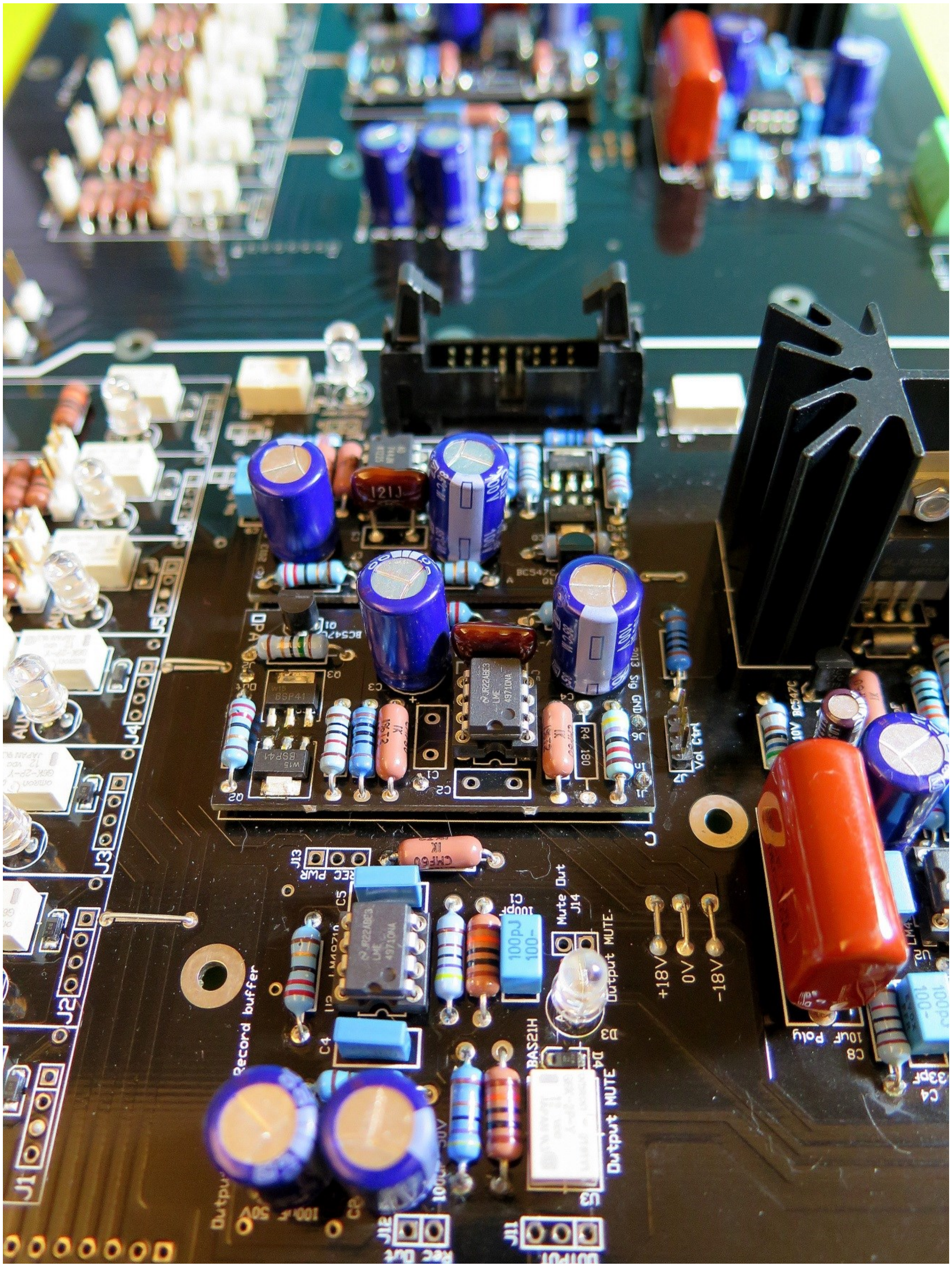
Part 1 - The Ovation 'Symphony' High Fidelity Preamplifier

A full function high fidelity line level preamplifier featuring 7 line inputs with jumper selectable -20 dB attenuation pads, MC/MM input facilities, Baxandall tone controls and a 2 W class A headphone amplifier. The design makes use of amplifier modules, using low noise opamps and in-loop discrete class A buffers to achieve distortion approaching 1ppm at 20 kHz at the rated output. The design targets -110 dBV noise performance and clips at 25V pk to pk into 600 Ωs, and 20 V into 200 Ωs.

The entire preamplifier is housed in a 450mm x 140mm x 300mm Aluminum Modushop chassis and weighs 10 kg.

Andrew C. Russell

October 2014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REC PWR
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+18V
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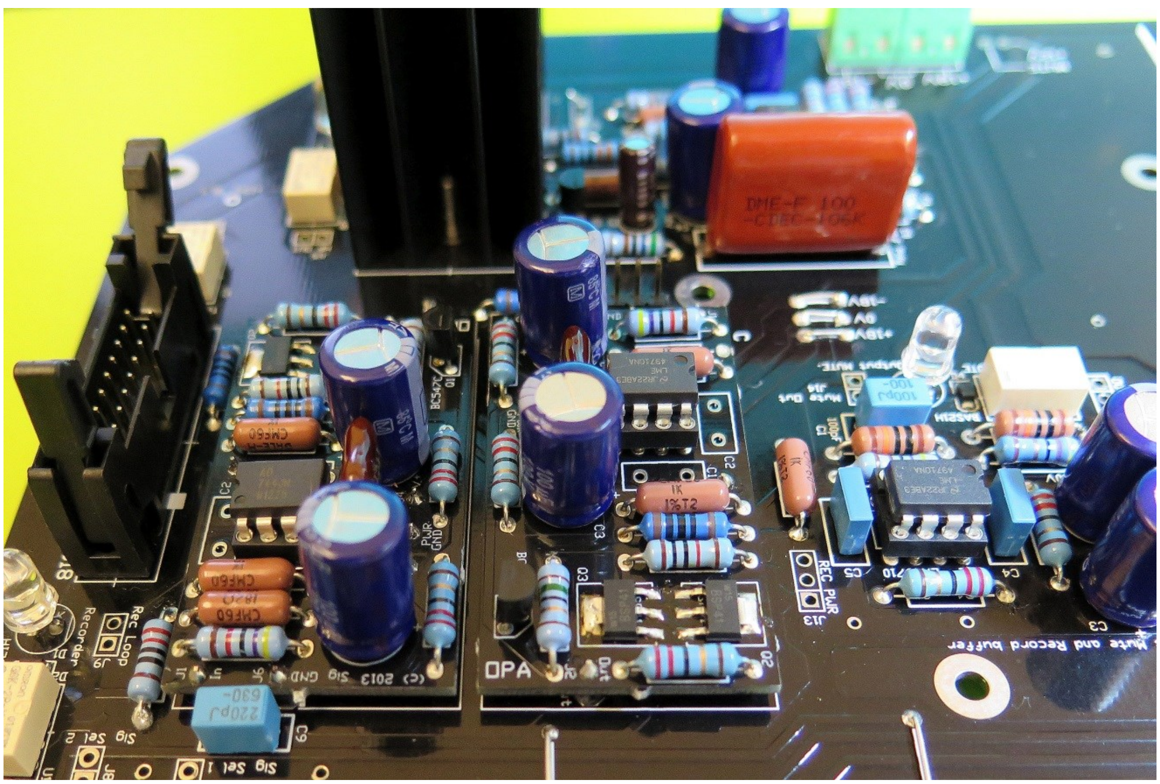
1. Introduction

I have designed and built two preamplifiers over the past few years, this being the third. The first of my recent efforts – after a 25 year layoff – was the X-Altra Mini One, which featured an ultra-simple signal chain based on an LM4562 op-amp. The second was the experimental SCA-1 which was an all-out top of the range IC based design using a TI PGA2320 chip configured in balanced mode, along with LM4562, LME49600 buffers and a headphone amplifier. This design had no tone controls of any sort, but featured a remote control and a 5" GUI TFT display, all controlled by an NXP LPC1768 ARM based mbed controller. I have not got around to housing this design yet – I guess the expense of a large custom case has put me off for the time being. This brings me to the current project, the Ovation 'Symphony One'. I designed and built a few pre-amps in my early 20's that featured Baxandall tone controls and headphone outputs, and after reading about Douglas Self's latest design in Elektor, I was inspired to try my hand again at a full function preamp, incorporating decent (i.e. Baxandall) tone controls, a class A headphone amplifier and an optional MC/MM input board, which would be designed at a later stage. With no pretensions to convenience, this design does not cater for remote control, and has no fancy 5" TFT display like the SCA-1. However, this preamp features the following:-

- The design uses AD797 and LM4562 low noise opamps in the main signal path, to achieve outstanding noise performance
- All the op-amps are buffered with discrete class A output stages and their outputs are also bootstrapped at around 600uA into class A mode. The buffers are inside the opamp feedback loop
- All class A operation means the supply lines carry only the fundamental of the output signal and low order harmonics – so no wide band harmonics as is the case with class AB operation, reducing noise and any impact on distortion performance due to magnetic coupling of these higher order components into sensitive circuit nodes; further, with this technique, HF excitation currents are kept off the supply rails, minimizing potential ringing on the supply lines (see Kendall Castor-Perry's articles on power supply decoupling for example)
- This design uses a very low noise PSU with a targeted wideband noise of less than 5 µV
- The design uses 'back terminated' input signal select switching to deliver very high input 'offness'
- Strict attention to physically separating the left and right channels keeps channel separation high
- Distortion at 1 V RMS output into 10 kΩ is in the region of 1ppm at 20 kHz, and at 8 V RMS out into 600 Ω better than 5 ppm, again at 20 kHz
- A Baxandall tone control (which can be completely bypassed) offers +-10 dB of boost and attenuation at 100 Hz and 10 kHz with distortion of < 15 ppm at 20 kHz and 10 V peak out
- This preamp features a very high performance headphone output that will drive 32 Ω to >3 V pk-pk and still remain in class A operation at less than 10 ppm distortion at 20 kHz
- 7 input sources and a switchable buffered tape loop
- The volume control is a front panel mount Goldpoint Mini-V 5k log taper unit for the ultimate in transparency and tactile feel

1. Design Approach: Some General Thoughts

My more recent (2008) X-Altra Mini One preamp is a minimalist design that eschews tone controls and really focuses on doing as little as possible with the source signal, other than providing source selection, volume control and some gain and buffering in order to match the typical 1V required to drive a modern power amp. Input source selection is based on small signal relays (Panasonic AGN series), which, along with a carefully designed power supply and layout, allows this design to achieve about 5 ppm at 20 kHz distortion at 1V RMS into 600 Ω, and <70 ppm at 6.5V RMS into 600 Ω, again at 20 kHz. If applied correctly (and that's easy to do if you just follow basic, simple layout and circuit design practice) the LME4562 is a wonderful sounding chip. Sloppy layout, bad decoupling and other avoidable design missteps can lead to problems – you are dealing with a device with an OLG of 140dB at LF and a ~55 MHz GBW. In the write up, I comment on the very good sound – open mid-range and top end along with first class imaging. Direct coupling means bass performance is not compromised, and there are no opportunities for electrolytic sonic intrusion, inasmuch as this is a problem with a carefully chosen component.



A year or two later, this was followed up with the Ovation SCA-1. Again, no tone controls, but the volume control and main gain element featured a TI PGA2320 configured in balanced mode, with buffering before and after using LM4562's and LME4900 unity gain buffers. The PGA2320 has taken quite heavy criticism in some quarters, with claims that they are sub optimal in sonic terms or noisy and so on, but my practical experience is different. You need to feed them from a low source impedance to get the best noise and distortion performance (so something well below 1 k Ω) and the outputs do need to be buffered – although I would say this applies to any op-amp based design that has 'high end' pretensions. In the SCA-1, the balanced input signal after the relay selection stage is buffered by a dual LM4562 opamp, which in turn drives each channel of the PGA2320 in balanced mode. The output of the PGA2320 then feeds into an inverting stage and an LM49600 high current buffer, which is inside the opamp's feedback loop. This configuration will drive a 20 V pk-pk balanced signal into 200 Ω with less than 3 ppm distortion at 20 kHz. At 1V RMS out into 600 Ω, the distortion is a few hundred ppb. A servo keeps output offsets to less than 50 µV. Most of my assessment of the SCA-1 (both subjective and in comparison with the X-Altra Mini, Marantz and various iPods and CD players) has been done with an assortment of headphones including Sennheiser, Audio Technica ATH-900, some Sony IE's and a very high end Stax tube based system along with some evaluation sessions on the Ovation 250. The design features a very open top end, great bass, imaging and low noise. As further independent evidence to the performance potential of the PGA2320, a 2008 review (and there are more recent iterations of the C-03 that still retain the PGA2320 as the main gain control element) of the top of the line C-03 Esoteric line preamp which uses this chip as the primary gain element and retails at over $10k, gave it top marks for sonics and overall audio performance. In fact, the reviewer claimed it was one of the *very best* line preamps he had ever heard. And, this was not the only review to similarly rate this Esoteric preamplifier in the top echelon – so did 6 Moons amongst others, and Stereophile also had very positive comments after hearing a system built around one. When thinking about high end audio, one's component and semiconductor device prejudices are best set aside I have found – whether you believe equipment reviewers or not.

Both of my recent designs (X-Altra Mini and the SCA-1) tell the brutal truth, and especially so with respect to recording quality. But, there is no doubt that the two biggest – by an order of magnitude or more – contributors to sound perception are the recording itself, and the speaker + room interaction. If you have a reasonably large record or CD collection, this can leave you with recordings that lack the right kind of balance given your specific room/speaker setup. Some commentators believe a good speaker will sound good in any environment, and if you think you have a sound problem, then your system is not up to standard. I take the view that in most cases the system is capable of very good performance and it's the room that's not always up to the task. I have about 50 CD's out of 500 that are almost perfect for my listening environment: the bass and treble are well balanced, good imaging, and the overall sound well integrated and pleasing to the ear. This leaves a lot of recordings in my listening environment which need some response balancing and the requirement for a decent tone control, which will be discussed in some detail a little later.

