# Part 2 - The Ovation 'Symphony' High Fidelity Preamplifier

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October 2014

# 1. Headphone Amplifier

Anyone who has listened to music through good headphones  will know that they can be exceeded only by a few high end speaker systems. I've owned quite a number of headphones over the years, with my first real hi-fi phones being the Sennheiser HD414's, bought in about 1978 or 1979 and noted for their remarkably open, transparent sound compared to other offerings at the time.  These were originally launched in about 1968, and changed the headphone landscape forever – there's still a loyal following and even a Facebook fan page, but they don't quite make the grade up against the better 'phones of today.

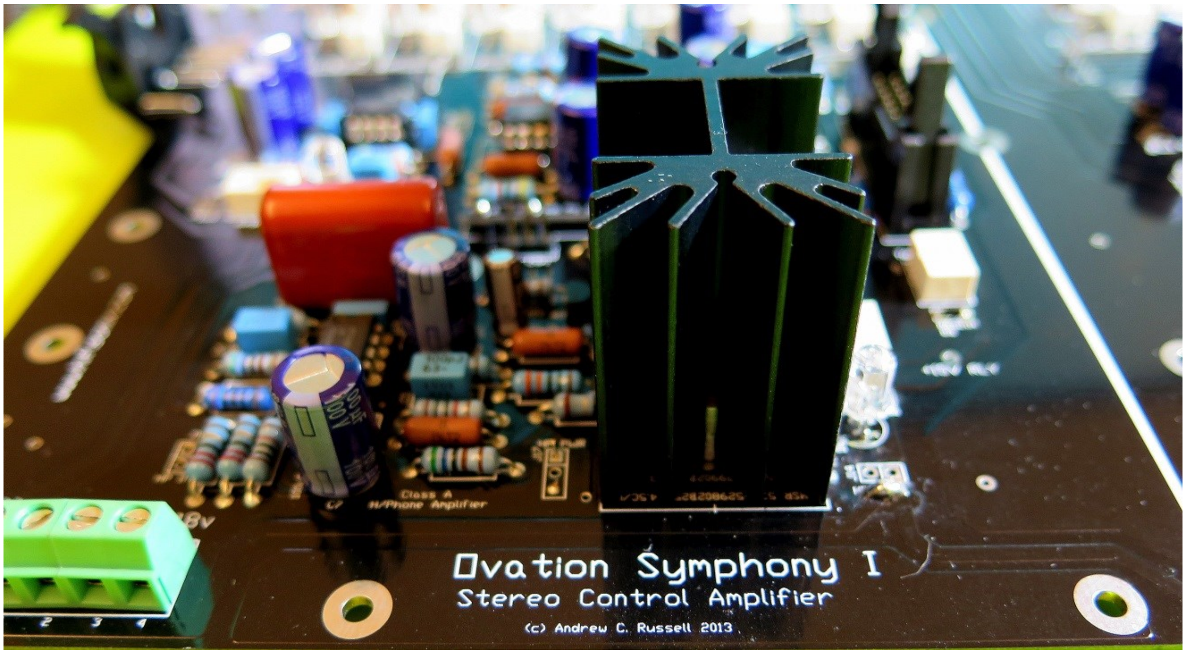
Later, I owned Beyer Dynamics DT212's which I found too bassy and too hissy in the top end, and more recently an assortment of Sennheiser, Bose and Sony noise canceling types. The Bose


noise cancellers are well made, comfortable and great for long haul flights – but they are most decidedly not hi-fi 'phones.  The little [Sony in ear devices] are the clear winner in both sonic and noise cancelling terms –


although they can get uncomfortable after a few hours. As far as high fi headphones go, one of the best, which I had on loan for about 6 months, was a tube based [Stax ESL] set-up, used as the application lab reference phones while I was based in Tokyo. I'd describe the sound as open, very relaxed and neutral with a smooth, sparkling top end. They did not go loud, and because of the open back construction, offered no attenuation of background noise. If I had a criticism, it would be the bass, which was a bit 'loose' for my liking.

Another pair I owned  that were really surprising were a Koss model targeting sports users which, rather than an over the head band, sported a neck band.  These had a sweet, open sound with great bass.  Unfortunately, they were badly designed and manufactured, and the clips on the driver capsule that attached to the band,  broke off after about 3 months and the cable sleeve perished. This brings me to my current every day headphones, the Audio Technica ATH-AD900. I bought these in Tokyo in

2011, after learning my lesson with the 'su per bass' Bose

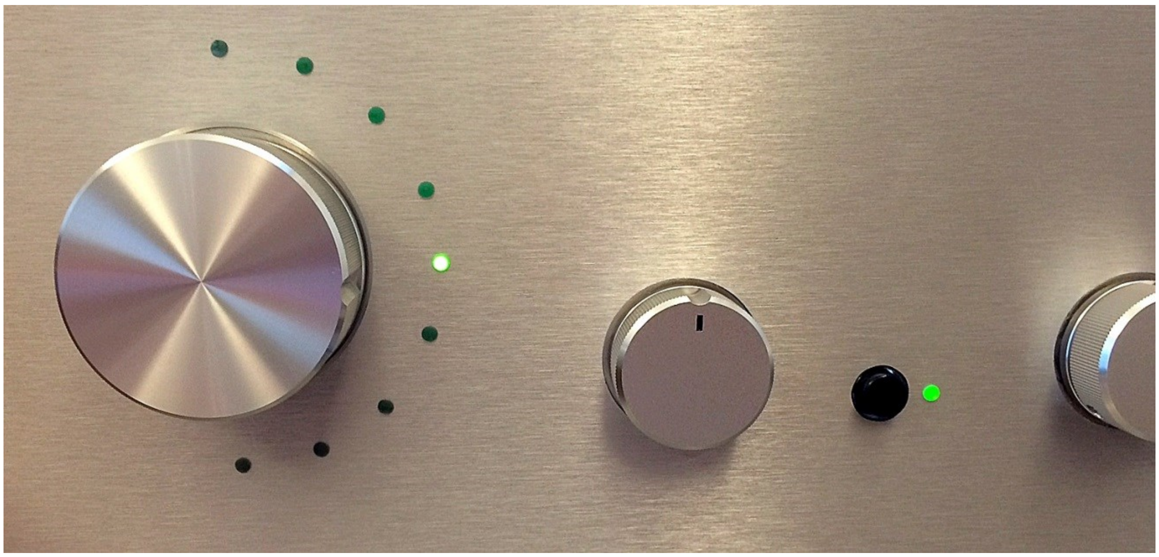



model I mentioned earlier, spending a morning listening to various brands, carefully comparing the sound vs the price. There were a more expensive pair of AT's at about $650 that I liked, but eventually I settled for the 900's at about half the price. These are, to my ears, fantastically open and clear and the user feedback around the internet at the time seemed to confirm this. However, the bass in some cases could do with some reinforcement – a point picked up in a few of the reviews, and this of course is where the tone control can help. When buying headphones, even more so than speakers, you really need to take a few hours out and listen, because some of the vaunted brands and models just do not cut it in my view and in a sequential (i.e. swapping the headphones out one after the other) some sound decidedly boxy, others have overblown bass while quite a few seem to emphasize frequency extremes at the expense of the midrange. And then of course, there's personal preference to consider.

Headphone sensitivity varies considerably from 80~100 dBm with around 90 dBm (figures quoted for 32 Ω) typical for most brands. I was not able to find any definitive study listing the spread on this spec but one can quickly get very anal on something like this. The bottom line is you should at least be able to hit 105dBm peaks with any 32 Ω Headphone. A rock concert is 115 dB SPL – I would not recommend anyone listen at these levels for extended periods: the maximum recommended exposure at 115dB SPL is 7 minutes, while at 105dB SPL its 1 hour. If we assume a very worst case headphone efficiency at 80dB/mW, we will require ~ 100mA and 3V to hit peaks of 105dB SPL, so this was the design target I set for the Symphony, assuming 32 Ω headphones.

The Symphony headphone amp can deliver around 2 W in class A into a 32 Ω load, and is powered off +-15V rails.

This headphone amplifier thus has enormous head room, and it's reflected in the very open, unconstrained sound when you compare it to something like a iPod, or other battery powered devices. For a more realistic 90dBm sensitivity headphone, at ~300mW input power to the 'phones, the achievable SPL is 115 dB – more than enough in my view. I did some listening tests, playing music very loud on the prototype on my ATH-AD900's, whilst monitoring the output with a scope and the absolute peaks never exceeded 1V, with something in the 100 to 300mV being more the norm.



