Pass X150

Owner's Manual

Now For Something Completely Different:

The X150 stereo amplifier embodies the design technology and refinements of the larger "X" series amplifiers including extensions of the patented Supersymmetry circuit.

The Supersymmetry circuit topology was granted a U.S. patent in 1994, and is the result of 19 years of effort by Nelson Pass. The amplifier uses highly matched components in a classically simple balanced Class A circuit. The amplifier contains only two simple stages: the first is a balanced single-ended Class A voltage gain stage. Its output drives a bank of high power Mosfets operated as voltage followers.

These are inherently low distortion types of circuits, but their performance is improved when operated in balanced mode through cancellation. Distortion and noise identical to both halves of a balanced circuit will disappear at the output, and in a well-matched symmetric circuit, most of the distortion and noise is identical.

Supersymmetry enhances this effect by providing a connection between the two halves of the balanced circuit that further perfects the match. Any distortion and noise not already identical to the two halves is made identical, and the result is improved cancellation at the output.

Unlike feedback techniques where the goal is to correct for the distortion by feeding a gain stage an inversely distorted signal, Supersymmetry seeks merely to create perfect matching.

Matched balanced power circuitry typically sees a distortion and noise reduction of about 90% (20 dB) through a balanced connection without any additional effort. The Supersymmetric circuit delivers another 90% reduction, so that the X series has about 1/100 of the distortion of a conventionally simple amplifier. Actually this ordinary distortion and noise can still be seen at the output of one half of the circuit, but since it is virtually identical on the other half, it goes away at the speaker terminals. This gives good measured performance, which because it is simple, also sounds excellent.

Previously these kinds of simple Class A circuits have been popular for their sound quality in low power amplifiers, but have not found application at high power levels due to excessive distortion and low efficiency. Supersymmetry overcomes this barrier, delivering the sweetness, staging, and detail of very simple circuitry up to kilowatt power levels and beyond.

The X Series amplifiers have the tremendous dynamic range (>150 dB) to do justice to the 24 bit recordings of the 21st Century. The simple but powerful circuitry moves easily from total silence to explosive transient and back to silence without a trace. It’s a spooky experience.

So relax and enjoy your amplifier. Call us if you ever have a problem or question. Nelson personally answers his email addressed to **nelson@passlabs.com**, and you are welcome to ask questions or offer comments. Thank you for buying our product.

Setup

You can position the amplifier anywhere you want, but it requires ventilation. We do not recommend placing it in enclosed cabinets or small closets without means for air to circulate freely. The amplifier idles at about 200 watts.

Let's talk about power requirements. The amplifier draws about 2 amps (continuous rms) out of the wall during normal audio operation, and this reflects mostly the idle current that we run through the output stage. If you are driving a low impedance load, you will draw more than this, but this will not be typical.

The X150 is provided with the more conventional AC line cord, which is rated at 15 amps. The circuit ground is attached to the chassis in the conventional manner.

Under no circumstances should you defeat the ground connection of the power cord. For your safety, the chassis of the amplifier should be earth grounded.

Looking at the rear panel you will see the AC power cord receptacle, a power switch, fuse holder, two pairs of output connectors, a pair of 5 way connectors for remote turn-on, two RCA input connectors and two XLR balanced input connectors.

Make sure that the power switch is off (down). Plug the AC cord into the back of the amplifier, and then into the wall. Then turn the switch on (up). The lights in your house will blink when the power supply charges the capacitors.

On the front panel, the "Standby" LED indicator should be glowing blue, indicating that the power is on. The "Power" LED should not be on. If the "Power" LED is on, don't get excited, just use the front panel stand-by button to go to stand-by mode, with the "Standby" LED on and the "Power " LED off.

OK, so the amplifier is sitting there in stand-by mode with just the single blue LED lit. No speaker connected yet. You can go ahead and connect the source and speakers.

The amplifier can be driven single-end or balanced, if driving the amplifier single-ended leave the supplied jumpers in place (between pins 1 & 3 on the XLR).