Douglas Self's two major DIY preamp designs (the first here) have featured another of Peter Baxandall's innovations, his Active Volume Control. This approach varies the feedback factor in an active gain stage to achieve a volume control range in the order of 100:1, or about 40dB and it achieves a log like response using standard linear pots which are always easier to get hold of, especially in the 1 k to 20 k range. There are clear advantages to this design, and the fact that the gain need only be as much as is required for ones desired listening level means that it always provides the best signal to noise ratio for a given output level. Some concerns with this configuration are that you have to hang a potentiometer off the sensitive summing node in an opamp feedback network, but the same criticism of course can be leveled at Baxandall's tone control, or any inverting, current summing circuit for that matter.

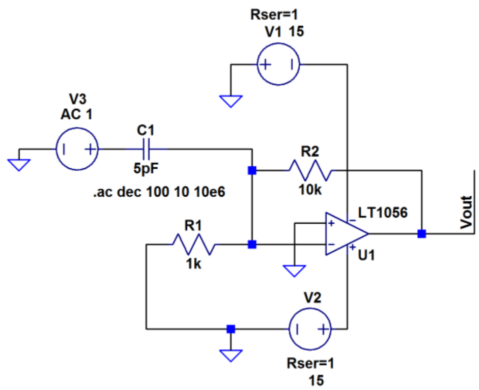


Figure 1 – Modelling noise susceptibility of summing junction circuit arrangement

It's important to realize that if noise finds its way *between* the summing junction of the feedback resistors and the -input in an inverting amplifier, that noise is amplified by the full loop gain of the opamp - with loop gains easily approaching 100 dB, 100nV noise becomes a 10mV problem on the amplifier output. This of course is mitigated by the use of low value feedback network resistors and careful, compact layout, which is one of the reasons Self's Elektor preamplifier implementation performs well in this regard.

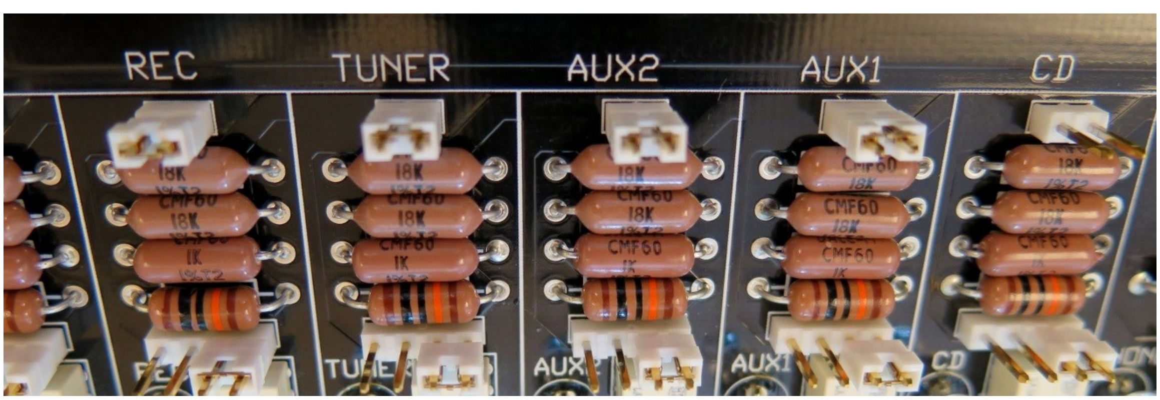
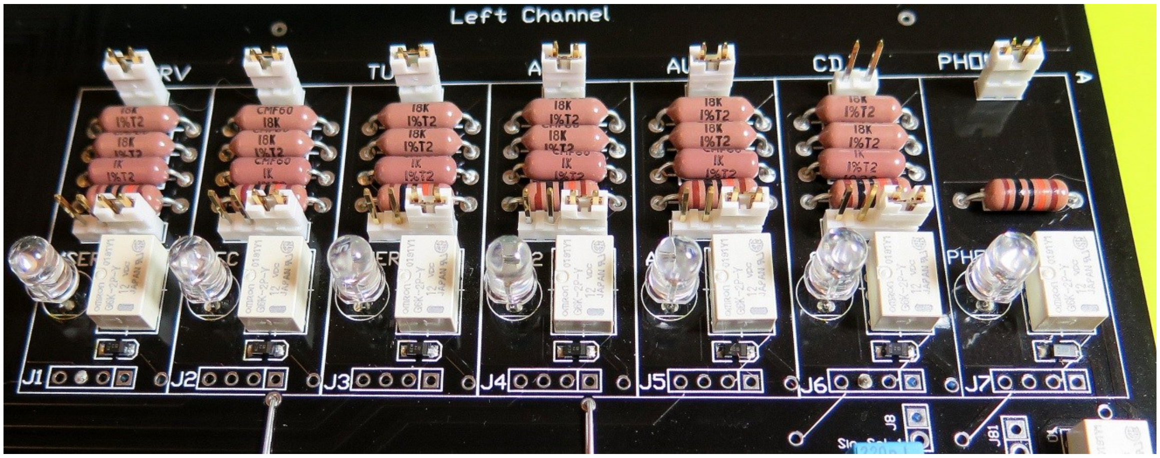


Figure 2 - Just 5 pF of coupling means at 10 kHz, noise performance is degraded to -50 dB with 1k + 10k feedback resistors (gain of 11)

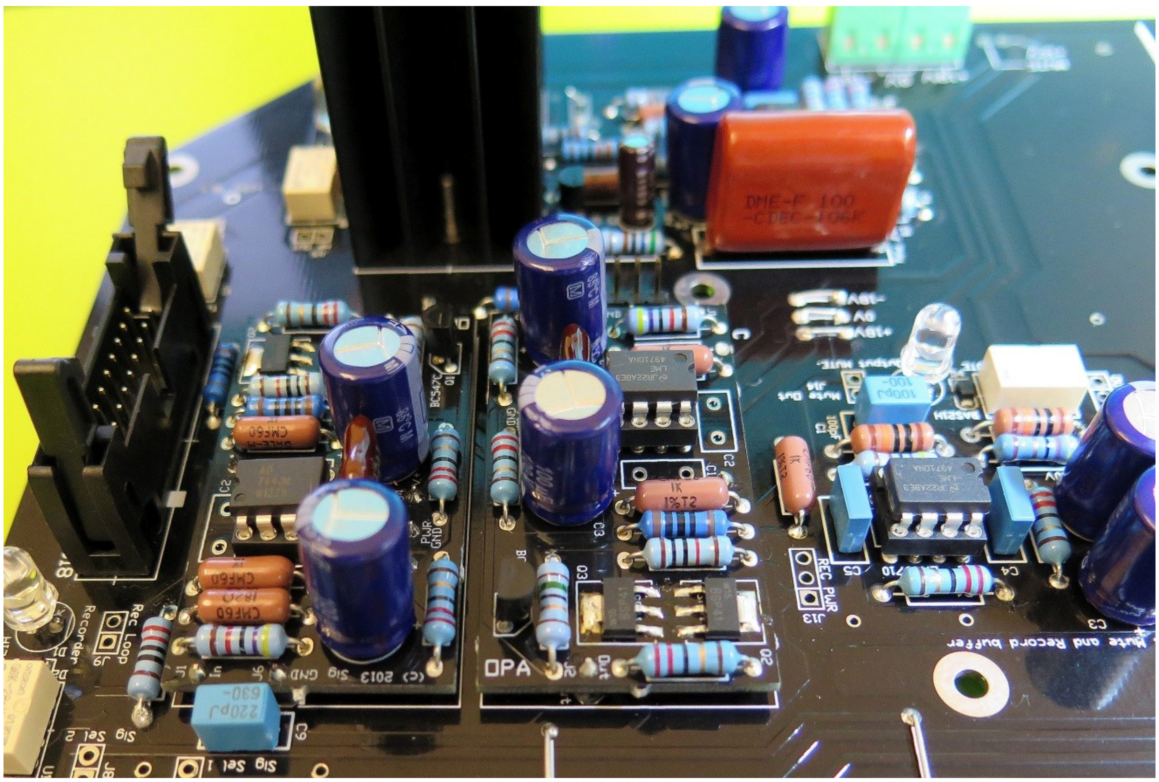
Figure 1 and 2 shows the problem more clearly. Dropping the feedback resistors by a factor of 10 offers a 20 dB improvement, and demonstrates a good reason to adopt the lowest value feedback resistors your application can tolerate. Careful layout, screening and the use of quality potentiometers will get a design the rest of the way to decent performance. However, non-summing junction topologies do not suffer from these issues, and any noise appearing between the feedback junction and the amplifier inverting input is amplified by the closed loop gain only, which in a high performance opamp based design, is theoretically a difference of over 100 dB compared to inverting variants. Of course, the issue I raise here applies equally to the Baxandall tone control, where the summing junction is fed from a resistor (from the bass side) and a capacitor (from the treble side). Again, careful layout is required to mitigate any problems. Let me stress, we are talking about noise pick up between the feedback network upper and lower resistors junction and the inverting input of the amplifier element, and not about the inherent noise performance of the inverting or non-inverting configuration.

On the Baxandall active gain stage, the summing junction input impedance at high gain settings can be very low, placing a heavy load on the opamp such that it would be exiting its class A region in the presence of very small output signal levels – just the opposite of what we would intuitively expect. I did some simulations, and the drive current required using a 1 k pot feedback element with maximum gain selected is indeed high at 10 mA. A good opamp (like an LM4562) can *easily* drive this type of load at 1-2V with single digit ppm distortion levels at 20 kHz; Thus, with say a 10mV output signal in this scenario you could expect 100 ppm distortion. However, I would not consider this a design flaw – maybe at worst an idiosyncrasy of the circuit. Besides, if it is of concern, it is a simple matter to place a class A discrete buffer following the opamp and ensure it is enclosed in the overall feedback loop. Self paralleled opamps in his design to reduce noise and this also solved the drive issue, so his Elektor preamp achieves very low distortion as a result.

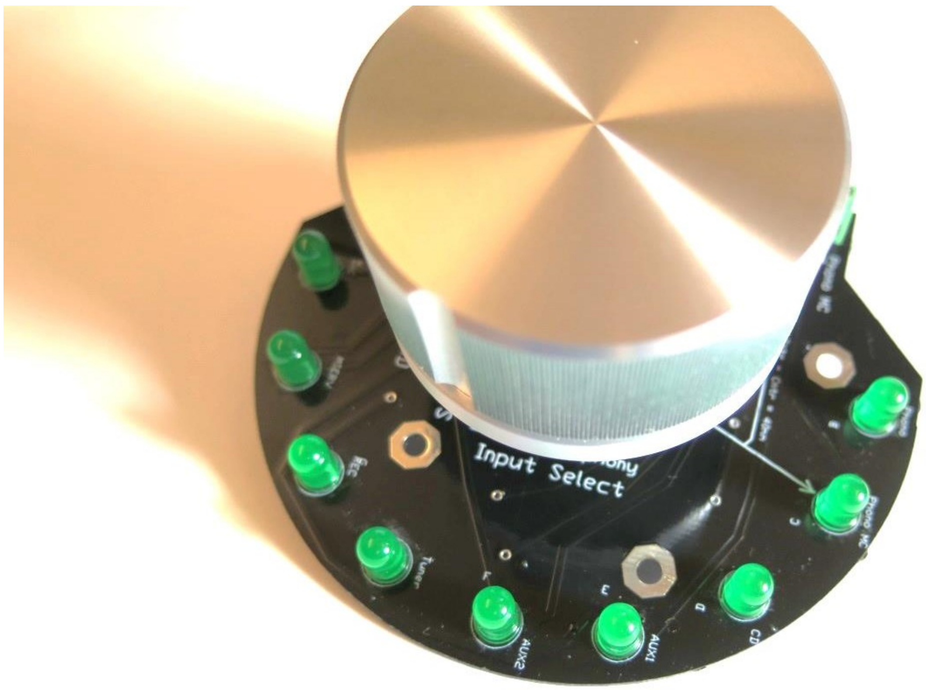


A more conventional approach might feed the input signal straight into a potentiometer (sometimes after buffering), placing a gain of circa 5x after this to provide signal level matching to the power amp, which typically would require 1V to drive it to full output power. In this scenario, we are placing a gain stage after the attenuation element, so the amplifier will always be contributing a fixed amount of output noise (ignoring for a minute the current noise contribution which will vary with potentiometer setting). So, at high attenuation factors (so low listening levels), the signal to noise ratio can be severely degraded if the gain element is not carefully chosen. The best signal to noise performance for this type of design is when the volume control pot is set to maximum, so the amplifier noise is masked by the high output signal levels; of course, the source will also probably have a low output Z, so the noise with no signal in is likely to be low in this situation in any event. Most pre-amplifiers use this approach, and with modern gain elements and volume potentiometers typically at about 10 k Ω, the overall subjective performance remains very good. A prime concern usually cited by designers who select this signal chain configuration is to do with overload capability. You can feed in a 2V CD signal into this type of preamp on one of the non-CD inputs (CD inputs are usually attenuated by 20 dB to bring them into line with tuner and recorder output levels which are 200 mV), and by simply adjusting the volume control can prevent any overload. This is how the X-Altra Mini One is configured, and how the attenuation in the PGA2320 is also accomplished (the PGA2320 also offers up to 31 dB of gain which is done by adjusting the feedback factor of the internal opamp gain stage above attenuator settings of +0dB) – this gives an improvement in S/N ratio at low attenuation settings when the opamp is running at unity gain. In my SCA-1 design, the maximum gain as set to 16 dB and the noise levels were extremely low.

