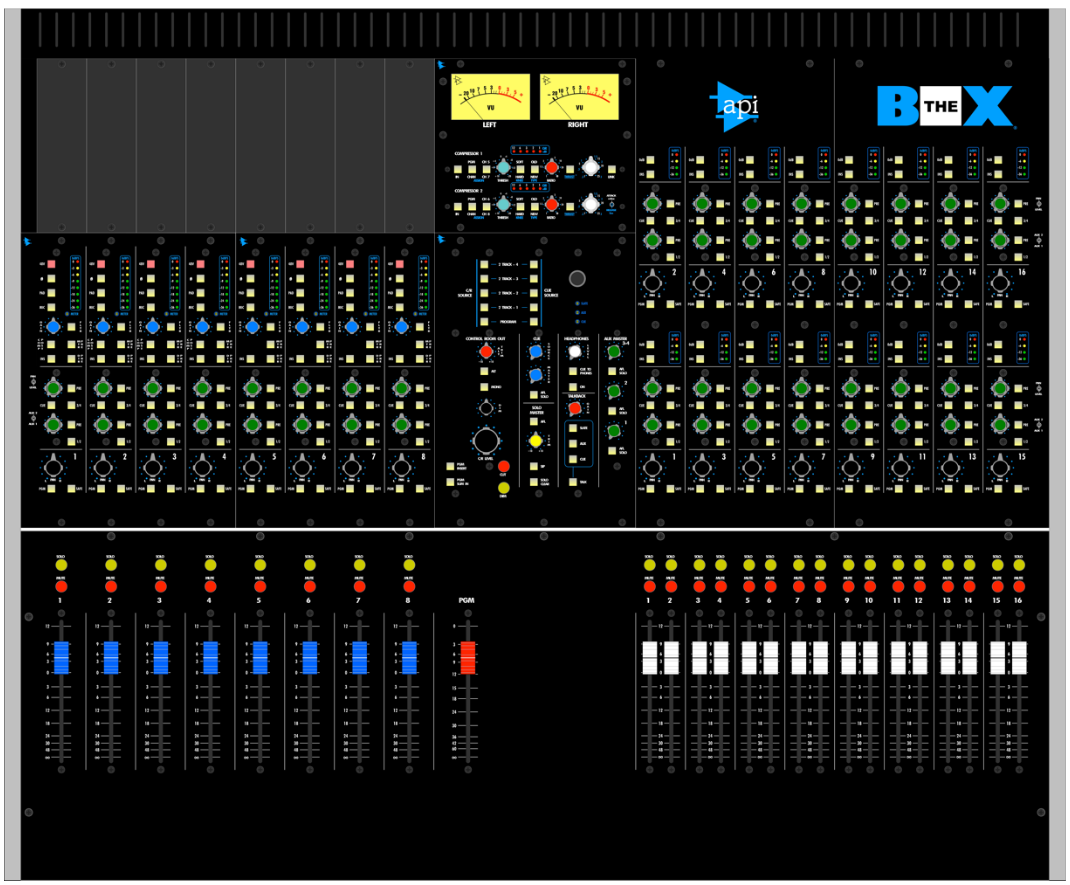
“THE BOX 8”
Recording & Mixing Console
Operator’s Manual

Automated Processes, Inc.



Written for Automated Processes Incorporated
by Dan Pfeifer
Rev. 21-2-2

THE BOX®

api®

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About This Manual

This manual explains the operation and applications of the 24-channel version of the API "**THE BOX**" recording console (a.k.a. "**THE BOX 8**"). This console is the next evolution of the original API "**THE BOX**" console that was released in 2013 and is identical in regard to program bus, cue bus, and auxiliary bus architecture, metering, and most master section functions and features. The primary difference between the original "**THE BOX**" and the "**THE BOX 8**" is the 24-channel consoles have eight (8) input channels instead of four (4). The original "**THE BOX**" console also had permanent 550A EQs on input channels 1-2 and only channels 3-4 had 500 Slots. "**THE BOX 8**" is equipped with eight (8) 500 Series module slots for greater flexibility.

Legend:

- **UPPER-CASE BOLD = SWITCHES, BUTTONS, POTS, & FADERS**
- UPPER-CASE = REAR PANEL CONNECTIONS

Manual Revision History: YY-MM-DD format
Original: 19-1-15
Revised: 19-2-24
Revised: 21-2-2

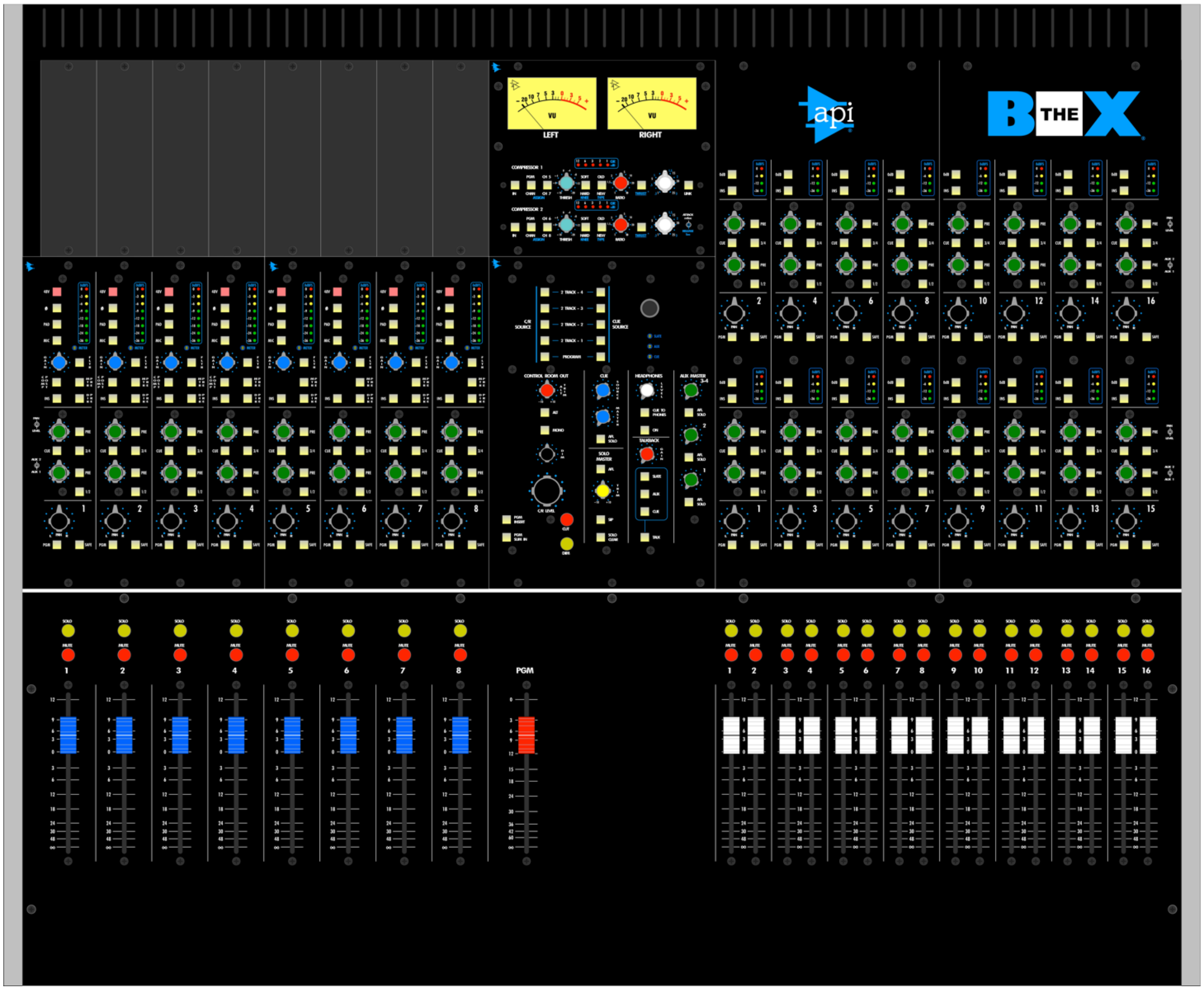
Important Safety Instructions

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 the AC power cord to the power supply 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



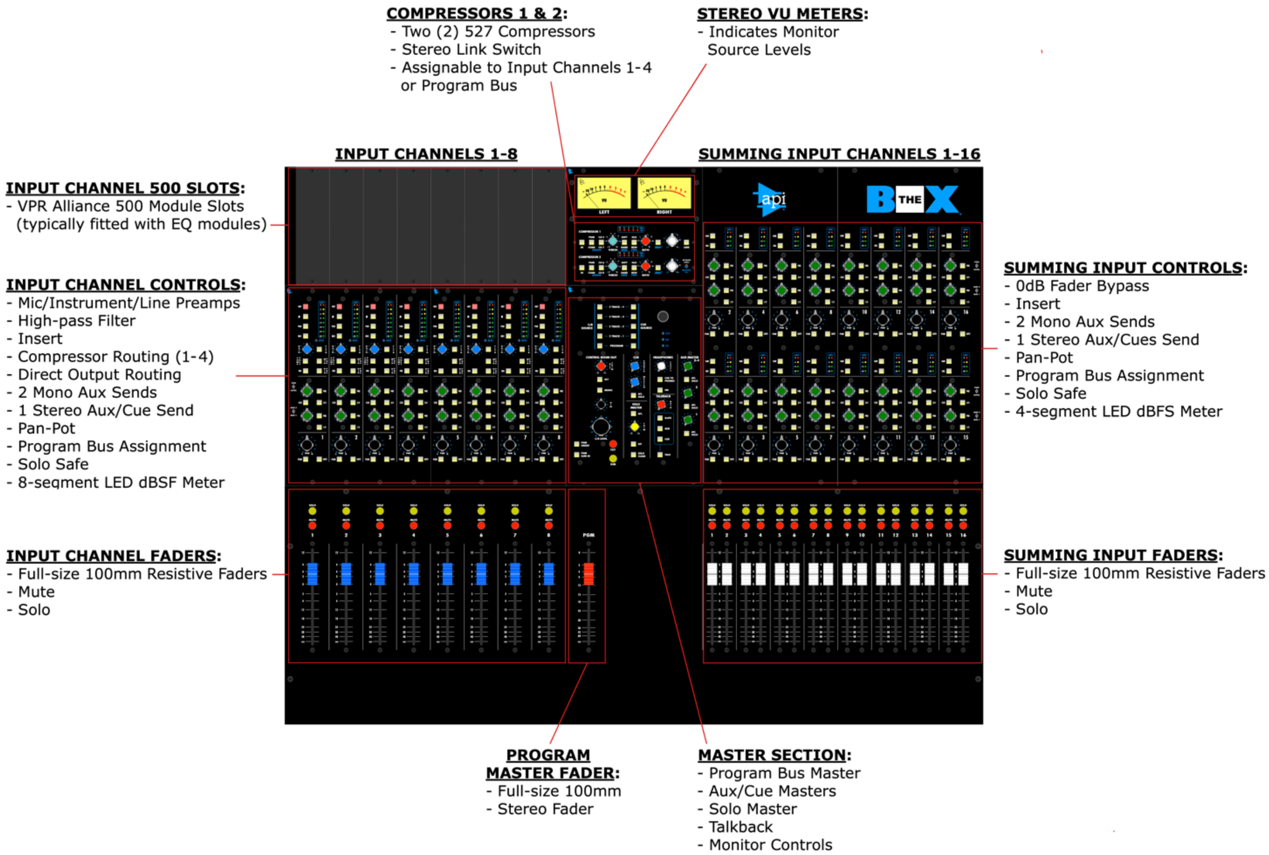
Building on API's rich heritage of extremely high-quality recording consoles, we introduce "**THE BOX 8**" a small-format recording/mixing console designed for professional project studios, home studios, and production facilities of all kinds. Optimized for the digital era, "**THE BOX 8**" handles all 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 8**" provides the legendary "discrete" API sound in an efficient, cost-effective package.

Features:

- Eight (8) input channels with mic/instrument/line preamp, HP filter, & 500 slot
- Sixteen (16) summing inputs (24 channels during mix)
- Stereo program bus with master fader, insert, and external input
- Two (2) compressors assignable to input channels or program bus with stereo link
- 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

1.1 Console Layout

"**THE BOX 8**" console provides eight (8) input channels, sixteen (16) summing inputs, two (2) mono auxiliary sends, one (1) stereo aux/cue send, assignable 527 compressors, a stereo program bus, and a comprehensive master section. This configuration provides eight input channels for recording and a total of 24 channels while mixing. The assignable 527 compressors, eight (8) 500 module slots, and internal routing options provide enhanced flexibility and sonic options for both recording and mixing workflows.



All Input channels are equipped with mic/line preamp, high-pass filter, 500 module slot, insert, aux/cue sends, and output routing. Input channels 1-4 are also equipped with instrument inputs. The two built-in 527 compressors in the master section can be assigned channels 1-4 for use during recording or mixing.

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 8**" is equipped with a comprehensive master section that includes controls for program bus insert and external sum input, 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 1-4 when recording and the program bus when mixing.

To help assure reliability and longevity, "**THE BOX 8**" is built to the same exacting API quality standards as our Vision, Legacy AXS, 2448, and 1608-II consoles.

2.0 Input Channels 1-8

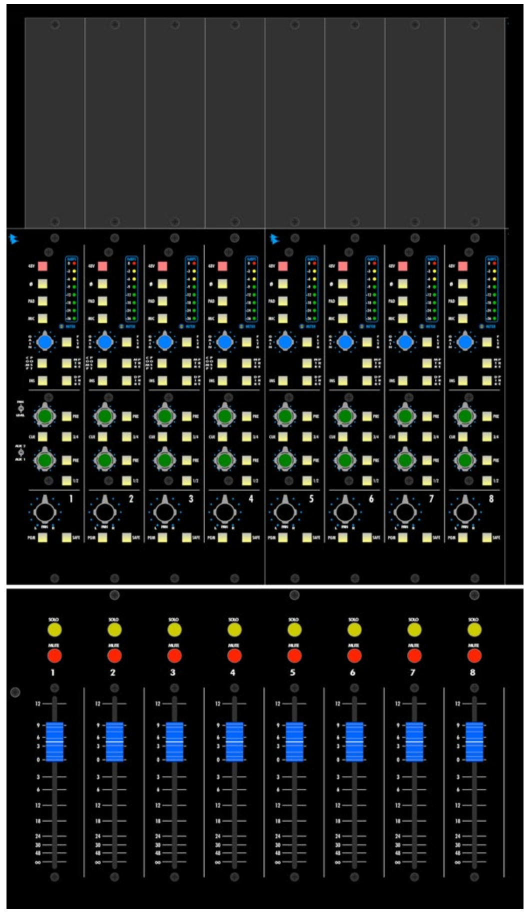
"**THE BOX 8**" provides eight (8) input channels on the left-hand side of the console. From top to bottom, each input channel has three sections: 500 Series module slots, input channels, and faders.

INPUT CHANNELS 1-8

500 Module Slots

Input Channels

Faders



Working together, these three sections form an integrated input channel that provides a single, comprehensive audio path for recording applications. When mixing, the input channels augment the sixteen summing inputs by providing eight (8) multitrack return channels with enhanced signal processing capabilities. Because of these enhanced capabilities, these channels are often used for delivering critical tracks, such as vocals, to the mix.

Input channel features include:

- Microphone/instrument/line preamplifier (INST IN channels 1-4 only)
- High-pass filter
- Integral 500 module slot
- Balanced insert send/return
- Two (2) routable 527 compressors (channels 1-4 only)
- Two (2) mono auxiliary sends
- One (1) stereo aux/cue send
- Fader and mute
- Pan-pot
- Solo and solo safe
- Routable 8-segment LED dBFS meter
- Direct output (post-fader or preamp output routing)
- Stereo program bus assignment

2.1 Input Channel Simplified Signal Flow

The signal flow is identical for all eight input channels, except for instrument inputs and compressor routing. Input channels 1-4 are equipped with switching instrument input jacks (DIs) that feed the preamp. These instrument inputs are located under the armrest (INST IN). The two 527 compressors in the master section can be individually assigned to channels 1-4. These compressors cannot be assigned to input channels 5-8.

The source for all input channels is the output of the preamp. The preamp can accept the following sources:

- **Microphone**: Rear panel MIC INPUT (**MIC**)
- **Instrument**: Switching INST IN jacks located under the armrest (channels 1-4 only)
- **Line**: Rear panel LINE INPUT

The output of all input channels can be routed as follows:

- **Direct Output**:
  - Post-fader: Default
  - Post-preamp/pre-processing (**DIR PRE** engaged)
- **Program Bus**: Stereo program bus, post-pan-pot

The direct output appears at the DIRECT OUTPUTS DB25 connector on the rear panel for interfacing with a DAW or another recorder.

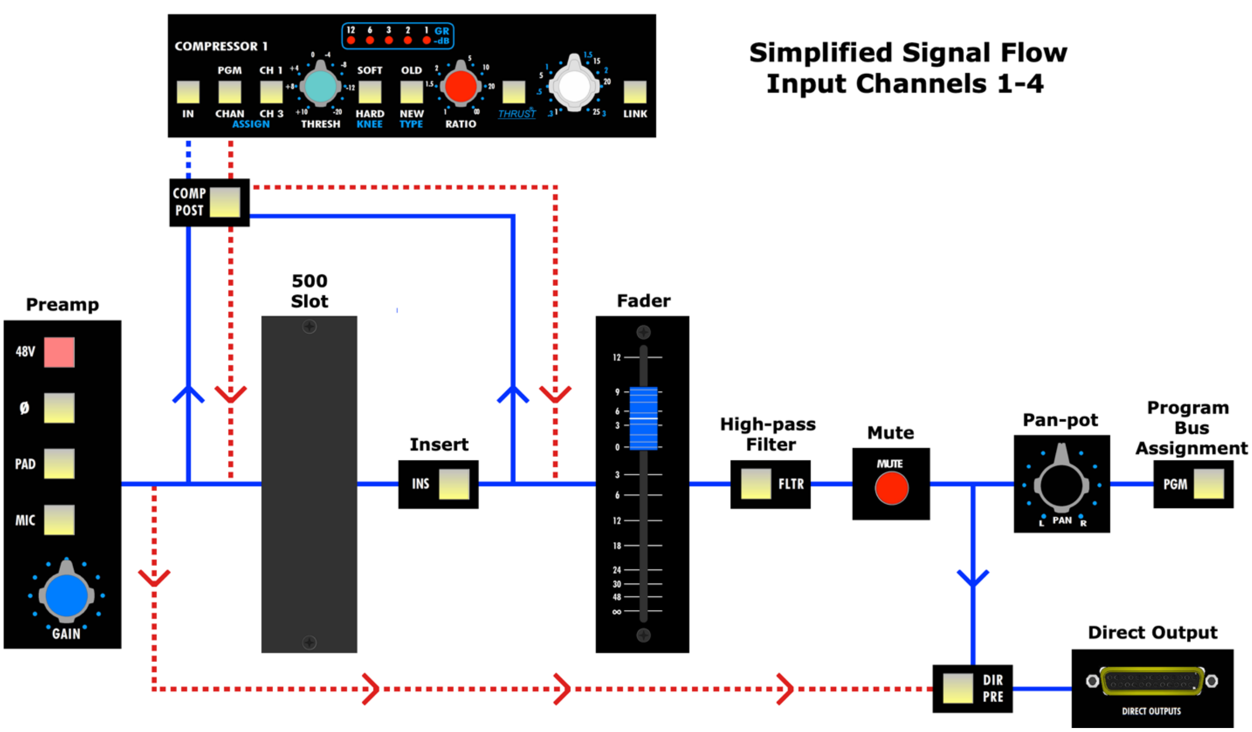
2.1.1 Input Channel 1-4 Signal Flow

The basic signal flow input channels 1-4 is as follows:

Preamplifier ⇒ compressor ⇒ 500 slot ⇒ insert ⇒ fader ⇒ filter ⇒ mute ⇒ output

The compressor can be moved to after the insert (COMP POST) to yield:

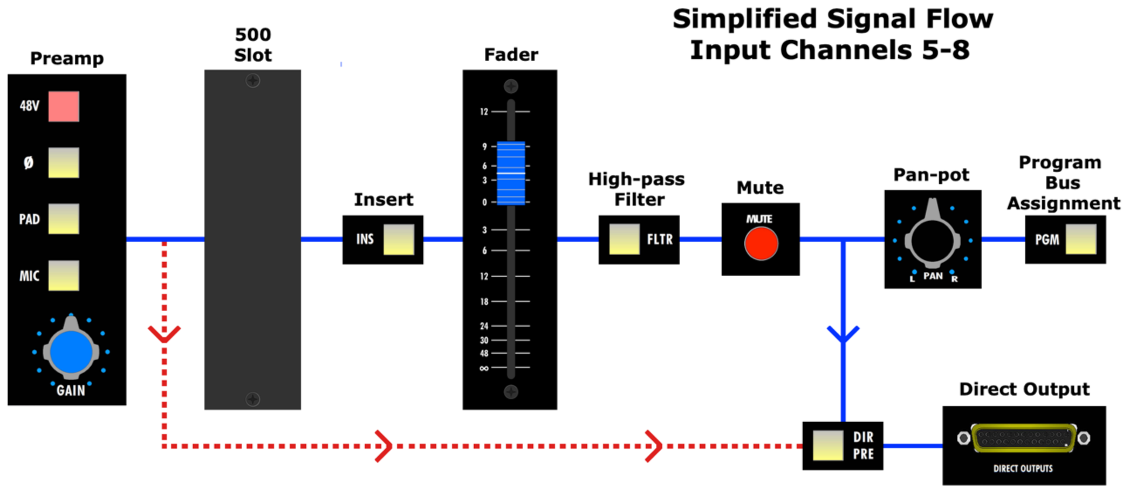
Preamplifier ⇒ 500 slot ⇒ insert ⇒ compressor ⇒ fader ⇒ filter ⇒ mute ⇒ output



2.1.2 Input Channel 5-8 Signal Flow

The basic signal flow input channels 5-8 is as follows:

Preamplifier ⇒ 500 slot ⇒ insert ⇒ fader ⇒ filter ⇒ mute ⇒ output

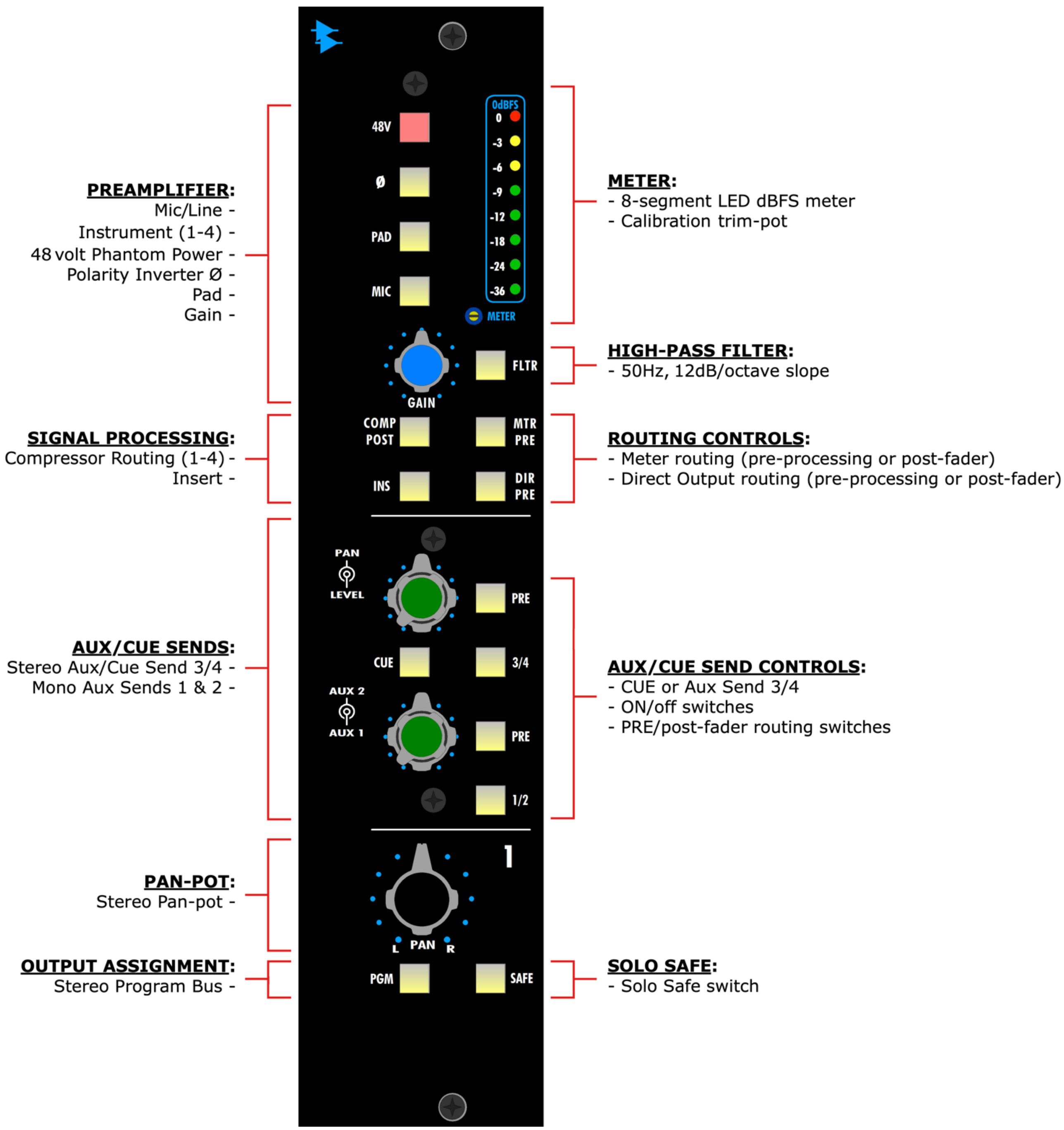


2.2 Input Channel Controls

The controls for the input channels are as follows:

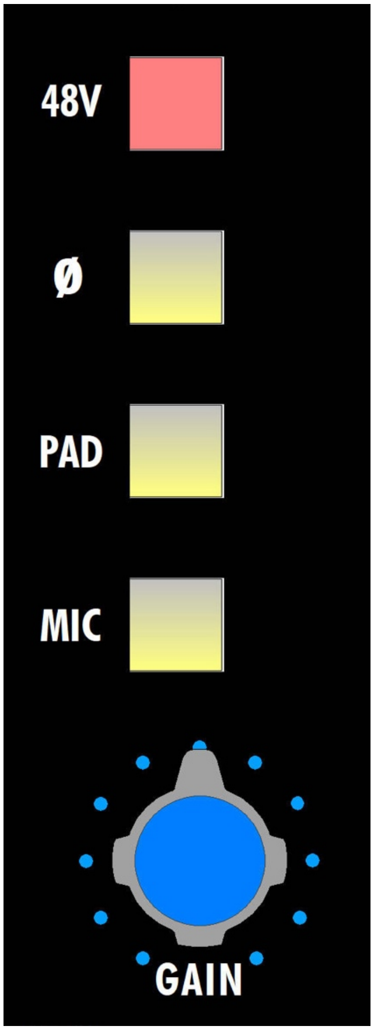
- **Preamplifier**: Microphone, instrument (1-4 only), and line inputs
- **Signal Processing**: 50Hz high-pass filter, 500 module slot, and insert
- **Aux/Cue Sends**: Mono aux 1-2, stereo aux 3/4, and stereo cue send
- **Output Routing**: Direct output, pan-pot, program bus assignment, with solo **SAFE**
- **Meter**: 8-segment LED dBFS full-scale meter

The controls for the input channels preamp, signal processing, auxiliary/cues sends, output routing, and metering are laid out as indicated in the diagram below.



