Practical Considerations on Designing and Measering LCR RIAA Filters

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Introduction

First off all, HiFi and vinyl reproduction are topics of my private interests. The hardware and software hardware/components presented here were bought paying the normal market prices. There was no commercial relationship to any of the companies or persons mentioned here. The underlying concept should also work with components supplied by different vendors. The challenges are often to convince someone to manufacture products that are not available in the current product line and even more challenging if they can be realized with reasonable investment in time and material.

It is well known that vinyl music reproduction needs equalization. During the cutting process low frequencies need to be attenuated and high frequencies need to be boosted. The cutting head has more or less constant velocity characteristics and recording amplitudes increase while the signal frequency decreases which would dramatically reduce recording times or even worse would result in touching or cutting in neighbour groove tracks. During playback the exact inverse filter characteristic must be applied to receive a flat frequency response. In the early days of vinyl production many companies used their own EQ filters for the cutting process until the famous RIAA became a worldwide standard. I will only focus on the RIAA standard here as it is in place for over 50 years and all new records are produced following this standard. Although, other filter characteristics could be implemented with this approach.

Every phono stage therefore needs an RIAA filter somewhere in the amplification chain. Many possibilities exist to realize such an filter. There are two main types of filters, active and passive. Active filters change the amplification factor frequency dependent of one or more amplification devices or use a frequency dependent feedback to mimic the required equalization. Passive filters only use passive components and have a distinct signal damping of about 20 dB at 1 kHz. Although passive filters have this supposed disadvantage of high damping so that the signal needs to be re-amplified many people report that they prefer passive equalization because of the better sound quality. It should be mentioned that an LCR-RIAA module belongs to the passive filters.

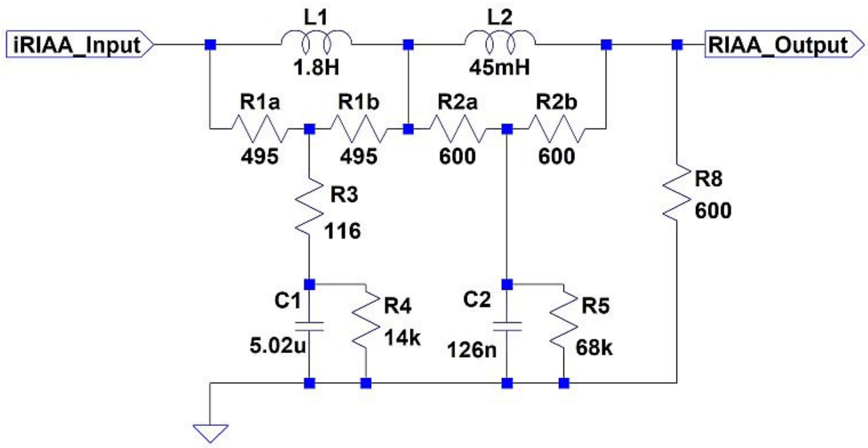
The simplest way to realize an passive RIAA filter is to use resistors and capacitors. I will not go into detail with respect to this as these type of filters have been discussed many times in literature and elsewhere in the web. But if we focus our view on the impedance of such an RC filter with respect of what the driving stage sees as load impedance we must recognize that this impedance is not constant. Many people will not care about this and it may not be an important issue. But some do and I assume that was one of the reasons why LCR filters came into interest for phono stages. They are based on bridged t-network filters which can be designed for constant impedance.

I have to claim that in this paper there are no really new theoretical findings or mathematical proofs. See it more as a guide or workflow around this topic. The theoretical credits go to others and I think that it is not necessary to provide references for well known facts except if it is useful to point the reader’s attention to valuable sources in the sense of this paper or for a deeper understanding. I have only used and combined well known things with a maybe different interpretation.

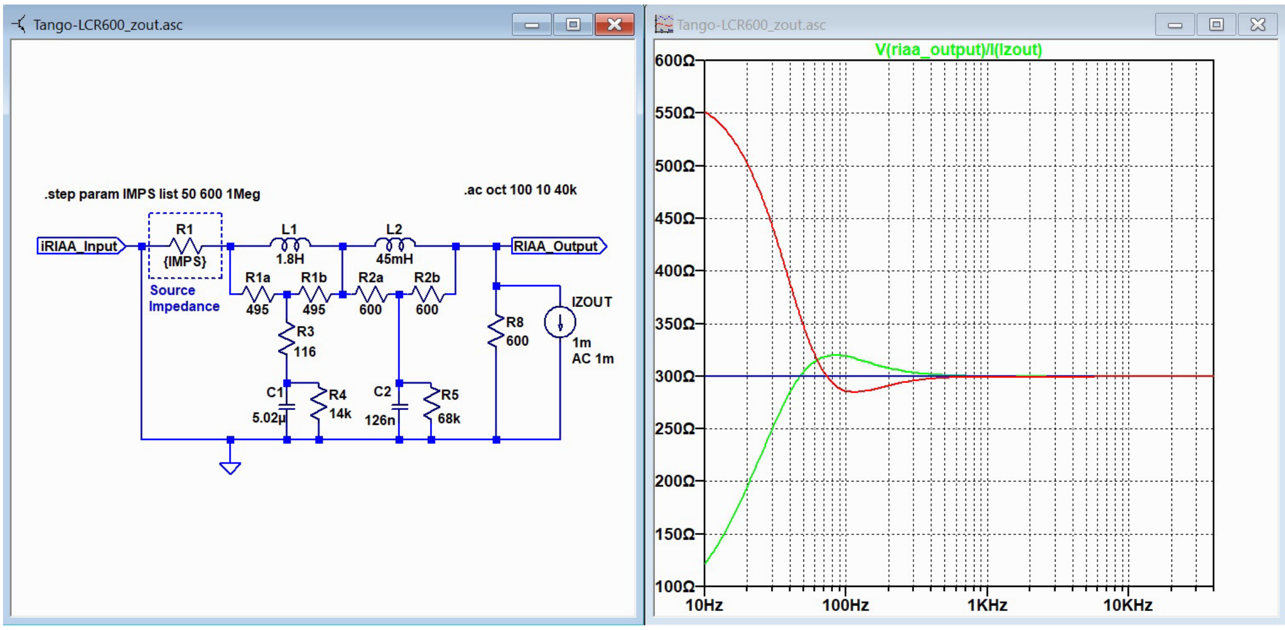
The main focus of my interest are tube amplifiers. Because of their usually higher internal impedances compared to solid state devices they need a special design to be able to drive a low impedance LCR RIAA filter.

