Pass Laboratories

X-2 Owner's Manual

Introduction

The X-2 is a balanced single ended Class A audio preamplifier combining new design thought applied to a traditional topology and the experience of over twenty five years of amplifier design.

This product design flows from a commitment to create the best sounding product: a simple circuit with the most natural characteristic.

This is the second version of the X-2. The first version used differential Mosfet circuits operated without feedback. This newer version uses the patented Super-Symmetric topology employed in the new generation of X power amplifiers from Pass Labs. The newer circuitry has much lower noise, lower distortion, and swings about three times the voltage.

The circuits integrate low noise JFET input devices coupled with Mosfet output devices in a balanced Class A circuit in order to deliver the most natural sound possible. The Super-Symmetric topology exploits inherent distortion cancellation to get extremely low distortion from very simple arrangements of parts and very little feedback. The combination of simplicity and excellent measured performance makes it easy for us to get very good sound without the annoying artifacts of complex transistorized designs.

The X-2 is very much a minimalist product. We look to pass the signal through as few components as possible and still achieve superior performance. The decision as to where to draw the line in performance versus parts is ultimately the result of extended listening time with the product, and if it doesn't give us the sound we're looking for, no measurement has any value. The result is that we can make it more simple or complex, but this is the sweet spot as determined by living with the product.

Thank you for purchasing this product. We at Pass Labs hope that it provides you with the sonic experience that you are seeking, and that your enjoyment of it is equal to the effort we put into its design and construction.

Nelson Pass

Setup

The preamplifier has four sets of input connections, two sets of output connections, an input selector and a level control.

The X-2 also has an IEC, AC line power connection and detachable supplied power cord. The amplifier's voltage and current rating are indicated on the rear. It will be either 240 volts, 220 Volts, 120 volts, or 100 volts. A .5 amp 3AG slow blow fuse is provided with 100-120 volt units, and a .25 amp slow blow fuse is provided with 220-240 volt units. The frequency rating of the power supply is 50 to 60 Hz. The preamplifier typically draws 25 watts during operation. Please verify that the indicated required voltage is consistant with the supplied power in your location

We have provided a standard AC power cord which fits into the IEC line receptacle at the rear. The preamplifier is equipped for operation with an earth ground provided by the users AC outlet. Do not defeat this ground. The chassis and circuit ground of the preamplifier are connected to earth through a power thermistor, which gives a ground connection but helps avoid ground loops.

The four input connections on the rear are pairs of XLR and RCA connectors with right and left channels indicated. If your signal source is balanced, you may use the XLR input connectors. On these connectors, pin 1 is ground, pin 2 is the positive signal input, and pin 3 is the negative signal input.

If your signal source is unbalanced, input will occur through the RCA input connector, which is in parallel with the XLR connections 1 and 2. For operation with unbalanced inputs, a shorting plug is provided between pins 1 and 3, shorting the negative input to ground, and providing optimal performance.

For best performance with single-ended input 1, be certain to use this jumper.

The unbalanced input impedance of the preamplifier is a nominal 10 kOhms. In balanced mode, the input impedance is higher, with a differential impedance of 20 kOhms.

Next to the inputs on the rear panel, the preamplifier offers tape outputs through RCA connectors. This output is a direct connection to inputs 1, 2, and 3 when they are selected from the front panel. Input 4 is deliberately not available through the tape output connection. Input 4 is designated for use with a tape recorder if you have one, and we have arranged that it will not place its output on the tape out, which will prevent you from accidentally creating a feedback connection with your tape machine.

At the left hand side (viewed from the rear) of the rear panel, two male XLR connectors and two RCA connectors are used for the main output. On the XLR, pin 1 is ground, pin 2 is positive, and pin 3 is negative. The RCA connector's ground is in parallel with pin 1 and the RCA hot is attached to pin 2. You may use either of these connectors for balanced or unbalanced operation.

Product Philosophy

When I started designing amplifiers 25 years ago, solid state amplifiers had just achieved a firm grasp on the market. Power and harmonic distortion numbers were king, and the largest audio magazine said that amplifiers with the same specs sounded the same.

We have heard Triodes, Pentodes, Bipolar, VFET, Mosfet, TFET valves, IGBT, Hybrids, THD distortion, IM distortion, TIM distortion, phase distortion, quantization, feedback, nested feedback, no feedback, feed forward, Stasis, harmonic time alignment, high slew, Class AB, Class A, Pure Class A, Class AA, Class A/AB, Class D, Class H, Constant bias, dynamic bias, optical bias, Real Life Bias, Sustained Plateau Bias, big supplies, smart supplies, regulated supplies, separate supplies, switching supplies, dynamic headroom, high current, balanced inputs and balanced outputs.

Apart from digitally recorded source material, things have not changed very much in twenty five years. Solid state amplifiers still dominate the market, the largest audio magazine still doesn't hear the difference, and many audiophiles are still hanging on to their tubes. Leaving aside the examples of marketing hype, we have a large number of attempts to improve the sound of amplifiers, each attempting to address a hypothesized flaw in the performance. Audiophiles have voted on the various designs with their pocketbooks, and products go down in history as classics or are forgotten. The used market speaks eloquently: Marantz 9's command a high price, while Dyna 120's are largely unwanted.

