Speaker Workshop(SW), is a freeware program that allows to measure, project and test a loudspeaker using the computer sound card; however the program needs some knowledge thus some time has to be spent in reading about the use of it. The first version of SW comes out in 1997 and is born from the mind of Mark Zachmann, that has been one of the founders of the ZSoft Corporation, manufacturer of the program PC Paintbrush from which comes Windows Paint; Mark also worked as a reviewer of PC Magazine, for the Kodak and has invented the graphical format PCX. In 1993 he founded the Audua that produces professional software for audio and video systems: SW is a part of this project. The last version of SW, the 1.06, is dated 2002, and is unknown if a newer version will ever come out. Let's see which are the system requirements for this software: SW supports Microsoft Windows 95, 98, Me, XP, and 2000; at least a Pentium II is needed; the sound card must be full-duplex, better if one of the last generation. It is important to emphasize the importance of the sound card quality that I suggest to be noiseless and with sample rate of at least 48k Hz; these are my preferred ones:

Sound Blaster Live

Sound Blaster Audigy II

Terratec DMX6 Fire 24/96

Using a well known brand of card also helps in case of problems: it's easier to find other SW users that have the same audio card, so that they can help us to solve our problem. Check in SW Audio DB, where I created a database of sound cards that works with SW and their main settings, thanks to the information given me by the Speaker Workshop community.

SW, that is available in five languages, has a good help, and another great source of aids is the forum of Speaker Workshop.

Here's what Speaker Workshop is able to perform:

Measurement: impedance and acoustic response of the driver/ speaker, harmonic distortion, transfer function, calculation of Thiele/Small parameters.

Project: closed or vented boxes, crossovers till the fourth order, impedance compensation net.

Besides SW has an audio generator with ten different kind of waves; it allows to measure passive components like resistors, caps and inductors but it can't substitute a LCR meter since the range of measurement is narrow.

It's available for download the **Speaker Workshop Manual**, by Jay Butterman, at this address: http://www.audiodiycentral.com/ that covers many aspects on building and testing a speaker.

Well, let's start with **Speaker Workshop setup !**

This is the Speaker Workshop sound cards database, created through the informations given by SW users. It will be updated everytime I receive new data.

BRAND	MODEL	CHIPSET	S/R USED	SW CLIPPING LEVEL	SW VUMETER USED LEVEL	SW VOLUME %	WIN MIXER RECORD LEVEL %	WIN MIXER PLAY LEVEL %	INTER-CHANNEL DELAY us.	LATENCY SETTING ms.	IMPEDANCE MEASUREMENT NEEDED AN AMP?	KNOWN ISSUES	TIPS	TESTED OPERATING SYSTEM
CREATIVE	SOUND BLASTER LIVE	CT 4620	48 K	26K	20K	100	100	81	-10,8	1	NO		BETTER RESULTS WITH EXTERNAL AMPLI	98 SE
CREATIVE	SOUND BLASTER LIVE	CT 4830	48 K	26K	20K	100	100	81	n/a	n/a	NO		BETTER RESULTS WITH EXTERNAL AMPLI	98 SE
CREATIVE	PCI 128	CT 4810	48 K	27500	18K-24K	80	50	100	-14,3	0	YES	INSTABILITY IN LATENCY AND MIXER VOLUMES, SOME DC OFFSET IN LEFT LINE IN	CHECK FOR NEWER CARD DRIVERS	98 SE
CREATIVE	SB MP3+ USB	n/a	48K	29000	17K-25K	75	50	65	0	8	YES	LATENCY SETTING IS VITAL, USB SOUNDCARD OWNERS SHOULD FIND THE CORRECT VALUE BY TRIAL	Do calibration very well; Latency setting is vital, USB soundcard owners should find the correct value by trial	XP
CREATIVE	SB AUDIGY 2 NX (USB)	n/a	96k	32k	26k	100	60	80	0	20	NO	Soundcard drivers need to be updated to v1.00.0009 for operating under 24-bit 96 kHz setting. Windows XP Service Pack 1 should have been installed beforehand.	If you installed creative software as well as the drivers then 24-bit 96 kHz setting should be selected from 'device control' under control panel first.	XP
CREATIVE	SB AUDIGY 2 VALUE	n/a	96k	32k	25k	100	30	90	0	0	NO	Be careful setting mixer levels. I got a squealing sound when any levels were too high	Use only the Line input and the Green output	XP
M-AUDIO	REVOLUTION 5.1	n/a	96k	31700	30k	100	30	90	0	5	NO	Do a fine calibration expecially the channels difference; not very good response below 100Hz; every time needs adjusting play-record levels from scratch	Select 2ch stereo operation and mute all unneeded monitor inputs and outputs	XP
QDI KINETIZ 7B	ON BOARD SOUND CARD	n/a	48K	32K	18K	100	15	30	0	-8.12	NO	FREQUENCY RESPONCE: 20-20k Hz -3dB; latency instability	DO A CAREFULL CALIBRATION PROCEDURE, EXPECIALLY ON VOLUME LEVELS	XP
TERRATEC	DMX 6 FIRE 24/96	ENVY 24	96K	>32K	22K	100	n/a	n/a	0	0	NO		USE TERRATEC CONTROL PANEL	Me, XP
TURTLE BEACH	MONTEGO II	AU8830A2	48 K	32 K	16K-27K	90	n/a	n/a	n/a	1.16	YES	Interchannel delay requires calibration for each project. Preemphasis causes high frequency roll off	Works best with its own mixer software. All unused inputs and outputs must be turned off at mixer	98
TURTLE BEACH	SANTA CRUZ	n/a	48 K	29 K	24 K	100	30	95	0	-1.6	NO	MUTE THE LINE-IN PLAYBACK	MUTE ALL PLAYBACK EXCEPT THE WAVE	2000

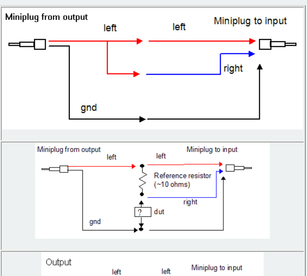
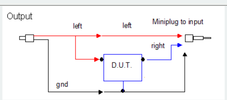
The setup of Speaker Workshop is an operation that demands time and patience, but is very important in order to obtain good results. It is begun with the construction of the hardware, that is the JIG that allows us to connect the input and output of the sound card (LINE IN and LINE OUT) and the device to test (DUT); practicaly there are two ways to build this JIG: one is the very used Eric Wallin's JIG II, that I briefly describe in this folder, the other is less beautiful to see and you will find it illustrated with many photos in the **Cables** folder. To you the choice!

With the JIG available, we begin to set the **Volumes** in order to understand until which level we can go before introducing distortion that would give us wrong results. This task is the more delicate part of the setup section, but there are many photos that guide you step by step in the setup; be patience in loading the page! Once we find the right volumes we will verify the goodness using them with the program **RMAA** (check the Download or Links folder) that will also give us useful information on which sampling frequency to use with SW.

After this we are ready for the **Calibration** of SW: it is necessary to have a good multimeter in order to measure exactly the values of the reference and calibration resistors; it is better to use low tolerance ones (1% 0.5 watt). You will need of 4 values: the reference one, 8-12 •, two for calibration that are one the half and the other the double of the reference value and finally one for the sound card calibration, whose value is of 10k •. I use: 5.6 / 11.1 / 22.1, obtained with series-parallel connections , but you can play as you like.

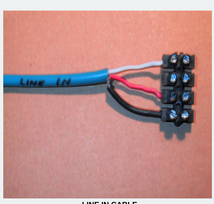
Finally we will **Test** some passive components, resistores, capacitors and inductances, in order to verify that everything is working! However we must remember that SW is not a LCR meter: the range of use is limited, therefore having a good multimeter is a must. Are you ready? Let's start with the construction of the cables!

In order to work SW needs a connection between the output and the input of the sound card, and the device to test (DUT). There are three types of connections depending on what we want to measure:

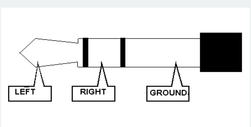
LOOP CABLE is used for channel calibration and for volumes setup	
IMPEDANCE CABLE is used to measure drivers impedance, to calibrate SW and to measure passive components	
ACUSTIC MEASURE CABLE is used to measure frequency response	

I suggest to use bi-polar shielded cable, 2 x 0.75 mm, well made connectors and keep short the cables that are to be connected to the sound card, *Line In and Line Out*, while there is no length problem with the DUT cable. Be careful when soldering the minijack: remember that SW use the left channel as REFERENCE and the right one as DATA.

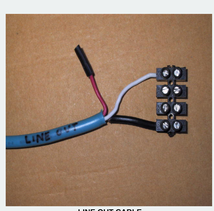
In the following pictures the red cable is right channel; the white is the left channel; the black is the ground while the light blue squared cable is the one connected to the DUT.



LINE IN CABLE



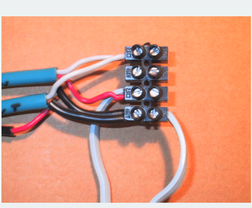
MINI JACK STEREO CONNECTIONS



LINE OUT CABLE

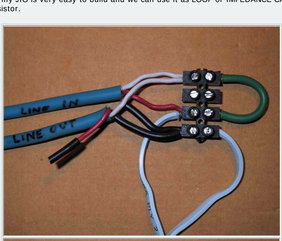
LINE IN + LINE OUT CABLES

DUT CABLE



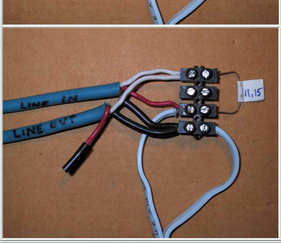
LINE IN + LINE OUT + DUT CABLES

As you can see my JIG is very easy to build and we can use it as LOOP or IMPEDANCE CABLE just changing a piece of cable with a resistor.



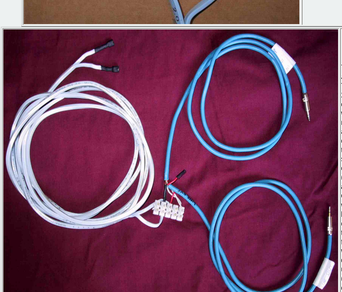
LOOP CABLE

Using the green cable we get a loop cable.



IMPEDANCE CABLE

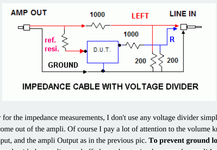
This cable is used for impedance measurement and is obtained connecting the reference resistors (11.15 ohm)



This is my JIG in Toto: the 2 blue cables with minijack are connected to the sound card (**LINE IN and LINE OUT**) while the light blue cable with two black faston is where the device under test (**DUT**) is connected

This JIG doesn't have any voltage divider resistor so be careful: if you have an amplified sound card you can fry it, during the volumes setup: the modern sound cards are just pre-amplified and, usually, supports up to 1.5-2.0 volts input, so there's no risk of damaging them; while old SC (Sound Blaster AWE) are amplified so check it up before connecting and be very careful during the volumes adjusting.

Pay attention at the quality of your pre-amplified sound card: being such it means that it is intended to work with high impedance loads and not with the typical low impedance of the drivers that we want to measure. This can induce the SC amplification transistor or IC to work badly, introducing distortions that would alter the measurements. In these cases you need to add a 1000 Ohm resistor in series with the LINE OUT or you can use an external amplifier with the knowledge of the risk of being able to burn the SC if a voltage divider resistor is not used as. An example of voltage divider using two resistors (1k and 200) per channel is the following:



IMPEDANCE CABLE WITH VOLTAGE DIVIDER

Personally, if I have to use the amplifier for the impedance measurements, I don't use any voltage divider simply because we have to work on small signals, thus 0,3-0.8 V are expected to come out of the ampli. Of course I pay a lot of attention to the volume knob! When using an amplifier, just connect the SC Line Out to the ampli Input, and the ampli Output as in the previous pic. **To prevent ground loop**, you can disconnect the sound card Line In ground, but only after you measured, with the ampli turned off, that a short exists between the ampli Input ground and its Output negative. If your SC has a front module, I suggest you not to connect the LINE IN and OUT through it but using the back connectors cause usually the front panel connections are noisier.

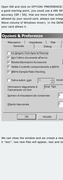
Let's go to the ***volumes setup.***

Opening Speaker Workshop and click:

I created the default file "SPKWS1.SWS" and give a new name to the project: "EST". To make it visible, close "SpkWS1" and open the "EST.SWS" file from the menu.

The main/electronic "Speaker Box" is created. Here we will do the most important during the program calibration:

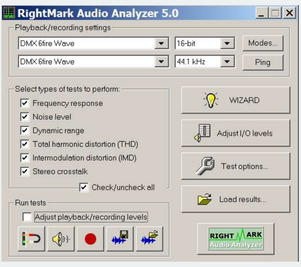
We can start the sound card to see if in a computer with SB / OPTIONAL WOWO CREEK SOUND CARD is not the soundcard checking wizard. Then we open the Windows Mixer and select the "Wave" and "Line" inputs and set the sliders to the middle. Then we open the Windows Mixer and select the audio properties in which right mouse click on the Volume control and select the first card starts checking. We select Windows Mixer and select the menu OPTIONS PROPERTIES, then click on all of the slide squares boxes to show all of the volume controls. Click OK. Now we have the Windows Mixer with all of the volume controls for those marked Volume Control and Wave. See the pack and click all of the boxes:



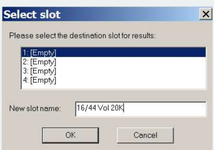
Open a second copy of Windows Mixer and select the menu item OPTIONS PROPERTIES then click on the radio button Recording. This should show all of the recording controls. Click OK. Now we have the Windows Mixer with all of the controls and Wave. These will be the only ones that we use.

