MAKING YOUR WORLD SOUND BETTER

CHANNEL™

MASTER PREAMP & INPUT PROCESSOR

THE COMPLETE CHANNEL STRIP FOR VOICE AND INSTRUMENTS

OWNER'S MANUAL

a DWV ENTERTAINMENT company

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.

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1.0 Controls and Indicators

POWER INDICATION
The Aphex logo above the power switch glows yellow during the tube preamp's warmup time and green when the unit is ready.

LOW NOISE MIC PRE
The Channel works with all professional microphones. The clean and stable 48V phantom supply is suitable for even the most expensive microphones. The 12dB/octave low cut filter starts at 70Hz to effectively reduce wind blast without diminishing voice body. Included is a selectable phase rotator following the principles of the Spectral Phase Refractor (SPR) circuit. The main bebefit of the Phase Rotator is that by making certain voice waveforms more symetrical, the compressors action becomes more efficient. This allows for a more optimized output level.

COMPRESSOR
The Channel includes a special version of Aphex's patented Easyrider™ Compressor which has proven to be outstanding. It is very simple to use with only two controls: Gain/Drive and Release. To get deeper compression, turn up more Gain. To manage the density and loudness, work with Release. Faster is louder and denser. Slower is more natural and open. Note that turning up the Gain/Drive to get more compression alters the overall gain structure through the device such that the output may need compensating adjustment.

LOGIC ASSISTED GATE™
Once triggered, even by a microscopic transient, the signal progresses fully through the attack, hold, and release sequence. This virtually eliminates the chatter experienced with other gate products. The attenuation depth and gating threshold are user adjustable to allow for varying requirements.

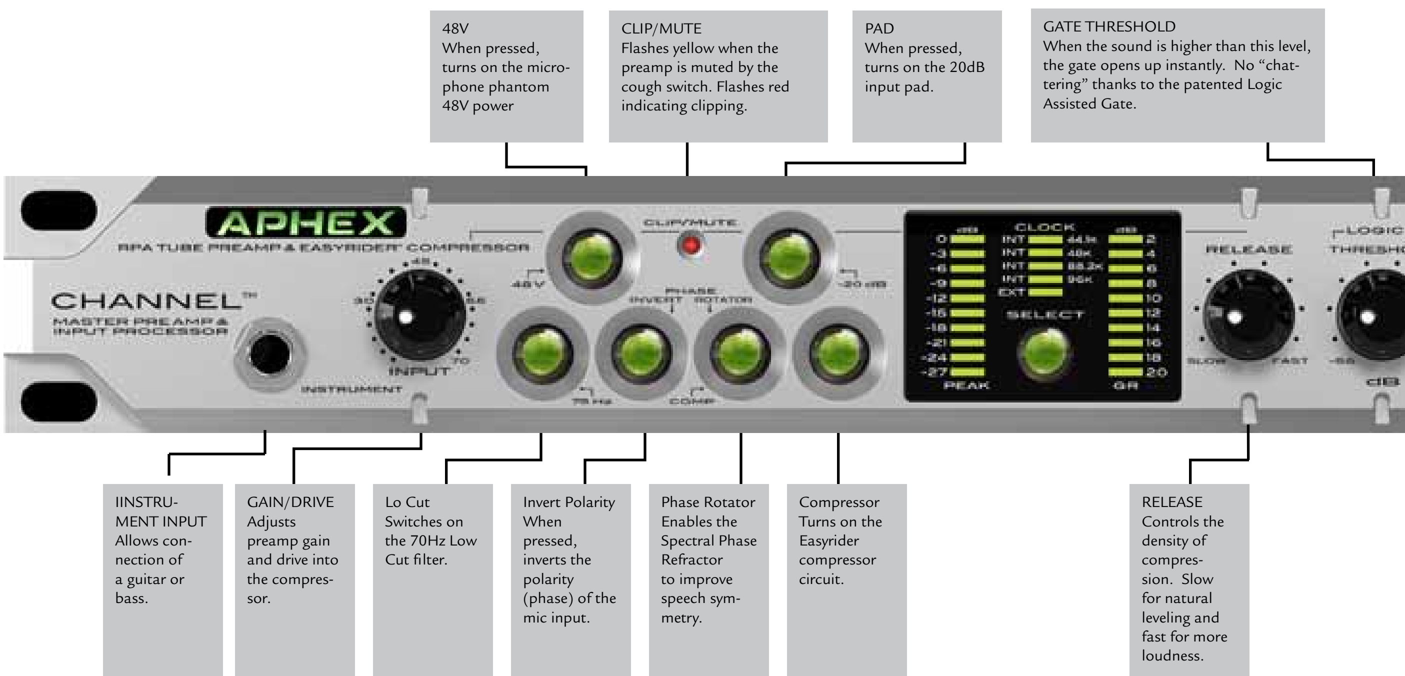
DE-ESSER
Using a split-band technique with Linkwitz-Riley crossovers, the voice remains bright and sharp, never losing presence while de-essing. The de-ess threshold control lets you chose the essing level that you want.