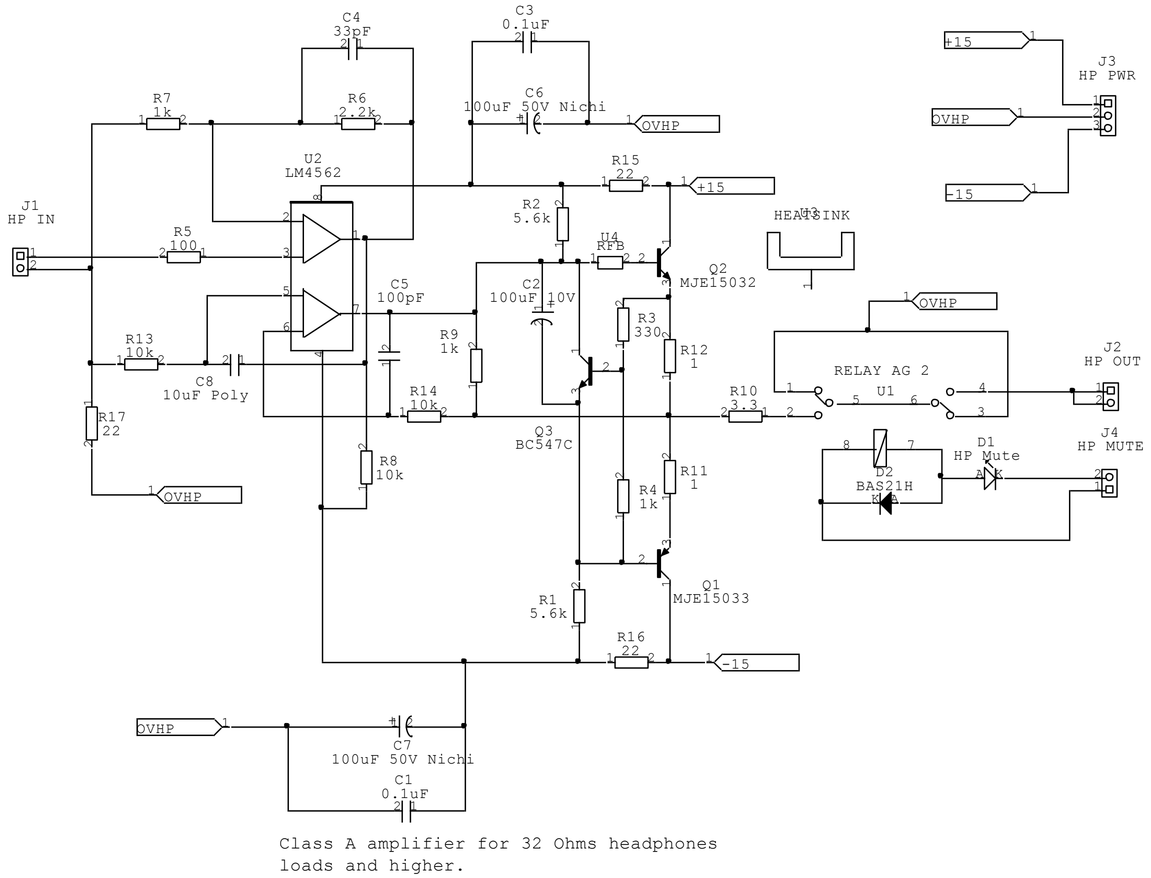



**Figure 19 - Symphony Preamp Class A Headphone Amp**

Fig. 19 details the class A headphone amplifier circuit. An LM4562 dual opamp forms the core, with the first half amplifying the line level signal by ~11 dB or 3.2x, set by R20 and R21, with C4 providing some HF response shaping. This is then coupled through C8 (10uF Poly cap) to the second stage – I did this because I did not want any DC offsets present (which are in the 100's of µV *worst case* at the line output) causing offsets on the output of the HP amp. R13 sets the lower -3 dB response at < 2 Hz. The second opamp drives a class A complementary push pull output stage (Q1 and Q2) with Q3 BC547C, the bias controller, setting the output stage standing current at ~140mA, while C2 acts to decouple Q3 at HF. The RFB is included because feeding the OPS from a low source impedance is important for the very low distortion I targeted and which I confirmed in my sims; I did not want to run the risk of HF oscillation – a potential issue on follower stages and class A push pull configurations like this where the devices have high fT's – the MJE15032/3 are up at 30 MHz and the presence of any excess inductance in the base and collector circuits can lead to problems. Like all the other opamp stages in this preamp, R8 and R9 keep the opamps in class A. The entire HP stage is very well decoupled to keep HF signal currents off the main supply rails, and to ensure the opamps don't run into any supply related parasitic oscillation. Relay U1, like the line output mute relay, is controlled by a delay circuit on the PSU board and ensures there are no switch on or off plops when the preamp is powered up or down. The distortion levels in this design are very low. Into 32 Ωs at 1V out at 20 kHz it simulates at about 600ppb – clearly as a result of that class A output stage - rising to about 2ppm at 3V peak while at 8V out its <60ppm. However, beyond 3V even on 80 dB/mV 'phones, you would have some serious ear ache. R15 and R16, the filter resistors also act to limit the maximum current to around 640 mA and are designed to offer some basic protection – nothing more.

# 2. Pulling All the Pieces Together

Figure 20 shows the top Level circuit for one channel. Since we have already discussed the different circuit blocks in detail, I will only cover a few key remaining points in this section.

***Relay switching.*** Firstly, all the input select relays are fitted with flywheel diodes located directly across the relay coil pins. With the prototype, there was a small but annoying click every time I selected an input with the volume control set to max, despite the presence of the flywheel diodes right at each of the relay coil terminals. A bit of sleuthing revealed that the fast edges, and contact bounce on the switch as power was released and then applied to the next relay coil generated fast rise time signals that coupled into the input signal bus. When the relay was released, the current spike through the flywheel diode also generated a click, albeit at a much lower level. The front panel mounted input selector switch was fitted with a 220 Ω resistor in series with the wiper, and directly across every relay coil I fitted a 0.1 µF 1206 SMD ceramic capacitor so that each coil sees an RC network of 220 Ωs and 0.1 µF as it is switched. The RC network and flywheel diodes as shown completely quench any audible switching noise. It should be noted that the connection from the front panel switch to the select relays is about 40 cm all told and made through a ribbon cable to the left hand edge of the main PCB – so there is an opportunity to radiate noisy switching events into the front end circuit if precautions like this are not taken. The lesson here is if you use relays for signal selection, you need to take precautions with the coil switching transients, and, if using a mechanical switch as I have done here to actuate the relays, the associated contact bounce.

Self discussed 'offness' in his 'Small Signal Analog Design' and you can see that capacitive signal coupling across the relay contacts can cause problems. To solve that problem here, as mentioned earlier, I use both relay contacts in a back to back arrangement so that when not selected, both the contact wipers connect to ground. This is extremely effective in preventing any signal bleed through and at full volume with music playing on one of the adjacent inputs, there is absolutely no audible signal on the speakers or headphones.

Its very important that the supply rail to the relays is clean, and as discussed in the PSU section, the relay supply (~15 V, of which 3.5 V gets dropped across LED's and the 220 Ω RC resistor) is fed from a ripple eater to prevent any noise from the relays as the coils are switched, getting onto the main supply rails. If you are targeting better than -110 dB noise performance as I am here, it will only take a few µV of induced noise/hum to cause a problem. As a general recommendation, small signal switching relays must always be fed from a quiet, well regulated supply to avoid coupling noise into the input signal via the relay coil.

***Record Buffer Amplifier.*** The circuit around U11 buffers the line in bus to drive the record loop. Most classic amplifiers never buffered the tape loop, which I always found strange, since I could clearly detect a change in the sound when the tape loop output was loaded and was switched in and out with the levels on the tape recorder carefully balanced. Someone remarked that today tape loops are redundant because everything is digital and 0 dBFS is 2V. I won't argue, but included the loop just in case I ever need it. Besides, if you are going to hang a digital recorder onto the output, the buffer isolates it nicely from the preamp line-in bus which is important. The output is tied to the –ve rail with a 4.7k resistor, so this buffer will drive ~5k in class A – heavier loads will cause it to transition to class AB. As with all the other stages, heavy localized decoupling and filtering keeps noise off the rails and the opamp operating stably.

***Input Attenuator.*** This consists of two 18k 1% metal film devices in the upper leg and a 1k in the lower leg of the divider for an exact 20 dB attenuator. If the link is placed between pins 1 and 2 on the header, the divider is bypassed, while if its inserted across pins 3 and 4, the attenuator is activated. On the Phono equalizer input (still to be designed) via relay U10, there is no attenuator as the output will be designed for ~200mV output

R33 and R17 were included at the output of the tone control buffer (U16) in case exact signal matching was required. This usually only happens if the tone control is not 0 dB when set to the flat position and I found during testing that is was absolutely 0 dB and flat across the audio band to the point that when switching the tone control in and out in this position, there is no audible difference. Further, these two resistors, along with the 22 Ω ones on the amplifier modules provide isolation from capacitive load.

