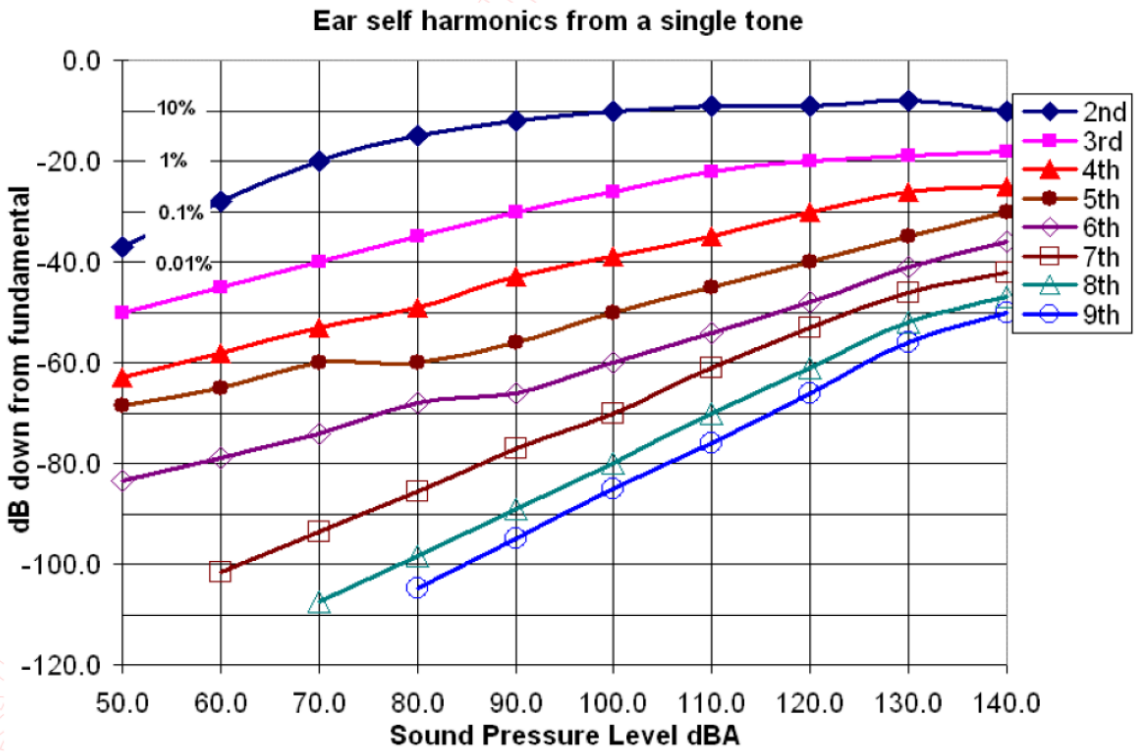
- **ACRUX WHITE PAPER** WTP 11/8/2022

Philosophy/Design Approach

The science of psycho acoustics has been searching for suitable measures of “sound quality” for many years.

The non-linear nature of the auditory system has been clearly demonstrated, and the existence of Acoustic Harmonic Masking is a proven phenomenon.

Masking occurs when the "ear", when presented with a sine wave, for example, self-generates a regularly decaying harmonic structure. This harmonic structure depends on amplitude, and extends to very high order harmonics (6th and above) until the amplitude falls below the auditory threshold at about 0 dB SPL or about 110 dB below the highest comfortable sound level.



(The above illustration was found on the web)

Any significant deviation from this harmonic structure will be audible, and the sensitivity to this deviation increases as the harmonic order increases. The rate of increase in this sensitivity is unclear, but there is evidence that several percent of 2nd harmonic distortion is acceptable, and 0.01 % of 9th harmonic which is not accompanied by an appropriate amount of lower order harmonics, is both audible and unacceptable.

In effect, as long as the harmonic structure of the reproduced music approximates the ears "desired" sequence the output is considered to be musical. If it does not, then it is considered to be discordant. This deviation in itself can be used as a basis for judging the quality of both transducers and amplifiers. Also, low frequency narrow band maskers apply over a broader

frequency range than higher frequency maskers. This implies that higher frequency audio signals are of more concern.

This understanding has led to a variety of acoustic metrics being developed- for example those by GedLee (2003), Cheever (2001), Tan (2004), Voishvillo (2006), Chalmers (2010) etc..
It has also validated the general worthlessness of common metrics such as Total Harmonic Distortion (THD) (especially at full power for the case of output amplifiers) and the usual two tone intermodulation measurements, except in specific cases such as transient intermodulation distortion (TIM/TID).

Despite this, the matter of what is a suitable audio metric is still highly contentious.
As mentioned previously, the initial efforts used simple distortion measures (THD, 2 tone IMD, 3 tone IMD), then more complex ones were developed that used weighted measures of single tone harmonics (Cheever, GedLee).

The most recent attempts split the audio range up into frequency bands that correspond to the ear's physical response. The behavior of the source to be assessed is then evaluated for distortion, both amplitude and temporal, and an overall weighted value is computed. This is a non-trivial exercise and there are no readily available tools to assess the result beyond producing a DSP post processing algorithm. Requests have been made to have Virtins Pro implement a more advanced metric beyond GedLee, but there have been no updates as yet.

None of the metrics except "Cheever" attempt to provide any guidance as to how to design an audio element for best sound, and that author has a very specific point of view as to how to approach the problem which is different from the path taken in this design.

However, some lessons seem to be clear- the minimum requirements to attempt to satisfy all of the various metrics and achieve objectively and subjectively neutral sound are for the amplification stages to have extremely low high order distortion and minimal temporal dispersion, especially at higher audio frequencies.

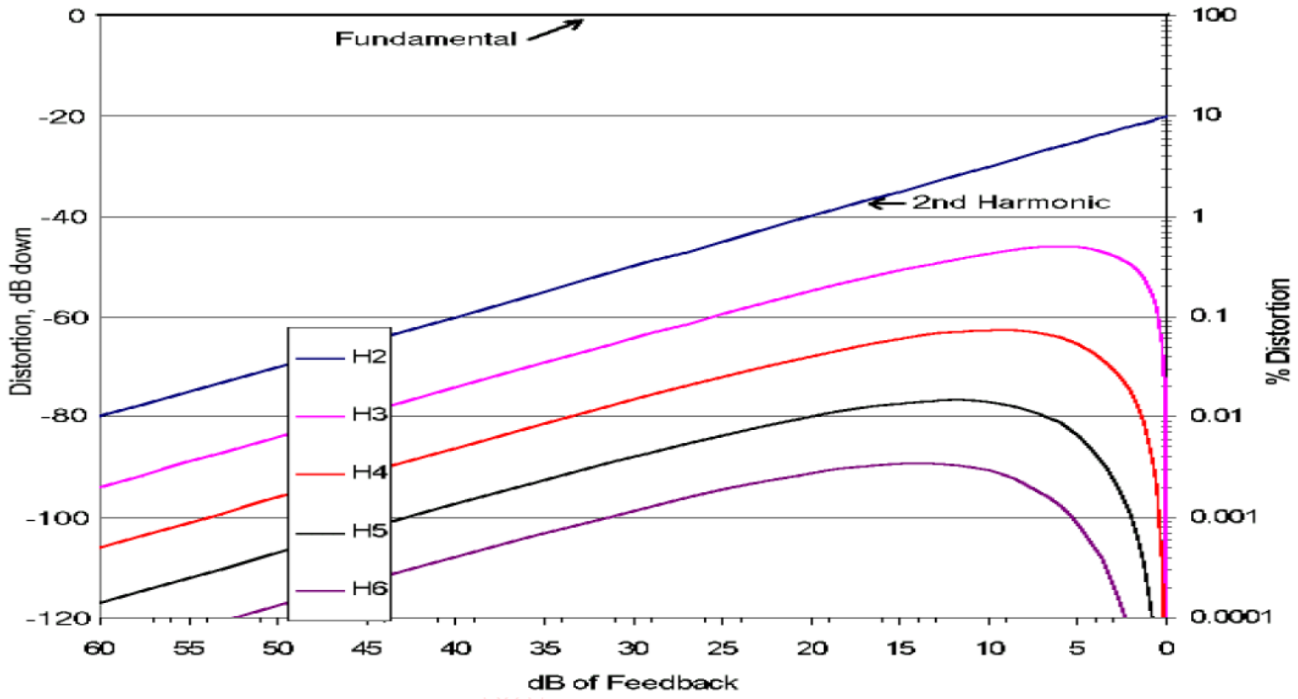
In any event, despite none of these existing metrics being particularly successful, some general observations can be made.

In essence, there are two classes of audio systems- those with significant negative feedback, either global or local, and those without.

Most transducers- such as phono cartridges, loudspeakers, microphones, are in the no negative feedback class and have a different set of distortion rules applied in comparison to audio components that employ significant amounts of negative feedback, as is the case for most solid state amplifiers.

Zero and very low amounts of negative feedback amplifiers, such as many tube amplifiers, are essentially in the same class as transducers.

Negative feedback generally acts to create the high order distortion components that the ear finds unacceptable, and this effect is at its peak at about 15 dB of negative feedback, and then declines with increasing feedback levels.



(The above illustration was found on the web)

It is in this way that non-feedback systems and transducers can be acceptable even if they have high levels of intrinsic distortion, whereas systems with low levels of low order harmonics but relatively high levels (compared to the masking threshold) of high order harmonics may not be acceptable.

However, there is an exception to this "rule". As mentioned before, if all of the amplifier generated harmonics are at a level which is substantially below the audibility threshold (at about -110 dB full scale) then the ear does not perceive them as being an add-on to the self generated harmonics and the acoustic "acceptability" will be equivalent to the ideally decaying self-generated harmonic case and will be fully masked.

This implies that if sufficient negative feedback is available to ensure that all distortion components are sufficiently low then the output will be indistinguishable from the self generated distortion case and it will be perceived as being fully transparent to the ear. As has been famously stated, “the problem is not that amps have too much negative feedback, but that they have too little”.

Most negative feedback systems have frequency dependent gain functions and hence feedback factors that are of an inverse frequency kind- i.e. the degree of gain/feedback factor falls as the frequency increases.

As a result, the amount of feedback available for correction of the high order harmonics within the audio band declines with increasing frequency which is contrary to what would be desirable. This could be problematic in phono stages as the required gain is so large, but as the overall gain requirement versus frequency also declines at approximately a 1/f rate, the problem is not as severe as it might at first appear to be.

The intent behind this phono stage is to reduce all known audio imperfections to an inaudible level. To do this the lowest level of negative feedback employed is 46 dB and a nominal value of 50 dB is targeted. This is a reflection of documented sensitivity and masking results based on a simple single stage distortion characteristic. This should sufficiently reduce all measured amplitude distortion- harmonic, standard two and three tone intermodulation, and many tone

intermodulation for example- and also indirectly contribute to very low levels of phase intermodulation distortion and low dispersion across the audio band.

The required level of feedback is also open to some local variation based on individual amplifier characteristics and signal levels, as long as the net about -110 dB distortion level is maintained over the entire amplifier chain.

The RIAA compliance must also be extremely tight. This is a reflection of the fact that many people can detect broad frequency response variations of <0.25 dB p-p, which is perceived as subtle changes in the musical character, whereas even trained listeners seemingly can not detect <0.1 dB p-p variations. So, the goal is to have <0.1 dB p-p RIAA error in order to produce an objectively and subjectively neutral frequency response. In addition, this minimizes the amplifier's temporal error- that is the variation in group delay over the audio band- relative to the original source prior to the application of RIAA pre-emphasis.

Using opamps with very high levels of feedback at 20 kHz also helps to minimize variations in RIAA compliance at that extreme caused by manufacturing tolerances in their gain bandwidth products. Such variations amount to +/- 20% or so of the nominal value.

The importance of many tone intermodulation performance, including ultrasonic frequencies up to 40 kHz or so, recognises that in the presence of complex (music) waveforms, intermodulation distortion of the signal components creates a noise like dynamic signal that can elevate the audible noise floor, reducing dynamic range, degrading attack characteristics, and creating a loss of clarity and imaging quality. As a result the distortion introduced by such a complex input has to be exceptionally low to be acceptable, and should meet the prior criterion of being below the audibility threshold.