EQUALIZATION BLOCK
A fully parametric equalizer band is provided along with the popular Aphex Big Bottom and Aural Exciter. These features increase power, punch and intelligibility without adding noise or an increase in output level. As a real plus, they are also very easy to adjust.

REAR INSERT JACK
Allows you to insert any kind of line level audio equipment into the signal path between the Channel's dynamics processing and the equalizer section. The operating level at this jack is 0dBu. You should set up your inserted outboard gear accordingly.

METERS
Meters are provided for peak output level (relevant to both analog and digital headroom at 0dB = max before clip) and the compressor's gain reduction. The output level bar graph will move upwards indicating level, and the gain reduction bar graph will move downwards indicating gain reduction.

OUTPUT CONTROL
Once all processing and equalization is set where you want it, the output level may need to be adjusted. Do not let the peak level frequently exceed -6dBFS. This will make sure the digital output carries well through subsequent mixing and processing. When using the analog output, adjust the level to produce the desired average output level (0VU) as seen on the outboard equipment's meters.



Rear Panel Analog Connections

MIC INPUT
Works with all standard microphones, phantom powered or passive. Phantom power activated from front panel.

BALANCED
TRS

Line Level
Processor,
i.e.,
Compressor, EQ

+4dBu
Balanced
XLR Output

COUGH
SWITCH
Allows you to silently mute the output on the fly using a handheld, desktop, or floor switch.

DIGITAL AUDIO
The Channel supplies industry standard stereo AES/EBU, S/PDIF, and TOSLINK digital audio outputs. The single mic channel appears on both stereo channels as a mono signal. Sample rate and word clock options are all available at the front panel. The word clock selection LEDs will flash when no clock signal is present and light solid when locked.

Rear Panel Digital Connections

AES/EBU
OUTPUT
Digital 110 Ohm cables only. (Do not use regular audio cables.)

S/PDIF &
OPTICAL
OUTPUTS
All outputs are active at the same time.

WORD CLOCK
BNC input and output jacks.

DEPTH
Lets you set how much attenuation the gate will deliver. Use the lowest attenuation that's needed to reduce room wetness, breath, or background noise. Usually 6 to 15dB is sufficient. Fully CCW is off.

PARAMETRIC EQ
Works like a standard parametric band with tune, Q, and boost/cut controls.

Why You Shouldn't Connect the Channel's Output to a Mic Level Input.
The bottom line: you will get an unsatisfactory noise level. The Channel is designed to generate a line level output from a mic level input. It optimizes the signal to noise ratio by giving you a strong signal far above the preamplifier's basic self noise level. This strong output signal can be up to 65dB higher than the mic signal. It is perfect for a standard line input of a mixer or console but can overload a mic input. If you have no other choice but to plug the Channel in to a mic input, turn that mic input's gain all the way down. If the signal still clips the input, try bringing down the Channel output level.

