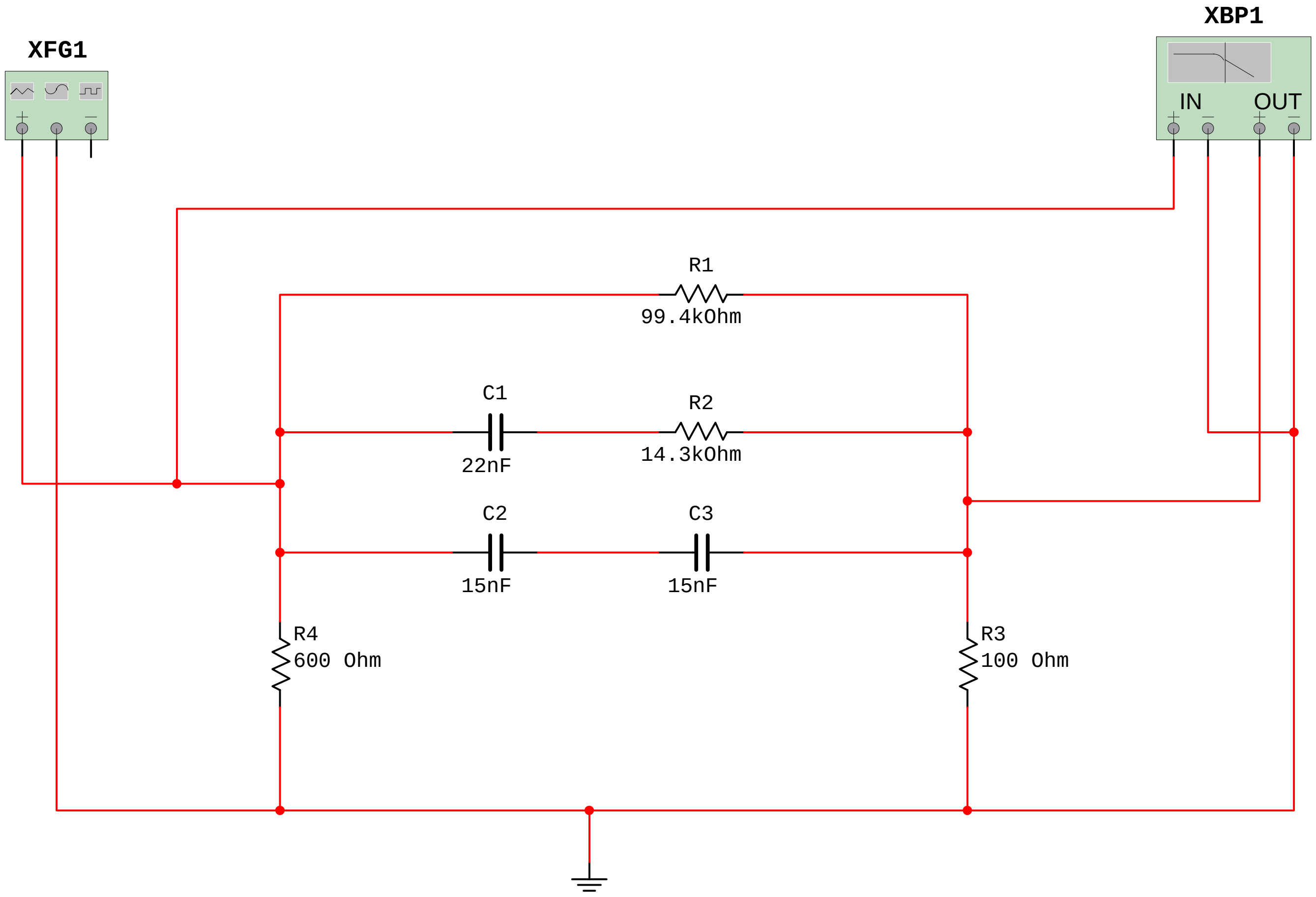
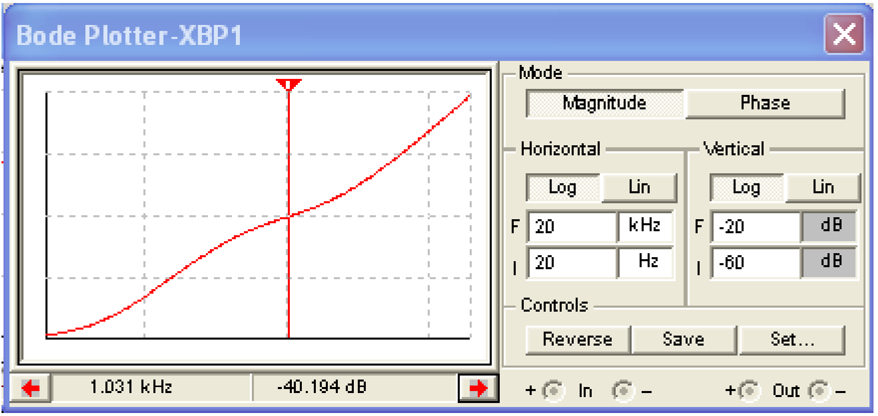
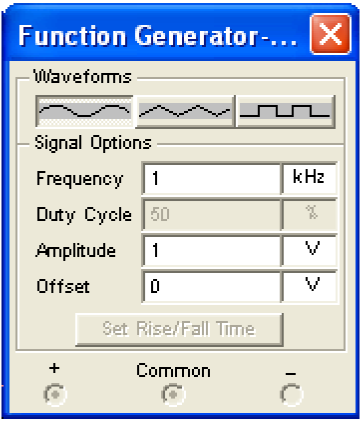
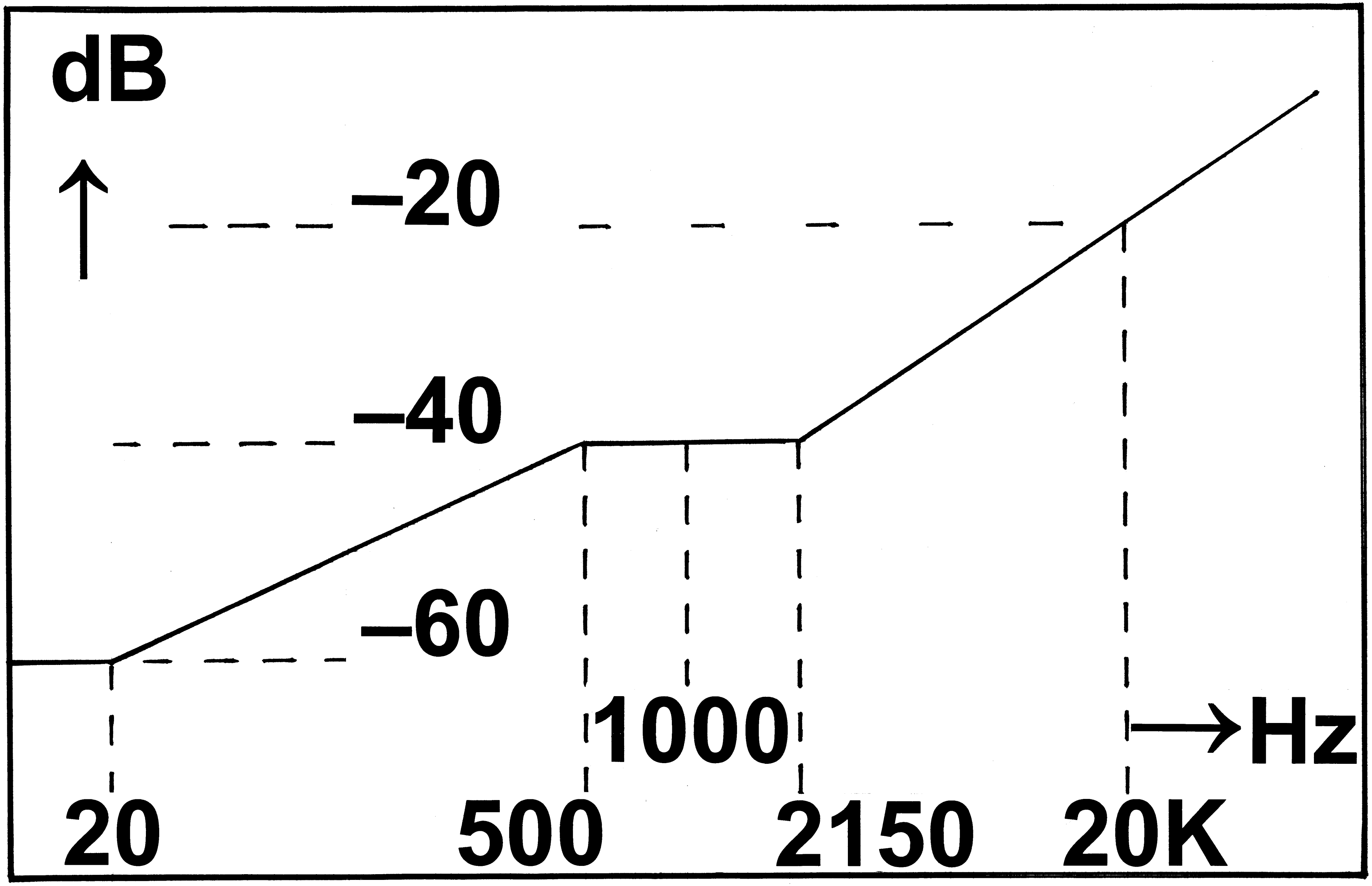