However, the third alternative is to amplify the signal first and then attenuate. Further, you can provide a higher input impedance load to the source components, like 47 k or 100 k rather than the 10 k of the input volume control potentiometer, which can be useful if for example you are driving the preamp from a tube based cathode follower where the output impedance may be many k Ω. The output of the amplifier stage can be buffered, biased into class A and thus drive a low value volume control pot, for example 1 or 2 k, which brings with it improvements in noise performance in the follow-on buffer stage. This carries the overload risk, but gets around the noise degradation. But, the overload risk is greatly exaggerated in my view with this type of design. Most digital signal source outputs today (2014) are 2 V, while legacy sources such as tape recorders, analog tuners and so forth, around 200mV. Almost any power amp available on the market will be driven to full output with a single ended 1 V input signal, with some requiring 1.5 V.



This leaves us with two options: attenuate the digital sources by c. 6dB to get the 1V full scale output and amplify the legacy sources by ~14 dB to get 1 V out from 150 mV to 200 mV input full scale. For the Ovation Symphony, I chose the latter course of action, since I will be providing an MM and MC input facility, and possibly a tuner in the future. Further, if you attenuate by 6 dB on the digital source inputs like CD players or music servers and you assume quite reasonable 10k input impedance is required for the pad, this leaves you with a 2.5k source resistance (parallel 5 k resistors with the pot set at the electrical mid-point). You could buffer the digital source first and then drive a low impedance divider, say 2 k for an overall source impedance of 500 Ω, but there is added complexity and the distortion and sonic contribution of an opamp is not zero, whereas good resistors can easily achieve <100 ppb and they don't require a power supply, decoupling and so on. However, if you go the 20 dB padding route to level all the signal sources to the 150mV ~ 200mV range before boosting them back up by 16 dB to the 1 V required for the power amplifier, then overload can be avoided and you end up with the best of both worlds: Noise is attenuated along with the input signal but unlike the Baxandall active gain stage, there are no potentiometers – with the risk of picking up noise - hanging off a sensitive, 16 dB gain, summing junction. A 10 k 20 dB pad offers a source resistance of ~900 Ω to the 1st stage buffer/amplifier which can get you to the benchmark >110dB SNR ref 1 V out provided the gain device has decent low noise performance.



Ovation Symphony System Level Block diagram (Fig. 3). All of the inputs have jumper selectable -20 dB pads, feeding a low noise 14.4dB gain stage which then drives a 5 k Goldpoint log volume potentiometer (assuming the tone control block is switched out of the signal chain). With a nominal 200 mV input signal and the opamp stages running off +-15 V, assuming 12 V pk-pk undistorted output swing, the overload is 20 dB – plenty when the amplifier can be driven to full power at 1V input level. The 20 dB pads are built around a 9 k+1 k divider, so the worst case source impedance seen by the gain stage is about 1000 Ω when selected including the source driving impedance, and when fed directly from a source, much lower than this – you can safely assume on modern equipment 50-100 Ω. There is a small noise penalty to pay for this type of divider arrangement, but in my assessment it does not detract from the overall subjective sound of this pre-amplifier. In the worst case mid resistance setting of the volume control (2.5 k), the parallel combination of the two halves of the potentiometer is ~1.25 k. Since there is no amplification taking place after the potentiometer, only buffering, the noise contribution is very small, and referred to the input, the buffer stage noise contribution is divided by the gain of the preceding stages. Of course, if the tone control is switched in, the noise contribution of the tone control has to be factored in. But, again, because of the way the signal chain is configured, and the use of 5k potentiometers in the tone control section, even with full treble boost, this preamplifier is still incredibly quiet. Further, since for most listening, the volume control will be set between the 9 o'clock and 1 o'clock positions, noise generated by the preceding stages will be attenuated: you get all the benefits of an active volume control with, dare I say, none of the drawbacks.

***Small signal relays*.** I've read a lot of commentary on the web about relays in high-end audio applications. There seems to be a fear that after a while they will fail, or the contacts will become corroded or damaged, affecting the signal seriously. Small signal relays like the ones used here are incredibly reliable – after 10 million operations with a load of 10uA, the contact resistance is specified within 10 mΩs of the original 30~65 mΩs of the sample set at the start of the test - the device is specified at 100mΩs contact resistance. The Omron G6K is rated for 100 000 switching operations at full load, and 50 million mechanical operations - i.e. no or very low contact load. Some relays specify potential contact problems if continuously powered up due to outgassing of the plastics and insulation used in the relay. My X-Altra Mini One is never powered down (on 24/7 for months at a time) – I've had no problems on the AGN type relays. Another specification that people have concerns with is the minimum rated contact current – a typical spec being 10 µV~10 mV at 10 µA load current where there is a fear or concern that you cannot switch lower levels without affecting the sound. These specs are normally limited by the test gear – not the relay contact performance. Measuring and testing 10 µV~10mV/10 µA contact performance in is no easy task – thermoelectric issues between the contacts under test and the measurement gear for one pose a significant challenge – and trying to do this in a high speed automated test set-up would be expensive. Switches have exactly the same issues and for the most part they are not even sealed. Sealed relays keep atmospheric contaminants away from contacts, and at low signal switching levels, the gold clad contacts stay clean providing consistent contact resistance performance. Let's also not forget the high frequency performance of small signal relays – they are generally also quite capable of switching RF up to 20 or 30 MHz with minimal loss and therefore qualify as very wide bandwidth devices. A further important benefit of relays is that you can locate the switching close to the signal – long PCB tracks or wires with potential cross talk problems are avoided, as is the case with switches.

For the input source select relays, I used Omron G6K2P series devices. A good reason for their great performance of course has to do with the small physical size, silver with gold clad contacts, and importantly, the fact that they are sealed – so no issues with the ingress of atmospheric contaminants. Another great relay for this type of application is the Panasonic AGN – this is physically smaller, but also features a fully sealed, silver with gold clad contact construction. Neither of these relays is cheap, but they offer a long life and consistent contact performance. In this design, the relays are configured in a 'back terminated' arrangement so that the wipers are grounded when an input is not selected, resulting in very high 'offness'. In conventional arrangements, if you turn the volume fully up with a source playing on a non-selected input, you are likely to just be able to hear bleed through, even on a good layout due to capacitive coupling across and around the contacts – and the problem gets worse as the receiving input impedance gets higher. With the arrangement shown here, there is zero signal bleed through - the technique is very robust in this regard. Of course, for balanced inputs, you will need two relays, rather than the one shown here, so it quickly becomes an expensive proposition. However, in a top end design, which is what I am targeting here, this is not an issue.

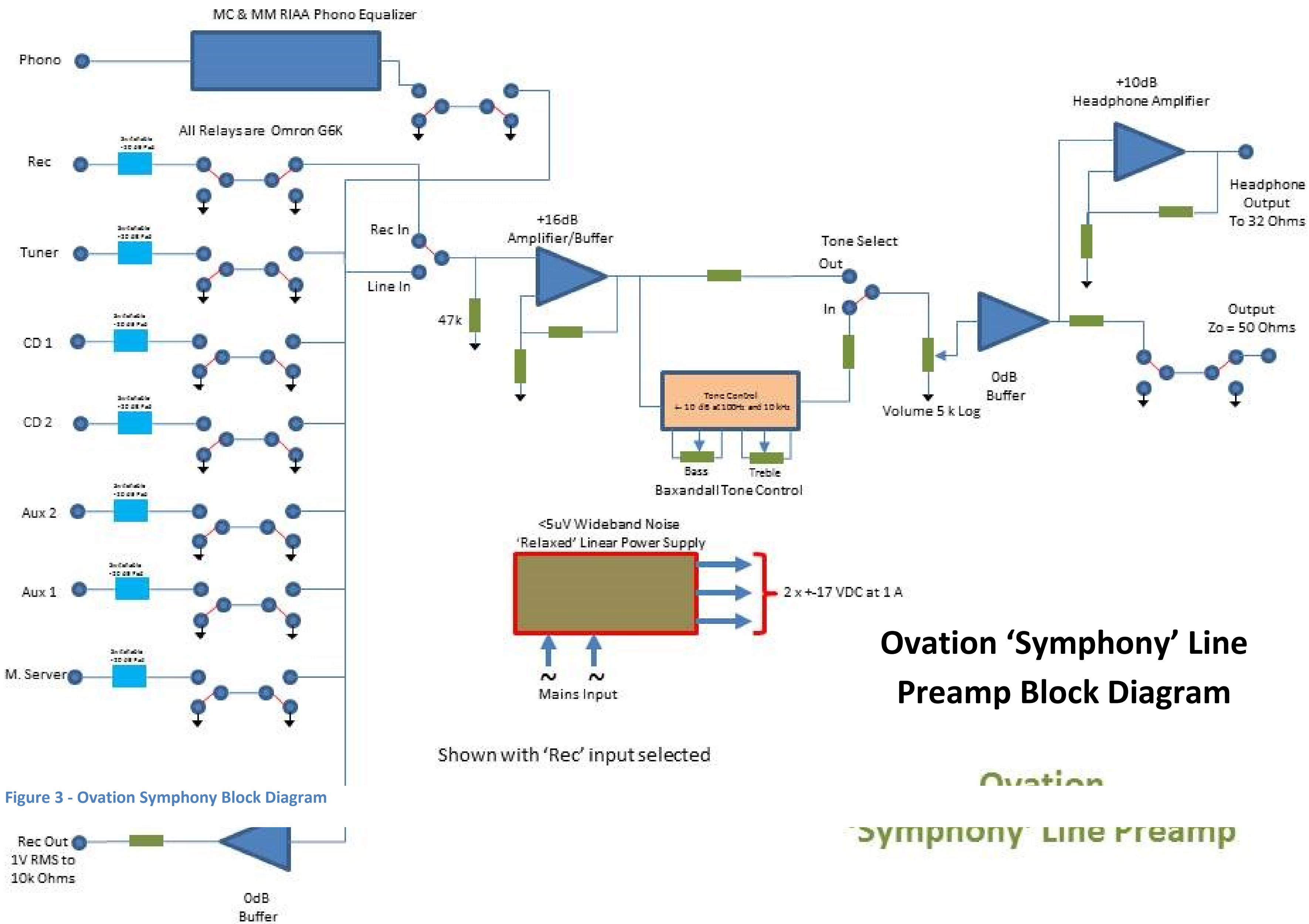
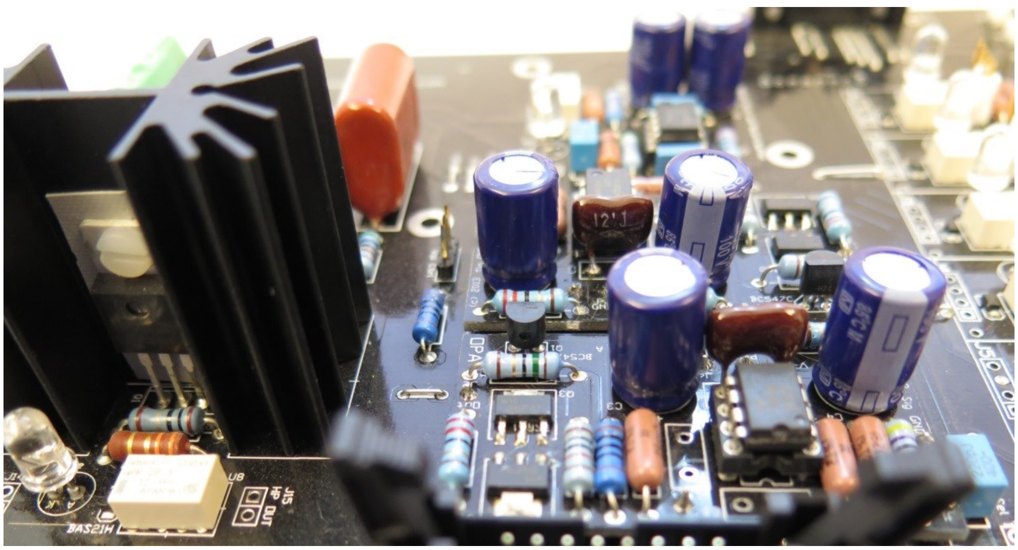


Figure 3 - Ovation Symphony Block Diagram

The tone control is located on a separate PCB, which also has the headphone socket, mute and tape loop switches. The tone defeat switch allows the tone control to be completely bypassed with the signal routed around the tone block. Note that no switches are used in any of the signal routing duties, including the tone control bypass – this is all done with the Omron – the front panel switch simply applies power to the tone bypass relay coil, located on the main board.

The output from the tone select relay feeds the Goldpoint 24 position log law attenuator, and from there it is routed to the output buffer and the headphone amplifier. The output buffer, like the input gain stage and tone control, is also an all class A stage capable of driving 200 Ωs to 10V pk in class A. For the headphone amplifier, I had the choice of going for an LM4990 buffer in an opamp feedback loop as I did on the SCA-1, but this is class AB, and my stated goal was 'all class A'. The end result is a 2W class A design that features under 10ppm distortion at 3 V output into 32 Ωs, while in class AB it can deliver ~4 W into 32 Ωs.