Now that the source component is connected, make sure there is no signal coming from it, probably by turning the volume all the way down.

With the speakers connected, push the front panel button to activate the amplifier. The "Power" LED will come on.

You are ready to play music.

Do everybody a favor and try not to have shorted output cables. It happens accidentally all the time, and the amplifier is designed to survive, but I wouldn't bet the farm on it.

Of course it's always possible that something could go wrong. If so, don't get excited, just relax. It's really aggravating when something like this doesn't work, we understand, but it will get fixed. We go to a lot of trouble to make products reliable, and the failure rate of our amplifiers is almost non-existent. This is small comfort to the few, but take it easy and give us a call if you have problems.

People are interested in how long it takes for these amplifiers to break in. It takes about an hour for them to warm up, and this is where we adjust them first. Then we adjust them again and again over a couple of days, keeping the bias and offset in the sweet spot. Our environment is about 23 degrees Centigrade, room temperature, and the heat sinks will rise to about 22 degrees C. above that, for a heat sink temperature of 45 degrees C.

In your setup the temperature may vary a bit due to line voltage and ventilation, but it is not a big deal. You should be able to put your hands on the heat sinks without discomfort.

The amplifier has a thermal cutout that will disconnect AC power if the temperature exceeds 75 degrees Centigrade. This should never occur in real life.

More things to know: You can remotely operate the stand-by mode by applying 12 volts DC to the single pair of 5 way connectors on the rear of the amplifier. The positive of the 12 volts DC goes to the red connector. This connection has an actual operating range of about 9 volts to 15 volts. This switching will override the front panel button, so if you want the button to operate, leave the rear connection open.

So much for essential information.

Speaker Interface

The X150 is optimized for loads nominally rated at 4 ohms and above. You can run the amplifiers into a lower nominal impedance without difficulty, and we are not aware of a speaker on the market that presents unusual difficulty with these amplifiers.

The X amplifiers do not care particularly about the reactivity of the load. Reactive loads typically will have slightly less distortion at a given voltage/current level than resistive loads, but will make the amplifier run a little hotter. The X circuit was designed to be quite happy driving electrostatic and other speakers.

When driving transformer-coupled loads directly, as in some electrostatic and ribbon designs, some attention must be paid to the DC character of the situation. If the transformer primary is being driven raw with no protection from DC and your source has DC voltage, or in cases where the small offset of the power amplifier is still too much, you may create distortion in the transformer and get less than optimal performance from it. Generally this is not the case with transformer coupled loudspeakers, but it does occasionally surface. In these cases, take special care that the source does not contain a differential DC component, and confirm the differential DC offset of the amplifier is sufficiently low. This is easily adjusted by a qualified technician armed with the service manual. Again, consult your dealer or call us.

Interconnects and Speaker Cables

We have a general recommendation about interconnects, which is that they should cost less than the amplifier. We have tried a lot of products and most of them work well, but as a practical matter we cannot make blanket recommendations.

The amplifier is not sensitive to source interconnects. It is also not sensitive to radio frequency pickup, which allows some flexibility in choosing source interconnects without shields.

We prefer speaker cables that are thick and short. Silver and copper are the preferred metals. If you find any cable made of gold, please send me a couple hundred feet.

Fortunately the amplifier is not sensitive to the capacitive/inductive character of some of the specialty speaker cables, so feel free to experiment.

We have found that about 90 per cent of bad sounding cables are really bad connections, and we recommend that special attention be paid to cleanliness of contact surfaces and tight fit.

Speaker cables should be firmly tightened down at the speaker output terminals, but do not use a wrench. They will not withstand 100 foot-lbs of torque. Hand tightening without excessive force is plenty.

Fun Hardware Facts

The X150 has a power transformer rated at 800 watts, continuous duty. Under actual conditions in the amplifier, it will do about 1500 watts for short duration.

