AXS LEGACY

Discrete
Mixing and Recording Console

OPERATOR'S MANUAL

V2.0

Written for Automated Processes Incorporated

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About this Manual

This manual is presented in several sections. Section one, "Introduction" provides a summary of the console features and attributes. Section two, "Console Configuration" provides an overview of various console configurations and along with a description of the available pre-delivery customization options. Likewise the rest of the manual is organized according to the various console systems such as "Channels, Multitrack Buses, Program Masters," and so on.

Sections one and two are useful to all users, and should be read thoroughly before purchase and delivery. Experienced engineers may get up and running quickly by simply checking the graphic overviews at the start of the sections. If you are new to API consoles and multitrack recording, it's recommended that you read the entire manual at least once and keep it at the console for reference.

NOTE: This version of this manual does not contain information regarding console automation. Console automation will be addressed in a separate manual or new version of this one sometime in the near future.

Introduction

Thank you for choosing the API Legacy AXS discrete analog recording and mixing console. The Legacy AXS console continues API's commitment to our world-famous analog signal path. Offering an expanded and powerful feature set, the Legacy AXS is designed to meet the needs of today's audio professionals.

Since the introduction of the first API Legacy Series console to Green Street Studios in Manhattan in late 1989, API has delivered over 75 Legacy Series consoles around the world. The astonishing statistic is that virtually every Legacy built is still in service.

We continue the API Legacy with the Legacy AXS Recording and Mixing Console. Featuring the traditional API analog signal path, the AXS also offers a new and powerful feature set, all at an attractive price for today's studios, artists, engineers and producers.

The Legacy AXS combines a traditional, easy to use in-line input/output architecture with flexible customization options and superior sound quality. The Legacy AXS is built to the same exacting standards of reliability, sonic performance, and investment grade audio of the API Legacy, Legacy Plus, and Vision consoles, setting the standard for modern recording consoles.

Features

- 32 to 80 channels
- API's discrete signal path with transformer outputs
- Numerous preamp and signal processing options
- VPR Alliance 500 Series compatible slot for EQ
- Six assignable stereo line-level returns
- Twenty-four multitrack summing buses
- Three independent stereo program buses and assignable Grand Master
- API 2500C stereo bus compressor
- Extensive routing features for parallel compression and stem mixing
- Twelve auxiliary buses
- Onboard or remote full patch bay (only 32 thru 64-channel consoles can be onboard)
- Highest quality finish and wood trim
- Comprehensive automation system
- Optional DAW/producer's desk
- Configured to client specification

Configuration Options

- Mainframes:
  - 32 to 80 channels (in 16-channel buckets)
  - Channel Width: 1.5"
  - Channel bucket Width: 24"
- 500 Series equalizer slot on each channel
  - VPR Alliance compliant
- Dual 200 Rows on each channel:
  - Upper slot for preamp
  - Lower slot for optional signal processing
  - Pre or post EQ lower 200 slot routing option (specified when ordered)

Channel Features

- Dual signal path architecture: Two independent audio paths:
  - Large Fader: 100mm automated fader
  - Small Fader: 45mm fader (non-automated)
- Line In or Preamp Out input sources on both fader paths
- Alternate Line Input source for Small Fader path
- Upper 200 slot preamp options (as specified):
  - 205 Direct Input
  - 212 Microphone Preamplifier
- Lower 200 slot signal processing options (as specified):
  - 215 Sweep Filter
  - 225 Compressor
  - 235 Noise Gate
- 500 Series slot and equalizer/filter options (as specified):
  - 550A 3-band parametric EQ
  - 550b 4-band parametric EQ
  - 560 10-band graphic EQ
  - 565 Filter Bank
  - Compatible with any VPR Alliance approved 500 module (except preamps)
- 40-600Hz, 12dB/8va high-pass sweep filter in the Large Fader path
- 50Hz, 12dB/8va fixed high-pass filter in the Small Fader path
- Balanced Insert in each path
- Phase reverse (polarity inverter) in each path
- Twelve auxiliary sends and buses: 6 mono, 3 stereo
- Channel VU metering for: Large Fader, Small Fader, Direct Output, and Multitrack Bus
- Peak indicator in each path (globally set)

Central Facilities

- Three control room monitor systems: stereo/surround MAINs and stereo ALT 1 & ALT 2
- Two stereo studio monitor systems
- Four stereo and two surround external playback sources
- Comprehensive stereo program and cue send monitoring
- Individual CUT control of all control room monitors (LEFT, RIGHT, CNTR, SURL, SURR, LFE)
- Monitor calibration level trim controls
- Comprehensive cue and talkback systems
- Solo System: Independent/Linkable AFL, PFL, and Solo-In-Place for Small and Large faders
- API 2500C stereo Bus Compressor
- Routable Sine-wave oscillator
- Four USER buttons (toggled contact closures on D-connector)

As with all API products, The Legacy AXS Console offers the same exacting craftsmanship featured on all API large format consoles, including a unique 5-year warranty on all parts.

Console Configuration

Each Legacy AXS console is built to meet the specifications determined by the owner when the console is ordered. With a flexible design, the console may be configured in many ways. Beyond frame size, users may configure each module with their preferred pre-amp and signal processor setup to include a variety of preamp, equalizer, filters, and dynamic processor possibilities.

While workflow, capacity, and patching may vary by configuration, the basic operational premise of all Legacy AXS consoles is the same.

Mainframe and Channel Slots

The Legacy AXS console mainframe comes in a variety of frame sizes consisting of a master bucket and two or more 16-channel channel buckets. The smallest frame size is 32 channels and the largest accommodates 80. The patch bay for consoles with frames for 32 thru 64 channels can be mounted onboard the console or externally. The patch bay for console frames over 64 channels must be externally mounted.

Channel Bucket

There are several module slots associated with each 16-channel bucket in the Legacy AXS mainframe. Certain module slots can be fitted with optional components.

From the meter-bridge to the armrest, the slots are in the following order (see Channel Strip Overview below):

- Upper 200 slot (preamps)
- Lower 200 slot (signal processing)
- 624 Bus Assignment Module slot
- 500 Series EQ slot
- 968 Input Module slot
- 948 Large Fader bay

The Bus Assignment slot can only be fitted with the 624 Bus Assignment Module.

The Input Module slot can only be fitted with a 968 Input Module.

The 200 slots and the 500 EQ slot offer customization options for inputs and signal processing.

Configuration of the upper 200 slot, lower 200 slot, the 500 EQ slot allows the user to customize the console for their desired workflow and use. The upper 200 slot may be configured with one of two preamp options, while the lower 200 slot may be fitted with one of three dynamic processors or filters. The 500 EQ slot can be fitted with one of four API EQ and filter options or any VPR Alliance approved 500 Series module (except preamps). Any or all of the slots may be left open for future expansion.

Configuration options are specified by the user in consultation with API engineering staff before the console is built at API.

Upper 200 Slot

The upper 200 slot is designed to house the channel preamp. The upper 200 slot is typically fitted with the API 212 Microphone Preamplifier module. Alternately the upper 200 slot can be fitted with an API 205 Direct Input. Most consoles are fitted primarily with 212 Mic Preamps, but usually include at least two 205 Direct Inputs to increase versatility and tonal variance when recording electric instruments.

The output of the preamp installed in the upper 200 slot is internally routed to the Small Fader path input by default. The preamp output can be routed to the Large Fader path input on individual channels by engaging the **FLIP** button on the 968 Input Module or globally by engaging the **FLIP** button in the Master Section. The input to the Small Fader path will switch to the channel LINE IN (multitrack return) when **FLIP** is engaged. The output of the upper 200 slot preamp cannot be directly routed to both fader paths simultaneously.

The microphone input to the upper 200 slot preamp is available on patch as MIC PREAMP INPUT. These patch points are fed from the MIC TIE-LINE patch points via a full-normal connection.

Lower 200 Slot

The lower 200 slot is designed to house additional channel signal processing and is typically fitted with an API 225 Compressor-Limiter. However, API provides two additional 200 Series signal processing module options:

- 215 Sweep Filter
- 235 Noise Gate/Expander

In the standard configuration, the lower 200 slot signal processor (COMP) is located pre-EQ in the Small Fader path. The lower 200 slot (COMP) can be moved to the Large Fader path on individual channels by engaging the COMP LRG button on the 968 Input Module or globally by engaging the COMP LRG button in the Master Section. Engaging the COMP BYP button in the Master Section will bypass the lower 200 slot (COMP) on all channels.

An internal link inside each 968 Input Module can be programmed to place the lower 200 slot (COMP) post-EQ instead of pre-EQ in its assigned path.

The signal processor in the lower 200 slot may be accessed using the COMPRESSOR INPUT, SIDE CHAIN, and COMPRESSOR OUTPUT patch points.

See the 200 Series signal processor descriptions in the “Channel” section later in this manual.

624 Bus Assignment Module Slot

The channel bucket contains dedicated slots designed to support API 624 Bus Assignment Modules. The 624 module handles assignments to Multitrack and Program Buses from the channel Large and Small Fader paths. It also facilitates pan-pot activation in each path. The assignment module slots can only be fitted with API 624 Bus Assignment Modules.

Multitrack Bus outputs and access to the program bus signal flow are available in the System patch bay.

500 Series Equalizer Slot

The 500 Series slot on each channel strip is intended to house the channel equalizer (EQ). This 500 Series slot complies with VPR Alliance standards so it can accept any API or VPR approved 500 Series module except preamps. This slot is not designed to house a preamp and does not have a 48v power supply for phantom power.

The 500 Series EQ slot is typically fitted with one of four API equalizer or filter modules:

- 550A 3-Band EQ
- 550b 4-Band EQ
- 560 10-Band Graphic EQ
- 565 Filter Bank

The 500 Series EQ slot is assigned into the Large Fader path by design. The EQ can be routed to the Small Fader Path by engaging the EQ SM button on the 968 Input Module.

968 Input Module Slot

The channel bucket contains dedicated slots designed to support API 968 Input Modules. The 968 is the main channel module and contains the Large and Small fader paths, internal channel routing, and associated controls. The 968 modules also contain the channel auxiliary sends. The input module slots can only be fitted with API 968 Input Modules.

Access to the channel signal flow is available in the Channel patch bay.

948 Fader Bay

Bays for 948 Fader Modules are located at the bottom of the channel buckets. These bays contain two 8-channel fader modules that support the 16 channels in the bucket.

Master Bucket

The master bucket of the Legacy AXS contains routing and controls for buses, summing, outputs, monitoring and other central console controls. From top to bottom the sections include:

- Program Meter Bridge
- 2500C Stereo Bus Compressor
- 624M Multitrack Masters
- 624SM Program Bus Masters
- 265 Auxiliary Bus/Cue Send Masters
- Six assignable Stereo Returns
- Master Section
- 946 Program and Group Master Faders and automation controller.

Patch Bay

All consoles include fully integrated, fully balanced TT (bantam) patch bays that provide access to the signal flow of various console and studio systems. In the standard configuration the “Channel Bay” contains 14 patch points in the channel signal flow. The “System Bay” contains patch points for program, multitrack, & auxiliary buses, stereo returns, 2500C Bus Compressor, monitoring, cues, effects, and external studio audio devices.

Patch bays for consoles with frames for 32 thru 64 channels can be mounted onboard the console or mounted externally. Patch bays for consoles frames larger that 64 channels must be externally mounted.

Optional Work Station-Producer’s Desk

An optional producer’s desk can be specified at the time the order for the console is placed. The producer’s desk is designed to support a DAW workstation and/or work surface to help integrate analog and digital audio workflow. Because the producer’s desk is built into the frame, it must be specified before the console is built.

FOR A CONCISE AND CONSISTENT DESCRIPTION OF THE CONSOLE, IT IS NECESSARY TO ASSUME ONE OF THE ABOVE CONFIGURATION CHOICES HAS BEEN MADE.

FOR THE REMAINDER OF THIS MANUAL IT IS ASSUMED THAT THE UPPER 200 SLOT IS EQUIPPED WITH A 212 MIC PRE, THE LOWER 200 SLOT IS EQUIPPED WITH A 225 COMPRESSOR, AND THE 500 SERIES SLOT IS FITTED WITH A 550A EQ.

Channel Signal Path Architecture

From top to bottom, the following channel bucket components described above make up a complete Legacy AXS channel strip:

- Upper 200 slot: preamp
- Lower 200 slot: signal processing
- 624 Bus Assignment Module
- 500 Series Slot: EQ
- 968 Input Module
- 948 Fader Module

The Legacy AXS is a "dual-channel" or "in-line" console. Each channel strip provides two independent, discrete audio paths for multitrack recording. One path routes microphones to a multitrack recorder and the other path is for mixing the multitrack return. After recording, both audio paths can be routed to the program buses to provide additional inputs during mix-down. On the Legacy AXS these audio paths are referred to as the Small Fader path and the Large Fader path.

Channel Input Selection

There are two primary choices of input sources, Preamp output and Line In, and one Alternate Line In.

Preamp: This is the output of the preamp installed in the upper 200 slot. There are two modules available to serve as a preamp:

- 212 Mic Preamp
- 205 Direct Input
- Feeds the Small Fader Path by default

Line In: This is the output from the multitrack recorder interfaced with the console.

- A -6 dB TAPE PAD can be engaged at the Line In
- Feeds the Large Fader Path by default

Alt Line In: This is an alternate line input for the Small Fader path, replacing the Preamp output.

- Feeds the Small Fader path input when the **ALT** button is engaged

The default input for the Large Fader path is Line In and the default input for the Small Fader path is the Preamp output. Each input source cannot feed both paths simultaneously, so the Preamp, Line In, or Alt Line In can only feed one path at a time.

