Owner's Manual

Model 1100 MkII
Dual Class A Discrete Tube Microphone Preamplifier with Integrated 24-Bit 192kHz A/D Converter

Document No. 999-4200

Aphex Thermionics - Division of Aphex Systems Ltd.

Manual designed and written by Donn Werrbach 02/03. Creation tool: Adobe InDesign 1.5 for Windows. Printed in U.S.A.

A MESSAGE FROM THE PRESIDENT

Dear Aphex Customer,

We are very pleased to present the new Mark II version of our highly acclaimed Model 1100 Microphone Preamplifier. The Mark II has been augmented with a new, cooler, power supply and extended features that were requested by our customers. The unmatched sonic qualities and high technical performance that brought so much acclaim to the original Model 1100 have not been changed with the Mark II.

As with all our products we are extremely proud of the ingenuity of design and the manufacturing quality of the Model 1100 MkII. We love to hear from you, our customers, about your experiences with any of our products. Our customer support is unmatched in the industry, so please do not hesitate to contact us.

Sincerely,

Safety Declarations

WARNING SHOCK HAZARD DO NOT OPEN AVIS: RISQUE DE CHOC ELECTRIQUE NE PAS OUVIR	TO PREVENT FIRE OR SHOCK HAZARD DO NOT EXPOSE THIS DEVICE TO RAIN OR MOISTURE. DO NOT REMOVE THE COVER. NO USER SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED PERSONNEL ONLY. ATTENTION: AFIN DÉVITER DES CHOCS ÉLECTRIQUES. N'ENLEVEZ PAS LE COUVERCLE. IL N'Y A AUCUNE PIECÈS D'ENTRETIEN À L'INTÉRIEUR. RÉFÉREZ LES RÉPARACIONES À UNE PERSONNE QUALIFÉE.

CAUTION: For protection against electric shock, do not remove the cover. No user serviceable parts inside.

WARNING: This equipment has been tested and found to comply with the limits for a Class A digital device pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the operating guide, may cause interference to radio communications. Operation of this equipment in a residential area is likely to cause interference in which case the user will be required to correct the interference at his own expense.

The user is cautioned that changes and modifications made to the equipment without approval of the manufacturer could void the user’s authority to operate this equipment.

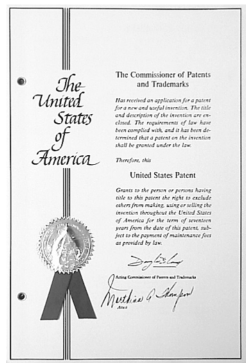
It is suggested that the user use only shielded and grounded cables to ensure compliance with FCC Rules.

C LISTED US
59887

Conforms to standards
UL60950 and EN60950.

Abstract From U.S. Patent Number 5,450,034

"A reflected plate amplifier (RPA) for use with electronic audio equipment. The RPA comprises an input circuit for receiving an input signal, a vacuum tube, a plate current reflector, and an output circuit for delivering an output signal. The vacuum tube has a control grid coupled to the input circuit for receiving the input signal, and a plate for delivering a plate current responsive to the input signal. The plate current reflector is a transistor having an input terminal coupled to the plate of the vacuum tube for receiving the plate current, and an output terminal for delivering a reflected current which is responsive to the plate current of the vacuum tube and therefore responsive to the input signal, while the plate current reflector holds the plate voltage of the vacuum tube substantially constant for a wide range of the plate current of the vacuum tube."


The United States of America

The Commissioner of Patents and Trademarks

United States Patent

That patent was issued to inventor Donn Werrbach and assigned to Aphex Systems Ltd. on September 12, 1995. A close examination of the patent reveals that it is a "method" patent and not just a patent on some particular circuit. That means it is possible to apply the patent teachings to a wide variety of applications and circuit variations with full patent protection. Other manufacturers will not be able to duplicate our RPA based products. Nevertheless, no one at first really knew just how significant that invention would prove to be.

The first product to apply an RPA circuit was our model 107 Thermionic Mic Preamplifier. The outstanding sonic quality of the unit has become world reknown and tens of thousands of units have been sold. Anything new and better is sure to stir up controversy. Not surprisingly, the success of our tube circuitry has alarmed other manufacturers of tube products. Some of them reacted with claims that our circuit is not "classic" and therefore not "real tube". The claim is ludicrous, but less than erudite reviewers hooked on conventionality have misreported it. For example, one particularly delusional reviewer rather condescendingly declared it nothing more than a "starved plate amplifier" (as if he ever saw one). Fortunately, these cases are greatly offset by the phenomenal word of mouth praise and promotion we have received for the RPA products from their actual owners over the years.

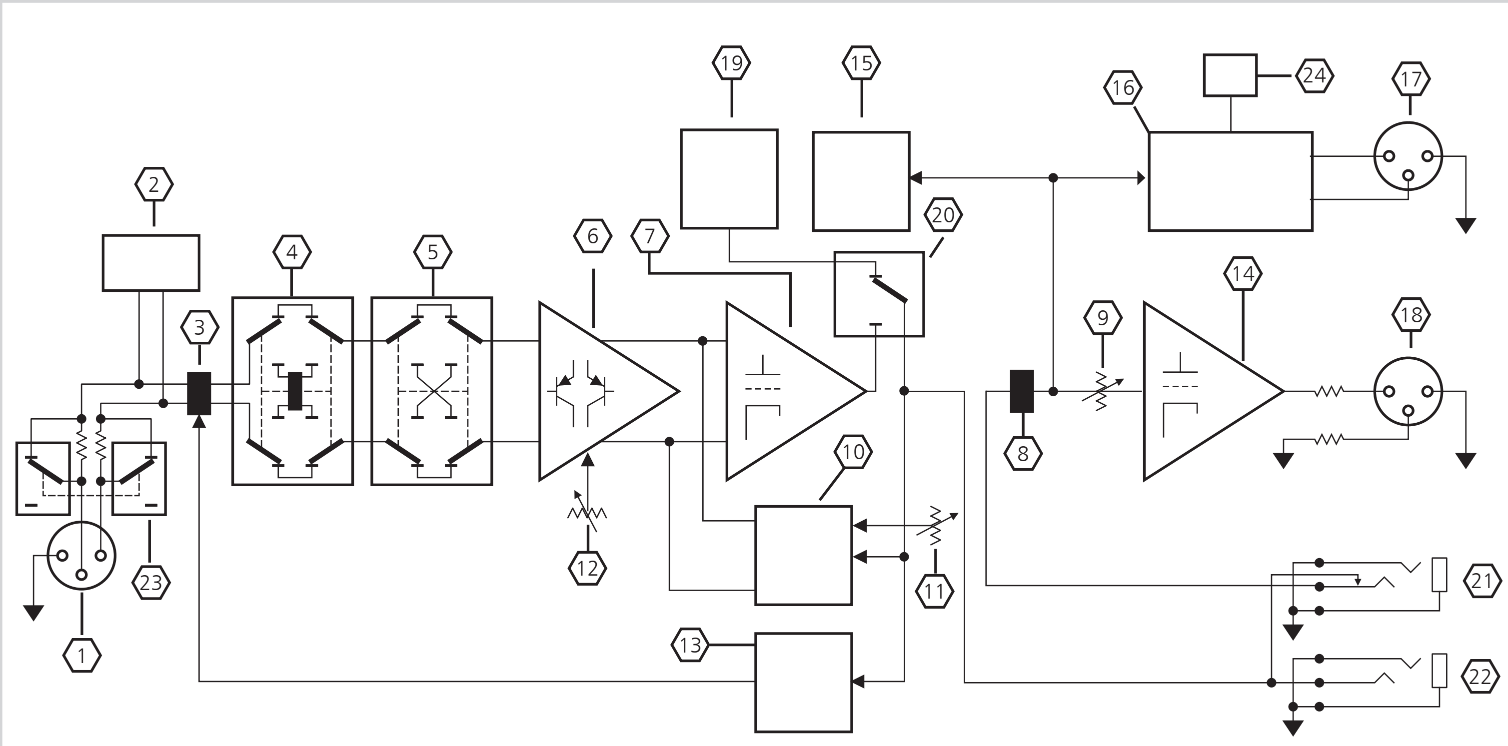
Since the original patent was filed, we have discovered new ways to create many useful circuits with RPA principles that would be impossible to realize with conventional tube circuitry. For example, the Model 1100 MkII Mic Preamp incorporates a very low noise second stage opamp using just one single triode RPA. Future products will feature many new tube circuits now in the laboratory. Aphex's unique servo balanced input and output stages translated into RPA implementations are examples.

It has become apparent that the principles of the RPA patent are the keys to gaining the unparalleled sonic benefits of tubes in an almost unlimited range of audio products. To fully distinguish our highest level of tube based audio products, we created the division of "Aphex Thermionics".

Watch for more exciting new products from AphexThermionics.


Dual Triode

***Thermionics** refers to the practical use of the "thermionic emission of electrons". Vacuum tubes use thermionic emission by heating the metal cathode to incandescence. Electrons are consequently emitted into space from the high temperature cathode surface. The emitted electrons become the source of current flow through the amplifying tube. A thermionic amplifier, then, is just another name for a vacuum tube amplifier.*



1. Mic Input
2. Phantom Power Source
3. MicLim Element
4. 20dB Pad
5. Phase Reverse
6. Class A PNP Input Stage
7. Reflected Plate Amplifier Second Stage
8. Mute Element
9. Line Output Calibrate
10. LoCaf™ Low Cut Cancellation Filter
11. LoCaf Tuning
12. Preamp Gain
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14. Reflected Plate Amplifier Output Stage
15. Headroom Meter
16. 24 Bit A/D Converter
17. AES/EBU Output
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19. 700Hz Tone Generator
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22. Send Jack
23. Impedance Compensator
24. S/PDIF and TOSLINK Digital Outputs

The Model 1100 MkII is an extraordinary microphone preamplifier.

Every detail met with laborious attention during development. Not until we exhaustively tested our design both in the field and in the laboratory did we resolve to release it to production.

To complement the technical achievements inside, we styled it with a beautiful new look. Chromed side brackets provide an attractive and very robust means for rack mounting. The control windows are illuminated for easy viewing in darkened studios. The front panel is 3-D machined from billet aluminum then finished in brightly polished deep blue anographics. The text and lines are deeply embedded in the anodizing and can never rub off. We sincerely hope that our styling efforts will enhance your enjoyment of this preamplifier.

By looking at the block diagram to the left of this page, you will see how the signal flows through the Model 1100 MkII. You may find it useful to refer back to this diagram later as you read through other sections of this manual.

You will notice that the microphone input is coupled to the preamp first stage by going through a series of relays. The first relay selects the 20dB pad while the second serves to flip the polarity of the signal lines.

The first stage is a class "A" PNP ultra low noise variable gain differential amplifier having a gain ranging from -3 to 41dB. The second stage, directly coupled to the first stage, is a patented "Reflected Plate Amplifier[1]" differential tube stage with a fixed gain of 21dB. The combination of these two stages provides a novel and unique front end yielding from 18 to 62dB of gain.

The mute element shown comprises a series-shunt dual optocoupler to provide noiseless switching and high attenuation.

The patented Reflected Plate Amplifier output stage adds 3dB of gain to the front end yielding the specified gain range of 21 to 65dB.

Because of graduated power rails, the front end clip point matches the output stage clip point very closely. Therefore, metering the output of the front end accurately displays the headroom of the total system. Also, connecting the A/D converter to the front end allows precise calibration of the digital dynamic range to that of the preamp.

You will notice the function called LoCaf™. This is an Aphex exclusive low cut cancellation filter that provides increased headroom for frequencies below the cutoff. Garbled overload from handling noise, wind, or breath is completely eliminated by LoCaf.

48 volt phantom power is sourced from a slowly ramping voltage generator that eliminates thumps when powering up or down.

Later in this manual you will be more fully taught about MicLim™. You will notice that there is a MicLim optocoupler element located immediately across the mic input lines. The MicLim element is controlled by the peak detector monitoring the front end output. When enabled, MicLim will make the Model 1100 MkII virtually overload proof. A switched input resistance has been added to the Model 1100 MkII to extend the MicLim headroom when using low impedance microphones such as Neumann Condensers.

The 700Hz tone generator is a unique feature with many useful purposes. Read about it later in this manual.