To avoid huge inrush of current during charge up, each of the transformer primary coils has its own inrush suppressor, which keeps the inrush down to 100 amps or so.

The X150 has 4 computer grade (the old large style computer capacitor cans, not the new dinky ones) capacitor cans at 31,000 uF and 50 volts each. These are used to create the unregulated output stage rails at plus and minus 32 volts at 20 amps.

In the X150, additional voltage for the front end is derived from separate windings on the main transformers. This extra front end supply lowers the distortion and noise of the system, and allows the front end to swing the output stage rail-to-rail with losses on the order of only a volt or so, extracting every last possible watt.

The circuit of the amplifier is completely DC, with no capacitors in the signal path. There are also no slew rate limiting capacitors in the circuit.

All the transistors in the product are power Mosfets, actually Hexfets from International Rectifier and Harris. These are hyper-matched parts, with gate voltages matched to 0.5% and all devices taken from the same lot codes (made on the same wafer). The speed and noise critical gain devices in the front end, (that is to say the actual balanced pair of transistors) are ultra low noise and distortion matched JFETs having a low (.02 S) transconductance figure. The JFETs are made on the same substrate for prefect matching.

The X150 has 40 output Mosfet power transistors in TO-3 plastic packages, again matched to 0.5% and drawn from the same lot codes for each type. The output stages can sustain transients of about 5,000 watts, but are not allowed to dissipate more than 1000 watts for any instant, even into a dead short.

So how long will this hardware last? It is my experience that, barring abuse or the odd failure of a component, the first things to go will be the power supply capacitors, and from experience, they will last 15 to 20 years. Fortunately they die gracefully and are easily replaced. After that, the longevity will depend on the number of operating thermal cycles, but I can say that I have had amplifiers operating in the field in excess of 20 years with no particular mortality except capacitors. The answer is, I don't have good information beyond that. More to the point, I would suggest that you not worry about it. This is a conservatively built industrial design, not a tweaky tube circuit run on the brink. If it breaks, we will simply get it fixed, so sleep well.

Warranty Information

This product is warranted for parts and labor from the date we ship it. Check with the factory authorized distributor in the country you are purchasing this product for specific warranty information.

Distributors are only required to offer warranty service on Pass products that they have sold. They are not obligated to offer warranty repair for products purchased from other distributors. Products purchased from other distributors should be returned to the country of purchase for warranty repair.

Supersymmetry: What it is, Where it came from, How it works, Why bother

(theory and philosophy by Pass that you can ignore)

Supersymmetry is the name given to a new type of amplifying circuit, which operates quite differently than the designs presently appearing in literature and the marketplace. I have been designing new amplifiers all my adult life, and patented several of them, but I regard this particular idea as the most interesting and profound. The name “supersymmetry” describes the circuit but is also the name of a theory from the field of particle physics that considers the ultimate nature of matter and forces in terms of symmetries.

A little history of the development of this idea might help to illuminate the concept. As far as I’m concerned, the progress in amplifier design has to do with making amplifiers better while making them simpler.

Numerous amplifier design techniques have been offered during this century, but the ideas that have stood the test of time have delivered much better performance in simple ways. Two of the best ideas have been negative feedback and push-pull operation. Negative feedback is a simple technique, which requires only a couple more parts, arranged simply, but it achieves dramatic improvements in performance. Similarly for push-pull operation, a couple more parts delivers incredibly greater efficiency and improved distortion at high power levels.

The concepts of negative feedback and push-pull operation in amplifiers were old enough in 1970 that some of their limitations were becoming apparent, at least with regard to audio amplifiers. In the hands of mediocre designers, feedback was often overused to cover up design sins elsewhere in the circuit, with the result that the amplifier did not sound very good, in spite of good distortion measurements. Push-pull circuits, while allowing high efficiency and cheap manufacture, did not improve the character of the sound at lower levels, where we do most of our listening, a deficiency which designers often use feedback to cover up.

