and a 90dB sensibility. Its measured Thiele/Small parameters are: Fs 44.9Hz, Qts 0.42, Qms 6.99, Qes 0.44, Re 6.06Ω, Vas

37.6 ltr, and Sd 176.7cm^2.

The 7K 4412 frequency response shows a typical rigid cone peak at its break-up frequency, around 2800Hz. I also measured a dip around 1100Hz that wasn't present in the Focal datasheet. I decided to investigate it, noting that in the woofer free air impedance response there was a peak at that frequency, to confirm the reliability of my measurement. So I contacted Focal in France to ask how a dip can disappear like magic.

They confirmed that the dip at 1100Hz really exists but is not visible in the datasheet because their measurement methodology (on infinite baffle) tends to rise up to the cone break-up point, somehow compensating for the dip at that particular frequency. They believe the dip is due to a slight mechanical termination mismatch between the polykevlar cone and the surround.

CABINET

Auri is a floor-standing tower speaker whose internal dimensions (959 × 188 × 218mm, H × W × D) are not in line with the golden ratio. Here's what I've done to try to make a deaf cabinet. All panels are 2cm thick MDF with the exception of the front and rear ones, whose thickness is 3cm. An internal frame is present all around the perimeter, which helps to harden the structure and at the same time serve as an end guide for the front and rear panels. The top, bottom, and lateral panels use L joints (**Photos 1-2**). The front panel has a step (**Photo 3**), so that it's partially stuck inside the cabinet, while its edges are rounded to reduce *cabinet edge diffraction.*

There are actually two schools of thought about how audible cabinet diffraction is. The first affirms that the change from 4π str to 2π str causes an image degradation, and some suggest the use of a compensation circuit. The other states that the room response compensates for any low frequency pressure drop (baffle step or diffraction loss) noticeable in the anechoic response. Moreover, the diffraction effects are very directive while, usually, the listening is made off-axis. I tend to take the middle ground, trying to minimize the diffractions, since eliminating them completely is impossible, by using a narrow front panel that contributes toward a better sound image, and by rounding its edges with a radius of at least 1″, as reported by J. D'Appolito and J. Moriyasu[3].

I also flush-mounted all the drivers to reduce diffraction at the driver's edge, and enlarged the woofer internal hole because the panel thickness (3cm) could have produced an unwanted resonance (**Photos 4-5**). On all internal sides, excluding the front one, I used a product called *Fonogel* from AZ Audiocomp, which is a viscous-elastic damping compound that's non-toxic and doesn't smell like solvent-based products. I also used a damping mat on all panels except the front one, choosing from the long fiber polyester and the rusticated expanded polyurethane. Each type comes with a damping factor chart. So knowing the cabinet internal dimensions and applying the formula $F = 345/2d$, where d is expressed in meters, I was able to use the better mat type for that particular resonance frequency.

Another issue I addressed is the transmission of vibrations from the woofer to the cabinet and the tweeter. First of all, I must say that various modes of panel vibration, or resonance, exist. The first mode is referred to as forward-backward panel movement; the second mode occurs when a half panel moves forward while the other half moves backward, divided on the vertical axis; the third mode is like the second one but on the

PHOTO 1: Cabinet internal frame.

PHOTO 2: Panels L joints.

PHOTO 3: Front panel internal step.

PHOTO 4: Front panel drivers flush mount.

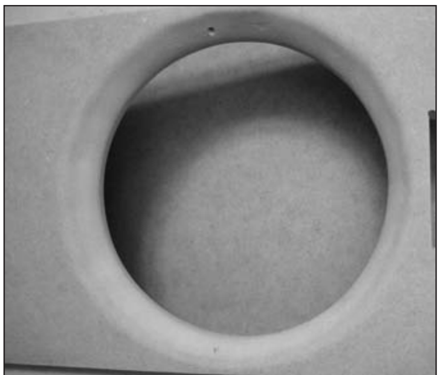

PHOTO 5: Enlarged woofer internal hole.

PHOTO 6: Rubber insulated rivet nuts.

horizontal axis; the fourth mode refers to a panel divided in three sections with the center section moving forward while the other two move backward; and so on.

To reduce the transmission of vibrations from the woofer to the baffle, I used rubber insulated rivet nuts named Well-Nuts or Flex-Loc (**Photo 6**). Actually it's not so easy to find speakers that use these kinds of nuts, even if Vance Dickason[4] has suggested their use for a long time and their cost is not an issue. If you are still not convinced of their utility, I suggest you read Jim Moriyasu's *audioXpress* article[5], in which he proves that using Well-Nuts brought about a reduction of second-third-fourth mode vibrations, while the first mode remained the same.

