Elliott Sound Products Project 06

Hi-Fi Phono Preamp (RIAA Equalisation)

Please Note: PCBs are available for this project. Click the image for details.

In an update from 2003 it was pointed out that for some unknown reason, some suppliers no longer stocked 82nF capacitors, and several constructors have had difficulty sourcing them. There is an answer, and it actually improves the EQ accuracy (albeit marginally). The situation has changed now, and 82nF caps are available (almost) everywhere, but the preferred network uses 750Ω and 100nF caps.

R8 (L&R) is changed to 750 ohms (a standard E24 value), and C4 (L&R) is now 100nF. If the 750 ohm resistors are not available from your supplier, use 2 x 1k5 resistors in parallel. Alternatively, you can use the originally specified 82nF cap with a 910 ohm resistor. The worst case error with any of these networks is less than 0.5dB, but 82nF and 910Ω or 100nF and 750Ω are close to perfect, with the 750Ω/ 100nF version having the edge by a few milli-dB. The original network was used before E24 resistor values became commonplace (820 ohms is an E12 value, which used to be all one could get easily).

Introduction

RIAA equalisation is the standard for vinyl disks. It's been in use for a long time (some time around 1954), and was 'tinkered' with by the IEC to tame the bottom end. An additional pole was added in 1976, at 20Hz (7,950µs), but this has *not* been included in the P06 as shown here. The 'amendment' by the IEC was (apparently) withdrawn in 2009. IMO it never worked, and never sounded right.

Many active EQ stages can't continue the rolloff much beyond 25kHz or so, because the gain of the amplifier stage can never be less than unity. A few use fully passive EQ in the belief that it somehow sounds 'better', but the stage featured here uses a combination of active *and* passive, in separate networks. The design was used by me long before the Internet, and the version shown (with a few minor updates along the way) was first published on the ESP website in 1999.

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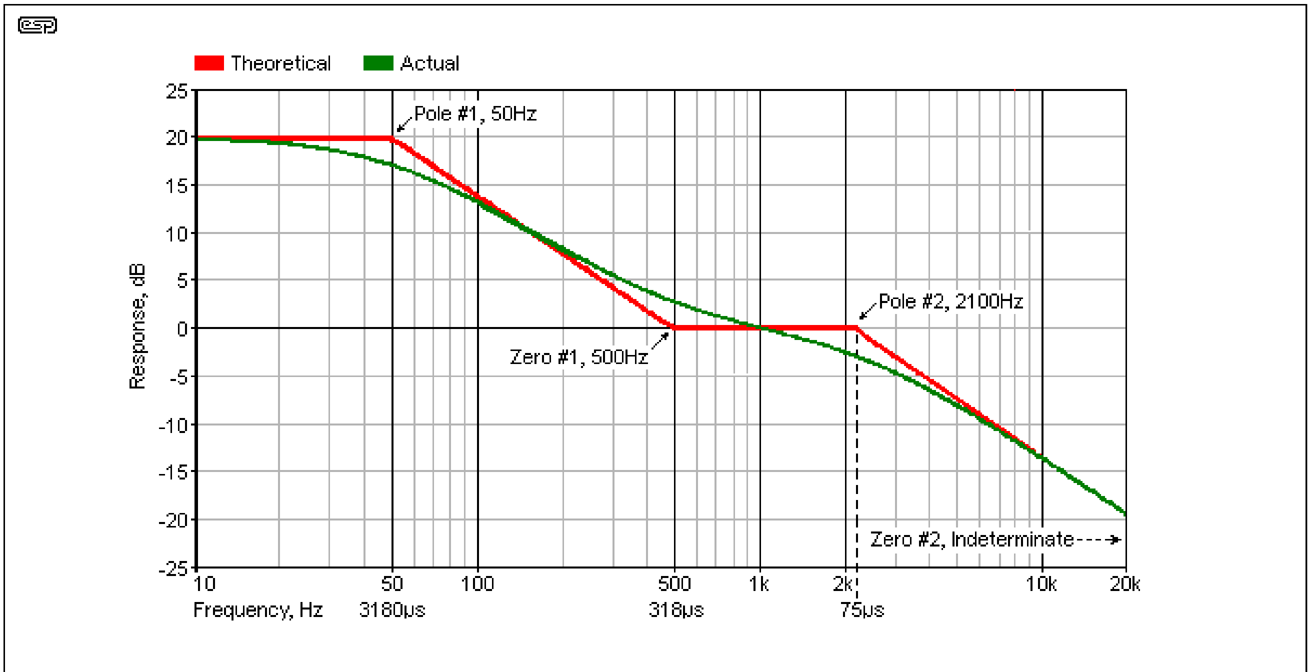