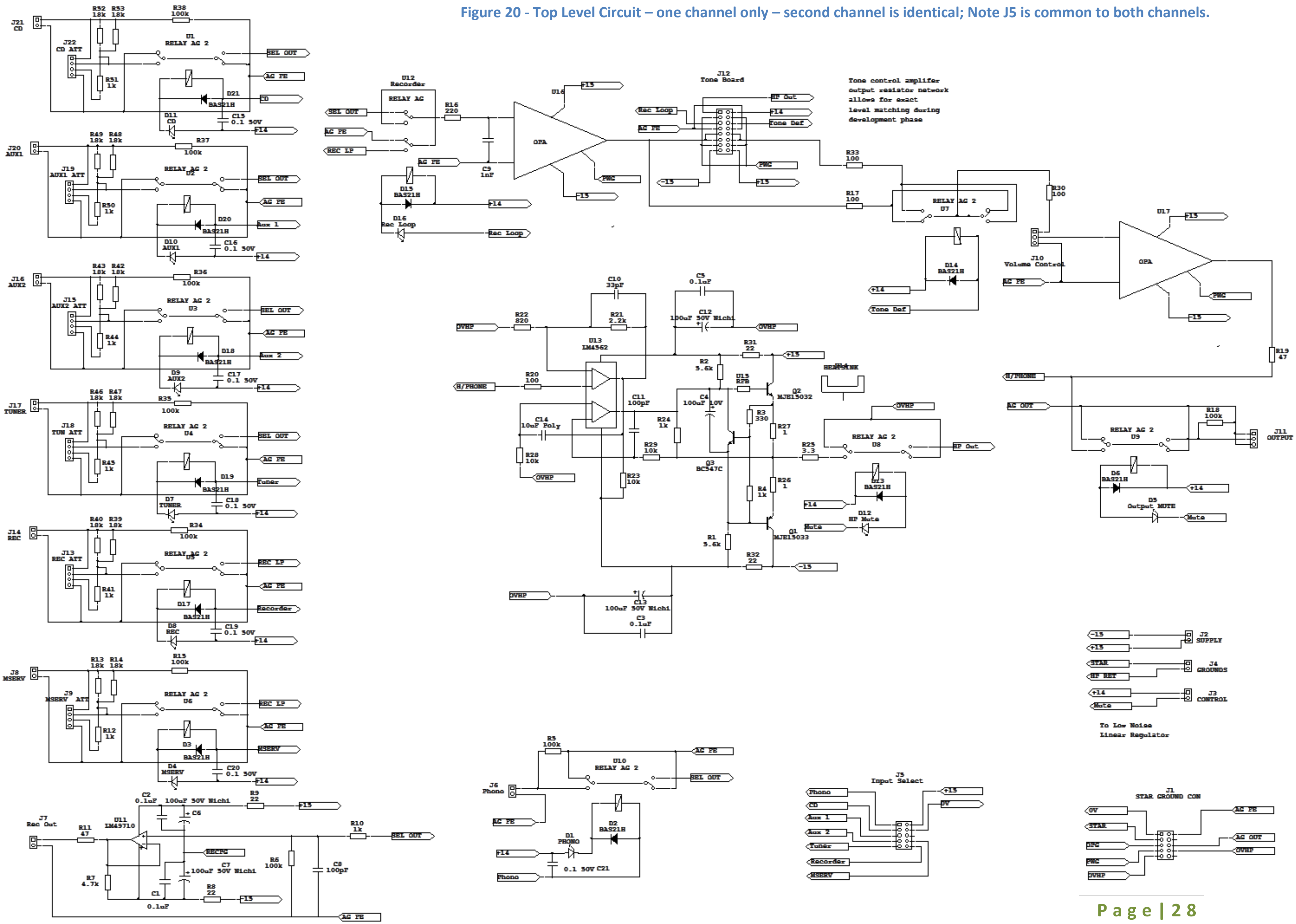
***Connection to the Tone Control Board.*** This is made using a 16 way ribbon cable assembly (one for each channel) which carries the +-15V supply rail, analog ground, power ground, the tape loop and the tone mute relay control signals. This approach gives a lot of freedom in the mechanical design, allowing the tone control assembly to be positioned more conveniently on the front plate. This would not be the case if the bass and treble were on the main board, and would probably have necessitated shaft extenders, Teflon front plate bushings and so forth. Furthermore, with the arrangement as shown, when the tone control is bypassed, there are no left-right channel tracks in close proximity and channel separation is preserved. When the tone control is switched in the channels only get physically close at the control potentiometers. Of course, I kept the supply rails to each half of the tone control separate as well – the left ribbon cable only carries left channel power, and the same holds for the right channel.

Also located on the board are the push button switches for the tape loop and the tone bypass relays, where I deployed good quality C&K devices which have a great positive tactile click to them.

For the tone control potentiometers, I had a choice between the Alps RK09 devices, which in physical size are similar to the Spectrol types used in Self's Elektor preamp, or the more upmarket and better Alps RK27. Naturally, I chose the RK27's – I've used one on the X-Altra Mini and in terms of tactile feel, they are really smooth with just the right amount of rotational torque needed with an appropriately sized dial. These are 5k linear taper devices with a central detent and I had to specially order them from Alps Japan – but, well worth it in my view.

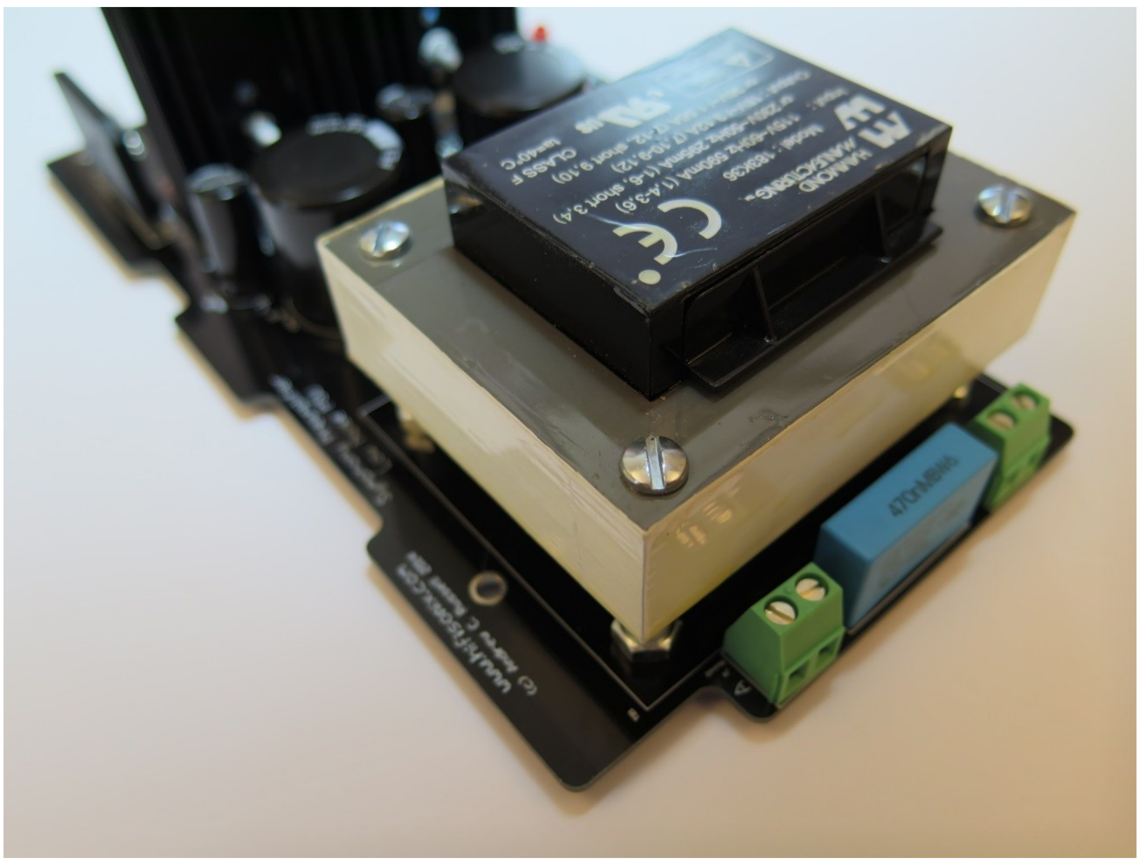
***Phono Equalizer.*** The phono input has no attenuator associated with it like the other inputs and will be on a separate board that will have both the MC and MM circuits. However, only one set of rear panel phono input sockets is provided, and MC or MM must be selected via a jumper on the equalizer board.

Figure 20 - Top Level Circuit – one channel only – second channel is identical; Note J5 is common to both channels.



# 3. Power Supply Design

Power supplies for audio preamplifiers need to be stable and quiet. A lot has been written about power supply design for line level stages.  On one end we have the very convenient, reliable and cost effective LM317/LM7815 type regulators, and on the other, Walt Jung's Super Reg and derivatives, which pop up in all sorts of DIY designs, and sets the benchmark when it comes to low noise and regulation. There is also a section within the professional and DIY audio community that swear by shunt regulators, and I have seen designs featuring a heavy (say 300~500mA) constant current source, feeding multiple parallel 'shunty s', as they are affectionately known.