GATING
Shows when the gate is closed or closing. When the light is off, the gate is fully open.

DE-ESSING
Shows when the the de-esser is working. When the light is on, sibilance is being controlled.

DE-ESS
THRESHOLD
Lets you set the level where you want your esses to level off.

BIG-BOTTOM
FREQ
Adjusts the frequency below which enhancement takes place.

BIG-BOTTOM
AMOUNT
Adjusts the strength (boost) of the Big Bottom effect.

AURAL EXCITER
FREQ
Adjusts frequency above which enhancement takes place.

AURAL EXCITER
AMOUNT
Adjusts the strength (boost) of the Aural Exciter effect.

BB/EQ/AX
ON
Switches the BB/EQ/AX on/off. Provides full true bypass of the circuits when off.

OUTPUT
Adjusts the final processed output level as seen on the VU meter.

2.0 Installation & Interfacing

2.1 INSTALLATION

The Channel occupies a single rack space (45mm or 1-3/4 inches) of a standard EIA equipment rack.

When rack mounting, use appropriate cushioned rack screws. Never restrict air flow through the device's vents. When installing the units into a rack, distribute the units evenly. Otherwise, hazardous conditions may be created by an uneven weight distribution. Connect the unit only to a properly rated supply circuit. Reliable earthing (grounding) of rack mounted equipment should be maintained. Try not to position the Channel directly above devices that generate excessive heat such as power amplifiers (unless adequately ventilated) or near equipment with heavy transformer hum fields.

2.2 REAR PANEL VIEW

2.3 AC LINE CONNECTION

Use only a power cord that carries approvals for use in your location. The Channel's internal power supply is designed to operate from all nominal power sources from 100 to 240 volts a.c. at 50/60Hz without requiring the user to change any settings. In case of failure, do not attempt to change the internal fuse because it will never blow unless the power supply fails catasrophically. The power supply will need to be serviced by a competent service technician in such a case.

2.4 MIC INPUT CONNECTION

The microphone input connector is located on the rear panel. It is the standard XLR-3F type. Use only properly wired balanced mic cables.

PROPER MICROPHONE CABLE WIRING

CAUTION: Some ribbon mics will be damaged by phantom power, some ribbon mics require phantom power and some just ignore it. Please consult your ribbon mic's manual before connecting it to the Channel.

CAUTION: Beware that 48 volt PHANTOM POWER may be applied to the microphone input, creating a potential shock hazard. Shut off the phantom power before plugging or unplugging microphones, waiting at least 10 seconds for the voltage to fall. This also protects sensitive microphones against power inrush.

2.5 INSERT JACKS

The Channel allows you to insert additional signal processing between the Channel's dynamics processing and the equalizer block. Both Send and Return jacks are balanced and run at 0dBu. This is a perfect place to insert an external reverb unit or profanity delay, but you have the option to put anything there that you wish as long as it returns a nominal level.

Direct feed-through occurs with the normalling contacts of the jacks. If you plug into either the SEND or RETURN jack, the internal path is interrupted. You need to be sure you have a viable send-return circuit externally or there will be no audio output from the Channel.

There is no insert bypass switch. Once plugged in, the insert is always inserted. If you can't get any output from the Channel, make sure the inserted gear is operating.

2.6 OUTPUT CONNECTORS

There are two output connectors located on the rear panel: one 1/4" TRS phone type and one XLR-3M type. They can be used at the same time to feed separate equipment.

The output level at the XLR is +4dBu impedance balanced, while the TRS balanced/unbalanced jack is switchable between -10dBV (consumer level) and +4dBU (professional level).

If you intend to make an unbalanced output from the XLR jack, simply take "hot" from pin 2 and use pin 1 for ground. Leave pin 3 unconnected or grounded. Never ground pin 2.