RIAA Equalisation - Theoretical And (Idealised) Actual

The above graph shows the theoretical and (idealised) actual response of an RIAA EQ stage, normalised to 0dB at 1kHz. Most RIAA equalised phono stages have an additional (and undesirable) zero at some frequency above 20kHz. This extra zero is avoided in the design described, because the circuit uses a passive low pass filter that continues to roll off the high frequency response above 20kHz, with the final rolloff limit somewhere well beyond 10MHz (depending on the capacitor's self inductance).

> The terms 'pole' and 'zero' need some (in this case simplistic) explanation. A single pole causes the signal to roll off at 6dB/ octave (20dB/ decade), and a single zero causes *boost* at the same rate. If a zero is introduced after a pole (as shown above), the effect is to stop the rolloff - back to flat response. The flat response is seen between 500Hz and 2,100Hz. The next pole (2,100Hz) causes the signal to roll off again. The 'indeterminate' zero above 20kHz is caused because many preamps cannot reduce their gain below some fixed value determined by the circuit (although the effect is often seen well before the gain falls that far). Not all have this issue, and it's not present in P06.

As noted further below and elsewhere on the ESP website, striving for 'perfect' accuracy is pointless, as so much depends on the pickup cartridge itself, the tone arm, and (of course) the recording. When you purchase vinyl, no-one tells you what EQ was applied during the mastering and cutting processes, the high frequency response degrades after the disc has been played many times, so ultimately you have to let your ears be the final judge of what sounds right to you.

Phono Preamp Circuit

The phono preamp described here has an accurate RIAA equalisation curve, is very quiet, and offers better sonic performance than the vast majority of those seen in magazines and application notes. (This is of course subjective, and is based on countless reports from those who've built it). Like most other ESP projects, it is very tolerant of opamps but the NE5532 dual op-amp is a good choice. This is a low noise, high speed device with excellent characteristics, and is inexpensive. It is ideally suited to this sort of application. Noise is extremely low, since any amplifier stage noise is rolled off above 2kHz with the passive filter. Other excellent opamps are the OPA2134 and LM4562, and OPA2134s are used in my own units.

One factor often overlooked with phono preamps is the capacitive loading on the opamp output at high frequencies. This is all but eliminated in this design, and since the NE5532, OPA2134 and LM4562 can all drive a 600 ohm load with ease, the 750 (or 820) ohm output resistor isolates the output stage from any capacitive loading. The first stage has 10k in series with the cap, so capacitive loading is not an issue.

Note that if a moving-coil phono cartridge is to be used, a step-up transformer or ultra low-noise preamplifier circuit is needed before the phono preamp. This circuit is intended for use with the standard moving magnet type phono pickup.

