

Pass Laboratories

Aleph P Owner's Manual

Introduction

The Aleph P is a single ended Class A audio preamplifier, the first produced by Pass Laboratories. It combines completely new design thought applied to a traditional topology and the experience of twenty five years of amplifier design. This manual applies to the second version of the Aleph P, which incorporated circuit improvements and remote control operation.

This preamplifier flows from a commitment to create the best sounding product: a simple circuit with the most natural characteristic. The Aleph P integrates power Mosfet devices and single ended Class A operation in a simple topology in order to deliver the most natural sound possible.

The circuitry of the Aleph P breaks new ground in simplicity and performance. Each gain stage consists of a single Mosfet power transistor operated "common source" mode. One such circuit amplifies each signal polarity of a balanced input signal, and their operation is inter-coupled to allow optimal operation with both balanced and unbalanced input signals while still providing balanced and unbalanced outputs. It retains the important simple single-ended characteristic while operating either single-ended or balanced.

It is unique in providing superior flexibility and performance with balanced and unbalanced inputs and outputs, converting one to the other as desired without switches or adapters.

The Aleph P absolutely minimizes the number of components in the signal path, and yet retains exemplary objective performance specifications. More importantly, it pushes the edge of the art in exploring how much subjective quality is obtainable with a new but very elementary gain stage.

A very few people are involved in the production of this product. I supervise all phases of the construction, and I test and listen to each preamplifier myself. If you have questions, comments, or problems, please feel free to contact me directly.

Thank you for purchasing this preamplifier. It is my sincere hope that you will enjoy its sound as much as I do.

Nelson Pass

Serial #

Date:

Next page: Distortion curve of your particular preamplifier at 1 kHz from .1 to 10 volts

Setup

The preamplifier has four sets of input connections, two sets of output connections, an input selector and three level controls. Optionally, it also has a remote control.

It also has an AC line power connection. The amplifier's voltage and current rating are indicated on the rear. It will be either 240 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.

We have provided a standard AC power cord which fits into the line receptacle at the rear. The preamplifier is equipped for operation with an earth ground provided by the AC outlet. Do not defeat this ground. The chassis and circuit ground of the preamplifier is 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 grounded, 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.

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 both XLR and 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.

The circuitry of the preamplifier is such that it will drive any impedance without distortion, however as the load impedance goes down, you will experience a reduction in gain. The preamplifier will act as a maximum plus and minus 20 ma current source into a dead short.

The operation of the front panel input selector is straight forward. The selected input is indicated by a blue LED above the knob, and the selection can be moved by rotating the knob. The knob is attached to an optical encoder indicating the direction of rotation to the microcontroller which drives the input relay switches.

The Left and Right input level controls and the Master level control are designed not only to provide level adjustment and balance adjustment, but also to optimize the performance of the circuit against various input levels.

The input of the gain stage is directly coupled to the input selected, without an intervening volume or balance control as seen in other preamplifiers. This provides the highest performance from the Mosfet gain devices.

Because there is a wide variety of possible source levels, the intrinsic gain of the circuit should be adjusted to provide optimal performance. This is something you will not see on other preamplifiers. The intrinsic gain of the circuit itself is adjusted by the Left and Right level controls. We set the gain lower for sources which have high levels, and higher for sources with low levels, minimizing the distortion and noise for each.

We have set the maximum input so that the circuit will remain linear with balanced input peaks up to about 18 volts. If you are concerned that your source will approach or exceed this level, you may use the internal attenuator switch to reduce the input by 12 dB.

The switch default position is 1,2,7,8 = ON and 3,4,5,6 = OFF. For a 12 dB input attenuation, 1,2,7,8 = OFF and 3,4,5,6 = ON.

This switch can also be used to effectively lower the gain of the preamp by 12 dB. It does not otherwise affect the performance of the preamp.

At the output of the active gain circuits is the master level control. This is a low impedance discrete level control which sits between the preamp active circuitry and the input to your power amplifier. It attenuates the output of the preamp in the same manner as you would experience with a "passive preamp" except that it has a precision 256 position 4 pole attenuator for both balanced channels matched to .1%. It also has a low impedance, which assures that the typical unbalanced output impedance is between 100 and 1000 ohms.

There are good reasons for having such an arrangement:

The gain stage operates at a constant level regardless of the setting of the master level control, and so the sound of the this circuit will not alter at various level settings.

The preamplifier will drive a low impedance load without alteration in the signal, in fact it can be used as a current source driving a 0 impedance mixer junction without any distortion.

Any noise characteristic of the preamp circuit is attenuated along with the signal, unlike circuits where the volume control is before the input.

If you have a concern that the output impedance may be too high to properly drive the capacitance of a cable, keep in mind that that 1000 ohms still will allow bandwidth to 160 kHz into 1000 pF of capacitance. The maximum output impedance only occurs at the volume control maximum, and quickly goes to a low figure below the maximum level position. In typical operation, the output impedance will be between 100 and 300 ohms.

The sixteen relays which operate the level control allow greater than 48 dB range and are driven by a microcontroller which reads the optical encoder which serves as a front panel volume control. In this manner tracking of the volume of the two balanced channels is possible with an accuracy unavailable on any ordinary volume control, assuring precise level steps and high common mode rejection in balanced circuits.