It appears that the human sense of hearing is more subtle in some ways than distortion measurement apparatus, and many audiophiles were dissatisfied with the results of the new breed of solid state amplifiers appearing in the 60’s and 70’s. These designs used lots of feedback to clean up their efficient push-pull circuits.

The innovative designers were beginning to consider some variations of and alternatives to these tried and mostly true approaches, and some new designs appeared.

Once it was recognized that excessive use of negative feedback was creating problems with the sound, several designers addressed the problem by simply reducing the amount of feedback and regaining the performance by paying more attention to the character of the amplifying circuit itself. Feedback stopped being a “something for nothing” idea, and became more like a credit card, which is OK to use as long as you can afford to pay when the statement arrives. In this case, the ability to pay involves the intrinsic quality of the amplifier circuit. The paradox is that feedback is best applied around circuits that need it the least.

One of the alternatives is the use of “no feedback”, or more accurately what is known as only local feedback. I say this because purists might argue that local feedback is still feedback. In point of fact, there is always some amount of feedback locally around any gain device by the nature of the device. So I will state here and now that I consider “no feedback” to be where feedback does not extend further than a single gain device or stage, so that circuits having “local feedback” are still considered “no feedback”. Anybody disagreeing with this should send me a diagram of a “true no feedback” circuit, and I will try to point out the hidden feedback.

On the push-pull front, a major improvement was offered by Class A operation, not a new concept, which delivered significantly better performance by sending a much larger amount of current idling through the gain devices. This lowered the distortion of the gain devices dramatically, but at the cost of high heat dissipation. Operating an amplifier in Class A mode was, and remains, an expensive proposition compared to conventional designs, not necessarily so much in wasted energy, but in the cost of the heavier hardware needed to deliver and dissipate the additional heat.

One of the important potential advantages of Class A operation is the possibility for simplified circuitry requiring little or no feedback because of the much more linear performance of gain devices biased to a high current. By the mid 1970’s the marketplace began to see high end solid state amplifiers offering varying degrees of Class A operation in their output devices, although as far as I can tell, at the time none of them took advantage of Class A operation to create simpler circuits with less feedback. Mine didn’t, in any case.

Also about this time Matti Ottala introduced the concept of Transient Intermodulation Distortion (TIM), in which the overuse of feedback, coupled with slow amplifier circuits was identified as the major culprit in bad sounding amplifiers. It was all the rage for a while, but is no longer touted with such enthusiasm. The solution to TIM is low amounts of feedback coupled with fast amplification (high slew rate).

In retrospect, the idea was at least half right, but I believe not completely for the following reasons: First, it presumed that there was really fast signal in music. Research conducted independently by Peter Walker and myself showed conclusively that real music contained very little signal with appreciable slew rate, therefore slew rate limiting on the order proposed by Ottala was pretty unlikely. Further, all those good sounding tube amplifiers had terrible slew rate figures.

However, while slew rate limitations of an amplifier might not be the cause of bad sound, it did correlate to sonic performance in the following manner. It turns out that there are two ways to make faster amplifiers, the first way being to make the circuit more complex. The second is to make it simpler. Video amplifiers, which must be very fast, are very simple. Tube circuits tend to be very simple also.

Rushing to market in the 70’s with their low TIM distortion designs, companies employed either simpler or more complex circuits to achieve high slew rates. The amplifiers that had simpler circuits with fewer parts tended to sound better than the amplifiers with complex circuits and a lot of parts. They also cost less and broke down less often, not an unimportant benefit.

Thus was a great principle of audio amplifier design reborn. Like the principle of Occam’s razor, if you have two amplifiers with similar performance numbers, the simpler one will sound better. Often the simpler one will sound better even if its measured distortion is higher.

Looking back on my amplifiers, I see a steady progression of simpler and simpler. Like the products of other young designers, my first commercial product had everything but the kitchen sink in it. Now I strive to be like Picasso, who could draw a woman with a single pencil stroke and create a masterpiece.

