# Designing a High-Accuracy Passive Inverse RIAA Filter, Why Build One at All?

Ala Electronics · 2025-07-05 6:46 pm

Ala Electronics
Member
Joined 2025

2025-07-05 6:46 pm #1

*[Update]*

*For easier navigation, here's a quick index of the posts in this thread:*



Hi everyone,

Just wanted to share the start of a project I've recently finished, a passive inverse RIAA filter that ended up being surprisingly useful (and a bit more involved than I expected). It's designed to help with testing both MM and MC phono preamps, staying accurate across the full audio band: 20 Hz to 20 kHz.

Laser-engraved dual-channel filter enclosure:


The motivation was pretty straightforward. Most IRN filters I came across, whether commercial or DIY, either lacked the precision I needed, didn't handle MC levels well, or relied on active stages that I wanted to avoid. If you're building or measuring phono stages, having a clean, analog-domain IRN filter is super helpful, especially when you want to check frequency response or verify your RIAA curve.

The original "The Audio Amateur" article, 1980:

# A High Accuracy Inverse RIAA Network

*by* STANLEY P. LIPSHITZ
*and* WALT JUNG

READERS WILL BE AWARE of the recent correspondence between the two authors regarding RIAA equalization in phono preamplifiers[1,2]. This has led to a re-examination of many familiar circuits, and has shown the need for changes and corrections to many of these in order to improve their accuracy. Extremely high accuracy can indeed be achieved by careful design, such as in the Jung-White modification of the Dynaco PAT-5[3], and many readers may be interested in having available an inverse RIAA circuit of sufficient precision to enable them to measure the deviations of phono preamplifiers to a high order of accuracy, as well as to conduct critical listening tests.

*TAA* published in 1971[4] an inverse RIAA design by Reg Williamson, shown in *Fig. 1*. In original form this circuit is fairly accurate, and indeed is suitable for many requirements. However, it can be improved by changing $R_1$ from 910k to the theoretically ideal 883.333k, or 887k (nearest 1% value). But, even then the termination resistor $R_3$ slightly displaces the middle RIAA time constant $T_2$ (ideally 318$\mu$s) from its desired value, because of the way in which the component values interact in determining the circuit's overall time constants. For a complete theoretical analysis of this see "On RIAA Equalization Networks"[5], copies of which are available, free of charge (while the supply lasts), from the first author at Dept. of Applied Mathematics, University of Waterloo, Waterloo, Ontario, N2L 3G1, Canada.

Ideally the frequency response of an inverse RIAA network should be as in *Fig. 2*. The corner at $f_4$ should not occur at all (i.e. $f_4 = \infty$ Hz); in practice it is unavoidable. In general it should be placed as high in frequency as possible, so as to minimize deviations occurring in the audio frequency range of interest, namely to above 20kHz.

In considering an improved inverse RIAA circuit, taking into account all significant interactions (e.g., the effect of $R_3$ and driving source impedance), we originally felt that an active pre-emphasis circuit might be necessary. Such a design, unfortunately, introduces as many problems as it circumvents. It was then decided that a slight modification to the Williamson circuit would be most practical. The circuit is shown in *Fig. 3*. Present owners of previously purchased Old Colony Williamson networks can easily adapt them to the new values, if they so desire.

FIG. 1

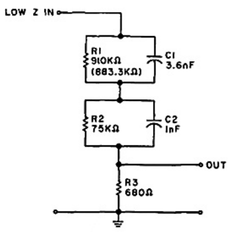
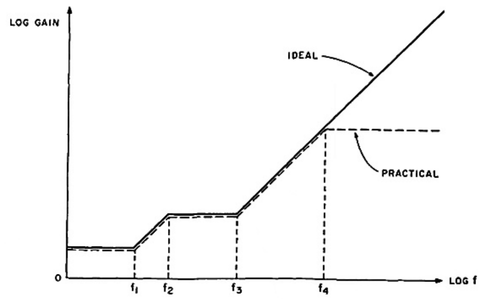



*Fig. 1. The original Reg Williamson circuit[4] and its modification.*

You will appreciate just how good this network really is if you examine the data in *Table 1*, which lists the relative accuracy of two of these networks for various test conditions. For some time Old Colony has been supplying the Williamson network with an ideal $R_1$ value of 883.3k and all 1% circuit components, as shown here. In the first column shown (listed as stock), this version of the network can be seen to offer better than ±0.1dB accuracy, with sample number 1 being appreciably better (due to values which fall closer to theoretical).

These measurements were taken with a 600 ohm load resistor, the kit value supplied for $R_3$. However, as noted above, the presence of $R_3$ not only places $f_4$ at

FIG. 2




Defining parameters (ideal case):
$f_1 = 50.05$ Hz; corresponding to time constant $T_1 = 3180\ \mu s$.
$f_2 = 500.5$ Hz; $T_2 = 318\ \mu s$.
$f_3 = 2122$ Hz; $T_3 = 75 \mu s$.
$f_4 = \infty$ Hz; $T_4 = 0 \mu s$.
*Note: In practice $f_4 \neq \infty$; $T_4 > 0\mu s$.*

*Fig. 2. The frequency response of an inverse RIAA network in ideal and practical cases.*



This one's based on the well-known Lipshitz & Jung design from Audio Amateur (1980), with a few ideas borrowed from more modern builds like the Hifisonix Accurate IRN. But it's been adapted and rebuilt from scratch with the following goals in mind:

•Dual outputs: –40 dB (MM) and –63 dB (MC)

•Fully passive RC topology, no active gain stages

• ±0.11 dB deviation from ideal RIAA (simulated)

•Separate shielded PCBs for left and right channels

•Premium hand-matched components for better tolerance

Resistors and capacitors were carefully hand-picked to ensure high precision and matched performance between boards:

I'll go into the schematic and how I structured the attenuation stages for the next post. Even if IRN filters aren't part of your regular toolkit, I'd love to hear what other folks here are using when testing phono preamps.

Cheers,
Alan

Drbulj, Tony Salsich, restorer-john and 2 others

Last edited: 2025-07-12 9:48 pm

**Mooly**
Moderator
Joined 2007

2025-07-05 6:51 pm #2

Welcome to diyAudio Alan 🙂

Installing and using the new 2023 version of LTspice

Ala Electronics

**Ala Electronics**
Member
Joined 2025

2025-07-05 7:26 pm #3

Thank you Mooly! Really glad to be part of this great community 🙂

**Ala Electronics**
Member
Joined 2025

2025-07-05 7:38 pm #4

Following up from the first post, here's a closer look at how the circuit was actually built. The filter uses a fully passive RC topology, simple in principle, but getting the precision right took a bit of effort.