Power is supplied by a linear voltage regulator assembly and a toroidal power transformer. The ac supply voltage can be selected by the user to work with all world standards.

You should find this manual easy to use. Please do use it even if you are quite familiar with microphone preamplifiers. There are special features we want you to properly understand and use.

Thank you for purchasing the Model 1100 MkII from Aphex Thermionics.

1. Reflected Plate Amplifier, U.S. Patent No. 5,450,034. See, in this manual, "Aphex Thermionics: The Story".

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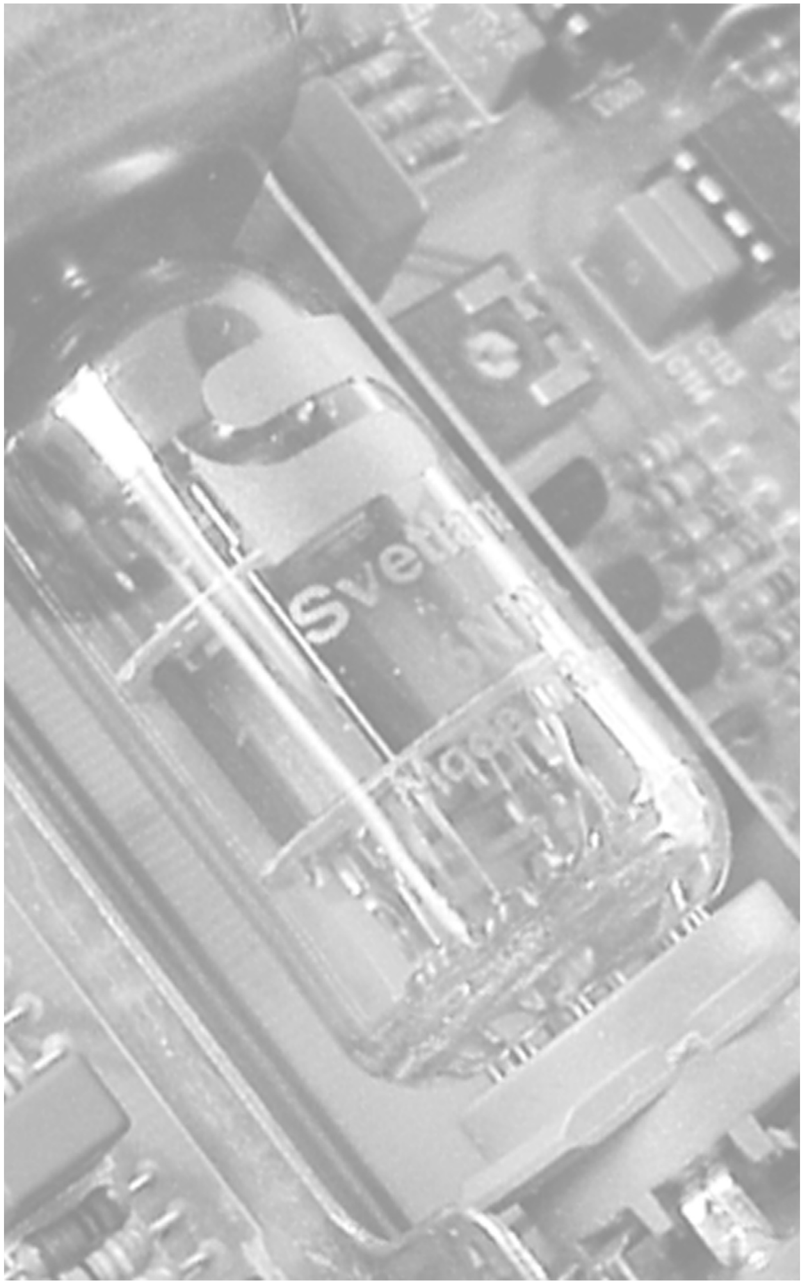


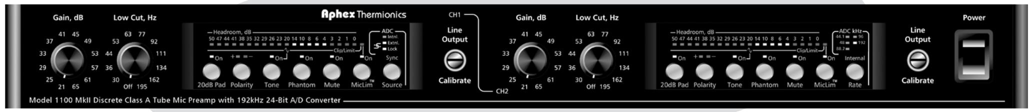


Figure 1-1 Front Panel

Perhaps the easiest way to learn this product is to study its front panel features. A glance at the panel snapshots should help bring into focus the commentary to follow.

1.1 The 20dB Pad

This button turns the pad on and off. A light above the button tells you when the pad is on. The pad is inserted ahead of the preamplifier input but in no way affects the phantom powering of the microphone. The switching is performed by gold contact bifurcated (multi contact) relays to avoid developing any distortion generating “diode effects”[1]. Use the 20dB Pad to desensitize the preamplifier for extremely high input signals whenever the lowest gain selection still yields clipping.

1.2 Polarity

This directly reverses the polarity of the microphone input lines. Like the 20dB Pad, the switching is performed by gold contact bifurcated relays to avoid contact distortion. Use this feature to correct the phase of improper mic cables, or to optimize the mixing of two or more spaced microphones.

Channel 1 Window
Channel 2

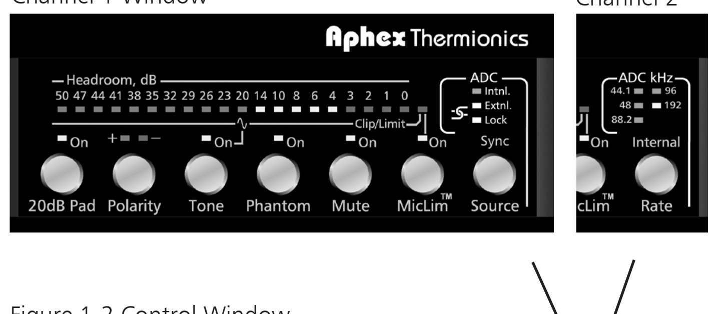


Figure 1-2 Control Window

1.3 Tone

This button turns on the -20dB, 700 hertz reference tone. For safety reasons[2], the button must be held down continuously for at least 1 second to trigger the event. Above the button is a light indicating when the tone is on, and a small sine wave symbol directly below the 20dB Headroom Meter segment shows you the correct tone level. While the tone is on, a quick touch of the button will immediately shut it off. Use this tone to calibrate various pieces of gear you are feeding from the preamp. For example, turn on the tone and then adjust the output calibration trim (and possibly the rear panel output normal switch) to get a 0dB indication on the VU meter of an analog recorder or console. This assures you will get 20dB of recording headroom along with the lowest possible noise. The tone is also useful for calibrating the drive to an external A/D converter.

1.4 Phantom

This button turns on and off the 48 volt phantom power to the microphone jack. The 48 volt source ramps up and down very slowly to avoid thumps. Slow ramping of power is very friendly to phantom powered microphones which can sometimes be damaged by sudden power application. Make sure the phantom power is off when using passive (unpowered) microphones.

1.5 Mute

An unusual feature, the Mute button activates a soft switch that silences the preamplifier output. This feature is very useful for testing and performer control. A rear panel jack allows remoted muting. If either the remote or front panel mute is activated, then the other cannot bring an unmute. Muting affects both the analog and digital outputs as well as the reference tone. Use it to allow talent to mute/unmute their own microphone (a “cough button”), for example.

1.6 MicLim

Only Aphex microphone preamplifiers have this unique and useful feature. When turned on by the push-button, a proprietary limiting module acting directly upon the microphone signal limits the peak level reaching the preamplifier input to completely prevent any clipping caused by excessive input levels up to approximately 20dB over the normal clipping level. When used mainly for protection against unexpected overload levels during live shows and recording, it is amazingly transparent. Use it moderately to help you record or reproduce consistent full maximum peak levels without fear of distortion. Read more about this powerful feature in section 5, “Understanding & Using MicLim”.

1.7 Headroom Meter

20 LED segments indicate the output signal’s peak level below the point of clipping, thus it shows you how much headroom is left at all times. In the case of digital audio, the scale also represents the digital dynamic range of the internal A/D converter, i.e., the meter is the same as a digital peak level meter referenced to 0dB(fs)[3]. This meter is useful in establishing the operating gain to attain consistently high but safe recording levels. You will quickly appreciate how it helps you to fully use this preamplifier’s extraordinary dynamic range.

1.8 Line Output Calibration

This recessed control attenuates the analog line output stage gain from 0 to -12dB, allowing you to match up the dynamic range of the preamp to your connected equipment. Use it to match the -20dB test tone to your -20dB reference level. In many cases, this will be done simply by getting a 0VU tone indication on your analog recorder or console with the record level or fader set to a nominal position. Once calibrated, the Model 1100’s headroom meter will closely match the headroom of your connected equipment and you can proceed to operate the preamp with complete confidence that you’re getting the best possible noise performance and dynamic range.

1.9 Clock Source

One of the two digital audio controls, this button selects the A/D converter’s clock source. When “internal” is selected, an internal low-jitter crystal oscillator supplies the clock reference. When “external” is selected, then word clock is taken from the rear panel BNC clock input jack. The 1100 MkII will go into a hunt mode until it finds the word clock parameters and locks on. The frequency indicator lights DO NOT come on to tell you the locked frequency. These lights are reserved to indicate clock frequencies during internal sync only. If external clock is not present, then the 1100 MkII will continue hunting for a lock indefinitely and the “locked” light will never come on. The digital audio output will be squelched, i.e., there will be no AES/EBU output at all. If the Clock Reference is present and valid, then the “locked” light will come on and the AES/EBU output will be present and synchronized to the clock source. The “locked” light will always come on whenever the “internal” mode is selected.

1.10 Internal Rate

The second of two digital audio controls, this push-button selects the A/D converter’s sample rate when the clock source is internal. The choices are 44.1kHz, 48KHz, 88.2kHz, 96kHz, and 192kHz. It has no function when the Clock Source is set for “external”.

1.11 A/D Converter Indicators

There are three lights related to the A/D converter in each of the channel windows. In the channel 1 window, you will see the lights relating to the Clock Source status and the “locked” status. In the channel 2 window you will see the internal rate indicators. The internal rate lights will all be dark when external clock is selected, but will be lighted in internal clock mode to indicate the chosen sample rate.

1.12 Gain

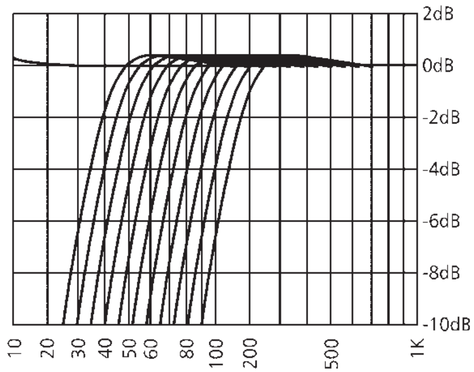
This switch sets the preamp gain in precise 4dB steps from 21dB to 65dB. Use it to optimize the operating level of the preamplifier. While observing the peak headroom meter, keep the meter peaking as high as possible without frequently hitting 0dB.

1.13 Low Cut (LoCaf™)

This rotary switch sets the low cut (high pass) corner frequency to one of 11 selected values as illustrated below. In the Off position, the low cut filter is tuned very low, around 1 Hertz, taking it completely out of the way. It comprises a unique servo cancellation circuit with a second order modified Butterworth response to optimize the sonic effectiveness of the filter. The servo cancellation method is so unique we have named it LoCaf.

With LoCaf you get as much as 20dB of improved headroom below the low cut corner frequency to reject input overload when compared with other products. Use this filter to eliminate wind or breath rumble, handling noise, or other low frequency problems.

Figure 1-3 Low Cut Characteristics



1.14 Power Switch

Pretty simple: it turns the power on and off. Power indication is by means of the window backlights. The windows will always be lighted when the power is on unless the fuse is blown or the electric supply is disconnected.