Engaging the **FLIP** button will swap the input sources between fader paths, so the Preamp will feed the Large Fader path and the Line In will feed the Small Fader. Engaging the **ALT** button will switch the Small Fader path input to the Alt Line In for recording or mixing additional line level sources. If the **ALT** and **FLIP** buttons are engaged, the Alt Line Input will feed the Large Fader path.

There is a **FLIP** button in the Master Section to reverse the inputs between the Small and Large Fader paths on a global basis. There is also a master **SM ALT LINE** button that routes the Alternate Line Input (ALT LINE INPUT) to the Small Fader input console wide. The master **ALL CLR** button clears all **FLIP** and **ALT** buttons globally and restores normal channel inputs.

Channel Output Routing

The Small Fader and Large Fader paths have access to all output assignment possibilities. The output possible assignments are:

- Direct Output
- Multitrack Buses 1-24
- Stereo Mix Bus A, B, and C

Direct Output

- Only one path can be assigned to feed the Direct Output at a time
- The Large Fader will normally feed the Direct Output.
- The Small Fader will feed the Direct Output when the **DIR SM** switch is engaged. This will defeat the Large Fader signal to the Direct Output.
- The pre-EQ and Insert signal of the selected fader path will feed the Direct Output when the **DIR PRE** switch is engaged.

Multitrack Buses

- There are twenty-four (24) Multitrack Buses which feed the Active Combining Amplifiers (ACA).
- The Large Fader or Small Fader can be assigned to any of the 24 Multitrack Buses in the 624 Bus Assignment Module.
- Only one audio path, Small Fader or Large Fader, can be assigned to the Multitrack Buses at a time.
- The default source for the Multitrack Buses is the Large Fader.
- The Small Fader can feed the Multitrack Buses by pressing the **24** button on the 968 module. This will defeat the Large Fader signal to the Multitrack Buses.
- There is a TRIM control and on/off switch for each ACA output.

Stereo Program Buses

- Both paths, Small Fader and Large Fader, can be assigned to any and all stereo Program Buses, A, B, and C.
- Program Bus assignments are made for both fader paths in the 624 Bus Assignment Module. There are two sets of STA, STB, and STC switches at the bottom of the 624 module. The left set of switches is for the Small Fader (as indicated by **SM** below the switches). The right set of switches is for the Large Fader (as indicated by **LG** below the switches).

Panning

- Pan-pots are not engaged by default in the either fader path. Each path has its own pan-pot, which must be engaged to be active.
- Panning assignments are made for both fader paths in the 624 Bus Assignment Module by using the two PAN switches.
- The left PAN switch is for the Small Fader (as indicated by **SM**). The right **PAN** switch is for the Large Fader (as indicated by **LG**)

Session Signal Flow

For those familiar with other in-line recording consoles, one fader path functions as an "mic" or "channel" path and the other fader path functions as a "monitor" or "mix" path. The fader used for each function depends on the selected signal functions. The operational controls allow for both paths to be used for either function.

Each path can provide a different function during the various types of recording sessions. The signal routing of input and output of each path is handled by the 968 Input Module and the 624 Bus Assignment Module.

<u>Multitrack Recording and Overdubs</u>: In a typical multitrack recording session, the channel strips function as follows:

- <u>Record Path (channel path, input path, mic path):</u> One fader path carries the signals from the microphone preamp, DI, or other input source to the multitrack recorder (DAW).
  - Routed via Direct Outputs or the 24 Multitrack Buses.
  - Signal processing is often recorded to the multitrack so filters, EQ, and dynamics are assigned to the record path as needed.
  - The fader used for this function (Small or Large) is determined by the user.
- <u>Monitor Path (mix path, return path):</u> The other fader path is used to mix signals from the multitrack recorder (DAW) to the stereo Program Buses for monitoring.
  - The fader used for this function (Small or Large) is determined by the user.

<u>Stereo Mixing</u>: In a typical mixing session, the channel strip functions as follows:

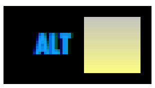
- The Large Fader path is the main mix path.
- The Small Fader path can also be used to route additional returns from multitrack recorders (DAW) or other sources such as effect units to the mix.
- Signal processing is often applied to channels contributing to the mix, so the compressor and
- EQ are assigned to the Large or Small Fader paths as needed while mixing.
- Stereo Returns are used to return mix stems and effects unit outputs.
- The channels and stereo returns contributing to the mix are routed to one or more of the three stereo Program Buses, A, B, C.
- The three stereo Program Buses, A, B, C can be summed in the Grand Master Program Bus.
- The 2500C Stereo Bus Compressor can be patched to any of the stereo Program Buses.

Default AXS Signal Flow: With no routing buttons engaged, the default channel signal flow is as follows:

- Small Fader path:
  - Input: output of the preamp (upper 200 slot)
  - Signal Processing: Lower 200 slot (COMP)
  - Output: Assignable (Direct Output, Multitrack Buses, & Program Buses)
- Large Fader path:
  - Input: INPUT (normalled to the MULTITRACK OUTPUT)
  - Signal Processing: 500 Series EQ slot
  - Output: Direct Out & by default and assignable (Multitrack Buses & Program Buses)

This default channel signal flow can be altered determined by changing the positions of the channel routing buttons labeled in **BLUE: FLIP**, **ALT**, **COMP LRG**, and **EQ SM** and the Direct Output routing buttons: **DIR SM** and **DIR PRE.**

FLIP: Swaps the Large and Small fader path inputs when engaged:
- Large Fader source: Preamp output
- Small Fader source: Line Input
  - Illuminates when engaged



ALT: Switches the Small Fader path input to the Alternate Line In (ALT) when engaged:
- Replaces the preamp output with the Alt Line In as the Small Fader path source
- Illuminates when engaged

COMP LRG (Compressor Large)**:** Moves the lower 200 slot signal processor (COMP) to the Large Fader path
- Illuminates when engaged

EQ SM (Equalizer Small)**:** Moves the 500 Series EQ slot to the Small Fader path
- Illuminates when engaged

DIR SM (Direct Small)**:** Routes the Small Fader to the Direct Output
- Disengages the Large Fader path from feeding the Direct Output
- Illuminates when engaged

DIR PRE (Direct Pre)**:** Routes the selected fader PRE signal to the Direct Output
- The Direct Output is fed post-COMP, pre-EQ & Insert instead of post-MUTE
- Illuminates when engaged

While either configuration is appropriate, most engineers will choose to use the Small Fader path as the “Record Path” to route microphones and other sources to the multitrack recorder and use the Large Fader path the “Mix Path” to monitor the multitrack return via the Program Buses. Using this method, the console will be in a basic mixing configuration when tracking is completed. Many engineers also prefer to have the multitrack sends on the small faders and use the large faders to mix the multitrack returns.

Small Fader to Multitrack

To use the Small Fader path as the “Record Path,” an output to the multitrack recorder must be assigned. The Direct Outputs provide the most direct audio output path and are normalled to the Multitrack Inputs in the standard configuration, so the Direct Output is often used when recording single source to a single track. Engaging the **DIR SM** button will route the Small Fader path to the Direct Output post the MUTE switch. If no console EQ or further gain adjustment is needed, also engaging the **DIR PRE** will route the Small Fader path signal to the Direct Output post-COMP and pre-EQ.

If two or more sources are to be summed together before being sent to the multitrack recorder, the Small Fader path must be routed to one or more of the twenty-four Multitrack Buses in the 624 Bus Assignment module. The Multitrack Buses are fed from the Large Fader path by default, so the **24** button must be engaged to reroute the Small Fader to feed the Multitrack Buses, followed by the desired bus assignment(s). If odd-even stereo panning is to be employed, the **PAN** button in the Small Fader section of the 624 module (**SM**) must be engaged.

In this configuration, the Large Fader path will be used to mix the multitrack returns and must be assigned to a stereo Program Bus for monitoring. Assign the Large Fader path to the desired stereo bus by engaging the **STA**, **STB**, or **STC** and the **PAN** buttons in the **LG** section on the 624 Bus Assignment module.

Large Fader to Multitrack

If it is preferable to use the Large Fader path as the "Record Path" to route sources to the multitrack recorder, engage the **FLIP** button to route the output of the 200 slot preamp to the Large Fader path input. The Direct Output and Multitrack Bus matrix are fed from the Large Fader by default, so only the desired multitrack bus/pan-pot assignments need to be made. If odd-even stereo panning is to be employed, the **PAN** button in the Large Fader (**LG**) section of the 624 module must be engaged. If no console EQ or further gain adjustment is needed, also engaging the **DIR PRE** only will route the Large Fader path signal to the Direct Output post-COMP and pre-EQ.

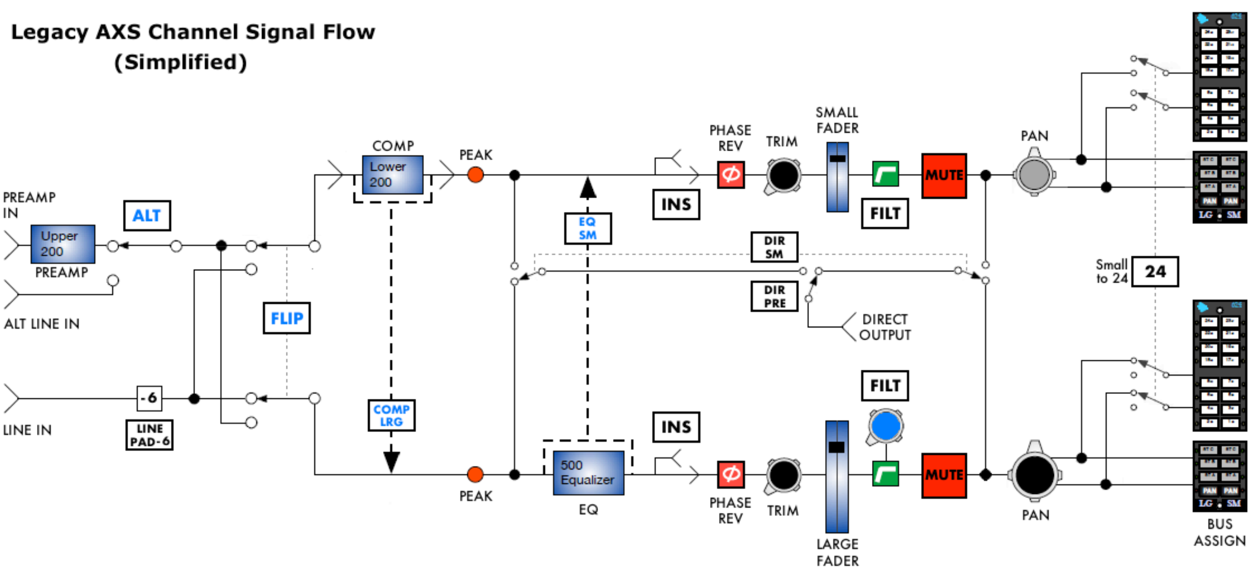
In this configuration, the Small Fader path will be used to mix the multitrack returns and must be assigned to a stereo Program Bus for monitoring. Assign the Small Fader path to the desired stereo bus by engaging the **STA**, **STB**, or **STC** and the **PAN** buttons in the Small Fader (**SM**) section on the 624 Bus Assignment module.

NOTE: The multitrack recorder may alternately be fed from the Multitrack Buses by patching the BUS OUTPUT patch points to the MULTITRACK INPUT patch points.

Signal Flow Block Diagram

The block diagram below indicates the signal flow through the Small and Large Fader signal paths. A more technical and detailed signal flow diagram is available in a separate document in the appendix of this manual.
The features of each path are shown in the order in which they occur within the path. The diagrams assume the following:

- A 212 Preamp is installed in the upper 200 slot
- A 225 Compressor/Limiter is installed in the lower 200 slot
- A 550A Equalizer is installed in the 500 Series EQ slot
- **FLIP**, **ALT**, **COMP LRG**, **EQ SM**, **DIR SM**, and **DIR PRE** are in default positions (not engaged)
- Pan-pots are engaged
- Auxiliary Sends, 0dB Fader Bypass, solo sends, and VU meter are not included



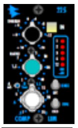
Fader Bypass

The faders in both signal paths can be bypassed allowing the audio to pass at unity gain (0dB) when the console is used with DAW automation and as a control surface. The Fader Bypass for the Large Fader path is handled via the console automation package. The Fader Bypass for the Small Fader is engaged globally by engaging the 0dB SM (0dB Small Fader) button in the master section. This option is not available on individual channels and is not shown in the diagram above.