Driving LCR Modules with Tube Amplifiers

The probably most famous LCR RIAA module is the Tango EQ-600P:



It has a constant impedance of 600 ohms at both input and output side. Quite a challenging load if using tube amplifiers to drive a load of 600 ohms. Therefore, we often see a step-down transformer before the LCR. This lowers the driving impedance but also reduces the signal level at the input of the LCR. One way to overcome this is to use a step-up transformer after the LCR. There are some examples of such an design in the web. Line level input transformers often have a rated impedance of 600 ohms and usually work well with driving impedances of 600 ohms or below. From my experience the damping of the input transformer is better if it is driven from a source near the rated impedance. But what is the output driving impedance of the Tango LCR-600 module - or any similar 600 ohms design - which is terminated with an 600 ohms resistor? Let’s take a look at the following picture:



Within Spice it is quite easy to simulate the impedance by introducing a current load and configure a specific AC stimulus (here with 1mA). The output voltage at this node divided by this current gives the impedance at this node. The driving source impedance was stepped in this calculation as green = 50 Ohms, blue = 600 Ohms, red = 1Meg. We can see that a following step-up transformer will encounter a constant driving impedance of 300 Ohms only if the filters driving source impedance is exactly 600 Ohms.

If we intend to use one of the available input transformers which are rated as 600 ohms devices why not use an LCR module which has an output driving impedance of 600 ohms together with the termination resistor? This was my initial question and the answer is of course easy: use an LCR-1200 module. This should provide a proper damping of the input transformer without too much loading it down at the secondary winding which would influence the LCR response. Besides this, some dB might be gained in this way by reducing the insertion loss because of a slightly less demanding load to drive for the first stage in our phono amplifier.

Fine for now, but if we search the transformer vendor market we will not find such an module. If we look for even more higher LCR impedances, i.e. 10k, like the well-known Audio Note M10 amplifier, the inductors become very large and should be hard to wind properly. Of course, there are LCR-10k modules but these are out of focus here.

The original Tango LCR-600 modules are no more availably as factory new devices. But there are other companies that still offer similar devices. But all designs seem to be nailed fix at 600 ohms. Theoretically, it is quite easy to change the design towards 1200 ohms as the inductors and resistors need to be doubled and the capacitors need to be halved.

The most demanding challenge here is to produce the inductor L1. What is not obvious at the first view that the designers of the Tango module did a really good job with respect to this. A real inductor has of course some stray components like parallel capacitances and leakage inductance but the most determining property with respect to the LCR, while the previously mentioned stray components having under control, is the series resistance of this L1 inductor to achieve the right bass equalization. The Tango modules L1 1.8H inductor has about 26 ohms copper resistance. What we will see later this is very near to the best theoretical value for flat frequency response. Not much less and not much more. There are some designs which fail to yield a low enough series resistance for this inductor and more than 100 ohms copper resistance for L1 in an LCR-600 is definitely too much in my opinion.

How to find the correct component values?

One way to search for optimal component values would be to choose intuitive start values into a Spice simulator and play around as long as the accuracy is acceptable. But this would be more or less try and error and maybe will never lead to an fully acceptable solution. Spicing is a really good method to check and understand certain things but more satisfactory would be a theoretical proof how to determine the best fitting values. If we do a search in the web about this issue we will not find much.

The most comprehensive work and at the same time I would call it a brilliant work was published in a special vendor forum and on diyaudio.com [1]:

http://www.intactaudio.com/forum/viewtopic.php?t=1306

http://www.diyaudio.com/forums/analogue-source/227736-riaa-lcr-mythos.html

This work covers a theoretical consideration and mathematical solution contain copper resistances and additional series resistors for the inductors. Resistors and capacitors are treated as ideal components. The mathematical constraint which simplifies solving the equations is that the driving impedance is set equal to the terminating impedance of the LCR. In fact, driving the LCR from a too low source impedance (the amplifiers first stage) will cause the bass equalization become slightly inaccurate. We can use a low driving impedance but should adapt the terminating resistor what we will see later.

Additionally, Claus provided an Excel sheet for designing your own LCR modules for any impedance you want. I used this Excel sheet to find the theoretical „best" component values for an LCR-1200 as a design starting point. A Solver is also provided to find the best set of real component values corresponding to the theoretical set of values.

A very interesting consideration is an inductor with nearly zero copper resistance which is or course not possible in reality but a very good assumption to understand some things. The resulting inductor value for L1 would at about 3.44H. Wait a minute, this is far below that of the double Tango value of 3.6H. How could this be? The series resistor of the inductor makes the difference here. It can be either the internal copper resistance or an additional series resistor or the sum of both. If we now type in 3.6H in the Excel solver the reported series resistance is about 47 Ohms. For an LCR-600 and an 1.8H inductor it is 23 Ohms. Now we are able to assess how good these Tango LCR modules were designed! An inductor of 3.8H needs an series resistor of 105 ohms. The solver has also an maximum error macro for every component set but this seem to work only with Microsoft Excel. It seems reasonable to keep the series resistor as small as possible. 5% copper resistance or less of the rated impedance of the module seems to be a reasonable target design point.

At this point you need to find someone who is able to wind such an inductor which should have of course also a quite stable inductance over the audio range. A longer search later I found a product at Sowter Audio Transformers in the UK. Many thanks to Brian Sowter and Brian Last who took the time to reduce the internal resistance from their existing design with an special wire and at the same time providing some taps to have additional options for fine tuning.

