The Ovation sx-Amplifier

The Ovation 'sx-Amplifier' is a low distortion, very wide bandwidth (DC to 530 kHz -3 dB) 15 W CFA topology class A power amplifier. This design achieves outstanding DC performance with offsets of <10mV within 2~3 minutes of switch-on, and 1~2 mV after 10 minutes without the use of servos or DC blocking capacitors. The maximum offset at switch-on is under 50 mV.

A simple, single transistor bias controller directly regulates the 1.4 A push-pull OPS standing current to ~4% over temperature, yet simultaneously allows the amplifier to transition to class AB mode, where peak currents of up to ±5 A can be supplied to the load. The sx-Amplifier distortion products are primarily 2^{nds} and 3^{rds} and clipping behavior is soft, somewhat emulating tube designs in this regard.

The amplifier and power supply use modern, readily available components and 2 channels and a PSU board can be built for about $100, excluding DSTHP PCB's.

Single sided versions of the amplifier PCB layout are also available on the www.hifisonix.com website in PDF format for constructors who want to etch their own boards. Each channel of this design requires a heatsink of not more than 0.4°C/W, with 0.3°C/W preferred.

With its lower distortion, wider bandwidth and rise/fall times of well under 1µs, the sx-Amplifier offers a significant upgrade for class A aficionados looking for a step up from JLH's classic 10 Watter from 1969, or Jean Hiraga's iconic 20 W design from the 1980's.

Andrew C. Russell

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V2.10

This document is maintained at http://hifisonix.com/the-sx-amplifier/

!Caution!

This amplifier uses a mains powered supply and involves wiring and connections to the mains voltage.

AC Mains is dangerous and mistakes can be lethal.

Exercise caution and seek expert help if you are not qualified or experienced to undertake this type of wiring.

You can buy a set of very high quality Double Sided Through Hole Plated Gold Flashed PCB's with silk screen and solder mask from Jim's Audio here:-

sx-Amplifer PCB Set

The PCB set consists of 2 amplifier PCB's and a PSU PCB

DIYAudio.com hosts an sx and nx-Amplifier build page where you can meet with other builders and get useful information on the sx-Amplifier here:-

DIYAudio.com sx-Amplifier Page

Ovation sx-Amplifier

A 15 Watt Symmetrical Class A Hi-Fi Amplifier

1. Introduction or 'Why Class A?'

Jean Hiraga's class A design [the link is to part 1 of his article] (See Fig. 1 later in this document) became something of a cult amplifier in the 1980s in the audio community and constructors from Europe, Australia, Japan and the USA praised it's sonic performance. In that design, Hiraga used a simplified current feedback amplifier (CFA) topology and low feedback to create an amplifier that was described somewhere as sounding 'liquid' and 'tube-like'. Hiraga has always been noted for his minimalist, idiosyncratic designs, and comments about the sonics aside, I was attracted to the simplicity – only 8 transistors in the original and a handful of discrete components (excluding the power supply of course) for a complete class A amplifier. Distortion was by any standards *very* high, topping out at about 1.8% at the rated power into 8 Ω, although he seems to have indicated it was of the 'good variety' due to its harmonic structure. However, Hiraga's amplifier (rated at 20 Watts per channel into 8 Ω) was the product of 1980's know-how and device technology and I found myself wondering what I could cook up with today's components and the insights afforded by circuit simulation tools that were unavailable back in the early eighties. I did quite a bit of research on Geoff Moss's excellent the class A amplifier site ('tcaas') and Rod Elliot's ESP site and the Death of Zen (DoZ) amplifier, modeled after JLH's classic 10 Watt design from 1969. Compared to the 1960's through to the 1980s, we now have some really good power transistors, and there is no shortage of cheap good quality small signal devices either, other than the ultra-low noise small signal bipolars' and JFETs from Toshiba in Japan of course, which unfortunately are now discontinued for the most part.

Looking at the reprint of JLH's original 1969 10W class A design[1], it is immediately apparent that the distortion vs. frequency graph at 9 W output is *flat* all the way out to the measurement limit, which was beyond 20 kHz. In a conventional Voltage Control Amplifier (VCA) class AB topology, when driving a load that causes the output stage to exit the very narrow class A bias region (ideally set at $.026/Re_{tot}$ where Re_{tot} is the total output transistor emitter degeneration resistor including the reflected base component, and typically ranging from 0.1 Ω through to 0.47 Ω), distortion kinks upwards at 40 dB/decade, with the kink point a few hundred Hz up to a few kHz. 20 dB/decade comes from the fact that the amplifier loop gain beyond the -3 dB point is decreasing at -20 dB/decade, while the THD contribution doubles with every octave because with every doubling of frequency, there is a doubling of cross over events, and hence distortion.

[1] Many are still being built, as judged by the discussion around this design on the internet.

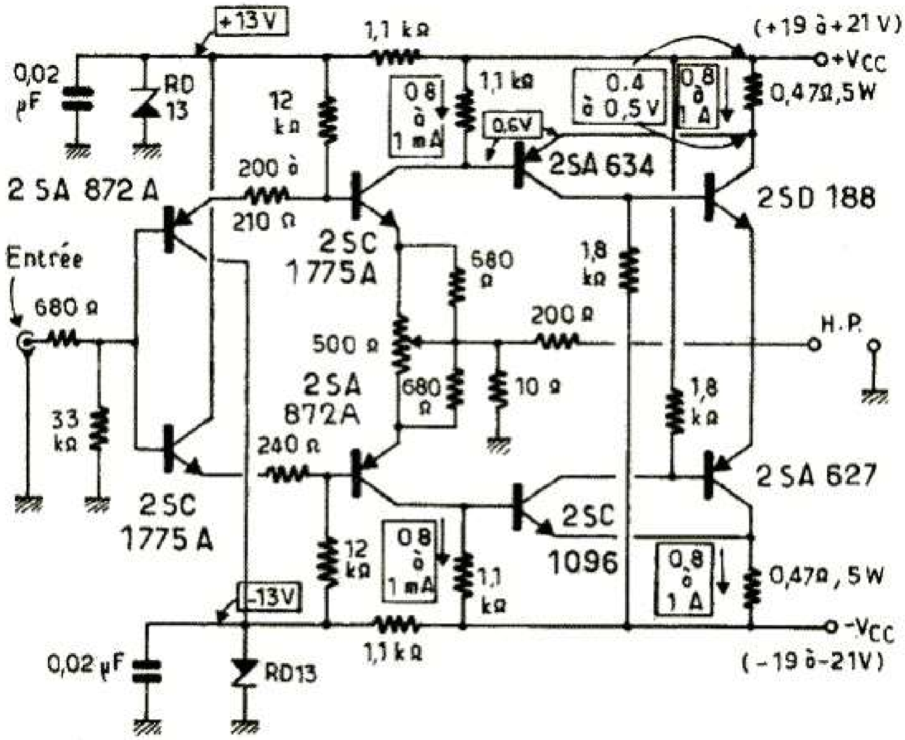


Fig. 1 : Diagram of the 20W amplifier, final version, including some slight modifications compared to the description published in Issues 10 and 11.

Figure 1 - Jean Hiraga's 1980s 20 W Class A Amplifier as Published in the French Magazine 'l'Audiophile'

The shape of the distortion curve over frequency tells us something about the nature of class AB amplifiers when required to move from their class A region into class B – feedback can only do so much to reduce the distortion, and if that feedback (loop gain) is already decreasing at 20 dB/decade within the audio band, you simply have to accept the uptick in the distortion vs. frequency plot. JLH's output stage design was class A of course, and so did not have to contend with crossover distortion and this is the fundamental advantage of class A: the major distortion component in any competently designed class AB amplifier is the crossover distortion, and if that can be removed as a factor, there are significant performance and sound benefits to be had. At low frequencies (i.e. below about 30 Hz), the distortion on JLH's amp rose rapidly because of the electrolytic coupling cap between the output and the speaker. Later enhancements of his design did away with the output coupling capacitor for the expense of a split rail supply and output offset adjustment.

Classic CFA vs classic Lin VFA topology: A very short primer for the uninitiated . . .

$$\frac{1}{Av} = \left(\frac{1}{Aol}\right) + \left(\frac{1}{G}\right) \quad where \quad G = 1 + \left(\frac{Rf}{Rg}\right)$$

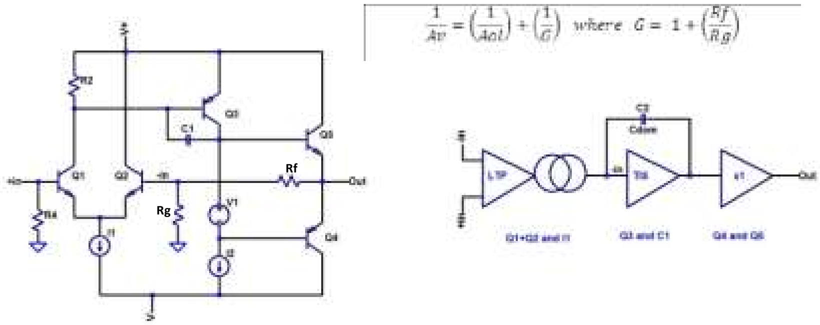


Exhibit 1 - Classic LIN VFA Topology Amplifier

There are many explanations about CFA topologies like this or this. Some plunge into math, loop gain equations and so forth, leaving the reader none the wiser, while this one (equations 1~4) from Hans Palouda is altogether easier to understand. A VFA typically has two active gain stages - the LTP and the TIS; the input stage LTP is usually designed to provide gains of 10 to 20 dB depending on the design specifics, with most of the open loop gain coming from the TIS. Both the inverting and non-inverting inputs are high impedance nodes in a VFA. On the right hand side diagram of Exhibit 1 you see a conceptual drawing of a classic VFA – differential input, driving an integrator (a TIS with C2 wrapped around it) followed by a buffer.

In a Lin VFA topology, a differential pair input stage (or Long Tail Pair 'LTP' - Q1 and Q2 in Exhibit 1), the tail current is *fixed* by a current source I1. The maximum output current of the diff amp stage available to drive the TIS (Q3 loaded by I2) and any compensation networks (MC, MIC, shunt, etc but in this example C1) is equal to this tail current I1, assuming the LTP is loaded with a mirror – if its resistively loaded as in this example, its lower and in a correctly balanced LTP about *half* the tail current. In a CFA (Exhibit 2), the input devices are arranged in a diamond buffer configuration (Q1~Q4) with unity gain – the non-inverting input is a high impedance node, and the buffer output is connected to a *low impedance* inverting input node at the junction of Rg and Rf. The output current of the diamond stage appears at the collectors of the level shifters Q3 and Q4 and is not limited by a current source as is the case in a VFA, but instead by the output voltage level and the *value of the feedback resistor* + Ro. Ro is usually very small (in IC's a fraction of an Ω, but in practical power amplifiers up to 10's of Ω's). Further, since in a CFA there is only one active gain stage (the mirror or as in this case the TIS), it has two major poles in the overall transfer function - the TIS and the output stage (Q7 and Q8). All the gain in a CFA is provided by the 2nd stage – I_{mirror} x R_{ad}. In a practical TIS based CFA like the sx-Amplifier, Rad would be the input resistance of the output stage in parallel with any further load to ground, including any shunt compensation networks. In CFA's where the 2nd stage is a mirror (typically in IC's), the 2nd stage pole is usually very high (can be MHz), but in TIS CFA's, the 2nd stage pole is much lower – typically 10's of KHz to 100's of KHz.

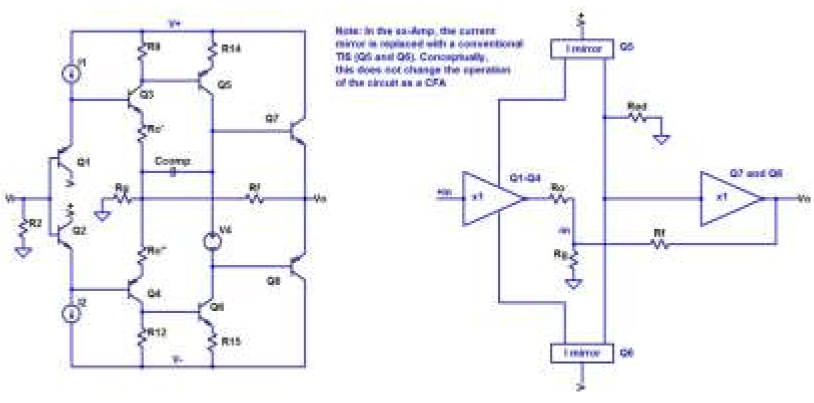


Exhibit 2 - CFA Topology Amplifier

$$\frac{1}{Av} = \frac{1}{G}\left(\frac{Rf}{Rad \times A1} + 1\right) \quad where \quad A1 = Imirror \times Rad$$

VFA's tend to have *much higher open loop gains, and hence loop gains* because of the additional gain provided by the LTP stage. However, in the open loop condition, the extra gain causes the OPS pole to locate lower than the unity gain frequency, causing excess phase shift to accumulate. That requires that *dominant pole compensation* (MC) be applied that results in a 20 dB/decade roll-off starting typically below 100 Hz and a ULGF intercept designed for somewhere between 1 MHz and 3 MHz – a technique that pushes the first pole down in frequency, and the second pole up beyond the ULGF - so called 'pole splitting'. The -3 dB closed loop bandwidth of a VFA is thus linked to its closed loop gain – the higher the gain, the lower the bandwidth and vice-a-versa: the constant gain bandwidth product is a feature of *any* dominant pole compensated amplifier – CFA or VFAs.

CFA's on the other hand have *lower open loop and hence loop gains*, with the *OPS pole often lying well above the UGF*, and thus do not require dominant pole compensation; as a result the -3 dB bandwidth is *not* linked to the closed loop gain (you can see that from Fig. 5 further on in the document) – closed loop gain can be varied over a reasonable range with very little impact on the 3 dB bandwidth of the amplifier which is in marked contrast to VFAs. In general, for low closed loop gain designs (typical in audio), this translates into wide loop gains – often > 20 kHz, and in the sx-Amp its ~60 kHz (see Fig. 13). The conventional way to compensate a CFA is to adjust the value of Rf to achieve the optimum gain and phase margins, usually by observing the overshoot and setting time to a square wave input stimulus. In practical power amplifiers however, capacitive shunt compensation is often employed, although there are more advanced techniques like Alexander compensation as used in the sx-Amplifier

The other major difference between the two topologies is the slew rate (SR) performance. In a VFA, the maximum rate of change of the output signal that can be supported is set by the compensation capacitor value and the LTP tail current from it=CV. As already pointed out, in a CFA the peak current available to the input of the 2nd stage is set by the maximum output voltage and the value of the feedback resistor+ Ro. In a correctly compensated CFA, this can be a factor of 10 higher than a VFA, and explains the big differences in SR between the two topologies. Importantly, in both topologies, the output stage phase shift limits the amount of feedback and the loop bandwidth of the amplifier which translates into setting the maximum upper ULGF around 1.5 MHz for an EF3 and 3 MHz for an EF2. For both the VFA and CFA, the closed loop gain is defined as A_{vcl} = 1+(Rf/Rg) where Rf is the resistor connected between the output and the inverting input, and Rg is the resistor between the inverting input and ground as shown in the Exhibits above.

Summary: VFA's have higher open loop gains and 2 major poles in their open loop response; CFA's have lower open loop gains and also 2 major poles in in their response. VFA's require dominant pole compensation ('pole splitting') in order deal with additional phase shift in their response due to their higher open loop gains, and this compensation links the closed loop -3 dB bandwidth to the closed loop gain. The loop gain in VFA's starts dropping off from the dominant pole – usually set at a few 10's of Hz. Classic CFA's do not require dominant pole compensation because their OPS pole usually lies above the unity gain frequency, and so typically yield wider closed and feedback loop bandwidths; furthermore, the SR in CFA's is not limited by the tail current, but by the total feedback resistance (Rg + Ro in Exhibit 2) and any capacitive load connected to the TIS output, allowing very high slew rates to be achieved as a matter of course. Both topologies can be exploited successfully for to create high performance, practical audio amplifiers, provided their associated shortcomings are suitably mitigated.

I have deliberately used the term 'classic VFA' in this explanation because there are alternative VFA compensation schemes that allow the it-CV SR limit in MC to be broken so that comparable closed loop bandwidth and SR performance are attainable by VFAs – you can read about some of these techniques in Bob Cordell's book 'Designing Audio Power Amplifiers' Chapter 4. See Also CFA vs. VFA

2. The PIMD Debate

There is quite a debate within the audio community about the importance of open loop bandwidths being higher than the audio band (so >20 kHz). One camp asserts that VFAs with low open loop bandwidths (i.e. the 10 Hz up to 100's of Hz mentioned above) produce PIMD (**P**hase **I**nter-**M**odulation **D**istortion), whilst the other camp believes it is not a problem, and is cured by providing sufficient levels of negative feedback – ergo high open loop gain, and hence high loop gain are what is prescribed[2]. Both parties have published papers supporting their claims. I am not going to take a hard position on this issue here, since this is another hi-fi audio debate that is unlikely to ever see the two factions arriving at an agreed conclusion.

It is important to note that if a CFA is designed with high open loop gains similar to that of a VFA, you will have to apply VFA type compensation schemes (MC, MIC, TMC, TPC etc) to deal with the greater phase accumulation and there are many designs on DIYAudio.com that deploy this philosophy. A study of this type of CFA will reveal that the loop gain bandwidth is much lower than the classic low OLG CFA at 3~10 kHz and designers therefore have a clear tradeoff to make: lower OLG CFA with very wide LG bandwidths, or high OLG and lower LG bandwidths. Of course, in both cases if PIMD is to be avoided, you don't want low OLG and low open loop bandwidth.