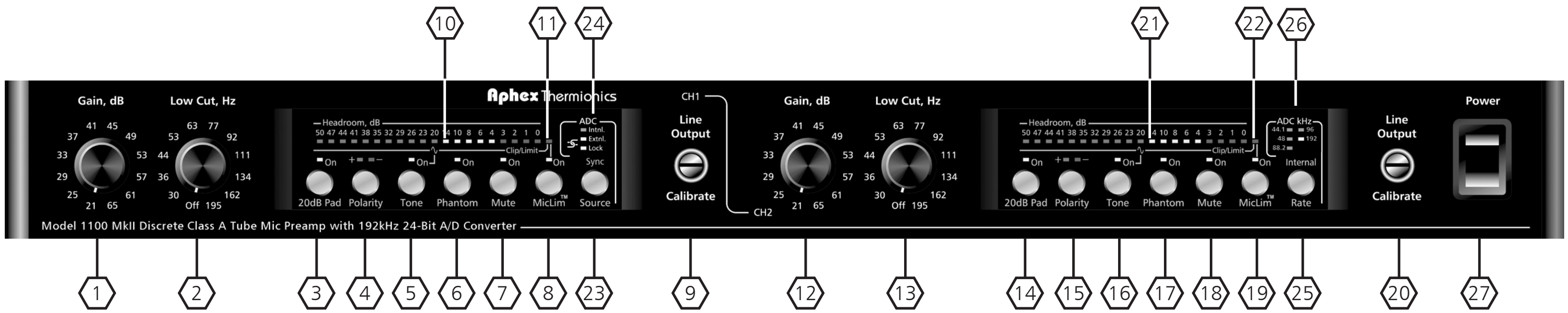
Notes:

1. Diode contact effects commonly occur whenever microphone signals are routed through ordinary mechanical switch contacts. Here's how that happens. The well established fact is that only a small number of points actually touch between any two facing contacts. Eventually, all metal surfaces begin to oxidize including these points of contact. When that happens, metal oxide point contact diodes are created that rectify the audio signal causing distortion. The extremely minute current from a microphone signal cannot break down these diodes and eventually the diode effect grows worse and worse until it is plainly audible. We chose to avoid the problem altogether , albeit at a higher cost, by using sealed bifurcated gold contact relays to do mic level switching. Bifurcation (multiple metal contacts for each circuit) overcomes the odds of getting a diode in the path, while gold contact surfaces oxidize at a rate thousands of times lower than other materials.

2. One wouldn't want the reference tone to be accidentally switched on during a session. That's why we put in a delayed reaction. You must intentionally hold down the button for a period of time to get the tone.

3. dB(fs) refers to "decibels in reference to full scale" and is the common digital audio meter reference. 0dB(fs) is the maximum peak value that can be converted to digital.

Model 1100 MkII Front Panel Summary



1 12 Input Gain: 4dB steps, 21 to 65dB

2 13 Low Cut Filter: Off, 30, 36, 44, 63, 77, 92, 111, 134, 162, 195 Hz.

3 14 20dB Pad: On/Off

4 15 Polarity: Normal, Reverse

5 16 Tone: -20dB reference level, 700Hz

6 17 Phantom Power: 48V On/Off

7 18 Mute: On/Off

8 19 MicLim On/Off

9 20 Line Output Calibrate: 3-Turns, 0 to -12dB

10 21 Headroom Meter: 0 to 50dB indicated

11 22 Clip/Limit LED: Flashes to indicate clip or MicLim action

23 Internal/External Sync (Clock) Source

24 A/D Converter Indicators: Internal, External, Locked

25 Clock Rate : 44.1, 48, 88.2, 96, and 192kHz

26 A/D Converter Indicators: Internal, External, Locked

27 Power: On/Off

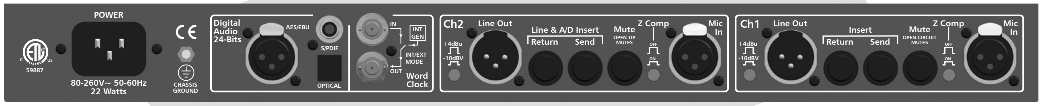


Figure 2-1 Back Panel

The rear panel hosts all power, input, and output connections of the Model 1100 MkII. This section teaches you how to properly connect and use each interface. Most users will probably find the back panel completely self explanatory, nevertheless we suggest that you read the commentary carefully to avoid any possible misuse or confusion and assure that you are getting the maximum performance from your microphone preamplifier.

2.1 Power Entry

The standard IEC power receptacle automatically accepts any normal a.c. power source from 80 to 260 volts, 50 to 60Hz. You don't have to set a voltage selector and there is no replaceable fuse. The internal power supply is self regulating and self protecting. If the internal fuse should ever blow out, then a catastrophic failure is indicated requiring service by a qualified technician.

2.2 Chassis Ground

Connecting this ground is not essential in most cases since the power cord will usually provide the proper safety ground to the unit. However, in case you encounter hums, buzzes, or radio interference, you may be able to eliminate them by connecting a separate, heavy gauge, system ground wire to this screw. This is especially helpful if the unit is not installed in a rack having grounded metal rails that would act as a chassis ground source. You can try connecting the chassis ground to the mix console ground, or the chassis of other equipment, for example.

To meet legal requirements, we have not provided a means to lift the power cord ground from the chassis. If you determine it is necessary to lift the power cord ground, you must use a ground lift adapter on the power plug. You will not be able to lift the power cord ground from inside the chassis.

2.3 AES/EBU Digital Audio Output

Preamplifier channels 1 & 2 are encoded into the respective AES/EBU channels (ch1=left, ch2=right). The standard 110 ohm transformer balanced output will interface with any standard AES/EBU input. Be sure you use professional grade digital audio cables to insure high quality transmission of data. Low grade cables or plain audio cables will severely limit the transmission distance, and, even worse, induce a large amount of jitter into the data stream.

In case you get no output signal, check the front panel to be sure the clock is locked. If external clock is selected and there is no external clock source, the AES/EBU output will be squelched and you'll get no output. In all other cases you should get an AES/EBU output signal if the unit is functioning properly. Before declaring that you have no output signal even though the clock is locked, be sure your receiving equipment is able to lock to the sample rate you are using. Many pieces of gear are not yet capable of receiving the 96KHz sample rate.

2.4 External Clock

Two BNC jacks are provided to permit multiple Model 1100 MkII's to be daisy-chained together from a master clock source. In trying to simplify the way things work, and we have made everything pretty much automatic for you. Nevertheless, please take the time to study their usage before planning your installation.

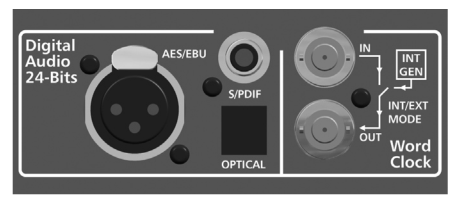


Figure 2-2 Digital I/O & Clock

2.4.1 Clock In

This BNC jack is provided to receive your master clock source. It will accept industry standard Word Clock, from 1 to 5V pulse amplitude. It does not accept AES/EBU or Superclock.

The Model 1100 MkII can operate equally well from a typical "brute force" 5V word clock output or from a matched impedance master clocking distribution. Please refer to the appendix for a thorough discussion of clock wiring and distribution systems.

When the Model 1100 MkII is operating in external clock mode, the Clock In jack is tied directly to the Clock Out jack through a metal relay contact. The input impedance is high enough to daisy-chain up to eight units by looping straight through. <u>The final unit should have a 50 or 75 ohm (as dictated by your distribution cable impedance) BNC terminator plug inserted in its Clock Out jack to properly terminate the transmission line.</u> If only one unit is connected, then simply place the terminator plug on the single unit's Clock Out jack.

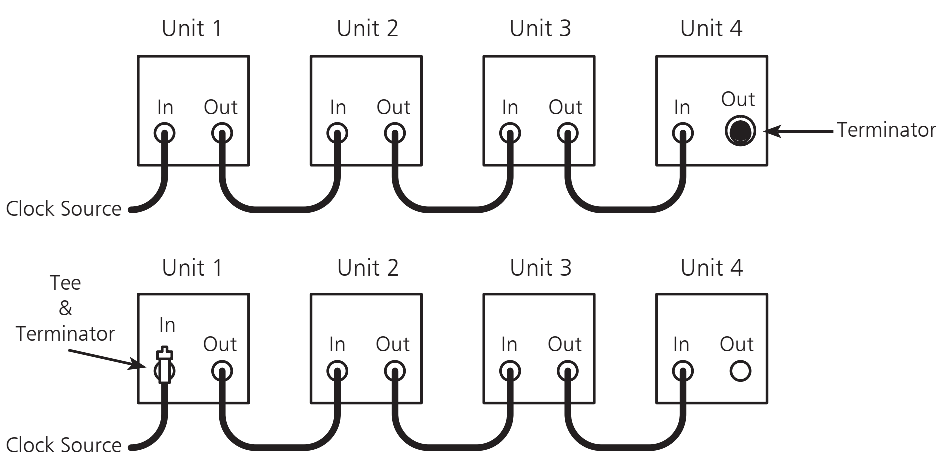
When the Model 1100 MkII is operating in the internal clock mode, the Clock In jack is isolated and unterminated. Be aware of this in case you intend to operate in both internal and external clock modes. You may need to terminate your distribution cable using a tee adapter up front instead of using the last Clock Out jack for that purpose. Refer to figure 2-3 for illustrations of the two most commonly used daisy-chaining options.

2.4.2 Clock Out

As previously described, this jack is directly tied to the Clock In jack to facilitate daisy-chaining multiple units to a master clock source when the unit is operating in external clock mode.

When the Model 1100 MkII is operating in internal clock mode, the Clock Out jack is disconnected from Clock In and instead is connected to the unit's internal clock source. Thus, any Model 1100 MkII operating in internal clock mode can serve as a master clock source when required. The internal clock is derived from a low jitter crystal oscillator and is fully competent as a master clock.

Figure 2-3 Daisy Chaining the External Master Clock



2.4.3 Local Clock Mastering

In the absence of a suitable external clock source, you can use one Model 1100 MkII as a clock master and slave everything else to it. For example, to synchronize a group of Model 1100 MkII's, simply set the first unit to internal clock mode and all the others to external. Daisy-chain the Clock Out of the first unit through the clock ins and outs of the remaining units. This way all digital audio outputs will be locked together to the first unit's clock reference. You can also loop the clock output of the last unit in line to your digital recorder or workstation, but remember to terminate the last unit in line.

2.5 Audio I/O

Both preamplifier channels comprise a similar back panel arrangement. We'll describe channel 1 in detail then you can apply that to channel 2 as well.

2.5.1 Line Output

The phase of the Model 1100 MkII is matched from input to output, i.e., pin 2 of the output is of the same signal polarity as pin 2 of the mic input jack.

The transformerless active balanced line output is provided through a standard 3-pin XLR jack.

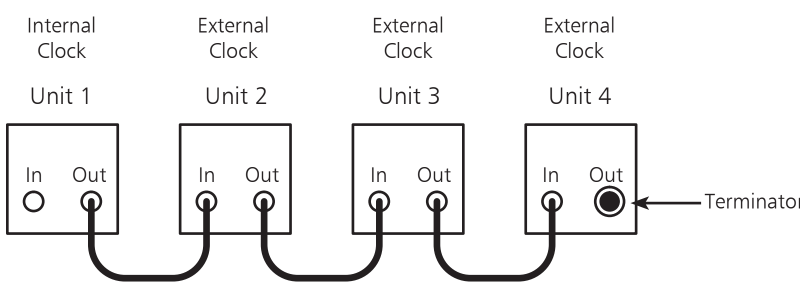
The Model 1100 MkII balanced output is a special type known as "impedance balanced" that eliminates many of the problems often encountered when interfacing with a variety of balanced and unbalanced studio gear.

When wiring the Model 1100 MkII to the balanced input of other equipment, there is nothing special to do; just use standard wiring. Your equipment will see the Model 1100 MkII's output as a perfect balanced source and you will receive all the hum and noise rejection afforded by a balanced line transmission. The fact is, however, there is signal driven only to pin 2. Pin 3 is returned to ground through an impedance equal to pin 2's output impedance. How, you may wonder, can that be equal to a balanced output circuit? The answer is a bit technical, and for those who really want to know, you can refer to the end of this section for a full explanation. For everyone else, please trust us. It works and it works very well.

2.5.2 Nominal Output Reference

This switch simply inserts a 12dB pad into the output line to reduce the line output level into the proper range for semi-pro gear typically connecting through RCA jacks. It converts the professional standard level of +4dBu to the IHF standard of -10dBV (316mV) . More and more equipment is moving towards or somewhere around the IHF operating levels, so be aware that you may need to set that switch accordingly. What you want is a good and proper level into your gear when the Model 1100 MkII's Headroom Meter is peaking within the proper range. To learn about the Headroom Meter, look in section 1 of this manual.