The product name is 1459e for the HF inductor with a nominal value of 92mH and 1462e for LF with a nominal value of 3.7H. The 1462e has a copper resistance of 59 ohms which is a really good fit to the target design. These inductors were exclusively manufactured for me and might undergo some slight adjustment of the taps based on my experience if there will arise other requests for these products.

Verification of the Results with Spice Simulations

Can we trust the values that are coming out from the Excel sheet? Let us put an example into spice and check the resulting cuve. For this purpose I generaly use the freely available LTspice software which is available here:

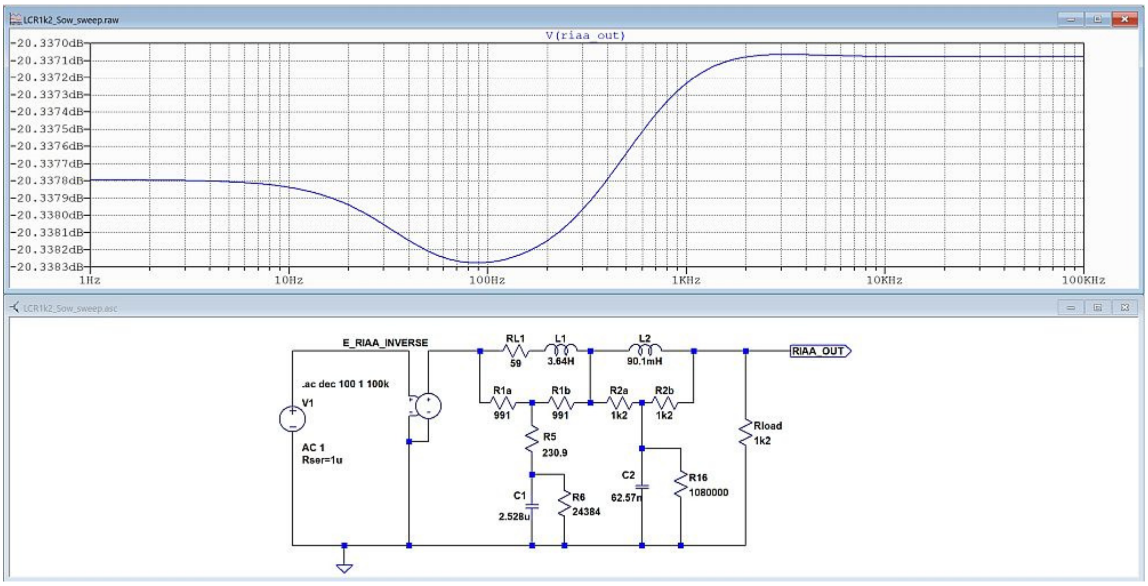
http://www.linear.com/designtools/software/#LTspice

We should use a mathematical correct inverse RIAA function and not some component values hacked together from the internet. How to do this is described here [2]. Simply use a voltage controled voltage source (which has the symbol "E" in Spice) with the Laplace function of the RIAA filter as input value. The function should be normalized to 0dB damping at 1kHz:

Laplace=0.101*(1+s*3180e-6)*(1+s*75e-6)/(1+s*318e-6)

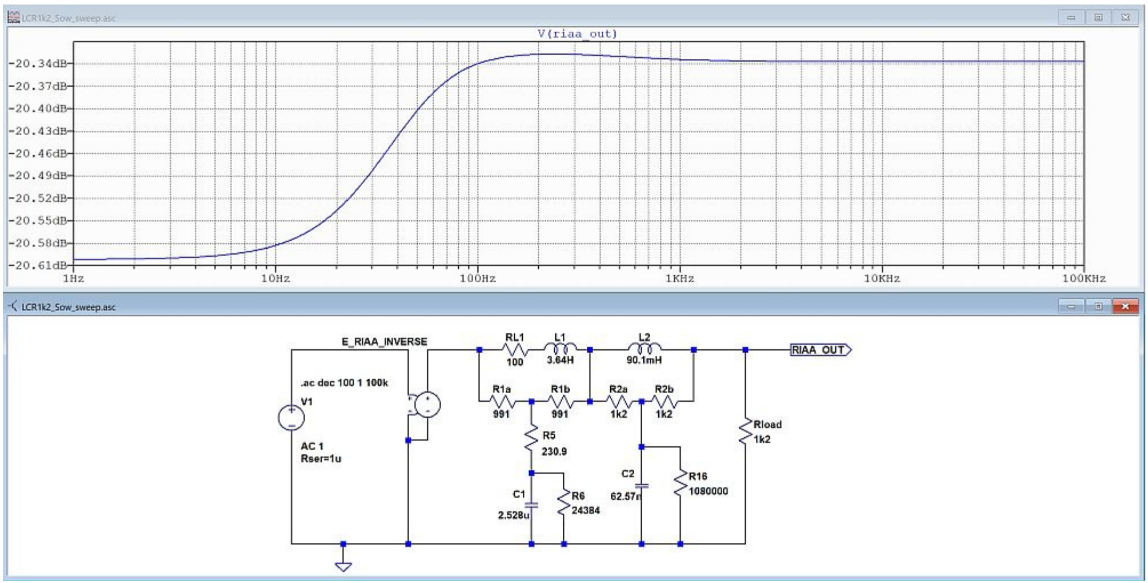
I did not use the additional single 50 kHz pole because the so called Neumann constant is a myth and was never used in any cutting machine in that way as a single pole. Anyone who is of other opinion can use the Excel sheet and also incorporate the "Neumann constant".

What would be the best fitting value for the inductor having 58 ohms internal copper resistance? The Excel solver gives 3.64H. So, one of the taps of the inductor should give something in the vicinity of this value. I will use this value for the Spice verification:



Don‘t become confused by the curved behaviour – look at the dB scale. I think there remains nothing more to say about the proof of the concept and the underlying math from Claus. The result is better than +-0.01dB in this simulation analysis. In reality you will not see such an perfect behaviour because inductors have leakage inductance and parallel capacitances. Real capacitors are also no perfect devices and even resistor are not as perfect many might believe.