Open the and click on OPTIONS PROPERTIES MEASUREMENT and move the cursor of Sample Rate and Sample Size all the way to the right. Its signal processing, you could say a little bit of a DSP. Click OK. Now we have the Windows Mixer with all of the controls and Wave. Set the Volume control, CD, Line-In, and a CD to accuracy. (DP / SG), that are more than sufficient in almost any situation. In case you want to use a sample rate lower than the maximum measurement, you can do so by selecting a lower sample rate in the Wave Out Mixer

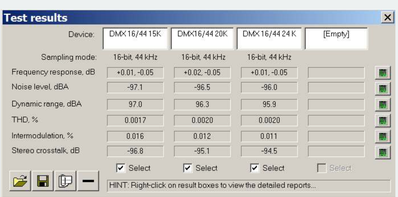
After closing SW we open *RightMark Audio Analyzer* and select our sound card in ***Playback/recording settings***, 16 bit e 44.1k Hz; de-select ***Adjust playback/recording levels***, since we are going to use our volume sets, the ones we saved with QuickMix:



With the SC connected through the **LOOP CABLE**, click on and after few seconds we will see a windows in which we can save the results: give a name that remembers the settings used (sample rate, resolution and volume setting):



RMAA allows the compare up to four different results so we can test others volume settings to see which one is better, keeping the same frequency and sample rate; so just recall the volume files saved in *QuickMix* and rerun the test saving the result with a different name:



Here we have three different volumes (15k, 20k e 24k) all at 16 bit and 44k Hz.

Frequency response has to be as close to 0 as possible: in this case we can notice how different volumes don't change the results.

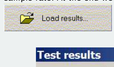
Noise level estimates a noise level in silence in the test chain. SNR is usually measured in dB FS A, which means sound level relative to the full-scale signal and is weighted with a special aural perception A-curve. Professional usage requires SNR more than –96 dB A, and in this case we see that the 15k set is the better one.

Dynamic range test estimates a noise level with a weak signal applied, and linearity of a sound device operating at low signal levels (which is very important for high-quality sound recording and playback): The higher the better. The 15k set wins again.

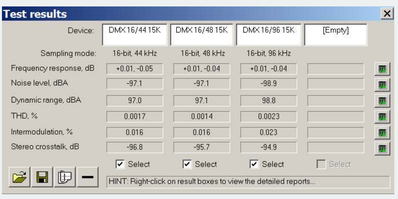
THD defines a level of unwanted harmonics generated in a sound device. Usually high-quality devices have a low THD value (lower than 0.002%), but there are exceptions. Many tube devices have a quite high THD level, which makes their sound "warm". But transistor devices must have a low THD value because their (odd) harmonics don't make sound pleasant. No big differences between the volume settings.

Intermodulation estimates amount of intermodulation distortions that occur when a complex signal passes through a test chain. The test signal consists of 2 harmonics of different frequencies. After passing through the test-chain, the resulting signal contains different harmonics and, possibly, a large number of harmonics resulting from D/A and A/D conversion oversampling filters and sample rate conversion inaccuracy. The lower the better, and our results are equals.

Stereo crosstalk estimates leakage of a signal from one channel to another for various frequencies. If a sound device has poor crosstalk results, you can't get a good stereo image from your sound recording. The higher the value the better, and the 15k set wins.

Once we know which is the best setting, save it clicking on . If our sound card supports higher than 44kHz sample rate, (SB Live! works at 48k Hz and Terratec DMX6Fire and Audigy at 96k Hz) it's a good practice to repeat the tests at those others S/R, always using the 3 different volumes, and save the best result for each sample rate. At the end we can compare the three best results closing the window *"Test Results"* and click on 

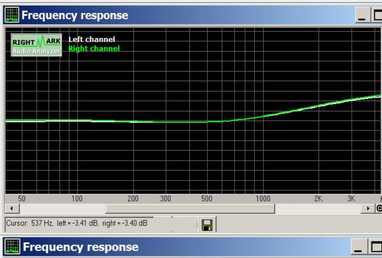
and select the saved files:



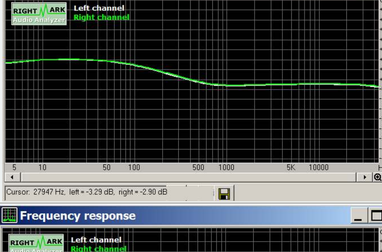
We see three different S/R (44k, 48k e 96k Hz) with the same volume (15k): the winner is the 96k Hz that will be used with Speaker Workshop.

Below I show some charts that show the frequency response we measured, but it's not a flat line as it should be: the tone control weren't in the flat position! Check it out, cause all the measurement will be invalided if this happens.

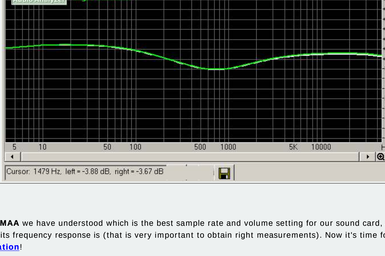
Flat response till 600 Hz, then it starts rising: the high control tone was at +4 dB



It's the opposite of the first chart: the bass control was at +6 dB

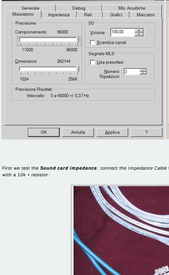


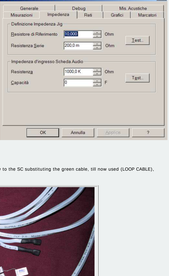
Here both the tone controls are in the + region, resulting in a wave curve



So using **RMAA** we have understood which is the best sample rate and volume setting for our sound card, and how linear its frequency response is (that is very important to obtain right measurements). Now it's time for the **calibration**!

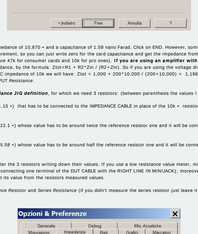
Using RMAA we understood which sample rate and volume we have to use in Speaker Workshop. First we call the chosen volume set through QuickMix, then we open SW and go to OPTIONS/ PREFERENCE/ MEASUREMENT and select the chosen SAMPLE RATE and slide to the maximum the SAMPLE SIZE; to improve the measurement increase to 3 the REPEAT COUNT. Now click on the IMPEDANCE tab:





First we test the ***Sound card impedance***: connect the *Impedance Cable* to the SC substituting the green cable, till now used (LOOP CABLE), with a 10k + resistor:

Before connecting the resistor measure it with the multimeter, and write down its value; then click on Test... and in the new window that will appear will be explained how to connect the resistor to the SC (as we already did); click on NEXT and write the value of the resistor as measured with the multimeter (in my case 9990 +).



Click on NEXT and after few seconds we will see :

So our sound card has an impedance of 10,870 + and a capacitance of 1,58 nano Farad. Click on END. However, some users reported wrong sound card impedance measurement, so you can just write zero for the card capacitance and get the impedance from the sound card technical specification (normal values are 47k for consumer cards and 10k for pro ones); **if you are using an amplifier with voltage divider**, you must add its value to the SC impedance, by the formula: Zist=R1 + R2*Zin / (R2+Zin). So if you are using the voltage divider I showed before (1k and 200 +) and assuming a SC impedance of 10k we will have: Zist = 1,000 + 200*10,000 / (200+10,000) = 1,196 ohm, that is the value to write in the SOUND CARD INPUT Resistance.

Now it's time for the ***Impedance JIG definition***, for which we need 3 resistors: (between parenthesis the values I used)

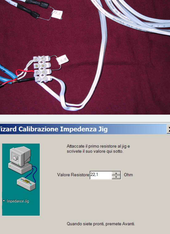
1) REFERENCE RESISTOR (11.15 +) that has to be connected to the IMPEDANCE CABLE in place of the 10k + resistor we used in the *Sound card impedance* test.

2) CALIBRATION RESISTOR (22.1 +) whose value has to be around twice the reference resistor one and it will be connected to the DUT CABLE connectors.

3) CALIBRATION RESISTOR (5.58 +) whose value has to be around half the reference resistor one and it will be connected to the DUT CABLE connectors.

We measure through the tester the 3 resistors writing down their values. If you use a low resistance value meter, measure the *Series Resistance* of the cable (measuring it by connecting one terminal of the DUT CABLE with the RIGHT LINE IN MINIJACK), moreover measure the multimeter cables resistance and subtract its value from the resistors measured values.

Back to SW, write the *Reference Resistor* and *Series Resistance* (if you didn't measure the series resistor just leave it to the default value of 0.2 +). Click on APPLY:

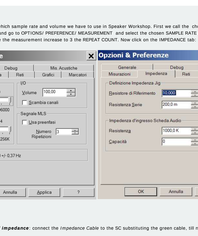


Then on Test... :

Click on NEXT and with the REFERENCE RESISTOR (11.15 +) connected to the IMPEDANCE CABLE and the first CALIBRATION RESISTOR (22.1 +) connected to the DUT CABLE, write in the SW window the value of the first Reference Resistor and then click on NEXT:

Wait for a while till the NEXT option is choosable again and substitute the first CALIBRATION RESISTOR (22.1) with the second one (5.58); again write the new value and then click on NEXT:

After a while a new window will appear where we can click on END :



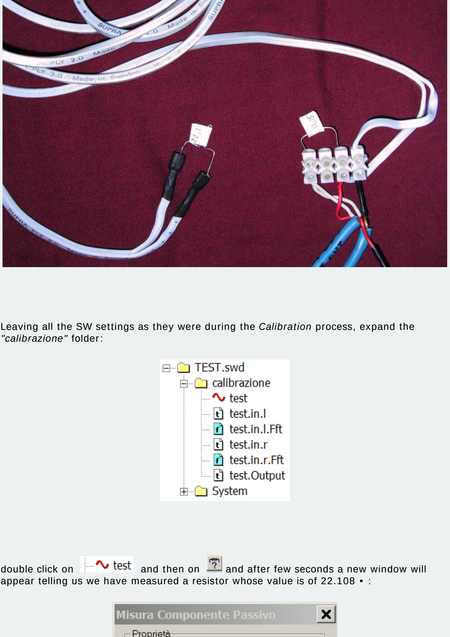
Now in the *OPTIONS & PREFERENCE* click on APPLY to make the new values effective:

If you get a negative resistance, that is impossible, or a too high *Series Resistance* it means that something went wrong; therefore check again the cables and the inserted values and repeat the procedure.

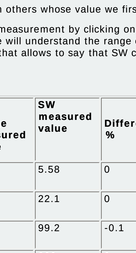
One last click on Ok and we are ready to test some passive components to be sure everything is set.

After the long calibration section, we now try to measure same passive components to verify that the calibration process was well done. We will continue to use the **IMPEDANCE CABLE** with the *reference resistor* (11.15 •), while to the **DUT CABLE** (the light blue one) we will connect the passive components that we are going to measure. From now on we aren't going to change this cable configuration, what will change will just be what we will connect to the **DUT CABLE**: now it will be resistors, capacitances and inductors, then it will be the driver and finally the speaker. If you use an amplifier the measurement range will be wider than not using it.

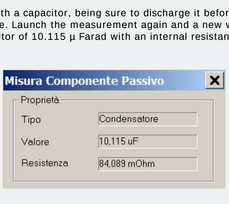
Let's start connecting the 22.1 • resistor to the **DUT CABLE**:



Leaving all the SW settings as they were during the *Calibration* process, expand the *"calibrazione"* folder:



double click on and then on and after few seconds a new window will appear telling us we have measured a resistor whose value is of 22.108 • :



Change the 22.1 • resistor with others whose value we first measured with the multimeter and repeat the SW measurement by clicking on ; in this way we can verify that SW is working right and we will understand the range of measurement we can trust. Here's a list of values I tested, that allows to say that SW can measure resistor till 1000 • (I haven't used the amplifier):

LCR bridge measured value	SW measured value	Difference %
5.58	5.58	0
22.1	22.1	0
99.3	99.2	-0.1
329	329	0
998	1010	1.2
2740	2850	4
9980	11646	16.6

Change the resistor with a capacitor, being sure to discharge it before connecting, in case it is an electrolytic type. Launch the measurement again and a new window will appear telling us that a capacitor of 10.115 µ Farad with an internal resistance of 84.089 milli • was measured:

From the following chart we can say that the range of use of SW in capacitors measuring is till 20 µ Farad:

LCR bridge measured value µ Farad	SW measured value µ Farad	Difference %
0.1	0.1	0
2.28	2.28	0
9.91	10.11	2
22	20.9	-5

Lately substitute capacitors with inductors and always clicking on , a new window will say that an inductor of 0.341 milli Henry that has an internal resistance of 412.319 milli • was measured:

Measuring different inductors I got:

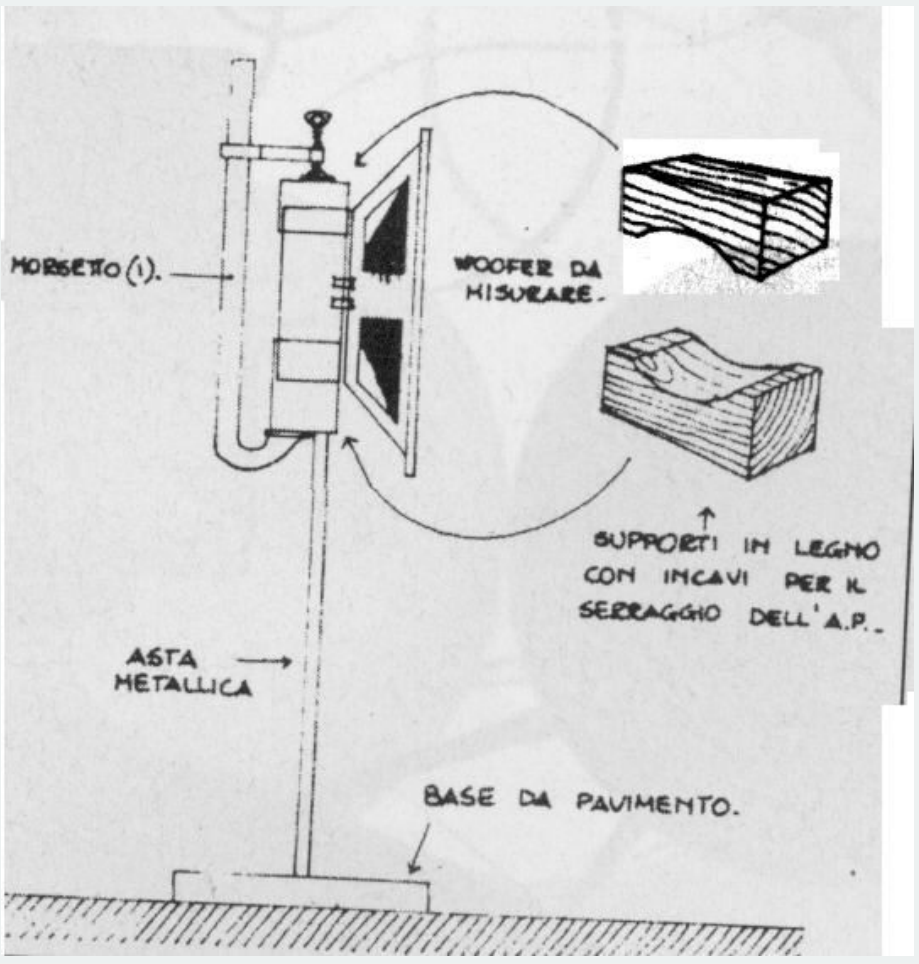
LCR bridge measured value mH	SW measured value mH	Difference %
0.189	0.199	5.2
0.330	0.341	3.3
1.43	1.47	2.7
13.1	12.11	7.6

It's important to test some passive components each time a new measurement session starts, to check everything is working; it would be even better to repeat all the calibration procedure, specially the *Channel Difference*, every time but only using Speaker Workshop will tell you if you need it or not.