Figure 2-4 Daisy Chaining the Local Master Clock



2.5.3 Mic Input

This is where you plug in the microphone. Pins 2 and 3 are phantom powered when the phantom power is turned on by the front panel button. There is little more to say about this jack. The input phase is matched to the output phase, i.e., pin 2 is positive at both jacks.

2.5.4 Z Compensation Switch

Some microphones have an output impedance appreciably lower than 150 ohms. The Model 1100 MkII is quite happy with these mics, except for the fact that MicLim can have only a limited amount of effect. To extend the limiter effectiveness, you should switch on the Z Comp to insert a small amount of resistance into the input circuit. This does not reduce the output level of the microphone nor does it affect the tonality. The only tradeoff is in the achievable noise floor at high preamp gain (>40dB). With Z Comp on, the noise floor may rise slightly, but typically less than 2dB.

While on the subject of noise, one point many people wonder about is why, when there is no microphone plugged into a microphone preamp (any preamp), is the output noise so much higher? The answer is not that cosmic rays are getting into the jack. Actually, it has to do with something called the "noise resistance". When the microphone is plugged in, its impedance creates a low noise resistance about equal to the mic's impedance, for example 150 ohms. Low resistances generate low noise. When the jack is open, the many times higher input circuit resistances become the dominating noise resistance, thus the generated noise is much higher. The preamp can't tell whether a mic is there or not, it just amplifies whatever signal is there even if it is just noise.

Some other pointers may help you get lower noise from this preamp. Do not leave an external mic pad on the line then goose up the gain later to get enough level. You should always use the strongest mic signal you can get and use the lowest mic preamp gain possible to keep the level right. Use the internal 20dB pad only when the input signal is approaching a line level. You will know when that is necessary if you switch the preamp gain all the way down to 21dB and the meter is still pegging.

Don't expect to be able to run mic cables directly along a.c. power cables. You are likely to pick up dimmer buzz. If you have to get near a.c. power cables then lay mic cables crossways to minimize induction. If you get a real pesky hum problem, try a different microphone, cable, channel, etc. Nothing beats substitution for tracking down a bad component.

Send & Return Jack Wiring
Tip = HIGH
Ring = LOW
Sleeve = GROUND

2.5.4 Insert Jacks (Send - Return)

Located next to the Line Out XLR connector, these 1/4" phone jacks allow you to insert outboard audio processors between the mic preamp and the A/D converter as well as between the preamp and the analog line output. The jacks are normally looped straight through ("normalled"). With a stereo plug inserted in the return jack , the signal will be routed through the outboard equipment. The Send jack may be used as an auxiliary output without breaking the normal link, but it is not fully isolated from the preamp and could cause noise or distortion problems if not properly used.

The operating level at the Send jack is approximately 0dBu when the preamp is operated with typical headroom allowances. It may run hotter if you are pushing the MicLim heavily.

You should use the input gain controls of your outboard equipment to match this level rather than turn down the preamp gain. The return audio level should be adjusted on the outboard equipment to match the send level. The front panel Headroom Meter indicates the level at the Return. This is also the A/D converter feedpoint, so the meter always indicates what the A/D converter is receiving, thus also indicating the converter's digital audio level.

2.6 Mute Jack

Plugging a switch of any sort into this jack allows you to remotely mute the preamp. Muting affects not only the line output, but also the digital audio. When the switch is open, the preamp is muted. A closed switch unmutes. The front panel mute button works in parallel with the jack.

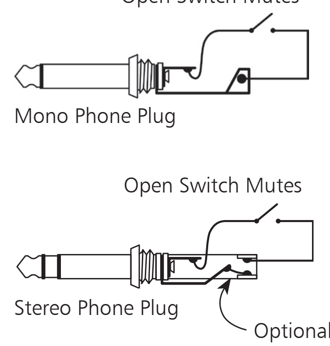


Figure 2-5 Remote Mute Plug Wiring

Whichever causes a mute takes precedence. The other cannot cause an unmute.

If no phone plug is inserted, then the jack normals to a closed condition, giving all mute control to the front panel button. Connections are to the tip and sleeve only. You can therefore use either a mono or stereo phone plug but it must be wired as mono, i.e., the stereo "ring" is not used or optionally it can be grounded to the sleeve.

2.7 Impedance Balancing and You

For the technically minded (and maybe just a bit skeptical) reader, here's how an impedance balanced output works. Your balanced input stage looks at the two wires and detects only the potential (voltage) difference between them. Anything that is the same on the two wires (for all practical purposes as seen measuring from ground) is called a common mode signal and is cancelled out by the differential amplifier. Figure 2-6 illustrates how the hum is induced into both wires equally and therefore is cancelled out.

The fact is that, as long as the hum and noise is induced into both wires equally, the hum will be rejected even if only one of the wires actually has the wanted signal on it. That may be a hard fact to swallow, but look at it this way. The reason we twist the wires together tightly is to make them occupy the same thin line in space on average. Now, if both wires have the same impedance to ground at both ends, then they will electrically appear like two identical antennas in space. Both wires will therefore pick up equal interference from all surrounding sources. Remember that when both wires carry identically the same interference, the interference will be cancelled.

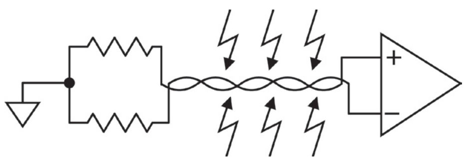


Figure 2-6 Common Noise Rejection Model

To carry the explanation one last step, look at figure 2-7. This is the model for an impedance balanced transmission line. Notice that one wire is fed from an output amplifier through a resistor while the other goes to ground through an equal resistor. If we consider that the amplifier's output is essentially a zero ohm source, then we can consider it to be the same as a ground connection in terms of impedance loading on the wire. This makes the circuit of figure 2-7 the same as figure 2-6 for the purpose of noise rejection.

So that explains how an impedance balanced output gives us full noise rejection, but how can you just drive one line and receive it OK? Well, that's really simple. If one wire is always at zero, and the other has a signal, then the difference between the two wires will be the signal, right? Right.

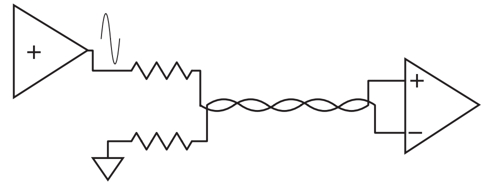


Figure 2-7 Impedance Balanced Line

Look at figure 2-8 showing a conventional line driver where there are two output amplifiers to drive the balanced line in counter phase through two equal resistors. The only real difference between the counter phase driver of figure 2-8 and the impedance balanced driver of figure 2-7 is that to make up for a zero signal on one wire, the impedance balanced driver must double the voltage output to the signal wire.

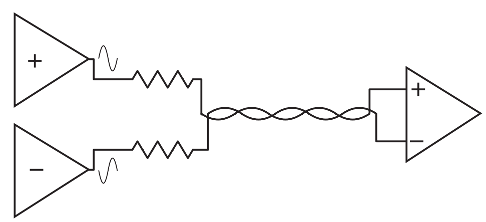


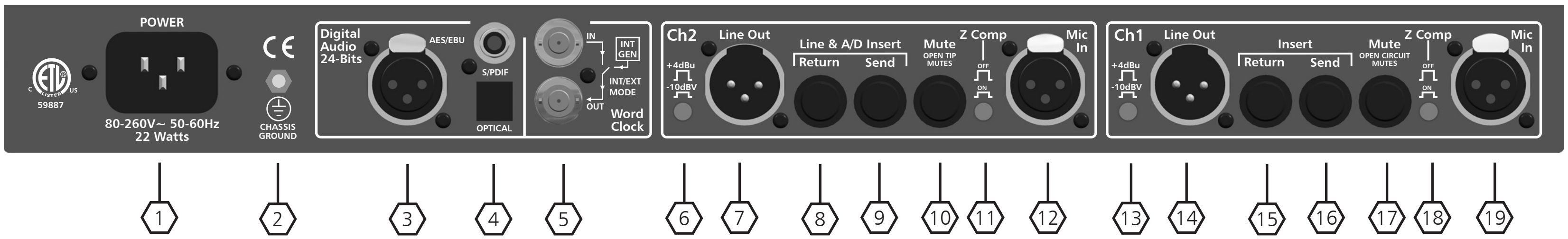
Figure 2-8 Conventional Balanced Line Driver

In other words, if we need two volts peak output balanced, a conventional output stage would drive negative one volt to one wire and positive one volt to the other wire. Our circuit just drives positive two volts to the signal wire. The balanced input circuit in the first case sees positive 1 volt minus negative one volt and gets 2 volts. In the second case, the balanced input stage sees positive two volts minus zero volts and still gets 2 volts.

Now here's why impedance balancing is such a good deal. Let's say you want to patch the output of the Model 1100 MkII into a line insert of your studio console. Some of these inserts are balanced, but most are unbalanced. With the Model 1100 MkII all you have to do is make sure pin 2 wires to the patch plug tip, and you've got full signal. You don't lose half the signal amplitude and with it 6dB of precious preamp gain. You also don't lose the 6dB of peak output headroom you lose with a counter phased output stage running unbalanced. In effect, the impedance balanced output stage acts a lot like a transformer balanced output but without the sonic degradation associated with audio transformers.

There is complete information on interfacing the impedance balanced output of the Model 1100 MkII in section 4 of this manual.

Model 1100 MkII Back Panel Summary



Power

1 Universal Power Receptacle. Voltage accepted: 80-265 VAC 50/60Hz

2 Chassis Ground Lug

Digital Audio

3 AES/EBU 24-Bit Digital Audio Output. Channel 1 = LEFT, Channel 2 = RIGHT. Sample Rates: 44.1, 48, 96, 192 KHz

4 S/PDIF and OPTICAL digital audio outputs.

5 Word Clock I/O Jacks. W. C. passes directly through in “external” mode. Output is from internal generator in “internal” mode.

Analog Audio

6 13 Output level noninal “0dBVu” switch. Provides +4dBu or -10dBV nominal reference level.

7 14 Impedance balanced line output XLR jack.

8 15 Insert return jack (balanced/unbalanced).

9 16 Insert send jack (balanced/unbalanced).

10 17 Remote mute jack. Tip to ground switch causes mute when open.

11 18 Z-Compensation switch. Use to extend MicLim range for mics with lower than 150Ω output impedance.

12 19 Microphone input jack.

With MicLim comes a new kind of freedom.

Can you imagine if you could have a preamp that wouldn't be overloaded by an unpredictable live performer? You could freely run higher gain to pick up the whispers, then sit back and relax while the orator rings out in mellifluous splendor. Ordinarily you'd have to ride preamp gain to avoid clipping during those loudest moments, but MicLim *does it for you instantly. Maybe you don't deal with live show performers, but maybe you have dreamed of being able to record whatever you want at hotter levels without any fear of overload. Learn how this exclusive feature works and how you can use it effectively for your own application.*

3.1 How MicLim Works, Basically

Microphone limiters traditionally consist of a mic preamp followed by a limiter circuit. Until MicLim was devised, that was the only practical way to do it. Everybody is familiar with the clipped overload that's heard when a performer suddenly gets on the mic too loudly. The compressor/limiter held down the P. A. level, but the preamp was clipping horribly. Until MicLim, there was no other choice except to constantly keep a hand on the preamp's gain knob.

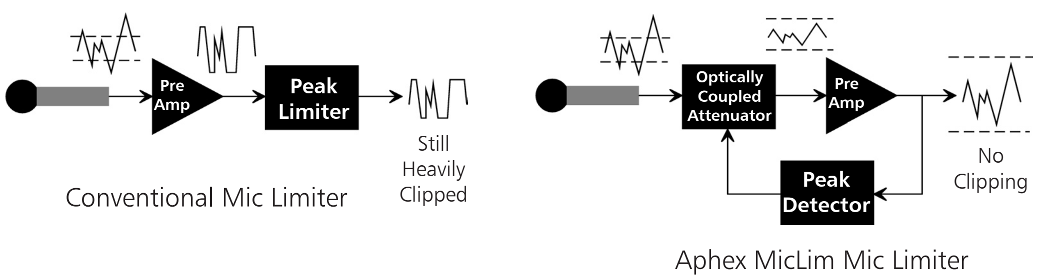
The difference with MicLim is that the limiter works with the microphone itself, before the preamplifier has any chance to clip at all. Patent pending technology had to be devised to make that possible since just about anything inserted between the mic and preamp will add noise or distortion. With MicLim now available you will seldom find it possible to overload the Model 1100 MkII mic preamp, even under severe conditions. That's because there is typically at least 16dB of limiting headroom afforded to any microphone of 150 ohms or higher. Effectively that means that, with MicLim, you get 16dB more preamp headroom before distortion.