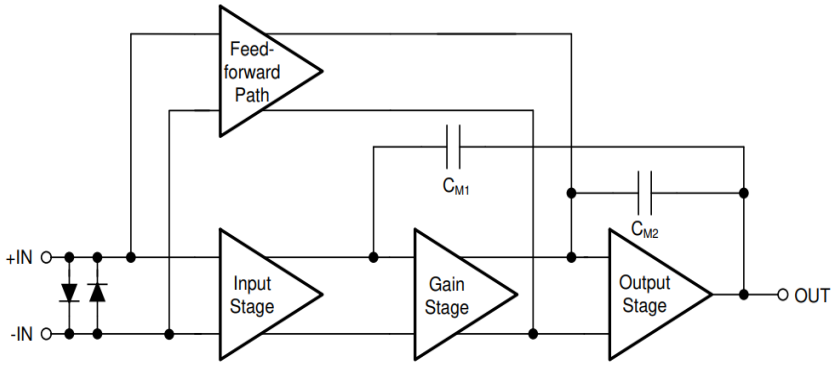
These are issues that are rarely ever mentioned in phono preamp design, and even more rarely measured as they have been for this design.

To achieve these goals and to maintain a 50 dB feedback factor, the overall open loop gain bandwidth product of a single amplifying stage used to construct the RIAA characteristic targeting MC cartridges with 64 dB gain at 1 kHz and 44 dB at 20 kHz, needs to be 180 dBHz- or 1 GHz.

Such an amplifier would also have to have an open loop gain of 114 dB at 1 kHz and 134 dB at 20 Hz to maintain the distortion characteristics, and it would also have to have very low input noise. Such a device does not seem to exist. The choice has to be two (or more) opamps.

Distortion in Opamps

It has been argued that opamps are not single stage amplifiers and so the simple distortion model does not apply.



The OPA1656 opamp functional diagram, taken from the TI data sheet.

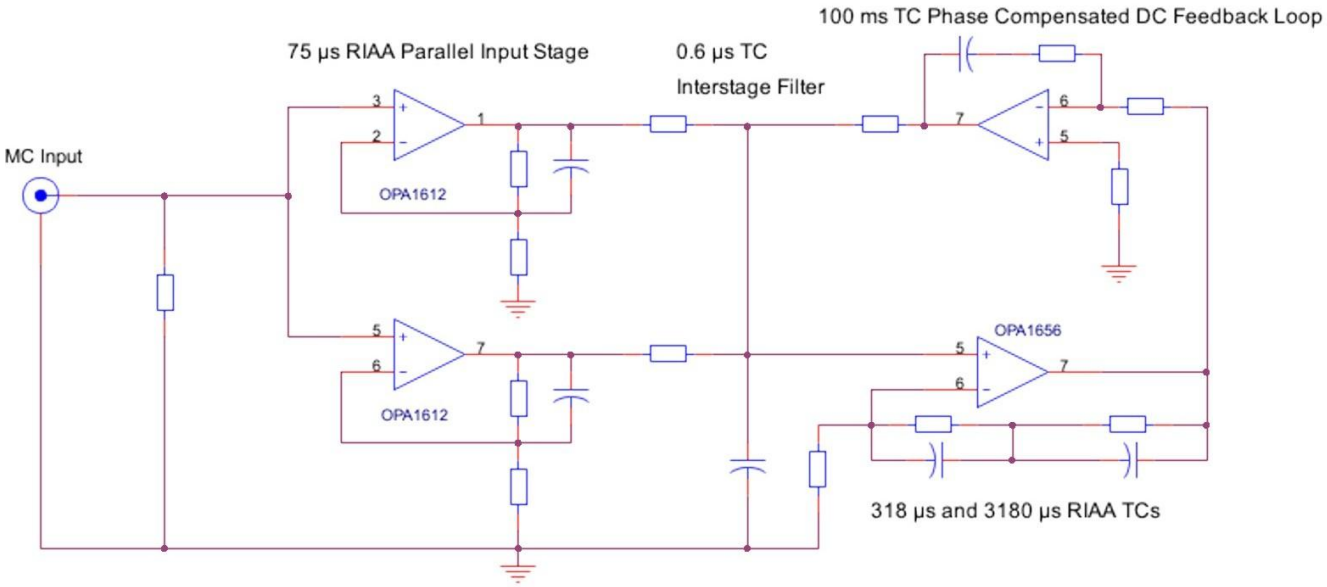
This is superficially correct. However, in modern low distortion opamps the general approach is to have an input stage that develops an error signal current and for that to be "transported" into a high impedance (the gain stage) to produce the voltage gain.

The input stage is, in effect, the single amplifying stage distortion that was earlier referred to. The second stage has an entirely different distortion mechanism that occurs under entirely different conditions to the input stage. It is the source of TIM (slew induced distortion) which was mentioned earlier. That distortion occurs when the amp cannot keep up internally with the maximum rate of change required for the signal. The opamps chosen can slew internally at a rate 40x the needed amount, which is sufficient to prevent this issue from occurring. The output from the gain stage is then applied to the output stage.

The distortion of the output stage is still different in kind to those of the other two stages, and is dependent on output signal level and load, and varies tremendously as a function of the specific details of the output stage. It can be minimized if care is taken in the choice of output amplitudes and loading, using the opamp datasheet as a guide.

Circuit topologies, such as the feedforward path shown above, can be implemented to reduce the intrinsic distortion and indeed, partial or even "complete" distortion cancellation can be produced. In addition, two or more non-identical distortion characteristics in cascade are not necessarily multiplicative, and feedback need not necessarily increase the number of higher order distortion products. For example, a novel design for a power amp employs a technique that essentially perfectly splits the analog signal into positive and negative halves for voltage gain and current scaling so that the distortion error is purely one of slope about the crossover point and does not produce harmonic multiplication when feedback is applied.

The RIAA Deemphasis Characteristic Implementation



The phono stage is configured as above.

For best case signal to noise performance there should be as much gain as possible in the first stage, and the opamps that are available with adequate noise and distortion include the OPA1612, which has a gain bandwidth product of 157 dBHz (about 70 MHz) for a closed loop gain of about 10 (20 dB). The nominal closed loop gain that this opamp can have at 20 kHz and still meet the 50 dB feedback factor is 157-86-50= 21 dB. This sets the initial 20 kHz gain.

This opamp was chosen for the input stage. As the output signal levels are quite small- about -10 dBv (about 0.3 v rms)- for this stage a somewhat higher gain can be tolerated and still keep the distortion levels adequately low, so 24 dB gain at 20 kHz was decided upon. As a result the second stage, using the same or a similar opamp, can have a gain of 20 dB at 20 kHz and meet the overall 44 dB gain goal for the RIAA stage, while satisfying the desired distortion goals.

The OPA1612 is also a low noise opamp, and the noise can be improved upon by using parallel stages. In this case two input amplifier stages are used to provide a 3 dB improvement over a single stage.

The RIAA de-emphasis characteristic has to implement three time constants (TCs), 75 us, 318 us and 3180 us. Sometimes other TCs are added, such as at 3.18 us, to try to emulate the roll off in the bandwidth of the record mastering lathe.

This was not done in this design essentially because there is no standard defining these characteristics that has been universally followed in the recording industry.

This design separates the 75 us from the other two TCs for implementation.
The 75 us TC requires about 20 dB higher gain at 1 kHz than at 20 kHz.
In order to have the best noise performance this TC should be placed in the first stage so that the input stage gain increases in the regions of highest audibility improving the audio band S/N ratio of the amplifier.

The input stage gain was set at 160 (44 dB) at 1 kHz and 16 (24 dB) at 20 kHz.

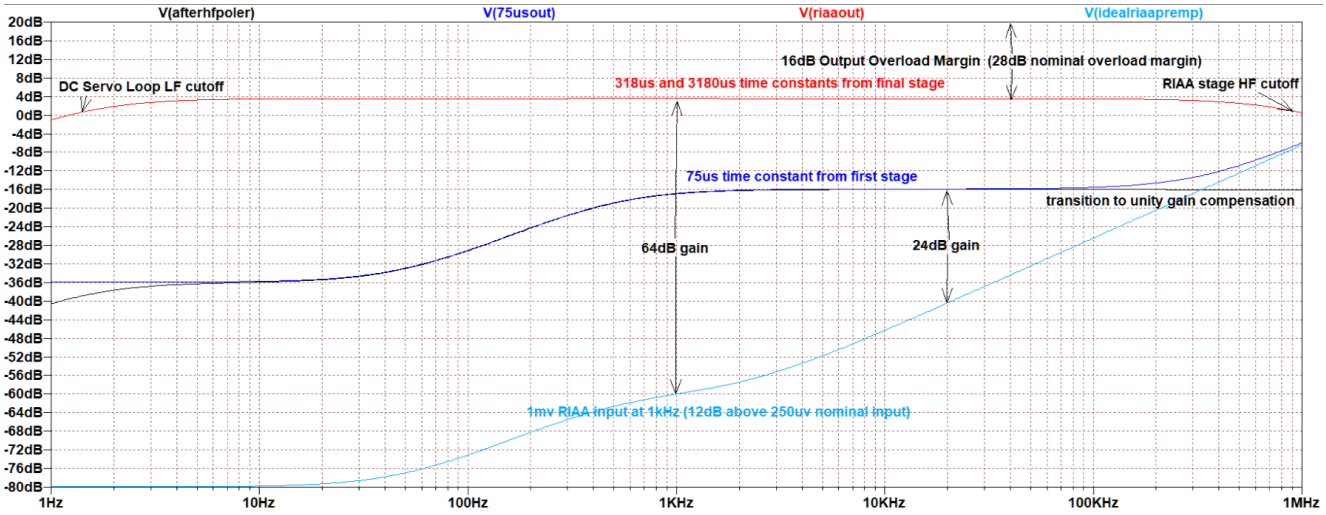
The feedback stage components were set by the availability of the low tolerance resistors (0.1%) and capacitors (1%) needed to fit the TC and the need to not excessively load the output of the opamp, which can create distortion.

One minor benefit of having two identical stages in parallel is that the feedback components are also in parallel. This acts to reduce the standard deviation of the error of the 75 us TC, which tightens the already exemplary deviation from the ideal RIAA characteristic even more.

The input stage is also non-inverting, the choice having been made to operate as a voltage rather than a transimpedance amplifier, so eventually the closed loop gain of this stage will fall to unity and will not fall any further, introducing an additional about 0.6 us TC to the response. Normally this is too high a frequency to matter for the RIAA characteristic, but in this case the decision was made to compensate for the extra TC in order to protect the second stage from potential overload at high frequencies due to any RF that might be introduced into the phono input, and to place the remaining two TCs and the remaining gain of +10 (20 dB) in the last stage. No unity gain transition occurs in the last stage. The components are also chosen to have extremely close tolerances.

An interstage RC network was added to provide the extra unity gain transition compensation TC.

Below is an illustration of the contributions of the various TCs to the gain breakdown of the design when an input with an ideal RIAA pre-emphasis is applied.



The levels are shown for the rated output of +10 dBv (about 3 v rms) , which occurs for 12 dB (4x) the 0.25 mv rms @1 kHz, 5 cm/sec sensitivity of the MC phono stage.

The reference RIAA pre-emphasis curve is shown, with a level of 1 mV at 1 kHz. As can be seen, the output from the input stage, with the 75 us TC, increases in level with increasing frequency up until 2 kHz when it flattens out. It will only start rising again at about 300 kHz as the first stage transitions to unity gain.

Beyond the first stage output, the additional TC kicks in, flattening its output response above 300 kHz and avoiding the routing of unwanted HF signals to the output stage from the input stage.

The final gain stage adds the remaining TCs, raising the response at low frequencies and providing an additional high frequency roll off.

The final result is an extremely small error with respect to the ideal RIAA characteristic over the entire audio band, with a desirable HF roll off occurring at about 1 MHz.

An additional interesting aspect of this approach is that the overload margin of the phono stage tracks the RIAA curve- that is the limiting/clipping within the audio band occurs at the output stage and not at an earlier stage, so that the input overload margin gets progressively higher as the frequency input increases. This still applies even when the first stage reaches unity gain, at least until that stage itself overloads. However, that doesn't occur until the frequency is in the MHz region, and the output stage is still protected, minimizing the impact of RF events occurring at the input.