DIGITAL OUTPUT DEFINITIONS	
XLR:	AES/EBU 110Ω @ 5Vp-p
RCA:	S/PDIF 75Ω @ 1Vp-p
OPTICAL:	Toslink S/PDIF encoded for optical fiber.

2.7 DIGITAL AUDIO OUTPUTS
The processed mic signal is converted to digital in both channels at equal level as a mono signal. There is no provision to externally drive one of the A/D Converter inputs.

2.8 WORD CLOCK

2.8.1 Clock Select
Synchronization is selected by the Clock Select button on the front panel. Sample rates of 44.1, 48, 88.2 or 96kHz are supported.

2.8.2 Word Clock IN
This BNC jack is provided to receive your master clock source. It will accept industry standard Word Clock, from less than 1 to over 5Vp-p pulse amplitude. It does not accept AES/EBU or Superclock.

2.8.3 Word Clock OUT
The Channel provides a very accurate internal word clock generator. When using the Channel as a clock master, the word clock signal is sent from the BNC word clock output. Note that the Channel does not support word clock THRU. External clock signals received at the BNC input are not routed to the BNC output. A word clock distribution device should be used in systems with multiple digital devices.

2.9 POWER SUPPLY

The Channel is internally powered from a standard IEC power receptacle on the rear panel. Be sure you use a power cord that is approved for use in your jurisdiction.

Soft-start, overload foldback limiting, and accros-the-line voltage spike protection is incorporated to protect the power supply from damage that might be caused by component failure or power line disturbances. If the internal fuse blows out, a catastrophic failure has occurred and simply replacing the fuse will not fix the problem. Due to the extensive protective measures used, it is highly unlikely a catastrophic power supply failure will ever occur. However, if it does, you should contact the factory or a competent service technician to affect repairs. There are no user serviceable parts inside.

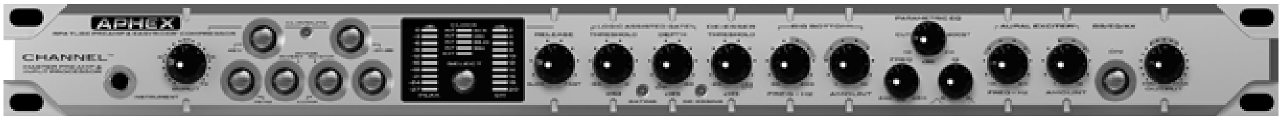
ACCEPTABLE POWER RANGE
100 to 240V~, 50 to 60Hz

2.10

Voice artists/actors often find it necessary to clear the throat, sip a beverage, or cough during narration. Thus, a convenient mute and unmute jack is provided. Any standard momentary or latching foot pedal or switch wired to a mono phone plug will work. Usually, a desktop box mounted silent pushbutton is preferred, as it can be accessed most readily.

3.0 Using the Channel

FRONT PANEL VIEW



3.1 USING THE MICROPHONE INPUT

The Channel is perfect for all types of microphones, either powered or not. We encourage you to try every mic you own with the Channel.

Many features of the Channel are standard with all professional preamps - polarity - pad - etc. Some features are unique and we hope you will fully exploit them.

3.2 USING THE INSTRUMENT INPUT

The Channel can also be used as a high quality direct box. When an instrument cable is plugged in to the Instrument input on the front panel, the rear panel XLR input is bypassed. This allows the user to keep a microphone connected to the rear panel and simply insert an instrument cable in to the front panel instrument input when D.I. functions are required. You will be able to acheive proper input levels no matter what kind of pickups are used. Single coil passive pickups will work just as well as active humbuckers.

3.3 USING PHANTOM POWER

Active microphones that take power through the standard mic cable fall into a class called “phantom powered” mics. The power is called “phantom” because it rides the mic cable invisibly, without interfering with the audio signal carried on the same wires.

The industry standard phantom power source is positive 48 volts d.c. supplied to pins 2 and 3 through precision low noise 6.81KΩ resistors. You may note from spec sheets that many mics rated for phantom power actually run at something less than 48 volts.   Don’t let these specifications confuse you. They all run perfectly well off the standard 48V phantom power source.