3.2 How MicLim Works in Detail

This discussion is especially for the technically astute reader. The rest of you can skip this until later (or forever) if you're not really all that interested in the small technical details of how this works.

The "optically coupled attenuator" of figure 3-1 comprises a specially developed optically coupled load resistor that is loaded across the mic input line to react against the microphone's self impedance, causing a variable loss of level depending upon the optocoupler drive current. Drive current is generated by a very fast peak detector. The maximum amount of attenuation available depends upon the microphone's self impedance, but typically at least 16dB is achieved with a standard 150 ohm microphone. Increased attenuation is available for microphones with higher impedances. That means that MicLim can cleanly limit microphone levels that are up to an incredible 16dB or more over the normal clipping point of the preamplifier!

Figure 3-1 Comparing MicLim



The peak detector operates in a feedback arrangement sensing the preamp's output level to anticipate accurately when any signal is approaching the clipping point. A progressive current is sent to the optocoupler as soon as the peak threshold is exceeded. The attack time of this limiter loop is designed to be fast enough to avoid any discernible clipping. Considerable inventiveness was required to devise a stable MicLim control loop with such a fast and accurate response. Previously there have been no successful fast attack optocoupler feedback limiters because the inherent latency and memory of Cadmium Sulfide (CdS) photocells causes oscillatory reactions. Only feed forward optocoupler limiters have been produced. For MicLim to work in this application, however, it has to operate in the feedback configuration.

Photocell

In addition to inventing a unique new way to control an optocoupler element, it was necessary to devise a new kind of CdS optocoupler that can be driven to reach a very low "on" resistance, typically 10 to 100 times lower than previously available units. Considerable research and cost was expended to develop that element. Finally, after nearly a year of research, we found success.

Cadmium Sulfide photocells have been used in popular "optical" studio compressors for decades. Their complex time constants and optical "memory" are known to provide unusually smooth compression under all types of program material. Although our new technology photocell achieves a much greater dynamic range than ever before, it shares in the timing complexity of its predecessors. That's one reason why MicLim so gracefully handles both transient and sustained limiting.

3.3 How To Use MicLim

Within each channel's control window there is a push-button for turning on and off the MicLim feature. All you have to do is turn MicLim on and it will instantly become armed. Nothing will happen, however, until an audio peak approaches clipping: then the limiter will act on the microphone signal to prevent the peak from reaching the clip point. Whenever the limiter hits a peak, the red Clip/Limit light in the channel control window will flash.

There are no special rules about making MicLim work for you. The concept is simple: run the preamp gain at a point where the peaks indicated on the Headroom Meter only reach the MicLim threshold infrequently or unexpectedly.

It may be evident to the astute that if one deliberately increased the preamp gain until peaks frequently limit, one could obtain a very dense output level by bringing the average signal so much closer to the peak ceiling. Although it is certainly possible to operate MicLim in that fashion, and often the effect can be desirable, we recommend moderation. MicLim was designed to be fast and unobtrusive for protection against accidental overloading, especially with voices. You may find the fast speed of this limiter too aggressive for general purpose limiting. Your favorite studio compressor/limiters can still be put to good use. That's why we include an insert path in the product.

3.4 MicLim and the Microphone

MicLim works on all microphones. The use of phantom power in no way inhibits the performance of MicLim.

Using The Z-Comp Switch

The 1100 MkII has an **impedance compensation (Z-Comp)** switch near each mic input jack to compensate for extremely low low mic impedances. and extend MicLim's effective range. If you are using powered mics that typically have 50 ohms or less output impedance, you will need to turn this switch on. There will not be any compromise in sound quality. However, the potential noise floor may rise by up to a scant 2dB for preamp gains over 40dB.

When using dynamic mics with impedances typically over 120 ohms, MicLim does not need the use of the Z-Comp switch. There will be no ill effect of having it on, however, other than the slightly increased noise floor as stated above.

3.5 Whenever in Doubt, Use It!

Not sure about the MicLim? Just use it. If you never need it then it won't have any effect at all. But when that big ass peak comes along, Bam! You're saved.

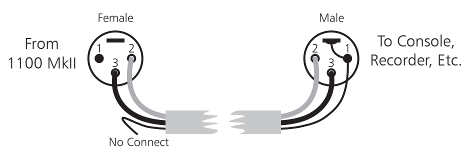
Because of its impedance balanced output stage, the Model 1100 MkII is unusually easy to interface with balanced and unbalanced outboard gear. We have subscribed to the "pin-2 high" AES wiring standard so the only problem you should possibly encounter would be with interfacing cables wired using the older and sometimes still seen "pin-3 high" practice. This would be a problem mainly when connecting to unbalanced inputs and some of the older console patch bay inserts. If you should experience problems with hum, buzz, no signal or very low level, check your cables against the diagrams shown in this section. These diagrams are guaranteed to work.

4.1 The Mic Cable

Connect all 3 wires pin for pin and connect pin 1 on each end to the shell ground as shown.

4.2 XLR to XLR Balanced Output

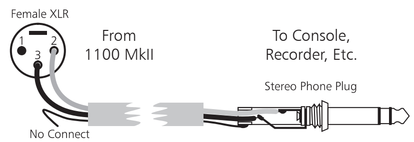
Normally, connect the shield only at the male end. Connect male connector shell to pin 1. In rare cases you may need to connect the drain wire to the female XLR pin-1 and shell similar to the mic



cable above. Usually this increases the chances for ground loop hum and radio interference, however.

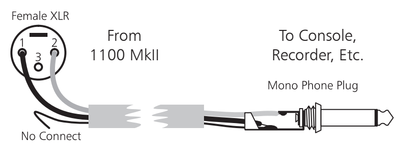
4.3 XLR to Phone Balanced

For proper phasing, connect the tip to pin 2, not pin 3. Follow the same rules for the drain (shield) wire as XLR to XLR, above.

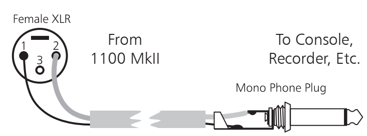


4.4 XLR to Phone Unbalanced

The preferred method to make an unbalanced cable is shown below. Note the use of balanced wire. This technique improves hum rejection over standard single conductor shielded wire.



The method for using single conductor shielded wire is shown below. You should use high grade shielded wire to assure the lowest hum pickup.



4.5 Grounding Practices

Over the years there have been various practices taught on proper grounding. Many studios subscribe to the idea of always grounding shields at both ends, or only at the sending end. These cases both violate the rules we are teaching.

Many years of experience with world class studios, broadcasters, and live sound companies have taught Aphex engineers to analyze the grounding problem scientifically, not shooting-from-the-hip as often encountered out in the field.

Except for microphone cabling, we have found compelling reasons why grounding all shields only at the receiving end of the cable is best in the majority of circumstances.

An analysis of this subject is rather lengthy, so we won't burden you with it here. We ask you to trust us on this, regardless of what you may have read in technical journals or other equipment manuals.

Does this mean that you have to rewire your whole studio complex? No, we don't expect that at all, and your Model 1100 MkII will probably work very well in any studio wiring scheme because its output is impedance balanced. However, should you experience chronic hum problems around your studio, you may want to consider rewiring with our recommended grounding practice.

4.6 Using Purchased Adapters & Cables

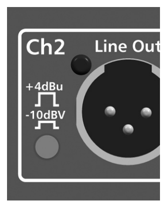
You may sometimes have trouble using the variety of connector adapters available through audio

pro shops, though most of the time they will work extremely well. If you get hum or no signal, use an ohmmeter or continuity tester to be sure they are wired according to the diagrams of this section. Some adapters are built with the outdated “pin-3 hot” practice which will not always work with the Model 1100, especially when converting to an unbalanced output. If you are in the practice of using standard mic cords (which are wired according to 4.1) for general purpose interconnecting of equipment, then the shield ground will flow through connecting both pin 1’s. That will violate the rules we have been outlining here, but don’t despair. You probably won’t experience any problems. If hum or interference does appear, simply try cutting the shield wire from the sending end of the cable.

4.7 Checking For Output

If, for some reason, you think there is no output from the Model 1100 MkII even though signal is indicated on the Headroom Meter, check the insert jacks. If they are not used, then the signal should pass through. If the Return jack has a plug inserted, be sure a signal is getting back to the 1100 MkII from the external equipment. In the event all of this seems ok, then be sure the output cable is correctly wired. There should be no troubles using balanced lines, but if you are feeding an unbalanced input, then be sure you have the 1100 MkII’s output wired as pin-2 hot, not pin-3.

4.8 Matching The Levels



Audio equipment interface levels can vary within the same room. The Model 1100 MkII accommodates both pro and consumer operating levels by providing a normal level switch on the back panel.

4.9 Mute Function Wiring

The rear panel mute jack accepts either a mono or stereo phone jack. Refer to section 2 on Back Panel Features for wiring diagrams and instructions for use.

4.10 Insert Jacks

The 1100 MkII’s rear panel send and return jacks are intended to allow you to insert a compressor or other processing device between the mic preamp and the A/D converter and line output stage.

The send output is impedance balanced and can therefore be readily wired out as balanced or unbalanced. The operating level is typically around 0dBm but may run hotter if you are running the preamp gain high enough to consistently push MicLim. If you desperately need an auxilliary audio output, you can plug into the send jack without breaking the 1100’s signal path. Just be aware that you can potentially introduce noise or distortion into the A/D converter and the main analog output if your interception is not clean and free of loading.

Inserting a plug into the return jack breaks the 1100 MkII’s internal signal path and gives you the entry point to the A/D converter and line output stage. The return input is semi-balanced, meaning it accepts a balanced input but is not fully balanced electronically. Usually, outboard equipment used with the 1100 MkII’s insert will be located nearby, so hum or buzz will never become a problem.

It is imperative that you adjust your external equipment’s output level to return an acceptable level to the 1100 MkII. The worst problem you can encounter is overdriving the A/D converter with too much level. You can use the 1100 MkII’s headroom meter to verify. It monitors the signal received at the return jack whether from the normal circuit or external equipment. Make sure you never drive the peak headroom meter all the way up to 0 from external equipment. When running without an insert, the 1100 MkII automatically protects the A/D converter from overdrive and hitting 0 on the meter will cause no harm other than potential distortion.

5.1 Unpacking

Your Model 1100 MkII was packed carefully at the factory in a container designed to protect the unit during shipment. Nevertheless, Aphex recommends making a careful inspection of the shipping carton and the contents for any signs of physical damage.

5.2 Damage & Claims

If damage is evident, do not discard the container or packing material. Contact your carrier immediately to file a claim for damages. Customarily, the carrier requires you, the consignee, to make all damage claims. It will be helpful to retain the shipping documents and the waybill number.

5.3 Mains Voltage And Fuses

The Model 1100 MkII's power supply automatically adapts to all a.c. power standards. There is no voltage selector or replaceable fuse.

5.4 Power Cord

The Model 1100 MkII uses a standard IEC power cord set. The appropriate mains plug for each country is normally shipped with each unit. If you must install or replace the plug, use the correct wiring code according to Table 1.

5.5 Mounting In A Rack

The Model 1100 MkII occupies one standard 19 in. by 1 3/4 in. rack space (1RU). Chassis depth is 9 inches not including connectors. Allow at least 3.5 inches additional space in back for wiring and connectors. The chassis is designed to be fully supported by front panel mounting alone. To avoid cosmetic damage to the panel, use the cushioned rack screws provided in the shipping kit or the equivalent.

Table 1: Power Cord Color Codes
USA Color Code
Black = Hot (live)
White = Neutral
Green = Ground
IEC/Continental Color Code
Brown = Hot (live)
Blue = Neutral
Yellow/Green = Ground

5.6 Proper Ventilation

The Model 1100 MkII is a Class A tube amplifier. It gets hot in use. Please be sure there is adequate clearance above and below the unit to allow for ventilation. One rack space is usually sufficient.