DC Offset Servo Loop

The overall amplifier is initially DC coupled, and has a DC closed loop gain of c. 84 dB or about 16,000. The input opamps have maximum voltage offsets in the order of 0.5 mV, so the output DC voltage from the complete phono stage could be as high as 8 v.

This is obviously unacceptable, so there are basically three different ways to deal with the problem.

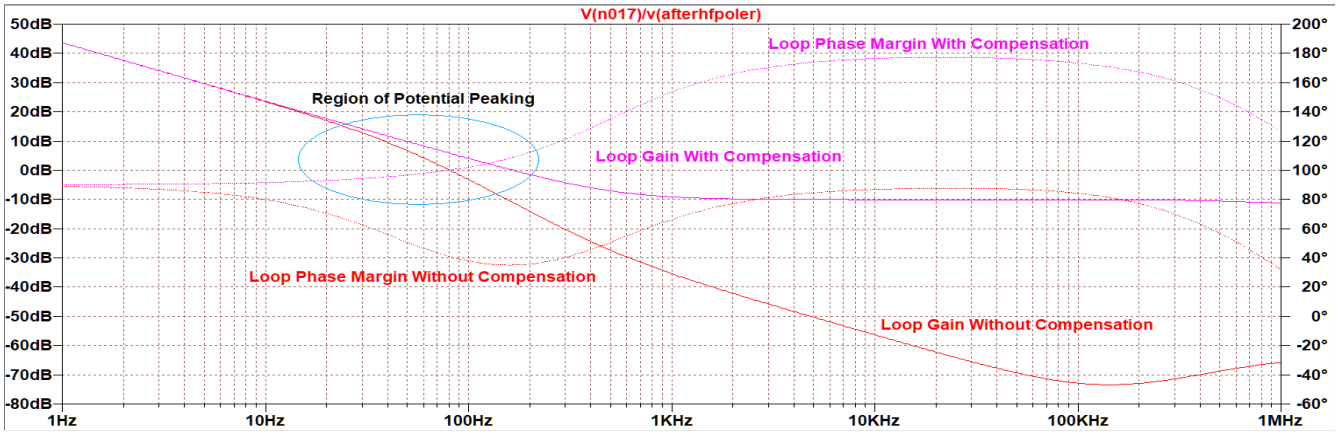
The output could be AC coupled by adding a large cap in series with the RIAA output. This was rejected as it makes the output cutoff sensitive to the external load and requires a high value series cap, which comes with a number of problems, such as a decrease in overload margin and a potential increase in distortion close to LF cutoff and an increased noise pickup sensitivity due to the physical size of the device etc..

The input of the second RIAA stage can be AC coupled by placing a large value cap in series with the resistor to ground. This eliminates the loss of overload margin, but retains the other problems associated with a large cap.

The third option is to use a DC feedback loop (DC Offset Servo Loop). This approach permits the use of low distortion low value capacitors and high value low distortion resistors, and does not place any additional elements directly in the signal path.

Because the RIAA output amplifier stage is non-inverting, an inverting, integrating, opamp can be used to provide the DC correction using negative feedback to the output stage input. The only possible downside is the presence of the two low frequency TCs in the output amp which introduce phase shifts which can cause low frequency peaking, and this can be compensated for.

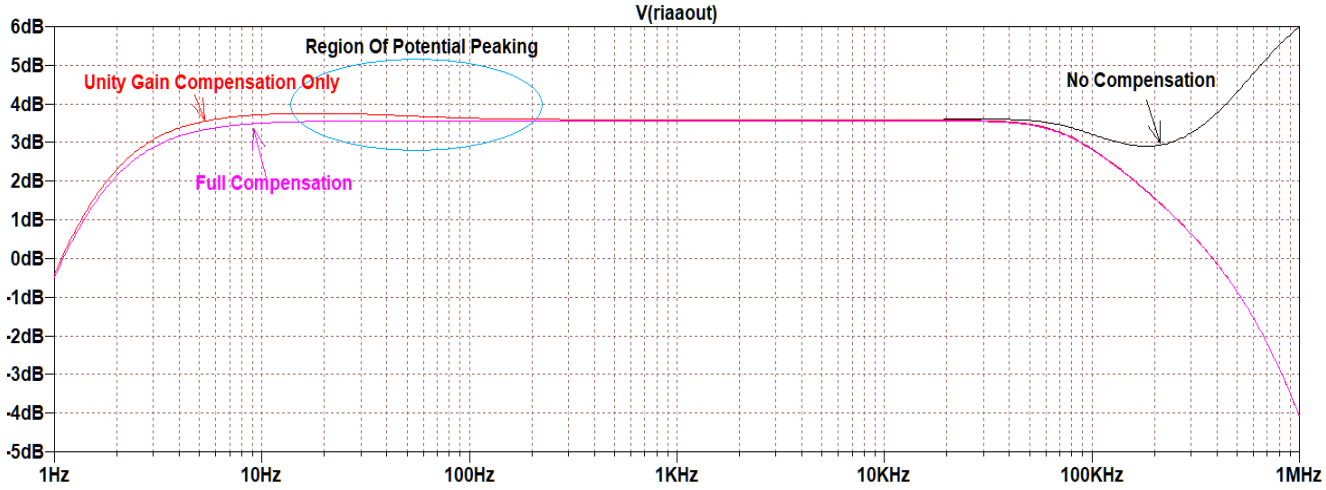
The feedback can be introduced directly into the interstage network through a resistor that also provides attenuation so that the maximum output of the opamp is just sufficient to correct for the worst case output offset of the input stage plus the input offset of the second stage. This attenuation is in the order of 40 dB, which prevents any distortion and noise introduced by the feedback stage from contributing measurably to the overall RIAA stage distortion and noise. Below is the Gain and Phase Margin of the DC loop with and without compensation.



The above Loop Gain and Phase curves show the relationships from the input of the output RIAA opamp to the output of the DC feedback opamp. The additional 40 dB attenuation between the integrator output and the input of the RIAA output opamp through the interstage compensation network ensures that the loop is stable. However, as can be seen from these, the uncompensated DC loop which includes the two output TCs and the feedback integrator has a region where the phase margin falls to less than 80 degrees and the gain is close to unity and rising, introducing the possibility of a small, but significant, amount of local peaking in the closed loop response.

Adding a compensating resistor in series with the integrator cap keeps the phase margin at a sufficient level to avoid this peaking.

The effect of the DC feedback compensation and the interstage filter compensation on the RIAA characteristic are shown below.



The black trace is with no compensation.There is peaking of about 0.15 dB at 20 Hz and there is an undesirable rise appearing above 100 kHz or so.

The red trace has the interstage filter added, eliminating the HF rise, but the LF peak remains.

The magenta trace is with both compensation networks in place.The LF peak is gone and the nominal simulated RIAA error over the audio band with nominal production resistor and capacitor values is 0.017 dB p-p.

At all frequencies in the audio band any noise or distortion introduced by the DC feedback loop opamp is reduced to a negligible amount.

The output DC value for the Phono section is set by the input offset of the feedback opamp which is an OPA2197 with a maximum value of +/- 0.2 mv. The overall maximum DC value for Acrux at the balanced outputs is +/- 5 mv and the overall low frequency response is 3 dB down at about 1.4 Hz.

Why two different phono stages for MM and MC?

The Acrux provides two separate inputs for MM and MC which go to two separate amplifiers. The two stages have identical architectures but the MM operates with 20 dB lower gain. It also uses different input opamps- OPA1656s rather than OPA1612s. OPA1656s have MOSFET rather than bipolar input devices and have very different noise characteristics as a result.

This choice was made in order to improve the Signal to Noise (S/N) ratio for high series resistance, high series inductance cartridges (MM) versus low series resistance, low series inductance ones (MC).

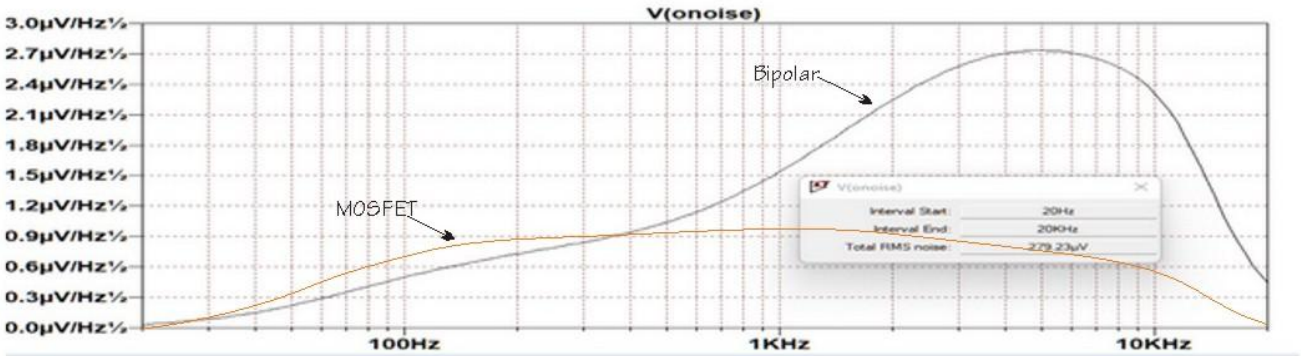
When low noise bipolar input opamps such as the OPA1612 are operated with MC cartridge impedances which are typically similar to 10 ohms in series with 10 uH, the total input voltage noise of the stage is essentially set by the voltage noise of the opamp input and the input current noise plays little part.

As the impedance rises due to the inductance of the cartridge the voltage noise caused by the input current noise rapidly increases and at about 1k ohm the total noise is more or less equally produced by the voltage and current noise source components of the opamp.

For the MM/FET input case the voltage noise is c.10 dB (3x) larger but the current noise is much, much smaller, essentially zero.

Below is the result of simulations intended to illustrate this issue.The two plots are of the A-weighted MM phono stage noise with a 1.3k ohm 500 mH inductance shorting the input and with either the bipolar or the FET input opamps used. The shorting values are typical for a MM cartridge such as the Shure V15 III.

The black curve is using the same OPA1612 devices in parallel as was used for the MC design, the orange curve is with these input opamps replaced with OPA1656s.



The total A-weighted noise in the audio band is 0.2792 mv for the bipolar input case and 0.0846 mv for the MOSFET input case. As a result, the A-weighted S/N ratio decreases by a factor of (279.2/84.6) = 10.4 dB when the lower voltage noise opamp is replaced by the lower current noise opamp.

With the MM cartridge attached, the simulated S/N ratio of the MOSFET amp is about 85 dBA relative to the full scale output of the preamp, which is subjectively inaudible and significantly quieter than the bipolar one.

These distinctions are usually not captured in the manufacturers specifications for phono stages. As a matter of course the noise is generally specified with the inputs shorted, which can be misleading, depending on the nature of the cartridge used.

Choice of capacitors and resistors.

The capacitors were chosen to be C0G ceramics for the critical signal path and 1206 X7R MLCC types for the supply decoupling networks, sourced from specific manufacturers.
The initial prototypes were constructed using Wima and Kemet polypropylene film caps throughout, and the switch was made only after exhaustive evaluation, both technical and subjective, was made.

The complete phono stage was built using both technologies, and careful measurements of distortion and frequency response occurred. No difference could be found between the two implementations.

A test fixture was constructed and the Dielectric Absorption (DA) of the C0Gs and the polypropylene caps was made. The two sources of polypropylene caps and the C0G caps exhibited similar DA behavior at under 0.1 %. This is consistent with prior measurements made and published, but not with the oft stated 6 % figure for C0G that seems to have an unknown provenance. C0Gs are virtually immune to the “microphonic” problems that ceramic caps often have.That effect is mostly due to piezoelectricity and C0Gs have almost none.

A 2007 technical paper from Kemet where a number of capacitor types were evaluated for microphonics concluded that C0G caps “have no noticeable acoustic effects”.
Informal “tap” tests with a pencil in the high impedance test jig used to test DA showed very small levels of noise produced, in fact better than the comparable polypropylene caps. Even this noise disappeared in relatively low impedance circuits, such as those used in the phono stage.

