Automated Processes, Inc.

Recording & Mixing Console Operator’s Manual

10-01-13

Written for Automated Processes Incorporated
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Important Safety Instructions

1. Please read these instructions
2. Keep this Information in a safe place
3. Do not use this console near water
4. Clean only with a dry cloth
5. Do not block any ventilation openings
6. Do not install near any heat sources such as radiators, heat registers, stoves, or other devices (including the power supply) that produce heat
7. Do not defeat the safety purpose of the polarized or grounding type AC plug
8. Protect both the AC power cord to the power supply and the DC cable between the supply and the console from being walked on or pinched
9. Use only attachments/accessories specified by the manufacturer
10. Unplug this device during lightning storms or when unused for long periods of time
11. Refer all service to qualified personnel

ATTENTION: Exposure to extremely high noise levels may cause permanent hearing loss or damage. Individuals vary considerably in susceptibility to noise-induced hearing loss, but nearly everyone will lose some hearing if exposed to sufficiently intense noise (this may include music) for a period of time. Be safe.

WARNING – To reduce the risk of fire or electric shock, do not expose this apparatus to rain or moisture.

1.0 Overview

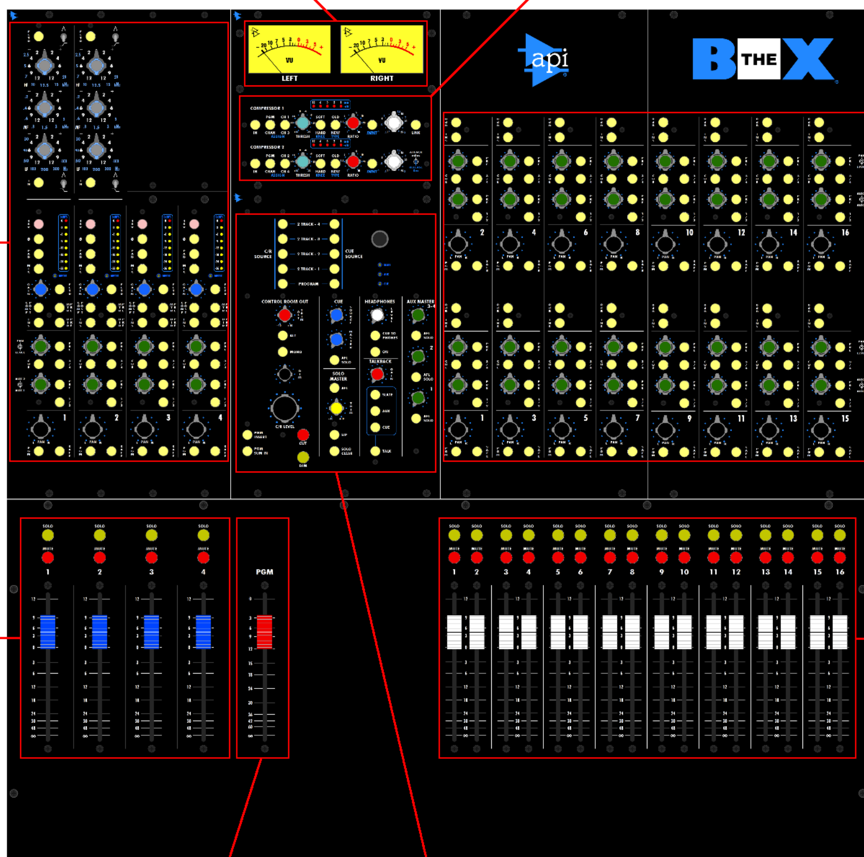
Building on API's rich heritage of extremely high-quality recording consoles, we introduce "The BOX," a small-format recording/mixing console designed for professional project studios, home studios, and production facilities of all kinds. Optimized for the digital era, "The BOX" handles all the functions needed for production not provided by most DAWs, including a preamp, input signal processing, high-quality mix bus, cue sends with talkback, monitor control, and more, without the redundant capacities of larger consoles. Most importantly, "The BOX" provides the legendary "all discrete" API sound in an efficient, cost-effective package.

Features

- Two (2) input channels with mic/instrument/line preamp, HP filter, & integral 550A EQ
- Two (2) input channels with mic/instrument/line preamp, HP filter, & 500 slot
- Two (2) compressors assignable to input channels or program bus with stereo link
- Sixteen (16) summing inputs (20 channels during mix)
- Stereo program bus with master fader, insert, and external input
- One (1) stereo and (2) mono auxiliary sends/buses
- Stereo cue send/bus & headphone system

- PFL, AFL, and solo-in-place solo modes with stereo solo bus
- Full-featured monitor section that supports two stereo monitor systems
- Talkback system
- Comprehensive rear panel connections with balanced inputs and outputs
- Integrated power supply

Console Layout



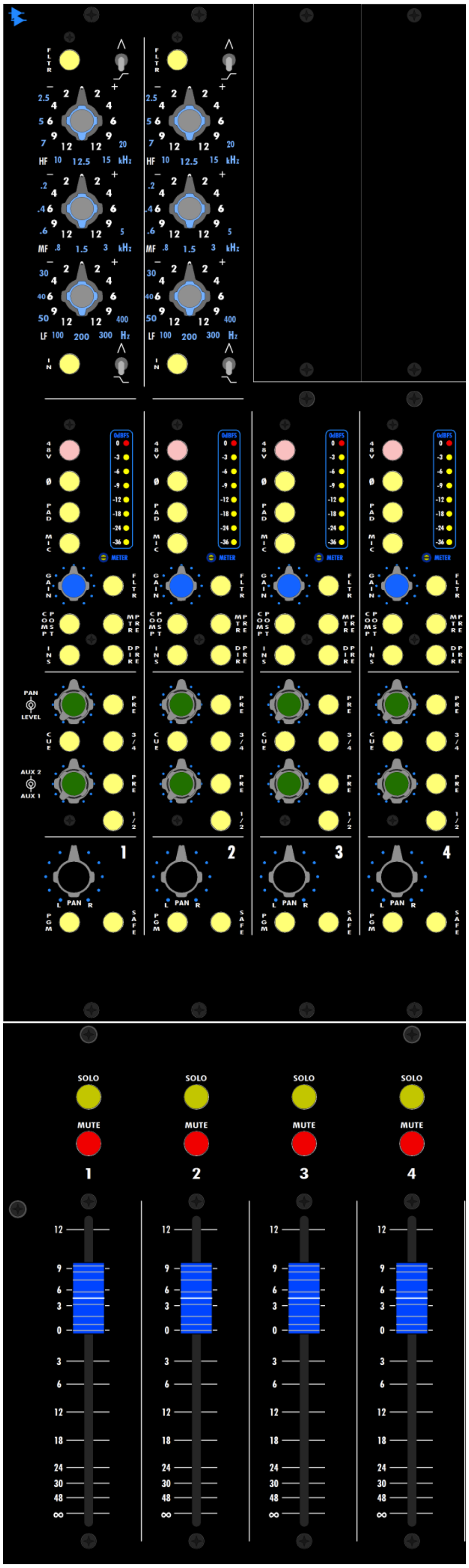
For recording, "The BOX" provides four (4) input channels. Channels 1 & 2 are equipped with preamp, filter, built-in 550A equalizer, insert, assignable compression, and output routing. Channels 3 & 4 provide the same features, but are each equipped with an API 500 series slot instead of the built-in EQ.

For mixing, “The BOX” provides sixteen (16) dedicated summing input channels, as well as the four (4) input channels, for a total of twenty (20) channels. Accordingly, all channels, input and summing, have a fader, mute, pan-pot, program bus assignment, insert, solo, solo safe, and aux/cue sends. Additionally, summing inputs have a “0dB” switch that bypasses the fader and delivers the input signal to the pan-pot at unity gain. This feature is very useful for summing individual DAW tracks that have fader & mute automation.

“The BOX” is equipped with a comprehensive master section that includes controls for program bus insert and external inputs, auxiliary and cue masters, solo master, talkback, VU meters, and a fully-featured monitor section that supports two (2) stereo speaker systems and headphones. In addition, the master section contains two (2) 527 compressors that can be assigned to the input channels when recording and the program bus when mixing.

To help assure reliability and longevity, “The BOX” is built to the same exacting API quality standards as our Vision, Legacy Plus, and 1608 consoles.

2.0 Input Channels 1-4



“The BOX” provides four (4) input channels that are designed primarily for recording, but can be used as additional channels while mixing.

All four channels are identical except for equalization. Channels 1 & 2 have a built-in 550A equalizer. Channels 3 & 4 have a 500 series slot, instead of the built-in EQ. The slot can accept any 500 series module that conforms to the VPR 500 Alliance standards. http://apiaudio.com/vpr_alliance.html

Together with the channel fader, each input channel provides a single, comprehensive audio path for recording and mixing applications. Features on all four input channels include:

- Microphone/instrument/line preamplifier
- Assignable compressor*
- Integral 550A EQ or 500 slot
- High-pass filter
- Balanced insert send/return
- Two (2) mono auxiliary sends
- One (1) stereo aux/cue send
- Fader and mute
- Pan-pot
- Solo and solo safe
- Direct or preamp output routing
- Program bus assignment

*The onboard compressors can be assigned to only two (2) input channels at a time

All aux/cue sends have ON/OFF switches and can be routed either pre* or post-fader.

*An internal jumper determines the source of the aux/cue feed when the PRE switch is engaged. Refer to section 2.5.4 Aux Pre Jumper for detailed information.

The channel output can be routed as follows:

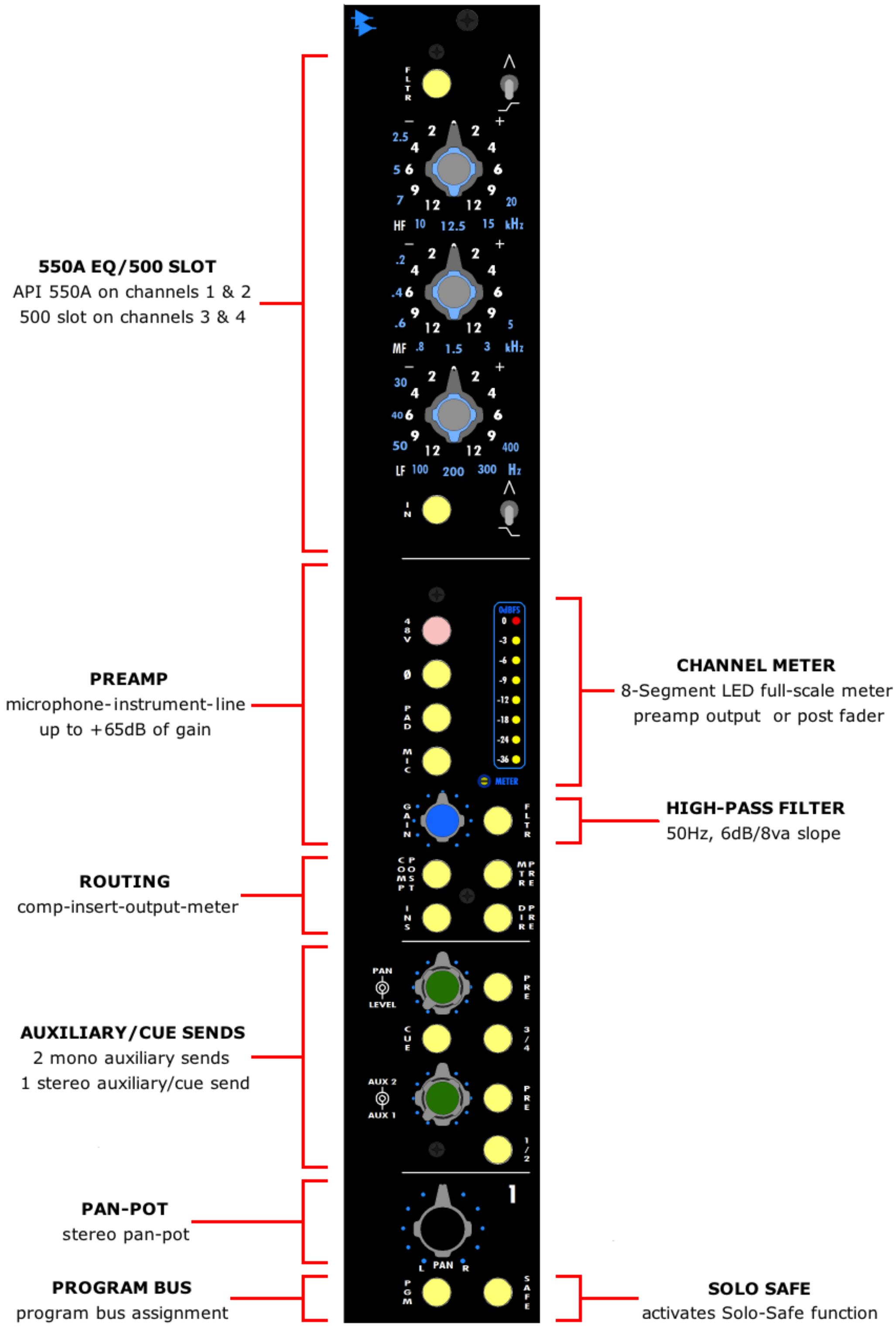
- Direct output (post-fader)
- Preamp output (pre-processing)
- Program bus (post-pan-pot)

The direct or preamp outputs as well as other outputs appear at the DIRECT OUTPUTS/PRE OUTPUTS DB-25 connector on the rear panel for interfacing with a DAW or other recorder.

The basic signal flow through the input channels module is as follows:
Preamplifier ⇒ compressor ⇒ EQ/500 slot ⇒ insert ⇒ fader/mute ⇒ filter ⇒ output

The compressor can be moved to after the insert (COMP POST) to yield:
Preamplifier ⇒ EQ/500 slot ⇒ insert ⇒ compressor ⇒ fader/mute ⇒ filter ⇒ output

Input Channel Sections



2.1 Preamplifier

The preamplifier section of the input channels provides a fully featured input stage that’s identical to the channel preamps in the API 1608. The preamp can accept the following input signals:

- Microphone
- Instrument
- Line

Microphone Input: Accepts balanced, low-impedance, microphone-level signals from the MIC INPUT female XLR connector on the rear panel. The preamplifier has the following features:

- +35dB of gain (+30dB to +65dB)
- -20 dB Pad
- Polarity Inverter (phase reverse)
- 48v Phantom Power

Note: The MIC switch must be engaged to use this input.

Note: The signal from the MIC INPUT is replaced with the signal from the Instrument Input when a 1/4” plug is inserted in the Instrument Input jack.

Line Input: Accepts balanced, low-impedance, line-level signals from the LINE INPUT connector. The Line Input has the following features:

- -6dB Pad
- Polarity Inverter (phase reverse)

Note: Line Input is the default input channel source and is available when MIC is not engaged.

Instrument Input: Accepts unbalanced, high-impedance, instrument-level signals from the ¼" INSTRUMENT INPUT connector. The Instrument Input serves the function of an external direct injection (DI) box and has the following features:

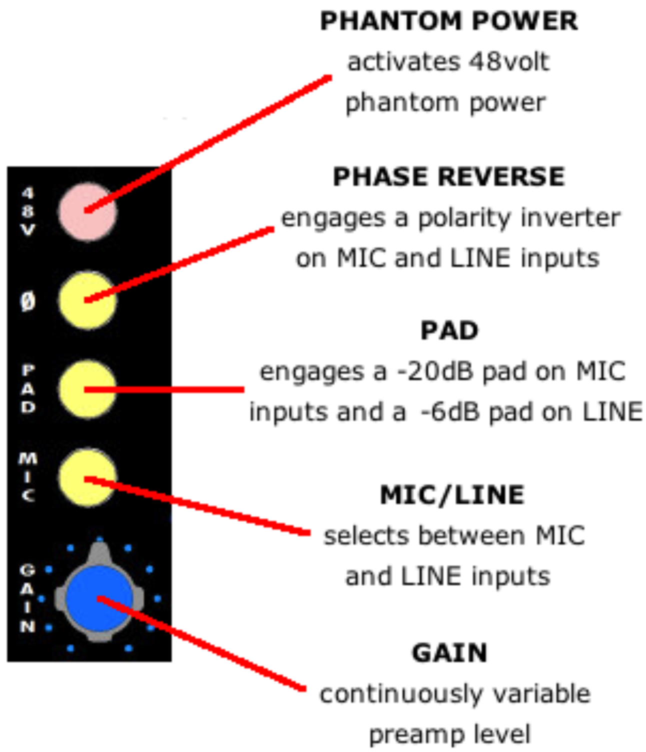
- +35dB of gain (+5dB to +40dB)

Note: The MIC switch must be engaged to use this input.

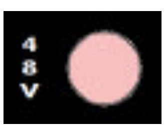
Note: INSTRUMENT INPUT *jack is a switching jack. The signal from the MIC INPUT is replaced with the signal from the* INSTRUMENT INPUT *whenever a ¼” plug is inserted in the* INSTRUMENT INPUT *jack.*

Note: The pad and polarity inverter (phase reverse) do not apply to the INSTRUMENT INPUT*. If these functions are needed, the use of an external direct injection (DI) box connected to the MIC* INPUT *is suggested.*

Preamp Controls:



<u>*No switches engaged*</u>: LINE INPUT is selected by default



<u>48V (Phantom Power)</u>: Applies 48 volt phantom power to the MIC INPUT XLR connector for use with condenser mics and active DI boxes

- Illuminates in red when engaged

<u>Ø (Phase Reverse)</u>: Inverts the polarity of the signal arriving at the MIC INPUT and LINE INPUT

- Illuminates when engaged
- Does not affect the instrument input

<u>PAD</u>: Attenuates the MIC INPUT by 20dB and the LINE INPUT by 6dB

- Illuminates when engaged
- Does not affect the instrument input

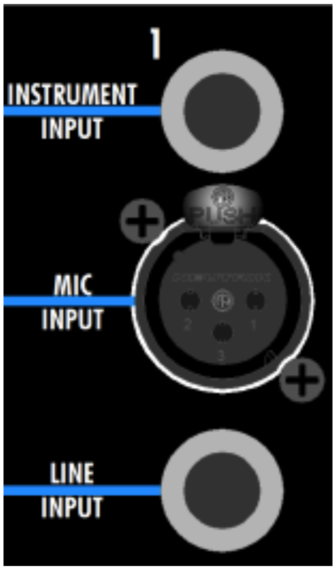
<u>MIC</u>: Selects the microphone signal as the input for the channel:

- MIC must be engaged for the INSTRUMENT INPUT to be used
- *The MIC INPUT signal is replaced by the INSTRUMENT INPUT signal when a 1/4" plug is inserted in the INSTRUMENT INPUT jack*

<u>GAIN</u>: Provides up to +65dB of preamp gain

Rear Panel Preamp Connections

All four input channels have the same preamp input connections on the rear panel.



INSTRUMENT INPUT: ¼" tip-sleeve, unbalanced, high-impedance

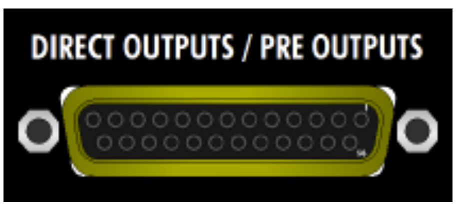
- MIC must be engaged to use this input
- Replaces MIC signal as preamp input

MIC INPUT: 3-pin female XLR, balanced, low-impedance

- MIC must be engaged to use this input
- Provides phantom power when the 48v switch is engaged

LINE INPUT: ¼" tip-ring-sleeve, balanced, low-impedance

- MIC must be disengaged to use this input

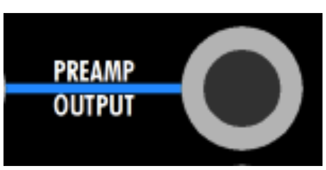


The DIRECT OUTPUTS / PRE OUTPUTS connector carries direct or preamp outputs from input channels 1-4, as well as the insert send from input channels 1 & 2 and the preamp outputs from input channels 3 & 4.

DIRECT OUTPUTS / PRE OUTPUTS: DB-25, balanced, low-impedance, line-level

- Outputs 1-4: Direct out (post-fader) or preamp output from input channels 1-4
  - Fed post-fader/mute by default (direct output)
  - Preamp output replaces post-fader output if DIR PRE is engaged
- Outputs 5 & 6: Insert send from input channels 1 & 2
- Outputs 7 & 8: Preamp outputs from input channels 3 & 4
- Standard pin-out: 1-4=direct or preamp outputs 1-4, 5-6=insert send 1-2, 7-8=preamp outputs 3-4

In addition to the DB-25 connector, input channels 3 & 4 have a separate preamp output that's part of an extra set of connections that support the 500 slots.



PREAMP OUTPUT: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

- Not available on input channels 1 & 2

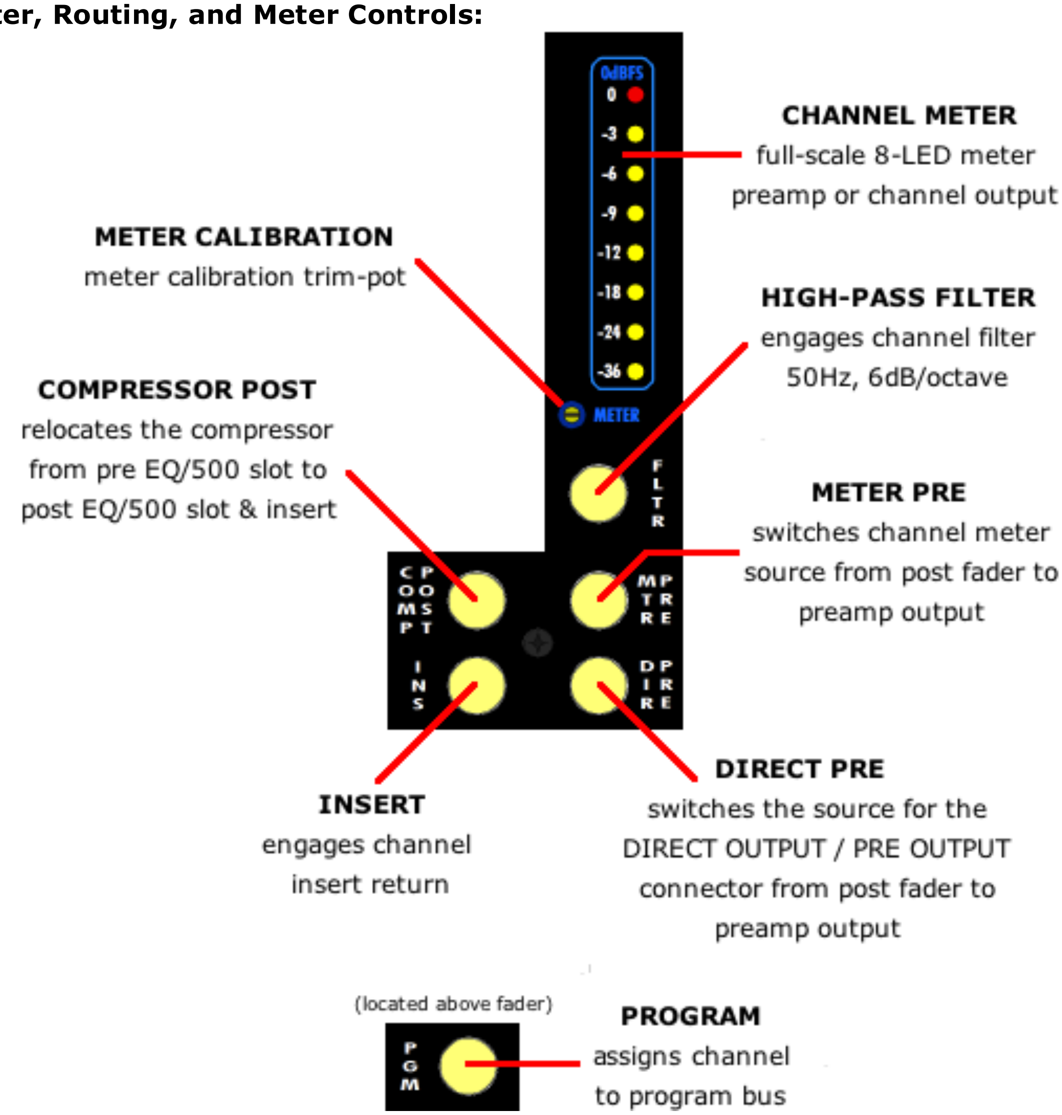
2.2 Filter, Routing, and Meter

Input channels provide a 50Hz high-pass filter, an 8-segment LED full-scale meter, and several routing possibilities.

Routing possibilities include:

- Compressor assignment to input channels
- Compressor placement in signal flow
- Insert
- Direct output routing
- Program bus assignment
- Meter routing

Filter, Routing, and Meter Controls:



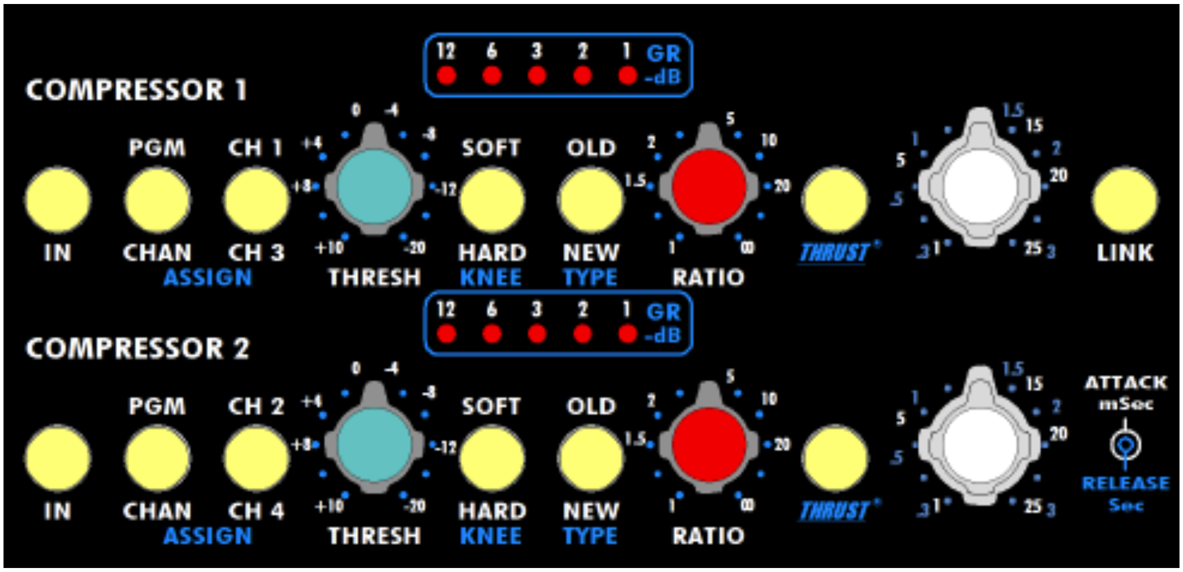
2.2.1 High-Pass Filter

FLTR (Filter): Engages high-pass filter

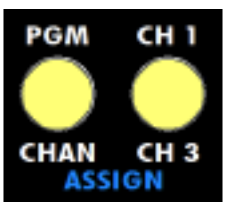
- -3 dB at 50Hz
- -6dB/octave slope
- Located post-fader
- Illuminates when engaged

2.2.2 Compressor Routing

"The BOX" includes two API 527 compressor/limiters in the center section of the console. These compressors are assigned to the program bus by default for mixing, but can be routed to the input channels while recording.



To use the compressors while recording, they need to be assigned to the input channels. Compressor 1 can be assigned to input channels 1 or 3. Likewise, Compressor 2 can be assigned to input channels 2 or 4.



Assignments to input channels are accomplished using the two ASSIGN switches on the compressors. Engaging the PGM-CHAN switch will move the compressor from the program bus to the input channels. When PGM-CHAN is engaged, compressor 1 is assigned to input channel 1 and compressor 2 is assigned to input channel 2 by default. Engaging CH 3 or CH 4 assigns the compressors to these channels. These switches illuminate when engaged.

When assigned to an input channel, the compressor will be located after the preamp and before the equalizer/500 slot and insert by default. It can be moved to after the equalizer/500 slot and insert by engaging the COMP POST switch. This switch illuminates when engaged.

The compressor will be bypassed unless the IN switch on the compressor is engaged. Engaging the IN switch will place the compressor in the signal path. This switch illuminates when engaged.

Stereo compression during recording is possible when both compressors are assigned to input channels and the LINK switch is engaged. This switch illuminates when engaged.

Refer to section 6.0 Compressors for more information regarding the compressors.

2.2.3 Insert

A balanced insert with switch is provided on each input channel.

INS (insert): Routes the insert return to the signal path

- The insert return is active only when the INS switch is engaged
- Illuminates when engaged

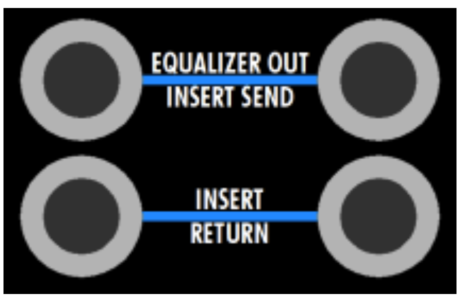
The insert send is fed pre-fader from the output of the EQ/500 slot and is always active.