I happen to like symmetrical designs, and did not want to go for a single ended variant like JLH's. Hiraga's topology was simple, but I wondered, might I be able to improve on the distortion performance? And then, what about PIMD? I have never heard or had pointed out to me PIMD or its artifacts, but what if the designers that claim this is an issue are correct? Well, it turns out, one of the features of conventional, low OLG *Current Feedback Amplifiers (CFA)* is their very wide open loop bandwidth; in the closed loop condition, they display a fairly constant -3 dB bandwidth even with quite substantial changes in the closed loop gain (See Fig 2). This contrasts markedly of course with classic Lin VFA topologies e.g. Douglas Self's 'blameless', where the open loop bandwidth is usually quite low - typically 10 Hz to a few hundred Hz - and a closed loop -3 dB bandwidth that is linked by fundamental topological constraints to closed loop gain, such that if you increase the closed loop gain, the -3 dB bandwidth decreases. In VFAs, as the LTP *gm* changes with the input signal, it modulates the -3 dB open loop corner frequency, which in turn can manifest itself as phase modulation of the loop gain *above* the -3 dB break frequency, and therefore ultimately the output signal in the closed loop condition. The higher the loop gain, the *smaller* this effect, because the gm changes needed to drive the forward amplification loop are much reduced. Returning to CFA topology amplifiers, another great feature is their inherent speed – both in bandwidth and slew rate terms. In a typical Miller Compensated (MC) Lin topology VFA, the front end LTP stage output is in the form of a current, usually in the 2-10 mA range on a power amplifier, depending on the design.

[2] For a widely accepted view on PIMD distortion, see Robert Cordell's article at http://www.cordellaudio.com/papers/phase_intermodulation_distortion.shtml

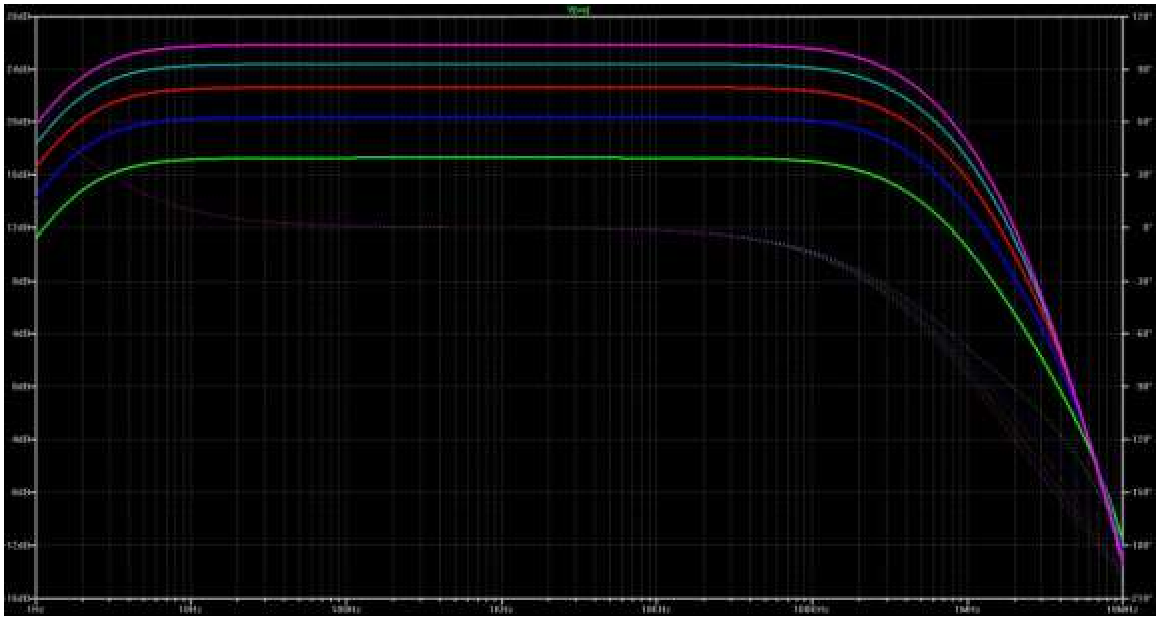


Figure 2 - sx-Amp Simulated Frequency Response. Closed loop bandwidth remains essentially constant even though the closed loop gain is changed from 18 dB to 26 dB, a feature of the CFA topology.

To convert this current to a voltage, the LTP output, which is in the form of a current, feeds into a high gain *integrating* amplifier stage, formed by connecting a low value capacitor, called Cdom, between its inverting input (usually the base of the TIS[3] input transistor) and output. It is this circuit structure that fundamentally *limits the closed loop bandwidth and it=CV slew rate* [4] of Lin VFA topology amplifiers. There are VFA compensation techniques, Miller Inclusive Compensation or MIC for example, which allow this constraint to be somewhat mitigated, allowing for higher slew rates for a given LTP tail current. However, CFA's have none of these issues – there is no pole splitting integrator stage involved – and, ignoring compensation required to meet overall loop stability for a second, bandwidth and slew rate are essentially limited by device parameters and the circuit node currents. In CFA's, the current through the feedback resistor (R12~R16 in fig. 5) can be seen as the equivalent of the VFA's LTP current. In the case of the sx-Amp, this is 110 mA peak (from $Vo_{peak}/Rf = 22\ V/200\ \Omega$) - an order of magnitude higher than would be found in a typical VFA topology amp like my e-Amp design for example, which has an LTP current of 10mA.

It follows therefore, that if a wide open loop bandwidth amplifier that can also deliver wide loop gain so that all error terms within the audio band are corrected by the same amount of feedback is a primary design goal, CFA topologies make a good design choice[5].

[3] I have adopted the term 'TIS' rather than the more widely used term 'VAS' for the 2nd stage in a typical VAF amplifier. VAS is in fact *not* the correct term for this stage, since it is converting a current from the LTP output into a voltage, thus it is correctly called a *Trans Impedance Stage*.

[4] In MC VFA amplifiers, Unity Gain Bandwidth and speed (slew rate) are set independently; Further, it is quite normal to have an open loop bandwidth of <10 Hz and yet have high slew rates of 10s of volts per microsecond. By way of contrast, some integrated CFA op-amps achieve SR's of >1000 V/us.

[5] Note here that wide open loop bandwidth and wide loop gain bandwidths are not prerequisites for good audio performance – the sx-Amplifier simply incorporates this design goal as part of the overall design philosophy

Distortion. The sx-Amp does not specifically target ultra-low distortion; with speaker systems producing upwards of 1 or 2 %, it's THD harmonic structure that's more important than ultra-low distortion. As noted earlier, to achieve lower distortion would require a few more components and added complexity. This design instead aims at high speed i.e. fast rise and fall times, and very wide bandwidth. For the harmonic structure of the distortion components, these are located in mainly the 3^{rd} with some 2^{nd} – since $2^{nd's}$ partially cancel in symmetrical designs. Higher order harmonics (5^{th} and 7^{th}) are considered by some commentators as particularly bad for music reproduction, and here the sx-Amp performs admirably below clipping levels. Once an amp clips (any amp), all bets are off of course, but things may be helped a bit by ensuring the clipping behavior is soft.

My objective for this design is to *keep it simple* with only as much complexity as needed to do the job well, operate in class A (8 Ω load) up to 15 W, and, with a nod and a doff of the hat to the pro wide-open-loopbandwidth protagonists, design it for exactly that. Importantly, as with all my designs, the final result is stable into any load and also passes the 8 Ω//0 up to 2.2 uF torture test with no oscillation or instability.

Figure 3 - Rear View of the 1^{st} build of the Ovation sx-Amplifier

3. Specifications

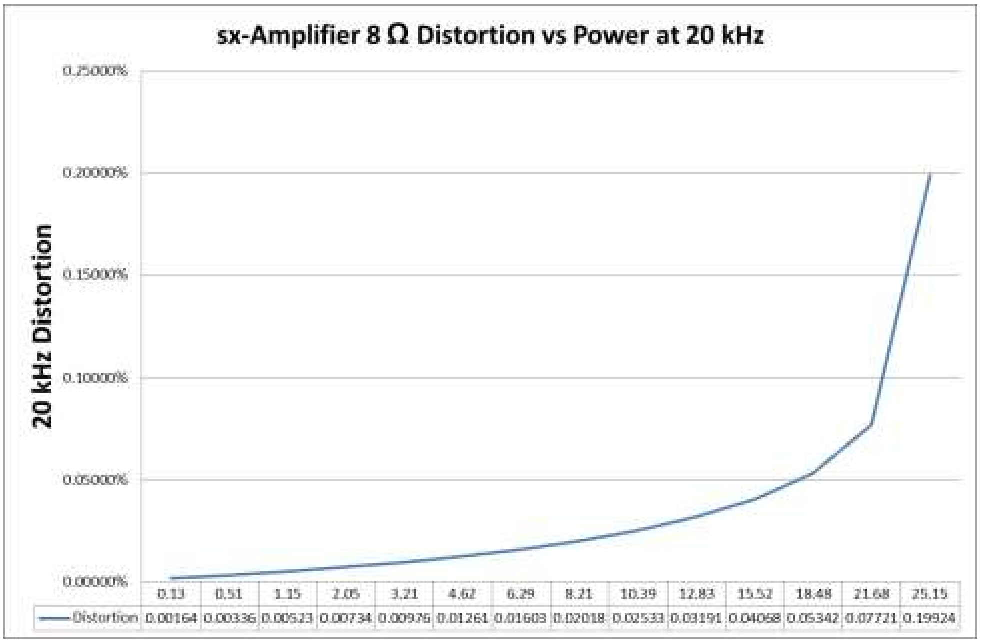
All specifications are into 8 Ω with +-22 V DC power supply rails unless otherwise stated

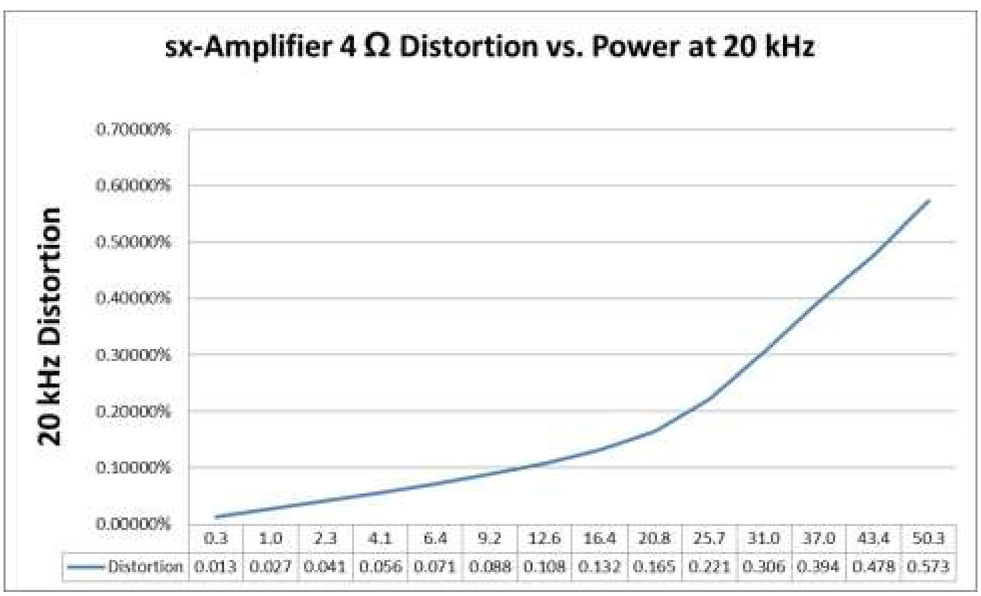
Power	15 W RMS class A; ~ 25 W peak class A; ~50 W peak into 4 Ohms class AB	
Frequency Response	DC – 530 kHz -3 dB 1 W into 8 Ω	
	DC – 1.5 MHz – front end filter disabled – 1 W into 8 Ω	
Distortion (see plots overleaf)	15 W 8 Ω	~0.05% at 20 KHz
	15 W 8 Ω	~0.05% at 1 kHz
	1 W 8 Ω	~0.005% at 20 KHz
	3 W 8 Ω	0.03% at 20 kHz
	3 W 8 Ω	0.02% at 1 kHz
	25 W 8 Ω	<0.3% at 20 kHz
	50 W 4 Ω	<0.6% at 20 kHz

Distortion components at all rated powers predominantly 2nd and 3rd harmonics.

Noise	<200 uV into 8 Ω (40 kHz bandwidth); -100 dB ref 20 Vpk output (input open circuit)
Closed Loop Gain	14.3x or 23 dB
Input Impedance	1 k Ω+ [10 k Ω// 220 pF]; circa 11 k Ω at 1 kHz
Loop Gain	34 dB DC to 60 kHz (+0 dB -3 dB)
	34 dB DC to 20 kHz -0.2 dB
Rise/Fall Time (10-90%)	<1 us with front end filter
	~250 ns with front end filter disabled
Small signal rise/fall time (10-90%)	~50ns 2V pk-pk
Slew Rate	>250 V/us (front end filter disabled)
	c. 140 V/us (front end filter enabled)
	(slew rate measured before the output LR network)
Peak Output Current	>4 A – limited by Power Supply; supply rails fused at 5A
Power Consumption	~60 W at idle per channel – 120 W for stereo amp
Maximum Power Consumption	250 VA (as built with 250VA transformer)

The two plots (simulated) on this page show the sx-Amplifier distortion vs power at 20 kHz for both 4 and 8 Ohm loads. The performance is considerably better than JLH's 10 W and Jean Hiraga's 20W classic designs. Into 4 Ohms, the sx-Amplifier can deliver around 50W peak at 0.6% distortion (5A rail fuses notwithstanding), while at the important 1 W industry benchmark, its around 50 ppm into 8 Ohms. These plots are at 20 kHz – the 1 kHz distortion is about 30~50% lower in all cases. As noted elsewhere, the distortion components are predominantly 2nd and 3rds - probably a good reason why this amplifier has a warm presentation that really can deliver a great listening experience with strings and acoustic material in general.





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4. Circuit Description and Design Discussion

Refer to Fig 5. The input signal is bandwidth limited by R17 (1 k Ω) and C13 (220 pF) before feeding into the unity voltage gain 'diamond buffer' stage comprising Q8 (BC547C NPN) and Q10 (BC557C PNP) which in turn drive the bases of the current summing stage (Q9 and Q11 – also BC547C and BC557C) which control the current feedback node at the junction of R28 and R29 (100 Ω each). The input transistors derive their base bias via R10 (10k) through R17, which also sets the input impedance of the amplifier to about 11 k Ω at 1 kHz. The front end emitter follower buffers (Q8 and Q10) are loaded via R34 and R35, which are fed from 10 V Zener regulators D3 and D4. This ensures that the quiescent operating current for the whole small signal part of the amplifier is firmly fixed even with shifts in supply voltage of +-20%. C6 and C12 provide heavy local decoupling of the Zener references and filtering of the supply rail ripple. R36 and R37 (150 Ω) provide a small ~150 mV stand-off voltage to the bases of Q9 and Q11 (BC547C NPN and BC557C PNP), so that the emitter currents of Q9 and Q11 can be set by resistors R28 and R29 (each at 100 Ω). The collectors of Q9 and Q11 (also referred to as the level shifters) drive the TIS stage (Q6 and Q7) bases via the 1 k R32 and R33 collector load resistors. In a conventional CFA topology, Q9 and Q11 would normally drive current mirrors, which would improve linearity, increase the open loop gain and isolate the front end from the output stage, but in my view avoiding that complication in this design is a virtue, since we really are aiming at a minimalist circuit. The collectors of the TIS transistors (Q6 KSC3503 NPN and Q7 KSA1381 PNP[6]) are tied together by the class A bias control amplifier, configured around Q3 (BC547C), R20, R6 and R21. This circuit simply measures the voltage across the emitter degeneration resistors (R26 and R27, 0.33 Ω) and clamps the output stage bases to control the output stage bias current, which in the sx-Amp is set to 700 mA per output pair, total 1.4 A, to deliver up to 15 Watts in class A operation. The current regulator performance is quite good with the variation in total quiescent current measuring <40 mA (i.e. < 3%) between 0 °C and 60 °C and with supply shifts of -10% to +10% around the nominal specified ±22 V. With supply changes of ±20%, the quiescent current shift is around 60 mA – still low given the 1.4 A standing current.

For the output stage, I chose MJL1381 and MJL1302 sustained beta devices (Q1, Q2, Q4 and Q5), using 2 pairs to improve thermal management, and to reduce distortion by minimizing beta droop when driving low speaker impedances. Using one pair is possible, but that will require a *very efficient* heat sink, and in my view still put too much stress on the output devices. You can use slower devices like the MJE21193/21194 - you however may have to adjust the value of C3 to stabilize the loop. Distortion with these transistors (simulated) will be about 1.5 ~ 2x higher, but I cannot say what effect, if any, it will have on the sx-Amps sound.

The sx-Amp is DC coupled and R3 is a 10 k 22 turn potentiometer which is used to dial out the output offset during initial calibration. C1 (10 uF MLCC) filters any remaining noise from the Zener regulators, ensuring the offset trim current does not inject any noise into the feedback summing junction. We will discuss the feedback and compensation a little later.

[6] Note that I have deliberately used high voltage devices for the TIS stage to take advantage of the high Early voltage, which translates into lower distortion.