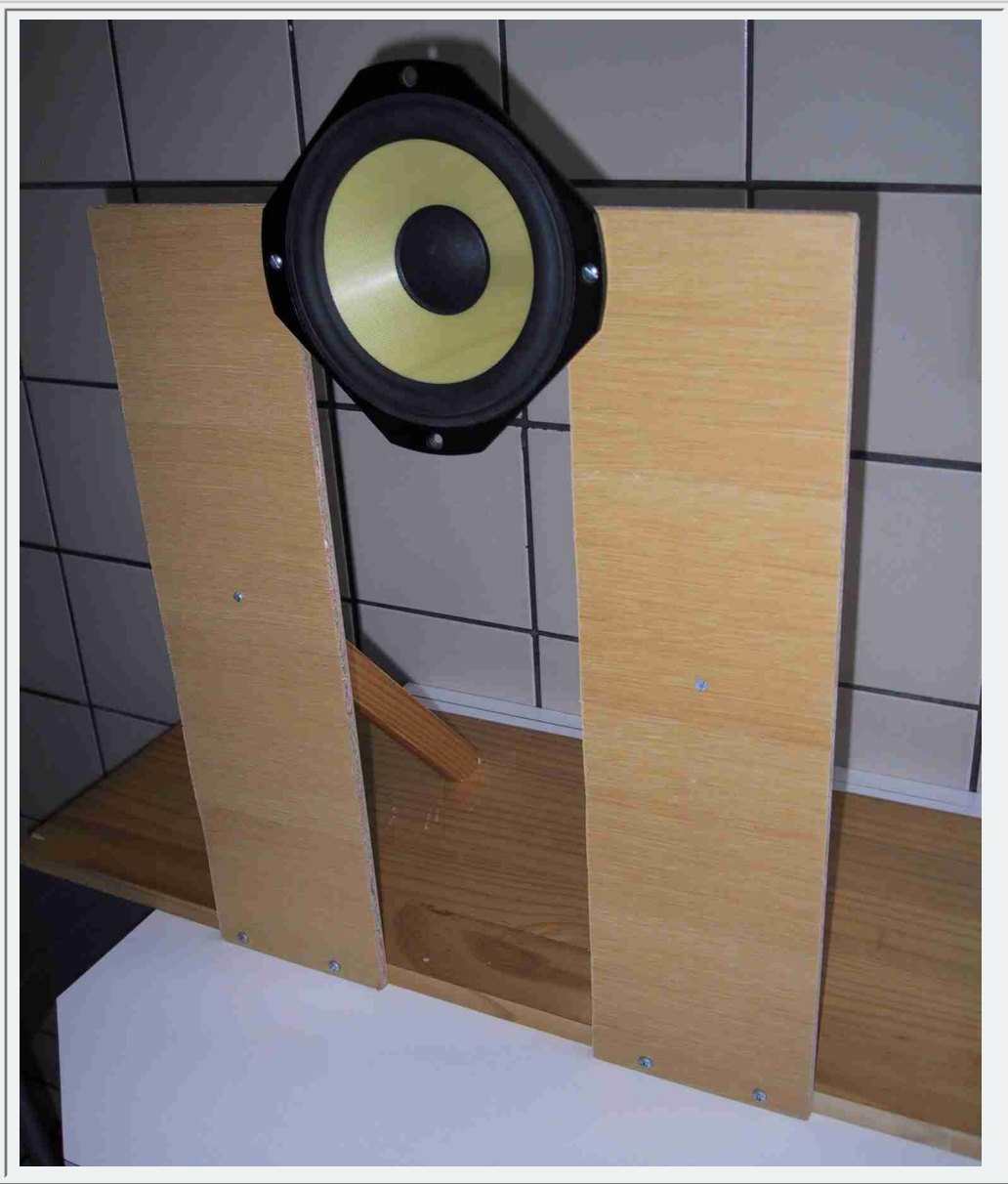
Finally we ended the setup section of Speaker Workshop: we prepared the cables, choose the best volume and sample rate for our sound card, calibrated SW and measured some passive components. Now we go on in measuring the drivers we are going to use in our project: it's time for *impedance* !

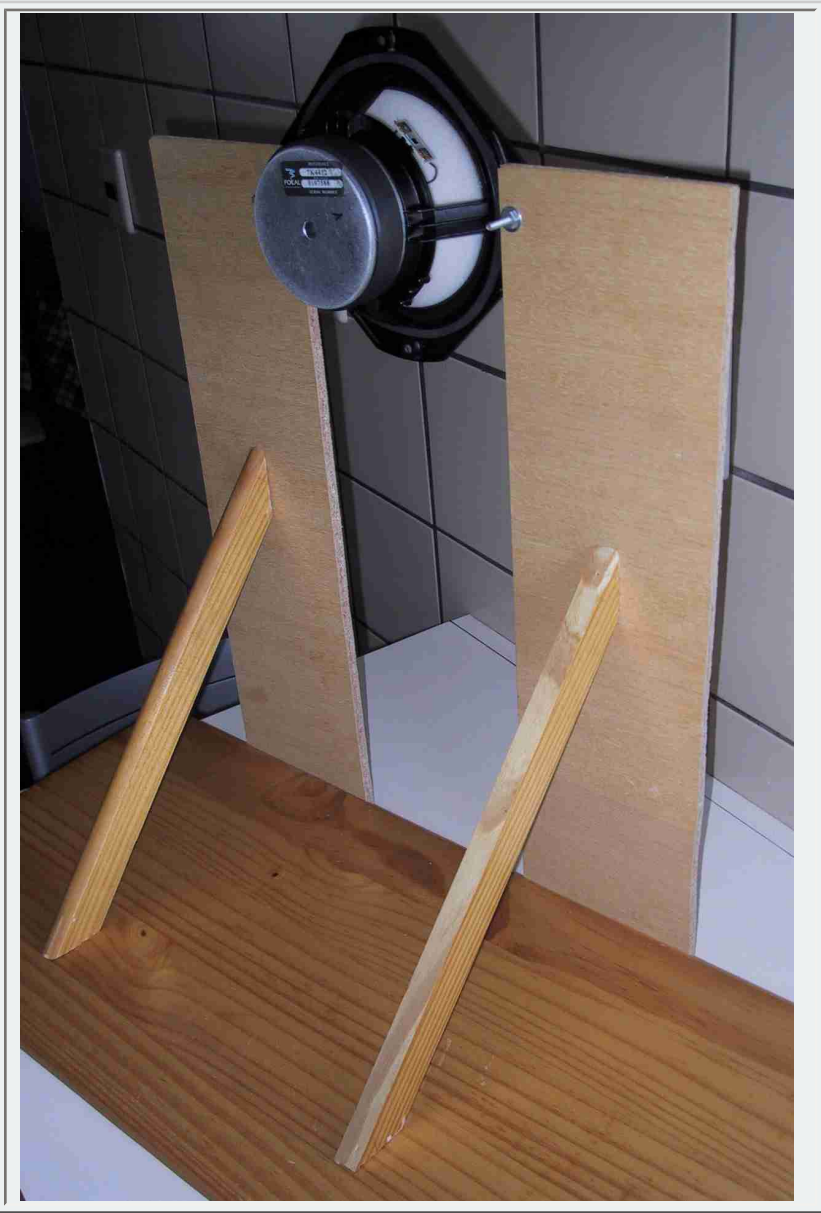
Here we are ready to measure Thiele/Small parameters of a Focal 7K4412 woofer, through the impedance response. The first thing to do is to break-in the driver, cause you can get up to 25% parameters differences between a new and broken-in driver. I use SW to generate a sweep signal from 10 to 100 Hz but you can test different ways to break-in. Of course you will need an amplifier if you use SW generator!

While the Focal is breaking-in, let's think on how to hold it during the measurement: we have to prevent unwanted vibrations! Here's two possible solutions:



It's important to test the driver vertically, as this will be it's final position once installed in the box; moreover keep it away from reflecting surfaces that could alter the measurement. This is what I used:

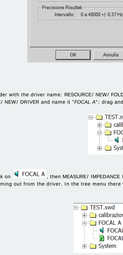




Once the break-in ended and the driver holder is ready we can start the impedance measurement. As a general rule it's advisable to let the driver play for 30 minutes then wait a couple of minutes (to let the coil cool down) and finally start the measurement. The **IMPEDANCE CABLE** with the *reference resistor* (11.15 •) will be used and the driver will be connected to the **DUT CABLE** (the light blue one) paying attention to respect the polarity. We are ready for the measurement .

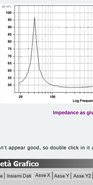
Recall the saved volume setting file and, if you are using an amplifier, turn its volume to get the desired voltage (between 0.2-0.7 Vrms AC) using SW sinewave generator. Then open Speaker Workshop and go to: OPTION/ PREFERENCE/ MEASUREMENT and check that the *Sample rate* is the chosen one and let the *Sample Size* to its maximum (256k in my example); increase the *Repeat Count* to 10 or more: in this way the results are more reliable even if the measurement time is longer. **Choosing the right sample rate is very important:** using 48k Hz SR and 131k of resolution will give a response from 0.37 to 24k Hz ± 0.37 Hz, more than sufficient in most cases: in some cases where a better precision is needed (for example when measuring a driver with low Fs and high Qms value), you can use 256k SS and even lower the SR but remember to increase the *Repeat Count* since SW is very sensitive to noise.

Click on .



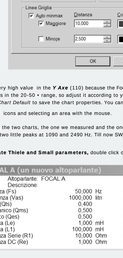
Crete a folder with the driver name: RESOURCE/ NEW/ FOLDER that I named *"FOCAL A"*; now create a driver: RESOURCE/ NEW/ DRIVER and name it *"FOCAL A"*; drag and drop the new icon into the *"FOCAL A"* folder:

Double click on then MEASURE/ IMPEDANCE IN FREE AIR or just click on and you will hear a whisper coming out from the driver. In the tree menu there will be a new file that is the free air impedance of the Focal A:



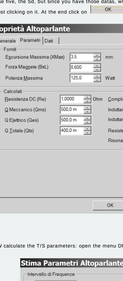
Double click on and its chart will appear (in my chart the blue line is the impedance while the yellow is the phase):

Free air impedance measured with Speaker Workshop



Impedance as given by Focal

Probably the chart doesn't appear good, so double click on it and make the changes as follows:



I used a very high value in the *Y Axe* (110) because the Focal woofer has a 101 + max impedance, while usually this value is in the 20-50 + range, so adjust it according to your driver maximum impedance. Remember to click on the *Make Chart Default* to save the chart properties. You can zoom in and out the chart or just a part of it using the icons and selecting an area with the mouse.

Comparing the two charts, the one we measured and the one published by Focal, we can see how similar they are: notice the two little peaks at 1090 and 2490 Hz. Till now SW is doing a great job!

To calculate Thiele and Small parameters, double click on and a new window open up:

Click inside of it with the right mouse button and select PROPERTIES and then DATA :

Write the drive dc resistance, measured with the multimeter, and check the box. In the *Impedance in free air* box there will appear the name of the file created during the measurement (*FOCAL A Aria Libera*). Select the tab *Parameters* and in the *Given area* write the parameters as given by the driver manufacturer: actually SW uses only one of these five, the Sd, but since you have those data, why not putting in. The measure units, if blued, can be changed just clicking on it. At the end click on .

Now let SW calculate the T/S parameters: open the menu DRIVER/ ESTIMATE PARAMETERS. :

If the measured curve is clean, as the one I measured, we can leave the first option; otherwise use an appropriate frequency range, making sure you make the upper cutoff high enough to show plenty of impedance or the estimates of the inductances could be pretty far off (which will not affect things much when it comes to designing a box). Click on and after few sec a message will appear: don't worry all the parameters will be calculated with the exception of the Vas:

And here is T/S driver parameters window:

Notice that a new file appear in the folder, that is just the impedance calculated on the estimated parameters. Open it and compare it with the chart: if the two charts are similar, it means that the estimate is right.

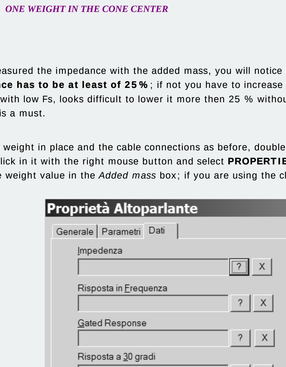
The next step is to compare the given parameters with the one SW generated; with this Focal there are 11 % differences in the resonance frequency, and 27 % in the Qts! Now you can see how important is to measure the driver we are going to use, cause manufacturer data often are not reliable. However keep in mind that the *Fs* and the *Q*'s are signal level dependant, that is at higher levels both *Fs* and the Q's lower; therefore to compare 2 different measurement system, at least the same signal level has to be used. Just remember that Thiele and Small parameters are intended for small signals, for power signals check Klippel papers.

Well, looks like we are done...not completely. We have to measure the second Focal! Even if some drivers, like the Focal, are sold in selected couple, you will see there are differences too. So create a new folder and driver, Focal B, and repeat the procedure.

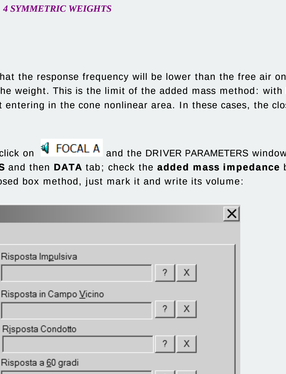
When finished, follow me in the Vas.