The insert return is located post-EQ (pre-fader) and is routed to the signal path only when the INS switch is engaged

Rear Panel Insert Connections

All insert sends and returns are balanced, low-impedance, line-level signals routed via ¼” tip-ring-sleeve jacks on the rear panel.

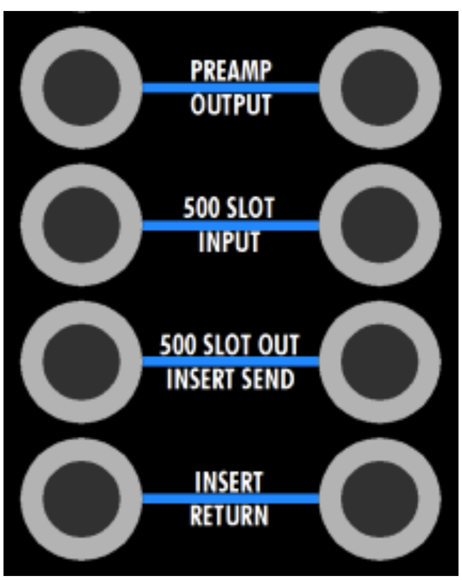
The insert send is post-EQ on input channels 1 & 2.



EQUALIZER OUT-INSERT SEND: ¼” tip-ring-sleeve, balanced, low-impedance, line-level

INSERT RETURN: ¼” tip-ring-sleeve, balanced, low-impedance, line-level

The insert send is post-500 slot on input channels 3 & 4. In addition to the insert send and return, the signal path on channels 3 & 4 can be accessed pre-500 slot via the PREAMP OUTPUT and 500 SLOT INPUT jacks on the rear panel.



PREAMP OUTPUT: ¼” tip-ring-sleeve, balanced, low-impedance, line-level

500 SLOT INPUT: ¼” tip-ring-sleeve, balanced, low-impedance, line-level

500 SLOT OUT- INSERT SEND: ¼” tip-ring-sleeve, balanced, low-impedance, line-level

INSERT RETURN: ¼” tip-ring-sleeve, balanced, low-impedance, line-level

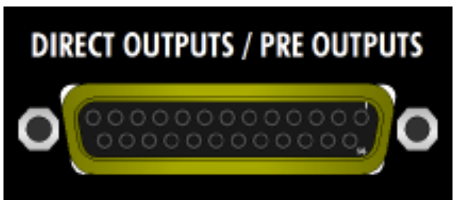
2.2.4 Output Routing and Pan-Pot

Direct Output

Each input channel has a direct output intended to feed a DAW analog input while recording. The direct output is fed post-fader/mute by default, but can be fed by the preamp output. The direct output is always active except when the channel MUTE is engaged.

DIR PRE (preamp direct output): Replaces the fader output with the preamp output at the DIRECT OUTPUTS / PRE OUTPUTS DB-25 connector on the rear panel

- Default feed is post-fader/mute if not engaged
- Illuminates when engaged



The DIRECT OUTPUTS / PRE OUTPUTS connector carries direct or preamp outputs from input channels 1-4, as well as the insert send from input channels 1 & 2 and the preamp outputs from input channels 3 & 4.

DIRECT OUTPUTS / PRE OUTPUTS: DB-25, balanced, low-impedance, line-level

- Outputs 1-4: Direct out (post-fader) or preamp output from input channels 1-4
  - Fed post-fader/mute by default (direct output)
  - Preamp output replaces post-fader output if DIR PRE is engaged
- Outputs 5 & 6: Insert send from input channels 1 & 2
- Outputs 7 & 8: Preamp outputs from input channels 3 & 4
- Standard pin-out: 1-4=direct or preamp outputs 1-4, 5-6=insert send 1-2, 7-8=preamp outputs 3-4

Program Bus Assignment

PGM (Program): Assigns the input channel output to the program bus when engaged

- Program bus is fed post-pan-pot
- Illuminates when engaged

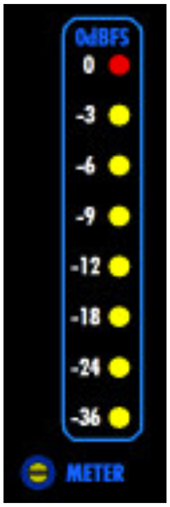
Stereo Pan-Pot

PAN (pan-pot): Continuously variable left-right stereo pan-pot

- Program bus is fed post-pan-pot
- -3dB pan law (-3dB per side when panned center)
- Center detent

2.2.5 Meter and Meter Routing

Input channels are equipped with an 8-segment LED full-scale meter. The meter displays the post-fader output from the channel by default, but can be switched to display the output of the preamp. A calibration trim-pot is provided so 0dBFS on the meter matches 0dBFS in the connected DAW.

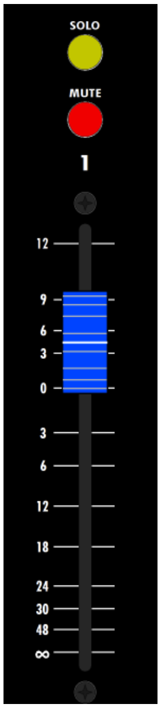


0dBFS: 8-segment LED full-scale meter
- Indicates post-fader, channel output level by default
- Indicates preamp output level when MTR PRE is engaged

METER (calibration): Meter calibration trim-pot
- Calibrates the meter to match 0dBFS on the DAW converter

MTR PRE (meter preamp): Replaces the output of the fader with the output of the preamp as the meter source
- Default meter feed is post-fader, channel output if not engaged
- Illuminates when engaged

2.3 Fader, Mute, Solo, and Solo-Safe

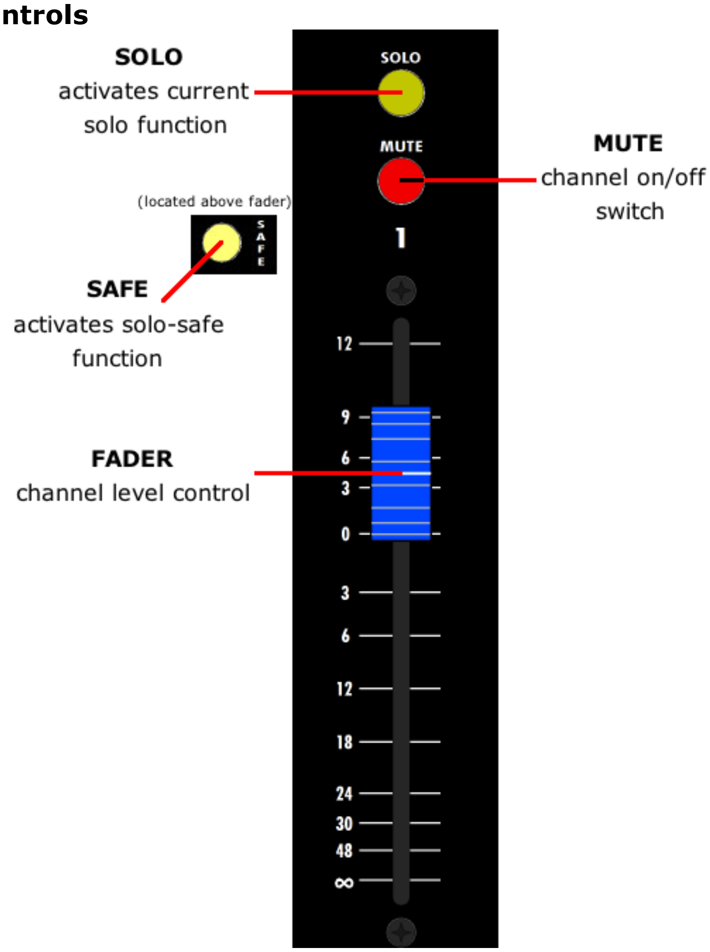


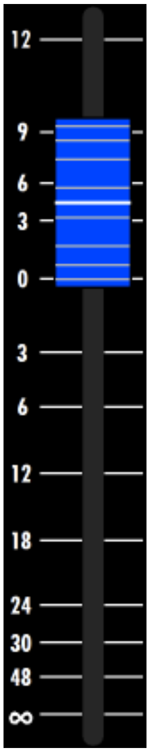
The fader module with blue fader caps contains the faders, MUTE and SOLO switches for input channels 1-4. These faders are the primary output level controls for the input channels.

The input channel faders can be routed to feed:

- Direct output
- Program bus
- Post-fader auxiliary/cue sends

Input Fader Controls





INPUT FADER: Primary output level control

- 100mm full-size audio-taper fader
- -∞dB to +12dB range
- 0dB is unity gain
- Blue fader cap

The input channel MUTE and SOLO switches are located just above the fader.

MUTE: Cuts the channel output

- Located post-fader in the signal flow
- Illuminates in red when engaged
- Activates on release

SOLO: Activates the current solo function on that channel

- PFL, AFL, or solo-in-place (PFL is default)
- Can be cleared by pressing the SOLO CLEAR switch
- Illuminates in orange when engaged

SAFE (solo safe): Activates the solo-safe function on that channel

- SAFE prevents muting when Solo-In-Place is active and another channel is soloed.
- Illuminates when engaged

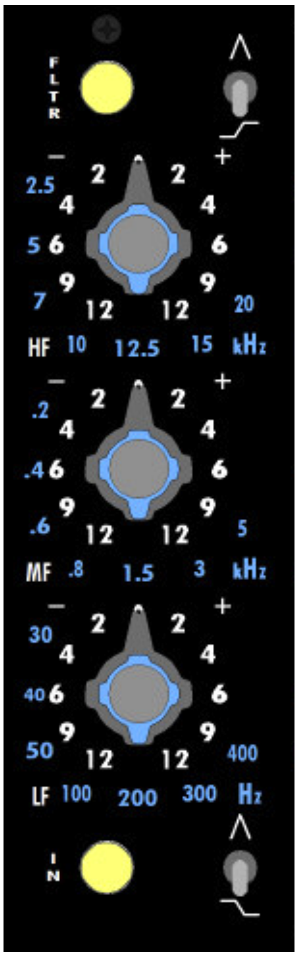
NOTE: The SAFE switch is located below the pan-pot.

Refer to section 9.0 Solo Master for more information regarding the solo system.

2.4 550A Equalizer – 500 Slot

Input channels 1 & 2 provide semi-parametric equalization using the legendary API 550A EQ. Input channels 3 & 4 provide two (2) 500 slots for modules that conform to the standards set forth by the API VRP Alliance. This combination provides the classic API sound as well as giving the owner modest means to customize their console with a very wide range of other API and third party products

2.4.1 550A Discrete 3-Band Equalizer



The API 550A Equalizer provides three (3) bands of equalization and a band-pass filter. It has been slightly repackaged for installation on input channels 1-2. Other than the repositioning of some controls these equalizers are identical to other 550A EQs.

Features

- Three (3) bands of classic API equalization
- Each band offers 7 API selected center frequencies
- Reciprocal and repeatable filtering
- Maximum 12dB of boost/cut per band
- HF and LF bands offer shelf/peak switching
- "Proportional Q" narrows filter Q at extremes
- Traditional API fully discrete circuit design
- High headroom +30dB clip level

Few equalizers enjoy the respect and admiration of the coveted API 550A. Designed by the now renowned Saul Walker in the late 60's, the discrete 550A was first used as a modular OEM equalizer. As the industry rapidly embraced the sonic quality of the 550A, it quickly found its way into many custom console designs by Frank DeMedio and other leading engineers. Many of these consoles are still in use today.

Over forty years later, the 550A remains the standard against which other EQs are measured, and it has played a major role in the recording industry for decades.

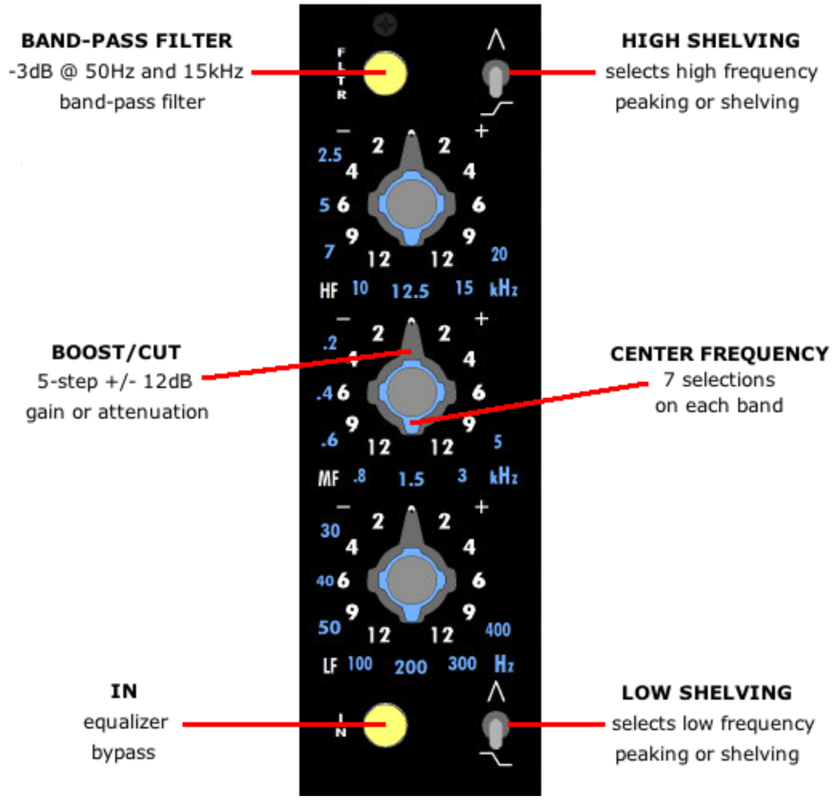
Still copied but never duplicated, the 550A became API's standard channel module EQ when the company began manufacturing consoles in 1971. With virtually all existing units spoken for, popular demand for this EQ resulted in API finally resuming production in 2004.

The 550A provides reciprocal equalization at 21 points in 5 steps of boost to a maximum of 12dB of gain at each point. The fifteen equalization points are divided into three overlapping ranges. The high and low frequency ranges are individually selectable as either peaking or shelving, and a band-pass filter may be inserted independently of all other selected equalization settings. Frequency ranges and boost/cut amounts are selected by three dual-concentric switches, and a push-button "IN" switch allows the EQ to be silently introduced to the signal path. A small push-button is used to insert the band-pass filter into the 550A.

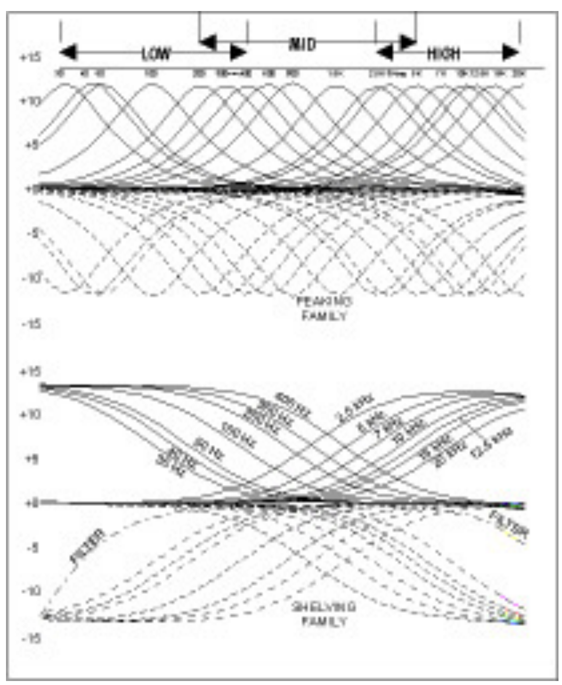
The combination of Walker's incomparable 2520 op-amp, and his "Proportional Q" circuitry gives the 550A user an uncomplicated way to generate acoustically superior equalization.

With the long-awaited reissue of this unit, an EQ that has had such a part in the history of recording is continuing to make history in today's music.

Equalizer Controls:

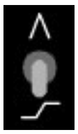


550A EQ Curves

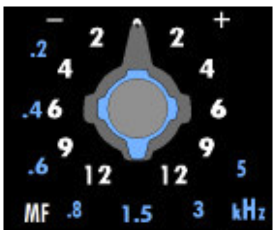


IN: Places the EQ in the signal path when engaged
- Silent operation
- Illuminates when engaged

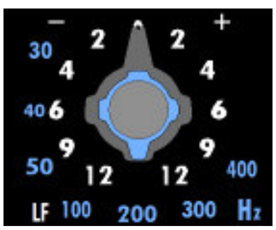
High Frequency:
- Center Frequency (knob & blue numbers):
  2.5kHz, 5kHz, 7kHz, 10kHz, 12.5kHz, 15kHz, 20kHz
- Boost/Cut (ring & white numbers):
  +/- 12dB in 5 steps each
- Peak or shelf



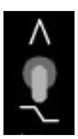
High Frequency Shelving: Changes the high frequency band from a peaking EQ to a shelving EQ
- All frequencies above the selected center frequency will be boost or cut



Mid Frequency:
- Center Frequency (knob & blue numbers):
  200Hz, 400Hz, 600Hz, 800Hz, 1.5kHz, 3kHz, 5kHz
- Boost/Cut (ring & white numbers):
  +/- 12dB in 5 steps each



Low Frequency:
- Center Frequency (knob & blue numbers):
  30Hz, 40Hz, 50Hz, 100Hz, 200Hz, 300Hz, 400Hz
- Boost/Cut (ring & white numbers):
  +/- 12dB in 5 steps each
- Peak or shelf



Low Frequency Shelving: Changes the low frequency band from a peaking EQ to a shelving EQ
- All frequencies below the selected center frequency will be boost or cut

FLTR (band-pass filter): Activates the band-pass filter
- -3dB @ 50Hz
- -3dB @ 15kHz
- Illuminates when engaged

NOTE: The band-pass filter in the equalizer should not be confused with the high-pass filter in the input channels.

Additional information about the 550A equalizer can be found at:
http://apiaudio.com/550a.html

2.4.2 API 500 Series Slot

Input channels 3 & 4 are each equipped with an API 500 series slot that can accept any module that conforms the standards set forth by the API VPR Alliance. These 500 series slots are built to power and interface any API or JDK Audio 500 series module. Suggested modules include:

- API 525 Compressor Re-Issue
- API 527 Compressor/Limiter
- API 550A 3-Band EQ
- API 550b 4-Band EQ
- API 560 10-band Graphic EQ
- API 565 Filter Set
- JDK V12 Single Channel Compressor
- JDK V14 4-Band EQ

More information about these 500 series modules can be found at: http://apiaudio.com and http://www.jdkaudio.com

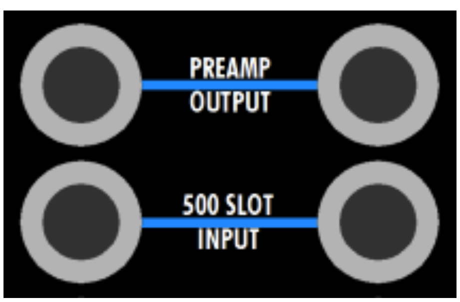
In addition to API and JDK modules, any third-party module that has been approved by API as part of the VPR Alliance can be fitted into the slots. Additional information about approved 500 series modules and the VPR Alliance is available at: http://apiaudio.com/vpr_alliance.html

IMPORTANT NOTE: Installation of any module that's not approved by the API VPR Alliance may void the console warranty!

If the slots remain empty, a jumper is installed in place of a module and the slot is covered with a blank panel.

API 500 Series Slot Rear Panel Connections

The signal path on channels 3 & 4 can be accessed via the PREAMP OUTPUT and 500 SLOT INPUT jacks on the rear panel.

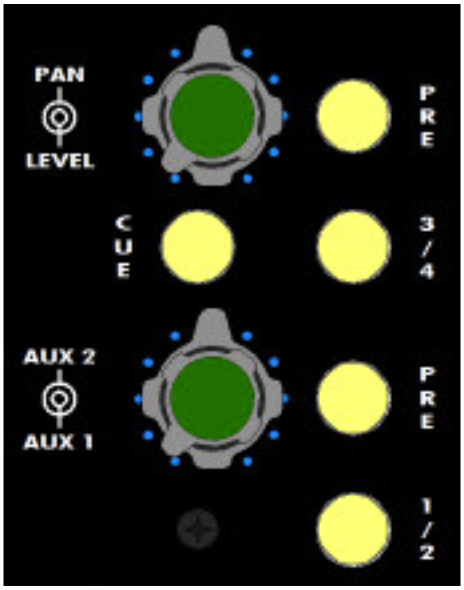


PREAMP OUTPUT: ¼" tip-ring-sleeve, jack* balanced, low-impedance, line-level

500 SLOT INPUT: ¼" tip-ring-sleeve, switching jack*, balanced, low-impedance, line-level

* The PREAMP OUTPUT and 500 SLOT INPUT jacks work as a "half-normalled" pair and the channel signal flows through these jacks. If a plug is inserted into the 500 SLOT INPUT jack, that signal replaces the signal from the PREAMP OUTPUT jack. The signal from the onboard preamp is broken and does not continue in the channel flow path.

2.5 Auxiliary/Cue Sends

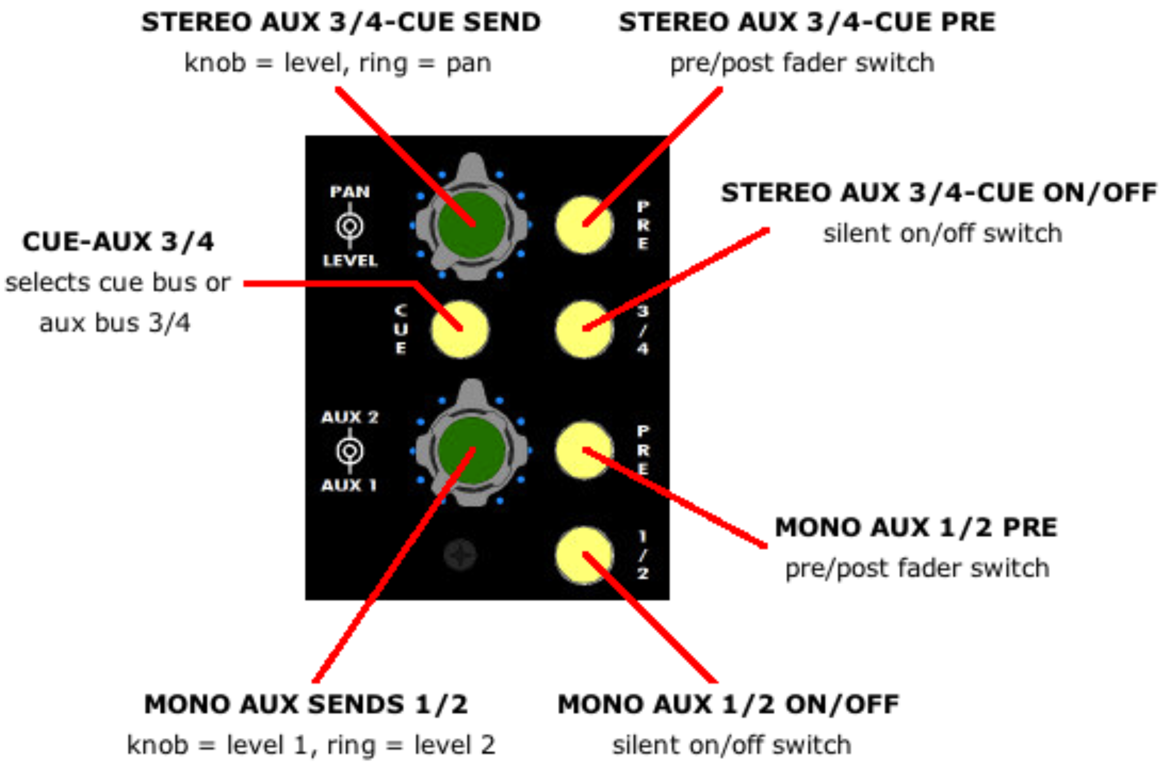


"The BOX" is equipped with four (4) auxiliary sends fed by two (2) mono auxiliary sends (1/2), and one (1) stereo auxiliary/cue send (3/4).

Each pair of sends has pre/post-fader routing and an on/off switch. The stereo aux send (3/4) can be routed to feed the stereo cue bus instead of aux buses 3 & 4.

To support these sends, "The BOX" is equipped with four auxiliary buses, a stereo cue bus, and corresponding masters with balanced outputs.

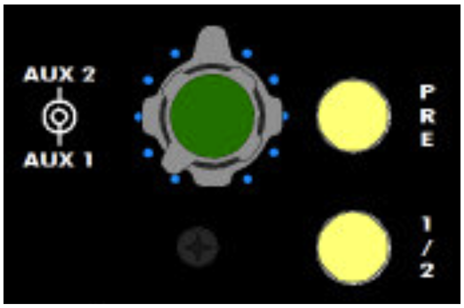
Auxiliary/Cue Controls



2.5.1 Mono Aux Sends 1/2

AUX 1 and AUX 2 are mono auxiliary sends that feed aux buses 1 and 2. These sends are commonly used as effects sends, but can be used for additional headphone feed or other purposes.

A common on/off switch is used to silently engage both mono sends. The post-fader signal feeds AUX 1/2 by default, but the pair can be switched to the pre-fader signal.



AUX 1-AUX 2: Level controls for AUX 1/2
- Knob = 1, ring = 2

PRE (pre-post): Routes the pre-fader source* to AUX 1/2 when engaged
- AUX 1/2 are fed post-fader by default
- Illuminates when engaged

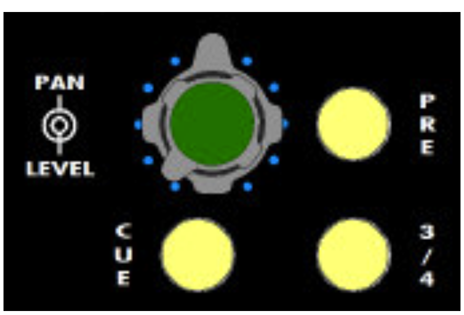
1/2 (on/off): On/off switches for AUX 1/2
- Illuminates when engaged

*An internal jumper determines the source of the aux/cue feed when the PRE switch is engaged. Refer to section 2.5.4 Aux Pre Jumper for detailed information.

2.5.2 Stereo Aux Send 3/4

AUX 3/4 is a stereo auxiliary send that feed aux buses 3 and 4 by default. These sends can also be routed to the stereo cue bus to quickly create a fully featured headphone feed when recording. When mixing, stereo AUX 3/4 or CUE can be used for effects sends.

A common on/off switch is used to silently engage the stereo send. The post-fader signal feeds AUX 3/4-CUE by default, but the pair can be switched to the pre-fader signal. In common practice, post-fader sends are often used for effects sends and pre-fader sends are typically used for headphone feeds.



AUX 3/4: Level controls for AUX 3/4 or CUE
- Knob = level, ring = pan

PRE (pre-post): Routes the pre-fader source* to AUX 3/4-CUE when engaged
- AUX 3/4-CUE are fed post-fader by default
- Illuminates when engaged

3/4 (on/off): On/off switches for AUX 3/4-CUE
- Illuminates when engaged

*An internal jumper determines the source of the aux/cue feed when the PRE switch is engaged. Refer to section 2.5.4 Aux Pre Jumper for detailed information.

2.5.3 Stereo Cue Sends

In addition to auxiliary buses 1-4, "The BOX" is equipped with a separate stereo cue bus, cue master, and balanced output. When recording, the primary purpose of this cue system is the creation of a headphone feed, so talkback and other sources can be routed to the cue master. When mixing, stereo send 3/4 can be routed to the cue bus instead of aux buses 3/4 to provide an alternate stereo effects send.

CUE: Routes aux send 3/4 to the stereo cue bus when engaged
- AUX 3/4 feeds auxiliary buses 3 and 4 by default
- The feed to auxiliary buses 3 and 4 is defeated when engaged
- Illuminates when engaged

2.5.4 Aux Pre Jumper

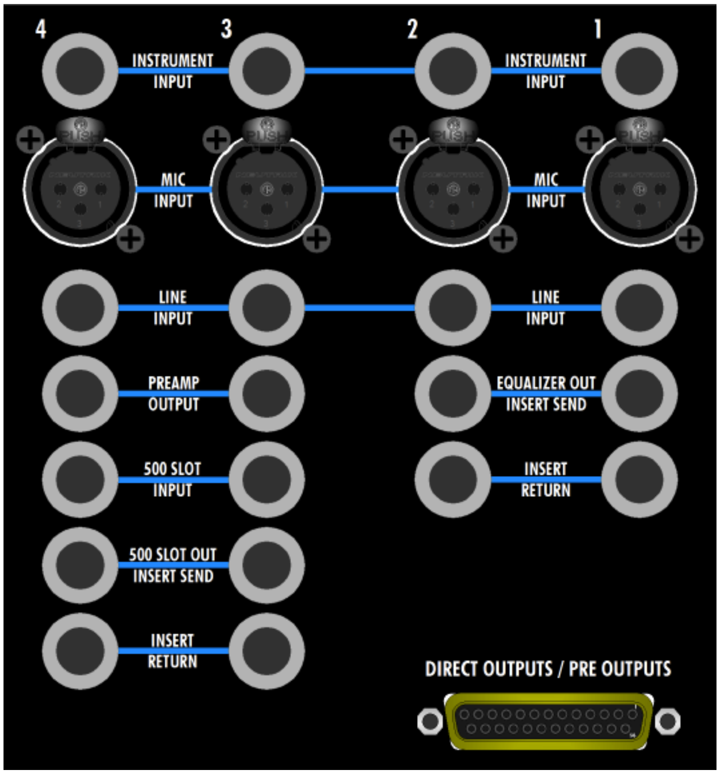
Aux/Cue sends set to PRE can be from two locations in the signal path, pre-EQ/500 slot or from the fader input. An internal AUX PRE routing jumper on each input channel determines the point in the signal path that feeds the aux/cue sends when the PRE switches are engaged.