XFG1
XBP1
IN
OUT
R1
99.4kOhm
C1
22nF
R2
14.3kOhm
C2
15nF
C3
15nF
R4
600 Ohm
R3
100 Ohm


Bode Plotter-XBP1
Mode
Magnitude
Phase
Horizontal
Log
Lin
F 20 kHz
I 20 Hz
Vertical
Log
Lin
F -20 dB
I -60 dB
Controls
Reverse
Save
Set...
1.031 kHz
-40.194 dB
+ In –
+ Out –


Function Generator-...
Waveforms
Signal Options
Frequency 1 kHz
Duty Cycle 50 %
Amplitude 1 V
Offset 0 V
Set Rise/Fall Time
+
Common
–


CALIF.


dB
–20
–40
–60
20
500
1000
2150
20K
Hz

100 Ω OUTPUT

REVERSE
RIAA
DUMMY
SIGNAL LEVEL
1:100

600 Ω INPUT

7 auxiliary-adapter voor MD-ingang

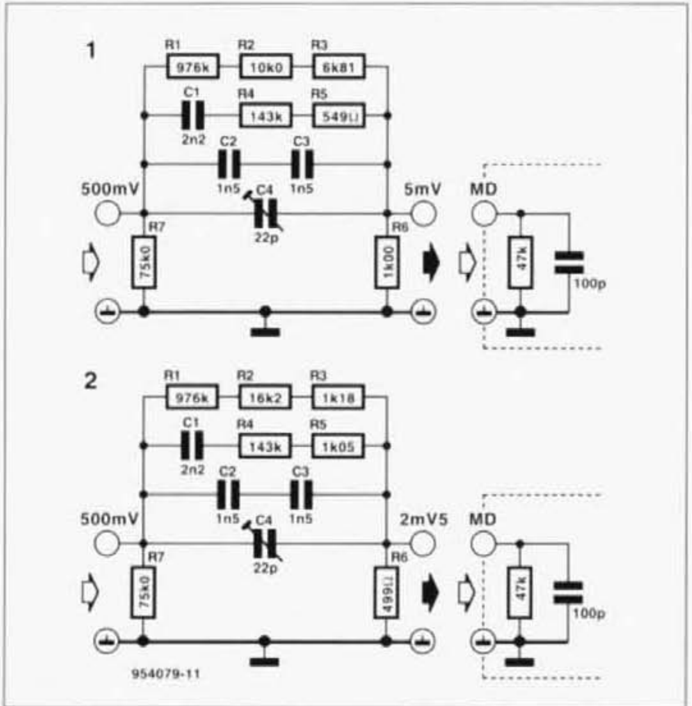
De meeste audio-systemen zijn nog steeds standaard voorzien van een of twee MD-ingangen voor een platenspeler. Gezien het feit dat platenspeler steeds minder gebruikt worden, blijft zo'n ingang ook steeds vaker ongebruikt. Gelijktijdig is er bij veel installaties een groot gebrek aan lijn-ingangen. Het kan dus voorkomen dat ondanks een tekort aan lijn-ingangen de MD-ingangen op de versterker ongebruikt blijven.