We have measured all the T/S parameters, by the impedance curve, with the exception of the *Vas;* to get this value we need to measure the driver impedance using a lossless closed box, whose volume should be equal to or less of the driver Vas, or with the added mass method, in which some nonmagnetic weights will be placed on the driver cone to lower its Fs. Measuring the Vas is very difficult, since depends on temperature and air density, thus it's variable in time. However is not necessary to have an high precision in this value: Joseph D'Appolito states that a 25% variation of the Vas results in less then ±1dB variation in the pass band response of a B_4 or QB_3 vented box alignment; in a closed box alignment, α>3 e Q_{tc}<1.1, that variation of the Vas brings an even less evident pass band response variation. **The closed box method**, suffers errors due to leakage of the box, driver-box, and even of the driver surround and dust cap; this method measures directly the Vas. **The added mass method** suffers errors due to suspension nonlinearities, thus this measurement is done at the lowest possible voltage level; an erroneous calculation of the driver S_d affects the result too, since it is present in the Vas formula squared; it also important not to use too heavy weights, never higher then the M_{ms} , not to enter in the suspension nonlinearities as well as not to change the Cms. Just attaching a mass to the cone causes a shift in the driver compliance, since the cone position is shifted: it's safe to apply a signal, equal to the driver resonance, to the driver with the added mass for a minute or so, to reset the suspension shift, and then perform the measurement. The added mass method calculates the Vas through the M_{ms} (cone mass) and the C_{ms} (compliance). D'Appolito reports a 4% variation of the same driver Vas measured with the two methods, to confirm the high tolerances of this value.

Here **I will show you the second method**, the added mass one. As for weight we can use the **Blue Tack** or some coins (use some bi-adhesive to stick them to the cone) but is very important to weight them with a precision balance like the one you find in jewelry. But of how much weight do we need? As reference use the Cone Mass, the *Mms;* if using a weight bigger then the Mms, don't put it in just one place of the cone, otherwise a misalignment of the cone will occur. If using blue tack, just model it to make a rubber ring, while if using coins put them symmetrically. The driver has to be measured horizontally, cause of the weights.



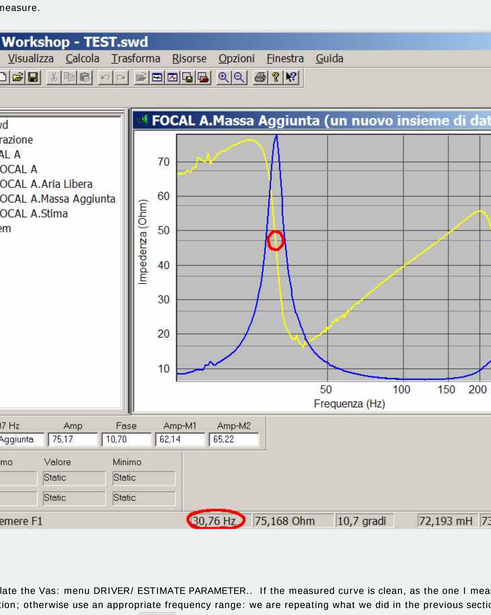
ONE WEIGHT IN THE CONE CENTER



4 SYMMETRIC WEIGHTS

Once measured the impedance with the added mass, you will notice that the response frequency will be lower than the free air one: **this difference has to be at least of 25%**; if not you have to increase the weight. This is the limit of the added mass method: with big woofers with low Fs, looks difficult to lower it more then 25 % without entering in the cone nonlinear area. In these cases, the closed box method is a must.

With the weight in place and the cable connections as before, double click on FOCAL A and the DRIVER PARAMETERS window opens; click in it with the right mouse button and select **PROPERTIES** and then **DATA** tab; check the **added mass impedance** box and write the weight value in the *Added mass* box; if you are using the closed box method, just mark it and write its volume:



Close the window clicking on OK. Then open menu MEASURE/ IMPEDANCE WITH ADDED MASS; again the driver will whisper and then a new file will be added in the folder: FOCAL A.Massa Aggiunta. Open it and check that the Fs is at least 25% lower then the free air Fs. The Fs value is easily seen if you look at the point where the phase (yellow line) pass for the zero, that is also the impedance (blue line) peak. Just pass the mouse cursor in that area and you will be able to read the Fs value at the bottom of SW window (be sure to have selected the STATUS BAR in the VIEW MENU). If the Fs difference is less then 25%, you have to increase the weight and repeat the added mass measure.

Now let SW calculate the Vas: menu DRIVER/ ESTIMATE PARAMETER.. If the measured curve is clean, as the one I measured, we can leave the first option; otherwise use an appropriate frequency range: we are repeating what we did in the previous section, but this time will not appear the Vas error window. Click on OK, and the Focal data window will show the Vas too, now:

We have almost finished, just repeat the procedure for the Focal B and then we can verify.

With the measured T/S parameters we can verify them using an Excel spreadsheet that you can download. Moreover other missing parameters will be calculated. Another option is to use a program written by *Bruno Dalle Carbonare*, that calculates T/S parameters starting from the free air impedance and the added mass or the closed box one.

Let's start with the Excel file:

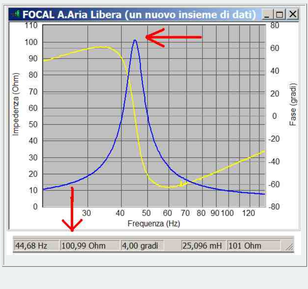
Parametri in aria libera misurati		
Focal 7 K 4412		
Fs	44,94	Hz
Re	6,06	Ohm
Zmax	100,99	Ohm
Res	94,93	Ohm
Sd	176,7	cm2
BxL	8,60	
Vas	37,65	litri
Qms	6,99	
Qes	0,44	
Qts	0,42	

Parametri con massa aggiunta misurati		
Focal 7 K 4412		
Fsm	31,13	Hz
Re	6,06	Ohm
Zmax	76,78	Ohm
Res	70,72	Ohm
Peso massa aggiunta	15,97	g

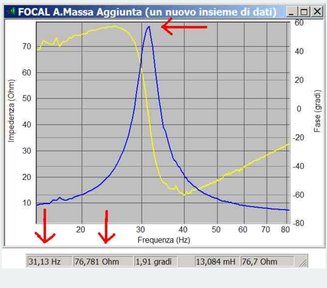
Parametri medi tra i 2 woofer			MEDIA
Fs	43,93	44,94	44,44
Vas	37,76	37,65	37,71
Qts	0,41	0,42	0,42
Qms	6,99	6,99	6,99
Qes	0,43	0,44	0,44
Re	6,12	6,06	6,09

Parametri calcolati				Var. %
Focal 7 K 4412				
Ro	16,67		Ohm	
R1	4,08		Ohm	
ZF1,ZF2	24,74		Ohm	
F1	33,50		Hz	
F2	59,75		Hz	
Fs	44,74		Hz	-0,45
Qms	6,96			-0,46
Qes	0,44			0,94
Qts	0,42	0,42		-0,59
Mmr	1,3506		g	
Mms	14,73		g	
Cms	0,000859	0,000850	m/N	
Cmes	0,000261	0,000261	Farad	-0,02
Bxl	7,53	7,58		-0,69
Mms + Mmr	16,08			
Vas	37,66	37,28	LITRI	1,02
Efficienza Watt	0,00729	0,00738		-1,25
Efficienza dB	90,83	90,88		-0,06

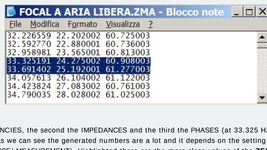
The numbers in red are the one we have to input, the blue one are the calculated. Just write the parameters SW calculated while for the Zmax, that is the impedance max value, we can get it from the FOCAL A.Aria Libera chart, zooming in the peak and passing the mouse cursor on it we can read its value in the STATUS BAR, as we did during the *Vas* measurement:



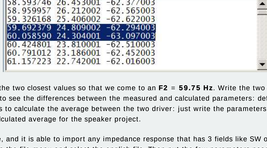
Repeat it for the FOCAL A.Massa Aggiunta chart, where we will read both the *Zmax* and the *Fsm* (Fs with added mass):



In the calculated parameter section of the spreadsheet we only need to input 2 values: **F1** e **F2**; to do this we read the value of **ZF1,ZF2** (that is the impedance at F1 e F2), and we must find in the FOCAL A.Aria Libera chart these two frequencies, one lower and the other higher then the Fs, at which the impedance is equal to the value of ZF1,ZF2 (24, 74 Ohm). You can use the *mouse cursor/status bar* procedure, or we can convert the impedance chart in numbers: with the chart opened click on **RESOURCE/ EXPORT**, and give a name to the file saving it; now open it with *wordpad on notepad* :

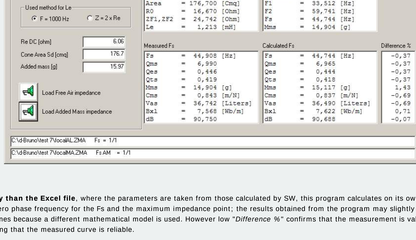


The first column are the FREQUENCIES, the second the IMPEDANCES and the third the PHASES (at 33.325 Hz the impedance is 24.27 Ohm with a phase of 60.90 degree); as we can see the generated numbers are a lot and it depends on the setting of the Sample Size in the SW setup (menu OPTION/ PREFERENCE/ MEASUREMENT). Highlighted there are the more close values of the **ZF1**; of course it will be impossible to find the exact value (24,74 •) so we interpolate the two closest values; so the first value, **F1**, is equal to **33,50 Hz**. Repeat the search for **ZF2** that will be found at a frequency higher than Fs (44.94):



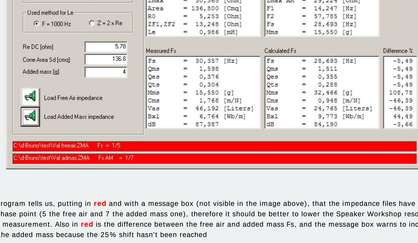
Here too we have to interpolate the two closest values so that we come to an **F2** = **59,75 Hz**. Write the two values, F1 and F2, in the Excel spreadsheet and we will be able to see the differences between the measured and calculated parameters: definitely a very good result! A last area in the spreadsheet allows us to calculate the average between the two driver: just write the parameters (Focal A and Focal B) as calculated by SW and use the calculated average for the speaker project.

Bruno's program is easy to use, and it is able to import any impedance response that has 3 fields like SW one. Once launched, if the language is not english, just open the file menu and select the englsh file. Then put the few parameters needed, and you will see:



Differently than the Excel file, where the parameters are taken from those calculated by SW, this program calculates on its own starting from the zero phase frequency for the Fs and the maximum impedance point; the results obtained from the program may slightly differ from the SW's ones because a different mathematical model is used. However how "*Difference %*" confirms that the measurement is valid, presupposing that the measured curve is reliable.

In case of mistakes, the program shows some warnings:

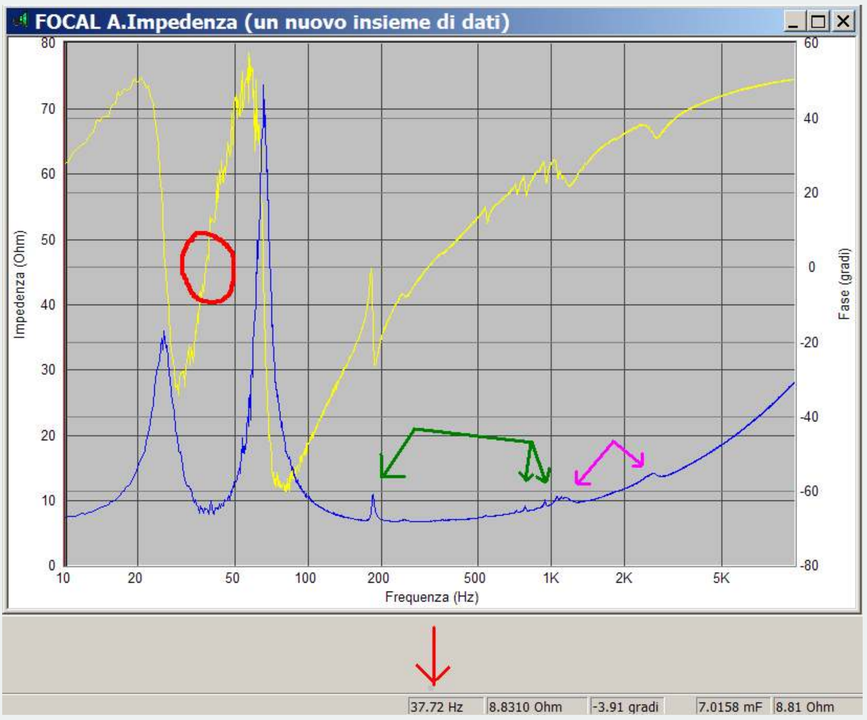


Here the program tells us, putting in **red** and with a message box (not visible in the image above), that the impedance files have more than one zero phase point (5 the free air and 7 the added mass one), therefore it should be better to lower the Speaker Workshop resolution and to repeat the measurement. Also in **red** is the difference between the free air and added mass Fs, and the message box warns to increase the weight of the added mass because the 25% shift hasn't been reached

With the program it's possible to use the closed box method, for the calculation of the *Vas*. You can take it from the download area.

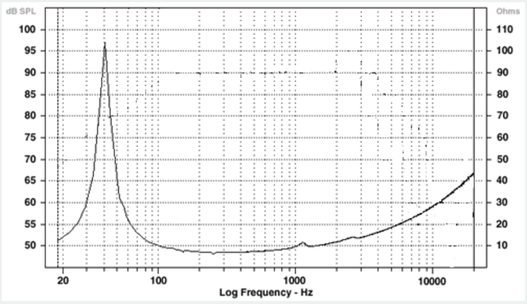
So now you have all the datas to start your project, so just go on! We will see again when the box is ready for the *speaker impedance measures* using Speaker Workshop.