The AUX PRE jumper can be set to one of two positions:
- PRE EQ (pre-equalizer): Pre-EQ/500 slot
  - Post-preamp, post-compressor (if routed)
  - Pre-EQ/500 slot, pre-insert send, pre-fader
- FDR IN (fader input): Pre-fader input
  - Post-preamp, post-compressor (if routed), post-EQ, post-insert return
  - Pre-fader

The default position for this routing jumper is factory-set to FDR IN (fader input). The jumper can be changed using the procedure outlined in a separate service document.

This routing option and jumper are available only on input channels 1-4. Aux/cue sends on summing inputs 1-16 are fed from the fader input when their PRE switches are engaged.

2.6 Input Channel Rear Panel Connections



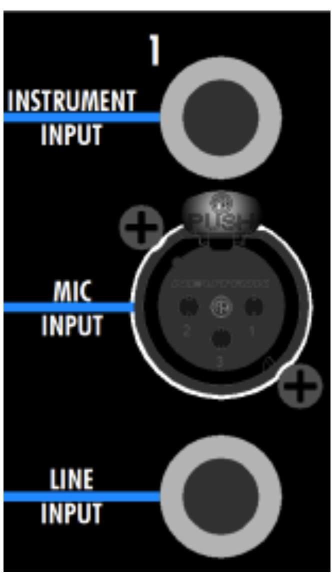
The rear panel provides a comprehensive and flexible set of input channel connections.

Connections include:

- Preamp inputs (mic, instrument, and line)
- Insert sends and returns
- Direct outputs / preamp outputs
- Preamp output / 500 slot input (channels 3 & 4 only)

2.6.1 Input Channel Preamp Connections

All four input channels have the same preamp input connections on the rear panel.



INSTRUMENT INPUT: ¼" tip-ring-sleeve, unbalanced, high-impedance

- MIC must be engaged to use this input
- Replaces MIC signal as preamp input

MIC INPUT: 3-pin female XLR, balanced, low-impedance

- MIC must be engaged to use this input
- Provides phantom power when the 48v switch is engaged

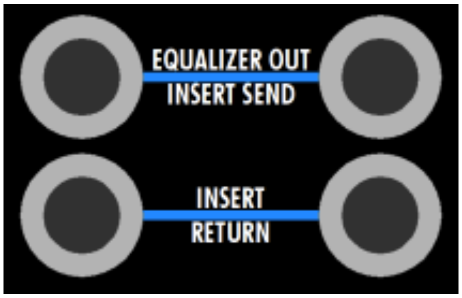
LINE INPUT: ¼" tip-ring-sleeve, balanced, low-impedance

- MIC must be disengaged to use this input

2.6.2 Input Channel Insert Connections

All insert sends and returns are balanced, low-impedance, line-level signals routed via ¼" tip-ring-sleeve jacks on the rear panel.

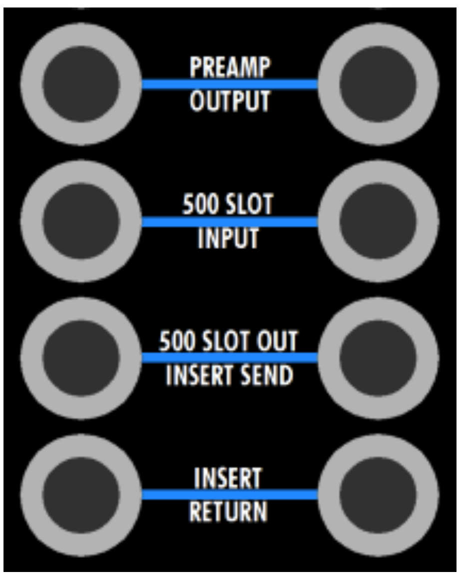
The insert send is post-EQ on channels 1 & 2.



EQUALIZER OUT- INSERT SEND: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

INSERT RETURN: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

The insert send is post-500 slot on channels 3 & 4. In addition to the insert send and return, the signal path on channels 3 & 4 can be accessed pre-500 slot via the PREAMP OUTPUT and 500 SLOT INPUT jacks on the rear panel.



PREAMP OUTPUT: ¼" tip-ring-sleeve jack* balanced, low-impedance, line-level

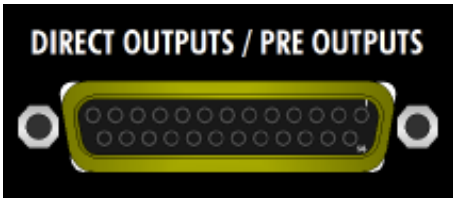
500 SLOT INPUT: ¼" tip-ring-sleeve, switching jack*, balanced, low-impedance, line-level

500 SLOT OUT- INSERT SEND: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

INSERT RETURN: ¼" tip-ring-sleeve, switching jack*, balanced, low-impedance, line-level

* The PREAMP OUTPUT and 500 SLOT INPUT jacks work as a "half-normalled" pair and the channel signal flows through these jacks. If a plug is inserted into the 500 SLOT INPUT jack, that signal replaces the signal from the PREAMP OUTPUT jack. The signal from the onboard preamp is broken and does not continue in the channel flow path.

2.6.3 Input Channel Direct and Preamp Output Connections

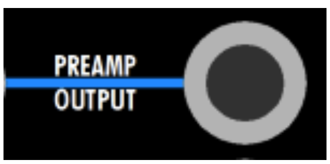


The DIRECT OUTPUTS / PRE OUTPUTS connector carries direct or preamp outputs from input channels 1-4, as well as the insert send from input channels 1 & 2 and the preamp outputs from input channels 3 & 4.

<u>DIRECT OUTPUTS / PRE OUTPUTS</u>: DB-25, balanced, low-impedance, line-level

- Outputs 1-4: Direct out (post-fader) or preamp output from input channels 1-4
    - Fed post-fader/mute by default (direct output)
    - Preamp output replaces post-fader output if DIR PRE is engaged
- Outputs 5 & 6: Insert send from input channels 1 & 2
- Outputs 7 & 8: Preamp outputs from input channels 3 & 4
- Standard pin-out: 1-4=direct or preamp outputs 1-4, 5-6=insert send 1-2, 7-8=preamp outputs 3-4

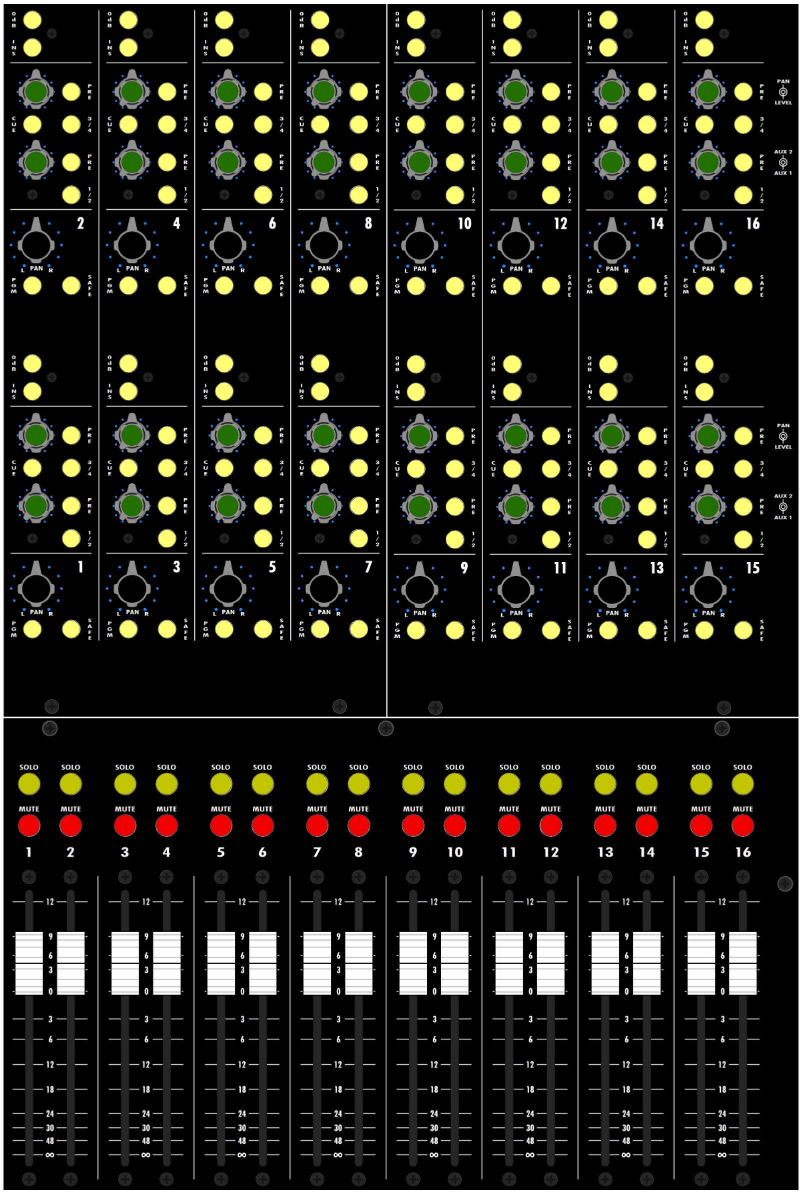
In addition to the DB-25 connector, input channels 3 & 4 have a separate preamp output that's part of an extra set of connections that support the 500 slots.



<u>PREAMP OUTPUT</u>: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

- Not available on input channels 1 & 2

3.0 Summing Input Channels 1-16



"The BOX" provides sixteen (16) summing input channels (summing inputs) that are optimized for mixing in a DAW environment. All sixteen summing inputs are identical.

Along with their associated faders, each summing input provides a simple, but appropriately featured audio path for mixing applications. In recognition of the signal processing power of most DAWs, the summing inputs do not contain filters, equalization or dynamic processing (compressor). However they do provide the essential elements and flexibility needed for mixing on an analog summing bus. Accordingly, all summing inputs include:

- 0dB fader bypass
- Balanced insert with switch
- 2 mono aux sends
- 1 stereo aux/cue send
- Fader and mute
- Pan-pot
- Program bus assignment
- Solo and solo safe

All aux/cue sends have ON/OFF switches and can be routed either pre or post-fader. The stereo send can be routed to aux buses 3/4 or the cue bus.

To make efficient use of space, summing inputs are arranged with two (2) channels per strip with the even number channels above the odd numbered channels. The fader controls for each pair of channels are situated side-by-side below the 2-channel strip. Each 2-channel strip contains two complete channels.

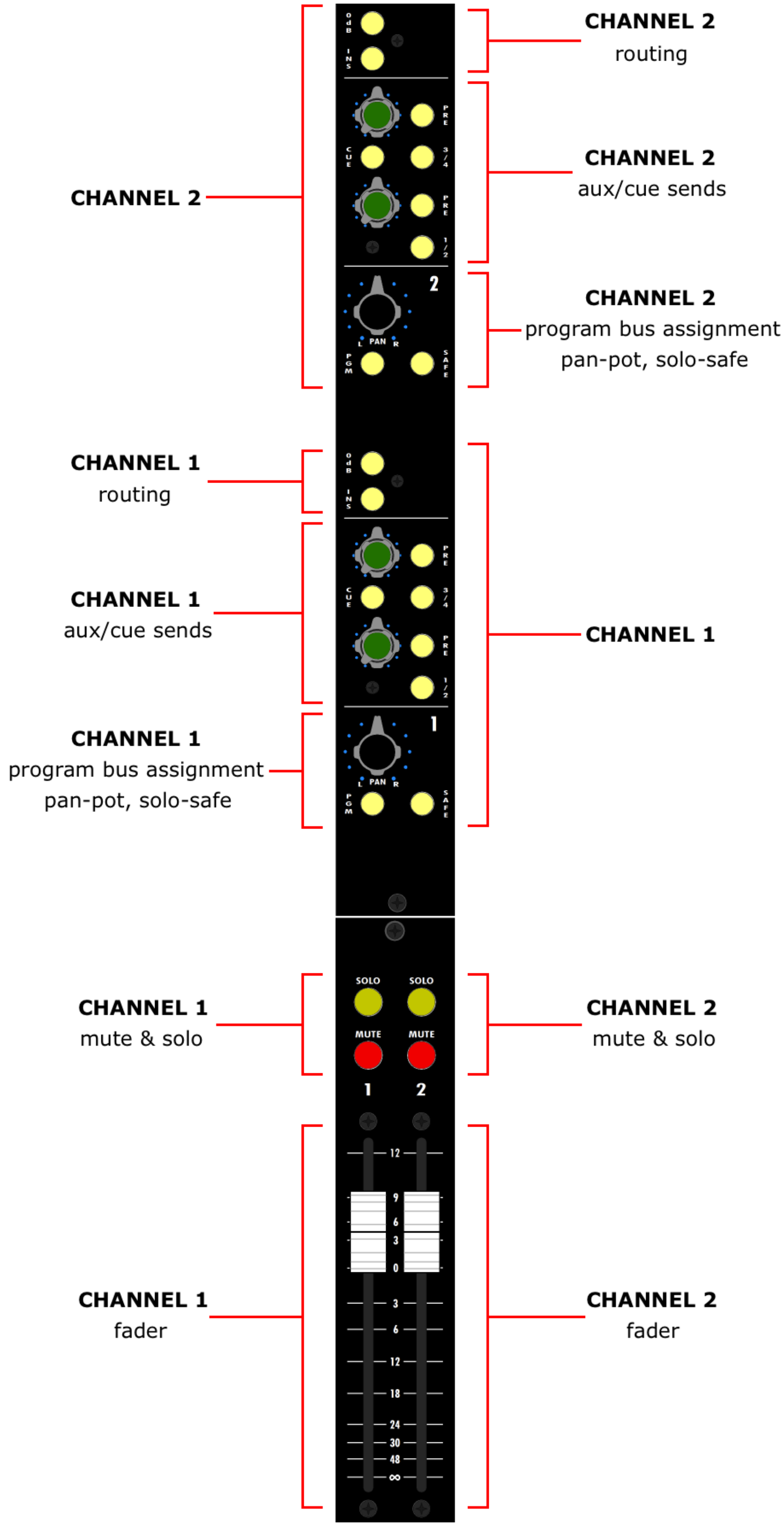
The default signal flow through the summing inputs is as follows:

Input ⇒ insert ⇒ fader ⇒ mute ⇒ pan-pot ⇒ program bus

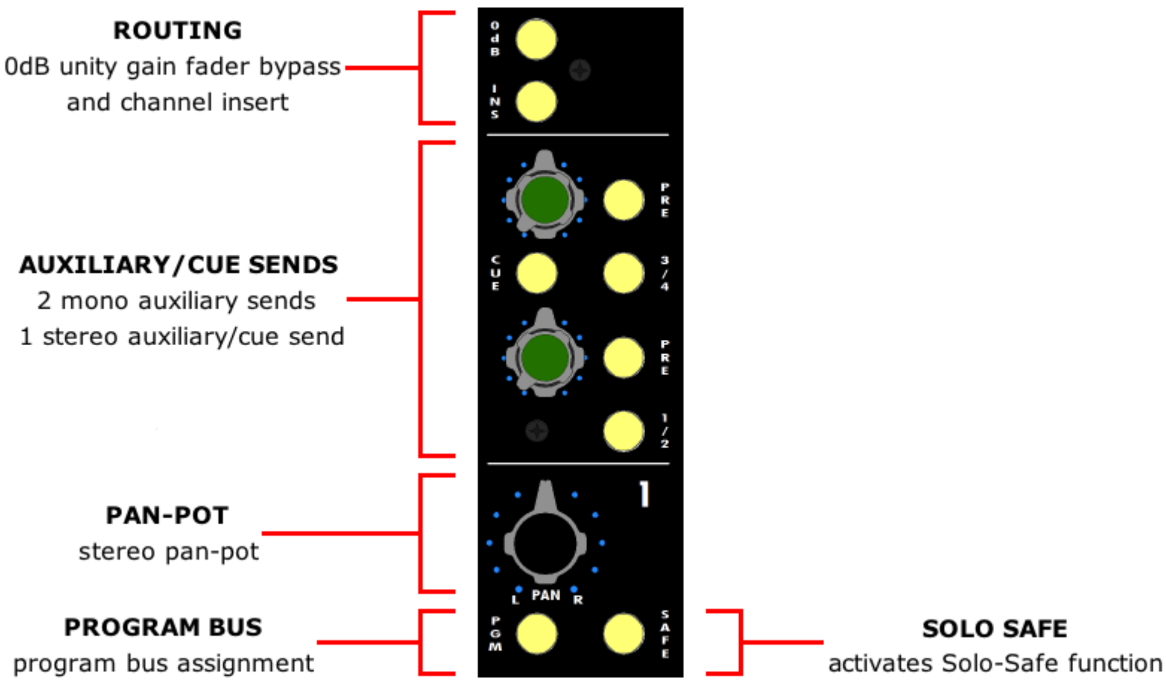
The fader can be bypassed (0dB) to yield:

Input ⇒ insert ⇒ mute ⇒ pan-pot ⇒ program bus

2-Channel Summing Input Strip Sections



Summing Input Controls



3.1 Input Routing

Summing inputs accept balanced, low-impedance, line-level signals from the SUMMING INPUTS 1-8 and SUMMING INPUTS 9-16 DB-25 connectors on the rear panel.

This line input is the primary summing channel input. Other than the insert return, it's the only input available.

3.1.1 Insert

A balanced insert with switch is provided on each summing input.

INS (insert): Routes the insert return to the signal path
- The insert return is active only when the INS switch is engaged
- Illuminates when engaged

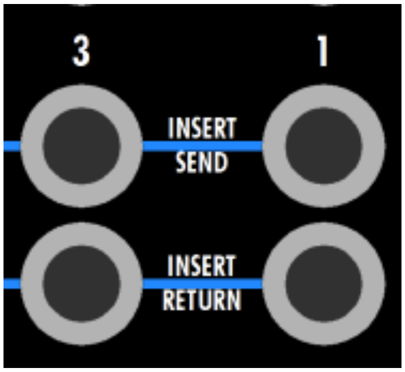
The insert send is fed pre-fader immediately after the line input and is always active.

The insert return is located before the fader bypass (0dB) and is routed to the signal path only when the INS switch is engaged

Rear Panel Insert Connections

All summing input insert sends and returns are balanced, low-impedance, line-level signals routed via ¼" tip-ring-sleeve jacks on the rear panel.

The insert send located immediately after the input on summing inputs 1-16.

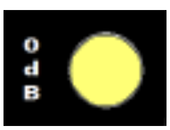


EQUALIZER OUT- INSERT SEND: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

INSERT RETURN: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

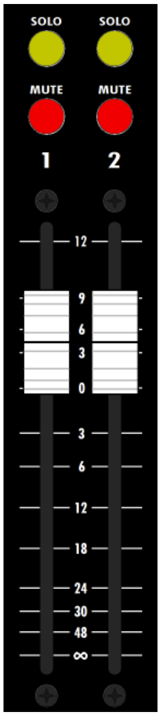
3.1.2 0dB (Unity Gain)

When mixing, track levels are often automated in the DAW and do not require further rebalancing on the console. Traditionally, engineers would set all faders to 0dB (unity gain) when mixing DAW returns on a console. To preserve the balance of automated tracks coming from the DAW, "The BOX" includes a fader bypass ("0dB" switch) on the summing inputs to route these tracks directly to the program bus at unity gain. Engaging the 0dB switch is equivalent to setting the fader to unity gain (0dB). Only the fader is bypassed and all other channel functions operate normally. This feature avoids the need to manually set the faders to 0dB, prevents errors and accidental fader movement, and slightly shortens the signal path, while providing full channel functionality.



0dB: Engages the unity gain fader bypass
- Silent operation
- Illuminates when engaged

3.2 Fader, Mute, Solo, and Solo-Safe



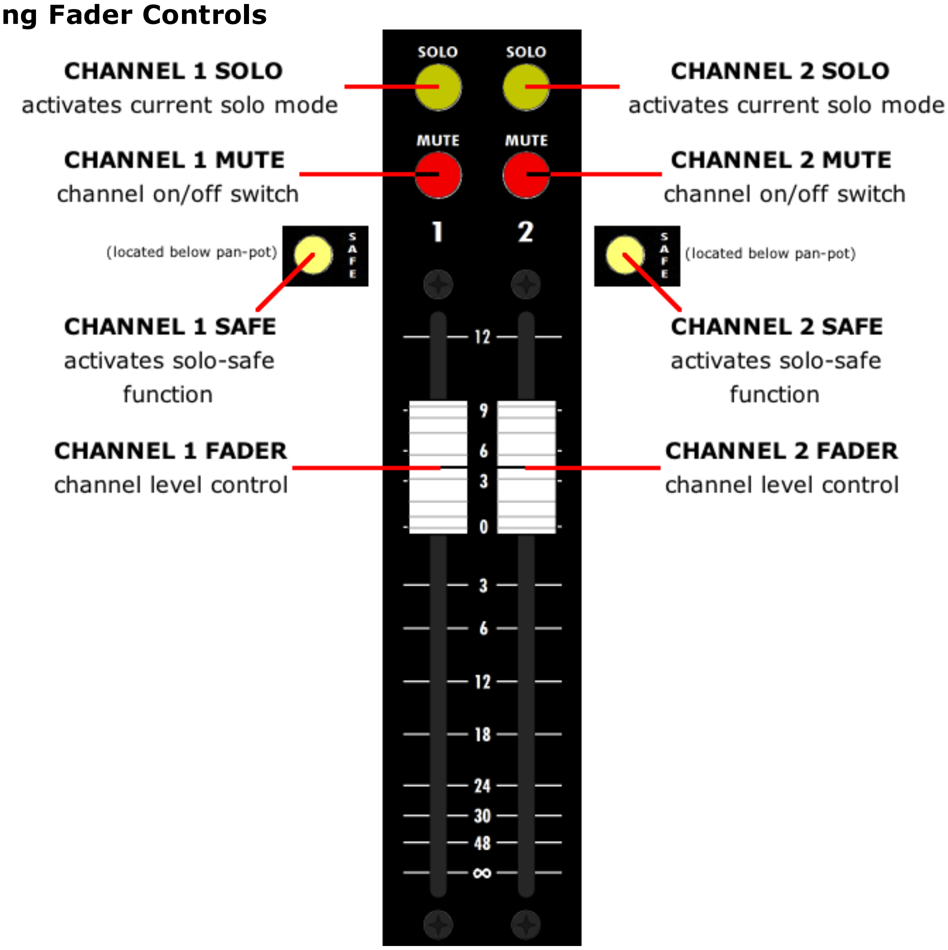
The fader modules with white fader caps contain the faders, MUTE and SOLO switches for summing inputs 1-16 (1 & 2 shown). The summing input fader controls for each pair of channels are arranged in a side-by-side configuration.

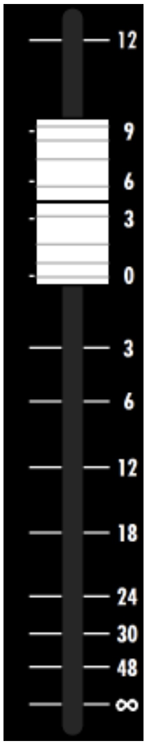
These faders are the primary output level controls for the summing inputs.

The summing input faders can be routed to feed:

- Program bus
- Post-fader auxiliary/cue sends

Summing Fader Controls





SUMMING FADER: Primary output level control
- 100mm full-size audio-taper fader
- -∞dB to +12dB range
- 0dB is unity gain
- White fader cap

The summing input MUTE and SOLO switches are located just above the fader.

MUTE: Cuts the channel output
- Located post-fader in the signal flow
- Illuminates in red when engaged
- Activates on release

SOLO: Activates the current solo function on that channel
- PFL, AFL, or solo-in-place (PFL is default)
- Can be cleared by pressing the SOLO CLEAR switch
- Illuminates in orange when engaged

SAFE (solo safe): Activates the solo-safe function on that channel
- SAFE prevents muting when Solo-In-Place is active and another channel is soloed.
- Illuminates when engaged

NOTE: The SAFE switch is located below the pan-pot.

Refer to section 9.0 Solo Master for more information regarding the solo system.

3.3 Output Routing and Pan-Pot

Each summing input can be routed to the program bus via the PGM assignment switch. The program bus is fed post-fader/mute and pan-pot.

Program Bus Assignment

PGM (Program): Assigns the summing channel output to the program bus when engaged

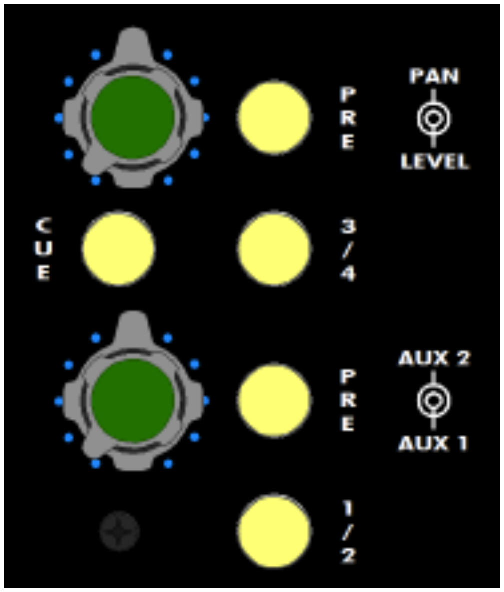
- Program bus is fed post-pan-pot
- Illuminates when engaged

Stereo Pan-Pot

PAN (pan-pot): Continuously variable left-right stereo pan-pot

- Program bus is fed post-pan-pot
- -3dB pan law (-3dB per side when panned center)
- Center detent

3.4 Auxiliary Sends

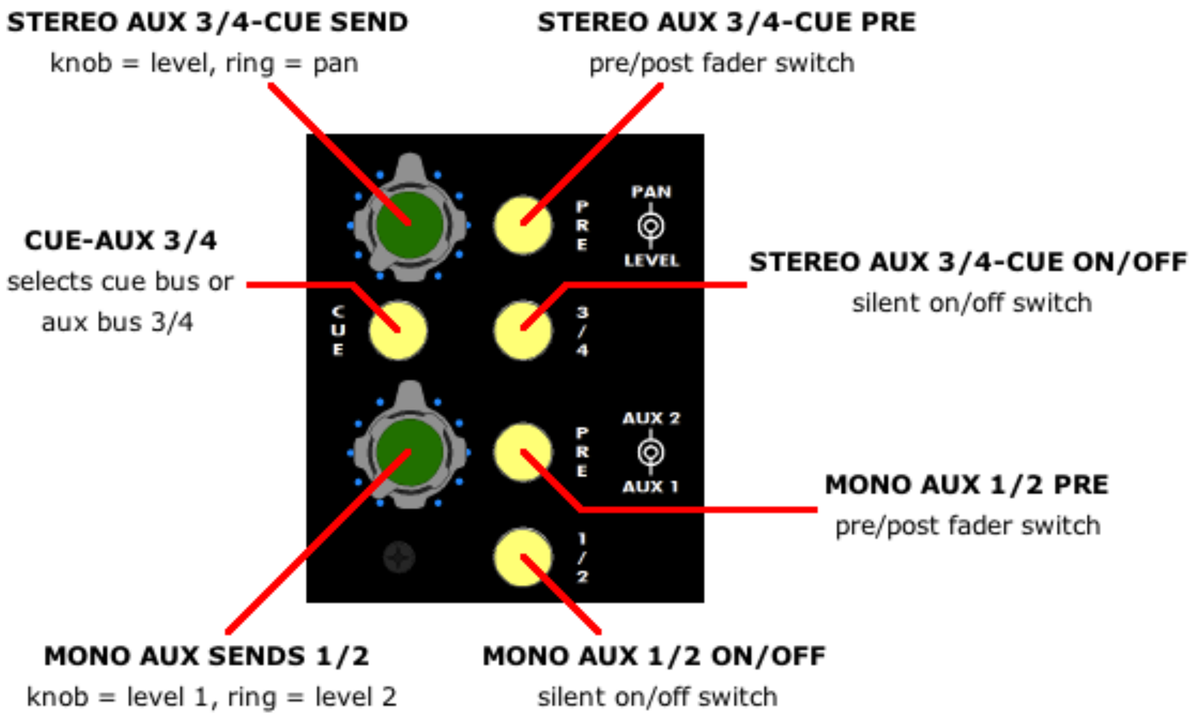


"The BOX" is equipped with four (4) auxiliary sends fed by two (2) mono auxiliary sends (1/2), and one (1) stereo auxiliary/cue send (3/4).

Each pair of sends has pre/post-fader routing and an on/off switch. The stereo aux send (3/4) can be routed to feed the stereo cue bus instead of aux buses 3 & 4.

To support these sends, "The BOX" is equipped with four auxiliary buses, a stereo cue bus, and corresponding masters with balanced outputs.