2. Specifications – Line Stages and Outputs

General Description:	High performance stereo line preamplifier featuring 7 single ended line inputs with jumper selectable 20 dB pad, MC/MM phono input, record loop, Baxandall tone controls (can be bypassed) and class A headphone amplifier; single ended output
1. Inputs	7 stereo single ended line level inputs with individual jumper selectable -20 dB 10 k Zin attenuator pads (located internally on main PCB)
	2 off stereo Phono inputs selectable on phono board for MM or MC (note, only one set of inputs can be active at a time - either MM or MC); Input selected via jumper on phono equalizer board (note: phono equalizer board has not been built as of June 2014)
2. Outputs	1 off single ended output (200 Ω max load)
3. Output Level	Nominal 1.5 V into 600 Ω at 1 ppm distortion 20 Hz to 20 kHz
	12 V into 600 Ω; 5 V into 200 Ωs at 10 ppm distortion
	1.3 V into 50 Ωs at 25 ppm distortion
4. Tone Controls	Baxandall active type +-9 dB cut and boost at 100 Hz and 10 kHz
	Tone control can be switched completely out of signal chain
5. Frequency Response	With tone control switched out: 20Hz to 20 kHz +0 dB, -0.1 dB; DC – 200 kHz -3 dB
6. Distortion	1 ppm (-120dB) at 1.5 V into 600 Ω 20 Hz- 20 kHz
	10ppm (-100 dB) at 12 V into 600 Ω 20 Hz to 20 kHz
	5ppm 1V RMS into 50 Ωs
7. Noise	-110 dB (~2μV) ref 1 V into 600 Ω (Zin source 50 Ω)
	Better than -130 dB ref 10 V into 600 Ω
8. Cross talk	Better than -80 dB at 20 kHz; better than -100 dB at 1 kHz
9. Headphone Amplifier	Max Output in class A is 2W RMS into 32 Ω. 3 V peak into 32 Ω class A at <3 ppm 20 Hz to 20 kHz; 6 V into 32 Ω class AB <13 ppm 20 Hz to 20 kHz; ~500 ppb distortion at 500 mV into 32 Ω; 12V peak into 32 Ω < 60ppm distortion; Freq. Response 2Hz to 150 kHz -3 dB
10. Front Panel Controls	Input select, Volume, bass, treble, tone control bypass, record loop monitor; 1 x headphone output socket (6.5 mm); Power On/Off located on rear panel.
11. Rear panel connections	RCA sockets: 7 x stereo line inputs (incl. 1 x record loop), 2 x stereo phono inputs (only 1 can be active); 1 set stereo outputs; 1 x record output (buffered, class A into 10 k Ω load); 1 x switched fused 3 pin IEC mains receptacle
12. Power Consumption	25 Watts
13. Dimensions	480mm x 150mm x 300mm
14. Weight	10 kg's

Specifications RIAA Phono Equalizer[1]

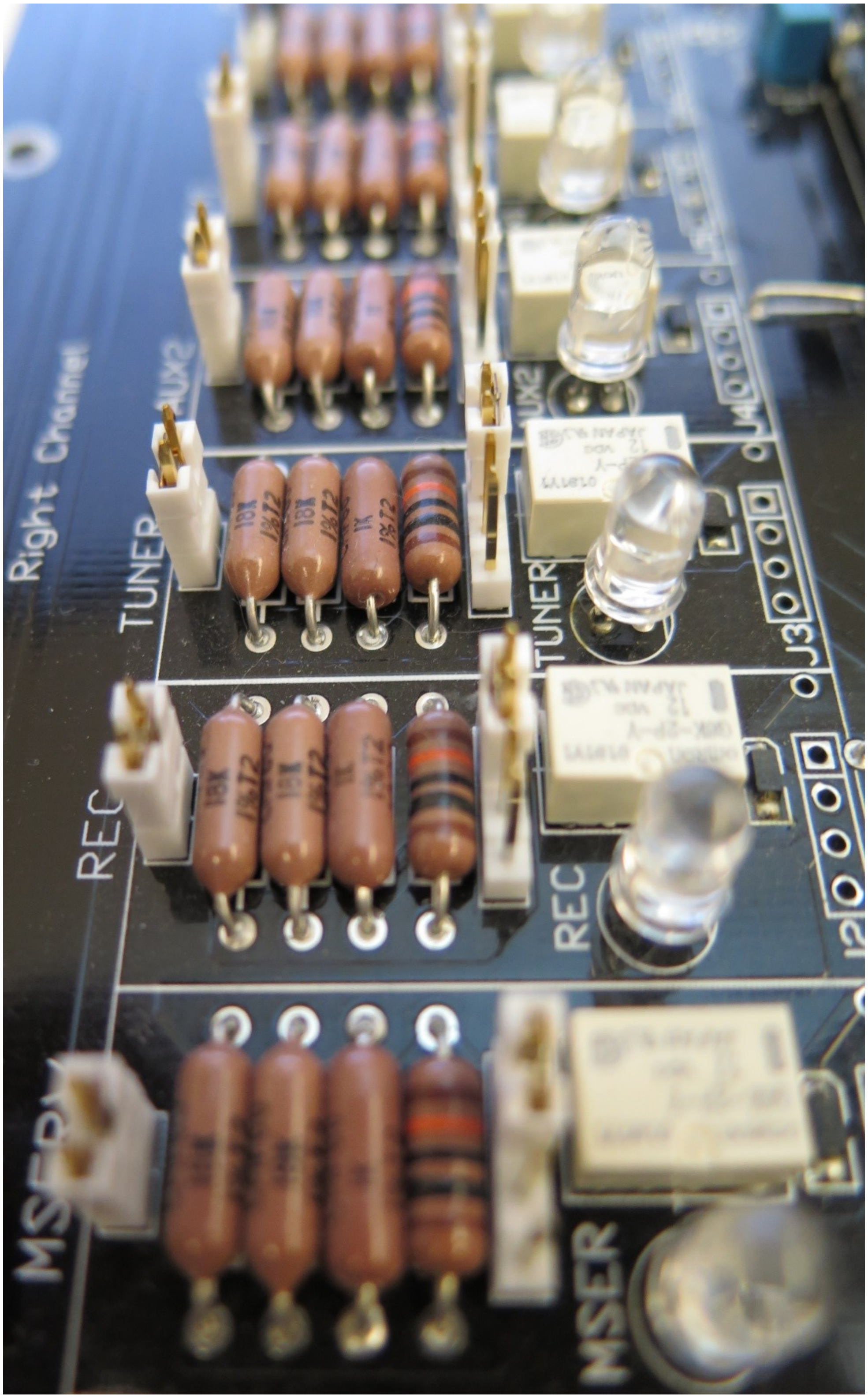
MM Inputs

1. RIAA Conformance	<0.15 dB 20 Hz to 20 kHz
2. Distortion	<2 ppm at 1 kHz 200 mV Output into 10 k Ω
	<3 ppm at 1 kHz 1.5 V Output into 600 Ω
3. Gain	DIP Switch selectable for 30, 36 dB or 42 dB
4. Output impedance	22 Ω at 1 kHz
5. Overload Capability	>40 dB ref 3 mV input 20 Hz to 20 kHz; >35 dB ref 5mV input 20 Hz to 20 kHz
6. Noise	- 82 dB ref 5 mV input (20 Hz to 20 kHz measurement bandwidth) un-weighted
7. Topology	Discrete JFET front end, fully active equalization driving in-loop class A buffer
8. Cartridge load	Capacitive 39, 68 100 220 pF via DIP switch on equalizer PCB
	Resistive 22k, 39 k 47 k, 68 k 100k k via DIP switch on equalizer PCB
9. Channel crosstalk	< -90 dB at 1 kHz (alternate input short circuited)
	< -70 dB at 20 kHz (alternate input short circuited)

MC Input

1. Gain	14, 20 or 26 dB DIP switch selectable
2. Distortion	< 500 ppb at 1 kHz ref 100 µV input and 3 mV output 20 Hz to 20 kHz
3. Cartridge load	Switch selectable 10, 22, 47, 100, 220 Ω; 47, 100, 220, 470 pF via DIP switch
4. Output impedance	2.2 Ω at 1 kHz
5. Frequency Response	20 Hz to 20 kHz +0 dB -0.2 dB; 20 Hz to 200 kHz -3 dB
5. Overload Capability	>60 dB (MC amplifier only); for complete RIAA MC signal chain, same as MM input above
6. Noise	<0.3 µV equivalent input noise (20 Hz to 20 kHz) 10 Ω Source impedance
7. Topology	Single ended discrete bipolar input transistor stage + Integrated circuit gain block

[1] Note: MM and MC equalizer board still to be built as of June 2014


Right Channel
AUX2
TUNER
REC
J3
J4

3. The Gain and Buffer Line Level Stages:

The Foundation of Good Sound

For the gain block to be used throughout this preamplifier, I wanted to keep my options open for possible future changes where there is a clear choice between discrete vs opamp, bipolar vs JFET, and balanced vs single ended. Given the low impedances involved to keep noise to respectable levels, output drive requirements would necessarily be high. Also to be carefully considered was the class A vs class B question. Self’s Elektor preamp exited class A operation at low signal levels – I decided as a design goal, I did not want this to happen at any signal level in any of the amplifier stages. Furthermore, as we will see in the power supply design section, running in class AB can spray a lot of HF current harmonics onto the supply rails which I also wanted to avoid – I problem all the more exacerbated with heavy loads.

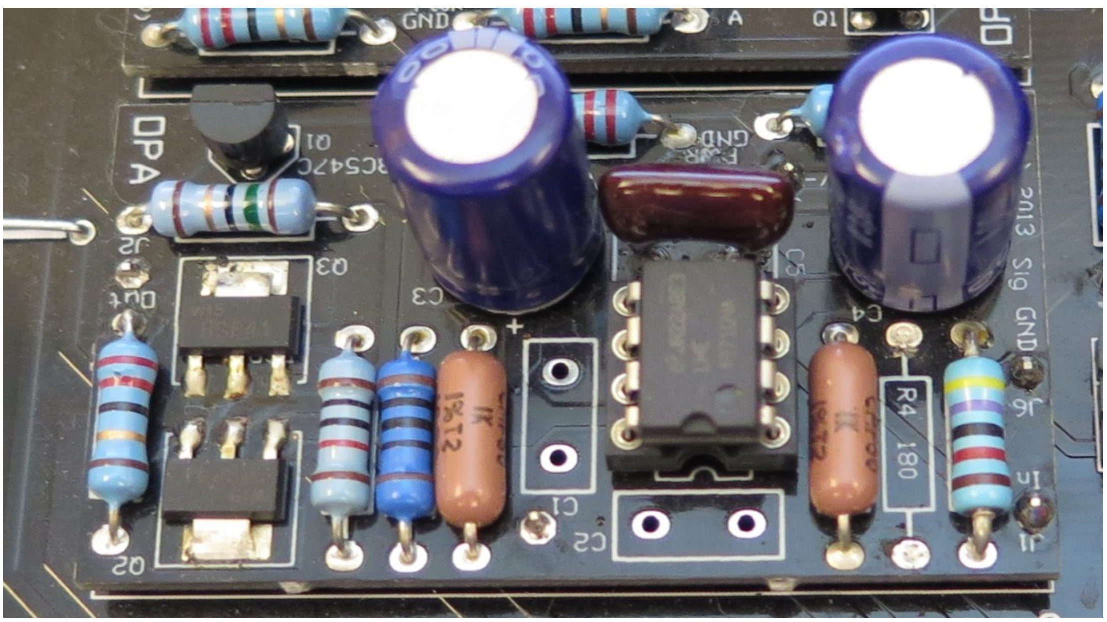


Photo 1 - Symphony Class A Opamp Gain Stage

With these points in mind, the Symphony preamplifier gain stage design goals evolved as follows:-

- Each gain stage would be in the form of a small module – final dimensions ~60mm by 25mm. This would allow me to experiment with both IC and discrete designs, although the initial iteration of the Symphony preamp which you are reading about here uses IC opamps
- For lowest distortion, the module uses an opamp bootstrapped into class A, followed by a discrete class A buffer which is inside the opamp feedback loop. At 2V out into 600 Ω, the amplifier module targets distortion well under 1ppm at 20 kHz, and will deliver 10V pk into 200 Ω whilst remaining in Class A

- As mentioned, Class A operation means the noise reflected into the supply rails is much lower, and its lower order harmonics, unlike class AB operation.

Opamp Based Design. (Refer Fig. 7) The PCB layout allows any 8 pin single opamp to be used as the compensation cap – if required – can be located across pins 1 and 5, or 5 and 8. No facilities for dialing out any offset are provided, since opamp offset and drift are so good nowadays, that this is not necessary. The output buffer section is a simple SE current source loaded design. The output current setting resistor (R3), can be adjusted to set the class A current, which in this preamp design is either 25mA or 50mA corresponding to 22 Ωs or 12 Ωs. The output devices (U2) are SOT223 BCP41 from NXP and with a small stick on heatsink, the module can comfortably dissipate about 1.5W – although I found during testing that the heatsink was not necessary even for the 50mA output version. As will be discussed in the power supply section, tight localized decoupling (22 Ω and 100uF electrolytic capacitors) around the while opamp module keeps HF noise off the power supply rails and reduces radiated noise from the supply lines. R5 is a base stopper to prevent potential HF oscillation, while R2 (1 k) bootstraps the opamp output stage to ~0.6mA, so that it also always operates in class A mode. The AD797 and LME49710 used in this design are high speed, wide bandwidth and very high gain devices, and as a result I spent quite some effort ensuring that the layout was optimized to minimize the stray capacitance at the inverting input. The modules do not use SMD resistors or capacitors, so lead dress issues on the through hole components had to be considered carefully.

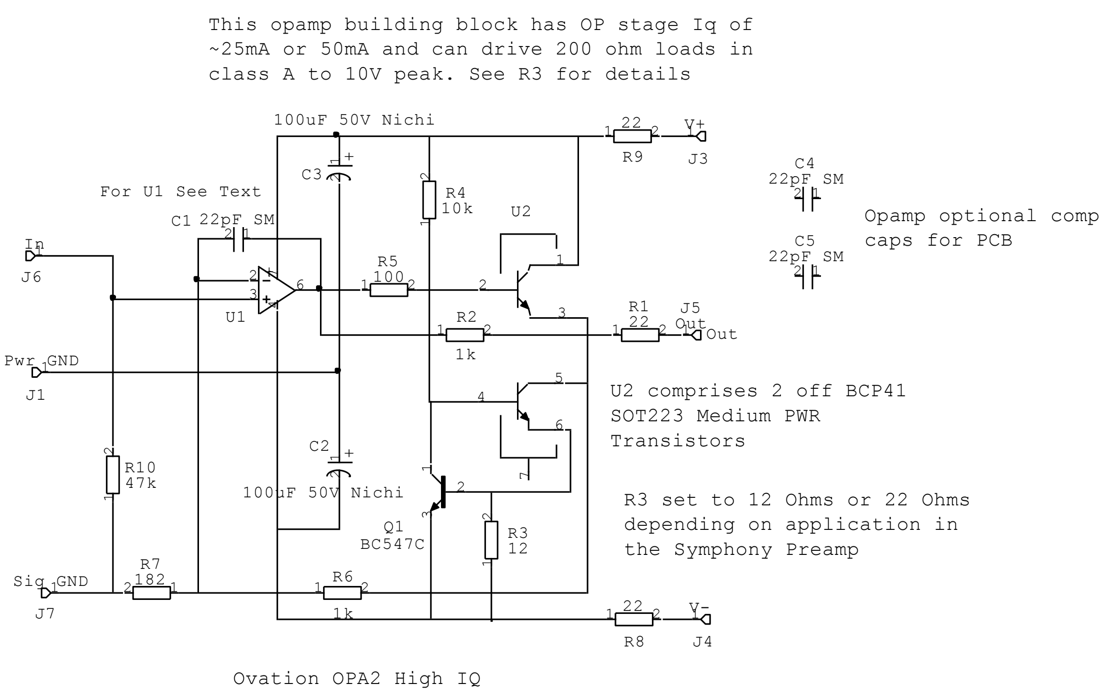


Figure 4 - Symphony Non-Inverting Opamp Based Gain Block

Figure 5 – A simulation of the power rail harmonics of the class A gain block (10V pk into 300 Ω at 20kHz)

Fig. 5 shows a simulation of the reflected power rail current harmonics with the amplifier stage driving 10V pk into a 300 Ω load – they are all innocuous low order which means they will couple less readily to adjacent circuitry than would be the case with higher order ‘spray’ harmonics – which is what happens with class B.