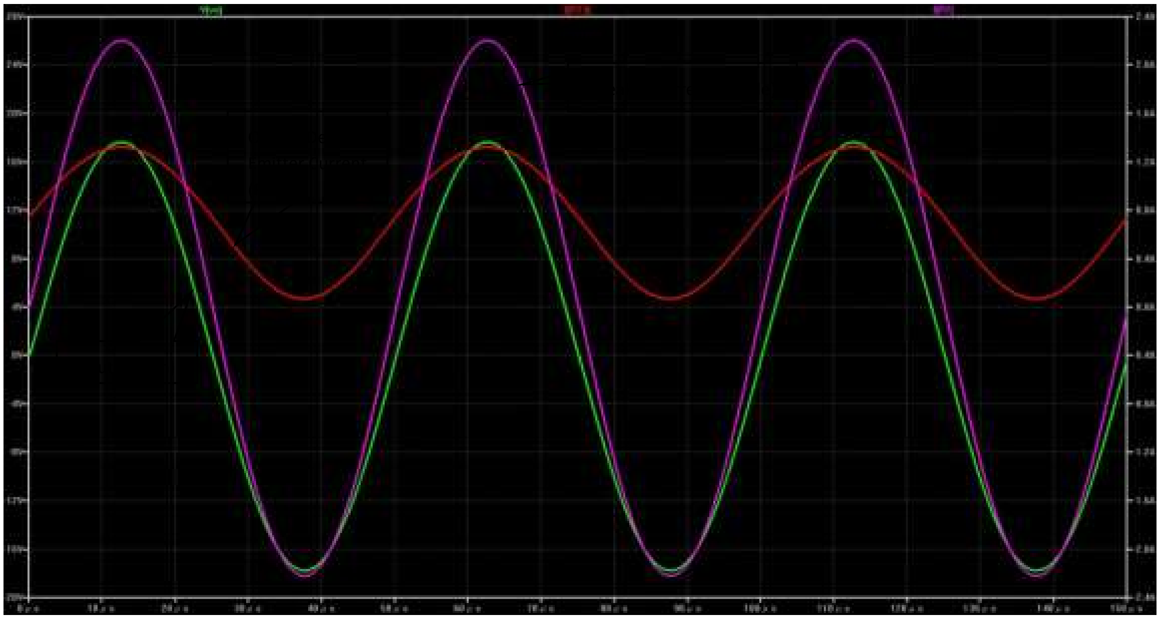


Figure 4 - sx-Amp Output Waveforms (8 Ω Load); output device emitter quiescent current is 700mA per pair, 1.4 A peak which delivers up to 2.3 A peak into the speaker load in class A. In class AB, it can deliver up to 5A, limited by the fuse rating

Figure 4 shows the output waveforms just prior to the onset of clipping, which for the nominal 20 V rails, takes place at around 18 V (note that in the practical implementation, I ran with 22 V rails). The total bias current for the amplifier (one channel) is set at 1.4 A – so 700 mA per pair of output devices. The output power is calculated from:-

$$P_o(ac) = \left(\frac{V_{CEQ}}{\sqrt{2}}\right)\left(\frac{I_{CQ}}{\sqrt{2}}\right) = \frac{1}{2}V_{CEQ}I_{CQ} = \frac{V_{PP}^2}{8R_L}$$

Which for this design gives us just over 20 W of class A power. The output stage quiescent current can be then calculated from

$$Po = I^2R$$

If we select Po = 15 W class A as the target, having duly considered the thermal requirements, power supply sizing and output stage drive capability, we end up with a total bias current of 1.37 A, which I rounded up to 1.4 A to provide a little margin. It is possible to bias this design up for 20 W of class A (so the quiescent current would then be circa 1.6 A), but distortion starts to increase rapidly as the drive requirements increase, which loads the TIS stage. For anything much above the 15 W design target, an EF2 or mosfet output stage with their attendant design challenges would be needed if performance is not to be compromised.

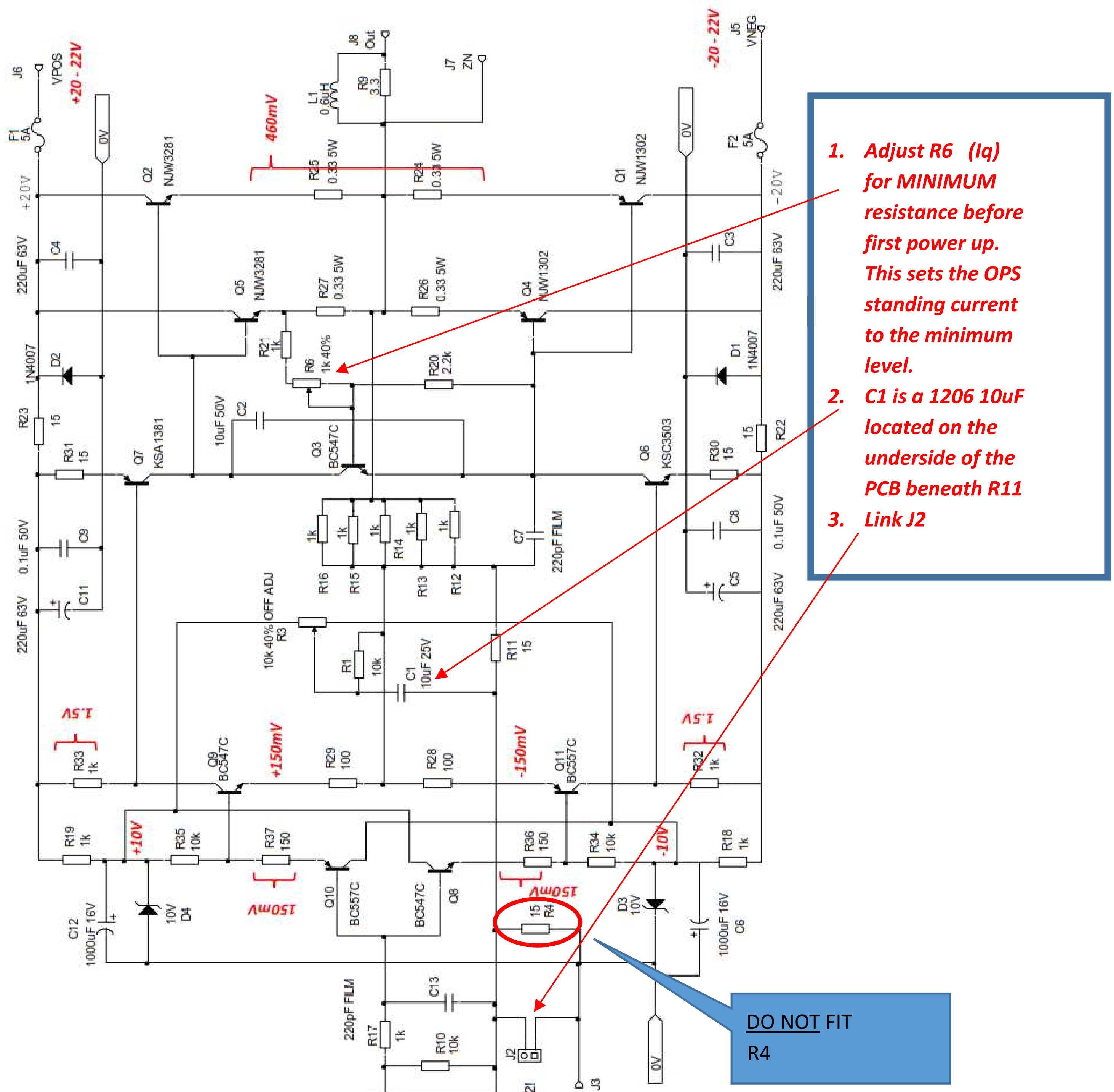


Fig 5 – sx-Amplifier Circuit Diagram

OPS Options. I did consider using an EF double on the output stage, and simulation indeed indicated that distortion would be about 15 dB lower at 0.007% for the 8 Ω rated output power at 20 kHz. However, this would necessitate a higher supply voltage and thus greater standing power dissipation. If this was to be averted, then a split supply with the output stage running off a lower rail would be an option, but then again, all this adds more components and complexity, so I also rejected this idea. There is the option of using a common emitter output stage, but even with fast output devices like the MJL1381/1302, you end up with a pole in loop response that can only be cured by reducing the loop gain or applying compensation that would reduce the loop gain bandwidth. The loop gain - which at 34 dB on the design you see in Fig 4 is not particularly high – and the higher output impedance of a CFP OPS only invites stability problems which would probably need a bigger output inductor. What about mosfets running in a common source configuration? Yes, this is a possibility, but getting two pairs to share current over load variations and temperature would not be trivial without a separate current regulator for each pair. For mosfets, I think Nelson Pass's approach where he runs a source follower and configures bottom half of the output stage as a current sink is an option here, but in the end, I stuck with the bipolars – a good fit for this project.

The sx-Amplifier Class A Bias Controller (Q3). On single ended emitter/source follower designs that are current source loaded as mentioned above, exceeding the class A current results in a hard stop – the current source half of the output stage limits the output power. It is for this reason that such designs, even for a modest 30 W class A amplifier, require enormous standing currents in order to deliver low distortion at rated power into low impedance loads. On the sx-Amp, if the load current peak exceeds the class A standing current of 1.4 A, either due to the peak voltage exceeding about 15.5 V, or the load being less than the nominal 8 Ω, the amplifier simply transitions to class AB, and will safely deliver peak currents of up to 5 A – the rail fuse limit. The advantage of the current regulator design in the sx-Amplifier is that it truly *regulates* the OPS standing current at the design value (+- some small 10's of mA variation due to thermal effects) by measuring the IxR drop across the OPS emitter degeneration resistors, and then offsetting the Vbe of the PNP output transistors by the Vbe of the bias controller transistor Q3. The Vbe of the bias controller transistor also acts as a temperature dependent reference voltage of ~600mV - the result is accurate direct regulation of the OPS standing current compared to conventional bias spreaders that rely on thermal tracking for regulation which is an indirect control of the OPS bias current. In conventional P-P class A amplifier bias controllers there is a lot of added complexity around the thermal & mechanical design of the bias generator– unless you take special care, thermal runaway, or big shifts in the standing current occur - it's a very 'loosey goosey' approach in my view. There are none of these issues in the sx-Amp – the bias controller fixes the OPS standing current over a very narrow range, but allows it to *transition gracefully to class AB on demand* and deliver high peak currents. By the way, this circuit can also be scaled up to control an EF2 OPS's standing current – simply insert a diode in the emitter of the bias generator to offset the driver Vbe, and scale up the feedback resistors accordingly.

Photo 1 - Internal View of the first Ovation sx-Amp built showing initial hook-up wiring

With my bias controller design, one of the areas I was particularly interested to investigate was the behavior of the amplifier as it moved from class A to class AB operation when driving a heavy load (4 Ω or less) or near maximum output swing. With the 22 V rails as built, the sx-Amp will deliver up to 25 W RMS in class AB, the first ~15 W RMS of course on class A. Under these conditions, given the heavy standing current of 1.4 A, it is very important that the transition from A to AB is clean with no spurious oscillation, or overhang when the output stage goes back into class A. I was happy to find that indeed this was the case looking at the scope trace; on simulation distortion increases to about 0.2% in class AB mode (I have conservatively rated it at <0.3% in the specifications) - a good result in my view given the low loop gain and simplicity of the circuit. Further, when driven into clipping, the amplifier should not suffer from any parasitic oscillation or rail sticking. As a result of the low loop gain, the sx-Amp clips quite softly when overdriven (see Fig 6 and Fig 7), emulating somewhat the behavior of tube amplifiers.

OPS Devices. Although the supply voltages on the sx-Amp as built are only +-22 V, I still elected to stick with high voltage TIS and output transistors, rather than use 80 or 100 V types which would have been more typical; besides, I had about 10 pairs in my parts cupboard looking for something to do. The benefit here of course is that there is no secondary breakdown to contend with. Since the combined conduction current capability of the devices when paralleled as shown is around 20 A, no special current limiting is needed – in fact, if you use the 250 VA transformer core (for both channels, 120 VA for each channel if you are going for dual mono construction) recommended for this design, the 5 A fuses in each supply rail of the amplifier modules are all you need for protection (of course, you must also fuse the mains input side of the completed amplifier to meet electrical safety and fire hazard regulations). During development, I shorted the output quite a few times, and the fuses simply popped, I replaced them, and the amp was ready to go again. In short, this is an extremely rugged little amplifier that can take some serious abuse.

PSRR and Front End Regulator. It's useful to point out that in a class A amplifier, the heavy quiescent current draw means that there is usually substantial ripple on the supply rails – an added whammy if the topology PSRR is not particularly good. One of the drawbacks with CFA topology amplifiers is their lower PSRR – something well designed VFA topology amplifiers really excel at. To get around this shortcoming, the front end stage is heavily filtered (R22, R23 along with C5, C8 and C9 and C11) and Zener regulated (D3 and D4) - the sx-Amplifier PSRR is in the region of 50 dB at LF using the recommended power supply. A ripple eater would also help significantly improve things here, but that will have to be something for another project – this one is about keeping it simple. JLH's 10 Watter suffered from very low PSRR primarily due to front end circuit arrangement and the bootstrap circuit and really needs a regulated power supply if it is to perform at its best whereas the sx-Amp, like Hiraga's original design, does not need a regulated PSU.

Speaker Coupling. The speaker is coupled to the output stage by means of L1, a 0.5 uH inductor in parallel with R9, a 3.3 Ω resistor. At high frequencies, this network isolates the amplifier from any capacitive loads, ensuring that it remains stable – important since the sx-Amplifier bandwidth is exceptionally wide at ~570 kHz. Further, directly off the output, before inductor, I have fitted a tab connector for a Zobel network, which is actually located on the PSU PCB. At high frequencies and in combination with L1, the Zobel network ensures the amplifier sees a load that approximates 8 Ω, also aiding in overall stability. Note that on the single sided PCB version, the Zobel is located on the amplifier PCB itself.

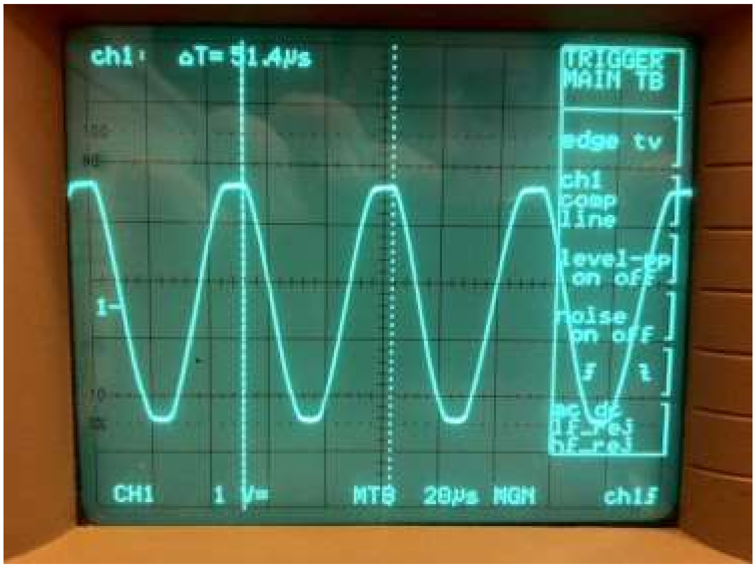


Figure 6 - sx-Amp Clipping Behavior is Soft (scope probe 10x)

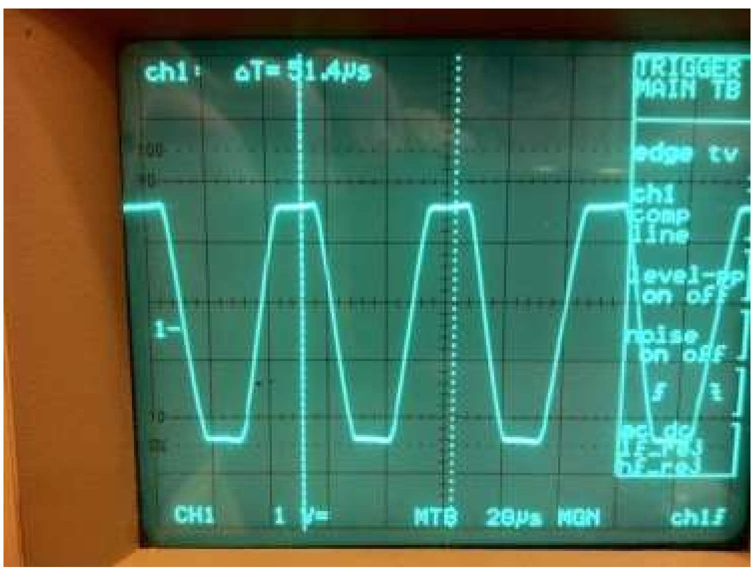


Figure 7 - Clipping Behavior with High Overdrive (Scope Probe 10x)

As can be seen from the above figures, at 40V pk to pk, the amp exits clipping cleanly with no rail sticking or parasitic ringing. For both of these figures, the amplifier load was 8 Ω.

In the event that one of the rail fuses blow open, D1 and D2 (1N4007) prevent the semiconductor junctions on the board from being reverse biased, which could lead to damage. This is a high speed design, so it is important that the supply impedance remains relatively flat and low out to beyond 500 kHz. C3 and C4 (220 uF 63 V) are electrolytic devices that are designed to mitigate the probable 100-200 nH PSU wiring inductance (which is -.5 ~ 1 Ω at 500 kHz) between the amplifier boards and the PSU.