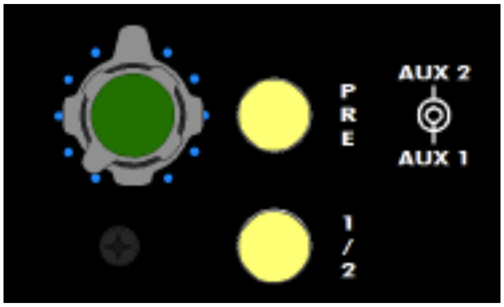
Auxiliary/Cue Controls



3.4.1 Mono Aux Sends 1/2

AUX 1 and AUX 2 are mono auxiliary sends that feed aux buses 1 and 2. These sends are commonly used as effects sends, but can be used for additional headphone feed or other purposes.

A common on/off switch is used to silently engage both mono sends. The post-fader signal feeds AUX 1/2 by default, but the pair can be switched to the pre-fader signal.



AUX 1-AUX 2: Level controls for AUX 1/2
- Knob = 1, ring = 2

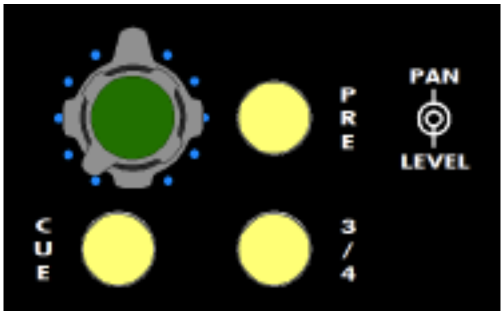
PRE (pre-post): Routes the pre-fader signal to AUX 1/2 when engaged
- AUX 1/2 are fed post-fader by default
- Illuminates when engaged

1/2 (on/off): On/off switches for AUX 1/2
- Illuminates when engaged

3.4.2 Stereo Aux Send 3/4

AUX 3/4 is a stereo auxiliary send that feeds aux buses 3 and 4 by default. These sends can also be routed to the stereo cue bus to quickly create a fully featured headphone feed when recording. When mixing, stereo AUX 3/4 or CUE can be used for effects sends.

A common on/off switch is used to silently engage the stereo send. The post-fader signal feeds AUX 3/4-CUE by default, but the pair can be switched to the pre-fader signal. In common practice, post-fader sends are often used for effects sends and pre-fader sends are typically used for headphone feeds.



AUX 3/4: Level controls for AUX 3/4 or CUE
- Knob = level, ring = pan

PRE (pre-post): Routes the pre-fader signal to AUX 3/4-CUE when engaged
- AUX 3/4-CUE are fed post-fader by default
- Illuminates when engaged

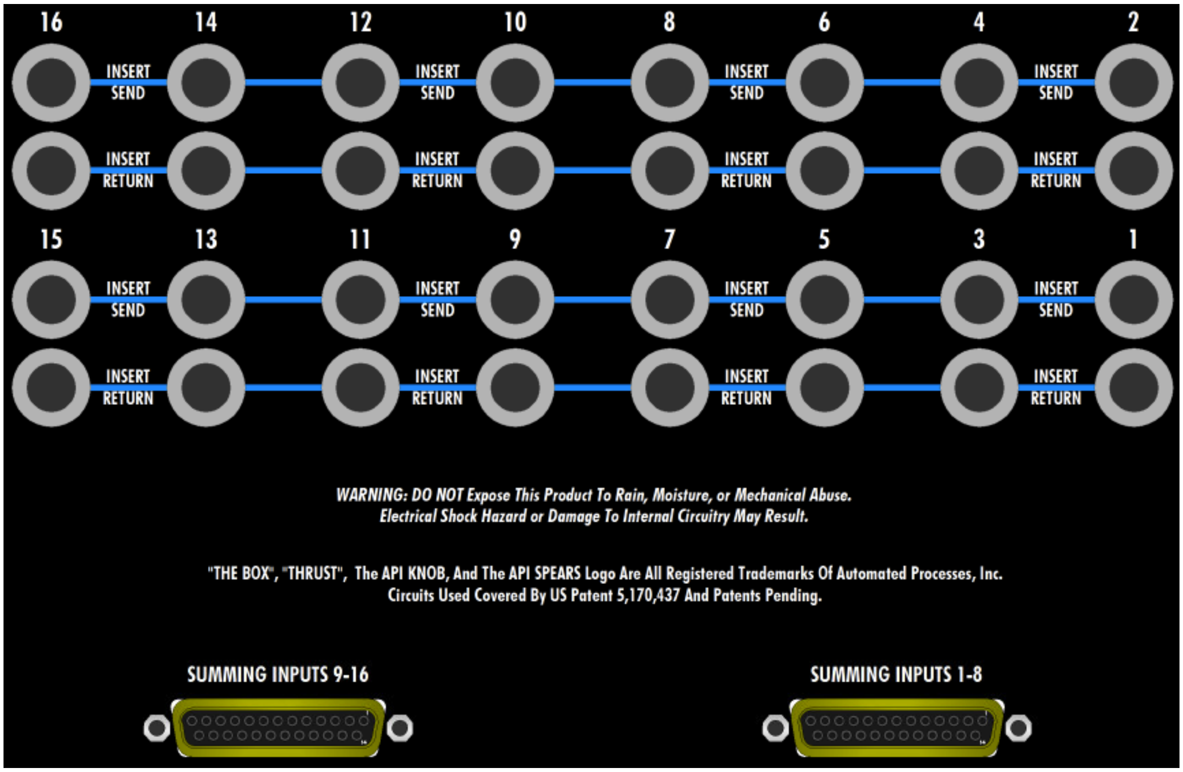
3/4 (on/off): On/off switches for AUX 3/4-CUE
- Illuminates when engaged

3.4.3 Stereo Cue Sends

In addition to auxiliary buses 1-4, "The BOX" is equipped with a separate stereo cue bus, cue master, and balanced output. When recording, the primary purpose of this cue system is the creation of a headphone feed, so talkback and other sources can be routed to the cue master. When mixing, stereo send 3/4 can be routed to the cue bus instead of aux buses 3/4 to provide an alternate stereo effects send.

CUE: Routes aux send 3/4 to the stereo cue bus when engaged
- AUX 3/4 feeds auxiliary buses 3 and 4 by default
- The feed to auxiliary buses 3 and 4 is defeated when engaged
- Illuminates when engaged

3.5 Summing Input Rear Panel Connections

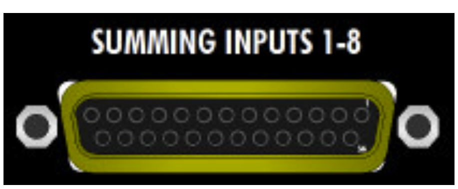


The rear panel provides a comprehensive and flexible set of summing input connections. Connections include:

- Line inputs
- Insert sends and returns

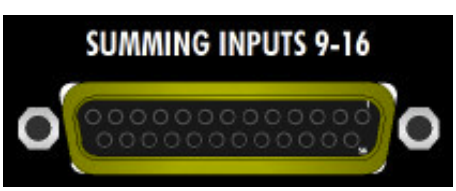
3.5.1 Summing Input Line Input Connections

The line inputs for the sixteen summing inputs are distributed across two DB-25 connectors.



SUMMING INPUTS 1-8: DB-25, balanced, low-impedance, line-level

- Standard pin-out: 1-8=summing inputs 1-8



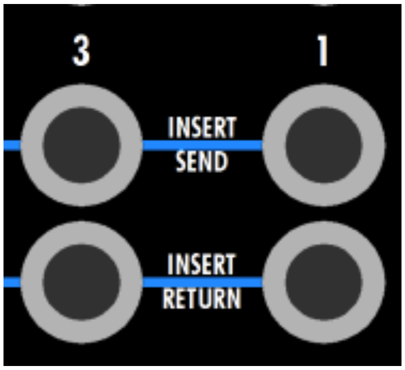
SUMMING INPUTS 9-16: DB-25, balanced, low-impedance, line-level

- Standard pin-out: 1-8=summing inputs 9-16

3.5.2 Summing Input Insert Connections

All insert sends and returns are balanced, low-impedance, line-level signals routed via ¼" tip-ring-sleeve jacks on the rear panel.

The insert send is fed immediately after the input on summing inputs 1-16.

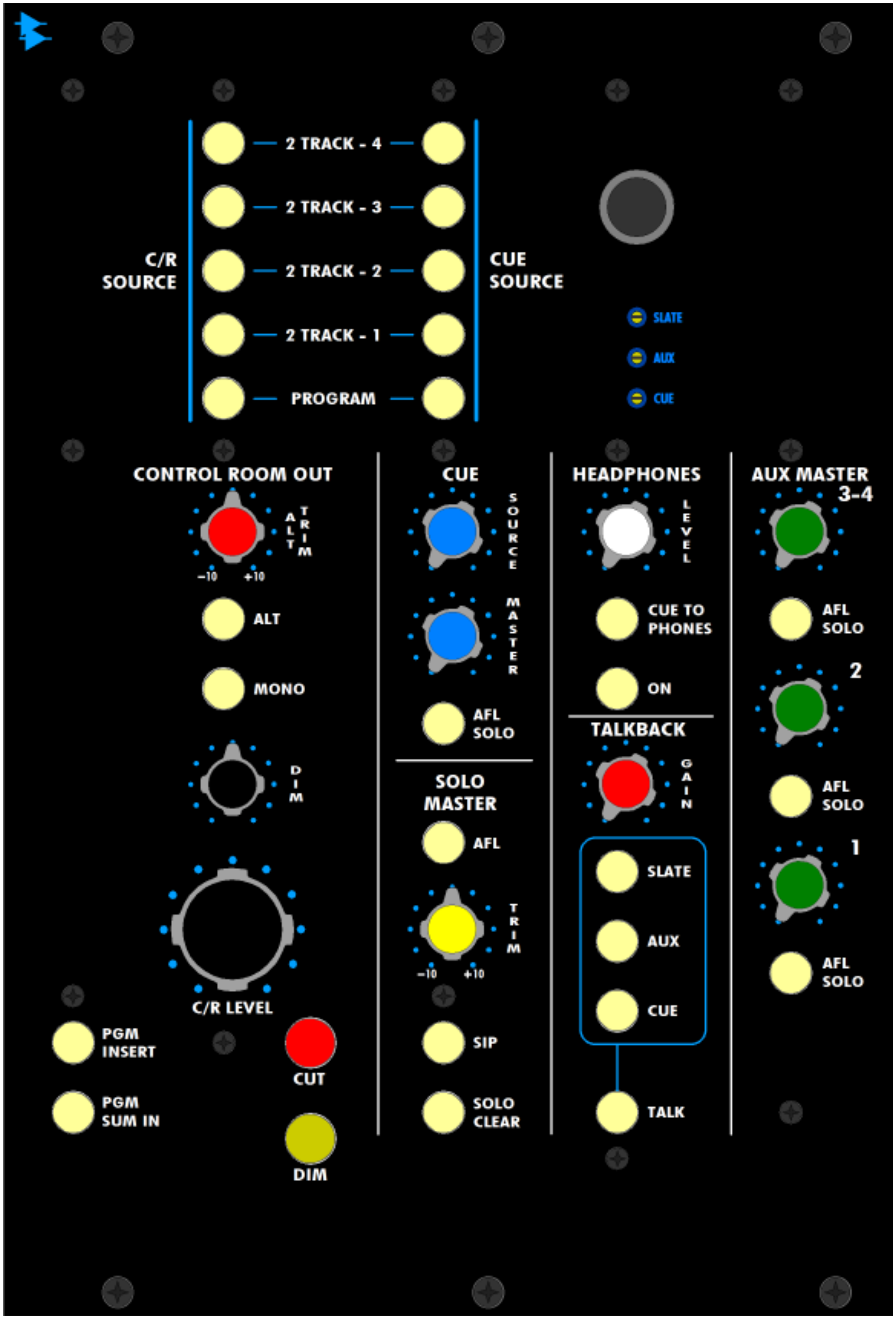


INSERT SEND: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

INSERT RETURN: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

4.0 Master Section

The master section of "The BOX" is equipped with a comprehensive set of components needed to support a wide range of recording and mixing applications.



The master section includes the following facilities:

- Stereo program bus master: PGM SUM IN, PGM INSERT, and master fader
- Auxiliary bus masters: AUX MASTER 1-4
- Cue master: CUE
- Solo master: SOLO MASTER
- Control room monitor master: CONTROL ROOM OUT
- Talkback master: TALKBACK
- Headphone master: HEADPHONES

Subsequent sections of this manual will explore each component of the master section in detail.

5.0 Stereo Program Master

"The BOX" stereo program bus employs the same "Active-Combining-Amplifier" (ACA) technology found in larger API consoles. This high-quality mix bus is a key feature of "The BOX" that offers depth, punch, and warmth that's only possible with analog summing.

The stereo program master is equipped with a comprehensive set of features that support a wide range of mixing applications. These features include:

- High-quality stereo mix bus (ACA)
- External summing input (PGM SUM INPUT) with switch
- Assignable stereo 527 compressor
- Balanced insert send/return with switch
- Stereo master fader
- Balanced line-level output

The default signal flow through the program master is as follows:

Program sum inputs
⇓
Channels ⇒ program bus ⇒ compressor ⇒ insert ⇒ master fader ⇒ output

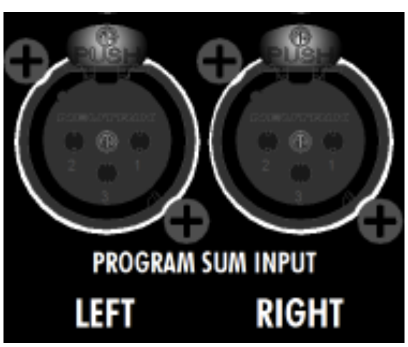
5.1 Program Sum Inputs

"The BOX" is equipped with a stereo pair of external inputs that sum with the program bus. This is useful for adding an external stereo source, such as a sub-mix to the program bus in "The BOX." These inputs are added to the ACA and sums with the program bus audio at unity gain.

PGM SUM IN (program sum input): Adds the PROGRAM SUM INPUTS on the rear panel to the program bus at unity gain

- Located pre-compressor and insert
- Illuminates when engaged

Rear Panel Program Sum Input Connections

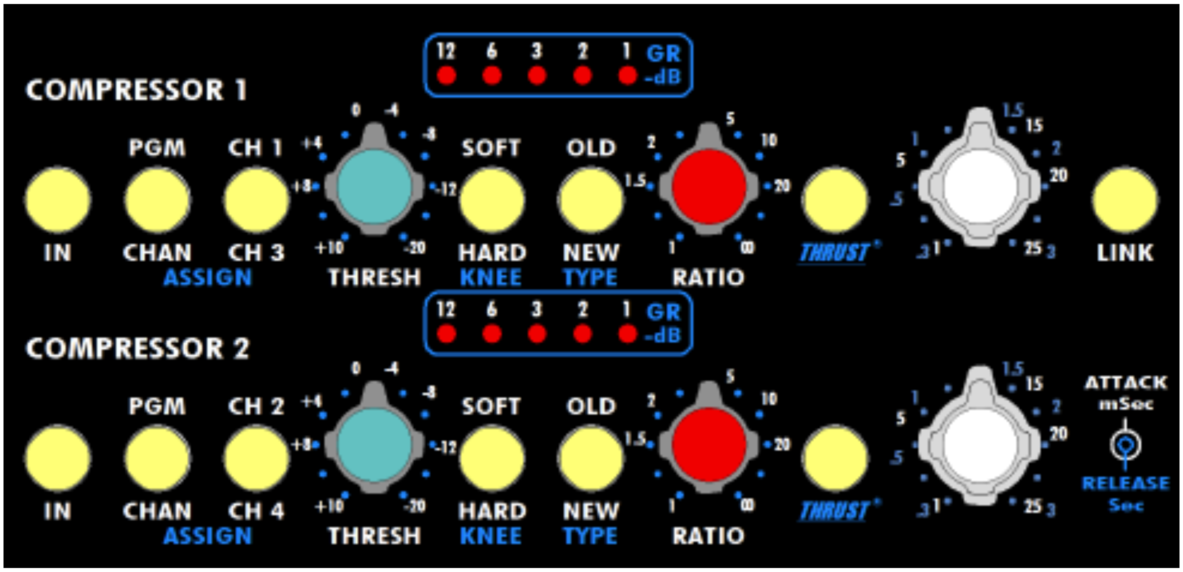


PROGRAM SUM INPUT (LEFT & RIGHT):
3-pin female XLR, balanced, low-impedance, line-level

- Located pre-insert and compressor

5.2 Program Bus Compression

"The BOX" includes two API 527 compressor/limiters in the center section of the console. These compressors are assigned to the program bus by default for mixing, but can be routed to the input channels while recording.



To use the compressors while mixing, they need to be assigned to the program bus, with Compressor 1 assigned to Left side of the program bus and Compressor 2 assigned to the Right. The compressors are located after the PROGRAM SUM INPUT and pre-insert.

The compressors are assigned to the program bus by default. When PGM-CHAN is engaged, the compressor is assigned to the selected input channel. These switches illuminate when engaged. Assignment to the program bus is accomplished by disengaging the PGM-CHAN switches on the compressor.

The compressor will be bypassed unless the IN switch on the compressor is engaged. Engaging the IN switch will place the compressor in the signal path. This switch illuminates when engaged.

Stereo compression during mixing is possible when both compressors are assigned to the program bus and the LINK switch is engaged. This switch illuminates when engaged.

NOTE: Both compressors must be IN, assigned to the program bus, with the link engaged to apply proper stereo mix compression. When LINK is engaged the gain reduction control voltage is summed. To assure that gain reduction is keyed equally by both left and right channels it is essential to set parameters on both compressors exactly the same.

Refer to section 6.0 Compressors for more information regarding the compressors.

5.3 Program Bus Inserts

The program bus is equipped with a stereo balanced insert with switch.

PGM INSERT (program insert): Routes the program insert return to the signal path

- The program insert return is active only when the PGM INSERT switch is engaged
- Illuminates when engaged

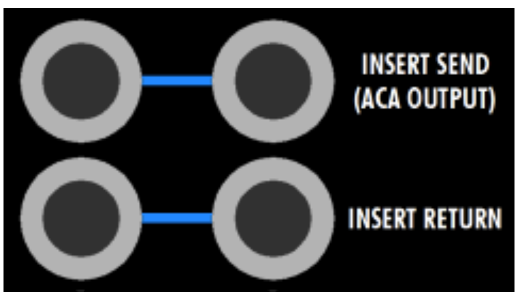
The program insert sends are fed pre-fader immediately after the PROGRAM SUM INPUTS and compressors and are always active.

The program insert returns are located before the master fader and are routed to the signal path only when the PGM INSERT switch is engaged.

Rear Panel Program Insert Connections

The program bus insert sends and returns are balanced, low-impedance, line-level signals routed via ¼" tip-ring-sleeve jacks on the rear panel.

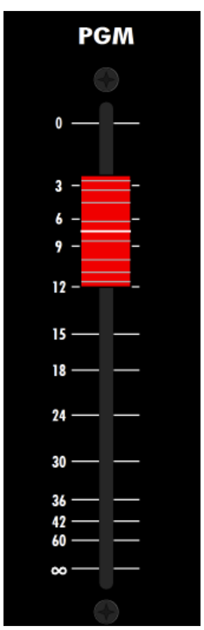
The insert send is located after the compressors on the left and right program buses.



INSERT SEND (ACA OUTPUT) (LEFT & RIGHT): ¼" tip-ring-sleeve, balanced, low-impedance, line-level

INSERT RETURN (LEFT & RIGHT): ¼" tip-ring-sleeve, balanced, low-impedance, line-level

5.4 Program Master Fader



The fader with the red fader cap below the master section is the stereo program master fader. The master fader handles the left and right program bus audio on a single stereo fader.

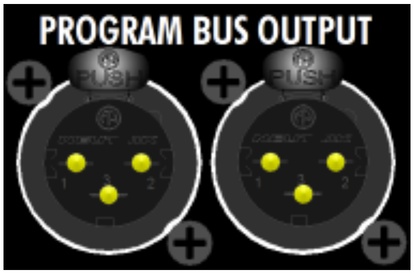
The master fader is located post-insert. It controls the primary console stereo program output level and is routed directly to the PROGRAM BUS OUTPUT connectors on the rear panel.

MASTER FADER: Primary stereo output level control

- 100mm full-size audio-taper fader
- -∞dB to 0dB range
- 0dB is unity gain
- Red fader cap

5.5 Program Bus Outputs

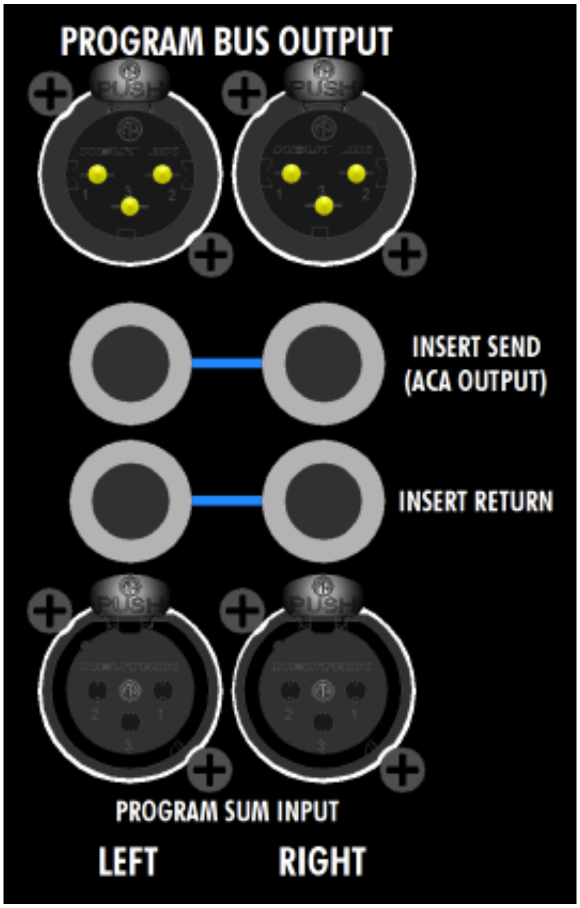
"The BOX" is equipped with stereo, balanced, low-impedance line-level program outputs terminated with XLR connectors.



PROGRAM BUS OUTPUT (LEFT & RIGHT): 3-pin male XLR, balanced, low-impedance, line-level

- Located post-master fader

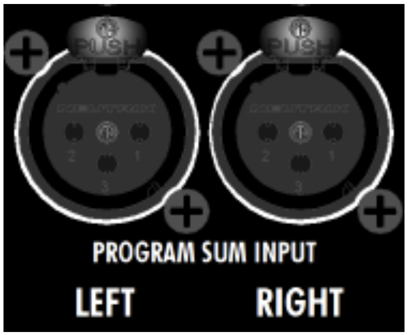
5.6 Program Bus Rear Panel Connections



The rear panel provides a comprehensive and flexible set of stereo program master connections. LEFT and RIGHT program bus connections include:

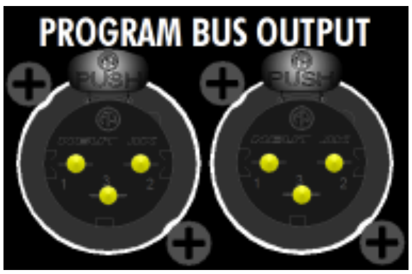
- Program bus outputs
- Insert sends
- Insert returns
- Program sum input

5.6.1 Program Sum Input Connections



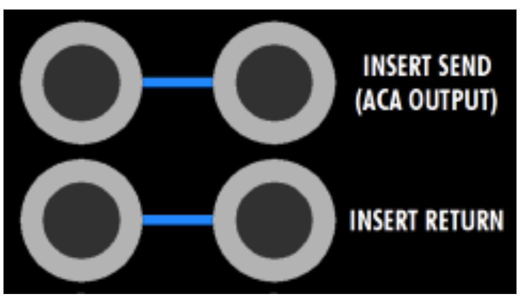
PROGRAM SUM INPUT (LEFT & RIGHT): 3-pin female XLR, balanced, low-impedance, line-level
- Located pre-insert and compressor

5.6.2 Program Bus Outputs



PROGRAM BUS OUTPUT (LEFT & RIGHT): 3-pin male XLR, balanced, low-impedance, line-level
- Located post-master fader

5.6.3 Rear Panel Program Insert Connections

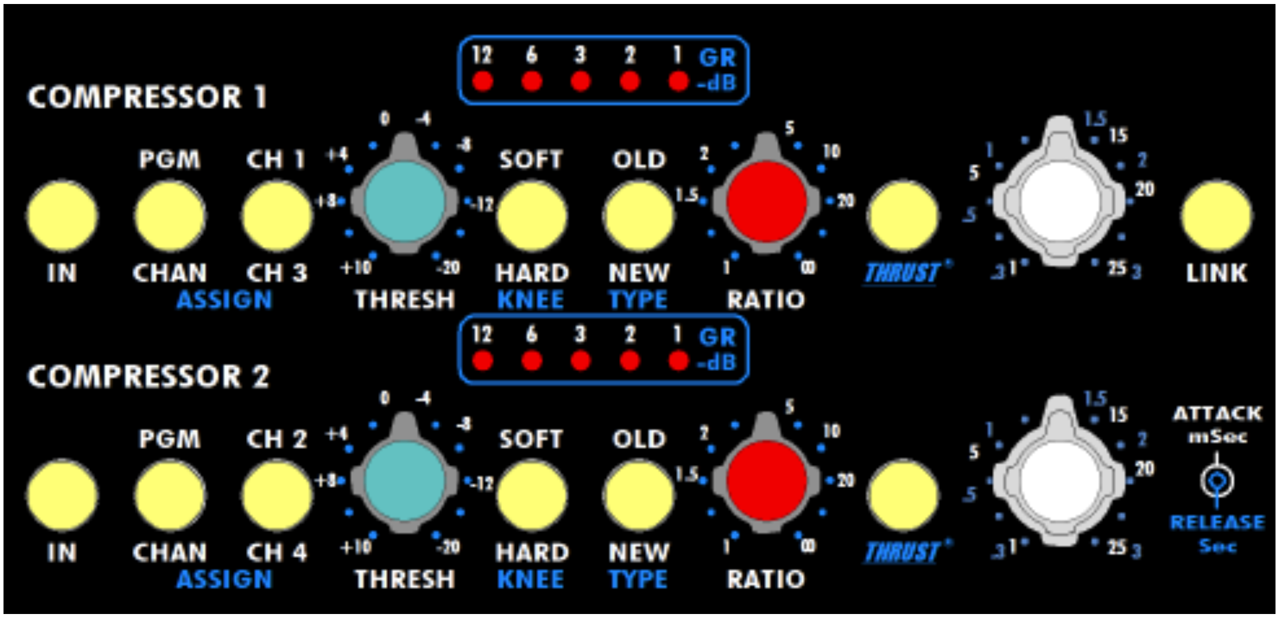


INSERT SEND (ACA OUTPUT) (LEFT & RIGHT): ¼" tip-ring-sleeve, balanced, low-impedance, line-level

INSERT RETURN (LEFT & RIGHT): ¼" tip-ring-sleeve, balanced, low-impedance, line-level

6.0 527 Compressors

"The BOX" is equipped with two API 527 compressor/limiters that are located above the master section of the console. These compressors are assigned to the program bus by default for mixing, but can be routed to the input channels while recording.



The API 527 Compressor takes it's place alongside the family of API VCA based compressors, the 225L and the 2500 Stereo Bus Compressor. Anyone familiar with those units will immediately be at home with the 527.

Features common to the line like "feed forward" (NEW) and "feed back" (OLD) gain reduction methods, selectable on the front panel, provide a choice of "that old way", or "the new way" of compression, for the highest level of flexibility in signal gain control. The "old way" or Feed-Back method is what most of the classic compressors used for the gain control circuit. The "new way" gain reduction is more typical of the newer VCA type compressors that rely on RMS detectors for the gain control voltage.

There is a "SOFT"/"HARD" KNEE switch for an "over-easy" type compression resulting in a very natural, uncompressed sound or a typical sharp knee type that lends itself to a much more severe limiting effect.

The patented ***THRUST***® function can be switched in and out as well, applying a high pass filter before the RMS detector circuit that preserves that punchy bottom end.

The output level remains fairly constant regardless of the threshold or ratio control, much like the "more/less" Ceiling control on the API 525 Compressor. This allows for live adjustments without any noticeable gain changes in the program level.

The 527 Compressor makes use of the 2520 discrete op-amps and exhibits the reliability, long life, and signature sound, which are characteristic of API products.

The 527 compressors in "The BOX" have the following features:

- Feed-forward or feed-back compression
- Hard or soft knee compression
- Patented ***THRUST***® switch for frequency dependent side chain control
- Continuously variable detented THRESHOLD control
- Continuously variable detented ATTACK and RELEASE controls
- Continuously variable detented RATIO control
- 5-segment gain reduction (GR) meter
- LINK switch for stereo compression
- Audio circuit uses the 2520 discrete op-amps

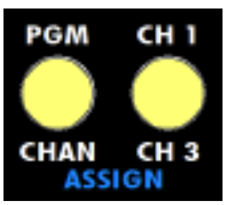
The 527 compressors in "The BOX" are identical to the API 527 modules and the 527 in API's "The Channel Strip," with the exception of the gain reduction meter and the a side-chain input on the TCS. The compressors in "The BOX" do not have a side-chain and the gain reduction meter has fewer LED segments.

The compressors in "The BOX" can be linked together allowing for stereo compression of the program bus when mixing or stereo inputs when recording.