Channel Strip Overview

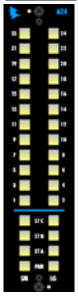
Upper 200 Slot

- 212 Microphone Preamplifier (standard configuration)
- Optional 205 Direct Input Module



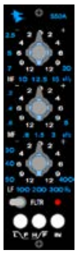
Lower 200 Slot - May be fitted with one of three processor options:

- 215 Sweep Filter
- 225 Compressor/Limiter (standard configuration)
- 235 Noise Gate/Expander



624 Bus Assignment Module

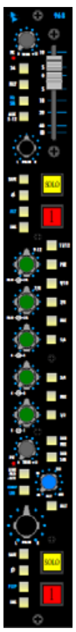
- Routes Small Fader or Large Fader to Multitrack Buses 1-24
- Routes Small Fader and Large Fader to stereo Program Buses (STA, STB, STC)
- PAN switch to engaged the pan-pot for both faders
    - Left/Right panning to stereo Program Buses
    - Odd/Even panning to Multitrack Buses



500 Series EQ Slot

Receives lower 200 slot output (COMP) when engaged (standard configuration)

- 500 series equalizers:
    - 550A 3-Band EQ
    - 550b 4-Band EQ
    - 560 10-Band Graphic EQ
    - 565 Filter Bank
- Any VPR Alliance approved 500 Series module (except preamps)



968 Input Module

<u>Channel Controls</u>

- Input Selection: Input **FLIP** and **ALT** Line In to Small Fader
- -6dB Line In Pad In/Out
- Signal Processing Routing: **COMP LRG** and **EQ SM**
- Output Routing:
    - Direct Output Routing: **DIR SM** and **DIR PRE**
    - Small Fader to Multitrack Buses: **24**
- Auxiliary Sends 1-12: Level, Pan, ON/off
    - Pre-fader Routing
    - Aux Send 5-12 from Small Fader

<u>Large and Small Fader Controls (on both paths)</u>

- Input **TRIM**
- **Ø**: Phase Reverse: Polarity inverter
- **FILT**: High-pass Filter In/Out: Sweep on Large Fader, fixed on Small Fader
- **INS**: Insert Return In/Out
- Pan-pot
- 45mm Small Fader
- SOLO
- Solo and channel routing SAFE mode
- MUTE

948 Large Fader

- 100mm long-throw automated resistive Large Fader

Channel Input

The channel has three audio input sources:

- Preamp Output: The output of the preamp in the upper 200 slot (typically 212 Mic Preamp)
- Line Input: Balanced line input, normalled to multitrack (DAW) returns
- Alternate Line Input (ALT): Alternate balanced line input

The Large and Small Fader paths can accept only one audio source at a time.

The output upper 200 slot Preamp is the default source the Small Fader path. The Line Input is the default source for the Large Fader. The inputs to the paths can be swapped on individual channels by engaging the **FLIP** button on the 968 Input Module. This will route the Preamp output to the Large Fader and the Line Input to the Small Fader. Engaging the **FLIP** button in the Master Section will swap fader path inputs on a global basis.

The Alternate Line Input can replace the Preamp output as the source for the Small Fader path. This provides additional balanced line-level inputs for recording and mixing. Engaging the **ALT** button on the 968 Input Module will route the Alternate Line Input to the Small Fader path on individual channels. Engaging the **SM ALT LINE** button in the Master Section will route the Alternate Line Inputs to the Small Fader paths on a global basis.

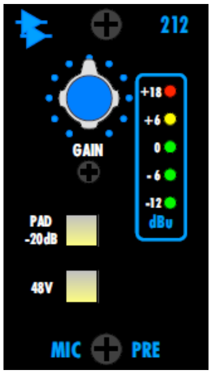
The Alternate Line Input can be routed to the Large Fader by engaging the **ALT** and **FLIP** buttons on individual channels or console-wide by the **SM ALT LINE** and **FLIP** buttons in the Master Section.

Upper 200 Slot

The upper 200 slot is designed to house the channel preamp. Two API 200 Series preamplifier modules are available for installation. Only the upper 200 slot is wired to accept a preamp, so only one preamp can be installed in each channel strip:

- 212 Microphone Preamplifier (standard configuration)
  OR
- 205 Direct Input

212 Mic Preamp



Features

- Wide range GAIN control
- -20dB Pad
- 48V phantom power switch
- Mic input transformer
- LED output Meter
- Uses API 2520 Op-Amp
- All discrete circuit
- Transformer coupled inputs & outputs to +28dBu

The tone of the 212 Mic Preamp finds its roots in the classic API 2488 series all discrete recording consoles, best known for the famed "LA" sound. This same mic preamp design is packaged into a compact module with features like a continuously adjustable Gain control and a 5-segment LED meter.

An audio chain is only as strong as its weakest link. Often, emphasis is placed on obtaining the highest quality microphones, while neglecting the integrity of the mic preamp. The API 212 incorporates the same circuit as the legendary API console input modules, dating from the 1970's up to and including the Model 1608, thus preserving its sonic qualities. This mic-pre articulates high frequencies with great detail, while delivering the big sounding, warm bottom end that API is famous for. Whether it's a live performance or the private setting of a studio session, the 212 will capture every nuance of the moment.

The 212 Mic Preamp's all discrete circuitry features the API 2520 op-amp and input and output transformers. A -20dB pad is provided, allowing the level feeding the mic transformer to be reduced by 20dB, while keeping the proper load on both the mic and transformer. 48V Phantom power can be

locally switched ON or OFF via the front panel switch. An LED meter indicates output level in dBu. The low noise floor, together with a clip point over +28dBu, makes this preamp suitable for the most extreme applications requiring the highest quality audio.
The controls for the 212 Mic Preamp function as follows:

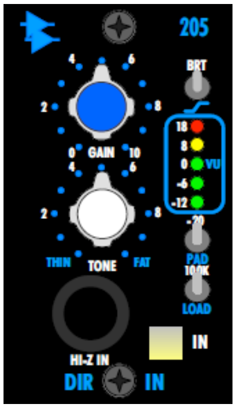
GAIN: Sets the preamplifier gain across a 55dB gain range

METER: The 5-segment LED meter indicates the mic pre output level in dBu

48V: Provides 48 Volt Phantom Power to the MIC PREAMP INPUT and MIC TIE-LINE patch points
- Illuminates when engaged

-20: Inserts a -20dB pad
- The level feeding the mic transformer is reduced by 20dB, while keeping the proper load on both the mic and transformer
- Illuminates when engaged

205 Direct Input



Features
- Wide range GAIN control
- Hi-Z ¼" input, like a tube amp
- Thin < > Fat TONE Control reduces low end mud
- BRIGHT Switch boosts the presence
- -20dB PAD lowers the input level
- 100 K LOAD switch changes the tone of instruments
- 5-segment LED Meter shows output level
- Transformer coupled output to +28dBu
- All Discrete Design

The API 205 Direct Instrument Input is specifically designed to accept a guitar or bass direct into it, without any loading on the pickups. The Gain Control is used to normalize the instrument's input level, up to a +4dBu output level. With the use of the unique TONE control, the low end "mud" can be reduced. Turning on the BRIGHT switch adds clarity, like the bright switch on any amp. The combination of these two functions may eliminate the need for any EQ on the instrument when recording, keeping the signal pure.

The input is designed specifically to load a pickup the same way that a tube guitar amp would, retaining all of the tone, and minimizing the loading effect on the pickup. This minimal path eliminates the need to use a standard direct box, which more times than not only converts the signal to a balanced mic level output, which still requires a mic input, equalizer, and fader to get the signal to a recordable level.

The 205 is capable of boosting any instrument to a hefty line level without additional amplifiers. With the 100 K LOAD switch, the tone of the pickup can be altered slightly, which tends to darken the high frequency content. The BRIGHT switch adds a treble boost in the same manner that most instrument amplifiers do, clarifying the instrument and adding presence to it without the need for outboard equalization. When inserting an instrument with an internal preamp or a line level instrument such as a sampler or a keyboard, the 205 has a -20dB PAD switch to drop the level without loading, thereby retaining the full range of the 205 and its features.

Also, the output of the 205 can be fed to multitrack or the input of a guitar amplifier. The artist can then be in the control room while the instrument amp is isolated in the studio. The 205 Direct Input makes use of the 2510 and 2520 op-amps and therefore exhibits the reliability, long life, and uniformity characteristic of API products.

The controls for the 205 Direct Input function as follows:

IN (in/out switch): Activates the module
- Illuminates when engaged

Hi-Z IN (¼" input jack): Instrument input jack
- High-impedance

- Unbalanced
- Blue illumination

-20 PAD: Inserts a -20dB pad at the instrument input

100K LOAD: Engages a 100K Ohm input load

- Changes the tone of the pickup and tends to darken the high frequency content.

GAIN: Sets the preamplifier gain from -40 to +20dB

TONE: Adjusts the tonal characteristics of inputs signal

- Adjusting the TONE control between “Thin” and “Fat” control can reduce “muddiness”
- THIN: -12dB at 150Hz
- FAT: Flat (+/- 0dB)

BRT (Bright): Activates a +6dB at 8kHz presence boost.

- The combination of the TONE and BRT functions may eliminate the need for any EQ on the instrument when recording, keeping the signal pure.

METER: The 5-segment LED meter indicates the 205 output level in dBu

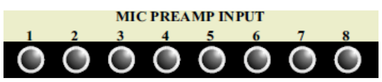
- Capable of boosting any instrument to a hefty line level without additional amplifiers.

Channel Input Patch Points

Depending on console configuration, there will be input patch points for 32, 48, 64, or 80 channels in the bay. In the interest of space, patch points for only 8 channels are shown below.

MIC TIE-LINE: Studio microphone outputs

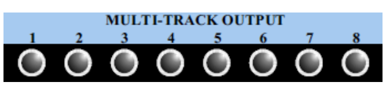
- Full-normal to MIC PREAMP INPUT
- Breaking output: Inserted patch cord will replace studio tie-line connection



MIC PREAMP INPUT: 212 Mic Preamp inputs

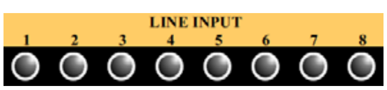
- Balanced, low-impedance, mic-level input
- Full-normal from MIC TIE-LINE patch points
  - Breaking input: Inserted patch cord will replace normalled connection
  - Not used with 205 Direct Input Module

CAUTION: *The MIC PREAMP INPUT patch point will carry 48-Volt phantom power when the **48V** button is engaged on the associated 212 Mic Preamp. Care should be used and **48V** should be always be disengaged before patching to these points.*



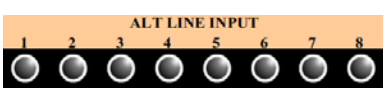
MULTITRACK OUTPUT: Multitrack recorder returns

- Half-normal to LINE INPUT
- Outputs split if patched



LINE INPUT: Main line-level inputs

- Balanced, low-impedance, line-level input
- Half-normal from MULTITRACK OUTPUT
  - Breaking input: Inserted patch cord will replace normalled connection



- ALT LINE INPUT: Secondary line-level inputs
- Balanced, low-impedance, line-level input
- Alternate input to the Small Fader path
  - Half-normal from owner specified source (optional)
  - Breaking input: Inserted patch cord will replace normalled connection

Patch points for alternate external sources (such as virtual instruments, extra DAW returns, etc.) normalled to the ALT LINE INPUT patch points are not provided in the standard configuration. However, the ALT LINE INPUTs are available via a multi-pin connector on the back of the patch bay to allow normal interfacing to these points.

Channel Signal Processing

Channel signal processing is provided by modules installed in the lower 200 slot and the 500 Series EQ slot. The lower 200 slot is usually designated for dynamic processing, although a filter set can be installed instead.

Lower 200 Slot

The lower 200 slot is designed to house one of the three API 200 Series signal processor modules:

- 225 Compressor/Limiter (standard configuration)
- 235 Noise Gate
- 215 Sweep Filter

The lower 200 slot is typically fitted with an API 225 Compressor/Limiter. Accordingly, the channel controls, master controls, and patch points associated with the routing of the lower 200 slot are labeled “COMP.” The channel signal flow is not designed to accommodate a preamp in this slot.

The two possible locations for the lower 200 slot are:

- Post-preamp/pre-EQ
- Post –EQ/pre Insert

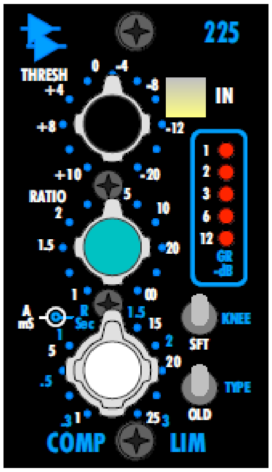
The lower 200 slot is routed to the Small Fader audio path by default. In the standard configuration, the lower 200 slot is located post-preamp and pre-EQ. As an option, the lower 200 slot can be located post-EQ and pre-Insert in the Small Fader signal flow. This option is set by internal links on the 968 Input Module and cannot be changed during normal operation. The initial locations of the lower 200 slots are specified by the owner when the console is ordered.

The processor in the lower 200 slot can be moved to the Large Fader path on individual channels by engaging the **COMP LRG** (compressor large) button on the 968 Input Module or console-wide by engaging the **COMP LRG** button in the Master Section.

The processor in the lower 200 slot can be bypassed console-wide by engaging the **COMP BYP** button in the Master Section. This provides a shorter signal path if the lower 200 slot signal processing is not needed. This option is not available on individual channels.

NOTE: The ***IN*** *button on the installed 200 module must be engaged for the installed 200 module to process audio. Disengaging the* ***IN*** *button will effectively bypass individual 200 modules on individual channels.*

225 Compressor/Limiter



Features

- Threshold control from +10 to -20dBu
- Variable compression ratio from 1 to infinity
- Release time: continuously variable from .3sec to 3sec
- Attack time: continuously variable from 1ms to 25ms
- "OLD" or "NEW" detection path and sound
- HARD or SOFT compression knee
- Auto-makeup Gain
- LED Gain Reduction meter
- API 2510 and API 2520 op-amps
- Transformer output to +28dBu

The API 225 Compressor/Limiter is ideal for all studio and broadcast applications. Regardless of the threshold or ratio settings, the output level always remains at unity. This unique feature allows for real-time adjustments without the need for changing the output level.