The relays may lack the high speed switching characteristic of the solid-state state switches used by other preamplifiers, but they also have none semiconductor distortions and noise which accompany analog gates and voltage controlled amplifiers. If it bothers you to hear the click of the relays as the level is adjusted, just consider the artifacts you won't hear after the level is adjusted.

Adjustment to optimize the level controls against an input level is easy. Select the most important input. Set the Right and Left level controls at minimum gain (counterclockwise) and adjust the Master level control to as high as you will want to listen. If you don't have enough gain, then increase the levels on the Left and Right level controls until you do.

For maximum performance, the idea is to keep the Master control as clockwise as possible, and the Left and Right level controls as counterclockwise as possible.

In other words, go ahead and turn it up.

This is not a big deal to worry about, but the ability to get the most performance is available in this preamp in a manner not seen elsewhere, and you will want to take advantage of it.

Using the remote control is very simple. Point it at the preamplifier and press one of the four buttons on the remote.

The side to side buttons will change the input selection.

The top to bottom buttons will change the master level.

The end.

Product Philosophy and Design Theory

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 single ended "pure" Class A, and because the signal levels are at small fractions of a watt, the efficiency of the circuit is not important.

Problems with push-pull amplifier designs associated with crossover distortion have been discussed elsewhere at length, and one of the primary results is non-monotonicity. Class B and many AB designs have distortion products that dramatically increase with decreasing signal. This is reduced greatly by Class A mode, but crossover distortion remains as a lower order discontinuity in the transfer curve.

A very important consideration in attempting to create an amplifier with a natural characteristic is the selection of the gain devices. A single ended Class A topology is appropriate, and we want a characteristic where the positive amplitude is very, very slightly greater than the negative. For a current gain device, that would mean gain that smoothly increases with current, and for a tube or field effect device a transconductance that smoothly increases with current.

Triodes and Mosfets share a useful characteristic: their transconductance tends to increase with current. Bipolar power devices have a slight gain increase until they hit about an amp or so, and then they decline at higher currents. In general the use of bipolar in a single ended Class A circuit is a poor fit.

Another performance advantage shared by Tubes and Fets is the high performance they deliver in simple Class A circuits. Bipolar designs on the market have between four and seven gain stages associated with the signal path, but with tubes and Mosfets good objective specifications are achievable with as few as one gain device in the signal path.

Regardless of the type of gain device, in systems where the utmost in natural reproduction is the goal, simple single ended Class A circuits are the topologies of choice.

We recommend the use of the balanced output mode whenever possible. It will retain the character of the input mode, but offers less distortion, less noise, more gain, and more voltage swing, all without compromising the sound in any way.

The Aleph P uses power Mosfets exclusively for its gain stage. These Mosfets were chosen because they have an excellent transfer curve for an asymmetric Class A design.

The gain Mosfets are rated at 40 watts each and peak currents in excess of 5 amps. Needless to say, they do not work very hard when sourcing 30 milliamps into a load. The use of such devices does provide very high transconductance and charge surface area over small gain devices, and this shows in the excellent linearity obtained with only one device operated without feedback.

Mosfets provide the widest bandwidth of solid state power devices, however they were not chosen for this reason. The design of the Aleph P does not seek to maximize the preamplifier bandwidth as such. The capacitances of the Mosfets provide a natural rolloff in conjunction with the resistive impedances found in the circuit, and the simplicity of the circuit allows for what is largely a single pole rolloff characteristic. Nevertheless, the bandwidth of the circuit will typically extend to about 200 kHz (-3dB).

There is no such thing as a slew rate for this circuit, as it will retain the linear RC characteristic for any input signal.

The common mode rejection of the preamp reflects the intrinsic common mode rejection of the topology, the matching of the gain devices, and the matching of the output attenuator channels. In this case we have been able to keep the total mismatch to about .1%, for a common mode rejection of approximately -60 dB.

The input system of the preamplifier will exhibit full common mode noise rejection with passive balanced sources, where the negative input is connected to ground at the source through the appropriate source impedance. This allows adaptation of unbalanced sources to balanced operation with passive cable connections in a manner that achieves the noise rejection of active balanced sources.

Load impedances do not make much difference to the character of the output. The intrinsic output of the impedance of the preamp is passive in nature, and no load will create nonlinearity.

The internal power supply for the Aleph P consists of a toroidal power transformer delivering an unregulated 85 volts which actively regulated before feeding passive filtering before powering the constant current sources which bias the gain stages. 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.

The chassis of the Aleph p is made entirely of machined aluminum. We mill and engrave the chassis components from solid aluminum material on computer controlled vertical milling machines. No sheet metal is employed.

The Aleph P 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	15 dB bal in / bal out	15 dB unbal in / bal out
	9 dB bal in / unbal out	9 dB unbal in / unbal out
Freq. Response	-3 dB @ 5 Hz -3 dB at > 100 kHz	
Distortion	< .1 % THD typically .01% @ 2 volts @ 1KHZ	
Maximum Output	15 volts rms. bal out	8 volts rms. unbal out
Output Impedance	0 - 1000 ohms unbalanced	
	0 - 2000 ohms balanced	
Input Impedance	10 kOhm unbalanced	20 kOhm balanced
CMRR	typical -80 Db, 20-20 KHz	
Crosstalk	typical -90 dB, 20-20 KHz	
Output Noise	Noise floor < -120 dBV, 20-20 KHz	
Power Consumption	25 watts	
Dimensions	19 " W x 11.5 " D x 4" H	
Shipping Weight	35 lb.	



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