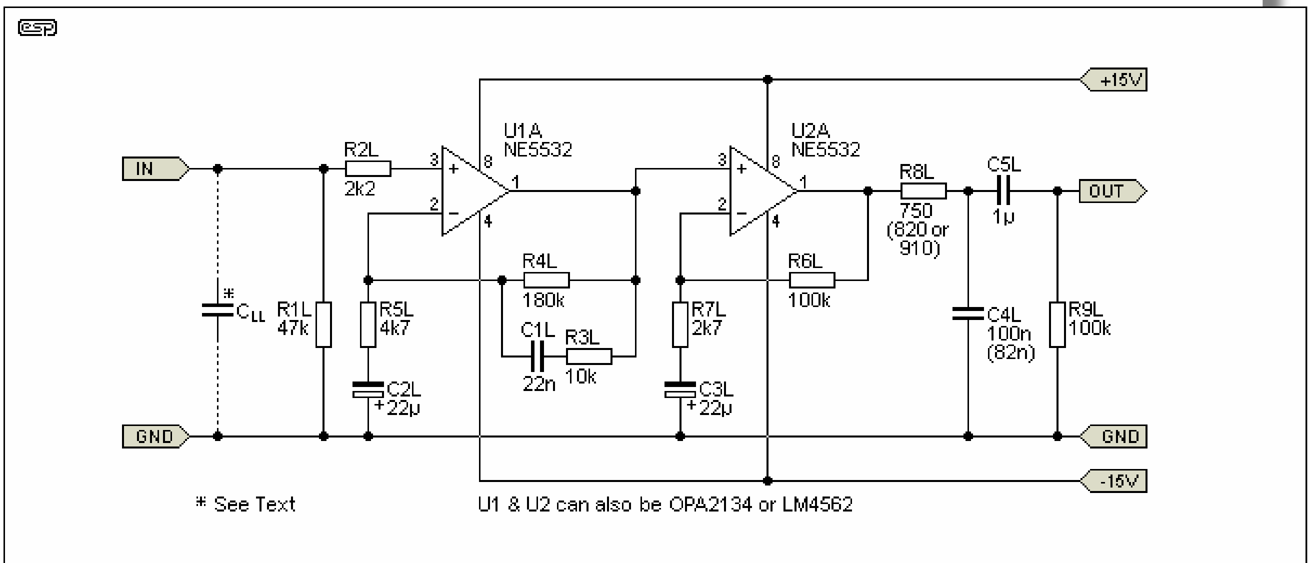

Figure 1 - Phono Preamplifier (RIAA Equalisation)

The cartridge loading capacitor marked * (CLL, and its equivalent on the right channel - CLR) is entirely optional. In almost all cases it isn't needed, because the cable capacitance between the phono cartridge headshell and the preamp will be (more than) sufficient. Some manufacturers specify a loading capacitance, but many do not. The vast majority of phono cartridges perform at their best with the lowest possible capacitance, and adding more rarely makes things better.

Few people have the ability to measure the capacitance of their interconnects or the internal tone arm cables, but it's usually in the vicinity of 100pF with typical cables. Should the cartridge maker suggest a higher capacitance, feel free to experiment with the value of CLL/R. It's best to locate these caps (if used) directly across the RIAA input sockets, rather than on the PCB, and for this reason there is no provision for the loading caps on the board. The caps also should be matched to within 1% so the Left and Right channels remain properly balanced.

There is some debate on the Net about cartridge loading impedance, with various suggestions that reducing the standard load from 47k to something lower (even as low as 10k) provides benefits. While this is plausible, I've not run any tests so cannot confirm or deny that there may be an advantage. If this is something you wish to try, I suggest that it be done at the phono RCA input socket (along with the capacitor if you want to try that too).

In general, I'd like to think that after all these years, the cartridge manufacturers would have a pretty fair idea of what they are doing. On that basis, I suggest that if you wish to experiment, do so by all means, but don't expect to get any 'magic' results. Bear in mind that any additional parts will also increase the input capacitance of the circuit, so you can easily end up much worse off than if you just left the circuit alone. See Phono Cartridge Loading for more information.

The high value capacitors could be non-polarised electrolytic types, since they will have (virtually) no DC voltage across them. However, these are quite large, and standard (polarised) electrolytics may be used instead. Polarised caps will function normally without DC bias, but do *not* use tantalum caps - they are my least favourite capacitor type, and are not recommended for use with zero DC bias. Standard aluminium electrolytics are actually perfectly alright with no bias (despite what you may have read), and if sufficiently large (in value) will contribute virtually no measurable distortion. The AC voltage across C2L/R and C3L/R will never exceed ~5mV at any frequency down to 10Hz, and these caps play no part in the equalisation process. Feel free to increase the value if you wish (100µF is not a problem).