Figure 6 – Simulation of Class AB Operation Supply Rail Harmonics (20 kHz @ 10V pk into 300 Ω with 3mA OPS bias current). The lower order harmonics are in the 100 to 300uA range.

Fig. 6 shows the supply current waveform with a class AB push-pull output stage with a respectable 3mA OPS bias current – a level already much higher than almost any commercially available opamp. Now, this state of affairs is of course because the rail current is being delivered as half have rectified pulses[2]. When playing a music signal, with widely varying frequency content, it clear that the rails will be full of load current hash extending into the 100's of kHz or even higher that can easily couple into sensitive circuit nodes – as we have seen earlier, just a few pF at HF can cause problems. It should also be clear why decoupling and layout is so important, and one could postulate may go some way to explain why experienced listeners are sometimes critical of opamp based designs where bad layout and deficient decoupling along with high loop gains (which in electrical engineering terms should be a virtue) can come together to create an ugly sonic cocktail. Comments like 'lacking micro-dynamics', 'glassy',' grainy', 'edgy' and so forth emerge despite what may have been lots of effort to reduce distortion in the design may be attributable to some of these problems. This is where distortion analysis tools like an AP, or a good sound card with associated software can really help to identify these issues.

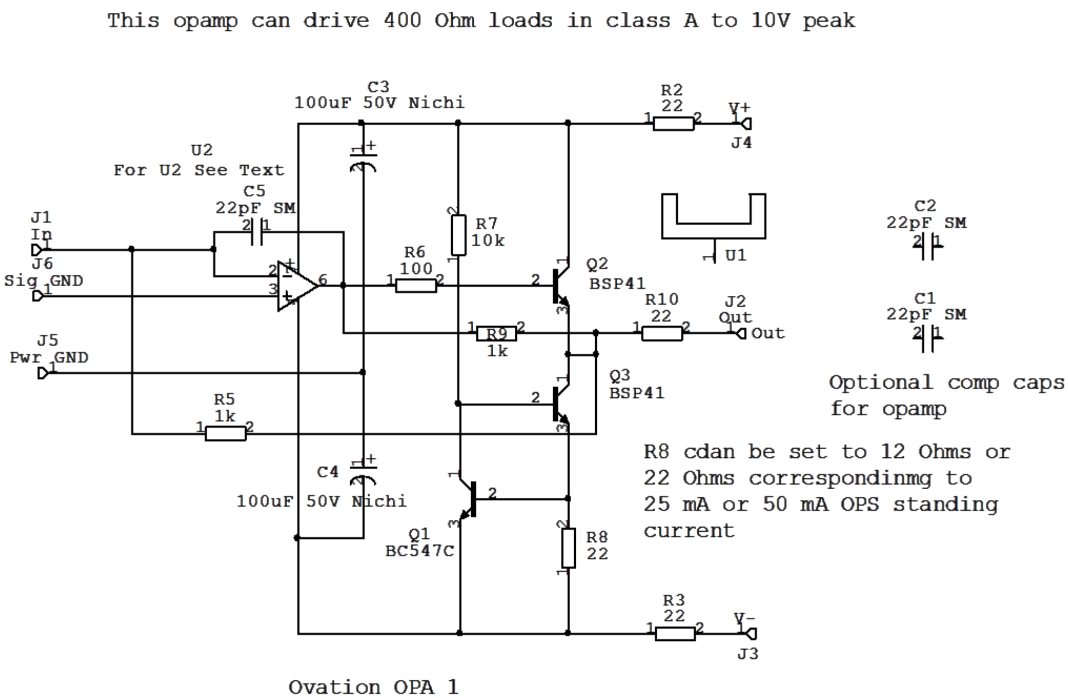


Figure 7 - The Inverting Opamp Module (details not described in the text, but operating principles are similar to those of the inverting opamp module shown in Fig. 5)

[2] Note: many opamp models do not accurately represent supply rail current – for these plots the opamp OPS was biased into class A to 1mA. The rail harmonics depicted in the plots are those arising purely from the discrete OPS.

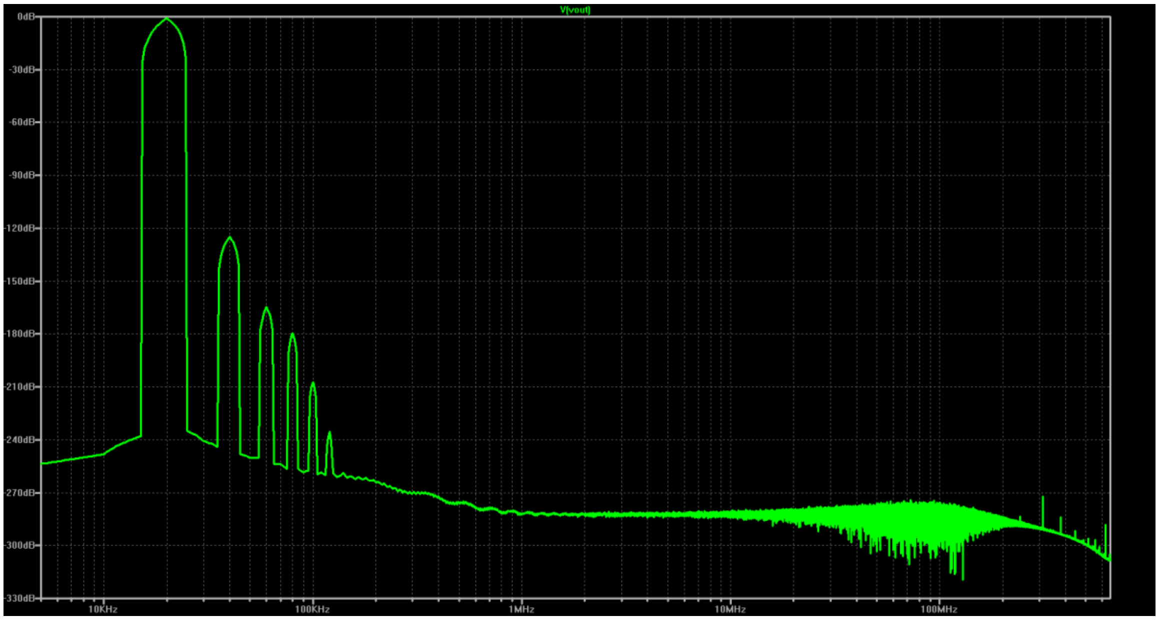


Figure 8 - Line Amplifier (Av = 6.5x) Stage Distortion Performance – 1.2 V into 700 Ωs (simulated)

Figure 8 shows the line amplifier (Av = 6.5x) FFT distortion performance driving a 700 Ω load at ~1.2V peak – the distortion is below 700 ppb (less than -120 dB). With a 700 Ω load and a peak output of 9 V the distortion around 7ppm. The output buffer stage (Av = 1) distortion is about 420 ppb or just a little shy of -130 dB into 700 Ωs, while with a typical power amplifier input load of 10 k Ω and 1.4V peak output, it is under 350 ppb, or a little over -130 dB down from the peak output. Finally, at 1.2V peak into 200 Ωs, the distortion is around 350 ppb. All these tests are at 20 kHz.

Noise considerations. Since we have a hefty class A output stage, the feedback resistor values can be low in value and we get reduced noise as a result. The Symphony preamp gain structure is shown in Fig. 9. The worst case noise occurs when the input attenuator is selected (900 Ω), the tone control is in the signal chain (assumed to be set to the flat position so that the input impedance is at 720 Ω) and the volume control is in the electrical mid position. The first stage is the high gain stage (Av = 6.5x). Any subsequent stages will have their noise contribution, referred to the input of U1, divided by U1 stage gain and any interposing stage gains – in this case, they are all unity (tone control set to flat or bypassed).

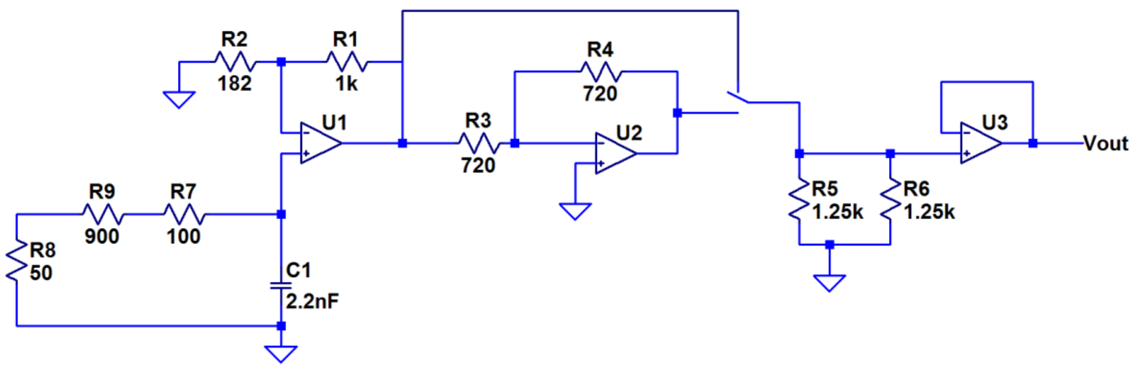


Figure 9 – Worst case Symphony preamp gain and source resistance structure (Line stages) – Tone Control set to flat position

The minimum noise contribution referred to the input source resistance for a given opamp is where the total input referred noise of the opamp (en_{TOT}) most closely approaches that of a pure resistor (see Fig. 10). To the left of this point, opamp input noise voltage dominates, and to the right, opamp noise current x source resistance dominates. So, it's at this point that the given opamp delivers the best S/N ratio referenced to a pure resistor. Note that the optimum source resistance for any op-amp is 0 Ω because this is where there is zero thermal noise contribution from the driving source.

I took Steve Hagman's very good 'Noise Visualizer' spread sheet from his EDN article and modified it so that it could accept 7 opamp noise inputs, as shown in Figs. 10 and 11 covering the LME49990, LM4562, LME49710, NE5534A, AD797, AD4898 and OPA627. Below 400 Ω, the AD797, LME49990 and the AD4898 are the winners with the AD4898 clearly ahead of the others at low R_{source} up to about 300 Ωs. In the 300 Ωs to - 2 kΩ range, it is joined by the AD797 and LME49990. These three devices therefore are good for buffering volume control pots of 5~10k Ω since at the worst case mid resistance position, the source impedance seen by the opamp is 1/4 the element resistance assuming you are using a low driving source - for a 10k pot, this is 2.5k. From 2000 Ω to 15k, surprisingly it's that old workhorse, the NE5534 with its low input noise current, despite its bipolar input, of 0.4nA/rt Hz making it the winner. It's a great pity that the IP bias currents on the 5534 are so high - 200nA typical and as high as 1uA. More modern opamps use input bias current cancellation techniques to achieve better performance in this regard. However, given its age, it is still a remarkable opamp.

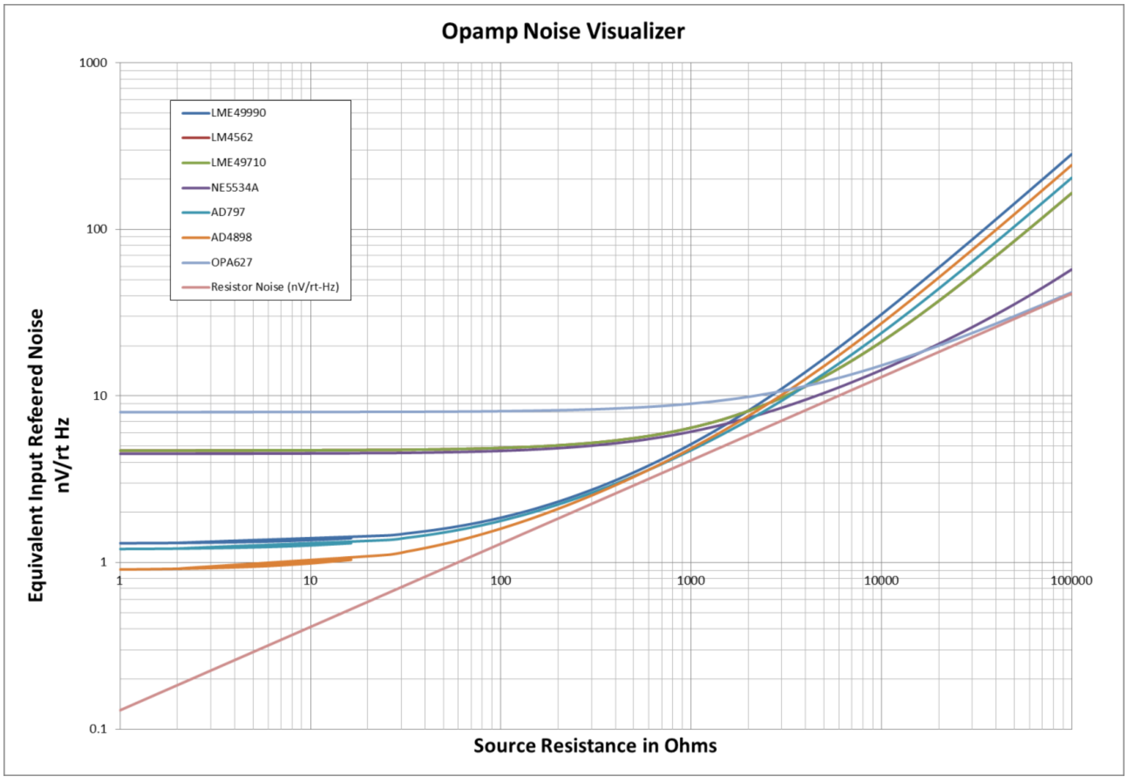


Figure 10 – Opamp Noise vs Source Resistance. The AD4898 is the quietest at 100 through to about 700 Ωs R_{source}; Between 700 and 2k Ωs, the AD797 is best while from 2 kΩ to 15k Ωs the 30 year old NE5534 is the best performer. Above 15 kΩ, the OPA627 is the best opamp.