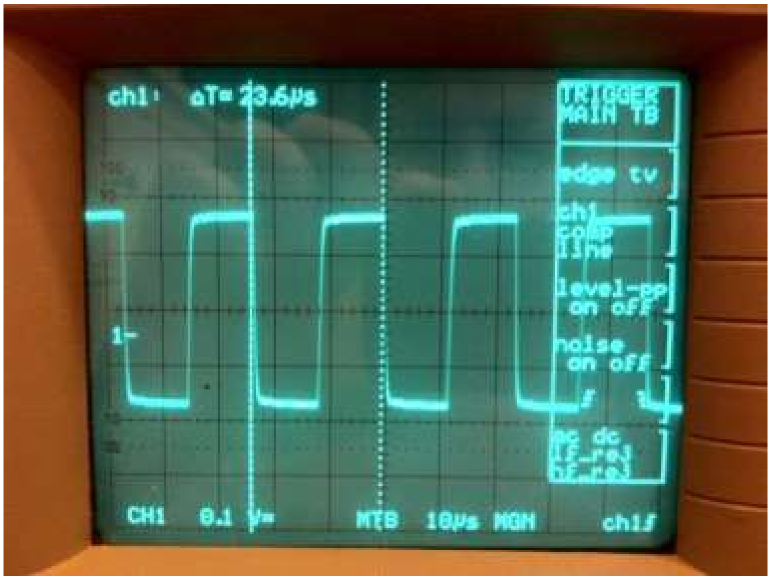


Figure 8 - 42 kHz into 8 Ω Square Wave Performance (Scope Probe 10x)

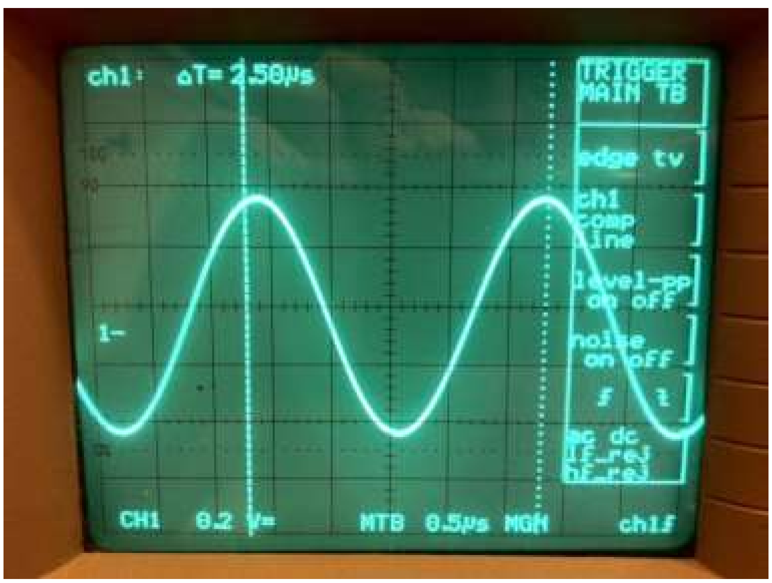


Figure 9 - 400 kHz into 8 Ω Performance 8V pk~pk (Scope Probe 10x)

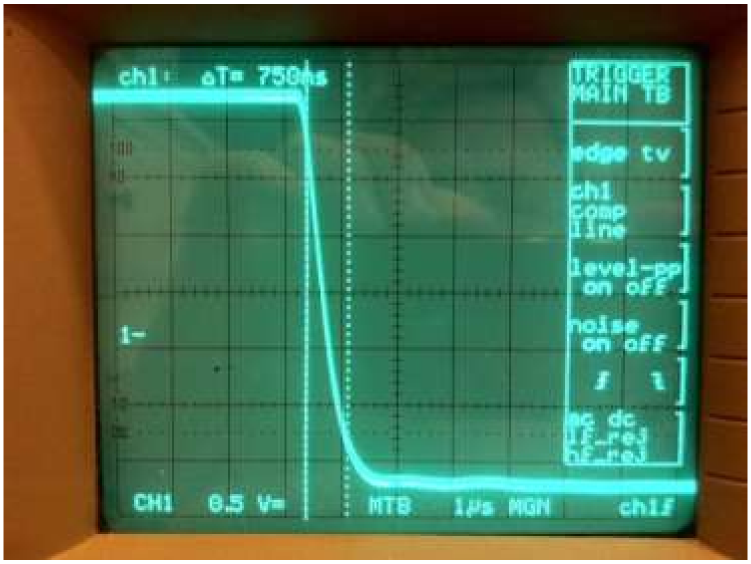


Figure 10 - sx-Amp Fall time is under 1 uS into 8 Ω (Front-end Filter in-situ, Scope Probe 10x)

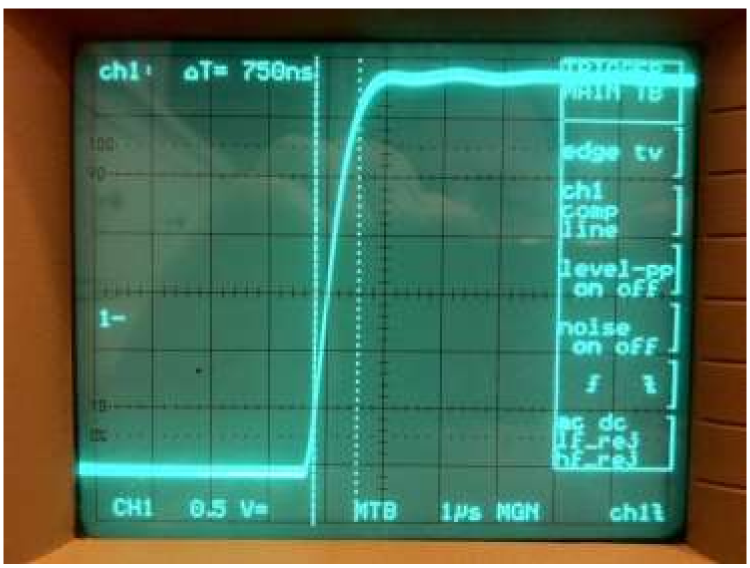


Figure 11 - Rise Time is also <1 us into 8 Ω (Front-end Filter in situ, Scope Probe 10x)

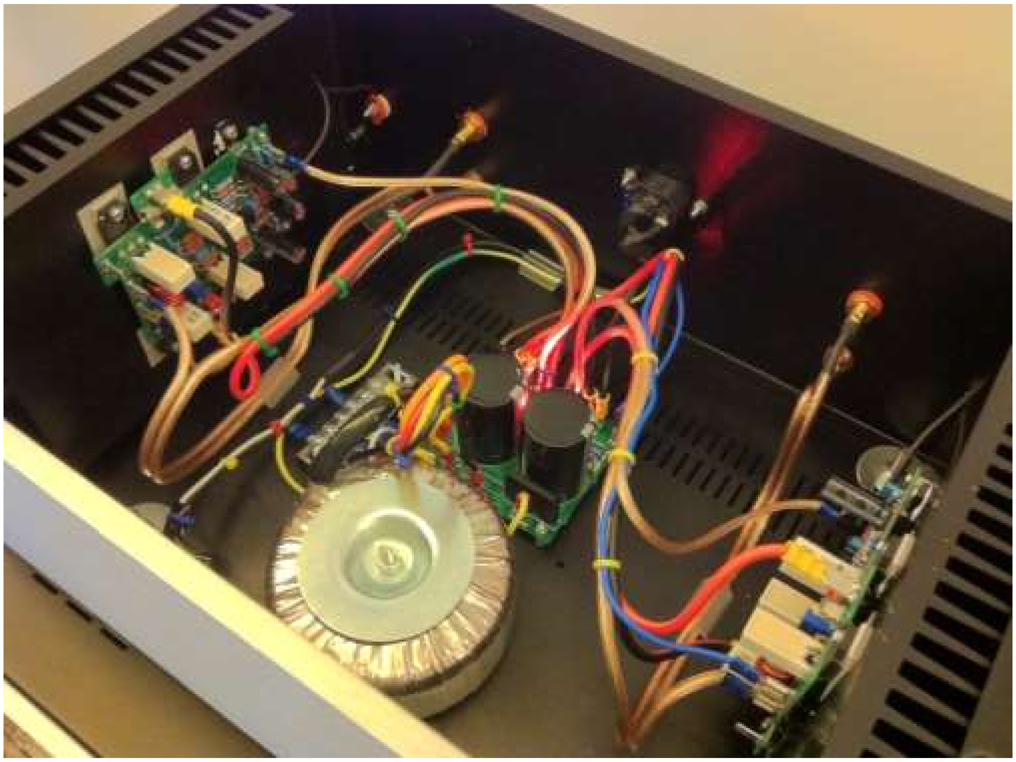


Photo 2 - Internal View of the Ovation sx-Amp - prototype hook up wiring

DC Performance and Offset Drift. The offset drift of this design is about 40 mV over the full operating temperature range. During initial set-up, the offset is dialed out once the heatsinks reach their operating temperature, which from an ambient of 27 °C (which was the temperature when I was doing my tests) takes about 20 ~ 30 minutes. For the 0.4 degree Celsius/W that I used, the final heatsink temperature is in the region of 60 °C. This means that at switch-on, the offset will be at maximum, and then as the amplifier warms up, the offset reduces to zero, with about 80% of the offset disappearing within the first 4~5 minutes. Given the simplicity of this design, the offset drift is remarkably low in my view, and once warmed up, stable to within 1 or 2 mV over many power-up and power down cycles. No doubt the close physical proximity of the input transistors promoting good thermal tracking, along with their high gain, helps in this regard. As far as I am aware, there are no 'blameless' or JFET input amplifier designs out there that can deliver this type of DC performance without a servo, DC blocking capacitors, or a combination of both. Like Hiraga's CFA design, and despite its simplicity, the sx-Amp is entirely DC coupled. ***For the reason, it's important that the sx-Amplifier source is DC free. If you have doubts, then you should AC couple the source to the sx-Amplifier with a 47uF or greater bipolar capacitor.***

Switch-on Transients. One of the other very pleasant surprises with this balanced design is the *complete lack* of any switch-on or switch-off thump[7] – quite a contrast compared to the Musical Fidelity A1 that I repaired for a friend a while back which was certainly very disconcerting in this regard. The large power supply filter capacitors, along with generous front end Zener regulator decoupling (C6 and C12) all help to slow the rate at with the supply rails come up, allowing the front end (Q8 through Q11) to assert DC control before turn-on transients can cause a problem.

[7] However, don't forget that if you turn your pre-amp ON or OFF while the sx-Amp is powered up, you will get a switch off thump from the pre-amp. Note also that the sx-Amp power supply takes about 15 seconds to power down to the point where it won't pass through any ON/OFF noises from the source.

5. Feedback and Compensation

The sx-Amp uses a moderate amount of feedback – about 34 dB (which is 12 dB lower than the 46 dB quoted by JLH in his original 10 W design from 1969), but importantly, and because of the CFA topology, the loop feedback is constant from DC all the way out to about 60 kHz (see Fig 5).

Fig 12 shows the closed loop response of the amplifier (simulated). The gain of the amplifier is set by the parallel combination of R12 through R16 (5 x 1 k in parallel = 200 Ω) and R11 (15 Ω) yielding a closed loop gain of 14.3x, or 23 dB. VFA lead or lag techniques (e.g. Cdom or a small capacitor across the feedback resistor) in a CFA run the risk of causing oscillation and overshoot, and especially so where there is high open loop gain and the risk of exceeding roll-off slopes of >20 dB per decade[9]. Because of the very wide bandwidths involved, paralleling the feedback resistor with a capacitor (lead compensation) can insert a zero in the feedback loop that also pushes the ULGF intercept up in frequency where there is likely to be more phase shift, and potential instability. The usual (recommended) technique used to compensate CFA's is to adjust the value of the feedback resistor, Rf; some approaches capacitively shunt the TIS output node to ground, although this is very sub-optimal in a power amplifier and it not without stability risks, while adjusting the value of Rf in a high power discrete amplifier is not a convenient approach. I do not recommend any of the above compensation approaches in CFA topology power amplifiers.

Instead, the method used in the sx-Amp couples the TIS output node back to the current summing junction via C7 (somewhat analogous to MIC in VAF topologies), and was first publicized by Mark Alexander in his Alexander Amplifer[10]. This technique improves the settling time and loop stability markedly over conventional CFA TIS capacitive shunt loading. At HF, this approach effectively encloses the front and TIS stages in a fast local feedback loop before the output stage poles get a chance to introduce enough phase shift to cause a problem. Its important that the transition frequency is kept well above the audio band in order to maximize the curative effect of feedback – if you set it too low, the result will be unnecessary levels of HF distortion (see Fig. 13 for a plot of the loop gain). Alexander remarks that it is possible to get away with much lower values of capacitance compared to simple TIS shunt loading, and my investigation during design and simulation confirmed this as the optimum major loop compensation configuration for discrete CFA topology designs.

I have erred on the side of caution in terms of the overall bandwidth on this amplifier. Experimentation showed that the main compensation capacitor, C7 220 pF, could be reduced to

[9] See Intersil Application note AN9787.1 and AN9420

[10] See appendix in his document for complete derivation

75 pF and the -3 dB point extended to around 1.5 MHz along with a higher slew rate. However, to do so invites problems with RF pickup. The front end filter is important; R17 and C13 set the -3 dB frequency at about 720 kHz, and when combined with the output filter network response dominated by the inductive reactance of L1, yields a measured overall response that is -3 dB down at 530 kHz (8 Ohm load). By any measure, this is a remarkably fast power amplifier both in terms of bandwidth and small signal rise/fall time (~50 ns) - this is entirely due to the CFA topology employed in this design of course.

Without this filter (and the output inductor and its associated damping resistor), the closed loop response extends out to around 4.5 MHz, and there is gain peaking and overshoot. The input filter forms an important part of the compensation of the sx-Amp and it will not perform to its published specs if omitted.

Because of its wide open loop bandwidth, the sx-Amp cannot suffer from PIMD in the audio band. Fig 13 shows the loop response is indeed less than -3 dB down at 60 kHz, even though the closed loop gain is varied from between 17 dB (GREEN trace) up to 26 dB (PURPLE trace).

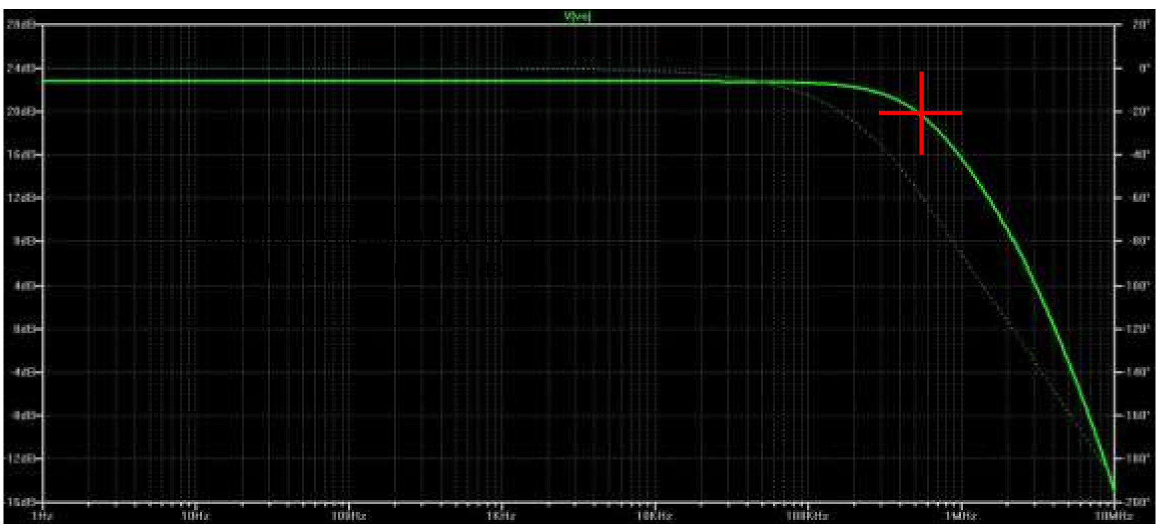


Figure 12 – sx-Amplifier Closed Loop Response

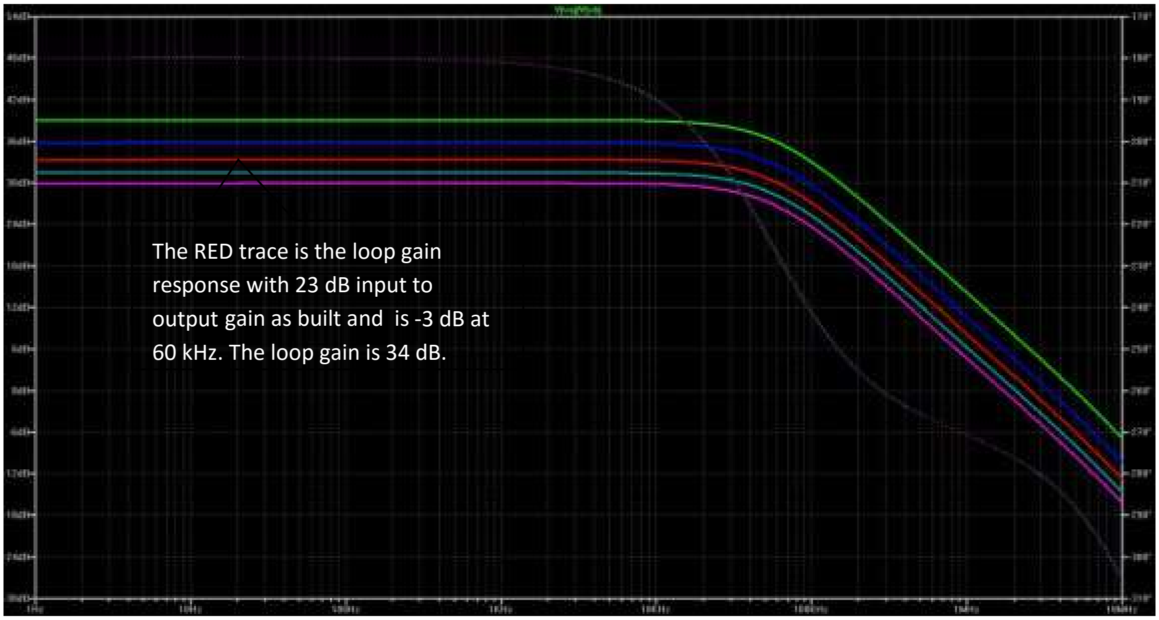


Figure 1 3 – sx-Amp Loop Gain Response for various closed loop gain settings.

Since one of the key design aims here was to achieve a uniform feedback factor across the audio band, I think it is safe to state that this indeed has been achieved. The sx-Amp rise/fall times without band limiting and using a minimal (but perfectly workable) 75 pF for C7 are in the region of 250 ns. Rise/fall times after compensating and band limiting the amplifier are in the region of 750 ns, which fulfills the other objective of building a fast, stable, wideband amplifier.