There has been a failure in the attempt to use specifications to characterize the subtleties of sonic performance. Amplifiers with similar measurements are not equal, and products with higher power, wider bandwidth, and lower distortion do not necessarily sound better. Historically, that amplifier offering the most power, or the lowest IM distortion, or the lowest THD, or the highest slew rate, or the lowest noise, has not become a classic or even been more than a modest success.

For a long time there has been faith in the technical community that eventually some objective analysis would reconcile critical listener's subjective experience with laboratory measurement. Perhaps this will occur, but in the meantime, audiophiles largely reject bench specifications as an indicator of audio quality. This is appropriate. Appreciation of audio is a completely subjective human experience. We should no more let numbers define audio quality than we would let chemical analysis be the arbiter of fine wines. Measurements can provide a measure of insight, but are no substitute for human judgment.

As in art, classic audio components are the results of individual efforts and reflect a coherent underlying philosophy. They make a subjective and an objective statement of quality which is meant to be appreciated. It is essential that the circuitry of an audio component reflects a philosophy which address the subjective nature of its performance first and foremost.

Lacking an ability to completely characterize performance in an objective manner, we should take a step back from the resulting waveform and take into account the process by which it has been achieved. The history of what has been done to the music is important and must be considered a part of the result. Everything that has been done to the signal is embedded in it, however subtly.

Experience correlating what sounds good to knowledge of component design yields some general guidelines as to what will sound good and what will not:

1) Simplicity and a minimum number of components is a key element, and is well reflected in the quality of tube designs. The fewer pieces in series with the signal path, the better. This often true even if adding just one more gain stage will improve the measured specs.

2) The characteristic of gain devices and their specific use is important. Individual variations in performance between like devices is important, as are differences in topological usage. All signal bearing devices contribute to the degradation, but there are some different characteristics are worth attention. Low order nonlinearities are largely additive in quality, bringing false warmth and coloration, while abrupt high order nonlinearities are additive and subtractive, adding harshness while losing information.

3) Maximum intrinsic linearity is desired. This is the performance of the gain stages before feedback is applied. Experience suggests that feedback is a subtractive process; it removes information from the signal. In many older designs, poor intrinsic linearity has been corrected out by large application of feedback, resulting in loss of warmth, space, and detail.

High idle current, or bias, is very desirable as a means of maximizing linearity, and gives an effect which is not only easily measured, but easily demonstrated: Take a Class A or other high bias amplifier and compare the sound with full bias and with bias reduced. (Bias adjustment is easily accomplished, as virtually every amplifier has a bias adjustment pot, but it should be done very carefully). As an experiment it has the virtue of only changing the bias and the expectations of the experimenter.

As the bias is reduced the perception of stage depth and ambiance will generally decrease. This perception of depth is influenced by the raw quantity of bias current.

If you continue to increase the bias current far beyond the operating point, it appears that improvements are made with bias currents which are much greater than the signal level. Typically the levels involved in most critical listening are only a few watts, but an amplifier biased for ten times that amount will generally sound better than one biased for the few watts.

For this reason, designs which operate in what has been referred to as "pure" Class A are preferred because their bias currents are much larger than the signal most of the time.

As mentioned, preamp gain stages and the front ends of power amplifiers are routinely "pure" Class A, and because the signal levels are at small fractions of a watt, the efficiency of the circuit is not important.

We recommend the use of the balanced interconnect where possible. It will retain the character of the musical signal, but offers less environmental noise without compromising the sound.

The internal power supply for the X-2 consists of a toroidal power transformer delivering an unregulated 84 volts which is actively regulated before passive filtering. The power supply noise reaching the circuit is on the order of a microvolt, and is differentially rejected at the output in a balanced system. The relays and control systems are regulated independently.

The output of the preamplifier is guarded by muting relays which delay connection during turn-on and shut off the output when insufficient power supply is available to maintain regulation. The preamplifier is designed to run constantly, and will exhibit optimum performance within an hour of turn-on.

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

(more theory and philosophy by Pass)

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.

The X-2 is warranted by Pass Laboratories to meet performance specifications for 3 years from date of manufacture. During that time, Pass Laboratories will provide free labor and parts at the manufacturing site. The warranty does not include damage due to misuse or abuse or modified products and also does not include consequential damage.

SPECIFICATIONS

Gain	16 dB bal out 10 dB unbal out
Freq. Response	-3 dB @ 5 Hz, -3 dB at > 100 kHz
Distortion	< .1 % THD typically .01% @ 2 volts @ 1KHZ
Maximum Output	30 volts rms. bal out 15 volts rms. unbal out
Output Impedance	220 ohms unbalanced 440 ohms balanced
Input Impedance	10 kOhm unbalanced 44kOhm Bal
CMRR	typically -60dB 20-20KHz
Crosstalk	typicall -80 dB, 20-20 KHz
Output Noise	Random Noise floor < -120 dBV,20-20 KHz
Power Consumption	25 watts
Dimensions	17 " W x 11.5 " D x 3.5" H
Shipping Weight	22 lb.

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