2.3 Input Channel Preamplifier

The input channel preamplifier provides a fully featured input stage that's identical to the channel preamps in the API 1608, 1608-II, and 2448 consoles.



The preamp can accept the following input signals:

- **Microphone**: Rear panel MIC INPUT 3-pin F-XLR
- **Instrument**: Switching ¼" INST IN (under channels 1-4)
- **Line**: Rear panel LINE INPUT ¼" TRS line input

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

- Up to +65dB of gain (+30dB to +65dB)
- -20dB pad
- Ø Polarity inverter (phase reverse)
- 48 volt phantom power

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

NOTE: The signal from the MIC INPUT is replaced with the INST IN signal when a ¼" plug is inserted in the jack.

Line Input: Accepts balanced, low-impedance, line-level signals from the ¼" LINE INPUT connector. The line input has the following features:

- -6dB pad
- Ø 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 switching ¼" INST IN connector under the armrest for input channels 1-4. The instrument input serves the function of an external direct injection (DI) box and has the following features:

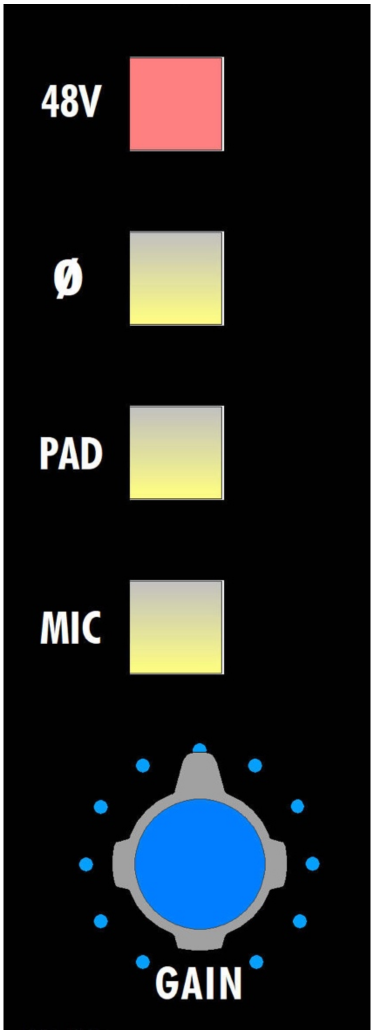
- Direct input from instruments such as electric guitars, basses, keyboards, etc.
- Up to +46dB of gain (+11dB to +46dB)
- Input channels 1-4 only

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

NOTE: INST IN jack is a switching jack. The signal from the MIC INPUT is replaced with the signal from the INST IN whenever a ¼" plug is inserted in the INST IN 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.

2.3.1 Preamp Controls



48V (Phantom Power): 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

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

- Does not affect the INST IN signal
- Illuminates when engaged

PAD:

- Attenuates the MIC INPUT by 20dB
- Attenuates the LINE INPUT by 6dB
- Does not affect the INST IN signal
- Illuminates when engaged

MIC: Selects the microphone signal as the preamp source:

- **MIC** must be engaged to use the INST IN

*NOTE: LINE INPUT is selected by default (**MIC** not engaged)*

NOTE: The MIC INPUT signal is replaced by the instrument input signal when a ¼" plug is inserted in the INST IN jack (input channels 1-4)

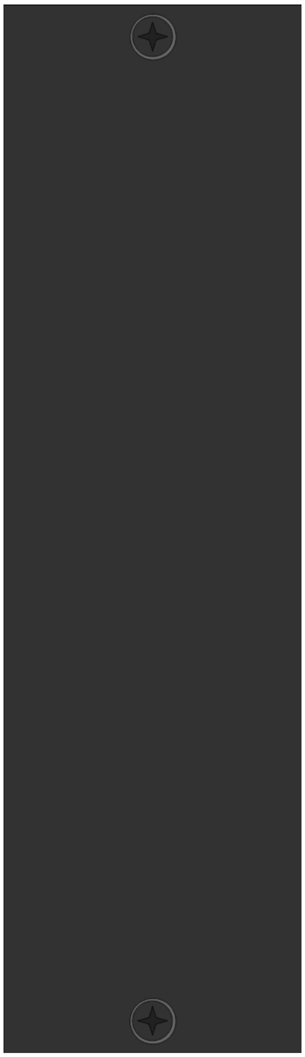
GAIN: Provides up to +65dB of preamp gain

- Mic: +30dB to +65dB range
- Instrument: +11dB to +46dB

2.4 Input Channel Signal Processing

In addition, the pad, gain, and polarity inverter provided by the preamp, input channels 1-8 provide a 500 Series module slot, a 50Hz high-pass filter, and a balanced insert for signal processing. Additionally, the two 527 compressors in the Master Section can be routed to input channels 1-4, pre or post the 500 slot and insert.

2.4.1 500 Series Slot



A built-in 500 series slot is provided on each input channel and is an integral part of the input channel signal flow. It allows input channel signal processing to be augmented with a 500 Series module. These slots are empty when the console is shipped, allowing the owner to add the “flavor” of signal processing of their choice on an “ala carte” basis. API equalizers are the most common choice, but other than preamps, the slots can accept any 500 Series module that conforms to the VPR 500 Alliance standards.

On input channels 1-4 the 500 slot is located immediately after the compressor (if routed) by default and always before the insert. If **COMP POST** is engaged, the 500 slot will be located immediately after the preamp (and the compressor will be after the insert).

On input channels 5-8 the 500 slot is always located immediately after the preamp and before the insert.

The 500 Series slots are built to power and interface any API or JDK Audio 500 series module. The 500 slots or often populated with EQ and spectral processing modules such as:

- API 550A 3-Band EQ
- API 550b 4-Band EQ
- API 560 10-band Graphic EQ
- API 565 Filter Bank
- JDK V14 4-Band EQ

In some cases it might be desirable to install dynamic processors such as:

- API 525 Compressor
- API 527 Compressor/Limiter

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

In addition to API and JDK modules, any third-party 500 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 will 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.

NOTE: The 500 slots are not configured to accept preamp modules.

2.4.2 High-pass Filter

A 50Hz high-pass filter is provided on each input channel.

FLTR (Filter): Engages the high-pass filter

- -3dB at 50Hz
- 12dB/octave slope
- Illuminates when engaged

The high-pass filter is located post-fader and pre-mute.

2.4.3 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

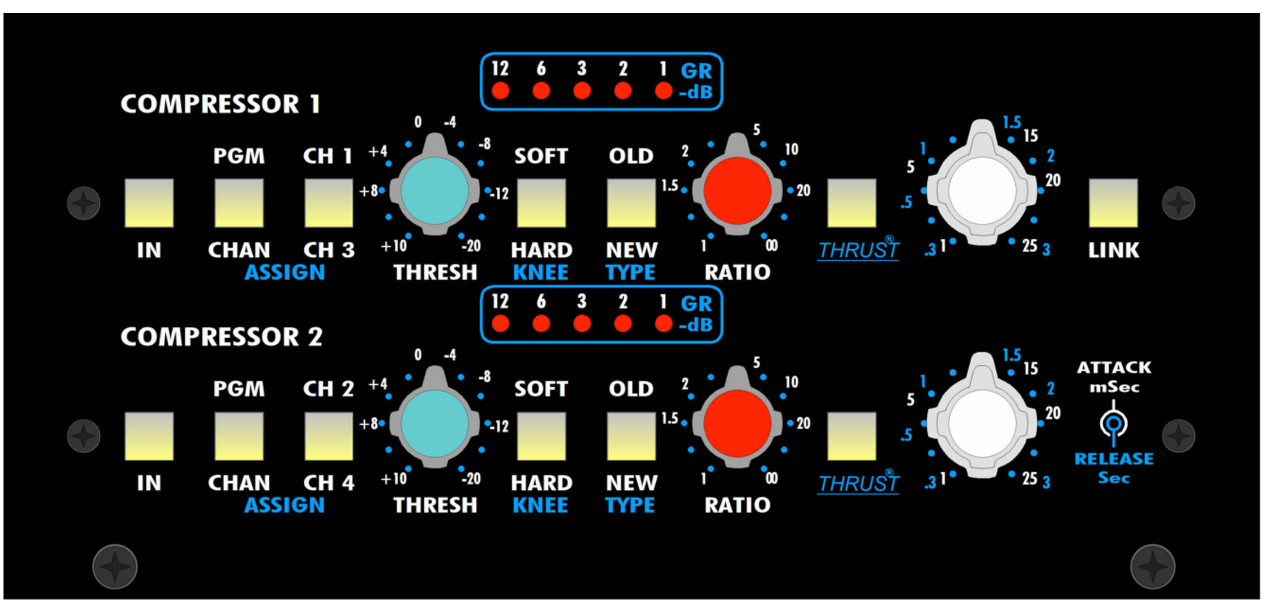
The insert is always located immediately after the 500 slot on all input channels. On channels 1-4 the insert is located before the fader by default. If **COMP POST** is engaged on input channels 1-4, the compressor will be located immediately after the insert and before the fader.

On input channels 5-8 the insert is always located immediately after the 500 slot and before the fader.

The insert signal is connected to the INSERT SEND jack on the rear panel and is always active. The signal from the INSERT RETURN jack on the rear panel is routed to the channel signal path only when the **INS** switch is engaged.

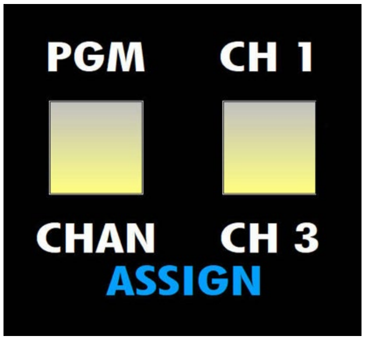
2.4.4 Compressor Routing

"**THE BOX 8**" is equipped with two API 527 compressor/limiters in 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 1-4 while recording or mixing.



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

Compressor 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.

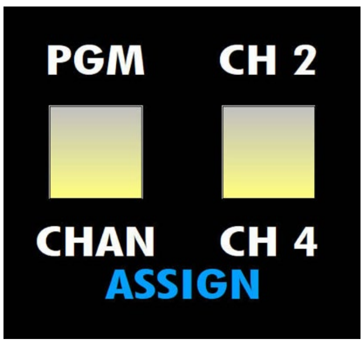


PGM - CHAN (Program – Channel): Routes Compressor 1 to input channels 1 or 3 when engaged

- Compressor 1 is routed to input channel 1 by default
- Illuminates when engaged

CH 1 – CH 3 (Channel 1 – Channel 3): Routes Compressor 1 to input channels 3 when engaged

- Routes Compressor 1 to input channel 3 instead of channel 1
- Compressor 1 **CHAN** must be engaged
- Illuminates when engaged



PGM - CHAN (Program – Channel): Routes Compressor 2 to input channels 2 or 4 when engaged

- Compressor 2 is routed to Channel 2 by default
- Illuminates when engaged

CH 2 – CH 4 (Channel 2 – Channel 4): Routes Compressor 2 to input channels 4 when engaged

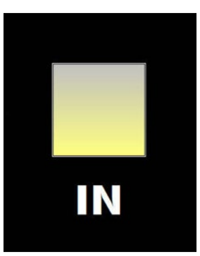
- Routes Compressor 2 to input channel 4 instead of channel 2
- Compressor 2 **CHAN** must be engaged
- Illuminates when engaged

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

COMP POST (Compressor Post): Routes the compressor post the 500 slot and insert on that channel when engaged

- Compressors are routed post-preamp and pre-processing by default
- Illuminates when engaged

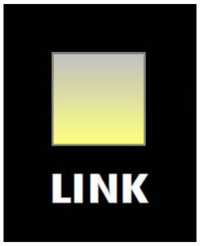
The compressor will be bypassed unless the **IN** switch on the compressor is engaged. Engaging the **IN** switch will place the compressor in the signal path.



IN: Places the compressor in the assigned signal path

- Illuminates when engaged

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

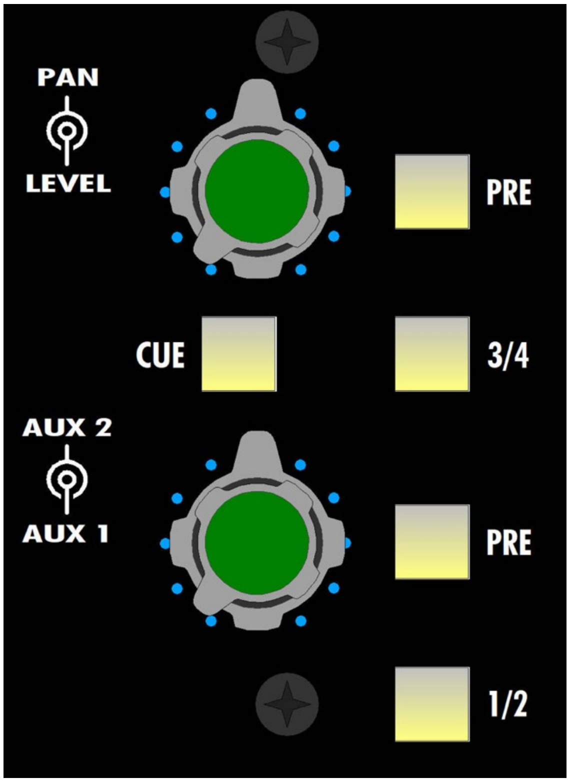


LINK: Sums the gain reduction control voltages of Compressor 1 and Compressor 2 for stereo processing.

- Illuminates when engaged

Refer to section 6.0 Compressors 1 and 2 for more information regarding the compressors.

2.5 Input Channel Auxiliary/Cue Sends



"**THE BOX 8**" is equipped with two (2) mono auxiliary buses (1/2), one (1) stereo auxiliary bus (3/4), and a stereo cue bus. Accordingly, all input channels are equipped with two mono aux sends (1 & 2) and one stereo (3/4) aux/cue send.

The stereo aux send (3/4) feeds stereo aux bus 3/4 by default, but can be routed to feed the stereo cue bus instead.

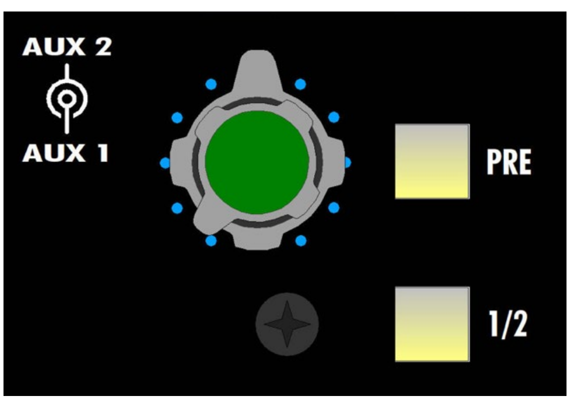
Each pair of aux sends (1/2, 3/4) has pre/post-fader routing and an ON/off switch.

In common practice, post-fader sends are often used for effects sends and pre-fader sends are typically used for headphone feeds.

2.5.1 Mono Auxiliary Sends 1/2

AUX 1 and AUX 2 are mono auxiliary sends that feed aux buses 1 and 2. These aux 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 engage both mono sends. AUX 1/2 is fed post-fader by default, but can be fed the pre-fader signal by engaging **PRE**.



<u>AUX 1 - AUX 2</u>: Level controls for mono aux sends 1/2

- Knob = 1
- Ring = 2

<u>PRE (Pre-fader):</u> Routes the pre-fader source to AUX 1/2 when engaged

- AUX 1/2 are fed post-fader by default
- Illuminates when engaged

<u>1/2 (ON/off)</u>: ON/off switch for AUX 1/2

- Illuminates when engaged

*NOTE: An internal jumper determines the source of the aux/cue feed when the **PRE** switch is engaged. Refer to section 2.5.3 Aux Pre Jumper for detailed information.*

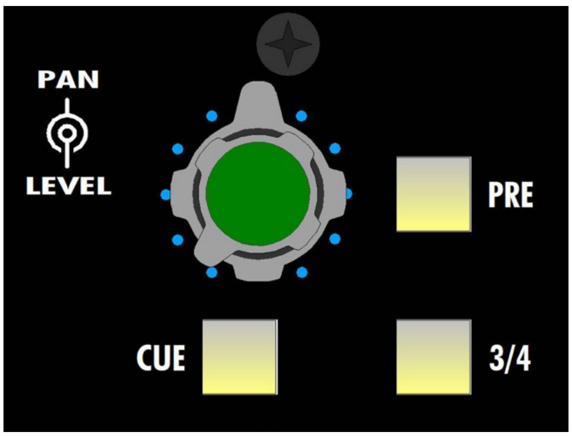
2.5.2 Stereo Auxiliary Sends 3/4

AUX 3/4 is a stereo auxiliary send that feeds aux buses 3 and 4 by default, but can be routed to feed the stereo cue bus instead. Aux sends 3/4 are commonly used for headphone sends while recording and effects sends while mixing, but can be used for other purposes.

In addition to auxiliary buses 1-4, "**THE BOX 8**" is equipped with a separate stereo cue bus, cue master, and balanced cue 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 and fed to the headphones. When mixing, AUX 3/4 can be routed to the cue bus instead of aux buses 3/4 to provide an alternate stereo effects send.

Engaging **CUE** will route AUX 3/4 to the stereo cue bus to allow a fully featured headphone feed to be created when recording. AUX 3/4 can be routed to aux buses 3/4 or the stereo cue bus for use as effects sends when mixing.

A common ON/off switch is used to engage the stereo send. AUX 3/4 is fed post-fader by default, but can be fed the pre-fader signal by engaging **PRE**.



AUX 3/4: Level and pan controls for stereo aux send 3/4-CUE
- Knob = level
- Ring = pan

PRE (Pre-fader): 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 switch for AUX 3/4-CUE
- Illuminates when engaged

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

*NOTE: An internal jumper determines the source of the aux/cue feed when the **PRE** switch is engaged. Refer to section 2.5.3 Aux Pre Jumper for detailed information.*

2.5.3 Aux Pre Jumper

Aux/Cue sends set to **PRE** can be fed from two locations in the signal path, pre-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-500 slot
  - Post-preamp, post-compressor (if routed)
  - Pre-500 slot, pre-insert send, pre-fader
- FDR IN (fader input): Pre-fader input
  - Post-preamp, post-compressor (if routed), post-500, post-insert return,

The default position for this routing jumper is factory-set to FDR IN (fader input). This routing option and jumper are available only on input channels 1-8. Aux/cue sends on summing inputs 1-16 are fed from the fader input when their **PRE** switches are engaged.

2.6 Input Channel Output Routing

All input channels can be routed to the following outputs:

- **<u>Direct Output</u>**:
  - Post-fader: Default
  - Post-preamp/pre-processing: **DIR PRE** engaged
- **<u>Program Bus</u>**: Stereo program bus, post-pan-pot

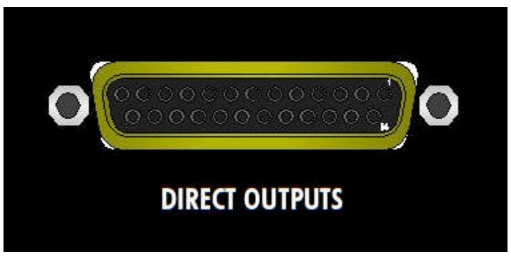
2.6.1 Direct Output

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

<u>DIR PRE (Direct Preamp Output)</u>: Replaces the post-mute signal with the preamp output at the DIRECT OUTPUTS connector on the rear panel.

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

The DIRECT OUTPUTS connector carries direct outputs from input channels 1-8.



<u>DIRECT OUTPUTS</u>: Line-level direct outputs from input channels 1-8

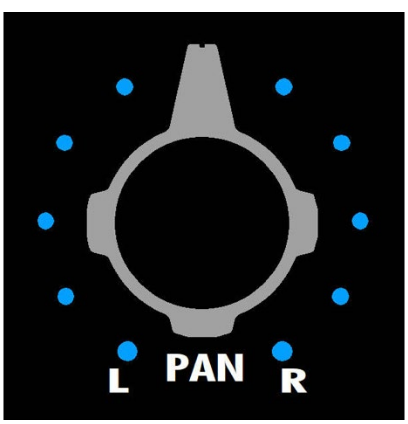
- Fed post-fader/mute by default (direct output)
- Preamp output replaces post-mute output if **DIR PRE** is engaged
- Balanced, low-impedance
- DB25 connect (standard pin-out)

2.6.2 Stereo Program Bus and Pan-pot

Input channels can be routed to the stereo program bus via the pan-pot when mixing, providing eight (8) multitrack return channels.

<u>PGM (Program)</u>: Assigns the input channel output to the program bus when engaged

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



<u>PAN (Pan-pot)</u>: Continuously variable stereo (left-right) pan-pot

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

2.6.3 Solo Safe

"**THE BOX 8**" is equipped with a "Solo-In-Place" function (SIP) that mutes input channels and summing inputs when **SOLO** is engaged on another channel or input. Engaging solo **SAFE** prevents that channel from being muted when a solo is engages in SIP mode.