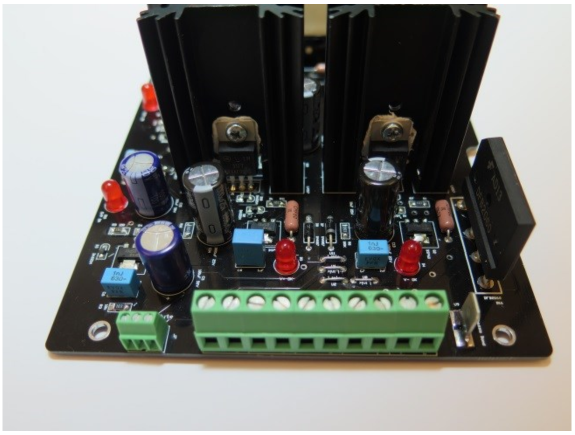


I have done quite a bit of modelling on LTspice and have come to the conclusion that the biggest issues affecting power supply performance outside of intrinsic device noise and load response are not the regulators themselves, but the passives around the regulators, trace inductance and layout. Kendall Castor-Perry did a series of articles in EDN looking at power supply performance,

with quite some focus on how the system behaves with real world components and layouts. I discovered through my own modelling efforts that if you model a simple super reg for example, PSRR and load transient performance are exemplary, until you add real world passives, trace inductances and so forth. What you end up with is less than stellar performance, despite lots of earnest engineering effort: for the most part it is misplaced and ultimately does not do anything for the performance beyond a certain point – the law of diminishing returns applies despite ones best intentions I am afraid. The requirements for low noise audio power supplies associated with line level preamps like this one are not the same as those for most other analog domains – e.g. instrumentation, video buffers, high speed A-D and D-A and there is a real opportunity here to rethink the approach from the ground up – which is what I did with the Symphony Preamp.

The quintessential linear regulators for small signal op-amp applications are the LM317 (positive supply) and LM337 (negative supply) IC regulators. With a typical output dual voltage of +-15V, you can expect around 450µV of wide band noise on each of the rails (yes, that is *very* noisy). If you follow the suggestion in the application note to heavily decouple the ref input pin to ground using a 10 to 100 µF capacitor, you can improve this by a further 20 dB or so, yielding a wide band noise figure of 45 µV. If you are prepared to decouple the ADJ pin using a really large capacitor – say 1000 µF – you can get a further 6-9 dB down on this, and here you are in *20 µV wideband noise* territory, which for most applications is not bad.

The 317/337 devices offer reasonable load an line regulation, thermal shutdown and current limiting and National Semi branded TO-220 1.5A devices can be found for about 60c (no doubt



lower price in high volumes). All-in-all, a pretty good bargain and no wonder many designers recommend them.

## Power Supply Noise Sources

**Regulator noise.** All semiconductors produce noise. In linear regulators like the LM3x7 series, this emanates mainly from the low voltage reference diode, the control amplifier inside the chip, and the feedback resistors. Very low noise regulators can be created using low noise op-amps and low noise references, but these quickly become complex, expensive and lack some of the protection features of an IC linear regulator. My earlier comments about final performance should be re-emphasized again here.

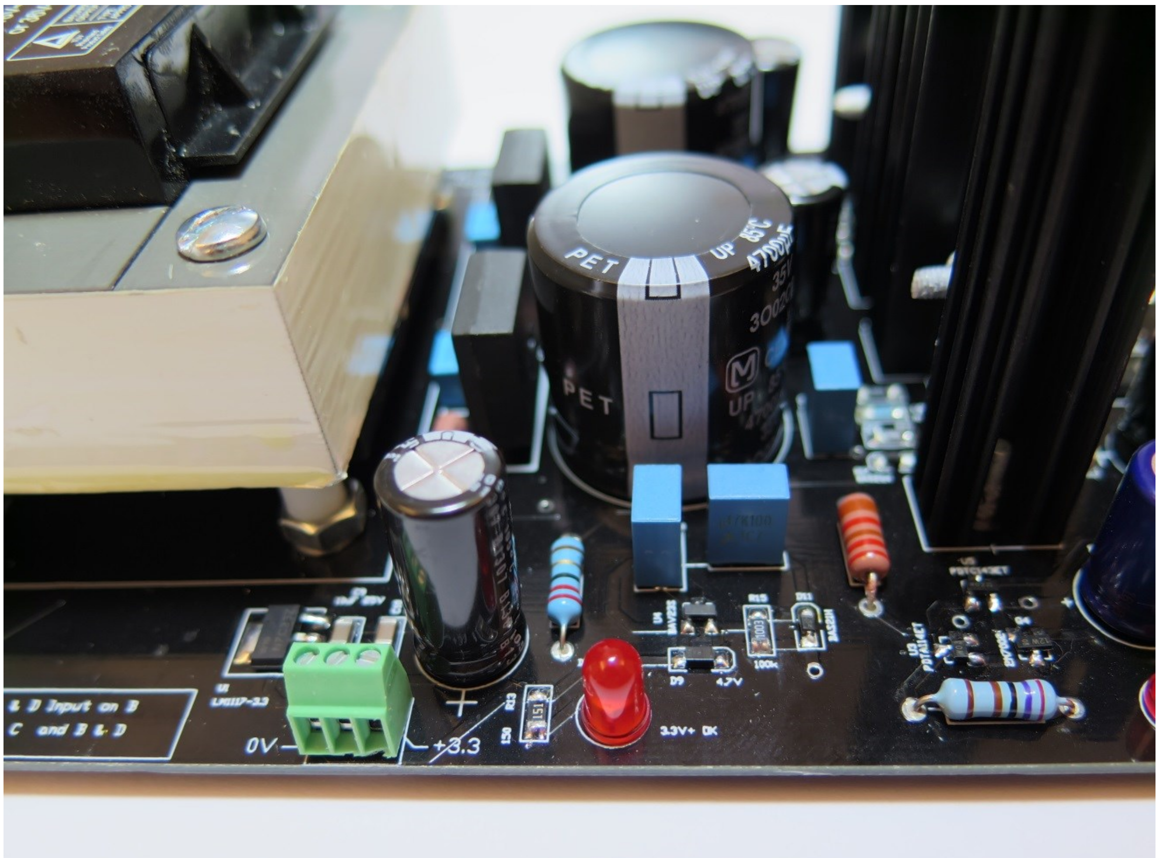
**Common impedance coupling.** Whenever AC current flows through a ground connection, it can introduce noise into any signal sources or feedback returns connected to it. The best way to mitigate this, conventionally, is to use a star ground, and upwards of 20 dB of ground impedance coupling isolation is easily achievable – so by way of an example, a very high 1 mV common impedance ground coupling noise problem can easily be reduced to well under 100 µV or lower. Bruno Putzey, writing in Linear Audio notes that this is a tough problem to solve without resorting to getting both the signal wire and its associated return off the ground – i.e. you need to float the ground return back to its source. Common impedance coupling also takes place in supply rails. If one of the loads attached to a supply rail creates a disturbance, others attached to that same rail will also see the disturbance. The trick to limit this, is to use localized decoupling.

**Magnetic coupling.** Any AC current flowing through a conductor generates a magnetic field, which then couples onto adjacent conductors. Screening, shielding and reduction of the AC component in the supply lines help. It is critical that the loop area between the driving source and the load be kept as small as possible to minimize coupling.

**Capacitive coupling.** Fast rising signal edges couple noise and extraneous signals into sensitive high impedance nodes. Best cure is to focus on layout, and minimize the number of traces carrying these types of signals – i.e. limit them to the signal paths only.

**Mains ripple.** Rectified and smoothed mains will always contain ripple voltage components at 2x the mains frequency (full wave rectification) along with additional harmonics and any AC load current at the PSU output and harmonics thereof. On a sub-optimally designed amplifier or preamplifier, driving their rated load, the supply lines will have harmonics from c. 100 Hz up to many 10's of KHz.

**Load and line disturbances.** Step changes in load and/or line voltages can only be corrected as fast as the regulator loop can respond. In a typical 3 terminal regulator, this is a few 10s of microseconds, and may involve a peak disturbance on the output of hundreds of mV following a heavy load transient. Localized decoupling at the regulator inputs and outputs, and in close proximity to the load disturbance helps to mitigate this, but zero disturbance is not possible and is dictated by the laws of physics.




If we examine these issues from a systems perspective, we see 3 basic guidelines emerge: (1) Make sure the power supply lines (+, - and GND) carrying DC have only a *small amount* of superimposed AC and certainly *no rectified class B components*, (2) keep all AC signal and power loops *localized* and as small and compact as possible and (3), ensure the power supply intrinsic noise is low, and can *easily be filtered out*. Applying these rules then leads to the following design outcomes:-

1. Run the opamps - both the IC and the buffer - in class A. This first step immediately means that the supply traces to the opamp carry a relatively small amount of superimposed AC at the output signal fundamental plus *low order harmonics*, reducing magnetic coupling and common impedance coupling as noise sources. You of course get

the major added benefit of class A operation which translates into lower distortion into heavy loads at HF - a worthy design goal in its own right.