Supersymmetry is not a single pencil stroke, but I am making progress. Its origin goes back to the late 1970's when I was examining the virtues and faults of so-called "error correcting amplifiers", an alternative to conventional feedback. In this approach, two amplifiers, a big one and a small one work together. The big one handles the big job of delivering power to the loudspeaker, and the little one sweeps up after it. The big amplifier, not having to worry about the details, delivers power like a supertanker crossing the ocean. The little amplifier is like a tugboat, which nudges it precisely into port. The concept is a good one, much of the credit going to Peter Walker, but it is a bit more complicated than we might want.

Thoughts about this approach on my part led to the Stasis amplifier, a simpler, if cruder, circuit in which the ocean liner could just about make it into port by itself with only minor damage, and the tugboat was capable of crossing the Atlantic, if not the Pacific. Threshold and Nakamichi have sold lots of these amplifiers for the last 19 years or so, and so it was pretty successful.

Yet it was always in the back of my head that there must be a better solution to the no-feedback performance problem, something even simpler and more elegant. I felt that symmetry and anti-symmetry in the character of signals and circuits held the key, but not having any idea how, I amused myself for the next 15 years by drawing topologies which might do something in this vein. One day in 1993 I drew a picture connecting two transistors, each with local feedback, and the concept fell into place. The following year I received a patent on the design.

The concept is actually very simple. Conventional feedback, local or not, is used to make the output of the circuit look like the input. In this circuit, feedback was not used to make the input look like the output in the conventional sense. Instead it works to make two halves of an already symmetric balanced circuit behave identically with respect to distortion and noise, dramatically lowering the differential distortion and noise but not the distortion and noise of each half of the circuit considered by itself.

If you build such a symmetric (balanced) circuit, you get much of this effect already. If you drive a matched differential pair of transistors without feedback with a balanced signal, you will see a balanced output whose distortion and noise is typically 1/10 that of either device alone, purely out of cancellation. With supersymmetry, the same differential pair's characteristic can be made so identical that the differential output will have only 1/100 the distortion and noise of either device alone.

Supersymmetry does not reduce the distortion and noise present in either half of the output of the balanced circuit. Comparing the distortion curves before and after the application of supersymmetry, we see essentially no difference in either half of the balanced pair considered alone. It is the balanced differential characteristic that improves dramatically, and that leads to one singular requirement of supersymmetric operation; it must be driven by a balanced input signal and it only produces a balanced output signal. You could drive it with a single-ended input and hook a speaker up to only one output and ground, but there would be no point to it at all.

Supersymmetry operates to make the two halves of the balanced circuit behave absolutely identically. Constructing the two halves of the circuit with identical topologies and matching the components precisely achieves a 20 dB or so reduction in distortion and noise, and local feedback with a Supersymmetric connection another 20 dB or so. This is easily accomplished with only one gain stage instead of the multiple stages required by conventional design, and so it results in only one "pole" of high frequency characteristic, and is unconditionally stable without compensation. In fact, if you build a supersymmetric circuit with multiple gain stages, it does not work as well.

In 1993 I attempted to build the first power amplifier using this principle, but it was not successful. Ironically, the supersymmetric concept not only allows for very simple gain circuits, but it requires them for good performance. My first efforts did not use a simple enough approach, although I didn't realize it at the time. A more modest version of the circuit found its way into a preamplifier, the Aleph P. Ultimately the power amplifier was set aside, as we were very busy building Aleph single-ended Class A amplifiers.

In 1997 I decided to build a state-of-the-art *very high power* amplifier, the X1000, a project not particularly appropriate for the single-ended Class A approach (believe me, you don't want to own an amplifier idling at 3000 watts per channel). So I pulled out the files on patent # 5,376,899 and took another look. Extensive testing of potential circuits revealed that the best topology for the front end of the amplifier is what we refer to as "balanced single-ended", a phrase I use to refer to differential use of two single-ended Class A gain devices. The classic differential pair of transistors (or tubes, for that matter) is just such a topology.