The low value capacitors should be 2.5% tolerance if obtainable, otherwise you may be able to measure a selection of standard tolerance caps to find those which are closest to the required value - preferably to within 1%. Some deviation from the ideal RIAA equalisation curve will occur if these caps are too far from the designated values. More important is matching between channels - this should be as accurate as possible.

Resistors (as always) should be 1% metal film for close tolerance and low noise. This design differs from most in that the low and high frequency equalisation are performed separately, with the LF being active and the HF passive. Because of the low value of the output resistor, a following stage input impedance down to 22k will cause little degradation of the EQ curve.

The customary 'flattening' of the curve at 50Hz has not been fully incorporated, since most listeners find that the bass sounds far more natural without this. In this respect it can be said that accuracy is lacking, but I am still using this arrangement, and have not found rumble or other low frequency 'noise' to be a problem.

Based on the RIAA specification, the table shows the performance with frequency - below 50Hz there is a marked (and deliberate) deviation, and 'accuracy' figures are not quoted.

Note that there is no provision for a 'rumble' (subsonic) filter, and the circuit as shown has a low frequency -3dB point of about 3Hz. A low rumble turntable is essential - especially if you use a subwoofer. A well damped and isolated turntable platform is an excellent idea, and I have had great success with a large concrete paving slab, neatly covered with speaker carpet or other material, and isolated using foam rubber. Some experimentation will be needed to get this exactly right. Usually, good results will be obtained when the foam support is compressed to 70% of its normal thickness with the weight of the concrete slab and turntable. A shelf attached to a wall is another good method of providing subsonic isolation.

If low-frequency noise is a problem, you will often see vigorous movement of the woofer cones even when there is no bass content. If this is an issue with your setup, I recommend that you include a Project 99 subsonic filter. The standard configuration is 36dB/octave, with a -3dB frequency of 17Hz. This will normally eliminate even the most intractable low frequency interference, typically caused by warped discs. It usually helps if you have LF feedback problems too, but they have to be below the cutoff frequency of the filter.

Freq - Hz	**TC**	**Gain - dB**	**Ideal - dB**	**Error - dB**
20	N/A	62.25	N/A	N/A
50	3180 µs	59.11	58.42	0.69
500	318 µs	43.87	43.85	-0.02
1000	N/A	41.2	Reference	
2100	75 µs	38.43	38.43	0
21 k	N/A	22.17	21.24	0.07

Table 1 - RIAA Equalisation Characteristics (750Ω and 100nF)

As can be seen from the table, accuracy is better than 1dB, and gain at 1kHz is about 40dB (100) so a nominal 5mV cartridge output will give 500mV output. This may be increased if necessary, by increasing the value of the 100k resistor in the second stage. Care is needed to ensure that the gain is not increased so far as to cause clipping of the signal - allowing for the worst possible case. As it stands, stage 2 has a gain of 38 (31dB). The 'TC' column shows the official time constants for each frequency 'break point' (a pole or a zero).

If the 100k resistor were to be increased to 220k, the total gain will be slightly more than doubled, at 38dB. An input signal to stage 2 at 17mV (5mV phono cartridge output) would then give a normal output at 1kHz (before the passive filter) of 1.12V RMS. The theoretical output at 20kHz is over 9.75V RMS, but this never happens because at 20kHz all recordings will be 15-20dB below the level at 1kHz (being very conservative). See Audio Level Vs. Frequency, below.

This means that *actual* output level at 20kHz will typically be around 1V RMS at the most. However, if the gain of the second stage is increased too far, there is a risk of clipping. This is an unlikely possibility due to the nature of music - there are very few fundamental frequencies of any instrument (other than a synthesiser) above 1kHz, and most harmonics roll off naturally at approximately 3 to 6dB per octave above about 2kHz, but it must be considered.