Both NEW ("feed-forward") and OLD ("feedback") methods are selectable, providing two choices of gain reduction. NEW gain reduction is typical of the newer feed-forward VCA type compressors that rely on RMS detectors for the gain control voltage. OLD or feedback method is what most of the classic

compressors used for the gain control circuit. There also is a SOFT or HARD selector. SOFT provides a more subtle compression resulting in a very natural, less compressed sound. HARD results in a more typical, sharp knee type compression that has a much more severe limiting effect.
Release time is adjusted by rotating the inner concentric R knob. Release time constants: continuously variable from .3sec to 3sec. Attack time is adjusted by rotating the outer concentric A knob. Attack time constants: continuously variable from 1ms to 25ms.

The 225 also has a side chain input for the detector amplifier.

The controls for the 225 Compressor/Limiter function as follows:

IN (in/out switch): Activates the module

- Illuminates when engaged

THRESH (Threshold): Sets the activation threshold from-20dB to +10dB

RATIO: Sets compression ratio

- 1 to ∞
- 1: 1:1…no gain reduction
- ∞: ∞:1…hard limiting

KNEE (HRD-SFT): Selects the "knee" of the onset of compression

- HRD (Hard): Fast curve (more useful for limiting applications)
- SFT (Soft): Slower curve (acts as an "over-easy" compressor)

AT (Attack Time): Sets attack time

- 1ms to 25ms

REL (Release Time): Sets release time

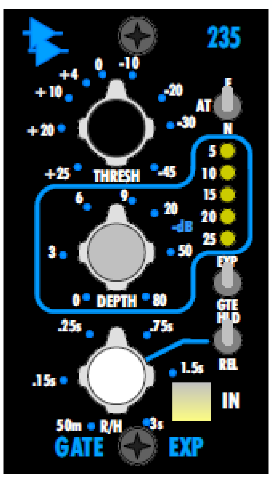
- .3 seconds to 3 seconds

TYPE: Selection between "new" and "old" types of compressors.

- NEW: "Feed-forward" compressor
- OLD: "Feed-back" compressor

-dB (Gain-reduction Meter): LED meter indicates the amount of gain reduction

235 Noise Gate / Expander



Features

- Threshold control from -45 to +25dBu
- Variable Fine Adjustable Depth to -80dB
- Adjustable Release/Hold Time, 50mS to 3sec
- 2:1 Expander or Noise Gate selection
- Release or Hold switch for release control knob
- Always at unity gain, regardless of settings
- LED Gain Reduction meter
- Uses the API 2510 and API 2520 Op-Amps
- Transformer Output, to +28dBu

The API 235 Noise Gate/Expander has the ability to reduce noise in any type of program. The 235 can open unusually fast, without losing any part of the sound. Its extreme flexibility, repeatable settings, and superb sound make it ideal for all studio, live sound, and broadcast applications.

The 235 is one of the fastest noise gates manufactured today. When an engineer is faced with program material with a high level of background noise, the 235 can either Gate the signal at a preset Threshold, or with the use of the Expander function, can reduce the background noise to an almost undetectable level. This can be achieved without the typical loss of the program material caused by slow triggering of the signal, or "pinging" caused by false triggering.

With a wide range of adjustment, the 235 can operate from -45 to +25dBu, fitting into any audio situation regardless of program level. Once the Threshold level is set, the Attack time can be selected to react faster than 100uS or slightly slower than 25mS to reduce false triggering. Although the Depth control has a full range of -80dB, the scale is expanded in the first half of rotation so 0 to 9dB is available for fine-tuning of subtle, undetectable gating. The second half of rotation is from 10 to 80dB for more drastic noise reduction.

The Expander function uses a 1:2 ratio, allowing the signal to "sneak up" to the full signal level without any loss of "under threshold" vocal or percussion nuances. Setting the threshold in the Gate function to the desired level, then switching to the Expander mode is the perfect setup.

For special gating functions such as the famed "gated snare reverb", the Release function can be switched to Hold, and the Release control becomes the Hold time. In both scenarios, the unselected function has a set time of 100mS.

The 235 makes use of the 2510 and 2520 op-amps and therefore exhibits the reliability, long life, and uniformity which are characteristic of all API products.

The controls for the 235 Noise Gate/Expander function as follows:

IN (in/out switch): Activates the module
- Illuminates when engaged

THRESH (Threshold): Sets the activation threshold
- Variable from +25 to -45dB.

DEPTH: Sets the amount of attenuation (gain reduction) from 0 to -80dB. Although the Depth control has a full range of -80dB, the scale is expanded in the first half of rotation so 0 to 9 dB is available for fine-tuning of subtle, undetectable gating. The second half of rotation is from 10 to 80dB for more drastic noise reduction.

AT (Attack): Selects attack time. Once the Threshold level is set, the Attack time can be selected to react faster than 100 micro seconds or slightly slower than 25 milliseconds to reduce false triggering.
- F: Fast (less than 100 microseconds)
- M: Medium (25ms)

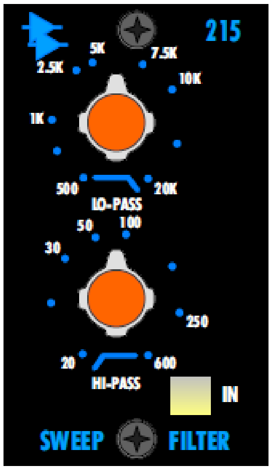
R/H (Release/Hold) Time: Sets release or hold time
- 50ms to 3 seconds
- The Release or Hold function is determined by the HLD/REL switch

HLD/REL (Hold/Release): The Release function can be switched to Hold, and the Release control becomes the Hold time. In both functions, the unselected one has a set time of 100ms.
- HLD (Hold): The R/H control determines Hold time (Release is set to 100ms)
- REL (Release): The R/H control determines Release time (Hold is set to 100ms)

EXP/GTE (Expander/Gate): Selects the Expander or Gate function
- EXP (Expander): Sets the ratio to 1:2
- GTE (Gate): Sets the gate function

-dB (Gain-reduction Meter): LED meter indicates the amount of gain reduction

215 High-Pass, Low-Pass Sweep Filter



Features

- Wide range 6dB per octave LO-PASS control
- Wide range 12dB per octave HI-PASS control
- Balanced input stage
- Passive filter design for smooth tone
- Unity gain throughout
- Hard Wire BYPASS switch
- Transformer coupled output to +28dBu
- All discrete design

The API 215 is a unique passive, sweepable filter, designed specifically to contour the sound to preserve the natural tone of the signal. Its extreme flexibility, repeatable settings and superb sound make it ideal for all studio, live sound and broadcast applications.

The 215 design is a passive low pass filter with a slope of only 6dB per octave, and a passive dual high pass filter with a slope of 12dB per octave. The filters are isolated from each other with the same discrete transistor buffer used in the famous 550 series equalizers. This minimizes interaction between the filters, as well as providing a low impedance source for the filter following the buffer.

The filters are both continuously adjustable, with a range from 20Hz to 20kHz in two bands. The low-pass filter has a range from 500Hz to 20kHz, and the high-pass filter has a range from 20Hz to 600Hz. This covers a broad range of frequencies throughout the entire audio spectrum.

Because of the subtle nature of the 215 filter, it finds a home with uses like rolling off the low end of a hi-hat, where a natural sound is desired, not the usual "phase-shifter" sound of a 18 to 24dB per octave filter found on most consoles. It can also be used to thin out room mics, without the complete loss of low end, which again usually results from steep filter circuits.

The controls for the 215 High-pass, Low-pass Sweep Filter function as follows:

IN (in/out switch): Activates the module

- Illuminates when engaged

LO-PASS: Sets the frequency of the low-pass filter – sweepable from 500Hz to 20kHz

HI-PASS: Sets the frequency of the high-pass filter – sweepable 20Hz to 600Hz

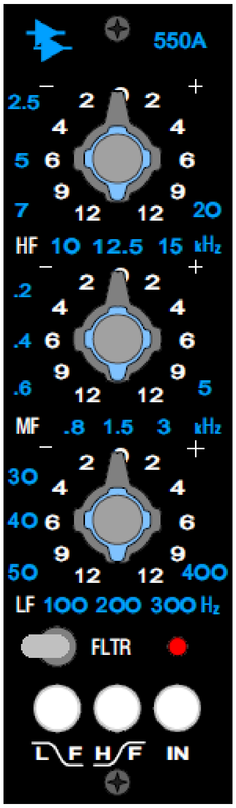
500 Series Equalizer Slot

The Legacy AXS channel strip is 1.5" inches wide in order to accommodate a VPR Alliance approved 500 Series slot. This slot is designed to house the channel equalizer. The 500 Series EQ slot may be fitted with one of the four API equalizer/filter choices or any VPR Alliance approved 500 Series module, except preamps (the channel signal flow is not designed to accommodate a preamp in this slot.)

API provides four spectral processor options for the 500 Series EQ slot:

- 550A Discrete 3-Band EQ (standard configuration for most channels)
- 550b Discrete 4-Band EQ
- 560 Graphic EQ (standard configuration for some channels)
- 565 Filter Bank

550A Discrete 3-Band Equalizer



Features

- 3 bands of classic API equalization
- Each band offers 7 API selected frequency centers
- Reciprocal and repeatable filtering
- Maximum 12dB of boost/cut per band
- High and Low EQ bands offer shelf/peak switching
- "Proportional Q" narrows filter Q at extremes
- Traditional API fully discrete circuit design
- High headroom: +30dB clip level

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

Forty years later, the 550A remains the standard against which other EQs are measured, and it has played a major role in the recording industry for decades. Still copied, but never duplicated, the 550A became API's standard channel module EQ when the company began manufacturing consoles in 1971. With virtually all existing units spoken for, popular demand for this EQ resulted in API finally resuming production in 2004.

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

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

The controls for the 550A EQ function as follows:

IN (in/out switch): Activates the channel EQ

- Hard bypass when not engaged
- A LED indicator illuminates when engaged

HF (High-Frequency): Peaking/Shelving Switch: Selects between peaking and shelving EQ curves for the high-frequency band.

Frequency Selection (Center Knob): Selects the center frequency for the selected band. Center frequencies vary by band:

- High: 2.5kHz, 5kHz, 7kHz, 10kHz, 12.5kHz, 15kHz, 20kHz
- Mid: 200Hz, 400Hz, 600Hz, 800Hz, 1.5kHz, 3kHz, 5kHz
- Low: 30Hz, 40Hz, 50Hz, 100Hz, 200Hz, 300Hz, 400Hz
- Frequencies are indicated in blue numbers

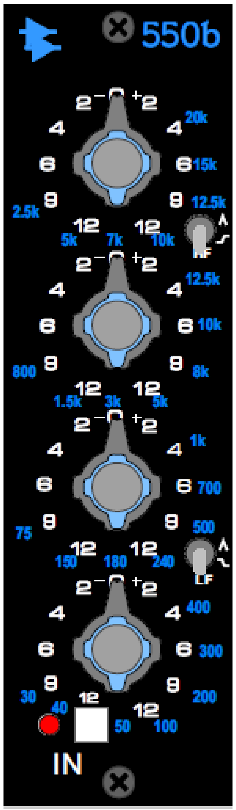
Boost and Cut (Ring): Sets the amount of boost or cut for the selected band.

- +12dB of boost or -12dB of cut
- Decibels are indicated in white numbers

LF (Low-Frequency): Peaking/Shelving Switch: Selects between peaking and shelving EQ curves for the low-frequency band

HF (High-Frequency): Peaking/Shelving Switch: Selects between peaking and shelving EQ curves for the high-frequency band

550b Discrete 4-Band EQ



Features

- 4 bands of the famous API equalization
- Each band offers 7 API selected frequency centers
- Reciprocal and repeatable filtering
- 12dB of boost/cut per band
- High and Low EQ Bands offer shelf/peak switching
- "Proportional Q" narrows filter Q at extremes
- Traditional API fully discrete circuit design
- High headroom: +30dB clip level

Based on API's original 550 from the late '60s, the API 550b is a continuation of the EQ that played a major role in the history of music recording, but with an additional filter band and several new frequencies. Incorporating API's exclusive circuitry and proprietary components (such as the legendary API 2520 Op Amp), the 550b artfully blends the past with the present. So many hit records still depend on the unique 550 sound that engineers around the world find it to be an invaluable tool. In fact, the 550b design has been taken from the original blue prints and spec control drawings from the API archives. It is unlike any other EQ you will ever use.

Rather than offer a huge assortment of complex features, the API 550b provides exactly the right number of controls. Its four EQ bands are overlapped significantly to aid in dual roles as problem solver and sweetening device with each band offering seven switchable filter frequencies that span four-to-five octaves. These frequencies, purposely selected to be musical rather than numeric, were selected by an experienced "who's who" list of the industry's most proficient engineers.

Making use of API's "Proportional Q," an innovation designed by Saul Walker in the '60s, the 550b intuitively widens the filter bandwidth at minimal settings and narrows it at higher settings without the need for additional bandwidth controls. This unique feature minimizes the "phase-shift" sound found in many equalizers. In addition, the reciprocal nature of the 550b enables the user to "undo" what has been done previously with exact precision.

The benefits of the API 550b are most obvious to those who work with EQ on a continuous basis. If major tonal restructuring is required, the extraordinary headroom made possible with API's 2520 Op Amp offers the predictable and warm analog performance, even under duress. With a surprisingly wide range of tonal variations, the 550b is an invaluable and professional audio tool with great flexibility and excellent sonic ability.