Additional information about the 527 compressor can be found at:
http://apiaudio.com/527.html

6.1 Input Channel Routing

To use the compressors while recording, they need to be assigned to the input channels. Compressor 1 can be assigned to input channels 1 or 3. Likewise, Compressor 2 can be assigned to input channels 2 or 4.



Assignments to input channels are accomplished using the two ASSIGN switches on the compressors. Engaging the PGM-CHAN switch will move the compressor from the program bus to the input channels. When PGM-CHAN is engaged, compressor 1 is assigned to input channel 1 and compressor 2 is assigned to input channel 2 by default. Engaging CH 3 or CH 4 assigns the compressors to these channels. These switches illuminate when engaged.

When assigned to an input channel, the compressor will be located after the preamp and before the equalizer/500 slot and insert by default. It can be moved to after the equalizer/500 slot and insert by engaging the COMP POST switch. This switch illuminates when engaged.

The compressor will be bypassed unless the IN switch on the compressor is engaged. Engaging the IN switch will place the compressor in the signal path. This switch illuminates when engaged.

Stereo compression during recording is possible when both compressors are assigned to input channels and the LINK switch is engaged. This switch illuminates when engaged.

6.2 Program Bus Routing & Link

To use the compressors while mixing, they need to be assigned to the program bus, with Compressor 1 assigned to Left side of the program bus and Compressor 2 assigned to the Right. The compressors are located after the PROGRAM SUM INPUT and before the insert send.

The compressors are assigned to the program bus by default. Assignment to the program bus is accomplished by disengaging the PGM-CHAN switches on the compressor. When PGM-CHAN is engaged, the compressor is assigned to the selected input channel. These switches illuminate when engaged.

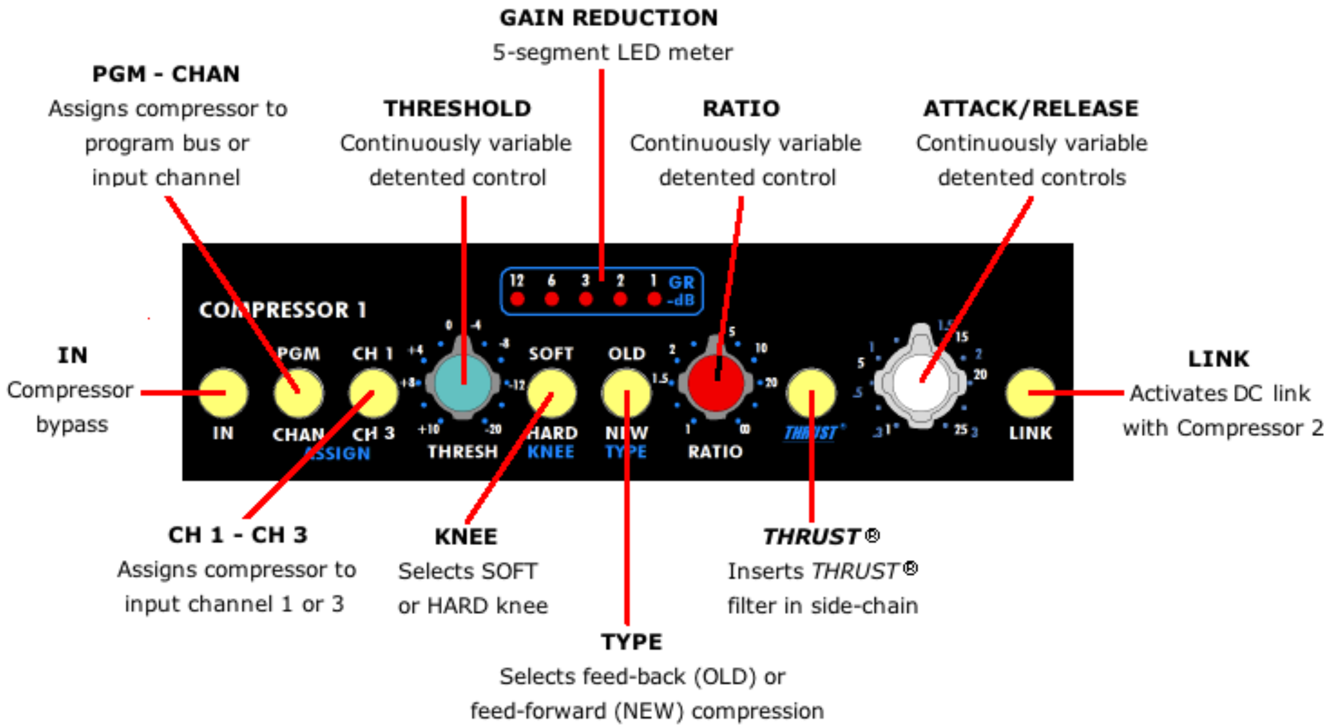
The compressor will be bypassed unless the IN switch on the compressor is engaged. Engaging the IN switch will place the compressor in the signal path. This switch illuminates when engaged.

Stereo compression during mixing is possible when both compressors are assigned to the program bus and the LINK switch is engaged. This switch illuminates when engaged.

NOTE: Both compressors must be IN, assigned to the program bus, with the link engaged to apply proper stereo mix compression. When LINK is engaged the gain reduction control voltage is summed. To assure that gain reduction is keyed equally by both left and right channels it's essential to set parameters on both compressors exactly the same.

6.3 Compressor Controls

The 527 compressor provide a complete set of functions and parameters, including the unique *THRUST*® side-chain filter.



6.3.1 Compressor Routing Controls

IN: Places the compressor in the signal path when engaged

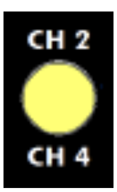
- Silent operation
- Illuminates when engaged

PGM-CHAN (program-channel): Toggles compressor routing between the program bus and input channels

- The compressor is assigned to the program bus when not engaged
- Assigns the compressor to the selected input channel when engaged
- Illuminates when engaged

CH 1 - CH 3 (channel 1 – channel 3): Toggles compressor 1 routing between input channels 1 and 3

- Compressor 1 is routed to input channel 1 when not engaged
- Compressor 1 is routed to input channel 3 when engaged
- Illuminates when engaged



CH 2 - CH 4 (channel 2 – channel 4): Toggles compressor 2 routing between input channels 2 and 4

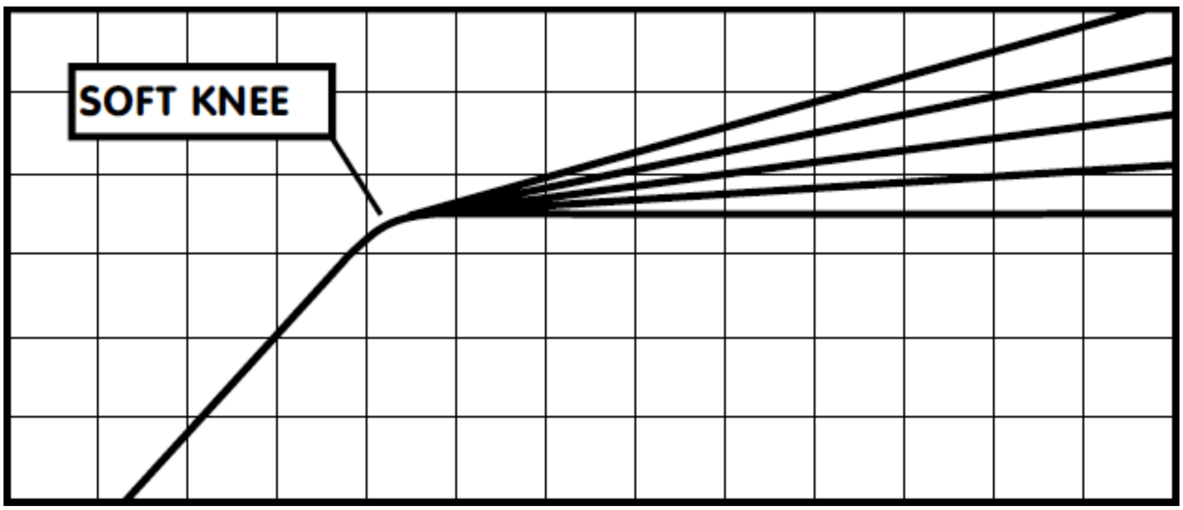
- Compressor 2 is routed to input channel 2 when not engaged
- Compressor 2 is routed to input channel 4 when engaged
- Illuminates when engaged

6.3.2 Compressor Configuration Controls

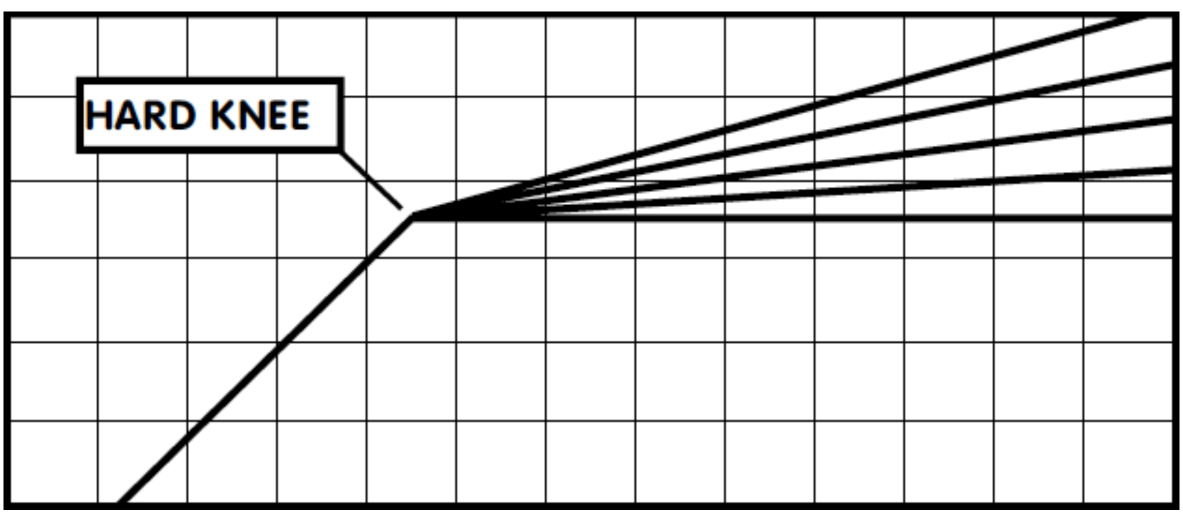
Knee

KNEE (SOFT-HARD): Toggles between soft and hard knee compression
- Changes the shape of the compression curve at the rotation point
- SOFT is selected when not engaged
- Hard is selected when engaged
- Illuminates when engaged

SOFT: Rounded response curve
- Gradual onset of compression (fade-in up to the set ratio)
- Similar to an "over-easy" type knee
- More transparent



HARD: Sharp response curve
- Immediate onset of compression (sudden transition to set ratio)
- More aggressive and noticeable



Compressor Circuit Topologies

The 527 compressors can be set to operate in two circuit topologies:

- OLD: Feed-back: The RMS detector receives the signal from after the VCA
- NEW: Feed-forward: The RMS detector receives the signal from before the VCA

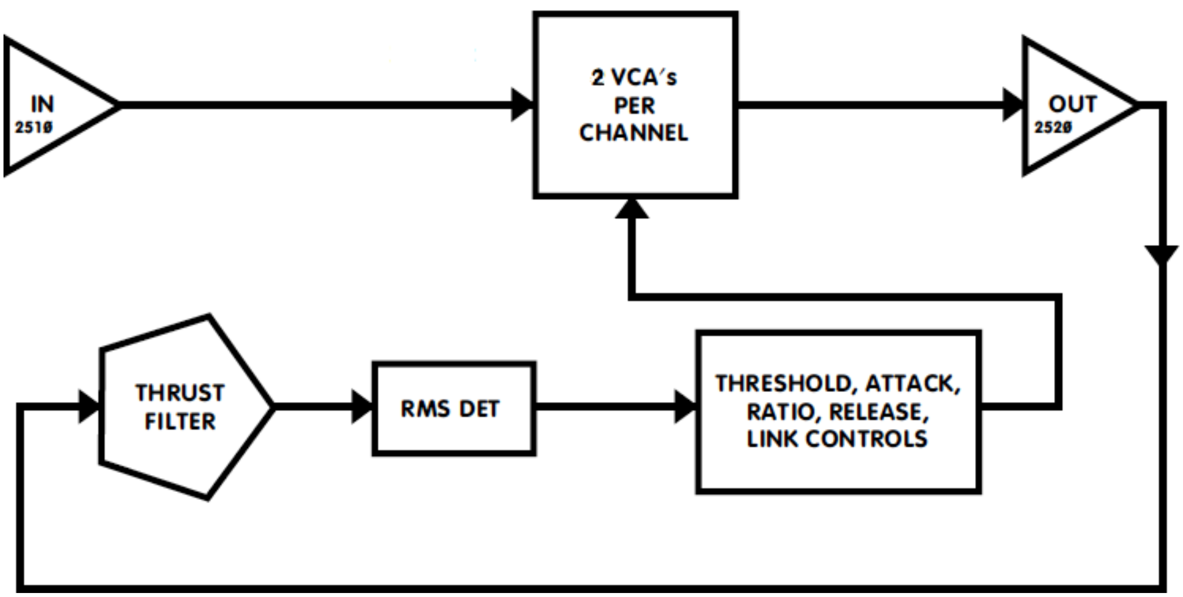
TYPE (NEW-OLD): Toggles between feed-back and feed-forward compressor topologies

- OLD feed-back is selected when not engaged
- NEW feed-forward is selected when engaged
- Illuminates when engaged

OLD: Feed-Back Compression

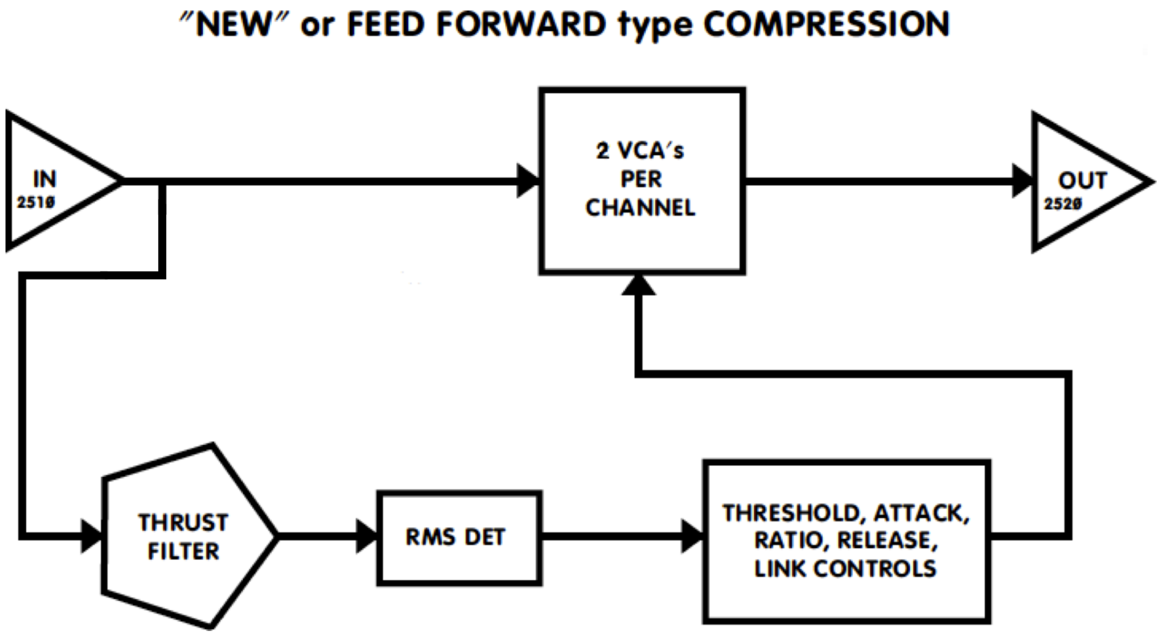
In a feed-back compressor, the RMS detector gets its signal from the output of the gain reduction device (VCA). This is how older API 525, 1176 type, and 660 type compressors work. This yields a smoother, softer, more transparent sound.

"OLD" or FEED BACK type COMPRESSION



NEW: Feed-Forward Compression

In a feed-forward compressor, the RMS detector normally gets its signal from a split of the input signal. (The detector path can alternately get its signal from a Side Chain Input.) With this method, the RMS detector sends a signal to the VCA that is an exact ratio of the desired compression set by the RATIO control. This is how many new VCA based compressors work. This can yield more aggressive compression and a harder, more affected sound.



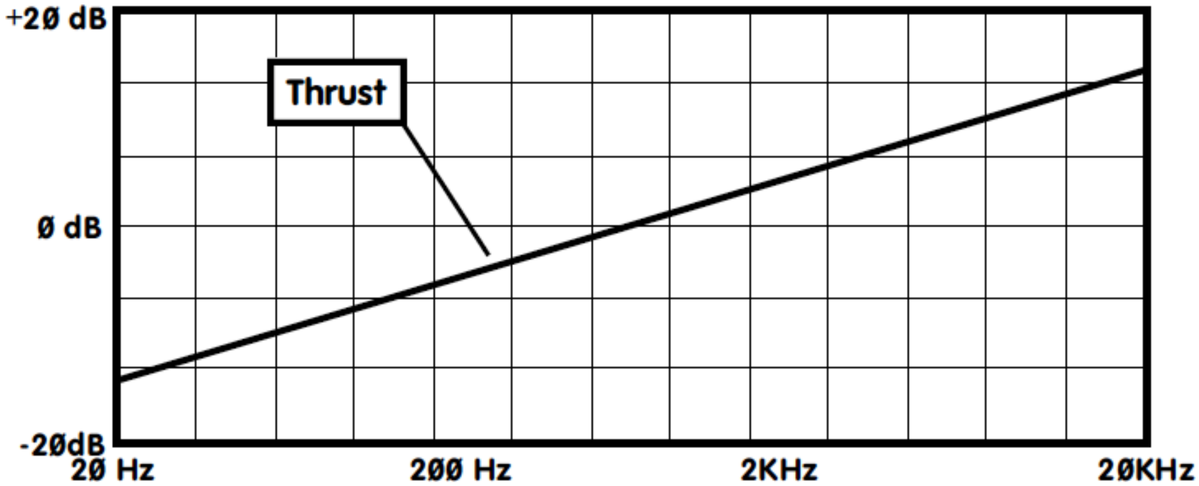
THRUST®

The 527 compressor includes API's patented ***THRUST***® circuit that can be switched in or out as needed. This places the ***THRUST***® filter before the RMS detector that decreases the compressor's reaction to low frequency content. The result is a noticeable increase of punch and low frequencies, but a uniformly compressed signal. It's the "little more punch" switch!

THRUST®: Inserts the *THRUST*® filter before the RMS detector
- Illuminates when engaged

The patented ***THRUST***® circuit has been used for many years in the famed API 2500 Stereo Compressor, ATI Paragon and Paragon II consoles, as well as the Pro-6 Input Strip. This circuit places a filter in front of the RMS detector with a slope of 10dB per decade (-3dB/8va), which is the inverse of the pink noise energy curve. In acoustics, the pink noise curve is used to equalize energy vs. frequency over the audio spectrum, as sound requires more low frequency energy than high frequency energy to sound correct to your ear. In Hi-fi equipment, a "LOUDNESS" contour is used to equalize the music at lower levels so it sounds correct. Even with this curve, there is still a substantial amount of low frequency information compared to high frequency information in the audio signal path. When that signal is fed into the RMS detector, the detector will process the signal into a DC control voltage based upon the those louder low frequencies, resulting in a control voltage that favors the low frequencies of the signal, causing pumping and a loss of punch. Sometimes, this is not desirable. By engaging the ***THRUST***® switch, this inverse filter is placed in front of the RMS detector, evening out the energy by lowering the energy in the low frequencies and increasing the energy in the high frequencies, so each octave has the same energy instead of each octave having half the energy as the one lower. This creates a unique compression effect that still reduces the overall gain, but the sound is much more punchy and the signal actually sounds less compressed.

THRUST® filter response



Stereo Link

LINK: Activated DC link between compressions 1 and 2

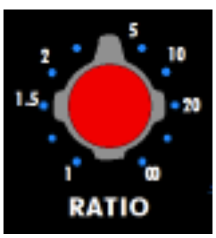
- Stereo compression when engaged
- Sums the DC gain reduction voltage of both units
- Illuminates when engaged

NOTE: Both compressors must be IN, assigned to the program bus, with the link engaged to apply proper stereo mix compression. When LINK is engaged the gain reduction control voltage is summed. To assure that gain reduction is keyed equally by both left and right channels it is essential to set parameters on both compressors exactly the same.

6.3.3 Compressor Parameter Controls

THRESH (threshold): Sets the level at which compression begins

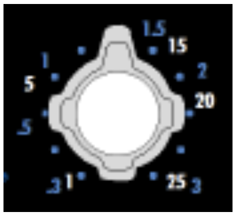
- Continuously variable between +10dBu and -20dBu
- Detented rotary pot for easy recall



RATIO: Sets the compression ratio of input vs. output levels for signals that fall above the set threshold

- Continuously variable between 1:1 and ∞:1
- A ratio of 10:1 or greater is generally considered to be limiting
- Detented rotary pot for easy recall

NOTE: Automatic make-up gain is applied so the compressor output remains fairly constant regardless of the threshold and ratio settings. This allows for live adjustments to be made with noticeable changes in perceived level.



Attack and release times are fully variable on the 527 compressor and share a dual-concentric potentiometer for control. Attack time is adjusted using the outer ring of the pot and Release time is adjusted using the inner knob.

<u>ATTACK (ring)</u>: Sets the time it takes the compressor to react when the input level exceeds the set threshold

- Continuously variable between 1 and 25 milliseconds (msec)
- Attack times are indicated with white numbers
- Detented rotary pot for easy recall

<u>RELEASE</u>: Sets the time it takes the compressor to recover to unity gain after the level falls below the set threshold

- Continuously variable between .3 and 3 seconds (sec)
- Release times are indicated with blue numbers
- Detented rotary pot for easy recall

6.3.4 Gain Reduction Meter

A 5-segment LED gain reduction (GR) meter is provided to indicate the amount of compression being applied.

When no gain reduction is being applied, none of the LED's are lit on the Gain Reduction meter (GR). As increasing amounts of compression occurs, the corresponding LED's illuminate to indicate the amount of gain reduction.

The following gain reduction increments are provided:

- -1dB
- -2dB
- -3dB
- -6dB
- -12dB

7.0 Auxiliary Masters

"The Box" is equipped with four (4) auxiliary buses arranged as two mono buses (1 and 2) and one stereo bus (3/4). Accordingly, the console is fitted with matching aux masters in the same configuration. The aux system provides effects sends while mixing, additional headphone feeds while recording, and extra routable sends for other applications.

The auxiliary signal flow is as follows:

Channel sends ⇒ aux bus ⇒ aux master ⇒ output

7.1 Auxiliary Master Controls

AUX MASTER 3-4: Output level control for auxiliary buses 3 and 4

- Stereo aux master with one ganged potentiometer
- Feeds the AUX OUTPUTS / CUE OUTPUTS DB-25 connector on the rear panel

AUX 3-4 AFL SOLO: Activates the AFL or PFL solo mode*

AUX MASTER 2: Output level control for auxiliary bus 2

- Mono aux master
- Feeds the AUX OUTPUTS / CUE OUTPUTS DB-25 connector on the rear panel

AUX 2 AFL SOLO: Activates the AFL or PFL solo mode*

AUX MASTER 1: Output level control for auxiliary bus 1

- Mono aux master
- Feeds the AUX OUTPUTS / CUE OUTPUTS DB-25 connector on the rear panel

AUX 1 AFL SOLO: Activates the AFL or PFL solo mode*

* Engaging an aux master AFL SOLO switch will activate the current PFL or AFL solo function. The solo function is determined by the position of the AFL switch in the solo master section. Engaging an aux master AFL SOLO while in *solo-in-place (SIP)* mode will not active the SIP solo function.

The aux masters provide +6dB of gain at the fully-clockwise position. For most applications the masters should be initially set to unity gain (approximately 2 o'clock). From here, the masters can be used to trim the aux outputs upward or downward as needed.

7.1.1 Aux Master Solos

Each aux master can be soloed using either the AFL or PFL solo mode.

When AFL is the current solo mode, mono aux masters 1 & 2 feed the LEFT and RIGHT sides of the stereo AFL solo bus in mono. Stereo aux master 3-4 feeds the output of aux bus 3 to LEFT and aux bus 4 to RIGHT. The AFL solo bus is fed immediately after the aux master potentiometers.

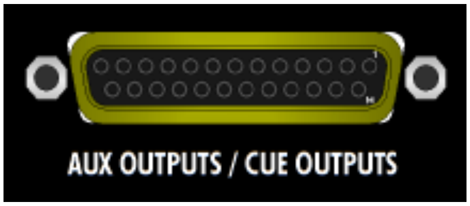
When PFL is the current solo mode, mono aux masters 1 & 2 feed the mono PFL solo bus in mono. The stereo aux master 3-4 outputs are summed together and fed to the PFL solo bus in mono. Even though the console is in PFL, the PFL solo bus is fed immediately after the AUX MASTER potentiometers.

NOTE: When solo-in-place (SIP) is the current solo mode, engaging an aux master AFL SOLO switch will activate the current PFL or AFL solo function. The solo function is determined by the position of the AFL switch in the solo master section. Engaging an aux master AFL SOLO while in SIP mode will not active SIP mutes on channels.

AFL SOLO: Activates the AFL or PFL solo function when engaged

- Labeled AFL because the selected solo bus is always fed after the aux master pot (no PFL solo function on aux masters)
- Can be cleared by pressing the SOLO CLEAR button
- Illuminates when engaged

7.2 Auxiliary Master Rear Panel Connections



AUX OUTPUTS / CUE OUTPUTS: DB-25, balanced, low-impedance, line-level

- Fed from the aux and cue masters
- Standard pin-out: 1-4=aux outputs 1-4, 5-6=cue outputs L-R

8.0 Cue Master

In addition to the four auxiliary buses, "The BOX" is equipped with a separate stereo cue bus and associated cue master. The cue system supports headphone feeds while recording and can be used as an alternate stereo effects send while mixing.

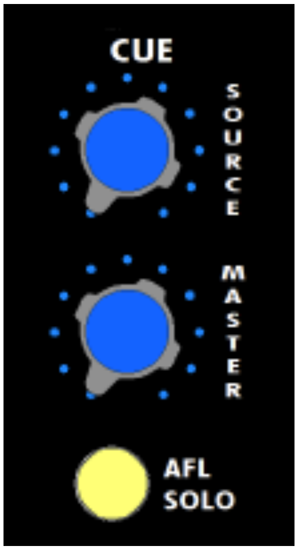
The cue system signal flow is as follows:

Cue source selectors ⇒ cue source pot
⇓
Channel sends ⇒ cue bus ⇒ cue master pot ⇒ output
⇑
talkback

The stereo cue bus is the primary source for the cue master. Additional stereo sources can be added using the CUE SOURCE selectors and level control. Sources are selected using the CUE SOURCE switches. The level of the selected sources feeding the CUE MASTER is controlled by the CUE SOURCE potentiometer. The CUE MASTER potentiometer is fed from the cue bus and the CUE SOURCE pot and controls the overall output of the cue system.

To support headphones (cue feeds) while recording, talkback can be added and the cue master output can be routed to the headphones master as well as the AUX OUTPUTS / CUE OUTPUTS connector on the rear panel.

8.1 Cue Master Controls



CUE SOURCE: Level control for the currently selected CUE SOURCE switches
- Stereo control with one ganged potentiometer
- Feeds the CUE MASTER pot

CUE MASTER: Overall cue output level control
- Stereo control with one ganged potentiometer
- Feeds the AUX OUTPUTS / CUE OUTPUTS DB-25 connector on the rear panel

CUE AFL SOLO: Activates the AFL or PFL solo mode*

* Engaging the cue master AFL SOLO switch will activate the current PFL or AFL solo function. The solo function is determined by the position of the AFL switch in the solo master section. Engaging the cue master AFL SOLO while in *solo-in-place (SIP)* mode will not active the SIP solo function.

The CUE MASTER provides +6dB of gain at the fully-clockwise position. For most applications the masters should be initially set to unity gain (approximately 2 o'clock). From here, the CUE MASTER can be used to trim the cue outputs upward or downward as needed.

8.1.1 Cue Master Solo

The cue master can be soled using either the AFL or PFL solo mode.

When AFL is the current solo mode, the stereo cue master feeds the AFL solo bus in stereo, immediately after the CUE MASTER potentiometer.

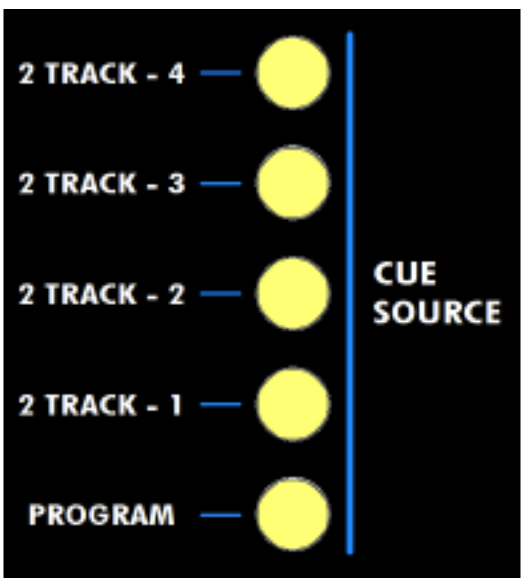
When PFL is the current solo mode, the stereo cue master output is summed together and fed to the PFL solo bus in mono. Even though the console in PFL, the PFL solo bus is fed immediately after the CUE MASTER potentiometer.