<u>SAFE (Solo Safe)</u>: 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

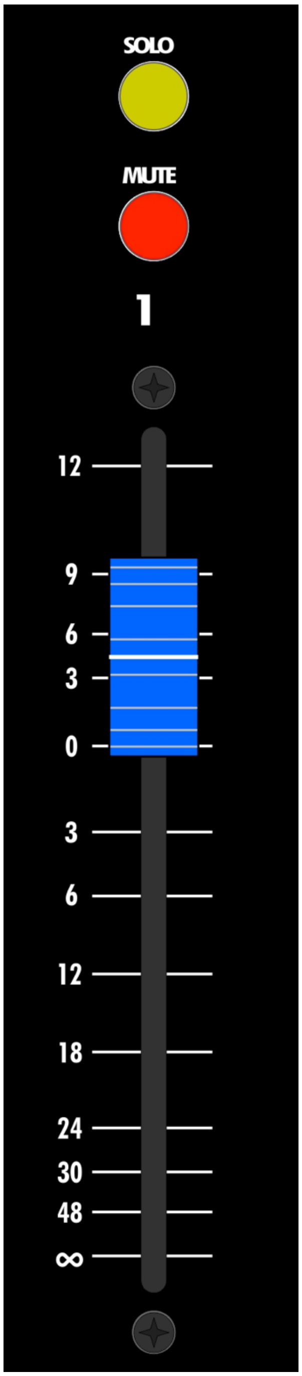
Refer to section 10.0 Solo Master for more information regarding the solo system.

2.7 Input Channel Faders

The fader section with blue fader caps contains the input channel faders, **MUTE**, and **SOLO** switches for input channels 1-8. These faders are the primary output level controls for the input channels.

The input channel faders can be routed to feed:

- **Direct Output**: Default DIRECT OUTPUT feed
- **Stereo Program Bus**: When **PGM** is engaged
- **Post-fader auxiliary/cue sends**: Default aux/cue send feed



SOLO: Activates the current solo function on that channel

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

MUTE: Cuts the channel output

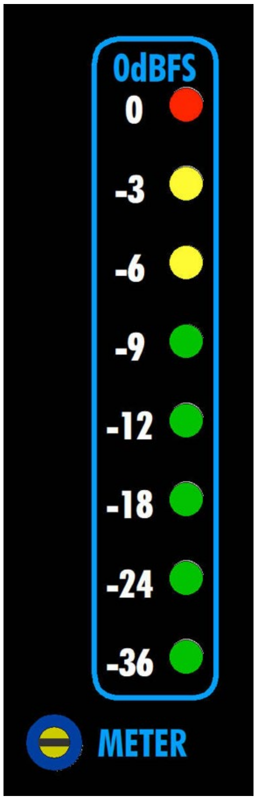
- Located post-fader, post-filter in the signal flow
- Illuminates in red when engaged
- Activates on release

INPUT FADER: Primary output level control

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

2.8 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.

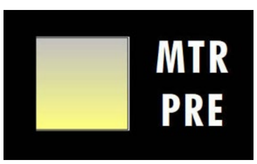


0dBFS: 8-segment LED full-scale meter

- 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/mute with the output of the preamp as the meter source

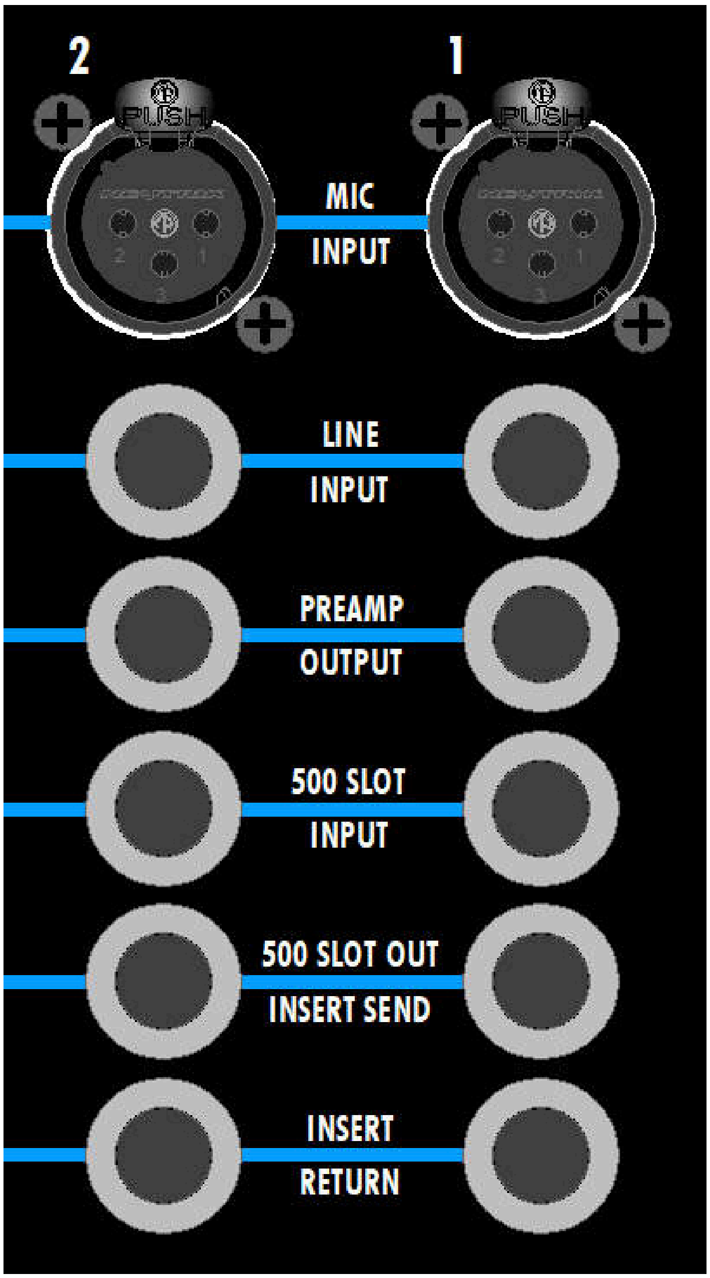
- Default meter feed is post-fader channel output if not engaged
- Illuminates when engaged

2.9 Input Channel Rear Panel Connections

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

Connections include:

- Preamp inputs:
  - Microphone input
  - Line input
  - Instrument input (under armrest of input channels 1-4)
- Preamp output
- 500 slot input
- 500 slot output/Insert send
- Insert return
- Direct output



MIC INPUT: Preamp microphone input

- Engage **MIC** to use this input
- Balanced, low-impedance
- Provides phantom power when the **48V** switch is engaged
- 3-pin female XLR connector

LINE INPUT: Preamp line input

- Disengage **MIC** to use this input
- Line-level, balanced, low-impedance
- ¼" tip-ring-sleeve

PREAMP OUTPUT: Preamp output

- Line-level, balanced, low-impedance
- ¼" tip-ring-sleeve

500 SLOT INPUT: 500 slot input

- Breaks feed from preamp/comp
- Line-level, balanced, low-impedance
- ¼" tip-ring-sleeve switching jack

500 SLOT OUTPUT/INSERT SEND: 500 slot output/insert output

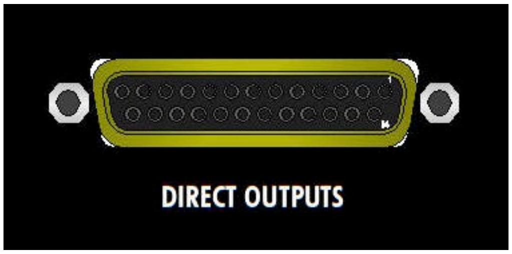
- Fed from 500 slot output
- Line-level, balanced, low-impedance
- ¼" tip-ring-sleeve

INSERT RETURN: Insert input

- Active when **INS** is engaged
- Line-level, balanced, low-impedance
- ¼" tip-ring-sleeve switching jack

NOTE: 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.

The DIRECT OUTPUTS connector carries direct outputs from input channels 1-8.



DIRECT OUTPUTS: Direct outputs from input channels 1-8

- Fed post-fader/mute by default (direct output)
- Preamp output replaces post-mute output if **DIR PRE** is engaged
- Line-level, balanced, low-impedance
- DB25 connector (standard pin-out)
  - 1-8=direct outputs 1-8

2.9.1 Instrument Input

Input channels 1-4 are equipped with an instrument input that accepts unbalanced, high-impedance, instrument-level signals from the switching ¼" INST IN connector under the armrest. The instrument input serves the function of an external direct injection (DI) box and has the following features:

- Direct input from instruments such as electric guitars, basses, keyboards, etc.
- Up to +46dB of gain (+11dB to +46dB)

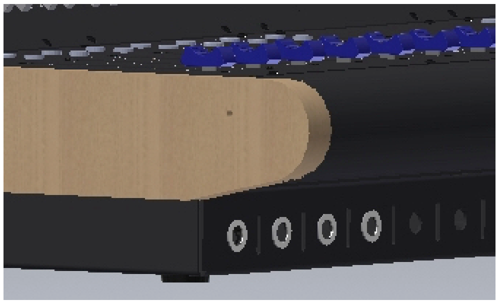
*NOTE: The **MIC** switch must be engaged to use this input.*

NOTE: INST IN jack is a switching jack. The signal from the MIC INPUT is replaced with the signal from the INST IN whenever a ¼" plug is inserted in the INST IN 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.

INST IN: Instrument-level preamp input

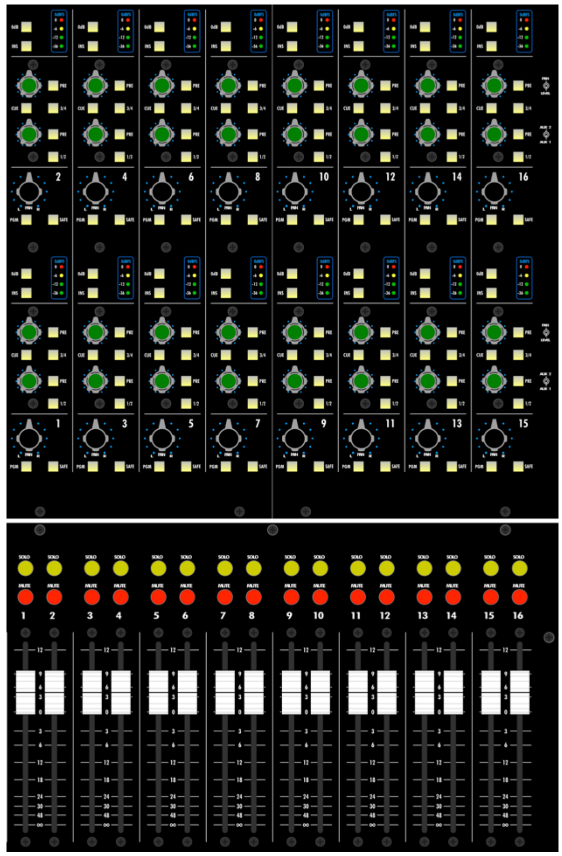
- Input channels 1-4 only
- Located under armrest
- Engage **MIC** to use this input
- Replaces MIC INPUT signal as the preamp input
- Unbalanced, high-impedance
- ¼" tip-ring switching jack



3.0 Summing Inputs 1-16

"**THE BOX 8**" provides sixteen (16) summing input channels (summing inputs) on the right-hand side of the console that are optimized for mixing in a DAW environment. All sixteen summing inputs are identical. Each summing input has two sections: summing inputs and faders.

SUMMING INPUTS 1-16



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.

Summing input channel features include:

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

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.

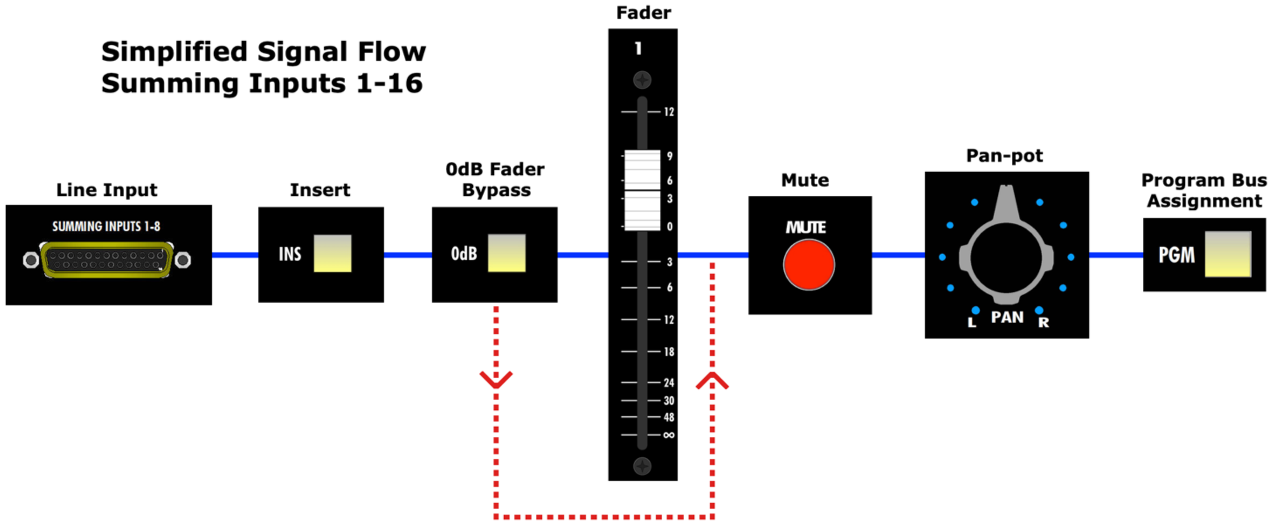
3.1 Summing Input Simplified Signal Flow

The signal flow is identical for all sixteen summing inputs.

3.1.1 Summing Input Channels 1-16 Signal Flow

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



The main source for all summing inputs is the line input signal from the SUMMING INPUTS 1-8 and SUMMING INPUTS 9-16 connectors on the rear panel.

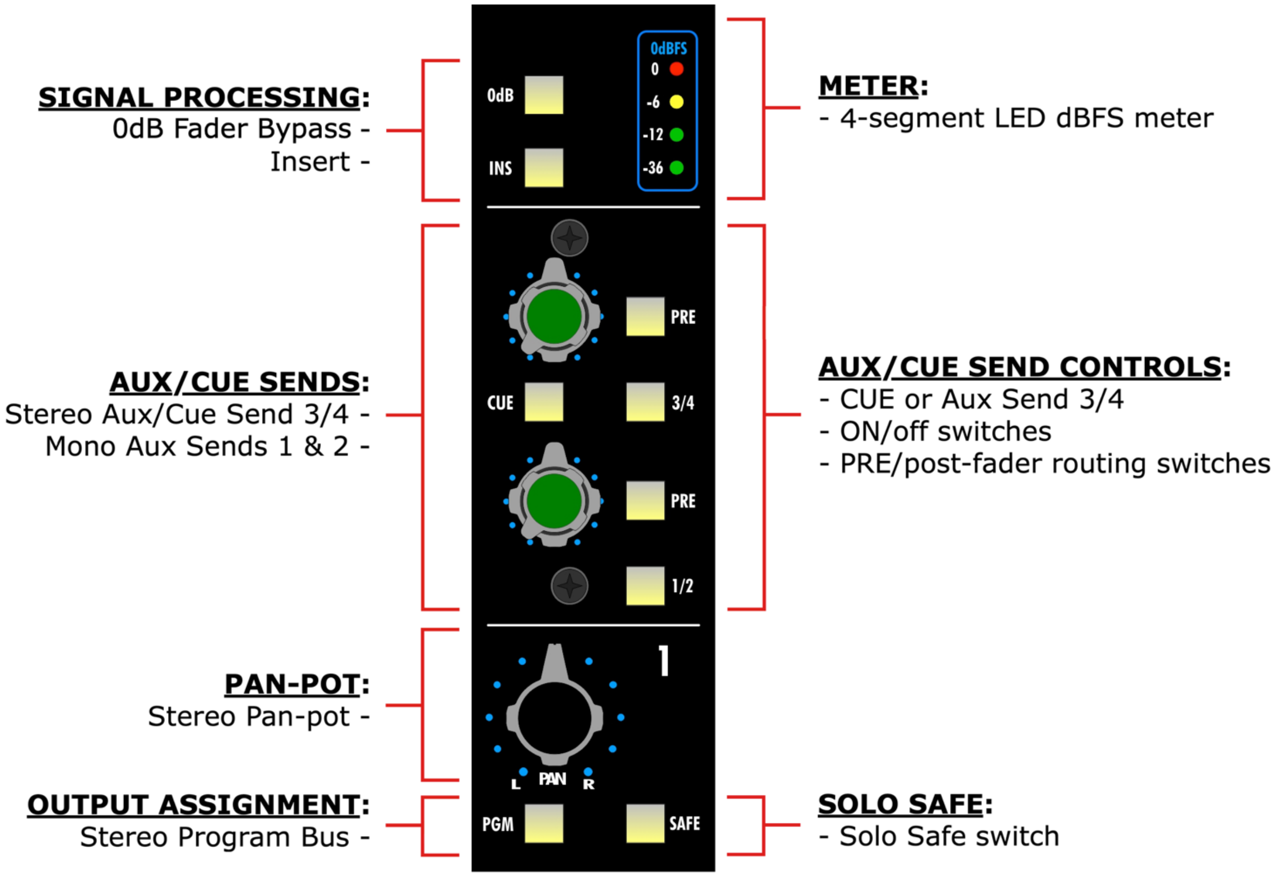
The output of summing inputs can be routed to the stereo program bus, post-pan-pot.

3.2 Summing Input Controls

The controls for the summing inputs are as follows:

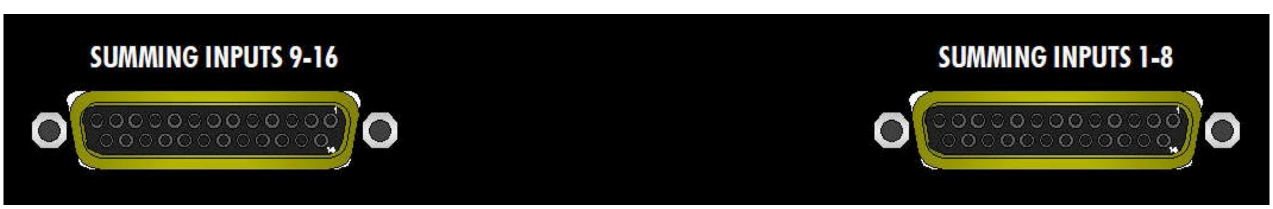
- **Signal Processing**: 0dB fader bypass and insert
- **Aux/Cue Sends**: Mono aux 1-2, stereo aux 3/4, and stereo cue send
- **Output Routing**: Pan-pot, program bus assignment, and solo **SAFE**
- **Meter**: 4-segment LED dBFS full-scale meter

The controls for the summing input signal processing, auxiliary/cues sends, and output routing are laid out as indicated in the diagram below.



3.3 Summing Input Line Input

Summing inputs accept balanced, low-impedance, line-level signals from the SUMMING INPUTS 1-8 and SUMMING INPUTS 9-16 DB25 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.

3.4 Summing Input Signal Processing

Summing inputs 1-16 provide a **0dB** fader bypass and a balanced insert for signal processing.

3.4.1 0dB Fader Bypass

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 8"** 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 (0dB Fader Bypass): Engages the unity gain fader bypass
- Illuminates when engaged

3.4.2 Insert

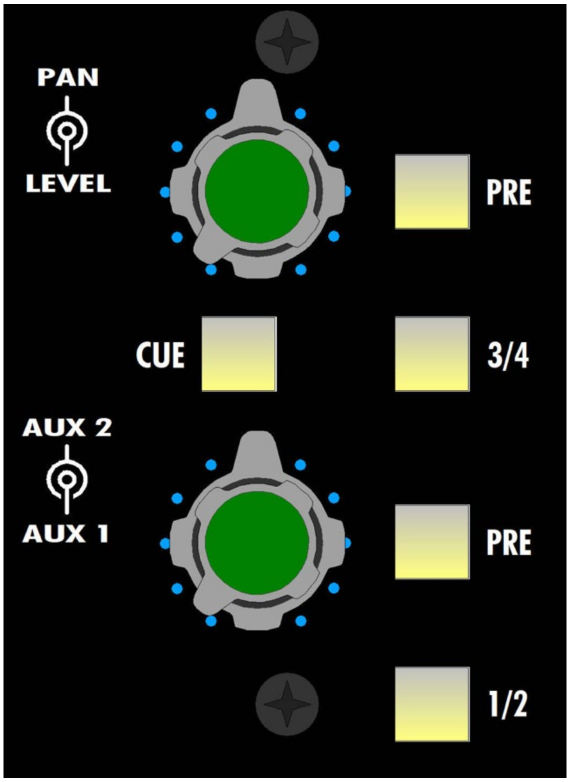
A balanced insert with switch is provided on each summing input.

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 directly from 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

3.5 Summing Input Auxiliary/Cue Sends



"**THE BOX 8**" is equipped with two (2) mono auxiliary buses (1/2), one (1) stereo auxiliary bus (3/4), and a stereo cue bus. Accordingly, all summing inputs are equipped with two mono aux sends (1 & 2) and one stereo (3/4) aux/cue send.

The stereo aux send (3/4) feeds stereo aux bus 3/4 by default, but can be routed to feed the stereo cue bus instead.

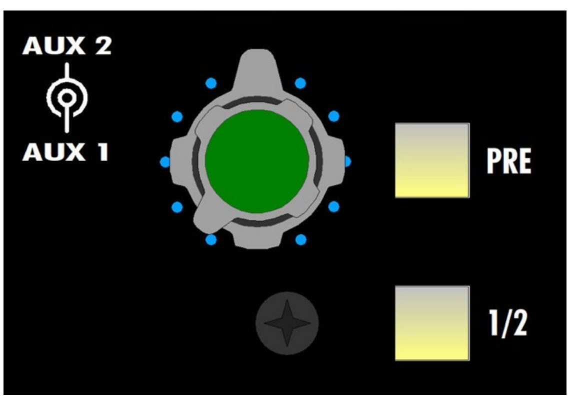
Each pair of aux sends (1/2, 3/4) has pre/post-fader routing and an ON/off switch.

In common practice, post-fader sends are often used for effects sends and pre-fader sends are typically used for headphone feeds.

3.5.1 Mono Auxiliary Sends 1/2

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

A common ON/off switch (**1/2**) is used to engage both mono sends. AUX 1/2 is fed post-fader by default, but can be fed the pre-fader signal by engaging **PRE**.



AUX 1 – AUX 2: Level controls for mono aux sends 1/2

- Knob = 1
- Ring = 2

PRE (Pre-fader): 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 switch for aux 1/2

- Illuminates when engaged

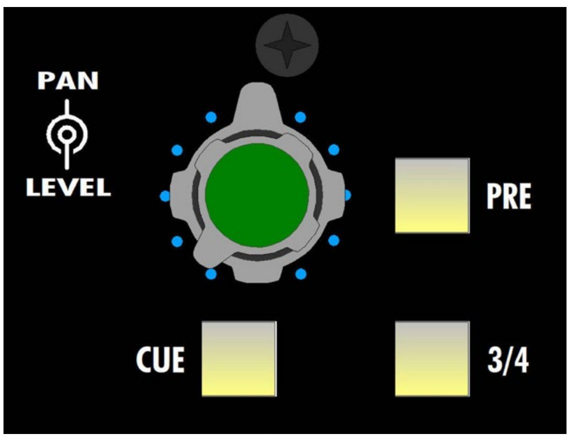
3.5.2 Stereo Auxiliary Sends 3/4

AUX 3/4 is a stereo auxiliary send that feeds aux buses 3 and 4 by default, but can be routed to feed the stereo cue bus instead. Aux sends 3/4 are commonly used for headphone sends while recording and effects sends while mixing, but can be used for other purposes.

In addition to auxiliary buses 1-4, "**THE BOX 8**" is equipped with a separate stereo cue bus, cue master, and balanced cue 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 and fed to the headphones. When mixing, AUX 3/4 can be routed to the cue bus instead of aux buses 3/4 to provide an alternate stereo effects send.

Engaging **CUE** will route AUX 3/4 to the stereo cue bus to allow a fully featured headphone feed to be created when recording. AUX 3/4 can be routed to aux buses 3/4 or the stereo cue bus for use as effects sends when mixing.