Plugging and Unplugging a microphone when phantom power is on can sometimes be dangerous. Some microphones can be damaged by power inrush. Switch off the phantom power first. Wait for the mic go silent before unplugging.

You should be aware of the shock hazard with the phantom power system. Long, open mic cables that are disconnected from the preamp while phantom is on can hold a d.c. charge for long periods of time, sometimes days, weeks or months. They will act as a storage capacitor and you can get shocked most rudely by holding the XLR plug and touching the pins inside. Also beware of microphone patch bays that may carry phantom voltage. Don’t hold the patch cord by the metal parts, only the plastic shell.

CAUTION: Some ribbon mics will be damaged by phantom power. Some ribbon mics require phantom power and some just ignore it. Please consult your manual before connecting a ribbon mic to the Channel.

3.4 USING THE POLARITY SWITCH
There will be times when you need to reverse the polarity (phase) of a mic signal. Vocalists monitoring themselves on headphones will hear a different sound when the phase is reversed. The reversed phase may sound fuller and more truthful or hollow and far away. That is because there is a cancellation of frequencies within the ear when the external sound from the headphone mixes with the sound directly conducted to the ear. “Flipping the phase” can make the problem either more or less noticeable.

Another time when phase reversing can be helpful is when using multiple microphones on the same source. For exapmple, using two microphones on a guitar speaker cabinet. You may get a “nasal” or hollow effect when both mics are on. Changing the polarity of one mic will often clear up the problem. It is always worth the time to experiment with mic polarity.

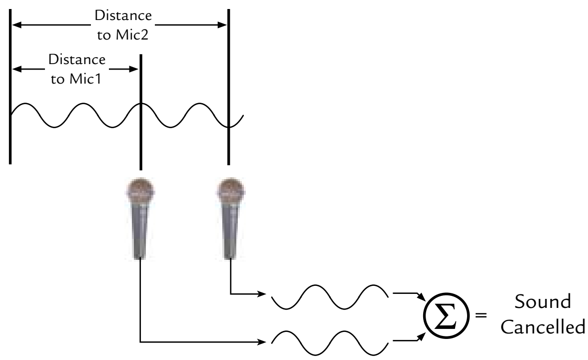


Figure 4-2 Phase Cancellation Effect

3.5 USING THE PAD

An input pad is nothing more than a resistive attenuator that drops the level coming from the input. Its purpose is to give you a way of preventing overload of the preamp when incoming signals become excessive.

In the Channel, we provided a pad of 20dB. That means when the pad is on, the net gain of the preamp is 20dB lower. There are times that the input may be overloaded even with the GAIN control set all the way down. For example, a bass player with active pickups and an agressive playing style might peak the instrument input. Or a microphone on a kick drum or snare drum could do it. Engage the pad and 20dB of headroom will be provided allowing you to better control the input signal.

3.6 USING LOW CUT

In the real world, mics pick up all sorts of unwanted low frequencies such as handling noise, wind rumble, or lectern thumps. We designed into the Channel a very effective way of cutting out these low frequencies. Switching on the LOW CUT FILTER rolls off all frequencies below 70Hz at 12dB per octave but places a slight compensation around 120Hz to improve the low end phase distortion and perception of remaining bass.

3.7 USING THE PHASE ROTATOR

The Channel's phase rotator is designed to help reduce high assymmetric peaks that can occur with certain audio waves. By reducing the amplitude of assymmetric peaks, the signal can ride louder through compressors and limiters.

Some words of advice: When doing voice work while wearing headphones, the Phase Rotator will affect how you hear yourself even though it is not affecting the actual sound of your voice. That's because the body-conducted sound mixes with the sound from the headphones. When the phase relationship of the two sounds changes, there will be partial to full cancellation at various frequencies. You should evaluate the phase rotator by auditioning recorded tracks made with and without the rotator.
It should also be noted that the effecs of the Phase Rotator are program dependant and results should be evaluated with any new signal.