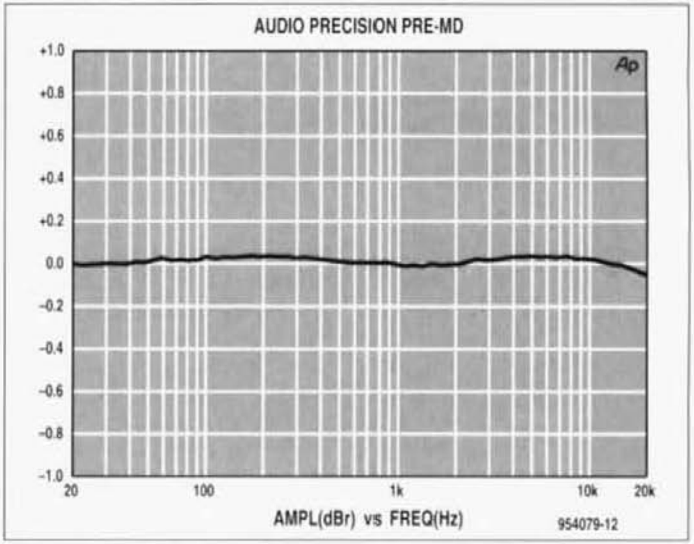
De schakeling die we hier voorstellen, maakt het mogelijk een ongebruikte MD-ingang alsnog te gebruiken als gewone lijningang. Een simpel passief netwerkje zorgt voor de noodzakelijke signaalaanpassing. In het schema worden twee voorbeelden gegeven, waarbij het tweede netwerkje het signaal meer verzwakt dan het eerste. Afhankelijk van de gevoeligheid van de MD-ingang (5 of 2,5 mV bij 1 kHz) kan met behulp van een van de twee omzetters een MD-ingang omgebouwd worden tot een lijn-ingang met een gevoeligheid van 500 mV.

De schakeling is met een extreem hoge precisie berekend, vandaar dat weerstanden met een tolerantie van 0,1% gebruikt worden. De kondensatoren dienen handmatig geselekteerd te worden via een nauwkeurige kapaciteitsmeter. De aldus verkregen schakeling is bijzonder nauwkeurig, zelfs zo nauwkeurig dat het zeker is dat het korrektienetwerk in de MD-versterker veel grotere afwijkingen heeft. De schakeling is dan ook bij uitstek geschikt om de MD-voorversterker door te meten en te testen op afwijkingen in de RIAA-korrektie.

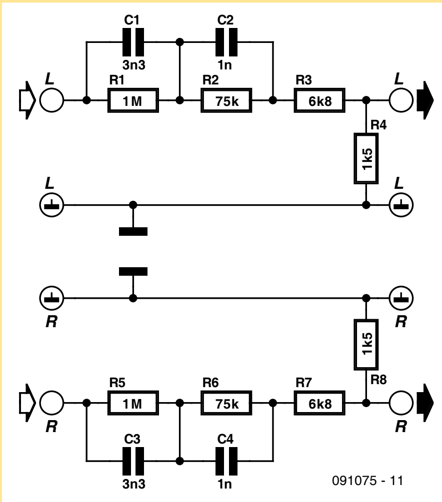
Ook als gewone komponenten met een tolerantie van 1% gebruikt worden, blijkt de schakeling het goed te doen. De grafiek laat de resultaten zien van de meting aan een omzetter die met standaard-komponenten uit het bakje is opgebouwd. De afwijkingen zijn minder dan ±0,05 dB. De theoretische afwijking van schakeling 1 is bij 20 kHz -0,05 dB, bij schakeling 2 is dat -0,012 dB.

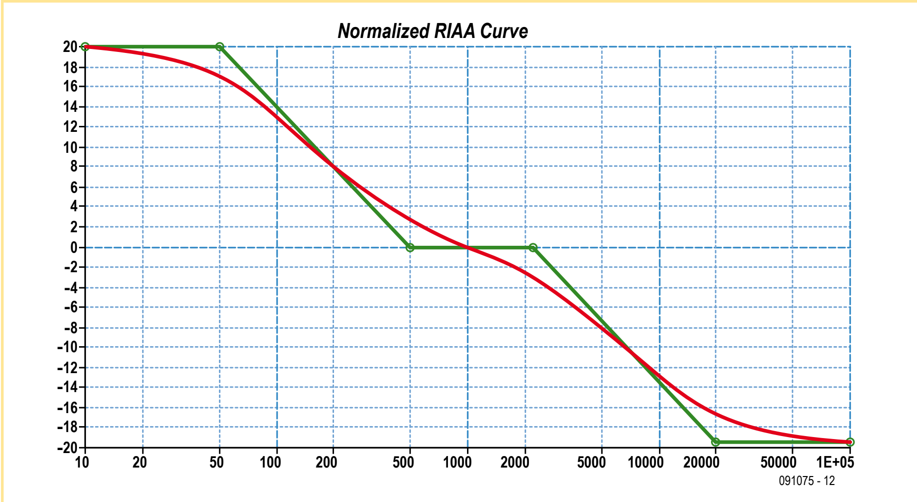
(954079)





Reverse RIAA Adaptor





Christian Tavernier (France)

If you're short of inputs on your amplifier, but it has an input for a magnetic pickup with RIAA correction, this very simple project will let you convert this into a high-level linear input, making it compatible with the outputs from all current audio sources. It won't have quite such perfect quality as a real line input, for two reasons.