2. Decouple each opamp + and – supply through a suitable isolation resistor and decoupling cap right at the op-amp supply pins. I use 22 Ω and 100 µF electrolytic with a -3 dB $f_o$ of ~72 Hz, which I find is a good compromise between size, decoupling and filtering effectiveness. Return the decoupling capacitor junctions to the system star ground through a dedicated power return trace to avoid contaminating any signal ground return lines. This technique isolates the opamp load dependent supply current changes from the main supply rail at mid to high frequencies.
3. Where possible, terminate the op-amp load return at the op-amp that is driving it – especially important if driving loads of more than a few mA. All AC current flowing into and out of the opamp pins is thus localized in small tight loops that minimize radiation and susceptibility to pickup.
4. Leverage the PSRR of opamps. Modern opamps achieve PSRR's of >60dB at 20KHz, with a few standout devices (e.g. AD797, LM4562, LM49710, LM49990) significantly better than this at 100 dB or more. If you couple their outstanding PSRR capabilities with the simple techniques described above, the effective equivalent *wide band* input referred noise is *below 10nV*. Opamp PSRR decreases at 20dB/decade, so at HF its not as good as it is at DC. Localized decoupling gets around the HF PSRR performance limitation (although still very good in modern devices), whilst at low frequencies, the opamp PSRR provides the noise rejection needed to get an overall 10nV input referred wide band *PSU noise.*
5. Use an LM317 for both the +ve and the –ve supplies The LM317 load and line regulation is considerably better than the LM337. On split rail supplies, rectify, smooth and regulate each half separately, and then only combine them to create the final, split supply. Decouple the LM317 ADJ pin – this gets you 20 dB lower noise (100uF cap).
6. Use a ripple eater on the output of the 3 terminal regulators as shown in Fig. 21 (Q2 and Q3 in Fig. 22). For preamp level work, it is quite easy with a Darlington pass transistor, a 1 kΩ resistor and a 1000 µF cap to have a pole of under 0.05 Hz, such that by 50 Hz there is >60 dB of attenuation of any regulator noise, and by 5 kHz, noise is down by -100dB. When this is coupled with (1) through (3) above, practical no-load wide band noise levels from the regulator below 5 µV become achievable as measured at the opamp supply pins. Real world layout parasitics, capacitor ESL and ESR mitigate against these numbers in absolute terms, but the point I am making is that supply performance and noise can be removed as a problem – essentially becoming 'blameless' to borrow from the lexicon of another practitioner.

You may question the wisdom of hanging a ripple eater on the output of a 3 terminal regulator, because the ripple eater is outside of the regulator control loop, so step load changes are going to cause significant output voltage disturbances. However, running all the op-amp stages heavily in class A mitigates this problem almost completely, as do the high load impedances (low current demands) associated with small signal conditioning circuits.

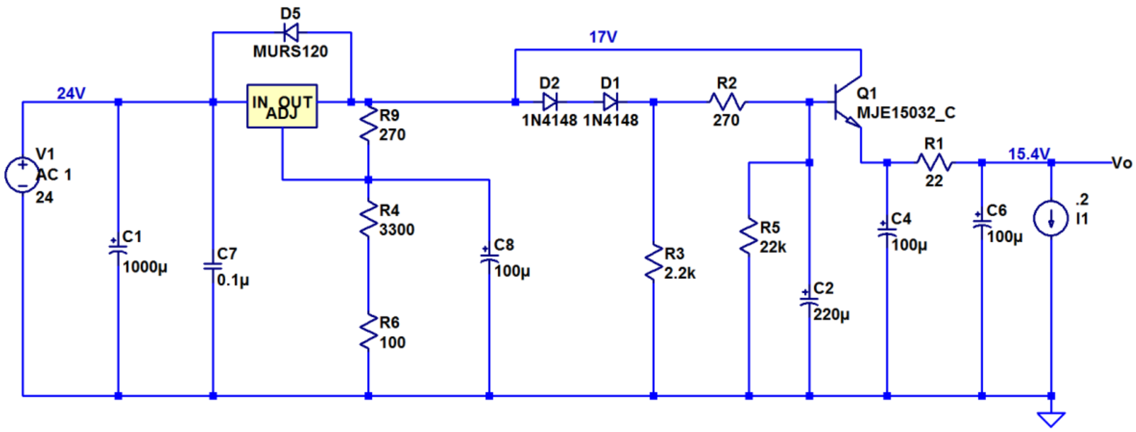



**Figure 21 - Simple Very Low Noise Regulator Concept Suitable for Class A Line Level Applications**

Its important to note that the techniques proposed above are generally not suitable for CFA stages, or low noise clock circuits[1]. CFA's need a stiff, low noise supply because their PSRR is usually 20 to 30dB worse than VFA designs – but this is a subject I may return to at some future date and an area where shunt regulators can really shine.

I have called the approach described in 1-6 above *'relaxed regulation'* power supply design. By deliberately running everything in class A, providing localized RC filtering on both rails, and leveraging the PSRR of opamps, we no longer need super regulators with low output impedances, or blindingly fast loop response for load and line perturbations. All that effort into selecting and using ultra low noise references you see on some super-reg designs is also not needed – the ripple eaters provide great filtering. Ringing does not take place on the supply rails, since they are *very heavily damped*, and the regulator load current very closely approaches that of a constant current source – the signal currents for the most part in everyday use are only a small fraction of the class A standing current within each of the stages. And, in this set-up, the humble LM317, another 30+ year old workhorse, does a very fine job of providing clean, low noise power, albeit at the expense of 1000 µF capacitor from the Adj pin to 0V and a hefty ripple eater output stage. Careful attention to layout with a separate decoupling capacitor ground return, keeps noise well away from the signal ground and rounds out the overall PSU low noise strategy.

This approach will not suit all types of analog small signal circuitry – but for high performance audio line/phono stage preamplifiers like the Symphony, I believe it is the best approach.

[1] In off-the-shelf crystal oscillator modules, a linear regulator is usually imbedded within the module to ensure good performance

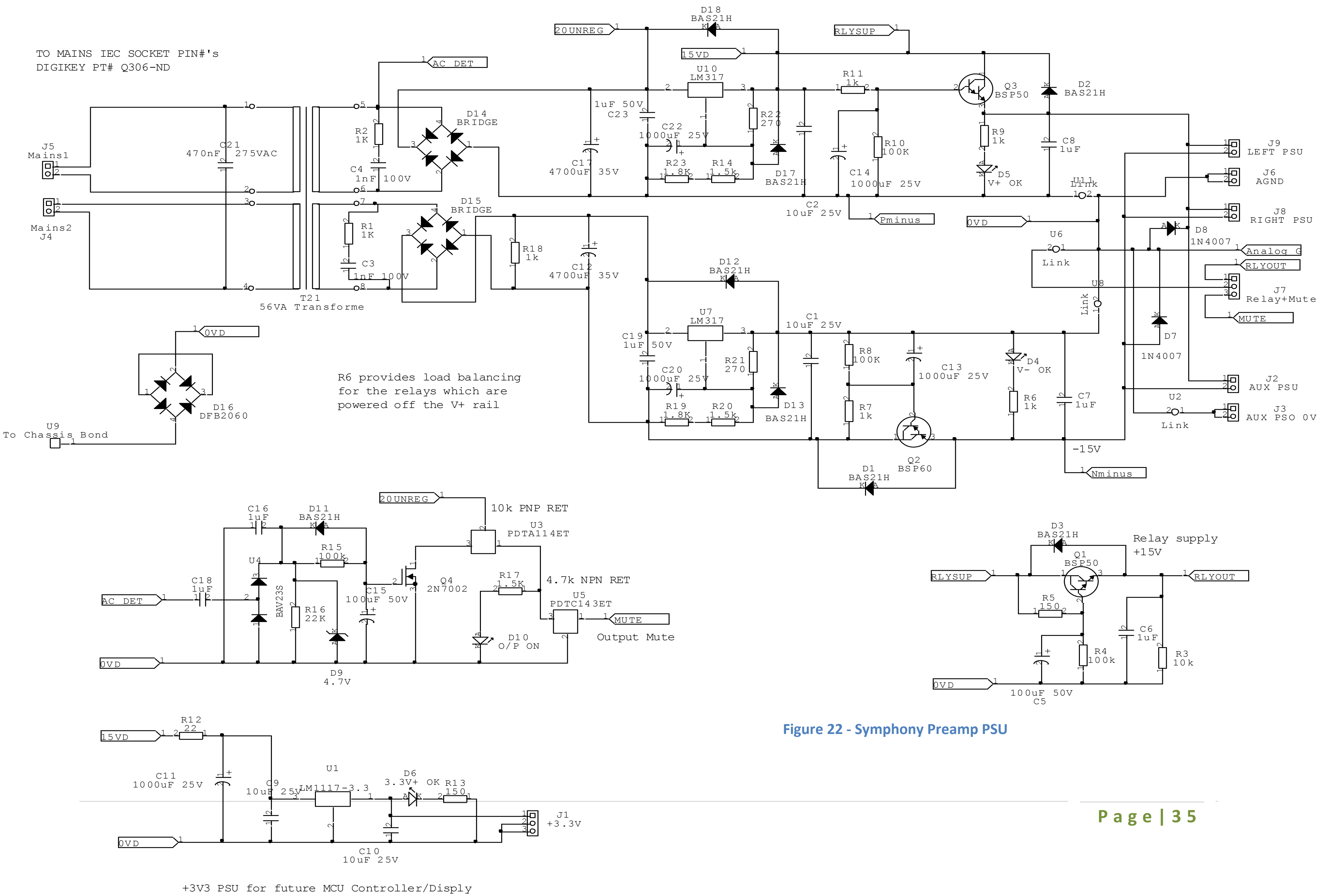



Figure 22 - Symphony Preamp PSU

Figure 22 shows the complete PSU circuit diagram.   The main PSU transformer is a Hammond 56 VA 18-0-18 split bobbin unit that features low inter-winding capacitance. Each secondary is separately rectified and regulated, as outlined in Fig. 23  by D14, D15, C17 and C12 and associated regulators and then finally being combined by links U8 and U11 at the main analog ground – the overall concept is depicted in Fig. 23 below. Doing it this way ensures there is no chance the capacitor charging currents can inject themselves into the star ground – in a line preamplifier, even 10 µV would be problematic and degrade the S/N ratio ref 1V output to a best case -100 dB. C21 provides some mains filtering, while R1, R2, C3 and C4 serve to damp any ringing that may occur as the rectifier bridge diodes switch – I have simply used the values recommended by Morgan Jones in his Linear Audio article.