The idea was to create two outputs, one at –40 dB for MM phono stages and another at –63 dB for MC inputs

Both paths follow the same curve, just scaled in level. Getting them to track each other closely required careful impedance planning, and quite a bit of part-matching, to be honest.

Many of the resistor and capacitor values didn't exist as single parts, so I ended up building combinations (series/parallel) to hit the targets more accurately. All components are 1% metal film resistors and polypropylene caps from WIMA and Nissei. A bit old-school, maybe, but reliable and consistent.

The layout is dual mono: each channel has its own board and its own enclosure. That decision wasn't just aesthetic, having the channels completely isolated actually helped with crosstalk performance when testing stereo phono stages.

Below is the schematic and a close-up from the build process. Would love to know what others here think, especially if you've designed something similar, or have tips on improving passive IRNs.

If resistors R13–R16 are changed to 562 Ω, and R17–R20 are changed to 40.2 Ω, the attenuation levels become:

1) –44 dB at 1 kHz for MM output

2) approximately –68 dB at 1 kHz for MC output

This modification may be useful if your phono stage has higher input sensitivity or if you want more headroom in measurements.

smndao and ranshdow

**Conrad Hoffman**
Member
Joined 2007

2025-07-05 9:37 pm #5

I built mine from the Hagerman article here- **https://www.hagtech.com/pdf/riaa.pdf** *(URL: https://www.hagtech.com/pdf/riaa.pdf)*
I admit to being lazy and only building one channel. Never really felt a need for two. I do have a construction advantage- the ability to measure small capacitance values with high accuracy and possession of a GR 1493 decade transformer. It turns out that measuring RIAA accuracy is harder to do than most people realize. If using a DVM, read the accuracy specs carefully.

Ala Electronics

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Z
**Zung**
Member
Joined 2005

2025-07-05 11:10 pm #6

Good job!
I use a different approach: I use a close but lower value component and patch it up with "shims", i.e. the 3.6nF can be a 3.3nF // 270pF // 33pF.
I also use a bridge type LCR meter to sort at least the main components; a **0.5% bridge** *(URL: https://www.aliexpress.com/item/1005008703519215.html?pdp_ext_f=%7B%22sku_id%22%3A%2212000046316591370%22%7D&sourceType=1&spm=a2g0o.wish-manage-detail.0.0)* cost only about $60.
Finally, polystyrene caps have a better tempco than MKP, but you have to hunt them down. I buy the Philips 1% type from eBay.

Monstercore and Drbulj

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S
**schiirrn**
Member
Joined 2018

2025-07-05 11:34 pm #7

About the "why build one at all": why don't you just use an inverse RIAA weighting filter in your measuring software?

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M
**MarcelvdG**
Member
Joined 2003

2025-07-06 12:19 am #8

Or if you measure it manually, an inverse RIAA spreadsheet?

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S
**stocktrader200**
Member
Joined 2008

2025-07-06 3:21 am #9

Most useful tool for building a preamp.

2025-07-06 8:13 am #10

**Ala Electronics**
Member
Joined 2025

> **Conrad Hoffman said:**
>
> I built mine from the Hagerman article

Thanks for sharing that. I actually went dual mono mostly to reduce crosstalk when testing stereo phono stages, but I agree that for most cases a single channel works just fine.

Your point about measurement accuracy is spot on. I'm currently using a DVM for quick voltage checks, but I know that's far from enough for validating the full curve. I'm setting up to do proper tests soon with a function generator and oscilloscope, and maybe some REW sweeps through a phono stage when that part’s ready.

And a GR 1493! That’s a serious piece of gear. Would love to see your build sometime if you have photos.

Thanks again for the insight.

2025-07-06 8:21 am #11

**Ala Electronics**
Member
Joined 2025

> **Zung said:**
>
> I also use a bridge type LCR meter to sort at least the main components;

Your shim approach is clever, I actually used something similar in a few spots where exact values weren't available. Combining caps or resistors gave me tighter results than single parts alone, especially with 1% tolerance limits.

I went with WIMA (FKP2) and Nissei film caps mainly because they're readily available, and I've had good experience with their long-term stability and consistency. Tracking down polystyrene is pretty difficult (and they tend to get expensive too), but I totally agree they're great for tempco and dielectric behavior.

That 0.5% LCR bridge you linked looks surprisingly good for the price. I'll definitely keep that in mind for future builds!

Ala Electronics
Member
Joined 2025

2025-07-06 8:43 am #12

> **schiirrn said:**
>
> About the "why build one at all": why don't you just use an inverse RIAA weighting filter in your measuring software?

Good point, and yeah, software-based inverse RIAA filters are definitely useful in many setups, especially when working entirely in the digital domain.

In my case, I wanted something hardware-based that could provide consistent, analog domain output levels for directly testing phono stages. A physical IRN lets me simulate real-world MM/MC signal levels and loading conditions without relying on digital compensation or assumptions.

It's not exactly a cartridge, of course, but it behaves close enough to help verify frequency response and gain accuracy with tools like an oscilloscope, sweep generator, or REW through a phono input.

And to be honest... I just enjoy building these things by hand 😄

sonicles and jan.didden

---

H

Hans Polak
Member
Joined 2005

2025-07-06 8:48 am #13

The source impedance driving this A-Riaa network will be part of the equation.
When not specified, accuracy will be affected.

When simulating the network in LTspice, and comparing it to a 100% accurate Laplace filtered source, it's easy to find the correct component values for a given source impedance.

In my case I'm using the 50R impedance signal generator that's included in my digital scope.

Hans

Ala Electronics

Ala Electronics
Member
Joined 2025

2025-07-06 8:51 am #14

> **MarcelvdG said:**
>
> Or if you measure it manually, an inverse RIAA spreadsheet?

Totally fair point, and I've definitely seen some great inverse RIAA spreadsheets out there.

I wanted something I could physically integrate into the measurement chain, especially when using sweep generators or REW through a phono input, where I'd rather not rely on post-measurement compensation.

Having a real-time analog filter also makes it easier to check things like channel balance and crosstalk stuff that's harder to spot if you're manually adjusting or re-referencing data afterwards.