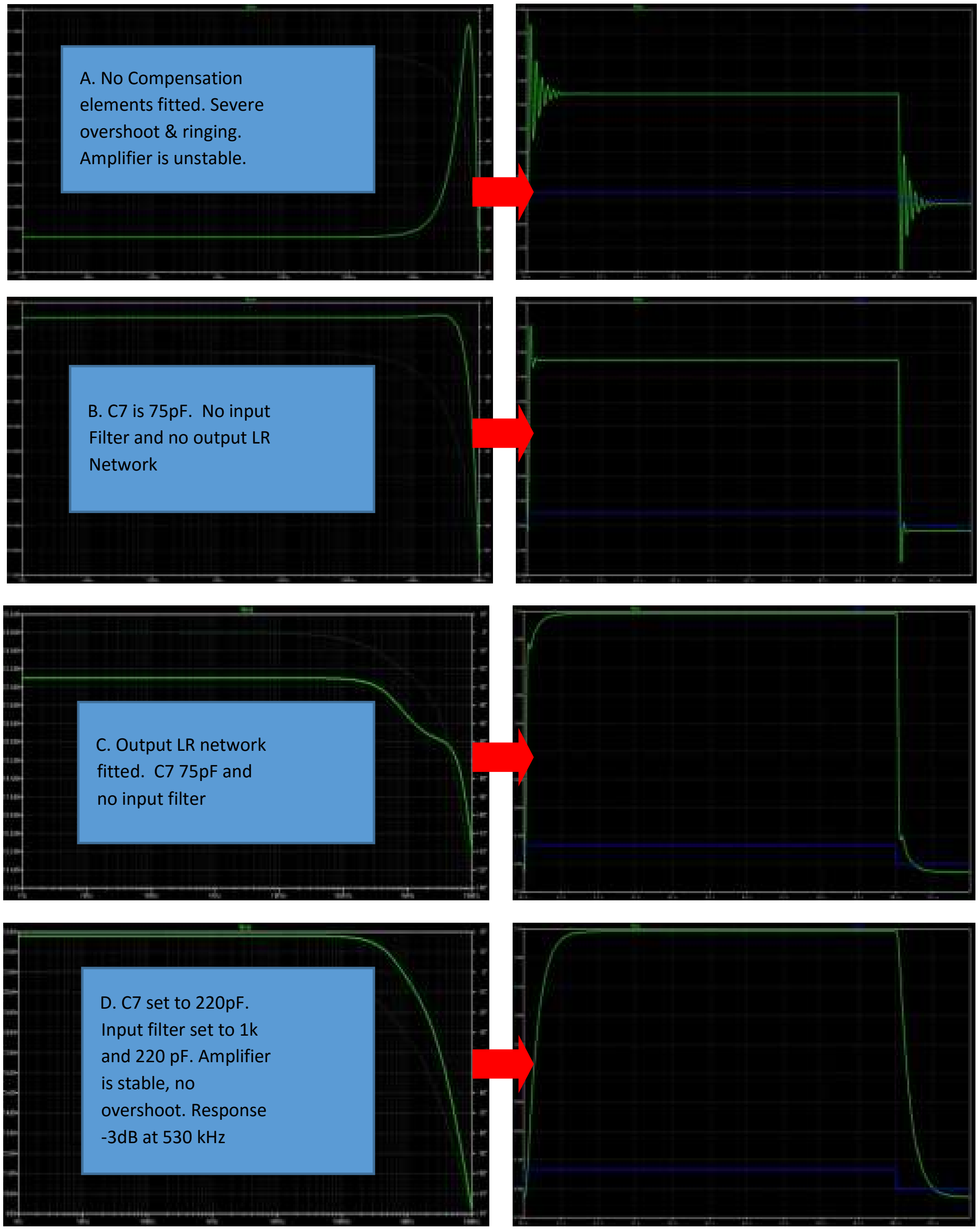
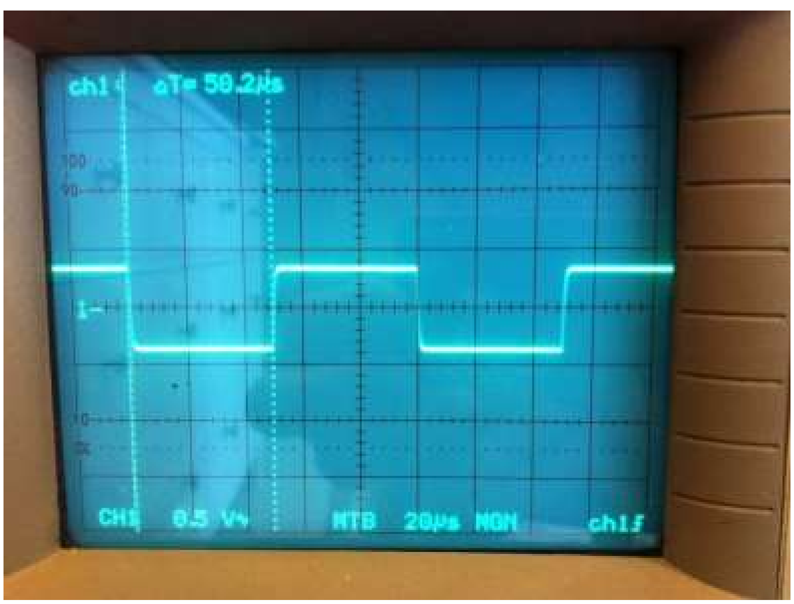
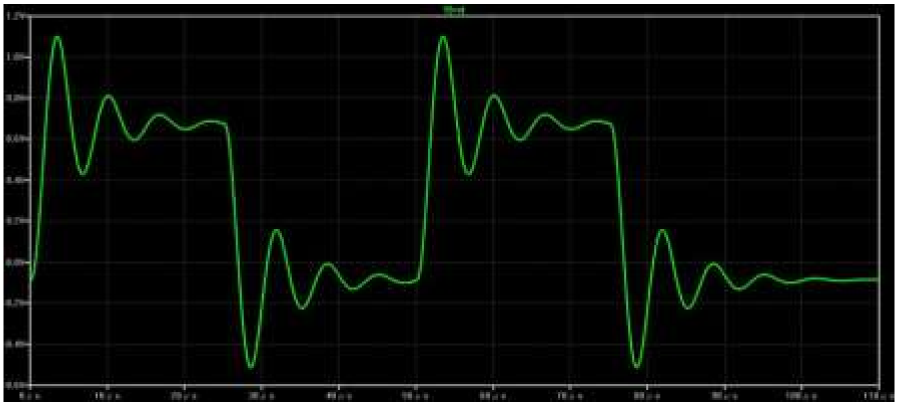


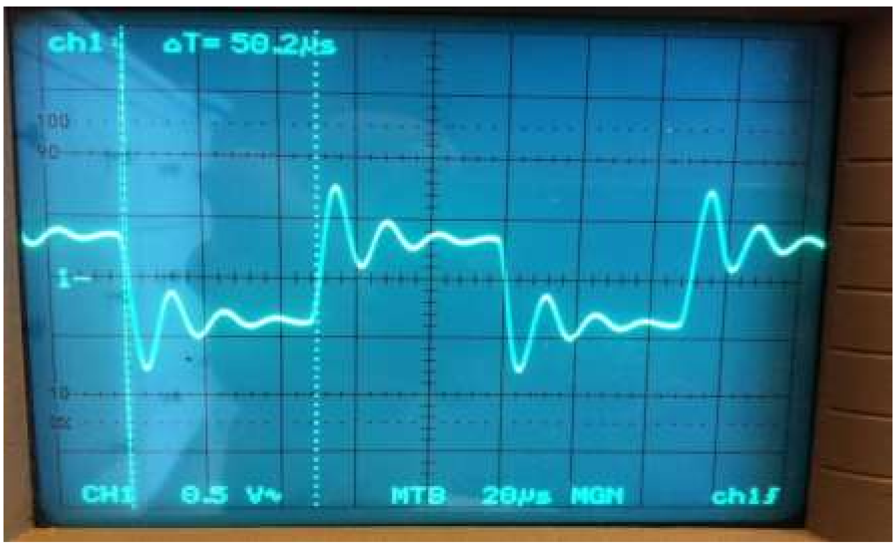
Figure 14 A to D. Left hand side graphs show the frequency response as the compensation design is evolved, while on the right hand side, the square wave response into an 8 Ω // 5 nF load is depicted. The input signal is a 0.1 V square wave with improbable 10 ns rise and fall time designed to provoke bad behavior and an ON period 10 us

Fig 14 A through D gives some idea of the compensation design evolution and how I arrived at the final response you see on the bottom graph (14D). For these simulations, a 5 nF capacitor was placed in parallel with the speaker load, and I used a very fast rise time of 10 ns – this to provoke any oscillation or ringing, though rise times of this order are never going to occur in practice. In (A) the response peaks at about 27 dB around 6 MHz with a -3 dB cut off frequency of 11 MHz. A response that wide and 'peaky' with rise/fall times << 100 ns (simulated) is not practical for all sorts of reasons – layout issues (trace inductances for instance), propensity for the amplifier to ring or break into oscillation in the presence of capacitive loads and RF ingress, to say nothing about the probable sonic impact as well. 14B shows the response when C7 is optimized to remove any peaking. Here, the -3 dB cut-off is 8 MHz with very slight peaking at about 0.2 dB – further fine tuning of C7 would allow this to be reduced further, but for these illustrative purposes this is quite satisfactory. 14C shows the response when the output inductor, L1, is included. I purposefully look at the response in this order (14A to 14C) in order to understand the response impact of the output inductor. You can see how, on the associated square wave depiction on the RHS, how the response anomaly places a strange little kink just before the output plateaus. The response drops off slightly at about 200 kHz, with the -3 dB point now lowered to about 4.5 MHz. Finally, in Fig 14D, you see the overall response with the front end filter (C13 = 220 pF) in place with the -3 dB point at 530 kHz and the rise/fall time of 750 ns into a resistive load and the 5nF parallel cap.

The photo below shows the small signal response of the sx-amp into an 8 Ohm resistive load after compensation. Rise times are well controlled with no overshoot or ringing.







The scope photo above shows the sx-Amp driving an 8 Ω//2 µF load, while the screen shot above it shows the LTSpice simulation – the correlation between the simulation and the practical result is indeed good. The output coil and the load capacitance (2 µF in this case) plus the speaker cabling and crossover in a real world set-up form a tank circuit that cause the ringing you see in these shots - and this is quite normal.

Compensation design conclusion. In conclusion, the sx-Amp response has been optimized to ensure it suffers no overshoot when driving resistive loads, and shows no tendency to break into HF oscillation with a wide range of capacitive loads (tested from 100 pF to 2 µF in parallel with 8 Ω). Despite the wide bandwidth capability of the design (about -3 dB at 1.5 MHz), the -3 dB bandwidth has been limited by design to 530 kHz. This ensures the design is tolerant of board layout, wiring and capacitive loads and suffers no RF ingress. The compensation transition frequency (via C7) has been set well above the audio band, as reflected in the loop gain plot in Fig. 13, thus ensuring feedback is applied equally across the whole audio band – one of the goals of this design.

6. Power Supply

Figure 15 shows the circuit for the sx-Amp power supply using a common transformer and capacitor bank for both channels. Class A amplifiers draw a heavy, continuous load, so the rectifier and capacitors need to be rated for the job, as does the transformer. Running a transformer at 80 or 90% of its rating for extended periods will lead to substantial temperature increases – you can expect over 40 °C temperature rise above ambient on a commercial specimen. This means you are going to need some good ventilation – and keep heat generating sources away from the filter capacitors in your final construction. To keep temperature issues under control, I used an oversized 250 VA Triad Magnetics VPT36-6940 250 VA 18-0-18 transformer with a split primary from Digikey. I have been very impressed with Triad's quality (I used a 1 kVA on my earlier e-Amp); their products are well made, and importantly, mechanically very quiet. Of course, they lack the screen and belly band that would make them an even better deal for audio, but the off the shelf unit gets us to 99% of where we need to get to.

Figure 15 – sx-Amp Power Supply

For a dual mono capacitor bank, a total value of 25 milli-Farad (25 thousand micro farads) per rail is required per channel, and if constructing the amp as a straight stereo design with a common power supply, then 47 milli-Farad per rail is required. These are very high values for a

15 W amplifier, but there is a continuous 1.4 A per channel load current being drawn and we also have to mitigate the lower PSRR performance of the CFA topology. Fortunately, you can use 25 V devices, and these are quite reasonable – 47 milli-Farad devices with the right ripple current ratings start at about $9 each at Mouser (Part # 667-ECO-S1EP473EA).Note however, that the 100/120 Hz charging currents spike at over 30 A, and are full of HF harmonics, so layout is critical if you are to avoid inductive or capacitive coupling into the amplifier circuitry. For the rectifier, I used a 25 A PCB mount Fairchild DFB2060 device. At full load, you can expect about 1 V of ripple with the capacitor values quoted above. You need to twist the +, -, 0V and speaker return line together within the amplifier housing to minimize the electromagnetic radiating area, and this can have a dramatic effect on reducing supply noise radiation into the sensitive amplifier circuitry.

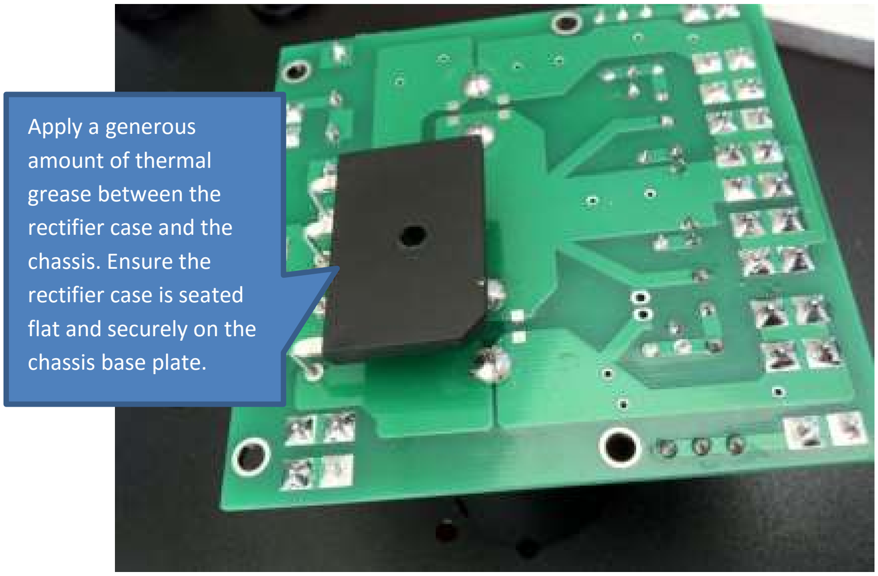


Photo 3 - Detail Showing How the Rectifier is Mounted on the Underside of the PSU PCB

Because of the *substantial heat dissipation in D4*, I ended up mounting it underneath the PCB, by folding back as shown in Photo 4. This allowed me to couple it to the bottom plate of the steel chassis for cooling using thermal grease.

It is important that the rectified output voltage is around 22 VDC +-2 V when idling. If you end up with voltages much higher than this, the result will be unnecessary dissipation and the risk of overheating, or, the requirement for an even larger heat sink.

Referring to Fig 14, the transformer AC secondary's are fed into a 25 A PCB mount rectifier D4. C4, C5, C7, and C8 (1 uF 50 V MLCC types) provide local decoupling for the large electrolytic filter capacitors C1 and C6. D1 is a ground lifter – ***make sure J3 is solidly connected to the metal chassis!*** This allows the amplifier circuitry to float at +- 2 diode drops around the chassis earth potential and normally resolves most noise problems if correctly implemented. On the chassis mount terminal block you can see in Photo 3 towards the top LHS of the transformer, I wired in a 22 Ohm 0.5 W resistor between the power supply 0 V and the chassis – i.e. in parallel with the ground lifter. The latest PSU updates have incorporated this resistor on the PCB.

D2, D3 and their associated resistors provide power indication and a bleed off load, and can lend a warm tube like glow to your final effort should you choose red LED's. Incidentally, with no load connected to the power supply, the LED's take a full 5 minutes to turn off after mains power is removed – there is a lot of charge stored in those filter capacitors! In this design, the Zobel networks (this is C2 and C3 and their associated 33 Ω resistors) are mounted on the PSU board. Its important that the Zobel ground return line does not get mixed up with the signal or decoupling ground, because it can under worst case conditions, inject HF noise into those ground nodes, causing distortion, or even instability in really bad layouts. Using the approach shown, helps prevent this type of ground contamination. The whole PSU board fits on a small 80 mm x 80 mm PCB, with all connections made using 5 mm push on tabs.

Due to the substantial load and large filter capacitor, the peak currents in the cabling from the transformer are very high. This makes wiring placement to avoid magnetic induction pickup quite critical, and I spent about two hours with a pair of headphones connected directly to the outputs to get it down to very low levels. With hindsight, a metal screen or brace over the transformer and power supply board, similar to my approach in the e-Amp and Ovation 250, would have made the job much easier and no doubt allowed me to get even lower noise levels. Let me add, with my ear right against my B&W 703 speakers, there is NO audible hum - either with the inputs open circuit, or when the pre-amp is plugged in– this amplifier is extremely quiet. However, if the cabling is moved around, for example in the vicinity of the transformer or the cables going to the power supply or terminal assembly on the base of the chassis, audible hum on the headphones is apparent. Cables need to be as short as possible and carefully dressed.

7. Component Matching.

I've had a lot of questions on both the 100 W nx-Amp, and the sx-Amp about matching, and whether or not it is important. To get the best out of the sx-Amp (and any low component count CFA for that matter), a bit more care is required during the build process around component matching. Since the loop gains are lower at DC, and the requirement to use low feedback resistor values, AC coupling of the feedback network is not a viable option, unless you are prepared of course to use huge capacitor values. Since the sx-Amp is all DC coupled and does not use a servo, its important that steps are taken to minimize drift and output offset. Further, this is a fully balanced design, so if some effort match the critical components is taken, the distortion will be minimized.