Once we have the speaker ready, we can measure the mounted woofer impedance that will give us useful information; we will be able to see the speaker tuning frequency, the interaction between the driver and the box. We will use the same connections used previously for the measurement of the impedance in free air with the only difference that DUT Cable will be connected to the in box mounted woofer; the volumes setup are those ones till now used, same for the sampling frequency; practically we use the same settings used during the measurement in free air, just check that the REPEAT COUNT is at 10 or more. Double click on FOCAL A and then on Z or MEASURE/ IMPEDANCE and wait till the measurement ends; then open the new created file *"FOCAL A.Impedance"* and the chart will be visible; the system used here is a tall reflex speaker (959x188x218 mm) with two tuning ports and without any acoustic absorbent:



The impedance curve of a driver mounted in **closed box** has one "peak" as it happens for the driver in free air, but this peak is no more placed at the resonance frequency (Fs) but is found at the *resonance frequency of the closed box* **(Fc)** that is always greater than the Fs and inversely proportional to the box volume. In a **reflex system** the impedance curve introduces *two peaks*, one *(Fl)* placed at a frequency lower than Fs, the other *(Fh)* at a frequency higher than Fs; between the two peaks there is a *dip* whose minimum corresponds to the **tuning frequency of the reflex box (Fb)**. A more precise way to know the system tuning frequency, is to look at the impedance phase: watching the chart, that refers to a reflex system, we can acknowledge which is the Fb placing the mouse cursor where the phase (yellow line) assumes zero value between the two peaks (red circle) and reading the value in the State Bar (red arrow); the phase assumes equal to zero value in three points that are the two impedance peaks *(Fl, Fh)* and the minimum of the dip *(Fm)*. Another way to calculate the Fb (J. E. Benson) is to carefully close the port (s) using a piece of wood with sealing material and measure its impedance; from the resulted chart read the value of the Fc and apply the formula: $\mathbf{Fb = (Fl^2 + Fh^2 - Fc^2)^{1/2}}$. In the Download area there is an Excel file "Data Check" that calculates for you the Fb and other parameters, like the box Q.

Knowing the Fb we now know if the chosen tuning has been respected or not; varying the port length and repeating the measurement we can catch up the wished tuning frequency; just keep in mind that a longer tube reduces the Fb, keeping box volume and port diameter the same. Looking at the chart, we can see (blue line) various peaks: we must try to avoid them, since colorations in sound will occur; let's try to understand what caused those peaks. Those marked with the green arrows are a result of the box standing waves; in fact if we apply the *F=345/2d* formula, where for d is one of the dimensions of the box expressed in meters, we will have **F1** =345/1.91=180.6 Hz; we repeat the calculation for the others two dimensions and have **F2** = 917,5, **F3** = 791.2. This is an example of how the impedance measurement can help us identify the interaction between the driver and the box, and it helps us to verify if the taken countermeasures (absorbent, wall reinforce etc.) have succeeded to diminish or to cancel the problem (because the resonances created by the box are a problem!). The two other peaks (fuchsia arrows), that correspond to the dimensions of 6.4 and 15.5 cm. approximately, are of the Focal itself as you can notice from the chart below, therefore we have to keep them.



Impedance as given by Focal

I think the impedance graph can give us a lot of useful information that help to understand the interaction driver-box and also let us know if the resonances cure worked out or not; besides if there is a peak in the impedance we will find it, as well, in the acoustic measure !

Assuming that we don't have an anechoic chamber, in order to perform the acoustic measures we will use the semi reverberant technique; clearly we will try to use a big room with the speaker in the center of it, in order to have all the reflecting walls as far as possible. The measurement happens in three stages: the **first** is the *near field*, that takes care of the low frequencies and is made possible placing the mic very close to driver so that diffractions and room effects are negligible. The **second** phase is the *far field*, where the microphone is placed to approximately one meter from the loudspeaker and, using the gating technique, only the frequency range free of reflections is used. Finally the **third** phase joins the two responses, near field and far field, to get the total driver response. The hardware needed for the acoustic measurement is different from the one used in the impedance measurement: we will need a microphone with his pre-amplifier, an amplifier and some meters of cable. Let's go directly to the cables !

The following scheme is the one I use for acoustic measures:

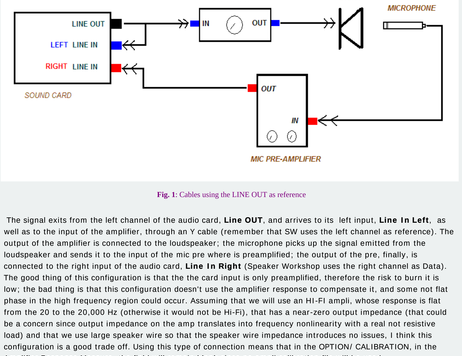


Fig. 1: Cables using the LINE OUT as reference

The signal exits from the left channel of the audio card, **Line OUT**, and arrives to its left input, **Line In Left**, as well as to the input of the amplifier, through an Y cable (remember that SW uses the left channel as reference). The output of the amplifier is connected to the loudspeaker; the microphone picks up the signal emitted from the loudspeaker and sends it to the input of the mic pre where is preamplified; the output of the pre, finally, is connected to the right input of the audio card, **Line In Right** (Speaker Workshop uses the right channel as Data). The good thing of this configuration is that the the card input is only preamplified, therefore the risk to burn it is low; the bad thing is that this configuration doesn't use the amplifier response to compensate it, and some not flat phase in the high frequency region could occur. Assuming that we will use an HI-FI ampli, whose response is flat from the 20 to the 20.000 Hz (otherwise it would not be Hi-Fi), that has a near-zero output impedance (that could be a concern since output impedance on the amp translates into frequency nonlinearity with a real not resistive load) and that we use large speaker wire so that the speaker wire impedance introduces no issues, I think this configuration is a good trade off. Using this type of connection means that in the OPTION/ CALIBRATION, in the Amplifier Response Measure, the field will remain blank since no ampli calibration file will be used.

Another procedure of connection exists, in which the *Line In Left* is connected to the output of the amplifier instead of the *Line Out*, but a voltage divider has to be used: you can use a 1k and 100 • resistors, to get a 1:10 voltage reduction:

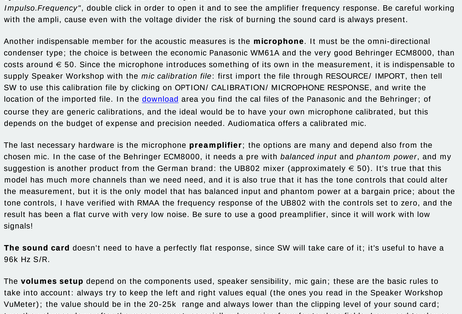


Fig. 2: Cables using the ampli as reference and a voltage divider

The advantage of this configuration is that it allows to use the ampli frequency response during the measurement. In order to measure it connect *Line Out* to the ampli input and its output to the sound card *Line In* left and right; then adjust the ampli volume in order to obtain, using an MLS signal, approximately 20k in the VuMeter of SW; then click on OPTION/ CALIBRATION/ TEST in the Amplifier Response Measure and follow the wizard; in the end the amplifier response file will appear automatically. To see the response open the *"Measurement.Impulse"* file in the *System* folder, then click on CALCULATE/ FREQUENCY RESPONSE and new file will be created: *"Measurement. Impulso.Frequency"*, double click in order to open it and to see the amplifier frequency response. Be careful working with the ampli, cause even with the voltage divider the risk of burning the sound card is always present.

Another indispensable member for the acoustic measures is the **microphone**. It must be the omni-directional condenser type; the choice is between the economic Panasonic WM61A and the very good Behringer ECM8000, than costs around € 50. Since the microphone introduces something of its own in the measurement, it is indispensable to supply Speaker Workshop with the *mic calibration file*: first import the file through RESOURCE/ IMPORT, then tell SW to use this calibration file by clicking on OPTION/ CALIBRATION/ MICROPHONE RESPONSE, and write the location of the imported file. In the download area you find the cal files of the Panasonic and the Behringer; of course they are generic calibrations, and the ideal would be to have your own microphone calibrated, but this depends on the budget of expense and precision needed. Audiomatica offers a calibrated mic.

The last necessary hardware is the microphone **preamplifier**; the options are many and depend also from the chosen mic. In the case of the Behringer ECM8000, it needs a pre with *balanced input* and *phantom power*, and my suggestion is another product from the German brand: the UB802 mixer (approximately € 50). It's true that this model has much more channels than we need need, and it is also true that it has the tone controls that could alter the measurement, but it is the only model that has balanced input and phantom power at a bargain price; about the tone controls, I have verified with RMAA the frequency response of the UB802 with the controls set to zero, and the result has been a flat curve with very low noise. Be sure to use a good preamplifier, since it will work with low signals!

The sound card doesn't need to have a perfectly flat response, since SW will take care of it; it's useful to have a 96k Hz S/R.

The **volumes setup** depend on the components used, speaker sensibility, mic gain; these are the basic rules to take into account: always try to keep the left and right values equal (the ones you read in the Speaker Workshop VuMeter); the value should be in the 20-25k range and always lower than the clipping level of your sound card; turn the volumes down after the measurement, especially when going from far to close fields. I am used to play an SineWave signal (you do remember how to do it, right?) and regulate the software volume to read 0.7 Vrms on the multimeter connected to the ampli input; then with the multimeter connected to the ampli output (and a 8 ohm resistor connected to the ampli) I play with the amplifier volume till I get 2.83 Vrms. The volt value is subjective cause depends on mic gain, driver efficiency; however just be sure not to overpass it too much for two reasons: the first is that the one volt you are sending to the ampli input, reaches the sound card left LINE IN too, so if it is too high it can make the card clipping or even burn it **(cables like in fig. 1)**; the second reason is that the one volt at the amplifier output goes first to the woofer (and it is not a problem) and secondly goes to the tweeter still unprotected by any crossover network! Be careful.

However remember that Speaker Workshop doesn't measure absolute SPL, thus the frequency response amplitude scale is not the real SPL: if you have a SPL meter, you can measure it and scale the chart (TRANSFORM/ SCALE) to get the right amplitude value. If you have a calibrated mic and you know your pre-mic gain, you can calculate the SPL as *Valerio Russo* showed:

SPL = 94 + GRAF – GAIN + 20 * Log (Vref / SENS)

where GRAF is the dB value that you read in the SW response chart; GAIN is the pre-mic gain in dB; Vref is the Vrms you have in the reference channel (left Line In) in millivolts; SENS is the mic sensibility measured in mV/PA; Log is the base 10 logarithmic calculation.

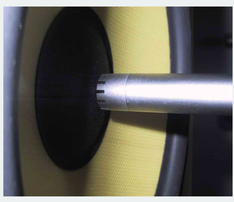
Here some pictures of my components:

This is the Y cable used to connect the LINE OUT to both the LEFT LINE IN and the Ampli Input

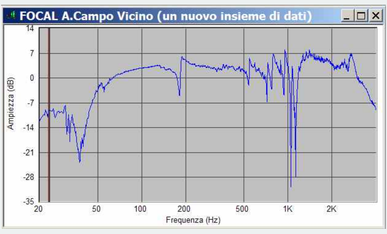
Well, now we are ready for the near field measurement.

The near-field measure is used to overcome the effects of low frequency standing waves caused by diffractions and room reflections; at this low frequencies the driver diaphragm acts like a rigid piston and the near-field measure is directly proportional to the far field one and is not affected by the environment in which the driver is. This technique is not full range, it's upper frequency limit is determined by the effective cone diameter : $F_{max} = 10950/D$ where D is in centimeter or $F_{max} = 4311/D$ with D in inches. To know **D** just look at the driver specification sheet, or calculate it from the Sd using the formula $D=2*(Sd/3.14)^{½}$ or measure the diameter with a ruler not considering the foam suspension. So in the **Focal** case, D=15 so $F_{max} = 730$ Hz that means that we can use the near-field response till this frequency. However this is valid for a driver mounted on infinite baffle, while lowers for a driver mounted on a panel, cause of the baffle effects, as reported by *C. J. Struck & S. F. Temme*. Just use the baffle dimension instead of the driver diameter, in the seen formulas.

The microphone has to be placed very close to the driver center, but not attached to it; the **maximum distance** between the two is given by the multiplication **0.055*D** as written by D. B. Keele, so the Focal center cone will be as close to the mic as possible but not more than 0.8 cm. Take a look at these pictures:



With everything connected as described in the cables section, and the volumes set to get a 20-25k (depending on your card clipping level) on both channels of Speaker Workshop VuMeter, Sample Rate and Sample Size to the maximum that works best with your card as you tested in the SWSetup section, choose a REPEAT COUNT of 5 (in OPTIONS/ PREFERENCE/ MEASUREMENTS) or more, depending on your test-room noise and used Sample Size (I use 20 repeat), and check that the USE PREEMPHASIS box is unmarked. Then click twice on FOCAL A and then on or click on MEASURE/ FREQUENCY RESPONSE/ NEAR FIELD; check SW VuMeter to be sure the values are almost equal for the left and right channel and in the 20-25k range; a new file will be created: ***"Focal A.Nearfield"***, open it and take a look at the chart.