And honestly, building the actual box was half the fun.

MarcelvdG

2025-07-06 8:57 am #15

**Ala Electronics**
Member
Joined 2025

> **stocktrader200 said:**
>
> Most useful tool for building a preamp.

I've definitely found it useful for checking gain accuracy and RIAA tracking early on, before diving into full listening tests. Having a consistent analog input makes it way easier to catch issues before they get baked into a design.

2025-07-06 9:15 am #16

M

**MarcelvdG**
Member
Joined 2003

When you use a piece of paper and a pen instead of LTSpice, you will soon find out that for resistive sources, the source resistance affects the poles in the exact same way as the load resistance. If you don't care much about the precise attenuation, you can reduce the combined value of R13...R16 to correct for the source resistance.

ejp and Ala Electronics

2025-07-06 9:20 am #17

**Ala Electronics**
Member
Joined 2025

> **Hans Polak said:**
>
> The source impedance driving this A-Riaa network will be part of the equation.
> When not specified, accuracy will be affected.

Great point, Hans.
Source impedance is definitely part of the network behavior, especially in passive IRN designs.

In my case, I assumed a 50Ω source impedance (to match typical signal generators), and selected components accordingly. The simulation was done in Multisim rather than LTspice, but I tried to match the RIAA curve with that loading condition as the baseline.

I've been thinking about testing the sensitivity of the network to variations in source Z, would love to hear more about how you approached it in LTspice with the Laplace reference.

2025-07-06 9:35 am #18

**Ala Electronics**
Member
Joined 2025

> **MarcelvdG said:**
>
> If you don't care much about the precise attenuation, you can reduce the combined value of R13...R16 to correct for the source resistance.

That's a great point, and I actually had an alternate resistor set in mind for that:

If R13–R16 are set to 562 Ω, and R17–R20 to 40.2 Ω, the 1 kHz attenuation shifts to –44 dB (MM) and ~–68 dB (MC), which better accommodates typical signal generator output impedances.

I has been mentioned on post #4 but glad you mentioned it.

Last edited: 2025-07-06 9:56 am

M

**Mark Tillotson**
Member
Joined 2018

2025-07-06 10:25 am #19

> **schiirrn said:**
>
> About the "why build one at all": why don't you just use an inverse RIAA weighting filter in your measuring software?

Because you can put a square wave through inverse-RIAA -> RIAA D.U.T. and see if the output is still square - very quick check of RIAA accuracy.

rayma, Zoji6645, Conrad Hoffman and 1 other person

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Z

**Zung**
Member
Joined 2005

2025-07-06 10:57 am #20

> **Conrad Hoffman said:**
>
> .... If using a DVM, read the accuracy specs carefully.

Most of the affordable DMM's are only OK-ish to about 1KHz.
If anybody knows any better, pls share the link.

Ala Electronics

M

**Mark Tillotson**
Member
Joined 2018

2025-07-06 11:05 am #21

My version of the RIAA circuit is 1/10th the impedance, using 88.8k || 36nF, 7.5k || 10nF, and a 91R : 10R output attenuator for MC. This gives a more representative source impedance for MC (and lower noise floor of 0.4nV/√Hz).

I added 1k + 0.45H impedance in series with the MM output to mimic typical cartridge impedance for more representative noise measurements too.

I further refinement was being able to switch in 10k instead of the RIAA section to give just a flat response attenuator too.

Ala Electronics

**Ala Electronics**
Member
Joined 2025

2025-07-06 12:19 pm #22

> **Mark Tillotson said:**
>
> I added 1k + 0.45H impedance in series with the MM output to mimic typical cartridge impedance for more representative noise measurements too.

Really clever move with the 1k + 0.45H. It mimics the complex impedance of an MM cartridge much more realistically than just resistive loading. That should give a more accurate look at input-stage noise performance in phono preamps. I kept the network at standard RIAA impedance levels mainly to match typical phono input loads and to preserve compatibility with basic bench gear, but I like the idea of building a low-Z version specifically for noise floor testing.

Out of curiosity, did you find any noticeable shift in frequency response compared to a flat resistive source? Or did the loading just mainly impact noise floor?

Inspiring approach overall 👍

M

**Mark Tillotson**
Member
Joined 2018

2025-07-06 12:22 pm #23

The 910 ohms of the divider for MC output swamps any MM output impedance which is at least 47k typically.

Ala Electronics

**Chris Hornbeck**
Member
Joined 2011

2025-07-06 12:47 pm #24

Do multiple samples in series/parallel taken from the (presumed) same production run actually average towards the bogie value, or do we still need "known" values and a way to measure them? Asking for a friend.

Lost in toleranceville,
Chris

ps: I'm sorry if the references are too obscure. You probably need to be old enough to be a John Prine fan and know his song *Dear Abby*, best on his 1988 live album.
Signed, Dear Abby

Last edited: 2025-07-06 12:53 pm

Ala Electronics

M

**MarcelvdG**
Member
Joined 2003

2025-07-06 1:07 pm #25

Mark, I hope you can short your 1 kohm and 450 mH for frequency response measurements, otherwise the effect of the LRC network will be counted twice, once by you and once by the cartridge manufacturer.

In principle, it's the cartridge manufacturer's responsibility to ensure a flattish response from the record to the cartridge output when the cartridge is loaded with the recommended load. That is, the combination of the mechanical and the electric (LRC) transfer has to be sort of flat. The phono preamplifier just has to have the right input impedance and the right (RIAA) response from the voltage across its input terminals to its output.

Ala Electronics and Chris Hornbeck

2025-07-06 1:11 pm #26

**Ala Electronics**
Member
Joined 2025

> **Chris Hornbeck said:**
>
> Do multiple samples in series/parallel taken from the (presumed) same production run actually average towards the bogie value, or do we still need "known" values and a way to measure them? Asking for a friend.

Dear Chris (Abby), statistically speaking, averaging samples from a consistent production run can help reduce random variation, assuming Gaussian distribution and no systemic drift. But in the real Toleranceville world, unless the bogie is tightly defined and verified, you're still flying blind without a known-good reference or calibration method.

Parallel hope, serial doubt.

Still measuring, not guessing. 🍻

Chris Hornbeck

2025-07-06 1:47 pm #27

**Chris Hornbeck**
Member
Joined 2011

I'm currently (and verrrrry lowly) in the process of building a ragtag prototype of the MvdG phono stage and a (better-than-my-1976-possibly-still 1dB ?) reverse RIAA lossy network, or better than is built into a Sound Technology 1200. Intent is to make a Lipshitz/Jung (883K) with 600 Ohm-ish R3. Can I assume (yes, I know) that I could reduce R3 arbitrarily without significant change to the important mid-band inflections?