The chamber's separation panel is 2cm thick, with rails on the lateral panels and embedded in the front-back ones (**Photos 7-8**). The cabinet is brush painted (**Photo 9**), and the dimensions diagram (expressed in mm) is visible in **Fig. 10**.


PHOTO 7: The chamber's separation panel.


PHOTO 8: Front panel internal view.


PHOTO 9: The complete speaker.

TUNING

Auri is tuned at about 40Hz in a net volume of 34.5 ltr, considering both chambers, through three ports (two external and one internal) with 5cm of diameter and 14.5cm of length. The chamber's volume ratio is 2:1, and has been obtained taking into account the drivers and crossover volumes, both located in the bigger chamber V_1.

CROSSOVER

I spent over six months on the crossover design of Auri, alternating computer simulations and listening and measurement, trying to keep the driver phase similar in the driver's crossover region, as well as a dip-free off-axis total response. In **Fig. 11** the starting point is depicted by the unfiltered driver's response, where the Focal 1000Hz and 2800Hz peaks are clearly visible, and the tweeter's 1500Hz peak is followed by a dip. With the woofer I used two parallel RLC cells in series with the signal to take care of the peaks, and a II° order filter. Actually in the second cell, which works on the 1000Hz peak, I didn't use any resistor because the measurement appeared more flat without it. The R2 resistor, which is in series with the C3 cap, is used to vary the filter *Q*.

With the tweeter, after the resistor R3 used to align the driver's sensibility, there is a III° order filter. All drivers are connected in phase with the signal, and their crossover frequency is about 2700Hz. The crossover schematic is depicted in **Fig. 12**.

SPEAKER MEASUREMENTS

Figure 13 shows the system impedance response with the three peaks present in all double-tuned loudspeakers. There are no unwanted resonances, and the load offered to the amplifier is of the easy kind, with the module always more than 6Ω and with an almost positive argument at its minimum. The impedance is a complex number formed by a real part (the module) and an imaginary one (the argument). With some calculations, you can obtain useful information.

Considering only the real part of the impedance and applying the formula *Rs = Z*cos(q)*, you can determine the speaker maximum load value. For example, from the impedance chart, frequency (f)= 41Hz, impedance (Z) = 9.9Ω, Phase (q) = +9.5°. Converting the phase degree into

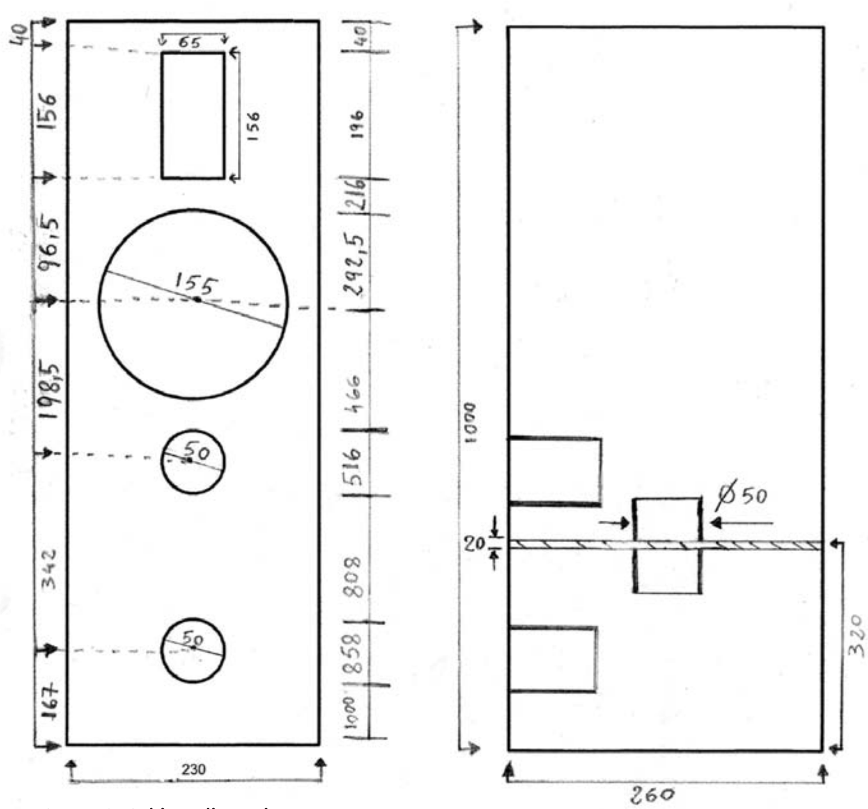

FIGURE 10: Cabinet dimensions.