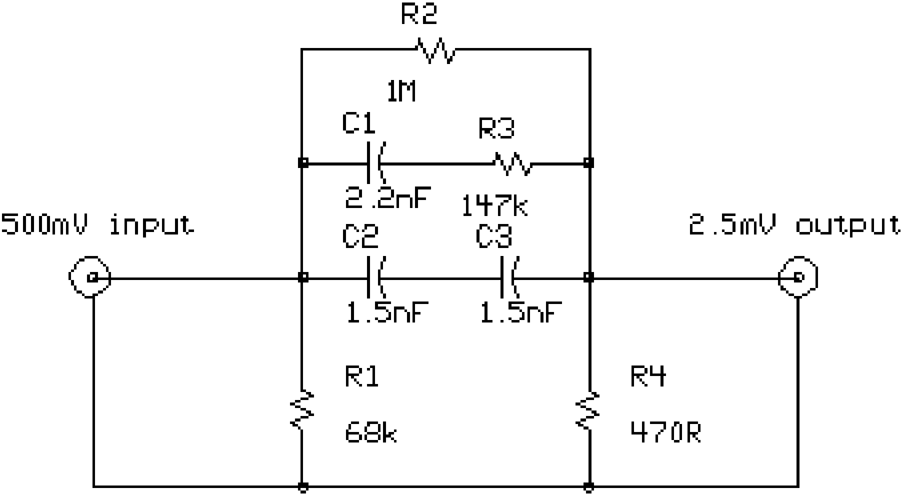
Firstly, our circuit is bound to introduce a slight reduction in the signal-to-noise ratio (SNR), as it attenuates a high-level signal and then amplifies it back up again. Secondly, minor linearity 'hiccups' are inevitable, as the correction it produces is not precisely the reverse of the RIAA correction applied by the preamplifier — but it is still perfectly acceptable, especially if it is only for playing MP3 signals!

Our circuit diagram is extremely simple, as it's just a simple passive filter whose components have been calculated to reproduce the inverse RIAA curve to that in the preamplifier, i.e. the same as that used when cutting discs. It's perfectly simple to build, but to avoid degrading the signal-to-noise ratio too much, we recommend using metal film resistors, which are less noisy than their carbon counterparts.

What's more, since the preamplifier magnetic pick-up input applies a great deal of bass amplification, because of the RIAA equalization, the circuit is extremely sensitive to induced interference, especially from AC powerlines, and so it will need to be very well screened. We built it 'in the air' and fitted it into a salvaged metal tube (a medicine container) which acts as both case and screen.

Given the components used, and although it does of course depend somewhat on the sensitivity of the magnetic pick-up input of the amplifier with which it is used, signals with an amplitude of 200–600 mV_{rms} can be applied to this circuit without fear of overloading the preamplifier.

(091075-I)



Line signal to phono input (MD/MC) converter ePanorama.net

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| Index | Schematics | Acronyms | Discussion | Mail to a friend | Post a message |

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Connect line-level signal to phono input

Copyright Tomi Engdahl 1997,1998

Circuit description

Introduction

Many amplifiers have phono inputs for connecting record players to the amplifier. Phono input is designed to take a up to few millivolt signal from phono pickup and amplify it. The amplifier stage does also some equalization based on standardized RIAA curve. The RIAA reproduction curve:

That RIAA equalization is used in the playback to reduce the high pitch noise and maximize bass dynamics in the phono playback. The audio material which is rocorded to the record has been pre-equalized so that the frequency response of the whole chain from the mixing desk to your speaker will give flat frequency response.

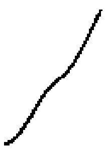
Nowadays phono inputs are largely unused because vinyl record players are getting rare. This circuit is is a simple converter to convert line level signal (0..1 V) to phono input levels (0..5 mV), which makes it possible to use those inputs as an extra line input. This circuit does the level conversion, impedance matching and inverse RIAA filtering. The inverse-RIAA filtering network in this circuit is needed, because all phono inputs have RIAA filter in them.

Circuit specifications

- Input impedance: > 10 kohm
- Output impedance 470 ohms
- Nominal input signal level: 500 mV
- Nominal output signal level: 2.5 mV
- Attenuation: 46 dB
- Frequency response: inverse-RIAA
- Phono input compatibility: moving magnet and high output moving coil type inputs (= compatible with most receivers and integrated amplifiers)

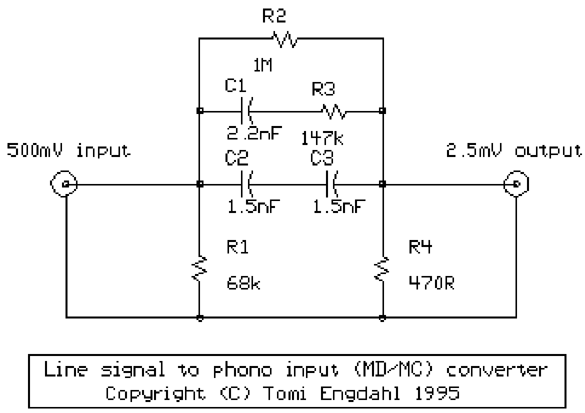
Circuit operation

The circuit does two functions: signal level attenuation and inverse RIAA filtering. The signal attenuation is needed to convert the 500 mV signal to 2.5 mV signal. The inverse-RIAA filtering is needed to make the frequency response of the system flat (same equalization that is used when music is transferred to vinyl in the studio). The picture blow shows the frequency response of the ideal inverse-RIAA filter:



Circuit diagram

The picture below shows the circuit diagram of one channel. The other channel (for stereo) is identical. The circuit is so simple that you can easily build it by just soldering the components together and fit them to small metal box with the RCA audio connectors.



This circuit is a simplified version of circuit published in Elektor Electronics (T.Giesberts, Pick-up input becomes line input, Elektor Electronics, December 1995, page 99). The basic circuit idea is same, but my circuit is simpler and it does not use high-precision components. The performance is not the same, but I think that this circuit build even from 5% components is adequate for many purposes.

Component list

```
C1    2.2 nF
C2    1.5 nF
C3    1.5 nF
R1     68 kohm
R2      1 Mohm
R3    147 kohm
R4    470 ohm
and two RCA connectors
```

This part list is for the one channels described in the schematic. Usually you want stereo sound, in which case you have make two identical circuits: one for left channel and one for right channel.

Using the circuit

When you connect this circuit to your amplifier, set the phono input to MM or MD position (if it has a selector). Keep in mind that phono input is much more prone to pick up interference, because signal levels in it are much lower than in line level inputs.

General information on phono input

When choosing a cartridge for a given phono section, it is the "gain" or output that have to match.

- Typically almost all receivers, and most integrateds have standard 'low-gain' phono preamplifying sections (30dB to 38dB of gain typically). These require the use of either moving magnet cartridges (typically in the 2.0 to 5.0mv range) or high output moving coil types (generally 1.5 to 2.0mV output).
- The better preamplifiers (and the odd integrated or receiver) accept low-output moving coil cartridges - which have outputs of .2 to .9mV. These high gain phono sections will typically have approx 55dB to 75dB of gain.
- Very old tube amplifiers (50 years old) have had CERAMIC cartridge input which is very high input and accepts around 100mV signal level.