All good fortune,
Chris

D

**Drbulj**

Member

Joined 2024

2025-07-06 3:08 pm

#28

Hi Alan,
Very nice and neatly executed project, congrats!
Since your are in precision, I can only drop a hint about components choice. For capacitors there are 1% polystyrene extended foil (extended foil means minimal inductance) caps, they are not in production for long time, but can be found, made by RIFA (PFE series) in 80's. Except they need to be found , they are also large, complicated all together but as far as I know the best filter cap ever.
Recently I finished preamp with passive RIAA and achieved max 0.04db mismatch between channels by using those, 0.05% resistors, and where I could not get exact value: hand picked from 2% resistors to reduce number in network.
Here is link showing deviation from left and right channel. I did not measure absolute riaa accuracy, but since components and math are right, there is no need to doubt. PS, in that post I mentioned 0.3db mismatch, think I was tired then, it is one zero more.
**https://www.diyaudio.com/community/...ono-preamplifier-thoughts.414816/post-8035424** *(URL: https://www.diyaudio.com/community/threads/fully-balanced-mc-phono-preamplifier-thoughts.414816/post-8035424)*

PS, Few good people here mentioned need of matching input and output impedance's here , IMHO excellent advice to make your tool even more usefull

Best,
Drazen

H

**Hans Polak**

Member

Joined 2005

2025-07-06 3:23 pm

#29

Alan,

I simulated your circuit in LTSpice, see example below with 70K load on the MM output showing the deviation from an ideal A-Riaa curve.
What comes out is that Rsource should be <= 50R to stay within 0.1dB up to 20Khz.
However, load impedance only affects overall attenuation but not the accuracy to stay within 0.1dB.

Hans

### Attachments

*(URL: /community/attachments/accinvriaa-jpg.1480376/)*

AccInvRiaa.jpg

364.4 KB · Views: 64

Drbulj

**Chris Hornbeck**

Member

Joined 2011

2025-07-06 3:42 pm

#30

Please retract my question in post 27. It's of course only a question of how much is too much, and meaningless.

Always good fortune,
Chris

Ala Electronics

S

**stellavox**

Member

Joined 2007

2025-07-06 4:33 pm

#31

As I do reel tape work, including preamps, Dave Slagle made me up an inverse board for NAB and IEC response.

Ala Electronics and Conrad Hoffman

Calvin

Member

Joined 2004

2025-07-06 4:43 pm #32

Hi,

as others already pointed out ... the source impedance should be taken into account.
I did exactly that in my own inverseRIAA .... switchable for source impedances of 0R, 50R, 550R and 600R ... as well as 1kHz reference levels of -40dB@600R and -60dB@60R (switch MM/MC) ... see attachments.
Later I designed a smaller and internally shielded SMD version for a fixed 50R source impedance ....since my DAAS32 as well as the QuantAsylum and most other signal generators feature 50R output impedance anyway.

I still find it a very useful device, even though I mostly use the user filter option of the QA.
The QA uses post filtering only and doesn´t allow prefiltering its generator part.
This may generate overload problems or a low signal level depending on the topology and dimensioning of Your Phono stage.
That is where the inverseRIAA shines

jauu
Calvin

### Attachments

*(URL: /community/attachments/inv-riaa-smd_b-top-silk-pdf.1480401/)*

inv RIAA - SMD_B - Top Silk.pdf

36.6 KB · Views: 27

*(URL: /community/attachments/inv-riaa-smd_b-schematics-pdf.1480400/)*

inv RIAA - SMD_B - schematics.pdf

21.9 KB · Views: 29

*(URL: /community/attachments/invriaa-01-small-jpg.1480399/)*

invRIAA 01 small.jpg

68.5 KB · Views: 70

*(URL: /community/attachments/inv-riaa-neumann-pcb-jpg.1480398/)*

inv RIAA-Neumann PCB.jpg

49.3 KB · Views: 51

*(URL: /community/attachments/inv-riaa-neumann-brd-jpg.1480397/)*

inv RIAA-Neumann brd.jpg

81.6 KB · Views: 72

*(URL: /community/attachments/inverse-riaa-rev_b-complete-schem-png.1480396/)*

inverse RIAA rev_B complete - schem.png

43.5 KB · Views: 72

Drbulj and Hans Polak

M

**Midnightmayhem**
Member
Joined 2012

2025-07-06 4:54 pm #33

Is this project available for others to build?
I could use a nice reverse RIAA network.

Ala Electronics

**Ala Electronics**
Member
Joined 2025

2025-07-06 4:58 pm #34

> **Chris Hornbeck said:**
>
> I'm currently (and verrrrry lowly) in the process of building a ragtag prototype of the MvdG phono stage and a (better-than-my-1976-possibly-still 1dB ?) reverse RIAA lossy network, or better than is built into a Sound Technology 1200. Intent is to make a Lipshitz/Jung (883K) with 600 Ohm-ish R3. Can I assume (yes, I know) that I could reduce R3 arbitrarily without significant change to the important mid-band inflections?
>
> All good fortune,
> Chris

You could reduce R3... but should you?
While mid-band inflections are fairly robust, R3 (in the Lipshitz/Jung network) contributes to both impedance shaping and precise time constants. Shrinking it too much risks underdamping the curve or skewing the slope, especially if the generator's output impedance is non negligible.

Think of R3 not just as a resistor, but a trusted partner in crime.
Bend him too far, and he'll mess with your curve behind your back.

Build bravely, but measure often.
All good fortune back at you,
Alan

2025-07-06 5:07 pm #35

**Ala Electronics**
Member
Joined 2025

**Hans Polak said:**

Alan,

I simulated your circuit in LTSpice, see example below with 70K load on the MM output showing the deviation from an ideal A-Riaa curve.
What comes out is that Rsource should be <= 50R to stay within 0.1dB up to 20Khz.
However, load impedance only affects overall attenuation but not the accuracy to stay within 0.1dB.

Hans

Thanks Hans, that's a really helpful confirmation. I actually ran my own simulations in Multisim as part of finalizing the attenuation structure, and your LTSpice result lines up closely with what I got.