If your manufacturer state that the requested LCR1200 can only be made with L1 having 100 ohms internal resistance see what happens then:



At least one order of magnitude worse but still acceptable. For 100 ohms copper resistance the best fitting inductance value is 3.78H. What we should recognize is that there is no complete right or wrong. If L1 will have a higher internal copper resistance it's inductance must simply be slightly higher and the other related component values need to be re-adjusted.

To summarize, what we have to look at in praxis is that the inductance value need to fit to the internal copper resistance of the inductor and at the same time as low as possible to minimize copper losses. Besides this, the RIAA inductor should have some taps to fine tune the inductance. The same holds for L2 but the internal resistance is less critical and generally much lower because of the much lower inductance.

How to Measure and Verify the LCR filter Modules?

Since a log time I have two passive inverse RIAA filter modules in my stock. One was bought and the other was self-build with accurately chosen component values after Lipshitz and Jung. I also own a quite cheap analogue function generator delivering max. 20 Volts output into 600 ohms. But two passive damping modules in series after the generator, the inverse RIAA module and then the LCR-RIAA, will almost give nothing as clean output signal after the LCR module. Noise is already overlaying the results.

A long time I was thinking about how to make the inverse RIAA actively or digitally in a simple way and at the same time absolutely accurate but had no idea how to achieve this. Just another interest of mine, measuring loudspeakers and designing cross-overs, gave me the right "click" how to do this.

John Mulcahy provides his fantastic REW or Room EQ Wizard software:

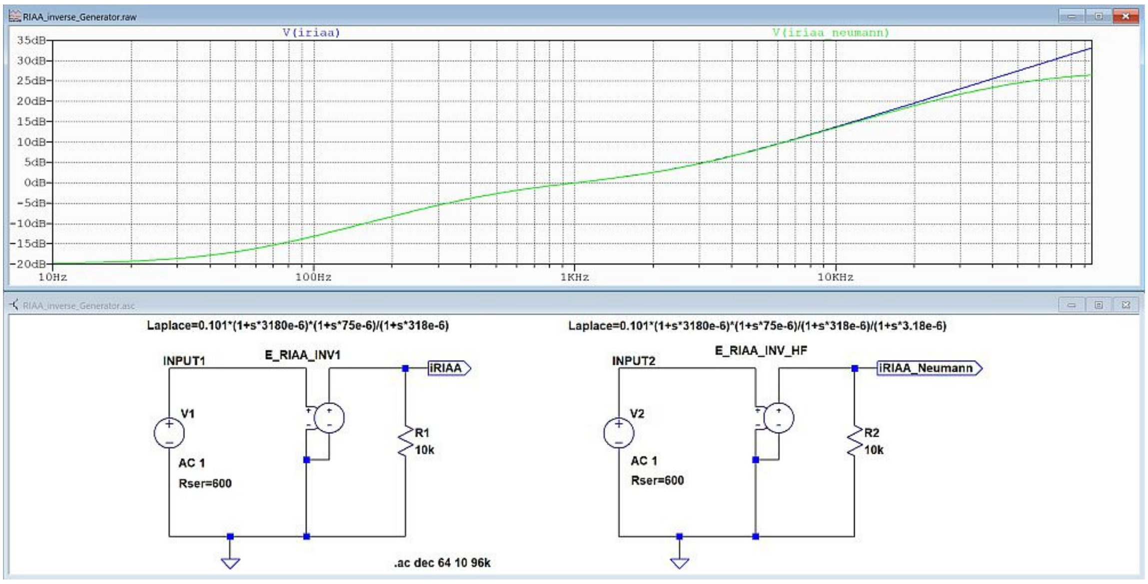
https://www.roomeqwizard.com/

I use an Focusrite Solo audio interface for this purpose with 192kHz sample rate which is already much more than sufficient for loudspeaker measurements. Within REW an calibration of the complete active amplification chain can be done. So, the output stage and the receiving input stage can be calibrated to a nearly flat response up to 96kHz if using 192kHz sample rate. I was really surprised how good the frequency response of this audio interface already is without calibration. with +-1dB from 10 to 50 kHz. After 50 kHz there is a more or less steep descent. Perfect for our use case.

The line output terminals of the audio interface will feed the LCR and from here (with or without a following input transformer) goes back to the instrument input which has a high enough input impedance not to influence the LCR termination very much. Bear in mind that the complete amplification chain was calibrated in advance. So, nearly every error except the LCR and input transformer is compensated for and we only see the pure properties of these.

But how to incorporate an inverse RIAA Filter in REW? We can do it afterwards internally in the REW software by adding a microphone calibration file. This option in REW is generally used to correct the microphone error. Exactly here we place a really brute force microphone calibration: the inverse RIAA filter – lossless! Again, the output stage of the audio interface feeds the LCR module and the input stage records the resulting voltage frequency dependent because we can do sweeps in REW. At the end of the process the inverse RIAA is added to the curve as microphone calibration. This works absolutely perfect! This is at the same time a very easy method the measure the frequency response of every phono stage.