The adj pin on both U10 and U7  are decoupled  with large 1000uF capacitors, reducing the wideband output noise to about 20 µV – a good figure indeed. Good localized decoupling on both the input and output of the regulators ensure stable operation and good load response.

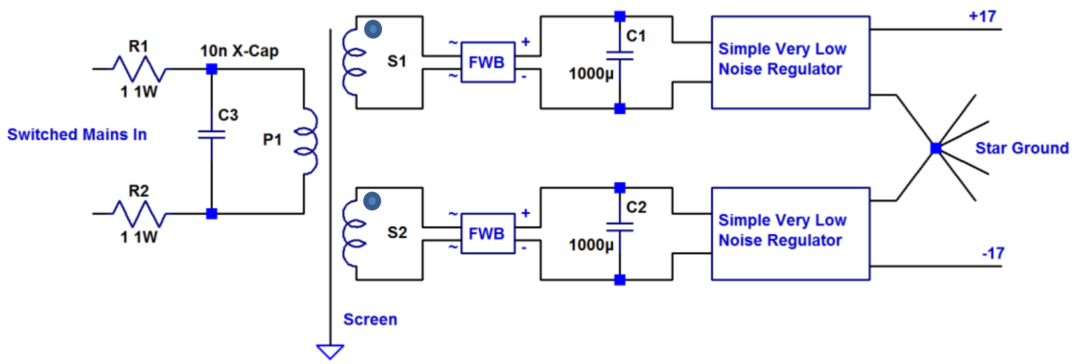



**Figure 23 - Low noise System PSU Architecture Starts by Separating the Two PSU Halves in split Rail Designs**

D2, D12 and D13 and D17 provide protection around the regulators to ensure their output and adj pins do not become reverse biased during power up or down, which may cause damage, while D7 and D8 are installed to prevent either of the rails from being pulled inadvertently to the opposite rail during power up/down events – in either case, this would result in latch-up and possible destruction of the regulators.  See Self's 'Small Signal Design' for a fuller expose on the subject.

Q2, Q3 (Darlington 1 A SOT223 transistors) and associated components form a simple ripple eater that removes any vestiges of noise from the supply rails. As already discussed at length, because the whole preamplifier runs in class A, there are no significant load transients, so this approach works really well.  Q1 is  designed to isolate the relay supply from the  main analog supply.  When relays switch off, despite the flyback diode and RC damping, they still dump a small HF spike onto the supply rail – I did not want any of this stuff on the supply to the opamps. I can confirm that the relay switching on this preamp is absolutely noiseless. You will notice many links in the circuit diagram (U2, U6, U8 and U11). These are vestiges of the layout process where I put a lot of effort into minimizing the potential common impedance coupling and earth loops – the links just make the layout process easier to achieve the correct layout. In the final PCB layout, I simply replaced them with PCB tracks.

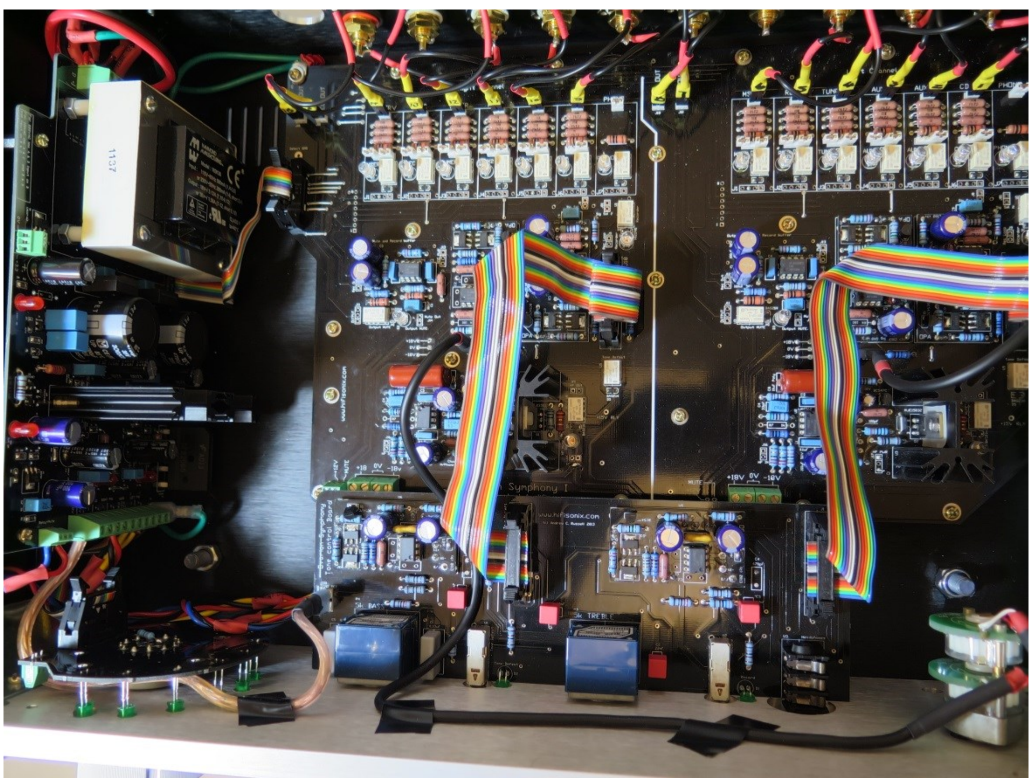
The delay circuit (Q4, U3, U5) and associated components) is based on the same design I used in the nx-Amplifier, but with the value of R7 raised to 22k following a suggestion from one of the nx-Amplifier builders – it works reliably offering a ~10 second turn on delay, and switches off in about 1 second. The output of U5 drives the output and headphone mute relays and ensures that there are absolutely no switch on or switch off plops, which many commercial products suffer from. I included a +3.3V 50mA (U1 and associated devices) supply as in the future I will use this PSU design to build an MCU with remote controller preamp – reuse where possible saves time.

The PSU board is built on a single PCB that is mounted on the LHS panel of the preamp. Originally I used a 25 VA dual 0-18 VAC secondary transformer from Amveco which I will have to admit was a little undersized for the job because it ran at over 65 Degrees C – quite acceptable as far as transformers go, but a clear sign that it is running at its rated capacity - the total DC load current is 600mA per rail. The power supply you see here is the redesigned version using a 56 VA Hammond split secondary 'E' core design[2].

Recently on DIY audio, there was some discussion about the merit of connecting the secondary's out of phase to improve noise. This proposition rests on the fact that the secondary's, when in phase, radiate a summed EM leakage field that is linked to the filter capacitor charging currents plus load current and if you wire them out of phase, you will get partial cancelation of the radiated fields. A reduction in EMR of up to 20 dB is claimed. The winding method (bifilar, layered etc) plays an important part on the effectiveness of this technique. It should also be noted that this technique will have no effect on the ripple voltage after filtering and regulation or on mains conducted noise coupling through the inter-winding capacitance – it only affects the radiated EM field due to the load current. In large power

[2] Note that some of the Symphony preamp photographs in this document still show the original power supply

supplies such as those found in power amplifiers, this can be a useful technique, even though the secondaries are hardly ever bifilar – the best way here is to try it out and see if there is a difference.



# 4. Housing Design and Construction

***Aesthetics.*** I struggle to appreciate a good circuit design or PCB layout when it's not housed professionally and appealingly. I think about those constructors doing wonderful restoration work on old cars, or building chopper motorcycles, or even airplanes. Finishing off a project professionally goes a long way to engendering pride of ownership and importantly, pride of effort.

This is an unashamedly retro design and would not look out of place in a late 1960's or early 70's very high end set-up. 10mm thick milled front panel, 50mm diameter solid aluminum 'Sato' turned knobs from Japan (they weigh 120 grams each – they are heavy) with C&K push button switches that have a reassuring click when depressed or released.

I used a 320mm deep 3U all aluminum Modushop Slimline case with 3mm aluminum panels and had the front and rear plates milled to my specification which cost around 170 Euros over and above the 90 Euros base cost. The input selector and $215 Goldpoint volume control are mounted directly on the front panel, and the corresponding knobs recessed from the front side by c. 3mm, which gives a very 'engineered' aesthetic to the whole preamp. I did not get the front panel engraved, but some cursive would probably set it off nicely – sort of reminds me of those classic Marantz integrateds' from the 1970's like the famous 1060.