Here's the simulation side of the passive IRN project from my end:

Once the attenuation structure was finalized, I ran a full frequency response test in Multisim to check how closely the circuit tracks the theoretical inverse RIAA curve, from 20 Hz to 20 kHz. The target was ±0.1 dB, the result came in at ±0.11 dB deviation, consistent across both channels.

The simulated output at 1 kHz landed at –40.08 dB for the MM path. The MC output follows the same curve, just 23 dB lower, as both share the same core network with the split at the output.

Here's the MM output curve compared with the ideal inverse RIAA. Really happy with how it held up across the full audio band.

2025-07-06 5:11 pm #36

Ala Electronics
Member
Joined 2025

> **Midnightmayhem said:**
>
> Is this project available for others to build?
> I could use a nice reverse RIAA network.

Yes, it's already built, and I'm now sharing the project step-by-step here (design theory, schematic, simulation, build process, etc.).

Once I've posted the remaining parts, I'll also include full documentation, BOM, and Altium files in case you or others would like to build your own version.

2025-07-06 5:17 pm #37

**Ala Electronics**
Member
Joined 2025

> **Drbulj said:**
>
> Hi Alan,
> Very nice and neatly executed project, congrats!
> Since your are in precision, I can only drop a hint about components choice. For capacitors there are 1% polystyrene extended foil (extended foil means minimal inductance) caps, they are not in production for long time, but can be found, made by RIFA (PFE series) in 80's. Except they need to be found, they are also large
>
> **Click to expand...** *(URL: )*

Thanks for your input, Drazen, I appreciate your words and the excellent capacitor tip!

Those extended-foil polystyrene caps are truly special. For this build, I went with WIMA FKP2s and NISSEI polypropylene film types, mainly for their availability and consistent mechanical construction, but I'll definitely keep an eye out for the old RIFA PFE series, especially if I go for a future ultra-low-noise version.

And that amazing 0.04 dB result! Really impressive work. 👍

Drbulj

M

**Midnightmayhem**

Member

Joined 2012

2025-07-06 5:21 pm

#38

> **Ala Electronics said:**
>
> Yes, it's already built, and I'm now sharing the project step-by-step here (design theory, schematic, simulation, build process, etc.).
>
> Once I've posted the remaining parts, I'll also include full documentation, BOM, and Altium files in case you or others would like to build your own version.

Thank you!

Ala Electronics

2025-07-06 5:25 pm #39

**Ala Electronics**
Member
Joined 2025

> **Calvin said:**
>
> Hi,
>
> as others already pointed out ... the source impedance should be taken into account.
> I did exactly that in my own inverseRIAA .... switchable for source impedances of 0R, 50R, 550R and 600R ... as
> well as 1kHz reference levels of -40dB@600R and -60dB@60R (switch MM/MC) ... see attachments
>
> **Click to expand...** *(URL: )*

Thanks Calvin, really appreciate the detailed explanation and the photos, that's an impressive setup.

The switchable source impedance feature is a great idea, especially for testing how different signal generators and phono inputs behave under realistic conditions. I kept mine optimized around 50 Ω, as that's what many signal generators aim for, but your implementation is clearly much more versatile.

I haven't worked directly with QuantAsylum gear myself, but I completely agree with your point about the limitations of post-filtering. Without proper analog pre-filtering, it's easy to run into overload or insufficient drive, exactly why I decided to build this standalone unit.

Thanks again for sharing your approach, very inspiring.

Last edited: 2025-07-06 5:33 pm

Calvin

**Ala Electronics**
Member
Joined 2025

2025-07-06 5:47 pm #40

With the schematic and simulations wrapped up, here's a closer look at the physical build, both the electronic and mechanical sides of the project.

This filter uses separate PCBs for left and right channels (dual mono). While not strictly required, it significantly reduces crosstalk when testing stereo phono stages. Each board sits inside its own aluminum enclosure.
All parts were hand-soldered, and key RC values were selected for tight matching. Polypropylene film capacitors from WIMA and Nissei were used throughout, and all resistors are 1% metal film. To hit target values more precisely, some components were combined in series or parallel, this helped stay within the ±0.11 dB window observed in simulation (post #35).

The PCBs were designed in Altium Designer, with a compact layout and fully star-grounded topology. Signal routing was kept short and direct to minimize coupling and maintain channel separation.

Photos below show the boards at various stages — from raw assembly to final enclosure. While dual enclosures weren't strictly necessary, early measurements showed a clear reduction in inter-channel coupling using this approach.

arjen6t8, sonicles, Deenoo and 3 others

---

M
**Mark Tillotson**
Member
Joined 2018

2025-07-07 8:15 pm #41

I fear plastic ends to the case will let the hum in...

Perhaps add a ground terminal to the metal part of the case and ensure the two halves are in electrical contact (anodized finish is an insulator).

---

Z
**Zung**
Member
Joined 2005

2025-07-07 8:24 pm #42

Hum has no fear of aluminum 🙂

Drbulj and Ala Electronics

2025-07-07 10:29 pm #43

**Ala Electronics**
Member
Joined 2025

> **Mark Tillotson said:**
>
> I fear plastic ends to the case will let the hum in...
>
> Perhaps add a ground terminal to the metal part of the case and ensure the two halves are in electrical contact (anodized finish is an insulator).

Good catch, both halves of the enclosure are electrically bonded, and I added a ground terminal. I also removed the anodizing at contact points to ensure continuity.

2025-07-07 10:35 pm #44

**Ala Electronics**
Member
Joined 2025

> **Zung said:**
>
> Hum has no fear of aluminum 🙂

Turns out hum reads datasheets too!

Drbulj

2025-07-07 10:49 pm #45

rayma
Moderator
Joined 2011

**Chris Hornbeck said:**

> Do multiple samples in series/parallel taken from the (presumed) same production run actually average towards the bogie value, or do we still need "known" values and a way to measure them? Asking for a friend.

Only if they are a truly random sample, whatever that is, but most will never order very many new parts at one time.
Often there is a clear bias toward smaller uF values, though you may get one or two dead-on parts out of a group of 10-20.

Ala Electronics

Ala Electronics
Member
Joined 2025

2025-07-08 11:23 am #46

Since many thoughtful ideas and questions have come up in this thread, I've gathered the key materials for anyone who wants to take a closer look or replicate the build.