Do you remember the peak at 180 Hz in the *Speaker impedance* graph? Here you can see that what we see in the impedance plot will be repeated in the frequency response; the DIP at around 38Hz is the vented system tuning frequency since, at this frequency, the tuned box-port combination put its maximum load on the woofer diaphragm. Take a look at the measured speaker impedance, and see what is the Fb (looking at the zero phase point between the two peaks): is should be equal to the DIP frequency! There's another anomaly just after 50 Hz, but I decided not to investigate since this speaker is just for testing; I will see if it is still there when I will start the *Double Chamber Reflex* measurement. Since reflex enclosures use port, we have to measure and add it; to better clarify the procedure, here's a prospect:

1) Measure the driver

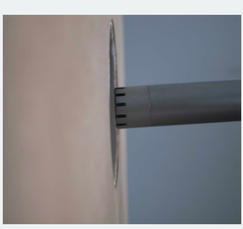
2) measure the first port with the same volume settings of item 1

3) (*optional*) measure the second port with the same volume settings of item 1

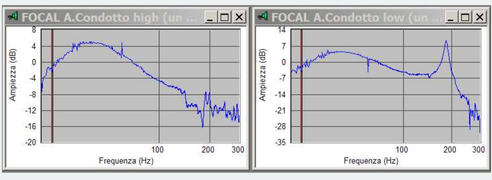
4) (*optional*) sum the two ports measures into one measure

5) combine item **1** with item **4,** in the case of two ports, or item **1** with item **2**, in the case of one port

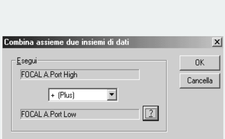
So after taking the near-field response of the driver, and not touching any of the volume settings used, put the microphone in the center of the port and flush with it (not inside the port):

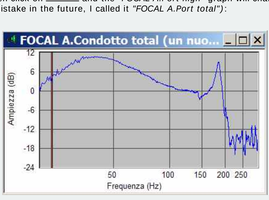


Then click on or click on or MEASURE/ FREQUENCY RESPONSE/ PORT RESPONSE; a new file will appear, *"FOCAL A.PORT"*. In case there is a **second port** rename the new file (*"FOCAL A.Port high"*) and place the mic on the second port; repeat the measure and rename the new file for *"FOCAL A.Port low"*. If only one port is used, jump down to **item 4** part:

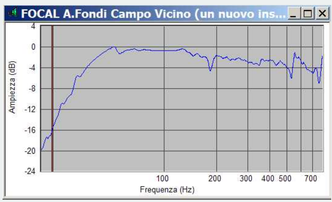


Now we have to sum the two ports response: highlight the first port chart, then click on CALCULATE/ COMBINE and choose ADD and select the second port file in the lower window then click on OK and the *"FOCAL A.Port high"* graph will change (I am used to rename the total ports file so I don't make mistake in the future, I called it *"FOCAL A.Port total"*):





Here we are at **item 4** where we have to merge driver and port measurements: double click on FOCAL A then DRIVER/ MERGE PORT RESPONSE; in the new window that appears, select the appropriate near-field driver and port responses; press the to select from the resource tree. In this case the two near-field files are the driver one (*"Focal A.Nearfield"*) and the port(s) one (*"FOCAL A.Port total"*); then in the PISTON AREA you will see the value of **Sd** as is in the driver properties (just check they are the same); finally you have to write the area of the port in the enclosure ($R^2*3.14$ *where R is the port radius*); remember to double the port area if are using two ports. Click on OK and a new file will be created, *"FOCAL A. Merge NearField"*:



That's it guys, our near-field response is ready but usable for a limited range, 730 Hz; in the chart the phase module is not visible but I will show you, in another moment, how useful it can be in deciding the network frequency. For now we need to measure the far field response.

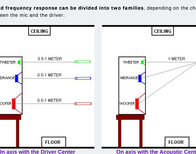
There are **two kinds of far-field measurement**: gated and not-gated one. In the **gated measurement** we set the markers so that we limit the data to be used to calculate the frequency response. The **not-gated far-field measurement**, instead, uses all the data including the room response, and is useful to see how the speakers will sound in the environment they will be used, helping us understand where to place them.

In the **far field measurement** the microphone is placed at a distance at which **the driver looks like a point source** and its sound pressure level is inversely proportional with its distance from the mic (6 dB drop for each doubling of distance); the industry standard sets this distance to 1 meter to allow integration of the various drivers in a front baffle, that is by 1 meter usually the woofer and tweeter response have added together so a single test gets both (however when testing speakers that uses line sources, like ribbon or array drivers, the mic-speaker distance is increased to 2 meters). If you are testing drivers separately then you can use any distance that works but be aware that if it's true that a less than 1 meter distance increases the signal to noise ratio, it's also true that a too close distance causes the driver not to act as a point source, thus the measurement cannot be considered far-field. Is common to use a distance equal to 4-5 times the driver diaphragm diameter, but not exceeding more than 1 meter: it rarely works for anechoic (gated) measurements cause the reflections are very close to the signal and you don't get any low frequency response (unless you have a huge measurement room, where floor-ceiling-sidewall reflections happens very late). I suggest you to measure the DUT at various distances, for example at 60–80-90-100 cm, and compare the responses: when the curve shapes are similar means that you are in the far field, thus that is the right microphone-baffle distance. Using 1 meter or more for far-field (not gated) in-room response is just fine.

In the **gated measurement** it is very important to place the DUT in the room center, so that all the parallel walls are equally far from it; I use to put the speaker on a table to center it between the floor and the ceiling. Also put some absorbing mat on the floor between the mic and the speaker, as well as on the microphone stand. Here's some pictures:

Let's start with the gated far field measurement !

The gated frequency response can be divided into two families, depending on the chosen axis between the mic and the driver:



On Axis with the Driver Center **On Axis with the Acoustic Center**

The On-Axis with the Driver Center (OA-DC) is made by placing the mic in line with the driver under test (when at a distance, measured from the front baffle, that varies from 0.5 to 1 meter depending on the driver. When measuring the driver off the driver cone's must be disconnected, thus only the OUT has to be fed with the signal. Each driver is measured using this technique, keeping the chosen mic-baffle distance the same for all drivers; as a result we will obtain the frequency responses of all the drivers in the best dispersion angle (on axis). Remember that the estimated phases of the drivers are not comparable with each other because the microphone location changes at every measurement, as well as the gating markers; therefore don't use the OA-DC phase to understand the driver offsets.

The next step is to measure each driver **Off-Axis at 30° - 60°** ; this is performed by placing the mic-driver center to one of those angles; I usually just leave the microphone always in the same position while orienting the speaker to reach the desired angle. These measures should be performed for each driver, giving as the frequency responses at their worst dispersion angles (30°-60°), the Off-Axis response is fundamental in choosing the right crossover frequency for the drivers used, so don't be lazy and do it.

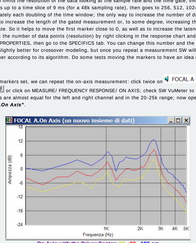
In the On-Axis with the Acoustic Center (OA-AC) the microphone is placed at ear height; for a seated listener; since the tweeter is usually found at this height, is a common rule assign the acoustic center to the tweeter center. This is the basis for the classic rule that works pretty well in any 2-3 way speaker; however remember that lots of possibilities exist, thus each project needs its own study. The OA-AC response is important as well, has to be sweep-linear phase (the acoustic response depend on it; a 1 meter distance works pretty well with a 2-3 way system but, again, every speaker needs its own study. The OA-AC is SUREMENT with all the drivers working together, this gives a real view of the speaker response; but since this measurement is made by leaving the microphone always in the same position, as well as the gating markers are unique, the phase response of each driver is comparable. It's also useful to perform the OA-AC with and without the crossover to see what happens adding different order/cut crossovers.

But let's start to **measure the On-Axis with the Driver Center using Speaker Workshop**! We are going to use the same cables-hardware seen in the near-field measurement, just you will need an higher mic pre-amplifier gain; remember that to measure low frequencies from the baffle, both everything connected and the volumes set to get a 20-25 (depending on your card clipping level) on both channels of Speaker Workshop Voltmeter. Sample Rate and Sample Size has to the maximum that works best with your card as you tested in the SWSetup section, choose a REPEAT COUNT of 1 (in OPTIONS/ PREFERENCES/ Measurements) with the Impulses rule that works pretty well in any 2-3 way speaker; however remember that lots of possibilities exist, thus each project needs its own study. Set the Output Level as well, has to be sweep-linear phase (the acoustic response depend on it; a 1 meter distance works pretty well with a 2-3 way system but, again, every speaker needs its own study. The OA-AC is SUREMENT with all the drivers working together, this gives a real view of the speaker response; but since this measurement is made by leaving the microphone always in the same position, as well as the gating markers are unique, the phase response of each driver is comparable. It's also useful to perform the OA-AC with and without the crossover to see what happens adding different order/cut crossovers.

Then click twice on F FOCAL A and then on F or click on MEASURE/ FREQUENCY RESPONSE/ ON AXIS; check SW VuMeter to be sure the values are almost equal for the left and right channel and in the 20-25 range; a new file will be created: *"Focal A On Axis"*, *"Focal A Left"* and *"Focal A Right"*.

As we can see the first impulse starts at around 8.5 ms and goes diminishing in amplitude till around 12.5 ms where second impulse makes its appearance: this is the first pulse that reaches the mic reflected by the wall, whose distance is equal to the time length of the second pulse divided per two (speed of sound meters) that is 12.5 (second pulse) – 8.5 (first pulse) = 4 / 3 =1.33 meters that, in my case, is the side wall distance from the mic. So we have to set the markers (the 2 vertical brown lines) to delimit the first pulse; the first marker should be set just before the starting of the first pulse at 8.5 ms. but I suggest you to leave it at 0 ms, as I will explain later on; the second marker should be positioned just before the second pulse starts, at 12.5 ms in this case. To move the markers just click on them and hold the left mouse bottom and move the mouse to the desired location; use the zoom option to better see the chart. In this case we will have a 4 ms of useful data, that is not a lot but enough for our purpose: a bigger room with a bigger floor-mic distance would have both and right channel identical. See the **download area** you can find an Excel file that allows to calculate when the floor bounce occurs knowing the microphone distances, as well as the expected impulse length and the points (resolution) by right clicking. To increase the time length, you could place the speaker laying on the floor with the drivers facing the ceiling, so that the first roof bounce will look silent; however place another microphone, but now you repeat a measure and the floor sine will reset this number according to its algorithm. Do some tests moving the markers to have an idea of how it works.

With the markers set, we can repeat the on-axis measurement: click twice on F FOCAL A and then on or click on MEASURE/ FREQUENCY RESPONSE/ ON AXIS; check SW VuMeter to be sure the values are almost equal for the left and right channel and in the 20-25k range; now open *"Focal A On Axis"*.

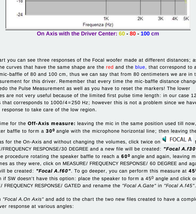


On Axis with the Driver Center: 60 - 80 - 100 cm

In the chart you can see three responses of the Focal woofer made at different distances; as you can see the curves that have the same shape are the red and the blue, that correspond to a distance between the mic of 80 and 100 cm, thus we can say that from 80 centimeters we are in the far-field measurement for this driver. Remember that every time the mic-baffle distance changes, you need to redo the Mark Measurement as well as you have to reset the markers. Where the lower frequencies are not very useful because of the limited first pulse time length: in our case 12.5-8.5=4 ms, that means a 250 Hz resolution; however this is not a problem since we have the near-field response to take care of the low region.

Now it's time for the **Off-Axis measure:** leaving the mic in the same position used till now, rotate the speaker baffle to form a **30°** angle with the microphone-horizontal line; then leaving the markers as for the On-Axis and without changing the volumes, click twice on F FOCAL A then on MEASURE/ FREQUENCY RESPONSE/30 DEGREE and a new file will be created: *"Focal A 30"*. Repeat the procedure rotating the speaker baffle to reach a **60°** angle and again, leaving markers and volumes as they were, click on MEASURE/ FREQUENCY RESPONSE/ 60 DEGREE and again a new file will be created: *"Focal A 60"*. To go deeper, you can perform this measure at **45°** as well, even if SW doesn't have this option: place the speaker to form a 45° angle, and again, leaving markers and volumes as they were, click on MEASURE/ FREQUENCY RESPONSE/ 45 DEGREE and again a new file will be created: *"Focal A 45"*.

Now open *"Focal A On Axis"* and add to the chart the two new files created to have a complete view of the driver response at various angles:



On Axis with the Driver Center + Off Axis at 30°-60°

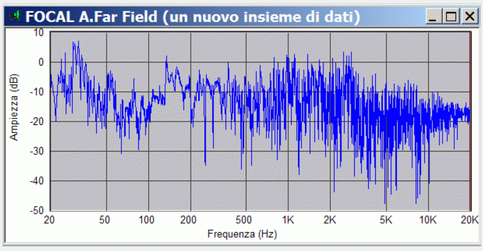
As you can see the Focal axis very well at 30° making possible a high cut crossover; the last chart On-Axis doesn't have a little less "sweetness alike" then the previous (312pts point) I choose to reached the 1024pts points of resolution increasing my card delay (the DMX Fire allows to do it from the right channel) and pushing the first marker at almost 0; also a sample rate of 96k helped to achieve it.

Now let's do the **On-Axis with the Acoustic Center**: Measuring all the speaker, with its crossover, will give us a response that combined with the near field one will show us the complete speaker response, like the ones we see in the magazine. What I will show you now, instead, is a procedure based on the technique described by Zaph Audio, to measure each driver response to correctly import them in a simulation program, with the aim to preview the filters to be used. Place the microphone-center at the end I came from (look at the drawing at the top of the page), and start with a distance of 1 meter; **connect all the drivers** and be careful with the amplifier volume since the tweeter doesn't have a crossover yet! Now click twice on F FOCAL A and then on or on MEASURE/ PULSE RESPONSE; check SW VuMeter to be sure the values are almost equal for the left and right channel and in the 20-25 range; open *"Focal A On Axis"* and set the markers as you already know; with the markers set, click twice on F FOCAL A and then on or click on MEASURE/ FREQUENCY RESPONSE/ GATED; check SW VuMeter to be sure the values are almost equal for the left and right channel and in the 20-25 range; open *"Focal A On Axis"* and add the new file created: *"Focal A Gate"*. Since all the drivers were sounding, rename the file in *"Focal A Gate Total"*. Now just leave connected one driver at a time and without moving the mic nor changing the volumes and the markers, repeat the procedure for each driver, remembering to rename each driver file to easily recognize it as the end.

In the **not gated far field** measurement, all the data coming from the microphone are used by Speaker Workshop to draw the response, therefore the environment interacts completely with the speaker. This kind of test is useful to better understand where to place the speakers in their final location, allowing us to try different positions and to compare the responses; however always try your ears as well, cause they will be your final judge! You can use the responses to practice your hearing as well: knowing, by the measurement, that one speakers position is more bass rich than another, try to figure it out by listening; to make it more hard, do the contrary: first listen and note the differences, then measure to prove your conclusion. As a general rule, a speaker close to the back wall causes an increase in the bass region, as well as if it stays close to corner walls. Same thing happens the more the woofer is close to the pavement.

To perform the not gated far field measurement, **use both the speakers** and place the microphone where you usually listen to music, thus recreate your music room with the difference that the mic will be your ears. To use both the speakers, try the MONO switch of your amplifier, otherwise use an Y cable to connect both the amplifier left and right input to the sound card left line out; do not connect both the speakers on the same channel of the amplifier output, unless you know what you are doing! With everything connected as in the gated measurement, and the volumes set to get a 20-25k (depending on your card clipping level) on both channels of Speaker Workshop VuMeter, Sample Rate and Sample Size to the maximum that works best with your card as you tested in the SWSetup section, choose a REPEAT COUNT of 5 (in OPTIONS/ PREFERENCE/ MEASUREMENTS) or more, depending on your test-room noise and used Sample Size (I use 20 repeat) and check that the USE PREEMPHASIS box is unmarked; then click twice on

and then on or click on MEASURE/ FREQUENCY RESPONSE/ FAR FIELD; check SW VuMeter to be sure the values are almost equal for the left and right channel and in the 20-25k range; a new file will be created: ***"Focal A.Far Field"***. Open it.