A common ON/off switch (**3/4**) is used to engage the stereo send. AUX 3/4 is fed post-fader by default, but can be fed the pre-fader signal by engaging **PRE**.



AUX 3/4: Level and pan controls for stereo aux send ¾-CUE

- Knob = level
- Ring = pan

PRE (Pre-fader): 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 switch for aux 3/4

- Illuminates when engaged

CUE: Routes aux send 3/4 to the stereo cue bus when engaged

- The feed to auxiliary buses 3 and 4 is defeated when engaged
- Illuminates when engaged

3.6 Summing Input Output Routing

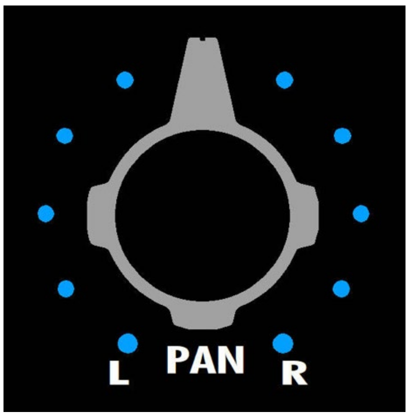
All summing inputs can be routed to the stereo program bus, post-pan-pot.

3.6.1 Stereo Program Bus and Pan-pot

Summing inputs can be routed to the stereo program bus via the pan-pot when mixing, providing sixteen (16) multitrack return channels.

PGM (Program): Assigns the summing input to the program bus when engaged.

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



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

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

3.6.2 Solo Safe

"**THE BOX 8**" is equipped with a "Solo-In-Place" function (SIP) that mutes input channels and summing inputs when **SOLO** is engaged on another channel or summing input. Engaging solo **SAFE** prevents that channel from being muted when a solo is engages in SIP mode.

SAFE (Solo Safe): Activates the solo-safe function on that summing input

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

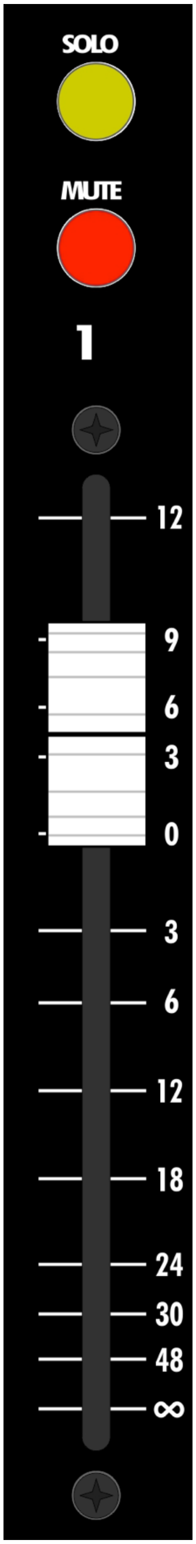
Refer to section 10.0 Solo Master for more information regarding the solo system.

3.7 Summing Input Faders

The fader section with white fader caps contains the summing input faders, **MUTE**, and **SOLO** switches for summing inputs 1-16. These faders are the primary output level controls for the summing inputs.

The summing input faders can be routed to feed:

- **Stereo Program Bus**: When **PGM** is engaged



SOLO: Activates the current solo function on that summing input

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

MUTE: Cuts the output of the summing input

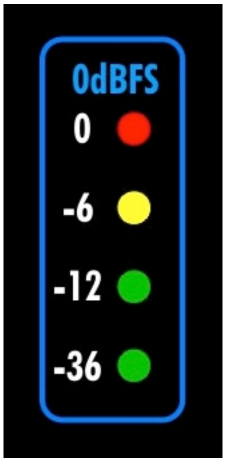
- Located post-fader in the signal flow
- Illuminates in red when engaged
- Activates on release

SUMMING INPUT FADER: Primary output level control for the summing input

- 100mm full-size audio-taper fader
- -∞dB to +12dB range
- 0dB is unity gain
- White fader cap
- 0dB unity gain fader bypass when **0dB** is engaged

3.8 Meter

Summing inputs are equipped with a 4-segment LED full-scale meter (0dBFS). The meter displays the post-mute output from the summing input.



0dBFS: 4-segment LED full-scale meter

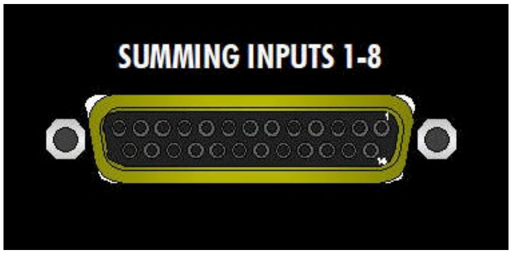
- Indicates post-mute, summing input level

3.9 Summing Input Rear Panel Connections

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

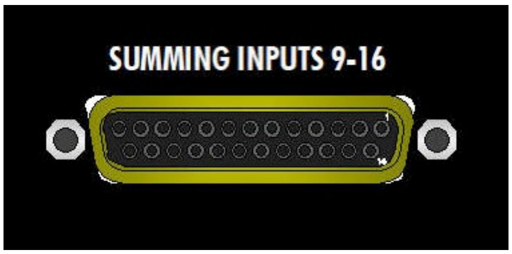
Connections include:

- Line inputs (1-8 and 9-16)
- Insert send (1-16)
- Insert return (1-16)



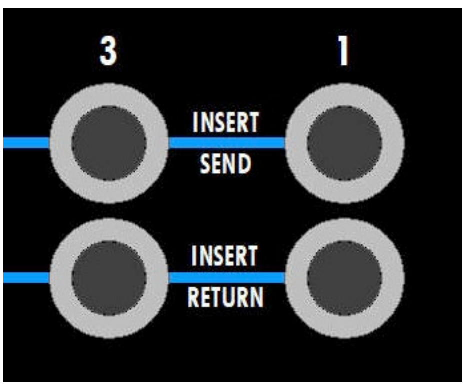
<u>SUMMING INPUTS 1-8</u>: Line input to summing inputs 1-8

- Line-level, balanced, low-impedance
- DB25 connect (standard pin-out)
    - 1-8=summing inputs 1-8



<u>SUMMING INPUTS 9-16</u>: Line input to summing inputs 9-16

- Line-level, balanced, low-impedance
- DB25 connect (standard pin-out)
    - 1-8=summing inputs 9-16



<u>INSERT SEND</u>: Insert output

- Line-level, balanced, low-impedance
- ¼" tip-ring-sleeve

<u>INSERT RETURN</u>: Insert input

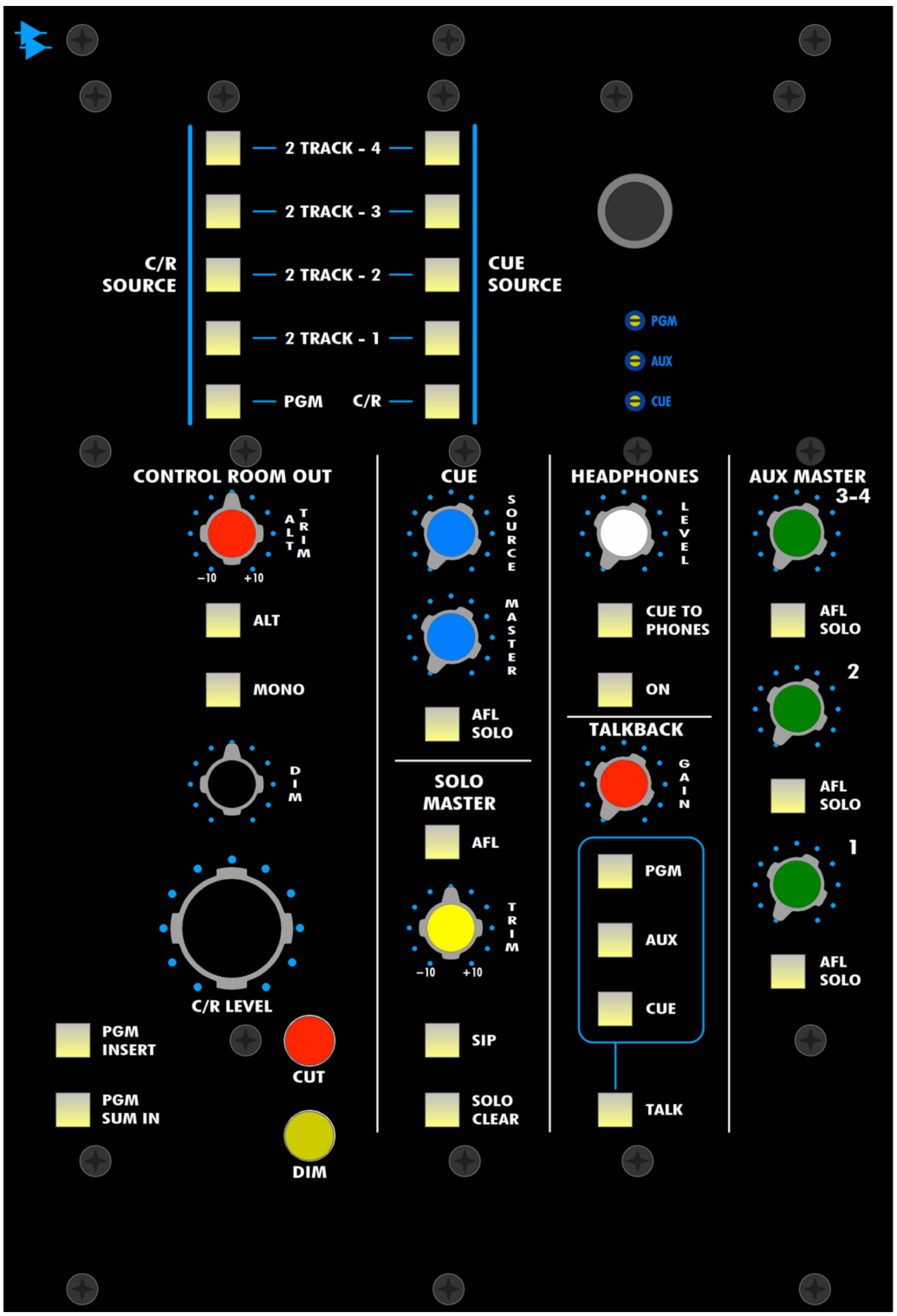
- Active when **INS** is engaged
- Line-level, balanced, low-impedance
- ¼" tip-ring-sleeve switching jack

4.0 Master Section

The master section of "**THE BOX 8**" 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**: External sum input, program insert, and master fader
- **AUX MASTER**: Mono aux masters 1 & 2 and stereo aux 3/4
- **CUE**: Stereo cue system master
- **SOLO MASTER**: Solo system master
- **CONTROL ROOM OUT**: Control room monitor controls
- **TALKBACK**: Talkback controls
- **HEADPHONES**: Headphone amplifier and controls



5.0 Stereo Program Master

"**THE BOX 8**" stereo program bus employs the same summing technology found in larger API consoles. This high-quality mix bus is a key feature of "**THE BOX 8**" that offers depth, punch, and warmth that's only possible with high-quality analog summing.

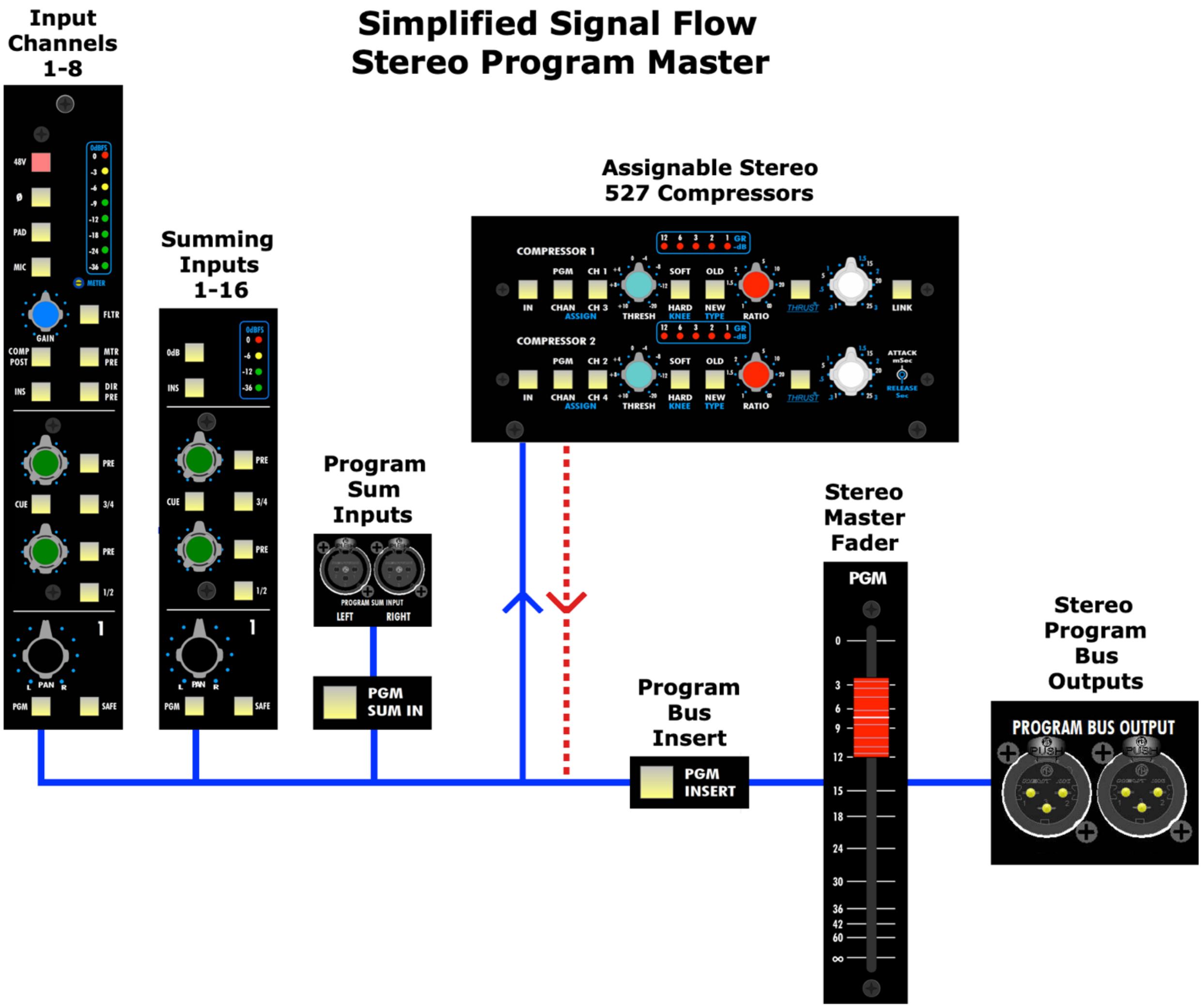
The stereo program master is equipped with a comprehensive set of features that support a wide range of mixing applications. These features include:

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

5.1 Stereo Program Master Simplified Signal Flow

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

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



5.2 Stereo Program Master Controls

The controls for the stereo program master are as follows:

- **PGM SUM IN (Program Sum Input)**: Stereo external summing input to the program bus
- **PGM INSERT (Program Bus Insert)**: Balanced stereo insert for the program bus
- **PGM (Program Master Fader)**: Main stereo mix output level control

5.2.1 Program Sum Input

"**THE BOX 8**" 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 8**." These inputs sum with the program bus audio at unity gain.

PGM SUM IN (Program Sum Input): Adds the signal from the PROGRAM SUM INPUT on the rear panel to the program bus at unity gain

- Located pre-compressor and insert
- Illuminates when engaged

5.2.2 Program Bus Insert

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

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

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

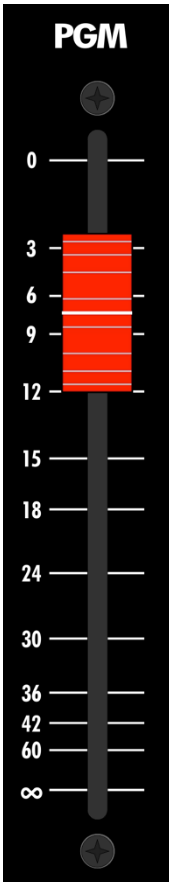
The program insert sends are fed pre-fader immediately after the PROGRAM SUM INPUTS and compressors and are always active.

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.

5.2.3 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.

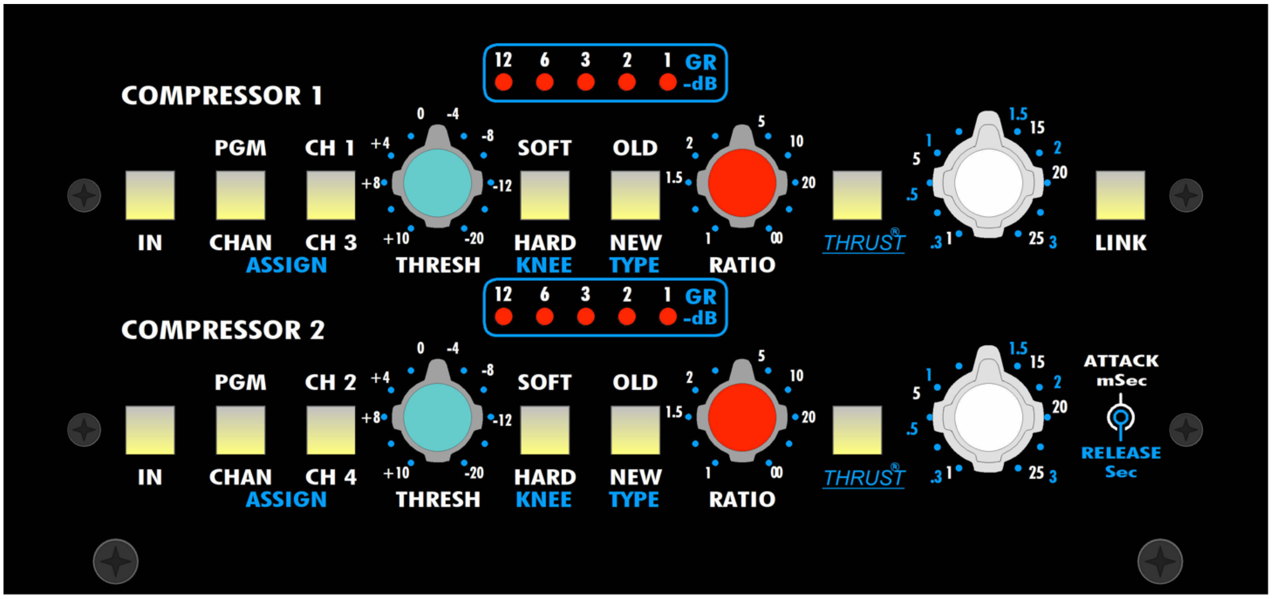


PGM (Program Master Fader): Primary stereo mix output level control

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

5.3 Program Bus Compression

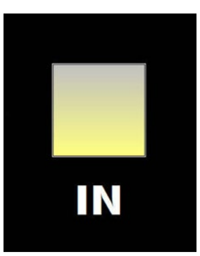
"**THE BOX 8**" includes two API 527 compressor/limiters in the center section of the console. These compressors are assigned to the program bus by default for mixing, but can be routed to the input channels while recording.



To apply the compressors to the mix, 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 in the program bus signal flow.

PGM-CHAN (Program-Channel): Assigns the compressor to the selected input channel when engaged

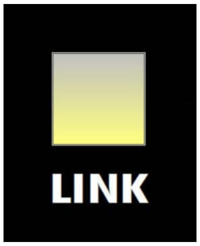
- Compressors are assigned to the program bus by default
- If needed, assignment to the program bus is accomplished by disengaging the **PGM-CHAN** switches on the compressor
- Illuminates when engaged



IN: Places the compressor in the signal path when engaged

- Compressors are bypassed unless the **IN** switch is engaged
- Illuminates when engaged

Stereo compression is possible when both compressors are assigned to either the program bus or a pair of input channels and the **IN** and **LINK** switches are engaged.



<u>LINK</u>: Links the compressors when engaged

- Sums the gain reduction control voltages when engaged
- Illuminates when engaged

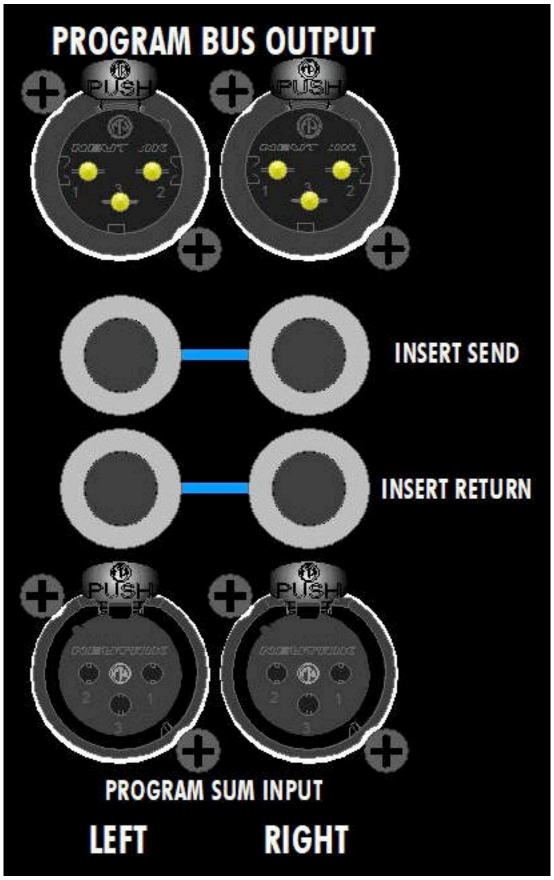
*NOTE: Both compressors must be **IN**, assigned to the program bus or a pair of input channels, with **LINK** engaged to apply proper stereo mix compression. When **LINK** is engaged the gain reduction control voltage is summed. To assure that gain reduction is keyed equally by both left and right channels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

Refer to section 6.0 Compressors 1 and 2 for more information regarding the compressors.

5.4 Program Master 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 OUTPUT**: Program bus main Left and Right outputs
- **INSERT SEND**: Stereo program bus insert sends
- **INSERT RETURN**: Stereo program bus insert returns
- **PROGRAM SUM INPUT**: Stereo external summing inputs to the program bus



PROGRAM BUS OUTPUT: Main stereo mix outputs

- Located post-master fader
- Line-level, balanced, low-impedance
- 3-pin male XLRs

INSERT SEND: Stereo program bus insert outputs

- Line-level, balanced, low-impedance
- ¼" tip-ring-sleeve

INSERT RETURN: Stereo insert inputs

- Active when **INS** is engaged
- Line-level, balanced, low-impedance
- ¼" tip-ring-sleeve switching jacks

PROGRAM SUM INPUT: Stereo inputs that sum with the program bus

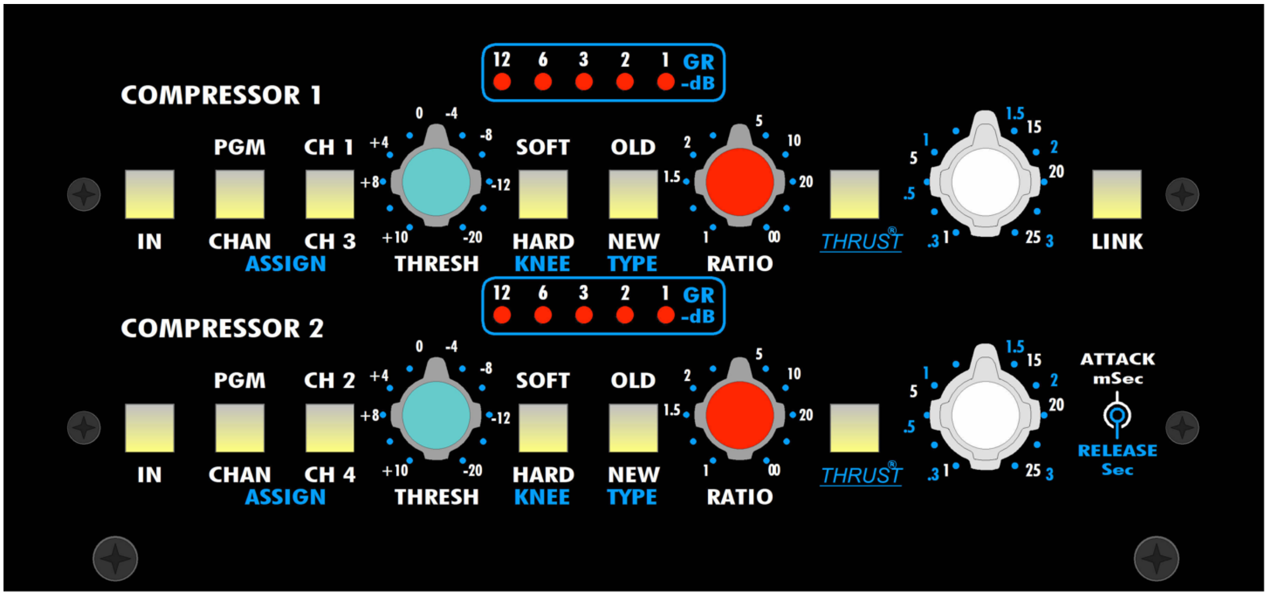
- Active when **PGM SUM IN** is engaged
- Located pre-insert and compressor
- Line-level, balanced, low-impedance
- 3-pin female XLRs

NOTE: *The PROGRAM SUM INPUTs are useful for bringing a stereo mix of external sources into to "**THE BOX 8**" for inclusion in the program bus. For example, two "**THE BOX 8**" consoles can be linked together by patching the PROGRAM BUS OUTPUTs of the slave console to the PROGRAM SUM INPUTs on the master console and engaging the **PGM SUM IN** switch on the master.*

NOTE: *The LEFT & RIGHT PROGRAM BUS OUTPUTs are paralleled with outputs 7 & 8 on the AUX OUTPUTS / CUE OUTPUTS DB25 connector.*

6.0 Compressors 1 and 2

"**THE BOX 8**" 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 its 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 8**" 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

Compressors 1 and 2 in "**THE BOX 8**" 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 side-chain input on the TCS. The compressors in "**THE BOX 8**" do not have a side-chain and the gain reduction meter has fewer LED segments.