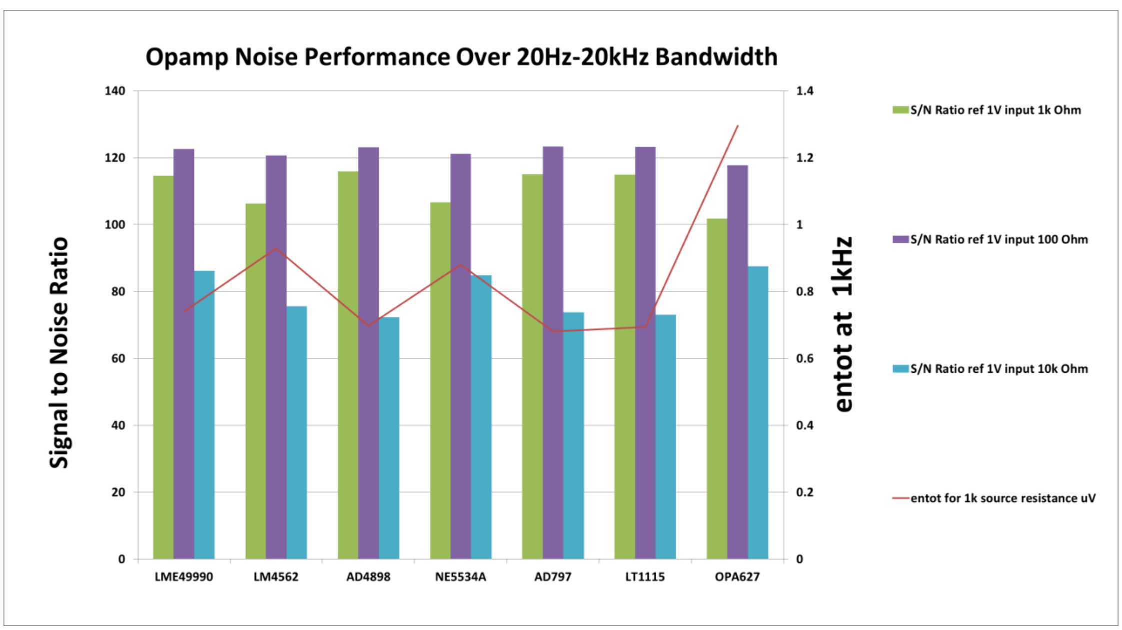
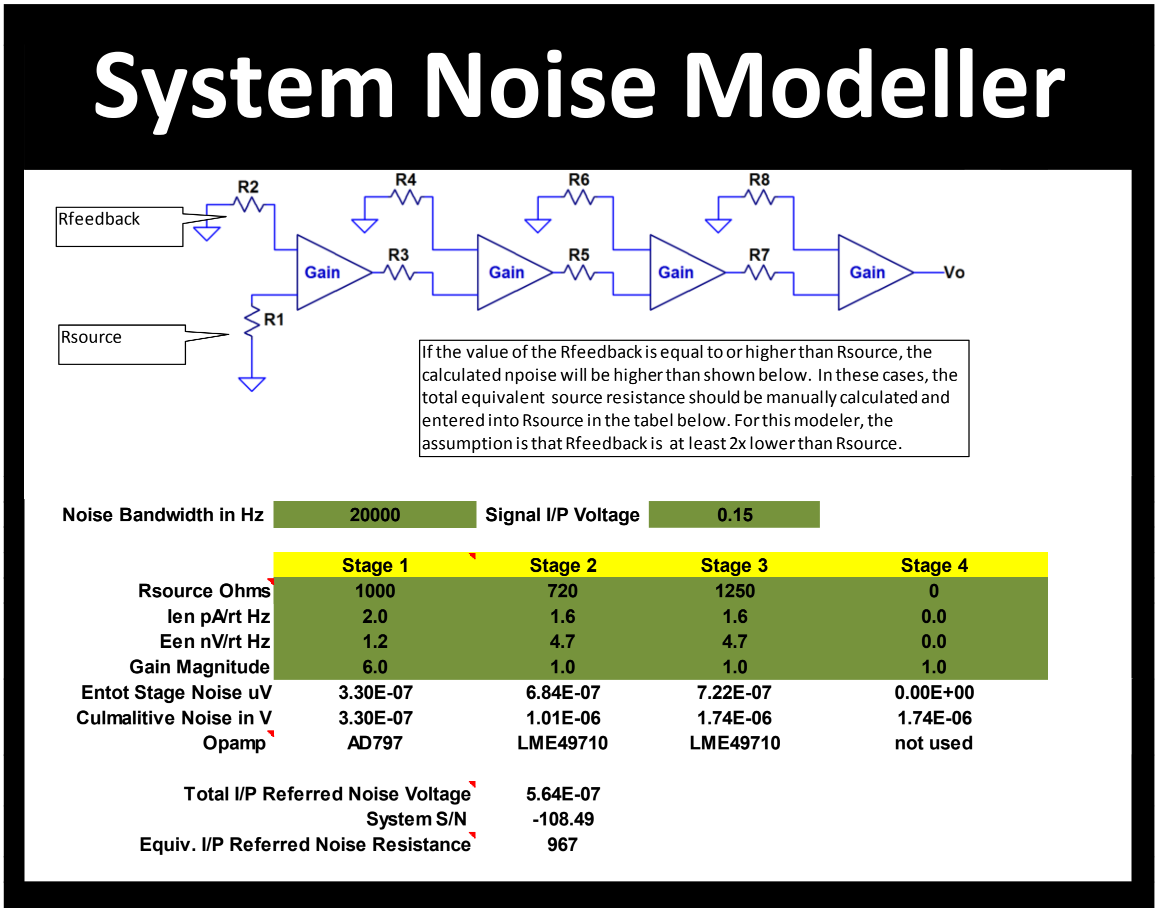


Figure 11 – S/N ratio for 7 high performance opamps at 1k, 100 and 10k Ωs source resistances (higher is better in this graph)

To support the design process, I developed a system noise modeler in the noise visualizer spread sheet – See Fig. 12 for a screen shot. This allows up to 4 gain stages to be modelled, and calculates the total input referred noise. You can download a copy of all of these tools here:- noise visualizer.



Noise Bandwidth in Hz	20000	Signal I/P Voltage	0.15	
	Stage 1	**Stage 2**	**Stage 3**	**Stage 4**
Rsource Ohms	1000	720	1250	0
Ien pA/rt Hz	2.0	1.6	1.6	0.0
Een nV/rt Hz	1.2	4.7	4.7	0.0
Gain Magnitude	6.0	1.0	1.0	1.0
Entot Stage Noise uV	3.30E-07	6.84E-07	7.22E-07	0.00E+00
Culmalitive Noise in V	3.30E-07	1.01E-06	1.74E-06	1.74E-06
Opamp	AD797	LME49710	LME49710	not used
Total I/P Referred Noise Voltage		5.64E-07		
System S/N		-108.49		
Equiv. I/P Referred Noise Resistance		967		

Figure 12 - Screen shot of the Noise Modeler

Opamp final choices. For the input buffer amplifier stage I selected the AD797, which features a worst case e_{en} of 1.2nV/rt Hz and I_{en} of 2pA/rt Hz. For input impedances of 700 Ω to about 2k Ωs, the AD797 is hard to beat - other opamps may feature lower e_{en}, but their i_{en}[3] is correspondingly higher. The newer AD4898 is about 1dB better than the AD797 at 1 kΩ, but is only available in SOIC. The TI LME49990 is also a very interesting device, but again, only available in SOIC and about 9dB better than the 49710/LME4562, which in the company of these low noise stars look decidedly pedestrian. The app note for the LME49990 indicates an RC network between from the output to ground may be needed for stability which means additional complexity. The AD797 is expensive at about $10 from Mouser, and since, referred to the input, the noise generated by succeeding stages is effectively divided by the gain of earlier stages, using this very expensive opamp in the tone control and buffer brings virtually no

[3] e_{en} = equivalent input noise voltage; i_{en} = equivalent input noise current

improvement – only about 1dB. From this, you can immediately see the benefits of placing high gain, low noise amplification at the front of the signal chain, which is what has been done here. As a result, the tone control and output buffer use the lower cost but somewhat noisier LME49710.

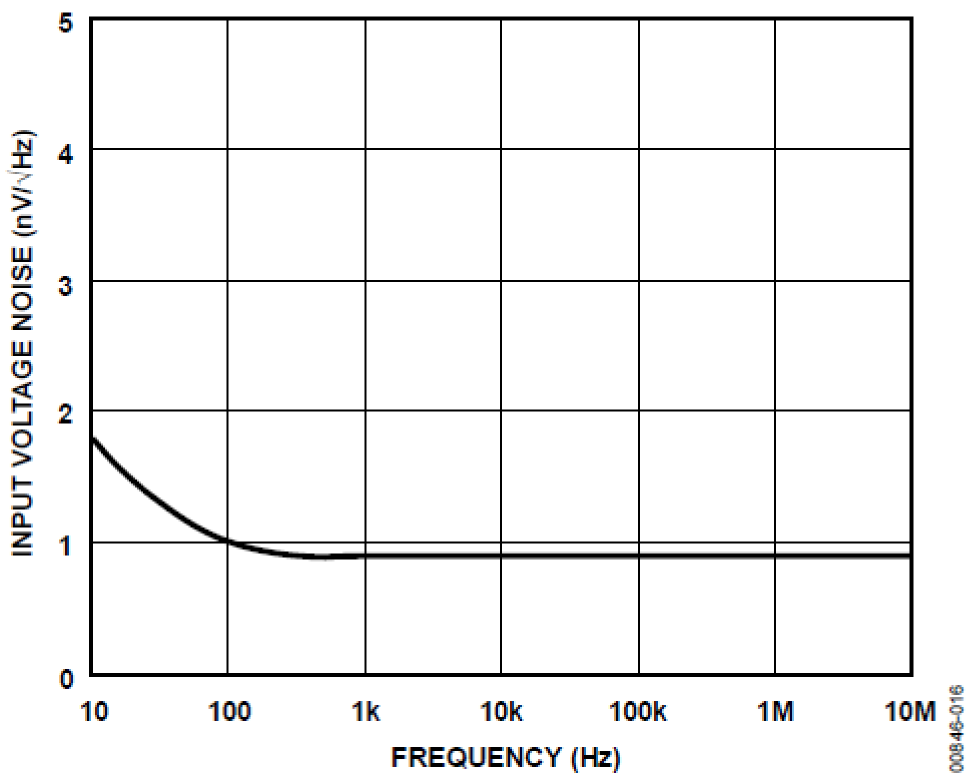


Figure 13 - AD797 Noise Performance

With a 150mV input reference signal, the Fig. 12 results point to an expected -111 dBV noise performance with a best case 50 Ω R_{source}. With a 1k source, as would be the case with the 20 dB pad selected, this drops to a still respectable 108 dB. If you try out the noise modeler, you will note that lowering the second or third stage source resistance to 1 Ω, makes only about a 1 dB difference to the overall noise performance, and this if course is because the noise of subsequent stages is swamped by that of the first. The best way to get any further dramatic improvement in the noise specification would be to go for a discrete JFET input stage using something like a LSK489 or LSK389 whose e_{en} matches that of the AD797, but feature negligible i_{en}: - but then some very dramatic tradeoffs in terms of distortion, PSRR and so forth would be required. Personally, I think I have the balance about right on this one . . .

What about distortion? The TI LME devices specify distortion level of 300 ppb at 3V into 600 Ω, while the AD797 data sheet indicates -110 dB *worst case* at 20 kHz, 3V RMS into 1 k Ω. Since we will be biasing the opamp up into class A[4], and will be driving a light 600 µA load, the distortion will be very low – however, this can only be confirmed with AP measurements which I hope to get done in due course. The *typical* distortion figure for the AD797 is quoted as -120 dB 3V RMS into a 1k load at 20 kHz.