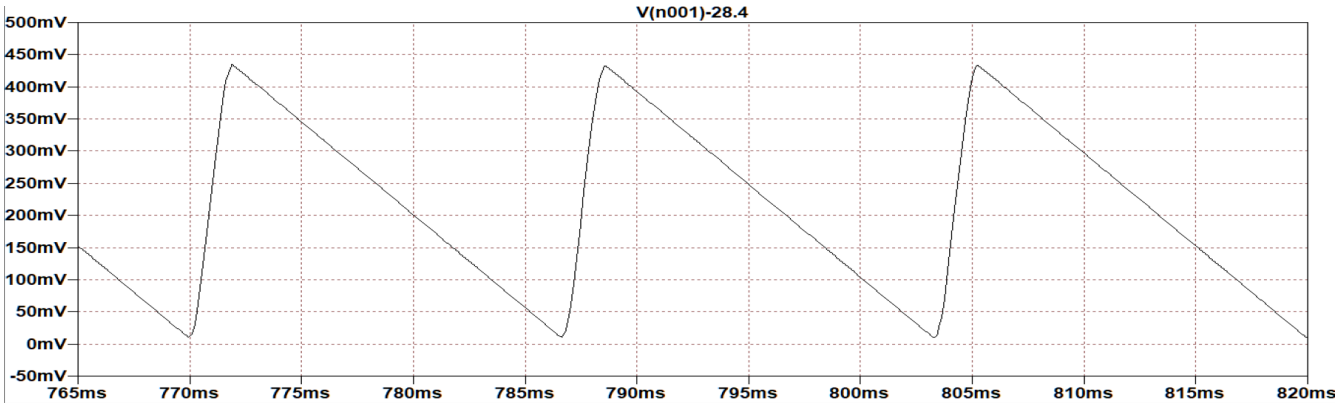
The original prototype used through hole metal film resistors.The Acrux resistors are all 0805 SMD thin film, with low temperature coefficient values in critical locations. SMD thin film resistors use essentially identical films to the standard metal film types.

No resistors are thermally stressed so it was unnecessary to go to the metal foil or MELF types.

The noise and distortion of the phono stages are not significantly different from the expected results, based on initial simulations and measurements, indicating that the resistor choices do not degrade the performance in any discernible way.

Design of the Power supply

The Acrux power supply consists of a conventional linear supply with IC regulation down to +/-18 v, enclosed in a separate chassis with an umbilical connection to the main preamp chassis where local IC regulators on each channel reduce the voltage to +/-15 v.



Above is the simulated power supply ripple at the input of the first regulator stage.

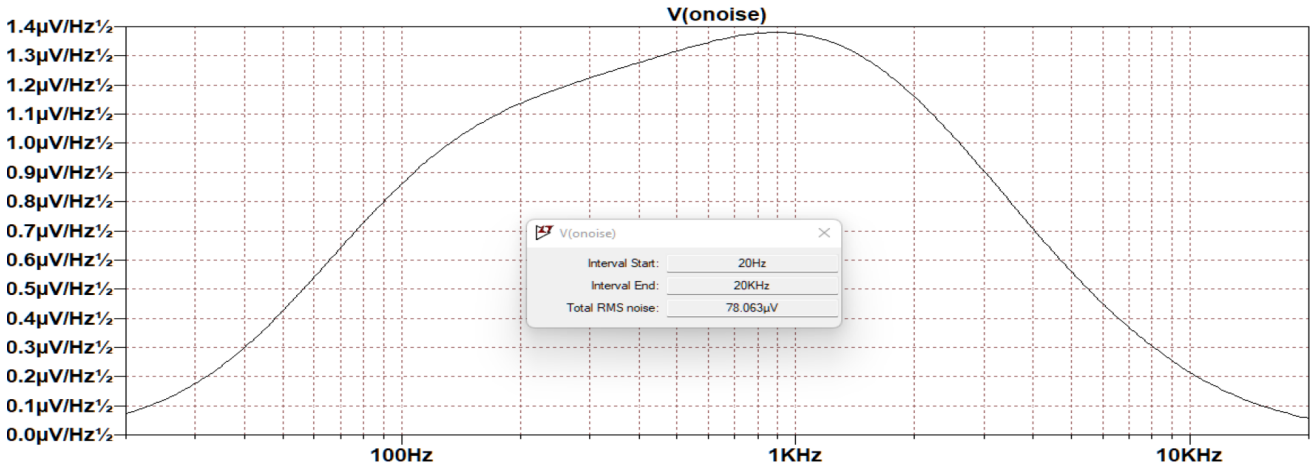
The supplies prior to the regulators have a measured ripple of c. 500 mv p-p, closely matching the simulations, and this drops to about 0.1 mv p-p after the first regulator stage. After the second bank of regulators the ripple is unmeasurably low, but is theoretically about 0.000300 mv p-p. The second bank of regulators does produce noise on the supplies. This amounts to a combined value of about 0.150 mv rms over a 100 kHz bandwidth.

Although there is some local filtering after the regulators, and at the opamp supply pins, the bulk of this noise is present on the supplies that are applied to the Acrux amplification stages. This may seem like a lot- after all, for example, it was previously shown that the Acrux noise for the MM stage measured at the output was a mere 0.085 mv rms A-weighted.

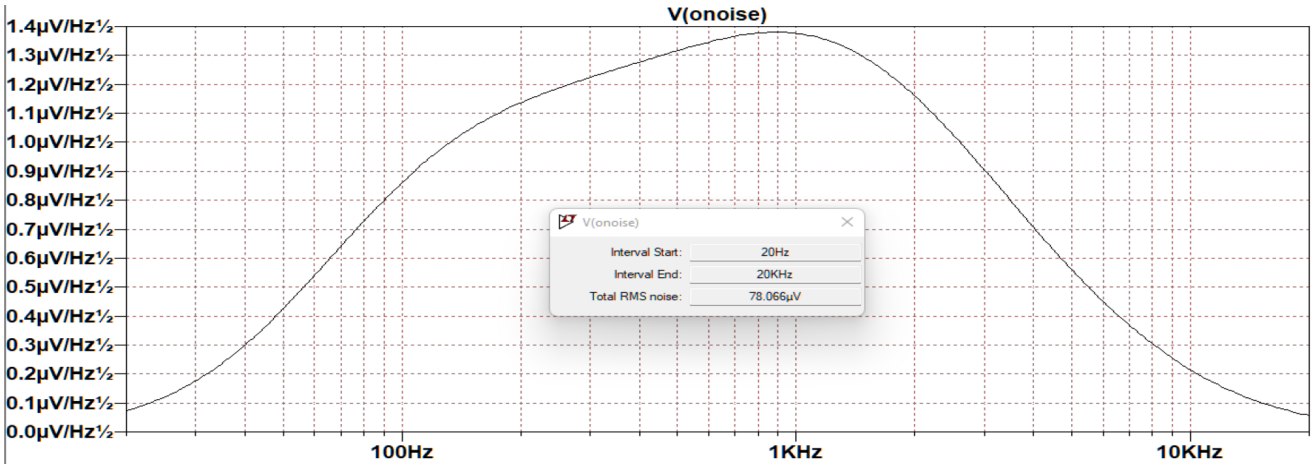
However, once again the use of large amounts of negative feedback comes to the rescue. Feedback acts to reduce, essentially proportionally, the effect of power supply ripple and noise on the output of gain stages.

Consider the MC phono stage. It has the largest amount of gain, and hence the lowest feedback factors. Below is a similar simulation to the noise evaluation of the MM stage.

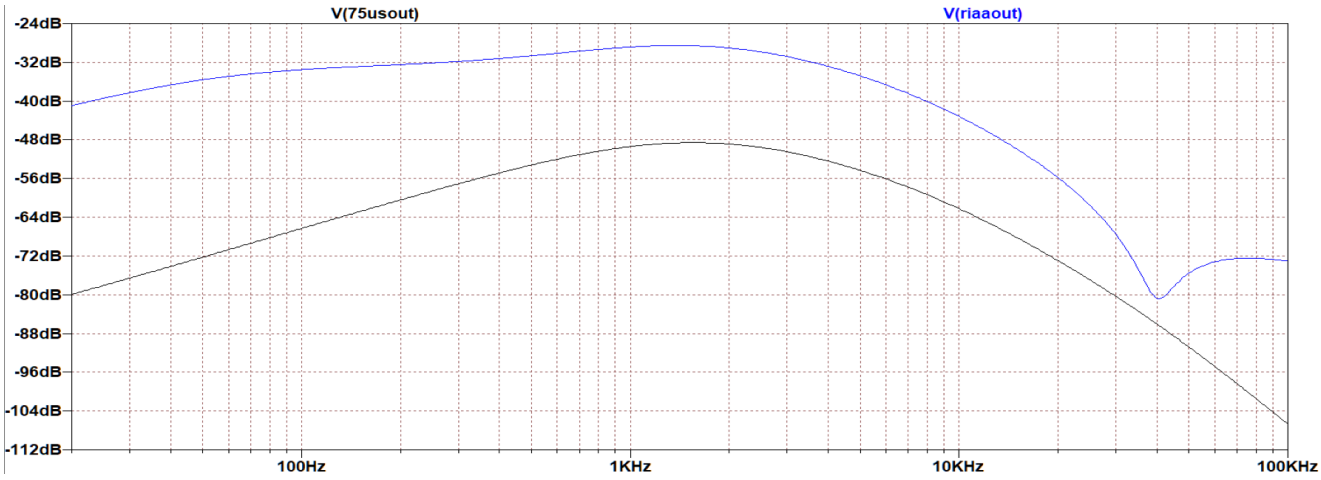
The result is the phono preamp output noise, A-weighted, with the input shorted, and no supply noise.



Below is the same simulation with the modeled power supply noise of 0.154 mv rms over a bandwidth of 100 kHz.



The result only differs in the last digit, increasing from 0.078063 mv to 0.078066 mv, an unmeasurable amount. This occurs because the opamps provide substantial power supply rejection. The plot below shows the power supply rejection versus frequency for the 75 us time constant (input amp) stage and the full MC phono preamp.



As can be seen, the input stage has a minimum of 48 dB of power supply rejection at 2 kHz (where the feedback factor is smallest) which increases both below and above this frequency. The full output has a similar characteristic, but with a minimum of 28 dB.

The extra gain at low frequency that implements the remaining TCs amplifies the noise from the first stage and causes the low frequency improvement to be reduced. Nevertheless, over the audio band the average rejection is about 40 dB which reduces the power supply noise that appears at the output to about 0.001 mv, which contributes only a tiny amount to the total output noise. It also reduces the hum pickup components by 30 dB.

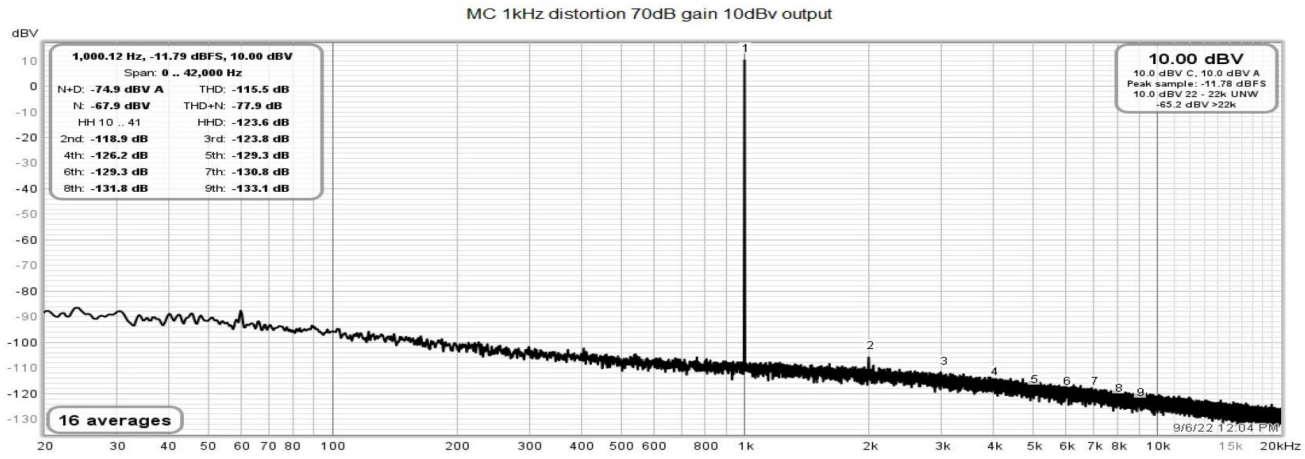
In the absence of these high feedback factors the power supply noise would indeed be a problem, and far more extreme measures would be required to deal with it.