Only one channel is shown, the other channel uses the remaining half of each op-amp, the pinouts of which are shown on the diagram. Remember that the +ve supply connects to pin 8, and the -ve supply to pin 4.

Op-amps are bypassed from each supply line to ground with a 10uF electrolytic and a 100nF polyester or ceramic capacitor to ensure stability. These parts are all provided for on the PCB.

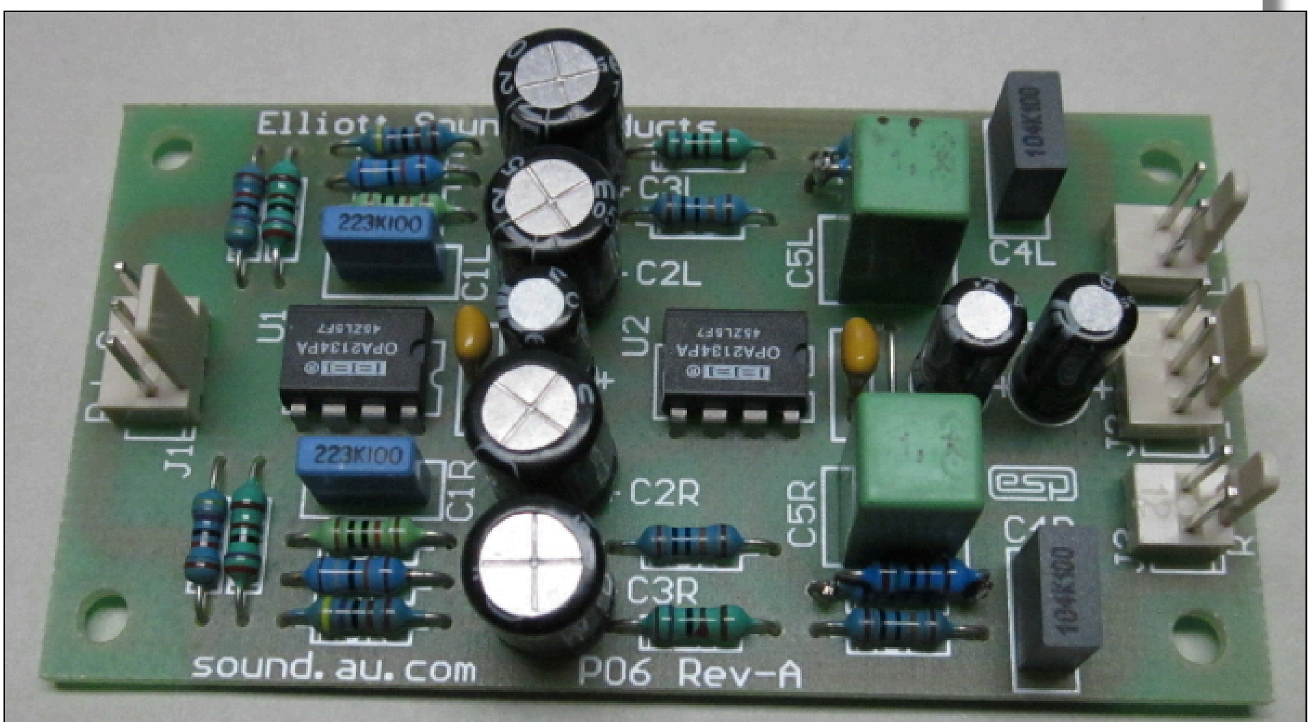

Photo of Completed Unit

The photograph above shows a complete phono preamp using the PCB. This is as shown in Figure 1, and is the latest version of the board. When I built it, I didn't have any 750Ω resistors in stock, so I used 2 × 1.5k in parallel instead. Note that the URL shown is now obsolete - it *should* read 'sound-au.com'.

A large proportion of P06 PCB sales are a direct result of recommendations from others who have built it and found (as I did when I first designed the circuit many, many years ago) that the overall sound is better than the average phono preamp. There's no reason I can think of that should make it sound any different from more conventional circuits, but it's hard to argue with hundreds of happy customers.