The controls for the 550b EQ function as follows:

IN (in/out switch): Activates the channel EQ

- Hard bypass when not engaged
- A LED indicator illuminates when engaged

HF (High-Frequency): Peaking/Shelving Switch: Selects between peaking and shelving EQ curves for the high-frequency band.

Frequency Selection (Center Knob): Selects the center frequency for the selected band. Center frequencies vary by band:

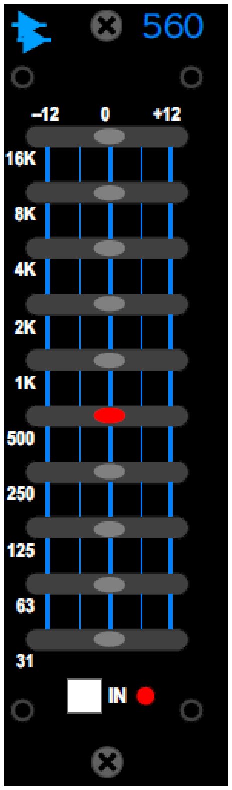
- High: 2.5kHz, 5kHz, 7kHz, 10kHz, 12.5kHz, 15kHz, 20kHz
- High-Mid: 800Hz, 1.5kHz, 3kHz, 5kHz, 8kHz, 10Hz, 12.5kHz
- Low-Mid: 75Hz, 150Hz, 180Hz, 240Hz, 500Hz, 700Hz, 1kHz
- Low: 30Hz, 40Hz, 50Hz, 100Hz, 200Hz, 300Hz, 400Hz
- Frequencies are indicated in blue numbers

Boost and Cut (Ring): Sets the amount of boost or cut for the selected band.

- +12dB of boost or -12dB of cut
- Decibels are indicated in white numbers

LF (Low-Frequency): Peaking/Shelving Switch: Selects between peaking and shelving EQ curves for the low-frequency band.

560 10-Band Graphic Equalizer



Features
- 10 bands of API proprietary equalization
- Familiar graphics operation on one octave centers
- 12dB of boost/cut per band
- "Proportional Q" narrows filter Q at extremes
- Additional resolution within the ±4dB region
- Center detent for reliable reset
- Silent bypass button
- Based on the original 1969 API 560 EQ

Originally conceived for use in API consoles of the '60s and '70s, the 560 is a unique device designed to accomplish tasks that no other EQ can. Extremely fast to set and reset using accurate zero detents, the curve shaping potential of the 560 remains unmatched. With a wide range of 500 mounting options, from racks to consoles, the 560 proves an invaluable asset to all critical performance applications. Based on API's original 560 EQ, the current production 560 has improved resolution in the ±4 dB area and possesses our exclusive circuitry and proprietary components, including the API 2520 Op Amp.

The extraordinary headroom made possible with the 2520 offers consistent analog performance even when using radical EQ curves. Of course, the 2520's ability to drive low impedance loads is key when it is paired with API's custom output transformers. The results are quite audible with better low frequency reproduction and tighter imaging, which gives you that legendary API "punch in your gut" sound.

The 10 precision EQ bands make the 560 ideal for signal sweetening and room tuning. A great companion to a parametric EQ, the 560 utilizes API's unique "Proportional Q" design introduced during the '60s. This design intuitively widens the filter bandwidth at lower boost/cut levels and narrows it at higher settings. Additionally, boost and cut characteristics are identical, allowing previous actions to be undone if desired.

Reliable, durable and uniform, the API 560 EQ delivers that "one-of-a-kind API sound" with precision easy set filtering and high headroom in a compact package. If you want the sound of classic American music in an easy-to-set graphic EQ package, you want the API 560.

10-Bands with the following center frequencies:
- 31Hz
- 63Hz
- 125Hz
- 250Hz
- 500Hz
- 1kHz
- 2kHz
- 4kHz
- 8kHz
- 16kHz

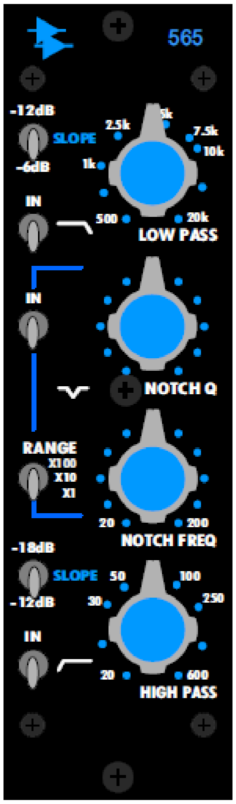
The controls for the 560 10-Band Graphic EQ function as follows:

IN (in/out switch): Activates the channel EQ
- Hard bypass when not engaged
- A LED indicator illuminates when engaged

Slider (Boost and Cut): Sets the amount of boost or cut for the selected band

- +12dB of boost and -12dB of cut
- 0dB detent

565 Filter Bank



Features

- Sweepable low-pass filter (500Hz to 20kHz, -6 or -12dB/8va slope)
- Sweepable high-pass filter (20Hz to 400Hz, -12 or -18dB/8va slope)
- Variable notch filter:
  - Adjustable Q
  - Fully sweepable between 20Hz and 20kHz
- Individual bypass switches for each filter section
- Transformer balanced output to +28dBu
- Traditional API fully discrete circuit design
- Uses API 2520 and 2510 Op-Amps as well as the same discrete transistor buffers used in the famous 550 series equalizers

The 565 Filter Bank module has circuits that are true to the musical filters of the famed 215 module found in large format API consoles.

The controls for the 565 Filter Bank function as follows:

IN (low-pass in/out switch): Activates the low-pass filter when engaged

- Hard bypass when not engaged

LOW PASS (frequency): Sets the frequency of the low-pass filter

- Sweepable from 500Hz to 20kHz

SLOPE (Low-pass): Sets the slope of the low-pass filter

- -6dB/8va or -12dB/8va

IN (Notch in/out switch): Activates the notch filter when engaged.

- Hard bypass when not engaged

NOTCH Q: Sets the Q (quality factor) of the notch filter

NOTCH RANGE: Multiplies the Notch FREQ by the following factors:

- x1
- x10
- x100

NOTCH FREQ (frequency): Sets the frequency of the notch filter.

- Sweepable from 20Hz to 20kHz
  Notch frequency determined by selected frequency and NOTCH RANGE factor

IN (High-pass in/out switch): Activates the high-pass filter when engaged.

- Hard bypass when not engaged

HIGH PASS: Sets the frequency of the high-pass filter.

- Sweepable 20Hz to 600Hz

SLOPE (High-pass): Sets the slope of the high-pass filter.

- -12dB/8va or -18dB/8va

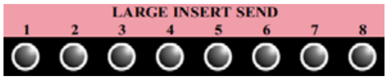
Channel Signal Processing Patch Points

The standard configuration provides a complete set of patch points for the lower 200 slot. There are no direct 500 Series EQ slot patch points.

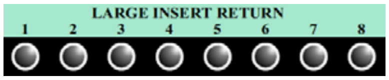
Depending on console configuration, there will be signal processing patch points for 32, 48, 64, or 80 channels in the bay. In the interest of space, patch points for only 8 channels are shown below.

Channel Insert Patch Points

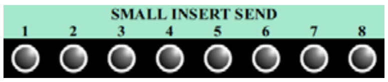
A set of INSERT SEND and INSERT RETURN patch points for each fader path (Large and Small) are provided in the Channel patch bay.



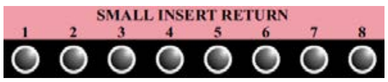
<u>LARGE INSERT SEND</u>: Insert Send from Large Fader path
- Always active
- Pre-fader, post-EQ



<u>LARGE INSERT RETURN</u>: Insert Return to the Large Fader path
- Patched signal is inserted in the Large Fader path when the **INS** button is engaged
  - Pre-fader, post-EQ



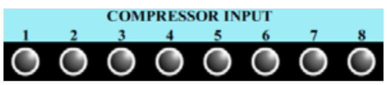
<u>SMALL INSERT SEND</u>: Insert Send from Small Fader path
- Always active
- Pre-fader, post-EQ



<u>SMALL INSERT RETURN</u>: Insert Return to the Small Fader path
- Patched signal is inserted in the Small Fader path when the **INS** button is engaged
  - Pre-fader, post-EQ

Lower 200 Slot Patch Points

COMP IN, SIDE CHAIN, and COMP OUT patch points in the Channel patch bay provide access to the lower 200 slot signal processor interfacing.



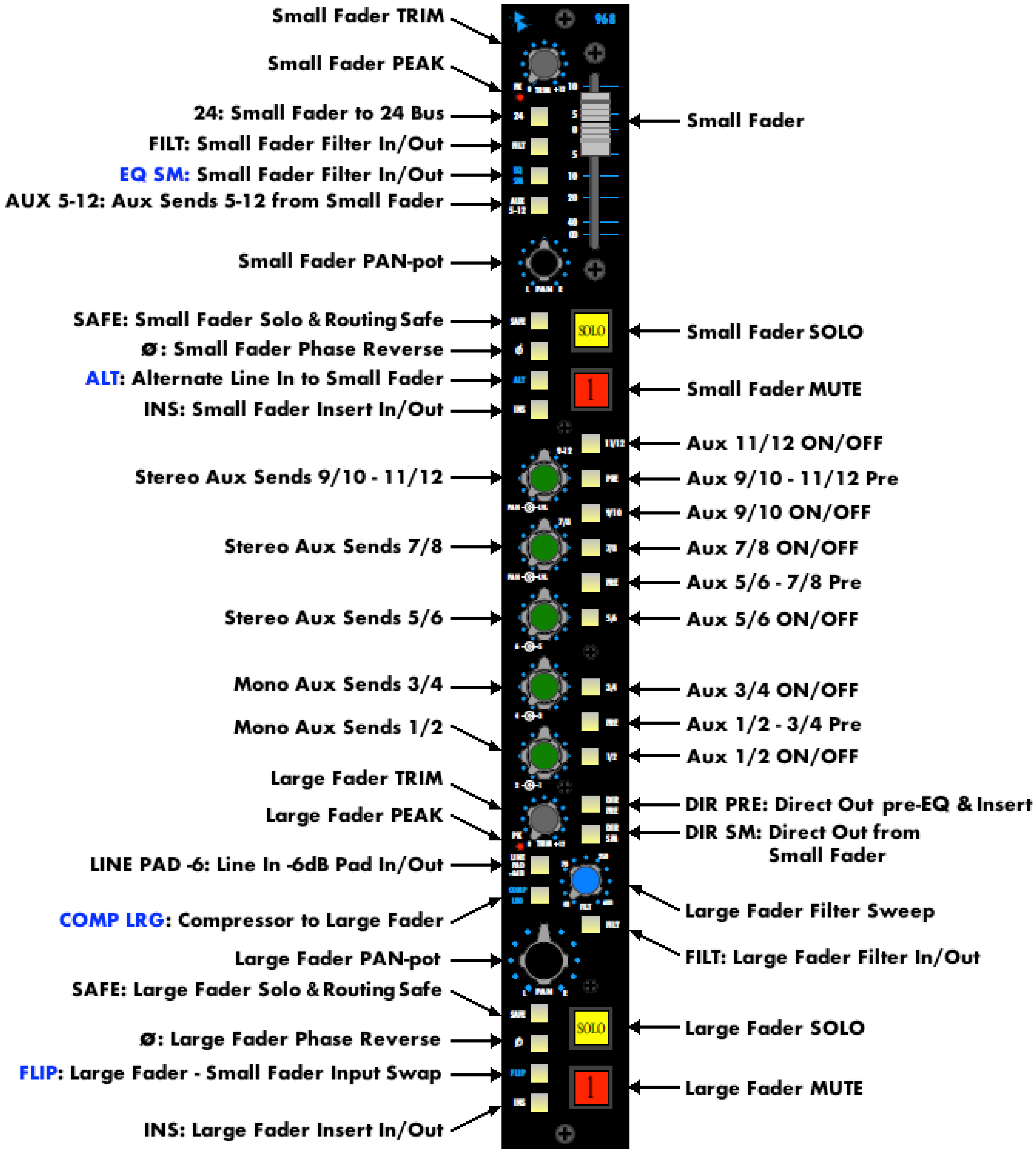
<u>COMPRESSOR INPUT:</u> Lower 200 slot input
- Balanced, low-impedance, line-level input
- Breaking input: Inserted patch cord will replace normalled connection

- <u>SIDE CHAIN (input):</u> Lower 200 slot detection path input
- Balanced, low-impedance, line-level input
- Full-normal from COMPRESSOR INPUT
  - Breaking input: Inserted patch cord will replace normalled connection

<u>COMPRESSOR OUTPUT:</u> Lower 200 slot output
- Balanced, low-impedance, line-level output
- Outputs split when patched

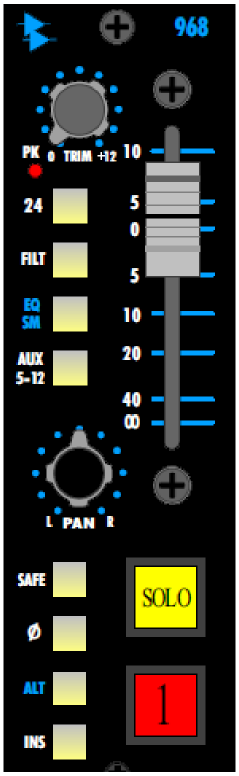
Channel Faders and Controls

968 Input Module



The illustration above shows the 968 Input Module with a brief description of each control. This serves as a quick reference guide. A more detailed description of each control follows.