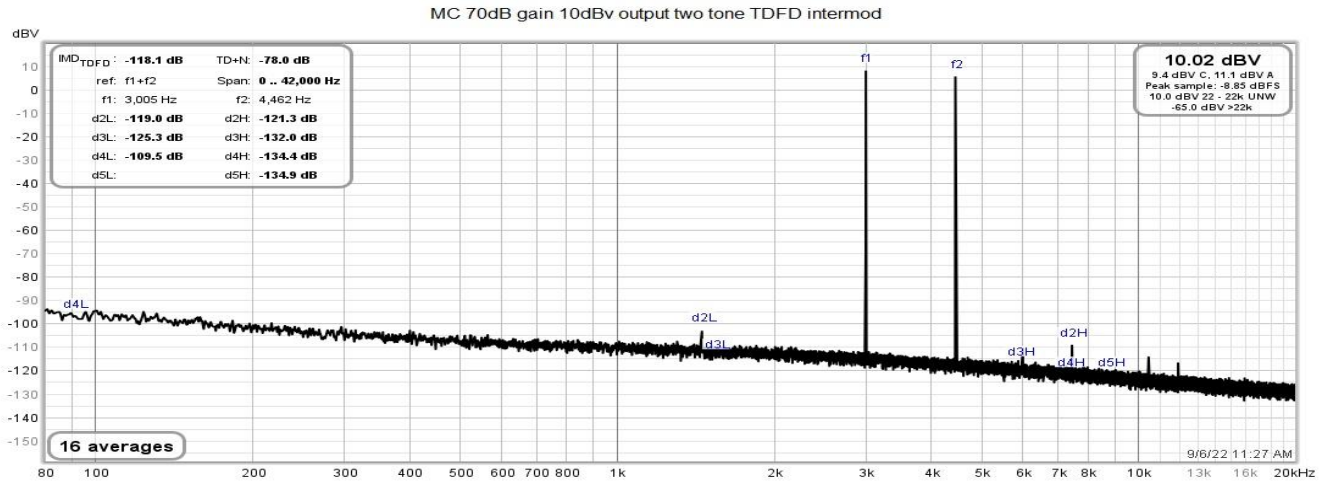
Measurement Results

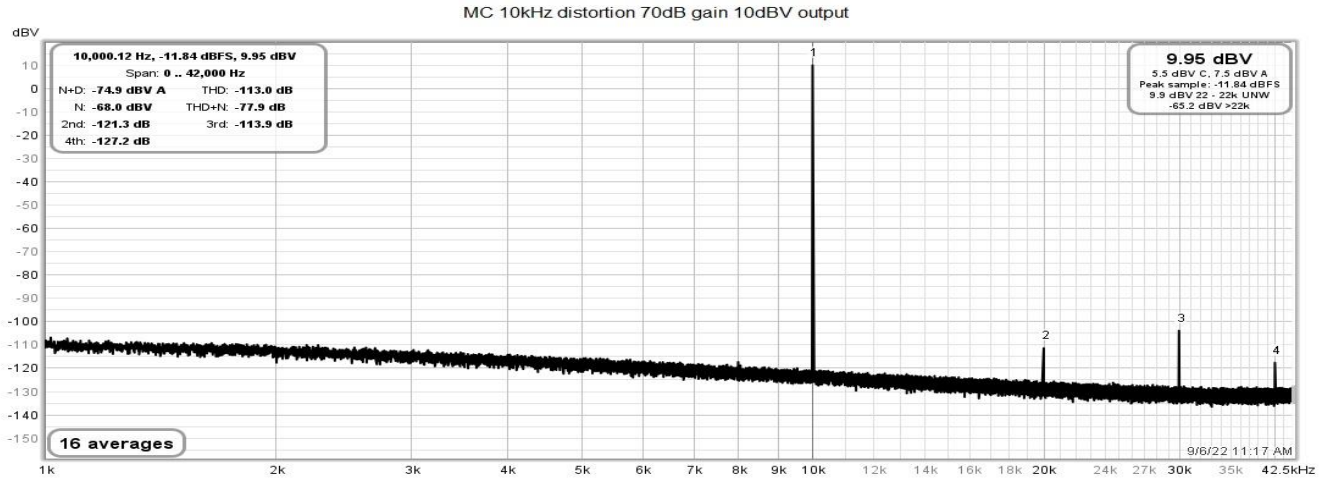
So far, all of this has been illustrated in simulation. How does the actual unit perform?

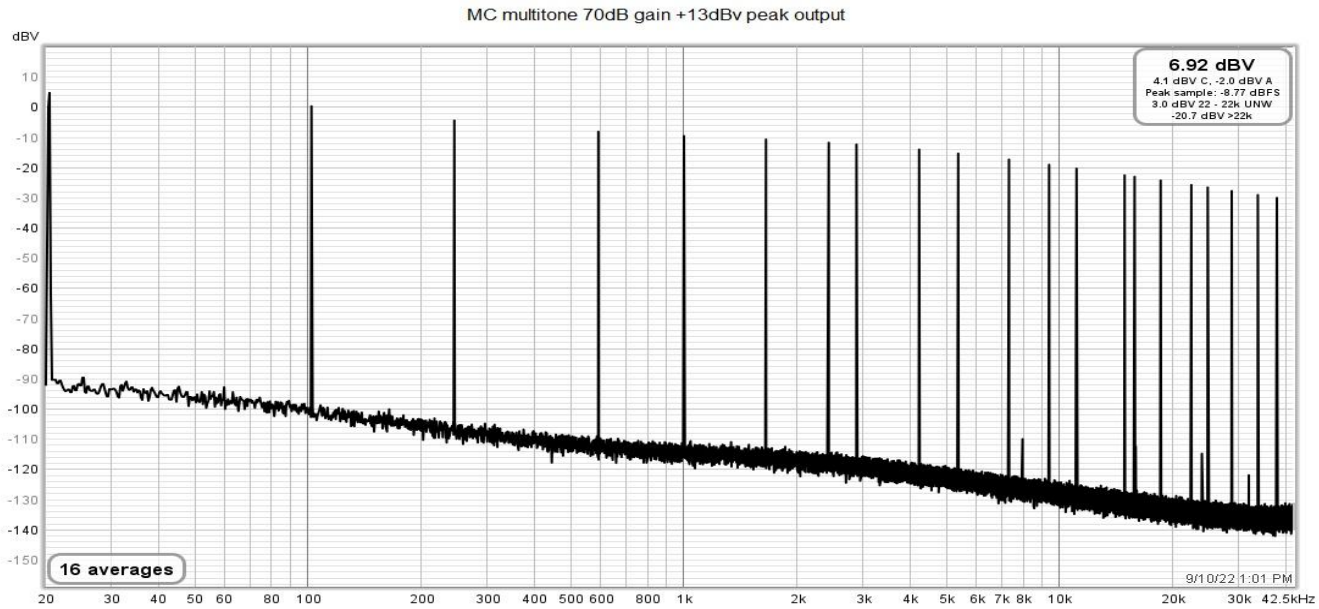
First, the MC amp, including phono stage, the switched out warp filter, and the gain block. Measured at the balanced outputs, with a 10 dBv full scale output. All measurements normalized, where needed, to a peak level of +13 dBv (about 9 v p-p).

Measurements were made using an RME ADI-2 PRO FS R BE ADC/DAC and custom attenuators.