Different cartridge types:

- MM (Moving Magnet) cartridges are designed to play into 47k ohms or higher input impedance. MM's will typically be loaded with capacitive loading in the pF range. Moving magnet phono cartridges have a typical output of ~3mv at 47K Ohm load. Movin gmagnet cartriges are the ones most commonly used.
- MC (Moving Coil) cartridges are typically designed to be loaded anywhere from 10 ohms to 1000 ohms.
- Ceramic catridges are high impedance signal sources. Their are not used nowadays because they were not "hifi quality".

Note to users of very old equipments

Some very old (typically TUBE) equipments had phono input designed for high impedance and high output level CERAMIC cartridge. These inputs were common 50 years ago, and disappeared with the introduction of stereo and transistors.

These inputs have no RIAA EQ (ceramic pickups are amplitude-sensitive, and the RIAA curve produces (very) roughly a constant-amplitude cut). A level of about 0.1 volt should be fine for those.

If you happen to have an equipment with CERAMIC cartridge input only, then you can't use the circuit described above. You should be able to do the conversion with much simpler circuit, just a 1:10 signal level attenuator, for example something like my speaker level to line level converter.

Tomi Engdahl <then@snakemail.hut.fi>

On Reference RIAA Networks

by Jim Hagerman

You'd think there would be nothing left to say. Everything you need to know about RIAA networks has already been published. However, a few years back I came across an interesting chapter in a vacuum tube book[1] which spoke of a mythical 3.18µs corner in the RIAA response. Huh? That's 50kHz. The seminal works[2][3] never mentioned this, equalization was to appear as Figure 1.

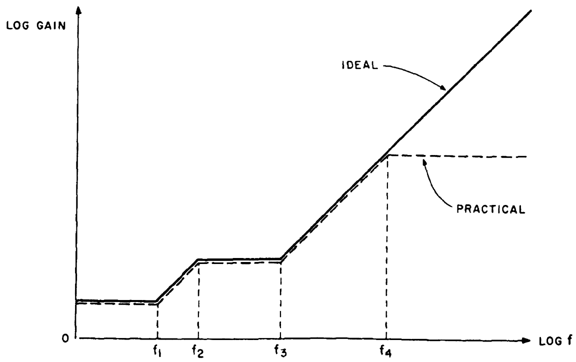


Figure 1. Desired inverse RIAA response from Lipshitz.

Hmmm. The mythical corner frequency is shown as f_4 but ideally should be missing. Or should it? To quote from Allen Wright's book[1]:

> "... look back at the graph of the recording EQ, they cut the LF and boost the HF. But do you really think they continue boosting to way out past whatever? Of course not, they'd burn out their cutter heads or something even more expensive ... This new 3dB point, according to a *Neumann* cutting amp manual, is set at 3.18µs – which equates to 50,048Hz ..."[1]

Not only does the cutting head response have a pole at f_4, but it must also have one at f_5! No amplifier has gain out to infinity. Hopefully, all cutting head manufacturers chose the same 3.18µs corner for limiting gain. Where is all this leading? It means that the RIAA equalization networks in our phono preamplifiers should have a zero at 3.18µs, putting back some gain before finally rolling off at higher frequencies. The legacy reference network[2] shown in Figure 2 has an f_4 pole at 337kHz – too high of frequency.

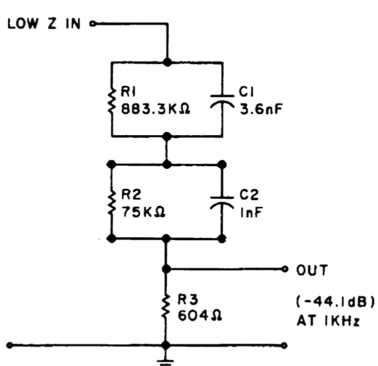


Figure 2. Legacy reference inverse RIAA network from Lipshitz.

Reference Inverse RIAA

What we need is a new modified RIAA reference curve to help us properly design phono equipment. I like using SPICE to simulate filter circuits and decided this would be a good way to generate a new standard. The generic filter section shown in Figure 3 is a simple lag-lead type with a zero and pole in the right places.

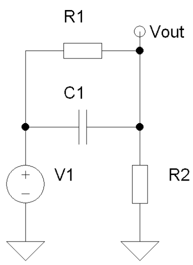


Figure 3. Lag-lead filter section.

Its voltage transfer function is given by

$$H(s) = \frac{s + \dfrac{1}{R_1 C_1}}{s + \dfrac{R_1 + R_2}{R_1 R_2 C_1}}$$

with zero and pole time constants determined by

$$t_z = R_1 C_1$$

$$t_p = \left(\frac{R_1 R_2}{R_1 + R_2} \right) C_1$$

The lower RIAA zero-pole pair is at 3180µs and 318µs. Using an arbitrary capacitance value of 1µF, the resistances are calculated as

$$R_1 = \frac{t_z}{C_1} = \frac{3180ms}{1mF} = 3.18k$$

$$R_2 = \frac{R_1 t_p}{R_1 C_1 - t_p} = \frac{(3.18k)(318ms)}{(3.18k)(1mF) - 318ms} = 353.3$$

The values for a second section (zero at 75µs and pole at 3.18µs) are 7.5k and 353.3 ohms respectively using a 10nF capacitor. The final circuit is shown in Figure 4. Note, I used a voltage controlled voltage source (E1) to decouple the responses of the two sections, otherwise the input impedance of the second section would load the first section and alter response.

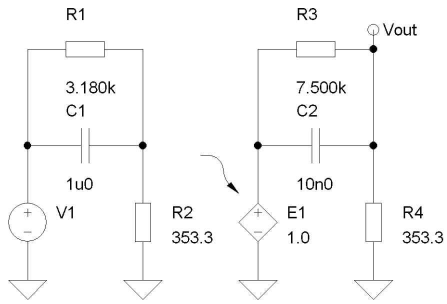


Figure 4. SPICE schematic for generating reference inverse RIAA curve.

Listing 1 is the input text file to my SPICE simulator. Figure 5 shows the resulting frequency response of the circuit, which is also given in tabular form in Listing 2. I offset the data so that gain at 1kHz would be 0dB. I find it helpful to sweep a wide frequency range of 1Hz to 1MHz as it gives a better view of the total response and what occurs outside of the 20 to 20,000Hz "audio band".