The only missing link up to here is the calibration file containing the inverse RIAA characteristics. But this can simply be done with the help of LTSpice:



By Clicking on the waveforms window and right-click "File" -> "Export data as text" we choose V(iRIAA) and save it as text file. Some editing is needed to bring the output to an for REW readable form. Here is only a list of some grid values how the file should look like:

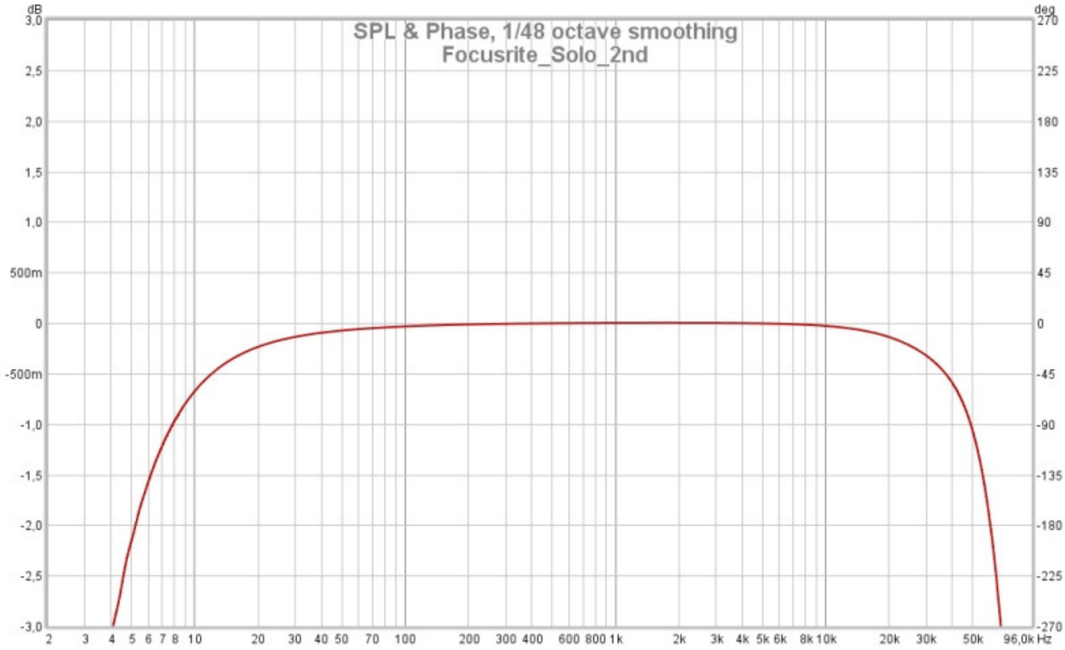
```
Sens Factor = 0.0dB, SERNO: iRIAA-RCA
10.0          19.75
20.0          19.28
504.3         2.61
1008.6        -0.03
2152.7        -2.93
10020.6       -13.75
20041.2       -19.64
96000.0       -33.20
```

Preparation of the Signal Measurement Chain

At first, we need to calibrate the amplification chain. What is not obvious at the first view that the cables used should have low capacitances between the signal wires. Especially with the tests including the step-up input transformer the cable used after the input transformer towards the instrument input of the audio interface should only have a few pF as any capacity appearing at the secondary side of the transformer are reflected with the square of the turns ratio to the primary side. For my first measurements I used a 5m shielded jack plug cable nearest to my hands and was very disappointed because of steep HF fall-off. Until I thought it would be better to check the capacity of the cable or more precise between the signal wires with an LCR meter. It showed 1.5nF signal to screen/ground! No wonder that he results were very much influenced by such an load, especially with the step-up transformer in the chain. After this experience I build my own short twisted pair cable without any shield adding only a few pF.

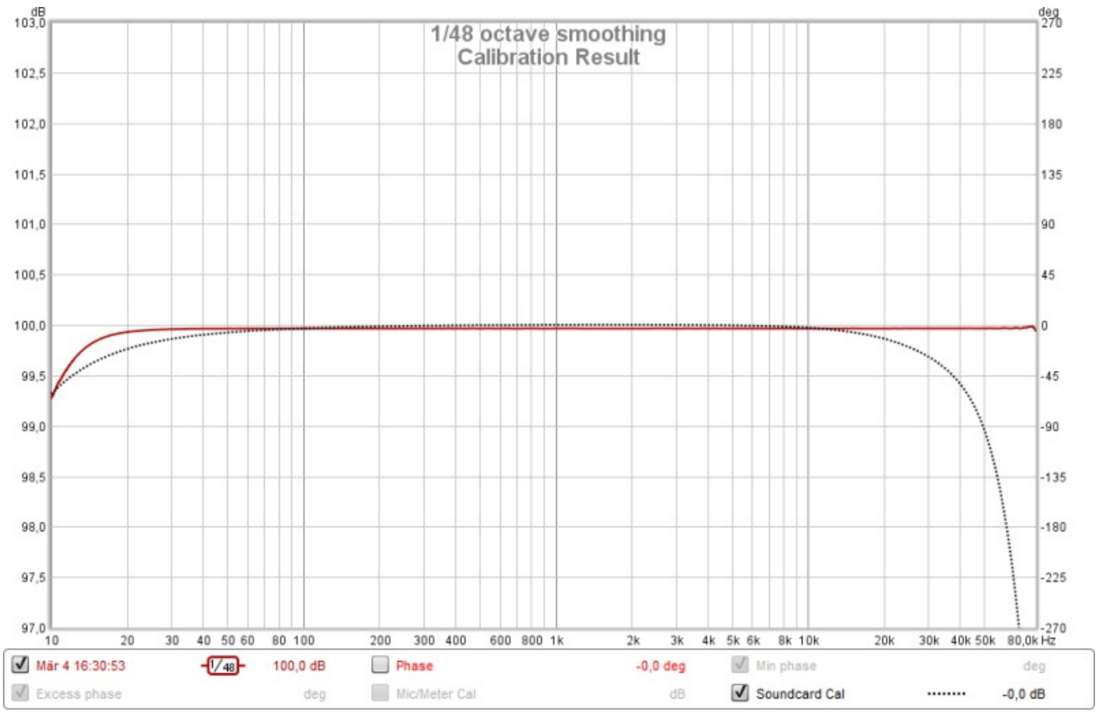
I tried to get the value of the input capacity of the instrument input from the vendor but they could not provide this to me. My guess is that it cannot be that much as the frequency response is too good even with higher impedances in the chain. I assume a value of about 50 pF including the short twisted pair unshielded cable of about 20cm especially prepared for this purpose.