The following component pairs should be matched

- Q8 and Q10 (diamond buffer transistors)
- Q9 and Q11 (diamond level shifters)
- Q6 and Q7 (TIS transistors)
- D4 and D3 (10V Zener Regulators)

For Q8 through Q11, match hFE to within 5% or better, and Vbe to within 5 mV – in both cases, tighter is better. This is easily done if you buy a bag of these devices and then sort first for hFE, and then after that group by Vbe.

For Q7 and Q8, match hFE to within 15% and Vbe to within 10 mV - if you can, tighter of course is also always better.

The Zeners are available as 1% and 2 % - I would recommend you purchase 1% types and then match more tightly if you can. I've checked a batch of 10 V Zeners and gotten matches to within 20 mV – i.e. 0.2%.

I do not specifically recommend that the resistors are matched. Nowadays, if you buy a reputable brand, the matching in general is extremely good – this is as a direct consequence of course of the improvements in processes and process control over the last 30 years or so in passives – and semi's are generally also very good in this regard.

Now, let me also add that if you *do not* match any of front end components, your sx-amplifier will still work: matching though, really ensures that you get every last ounce of performance out of the design.

You should always use the same gain grade for the front end and TIS transistors. If you use mismatched gain grades, you will get higher distortion and greater offsets – although in this latter case, you normally should still be able to dial it out.

8. Construction and Components

The amplifier modules are constructed on small PCBs measuring about 80 mm x 70 mm. I usually sandwich the output devices between the PCB and the heat sink, but in this design, to keep the board as small as practicable, the devices simply connect to the edge of the board. Note however, that the PCB actually rests on the bottom edge of the output devices (about 5 mm of overhang), so the output transistors act like spacers. All connections to the board are via 5 mm push on tabs, except for the input connections where I used a miniature Phoenix screw terminal connector, which allows for easy removal if you ever need to service it.

The sx-amplifier boards utilize some SMD components. I did this because they are convenient and save a lot of space. Having said that, it's the 4.7 Ω 1W base stoppers and two decoupling capacitors that are SMD. To solder SMD devices, you will need a good quality, *fine tipped soldering iron*, and a pair of fine tipped tweezers and some *1 mm or less* diameter solder. Take a look at Dave's EEVblog Youtube video here for some inspiration and guidance. Once you get the technique, soldering SMDs' are great fun.

Each channel of the complete sx-Amplifier is constructed on a small PCB with the exact same dimensions and mounting hole locations as the nx-Amplifier. I usually like to sandwich the output devices between the PCB and the heat sink, but in this design, to keep the board as small as practicable, the devices simply connect to the edge of the board – note however that the PCB actually rests on the edge of the packages of the output devices, so they effectively act as spacers. All connections to the board are via 5 mm push on tabs, except for the input connections where I used a small 2.54 mm pitch Phoenix screw terminal connector, which allows for easy removal if you ever need to service it.

To start off, mark the holes on your heatsink using the finished boards as a guide and measuring to where the TO3-P and TO-126 mounting holes should be after you have bent the leads (see Fig. 17). Drill and tap to M3. Take care with this procedure as breaking off a tap in a heatsink generally means you have to scrap the heatsink, or move the location of the PCB – either way, it's a botched job, so take extreme care.

For assembly, you should start with the SMD components first on all the boards (both sides), followed by the conventional resistors then capacitors and finally the small signal transistors, but NOT any of the power devices including the TO-126 drivers and TIS transistors. Once this is done, and you have checked your work against the circuit diagram and component overlay, you can proceed to fitting the power transistors as described below.

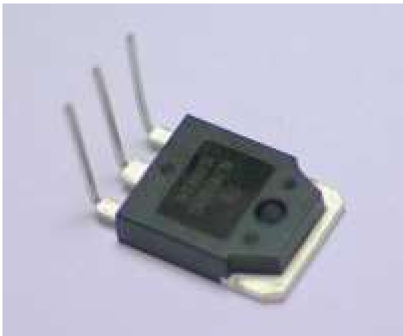


Figure 17 - TO3P Lead Bend

To mount the TO3P power devices, start by bending the leads at 90 degree angle as shown in Fig 17. Now mount the power devices onto the heatsink *in their correct locations wrt the PCB*, screwing them in place, but not tightly at this stage. Next, carefully fit the PCB in place. This is quite tricky, because you have to align the holes on the PCB with the transistor leads which are facing up at 90 degrees. Once in then place, make sure the transistors are nice and straight and screw them firmly in place (i.e. so they don't move); next screw the PCB lightly in place via the mounting holes – **but make sure the PCB is not flexed in any way** – you simply want to keep it in place for the next operation. Your board should look like that in Fig. 16. Now solder the transistor leads in place. Apply some pressure near to where you are soldering to ensure the PCB is seated flat on the power device case, which acts as a spacer – see Fig 19. During this process, make sure the PCB is neatly aligned with the power devices **this** and they remain orthogonally oriented with respect to the PCB.

Figure 16 - PCB + Heatsink assembly prior to soldering the power devices. In photo, it is the nx-Amplifier board that is shown – the sx-Amplifier assembly is exactly the same

Figure 19 - Apply pressure to PCB when soldering

Now neatly clip the excess lead length from the soldered power devices; unscrew all the power devices and the PCB and remove the assembly from the heatsink. Next, bend the leads of the TO-126 devices at 90 degrees, (See Fig 18) making sure the distance between the device mounting hole and the lead bend is the same as the distance between the heatsink mounting hole and PCB holes where the leads will go into. Generally, this means bending them at the pin shoulder, as shown in Fig 18. Mount the TO-126 devices on the heatsink, making sure they are in the correct location wrt the PCB and are aligned to accept the PCB. Put your heatsink washers underneath the main output devices (this is important, so do not skip this step). Carefully align the PCB over the TO-126 devices so that the leads protrude through the PCB holes. Screw the PCB in place via the power devices and the PCB locating holes.

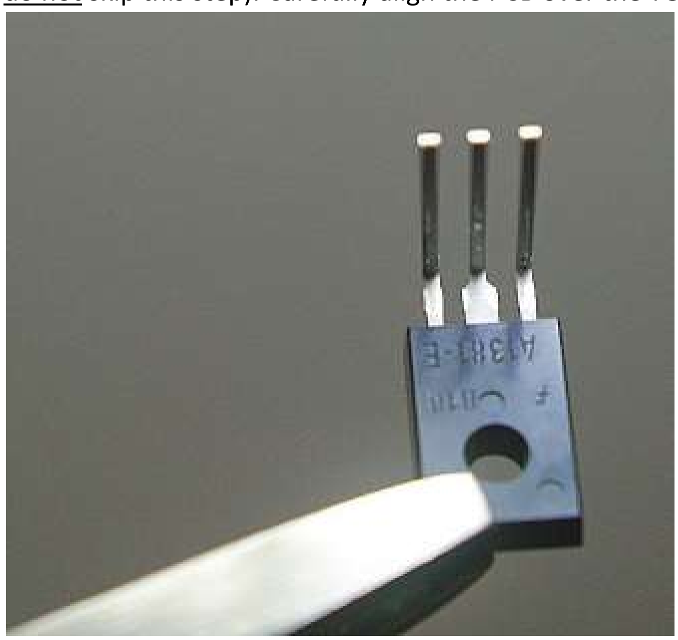


Figure 18 - TO-126 lead bend

Check to see that the TO-126 devices are absolutely flat and flush on the PCB – you may have to apply a bit of pressure on their cases using a screw driver to ensure this, after which you can solder them in place.

The finished PCB should look like Fig 20, which are now nearly ready to be mounted on their heatsinks in preparation for testing. I do not recommend that you mount untested boards in your chassis – fully test all of the boards, do any necessary debugging, and only after this move to final assembly and wiring.

Component Selection

This amp will dissipate about 60 W per channel at idle. **This is a lot of power and you need a *very substantial* heat sink – in the region of 0.4 °C/W absolute minimum with 0.2 ~ 0.3 °C/W preferred**. Once powered up and running for about 1-2 hours, the heatsink temperate should not exceed 55C to 60C. If it is much above this, your heatsinks are undersized, and, if you cannot get bigger heatsinks, my suggestion is that you dial the output stage bias current down, to keep within the limits suggested above. Incidentally, this does not mean you can dial the output stage bias current up if you happen to be using really oversize heatsinks – the upper total maximum bias current for the amplifier is still 1.4A. Again, in short, make sure you use adequately sized heatsinks of 0.4C/W absolute minimum. To guide you, for each channel, an aluminium finned heatsink of approximately the correct size should weigh about 2.5 kg.

Photo 4 - The ON/OFF Switch is Located Underneath the Amplifier Just Behind the Front Panel

I used good quality metal film resistors in throughout this amplifier. The feedback resistor (R11) is made up of 5 x 1 k MF resistors in parallel, since it will dissipate 2 W when driving at full power and if allowed to heat up, can lead to non-linearity. C7 and C13 should be decent quality silver mica, NPO ceramic[8] capacitors, or polystyrene, which is what I used. For the TIS transistors, stick with the KSA1381 and KSC3503. Lower voltage devices like the BD139/140 with higher Cob would be problematic for stability reasons and lower Early voltage means distortion would also be higher.

The simplicity of the sx-Amp means, other than the power supply and housing, the amplifier boards can be built for around $20 each, excluding the double sided PCB's. For a standard power supply for two channels, you can factor in about $160 including a 250 VA transformer. If you are going to get a transformer wound for this project, specify a belly band and a screen – these usually add a few $ to the overall cost, but pay handsome dividends in the long run. With class A amplifiers, the radiated transformer fields – due to the heavy bias current - are much more apparent than class AB amplifiers, so anything to mitigate this, like a belly band – is useful.

[8] NPO ceramic capacitors are indeed very good for audio applications

Unless you are creative and practical, housings will always be the most expensive part of a project like this, so another option is to go for a ready-made off the shelf chassis from Modushop or Fischer with integrated heat sinks – but this is going to set you back about $200, to say nothing of the shipping costs. I discovered that there are quite a few kit houses operating out of Hong Kong that sell pretty decent looking housings, and they take Paypal – this is yet another option.

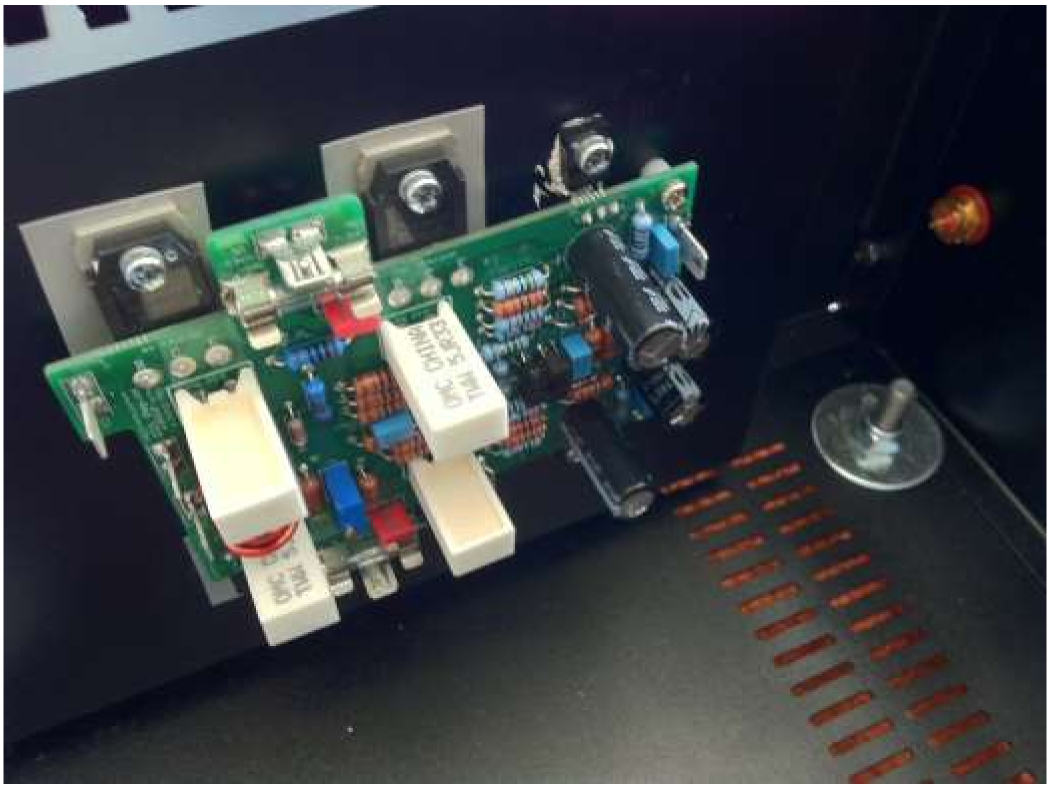


Photo 5 - A view of the Amplifier Module on the Heatsink Prior to Wiring

Make sure you earth the amplifier solidly. Use a 3pin IEC FUSED mains power receptacle. Connect the earth pin on the receptacle securely to the chassis using an M4 bolt and serrated washer. Use your Ohm meter to check that all parts of the chassis are connected solidly (means < 100 milli Ω resistance) to the earth pin on the receptacle.

Don't take chances with mains – carelessness and not sticking to recommended international wiring standards is hazardous.

Photo 6 - General Arrangement of Components – the ON/OFF Switch is Located at the Bottom Left of Center (this example is of the first of two sx-Amps built)

Photo 7 - Amplifier Housing Feet - Large Rubber Bumpers Ensure Good Airflow Under the Chassis

I used heavy duty speaker cable for the internal speaker wiring (2.5 mm diameter multi-strand) and 2 mm diameter multi-strand for the power and earth wiring. For this project, I invested about $100 on a good quality crimper, and can attest to the fact that it makes assembly and wiring a breeze – on my Ovation e-Amp and Ovation 250 power amplifiers, I used push on tabs as well, but those were all hand soldered and then dressed off with heat shrink which was very time consuming.

I used high performance Bergqvist Sil-pad thermal sheet between the output and driver transistors and the heastsink. The Mouser part number is 951-SP400-009-00-90 and they cost about 30c each.

Figure 20 - Final Wiring and General Arrangement

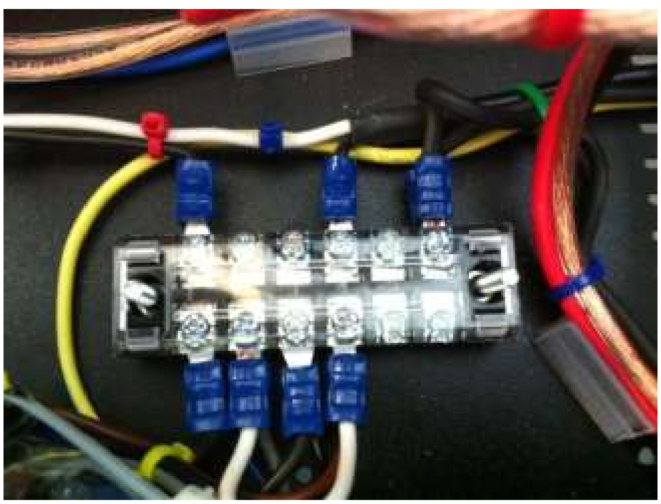


Figure 21 - Chassis Mount Terminal Block

I used a good quality chassis mount terminal block to wire the mains up to the transformer primary. The clear plastic cover ensures mains wiring and connections are visible but covered securely. Exercise extreme caution when wiring up the mains.

Testing and set-up

Before assembling the amplifier, set-up and test both amplifier modules thoroughly as detailed below. Do not assemble and/or wire up the amplifier before testing as if there are any problems, you will simply have to dis-assemble everything.

You will need 6 off ~500mm Crocodile leads, a 3.5 digit DVM with 200mV minimum voltage range, and at least a 2A current range, a fine tipped screw driver to adjust the pots, 4 x 1A 5x20 fuses and 4 x 5A 5x20 fuses.

1. Thoroughly check each board against the overlay, ensuring that all component values are in their correct locations.
2. Check the PSU board carefully, making sure the rectifiers are in the right way around, and especially check that the filter capacitors are installed correctly wrt polarity. A mistake here will be expensive, messy and possibly dangerous.
3. Measure across R6 (the output stage quiescent current set resistor) and adjust for its ***minimum*** value – i.e. 0 Ohms. This is important!

I recommend that you assemble your chassis, but do NOT mount the heatsinks yet – you will need these to be freestanding for the amplifier module test.

PSU Test

1. Wire the PSU board up to the mains – exercise extreme caution when doing this – mains kills. Do not connect anything to the PSU output.
2. Power up the PSU – the two LED's should illuminate – this will tell you that you have the rectifiers around the correct way.
3. Measure the output voltage – it should be between +-20V to +-22V. A volt or two higher or lower will be ok, but it is much above this, you will have power dissipation issues.
4. Make sure nothing gets hot – you should already have checked that the main filter capacitors were correctly positioned with respect to polarity.
5. If everything is ok, mount the PSU board onto your chassis, *making sure the main rectifier is coupled with heatsink grease to the base of the chassis* (chassis must be steel or at least 3mm thick aluminum).
6. Wire up the PSU to the mains through the fuse and power switch. Make sure the ground lifter connection to the chassis is in place.
7. Check all your wiring carefully
8. Measure between the chassis and the PSU 0V. You should read 22 Ohms. Check that there are no shorts between the PSU +-22V and 0V and the chassis.
9. If everything checks out, you can apply power and recheck the output voltages.