NOTE: When solo-in-place (SIP) is the current solo mode, engaging the cue master AFL SOLO switch will activate the current PFL or AFL solo function. The solo function is determined by the position of the AFL switch in the solo master section. Engaging the cue master AFL SOLO while in SIP mode will not active SIP mutes on channels.

AFL SOLO: Activates the AFL or PFL solo function when engaged

- Labeled AFL because the selected solo bus is always fed after the cue master pot (no PFL solo function on the cue master)
- Can be cleared by pressing the SOLO CLEAR button
- Illuminates when engaged

Cue Source Selection



In addition the stereo cue bus, the C/R SOURCE selections (PROGRAM*) and up to four (4) external stereo sources can be routed to the cue master. These additional cue sources are selected using the CUE SOURCE selector switches and routed to the cue master via CUE SOURCE level control.

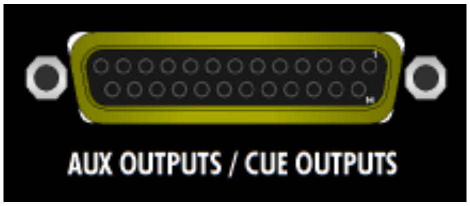
CUE SOURCE: Routes the selected source to the CUE SOURCE level control in the cue master

- PROGRAM*: Output of the C/R SOURCE selectors
- 2 TRACK 1-4: External sources 1-4
- Selections are additive
- Illuminate when engaged

* When PROGRAM is selected, the output of the C/R SOURCE selectors is routed to the CUE SOURCE level control in the cue master. This feed will include the program master outputs only if PROGRAM is selected as the C/R SOURCE and CUE SOURCE. Refer to section 11.1 Control Room Source Selectors for additional information.

Cue source selections are additive, so more than one source can be selected. Each source is added in stereo at unity gain. CUE SOURCE selections are fed to the CUE MASTER via the CUE SOURCE level control.

8.2 Cue Master Rear Panel Connections



AUX OUTPUTS / CUE OUTPUTS: DB-25, balanced, low-impedance, line-level

- Fed from the aux and cue masters
- Standard pin-out: 1-4=aux outputs 1-4, 5-6=cue outputs L-R

9.0 Talkback

“The BOX” is equipped with a complete talkback system for use when recording. The talkback systems includes:

- Internal electret microphone
- Level control
- Momentary talk button
- Preset routing

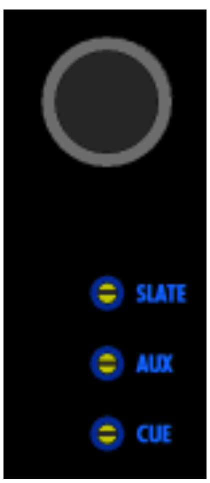
Talkback can be routed to the following:

- <u>SLATE</u>: Program bus
- <u>AUX</u>: Auxiliary masters
- <u>CUE</u>: Cue master

The talkback signal flow is as follows:

⇒ ⇒ ⇒ SLATE ⇒ trim ⇒ program bus
⇑
Talkback mic ⇒ preamp ⇒ TALK button ⇒ routing ⇒ ⇒ AUX ⇒ trim ⇒ aux masters
⇓
⇒ ⇒ ⇒ CUE ⇒ trim ⇒ cue master

9.1 Talkback Microphone and Trims



<u>MICROPHONE</u>: Internal talkback microphone

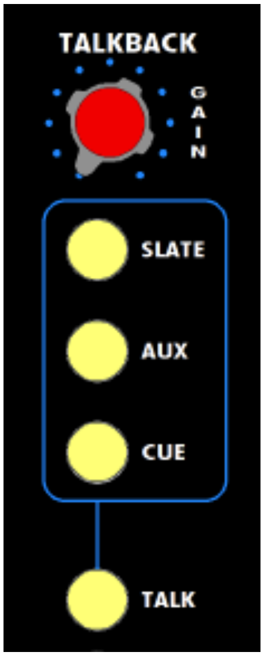
- Electret condenser mic

<u>SLATE</u>: Talkback to program bus calibration trim-pot

<u>AUX</u>: Talkback to auxiliary master calibration trim-pot

<u>CUE</u>: Talkback to cue master calibration trim-pot

9.2 Talkback Routing and Controls



GAIN: Talkback level control

SLATE: Enables talkback routing to the program bus
- Illuminates when engaged

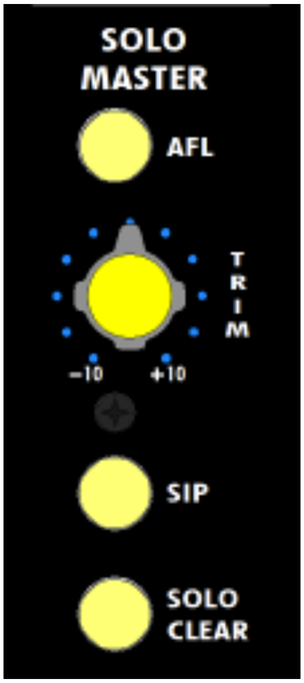
AUX: Enables talkback routing to the auxiliary master
- Illuminates when engaged

CUE: Enables talkback routing to the cue master
- Illuminates when engaged

TALK: Activates talkback to the enabled destination
- Activates loudspeaker DIM function when engaged
- Momentary switch
- Illuminates when engaged

Talkback routing to the desired destination(s) is enabled using the SLATE, AUX, and CUE switches. Talk back to these destinations is activated when the TALK switch is pressed. The overall level to the enabled destinations is set using the GAIN control. Levels to each destination can be trimmed using the SLATE, AUX, and CUE trim-pots.

10.0 Solo System



"The BOX" is equipped with a fully featured solo system that allows the engineer to isolate exactly what needs to be heard in three different ways:

- PFL: Pre-Fader-Listen (default solo mode)
- AFL: After-Fader-Listen
- SIP: Solo-In-Place

The solo system has the following components:

- Channel and master solo switches
- Stereo AFL solo bus
- Mono PFL solo bus
- Solo master

SOLO switches on the input channels and summing inputs activate the current solo mode. AFL SOLO switches on the aux/cue masters also activate the current solo mode, but are useful only when the console is in PFL or AFL solo modes. Refer to sections 7.1.1 and 8.1.1 for additional information.

10.1 Solo Modes

"The BOX" has three solo modes as explained below:

PFL (Pre-Fader-Listen)

Pre-Fader-Listen (PFL) is a mono, non-destructive solo mode. PFL is the default solo mode and is enabled when the AFL and SIP switches are not engaged.

When PFL is the current solo mode and a SOLO switch is engaged on a channel or master, the pre-fader signal is routed to the mono PFL solo bus. The output of the solo master replaces the current control room monitor source and the soloed audio is heard in the loudspeakers in mono. The program bus outputs are unaffected when a PFL solo is engaged (non-destructive).

Since the PFL solo bus is fed pre-fader, the level of the soloed channel may be louder or softer than the unsoloed level since the fader attenuation/gain is not applied. The overall solo level is controlled with the solo TRIM pot.

AFL (After-Fader-Listen)

After-Fader-Listen (AFL) is a stereo, non-destructive solo mode. AFL is active when the AFL switch is engaged and the SIP switch is not. AFL solo can be activated from input channels, summing inputs, and the aux/cue masters.

When AFL is the current solo mode and a SOLO switch is engaged on a channel or master, the post-fader (post-pan-pot) signal is routed to the stereo AFL solo bus. The output of the solo master replaces the current control room monitor source and the soloed audio is heard in the loudspeakers in stereo. The program bus outputs are unaffected when an AFL solo is engaged (non-destructive).

The overall solo level is controlled with the solo TRIM pot.

SIP (Solo-In-Place)

Solo-In-Place (SIP) is a stereo, destructive solo mode. SIP is active when the SIP switch is engaged. SIP solo can be activated from input channels and summing inputs.

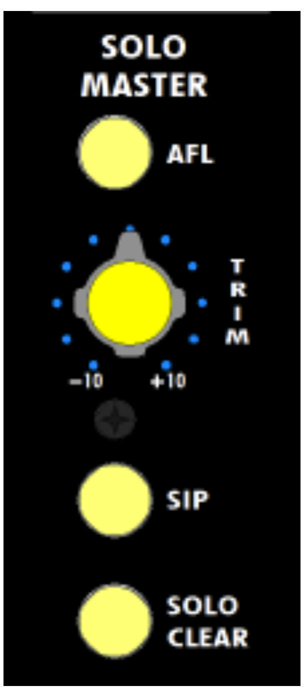
When SIP is the current solo mode and a SOLO switch is engaged on an input channel or summing channel, all non-soloed channels are muted and the soloed channels are heard in the loudspeakers via the program bus. Since all non-soloed channels are muted when a SOLO is engaged, this mode is considered to be “destructive.” Neither solo bus is used and the current control room monitor source is not replaced in SIP mode.

PFL and AFL solo modes are not available when SIP is engaged.

NOTE: When solo-in-place (SIP) is the current solo mode, engaging an aux or cue master AFL SOLO switch will activate the current PFL or AFL solo function. The solo function is determined by the position of the AFL switch in the solo master section. Engaging an aux or cue master AFL SOLO while in SIP mode will not active the SIP solo mode.

Engaging the SAFE switch on a input channel or summing input will prevent it from being muted when a SIP SOLO is engaged on another channel. This is useful on input channels when recording, when channel(s) are being used as effects return, and other applications.

10.2 Solo Master Controls



AFL (after-fader-listen): Enables the after-fader-listen solo mode

- Illuminates when engaged

TRIM: Stereo solo bus level trim-pot

- -10dB to +10dB
- Center detent is 0dB (unity gain)

SIP (solo-in-place): Enables the solo-in-place solo mode

- Illuminates when engaged

SOLO CLEAR: Disengages all currently engaged SOLO switches

- Momentary switch
- Illuminates when any channel or master SOLO switch is engaged

11.0 Monitoring Systems

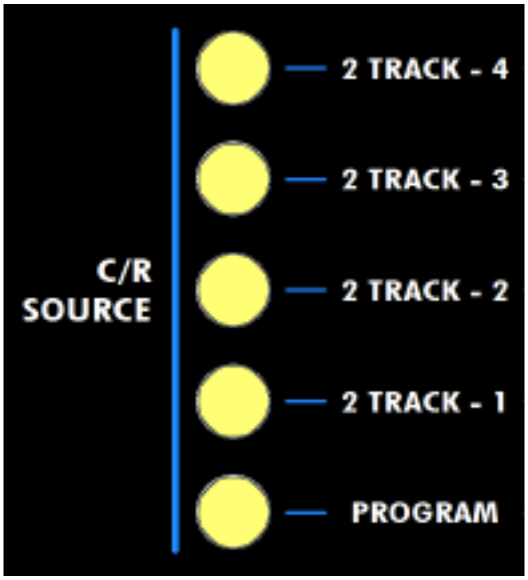
"The BOX" is equipped with a comprehensive stereo control room monitoring system that supports two (2) sets of loudspeakers, In addition, the console provides a fully routable cue and headphone system for studio and control room monitoring. Both systems employ many of the features found on larger consoles.

The program bus and up to four external stereo sources can routed to the loudspeaker, headphone, and cue outputs.

In total, the monitoring systems has the following components:
- Control room source selectors
- Control room monitor controls and outputs
- Headphone master and outputs

11.1 Control Room Source Selection

The output of the program master and up to four (4) external stereo sources can be routed to the control room monitors. The C/R SOURCE selections provide the feed for the active loudspeaker system and VU meters, as well as the default source for headphones.



Control room sources are selected using the C/R SOURCE selectors.

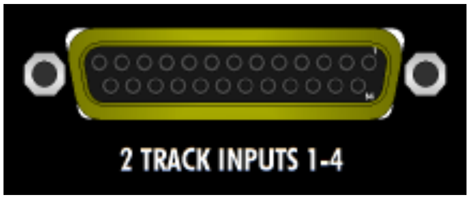
<u>C/R SOURCE (control room source)</u>: Routes the selected sources to the control room master
- <u>PROGRAM</u>: Program master output
- <u>2 TRACK 1-4</u>: External sources 1-4
- Illuminate when engaged

Nothing is routed to the control room loudspeakers if none of the C/R SOURCE switches are engaged.

Any time PROGRAM is engaged, the program master output is the only C/R SOURCE. 2TRACK sources are blocked even if their switches are engaged.

2 TRACK source selections 1-4 are additive, so more than one external source can be selected simultaneously. Each source is summed at unity gain when multiple sources are selected. Engaging PROGRAM will override any 2 TRACK source selections. The illuminated switches indicate the actual active source.

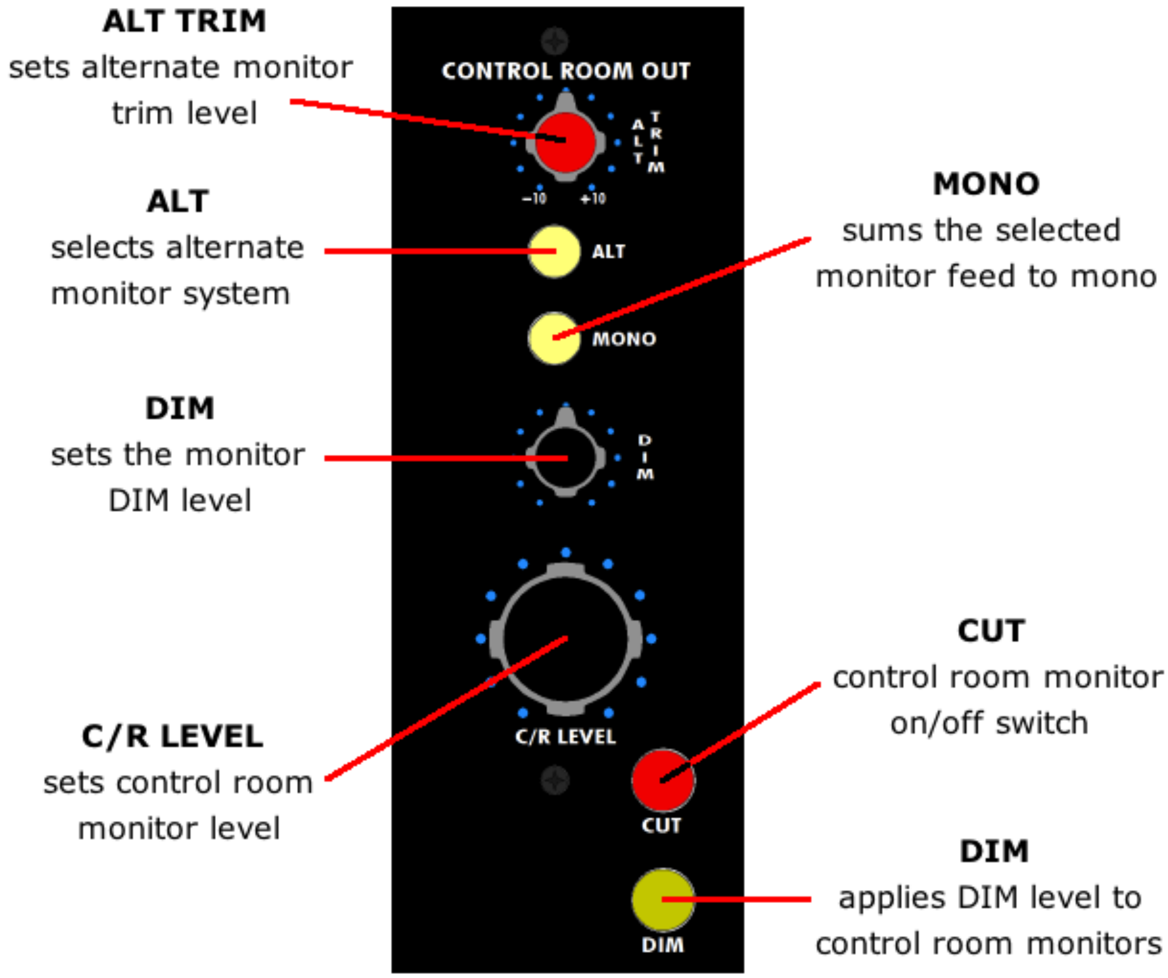
Rear Panel Control Room Source Input Connections



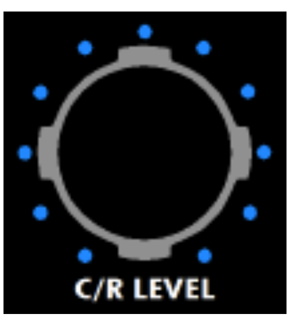
<u>2 TRACK INPUTS 1-4</u>: DB-25, balanced, low-impedance, line-level
- Standard pin-out: 1-2=2-TRACK 1, 3-4= 2-TRACK 2, 5-6=2-TRACK 3, 8-7=2-TRACK 4
- Routed to the C/R SOURCE and CUE SOURCE selectors

11.2 Control Room Monitor Controls

The CONTROL ROOM OUT master section provides a full set of controls for two (2) sets of loudspeakers.



Level and Cut



<u>C/R LEVEL (control room level)</u>: Sets the output level of the control room loudspeakers

WARNING: This is largest and most dangerous knob on the console…please be careful and use it wisely!

<u>CUT</u>: Mutes the selected control room loudspeakers when engaged

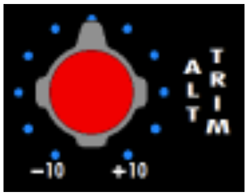
- Monitor output on/off switch
- Illuminates in red when engaged

Main and Alternate Control Room Monitors

The selected C/R SOURCE can be routed to one of two sets of stereo loudspeakers. Main and alternate monitors are selected using the ALT switch, with the main system selected by default. The output level for the alternate monitors can be trimmed to match the mains.

ALT (alternate): Toggles between the main and alternate control room loudspeakers outputs

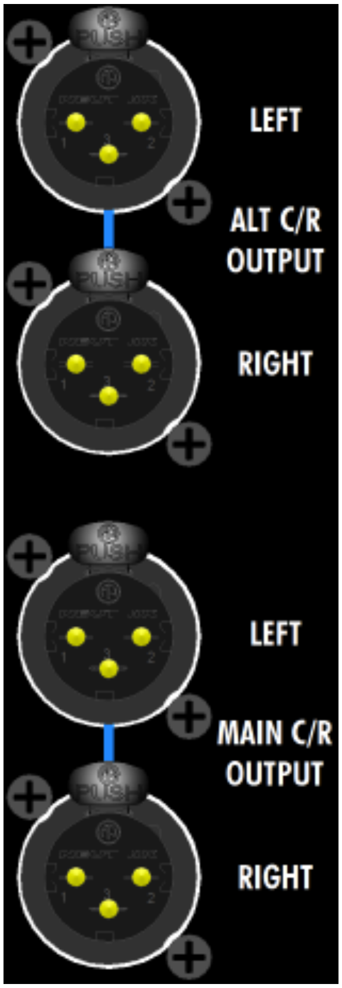
- Main outputs are cut when engaged
- Illuminates when engaged



ALT TRIM (alternate trim): Trims the alternate control room loudspeaker outputs

- -10dB to +10dB
- Center is 0dB (unity gain)

Control Room Monitor Output Rear Panel Connections



ALT C/R OUTPUT (alternate control room outputs): 3-pin male XLR, balanced, low-impedance, line-level

- LEFT & RIGHT stereo outputs
- Fed from the C/R LEVEL and ALT TRIM when ALT is engaged

MAIN C/R OUTPUT (main control room outputs): 3-pin male XLR, balanced, low-impedance, line-level

- LEFT & RIGHT stereo outputs
- Fed from C/R LEVEL when ALT is not engaged

Loudspeaker DIM Function

DIM: Activates the control room loudspeaker DIM function when engaged
- Monitor output is attenuated by the amount set with the DIM pot
- DIM activates when the TALK button is engaged
- Illuminates in orange when engaged

DIM: Sets the control room loudspeaker DIM level
- Controls the amount of attenuation applied to the loudspeaker outputs when DIM is engaged

Stereo and Mono Monitoring

The selected C/R SOURCE can be monitored in stereo or mono. Stereo is the default setting and mono can be selected by engaging the MONO switch.

MONO: Sums the stereo control room loudspeaker outputs to mono
- Left and Right C/R SOURCE selections are summed to mono when engaged
- Illuminates when engaged

11.3 Headphone Amplifier

In addition to the two control room monitor controls, "The Box" is equipped with separate headphone amplifier and associated controls. The system supports a single stereo headphone feed to two (2) headphone jacks, one on the rear panel and one under the armrest. The cue feed is typically routed to the headphones while recording and the control room source (program bus) is sent while mixing.

The headphone amp is fed from the C/R SOURCE selectors by default. This feed can be replaced by the output of the cue master by engaging the CUE TO PHONES switch. To support headphone feeds (cue feeds) while recording, talkback can be added to the cue master output before it is routed to headphones.

LEVEL: Stereo level control for the headphone amplifier

- Stereo level control with one ganged potentiometer
- Feeds the HEADPHONES jacks under the arm rest and on the rear panel

CUE TO PHONES: Routes the cue master outputs to the headphone amp inputs

- Replaces the C/R SOURCE selections as headphone amp inputs when engaged
- Illuminates when engaged

ON: Activates headphone amp outputs

- Illuminates when engaged

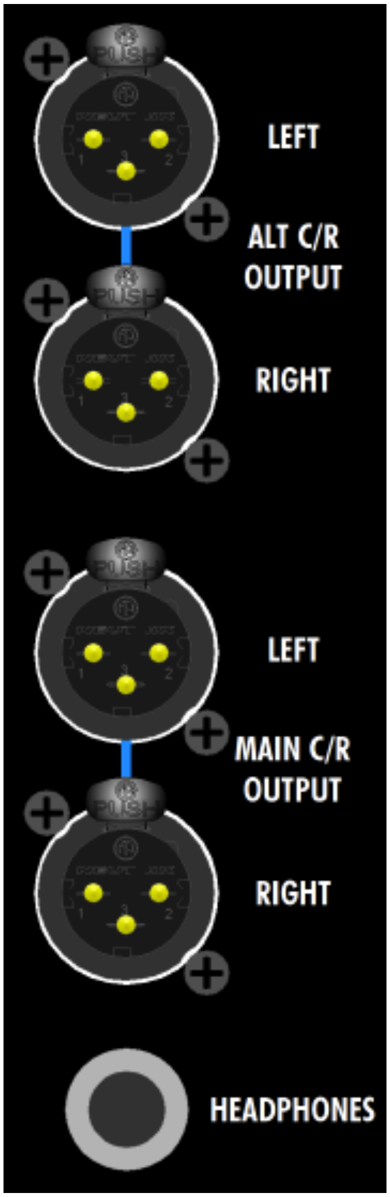
Headphone Jacks

HEADPHONES: ¼" tip-ring-sleeve headphone jacks

- Fed from headphone amp outputs when ON is engaged in the headphone master
- LEFT & RIGHT stereo output
- Headphone jack is located under armrest
- Second headphone jack on rear panel

11.4 Monitor System Rear Panel Connections

Loudspeaker and Headphone Outputs



ALT C/R OUTPUT (alternate control room outputs): 3-pin male XLR, balanced, low-impedance, line-level

- LEFT & RIGHT stereo outputs
- Fed from the C/R LEVEL and ALT TRIM when ALT is engaged

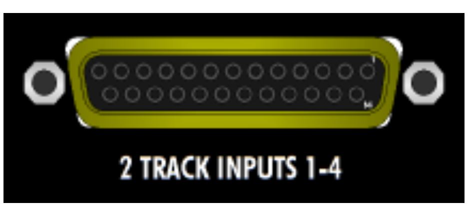
MAIN C/R OUTPUT (main control room outputs): 3-pin male XLR, balanced, low-impedance, line-level

- LEFT & RIGHT stereo outputs
- Fed from the C/R LEVEL when ALT is not engaged

HEADPHONES: ¼" tip-ring-sleeve headphone jack

- LEFT & RIGHT stereo output
- Fed from headphone amp outputs when ON is engaged
- Second headphone jack under armrest

External 2-Track Inputs

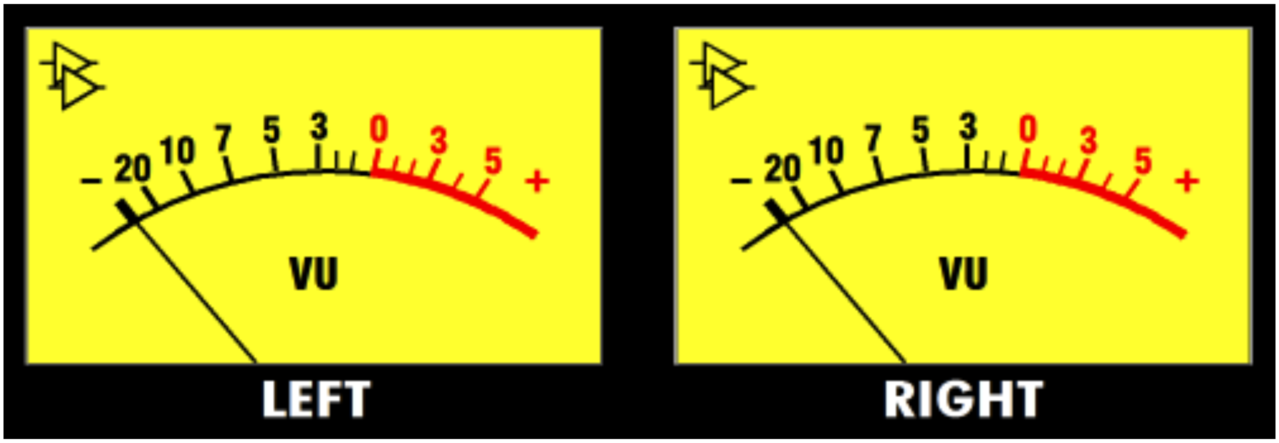


2 TRACK INPUTS 1-4: DB-25, balanced, low-impedance, line-level

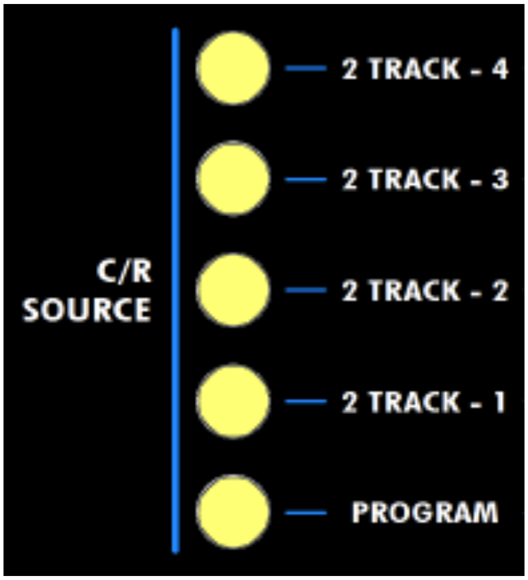
- Standard pin-out: 1-2=2-TRACK 1, 3-4= 2-TRACK 2, 5-6=2-TRACK 3, 8-7=2-TRACK 4
- Routed to the C/R SOURCE and CUE SOURCE selectors

12.0 VU Meters

"The BOX" come equipped with a pair of LEFT and RIGHT stereo VU meters mounted above the 527 compressors. These VU meters are back-lit and have an extended range to +5 to accommodate high output content. 0VU = +4dBu



The meters are fed from the C/R SOURCE selectors above the control room monitors controls.

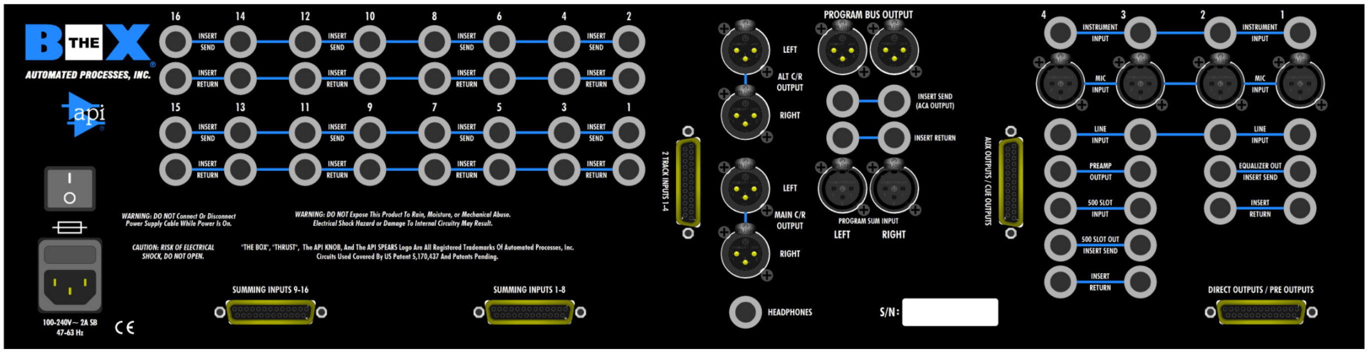


The stereo VU meters display the levels of the sources selected for control room monitoring (C/R SOURCE).

Refer to section 11.1 Control Room Source Selection for additional information regarding VU meter source selection.