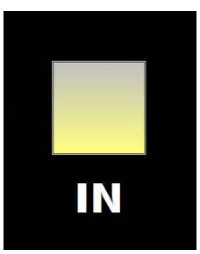
The compressors in "**THE BOX 8**" can be linked together allowing for stereo compression of the program bus when mixing or a pair of input channels when recording.

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

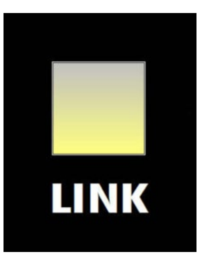
6.1 Compressor Program Bus Routing

To apply the compressors to the mix, 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 in the program bus signal flow.

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.



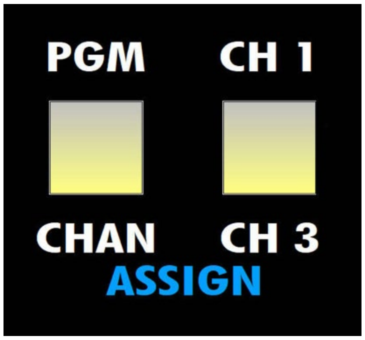
Stereo mix compression is possible when both compressors are assigned to the program bus and the **IN** and **LINK** switches are engaged. This switch illuminates when engaged.

*NOTE: Both compressors must be **IN**, assigned to the program bus or a pair of input channels, 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.*

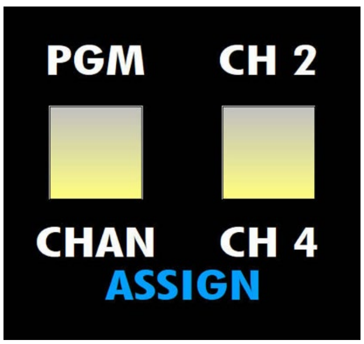
6.2 Compressor Input Channel Routing

The compressors can be assigned to the input channels for use during recording or mixing. The compressors can be assigned to input channels 1-4 only. Compressor 1 can only be assigned to input channels 1 or 3. Likewise, Compressor 2 can only be assigned to channels 2 or 4.

Compressor assignments to input channels are accomplished using the two **ASSIGN** switches on the compressors.



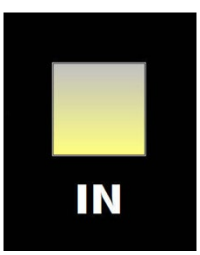
Engaging the **PGM**-**CHAN** switch on Compressor 1 will move Compressor 1 from the left side of the program bus to input channel 1 by default. Engaging **CH 3** will assign Compressor 1 to input channel 3. These switches illuminate when engaged.



Engaging the **PGM**-**CHAN** switch on Compressor 2 will move Compressor 2 from the right side of the program bus to input channel 2 by default. Engaging **CH 4** will assign Compressor 2 to input channel 4. These switches illuminate when engaged.

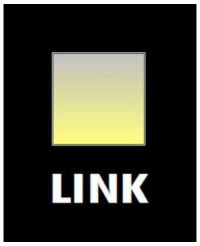
When assigned to an input channel, the compressor will be located after the preamp and before the 500 slot and insert by default. It can be moved to after the 500 slot and insert by engaging the **COMP POST** switch.

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.



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 or mixing with input channels is possible when both compressors are assigned to input channels and the **LINK** switch is engaged.

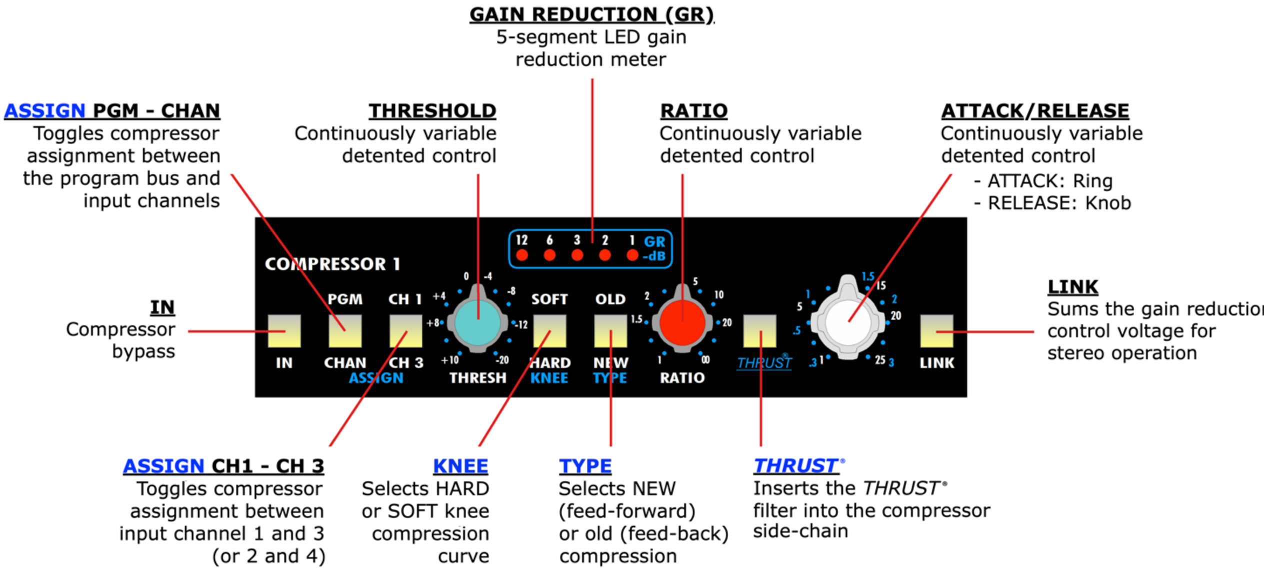


Stereo track compression is possible when both compressors are assigned to input channels and the **IN** and **LINK** switches are engaged. This switch illuminates when engaged.

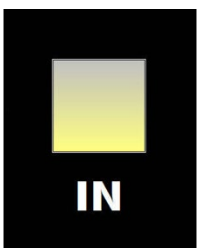
*NOTE: Both compressors must be **IN**, assigned to the program bus or a pair of input channels, with **LINK** engaged to apply proper stereo mix compression. When **LINK** is engaged the gain reduction control voltage is summed. To assure that gain reduction is keyed equally by both left and right channels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

6.3 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

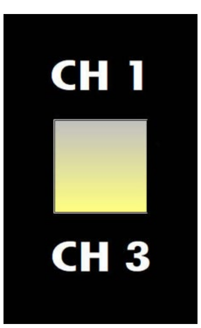


IN: Places the compressor in the signal path when engaged

- Illuminates when engaged

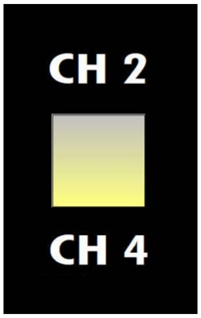
ASSIGN PGM-CHAN (Program - Channel): Toggles compressor routing between the program bus and input channels

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



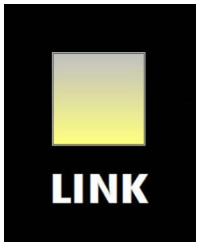
ASSIGN CH 1 - CH 3 (Channel 1 – Channel 3): Toggles Compressor 1 routing between input channels 1 and 3

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



ASSIGN 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



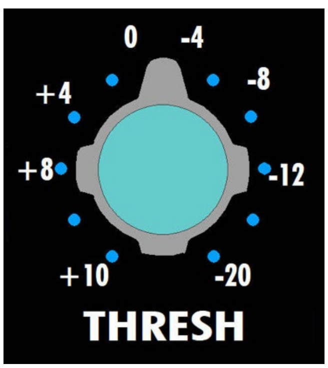
LINK: Activates DC link between Compressors 1 and 2

- 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 **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.*

6.4 Compressor Parameters

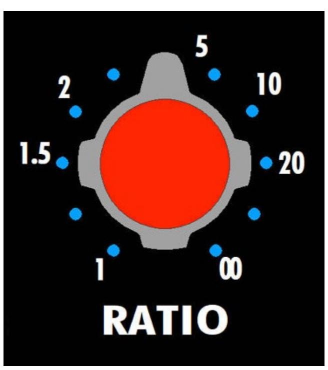
6.4.1 Threshold



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

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

6.4.2 Ratio



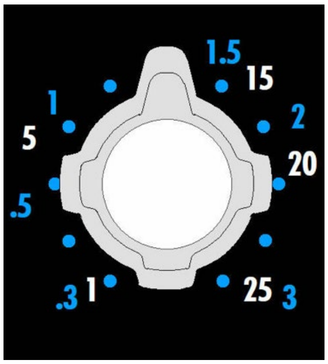
RATIO: Sets the compression ratio of input vs. output levels for signals that fall above the set threshold

- 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.

6.4.3 Attack and Release

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.



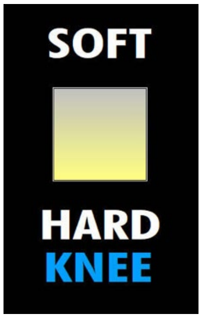
ATTACK (Ring): Sets the time it takes the compressor to react when the input level exceeds the set threshold

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

RELEASE (Knob): 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

6.4.4 Knee

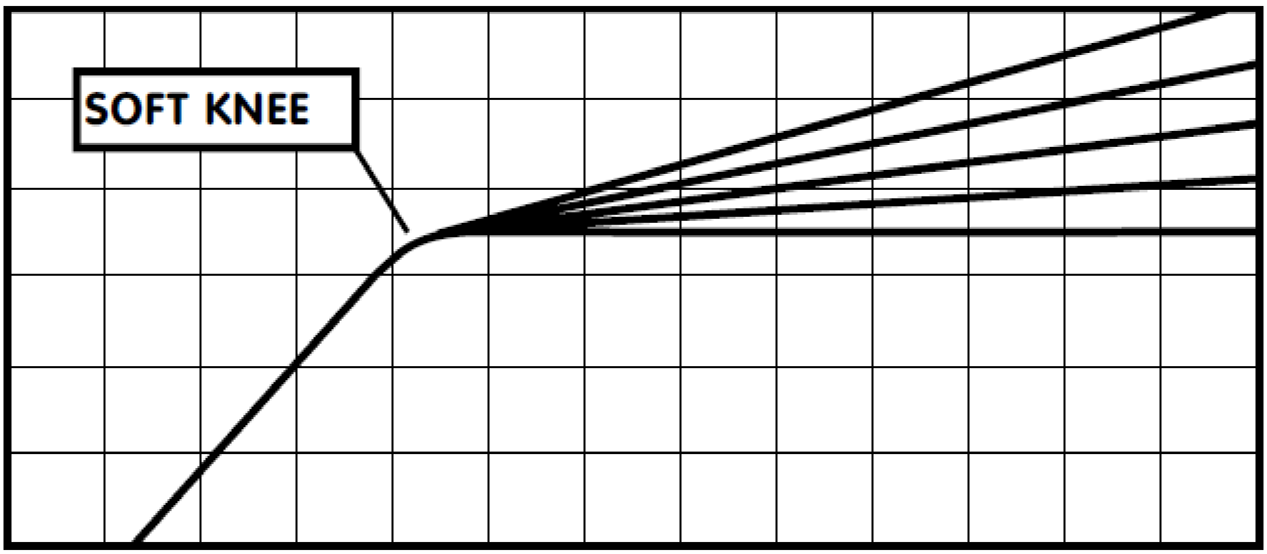


KNEE (SOFT-HARD): Toggles between soft and hard knee compression

- 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

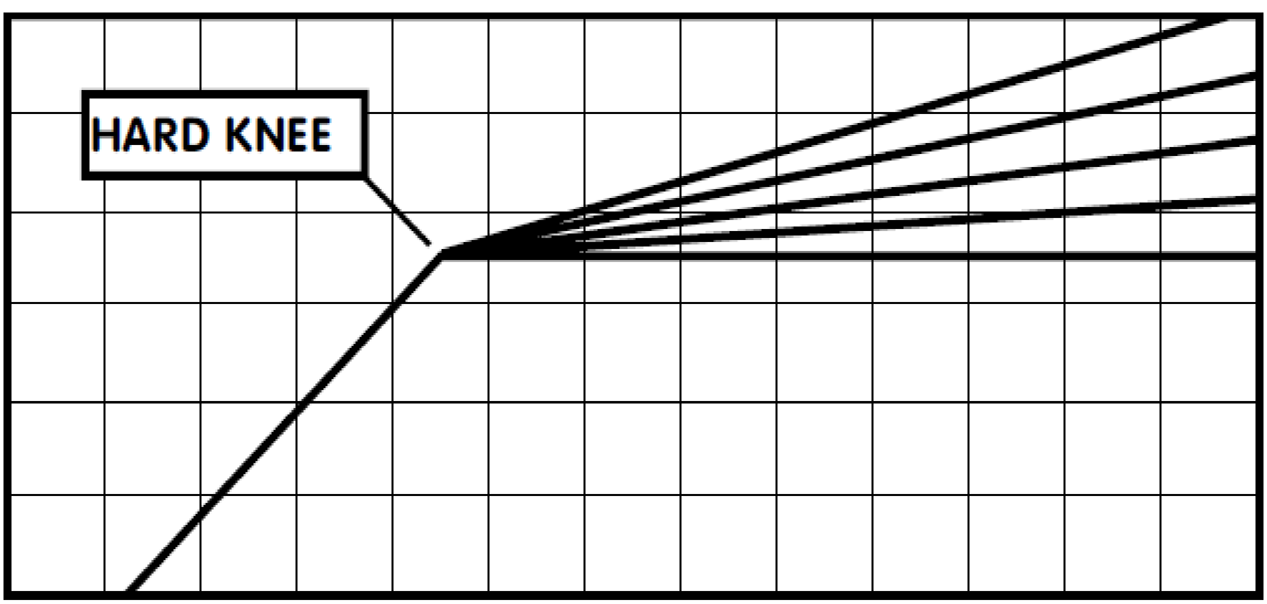
SOFT: Rounded response curve

- 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



6.4.5 Compressor Circuit Topologies

The 527 compressors can be set to operate in two types 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

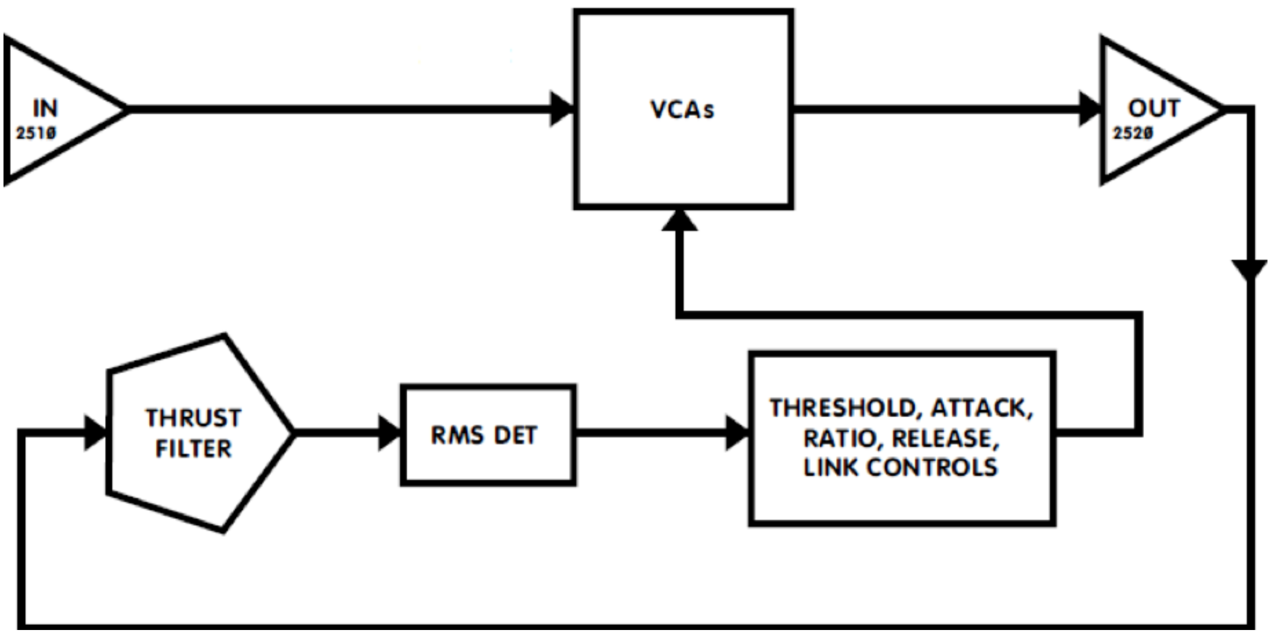
TYPE (OLD-NEW): Toggles between feed-back and feed-forward compressor topologies

- 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.

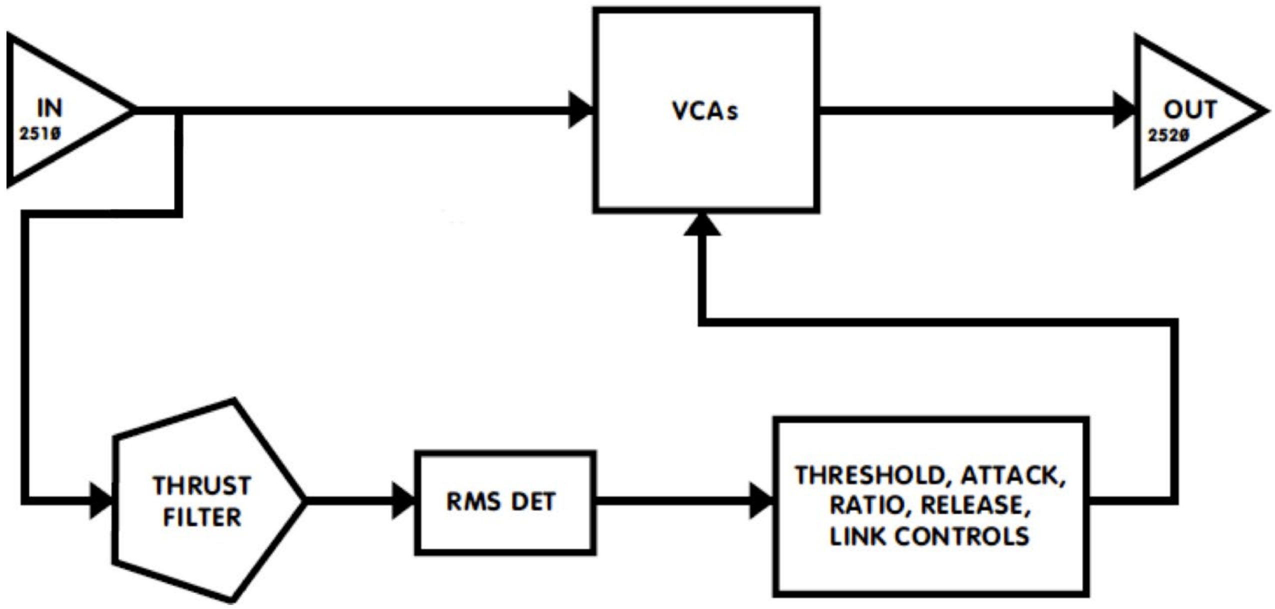
"OLD" or FEED BACK type COMPRESSION



NEW: Feed-Forward Compression

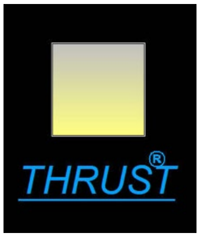
In a feed-forward compressor, the RMS detector normally gets its signal from a split of the input signal. With this method, the RMS detector sends a signal to the VCA that is an exact ratio of the desired compression set by the RATIO control. This is how many new VCA based compressors work. This can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

"NEW" or FEED FORWARD type COMPRESSION



6.4.6 *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!



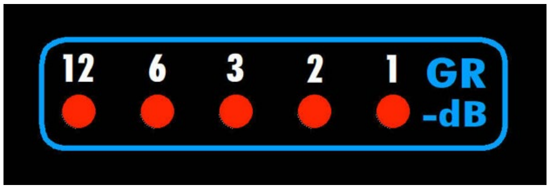
THRUST®: Inserts the ***THRUST®*** filter before the RMS detector

- 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. Without ***THRUST®***, the detector will process the signal into a DC control voltage based upon 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.

6.4.7 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

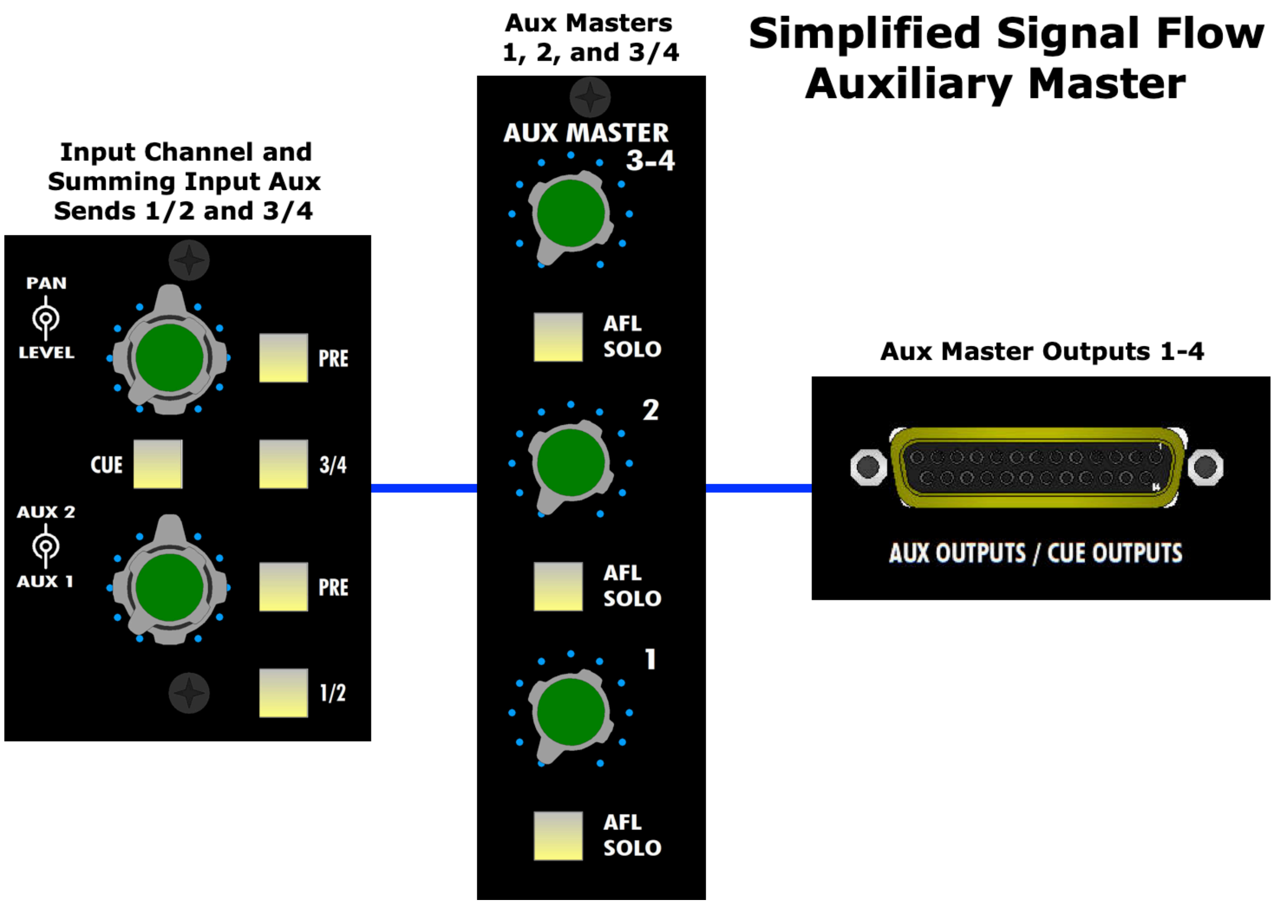
7.0 Auxiliary Masters

"**THE BOX 8**" 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.