Audio Level Vs. Frequency

There is very little on the Net or elsewhere that gives anyone an idea of the level they should expect at any frequency. The image below was captured using 'Visual Analyzer' - one of many PC based FFT programs that are available. The signal was taken from an FM tuner - you can see the sharp rolloff above 15kHz and the 19kHz pilot tone used to decode the 38kHz FM sub-carrier. The capture was taken off-air, from an Australian 'alternative' radio station, so includes several different genres of music, as well as speech.

The capture was set up to hold the maximum level detected over the sample time (over 2 hours), so represents the highest level recorded at any frequency across the band. Although everything above 15kHz is removed, the overall trend is clearly visible. While there will always be deviations and exceptions with different musical styles, I have run this test before and used different programme material. The general trend is valid over a wide range of music styles. No equalisation was used on the received signal - it is captured directly off-air.

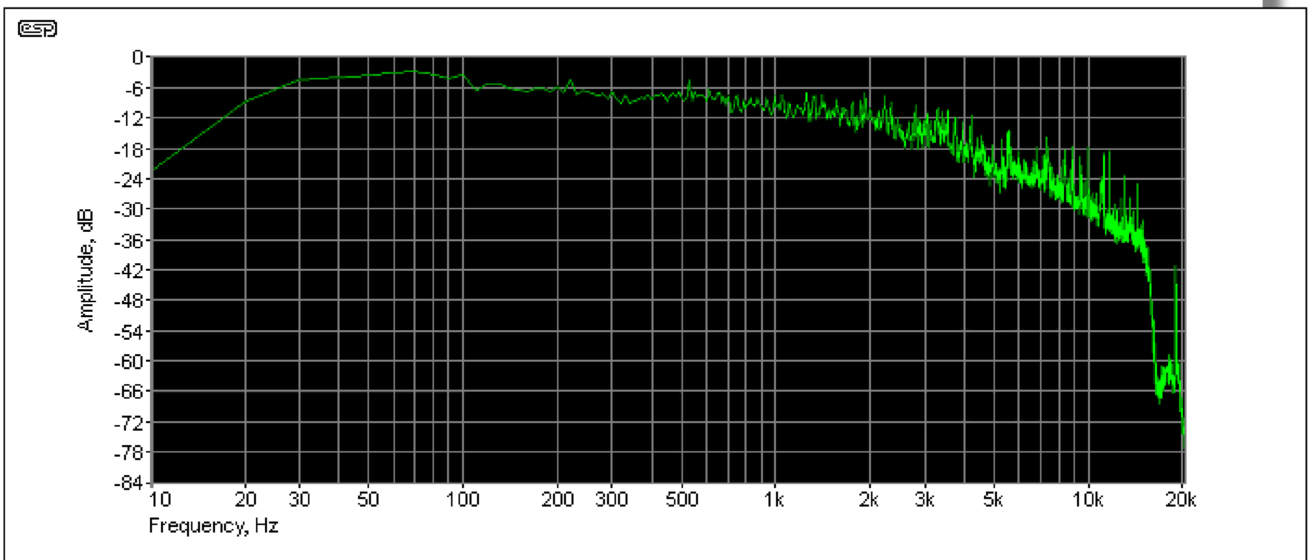

Figure 2 - Amplitude Vs. Frequency of 'Typical' Audio

The 'reference' level is -9dB at 1kHz. The maximum peak levels are seen between 30Hz and 100Hz (this is definitely programme material dependent!), and the level between 200Hz and 2kHz is reasonably flat, showing roughly 3dB fall over that frequency range. There is a ~6dB rolloff in the octave from 2k-4kHz, followed by a ~10dB rolloff between 4k-8kHz. What is of greater interest is the amplitude of the highest peaks, because overload will occur on peaks, not average levels. At 10kHz and just above, there are peaks at -18dB and some additional peaks (-24dB) at just below 15kHz.