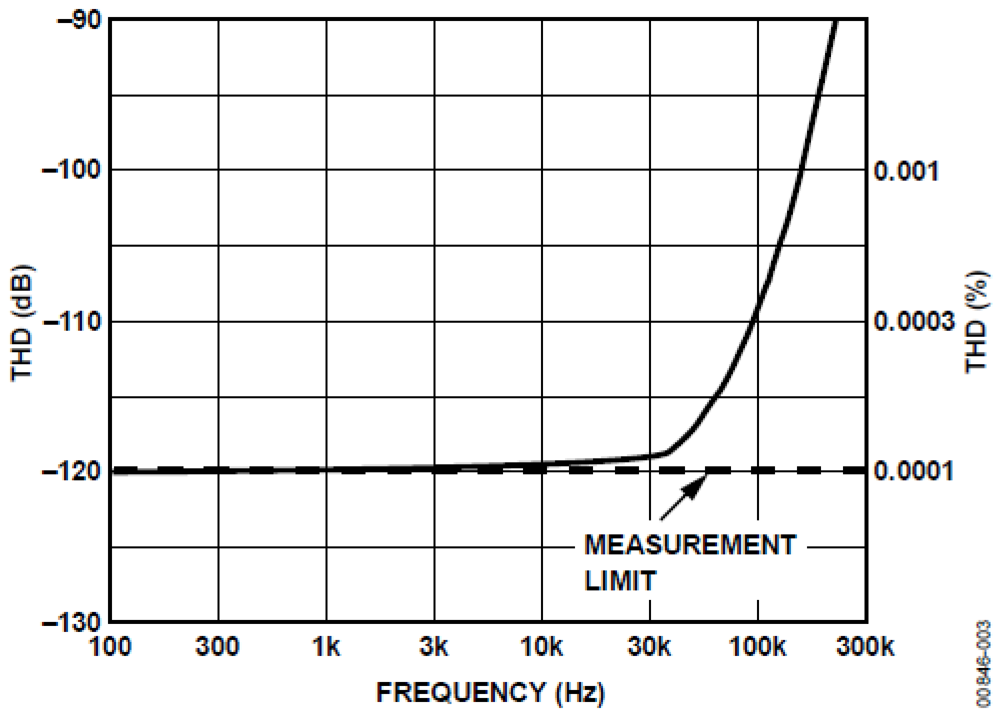


Figure 14 - AD797 Distortion Performance- the measurement limit of the test setup is 1ppm

The best distortion performance point vs output voltage swing is not at the lowest output voltage because it is swamped by the measurement equipment noise floor and non-signal level related residual distortion of the DUT - you can see that effect in Fig 14 and also in the measurement results in the X-Altra Mini One write up. Details on exactly how the AD797 distortion was measured are not given, but from Fig. 14 which was extracted from the data sheet, its clear that it is extremely low.

I did not fit any distortion reduction capacitor on the AD797 as detailed in the app note – I can't hear the difference between -110 dB and -120 dB distortion, and neither can anyone else for that matter. Simply adding components that may show a small difference on an AP distortion plot does not add any real value in my view.

You may also note that each module output exits the board via a 22 Ω series resistor. This is included to help isolate the module from capacitive loads. In this preamp, the module loads amount to few hundred pF maximum, but I felt it was best to take precautions. The opamps

[4] I assume here almost all the distortion in a well designed opamp is from the class AB output stage as it transitions from class A to class B

used are all very high gain, very high GBWP devices and it's quite easy to get them to oscillate at a few MHz if you are not careful about any capacitive load and capacitance between the inverting input and ground. As a general point, it should be noted that as opamp open loop gains get larger, and the bandwidths get wider, greater care in general is required with layout, decoupling and so forth. There are plenty of things you can get away with on a 5532/34 or a 301 that will cripple a design using some of the modern devices discussed above because of this. See this write-up for some ideas on what to look out for.

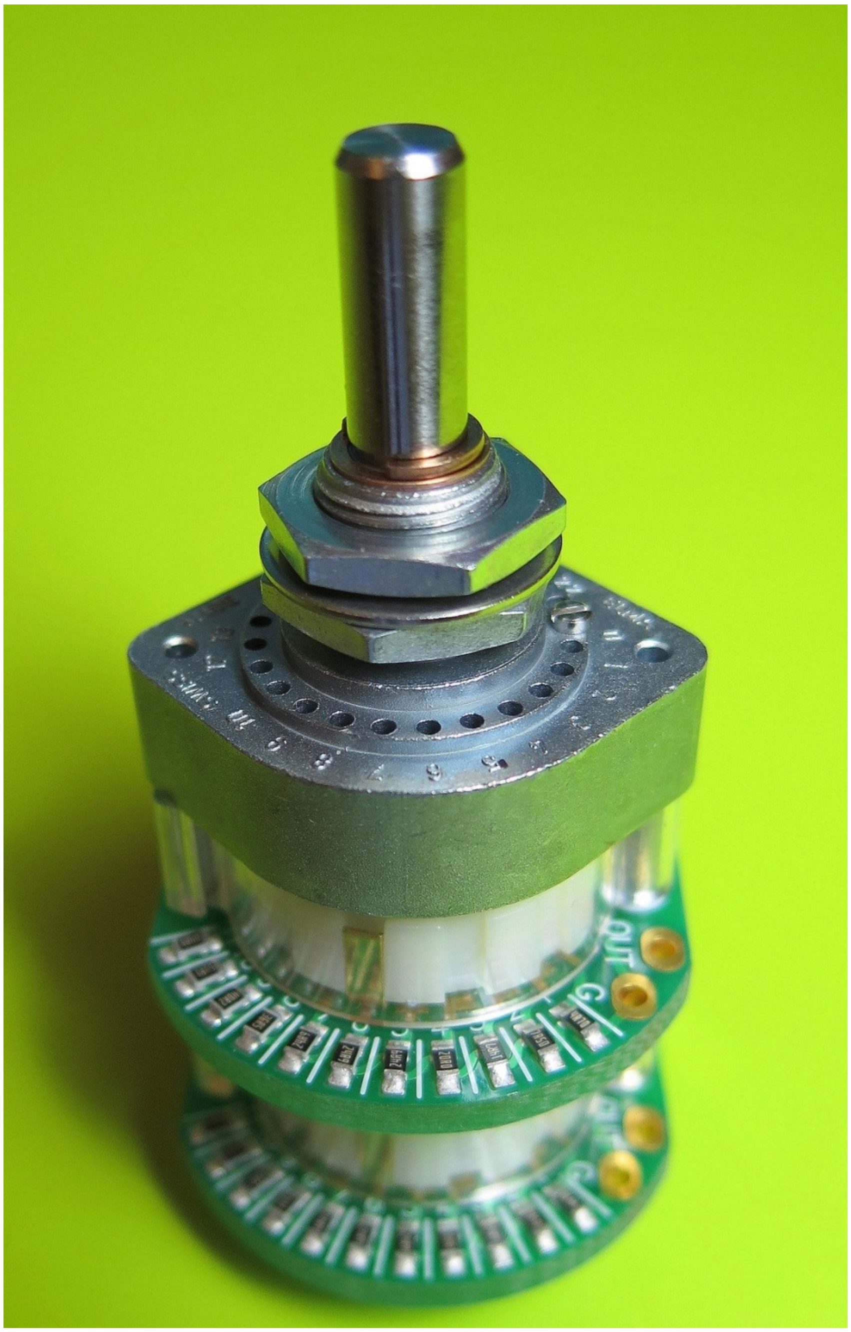


Figure 15 - Goldpoint Level Controls 5k Log Law Attenuator uses precision thin metal film resistors configured in a 24 position ladder control arrangement for ultra low distortion

4. Tone Controls: Getting the Balance Right

Many recordings are 'light' and what I mean by that is that they can lack some weight in the bass registers, while fewer recordings seem to have treble issues, although some are overly bright. Foremost amongst tone control circuits is the Baxandall configuration which has gone through 50 years of gradual refinement and simplification. Boost, cut and bass/treble response are all symmetrical, and use readily available linear potentiometers.

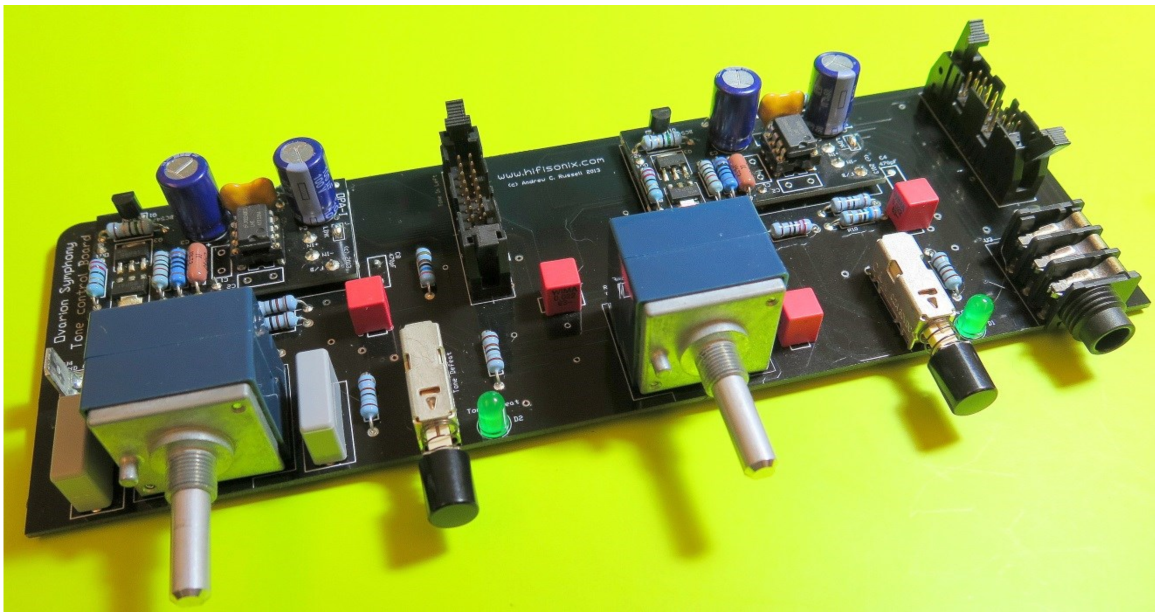


Photo 2 - Symphony Baxandall Tone Control Board. The PB switches activate relays on the main board for signal switching via separate ribbon cables for left and right channels. The tone control board uses an inverting configuration amplifier stage with the same class A buffer as the non-inverting stage

It is important to acknowledge that some tone control designs from the early age of transistors and opamps were very sub-optimal, and this indeed would have lead audiophiles to turn against them. On discrete based designs, passive tone controls were often used, and these required high gain circuits after the passive network that were noisy and distorted to say nothing of the fact that they required log pots and were rarely symmetrical in terms of boost and cut, or the frequency extremes – so you might typically get more boost than cut at bass, and in the treble registers, a significantly different response curve. Log pots are needed because the central position of the passive tone control pot is not the flat response position, unlike the Baxandall. Sometime after the Japanese hi-fi wave, which started in the 1970's, came the single opamp tone control, wherein the frequency shaping network was incorporated into the feedback loop of a non-inverting stage, allowing the input signal to be fed directly into the opamp non-inverting stage, thus providing a high input impedance, via a volume control potentiometer. Neat, since it got around the low input impedance of the Baxandall design (which almost always necessitates a preceding buffer) and it was very cheap, especially if built around a 60 cent (production volume price at the time) dual opamp (one per channel of course . . . or one half for the RIAA and the other for the tone control). Unfortunately, like the passive tone control, it requires special log pots, and boost and cut are not symmetrical either. It is noisy (since the potentiometer values need to be kept high to reduce the load on the opamp output) and suffers from HF distortion, although both these points can be mitigated somewhat if a buffer stage is interposed between the op-amp output and feedback network – however, since cost was always an issue, this was rarely the case. I experimented with this approach around the time I built my first preamps in the late 1970's and concluded in the final analysis that it was a novel design but not one that could seriously compete with the Baxandall circuit, which remains the standard by which all others are measured.

The original design by Peter Baxandall was published in Wirless World in October of 1952, when he was 31 years old (an earlier version published around 1950 had already earned him a prize in a design competition). Self discusses the state of tone control design in his book 'Small Signal Audio Design' and notes that prior to this it was passive based and basically a shambles with the requirement for switches and log pots. Baxandall's insight was to think about the problem differently, recognizing that by altering the transfer function of a feedback network around an active element, the overall frequency response could be adjusted – the summing junction approach meant he could use linear pots and the cut and boost were symmetrical – something that was near impossible before the advent of this very elegant design. The Baxandall tone control has remained the best analog solution for bass and treble control, and I don't see it being usurped any time soon. A key further advantage is its extreme simplicity: with two pots, an opamp and seven passive components, you can create a high performance tone control, that at full boost (+10 dB) at 10 V peak out at 20 kHz will still deliver < 50 ppm distortion.

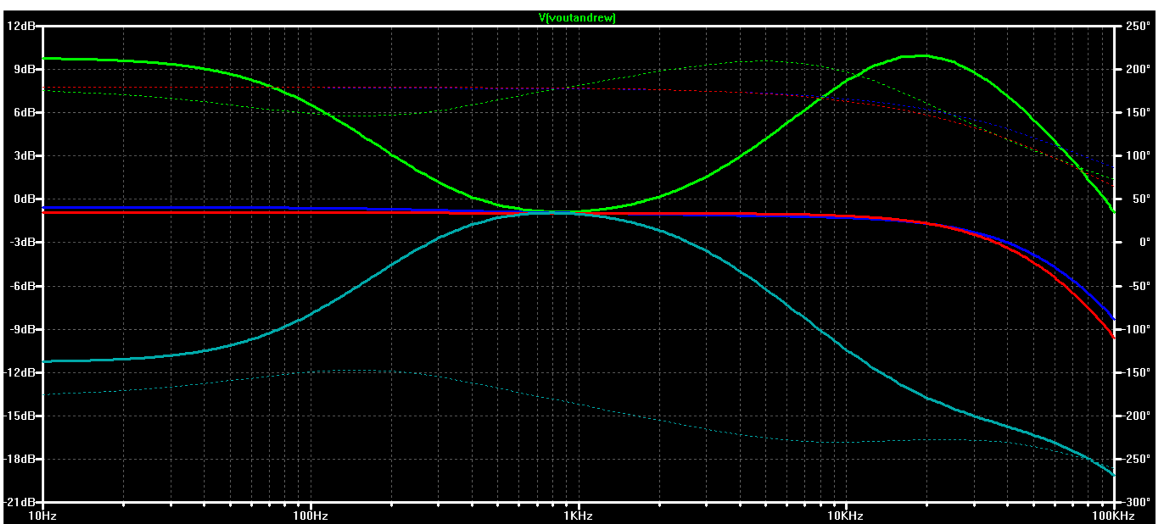
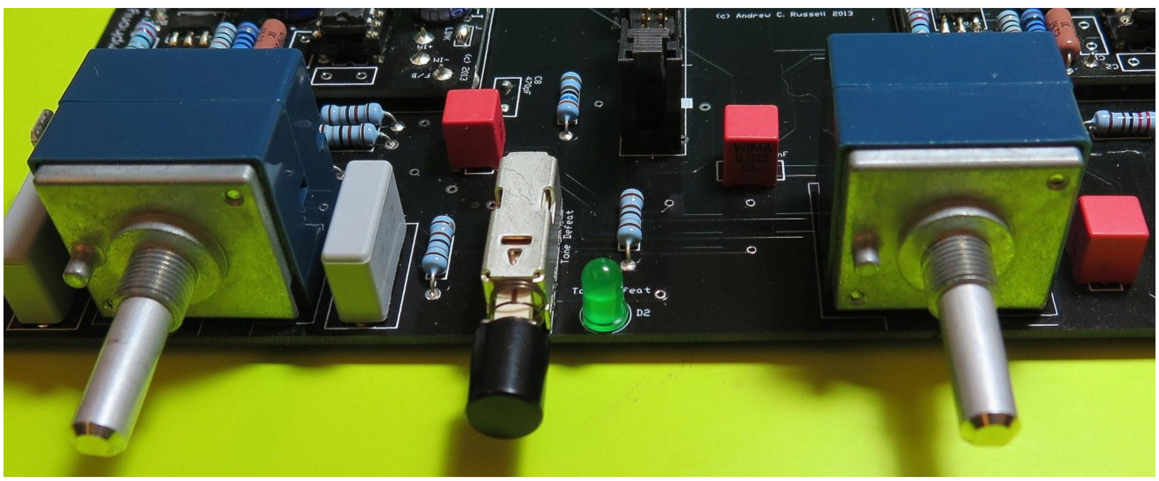