3.8 USING THE COMPRESSOR

The Channel's compressor is very simple to use. Nevertheless, it is more sophisticated than compressors with many more controls. The Easyrider compression automatically adapts to audio waves in a manner that greatly reduces any pumping effect while it tightens the average level very flatteringly.

There is only one obvious control: RELEASE. However, the mic pre's GAIN control doubles as the compression drive adjustment. To get more compression depth, run up more gain. The RELEASE control allows you to chose the aggressiveness of the compression. For thick and loud, go faster. For more natural and "open", go slower.

The gain reduction is displayed on the Channel's 10 segment bargraph meter. You may need to boost the output level to compensate for the gain reduction of the signal.

3.9 USING THE LOGIC ASSISTED GATE

Aphex's Logic Assisted Gate solves many common workflow problems when using gates. The Channel's gate trigger is absolutely positive because it's independent from the energy content of the sound peak. The slightest exceeding of the threshold by the soundwave triggers the logic that forces the gate's attack-hold-release sequences to perform completely and repeatably. That makes finding the right threshold fast and easy. All you need to set are the threshold and depth of gating.

If you want complete silence between phrases, then use the maximum depth. However, if you simply want to bring down ambience pickup as with multiple open mics in a room, then use minimal depth. The Gating LED is triggered when the audio signal is below the Threshold. If the Threshold is turned all the way up, the LED may never light. If the Depth is turned all the way counter clockwise no gating will occur. However, because the LED is triggered from the Threshold the LED can be on even though the gate is inactive.

3.10 USING THE DE-ESSER

Certain mics are too harsh in the upper range and some sounds tend to whistle or splatter. Conventional de-essers simply detect the presence of any frequency above a certain tuning point and duck the whole signal accordingly. The Channel's de-esser is different. It uses split band techniques to attenuate only the sibilance, while leaving the body of the sound alone.
Operation of the de-esser is simple. Just set the threshold to the point where you want the esses to limit out. Reducing the threshold setting brings down the level of the esses dynamically. In other words, it's like an automatic downward shelving equalizer. It stays flat until the ess level gets too high and then introduces the shelf at the level needed to limit the sound to the threshold level. When there is de-essing, the LED is lit.

3.11 USING THE TONE ENHANCEMENT BLOCK

Once the signal passes through the compressor, gate and de-esser, it encounters the Big Bottom low frequency enhancer, parametric peak/dip section, and the Aural Exciter top end enhancer. The whole block is bypassable with the BB/EQ/AX on/off pushbutton.

3.11.1 Big Bottom
Some signals have no low bottom end. In such cases, the Big Bottom won't synthesize a new low end for you and it should not be used. However, voices that contain a deep chest resonance can be augmented by the Big Bottom.

Start by turning up the BB Amount to 12:00. Then adjust the BB FREQ to find a frequency that lifts the bottom. Last, reduce the BB Amount until just the right touch of bass enhancement is felt.

3.11.2 Parametric Equalizer
This is a familiar and conventional EQ section. You can adjust the boost/cut, frequency selection and Q.

3.11.3 Aural Exciter
Clarity, presence, and loudness can all be enhanced by the Aural Exciter.

Start with the AX Ampount at 12:00. Next, sweep the FREQ to find the best tonal balance. Presence is best augmented with lower settingss. Air is added with higher settings. Finally, readjust the AX Amount for the right amount of brilliance. Be conservative. Use the BB/EQ/AX on/off switch to compare the original signal to the enhanced signal.

3.12 USING THE OUTPUT LEVEL CONTROL

Once all the processing is set, the output level may need to be adjusted. Change the OUTPUT control to obtain peaks that don't go above -6dBFS on the output meter. Check the input meter of the device the Channels output is plugged in to. If the input device is cliping, first turn the input devices gain control down. If the input is still cliping, bring the OUTPUT level of the Channel down to compensate. Be sure that the operation level of each device is set properly at either -10dBV or +4dBu.