Small Fader Path Controls



The 968 Small Fader audio path is a complete audio channel designed to serve the following functions:

- Routing microphone and instrument inputs to a multitrack recorder OR monitor mix during multitrack recording
- Additional inputs during mix down
- Retuning effects, virtual tracks, and other sources during mix down

The Small Fader path is fed from the upper 200 slot Preamp Output by default.

Accordingly, the Small Fader audio path is equipped with the following features:

- Alternate Line Input source
- Channel TRIM
- Ø Phase Reverse (polarity inverter)
- Default path for lower 200 slot (COMP)
- Assignable EQ
- Insert Send and Return
- High-pass filter (50Hz, 12dB/8va)
- Assignable to Direct Output
- Assignable to Multitrack Bus access
- Odd-even panning across Multitrack Buses
- Assignable to stereo Program Buses A, B, and C
- Panning to stereo Program Buses
- 45mm Small Fader
- MUTE
- SOLO
- Solo and channel routing SAFE mode
- Peak Indicator
    - Access to Channel Meter (depending on Master Meter Selection)

The controls for the Small Fader path function as follows:

TRIM: Sets the amount of pre-fader level Trim (boost)

- 0 to +12dB of gain
- Located pre-fader, so the added gain is reflected in all post-fader stages (Aux Sends, Solos, Bus Assignments and Direct \Output)

PK (Peak): A red LED illuminates when the preset Peak Reference level is reached in the Small Fader audio path

- The peak level is selected by the PEAK REFERENCE selector in the Master Section

24: Routes the Small Fader Path to the Multitrack Bus assignments

- Disables the Large Fader signal routed to the Multitrack Bus assignments
- Illuminates when engaged

FILT (Filter): Inserts a 50Hz high-pass filter in the Small Fader path

- 2nd order filter: 12dB/8va slope
- Illuminates when engaged

EQ SM: Moves the 500 Series equalizer into the Small Fader path

- The 500 Series EQ slot is routed to the Large Fader path by default
- The EQ is always in one path or the other, but the **IN** button must be engaged to be used
- Illuminates when engaged

AUX 5-12: Routes the Small Fader Path to Stereo Auxiliary Sends 5-12

- Aux Sends 5-12 are fed from the Large Fader path by default
- Provides Aux Sends from the Small Fader path
- Illuminates when engaged

SAFE: Activates the Safe mode for the Small Fader path

- The SAFE button protects the Small Fader from being muted when the Solo-In-Place function is active and another channel is soloed
- Protects from global changes to channel routing (input and compressor)
- Illuminates when engaged

Ø (Phase Reverse): Inserts a polarity inverter into the Small Fader path

- Illuminates when engaged

PAN: This is the pan pot control for the Small Fader path

- -2.5dB pan law
- Left/Right panning to stereo Program Buses A, B, and C
- Odd/Even panning to Multitrack Buses 1-24
- The **SM PAN** button in the 624 Bus Assignment module must be engaged for the pan-pot to work

ALT: Routes Alternate Line Input to the Small Fader path input

- The upper 200 slot Preamp Output is the default input for the Small Fader path
- Will engage on all channels when the **SM ALT LINE** button is engaged in the Master Section
- Illuminates when engaged

INS (Insert): Activates the Small Fader path Insert Return

- Insert Send is fed from the output of the 500 Series EQ slot
- Insert Send is always active
- Insert Return is located pre the Small Fader
- The Insert Return is active only when the **INS** button is engaged
- Controlled by automation
- Illuminates when engaged

SMALL FADER: Level control for the Small Fader path

- 45mm resistive fader
- +10dB of gain
- -∞dB of attenuation
- Unity gain when set to 0dB
- Global 0dB Fader Bypass (in Master Section)
- Not automatable

*NOTE: A 0dB Small Fader bypass can be applied to all channels by engaging the **0dB SM** button in the Master Section. This feature provides is useful an additional Small Fader gain stage is not needed or when using DAW automation.*

SOLO: Activates the selected solo function for the Small Fader path. The following solo functions may be selected via the Master Section:

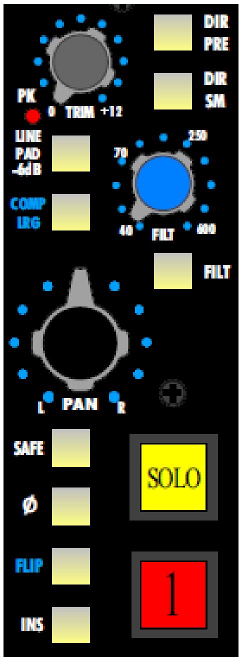
- Pre-fader Listen (PFL): Non-destructive, mono
- Pre-EQ & Insert Listen (PFL2): Non-destructive, mono
- After Fader Listen (AFL): Non-destructive, panned
- Solo-In-Place (SIP): Destructive, panned, post-fader
- Will disengage when the **SOLO CLR** button is engaged in the Master Section
- Illuminates in yellow when engaged

NOTE: AFL is the default solo mode.

MUTE: Cuts the Small Fader path signal

- The MUTE button is the on/off switch for the Small Fader audio path
- Can be engaged by Solo-In-Place solos
- Controlled by automation
- Illuminates in red when engaged

Large Fader Path and Channel Routing Controls



The 968 Large Fader audio path is the primary audio channel designed to serve the following functions:

- Primary signal path during mixing with 100mm, automated fader
- Monitor mix during multitrack recording

OR

- Routing microphone and instrument inputs to a multitrack recorder
- Retuning effects, virtual tracks, and other sources during mix-down

The Large Fader path is fed from the Line Input by default.

Accordingly, the Large Fader audio path is equipped with the following features:

- -6dB Line Input Pad
- Channel Trim
- Ø Phase Reverse (Polarity Inverter)
- Default path for EQ
- Assignable lower 200 slot (COMP)
- Insert Send and Return
- Sweepable high-pass filter
- Default source for Direct Output
- Default source Multitrack Bus assignments
- Odd-even panning across Multitrack Buses
- Assignable to stereo Program Buses A, B, and C
- Panning to stereo Program Buses
  - 100mm, long-throw, automated Large Fader
  - MUTE
  - SOLO
  - Solo and channel routing SAFE
  - Peak Indicator
  - Access to Channel Meter (depending on Master Meter Selection)

The channel routing controls are also included in this section and function as follows:

FLIP: Swaps the input sources between the Large and Small Fader paths

- The upper 200 slot Preamp Output is the default source for the Small Fader path
- The Line Input is the default source for the Large Fader path
- Will engage on all channels when the **FLIP** button is engaged in the Master Section
- Illuminates when engaged

DIR PRE (Direct Output Pre): Routes the pre-EQ & Insert signal from the assigned fader path to the Direct Output.

- The Direct Output is the fed post-fader signal from the assigned path by default
- Illuminates when engaged

DIR SM (Direct Output Small Fader): Routes the Small Fader path to the Direct Output

- The Direct Output is fed from the Large Fader path by default
- Illuminates when engaged

The controls for the Large Fader path function as follows:

TRIM: Sets the amount of pre-fader level Trim (boost)

- 0 to +12dB of gain
- Located pre-fader, so the added gain is reflected in all post-fader stages (Aux Sends, Solos, Bus Assignments and Direct \Output)

PK (Peak): A red LED illuminates when the preset Peak Reference level is reached in the Small Fader audio path

- The peak level is selected by the PEAK REFERENCE selector in the Master Section

LINE PAD -6dB: Inserts a -6dB pad immediately after Line Input

- The -6dB Line Pad will reduce distortion and increase headroom in the audio path when the level from the multitrack recorder return is hot enough to overload the Line Input
- Illuminates when engaged

COMP LRG (Compressor to Large Fader): Moves the 200 slot processor into the Large Fader path

- The lower 200 slot is routed to the Small Fader path by default
- Will engage on all channels when the **COMP LRG** button is engaged in the Master Section
- Illuminates when engaged

*NOTE: Engaging the **COMP BYP** button in the Master Section will bypass the lower 200 slot (COMP) on all channels console-wide.*

PAN: This is the pan pot control for the Large Fader path

- -3.5dB pan law
- Left/Right panning to stereo Program Buses A, B, and C
- Odd/Even panning to Multitrack Buses 1-24
- The **LG PAN** button in the 624 Bus Assignment module must be engaged for the pan-pot to work

SAFE: Activates the Safe mode for the Large Fader path

- The SAFE button protects the Large Fader from being muted when the Solo-In-Place function is active and another channel is soloed
- Protects from global changes to channel input source
- Illuminates when engaged

Ø (Phase Reverse): Inserts a polarity inverter into the Large Fader path

- Illuminates when engaged

INS (Insert): Activates the Large Fader path Insert Return

- Insert Send is fed from the output of the 500 Series EQ slot
- Insert Send is always active
- Insert Return is located pre the Large Fader
- The Insert Return is active only when the **INS** button is engaged
- Controlled by automation
- Illuminates when engaged

FILT (Filter): Inserts a high-pass filter in the Large Fader path

- 2^{nd} order filter: 12dB/8va slope
- Sweepable 40Hz-600Hz
- Illuminates when engaged

SOLO: Activates the selected solo function for the Large Fader path. The following solo functions may be selected via the Master Section:

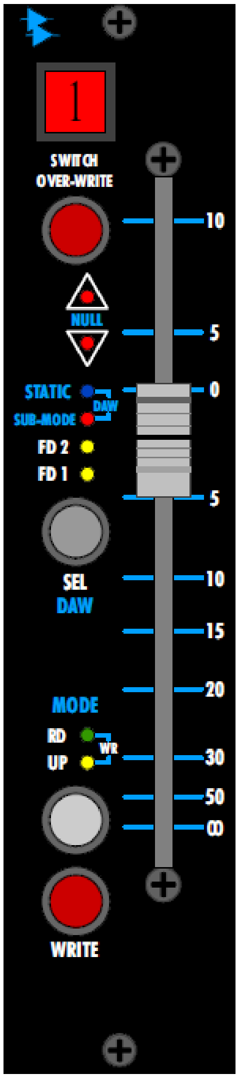
- Pre-fader Listen (PFL): Non-destructive, mono
- Pre-EQ and Insert Listen (PFL2): Non-destructive, mono
- After Fader Listen (AFL): Non-destructive, panned
- Solo-In-Place (SIP): Destructive, panned, post-fader
- Illuminates in yellow when engaged

NOTE: AFL is the default solo mode.

MUTE: Cuts the Large Fader path signal

- The MUTE button is the on/off switch for the Large Fader audio path
- Can be engaged by Solo-In-Place solos
- Controlled by automation
- Illuminates in red when engaged

Large Fader



<u>AUTOMATION MUTE:</u> (Channel Mute) Cuts the Large Fader audio output

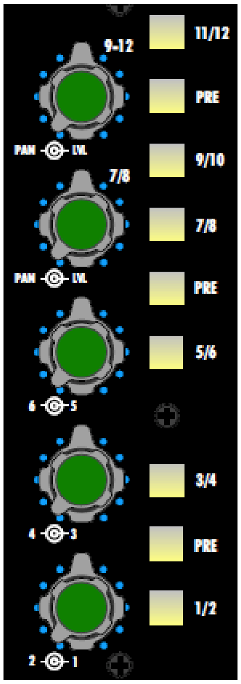
- The MUTE button is the on/off switch for the Large Fader
- Activates the 968 MUTE button
- Not activated by the 968 MUTE button
- Controlled by automation
- Illuminates in red when engaged

<u>LARGE FADER:</u> Controls the output level of the Large Fader audio path

- Full-size 100mm resistive fader
- +10dB of gain
- -∞dB of attenuation
- Unity gain when set to 0dB
- 0dB fader bypass (in via automation system)
- Controlled by automation

NOTE: A 0dB Large Fader bypass can be applied via the automation system. This feature is useful if an additional Large Fader gain stage is not needed or when using DAW automation/worksurface control.

Auxiliary Sends



The Legacy AXS console provides a powerful Auxiliary Send system that provides a complete set of options for Cue Sends (headphone feeds) and Effects Sends.

There are twelve (12) Auxiliary Buses, all of which may be used simultaneously. The 12 Aux Buses are fed from the channels using the following sends:

- Six (6) Mono Sends (1-6)
- Three (3) Stereo Sends (7-12)

Each of the Auxiliary Sends feed their respective Auxiliary Summing Bus.

Aux Sends 9/10 and 11/12 can be used simultaneously, but will share the same level and pan-pot.

All Aux Sends are sourced post-fader from the Large Fader path by default.

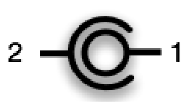
All Aux Sends may be switched to pre-fader in groups of four sends (1-4, 5-8, 9-12) by engaging the **PRE** button associate with that group of four sends.

Aux Sends 5-12 may be switched from the Small Fader path on individual channels by engaging the **AUX 5-12** button in the 968 Small Fader controls.

Mono Auxiliary Sends 1-6 feed the 265 Mono Aux Masters in the Master Section 1-6. Stereo Auxiliary Sends 7-12 feed the 265 Stereo Aux/Cue Masters 7/8, 9/10, and 11/12 in the Master Section.

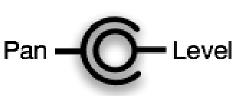
Any Aux Send can be used for any purpose. Mono Aux Sends 1-6 are primarily used as effects sends during recording and mixing. Stereo Aux Sends 7/8, 9/10, and 11/12 are typically used as cue sends during recording and as stereo effects sends during mixing.