PSPICE Input

```
Inverse RIAA Curve

Vin 1 0 ac 1

R1 1 2 3.18k
R2 2 0 353.33
C1 1 2 1u

E1 3 0 2 0 1.0
R3 3 4 7.5k
R4 4 0 353.33
C2 3 4 10n

.ac dec 20 1 1000k
.print vm(4)
.probe
.end
```

Listing 1. SPICE input file for generating modified reference curve.

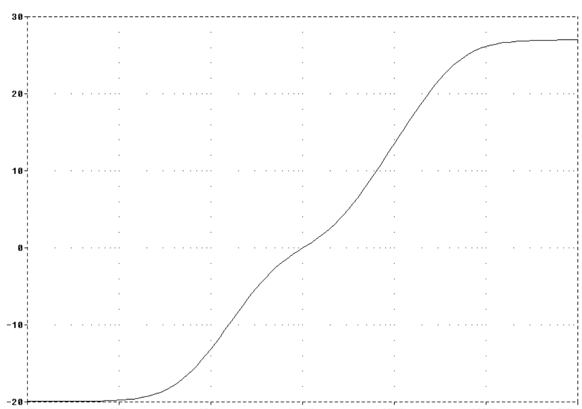


Figure 5. SPICE output of reference inverse RIAA curve with 1kHz set to 0dB.

Modified Inverse RIAA					
frequency	dB	frequency	dB	frequency	dB
10.00	-19.74	223.9	-7.43	5012	8.18
11.22	-19.70	251.2	-6.64	5623	9.04
12.59	-19.64	281.8	-5.88	6310	9.92
14.13	-19.58	316.2	-5.15	7079	10.81
15.85	-19.50	354.8	-4.46	7943	11.72
17.78	-19.40	398.1	-3.81	8913	12.63
19.95	-19.27	446.7	-3.20	10000	13.55
22.39	-19.12	501.2	-2.63	11220	14.46
25.12	-18.94	562.3	-2.11	12590	15.37
28.18	-18.72	631.0	-1.63	14130	16.27
31.62	-18.46	707.9	-1.19	15850	17.17
35.48	-18.16	794.3	-0.77	17780	18.04
39.81	-17.80	891.3	-0.38	19950	18.89
44.67	-17.40	1000	0.00	22390	19.71
50.12	-16.93	1122	0.38	25120	20.50
56.23	-16.41	1259	0.77	28180	21.25
63.10	-15.84	1413	1.17	31620	21.96
70.79	-15.22	1585	1.60	35480	22.62
79.43	-14.55	1778	2.07	39810	23.23
89.13	-13.83	1995	2.57	44670	23.79
100.0	-13.09	2239	3.12	50120	24.28
112.2	-12.31	2512	3.72	56230	24.72
125.9	-11.51	2818	4.36	63100	25.11
141.3	-10.70	3162	5.05	70790	25.44
158.5	-9.88	3548	5.78	79430	25.72
177.8	-9.05	3981	6.55	89130	25.96
199.5	-8.23	4467	7.35	100000	26.16

Listing 2. Results from SPICE simulation.

New Network Design

Figure 2 can be modified to achieve the desired results. In order to shift the high frequency pole the value of R3 must change. Unfortunately, this also changes the gain of the network. However, by moving part of R3 to the input side, we can control both gain and pole frequency independently.

A reference network should interface nicely between test equipment and phono preamplifiers. Therefore, I selected the following design parameters:

- 50 ohm source impedance (many generators do not use the 600 ohm audio standard)
- 600 ohm output impedance
- Dual output gains of –40dB and –60dB @1kHz
- Standard capacitance values.

Figure 6 shows my new modified RIAA network.

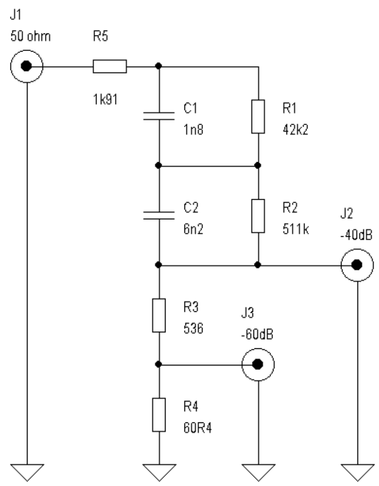


Figure 6. Modified inverse RIAA circuit.

Optimizing component values is easily done by iterative SPICE simulations, but a good starting point is needed. Approximate values can be calculated by utilizing known boundary conditions. Since output impedance should be 600 ohms, and the two outputs are 20dB apart, we can write

$$R_3 + R_4 = 600$$
$$\frac{R_4}{R_3 + R_4} = -20dB = 0.1$$

Solving we get

$$R_4 = (0.1)(600) = 60$$
$$R_3 = 600 - 60 = 540$$

At high frequency the capacitors appear as short circuits and the network simplifies to a resistor divider comprised of R5, R3, and R4. From Figure 5 we see that high frequency gain is about 27dB higher than at 1kHz. Since desired 1kHz gain is –40dB, our high frequency gain will be about –13dB. The gain equation

$$dB = 20\log A_{HF}$$

is rewritten and solved as

$$A_{HF} = 10^{\left(\frac{dB}{20}\right)} = 10^{\left(\frac{-13}{20}\right)} = 0.22$$

High frequency divider gain is then given by

$$A_{HF} = \frac{R_3 + R_4}{R_3 + R_4 + R_5} = 0.22$$

and we can solve for R5

$$R_5 = \frac{R_3 + R_4}{A_{HF}} - R_3 - R_4 = \frac{600}{0.22} - 600 = 2.1k$$

At low frequency we have the opposite effect and the capacitors appear as open circuits. The divider again is resistive and has a gain of –60dB. This is written as

$$A_{LF} = \frac{R_3 + R_4}{R_1 + R_2 + R_3 + R_4 + R_5} = 0.001$$

and the equivalent series resistance of R1 and R2 is solved as

$$R_1 + R_2 = 999(R_3 + R_4) - R_5 = 597k = R_{eq}$$

Our high frequency pole at 3.18µs is equal to the series resistance of R3, R4, and R5 times the equivalent series capacitance of C1 and C2. This capacitance is given by

$$C_{eq} = \left(\frac{C_1 C_2}{C_1 + C_2}\right) = \frac{3.18ms}{R_3 + R_4 + R_5} = \frac{(3.18ms)}{600 + 2.1k} = 1.2nF$$