13.0 Rear Panel Connections

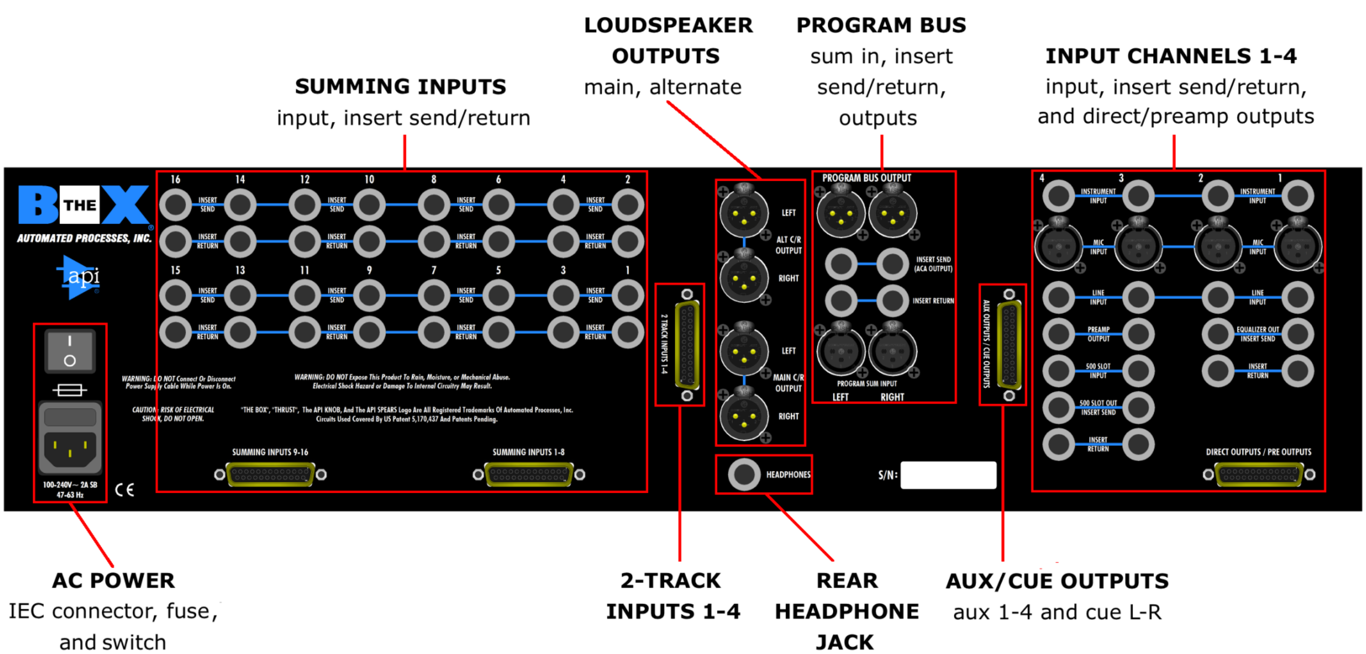
"The BOX" rear panel contains all the connections needed to interface the console with other studio equipment.



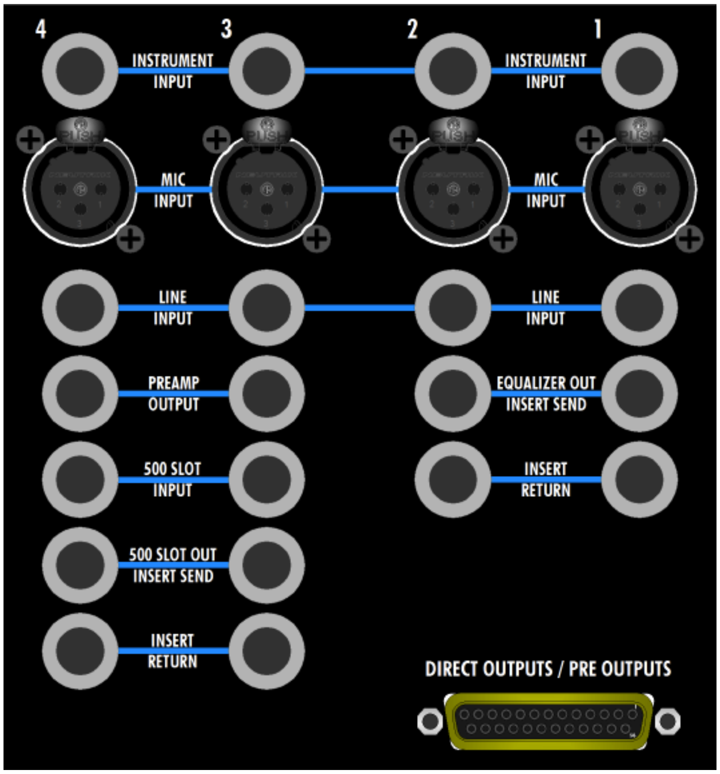
Connections are organized in several sections:

- Input Channels 1-2: Preamp inputs, insert sent/return, direct/pre out
- Input Channels 3-4: Preamp inputs, preamp output, 500 slot input, insert sent/return, direct/pre out
- Summing Inputs 1-16: Line inputs, insert send/return
- Program Bus: Sum inputs, insert send/return, outputs
- Aux/Cue: outputs
- Control Room/Cue Source 1-4: 2-tracks 1-4
- Control Room Loudspeakers: Main and alternate outputs
- Headphones: Headphone amp output
- AC Power: IEC connector, fuse, and power switch

Rear Panel Sections



13.1 Input Channels 1-4 Connections



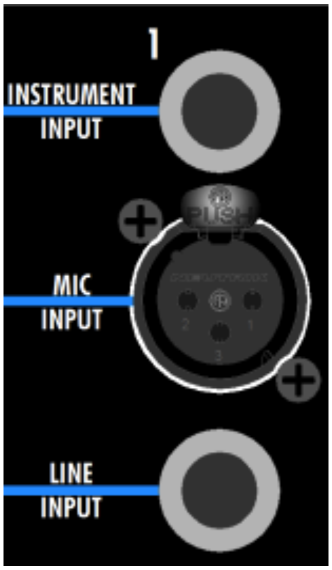
The rear panel provides a comprehensive and flexible set of input channel connections.

Connections include:

- Preamp inputs (mic, instrument, and line)
- Insert sends and returns
- Direct outputs / preamp outputs
- Preamp output / 500 slot input (channels 3 & 4 only)

Input Channel Preamp Connections

All four input channels have the same preamp input connections on the rear panel.



INSTRUMENT INPUT: ¼" tip-sleeve, unbalanced, high-impedance

- MIC must be engaged to use this input
- Replaces MIC signal as preamp input

MIC INPUT: 3-pin female XLR, balanced, low-impedance

- MIC must be engaged to use this input
- Provides phantom power when the 48v switch is engaged

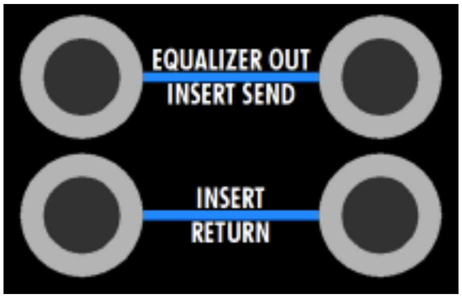
LINE INPUT: ¼" tip-ring-sleeve, balanced, low-impedance

- MIC must be disengaged to use this input

Input Channel Insert Connections

All insert sends and returns are balanced, low-impedance, line-level signals routed via ¼" tip-ring-sleeve jacks on the rear panel.

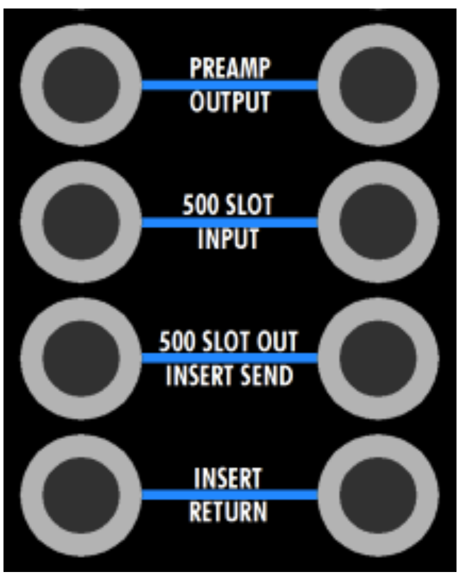
The insert send is post-EQ on channels 1 & 2.



EQUALIZER OUT- INSERT SEND: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

INSERT RETURN: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

The insert send is post-500 slot on channels 3 & 4. In addition to the insert send and return, the signal path on channels 3 & 4 can be accessed pre-500 slot via the PREAMP OUTPUT and 500 SLOT INPUT jacks on the rear panel.



PREAMP OUTPUT: ¼" tip-ring-sleeve, jack*, balanced, low-impedance, line-level

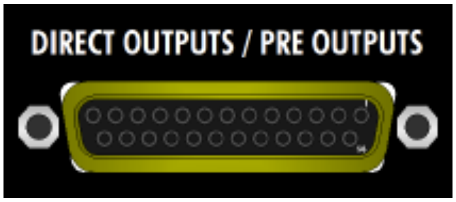
500 SLOT INPUT: ¼" tip-ring-sleeve, switching jack*, balanced, low-impedance, line-level

500 SLOT OUT-INSERT SEND: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

INSERT RETURN: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

* The PREAMP OUTPUT and 500 SLOT INPUT jacks work as a "half-normalled" pair and the channel signal flows through these jacks. If a plug is inserted into the 500 SLOT INPUT jack, that signal replaces the signal from the PREAMP OUTPUT jack. The signal from the onboard preamp is broken and does not continue in the channel flow path.

Input Channel Direct and Preamp Output Connections

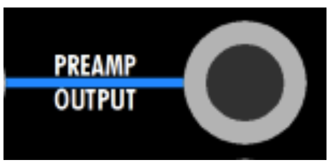


The DIRECT OUTPUTS / PRE OUTPUTS connector carries direct or preamp outputs from input channels 1-4, as well as the insert send from input channels 1 & 2 and the preamp outputs from input channels 3 & 4.

DIRECT OUTPUTS / PRE OUTPUTS: DB-25, balanced, low-impedance, line-level

- Outputs 1-4: Direct out (post-fader) or preamp output from input channels 1-4
  - Fed post-fader/mute by default (direct output)
  - Preamp output replaces post-fader output if DIR PRE is engaged
- Outputs 5 & 6: Insert send from input channels 1 & 2
- Outputs 7 & 8: Preamp outputs from input channels 3 & 4
- Standard pin-out: 1-4=direct or preamp outputs 1-4, 5-6=insert send 1-2, 7-8=preamp outputs 3-4

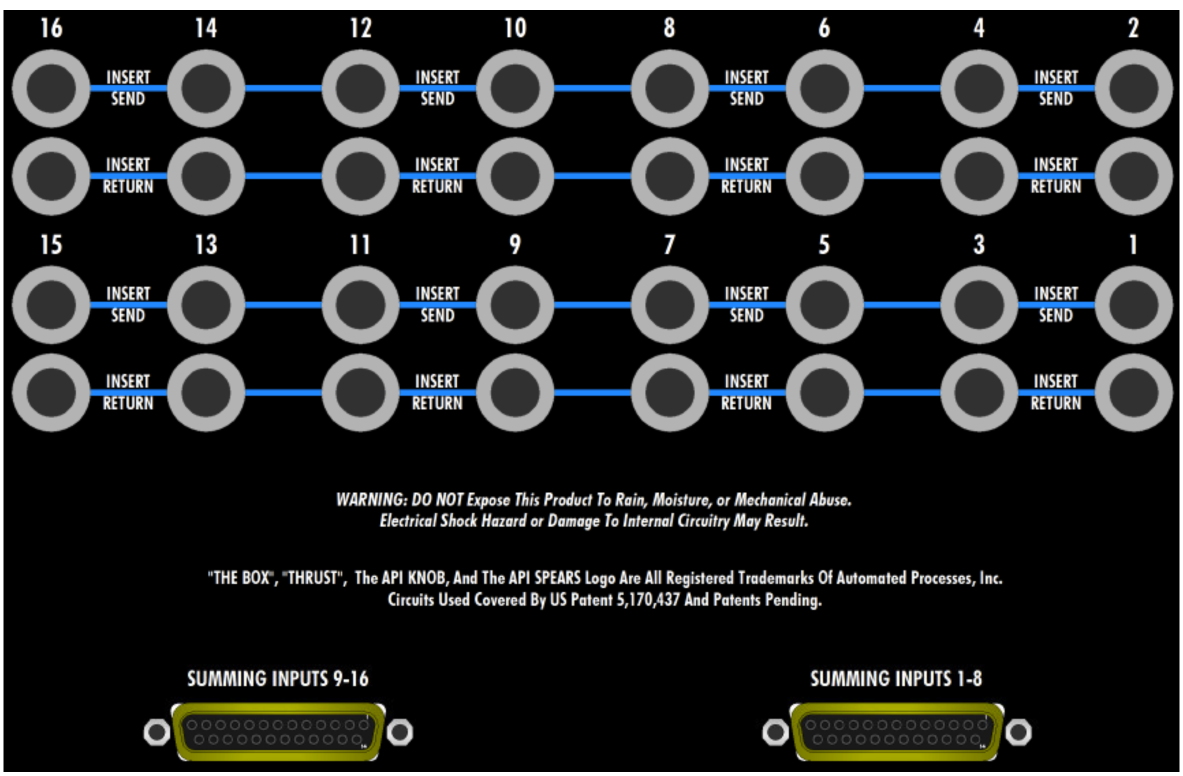
In addition to the DB-25 connector, input channels 3 & 4 have a separate preamp output that's part of an extra set of connections that support the 500 slots.



PREAMP OUTPUT: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

- ¼" preamp output is available on input channels 3 & 4

13.2 Summing Inputs 1-16 Connections

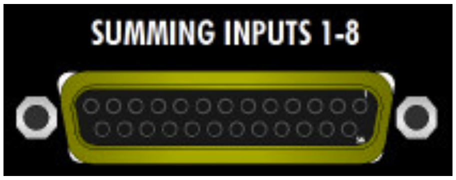


The rear panel provides a comprehensive and flexible set of summing input connections. Connections include:

- Line inputs
- Insert sends and returns

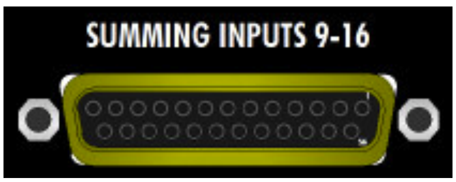
Summing Input Line Input Connections

The line inputs for the sixteen summing inputs are distributed on two DB-25 connectors.



SUMMING INPUTS 1-8: DB-25, balanced, low-impedance, line-level

- Standard pin-out: 1-8=summing inputs 1-8



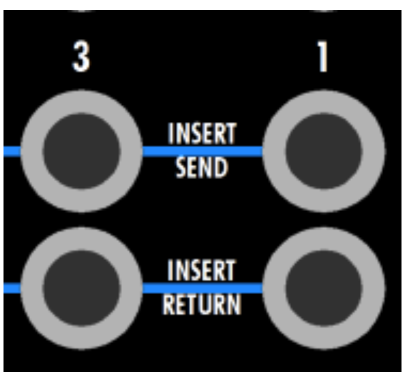
SUMMING INPUTS 9-16: DB-25, balanced, low-impedance, line-level

- Standard pin-out: 1-8=summing inputs 9-16

Summing Input Insert Connections

All insert sends and returns are balanced, low-impedance, line-level signals routed via ¼" tip-ring-sleeve jacks on the rear panel.

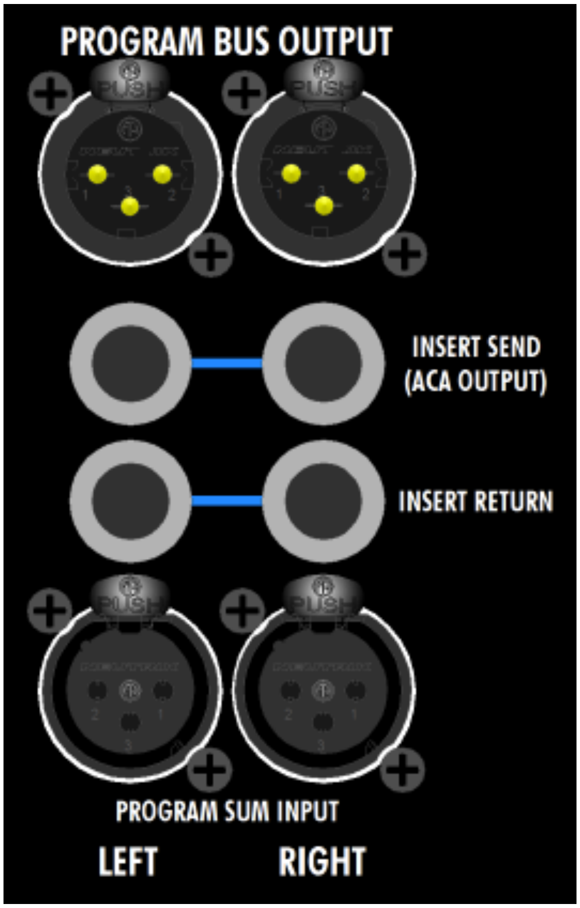
The insert send is fed immediately after the input on summing inputs 1-16.



INSERT SEND: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

INSERT RETURN: ¼" tip-ring-sleeve, balanced, low-impedance, line-level

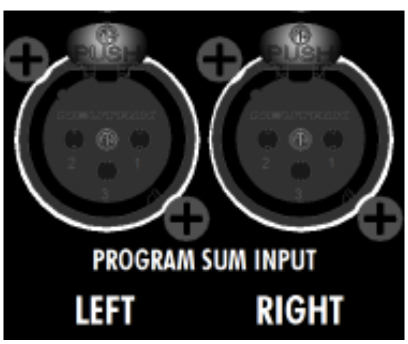
13.3 Program Bus Connections



The rear panel provides a comprehensive and flexible set of stereo program master connections. LEFT and RIGHT program bus connections include:

- Program bus outputs
- Insert sends
- Insert returns
- Program sum input

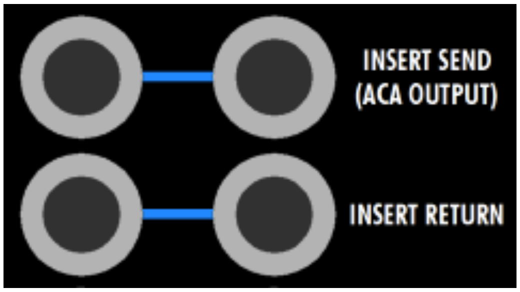
Program Bus Sum Input Connections



PROGRAM SUM INPUT (LEFT & RIGHT): 3-pin female XLR, balanced, low-impedance, line-level

- Located pre-insert

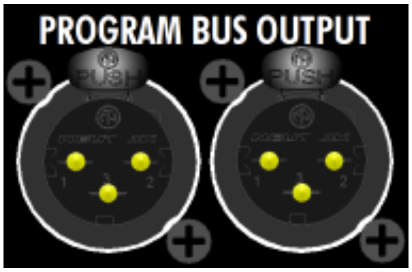
Program Bus Insert Connections



INSERT SEND (ACA OUTPUT) (LEFT & RIGHT): ¼" tip-ring-sleeve, balanced, low-impedance, line-level

INSERT RETURN (LEFT & RIGHT): ¼" tip-ring-sleeve, balanced, low-impedance, line-level

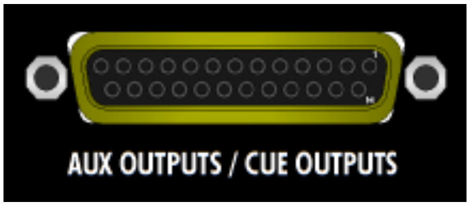
Program Bus Output Connections



PROGRAM BUS OUTPUT (LEFT & RIGHT): 3-pin male XLR, balanced, low-impedance, line-level

- Located post-master fader

13.4 Auxiliary & Cue Connections

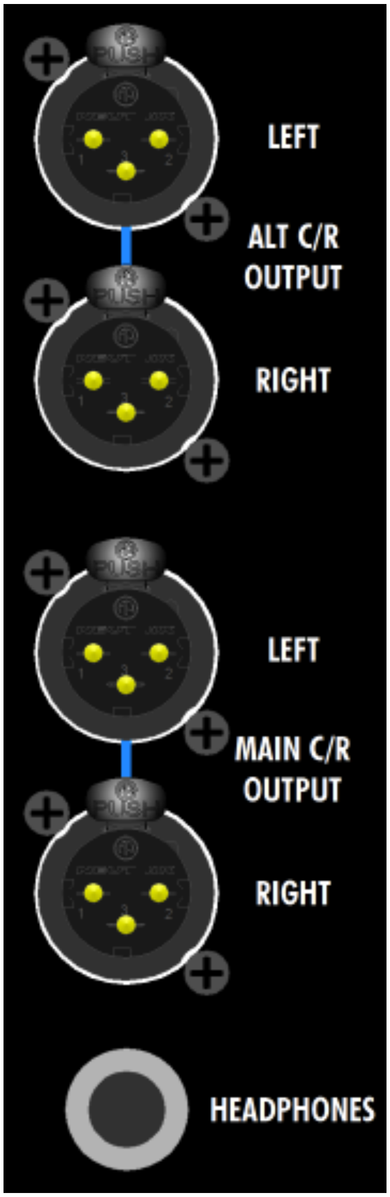


AUX OUTPUTS / CUE OUTPUTS: DB-25, balanced, low-impedance, line-level

- Fed from the aux and cue masters
- Standard pin-out: 1-4=aux outputs 1-4, 5-6=cue outputs L-R

13.5 Control Room Monitor/Headphone Output Connections

Loudspeaker and Headphone Output Connections

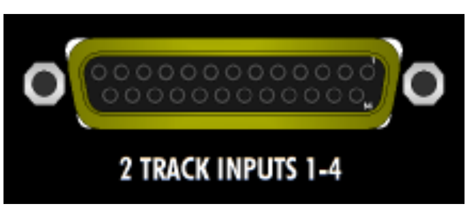


ALT C/R OUTPUT (alternate control room outputs): 3-pin male XLR, balanced, low-impedance, line-level
- LEFT & RIGHT stereo outputs
- Fed from the C/R LEVEL and ALT TRIM when ALT is engaged

MAIN C/R OUTPUT (main control room outputs): 3-pin male XLR, balanced, low-impedance, line-level
- LEFT & RIGHT stereo outputs
- Fed from the C/R LEVEL when ALT is not engaged

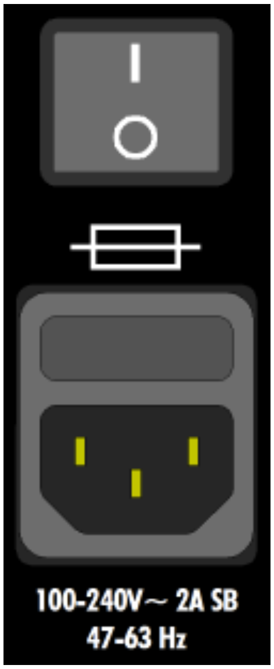
HEADPHONES: ¼" tip-ring-sleeve headphone jack
- LEFT & RIGHT stereo output
- Fed from headphone amp outputs when ON is engaged
- Second headphone jack under armrest

External 2-Track Input Connections



2 TRACK INPUTS 1-4: DB-25, balanced, low-impedance, line-level
- Standard pin-out: 1-2=2-TRACK 1, 3-4= 2-TRACK 2, 5-6=2-TRACK 3, 8-7=2-TRACK 4
- Routed to the C/R SOURCE and CUE SOURCE selectors

13.6 AC Power Connections



POWER SWITCH: On/off AC power switch

FUSE: 2 amp slow-blow

IEC Connector:

- Connection to AC power mains
- 100-240v
- 47-63Hz

14.0 Application Guide

"The BOX" is a highly versatile audio production console and is at home with a wide variety of applications including:

- Multitrack Recording
- Overdubbing
- Mixing
- Broadcasting
- Post Production
- Live Sound (with the addition of external preamps)

API realizes the owners of "The BOX" are professional users, so this section of the manual is designed to provide overviews of quick and easy setups for basic applications.

Note: These setups are intended to be a very basic "Getting Started" guide and should not be considered to be comprehensive.

Note: Any recording device with analog interfaces can be used with "The BOX." Since most users will record and mix with a (DAW) Digital Audio Workstation, a DAW will be used in the examples below.

14.1 Recording

This setup uses "The BOX" in a traditional recording fashion, with modern options. Input channels 1-4 are used as signal paths to a DAW for recording up to four mono or two stereo sources. Summing inputs are used to simultaneous mix up to sixteen individual multitrack returns from the DAW. The summing input returns can be routed to the program bus while mixing and recording, as well as cue/aux sends for headphone feeds and effects. Control room monitors are fed from the program master outputs and headphones are fed cue with talkback.

14.1.1 Input Channels While Recording

1) Connect the audio sources to be recorded to the input channel preamp inputs: MIC, INSTRUMENT, or LINE inputs on the rear panel.
2) Select the appropriate preamp input for the sources to be recorded (mic, instrument, or line):
    - MIC: Engage the MIC switch and 48V if needed
    - INSTRUMENT: Insert a ¼" plug from an instrument into the INSTRUMENT INPUT jack and engage the MIC switch to use the instrument input.
    - LINE: Disengage the MIC switch (LINE is selected by default).
3) Connect the input channel DIRECT OUTPUT/PRE OUTPUT to the inputs of the DAW.
    - DIRECT OUTPUT is fed post-fader including signal processing and insert.
    - Engage the DIR PRE switch to feed the DAW from the preamp output.
4) Employ the onboard signal processing as needed:
    - High-Pass Filter: Refer to section 2.2.1 High-Pass Filter.
    - 527 COMPRESSOR: Refer to section 2.2.2 Compressor Routing.
    - INSERT: External processors. Refer to section 2.2.3 Insert.
    - 550A Equalizer: Channels 1 & 2 only. Refer to section 2.4.1 550A EQ.
    - API 500 Series Slot: Channels 3 & 4 only. For processing with installed 500 modules, refer to section 2.4.2 500 Slot.

- <u>API 500 Series Slot Patches</u>: Channels 3 & 4 only. In additional to the insert, external signal processor may be connected via the PREAMP OUTPUT and 500 SLOT INPUT connections on the rear panel. Refer to section 2.4.2 500 Slot.

5) Set the input channel fader to 0dB (unity gain)
6) Use the preamp to set the channel level and polarity
    - Make fader adjustments as needed*
7) The DAW feed will be controlled by the channel fader and mute

* The preamp is built with a transformer output. When set to high levels, the preamp can drive the transformer to "magnetic saturation" before unpleasant distortion is heard. When done carefully, the top of the input/output transfer curve starts to flatten-out, giving a slightly compressed sound. Since this characteristic is magnetically based, the sound can be loosely associated with the "punch" of magnetic tape. However, setting the preamp at such a high level will likely be too high for the DAW analog-to-digital converters. To achieve the saturated sound without A-to-D overload, use the channel fader to attenuate the channel output to an appropriate level.

CAUTION: This effect is subtle and it may take some time to learn to recognize the sound and set it up properly. Care should be taken not to induce unwanted distortion in the preamp or DAW converters. If you're not familiar with this technique, experiment with a percussive source and listen to what happens to the attack as you drive the transformer harder. Also listen for undesirable distortion (don't go too far!). The waveform display in the DAW will visually display the effect and reinforce what you're hearing.

14.1.2 Summing Inputs While Recording

1) Use summing inputs 1-16 and unused input channels to route DAW multitrack returns to the program bus for control room monitoring and the cue/aux buses for headphone feeds and effects sends.
2) Connect the outputs of the DAW to the SUMMING INPUTS 1-8 and SUMMING INPUTS 9-16 connectors, as well as to the LINE INPUTS on input channels not used for recording.
3) Route the previously recorded tracks, as well as the new tracks to be recorded to the DAW outputs according to their console connections.
    - If the session has a large number of tracks, the DAW tracks can be sub-mixed to stereo or larger stems before assignment to DAW outputs.
4) Set DAW faders to 0dB (unity gain) and turn off DAW fader automation
5) With summing faders down, engage the PGM switch on the summing inputs and input channels carrying DAW returns to assign them to the program bus.
6) Set the program master fader to 0dB (unity gain).
7) Select the PROGRAM as the C/R SOURCE and set the loudspeakers to a modest level (no more than 85dB please!).
8) Use faders and pans to carefully push up a balanced stereo mix.
9) Use cue sends to create a mix for headphones or send the program bus to cues.
10) Use the aux sends for effects sends.

If a desirable automated mix has already been created in the DAW, use the following procedure to return the automated tracks to the program bus:

1) Recreate the panning used in the DAW mix on the corresponding summing inputs.
2) Route the automated tracks in the DAW to the corresponding summing inputs.
3) Engage the 0dB switches. This will bypass the console faders.
4) Engage the PGM switch to route the DAW tracks to the program bus at unity gain. The automated mix balance will be retained (assuming proper DAW calibration and matching pan-laws).
5) Use cue sends to create a mix for headphones or send the program bus to cues.
6) Use the aux sends for effects sends.