Mono Sends (1-6)



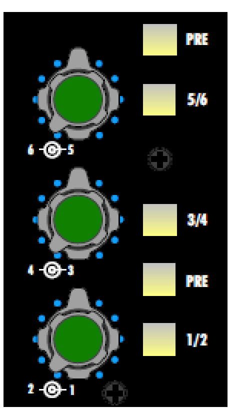
Each pair of mono Aux Sends uses dual mono “concentric” pots for level control. The odd numbered send is controlled by the inside or top pot and the even numbered send is controlled by the outside ring or bottom pot.

Stereo Sends 7-12



Stereo Aux Sends use “concentric” pots for level and pan. The stereo send LEVEL is controlled by the inside or top pot and the PAN is controlled by the outside ring or bottom pan-pot. Because stereo Aux Sends pass through a pan-pot, their levels at the extremes of the pan control are equal to that of the other sends, and are 2dB or 2.5dB down when the pan-pot is centered.

Mono Aux Sends 1-6



Aux Sends 1-6 are mono auxiliary sends typically used as effects sends.

Aux Sends 1-6 are fed post-fader from the Large Fader path by default.

<u>**ON** (on/off):</u> On/off switch for each pair of mono Aux Sends

- Illuminates when the pair is ON

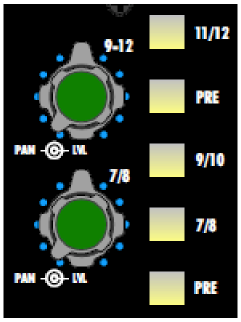
<u>**1-6** (Level):</u> Mono level potentiometer for each mono send

<u>**PRE** (Pre-Fader):</u> Engaging the **PRE** button will switch sends 1-4 and 5-8 to be sourced pre-fader from the assigned path.

- Normally fed post-fader from the Large Fader path
- Illuminates when engaged

*NOTE: Aux Sends 5-12 can be fed from the Small Fader path by engaging the **AUX 5-12** button in the 968 Small Fader controls.*

Stereo Aux Sends 7-12



Aux Sends 7-12 are organized as three stereo Aux Sends that may be used as a cue sends for headphones during recording or effects sends during mix down.

Stereo Aux Sends 9/10 and 11/12 can be used simultaneously, but will share the same level and pan-pot.

Aux Sends 7-12 are fed post-fader from the Large Fader signal by default.

ON (on/off): On/off switch for each stereo Aux Send
- Illuminates when the stereo send is ON

7/8, 9-12 (LVL): Stereo level potentiometer for each stereo send
- The inside or top pot is the level control for each stereo pair

7/8 (PAN): Stereo pan-potentiometer for each stereo send
- The outside ring or bottom pot is the pan-pot for each stereo send
- -2.5dB pan law

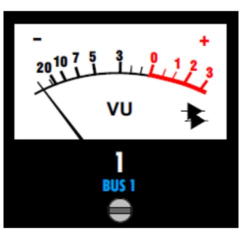
9-12 (PAN): Stereo pan-potentiometer for each stereo send
- The outside ring or bottom pot is the pan-pot for each stereo send
- -2.0dB pan law

PRE (Pre-Fader): Engaging the **PRE** button will switch sends 5/6-7/8 or 9/10-11/12 to be sourced pre-fader from the assigned path
- Normally fed post-fader from the Large Fader path
- Illuminates when engaged

NOTE: Aux Sends 5-12 can be fed from the Small Fader path by engaging the AUX 5-12 button in the 968 Small Fader controls.

Channel VU Meters

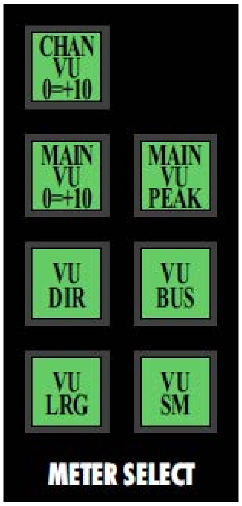
A VU Meter is provided for each channel, along with a LED Peak Indicator for each audio path (Small Fader and Large Fader).



VU (Volume Unit)**:** Channel level indicator

The channel VU Meter can be fed from the following points:
- Small Fader path input (pre-EQ & Insert)
- Large Fader path input (pre-EQ & Insert)
- Channel Direct Output
- Associated Multitrack Bus Output
- Stereo Program Bus Output (on channel VU meters 25-32)

The feed to the channel VU Meter is determined by the selection made using the VU SELECT controls in the Master Section. These selections are global and apply to all channel VU Meters.



The METER SELECT controls for the channel VU Meters function as follows:

VU LRG (Large Fader): Routes the Large Fader path input to the VU meters
- Illuminates when engaged

VU SM (Small Fader): Routes the Small Fader path input to the VU meters
- Illuminates when engaged

VU DIR (Direct Output): Routes the Direct Output to the VU meters
- Illuminates when engaged

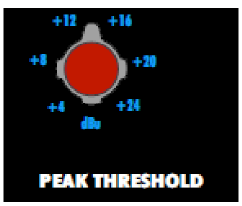
VU BUS (Bus Outputs): Routes the output of the Multitrack Buses 1-24 and the Program Bus STA, STB, STC, and GM to the VU meters
- Multitrack Buses 1-24 on channel VU meters 1-24
- STA, STB, STC, and GM on channel VU meters 25-32
- Illuminates when engaged

CHAN VU 0=+10: Inserts a -10dB pad before the channel VU Meter input
- Use to meter high level signals
- Illuminates when engaged

Large Fader and Small Fader Peak Indicators

Each fader path is equipped with a LED Peak Indictor. The peak reference level is set on a global basis by the PEAK THRESHOLD selector in the Master Section.

PK (Peak): LED peak indicator for Small and Large Fader paths
- A red LED illuminates when the selected peak threshold is reached in the audio path



PEAK THRESHOLD: Sets the peak reference level for the channel Peak Indicators console-wide
- +4dBu to +24dBu range
- Located in the Master Section

Channel Output Routing and Bus Assignment

The output of each audio path (Small Fader and Large Fader) may be routed to the following destinations:

- Channel Direct Output
- Multitrack Buses 1-24
- Stereo Program Bus A
- Stereo Program Bus B
- Stereo Program Bus C

IMPORTANT NOTE: If no output assignments have been made, the output of the Large Fader will feed the Channel Direct Output and the Small Fader will not be routed anywhere.

In a tracking session, it is typical for the Small Fader to be assigned to the Direct Output or a Multitrack Bus and the Large Fader to feed the Stereo A (STA) Program Bus. In a mixing session, it is typical for both paths on most channels to be assigned to one or more Program Bus (Stereo A, B, C).

Channel Direct Output

The Direct Output may be fed from the following sources:

- Large Fader path, post-fader, post-mute (default source)
- Large Fader path, post-COMP (lower 200 slot), pre-EQ
- Small Fader path, post-fader, post-mute
- Small Fader path, post-COMP (lower 200 slot), pre-EQ

The Direct Output is fed post-mute from the Large Fader path by default. The Direct Output cannot feed more than one of these destinations simultaneously.

The source of the Direct Output is determined by the **DIR PRE** and **DIR SM** buttons on the 968 Input Module.

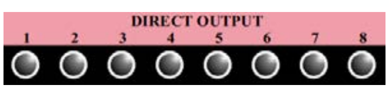
DIR PRE (Direct Output Pre-fader): Routes the Direct Output pre-fader from the assigned fader path.

- The Direct Output is routed post-fader by default
  - Output is routed post-lower 200 slot (COMP), pre-EQ
  - Illuminates when engaged

DIR SM (Direct Output Small Fader): Routes the Direct Output from the Small Fader path.

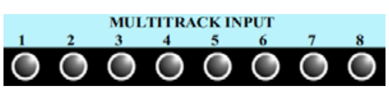
- The Direct Output is routed from the Large Fader path by default
  - Output is routed post-mute (post-fader)
  - Illuminates when engaged

The Direct Output feeds the DIRECT OUTPUT patch points.



DIRECT OUTPUT: Channel Direct Output (as assigned)

- Fed post-fader from the Large Fader by default
- Half-normalled to the MULTITRACK INPUT
  - Outputs split if patched

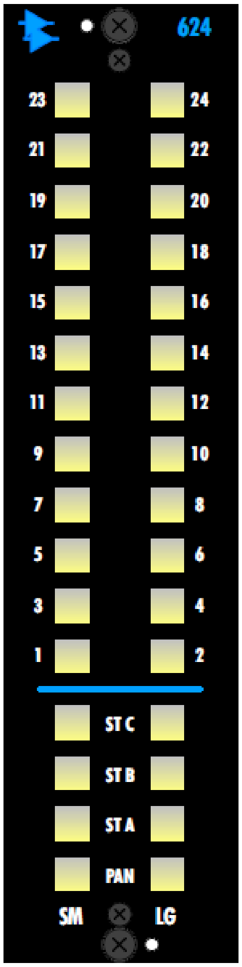


MULTITRACK INPUT: Connection to multitrack recorder inputs

- Half-normalled from the DIRECT OUTPUT
- Breaking input: Inserted patch cord will replace normalled connection

NOTE: The multitrack recorder may alternately be fed from the Multitrack Buses by patching the BUS OUTPUT patch points to the MULTITRACK INPUT patch points.

624 Bus Assignment Module



The 642 Bus Assignment Module provides comprehensive output signal flow routing for both fader paths and allows the assignment of the Large Fader and Small Fader path outputs to the following buses:

- Multitrack Buses 1-24
- Stereo Program Bus A
- Stereo Program Bus B
- Stereo Program Bus C

Only one fader path at a time (Large or Small) can feed the Multitrack buses (1-24) on a given channel. By default, the Large Fader will feed the Multitrack Buses.

The Small Fader path can be routed to the Multitrack Buses by pressing the **24** button in the Small Fader controls on the 968 Input Module. This will replace the Large Fader path signal to the Multitrack Buses.

To make Multitrack Bus assignments from the Large Fader path, engage numbered button (1-24) for the desired bus. The button will illuminate indicating the assignment has been made to that bus.

Either or both fader paths (Large or Small) can feed the stereo Program Buses. Neither path is assigned by default. At the bottom of the 624 module there are two rows of **STA**, **STB**, and **STC** assignment buttons. The left column of switches is for the Small Fader path (as indicated by “SM” at the bottom of the module). The right switches are for the Large Fader path (as indicated by “LG”).

To make stereo Program Bus assignments from either fader path, engage the buttons for the desired Program Bus (**STA**, **STB**, and/or **STC)**. The engaged button(s) will illuminate indicating the assignment has been made to that bus.

Channel Pan-Pots

The pan-pots are not engaged by default in either fader path. Without the **PAN** button engaged, all multitrack and program bus assignments are mono. At the bottom of the 624 module are two **PAN** switches, one for each fader path (SM or LG). The **PAN** switch must be engaged for each path for the pan-pot to be in the signal path

When the pan control is active, signal levels at the extremes of the pan control will be at the same level as they would be if the pan control was not active. According to the applied pan law, to avoid center channel build-up, signals are attenuated by 2.5dB in the Small Fader pan-pot and 3.5dB in the Large Fader pan-pot when the pan-pot is centered.

Depending on routing and assignments either pan-pot may be used for Left/Right panning to stereo program buses or odd/even panning to Multitrack Buses.

Panning across odd/even pairs of Multitrack Buses can be useful when it is desired to mix several sources together to a stereo pair of tracks on the multitrack recorder (DAW). For example, you may wish to record five tom mics to two tracks of the recorder and maintain the stereo position of each tom.

To pan across odd/even Multitrack Buses:

- Press **24** on the mic channels to feed the Small Fader to the Multitrack Buses
- Assign the Small Faders to a pair of odd/even Multitrack Buses (5-6 for example)
- Engage the **PAN** button in the “SM” column of the 624 Bus Assignment Module
- The Small Fader pan-pot can be used to “place” tracks in the stereo image on tracks 5 & 6

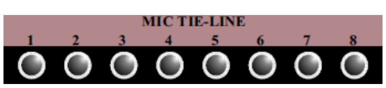
Channel Patch Bay

The Legacy AXS has 14 patch points associated with channel signal flow. These points are located in the Channel patch bay.

The 14 channel patch points are as follows:

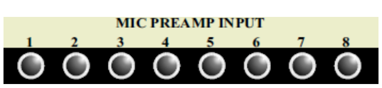
- MIC TIE-LINE
- MIC PREAMP INPUT (upper 200 slot mic preamp in)
- ALT LINE INPUT
- COMPRESSOR INPUT (lower 200 slot)
- SIDE CHAIN (lower 200 slot dynamic processor key input)
- COMPRESSOR OUTPUT (lower 200 slot)
- SMALL INSERT SEND (Small Fader path)
- SMALL INSERT RETURN (Small Fader path)
- MULTITRACK OUTPUT
- LINE INPUT (Large Fader path)
- LARGE INSERT SEND (Large Fader path)
- LARGE INSERT RETURN (Large Fader path)
- DIRECT OUTPUT
- MULTITRACK INPUT

The following section outlines the channel patch points in the order they appear in the Channel Bay. Depending on console configuration, there will be patch points for 32, 48, 64, or 80 channels in the bay. In the interest of space, patch points for only 8 channels are shown below.