10. The PSU is now ready to be used to check out and set up the 2 sx-amplifier modules.

Amplifier Module Tests

1. Check again that R6 on each of the amplifier modules is set for ***minimum*** resistance of c. 0 Ω. You can do this by measuring across the B-E of Q3 (BC547C) - it should read 2.2k Ω - make sure you do not have your DVM set to diode mode when you take this reading.
2. Mount the amplifier modules onto the heatsinks, following the board mounting instructions detailed earlier
3. Use the DVM and check that you have no shorts between the power transistor tabs and the heatsink
4. Check that there are no shorts between the amplifier module 0V and the heatsinks

Next, we are going to connect each amplifier module in turn to the PSU, and make the necessary adjustments. Fit each amplifier module V- negative rail with 1A fuses.

5. With the PSU still powered off, connect the V+ and V- on the amplifier modules to the PSU using the crocodile leads – make very sure you get the polarity right.
6. Connect the PSU 0V to the amplifier module
7. Set your DVM to 2A current range and use the crocodile clips to connect the DVM across the V+ fuse.
8. Apply power
9. The DVM will show between 300 and 400mA
10. Adjust R6 for a total amplifier supply current of 1.4A
11. Wait a few minutes and monitor the supply current – it should be stable and not drift by more than a few 10's of mA after adjustment.
12. Power down, and wait for the LED's to extinguish. Once done, insert a 1A fuse into the V+ fuse holder
13. Set you meter to the volts scale and power up again.
14. Measure between 0V and the amplifier output. You should read +-500 mV maximum. If ok, switch your meter to the 200mV range and adjust R1 for a reading of 0.00 mV +-2mV. This will bounce around by 1-2mV – its ok and confirms the amplifier is working OK.
15. Next, set you meter to the 20 V range, and measure the voltage around the circuit shown in RED in Fig 5. The readings should all be within 5% of those shown – if not, you have a problem. Specifically, the 10V reading across the Zeners must be within 100mV of each other – this is important to ensure minimal offsets.
16. Assuming all is ok, repeat steps 5~15 above for the second module.

Once fully assembled, leave powered up for 30 minutes and then repeat steps 5 through 15 on both amplifier modules. Once done, your amplifier is ready for music and sound testing.

9. How Does it Sound? A Few Final Words

I have to admit I started this project with some apprehension about what a small amp would sound like – especially since I am used to listening to a big amp (250 W per channel which you can read about here Ovation Amplifier and a 180 Watter called the Ovation e-Amp). Why design a 15 W class A amplifier, and especially one that is decidedly minimalist, after two big, complex, powerful, high performance ones? Well, there are a few reasons. Firstly, designing single digit, or sub 1 ppm distortion amplifiers can provide a rewarding intellectual experience, if only to later be thwarted by practical execution that impacts both performance and build aesthetics. Secondly, reductions in distortion below about 0.5% offer *little or no further* *improvements* in the perceived sound quality of an amplifier in controlled testing[9] and there is a lot of academic material in support of this contention. Anecdotally, Nelson Pass has built a name for himself with class A amplifiers that for the most part never see the south side of 0.1%, and yet are highly rated by the cognoscenti. What about power levels? Big amplifiers sound wonderfully at ease with themselves - they are unflappable and handle music dynamics well. Low power amplifiers, like the sx-Amp under discussion here, really need to offer something unique to justify the effort in construction, and exploring this territory is the 3rd reason for undertaking this project: is there a magic class A sound that makes building something like this worth it?

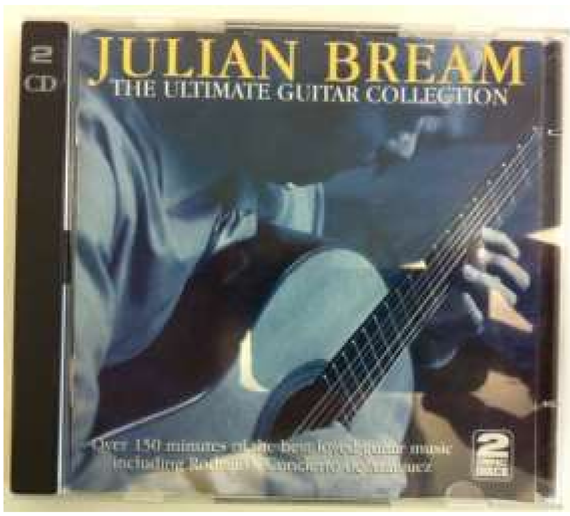
Speakers are notoriously non-linear often 3 to 5 times higher than the 0.5% I quoted above at realistic listening levels. Now, none of this can serve as an excuse for badly designed, sloppily engineered, high distortion amplifiers. We know that the ear is much more sensitive to some types of distortion (crossover for example) and to higher order harmonic content in particular. If these things are taken as considered inputs into the design process, we then have some latitude in design philosophy: Challenging, all out, ultra-low distortion designs, or something a little less demanding, but that ticks all of the right boxes given our *knowledge about how the ear/brain system works* and therefore sounds good because we avoid the major pitfalls, and is *FUN TO BUILD!* This is exactly where the Ovation sx-Amp is positioned.

Once testing and set up were completed, I was very excited to hear what this thing could do. For initial listening I chose a few classical CD's – a Lexus classical CD (freebie), Julian Bream's 'The Ultimate Guitar Collection', a wonderful 'LSO Live' sampler from Hi-Fi News, A Philips Sampler from the 1990's 'Introducing Mozart', followed by two jazz CD's: Michel Petrucciani's 'Both Worlds' and 'Time Out' from The Dave Brubeck Quartet.

I've had this LSO sampler for about seven or eight years. The tracks date from performances made between 1999 through 2002. All tracks superbly recorded with tremendous space (holographic) and dynamic range. The sx-Amp produced a wonderful three dimensional sound stage that extended well beyond the speakers, very deep and layered front to back. If you are ever looking for a classical demo CD – this has to be it! To be sure, a big part of this is the quality of the recording, but no doubt the class A magic also played

[9] This means DBT or similar in a controlled laboratory environment. Note, that my statement above does not imply any endorsement or support for high distortion low feedback or zero global feedback designs that as a result have high 2nd and 3rd harmonic distortion products. On the feedback 'to have or to not have' question, I am a declared agnostic – your ears should be the final judge.

an important role in what I was hearing. Strings have that 'bite' to them in their lower registers and the top end shimmers marvelously; brass has the leading edge snap followed by the tizz that you only get from a really good recording played through a sympathetic signal chain. The top end on this amplifier is very beguiling without any hint of harshness and the overriding sensation is one of smoothness and relaxed detachment. The scale on the Brahm's piece ('Denn Alles Fleisch Est Wie Gras') was very well reproduced which was surprising to me given the fact that only 15 W was on hand.

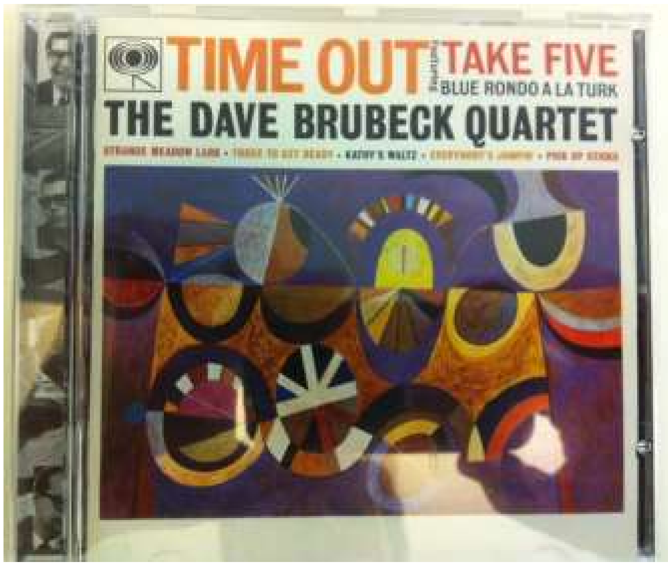


This double CD collection of Julian Bream recordings covers the four decades from 1959 through 1982. Some of the early recordings are a little noisy (tape hiss) but the sound is very spacey and the notes wonderfully rounded and resonant. My favorite is disc 2, which was recorded in 1982/83 and consists entirely of solo guitar and lute pieces. Here again, the sound staging and recording venue are beautifully captured and easily re-created by the sx-Amp.

I have had this /Mozart CD for about 20 years. There are 19 tracks and the recordings vary from good to outstanding. One of my favorite tracks on this sampler is the horn concerto in E-flat. I think Sir Neville Mariner's recording is one of the

best – the horn really floats out above the orchestra and the reverb and scale of the recording space make for an incredibly immersive experience. The whole piece is energetically played - I have found some other recordings, because of the arrangements and the conducting no doubt, to be laborious, plodding and acoustically flat by comparison. This recording is one of the better ones on this CD – I think some of the tracks are a little bright (maybe that's just because it's Mozart!), but the horn concerto is beautifully balanced.

The sx-Amp presented a very smooth, rounded sound with no hint of harshness. The layering front to back was very precise, and the left to right sound stage wide, though not as far beyond the edges of the speakers as the LSO CD – a wonderful listen however.



Dave Brubeck's 'Time Out' always amazes be because it was recorded in 1959 (like some of Julian Bream's recordings mentioned above) and you can hear the tape hiss and one or two other minor imperfections, and yet the sound is absolutely palpable. This is a re-mastered re-release but is has lost none of the quality of the original. The cymbals, always a very difficult sound to reproduce accurately, are as smooth as silk and seem to hang in the air – I've heard more recent recordings where they sound quite flat by comparison. Paul Desmond's alto sax and the bass, played by Eugene Wright, have some wonderful space around them on 'Strange Meadow Lark', one of my favorite tracks on this CD. The sx-Amp is able to convey the sparseness of the music, and the recording, reproducing the very wide and deep sound stage – very three dimensional. Again, as with the other recordings, there is a sense of a very relaxed, effortless, smooth sound.

Most of the tracks on Michael Petrucciani's 'Both Worlds' are spaciously recorded and the sound staging is good. The sx-Amp again did a great job of conveying the space around the musicians. There are a few tracks where the brass is set well back in the mix and this lends great depth to the recording, although in general the sound stage is not particularly wide. I was pleased to discover the sx-Amp could give the same sensation of depth as the Ovation 250 and the e-Amp, which offer a first class listening experience in this regard.

The following CD came to me by way of the glove compartment of my new car about a year ago.

The first track is Handel's 'The Arrival of the Queen of Sheba'. The recording is very bright, but the sx-Amp showed no signs of edginess or harshness. The other track that I really enjoy is the Beethoven Romance No. 2 in F. This is wonderfully recorded with great depth and a very wide sound stage with lots of mellifluous sounding strings that really help show off what the class A magic is all about. Track 9 – Debussy's 'Clair de lune' is another very old recording - this time from 1961 – that sounds incredibly spacey with a great stereo image.

I have not said much about the bass performance of this amplifier. You'd expect a 15 W amplifier like this lack the scale of higher power examples, but I was pleasantly surprised at how realistic the bass reproduction was. Importantly, it had weight and the notes were well sustained. I've heard a lot of systems where the bass is very lumpy. No doubt the speakers and recording play a role here, but if there are any shortcomings in the amplifier's ability to reproduce bass notes exactly as they are recorded, or drive the speakers effectively, you can bet the overall sound is going to be disappointing. Bass plays an important part in imparting space and weight to a piece of reproduced music – this is one of the reasons sub's often seem to bring a system to life, despite the fact that they are producing little or no acoustic output above 100 Hz or so.

My B&W 703s are moderately efficient at about 90 dB/W, and they are a relatively easy load to drive, so getting reasonable SPLs out of this set up is doable. The sx-Amp output stage is hefty, so up to the limits of the power supply voltage, it has no problem delivering plenty of current when required.

Of course, this is not an amplifier for a 'head banger' music set-up - the sx-Amp is better suited to jazz, classical and acoustic music. If you want 3D sound staging and shimmering highs on strings, this amp does it. If you have some efficient horns or suchlike (96 dB/W and above), then 15 W (~25 W peak class A) is going to allow you to get realistic orchestral levels, although I never found this to be an issue on the material I tried on my speakers as described above.

Figure 22 – Ovation sx-Amp Undergoing Listening Tests on Top of the 180 W Class AB e-Amp

The Ovation sx-Amp has achieved all of the goals I set when I started this project: a simple design using modern, readily available components with wide bandwidth and speed (i.e. fast rise/fall times). The design goal called for wide loop gain, which was achieved through the selection of the CFA topology. I was not expecting any huge surprises sonically, but after completing a few hours of listening tests, I can say the sx-Amp is wonderfully smooth, open and has a very relaxed, non-fatiguing sound - not what I was expecting at all, and a really pleasant surprise.

The earlier designs from JLH and Hiraga are highly regarded, and with JLH's approaching 45 years, and Hiraga's close to 35, they clearly have stood the test of time, with constructors returning time and again to their simplicity, circuit elegance and sound. Nelson Pass's mosfet based designs, some of which date back 25 or 30 years, feature very simple circuits, and much of the effort is focused on the harmonic structure of the distortion – his class A amplifiers are legendary within the DIY community and noted for their sonics.

I hope that the sx-Amplifier joins this august group of DIY amplifiers, and emerges as a 'modern take' on what is ultimately a very specialist and esoteric audiophile segment: minimalist low power class A amplifiers that focus on listenability.

Appendix 1 – BOM list for 1 off sx-Amplifier Board (updated September 2016)

Item	Count	Label-Value	Attributes	Designation	Disti Part Number (Where Applicable)	Distributor
1	1	10uF 6.3V	1206 MLCC	C1	80-C1206C106M3RACTU	Mouser
2	4	220uF 63V	RB.2/.4	C3,C4,C5,C11	667-EEU-FR1J221LB	Mouser
3	2	1000uF 16V	RB.2/.4	C6,C12	80-ESH108M016AH2AA	Mouser
4	2	220pF Polyprop	RAD0.2	C7,C13	80-PFR5221J400J11L4	Mouser
5	2	0.1uF 50V	RAD0.2	C8,C9	871-B32529C1104J	Mouser
6	2	1N4007	DIODE0.4	D1,D2	512-1N4007	Mouser
7	2	10V	DIODE0.4	D3,D4	78-TZX10C	Mouser
8	2	5A	5x20 F/Holder	F1,F2	504-HTC-201M	Mouser
9	1	CONN	SIP2	J2	538-22-10-2021	Mouser
10	1	PWR GND	TAB Connector	J3	571-638491	Mouser
11	1	Input	MPT 0.5/3-2.54	J4	651-1725669	Mouser
12	1	VNEG	TAB Connector	J5	571-638491	Mouser
13	1	VPOS	TAB Connector	J6	571-638492	Mouser
14	1	ZN	TAB Connector	J7	571-638493	Mouser
15	1	Out	TAB Connector	J8	571-638494	Mouser
16	1	0.6uH	INDUCTOR2	L1	See winding details in Text	Mouser
17	2	NJW1302	TO-3P	Q1,Q4	863-NJW1302G	Mouser
18	2	NJW3281	TO-3P	Q2,Q5	863-NJW3281G	Mouser
19	1	KSC3503	TO-2SCAH	Q6	512-KSC3503DSTU	Mouser
20	1	KSA1381	TO-2SCAH	Q7	512-KSA1381ESTU	Mouser
21	2	BC547	BC547	Q8,Q9, Q3	63-BC547CG	Mouser
22	2	BC557	BC557	Q10,Q11	63-BC547CG	Mouser
23	4	10k	AXIAL0.5	R1,R10,R34,R35	660-MF1/2DC1002F	Mouser
24	1	10k var	Offset Adjust 10T	R3	652-3296W-1-103LF	Mouser
25	1	1k var	Iq Adjust 10T	R6	652-3296W-1-102LF	Mouser
26	1	3.3	AXIAL0.5	R9	603-FRM-50JT-52-3R3	Mouser
27	7	15	AXIAL0.5	R11,R22,R23,R28,R29,R30, R31, R4	660-MF1/2DC15R0F	Mouser
28	11	1k	AXIAL0.5	R12,R13,R14,R15,R16,R17, R18,R19,R21,R32,R	660-MF1/2DC1001F	Mouser
29				33		Mouser
30	1	2.2k	AXIAL0.5	R20	660-MF1/2DC2201F	Mouser
31	4	0.33 5W	RES7WUP	R24,R25,R26,R27	588-TUW5JR33E	Mouser
32	2	150	AXIAL0.5	R36,R37	660-MF1/2DC1500F	Mouser
33	2	100	AXIAL0.5	R28, R29	660-MF1/2DC1000F	Mouser

See note below

Note: the table above shows the quantities for ONE amplifier PCB only. For a stereo amplifier, the quantities must be doubled.