Outside of the rack, do not stack units directly on top of one another. Use some type of blocking to leave at least an inch of airspace between units.

5.7 Safety Considerations

To minimize the risk of shock or fire, do not expose the unit to moisture. Allow adequate ventilation for cooling. Do not open the chassis cover: there are no user serviceable parts inside.

Installation should be performed only by qualified individuals. It is the installer's responsibility to insure his personal safety and the safety of others in the work area. It is never a good idea to work alone in the vicinity of high power electrical and radio frequency equipment.

5.8 Avoiding Hum and Noise

All microphone preamplifiers are sensitive devices that can pick up hum from the power transformers of other equipment. You should never install the Model 1100 MkII near high current equipment such as a power amplifier or console power supply. If you are experiencing a hum problem with a rack mounted installation, try moving the Model 1100 MkII to another rack space or move other equipment around so you can place the Model 1100 MkII the farthest away from the noise source.

You should be able to rack mount multiple Model 1100 MkII's immediately above and below each other (separated by a space for ventilation) without experiencing hum problems.

5.9 Lightning Protection

The Model 1100 MkII is not any more prone to spike damage than other studio equipment, so following good standard practices for lightning protection should prove sufficient.

If you are in an area that is frequently hit by lightning storms you should use lightning arrestors on all ac power lines and have local spike protectors where equipment is plugged into the power source. In areas where audio cables may be subjected to lightning spikes, you should pay close attention to how equipment is grounded making sure the lightning rod cable does not travel parallel to or near to any of your audio wiring.

5.10 Preparing Your Cables

Please refer to section 4, Interfacing & Cables, for instructions on proper wiring.

5.11 Overcoming Ground & Hum Problems

In studio and live sound venues, it is often a problem to establish a good ground system. You may encounter existing systems where grounding was never planned, or you may encounter, simply, a wrong grounding strategy extensively implemented.

Wherever hum and noise problems are endemic you may be sure the ground system is faulty. Dealing with that can be treacherous but with simple understandings you can usually get control.

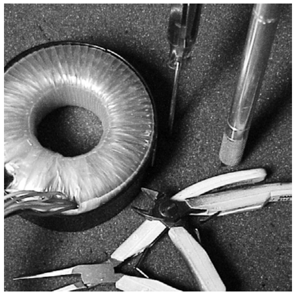
The most common ground problems come from the a.c. house wiring. The wall outlets of adjacent walls in a room are normally sourced from alternate phases of the power distribution panel. When equipment is plugged into both sets of wall outlets, there can be a voltage difference between the neutrals and outlet grounds. That voltage difference generates ground current loops through your equipment and consequently through your audio wiring. The best way to stop these problems is to reassign the power distribution panel so all outlets in the same room are from the same power phase. If that is not possible, then you should consider placing suitable power isolation transformers in your room to isolate the equipment power from the house wiring.

When possible, you can plan the house wiring to include insulated ground wires and use hospital grade outlets that keep the outlet ground isolated from the conduit and box. With all insulated ground wires bonded to the distribution panel ground bus, a “star ground” is established that eliminates most of the ground loops.

In the worst case, when you cannot in any way correct the house wiring problems, then you may need to distribute power to your equipment only from the outlets of one wall by using power strips. If you do this, be very sure the circuit does not become overloaded. Check the rating of the circuit breaker servicing those outlets and survey the total load including any appliances that may be in another room on the same breaker. You may become lucky and find a pesky hum problem can be eliminated simply by getting the right two pieces of equipment onto the same outlet and leaving everything else alone.

5.12 Radio Interference (RFI)

We designed the Model 1100 MkII to reject radio interference, but if the level of interference gets strong enough it will break through. If you experience TV “sync buzz” or taxicab calls in your audio, then you must isolate the source of the RFI. It may come through the mic cable, but it can also get in through the power cord or other cables plugged into the unit such as the insert send and return lines. Try moving the power cord and cabling around to minimize the effect. You may need to install a power line filter if you are located near a broadcast station. Keep all wiring as short as possible.



Aphex Systems supports its customers with spare parts and technical assistance. You may contact us by phone, fax, and the Internet. Out-of-warranty repair work should be performed by qualified service personnel only. We recommend contacting the factory or other authorized service agency for all repair work.

6.1 Obtaining Service

Units should not be shipped to Aphex for service without first obtaining an RMA (returned material authorization). Equipment received without an RMA may be refused for delivery and returned to the sender. Contact Aphex customer support for an RMA. The RMA number must be placed on the outside of the shipping carton to identify the unit. Please also include within the container a brief letter describing the defect or problem, your name and return shipping address, and a phone number where we may reach you or someone else familiar with the problem.

You may contact Aphex customer support through:

Telephone 1-818-767-2929
Fax: 1-818-767-2641
Internet: techsup@aphex.com

Outside the USA, contact your local authorized Aphex distributor or dealer for service. You can find the appropriate world-wide service agencies by contacting Aphex Systems by phone, fax, or on the Internet.

6.2 Warranty Claims

All warranty claims must be presented to the Aphex factory customer support department or to an authorized dealer, distributor, or agency for processing. Unauthorized repairs and modifications to the unit may void the warranty at the sole discretion of Aphex Systems.

Qualification for warranty service will be determined by the unit's serial number and purchase date. Mailing the registration certificate or registering by the internet is recommended to insure your warranty rights. In some instances you may be required to furnish proof of the purchase date or proof of ownership to obtain warranty service. Needless to say, units obtained by fraudulent means or known to be stolen will not be honored.

Flip the page for warranty information.

Limited Warranty

PERIOD
One year from date of purchase

SCOPE
All defects in workmanship and materials. The following are not covered:
a. Voltage conversions
b. Units on which the serial number has been defaced, modified,or removed
c. Damage or deterioration:
1. Resulting from installation and/or removal of the unit.
2. Resulting from accident, misuse, abuse, neglect, unauthorized product modification or failure to follow instructions contained in the User's Manual.
3. Resulting from repair or attempted repair by anyone not authorized by Aphex Systems.
4. Occurring from shipping (claims must be presented to shipper).

WHO IS PROTECTED
This warranty will be enforceable by the original purchaser and by any subsequent owner(s) during the warranty period, so long as a copy of the original Bill of Sale is submitted whenever warranty service is required.

WHAT WE WILL PAY FOR
We will pay for all labor and material expenses for covered items. We will pay return shipping charges if the repairs are covered by the warranty.

LIMITATION OF WARRANTY
No warranty is made, either expressed or implied, as to the merchantability and fitness for any particular purpose. Any and all warranties are limited to the duration of the warranty stated above.

EXCLUSION OF CERTAIN DAMAGES
Aphex Systems' liability for any defective unit is limited to the repair or replacement of said unit, at our option, and shall not include damages of any other kind, whether incidental, consequential, or otherwise.

Some States do not allow limitations on how long an implied warranty lasts and/or do not allow the exclusion or limitation of incidental or consequential damages, so the above limitations and exclusions may not apply to you.

This warranty gives you specific legal rights, and you may also have other rights which vary from State to State.

These specifications are ascertained mostly through laboratory measurements and are believed by the manufacturer to be true and accurate. Nevertheless, the manufacturer disclaims any liability for damages, direct or consequential, resulting from errors. It is the sole responsibility of the user to determine the suitability of the product. Aphex Thermionics reserves the right to improve or modify its products and all specifications are subject to change without notice.

Physical
Chassis Size: 19" wide x 9" deep x 1.75 " high
Front Panel Projections: 0.75" max.
Net Weight: 8.2 lbs.
Packed Weight: 10 lbs

Environmental
Operating Temp: 32-122 deg. F (0-50 deg C)
Humidity: 0 to 95% RH, non-condensing

Power
Power Requirements: 22 Watts
Line Voltage Input: 80-260 VAC 50/60Hz
Power Receptacle: Standard IEC Type

Analog Audio Specifications
Mute Function: Optical soft switch, greater than 60dB attenuation
Low Cut Filter Frequencies: Off, 30, 36, 44, 53 ,63, 77, 92, 111, 134, 162, 195 Hertz
Low Cut Filter Type: Second order butterworth servo cancellation technique
Phantom Voltage: +48VDC
Phantom Build-Out Resistances: 6.81K ohms to pins 2 and 3
Phantom Rise/Fall: Slow ramp up and down, approximately 5 seconds
Preamp Gain: 21dB to 65dB in 4dB steps (presuming an unloaded output and null or unity gain insert)
Output Match Trim: 3-Turn precision potentiometer 0 to -12dB range
Maximum Output Level (MOL): +27dBu unloaded, +25dBu into 600 ohms
Maximum Input Level (MIL): +27dBu minus preamp gain value plus pad value
Equivalent Input Noise: Gain=65dB, input shorted, BW=100Hz to 22KHz, EIN=-135dBu
THD: Output level=+15dBu, 0.09%; output level=+4dBu, 0.05%
IMD: Output level=+15dBu, 0.12%; output level=+4dBu, 0.08%
DIM: Below noise floor at all possible measurement levels
Slew Rate: 22V/uSec
Analog Dynamic Range: 97dB to 101dB depending upon preamp gain setting
Analog Output Noise: -70dBu to -74dBu depending upon preamp gain setting
Analog Signal To Noise Ratio (SNR): 76dB typical, for ref level = +4dBu

Digital Audio Specifications

Output Resolution: 24-Bits

Output Sample Rate: 44.1, 48, 88.2, 96, 192 KHz Selectable

Clock Source: Internal Low Jitter Crystal and external word clock

Digital SNR: 98dB typical for ref level = 0dB(fs), 80dB typical for ref level = -18dB(fs)

Dithering: Due to the microphone and preamp noise floor, dithering is inherent. You don’t have to add any dither when recording at 16 bit resolution.

Front Panel Controls

Preamp Gain (each channel): 21, 25, 29, 33, 37, 41, 45, 49, 53, 57, 61, 65 dB

Low Cut Frequency (each channel): Off, 30, 36, 44, 53 ,63, 77, 92, 111, 134, 162, 195 Hz

Line Output Calibration (each channel): 3-Turn precision potentiometer 0 to -12dB range

20dB Pad (each channel): on/off

Polarity (each channel): positive/negative

Tone (each channel): on/off

Phantom (each channel): on/off

Mute (each channel): on/off

MicLim™ (each channel): on/off

Clock Source: Internal/External

Internal Rate: 44.1/48/88.2/96/192 KHz

Power: on/off

Rear Panel Controls

Line Output Normal: +4dBu/-10dBV switch

Z Compensation: On/Off

Rear Panel Connectors

CH1&2 Line Output, XLR-3M

CH1&2 Insert Send, TRS Phone Jack

CH1&2 Insert Return, TRS Phone Jack

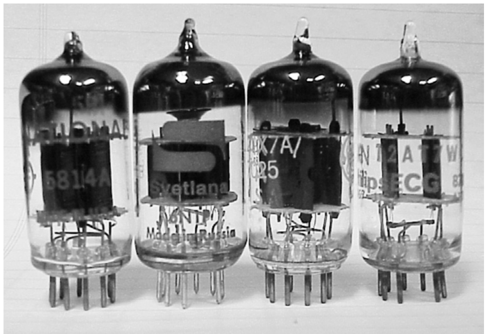
CH1&2 Mic Input, XLR-3F

CH1&2 Mute, TS Phone Jack

Word Clock In, BNC

Word Clock Out, BNC

Power Receptacle, IEC Standard



Representative Tubes

The Model 1100 MkII contains two dual triode tubes, one per channel. One triode of each tube is used in the low noise second stage and the other is used in the impedance balanced output stage. Thanks to the Aphex exclusive Reflected Plate Amplifier (RPA) circuits, you can substitute different tube types with equal success.

8.1 Tube Choice

The design of the Model 1100 MkII centered around a Russian tube called the 6N1P having a 6 volt filament. We found that the more readily available American made 6DJ8 performs identically in our listening tests and for all measurements. Therefore we reserve the right to ship units tubed with either type as their availability shall dictate.

Should you decide to experiment with different tube types, you will find we have made that very convenient for you. There may be differences in sound between different types and we feel some people will select a favorite. We invite you to experiment with tubes and tell us about your experience.

8.2 Filament Voltage

From a practical view, it only made sense during the design phase to accommodate as many tube types as possible in the Model 1100. The future supply of tubes is not completely certain, and we have recently seen some high demand tubes come into scarce supply such as the omnipresent 12AX7. To accommodate as many tube types as possible, we supplied a tube filament switch for 6 and 12 volts. This allows you to put many 6 or 12 volt types into the same socket.