7.1 Auxiliary Master Simplified Signal Flow

The auxiliary signal flow is as follows:

Channel sends ⇒ aux bus ⇒ aux master ⇒ output



7.2 Auxiliary Master Controls

The controls for the auxiliary masters are as follows:

- **AUX MASTER 3-4**: Stereo output level and AFL solo
- **AUX MASTER 2**: Mono output level and AFL solo
- **AUX MASTER 1**: Mono output level and AFL solo

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

- Stereo aux master control potentiometer
- Feeds the AUX OUTPUTS / CUE OUTPUTS DB25 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 DB25 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 DB25 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.

7.2.1 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.

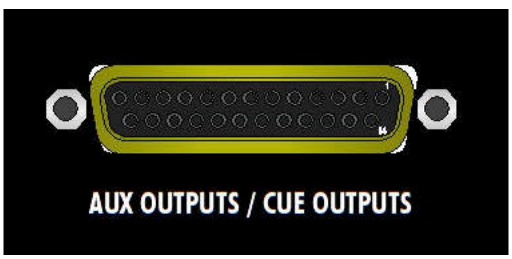
When PFL is the current solo mode, mono aux masters 1 & 2 feed the mono PFL solo bus in mono. The stereo aux master 3-4 outputs are summed together and fed to the PFL solo bus in mono. Even though the console is in PFL, the PFL solo bus is fed immediately after the AUX MASTER potentiometers.

*NOTE: When solo-in-place (SIP) is the current solo mode, engaging an aux master **AFL SOLO** 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

7.3 Auxiliary Master Rear Panel Connections



AUX OUTPUTS / CUE OUTPUTS: Aux master outputs 1-4

- 1-6 are fed from the aux and cue masters
- 7-8 are fed from stereo program masters
- Balanced, low-impedance, line-level
- DB25 connector:
  - Standard pin-out: 1-4=aux outputs 1-4, 5-6=cue outputs L-R, 7-8=PGM outputs L-R

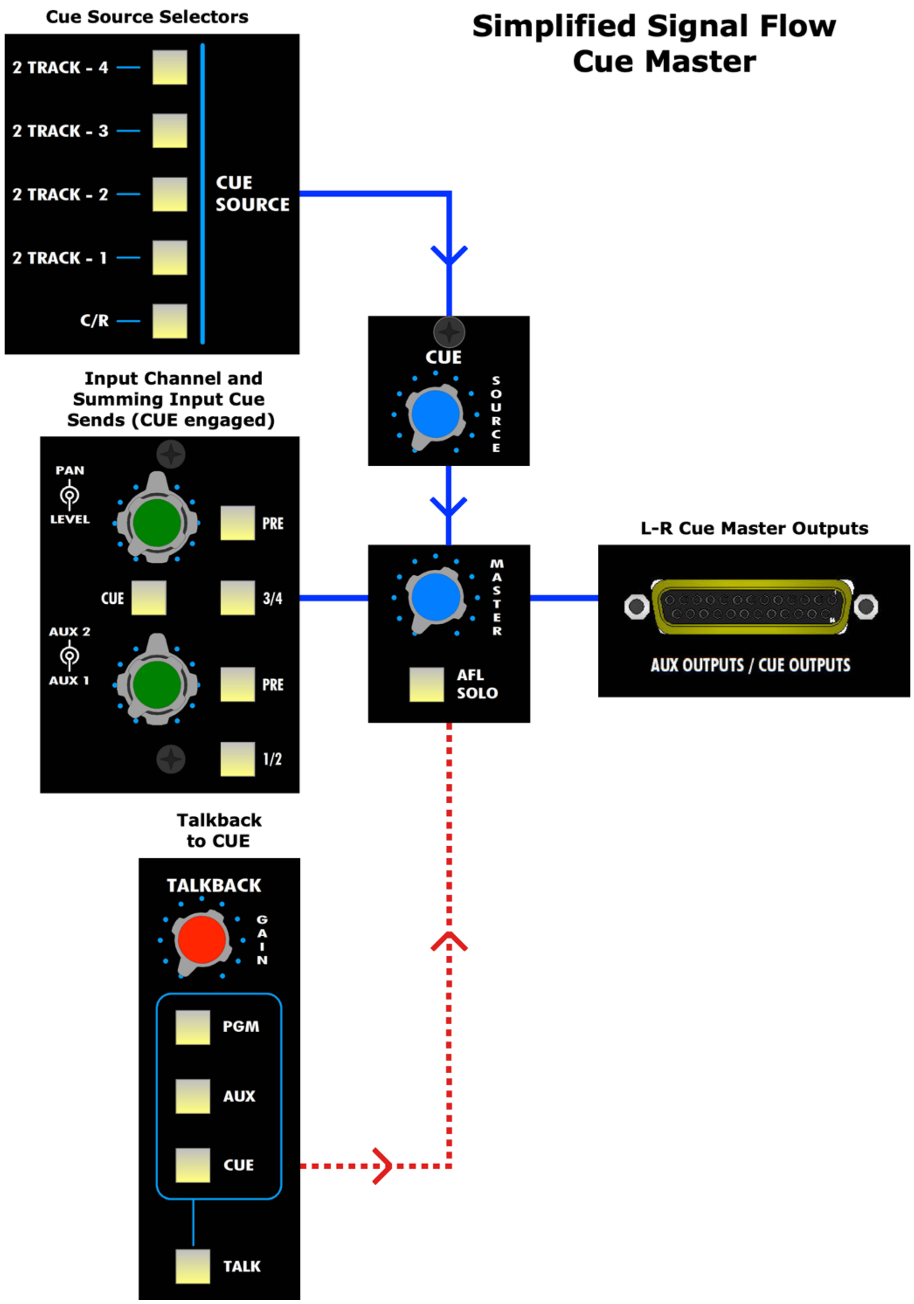
8.0 Cue Master

In addition to the four auxiliary buses, "**THE BOX 8**" 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.

8.1 Cue Master Simplified Signal Flow

The cue system signal flow is as follows:

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



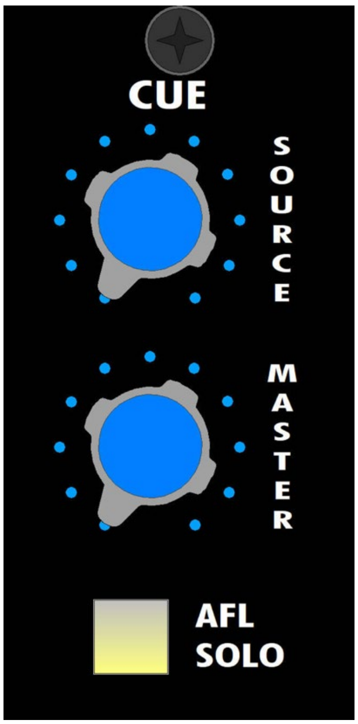
The stereo cue bus is the primary source for the cue master. Additional stereo sources can be added using the **CUE SOURCE** selectors and level control. Sources are selected using the **CUE SOURCE** switches. The level of the selected sources feeding the **CUE MASTER** is controlled by the **CUE SOURCE** 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.

8.2 Cue Master Controls

The controls for the cue master are as follows:

- **SOURCE**: Level of the selected stereo **CUE SOURCE** selectors
- **MASTER**: Cue output level and AFL solo



CUE SOURCE: Level control for the currently selected **CUE SOURCE** selection

- Stereo control potentiometer
- Feeds the **CUE MASTER** pot

CUE MASTER: Overall cue output level control

- Stereo control potentiometer
- Feeds the AUX OUTPUTS / CUE OUTPUTS DB25 connector on the rear panel

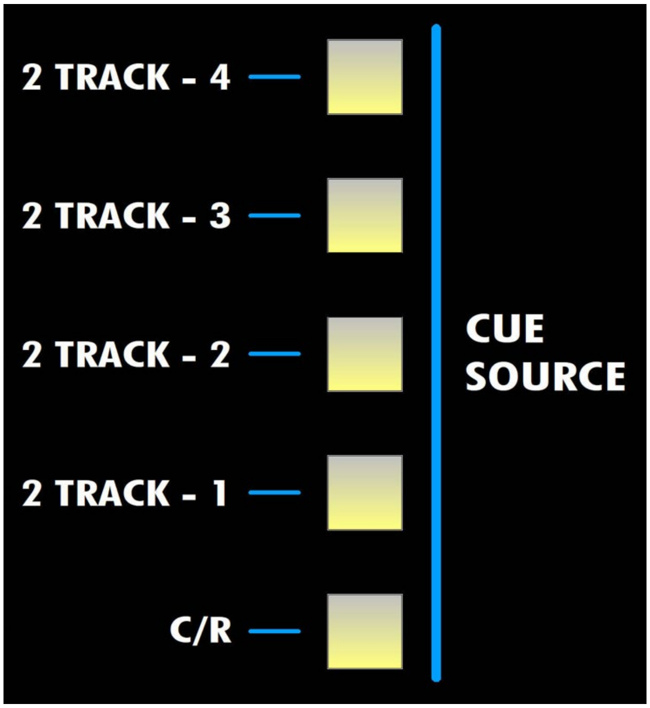
CUE AFL SOLO: Activates the AFL or PFL solo mode*

* 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.

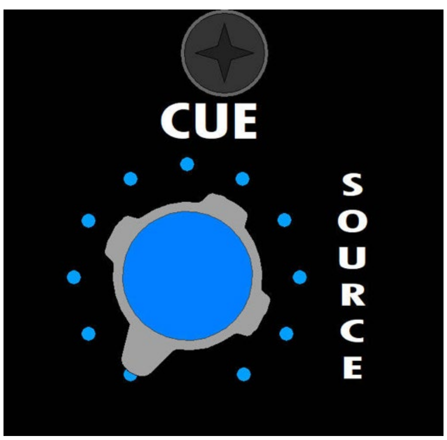
8.2.1 Cue Source Selectors

In addition the stereo cue bus, the **C/R SOURCE** selections, the output of the stereo program master 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(s) to the **CUE SOURCE** level control in the cue master

- **C/R (Control Room)**: Output of the **C/R SOURCE** selectors*
- **2 TRACK 1-4**: External stereo sources 1-4
- Selectors are additive
- Illuminate when engaged



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.

* When **C/R** 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 **PGM** is selected as the **C/R SOURCE** and **C/R** is selected as the **CUE SOURCE**. Refer to section 11.2 Control Room Source Selectors for additional information.

8.2.2 Cue Master Solo

The cue master can be soloed 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.

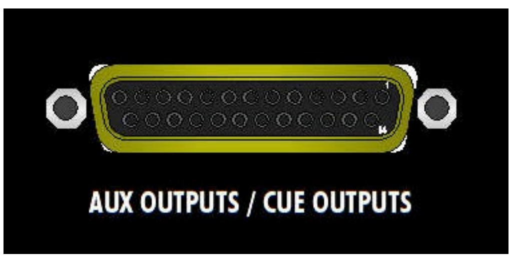
When PFL is the current solo mode, the stereo cue master output is summed together and fed to the PFL solo bus in mono. Even though the console in PFL, the PFL solo bus is fed immediately after the **CUE MASTER** potentiometer.

*NOTE: When solo-in-place (SIP) is the current solo mode, engaging the cue master **AFL SOLO** while in SIP mode will not active SIP mutes on channels.*

<u>AFL SOLO</u>: 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 cue masters)
- Can be cleared by pressing the **SOLO CLEAR** button
- Illuminates when engaged

8.3 Cue Master Rear Panel Connections



<u>AUX OUTPUTS / CUE OUTPUTS</u>: Left-Right cue master outputs

- 1-6 are fed from the aux and cue masters
- 7-8 are fed from stereo program masters
- Balanced, low-impedance, line-level
- DB25 connector
  - Standard pin-out: 1-4=aux outputs 1-4, 5-6=cue outputs L-R, 7-8=PGM outputs L-R

9.0 Talkback

"**THE BOX 8**" is equipped with a complete talkback system for use when recording. The talkback systems includes:

- Internal electret microphone
- Talkback mic preamp gain
- Trim controls
- Momentary talk button
- Preset routing

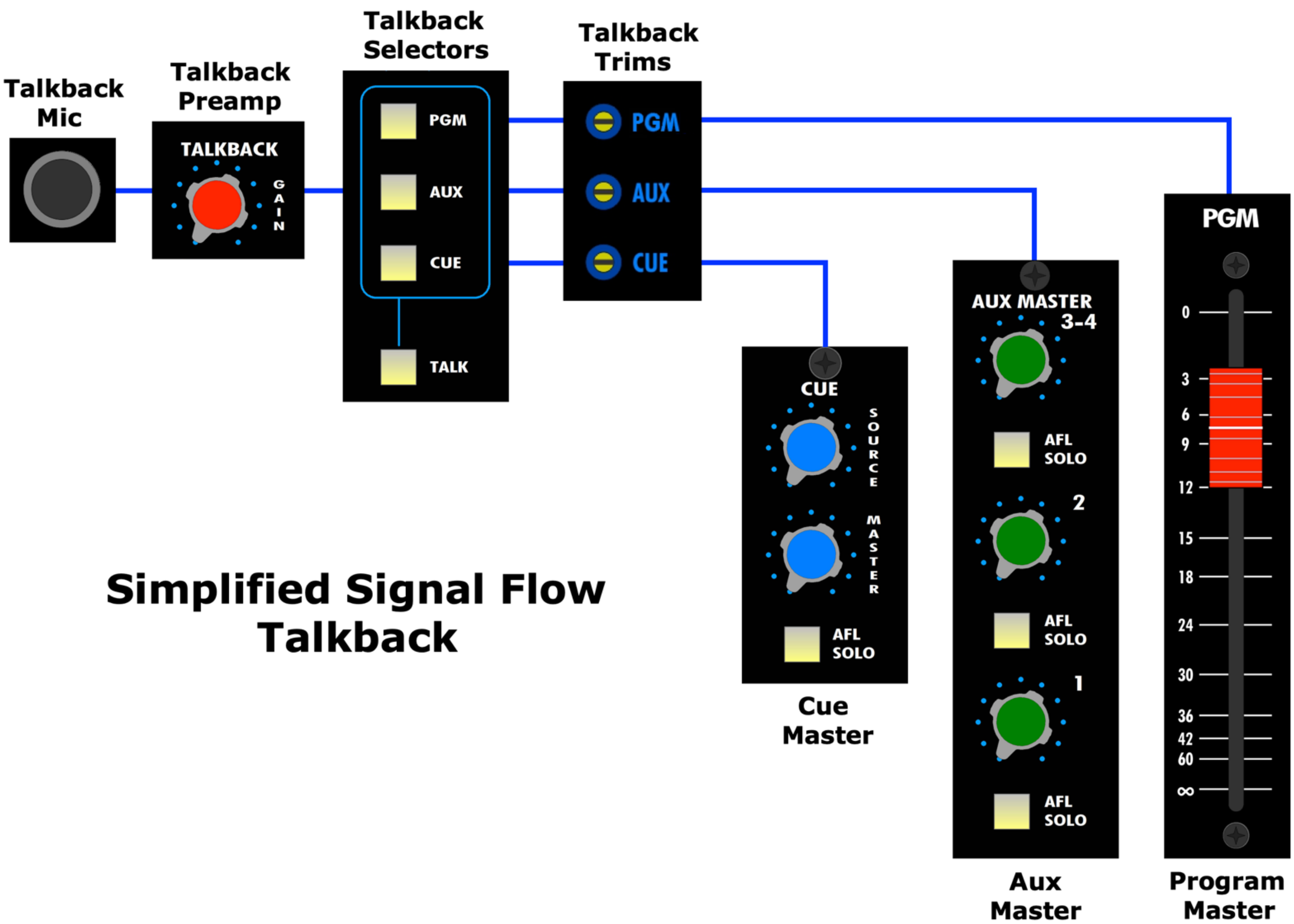
Talkback can be routed to the following:

- **PGM**: Program bus masters
- **AUX**: Auxiliary masters
- **CUE**: Cue master

9.1 Talkback Simplified Signal Flow

The talkback signal flow is as follows:

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



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

9.2 Talkback Routing and Controls

Talkback components and controls are as follows:

- **Microphone**: Built-in talkback mic
- **GAIN**: Talkback mic preamp
- **Trims**: **PGM**, **AUX**, and **CUE**
- **Routing Selectors**: **PGM**, **AUX**, and **CUE**
- **TALK**: Push-to-talk talkback button

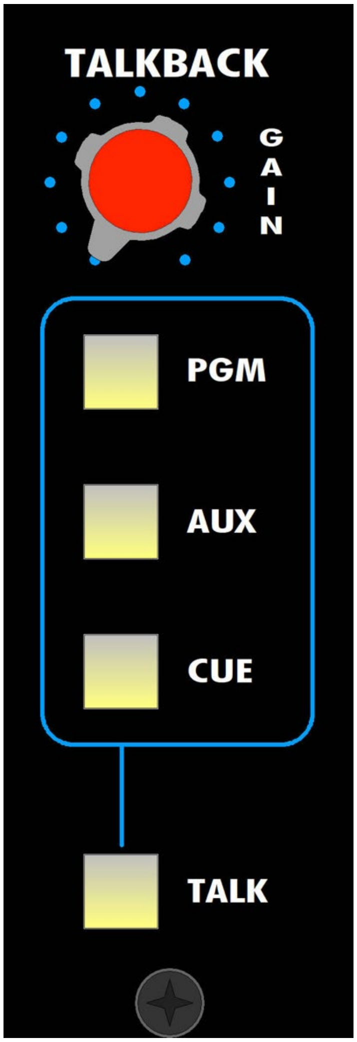
TALKBACK MICROPHONE: Internal talkback microphone

- Electret condenser mic

PGM: Talkback to program bus level trim-pot

AUX: Talkback to auxiliary master level trim-pot

CUE: Talkback to cue master level trim-pot



GAIN: Talkback level control (preamp)

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

AUX: Enables talkback routing to the auxiliary masters
- 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

10.0 Solo System

"**THE BOX 8**" 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:

- Input channel, summing inputs, 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.

10.1 Solo Modes

"**THE BOX 8**" has three solo modes as explained below:

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

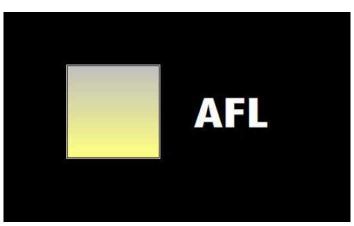
10.1.1 PFL (Pre-Fader-Listen)

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.

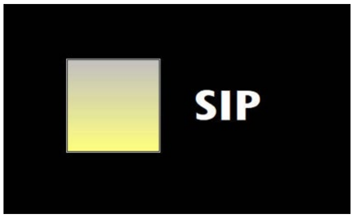
10.1.2 AFL (After-Fader-Listen)



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 AFL solo level is controlled with the solo **TRIM** pot.

10.1.3 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 only 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. Because all non-soloed channels are muted when a **SOLO** is engaged, this mode is considered to be "destructive." The AFL and PFL solo buses are not 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.*

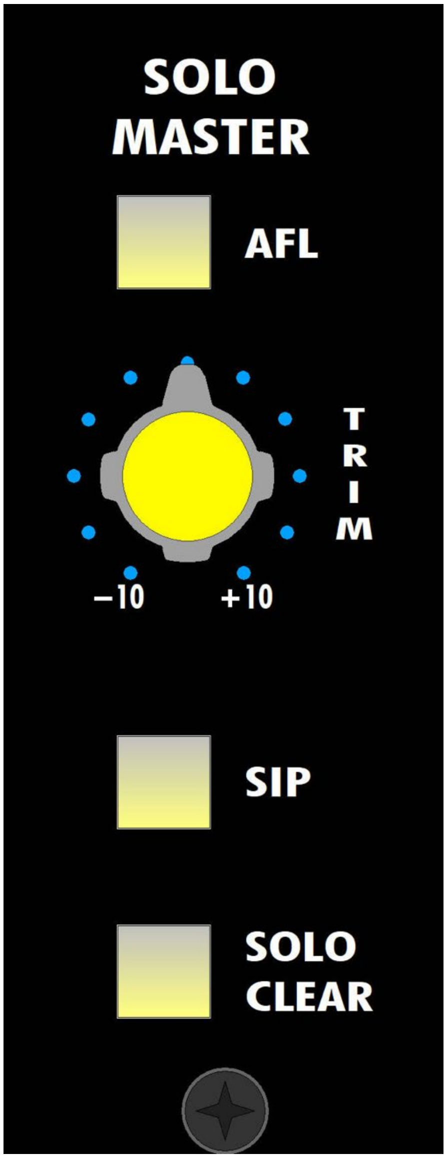
10.1.4 Solo Safe

Engaging the **SAFE** switch on an 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

The controls for the solo master are as follows:

- **AFL**: After-fader-listen solo mode
- **TRIM**: AFL and PLF solo bus level
- **SIP**: Solo-In-Place solo mode
- **SOLO CLEAR**: Clears all engaged **SOLO** switches



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

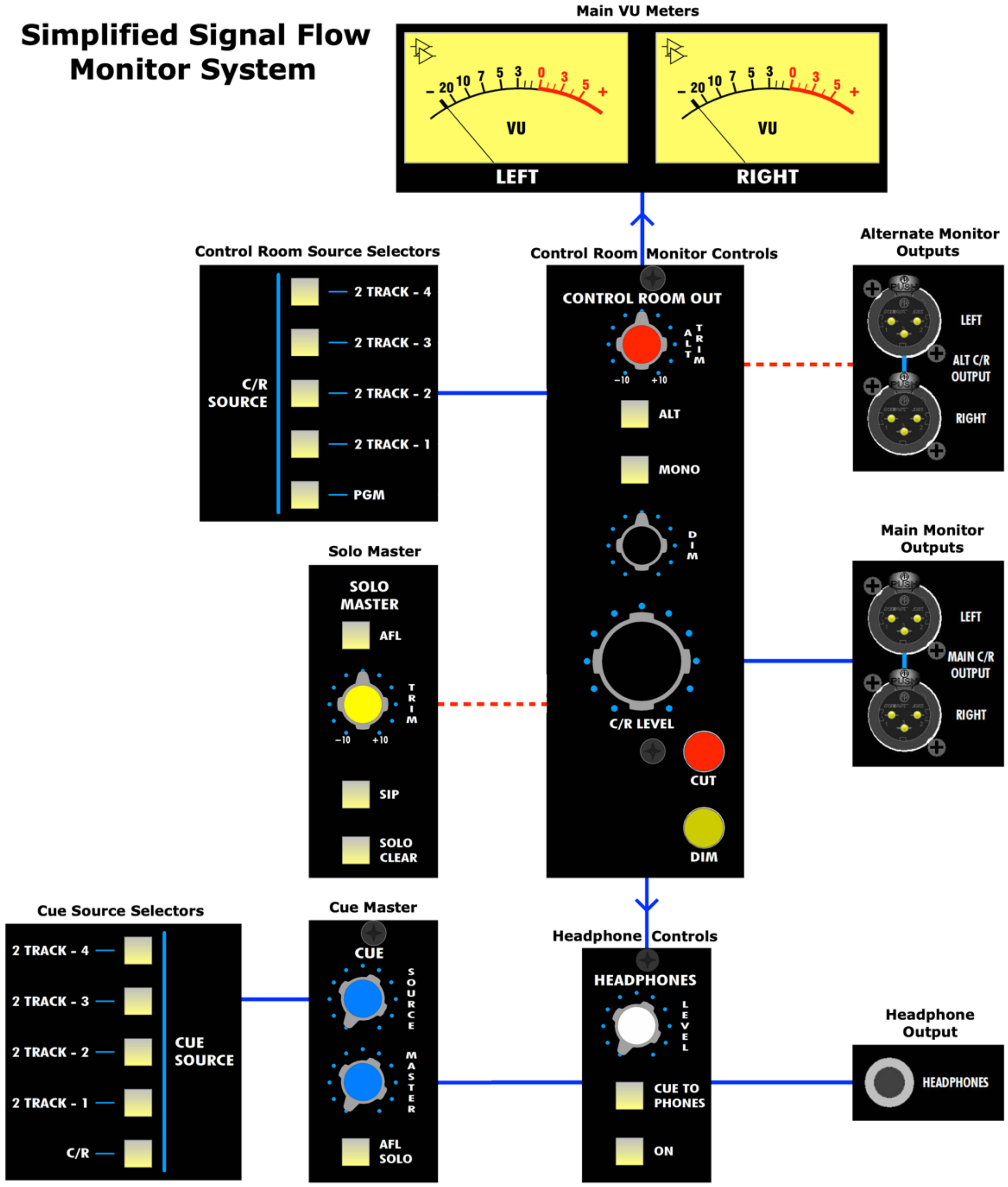
11.0 Monitoring Systems

"**THE BOX 8**" 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 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.