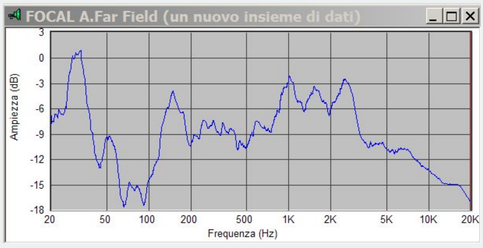


SW Not-Gated Far Field

As you can see it is not very readable, therefore **we need to smooth it**: click on TRANSFORM/ SMOOTH and select 1/3 octave then

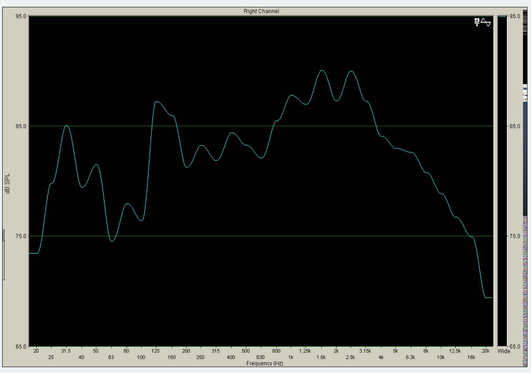
to see the result immediately; if you press CANCEL the smoothing will not be applied while clicking on

will make it permanent:



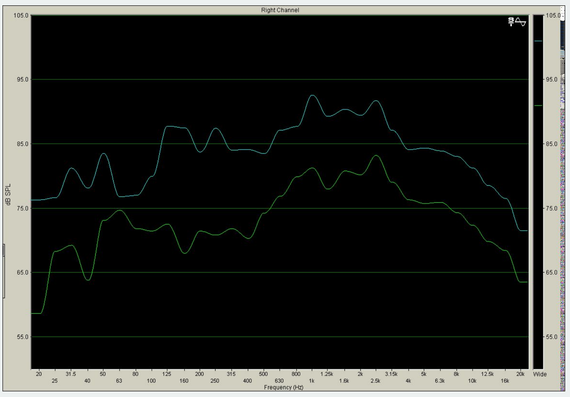
SW Not-Gated Far Field : Smoothed

Actually a better and faster way to perform this measurement is to use a **Real Time Analyzer**, like SpectraRTA; I did it and let's compare the results:



Far Field with RTA using Pink Noise

As you can see they are almost the same, telling us that there is a big hole in the 30-150 Hz region; the aim is to reach a not too bounced response, with the high frequencies gently rolling down. To give you a better idea of what the speakers positions can cause, I compared two measurements: with the speaker on the floor and with the speaker on a table at 80 cm from the floor. Here it comes:



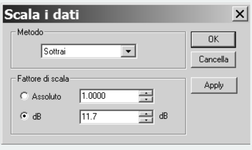
Far Field with RTA: on the floor - 80 cm. from the floor

As I said before, a woofer closer to the floor causes a boost in the lower region, that in our case helped to normalize a bit the response. Well, after this long journey together, it's time to start our own project, mine is the Double Chamber Reflex!

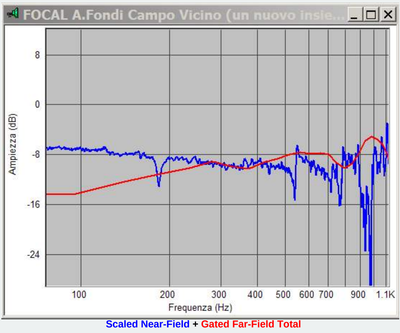
We are arrived, finally, at the last point in which we will **splice** together the near-field response with the far-field one to obtain the complete speaker response as you see in the Hi-Fi magazine. Since the two responses have different levels we need to scale down the near-field one, so open it and, by double clicking in it, ADD the gated far-field response, the one measured *On Axis with the Acoustic Center* and with all the drivers working:

Near-Field + Gated Far-Field Total

Before scaling, we have to decide at which frequency the splice will occur, knowing that it has to happen in the range delimited by the far-field lower limit (250Hz) and the near-field higher limit (730Hz). So taking a look at the blue response, the 200-500 Hz region is flat therefore ideal for our purpose; the red line shows a possible point in the 300-400 Hz area, so I decide for a joint frequency of 350 Hz. Once decided the frequency, we can **scale down**: click on TRANSFORM/ SCALE :

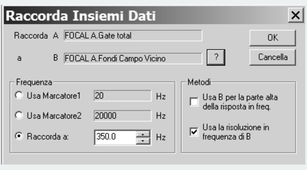


Since the near-field has an higher amplitude, we have to choose SUBTRACT and in the dB write a value of 10 to start, that is a 10 db amplitude subtraction is performed on the NF graph. Click on Apply to instantly see the result; keep changing the dB value till you are satisfied with the result; use the ZOOM option to limit the area of interest to better see it. When done, click on OK to make permanent the scaling:



Scaled Near-Field + Gated Far-Field Total

Now eliminate the Far-field response from the near-field chart and close it. Then open the gated far-field response ***"Focal A.Gate Total"*** and click on CALCULATE/ SPLICE :

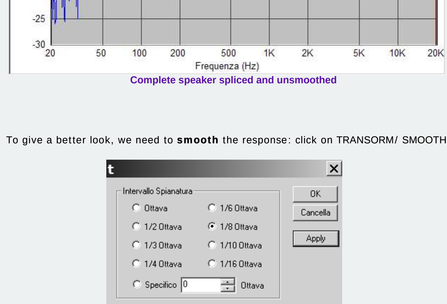


The **A** is the source dataset (the dataset currently being viewed) and is not editable.

The **B** is the near-field response, click on ? and select the file.

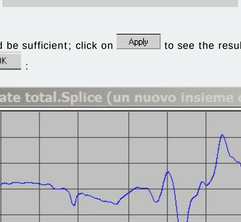
Frequency: select the frequency at which the splice takes place.

Methods: either A or B can be the high frequency portion of the result. Also, the frequency resolution can be based on A or B data. Check the appropriate boxes to use B rather than A (the default). In our case, A has the higher frequency, while B the better resolution. Select OK to perform the splice, and a new file with a Splice suffix is created: ***"Focal A.Gate Total.Splice"***:



Complete speaker spliced and unsmoothed

To give a better look, we need to **smooth** the response: click on TRANSORM/ SMOOTH :



A 1/8 octave should be sufficient; click on Apply to see the result that isn't permanent till you click on OK :

Complete speaker spliced and smoothed

The chart is not very nice for sure, but as I said before this was just a test, whose intent was to show how to use Speaker Workshop. One more measurement is waiting for us, the not-gated one!

A critic part of Speaker Workshop is the Phase, since **it's possible to have latency changes**, therefore the starting point of the impulse response can vary: to see it, just repeat many times a impulse response and check if the the first impulse always starts at the same time point. If the start of the impulse is different during two measurements, the phase too will be different. With the Terratec DMX the variations happen inside 0.01 ms (remember that 0.1 ms corresponds to 3 cm). This is a limitation of SW and of the sound card, therefore is better to use WDM drivers that gives some stability. Knowing this, **it will be impossible to know exactly the real phase, nor the time of flight, nor the real offset between the drivers**. Moreover, **the obtained relative phases of different drivers are not comparable for crossover design**, dependign on how much the impulse start jumps. To show you the problem, I made an acoustic measurement that I used as reference; then I repeated the measurement 3 times, without changing anything. I compared each measurement to the reference one, and I made a chart in which the phase differences was shown:

Phase variations on 3 measurements: woofer

All the measurements are gated far field, with 10 repetitions: the Focal woofer has a phase change lower than 1° till 3000 Hz, therefore it's not a big problem. I repeated the comparison with the ATD tweeter, and the results are positive too:

Phase variations on 3 measurements: tweeter

This is an ideal situation, that shouldn't create problem during the filter simulation. To prove it, I imported, in a crossover simulation program, the measured drivers response on acoustic axis, and I changed, in the simulation program, the woofer offset till the measured total response (all the drivers working) and the simulated total response were coincident. Then I connected a II° order filter on both the drivers, crossed at 3000 Hz, and compared the speaker simulated response with the speaker measured response and here's the positive result:

Speaker Freq. response with filters: measured - simulated

An even more precise method of measurement is to:

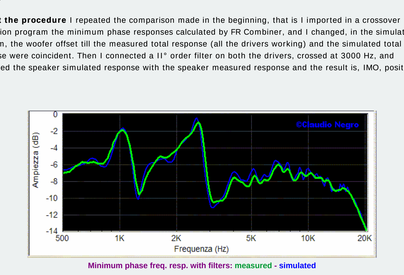
1) Measure the first driver and check its *reference* impulse response (file *Measurement.in.I* that is located in the SYSTEM folder) to know at how many ms the first impulse starts: write it down.

2) Measure the second driver and again check its *reference* impulse response looking at the file *Measurement.in.I* located in the SYSTEM folder to know at how many ms the first impulse starts: it has to start at the same ms of the first driver. If it doesn't, repeat the second driver measurement till the two values are equal.

The above method is valid only if the input channels (left and right) of your sound card have equal or constant latency: to know it, connect the *Line In* and *Line Out* of the sound card with a *Loop Cable* (left *Line Out* connected to both left and right *Line In*); create a new SIGNAL by clicking on RESOURCE / NEW / SIGNAL or use one that you already have; double click on the SIGNAL then MEASURE / FREQUENCY RESPONSE. Always verify in the SW Vu-Meter that you are not clipping the signal; now double click on *Measurement.in.I* located in the SYSTEM folder and a chart will open; right mouse click inside the chart and ADD the *Measurement.in.I* located in the SYSTEM folder too. Zoom in the area were the impulses start: if the two impulses start at the same time, you are a lucky man! **Probably the two impulses do not begin at the same time**, therefore calculate of how many ms is the difference between the two impulses start and write down the value; now repeat this measurement many times (double click on the SIGNAL then MEASURE / FREQUENCY RESPONSE), and each time write down the impulses start difference: if this difference value is constant, thus the two channels move together even if shifted, you can use the described method.

So till now we have seen that if the latency problem, that makes the first impulse not to remain stable at one position (it sort of jumps around on the horizontal time axis), is small, 0.01 ms in my case, there will be no problem in the crossover simulation. Further more, we have seen a method that allows an even more precise phase measurement, at the cost of an extra measurement time. But what happens if our sound card is too jumpy in latency? For cases where the first impulse moves very much, I (with *Bruno Dalla Carbonare* and *Valerio Russo)* have studied a procedure to calculate the minimum phase for each driver: doing this, we can overcame a changing impulse start problem since the minimum phase is calculated through the driver frequency response and *Hilbert transform*. First thing to do, is to export the response from SW, then convert it into logarithmic scale by using a program that is found in the Download folder; then import the file in *FR Combiner* where the minimum phase is calculated; finally export the new file.

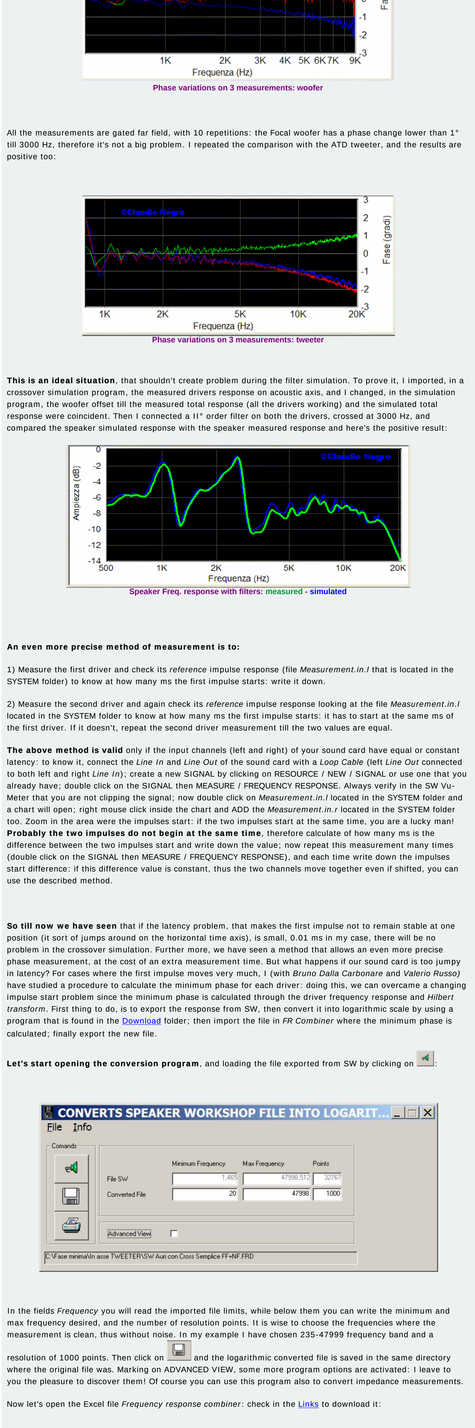
Let's start opening the conversion program, and loading the file exported from SW by clicking on :



In the fields *Frequency* you will read the imported file limits, while below them you can write the minimum and max frequency desired, and the number of resolution points. It is wise to choose the frequencies where the measurement is clean, thus without noise. In my example I have chosen 235-47999 frequency band and a

resolution of 1000 points. Then click on and the logarithmic converted file is saved in the same directory where the original file was. Marking on ADVANCED VIEW, some more program options are activated: I leave to you the pleasure to discover them! Of course you can use this program also to convert impedance measurements.

Now let's open the Excel file *Frequency response combiner:* check in the Links to download it:



1) Load the logarithmic converted file by clicking on EXECUTE. In this column you will read also the file frequency limits and number of points, that has to be equal to the ones chosen during the conversion.

2) Select *1Drv Base* e *Precise Phase & Group Delay* then click on *Extract Minimum Phase* and after some seconds the minimum phase will be calculated through Hilbert transform. To save the new file, go back to **circle 1** and click on LOAD so that it changes to SAVE and then EXECUTE: you will be asked where to save the new file.