Here is the raw frequency response of the Focusrite solo (2nd generation) cinch output to instrument input:



This looks already really good +-1 dB from 8 to 50 kHz. The instrument input was used because it has 1Meg ohms input impedance. The line inputs which could also be used in balanced mode have 52k Ohms input impedance. Together with the input transformer in place this would already influence our results when reflected down to the primary side with the square of the turns ratio.

For more precision we want to put the amplification chain out of the results. REW can do a calibration which will be overlaid to all measurements once stored internally. This is the result of a new measurement with added calibration correction:

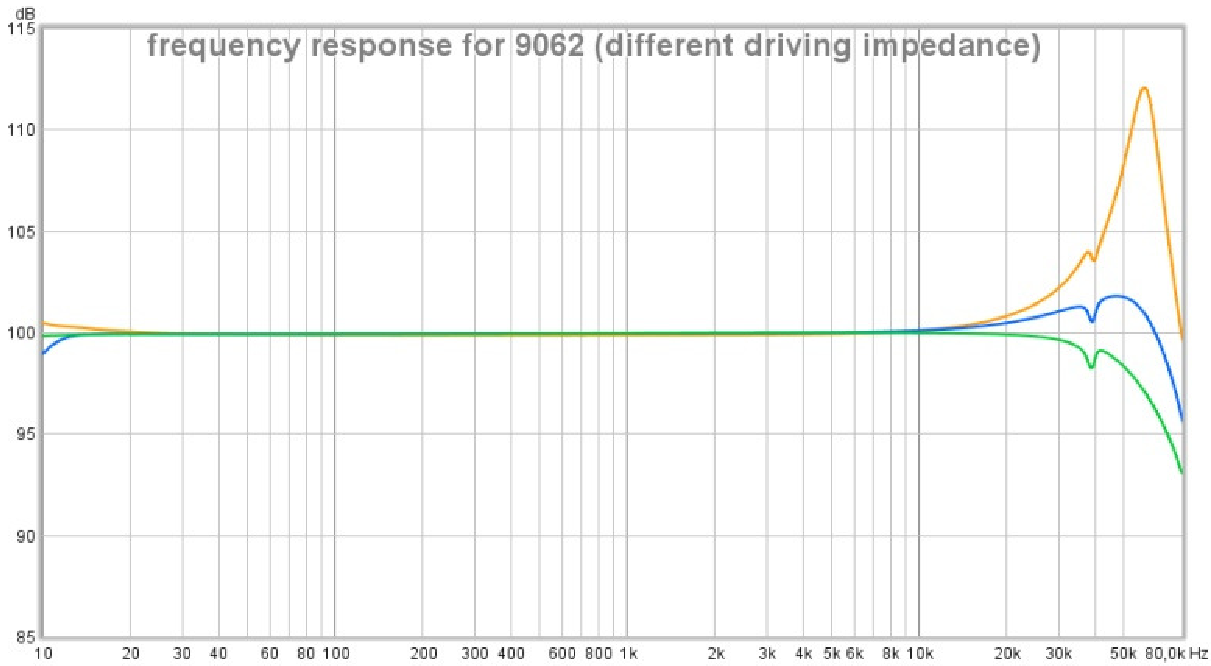


With this setup we can be quite sure that what we see between 20 Hz and 80 kHz will be the pure behaviour of the components tested and any influence of the underlying amplification chain can be neglected. Some more words about the dB scale. The relative dB values should be accurate but the absolute values are chosen with respect to better comparison between measurements. Within REW it is easy to add an offset to a graph.

The Line Step-up Transformer

To work together with the LCR unit the Sowter 9062 line step-up transformer was selected. This transformer has a higher internal impedance and is rated as 600/40k. It can be wired as 1:8 which gives the rated nominal impedances but I will use them as 1:4 with the secondaries in parallel. This is similar to 600/10k.

When we use step-up transformers we always have to pay attention to internal resonances and the interaction of the winding inductance or leakage inductance with external capacities. Series resistance can damp these resonances quite well but too much will cause HF fall-off and insertion loss. Just to use just the right driving impedance should be a good idea. But let's start with simply hook-up the 9062 in our chain without secondary termination. The Focusrite Solo has a quite low output impedance of only 47 ohms at cinch output. The result is a huge HF resonance (orange curve):



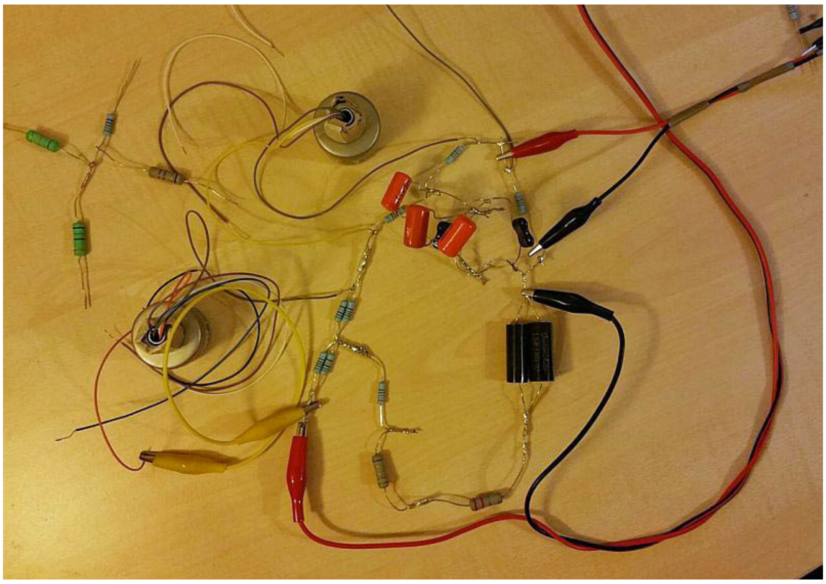
Our LCR-1200 has an output impedance of 600 ohms. The blue curve shows an added 560 ohms resistor in series before the 9062. The big 12 dB peak is now largely reduced to about 2 dB. If we think about a following tube stage with Miller capacitance this peak will be damped further. But how far can raise the driving impedance until the fall-off becomes too much? A driving impedance of 1k looks still very good (green curve) but any more might be too much damping. The other option is to use a termination resistor on the secondary side of the transformer but we have already the LCR1200 which has 600 ohms driving impedance. A too low secondary termination makes our RIAA curve inaccurate.