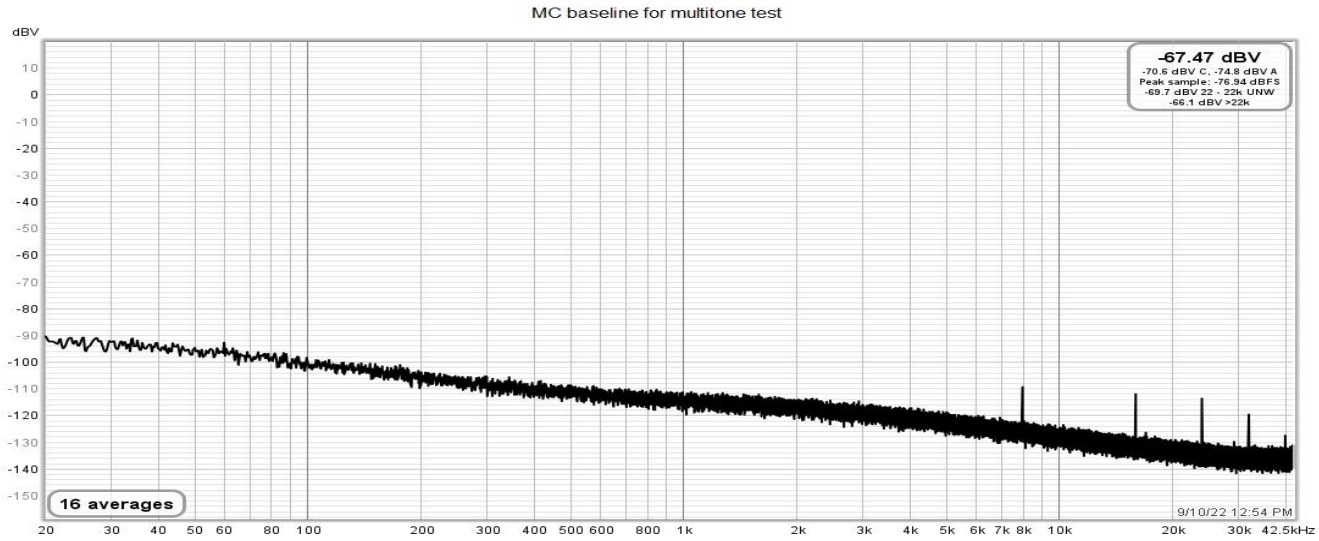




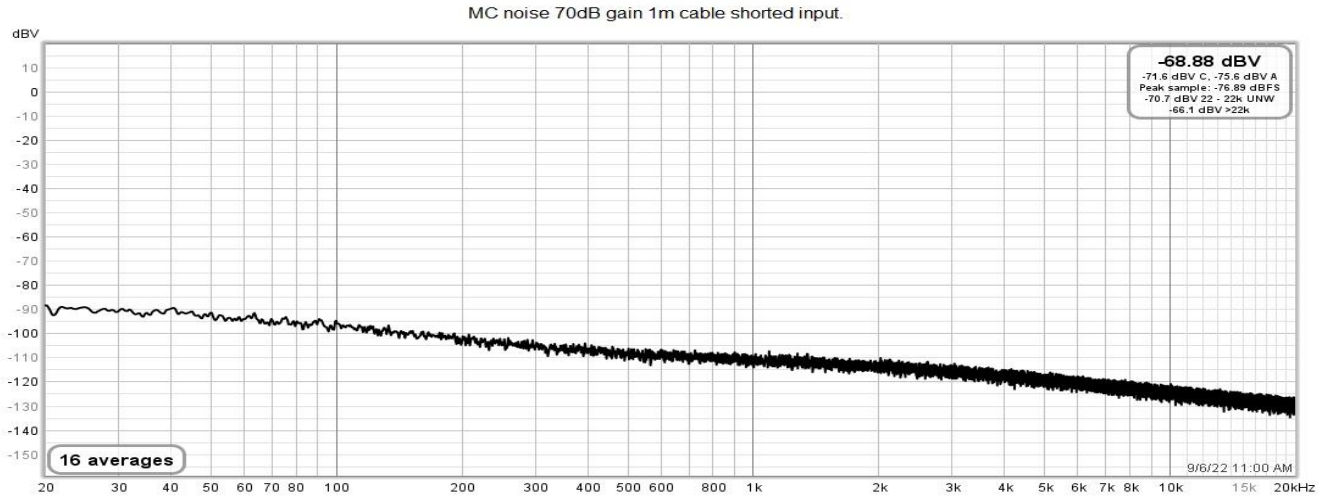




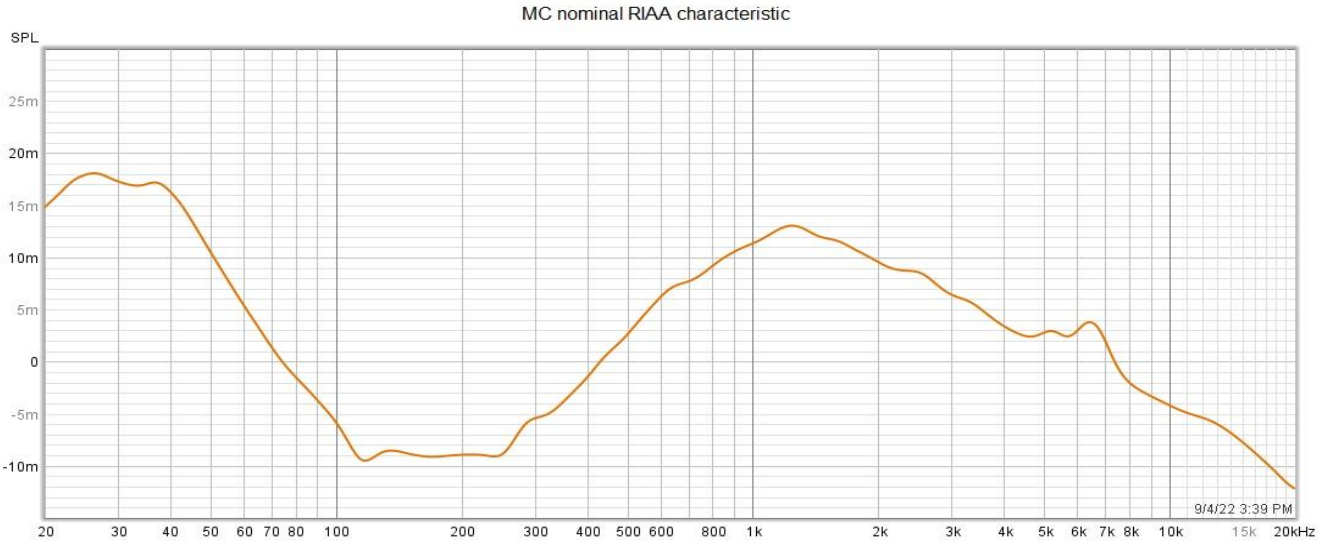
The multitone test was configured to closely match the 1/f spectrum of music, while matching the peak output signal level of the standard single tone input. No measurable additional tones were produced, indicating a very high level of linearity in the presence of complex signals.



This measurement is made with the signal generator present. Some USB coupling exists.



This was measured with a shorted 1 m input cable and no signal generator. No hum was present.

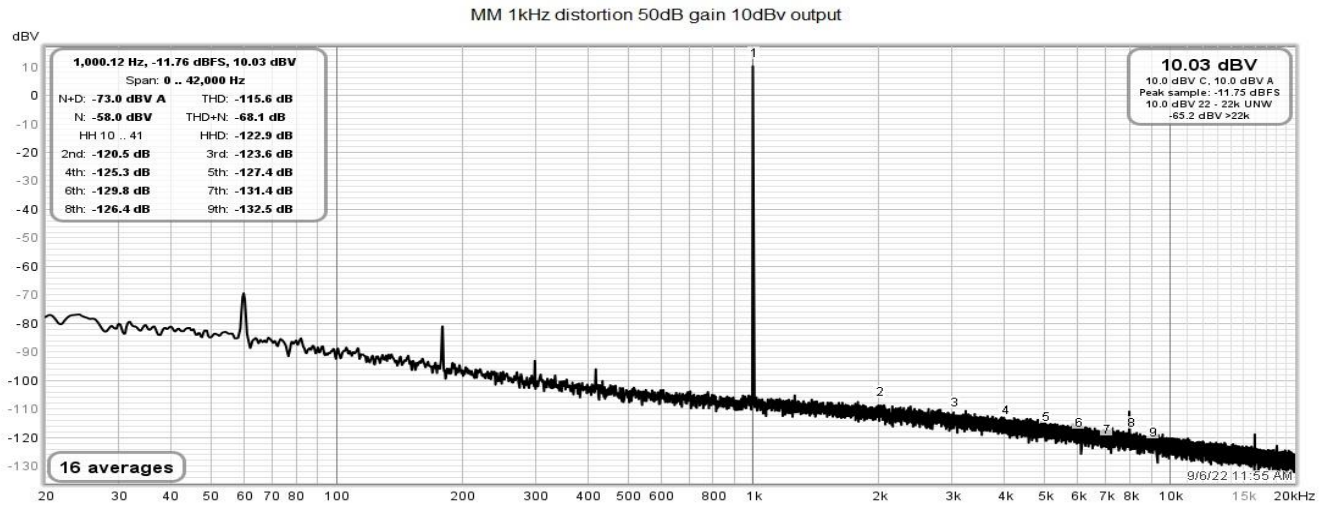


The above measured RIAA characteristic has about 0.032 dB p-p variation over the audio band. As can be seen, the MC stage meets all of the distortion and RIAA compliance target values.

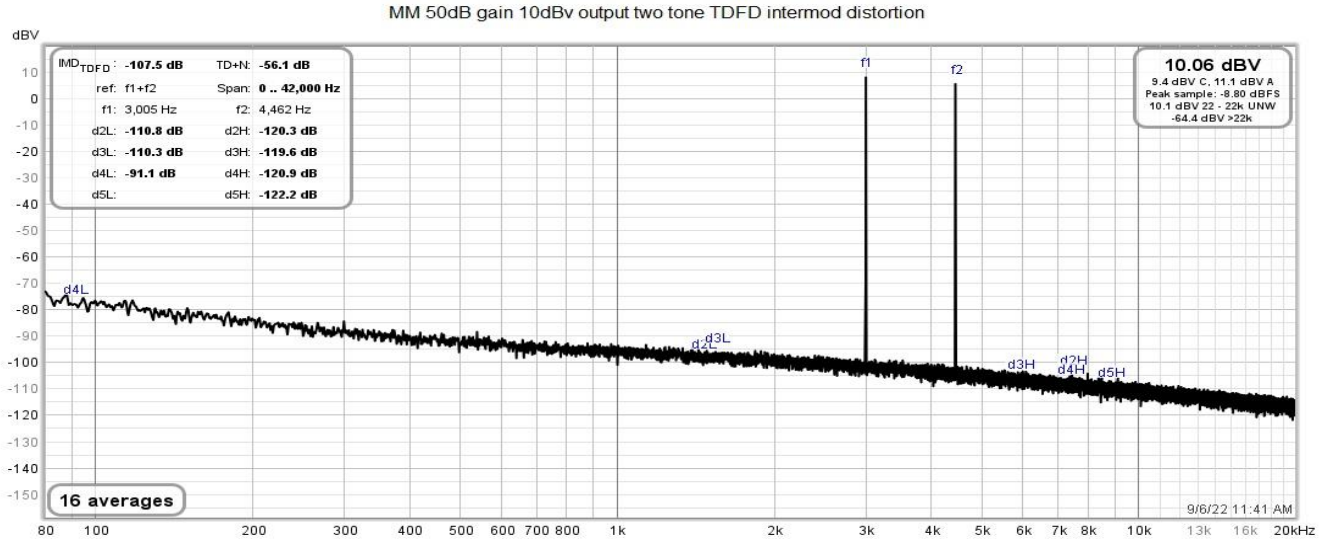
The A-weighted output S/N ratio is measured at 85.6 dBA with no measurable hum components.

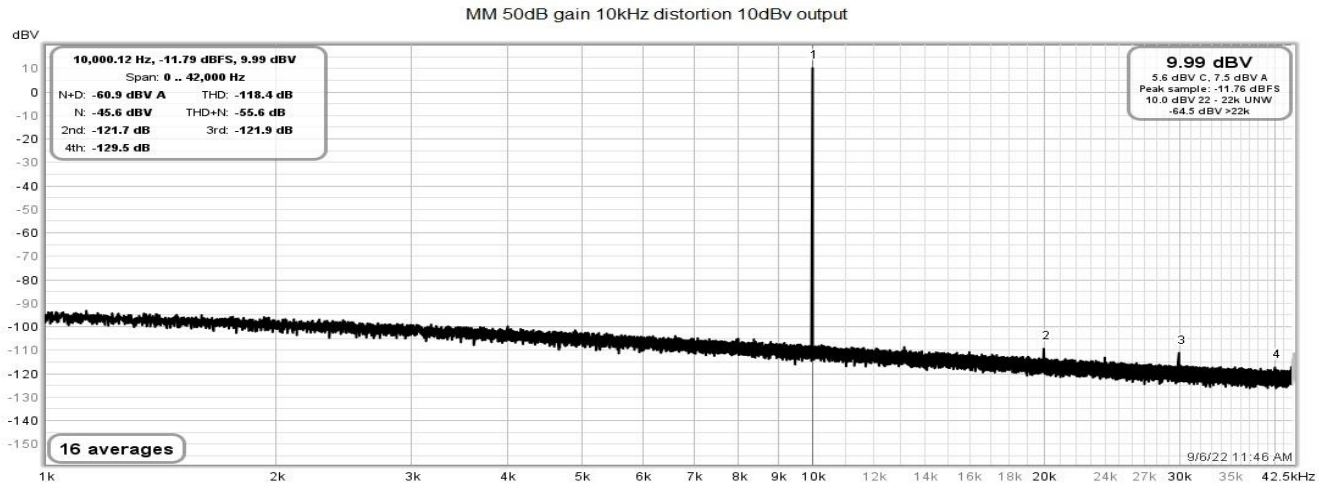
The simulated noise result, using our “optimized” opamp models, ideal resistors, and with the supply regulator noise modeled, was 86 dBA, which is extremely close to the measured result.

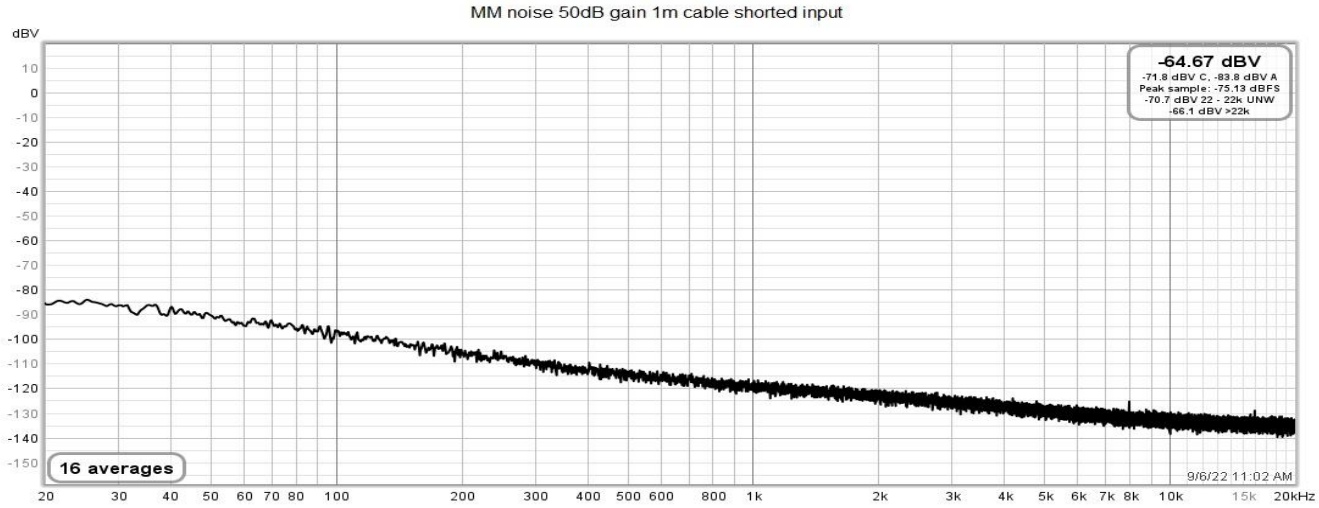
Now, the MM stage. Measured under similar conditions to the MC stage.