It's important to know that this method works well if the drivers response is well extended in frequency: **David L. Ralph** illustrates the problem in his "*Hilbert Transform Generated Minimum Phase & Errors*" and "*Driver Model Accuracy and its Impact on Phase.* So it is important to use the complete response for the woofer (near field + far field) and to get a well over 22k Hz response for the tweeter: if your sound card reaches 96k Hz of S/R, the problem is solved since it will allow you to get a measurement till 48k Hz. If your response reaches only 24k Hz (sound card with a S/R of 48k Hz), you can use the NORMALIZER, to extend the response both in the low and high region, with the desired slope. Remember to calculate the minimum phase after having used the NORMALIZER as showed in the **circle 2**. It's useful to use a calibrated mic with a linear extension well beyond the 20k Hz.

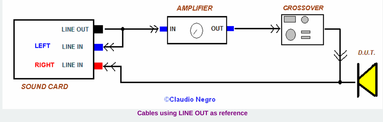
To test the procedure I repeated the comparison made in the beginning, that is I imported in a crossover simulation program the minimum phase responses calculated by FR Combiner, and I changed, in the simulation program, the woofer offset till the measured total response (all the drivers working) and the simulated total response were coincident. Then I connected a II° order filter on both the drivers, crossed at 3000 Hz, and compared the speaker simulated response with the speaker measured response and the result is, IMO, positive:

Minimum phase freq. resp. with filters: measured - simulated

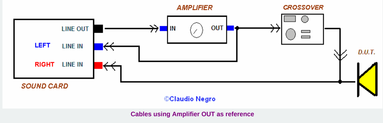
Now that I showed you the way to go, experiment yourself and have fun!

It is possible to measure, through Speaker Workshop, **the parallel Network Transfer Function** (NTF), that can help you out in the crossover construction. The procedure was first introduced, as far as I know, by *Guglielmo Caioli*, and then studied by *Valerio Russo*, to which go our thanks. Let's start looking at the necessary connections: basically it's like doing a frequency response measurement, but the amplifier output is connected to the filter input and the filter output is connected to the driver **and to the DATA channel of the sound card** (LINE IN right); for the Reference channel (LINE IN left) we can use the sound card Line Out or the amplifier Output: don't use more than 500 mV from the amplifier Output, or you can burn the sound card. **Using the amplifier Out as Reference**, brings some advantages: since SW measures the voltage difference between the Data and Reference channel, the amplifier gain will not be taken into account, therefore the chart will show 0 dB of filter attenuation in all the range where the filter is not working! Moreover if we sum the NTF chart to the driver unfiltered response chart, we obtain the filtered driver response.

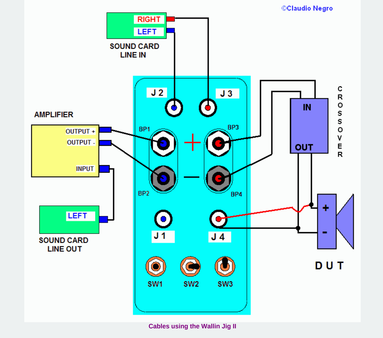
With Guglielmo's method we measure **the Network Transfer Function with the real driver impedance**: thus we can verify if our filter is right as we wanted, since it is very common that, in complex filters, the components tolerance brings to different than needed filters.



Cables using LINE OUT as reference

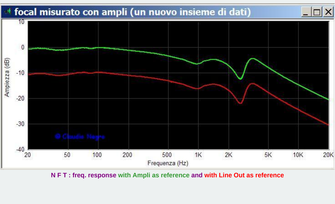


Cables using Amplifier OUT as reference



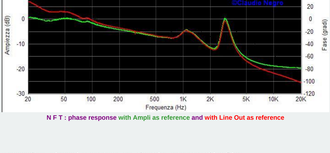
Cables using the Wallin Jig II

Once the cables are ready, put the amplifier volume to its minimum, and connect a voltmeter to its output; then create a new signal, RESOURCE/NEW/SIGNAL, or use one you already have; double click on it and then MEASURE/ FREQUENCY RESPONSE and increase the ampli volume till you read a 300-500 mV AC on the voltmeter. Repeat the measurement till you get the desired volt reading, and pay attention not to try the sound card, especially if you are using the amplifier output as Reference. Once the volume level is OK, repeat the measurement and open the newly created file, that is the Network Transfer Function: in the next chart I compared the NTF of the same filter measured using, as reference, the Amplifier Output and the Line Out. As you can see, the only difference is in the amplitude level. The filter is a second order pass bass, with 2 parallel RLC in series with the Focal woofer



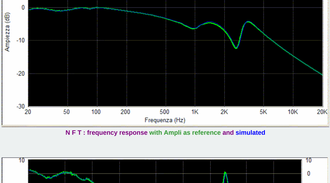
N F T : freq. response with Ampli as reference and with Line Out as reference

Now let's see the phases: here the limit of the measurement made with the LINE OUT as reference comes out; in the band extreme range, the phase is wrong.



N F T : phase response with Ampli as reference and with Line Out as reference

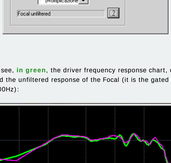
Next is a comparison between the measured and simulated responses of the filter:



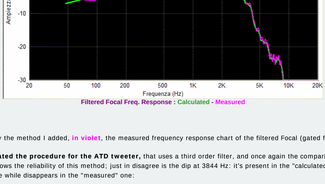
N F T : frequency response with Ampli as reference and simulated

N F T : phase response with Ampli as reference and simulated

If we have used the Ampli Output as reference, we can **sum the NTF and the unfiltered driver response to get the filtered driver response:** open the NTF chart (the one measured using the amplifier as reference) and click on CALCULATE/ COMBINE; a window will appear and choose MULTIPLY and then click on [?] to select the unfiltered measured driver response file:



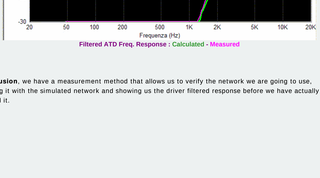
After clicking on [OK] we will see, **in green**, the driver frequency response chart, obtained through the sum of the network transfer function and the unfiltered response of the Focal (it is the gated far field measurement, therefore it's not useful below 300Hz):



Filtered Focal Freq. Response : Calculated - Measured

To verify the method I added, **in violet**, the measured frequency response chart of the filtered Focal (gated far field).

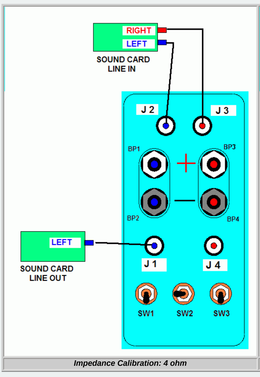
I repeated the procedure for the ATD tweeter, that uses a third order filter, and once again the comparison chart shows the reliability of this method; just in disagree is the dip at 3844 Hz: it's present in the "calculated" response while disappears in the "measured" one:



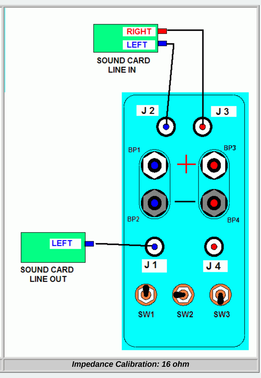
Filtered ATD Freq. Response : Calculated - Measured

In conclusion, we have a measurement method that allows us to verify the network we are going to use, comparing it with the simulated network and showing us the driver filtered response before we have actually measured it.

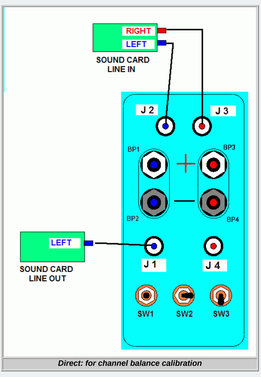
I don't use the **Wallin's Jig II**, as you already know; nevertheless, with the help of Nelson Wood, I decided to show some information and pictures since it is the most used Jig by the Speaker Workshop users, hoping to be helpful to someone. Anything about Wallin's Jig II can be found in Eric's home page, so I just add some additional information about his write up for those **who don't use the Ensoniq AudioPCI** sound card, but a newer card: do not be concerned with getting the card to output 2.83 Volt as Eric states, since modern SC hardly reach 2 Volt as they have just a pre-amplifier. Instead use the Speaker Workshop VU meter feature to get a sine wave signal in the 14-15k range and an MLS signal in the 20-22k, as you can find in the Calibration folder. Moreover, do not short out BP3 and BP4 when doing the calibrations and setting the volume levels, since it doesn't work. About the **first version of Wallin's Jig**, using the Protection LEDs caused problems, so just take them out, but you must be very careful with the volume levels and double check the connections before doing a test or you can ruin a card very easily, since there will be no over voltage protection.



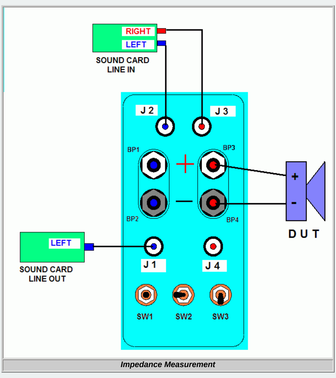
Impedance Calibration: 4 ohm



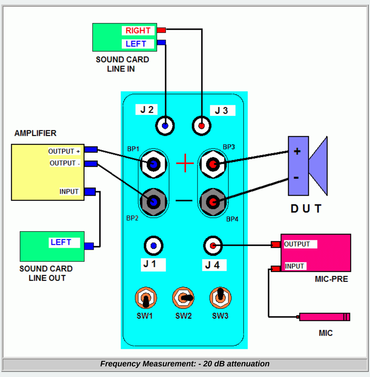
Impedance Calibration: 16 ohm



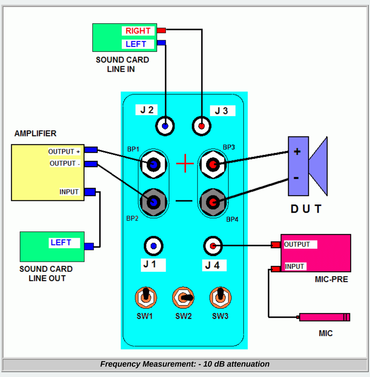
Direct: for channel balance calibration



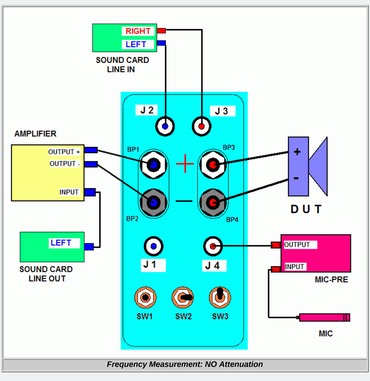
Impedance Measurement



Frequency Measurement: - 20 dB attenuation



Frequency Measurement: - 10 dB attenuation



Frequency Measurement: NO Attenuation

USEFUL LINKS

SPEAKER WORKSHOP

Website: http://www.speakerworkshop.com

Description: Speaker Workshop Home Page.

RIGHT MARK AUDIO ANALIZER

Website: http://audio.rightmark.org/index.html

Description: Program that tests your sound card, useful in choosing the right volumes setup.

QUICKMIX

Website: http://www.prodtechpart.co.uk/quickmix/index.htm

Description: Program that allows to save and recall the Windows mixer settings.

ERIC WALLIN

Website: http://mysite.verizon.net/tammie_eric/ericindex.html

Description: Here is a good tutorial SW using and gives the instructions on how to build the JIG and a mic preamplifier.

JAN'S SPEAKER PROJECT

Website: http://home.hccnet.nl/ine.dick/

Description: Project and measurement of a two way speaker using SW.

LEO'S HOME PAGE

Website: http://bellsouthpwp.net/l/j/ljfrank/Speaker_Tips.html

Description: Page about speaker building and Projects including a SW tutorial on taking measurements and using the crossover portion of the program.

Bob's Speaker Stuff

Website: http://www.geocities.com/rbrines1/

Description: Many single drive speakers projects in here, with measurement made by Speaker Workshop.

Frequency Response Combiner

WebSite: http://www.pvconsultants.com/audio/utility/frc.htm

Description: Excel file that allows to import frequency responses and calculate the minimum phase.

Unified Box Model for Loudspeaker Design

Website: http://home20.inet.tele.dk/kou/index.html

Description: Useful Excel spreadsheet program for calculation and simulation of closed, ported, passive radiator and band pass systems.

DOWNLOAD AREA

File name: SW tutorial ENG.zip

Description: My Speaker Workshop tutorial converted in PDF (ver. 2.1).

File name: sw 1_06 int.zip

Description: Speaker Workshop vers. 1.06 international.

File name: data check.xls

Description: Excel format file that allows to check the measured driver parameters. It calculates the reflex box tuning frequency and Qb, and others parameters. The instructions are in the folder **SW Impedance/Verify**

File name: acoustic measure.xls

Description: Excel format file that allows to calculate the mic-driver distance for the near-field measurement and the resonance for the far- field.

File name: Verify TS.exe

Description: Program written by *Bruno Dalle Carbonare*, that calculates T/S parameters starting from the free air impedance and the added mass or closed box one and allow to verify the measurements. The instructions are in the folder **SW Impedance/Verify**.

File name: SW logaritmic converter.zip

Description: Program written by *Bruno Dalle Carbonare* to convert into logaritmic scale the impedance and acoustic measurements made with Speaker Workshop.

File name: ecm8000cal.frd

Description: Generic calibration file for the Behringer ECM8000 mic. The file was given from Bob, that had his mic calibrated. Remember that this calibration file is perfect just for Bob's mic, for the rest of us its just a generic calibration file!

File name: WM61A.frd

Description: Generic calibration file for the Panasonic WM61A.

File name: WM60A.frd

Description: Generic calibration file for the Panasonic WM60A.

File name: QuickMixIn.exe

Description: Quick Mix Version 1.05. Setting your mixer up for use with Speaker Workshop is a rather tedious affair. Luckily someone (Martin Saxon) has built a neat little program called QuickMix that lets you do this easily.

File name: RMAA53.exe

Description: Right Mark Audio Analizer version 5.3. To test the audio card to verify the better volume settings.

File name: unibox407.xls

Description: Useful Excel spreadsheet program for calculation and simulation of closed, ported, passive radiator and band pass systems.