Some words about the resonance seen around 40kHz. I am not entirely confident if this is an artefact of my measurement setup. I have seen similar effects of an energy saving lamp in the vicinity of an input transformer causing strange effects in this frequency region! The power line noise encountered these days is a challenge that did not existed decades before but will not be discussed in more detail here. This issue was also discussed with Sowter and it could be caused by a very small internal resonance or asymmetry how different windings are manufactured around the core. Anyway, the effect is very small and should be totally inaudible.

The LCR-1200 Module Assembled

If we assess the overall accuracy of an phono amplifier we start with passive component deviations which are quite small, maybe in the range of 1-2%. The active components should have the biggest deviations for tube amplifiers. My personal opinion is that we do not need this high accuracy of the LCR as shown above and can make some small simplifications. A still reasonable good agreement with the RIAA filter would be achieved without using the two parallel resistors R4 and R5.

When I received the RIAA inductors a quick and dirty LCR was hooked up first by putting the required components together not really perfectly resembling the theoretical values. I would guess the deviation of the parts from the theoretical optimum to about 2-3%. This is a close-up of the loosely connected parts:

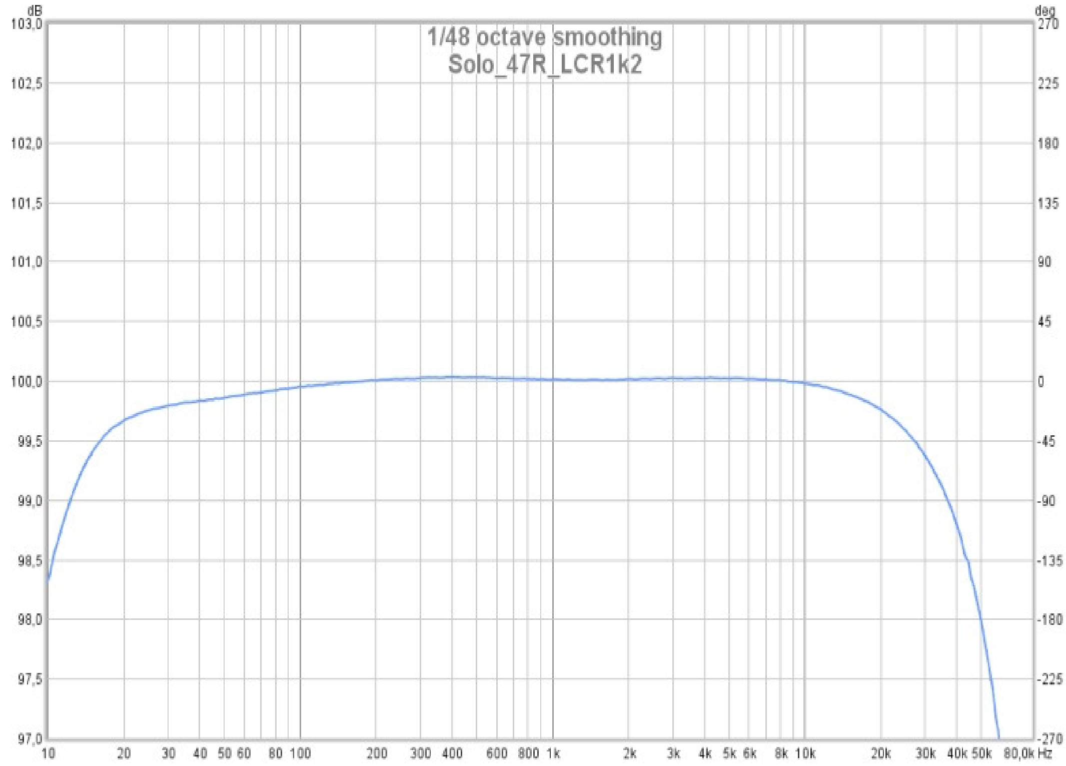


Then I attached my cheap Voltcraft LCR meter to do some first impedance shots at different anchor frequencies. The Voltcraft is able to show the impedance of an attached device at 100Hz, 1kHz, 10kHz and finally 100kHz. The result can be seen in the following table (values measured from the source of the filter:

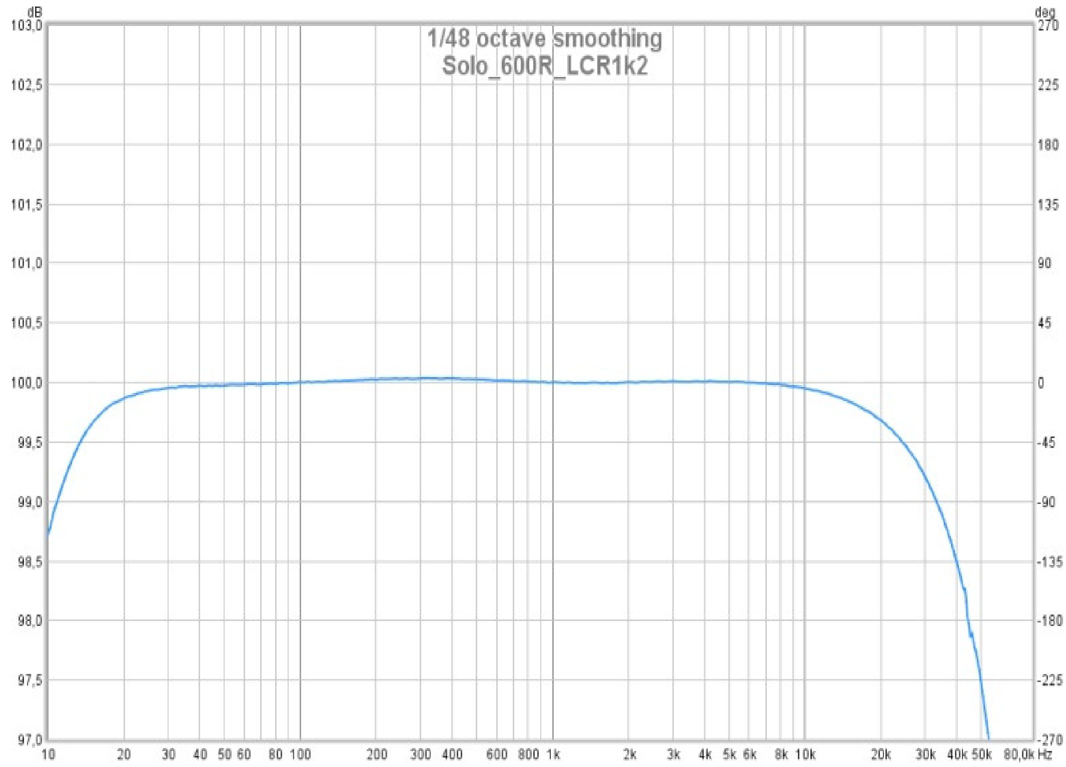
100Hz	1kHz	10kHz	100kHz
1200 ohms	1198 ohms	1198 ohms	1193 ohms

Ok, the measurement instrument must show garbage. This cannot be! Too good to trust to? On the other hand it would mean that the RIAA inductors must have a very small parallel capacitance and would be more than acceptable in our application.

But the first REW shot told me how good this approach is working quite nicely! The Focusrite Solo with its 47 ohms output impedance drives the LCR-1200:



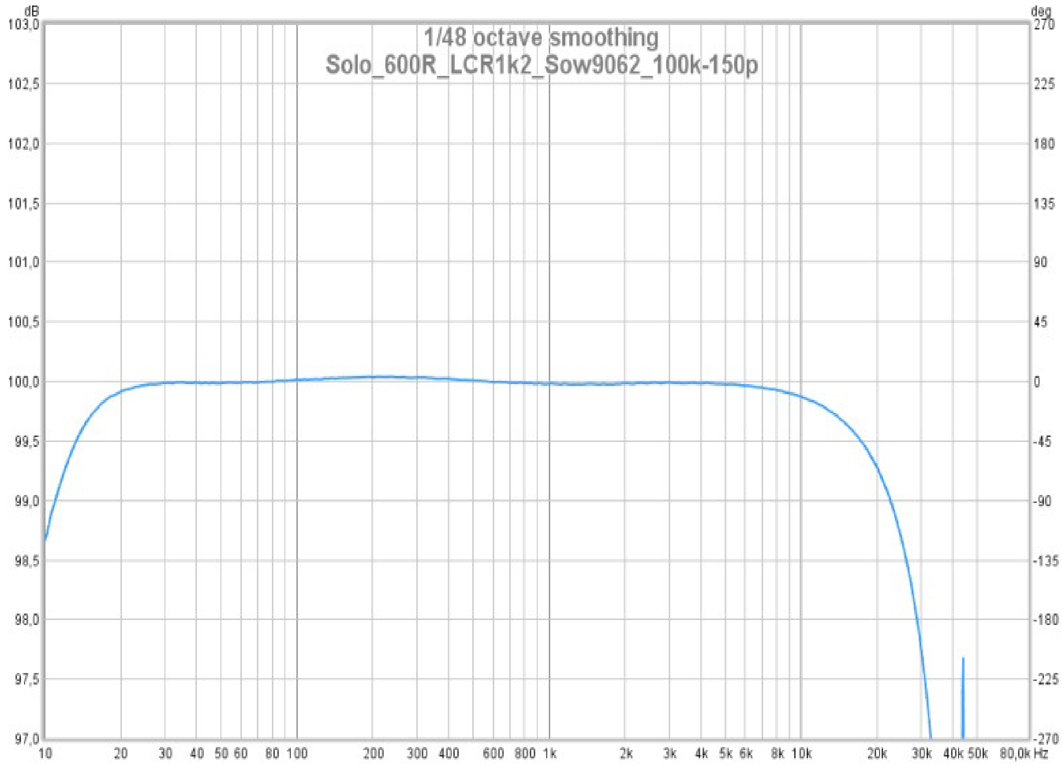
And in the next picture with added resistance to yield approximately 600 ohms driving impedance:



What we see is either that the LCR RIAA filter already works nearly perfect and otherwise that the driving impedance should not to be too far away from the filter impedance to get the right bass equalization. Although, one could reduce the terminating resistance slightly to lift the bass response if the driving impedance is too low.

The LCR-1200 module with step-up transformer

By introducing a step-up transformer after the LCR-1200 module we can get back some of the filter losses. But nothing comes for free. Although, I would rate the result as more than acceptable:



The secondary side of the step-up was slightly loaded with a parallel 100k resistor and a parallel 150pF capacitor. This should show fairly accurate the influence of a following tube amplification device. We again see this slight resonance around 40k and we have to pay with a little bit more HF fall-off. Who wants to further improve these results could bypass the whole LCR module with an RC combination. A 180k resistor in series with 150pF is a good starting point.

Conclusion

It is straight forward to adapt the famous Tango LCR module to other impedances but internal copper resistances of the inductors need to fit to the corresponding inductance value. The inverse RIAA filter could be realized by using REWs microphone calibration option. The output impedance of an LCR-1200 module with its terminating resistor of 1200 ohms is an reasonable driving impedance for 600 ohms rated line step-up transformers.

Literature

[1] Claus Weber, „RIAA LCR Filter Mythos Or How to Design“
[2] Alexander Potchinkov, „Simulation von Röhrenverstärkern mit SPICE"