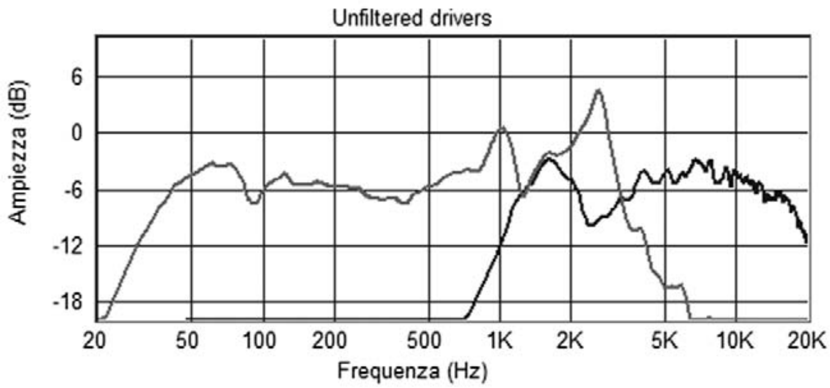

FIGURE 11: The unfiltered driver's response.

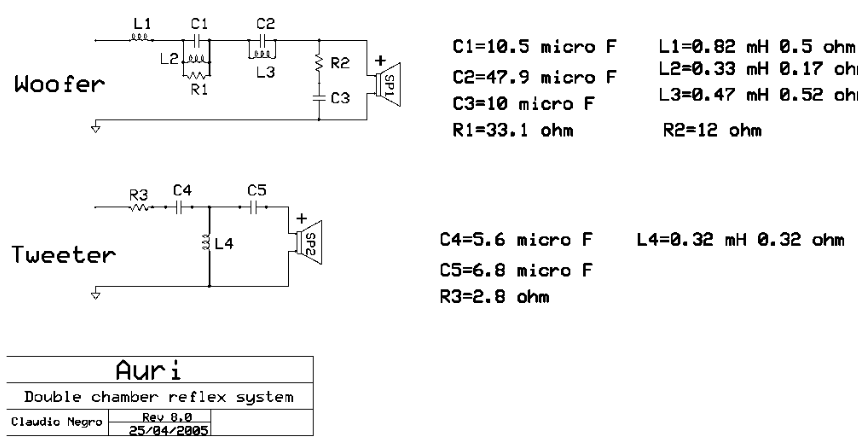

FIGURE 12: The crossover schematic.

radians and applying the formula gives Rs = 9.9*cos(0.16) = 9.7Ω. Thus at 41Hz it's as though the amplifier is connected to a 9.7Ω resistive load.

The Auri maximum load situation happens at 2819Hz with a resistive load of 6.2Ω. You can import your impedance into a spreadsheet, such as Excel and let it do the calculation for you. Moreover, considering both the real and imaginary parts of the impedance, you can calculate the energy requested by the speaker at each frequency by applying the formula $W = cos(q)/(|Z|*2\pi f)$, allowing you a better vision of what the amplifier will need to give. W = cos(0.16)/(9.9*257.4) = 0.000386 joule at 1V RMS.

I suggest you draw a graph with the calculated active energy values up to 150Hz, because the energy request diminishes as frequency increases, and look at the curve peak in **Fig. 14**. A narrow tall peak means a quick energy request, which is more difficult for the amplifier to handle than a wider curve. Special thanks to Valerio Maglietta, whose help has been instrumental both in the impedance understanding and in the active energy equation.

The one meter semi-reverberant anechoic response is in **Fig. 15**. The first dip is of the DCR itself, while the one at around 1200Hz belongs to the Focal 7K4412.

As a matter of fact, I give more importance to the in-room response, because it better depicts what arrives at our ear. I positioned both speakers in their final room position, along with the mike in the

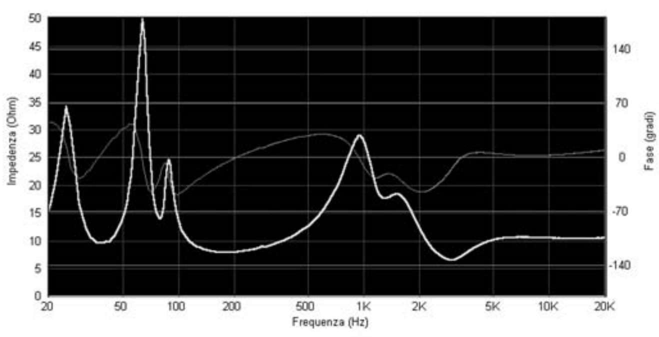

FIGURE 13: Loudspeaker impedance response.