Here is the original Altium Designer project, which includes the schematic and PCB layout exactly as used to fabricate the boards. All layers are included and the output jobs are set up so you can directly generate Gerbers for production if needed. And also a 12-page PDF that documents the project in detail, covering circuit theory, attenuation structure, simulation results, layout photos, BOM, mechanical design, and final assembly.

The goal of sharing these is to make the design transparent and reusable, whether for direct use or as a starting point for your own variation. It's not a commercial project, just a personal tool developed for precise phono stage testing, but if it's useful to others, all the better.

Download links:

**Altium project – ready for PCB fabrication** *(URL: https://github.com/Ala-Vala/My-uploads/raw/refs/heads/main/Inverse%20RIAA%20Filter.pcbdoc)*

**Project documentation PDF – full design and build notes** *(URL: https://github.com/Ala-Vala/My-uploads/raw/refs/heads/main/Passive%20Inverse%20RIAA%20Filter.pdf)*

Last edited: 2025-07-08 11:43 am

Z

Zung
Member
Joined 2005

2025-07-08 12:06 pm #47

> **Ala Electronics said:**
>
> Turns out hum reads datasheets too!

I'm only half joking.
In spite of the full cast alu box, I have a hard time finding THE sweet spot on my bench where hum is not too intrusive.

Ala Electronics
Member
Joined 2025

2025-07-08 7:00 pm #48

> **Zung said:**
>
> In spite of the full cast alu box, I have a hard time finding THE sweet spot on my bench where hum is not too intrusive.

I get that, sometimes it feels like hum has a mind of its own. Glad I'm not the only one playing 'find the hum-free zone' 👍

M

Monstercore
Member
Joined 2024

2025-07-08 7:25 pm #49

Good to have a reverse riaa.
I use this one for decades.

Never used enclosure, no humm.
Selected parts by hand from big batch.
Build several phonostages with good respons. See measurements below.

My generator has 50 ohm output impedance.

Recap, good to have your own reverse RIAA.
This one is build by hundreds DIY's.
Cost just a view euro,s.

Intellectuals solve problems, geniuses prevent them.

Ala Electronics

rayma
Moderator
Joined 2011

2025-07-08 7:25 pm #50

Sit the unit in back or on top of the preamp, and use very short cables between them.

Ala Electronics and Monstercore

M

**MarcelvdG**
Member
Joined 2003

2025-07-08 8:31 pm #51

Regarding hum, aluminium or copper enclosures don't shield low-frequency magnetic fields well unless they are much thicker than the skin depth, which is of the order of 1 centimetre in copper at the usual mains frequencies. They do shield low-frequency electric fields well. To reduce the sensitivity to low-frequency magnetic fields, you can try to keep loop areas as small as possible, particularly the loop from resistors 13...20 in post #4 to the output connector and back.

Ala Electronics

**Conrad Hoffman**
Member
Joined 2007

2025-07-08 9:29 pm #52

There's accurate and there's pretty. Some people have time to build both but I just needed to git 'er done.

gpapag, Ala Electronics and Monstercore

**Ala Electronics**
Member
Joined 2025

2025-07-08 11:39 pm #53

> **Monstercore said:**
>
> Good to have a reverse riaa.
> I use this one for decades.
> **View attachment 1481102** *(URL: https://www.diyaudio.com/community/attachments/1481102/)*
> **View attachment 1481103** *(URL: https://www.diyaudio.com/community/attachments/1481103/)*
>
> **Click to expand...** *(URL: )*

Great to see your version. Selecting parts by hand from a large batch really shows in the response. That's a true DIY spirit right there. Also the variety of approaches is what makes DIY so interesting.

Thanks for sharing your build and results 🙏

**Ala Electronics**
Member
Joined 2025

2025-07-08 11:51 pm #54

> **MarcelvdG said:**
>
> Regarding hum, aluminium or copper enclosures don't shield low-frequency magnetic fields well unless they are much thicker than the skin depth, which is of the order of 1 centimetre in copper at the usual mains frequencies. They do shield low-frequency electric fields well. To reduce the sensitivity to low-frequency magnetic fields, you can try to keep loop areas as small as possible, particularly the loop from resistors 13...20 in post #4 to the output connector and back.

It's interesting how surprisingly deep the skin depth is at 50/60 Hz.
I tried to keep return paths tight and loop areas minimal, especially around R13–R20 and the output jacks. I'll revisit that area again to double-check for possible improvements.
Appreciate the practical reminder👍

**Calvin**
Member
Joined 2004

2025-07-09 6:41 am #55

Hi,

interestingly regarding hum I typically can see large hum peaks when measuring with the QA401.
So far nothing helped, no moving positions, no switching lights of, no different connections, resp. cables etc.
The only option I see left seems a powered USB cable for the QA401, though nothing changed when I supplied it through an laptop instead of the PC.
I don´t see those big peaks with he DAAS32.
Connected in a listening setup its immideately obvious, that the phono-stages are not the source of the hum.
So if You measure through the inverseRIAA take the resullts with a grain of salt.
If large hum artefacts show up, it might not necessarily be audible hum, but just inverseRIAA related.

jauu
Calvin

Ala Electronics and Midnightmayhem

2025-07-09 7:19 am #56

Ala Electronics
Member
Joined 2025

> **Conrad Hoffman said:**
>
> There's accurate and there's pretty. Some people have time to build both but I just needed to git 'er done.
> **View attachment 1481151** *(URL: https://www.diyaudio.com/community/attachments/1481151/)*
>
> **View attachment 1481152** *(URL: https://www.diyaudio.com/community/attachments/1481152/)*

Sometimes just getting it finished matters more than anything, especially when the circuit works as it should. Your build definitely gets the job done.

Conrad Hoffman

Ala Electronics
Member
Joined 2025

2025-07-09 7:20 pm #57

> **Calvin said:**
>
> Hi,
>
> interestingly regarding hum I typically can see large hum peaks when measuring with the QA401.
> So far nothing helped, no moving positions, no switching lights of, no different connections, resp. cables etc.
> The only option I see left seems a powered USB cable for the QA401, though nothing changed when I supplied it
>
> **Click to expand...** *(URL: )*

Personally, I've found that inverse RIAA filters are best reserved for checking frequency response and crosstalk. Once we start analyzing noise, THD, or intermodulation, things can get unreliable pretty quickly, too many variables introduced by passive loading, grounding, and even loop area effects.

I see the IRN more as a tool for linearity tests than full spectrum analysis. Still incredibly useful, just within its sweet spot.