Finally, we have four equations and four unknowns

$$R_1 C_1 = 75ms$$
$$R_2 C_2 = 3180ms$$
$$C_{eq} = 1.2nF$$
$$R_{eq} = 597k$$

I'll spare you the math, R1 is solved as

$$R_1 = \frac{\frac{(R_1 C_1)(R_2 C_2)}{C_{eq}} - (R_1 C_1) R_{eq}}{(R_2 C_2) - (R_1 C_1)} = 50k$$

and R2 as

$$R_2 = R_{eq} - R_1 = 597k - 50k = 547k$$

and the capacitors as

$$C_1 = \frac{75ms}{R_1} = 1.5nF$$
$$C_2 = \frac{3180ms}{R_2} = 5.8nF$$

These are only starting values. For a best fit real world design I optimized for the nearest 1% resistor and standard capacitor values (as shown in Figure 6).

Comparison: Old vs. New

As a reference network the performance must be pretty good. Figure 7 shows the frequency response errors relative to Listing 2. My new network is accurate to within +/-0.1dB over the full frequency range. For comparison I ran an error simulation of the Lipshitz network in Figure 2. It is extremely good up to about 10kHz where it drops off as expected due to the missing 3.18µs corner. The Lipshitz circuit can, however, be modified by adding a 3.83k resistor as R5 at the input to reposition the high frequency pole.

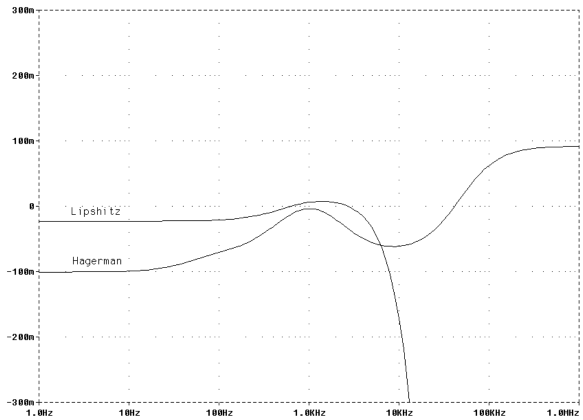


Figure 7. SPICE generated error responses for Lipshitz and Hagerman networks.

To check the sensitivity of component values I ran a few more simulations varying capacitances by 5%. Errors remained within +/-0.3dB.

I hope this information is helpful to designers of high end audio equipment.

References

[1] A. Wright, “The Tube Preamp Cookbook”, *Vacuum State Electronics*, 1995.
[2] S. Lipshitz, and W. Jung, “A High Accuracy Inverse RIAA Network”, *Audio Amateur*, 1980.
[3] S. Lipshitz, “On RIAA Equalization Networks”, *JAES* 1979.
[4] “AN-124: Three High Accuracy RIAA/IEC MC and MM Phono Preamplifiers”, *Analog Devices*, 1992
[5] M. Giles, “Audio/Radio Handbook”, *National Semiconductor*, 1980
[6] A. Wright, “Secrets of the Phono Stage”, *Sound Practices* #15, 1998

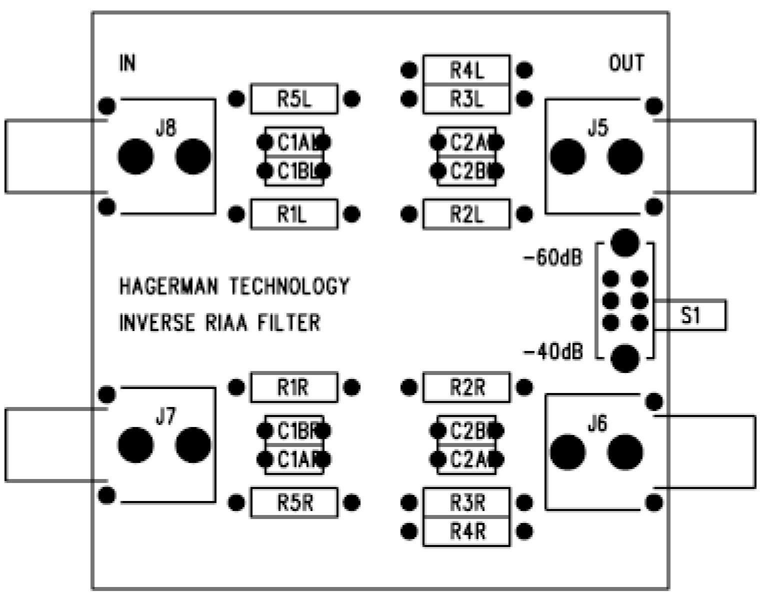
Sources

The network of Figure 6 is available as a kit (order #KF-1) from Old Colony Sound Lab for $25. The two channel kit comes complete with PCB, 1% metal film resistors, 2% polypropylene capacitors, connectors, and instructions.

Old Colony Sound Lab
PO Box 876
Peterborough, NH 03458-0876
603-924-6371
http://www.audioxpress.com