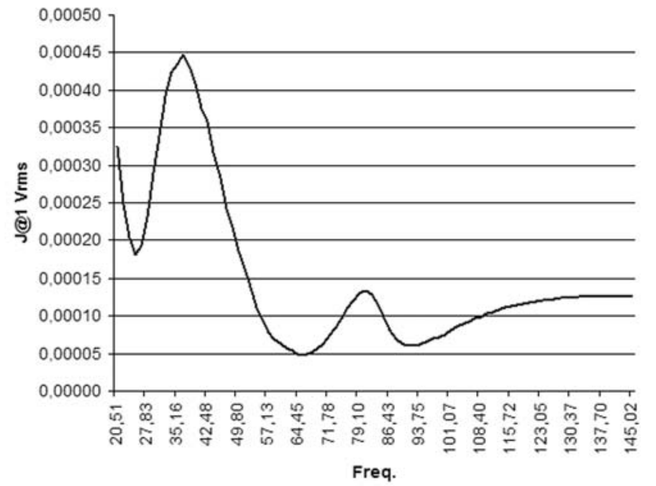

FIGURE 14: Loudspeaker active energy graph.

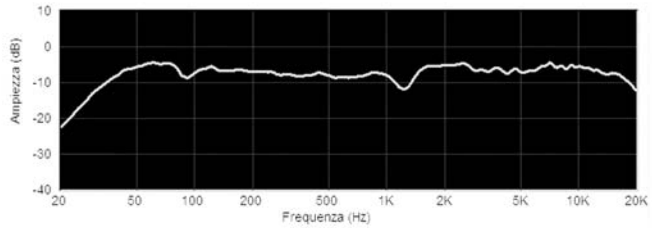

FIGURE 15: Loudspeaker semi-reverberant response.

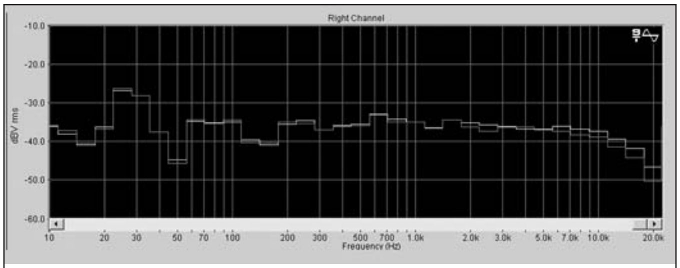

FIGURE 16: Loudspeaker in-room response, on-axis and 30° off-axis.

listening place. Both speakers received a ⅓ octave pink noise signal from two generators, thus each speaker was playing a different signal. In case you don't have two signal generators, you can produce a wave file with the program Sound Forge (which is capable of generating a not cyclic track, different for each channel), and then burn a CD with it to use your CD player as a double pink noise generator. If you are interested, you can download a ready-to-use wave file from my home page[6].

I also find it important to have a reference response for comparison, because it helps understand what is due to the speaker and what to the room acoustic: I used a pair of Dynaudio Contour 1.8 MKIIs for this purpose.

Another big reference help comes from the Henning Møller curve[7], which shows the optimum curve for hi-fi equipment measured in listening rooms. Even though the Møller study was presented in 1974, I think it has been forgotten by most! I have been lucky to know Renato Giussani, the Aedon Audio NPS-1000 designer, journalist for an Italian hi-fi magazine at that time, who personally assisted at the Møller workshop and keeps transmitting this precious information to people, like me, who were too young to remember these AES papers.

Finally, in **Fig. 16**, you can see the in-room ⅓ octave pink noise response, measured on-axis and 30° off-axis with this method.

CONCLUSION

I am satisfied with music coming out of Auri, especially the bass region where the double chamber reflex works very well. I liked the rich sound from the ribbon tweeter, but I will probably go back to using a classic dome for my next project, because I am more used to its sound. I found the best in-room position of the speakers by placing them at least 50cm from the back wall and oriented parallel to each other. Unfortunately, Focal is not selling home components anymore, so it's up to you to search the dealer's stock or to find a valid substitute for the 7K4412. *aX*

***Claudio Negro** is a retired commercial airplane pilot, and has been interested in speaker building since the age of 13 when he built a zero offset three-way speaker with a self made tweeter acoustic lens. He also became interested in electronics, building a mixer based on a Radford preamp and a Marantz MM pre-schematics, with separate PSU enclosure. In 1985 he was classified sixth in an Italian magazine DIY speaker contest, after which he took a long break from DIY audio. Recently, he has studied speaker measurements software, publishing a tutorial on Speaker Workshop. His other interests are listening and playing hard rock music, photography, computers, and online car racing.*

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