The Sato dials I used were very expensive – around $30 each for the larger 50mm items, and $12 for each for the 30mm bass and treble controls from Tang Hill in Taipei. However, its worth it because a big, heavy dial gives a rotary switch or attenuator a very solid, tactile feel; it looks and feels expensive. I read an article by hi-fi reviewer Ken Kessler a few years ago in which he said dials can make or break equipment – I agree and therefore avoid anything plastic or light in weight in my projects. Keep it heavy and as big as you can find. My only regret with this preamp is that I could not find anything at 60 or 70mm in diameter. Modushop used to do some very fine looking 50mm diameter dials, but they no longer supply these, which is a pity. I emailed Andrea Bettazoni, the Modushop CEO a few times but he is not budging – he told me his supplier will no longer manufacture them – so I assume he subcontracts these components out.

Despite having had the front panel machined, there was still a lot of work to be done to recess the LED's into the back of the panel. I used 3mm diameter LED light pipe inserts from Mouser to finish the holes off – these really look professional and I would recommend them for anyone doing a similar project. The hole size is important – 2.9mm +0.1 and nothing else will do since they are a friction fit.

I unfortunately I made a big error in my recesses and the mounting shafts for the potentiometers and switches were still too short, such that the associated shaft nuts could not be screwed on. I ended up using a large countersink bit to open them up – this solved the problem.

The main PCB is mounted onto the base plate with 15 off 15mm standoffs. That sounds like a lot, but the PCB is rather large at 270mm x 190mm and I did not want there to be any chance of the board mechanically flexing or vibrating.

***Wiring and interconnections.*** The connections from the main board to the tone control board are made through two 16 way ribbon cables which carry the power and signal. Interposing the power and analog ground wires between the power and signal lines keeps noise off the signal lines. I am very pleased with the noise performance – these cables and the interconnect structure work well. You will note a thick 2.5mm cross section areas cable on the RHS of the tone control board – this is the ground return from the headphone socket which is returned directly to the PSU star ground – you absolutely don't want to be simply tacking that onto one of the local signal or power grounds because you would then be injecting large signal currents into the small signal ground system – a clear opportunity for common impedance coupling problems: distortion, loss of channel separation and noise. As with the X-Altra Mini One, the headphone socket protrudes out of the front panel by about 1mm. This ensures the headphone plug never makes mechanical contact with the front plate, avoiding scratches and unsightly marks with use. The tone control board is held in position by the bass and treble pot shaft nuts. In retrospect, this is probably not ideal and there should have been additional brackets to secure the PCB to the front plate. However, there is no stress on any of the solder joints and mechanically the arrangement is remarkably solid.

The PSU PCB assembly is held in place on the LHS of the preamp with 4 off 20mm stand-offsand the PCB is shaped such that it rests on the base plate of the preamplifier – it weighs about 1 kg which is far too heavy to be supported just by the standoffs. The standoffs keep it well away from any of the small signal circuits, and the overall noise performance is excellent. All connections to and from the PSU board are made through Phoenix screw connectors, the only exception being the earth connection to the ground lifter which is via a 5mm push on tab connector.

I used high quality Neutrik phono sockets for all the inputs and outputs. These are gold plated and very well made – but, they are bulky and take up a lot of room. Secondly, there are a lot of wires between the rear panel and the main PCB – It took about 2 hours to make up the interconnect cables which are terminated in Molex 2 way 2.54mm pitch push on connectors. I dressed the finished housings off with a double layer of heatshrink, which grips the cable and holds it firmly in place to the connector shroud – it's unlikely that these will get pulled out or break , or flex excessively during assembly. However, if I were to do this preamplifier again, I would mount the relays on a rear terminal board, and use PCB mount phono sockets. This would then mean only the output and rec loop output connections between the main board and the rear terminal panel, plus of course the relay supply control signals – a job for a ribbon cable.

The input select switch is mounted on a PCB and held in position simply by the switch mount to the front panel. A 16 way ribbon cable runs from the switch PCB to the main board, and I routed it between the LHS panel the PSU to keep it neat, and keep relay coil switching transients away from the analog circuits. Each relay coil has the front plate LED and an on board LED wired in series with it. I'll be the first to admit I overdid it on the main board mounted LED's – but, they are red, and at night the whole thing has a warm tubey glow that goes exceedingly well with a double scotch. The preamp uses 12 V relays, and the supply voltage is about 15V to cater for the LED and 220 Ω series resistor drop and are therefore powered off about 11 volts in practice.

I used a switched, fused IEC 3 pin mains receptacle for the rear panel. I don't take shortcuts when it comes to mains – you need to use good quality inlet and it has to be fused – 215mA in this case. All the mains wiring is made with 1.5mm multi-strand wires, and each connection is dressed off with heat shrink. Earthing is important for safety and to keep noise to a minimum – the incoming earth connected to a single chassis bond point at the rear LHS of the chassis. A ground lifter (a PCB mount FWB rectifier rated at 25A) is interposed between the chassis ground and the entire preamp circuit, allowing the electronics to 'float' around the earth voltage at +-~1.2V – if mains should find its way in via any of the inter-connections, it will be clamped by the rectifier to earth, tripping the MCB – for this reason, the preamplifier is solidly earthed as well through the 3 pin IEC cable.

# 5. Sound: the Subjective Assessment

The comments that follow are of course primarily subjective and my personal opinion – but I will not be proposing some esoteric non-engineering or meta-physical reasons for my perceptions. I try to keep my feet firmly rooted in practical engineering when it comes to these things.

Initially, with the bass, treble and volume at full, there was clear hiss with my ear against the tweeter and a wave of disappointment swept over me – was this thing oscillating, or worse, had I completely screwed up the low noise design and associated calculations?

I soon isolated the problem down to the CD players (Oppo BD103 and Pioneer DV655A) – both have audible hiss when placed into the pause (i.e. 'II') position. When switched to an unused input and the 20 dB attenuator selected so the input source impedance is ~900 Ωs, the Symphony preamp is the quietest preamp I have built to date in terms of thermal noise – *its absolutely silent* at any volume setting. Further, none of the other preamps I have built have had any audible hum, and neither does this design.

Certainly the AD797 front end gain stage with its ~0.9nV rt/Hz (1.2 nV rt/Hz worst case) noise voltage and the low front end source impedance of ~900 Ωs worst case play an important part in this – as does the signal chain structure with the volume control placed after the gain stage which means as the signal is attenuated, so is the noise. As I noted earlier, gain stages subsequent to the front end line amplifier stage contribute very little noise because their gains are 0dB. This of course does not apply when the treble is set to full boost (~10 dB at 10kHz) – but, I was very pleased with the noise performance in this regard – the Symphony is still considerably quieter than the X-Altra Mini One, which uses LM4562's and a 10 k Alps potentiometer and when its volume control pot is set to mid level, - so around 2.5 k Ωs source impedance - the noise from my 89 dB/W B&W 702 tweeters with the X-Altra mini is perceptible. The SCA-1 is slightly noisier than my X-Altra Mini 1 – but let me stress, that in both cases you have to put your ear *right against the tweeter* – as a listener, you would certainly not pick up anything from a normal listening position some 2 to 3 meters away. As an experiment, I plugged AD744 devices into the front end gain stages and the noise was then perceptible – these devices feature an en of about 4.5 nV/rt Hz which is about 5x that of the AD797. It makes a difference at these low $R_{source}$ resistances, and this is where the AD797 really shows its superior low noise performance.

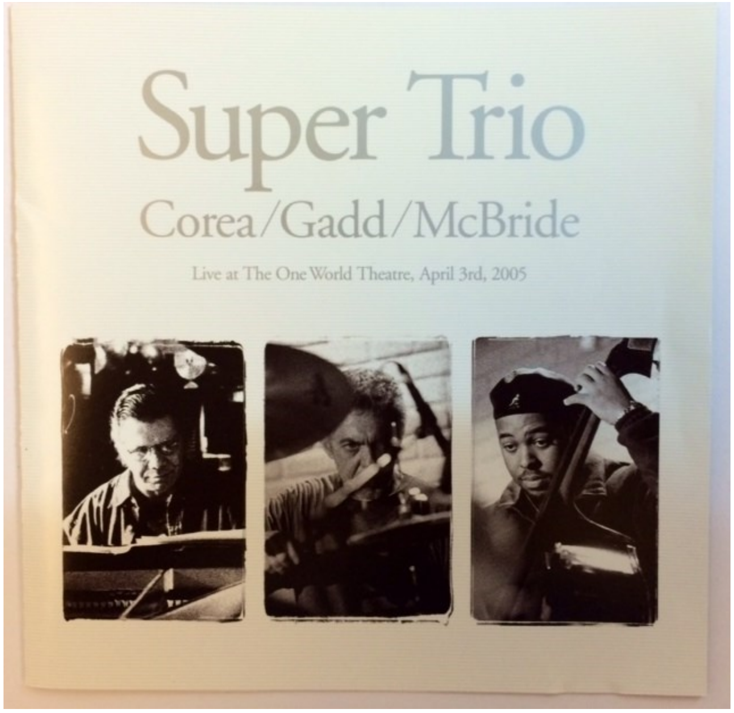
I spent quite some effort optimizing the layout and sticking strictly to a star and 'T' grounding system – as discussed in the power supply section. At full gain, there are no hum components discernable on the output with my scope (2mV/division sensitivity), and with headphones plugged in, none either.