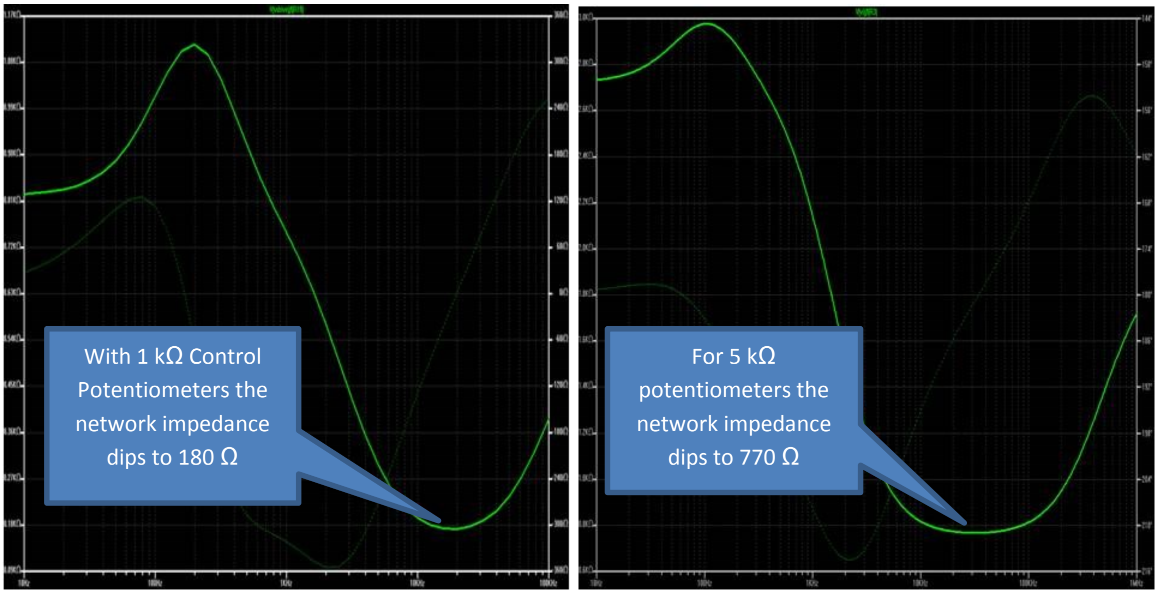
Figure 16 – Ovation Symphony Tone Control Performance

For the Symphony preamp (refer Fig 16), I elected to provide ~10 dB of boost and cut at 20 Hz and 20 kHz respectively (Fig. 15), with the shelving frequencies at about 80 Hz and 10 kHz respectively. There are designs out there that provide up to 20 dB boost and cut, but with decent speakers, this is really not necessary – event the 10 dB cut and boost you see here is much more than you will need. On some of my CD's, I am turning the bass up to 2 o'clock, while the treble is on some recordings advanced to between 1 and 2 o'clock. Further, the shape of the boost curves and the location of the central non boost band are also important design considerations. For the bass, the idea is to avoid changing things much above about 200 Hz, while on the treble side, boost really only kicks in from about 2 ~ 3 kHz. When set to the flat position, there should be *no audible* difference when you switch the tone control in and out. The overall approach is to provide a moderate range of adjustment at the frequency extremes to help 1st order compensate for differences in recordings and/or listening position where personal experience shows there is a huge range in the tonal balance.

To my ears, I find many older pop and classical CD's a little bass light on my B&W 703's in my listening room, with many jazz and modern classical and pop recordings seeming to get it a bit more right. However, it should also be noted that if I move my chair, I can easily really get a boost in the bass registers, but the imaging then suffers. As I mentioned earlier, more often than not, it's the room that's not up to the task.



The input impedance of the Baxandall tone control varies considerably over the control range. Fig 17 plots the worst case input impedance vs frequency curve which occurs at full boost for 1 kΩ control pots (LHS panel) and 5 kΩ pots (RHS panel). So, to the left of 1 kHz x-axis marker we have the LF input impedance, and to the right the HF input impedance. At 1 kHz, the input impedance is around 800 Ω, and dominated by the bass control input resistor R4 and C8 reactance in Fig. 18. With the values selected for the control networks, this drops to about 180 Ω between 15 and 30 kHz. I was concerned about noise, and this of course meant the network impedance had to be low, which is why the control pots are specified at 5k Ω. The low input impedance of course means that you will need plenty of drive capability and this is provided for by the front end amplifier buffer (U2, Q4 and Q5) in Fig. 7). In the final design, the tone control can be completely bypassed, such that the front end amplifier/buffer drives the 5 k Ω Goldpoint volume control directly.



These two panels show the input impedance of the Baxandall tone control when set to maximum boost, which is the worst case situation for the *driving* source. The LHS panel shows the case when 1 kΩ control pots are used – the impedance dips to around 180 Ω 10 kHz ~ 100 kHz. For the Ovation Symphony, I chose 5k Ω control pots, and the worst case load 10 kHz ~ 100 kHz is 770 Ω. Note that in the full cut position, the tone control opamp sees this heavy load as well – so both the source and the tone control opamps need to be fully capable of driving the network load.

Figure 17 - Baxandall Tone control Input impedance vs Frequency for 1k Ω and 5k Ω control pots

From Fig. 17, its easy to see why most opamp based designs would specify higher value pots than 1k, with concomitantly higher input impedances. In Self's design, he paralleled a number of opamps to provide enough low distortion drive into the very low input impedance, dictated of course, by the 1 k Ω pots he used, and he split the bass and treble control elements. For this reason, the Symphony buffer and tone control amplifier blocks are class A and able to drive 200 Ω at 10V pk whilst still remaining in class A and delivering single digit ppm distortion levels. It is important by the way, to use a *less than unity gain* stable opamp for the tone control. This necessitates testing the opamp in the prototype, since some devices are unity gain stable, but for inverting gains approaching 0.2 or less, they are not and the result is HF oscillation that affects the sound.

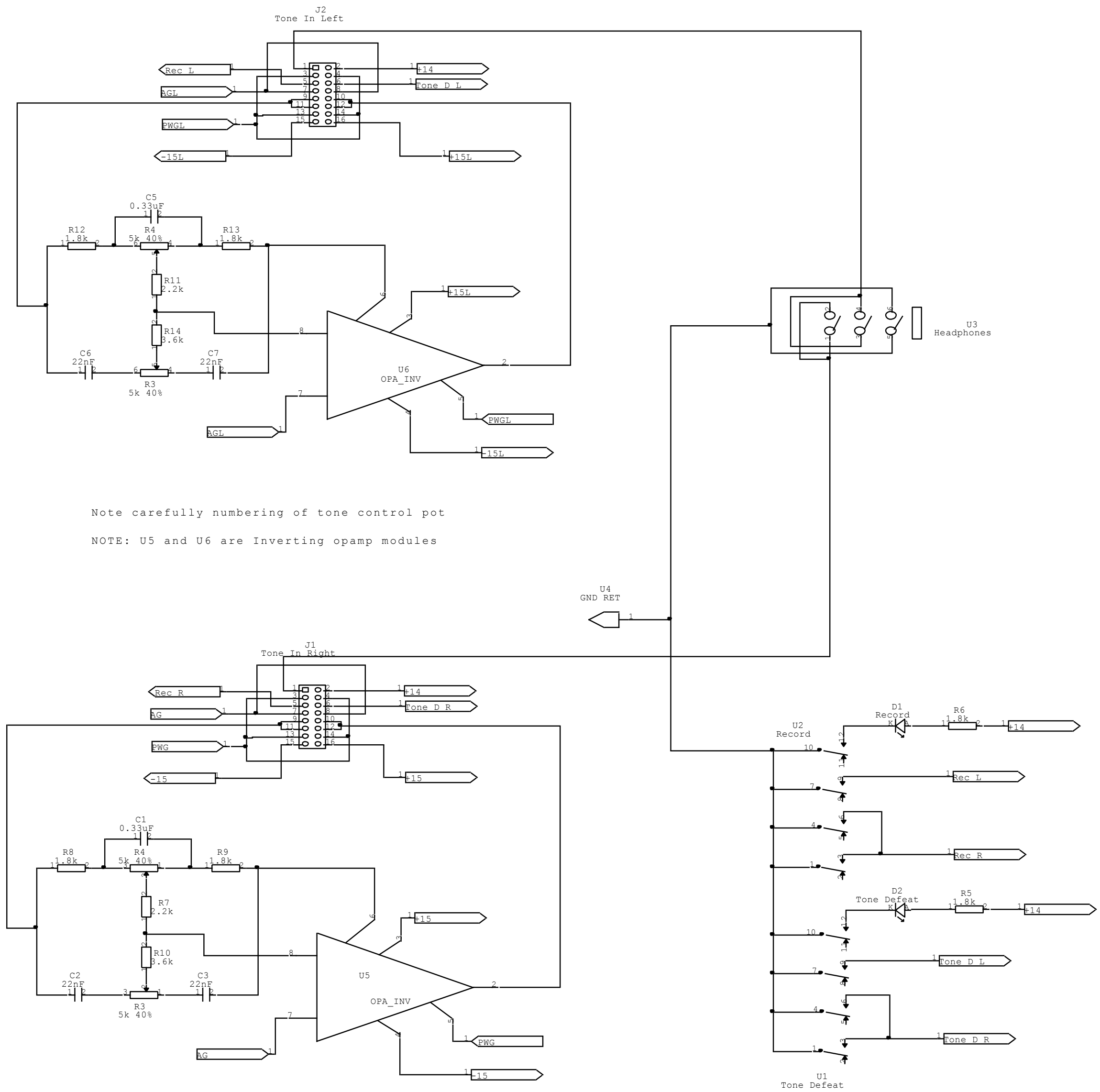


Figure 18 - Symphony Baxandall Tone Control Circuit

Fig. 18 shows the circuit details. U5 and U6 are inverting opamp modules (see Fig. 7) set for 27mA OPS standing current. 5k Alps RK27 linear pots with a central indent are used for the bass and treble control elements and are a good compromise between network input and feedback load and noise. The whole board is connected to the main analog board via two 16 way IDC cables, which carry the signal to and from the tone control board, along with the headphone signal. The tone control defeat switch (U1) switches power to a bypass relay on the main analog board – it does not switch the actual signal. The record loop pushbutton switch (U2) is also located on this board – again, switching power to the coil of the rec loop relay. As with all the opamp modules, each module supply rail is decoupled with a 22 Ω resistor and a 100 µ capacitor to localize supply loop currents, provide decoupling, and filter HF noise on the supply rails. The supply rails to each tone control channel are completely separate as well – this is to ensure that the effort put into maximizing channel separation is in no way compromised through any common impedance coupling.

How does it sound? With a well designed tone control and a good recording, the treble will appear shift the placement of the HF, a good example being cymbals – all that you should be aware of as you move the control around is the they move back into the soundstage as treble is cut, or they are pulled forward as it is boosted. There should be no change in the mid range registers – a good test being solo piano music where you should only hear a small change at the pot extremes and usually only in the cut position. There should be no change in tonal character and no sibilance should be apparent (assuming there is none on the recording) on female voices. Similarly, the effect of the bass control through the first half of its travel either side of the center position should be very subtle, and you should just feel the sound in the lower registers firm up. Depending on room acoustics, bass can excite room resonances, but changing your speakers is hardly the cure – repositioning them, and using some acoustic dampers can help. Since the ear is not good at locating bass sounds, you will not get the apparent depth location shift you get with treble. Bass controls that provide high levels of boost and are centered around 1 kHz, rather than the 800 Hz in this design, will thicken the midrange as they are advanced, which many listeners find objectionable. I would agree – a tone control amplifier is about subtly reinforcing or taming the frequency extremes and if you are lucky enough to be using decent speakers in a sympathetic listening space, +-3 dB is all you will probably need on for the most part. A little later we will discuss the headphone amplifier, and here the variation in response is considerably wider than speakers – my findings indicate +-6 dB of tone adjustment range is required, although on my Bose non-cancelling phones, a full 10 dB of cut still leaves the sound uncomfortably bass heavy. There is no doubt in my experience that a little cut or boost provided by a high performance tone control can significantly enhance the listening experience. What about noise, and specifically hiss? Even at full treble boost, this tone control circuit is quiet with only the very faintest hiss from the treble at full volume with my ear right up against the tweeter.

Purists may wince at tone controls. In that case, hit the tone control bypass button.