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

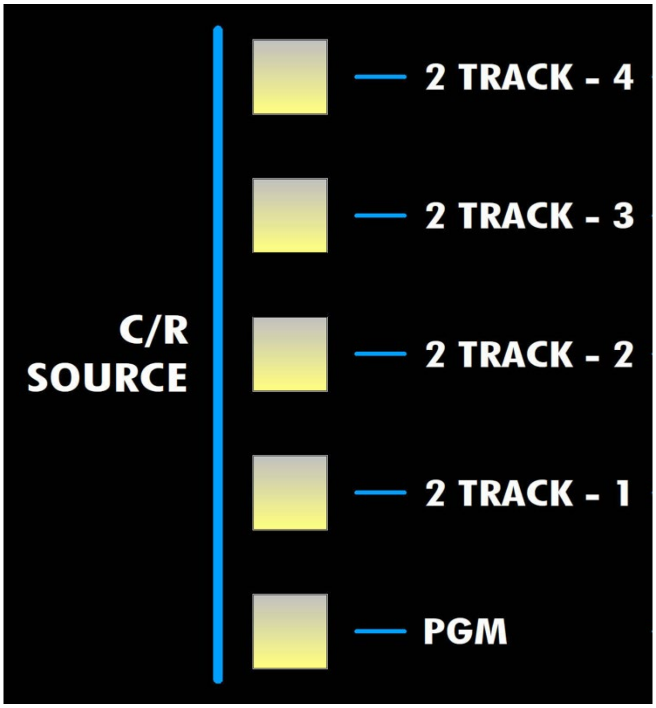
In total, the monitoring systems has the following components:
- Control room source selectors
- Control room monitor controls and outputs
- Headphone amplifier and outputs

11.1 Monitor System Simplified Signal Flow



11.2 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 monitor 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>PGM (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.

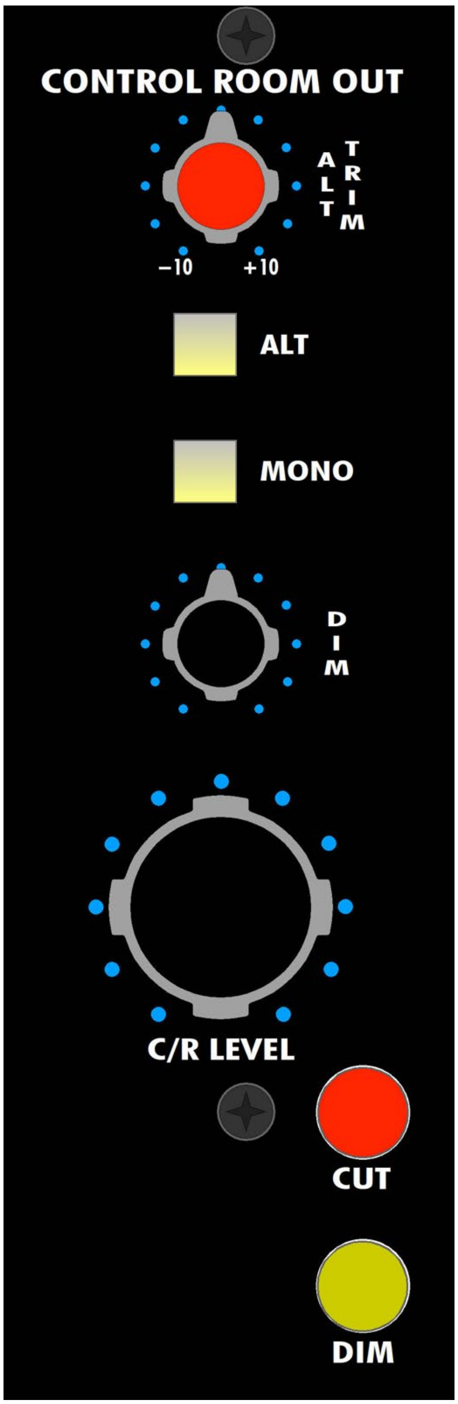
Any time **PGM** (Program) is engaged, the program master output is the only **C/R SOURCE**. **2 TRACK** sources are blocked even if their switches are engaged.

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 **PGM** will override any **2 TRACK** source selections. Illuminated switches indicate the actual active source.

11.3 Control Room Monitor Controls

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.

The controls for the control room monitors are as follows:
- **C/R LEVEL**: Control room monitor level
- **CUT**: Control room monitor mute
- **DIM**: Control room monitor dim (attenuation) function
- **MONO**: Sums the control room source to mono
- **ALT**: Alternate control room monitor system controls



ALT TRIM (Alternate Monitor Trim): Trims the alternate control room loudspeaker outputs
- -10dB to +10dB
- Center is 0dB (unity gain)

ALT (Alternate Monitors): Toggles between the main and alternate control room loudspeakers outputs
- Main outputs are cut when engaged
- Illuminates when engaged

MONO: Sums the selected control room source to mono
- Left and Right control room outputs are summed to mono when engaged
- Illuminates when engaged

DIM: Sets the control room loudspeaker **DIM** level
- Controls the amount of attenuation applied to the loudspeaker outputs when DIM is engaged
- Range from 0dB to -20dB

C/R LEVEL (Control Room Level): 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!***

CUT: Mutes the selected control room loudspeakers when engaged
- Monitor output ON/off switch
- Illuminates in red when engaged

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

11.4 Headphone Amplifier

In addition to the two control room monitor controls, "**THE BOX 8**" 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 control 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

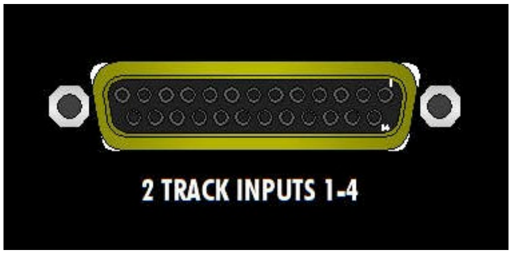
- Illuminates when engaged

HEADPHONES: LEFT & RIGHT stereo headphone output

- Fed from headphone amp outputs when **ON** is engaged in the headphone master
- Headphone jack is located under armrest
- Second headphone jack on rear panel
- ¼" tip-ring-sleeve headphone jacks

11.5 Monitor System Rear Panel Connections

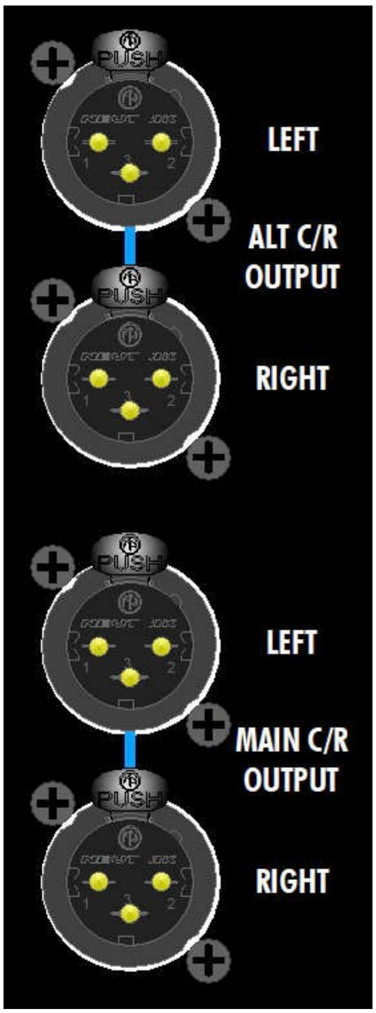
11.5.1 External 2 Track Inputs



2 TRACK INPUTS 1-4: External stereo monitor and cue sources

- Routed to the **C/R SOURCE** and **CUE SOURCE** selectors
- Balanced, low-impedance, line-level
- DB25 connector (standard pin-out)
    - 1-2=2 TRACK 1, 3-4=2 TRACK 2, 5-6=2 TRACK 3, 8-7=2 TRACK 4

11.5.2 Main and Alternate Monitor Outputs



ALT C/R OUTPUT (Alternate Control Room Outputs): LEFT & RIGHT alternate control room monitor outputs

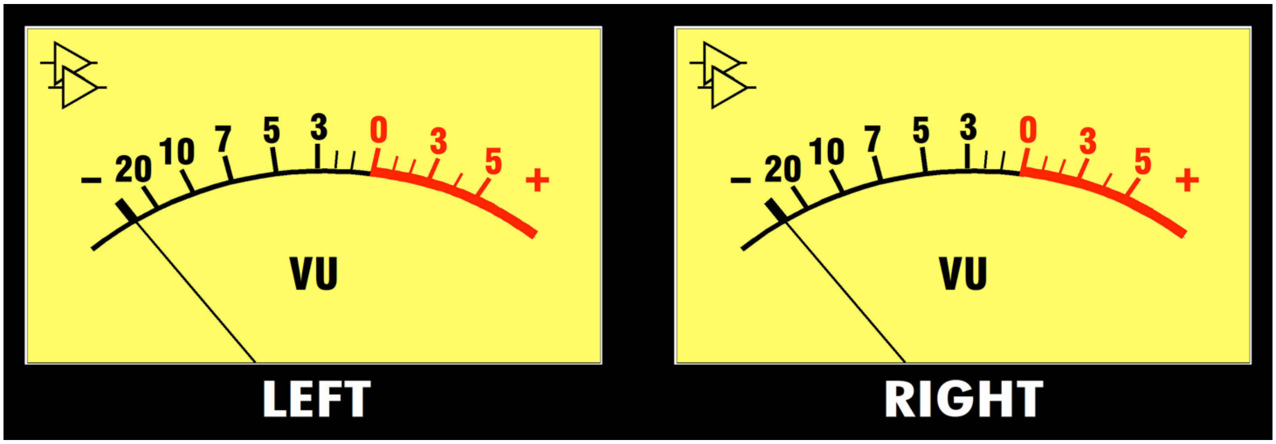
- Fed from the **C/R LEVEL** and **ALT TRIM** when **ALT** is engaged
- Balanced, low-impedance, line-level
- 3-pin male XLR

MAIN C/R OUTPUT (Main Control Room Outputs): LEFT & RIGHT main control room monitor outputs

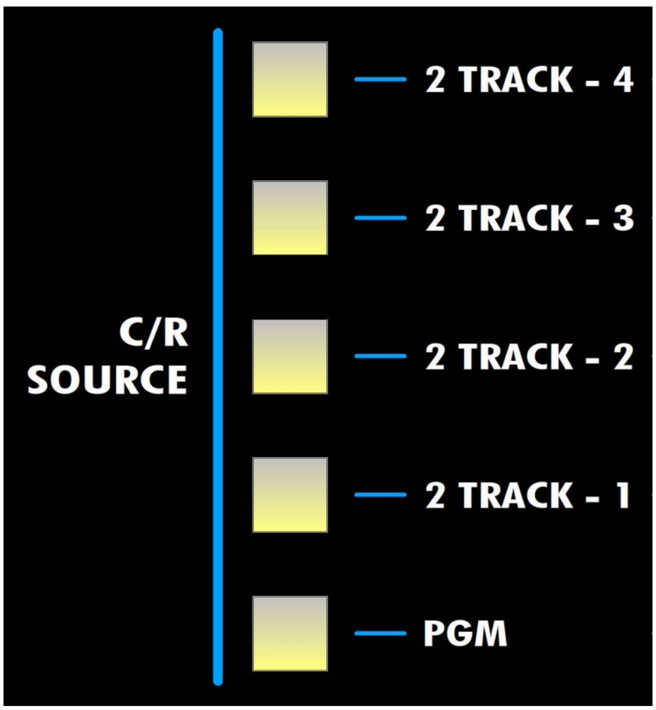
- Fed from the **C/R LEVEL** when **ALT** is not engaged
- Balanced, low-impedance, line-level
- 3-pin male XLR

12.0 VU Meters

"**THE BOX 8**" is equipped with a pair of LEFT and RIGHT stereo VU meters mounted above Compressors 1 & 2. 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.



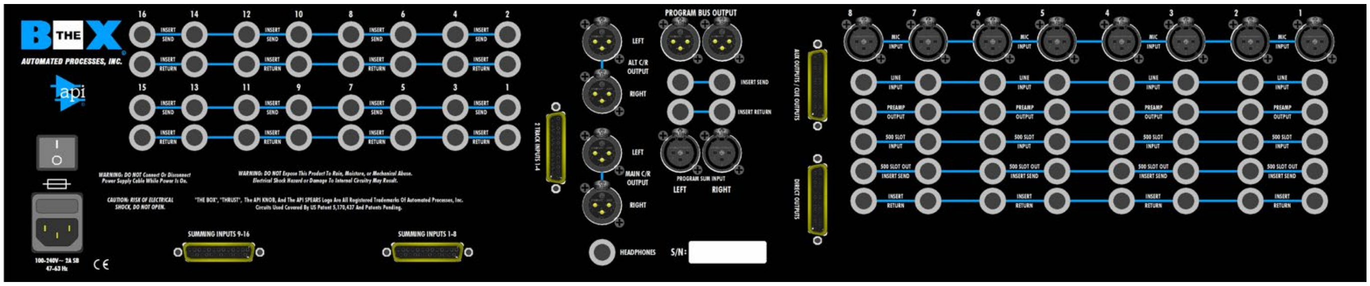
The stereo VU meters display the levels of the sources selected for control room monitoring (**C/R SOURCE**).

When a SOLO is engaged, the VU meters display the solo level.

Refer to section 11.2 Control Room Source Selection for additional information regarding control room monitor source selection.

13.0 Rear Panel Connections

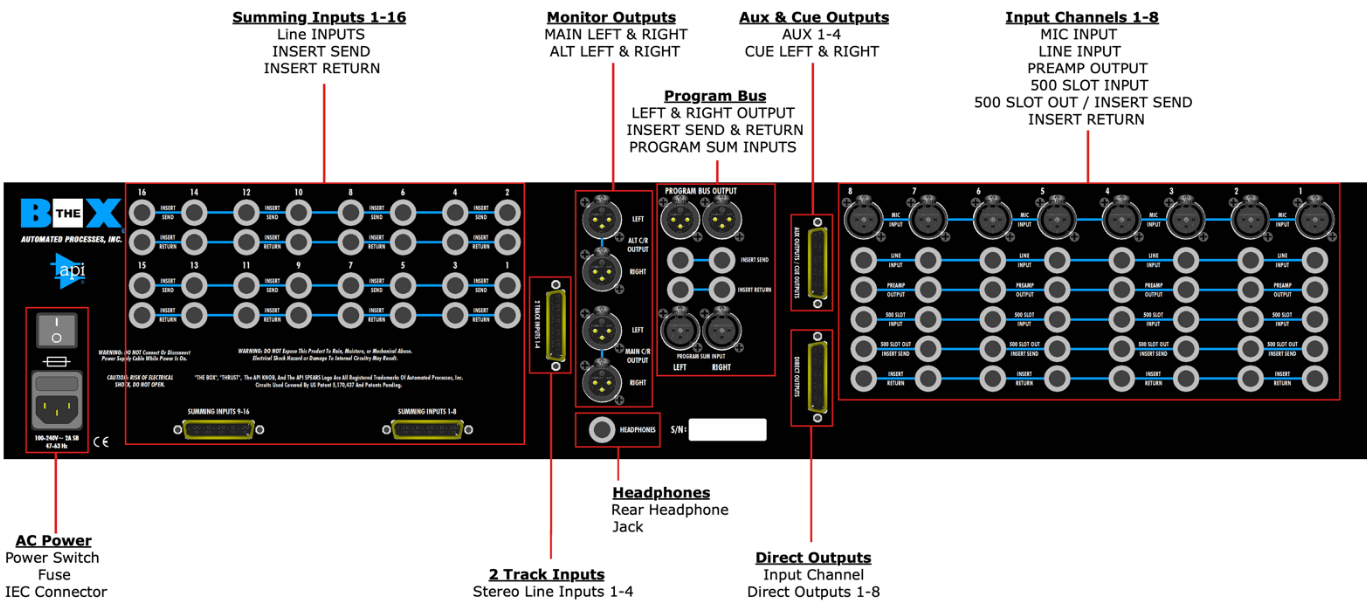
"**THE BOX 8**" rear panel contains all the connections needed to interface the console with other studio equipment.



Connections are organized in several sections:

- **Input Channels 1-8**: Preamp inputs, insert sent/return, direct/pre out
- **Summing Inputs 1-16**: Line inputs, insert send/return
- **Program Master**: Sum inputs, insert send/return, outputs
- **Aux/Cue Master**: outputs
- **Control Room/Cue Source 1-4**: 2 tracks 1-4 inputs
- **Control Room Loudspeakers**: Main and alternate outputs
- **Headphones**: Headphone amp output
- **AC Power**: IEC connector, fuse, and power switch

13.1 Rear Panel Sections

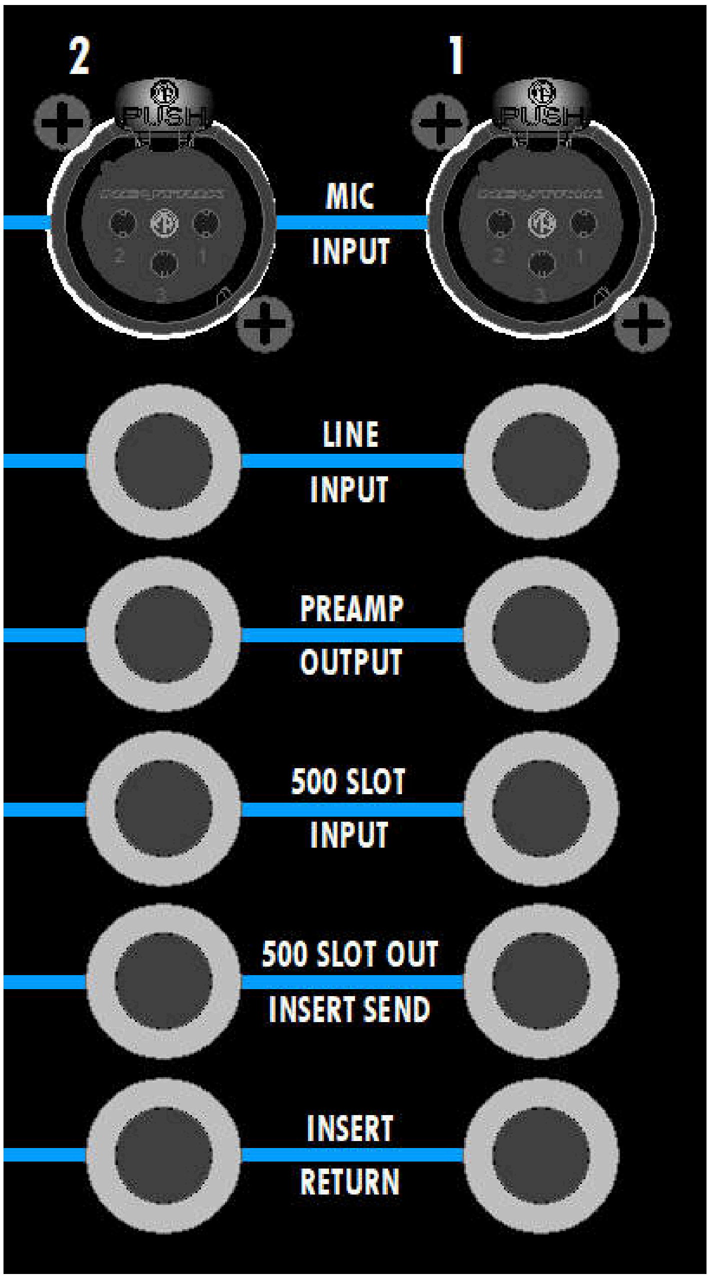


13.2 Input Channels 1-8 Connections

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

Connections include:

- Preamp inputs:
    - Microphone input
    - Line input
    - Instrument input (under armrest of input channels 1-4)
- Preamp output
- 500 slot input
- 500 slot output/Insert send
- Insert return
- Direct output



MIC INPUT: Preamp microphone input

- Engage **MIC** to use this input
- Balanced, low-impedance
- Provides phantom power when the **48V** switch is engaged
- 3-pin female XLR connector

LINE INPUT: Preamp line input

- Disengage **MIC** to use this input
- Line-level, balanced, low-impedance
- ¼″ tip-ring-sleeve

PREAMP OUTPUT: Preamp output

- Line-level, balanced, low-impedance
- ¼″ tip-ring-sleeve

500 SLOT INPUT: 500 slot input

- Breaks feed from preamp/comp
- Line-level, balanced, low-impedance
- ¼″ tip-ring-sleeve switching jack

500 SLOT OUTPUT/INSERT SEND: 500 slot output/insert output

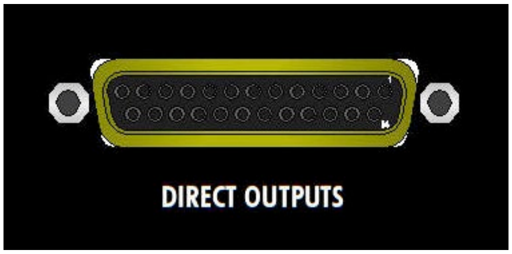
- Fed from 500 slot output
- Line-level, balanced, low-impedance
- ¼″ tip-ring-sleeve

INSERT RETURN: Insert input

- Active when **INS** is engaged
- Line-level, balanced, low-impedance
- ¼″ tip-ring-sleeve switching jack

NOTE: 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.

The DIRECT OUTPUTS connector carries direct outputs from input channels 1-8.



<u>DIRECT OUTPUTS</u>: Direct outputs from input channels 1-8
- Fed post-fader/mute by default (direct output)
- Preamp output replaces post-mute output if **DIR PRE** is engaged
- Line-level, balanced, low-impedance
- DB25 connector (standard pin-out)
    - 1-8=direct outputs 1-8

13.2.1 Instrument Input

Input channels 1-4 are equipped with an instrument input that accepts unbalanced, high-impedance, instrument-level signals from the switching ¼" INST IN connector under the armrest. The instrument input serves the function of an external direct injection (DI) box and has the following features:

- Direct input from instruments such as electric guitars, basses, keyboards, etc.
- Up to +46dB of gain (+11dB to +46dB)

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

NOTE: INST IN jack is a switching jack. The signal from the MIC INPUT is replaced with the signal from the INST IN whenever a ¼" plug is inserted in the INST IN jack.