GKTAUDIO

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Z
Zoran
Member
Joined 2004

2025-07-09 7:54 pm #58

Hi nice buildings.
I am using Jim Hagerman inverse riaa net for measurements.
**http://www.hagtech.com/pdf/iriaa2.pdf** *(URL: http://www.hagtech.com/pdf/iriaa2.pdf)*
**https://cdn.shopify.com/s/files/1/0433/2441/files/riaa.pdf** *(URL: https://cdn.shopify.com/s/files/1/0433/2441/files/riaa.pdf)*
cheers

Ala Electronics

2025-07-10 3:05 pm #59

**Ala Electronics**
Member
Joined 2025

Here’s a quick follow-up on how I ended up testing the passive inverse RIAA filter using RMAA, not a perfect tool for low-level measurement, but workable with a bit of creativity.

Since RMAA requires relatively high signal levels, I modified one of my own line preamps to achieve around +50 dB of gain (by changing the feedback resistor from 3k to 300k). This allowed the –40 dB IRN output to reach a level usable for frequency response analysis. The input impedance of the line preamp is 50k.

I placed the preamp alone in the RMAA test chain to log its native frequency response and offset behavior. This gave me a baseline to subtract from the combined IRN+preamp measurement later. Then I connected:
→ Audio interface output → IRN input
→ IRN output → high-gain preamp → audio interface input

For cross-validation, I simulated the same IRN circuit in Multisim with a 50-ohm source impedance and extracted the same frequency points.

The results were surprisingly tight, differences between the simulation and RMAA-corrected values were typically under ±0.06 dB across the 20 Hz to 20 kHz band. Both units exhibited identical frequency response in measurements, which aligns with their identical layout and matched components.

While the simulation assumed a 50-ohm source, the real test used a Xonar D2 interface, whose output impedance is estimated to be around 100–150 ohms. This mismatch likely accounts for the tiny deviations, especially at higher frequencies where source resistance slightly shifts the pole locations.

Attached below is the measurement table along with screenshots and a video from the test setup and simulation.

The frequency response of the flat preamp:

The frequency response of the IRN filter + flat preamp:

Last edited: 2025-07-10 3:15 pm

Conrad Hoffman

2025-07-11 8:08 pm #60

**Ala Electronics**
Member
Joined 2025

Quick update:

I connected the IRN filter to my PS-101 phono preamp and ran a frequency response test with RMAA. The result was a clean, flat curve, just as expected. Always good to confirm that the real-world performance matches the design and simulation.

Captured a quick video of the RMAA screen during the test, attaching it here in case it’s useful for others working on similar setups.

Appreciate all the input so far.

H

Hans Polak

Member

Joined 2005

2025-07-11 10:11 pm #61

> **Ala Electronics said:**
>
> Since RMAA requires relatively high signal levels,

Rmaa can cope with any levels, however your soundcard is the limiting factor.

Hans

Ala Electronics

2025-07-12 7:09 am #62

**Ala Electronics**
Member
Joined 2025

> **Hans Polak said:**
>
> Rmaa can cope with any levels, however your soundcard is the limiting factor.

You're absolutely right.

By mentioning RMAA requiring relatively high levels, I was referring to the practical limitations of the soundcard. In real-world use, it depends entirely on the card's input handling and SNR. In my case, I had to boost the signal upstream to avoid noise and get a clean response.

Alan

**Ala Electronics**
Member
Joined 2025

2025-07-12 5:37 pm #63

> **Zoran said:**
>
> I am using Jim Hagerman inverse riaa net for measurements.

I've come across Hagerman's IRIAA circuit before. That extra HF roll-off near 50 kHz is a smart nod to real-world cutting head behavior.

That said, I intentionally chose to leave that out in my version. My focus was on building a transparent, strictly passive inverse RIAA filter for precision testing in the audible band (20 Hz – 20 kHz), especially for checking frequency response flatness and stereo crosstalk.

The added ultrasonic pole makes sense when trying to fully mimic LP cutting characteristics, as discussed in the "Cut and Thrust" article in Stereophile, and other papers. But for my goals, mainly focused on passive, and verifiable behavior within the standard audio range, I preferred to keep the topology minimal and direct.

Still, I've been tempted to try a variant with that extra roll-off just to compare how much it impacts practical measurements, especially when transformers or high-bandwidth stages are involved.

M

**MarcelvdG**
Member
Joined 2003

2025-07-12 5:54 pm #64

According to Wikipedia, that 50 kHz roll-off is nonsense. Of course the recording RIAA correction cannot keep increasing indefinitely, but there is no reason why the roll-off with respect to the ideal RIAA curve would have to be first order.

**https://en.m.wikipedia.org/wiki/RIAA_equalization#The_Mythical_"Neumann_pole"** *(URL: https://en.m.wikipedia.org/wiki/RIAA_equalization#The_Mythical_%22Neumann_pole%22)*

ejp and Ala Electronics

2025-07-12 9:03 pm #65

**Ala Electronics**
Member
Joined 2025

For anyone curious to see the full project take shape, I've just uploaded a short video walk-through of the passive inverse RIAA filter to YouTube.

The video offers a quick but complete look at the design process, covering the goals, PCB layout, component selection, assembly stages, and final aluminum enclosures. It's about 2 minutes long and narrated entirely via subtitles, no voice-over.

Watch here:

This build has been all about precision, transparency, and sharing, both the technical details and the hands-on process that photos alone don't fully capture.

Really appreciated the ideas, feedback, and comparisons shared throughout the thread. It's always inspiring to see how each person approaches these challenges in their own way.

Cheers,
Alan

Last edited: 2025-07-12 9:13 pm

E

**ejp**

Member

Joined 2007

2025-07-13 1:29 am #66

> **Ala Electronics said:**
>
> The added ultrasonic pole

The so-called 'added ultrasonic pole' originated with Allen Wright, who graphed not a pole but a zero, although he doesn't use either term. That will give you some idea of how much nonsense has accreted around this myth. His central claim about the Neumann cutter manual was debunked by Doug Self years ago by reference to that manual, which he states gives not a zero as claimed but a 3rd-order Butterworth pole, at a different frequency. As **@MarcelvdG** *(URL: https://www.diyaudio.com/community/members/4072/)* says, why anyone would need to compensate for this even if real is another mystery.