3.13 CLIP/MUTE LIGHT

If the LED is flashing RED, then the internal operating level is too hot. This can only occur if the insert return signal is too hot or if the parametric equalizer is boosted way too much. This same LED will also flash yellow while the Channel is in the MUTED state activated by the COUGH SWITCH (rear panel jack).

4.0 Warranty & Service

5.1 Limited Warranty

PERIOD
One year from date of original purchase.

SCOPE
All defects in materials and workmanship.

The following are not covered:

Voltage conversions
Units on which the serial number has been defaced, modified or removed.
Damage or deterioration resulting from: Installation and/or removal of the unit; Accident, misuse, neglect, unauthorized product modification; Failure to follow instructions in the Owner's Manual, User Guide or other official Aphex documentation; Repair or attempted repair by anyone not authorized by Aphex; Shipping damage – claims must be presented to the shipper
WHO IS PROTECTED
This warranty will be enforceable by the original purchaser and by any subsequent owner during the warranty period, so long as a copy of the original Bill of Sale is submitted whenever warranty service is required.

WHAT APHEX WILL PAY FOR
All labor and material expenses for covered items. Aphex will pay all 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 warrant stated above.

EXCLUSION OF CERTAIN DAMAGES
Aphex liability for any defective unit is limited to the repair or replacement of said unit, out our option, and shall not include damages of any 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 rights which vary from state to state.

4.2 SERVICE INFORMATION

If it becomes necessary to return this unit for repair, you must first contact Aphex Systems, Ltd. for a Return Authorization (RMA number), which will need to be included with your shipment for proper identification. If available, repack this unit in its original carton and packing material. Otherwise, pack the equipment in a strong carton containing at least 2 inches of padding on all sides. Be sure the unit cannot shift around inside the carton. Include a letter explaining the symptoms and/or defect(s). Be sure to reference the RMA number in your letter and mark the RMA number on the outside of the carton. If you believe the problem should be covered under the terms of the warranty, you must also include proof of purchase. Insure your shipment and send it to:

Aphex
11068 Randall Street
Sun Valley, CA. 91352
PH: (818) 767-2929 FAX: (818) 767 -2641