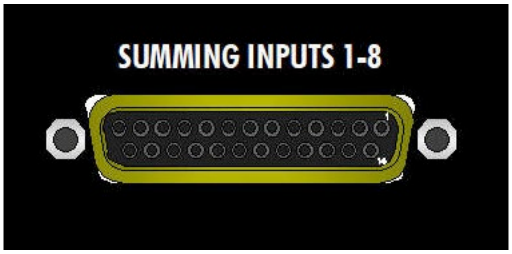
NOTE: The pad and polarity inverter (phase reverse) do not apply to the instrument input. If these functions are needed, the use of an external direct injection (DI) box connected to the MIC INPUT is suggested.

<u>INST IN</u>: Instrument-level preamp input
- Input channels 1-4 only
- Located under armrest
- Engage **MIC** to use this input
- Replaces MIC INPUT signal as the preamp input
- Unbalanced, high-impedance
- ¼" tip-ring switching jack

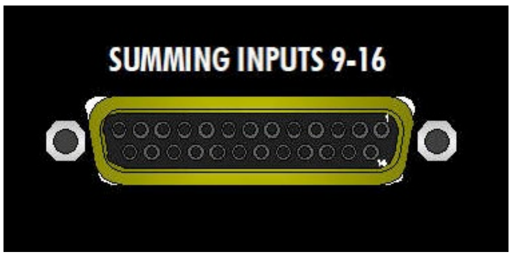
13.3 Summing Inputs 1-16 Connections

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

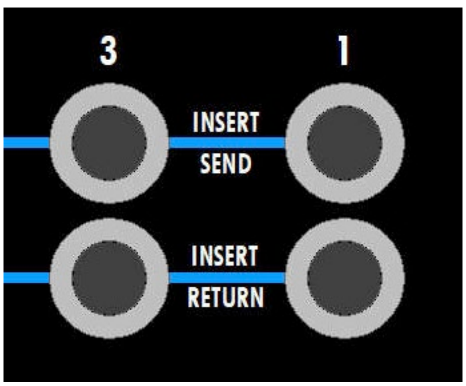
Connections include:
- Line inputs (1-8 and 9-16)
- Insert send (1-16)
- Insert return (1-16)



SUMMING INPUTS 1-8: Line input to summing inputs 1-8
- Line-level, balanced, low-impedance
- DB25 connector (standard pin-out)
  - 1-8=summing inputs 1-8



SUMMING INPUTS 9-16: Line input to summing inputs 9-16
- Line-level, balanced, low-impedance
- DB25 connector (standard pin-out)
  - 1-8=summing inputs 9-16



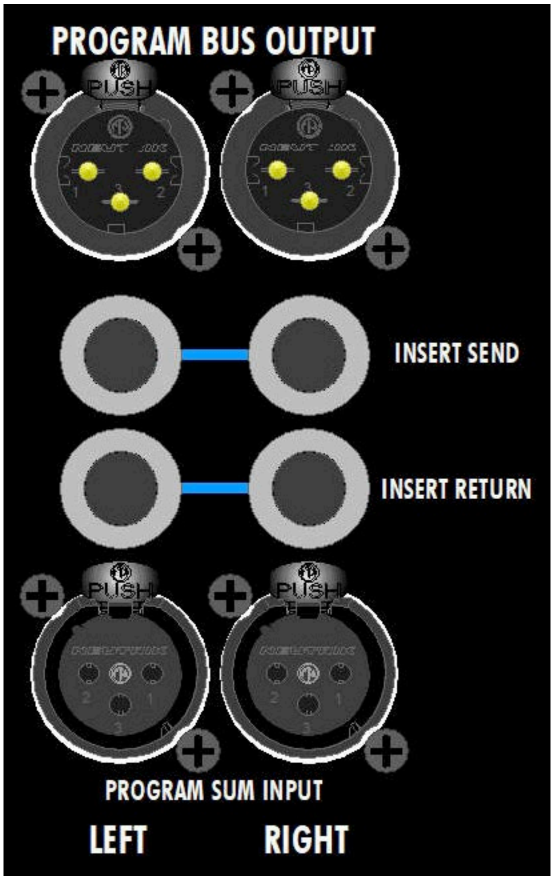
INSERT SEND: Insert output
- Line-level, balanced, low-impedance
- ¼" tip-ring-sleeve

INSERT RETURN: Insert input
- Active when **INS** is engaged
- Line-level, balanced, low-impedance
- ¼" tip-ring-sleeve switching jack

13.4 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 OUTPUT**: Program bus main Left and Right outputs
- **INSERT SEND**: Stereo program bus insert sends
- **INSERT RETURN**: Stereo program bus insert returns
- **PROGRAM SUM INPUT**: Stereo external summing inputs to the program bus



PROGRAM BUS OUTPUT: Main stereo mix outputs

- Located post-master fader
- Line-level, balanced, low-impedance
- 3-pin male XLRs

INSERT SEND: Stereo program bus insert outputs

- Line-level, balanced, low-impedance
- ¼" tip-ring-sleeve

INSERT RETURN: Stereo insert inputs

- Active when **INS** is engaged
- Line-level, balanced, low-impedance
- ¼" tip-ring-sleeve switching jacks

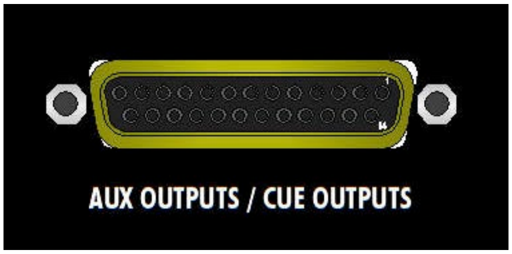
PROGRAM SUM INPUT: Stereo inputs that sum with the program bus

- Active when **PGM SUM IN** is engaged
- Located pre-insert and compressor
- Line-level, balanced, low-impedance
- 3-pin female XLRs

NOTE: *The PROGRAM SUM INPUTs are useful for bringing a stereo mix of external sources into to "**THE BOX 8**" for inclusion in the program bus. For example, two "**THE BOX 8**" consoles can be linked together by patching the PROGRAM BUS OUTPUTs of the slave console to the PROGRAM SUM INPUTs on the master console and engaging the **PGM SUM IN** switch on the master.*

NOTE: *The LEFT & RIGHT PROGRAM BUS OUTPUTs are paralleled with outputs 7 & 8 on the AUX OUTPUTS / CUE OUTPUTS DB25 connector.*

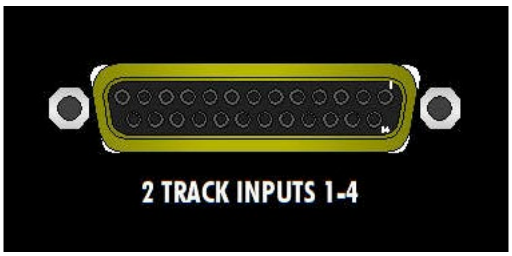
13.5 Auxiliary & Cue Connections



AUX OUTPUTS / CUE OUTPUTS: Left-Right cue master outputs, plus additional program master outputs

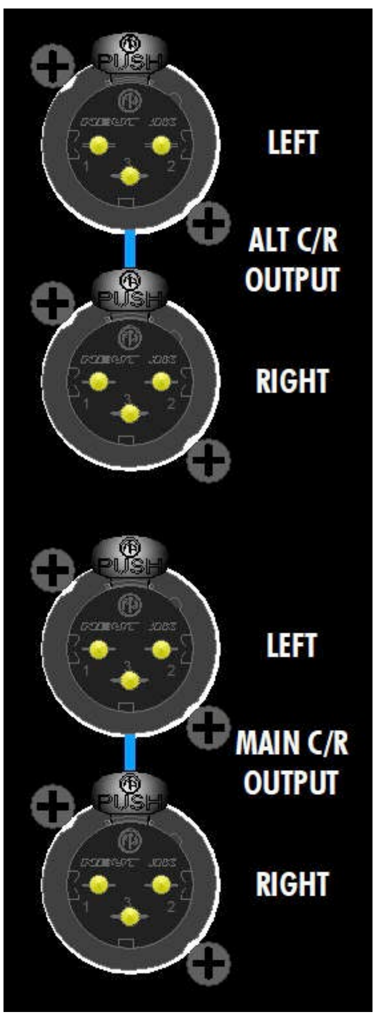
- 1-6 are fed from the aux and cue masters
- 7-8 are fed from stereo program masters
- Balanced, low-impedance, line-level
- DB25 connectors (standard pin-out)
  - 1-4=aux outputs 1-4, 5-6=cue outputs L-R, 7-8=PGM outputs L-R

13.6 Control Room Monitor System Connections



2 TRACK INPUTS 1-4: External stereo monitor and cue sources

- Routed to the **C/R SOURCE** and **CUE SOURCE** selectors
- Balanced, low-impedance, line-level
- DB25 connector (standard pin-out)
  - 1-2=2 TRACK 1, 3-4=2 TRACK 2, 5-6=2 TRACK 3, 8-7=2 TRACK 4



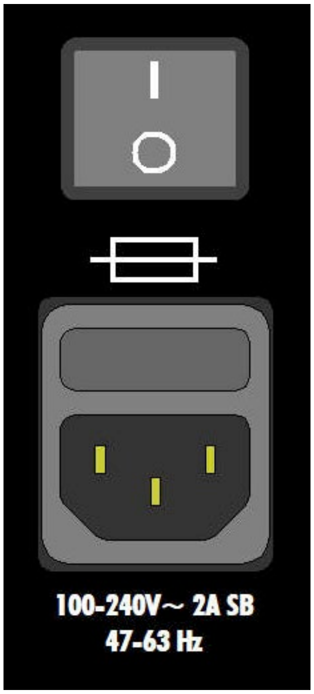
ALT C/R OUTPUT (Alternate Control Room Outputs): LEFT & RIGHT alternate control room monitor outputs

- Fed from the **C/R LEVEL** and **ALT TRIM** when **ALT** is engaged
- Balanced, low-impedance, line-level
- 3-pin male XLR

MAIN C/R OUTPUT (Main Control Room Outputs): LEFT & RIGHT main control room monitor outputs

- Fed from the **C/R LEVEL** when **ALT** is not engaged
- Balanced, low-impedance, line-level
- 3-pin male XLR

13.7 AC Power Switch and 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 8**" 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 8**" 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 8**." 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 8**" in a traditional recording fashion, with modern options. Input channels 1-8 are used as signal paths to a DAW for recording up to eight mono or four 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, INST IN or LINE inputs.

2) Select the appropriate preamp input for the sources to be recorded (mic or line):
   - **<u>MIC</u>**: Engage the **MIC** switch and the **48V** switch if needed
   - **<u>INST IN</u>**: Insert a ¼" plug from an instrument into the INST IN jack and engage the **MIC** switch to use the instrument input
   - **<u>LINE</u>**: Disengage the **MIC** switch (LINE is selected by default)

3) Connect the input channel DIRECT OUTPUTs to the inputs of the DAW.
   - DIRECT OUTPUTs are 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:
   - **<u>High-Pass Filter</u>**: Refer to section 2.2.2 High-Pass Filter
   - **<u>Insert</u>**: External processors. Refer to section 2.2.3 Insert
   - **<u>527 COMPRESSOR</u>**: Input channels 1-4; Refer to section 2.4.4 Compressor Routing
   - **<u>500 Series Slot</u>**: For processing with installed 500 modules, refer to section 2.4.1 500 Slot and the user guide for the installed modules

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 **PGM** 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 SOURCE** 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 **PGM** switch.

3) Using the **CUE SOURCE** selectors, engage the **C/R** 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 sent 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 **PGM** as the **C/R SOURCE** and **C/R** 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** 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 to the 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>PGM</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 1-8 to return up to twenty-four (24) 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 1-8 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) 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.

3) Set DAW faders to 0dB (unity gain) and turn off DAW fader automation

4) With channel faders down, engage the **PGM** switch.

5) Set the program master fader to 0dB (unity gain).

6) Select the **PGM** as the **C/R SOURCE** and set the loudspeakers to a modest level (no more than 85dB please!).

7) Use faders and pans to carefully push up a balanced stereo mix.

8) Employ the onboard signal processing as needed:
    - **<u>High-Pass Filter</u>**: Refer to section 2.2.2 High-Pass Filter
    - **<u>Insert</u>**: External processors. Refer to section 2.2.3 Insert
    - **<u>527 COMPRESSOR</u>**: Input channels 1-4; Refer to section 2.4.4 Compressor Routing
    - **<u>500 Series Slot</u>**: For processing with installed 500 modules, refer to section 2.4.1 500 Slot and the user guide for the installed modules

9) 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 5.3 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

A1 Specifications

A2 Block Diagram

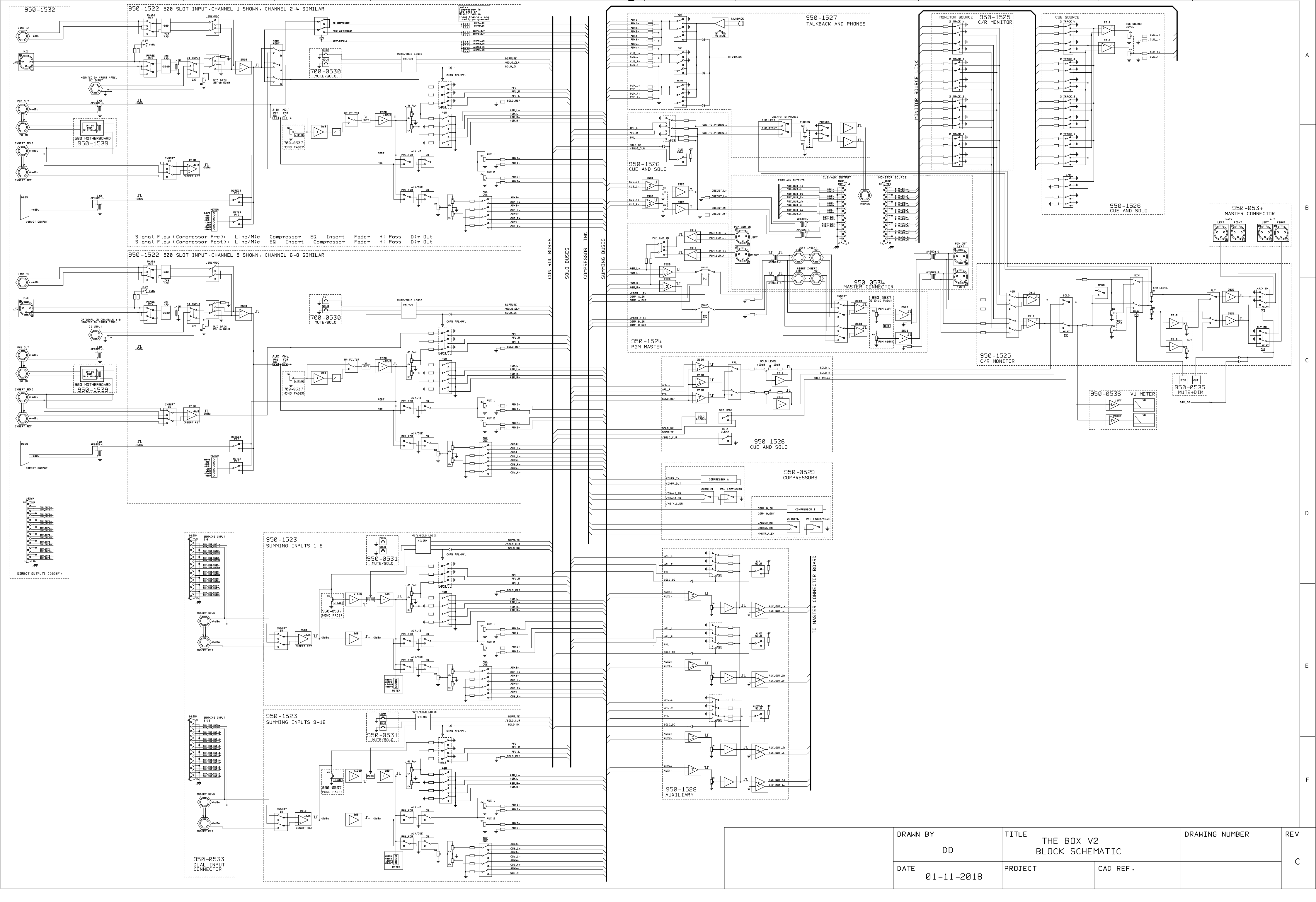
A3 DB25 Pin-out Diagrams

A4 Limited Warranty and Service

A1 Specifications

“The BOX 8” Specifications	
Input Specifications	
Line Input Maximum Level:	+28dBu (+34dBu w/PAD)
Microphone Input Gain Range:	+30dB to +65dB (+10dB to +45dB w/PAD)
Microphone Maximum Level:	-2dBu (+18dBu w/PAD)
Instrument Input Gain Range:	+11dB to +46dB
Input PAD	-20dB Mic, -6dB Line
Direct Output Maximum Level:	+28dBu
Direct Output Frequency Response:	+0/-0. dB, 20Hz to 50kHz
Signal to Noise Ratio (from max):	-116dB
Output Specifications	
Program Output Maximum Level:	+28dBu
Program Output Frequency Response:	+/- 0.5dB, 20Hz to 50kHz
Signal to Noise Ratio (from max):	-118dB
General Specifications	
Dimensions:	32.5" x 26.4" x 13.2"
Weight:	90 lbs.
Power:	120 Watts

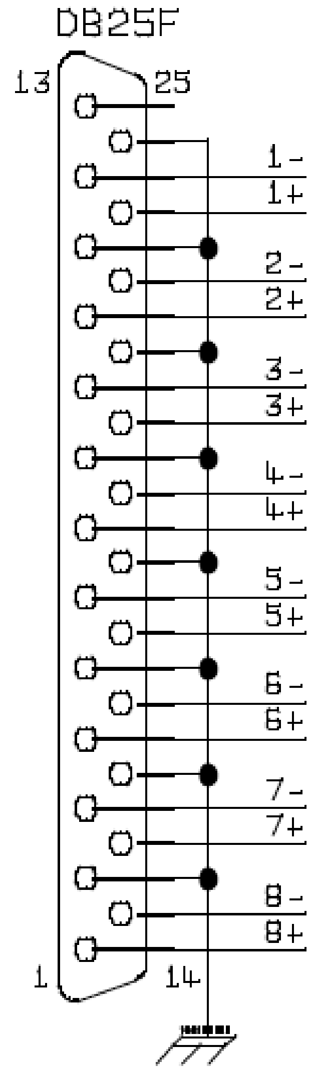
A2 Block Diagram



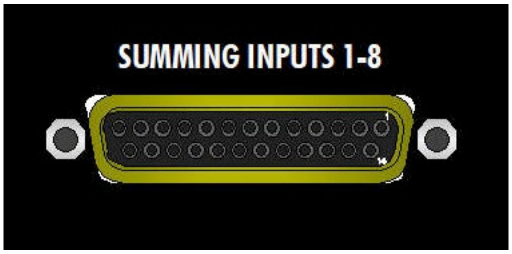
A3 DB25 Pin-Out Diagrams

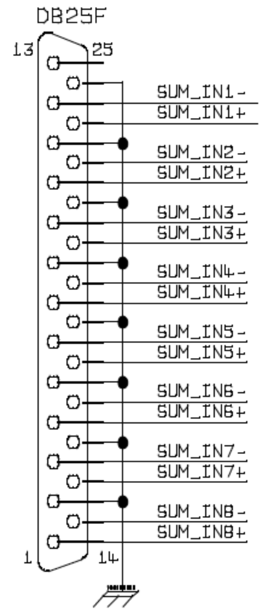
The DB25 connectors use a typical 8-channel pin-out standard.

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



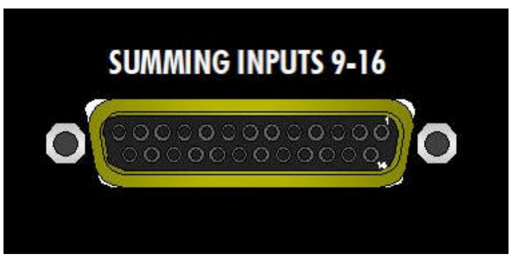
A3.1 Summing Input Line Input Connections

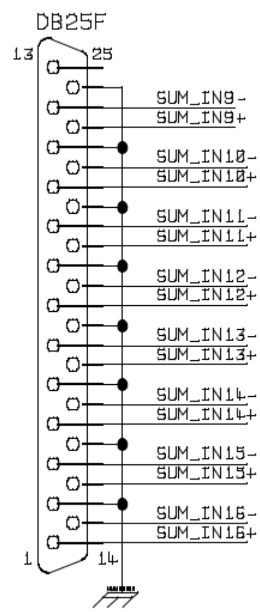




<u>SUMMING INPUTS 1-8</u>: Standard = API

- 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

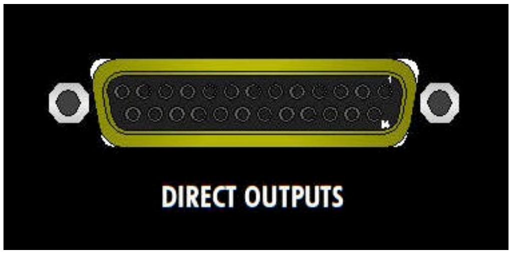


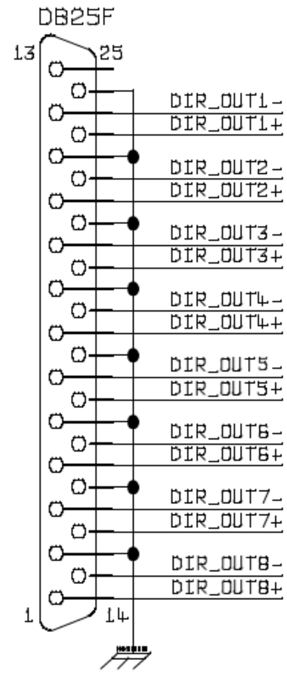


<u>SUMMING INPUTS 9-16</u>: Standard = API

- 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

A3.2 Direct Output Connections

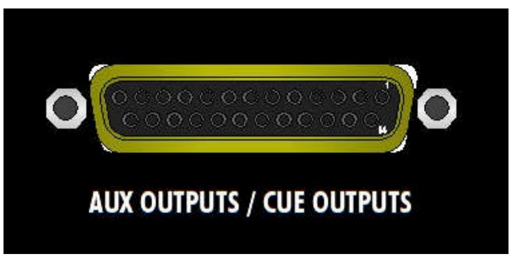


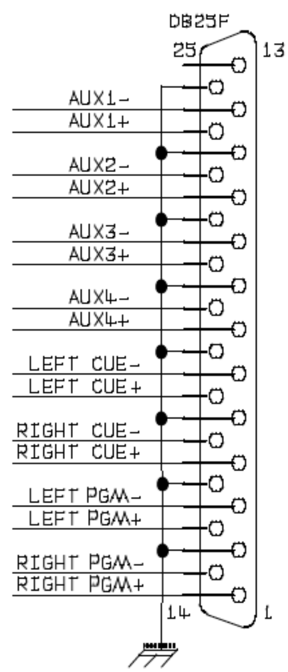


<u>DIRECT OUTPUTS</u>: Standard = API

- 1 = Direct Output 1
- 2 = Direct Output 2
- 3 = Direct Output 3
- 4 = Direct Output 4
- 5 = Direct Output 5
- 6 = Direct Output 6
- 7 = Direct Output 7
- 8 = Direct Output 8

A3.3 Auxiliary/Cue Output Connections

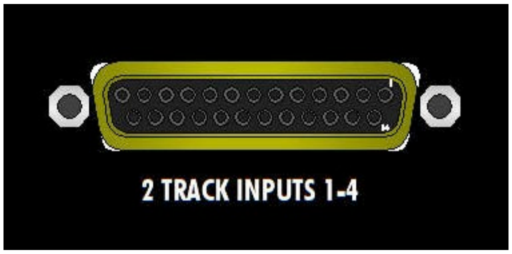


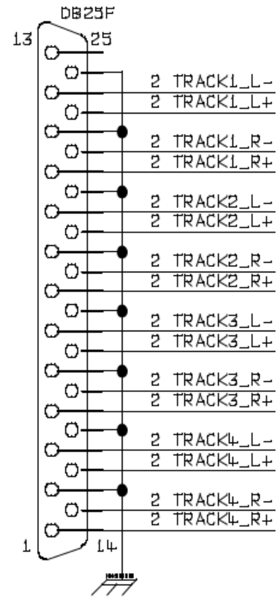


<u>AUX OUTPUTS / CUE OUTPUTS</u>: API = Standard

- 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 = Program Output LEFT
- 8 = Program Output RIGHT

A3.4 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

A4 API Limited Warranty and Service

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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