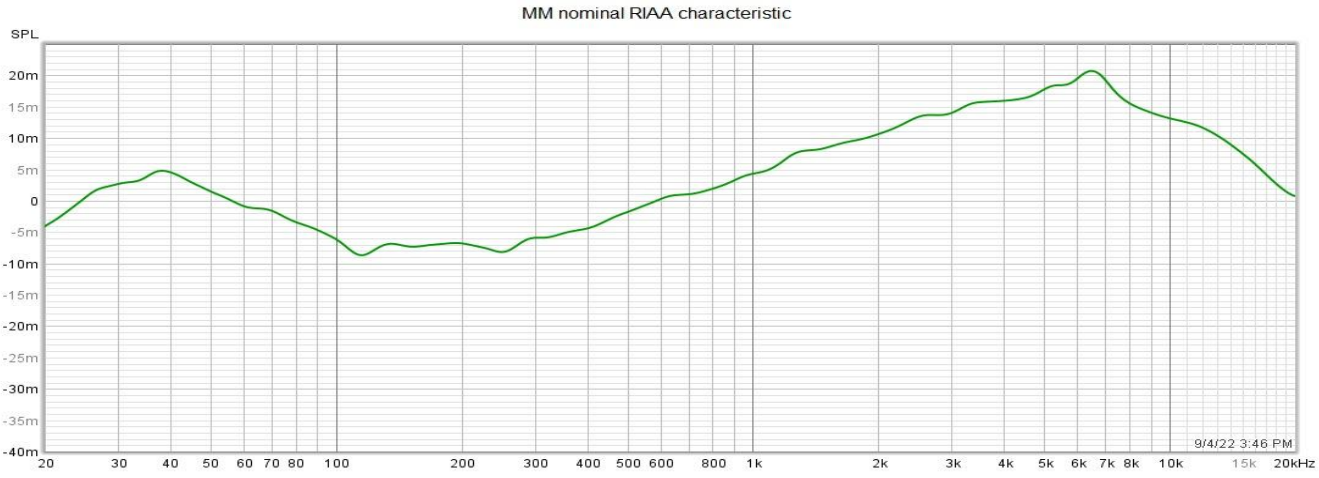
The hum components are due to ground loop issues with the signal generator/digitizer.







No hum components present.



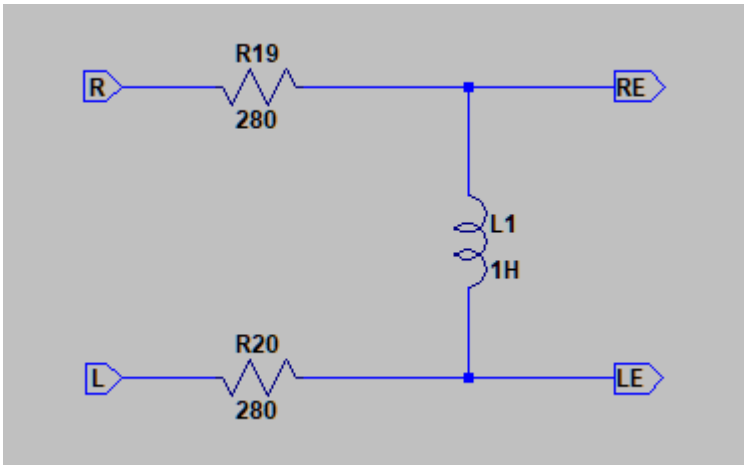
The above measured RIAA characteristic has about 0.030 dB p-p variation over the audio band. Like the MC the MM meets all of the objectives for distortion, noise and RIAA compliance.
The MM S/N ratio with an input short was measured at 93 dBA which is several dB better than the simulated value of 85 dBA.

The “warp” filter

The warp filter implements a function that comes and goes in phono stage designs, almost faddishly. It is intended as a substitute for the more or less useless rumble filter that many designs have.

It recognises that modern turntables do not exhibit rumble to any extent, and that the most significant low frequency problem with vinyl playback is due to LP warps that cause vertical displacement of the stylus and the resulting periodic low frequency “bumps”. This can be greatly reduced by selectively filtering the low frequency vertical component of the cartridge output.

How this operates is simple. It emulates the operation of the iconic “elliptic filter” that was long ago introduced in the vinyl mastering process.



An “elliptic” mastering filter.

The idea is to combine the LF components as mono (lateral) modulation while keeping the HF components as stereo (vertical) modulation. It can also be implemented by replacing the Rs with Cs and the L with an R.

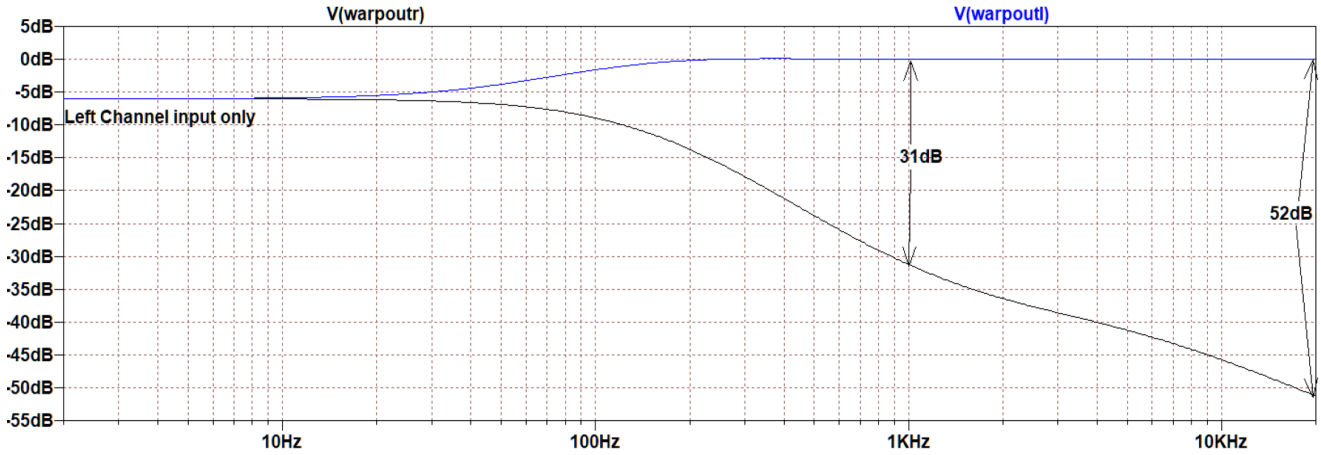
However, the complexity of the Acrux design has been significantly increased from the simple RL or CR prototype, and it now uses a second order active filter and a differencing circuit.

There were a couple of reasons for this change. Firstly, the RL or CR filter, when designed with a 90 Hz cutoff provides 33 dB rejection at 2Hz, and 26 dB channel separation at 1 kHz.

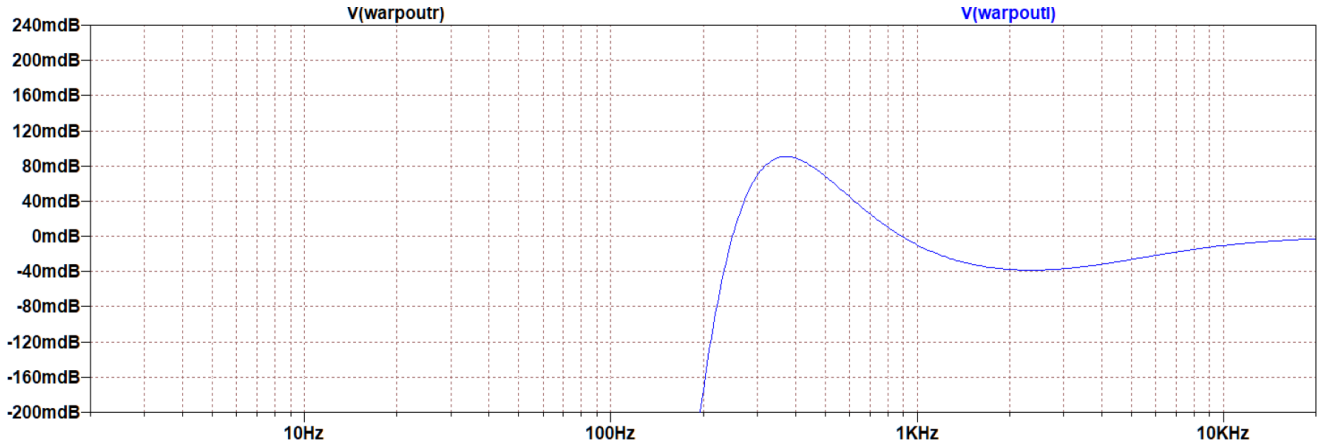
This channel separation is below the target value of 30 dB which is what high quality phono cartridges provide, and the more complex circuit increases the separation to greater than 30 dB at 1 kHz and above.

Secondly, the more complex filter requires additional active gain stages. The new circuit subtracts the Low Pass filtered unwanted signal from the total signal rather than operating in series. This ensures that the filter does not contribute significant distortion and noise to the desired signal in the critical 300 Hz and above region.

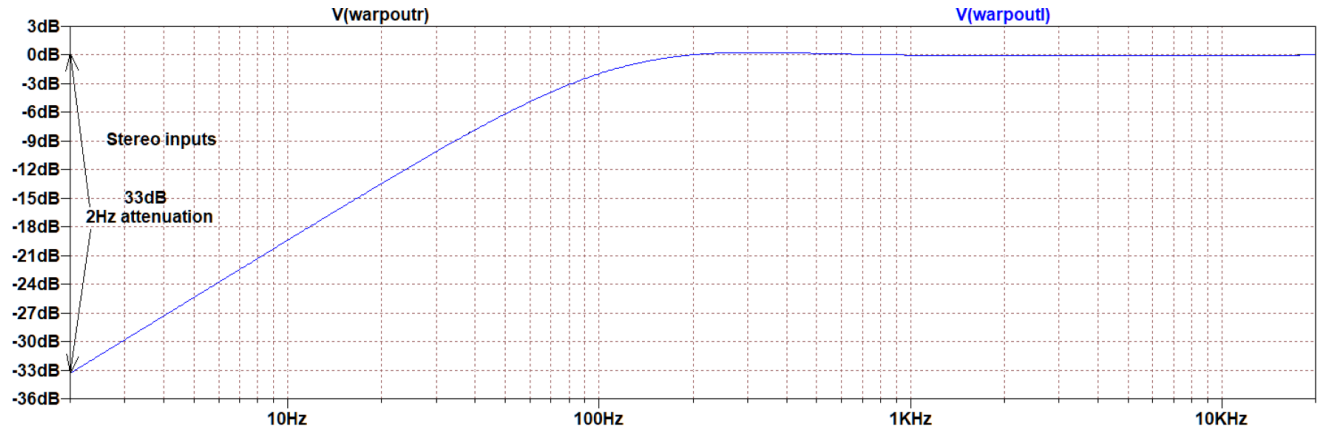
Below is a plot showing the simulated stereo channel separation for a single channel input.



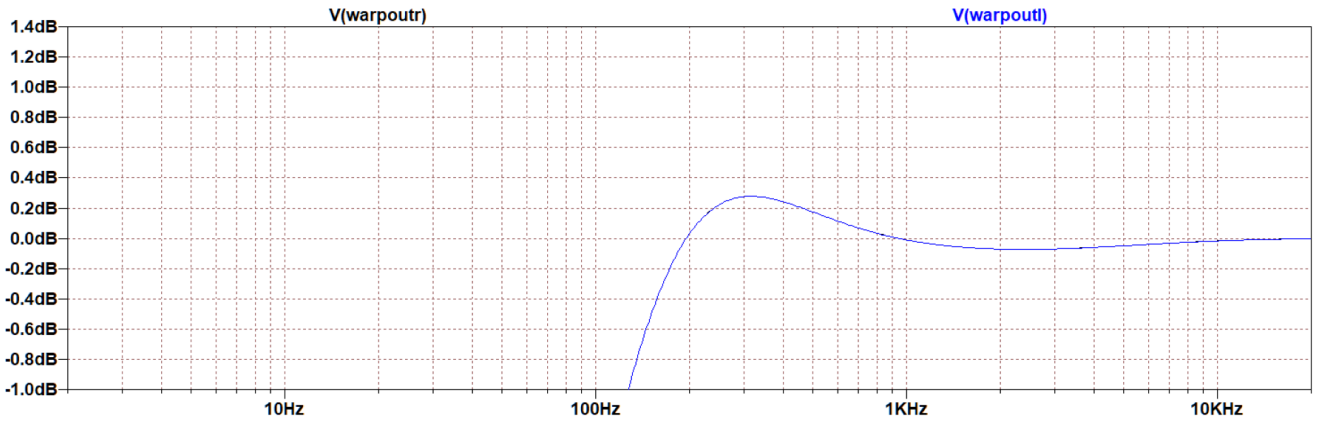
The circuit also lowers the frequency of cut off where the gain roll off crosses the audibility threshold of about 0.25 dB p-p variation, from about 360 Hz to 180 Hz, at the expense of a small amount of added ripple in the frequency response.