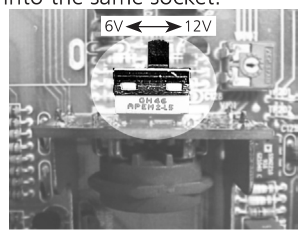


Figure 9-1 Filament Voltage Switch

Set this switch as appropriate when replacing or exchanging tubes.

Having the switch in the wrong position will simply result in no light-up but will cause no harm.

8.3 Consideration of Tube Types

We selected the 6N1P and 6DJ8 types because of their low noise, rugged construction, and low microphonics. They are readily available in premium quality from new old stock and not very popular for audio equipment making their continuing supply well assured.

Bear in mind, as already stated, the difference between tubes will be sonically very subtle, not as great as heard when substituting tube types in other equipment using conventional circuitry. For that reason, we expect that future tube replacement will not be an agonizing judgment call. Simply replace with any suitable type that is currently available. Of course, we recommend attempting to obtain the original type if possible. We maintain a stock of qualified tubes at the factory, and we recommend letting us supply replacements to you if ever needed.

8.4 Suitable Tube Types

We prefer not to comment upon the relative merits or sonic differences between tubes since those judgments are mainly subjective. Many people will probably detect no difference at all. We therefore offer these suggestions only on their merits as technically operable in the circuit.

List 1: Known To Work

6DJ8, 6N1P, 12AT7, 12AU7, 12AX7, 5751, 5814A, 6679, 6680, 7025

List 2: Believed To Work

6BC8, 6BK7, 6BS8, 6BX8, 6BZ7, 6BZ8, 6FW8, 12AV7, 12AY7, 12AZ7, 12BZ7, 12DM7, 12DW7, 12U7, 6072, 6211, 6679, 6829, 6851, 6955, 7247, 7318, 7728, 7729

8.5 Tube Life

It is unlikely you will ever have to replace the tubes in your Model 1100 MkII. Here's why.

The Model 1100 MkII regulates the filament supply voltage very tightly to about 20% less than the tube rating. This extends the filament life indefinitely. More than sufficient cathodic emission is still generated for use by the RPA circuits. The only failure mechanism will be expansion and contraction fatigue when the power is switched on and off. There will be a small percentage of tubes that fail earlier but most will survive indefinitely.

Another design feature that extends the tube life is the very low plate dissipation of an RPA circuit. Virtually all the heat generated by the tube is from the filament temperature alone. In conventional circuits, the plate power dissipation is considerable and that is what eventually causes a weak or gassy tube. In our RPA circuits, a well manufactured tube will probably never become weak or gassy even in 100 years of use.

Strangely, a benefit of how gracefully the RPA method embraces the tube function is that a tube testing too weak for a conventional circuit will probably work just fine in the Model 1100 MkII unless it has somehow developed excessive noise. You are probably asking, "How is that possible?"

The answer is that the Aphex patented Reflected Plate Amplifier takes full advantage of thermionic tube amplification at a relatively low plate voltage and current. With the RPA, the plate voltage is held essentially constant while the low plate current is extracted, reflected, and magnified by a linear current mirror. It is the reflected plate current that generates the output voltage. The tube runs much cooler yet still generates the traditional transconductance curve. In contrast, traditional tube stages have to operate at a high plate voltage to permit a useful signal swing to develop as a function of the tube's plate resistance, plate load resistor, and transconductance.

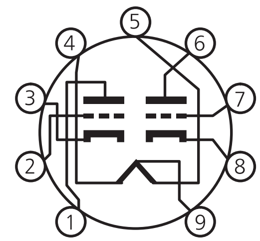
With the RPA, the tube's control grid receives the input signal and controls the electron flow by the law of the tube's transconductance curve just like a traditional stage. The tube's dynamics still exist, giving us the great tube sound from a much more efficient circuit.

8.6 Relevant Base Diagrams

The tubes that work in the Model 1100 MkII follow the industry standard base diagrams 9A and 9AJ. You will see that the only difference is the filament center tap at pin 9. We use the center tap provided on 12 volt tubes to allow their use with a 6 volt filament supply. When you change the filament switch from 6 to 12 volts, the two filament halves are placed in parallel, creating a 6 volt filament. This is a widely accepted practice having no deleterious effects.

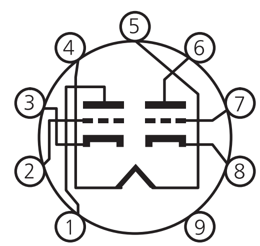
Tube Base Diagrams

12 Volt Tubes



9A

6 Volt Tubes



9AJ

More Gain. No Pain.

Synopsis: An extraordinary mic preamp combining new design philosophies allows you to safely run at higher gains without the pains of noise and overload distortion.

I. The Value of a Wide Dynamic Range

Consider dynamic range as a window. The top is the maximum peak level and the bottom is the noise floor. These are physical limits and they exist in the both the analog and the digital worlds. Ideally, the window is wide enough to accommodate the highest input level without any overload distortion while adding as little noise as possible to the signal. Wide dynamic range for a microphone preamplifier is particularly important inasmuch as the level of the input into the microphone can vary greatly. In order to accommodate these variations the gain in the preamplifier is set so there is no overload distortion on the highest peaks. The difference between that nominal gain setting and the maximum peak level is headroom. Setting the gain too low in the preamplifier, however, will require gain in a later stage. That means that any increase in gain in the later stage will also boost the noise from the preamplifier. Obviously, the lower the noise floor in the preamplifier, the lower the noise on the final output. If the output of the preamplifier is digitized at too low a level, the conversion will have low resolution. One bit represents 6dB of dynamic range in the digital domain. If the input is converted at -24dBfs the resolution will be four bits less than full resolution. Once the signal is converted there is no way to increase the resolution.

II. Setting Up a Conventional Microphone Preamplifier

A microphone is almost always used to pick up a live acoustic source, e.g.- a voice, an instrument, or ambient sound. Since level variations from these sources can be quite high, it is imperative that a great amount of headroom be set in the conventional preamplifier. This reduces the chances that the preamplifier will be overloaded due to an unexpected increase in input level, but the nominal output level will consequently be very low.

That low output level will have to be boosted in a following gain stage. This, however, shifts the problem of overload distortion to the following stage. That is why it is quite common to see a compressor or a limiter in between the preamplifier and the following stage.

The problem of noise build up, however, becomes quite apparent. As mentioned above, any gain taken on the signal after the preamplifier increases the noise from the preamplifier by the amount of gain in the second stage. In addition, the noise of the second stage itself combines with the input noise. For example, if the noise of the preamplifier is -60dBu and the noise of the following stage is -60dBu with 10dB of gain, the noise at the output of the second stage will be -57dB. Note: When two equal non-correlated noise sources are summed the noise is increased by 3dB.

When a compressor is used, it brings up the lower level signals (including noise) by whatever make up gain is set in the compressor. Adding to the noise in the output of the compressor is the noise of the compressor itself.

As you can see, noise builds up very quickly if the dynamic range of each gain stage is not maximized. That is why it is essential to choose the equipment with the widest possible dynamic range and use that equipment properly. And the most important gain stage is the first gain stage- the microphone preamplifier.

III. Determining the EIN and the Dynamic Range of a Microphone Preamplifier

A very important specification for any microphone preamplifier is the equivalent input noise (EIN). The noise is measured with the input shorted and at a specific gain. That figure is added to the gain. For example, a preamplifier with 60dB of gain has a noise floor of -68dBu. Adding the noise to the gain gives that circuit an EIN of -128dBu.

The dynamic range of that preamplifier at that gain setting is computed by adding the noise and the maximum output level. For example, if the preamplifier has a maximum output level of +27dBu the dynamic range of the preamplifier is 95dB (68 + 27) at that gain setting.

IV. Designing the Aphex Model 1100 MkII Tube Microphone Preamplifier

1. Noise

One of the primary design goals of the Model 1100 MkII was to have as wide a dynamic range as possible. Several key inventions combined with a no-compromise selection of components (see below) create a microphone preamplifier with unprecedented performance. The EIN with 65dB of gain is an incredible -135dBu. That means that the Model 1100 MkII adds less that 1dB of noise to the natural self noise of a 150 ohm microphone. The worst case dynamic range is 97dB and is a high as 101dB. But low noise is only part of the story.

2. 20dB extra headroom

As described above, a conventional preamplifier must be set for sufficient headroom in order to avoid overload. The Model 1100 MkII has two inventions that actually provide up to 20dB of extra input headroom so that it is virtually impossible to overload the preamplifier. A third invention does not directly increase headroom, but maximizes available headroom in the digital domain.

a. MicLim

The first invention is the Microphone Limiter (MicLim), first used on the Model 1788. It comprises a custom designed optical attenuator directly on the microphone input line. It smoothly limits the microphone output signal prior to the preamplification by up to 20dB. The peak limit detector is located after the preamplifier input stage and feeds a control current back to the attenuator so that the input signal remains below clipping. MicLim has no effect whatsoever on the input signal until the preamplifier's output approaches clipping.

b. Low Frequency Cancellation Filter (LoCaf)

The second invention is a tunable low frequency cancellation filter (LoCaf). It is a second order (12dB/octave) modified Butterworth filter meshed into the nodal intersections of the first and second amplifying stages in a servo configuration. The servo affects only frequencies below the corner frequency, thus it contributes nothing to the audible signal. Imposing the servo filter in such a manner gives the preamplifier about 20dB more overload headroom in the low cut range as compared to conventional techniques. Additionally, the added low frequency headroom eliminates the need for the MicLim to trigger earlier than necessary from excessive low frequency energy.

c. Drift Stabilized A/D Converter

Conventional analog to digital converters utilize high pass filters in the digital domain to block any DC generated in the conversion process or already in input signal. While this is effective in eliminating the DC, it requires extra headroom in the converter to allow for the DC offset. The patented drift stabilized A/D eliminates the DC offset in the analog domain so that the input peaks of both polarities can reach the true maximum level. Since there is no high pass filter in the digital domain, all ringing and time delays from that filter are also eliminated.

3. Other Features

a. Full Featured AES/EBU Digital Audio Output

AES/EBU XLR output is standard. Clock synchronization options allow for locking to standard "word clock" received at the standard BNC clock input jack. Internal clock options provide low jitter 44.1, 48, 88.2, 96, and 192kHz sample rates. When a unit is set for internal clock, its internal word clock reference is sent to the rear panel word clock BNC output jack to serve as clock reference to other units. When the unit is set for an external clock reference, the clock input BNC jack is directly tied to the output clock BNC jack for easy daisy-chaining of model 1100 MkII units from the master clock source. All digital audio settings are controlled and displayed on the front panel.

The A/D converter receives signal from the soft mute stage just prior to the analog output level control and triode output stage. This means you can use both the digital and analog outputs independently, with full and proper calibration of both regardless of the analog output level settings. The analog and digital outputs respond equally to the input gain, low-cut filter, and all front-end conditioning effects.

d. Bifurcated 20dB Pad and Phase Reverse

Many preamplifiers, even the more expensive models, switch microphone level signals directly through switch contacts. It is well known that even the best quality switches will eventually suffer from dry contact diode effects causing noise and dis-

tortion. The Model 1100 MkII uses high-grade, bifurcated (milti-finger) gold contact relays which do not develop these problems. The Gain and Low Cut controls are sealed gold contact rotary witches.

e. Precision Three Turn Output Level Attenuator

In order to match the analog output of the model 1100 MkII to the user's system level, the output gain is adjustable from zero dB (max gain) to -14dB. The user will appreciate the smoothness and precision afforded by the front panel 3-turn high-grade potentiometer adjustment.

f. 48-volt Phantom Power Circuit

Very slow rise and fall of the phantom voltage is used to eliminate turn-on and turn-off thumps. Industry standard resistances of 6.81 K-Ohms supply the highly filtered 48-volt source to pin 2 and pin 3 through a voltage ramping active buffer. The phantom powering system can withstand a short circuit to ground on both microphone jacks indefinitely.

g. Series-Shunt, Optical Soft Mute Attenuator

The second-stage output signal passes through a specialized series-shunt optocoupler circuit to provide a soft mute while introducing no distortion or noise.

h. Front Panel Peak Headroom Meter and Function Controls

Each channel contains a 20-segment LED headroom meter, making it easy to optimize the performance of the preamplifier. The headroom meter is calibrated in decibels below clipping, where 0dB is the analog clipping point. This coincides with the A/D converter's maximum input level, so the headroom meter also indicates the digital audio level accurately. Each channel contains its own independent controls over every function.

i. Rear Panel Mute Jack

The mute function may be activated by the front panel push-button, or by a remote switch plugged into the mute jack. In the absence of a phone plug, the mute jack serves as a closed circuit, and only the front panel push-button has control. In the presence of a phone plug, an open circuit mutes the preamplifier while a closed circuit un-mutes. This facilitates the convenient use of mute controls such as floor mat switches (step on to un-mute) and musician's footswitches.

j. Internal Switchmode Low Noise Power Supply

One of the technology improvements comprising the model 1100 MkII is a highly efficient switchmode/linear hybrid power supply. Switchmode design is used to very efficiently convert the a.c. primary power into secondary d.c. voltages. Great effort was made to eliminate switching noise and we phenomenally succeeded. All preamp supply voltages are further stabilized through linear regulators to maintain the highest audio performance. The tube filaments are run from regulated d.c. for extremely long life. The whole power supply is magnetically shielded. With this new power supply design, the 1100 MkII is lighter, runs cooler, and performs better.