Heatsink thermal pads are Bergqvist Sil Pad Mouser Pt# [951-SP400-009-00-90]

KSA1381/2SA1381

h_{FE} Classification

Classification	C	D	E	F
h_{FE}	40 ~ 80	60 ~ 120	100 ~ 200	160 ~ 320

KSC3503/2SA3503

h_{FE} Classification

Classification	C	D	E	F
h_{FE}	40 ~ 80	60 ~ 120	100 ~ 200	160 ~ 320

Any gain grade can be used, but both devices must use the same gain grade. Note, you may have to shop around to get the same gain grades

If you cannot source BC547 or BC557 you can use BC550 and BC560 in their place as follows

BC547C	use	BC550C	512-BC550CTA	Mouser
BC557C	use	BC560C	512-BC560CTA	Mouser

Use the SAME pairs i.e. either 547/557 or 550/560 – no not mix them

Appendix 2 – BOM list for sx-Amplifier PSU Board (Updated September 2016)

Item	Count	Label-Value	Attributes	Designation	Disti Part Number (Where Applicable)	Distributor
1	2	47mF 25V	RB.4/1.4	C1,C6	667-ECO-S1EP473EA	Mouser
2	2	0.1uF 100V	RAD0.2	C2,C3	871-B32529C1104J	Mouser
3	2	20A BRIDGE	DFB2060	D1,D4	512-DFB2060	Mouser
4	2	LED1	LEDT5	D2,D3	941-C566CRFNCT0W0BB2	Mouser
5	2	SIG OV	TAB Connector	J1,J2	571-638491	Mouser
6	1	GND	TAB Connector	J3	571-638492	Mouser
7	2	OVC	TAB Connector	J4,J14	571-638493	Mouser
8	1	ZNB	TAB Connector	J5	571-638494	Mouser
9	1	ZNA	TAB Connector	J6	571-638495	Mouser
10	2	Vplus	TAB Connector	J7,J8	571-638496	Mouser
11	2	Vneg	TAB Connector	J9,J10	571-638497	Mouser
12	3	OV	TAB Connector	J11,J12,J13	571-638498	Mouser
13	1	AC2	TAB Connector	J15	571-638499	Mouser
14	1	AC1	TAB Connector	J16	571-638500	Mouser
15	6	33 2W	AXIAL0.5	R1,R2,R3,R4,R7,R8	594-5083NW33R00J	Mouser
16	2	2.7k	AXIAL0.5	R5,R6	273-2.7K-RC	Mouser

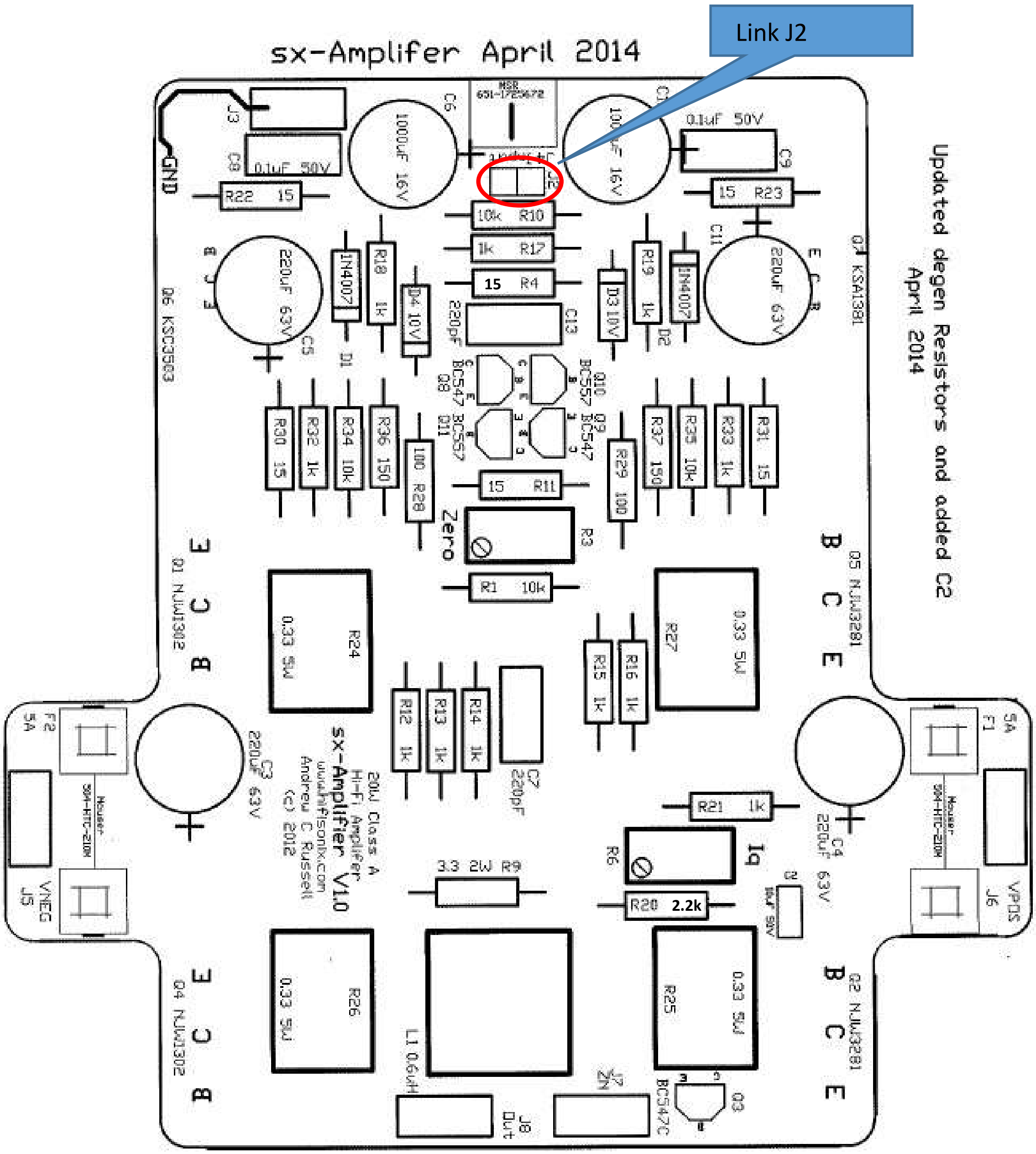
Note: you only need ONE PSU board PCB per stereo system for the sx-Amplifier, unless you plan to build a dual mono version, in which case, you must double the quantities shown and reduce C1 and C6 to 25 000 microfarads 25V

For the transformer, use a 200 or 250 VA device with a 16-0-16 minimum to 18-0-18 maximum secondary rating. The optimum secondary voltage under load is 22V.

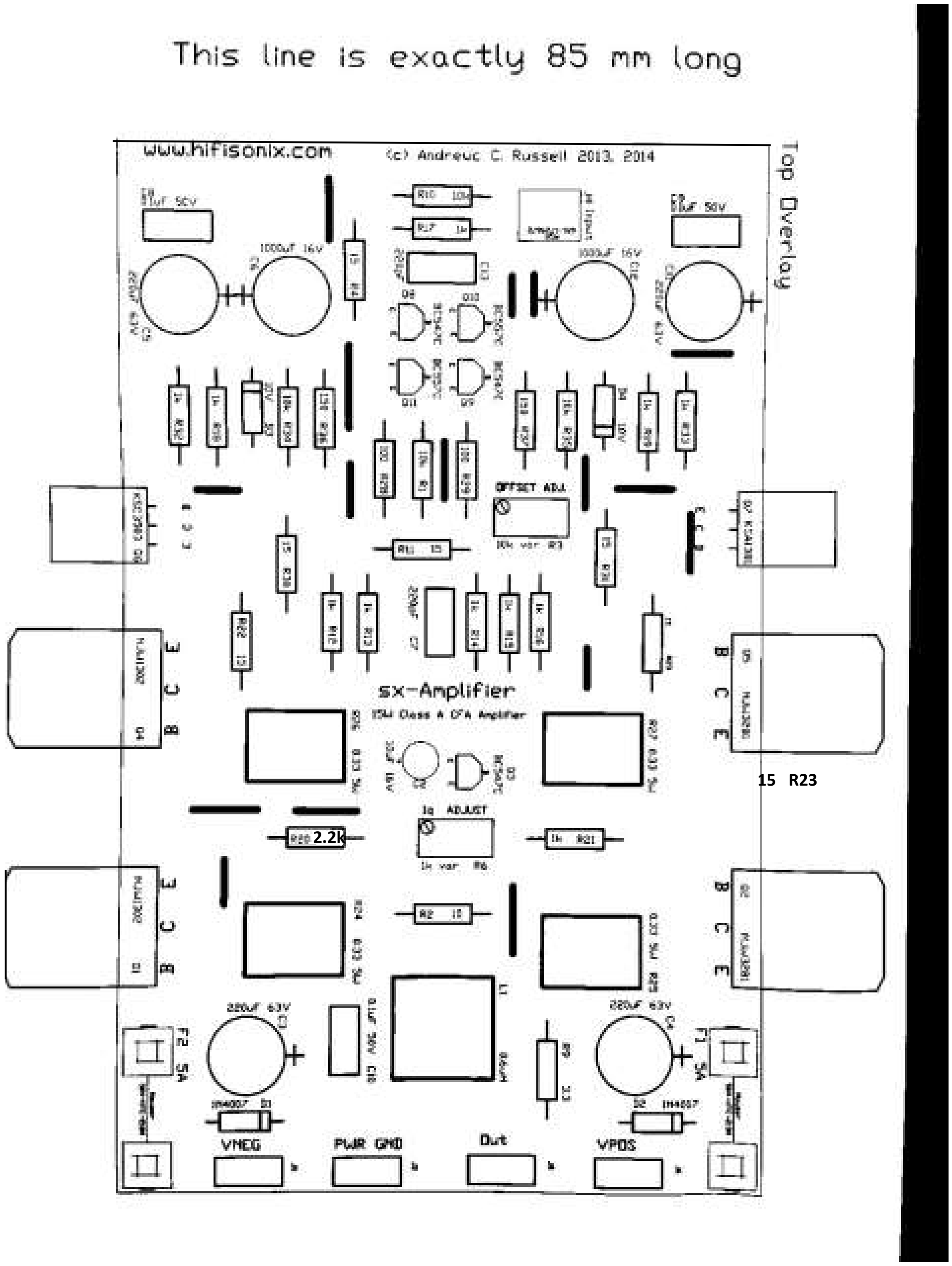
Note carefully the comments about heatsinking in the text – 0.3˚C/W is the recommended thermal performance. Higher than this will result in higher temperatures. Whatever heatsink you use, the temperature after 1 hour should not exceed 65˚C, with 55~60˚C or lower a far more preferable figure.

Note: there is no single sided PSU PCB layout available at this time.

Appendix 3 – sx-Amplifier DSTHP Component Overlay – Amplifier Board

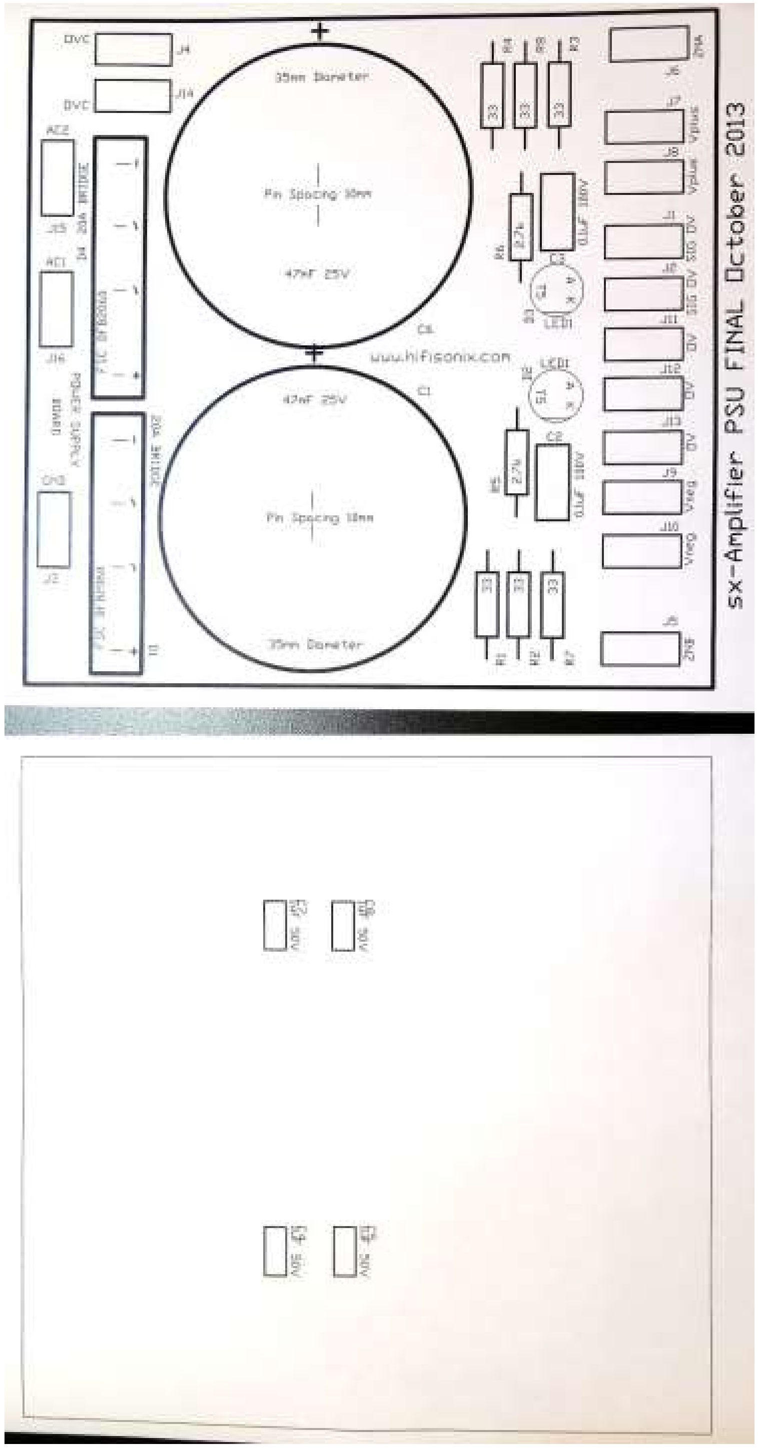


Appendix 4 - Single Sided sx-Amplifier Overlay

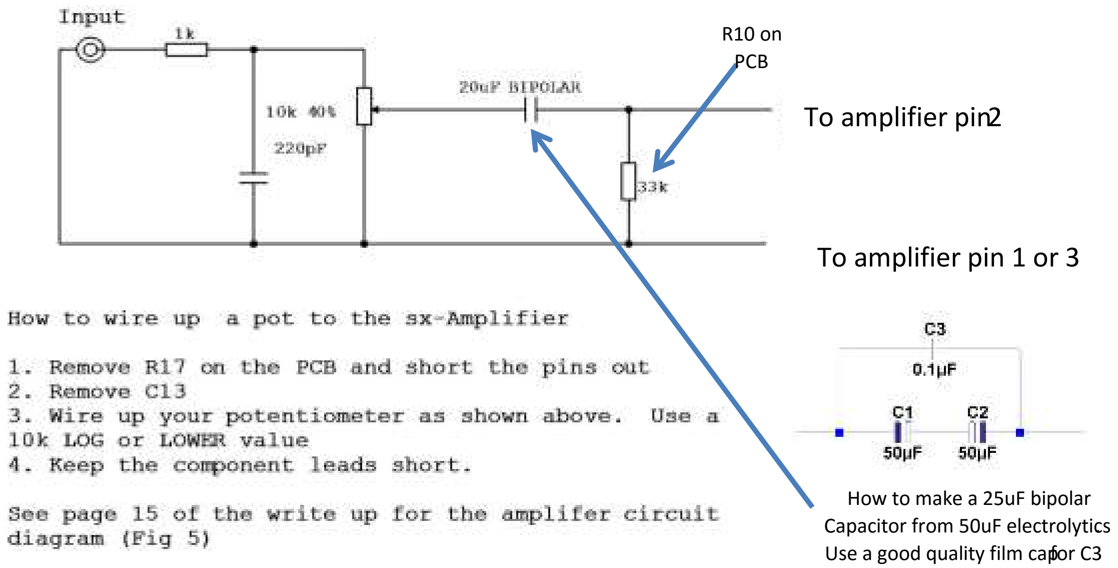


This is the layout for the **SINGLE SIDED** sx-Amplifier PCB. The component numbering is the same as the double sided version, and the same circuit diagram is used

Appendix 5 – Basic PSU Board



Appendix 6 – Attaching a Volume Pot to the sx-Amplifier Input



Quite a few people have asked me how to connect a volume control potentiometer to the input of an sx-Amplifier (the technique shown here will work equally as well on the nx-Amplifier) - the diagram above explains how to do it correctly. If you just wire up a pot to the front end, you will interfere with the amplifiers frequency response as the source resistance of the pot changes with various wiper (i.e. volume) settings, and secondly, you are likely to get 'pot scratch' noise as the amplifier bias currents flow between the track and the wiper. If your source has a DC offset, this will also cause pot scratch noise – the assumption here is that the source has no DC offset, or only a few mV worst case in which case there will be no problem.

In the diagram above, the amplifier bandwidth limiting filter is *removed from the PCB and placed in front of the pot*, filtering RF out and limiting the upper response before the potentiometer. A 20 uF bipolar capacitor keeps IP stage bias currents out of the pot, minimizing pot scratch noise. Replace R10 on the PCB with 33k - this ensures that the load on the potentiometer is not too high, as would be the case with the original value of 10k. Do not use a higher pot value than 10k – and a lower value like 5k is better still, although you do have to consider the load on the driving source. For modern CD players, DAB tuners, music servers, iPods and so forth, 5k is quite ok, but if you are feeding the sx-Amp from a tube stage, you will need to buffer it first.

Very Important: Make sure you keep the input wiring, the wiring around the pot and the connection from the pot to the amplifier PCB's WELL AWAY from all other wiring to avoid hum pickup.

Finally, if you cannot get hold of a bipolar cap, you can make one up using standard electrolytics as shown above.

Ovation sx-Amp Document History

V1.00 September 2012	**Initial Release**
V1.01 September 2012	**Minor grammatical edits to section 8**
V1.01 September 2012	**Page 6 - Corrected footnote 4 – Unity Gain Bandwidth**
	Page 23 – Corrected picture references
V1.02 September 2012	**Pages 6,7 & 19 – Clarified Comments on Compensation and CFA slew rate and bandwidth; numerous other edits and minor corrections.**
V1.03 June 2013	**Added pictures from second Ovation sx-Amplifier built in June 2013**
V2.0 October 2013	**Expanded design discussion, corrected circuit diagram, added notes about matching**
V2.08 February 2014	**Added appendices with component overlays on pages 45 and 46**
V2.09 July 2014	**Incorporated updates to diamond buffer emitter degeneration resistor values; corrected loop gain value (its 34 dB and not 28 dB as stated in the earlier versions); Added reference voltages around the circuit diagram**
V2.10	**Corrected value of R20 2.2k. Corrected circuit diagram to match PCB Layout around the bias controller – specifically the base connection of Q3. Added distortion plots; added single sided layout. Updated BOM September 2016**

Last document save: The sx-Amplifier V2.10.docx March 2016