14.1.3 Using DAW Mixes While Recording

Often during production, a stereo mix is created in the DAW that's satisfactory for use while overdubbing and recording additional tracks. Use of this mix can be an important time saver, but can have limitations while recording. If a desirable mix has already been created in the DAW, the stereo output of the DAW mix can be routed to a pair of summing inputs or a 2-TRACK input for monitoring while recording.

DAW Mix to Summing Inputs

Use the following procedure to use a pair of summing inputs to return a complete stereo mix from the DAW while recording:

1) Connect a pair of DAW outputs to one of the SUMMING INPUT connectors on the rear panel (typically channels 1 and 2).
2) Route the DAW mix to the DAW outputs connected to the summing inputs.
3) Engage the 0dB switches on these channels. This will bypass the console faders.
4) Pan the summing inputs hard LEFT and RIGHT respectfully.
5) Engage the PGM switch to route the DAW tracks to the program bus at unity gain. The DAW mix will be send to the program bus. Any changes to the mix must take place in the DAW.
6) Use the summing input cue sends to create headphone feeds or route the program master to directly to headphones (see below).
7) Any changes to the mix must take place in the DAW.

DAW Mix as a C/R and CUE SOURCE

Use the following procedure to return a complete stereo DAW mix as a C/R and CUE SOURCE while recording:

1) Connect a pair of DAW outputs to the 2-TRACK INPUT connector on the rear panel
2) Select the corresponding 2-TRACK selector to route it as the C/R SOURCE for the control room loudspeakers.
3) Select the corresponding 2-TRACK select to route it as the CUE SOURCE for a headphone feed.
4) The selected source(s) will be routed to the cue master via the CUE SOURCE level control.
5) Cue output will be fed from the CUE MASTER level.
6) With the headphone LEVEL turned down, engage the CUE TO PHONES switch on the headphone master.
7) Adjust the headphone master LEVEL to a comfortable volume.
8) Adjust the CUE SOURCE and CUE MASTER levels as needed.
9) Any changes to the mix must take place in the DAW.

14.1.4 Cue and Headphones

The cue system is designed to support headphone feeds while recording. The stereo cue master can be fed from the cue sends on all channels as well as the output of the CUE SOURCE selectors.

Headphone Feeds From Program Bus/External Source

1) Route the program master outputs and/or up to 4 external 2-track sources to the cue master
2) Using the C/R SOURCE selectors, engage the PROGRAM switch.
3) Using the CUE SOURCE selectors, engage the PROGRAM switch and any 2-TRACK sources needed in the headphones.
4) With the headphone LEVEL turned down, engage the CUE TO PHONES switch on the headphone master.
5) Turn up the CUE SOURCE control to a nominal level.
6) Turn up the CUE MASTER control to a nominal level.
7) Connect a pair of headphones to one of the console HEADPHONE jacks.
8) Engage the ON switch on the headphones master.
9) Adjust the headphone master LEVEL for a comfortable volume
10) Adjust the CUE SOURCE and CUE MASTER levels as needed.
11) Engage the CUE talkback assignment switch and adjust the GAIN as needed.

Headphone Feeds From Cue Sends

1) Use the cue system to create a primary headphone feed.
2) With the 3/4 send LEVEL turned down, engage the CUE, PRE, and 3/4 on/off switches on summing inputs and input channels carrying DAW returns.
    - CUE assigns the 3/4 send to the stereo cue bus
    - PRE is selected for cue sends so the bus is fed pre-fader. This allows the program bus mix to be changed without affecting the headphone feed.
3) With the headphone LEVEL turned down, engage the CUE TO PHONES switch on the headphone master.
4) Turn up the CUE MASTER level control to a nominal level.
5) Connect a pair of headphones to one of the console HEADPHONE jacks.
6) Engage the ON switch on the headphones master.
7) Adjust the LEVEL to a modest volume on the headphone master.
8) Use the 3/4 LEVEL and PAN controls on the summing inputs to carefully create an independent headphone mix.
9) Adjust the headphone master LEVEL to a comfortable volume.
10) Adjust the CUE MASTER level as needed.
11) Engage the CUE talkback assignment switch and adjust the GAIN as needed.

NOTE: A second independent headphone feed can be created using the auxiliary sends. Since channels can be routed only to the cue send or aux send 3/4, use the mono aux sends to create the second headphone feed if the same channel needs to be send to both headphone feeds.

"More Me" Headphone Feeds From Cue Sends and Program Bus

When recording a simple overdub, the program bus is all that is needed in the headphone feed. However it is sometimes desirable to add more of the tracks being recorded to the headphone feed. This can be done using the cue sends on those summing inputs. This is very useful when the talent asks for "More me!"

1) Set up the headphones to be fed from the program bus as describe previously.
2) With the 3/4 send LEVEL turned down, engage the CUE, PRE, and 3/4 on/off switches on the summing inputs carrying DAW returns of the newly recorded tracks.
    - CUE assigns the 3/4 send to the stereo cue bus
    - PRE is selected for cue sends so the bus is fed pre-fader. This allows the program bus mix to be changed without affecting the headphone feed.
3) Select PROGRAM as the C/R SOURCE and CUE SOURCE.
4) With the headphone LEVEL turned down, engage the CUE TO PHONES switch on the headphone master.
5) Turn up the CUE SOURCE control to a nominal level.
6) Turn up the CUE MASTER control to a nominal level.
   CUE SOURCE will control the level of the program bus and CUE MASTER will control the overall cue output level, including the cue bus.
7) Adjust the LEVEL to a modest volume on the headphone master.
8) Use the 3/4 LEVEL and PAN controls on the summing inputs to carefully add more of the new tracks to the headphone mix.
9) Adjust the headphone master LEVEL to a comfortable volume.
10) Adjust the CUE SOURCE and CUE MASTER levels as needed.
11) Engage the CUE talkback assignment switch and adjust the GAIN as needed.

NOTE: In addition to routing to headphones, the output of the cue master appears on the rear panel as well. This can be connected to external headphone amps for further distribution.

Headphone Feeds From DAW Via Cue Source

Many times a headphone mix will be created using aux sends/tracks in the DAW and it is desirable to use this mix while recording. This can be useful when working on a project over time or in multiple locations. The cue mix stays with the DAW session, but can be modified during production to suit the needs of each performer. However, most DAWs do not have talkback capabilities, so the DAW headphone mix cannot be routed directly to headphones. Use the following procedure to return a DAW headphone mix to a 2-TRACK INPUT and add talkback.

1) Connect a pair of DAW outputs to the 2-TRACK INPUT connector on the rear panel
2) Route the headphone mix in the DAW to the corresponding DAW outputs.
3) Select the connected 2-TRACK as the CUE SOURCE.
4) With the headphone LEVEL turned down, engage the CUE TO PHONES switch on the headphone master.
5) Turn up the CUE SOURCE level control to a nominal level.
6) Turn up the CUE MASTER control to a nominal level.
   CUE SOURCE will control the level of the DAW cue mix and CUE MASTER will control the overall cue output level, including the cue bus.
7) Connect a pair of headphones to one of the console HEADPHONE jacks.
8) Engage the ON switch on the headphones master.
9) Adjust the headphone master LEVEL to a comfortable volume.
10) Adjust the CUE SOURCE and CUE MASTER levels as needed.

11) Engage the CUE talkback assignment switch and adjust the GAIN as needed.

Headphone Feeds From DAW Via Summing Inputs or Input Channels
Use the following procedure to return a DAW headphone mix to a pair of channels and add talkback.

1) Connect a pair of DAW outputs to the LINE INPUT connectors for a pair of channels
2) Route the headphone mix in the DAW to the corresponding DAW outputs.
3) With the 3/4 send LEVEL turned down, engage the CUE, PRE, and 3/4 on/off switches on the channels carrying the DAW headphone mix.
    - CUE assigns the 3/4 send to the stereo cue bus
    - PRE is selected for cue sends so the bus is fed pre-fader. This allows the program bus mix to be changed without affecting the headphone feed.
4) Using the cue/aux PAN controls pan the headphone mix returns hard LEFT and RIGHT.
5) Using the cue/aux LEVEL control, turn up the send to a nominal level.
6) With the headphone LEVEL turned down, engage the CUE TO PHONES switch on the headphone master.
7) Turn up the CUE MASTER level control to a nominal level.
8) Connect a pair of headphones to one of the console HEADPHONE jacks.
9) Engage the ON switch on the headphones master.
10) Adjust the headphone master LEVEL to a comfortable volume.
11) Adjust the CUE SOURCE and CUE MASTER levels as needed.
12) Engage the CUE talkback assignment switch and adjust the GAIN as needed.

14.1.5 Talkback

Talkback can be routed the to cue master, aux masters, and program bus to facilitate communication while recording. When routed to CUE, talkback can be added to headphones. Talkback can be routed to the aux masters when aux sends are used for a second headphone feed. When routed to the program bus, talkback can be used to "slate" multiple takes while recording.

To route talkback while recording, use the following procedure:

1) Assign talkback to the desired destination(s). Three assignments can be made:
    - <u>CUE</u>: Routes talkback to the cue master
    - <u>AUX</u>: Routes talkback to the auxiliary masters
    - <u>SLATE</u>: Routes talkback to the program bus
2) Turn up the talkback GAIN to a nominal level.
3) Press the TALK button to send the mic to the selected destination(s).
4) Adjust the talkback GAIN as needed.

14.2 Mixing

This setup uses summing inputs 1-16 and input channels 1-4 to return up to twenty (20) individual multitrack returns from the DAW, effects processors, and other sources. These returns can be routed to the program bus, as well as cue/aux sends for effects. Control room monitors and headphones are fed the program bus outputs.

1) Use summing inputs 1-16 and input channels 1-4 to route DAW multitrack returns to the program bus for control room monitoring and the cue/aux buses for headphone feeds and effects sends.
2) Connect the outputs of the DAW to the SUMMING INPUTS 1-8 and SUMMING INPUTS 9-16 connectors, as well as to the LINE INPUTS on input channels 1-4.
3) Route the previously recorded tracks to the DAW outputs according to their console connections.
    - If the session has a large number of tracks, the DAW tracks can be sub-mixed to stereo or larger stems before assignment to DAW outputs.
    - While most signal processing will be done in the DAW, critical tracks can be routed to input channels if additional processing is needed.
4) Set DAW faders to 0dB (unity gain) and turn off DAW fader automation
5) With channel faders down, engage the PGM switch.
6) Set the program master fader to 0dB (unity gain).
7) Select the PROGRAM as the C/R SOURCE and set the loudspeakers to a modest level (no more than 85dB please!).
8) Use faders and pans to carefully push up a balanced stereo mix.
9) Employ the onboard signal processing as needed:
    - High-Pass Filter: Refer to section 2.2.1 High-Pass Filter.
    - 527 COMPRESSOR: Refer to section 2.2.2 Compressor Routing.
    - INSERT: External processors. Refer to section 2.2.3 and 3.1.1 Insert.
    - 550A Equalizer: Channels 1 & 2 only. Refer to section 2.4.1 550A EQ.
    - API 500 Series Slot: Channels 3 & 4 only. Processing with installed 500 modules. Refer to section 2.4.2 500 Slot.
    - API 500 Series Slot Patches: Channels 3 & 4 only. An additional external signal processor may be connected via the PREAMP OUTPUT and 500 SLOT INPUT connections on the rear panel. Refer to section 2.4.2 500 Slot.
10) Use the aux and cue sends for effects sends.

If a desirable automated mix has already been created in the DAW, use the following procedure to return the automated tracks to the program bus:

1) Recreate the panning used in the DAW mix on the corresponding summing inputs.
2) Route the automated tracks in the DAW to the corresponding summing inputs.
3) Engage the 0dB switches. This will bypass the console faders.
4) Engage the PGM switch to route the DAW tracks to the program bus at unity gain. The automated mix balance will be retained (assuming proper DAW calibration and matching pan-laws).
5) Use cue sends to create a mix for headphones or send the program bus to cues.
6) Use the aux sends for effects sends.

14.2.1 Program Bus Compression

Use the following procedure to use stereo program bus compression:

1) Disengage the PGM-CHAN switches on compressors 1 and 2.
2) Engage the LINK switch on the compressor. This will sum the DC gain reduction control voltage for stereo processing.
3) Adjust for the desired compression and be sure to set the controls on both compressors exactly the same. This assures that both sides of the stereo program trigger the gain reduction based on the same values.
4) Refer to section 5.2 Program Bus Compression.

14.2.2 Effects Sends and Returns

Sends and returns for outboard effects processors can easily be used while mixing using auxiliary/cue sends and masters. Use the following procedure to create effects sends:

1) Connect AUX/CUE OUTPUT connectors to the outboard effects processor inputs.
2) Connect the outboard effects processor outputs to summing input or input channel LINE INPUTS.
3) With the aux send LEVELs turned down, engage the 1/2 and 3/4 on/off switches.
    - Sends will be post-fader so the send output changes with the program bus contribution. This will retain the perceived "wet-dry" ratio as the channel fader is moved.
4) The channel cue sends can be used as an alternate stereo effect send when the CUE switch is engaged. When CUE is engaged, the 3/4 send is routed to the cue bus instead of aux bus 3/4.
5) Turn up the AUX MASTER level controls to nominal levels.
6) Turn up the CUE MASTER level control to a nominal level.
7) Engage PGM on channels carrying the returns from the outboard effects processors.
8) PAN the effects return channels as needed.
9) Set the faders on the effects return channels to a nominal level.
10) Use the channel aux and cue sends to create the needed effects processor feeds.

NOTE: To minimize noise when mixing with external effects processions, it is best to maximize the signal-to-noise ratio in the processor and return its output to the program bus at the lowest level possible. In other words, set the processor input/output to unity gain, use the sends to drive the processor inputs as high as possible without distortion, and set the return faders for the needed mix contribution. This requires careful gain setting from the send level, master level, device input/output levels, and return channel fader levels, but will yield better processor performance and lower noise, especially when multiple processors are contributing to the mix.

APPENDIX

A.1 API Limited Warranty and Service Information

a. **Warranty Information:** This product carries a one year labor and a five year parts warranty from date of purchase. API (Automated Processes, Incorporated) does not cover claims for damage due to alteration and/or abuse. This warranty is limited to failures during normal use, which are due to defects in material or workmanship. If any defects are found in the materials or workmanship, or if the product fails to function properly during the applicable warranty period, API, at its option, will repair or replace the product.

b. **PLEASE NOTE**: The design or quality of any non-authorized third party service or vendor is beyond the control of API. Therefore, use of **NON-API VPR Alliance modules in any API product – including consoles** may **VOID this warranty.** Also, service or modification of any API unit except by an authorized API representative may **VOID this warranty.**

c. API reserves the right to inspect any products that may be the subject of any warranty claims before repair or replacement is carried out. Final determination of warranty coverage lies solely with API.

d. This warranty is extended to the original purchaser and to anyone who may subsequently purchase this product within the applicable warranty period. Proof of purchase may be required.

e. **For questions regarding operation, interfacing or service of your API product, <u>please contact your API dealer from whom you purchased the unit.</u> Many times your authorized API dealer is the fastest and most cost-effective way to maintain and service your product.**

f. You may also contact API's Service Department directly.

a. Call API at 301-776-7879 (ext. 252) between 8:30 AM and 5:00 PM Monday through Friday (Eastern Time) to get a Return Authorization (RA). Products returned without an RA number may not be accepted.

b. Pack the defective part by wrapping in plastic and cushioning material. Seal securely in an approved shipping container. If you do not have a sufficient shipping container, ask API for advice when calling for the RA number.

c. Include a note explaining the problem and conditions of the service request. Include your complete return address (no P.O. Boxes, please).

d. Ship the product freight prepaid to:

API
8301 Patuxent Range Road
Jessup, MD 20794

IMPORTANT: Be sure the RA number is plainly written on the shipping carton.

g. This is your sole warranty. API does not authorize any third party, including any dealer or sales representative, to assume liability on behalf of API or to make any warranty for API.

h. THE WARRANTY GIVEN ON THIS PAGE IS THE SOLE WARRANTY GIVEN BY API AND IS IN LIEU OF ALL OTHER WARRANTIES, EXPRESS AND IMPLIED, INCLUDING THE WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE. THE WARRANTY GIVEN ON THIS PAGE SHALL BE STRICTLY LIMITED IN DURATION TO FIVE (5) YEARS FROM THE DATE OF THE ORIGINAL PURCHASE FROM API OR AN AUTHORIZED API DEALER. UPON EXPIRATION OF THE APPLICABLE WARRANTY PERIOD API SHALL HAVE NO FURTHER WARRANTY OBLIGATION OF ANY KIND. API SHALL NOT BE LIABLE FOR ANY INCIDENTAL, SPECIAL, OR CONSEQUENTIAL DAMAGES THAT MAY RESULT FROM ANY DEFECT IN THE API PRODUCT OR ANY WARRANTY CLAIM.

i. This warranty provides specific legal rights and you may have other rights, which vary from state to state.

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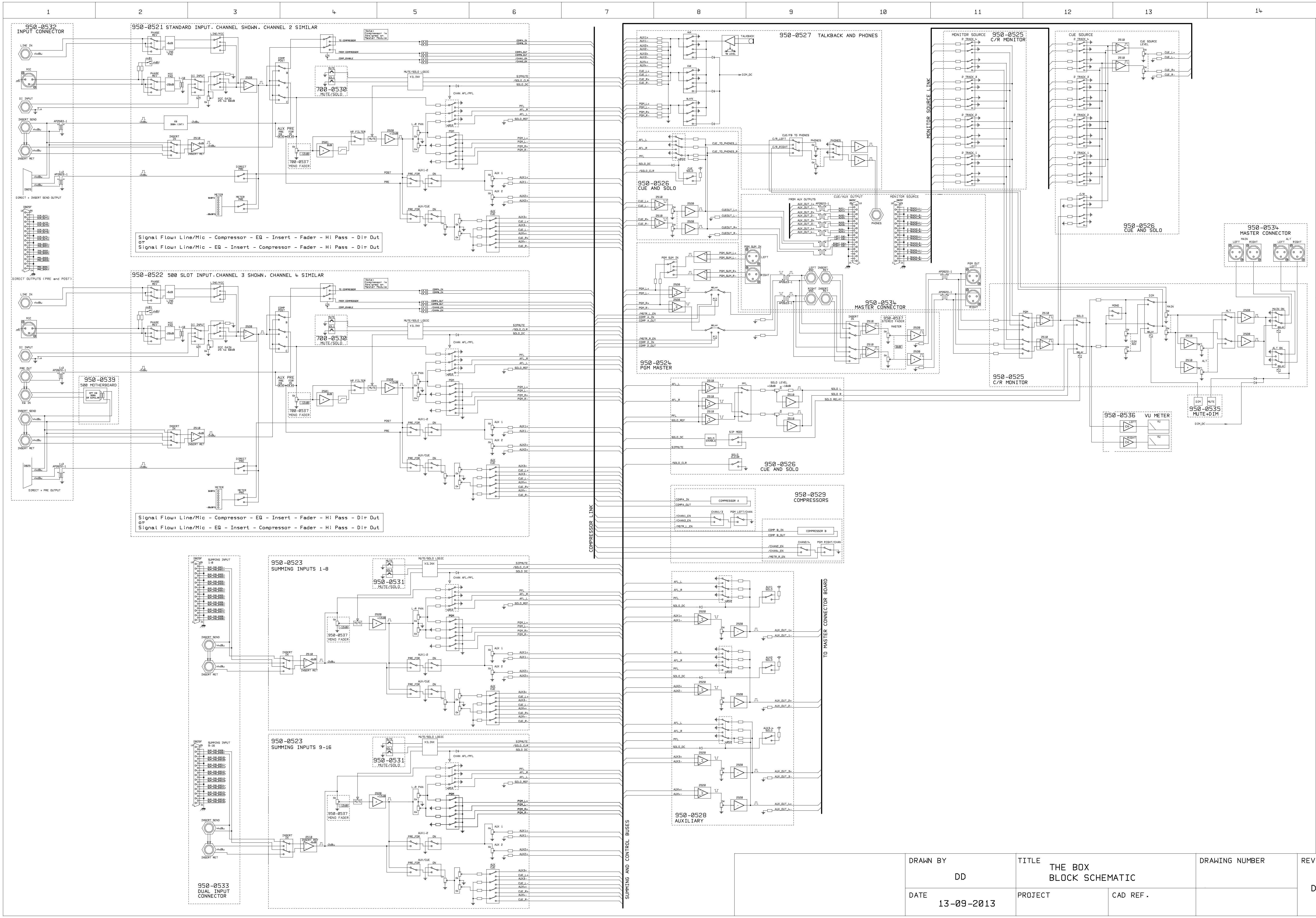
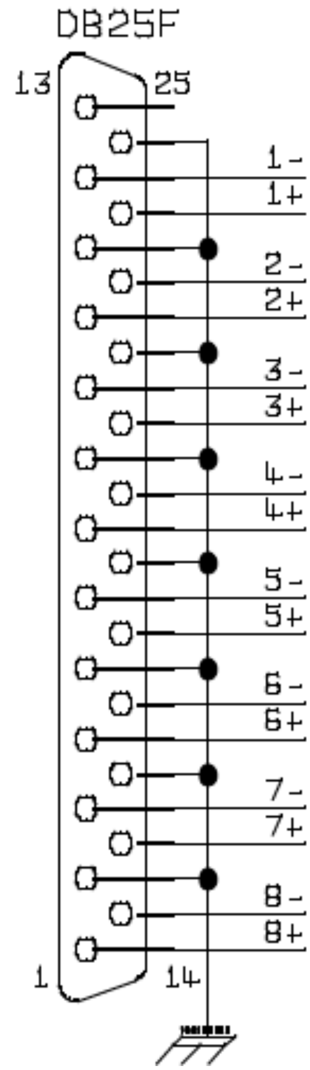


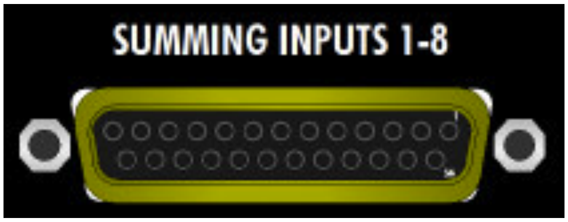
A.3 DB-25 Pin-Out Diagrams

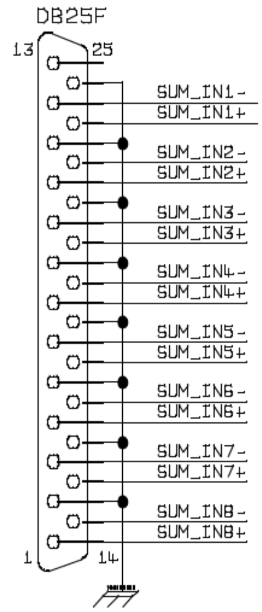
The DB-25 connectors use a typical 8-channel pin-out standard.

Channel	Signal	Pin
Channel 1:	High:	24
	Low:	12
	Ground:	25
Channel 2:	High:	10
	Low:	23
	Ground:	11
Channel 3:	High:	21
	Low:	9
	Ground:	22
Channel 4:	High:	7
	Low:	20
	Ground:	8
Channel 5:	High:	18
	Low:	6
	Ground:	19
Channel 6:	High:	4
	Low:	17
	Ground:	5
Channel 7:	High:	15
	Low:	3
	Ground:	16
Channel 8:	High:	1
	Low:	14
	Ground:	2



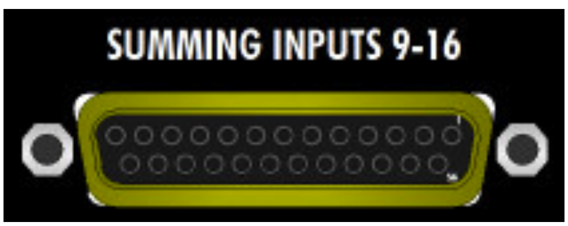
Summing Input Line Input Connections

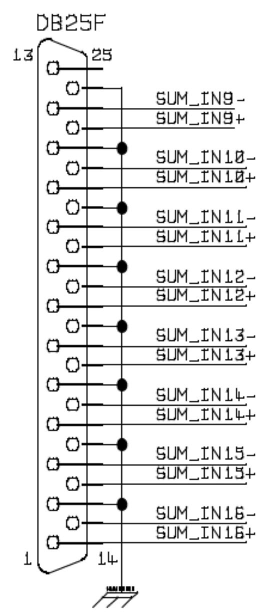




<u>SUMMING INPUTS 1-8</u>: Standard = API

- 1 = Summing Line Input 1
- 2 = Summing Line Input 2
- 3 = Summing Line Input 3
- 4 = Summing Line Input 4
- 5 = Summing Line Input 5
- 6 = Summing Line Input 6
- 7 = Summing Line Input 7
- 8 = Summing Line Input 8

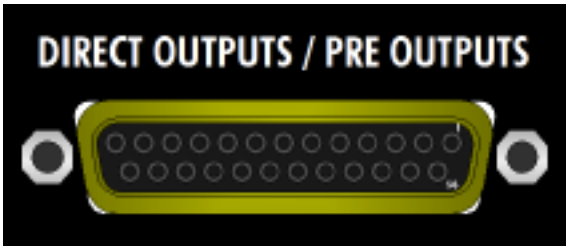


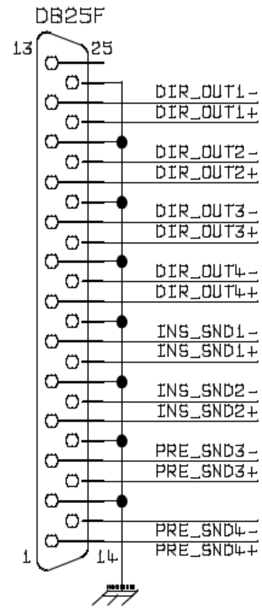


<u>SUMMING INPUTS 9-16</u>: Standard = API

- 1 = Summing Line Input 9
- 2 = Summing Line Input 10
- 3 = Summing Line Input 11
- 4 = Summing Line Input 12
- 5 = Summing Line Input 13
- 6 = Summing Line Input 14
- 7 = Summing Line Input 15
- 8 = Summing Line Input 16

Direct/Preamp Output Connections

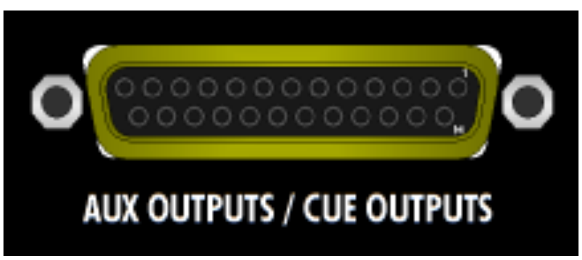


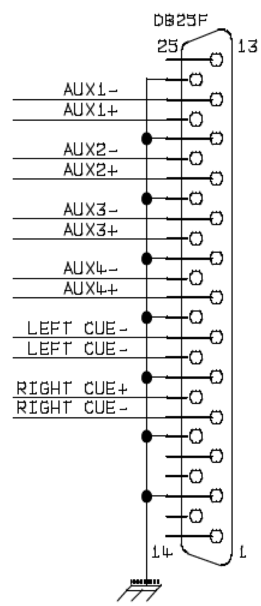


<u>DIRECT OUTPUTS/PRE OUTPUTS</u>: Standard = API

- 1 = Direct Output / Pre Output 1
- 2 = Direct Output / Pre Output 2
- 3 = Direct Output / Pre Output 3
- 4 = Direct Output / Pre Output 4
- 5 = Insert Send 1
- 6 = Insert Send 2
- 7 = Preamp Output 3
- 8 = Preamp Output 4

Auxiliary/Cue Output Connections

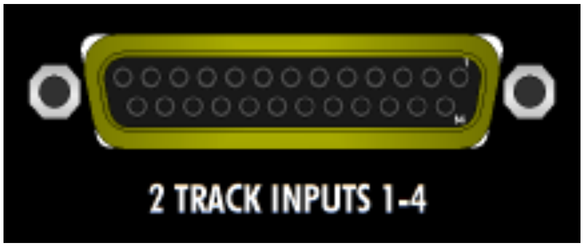


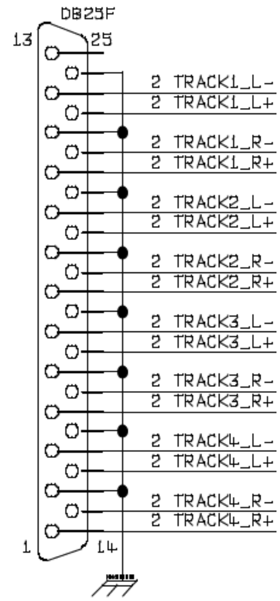


<u>AUX OUTPUTS / CUE OUTPUTS</u>: API = Standard

- 1 = Aux Output 1
- 2 = Aux Output 2
- 3 = Aux Output 3
- 4 = Aux Output 4
- 5 = Cue Output LEFT
- 6 = Cue Output RIGHT
- 7 =
- 8 =

External 2-Track Input Connections





<u>2 TRACK INPUTS 1-4</u>:

- 1 = 2-TRACK 1 Input LEFT
- 2 = 2-TRACK 1 Input RIGHT
- 3 = 2-TRACK 2 Input LEFT
- 4 = 2-TRACK 2 Input RIGHT
- 5 = 2-TRACK 3 Input LEFT
- 6 = 2-TRACK 3 Input RIGHT
- 7 = 2-TRACK 4 Input LEFT
- 8 = 2-TRACK 5 Input RIGHT

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