Based on this, it's reasonable to expect that the worst case level at above 15kHz will never exceed -30dB, and this is 21dB below the level at 1kHz (a little less than 1/10th). A cartridge with 5mV output at reference level 1kHz will therefore have no more than 5mV output at any frequency around 20kHz - this is the highest level we can expect. With the recommended component values for the RIAA equaliser, the maximum possible level from the output of the second stage is around 1V RMS - well within the capabilities of the suggested opamps. Even if the maximum level were to be 50mV (same output at 20kHz as at 1kHz), the second stage is still below the clipping level. Further increases of gain are not recommended unless you understand the likely outcome.

Overall Response

If the circuit is driven from an inverse RIAA network, the overall response should be flat. It's already been stated that P06 has a small low frequency boost, and that can be seen in the following graph. If you wanted to build your own inverse RIAA equaliser, see Project 80. It's a contributed design, and is as close to a true reverse RIAA EQ stage as you are likely to find. The plots below were done using the 'ideal' values in the P80 article for the signal source.

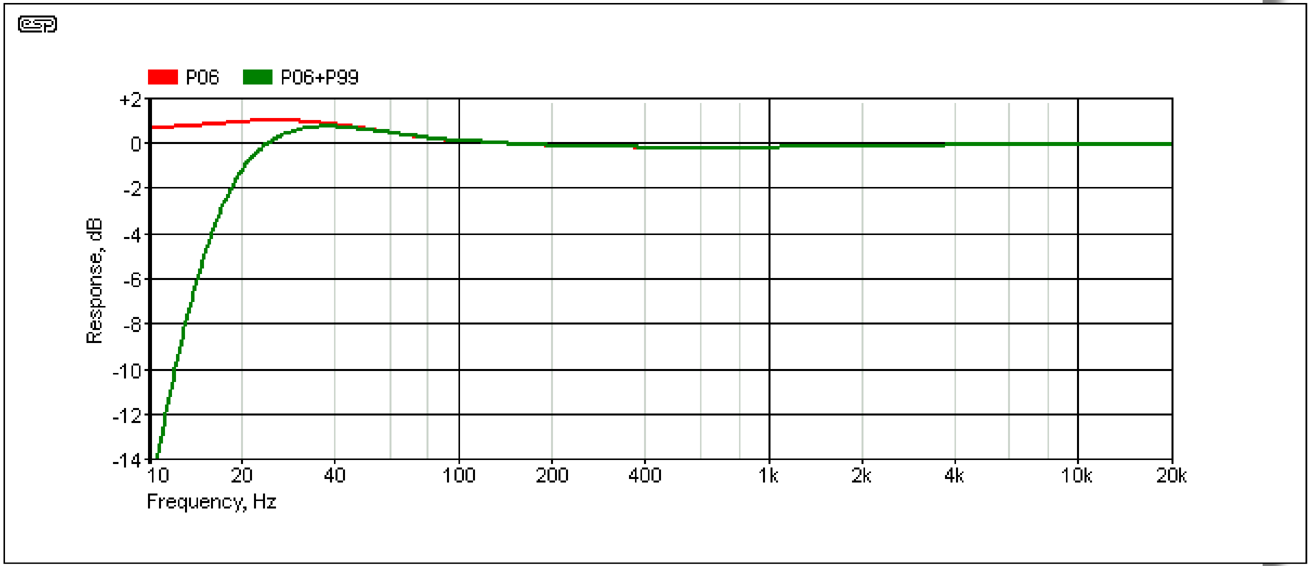

Figure 3 - P06 Response With & Without P99

When mated to the P99 'rumble' filter (which follows P06) the response is shown in green, with the P06 response by itself in red. The end result is a 1dB boost at 40Hz, with the response falling off at 36dB/ octave below 20 Hz. The Project 99 article has more details, and provides additional options for the low frequency cutoff.

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Updated Aug 2003./ Sept 08 - Figure 2 redrawn to match original circuit./ Dec 2011 - Added info on loading capacitor and/or resistor./ Feb 13 - added Figure 2 and associated info./ Jul 17 - included RIAA curve and explanations./ April 2020 - updated PCB photo.