Ala Electronics

Ala Electronics
Member
Joined 2025

2025-07-13 7:38 am #67

> **ejp said:**
>
> His central claim about the Neumann cutter manual was debunked by Doug Self years ago by reference to that manual, which he states gives not a zero as claimed but a 3rd-order Butterworth pole, at a different frequency.

Even if such a pole existed in certain cutter amp designs, it's unclear whether it was ever standardized or consistently applied across different mastering chains. As Doug Self points out in his analysis of the Neumann documentation, the actual implementation was a 3rd-order Butterworth low-pass, not a simple pole, and not necessarily part of the standard RIAA spec. Phono preamps are rarely judged beyond 20 kHz, and most cartridges roll off well around that.

The Lipshitz/Jung network remains a rigorously documented and repeatable approach, grounded in known references and measurable performance within the audio band.

ejp

M
MarcelvdG
Member
Joined 2003

2025-07-13 1:42 pm #68

It wasn't standardized as far as I know. When you allow active filter circuitry, it's very easy to make a response that is equivalent to the ideal RIAA recording curve cascaded with any odd-order low-pass response you like. Just use the roll-off of the RIAA recording curve network as the first-order section and add the appropriate second-order sections. For third-order Butterworth at 50 kHz, that would be one second-order section with $f_n$ = 50 kHz, $Q$ = 1.

A first-order roll-off from 50 kHz would result in a -0.6446 dB error at 20 kHz.

A third-order Butterworth low-pass at 50 kHz would result in a -0.0178 dB error at 20 kHz.

inyashd, ejp and Ala Electronics

E

ejp

Member

Joined 2007

2025-07-13 10:17 pm

New #69

Well yes but we're talking about what happens at the cutter. There is an HP slope in effect from the 75 uS time constant, so one of Doug's three poles cancels that, and the other two provide a second order roll off.

To undo all that in replay would also require three poles, HP of course. Not just one 'added ultrasonic pole'. All that could do would be to match Wright's imaginary zero.

Why any of this was ever thought to be a thing they needed undoing beats me. It seems to me that Wright has read Lipshitz but not really understood it.

Ala Electronics

M

MarcelvdG

Member

Joined 2003

2025-07-13 10:47 pm

New #70

My point is that assuming that the cutter (from whatever brand it may be) has been designed by people who have (or had) some basic understanding of filter theory, it would be very easy for them to limit the bandwidth in a much more effective way than with just one extra pole, with far less far ultrasonics and with negligible effect on the magnitude response below 20 kHz. That alone already makes the "Neumann pole" story unlikely.

> **ejp said:**
>
> To undo all that in replay would also require three poles, HP of course. Not just one 'added ultrasonic pole'. All that could do would be to match Wright's imaginary zero.

Three zeros you mean, three zeros that cover the three added Butterworth poles? Then again, filter experts often look at the attenuation instead of the transfer, the transfer poles then become attenuation zeros and the other way around. In any case, with an error of -0.0178 dB, there is not much practical need to correct for it.

Ala Electronics

E

ejp

Member

Joined 2007

2025-07-14 4:02 am

New #71

I agree entirely. Who on earth would put in a mere zero when what is clearly needed is a high-order lowpass filter? And who on earth would invent the necessity to compensate for it?

Last edited: 2025-07-14 4:21 am

Ala Electronics

Ala Electronics
Member
Joined 2025

2025-07-14 7:02 am New #72

> **MarcelvdG said:**
>
> In any case, with an error of -0.0178 dB, there is not much practical need to correct for it.

While some designs include an HF correction pole around 66.4 kHz to compensate for excess gain near the upper audio band, it's important to clarify the context.

Doug Self explains that such a correction is necessary only in active RIAA implementations, particularly when the gain of the first stage is deliberately kept low (e.g., ~+30 dB) to maximize overload margin. In those cases, the natural 6 dB/octave roll-off of the RIAA response doesn't continue far enough into the ultrasonic region, leading to a measurable +0.38 dB error at 20 kHz due to the zero at ~66.4 kHz.

To correct this, an additional low-pass pole (usually placed after the first stage) brings the frequency response back in line with the ideal RIAA curve.

"...but only required when active gain stages alter the intended RIAA response, especially at high frequencies due to reduced stage gain."

In contrast, passive RIAA networks — such as the classic Lipshitz/Jung topology — don't require this correction, since the attenuation profile is fixed by the passive network itself, and the gain stages that follow don't interfere with the curve shape.

That's why, for passive inverse RIAA filters like the one discussed here, no HF correction pole is needed or appropriate. The goal remains a clean and accurate response within 20 Hz–20 kHz, without introducing unnecessary ultrasonic shaping that was never standardized.

Drbulj

---

M

MarcelvdG
Member
Joined 2003

2025-07-14 8:31 am New #73

That's a different matter altogether. It's a response error in the RIAA preamplifier itself due to the + 1 term in the gain equation of a non-inverting amplifier.

ejp and Ala Electronics

**Ala Electronics**
Member
Joined 2025

2025-07-14 10:34 am New #74

> **MarcelvdG said:**
>
> It's a response error in the RIAA preamplifier itself due to the + 1 term in the gain equation of a non-inverting amplifier.

You're absolutely right to point out that the +1 term in the non-inverting amplifier gain equation is a key factor in the HF deviation, especially when stage gain is kept low. That's indeed what Doug Self emphasizes in Small Signal Audio Design, and it's an important distinction to make.

My earlier mention of the HF correction pole was more in the context of how that deviation manifests in practice within active RIAA circuits, particularly when trying to preserve headroom by limiting first-stage gain. Appreciate the refinement.

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**Chris Hornbeck**
Member
Joined 2011

2025-07-15 5:07 am New #75

It might be possible to define a model of a dimentionless cutting stylus and a similarly small playback stylus, but 12" vinyl at 33 rpm has its own limitations (of course). I only mean to say that dimentions dominate in phono playback. And, PV=RT.

All good fortune,
Chris

Ala Electronics

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E
**ejp**
Member
Joined 2007

2025-07-15 6:01 am New #76

The matter of the mythical 'Neumann pole' is rather thoroughly debunked with schematic extracts and Bode plots **here** *(URL: https://pspatialaudio.com/neumann_pole.htm)*.

arjen6t8, Ala Electronics and MarcelvdG

---

A
**arjen6t8**
Member
Joined 2024

2025-07-16 11:20 am New #77

I really like that kind of article. I have been reading this thread and thinking there must be documentation on these cutting lathes and here it is.

Ala Electronics