"Balanced single-ended" is an oxymoron in the sense that most single-ended enthusiasts believe that the most desirable characteristic of single-ended circuits is their generation of even-order distortion components by virtue of their asymmetry. Purists will point out that a balanced version of a single-ended circuit will experience cancellation of noise and even-order components. Just so. Interestingly, the single-ended nature of each half of the balanced circuit doesn't give rise to much in the way of odd-order distortion, and when the even-order components and noise are cancelled there isn't much distortion and noise left. In any case, "Balanced single-ended" is a phrase that accurately describes the circuit.

For the amplifier's front end, a balanced single-ended gain stage was developed which used just a differential pair of Mosfet gain devices. These were biased by constant current sources and cascoded for maximum performance and given local feedback and a Supersymmetric connection. After years of trying alternative arrangements, it ended up virtually identical to the schematic on the cover page of the patent, which is reproduced later in this manual.

The front end, which develops all the voltage gain for the amplifier, then presents this voltage to a large bank of follower Mosfet power transistors. Originally it was assumed that we would have to enclose this output stage in a feedback loop to get the performance we wanted, but ultimately we found that we could operate it without feedback as long as we put a healthy bias current through it. For these amplifiers this is about 600 watts worth. This is not pure Class A operation in the context of 1000 watts output, but it has proven to be the appropriate amount.

The result is a series of amplifiers using the supersymmetric topology delivering up to 1000 watts per channel into 8 ohms with good distortion and noise figures. If you are a little less fussy about distortion, you will get twice that into 4 ohms. This is accomplished with only two gain stages and no feedback.

People inevitably will ask how this relates to bridged amplifiers in general, and the balanced amplifier offerings of other companies. It is similar in that both terminals of the output to the speaker are "live"; neither of them is grounded.

The supersymmetric amplifier is a special subset of balanced amplifiers, unique and covered by U.S. patent. Supersymmetry is an approach that truly takes advantage of balanced operation like no other and requires a balanced input to retain the precisely matched behavior.

Supersymmetry is ideally used to obtain high quality performance from very simple circuit topologies, avoiding the high order distortion character and feedback instabilities of complex circuits. A single gain stage amplifier using this approach can perform as well as a two gain stage design, and a two gain stage version of this topology can outperform the four or five stages of a conventional amplifier.

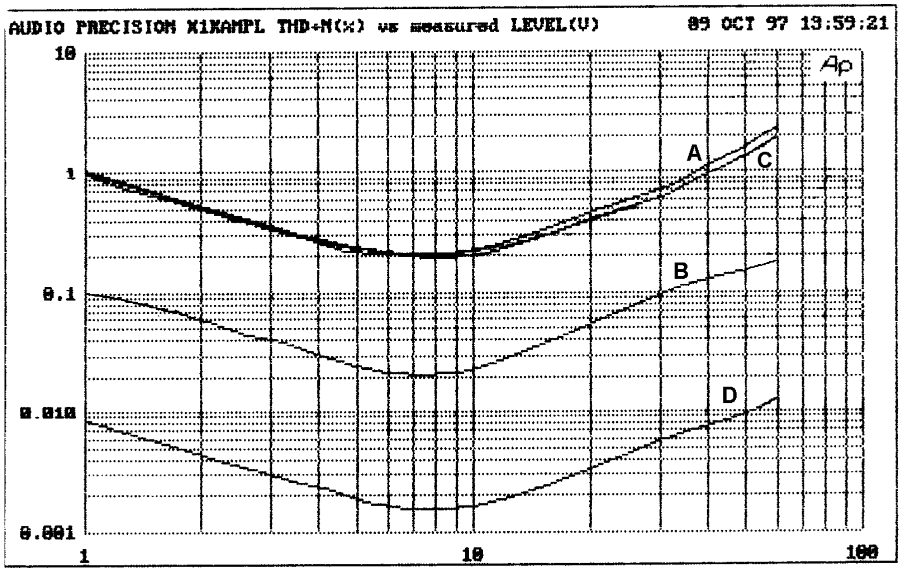
Here is some more explanation of the details of its operation:

The supersymmetry topology does not use operational amplifiers as building blocks, nor can it be represented with operational amplifiers. It has two negative inputs and two positive outputs and consists of two matched gain blocks coupled at one central point where the voltage is ideally zero. The topology is unique in that at this point, the distortion contributed by each half appears out of phase with the signal, and we use this to reinforce the desired signal and cancel noise and distortion. This occurs mutually between the two halves of the circuit, and the result is signal symmetry with respect to both the voltage and current axis, and anti-symmetry for distortion and noise. This means that the distortion and noise of each half appears identically and cancels.

The diagram on the patent cover sheet shows an example of this topology. Each of the two input devices 20 and 21 are driven by an input signal, and their outputs run through a folded cascode formed by devices 30 and 31 to develop voltages across current sources 34 and 35. The sources 20 and 21 are coupled through resistor 40 which is the sole connection between the two halves and which also sets the gain of the circuit.

The gates of the input devices 20, 21 are virtual grounds, and ideally would be at absolutely zero voltage. However, as the gain stage is not perfect, finite distortion and noise voltages appear at these points. These appear at the other side through resistor 40, in phase at the output of the other half of the system, where they match the distortion and noise of the first half.

By actual measurement, this circuit does essentially nothing to reduce the distortion and noise in each half. Distortion curves before and after supersymmetry is applied are nearly identical. The distortion curves of the circuit from the patent cover sheet show: (a) the intrinsic distortion of each half of the example circuit, (b) the distortion of the differential output lowered due to the intrinsic matching between the circuits, (c) the distortion of each half with supersymmetry, and (d) the differential distortion with supersymmetry.



On this curve (B) we can clearly see that intrinsic symmetry due to the matching of the two halves reduces the distortion by a factor of 10. Supersymmetry (D) creates a more perfect match, and results in an additional reduction by a factor of 10. However there is essentially no difference in the distortion figures at the output (C) of each half of the circuit considered alone. Supersymmetry does not work by reducing the distortion per se, rather it works to precisely match the two halves of the circuit and lets the balanced output ignore the unwanted components. As long as the two halves are matched, this performance tends to be frequency independent, and does not deteriorate over the audio band. With mid-level distortion figures on the order of .002%, this is very high performance for a single balanced gain stage.

The following pages include, a typical distortion curve of the amplifier, a list of specifications for the amplifier, and where to reach us.

If you have questions, or we can help you, please feel free to contact us. Again, you can easily contact Nelson Pass by email addressed to **nelson@passlabs.com**, and you are welcome to ask questions or offer comments. Other personnel are available through the website **www.passlabs.com**, as are copies of patents, DIY articles, and product information.

X150 SPECIFICATIONS

All figures obtained after 1 hour warmup, with regulated 120 VAC power line. See manual notes about AC power line regulation.

Gain	30 dB
Freq. Response	0 dB at DC, -3 dB at 100 kHz
Power Output	150 watts maximum @ 1% THD, 1 kHz, 8 ohms
Maximum Output Voltage	plus, minus 50 volts
Maximum Output Current	plus, minus 20 amps
Input Impedance	22 kohm balanced
Damping factor	250 ref 8 ohms nominal
Slew rate	approx. plus, minus 50 V/uS
Output Noise	approx. 300 uV unweighted 20-20 kHz
Random noise floor	approximately 2 uV
Dynamic range Balanced	145 dB (random noise floor to peak output)
Balanced CMRR	approx. -85 dB @ 1 kHz
DC offset	< 100 mv
Power Consumption	200 watts idle, 600 watts maximum
Dimensions	19 " W x 20" D x 7" H
Weight	70 lbs.

PASS

Pass Labs
PO Box 12878
Reno NV 89510-2878

www.passlabs.com
tel: (530) 367 3690
fax: (530) 367 2193