V. Sound Quality by Design

While specifications and functions are important, the most important characteristic of a piece of audio gear, particularly a microphone preamplifier, is how it sounds. The sound of the Model 1100 MkII is clean, clear, present, open and solid. It is extraordinarily detailed and spacious. The low end stands up without any muddiness and the high end is very extended without any harshness.

This sound is achieved through the use of proprietary designs, careful engineering and the highest quality components. It is a combination of Class A discrete components and patented tube circuitry as briefly described below.

1. Ultra Low Noise, Transformerless, Discrete Class A, Bipolar PNP, Variable Gain Differential Input Stage

No outer feedback is used, thus eliminating the possibility of any dynamic interaction with the microphone's self-impedance. The input impedance remains passive, providing an optimal load for any microphone. The solid state, class-A PNP bipolar design achieves high common-mode rejection with extremely low noise, wide bandwidth, and low distortion at all gain settings. Robust input overload protection assures that all performance features will be retained indefinitely. The gain is adjustable in 4dB precision steps from 21 to 65dB.

2. Tube, Discrete Class A Differential Second Stage

The unique, Aphex patented, "Reflected Plate Amplifier" tube circuit is configured as a single-triode differential opamp to further enhance the preamplifier's common mode rejection. This novel circuit topology subtly imparts the tube's sonic warmth and character while retaining relatively long and stable operating life.

3. Tube, Discrete Class A Output Stage

An Aphex patented "Reflected Plate Amplifier" tube circuit is configured as a low distortion triode buffer having a very low output impedance and high output current drive. The maximum output level of +27dBu meets the needs of any professional application. Matched-impedance balancing assures peak performance whether driving balanced or unbalanced lines. A rear panel switch is assigned to insert a 12dB low impedance pad into the output line for systems based on IHF (semi-pro) operating levels. Rear panel XLR and quarter-inch phone jacks are both provided for balanced output.

VI. Summary

Every circuit and component that went into the Model 1100 MkII was studied and scrutinized for optimum performance. The result of the innovations and careful engineering is a uniquely excellent preamplifier.

Word Clock Cabling

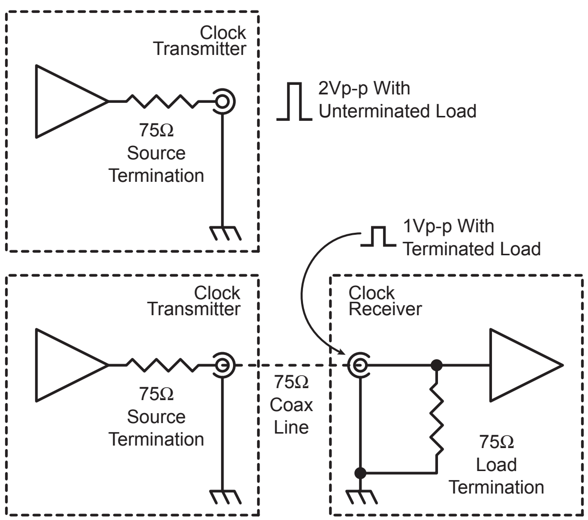
Transmission Systems

There are basically two systems in use for distributing word clock by coaxial lines: 75Ω matched, and unmatched with unspecified impedance. We sometimes call the unmatched method "brute force". You'll see why as you go through this tutorial.

Matched Coaxial Transmission

This is by far the better way to send pulse shaped waveforms over a long distance. Coaxial cable is manufactured to several standard impedance characteristics. The most common are 50 and 75 ohms (Ω). 50Ω cable has traditionally been popular around radio transmission apparatus, but television has long adopted 75Ω cable to carry video signals. Industry has also adopted 75Ω for the lead-in and internal house wiring for cable TV. More recently, the AES adopted, by standard AES3-ID 2001, 75Ω cable for distribution of digital audio.

Figure 1
Principle of 75Ω Matched Transmission



For practical purposes, transmission line theory need not be studied at great length. The basic idea is very simple. To create a matched transmission line, you terminate both ends of the line by a resistance equal to the line's own characteristic impedance.

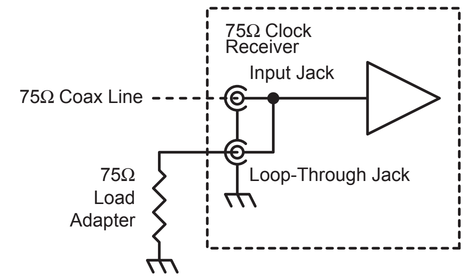
This configuration causes all the power launched from the sending end to be absorbed at the receiving end. The useful bandwidth of the line is maximized under these conditions, and the pulse waveform is best preserved.

That is not to say the pulse waveform won't undergo some degradation. All matched transmission lines have losses, especially at high frequencies. When the cable exceeds a certain length, depending on the cable's quality, cable equalization may be required to recover additional bandwith. Nevertheless, a general rule is that matched transmission lines will always have better bandwidth and less jitter than unmatched lines of equal length.

It should be noted from figure 1 that when a matching line driver is not loaded by the line, its output voltage rises by double. This is something to take into account when measuring or specifying word clock output levels.

A slight revision of figure 1 is shown in figure 2. The internal termination resistor is replaced by a "loop-through" jack and an external load adapter. The principle is the same, but the loop-through jack allows daisy-chaining of many clock recipients from a single clock source over a single clock cable. As you could imagine, this can greatly simplify clock distribution among compatible equipment.

Figure 2
75Ω Transmission Loop-Through Jacks



Daisy-Chain Example

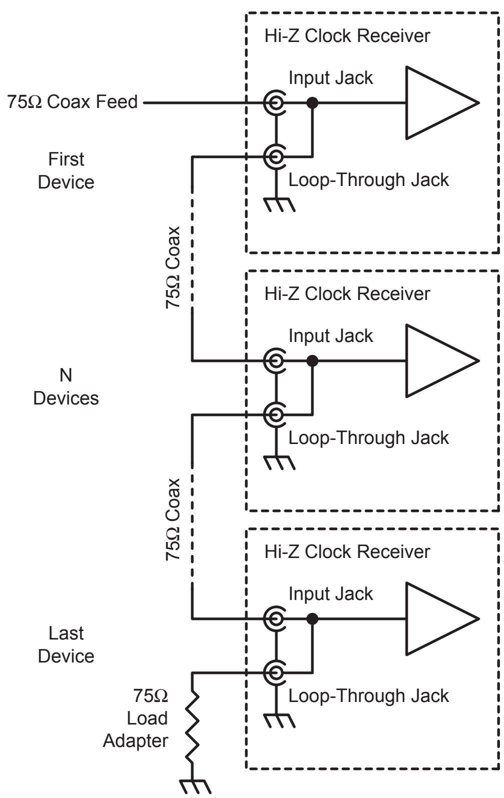
Figure 3 expands upon figure 2 to show the principle of daisy-chaining a matched transmission. Note that only the last unit in a daisy-chain needs a load adapter to properly terminate the 75Ω line.

Unmatched Lines

In the simplest context, an unmatched line is one that is not terminated in its characteristic impedance. In most cases, unmatched word clock lines are driven by a low output impedance voltage driver and loaded by a high impedance receiver.

A length of coax driven by a brute forced output driver will cause edge ringing and wave distortion of the clock pulses. If the length is too long, the distortion can cause clock jitter. It is important to minimize the cable length.

Figure 3
Transmission Daisy-Chain



Brute Force Clocking

It is an unfortunate fact that transmission line techniques for word clock have not been implemented in much of the digital audio equipment presently available. This leads to real clocking compatibility problems in today's studios. Obvious problems are usually seen as "no lock" but less obvious is sound degrading jitter.

Though ugly from an engineering viewpoint, the only solution is such cases is to apply unmatched transmission lines as best as one can and then rely on "brute force" clocking to hold it all together.

With brute force clocking, 4-5Vp-p clock pulses are fed out from a low impedance line driver with no intention of impedance matching the coaxial cable. Daisy chaining is not usually possible because the magnitude of jitter rapidly rises. Very short cable runs are necessary. Every piece of gear gets a separately driven feed, and no terminating load is used. Figure 4 above illustrates the Brute Force clock principle. We have found that many of the commercially available clock detangler and regenerator/driver boxes actually source out brute force clock as their hedge against unknown studio equipment characteristics.

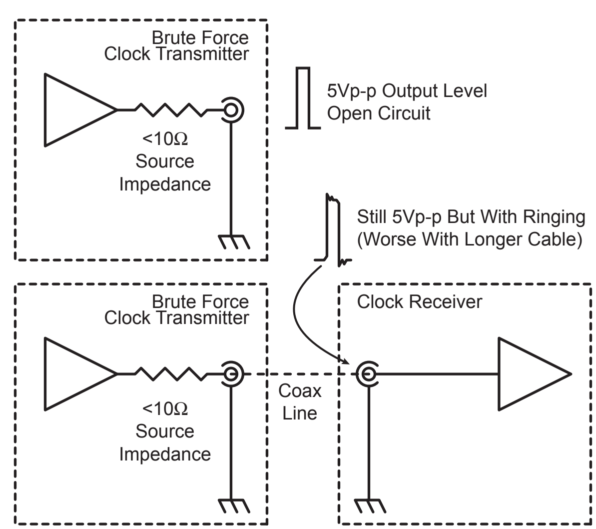
Clocking The 1100 MkII With 75Ω Matched Lines

First, you need to be aware that when switched to the external clock (W.C.) mode, the 1100 MkII automatically connects the word clock input jack to the word clock output jack through a metal relay contact. This permits direct feed-through of clock with almost no loading from the 1100 MkII. If using a 75Ω, 1Vp-p clock system, then you can daisy-chain the W.C. input to additional 1100 MkII's or other equipment. If the 1100 MkII is the last unit in line, the best option for terminating the line as shown in figure 5.

Clocking The 1100 MkII With Brute Force

Not knowing what the 1100 MkII's word clock input may be driven by, we made it automatically accept the brute force voltage levels.

Figure 4
Principle of Unmatched Lines
And Brute Force Clocking

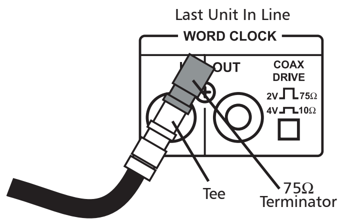


When accepting brute force clock, do not install a termination adapter - it would be of no benefit. When the driving source is not matched to the line impedance, then adding a load impedance will not improve

the transmission characteristics.

Maximum Brute Force cable length depends on several factors, but you probably shouldn't attempt distances greater than 4 feet.

Figure 5
End-of-Line Termination



NOTE ON CONVERTING TO A MATCHED LINE: because of the 1100 MkII's automatic input level adaptation, you can convert a brute forced clock source into a workable matched line by inserting an in-line 75Ω, 10dB pad **at the sending end of the cable**, and then installing a 75Ω termination adapter at the receiving end. This may reduce or eliminate jitter problems when the word clock distribution cable is of appreciable length

Typical 75Ω Attenuator

Notes