iRIAA2

Inverse Filter



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Description

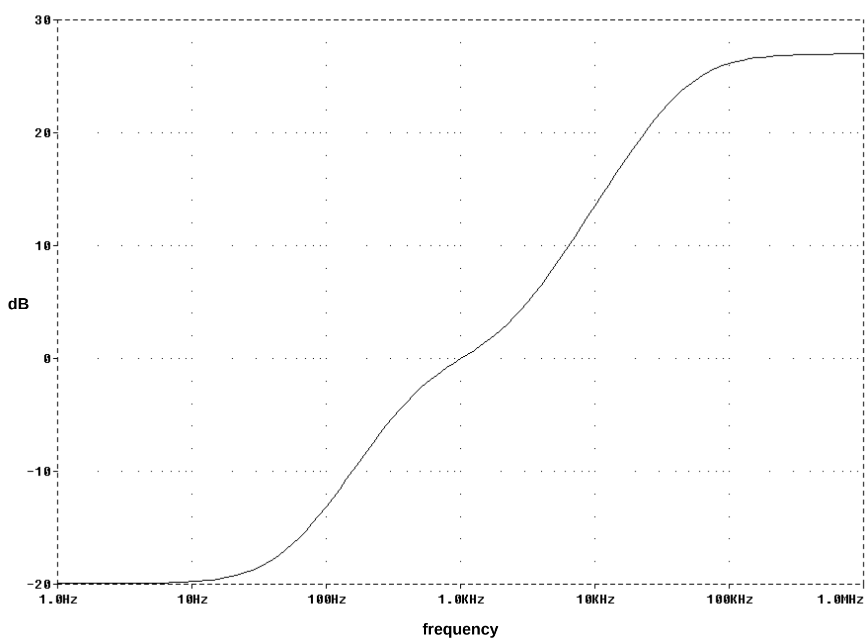
The inverse RIAA Filter is an accurate reference for testing modern phonostages. It provides the inverse transfer function of the RIAA response and level shifting so that a line level signal is converted into exactly what a phonostage input expects. Separate output levels of –40dB and –60dB (referenced to 1kHz) accommodate moving magnet and moving coil stages, respectively.

Specifications

Item	Specification
Accuracy	+/-0.5dB
Gain	-40dB / -60dB @ 1kHz
Output Impedance	660 / 60 ohms

Frequency Response

Each channel matches the ideal inverse RIAA transfer function (with corner added at 3.18us) to within +/-0.5dB.



Tabular Values

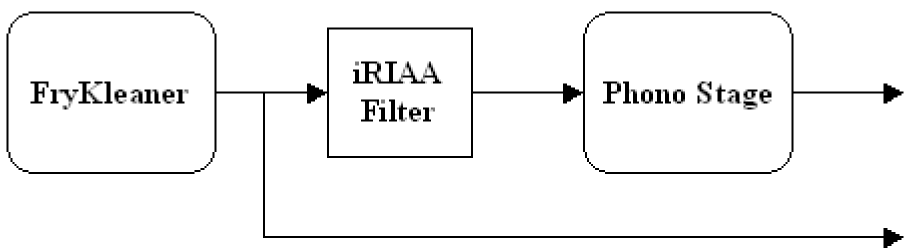
Frequency	dB	Frequency	dB	Frequency	dB
10.00	-19.74	223.9	-7.43	5012	8.18
11.22	-19.70	251.2	-6.64	5623	9.04
12.59	-19.64	281.8	-5.88	6310	9.92
14.13	-19.58	316.2	-5.15	7079	10.81
15.85	-19.50	354.8	-4.46	7943	11.72
17.78	-19.40	398.1	-3.81	8913	12.64
19.95	-19.27	446.7	-3.20	10000	13.56
22.39	-19.12	501.2	-2.63	11220	14.48
25.12	-18.94	562.3	-2.11	12590	15.40
28.18	-18.72	631.0	-1.63	14130	16.31
31.62	-18.46	707.9	-1.19	15850	17.23
35.48	-18.16	794.3	-0.77	17780	18.10
39.81	-17.80	891.3	-0.38	19950	18.96
44.67	-17.40	1000	0.00	22390	19.80
50.12	-16.93	1122	0.38	25120	20.61
56.23	-16.41	1259	0.77	28180	21.38
63.10	-15.84	1413	1.17	31620	22.11
70.79	-15.22	1585	1.60	35480	22.80
79.43	-14.55	1778	2.07	39810	23.44
89.13	-13.83	1995	2.57	44670	24.02
100.0	-13.09	2239	3.12	50120	24.54
112.2	-12.31	2512	3.72	56230	25.01
125.9	-11.51	2818	4.36	63100	25.43
141.3	-10.70	3162	5.05	70790	25.78
158.5	-9.88	3548	5.78	79430	26.09
177.8	-9.05	3981	6.55	89130	26.35
199.5	-8.23	4467	7.35	100000	26.57

Installation

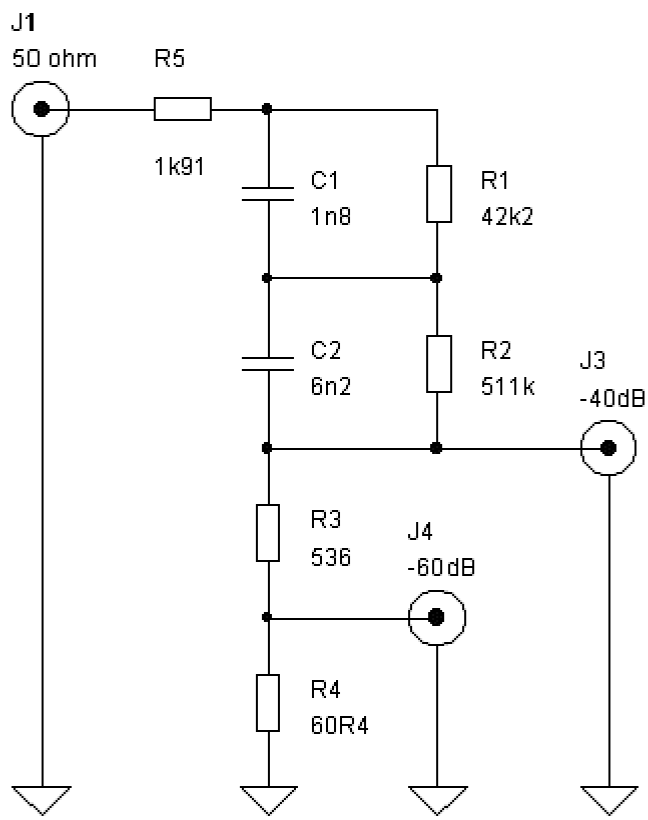
Connect the iRIAA2 filter between your line level source and the inputs to your phonostage.

Break-In

For break-in of phonostages, connect a line level signal source (CD player or FryKleaner) directly to the inputs. The –40dB outputs will mimic a stereo moving magnet cartridge and the –60dB outputs a moving coil cartridge. The output of the phonostage under test (PUT) should be a line level signal virtually identical to the original. If not, then there is a phase or frequency response aberration in the PUT.



Schematic


J1
50 ohm
R5
1k91
C1
1n8
R1
42k2
C2
6n2
R2
511k
J3
-40dB
R3
536
J4
-60dB
R4
60R4