Getting better noise performance would mean paralleling opamps, like Douglas Self has done, or going for a discrete design. But, in the latter of these two options, the design in my view will have to tradeoff significant distortion and PSRR performance.

***Subjective sound quality.*** I have had a lot of pleasure listening to my X-Alta Mini over the last 5 years or so, but the new design is altogether more spacious and 'silkier'. There's definitely a significant step up in the imaging and the top end is the smoothest out of the three preamps I have on hand. I would suggest that this may be to do with the lower noise power supply, stricter adherence to good layout principles, which gets better with practice, and of course the all class A configuration. I am not suggesting that opamps cannot sound superb in a more conventional, un-buffered class AB small signal design, but class A and buffering seems worth the effort to me – and from a purely technical position, this is clearly evidenced in the numbers as I showed in my simple line buffer from a few years ago where from 1~3V output, distortion is at or below the AP SYS272 noise floor.


For the listening test, I pulled out 'Songs of Joy and Peace' by Yo-Yo Ma and Friends. In terms of sound staging, this has to be one of the best CD's I own - its holographic and via the Oppo BD-103 and my nx-Amp, the new pre did not disappoint. Track number 12 ('Give us Peace') features Brazilians Sergio and Odair Assad along with Ma on the Cello and Edgar Meyer on the double bass. This particular track has a fantastic space around the players and the sound stage depth and width is truly amazing, stretching well to the left and right of the speakers and very far back behind them. Anyone who has reservations about what can be achieved on the standard CD format, should take a listen. There is a delightful 'thwang' at the end of the track on the double bass that has a wonderful resonance to it, and the imaging is so good on this CD, you can pinpoint exactly where Meyer is located within the sound stage. I set the bass control to the 1:30 clock position, while the treble was at 1 o'clock and this really brought the piece alive – a bit of boost created some real drama and excitement.

Another favorite of mine is 'Super Trio', featuring Chick Corea, Steve Gadd and Christian McBride on bass. I like this recording because it is very natural, and does not seem to have




been fiddled with in any way – on some of the tracks the piano notes are especially warm and almost seem to blossom, before fading and there is not a hint of harshness or sibilance anywhere on the CD. The sound staging is not quite as good as the Yo-Yo Ma one mentioned above, but it's a great listen. Here, the preamplifier again delivered a very solid, smooth sound. Gadd hits the bass drum with the kick pedal a few times on the first track and the depth, attack and weight of the note is absolutely tremendous. This is a live CD, so the audience interaction adds to the magic and the clapping and laughter are accurately reproduced with very good image depth.

Chris Botti's 'This is Chris Botti'. Reproducing trumpet accurately is quite hard to do because


you've got a lot going on sound wise with plenty of harmonics and in many cases the breathiness and 'tizz' that often seem disembodied from the rest of the instrument's sound. This CD is very well recorded and the Ovation Symphony did a great job of conveying the venue and audience size. There's lots of beautiful, mellifluous strings and it's easy to detect that Botti's position in the sound stage moves during some of the tracks as he walks around the stage. The bass is a little light, but just dialing it up a bit balanced things out nicely and added a bit of extra weight to the proceedings.


I have remarked elsewhere that many of my classical CD's could do with a bit more bass and in some cases some top end as well. I've attended a few classical concerts in my time, and the first thing you will notice if you've also been to a rock concert, is the sound level is a lot lower. Recording engineers in this regime don't have the luxury of dialing up the bass section or the violins and any post processing adjustment has to be done with care if the authenticity of the performance is to be preserved. But, when I listen to classical, I have to contend with a less than ideal listening space and I also want to inject that all important excitement into the music.

Stephen Kovacevich's recording of Beethoven's Piano Concerto No. 5 is a well recorded CD with the original done on analog tape I believe and from there, digitally re-mastered. The violins and brass are well captured, although when compared to some of the better recordings, perhaps a little less defined on crescendos - good solid state amplification *will never* gloss over sub par recordings . . . and neither should it. However, the sound stage is nice and wide, but not as deep as I would have liked; the strings and piano are very well captured however. The sound compared to my other two preamps is more solid and fleshed out, with just a little bass boost adding some weight to the lower registers.


The next CD I want to mention is the HiFi News LSO sampler (the same one I played when I took a first listen to my sx-Amp about a year or two ago). This is another holographic recording with wonderful midrange and great bass and treble extension. The Symphony preamp did a first class job of reproducing a big, wide, deep sound stage that had me glued to my seat for about 30 minutes as I enjoyed the performance.


Jack Johnson's 'In Between Dreams' is basic and bare – just a bunch of guys enjoying themselves, but it is well recorded with wonderful imaging. Track number 7, 'Staple It together' has a fantastic bass line, and the drums have been superbly recorded so that you get the great propulsive rhythm over which the electric guitar is overlaid. The thwack of the sticks on the rim and the snare are sharp an resonant and if you play it at the right volume – i.e. loud, you can really feel the music – but, because the piece only has a handful of players and its well recorded, you also get the feeling of space around the musicians. On this recording, there is no need for tone controls, and I just got the sense that the Symphony had a better overall presentation than the other two preamps – the lower registers seemed more extended, and the treble was to my ears, much cleaner and with the mid-range, without any strain – a very relaxed sound.


Maria Bethânia is a Brazilian bossa nova and popular music star. 'Pirata' is a CD of light, acoustic music that is superbly and intimately recorded. The sound stage is very wide and deep, and the upper registers on this CD are particularly crisp and sparkly in the nicest possible way. Bethânia has very husky, evocative voice that has been beautifully captured in my view. The new preamp did a fine job in projecting the sound stage to the left and right of the speakers, with a strong impression of depth as well. The bass balance is absolutely perfect on this CD and needed no reinforcement. There are also one or two tracks on this CD with a very deep, loud bass drum that emerge extraordinarily weighty and resonant from the speakers.


This Linn 'the hi-fi collection' sampler dates back about 10 years IIRC and showcases some of their superb catalog. All of the tracks are well recorded, with my standout favorites being 'Come fly with Me' by Tina May, and Claire Martin's 'Black Coffee'. In the latter recording there is a delicious organ solo about half way through the track and I am always compelled to turn it up. The Symphony did a first rate job of holding the big, deep sound stage together, while the midrange was clear and open, really doing her voice justice. The Tina May track has a great interaction going on between the keyboard and the sax and I was very pleased with the way this was presented on my system. Neither of these recordings needed any bass or treble adjustment – they are superbly balanced in my view.


The last CD, Paul Simon's 'Rhythm of the Saints' dates back to 1989 or so, and I like it because it's a very smooth, rounded recording in a way that many modern ones are not. Secondly, there's lots of percussion on this CD that are really spacey and expansive – you've probably guessed by now that I place sound staging above all else in my system performance – its probably the single most important factor in musical enjoyment in my view, and especially so with acoustic music. Here, the Symphony preamp was able to cast a sound stage well beyond the edges of the speakers, and with great depth as well. Some of the tracks, like 'Proof' have a solid electric theme to them, with a fantastic guitar 'duel' between George Seba and Martin Atangana along with a very powerful brass synth interjection every now and then, all underpinned by a rock solid bass kick drum beat. All of the players were clearly delineated in their space and the bass was very crisp and forceful.

***Headphone Sound.*** In the X-Altra Mini, I used a discrete class AB diamond buffer inside an LM4562 opamp feedback loop, capable of delivering up to 300mV in class A. The later SCA-1 used an LME49710 opamp driving LME49600 buffers – also in the feedback loop. This was a big step up from the discrete diamond buffer and there's no question that the National (now TI) LME49710 + LME49600 pairing are made for each other and sound fantastic –there are quite a few designs on the internet and they are relatively easy to use. But, the class A headphone amp in the Symphony brings all of the LME49xxx pairing sound to the party, but with a big extra measure of smoothness and detail that the former cannot match – you can just keep cranking up the volume, and it sounds good with not a hint of compression or treble harshness – my ears and the headphones gave in first. It has a tremendous amount of clean headroom and a very open, spacious sound that is not there on the other designs. Even under loud listening conditions, the headphone amp never exits class A all the way up to 12 V pk output, delivering around 370 mA peak into 32 Ω phones. The combination of clean, very low distortion, class A headroom make this the most compelling, open sounding and smoothest headphone amp I've designed to date. For comparison, I pulled up a few of the same tracks on my iPod and took a listen and the contrast was stark – clearly a low supply voltage class AB (and in many portable electronics 'inductorless output class D') cannot compete with a purpose designed hi-fidelity headphone amp where into 32 Ωs, the distortion at 1V out is about 600ppb, while at 3V out it is around 3 ppm.

# Document History

| Date | Release |
|---|---|
| 16th July 2014 | Initial publication - V1.0 |
| 4th October 2014 | Updated document with the new 56W low noise PSU V1.1 |