5.0 Specifications

5.1 GENERAL SPECIFICATIONS

INPUT		
	Connector:	XLR-3F
	Type:	Transformerless, NPN active balanced, tube second stage
	Input Z:	2KΩ nominal
	Instrument Connector:	1/4” TS
	Instrument Input Z:	10MΩ nominal
	Type:	Transformerless, NPN active balanced, tube second stage
	Maximum Input Level (MIL):	0dBu
	CMRR:	Greater than 70dB @ 60Hz
	Nominal Preamp Gain:	20 to 65dB
	Phantom Power:	+48VDC
	Pad:	20dB
OUTPUT		
	Connector:	XLR-3M and TRS 1/4” phone jack
	Type:	XLR is Impedance Balanced (may be used unbalanced); TRS is unbalanced.
	Output Z Balanced:	XLR: 66Ω
	Output Z Unbalanced:	XLR: 33Ω – TRS: 600Ω
	Nominal Level	XLR: +4dBu; TRS: -10dBV
	Maximum Output Level (MOL):	XLR: +25dBu Unloaded; TRS: +11dBV
COMPRESSOR		
	Attack/Release:	Program dependent, user variable release baseline.
	Ratio:	4:1
	Threshold:	Fixed
	Knee:	Medium Hard
GATE		
	Attack:	0.1 millisecond
	Hold/Release:	300 milliseconds/400 milliseconds
	Threshold:	Variable -50 to +20dB
	Depth:	Variable, 1 to 58dB
DE-ESSER		
	Attack:	0.1 millisecond
	Release:	100 milliseconds
	Threshold:	Variable -20 to +20dB
	Raio:	5:1
	Active Band:	4.KHz to 20KHz Linkwitz-Riley 24dB/octave crossover
INSERT		
	Connector Type Send:	1/4” TRS Phone Jack, Balanced
	Connector Type Return:	1/4” TRS Phone Jack, Balanced
	Nominal Operating Level:	0dBu
	Point of Insertion:	Between dynamics processing and equalizers.
BIG BOTTOM		
	Frequency Tune:	50Hz to 280Hz
	MIx:	OFF to +12dB
PARAMETRIC EQ		
	Frequency Tune:	240Hz to 8KHz
	Peak/Dip:	+/- 12dB
	Q Range:	0.5 to 5
AURAL EXCITER		
	Frequency Tune:	500Hz to 5KHz
	Mix:	OFF to +12dB
ANALOG AUDIO		
	THD:	<.01% @ +4dBu Out
	IMD:	<.01% @ +4dBu Out
	Freq Resp (FLAT):	18Hz to 24KHz +/- 1dB
DIGITAL AUDIO		
	Internal Sample Rates:	44.1KHz, 48KHz, 88.2KHz, 96KHz
	External Sample Rates:	Automatically syncs to any word clock between 32KHz and 96KHz
	Resolution:	24 Bits
	Word Clock Input:	BNC Jack, High Z, Captures <1Vp-p to 5Vp-p
	Word Clock Output:	BNC Jack, 75 Ohms, 5Vp-p
	Dynamic Range:	Digital dynamic range greater than analog front end.
	Noise Dither:	Dithered by analog preamp noise floor. Equivalent to 16-bit digital audio dither.
	Level Equivalency:	-20dBFS Digital = +4dBu Analog
OTHER SPECS		
	Power requirements:	85 to 260V~, 50-60Hz
	Power Consumption (maximum):	12 Watts
	Dimensions:	19” W x 1.75” H x 8.25” overall depth (482.6mm W x 445mm H x 209.6mm overall depth)
	Depth Behind Front Panel:	7.5” (190.5mm)
	Net Weight:	Rack-mounted: 6lbs. (2.73kg)
	Shipping Weight:	9lbs. (4.1kg)
	Tube Type:	12AT7/ECC81 Dual Triode

All specifications are subject to change without notice.

5.2 ARCHITECTURAL SPECIFICATIONS

Basic Description

A single channel processor comprising microphone and nstrument inputs, transformerless tube microphone preamp, a dynamics processing section and a tone controlling section, in that order. An insertion path shall be provided between the dynamics procesisng section and the tone controlling section.

The microphone preamp shall comprise the following selectable functions: 1.) +48VDC Phantom Power; 2.) Polarity Reversal; 3.) Selectable 20dB Pad; 4.) Selectable 70Hz 12dB/Octave Low Cut Filter; 5.) Continuous Gain Control; 6.) Phase Rotator.

The dynamics processing section shall comprise an adaptive dynamic range compressor, a logic assisted noise gate, and a split-band de-esser.

The tone control section shall comprise a Big-Bottom bass enhancer, an Aural Exciter treble enhancer, and a single band parametric equalizer.

Physical Properties

The device shall be packaged in an all metal chassis measuring 19” (482.23mm) wide, 1.75” (44.42mm) high, with an overall depth of 8.25” (210mm). Depth behind the front panel shall be approximately 7” (178mm). The device shall have a net weight of approximately 6lbs. (2.73kg) and is capable of mounting in a standard electronic equipment rack.

Power

The unit shall have a self contained power supply operating from the ac line. Primary voltage, connectorization and agency listings shall be appropriate to meet local requirements.

Patent Notice

This product is protected under one or more of the following Aphex patents.

4,578,648
4,633,501
4,843,626
4,939,471
5,115,471
5,155,769
5,334,947
5,359,665
5,422,602
5,424,488
5,450,034
5,463,695
5,483,600
5,485,077
5,612,612
5,737,432
5,848,167
5,896,458
5,898,395
5,930,374
6,266,423