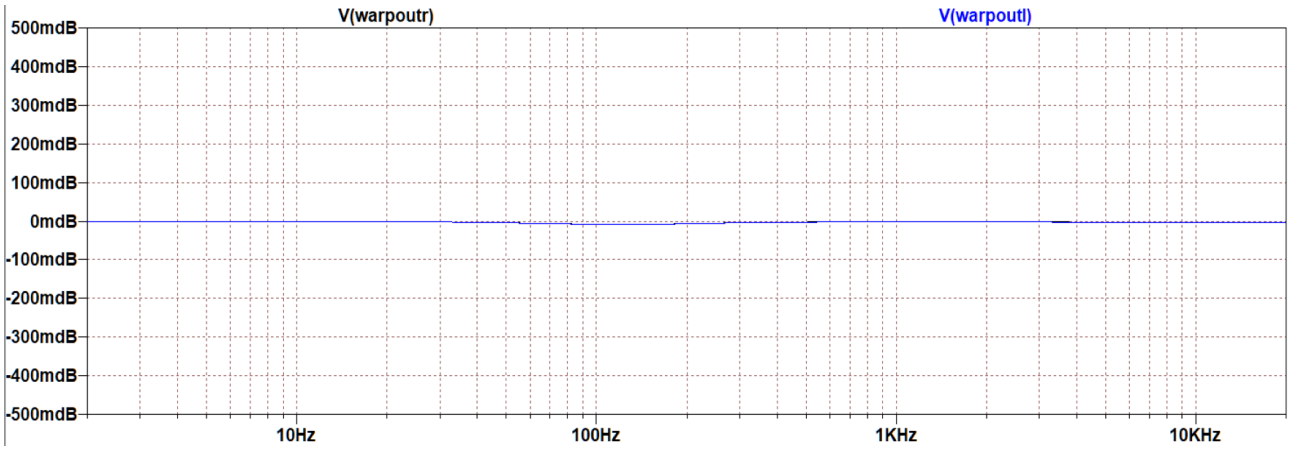
Above is the ripple in the single channel input warp filter out. It amounts to about 0.125 dB p-p.



The above shows the LF rejection of the warp frequencies, starting at 2 Hz.



The above shows the ripple for low frequency vertically modulated signals at about 0.3 dB p-p.



The above shows the lateral modulation frequency response. There is essentially zero frequency response error.

All of these elements in combination ensure that the warp filter is extremely acoustically transparent when in use, even when large amounts of warp related infrasonic signal is present at the input.

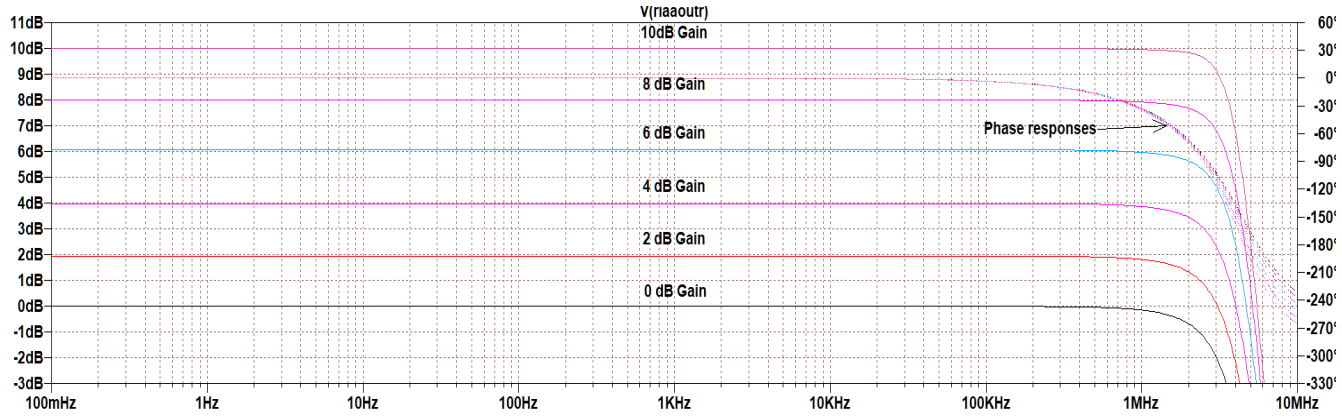
This feature has turned out to be surprisingly useful in eliminating unwanted woofer excursions and unnecessary, and potentially harmful, loading spikes on power amps with low frequency cut offs well below the audio band.

Some users just leave the feature on permanently and can detect no negative effects.

Gain Stage and Balanced Output Driver

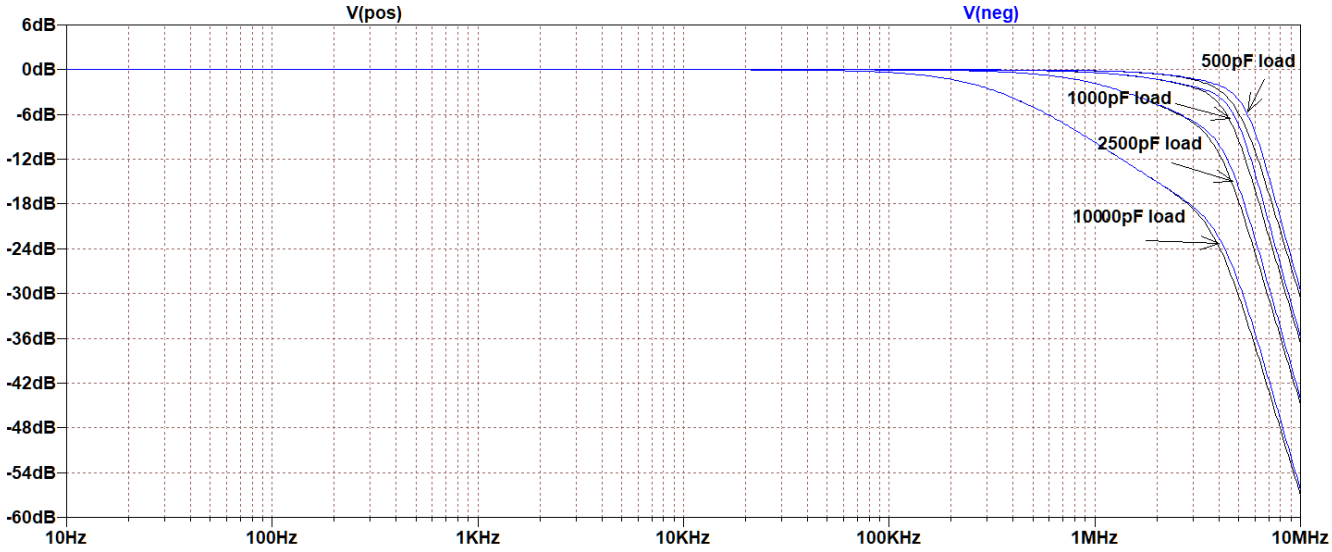
The Acrux has a six step gain control, incrementing from 0 dB to 10 dB in 2 dB steps.This block is placed between the phono stage and the balanced output driver.

The stage does not measurably degrade the S/N ratio or the distortion, but does degrade the overload margin by whatever the gain setting is, so it should be used with caution.

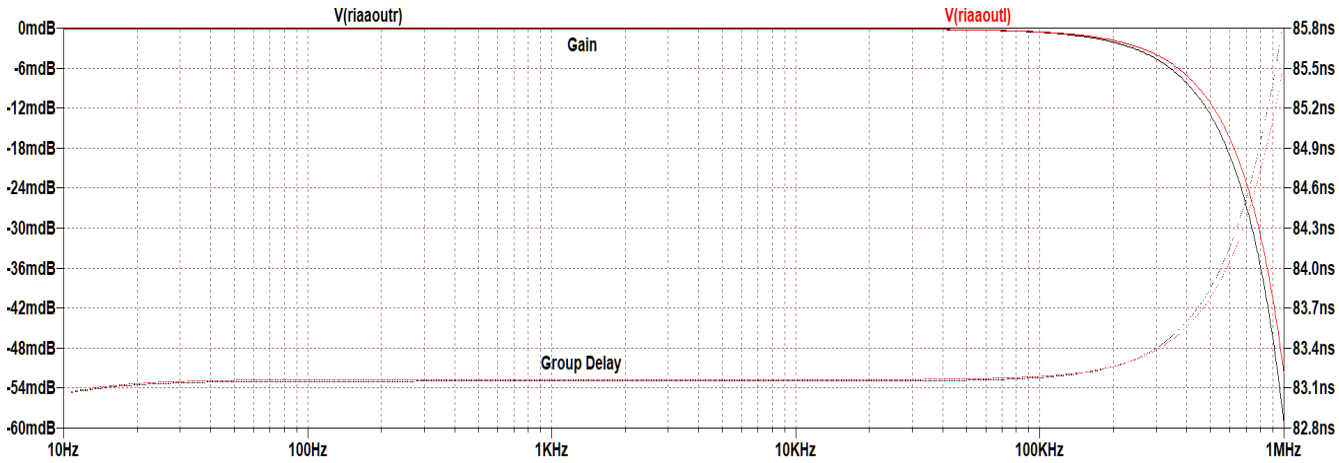


The balanced output driver provides an additional 6 dB of gain into a balanced input receiver. The positive phase is used as the single ended output and does not provide the extra gain. The stage provides closely balanced outputs which are insensitive to the termination, remaining stable into any capacitive load and able to drive 600 ohms with some degradation of distortion.

The output impedance is about 50 ohms/side, and the complementary driver paths have well matched gain and group delay characteristics up to frequencies well beyond the audio band.



Above is the balanced driver responses into capacitors varying from 500 pF to 10,000 pF. The output bandwidth declines as the cap load increases, but the stage remains stable and the performance in the audio band is uncompromised.



Above is the gain and group delay for the single ended gain stage output to the two balanced output phases driving a 10k ohm differential load. As can be seen the matching between the two phases is excellent.

Conclusions

The design of the phono stage embodies recent understanding in the area of psycho acoustics. It employs extremely high levels of negative feedback in order to reduce distortion levels across the entire audio band below the recognised thresholds of inaudibility.

This ensures that the “ear” does not hear anything that is distinguishable from its own self-distortion, while contradicting the common audiophile fear of negative feedback.
In effect it demonstrates that the problem with negative feedback is not that there is too much but that there is too little.

The design also reduces RIAA frequency response errors below the audible threshold of ±0.1 dB p-p while integrating a DC servo loop.

The DC servo loop reduces the output DC level below 5 mv and avoids the need for large coupling capacitors without audibly disturbing the signal in the audio band due to added distortion or peaking.

The overall RIAA accuracy and focus on wide bandwidth low dispersion designs in all stages also ensures that the design is extremely “temporally coherent” within the audio band and does not significantly contribute to the coherence limitations inherent in the vinyl record production process.

The design also recognises that MM and MC cartridges have very different amplifier noise sensitivities and must be dealt with accordingly. What works best for reducing voltage noise does not work best for current noise and vice-versa. It is essentially impossible to optimize a single input stage that will work equally well with both MM and MC cartridges without having extremely large input FETs in a discrete gain stage, which comes with its own set of compromises and problems.

The Acrux avoids this conundrum by utilizing separate input stages for MM and MC cartridges and taking advantage of the availability of devices which best serve the needs of the two different channel types. It also parallels up the input stages to obtain excellent noise performance for all classes of cartridge.

The power supply was designed in tandem with the phono gain stages to produce extremely low hum and noise by understanding the interactions between the two.

Taken all together, the result is a phono stage that is extremely transparent, neutral, and dynamic that sounds as good as it measures.