MIC TIE-LINE: Studio microphone outputs

- Full-normal to MIC PREAMP INPUT
- Breaking output: Inserted patch cord will replace studio tie-line connection

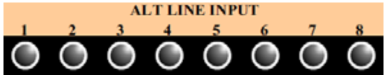


MIC PREAMP INPUT: 212 Mic Preamp inputs

- Balanced, low-impedance, mic-level input
- Full-normal from MIC TIE-LINE patch points

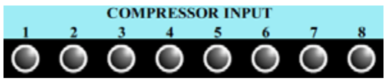
- Breaking input: Inserted patch cord will replace normalled connection
- Not used with 205 Direct Input Module

CAUTION: *The MIC PREAMP INPUT patch point will carry 48-Volt phantom power when the **48V** button is engaged on the associated 212 Mic Preamp. Care should be used and **48V** should be always be disengaged when patching to these points.*



- ALT LINE INPUT: Secondary line-level inputs
- Balanced, low-impedance, line-level input
- Alternate input to the Small Fader path
  - Half-normal from owner specified source (patch optional)
  - Breaking input: Inserted patch cord will replace normal input

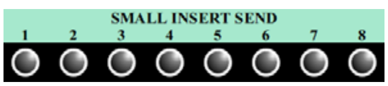
Patch points for alternate external sources (such as virtual instruments, extra DAW returns, etc.) normalled to the ALT LINE INPUT patch points are not provided in the standard configuration. However, the ALT LINE INPUTs are available via a multi-pin connector on the back of the patch bay to allow normal interfacing to these points.



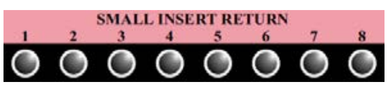
COMPRESSOR INPUT: Lower 200 slot input
- Balanced, low-impedance, line-level input
- Breaking input: Inserted patch cord will replace normalled connection

- SIDE CHAIN (input): Lower 200 slot detection path input
- Balanced, low-impedance, line-level input
- Full-normal from COMPRESSOR INPUT
  - Breaking input: Inserted patch cord will replace normalled connection

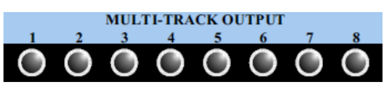
COMPRESSOR OUTPUT: Lower 200 slot output
- Balanced, low-impedance, line-level output
- Outputs split if patched



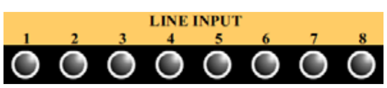
SMALL INSERT SEND: Insert Send from Small Fader path
- Always active
- Pre-fader, post-EQ



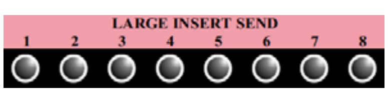
SMALL INSERT RETURN: Insert Return to the Small Fader path
- Patched signal is inserted in the Small Fader path when the **INS** button is engaged
  - Pre-fader, post-EQ



MULTITRACK OUTPUT: Multitrack recorder returns
- Half-normal to LINE INPUT
- Outputs split if patched



LINE INPUT: Main line-level inputs
- Balanced, low-impedance, line-level input
- Half-normal from MULTITRACK OUTPUT
  - Breaking input: Inserted patch cord will replace normalled connection



LARGE INSERT SEND: Insert Send from Large Fader path
- Always active
- Pre-fader, post-EQ

LARGE INSERT RETURN: Insert Return to the Large Fader path
- Patched signal is inserted in the Large Fader path when the **INS** button is engaged
  - Pre-fader, post-EQ

DIRECT OUTPUT: Channel Direct Output (as assigned)

- Fed post-fader from the Large Fader by default
- Half-normalled to the MULTITRACK INPUT
  - Outputs split if patched

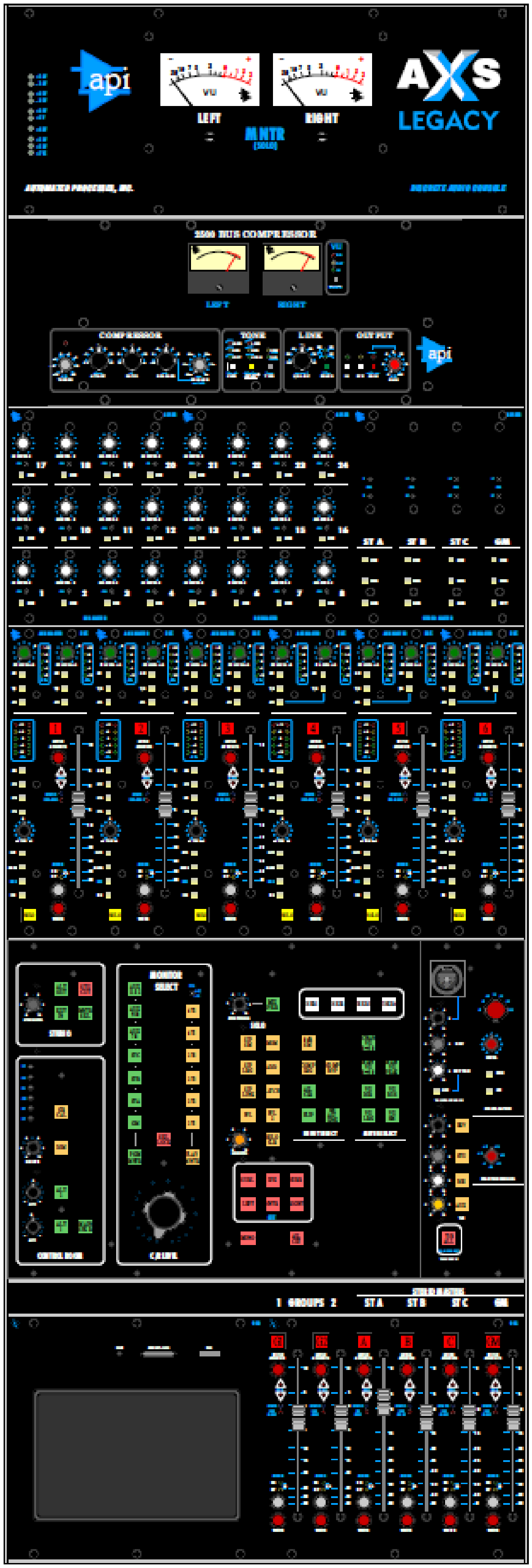
MULTITRACK INPUT: Connection to multitrack recorder inputs

- Half-normalled from the DIRECT OUTPUT
- Breaking input: Inserted patch cord will replace normalled connection

NOTE: The multitrack recorder may alternately be fed from the Multitrack Buses by patching the BUS OUTPUT patch points to the MULTITRACK INPUT patch points.

Master Bucket

The AXS master bucket accommodates all the central facilities and modules that work in concert with the channels to provide a complete, flexible, and comprehensive recording and production system.

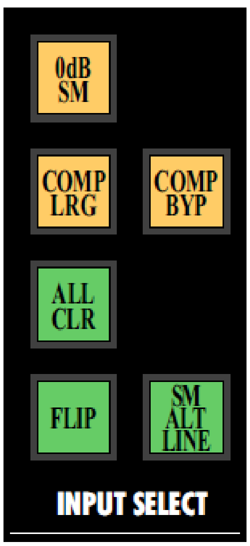


The Legacy AXS master bucket provides a comprehensive suite of master controls, facilities, and features that address all the needs of modern audio production. These facilities include:

- Master Channel Routing Controls
- 6 Assignable Stereo Returns
- 3 Stereo Program Bus Masters and Master Faders (A, B, C)
- Stereo Grand Master and Fader (GM)
- Main Stereo VU Meters
- 2500C Stereo Bus Compressor
- Automation Controller
- 2 Automation Group Master Faders
- 24 Multitrack Bus Masters
- 12 Auxiliary/Cue Masters
- Talkback Controls
- Solo Controls
- Meter and Peak Indicator Controls
- Control Room Monitor Controls
- Studio Monitor Controls
- Oscillator
- PSU Indicators
- User Switches

Master Channel Routing Controls

The channel signal flow can be altered console-wide using the controls in the INPUT SELECT area of the Master Section.



The INPUT SELECT controls function as follows:

FLIP: Swaps the input sources between Large and Small Fader paths globally

- The Preamp Output will be routed to the Large Fader path
- The Line Input will be routed to the Small Fader path
- Input sources on individual channels can be reset by engaging the **FLIP** button on the 968 Input Module
- Illuminates briefly when engaged to acknowledge activation, but then goes out

SM ALT LINE: Switches the Small Fader path input to the Alternate Line In (ALT) globally

- Replaces the preamp output with the Alt Line In as the Small Fader path
  source on all channels
- The Small Fader input source on individual channels can be reset by engaging the **ALT** button on the 968 Input Module
- Illuminates briefly when engaged to acknowledge activation, but then goes out

ALL CLR (All Clear)**:** Returns all channel input routing to the default positions:

- Resets the **FLIP** and **ALT** buttons on all channels
- The Small Fader source will be the Preamp output
- The Large Fader source will be the Line Input
- Illuminates only when pressed

COMP LRG (Compressor to Large Fader): Routes the 200 slot processor into the Large Fader path globally

- The Compressor routing on individual channels can be reset by engaging the **COMP LRG** button on the 968 Input Module
- Illuminates when engaged

COMP BYP (Compressor Bypass): Bypasses the 200 slot processor on all channels

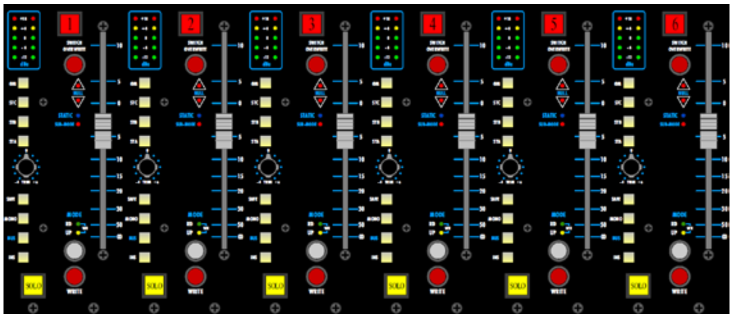
- Illuminates when engaged

0dB SM (Small Fader): Engages a 0dB fader bypass for Small Faders globally

- Illuminates when engaged

Stereo Return Modules 1-6

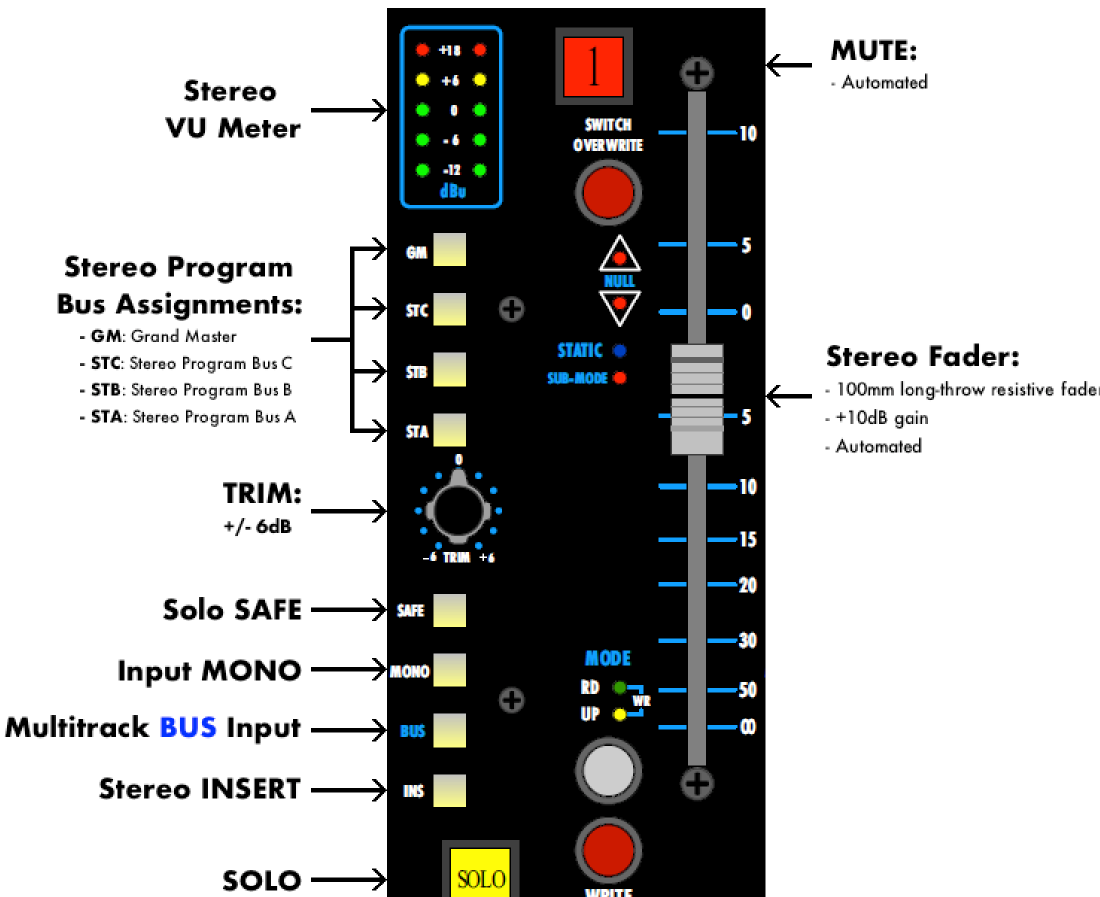
Every Legacy AXS is equipped with six Stereo Return modules, located in the master bucket.



The six Stereo Returns are designed to provide extra capacity and flexibility during mixing and support techniques such as returning stereo stems to a mix, returning virtual instruments and tracks, parallel compression, effects returns and other applications. Each Stereo Return can be assigned to any or all the three stereo Program Buses as well as directly to the program Grand Master.

Each Stereo Return is equipped as follows:

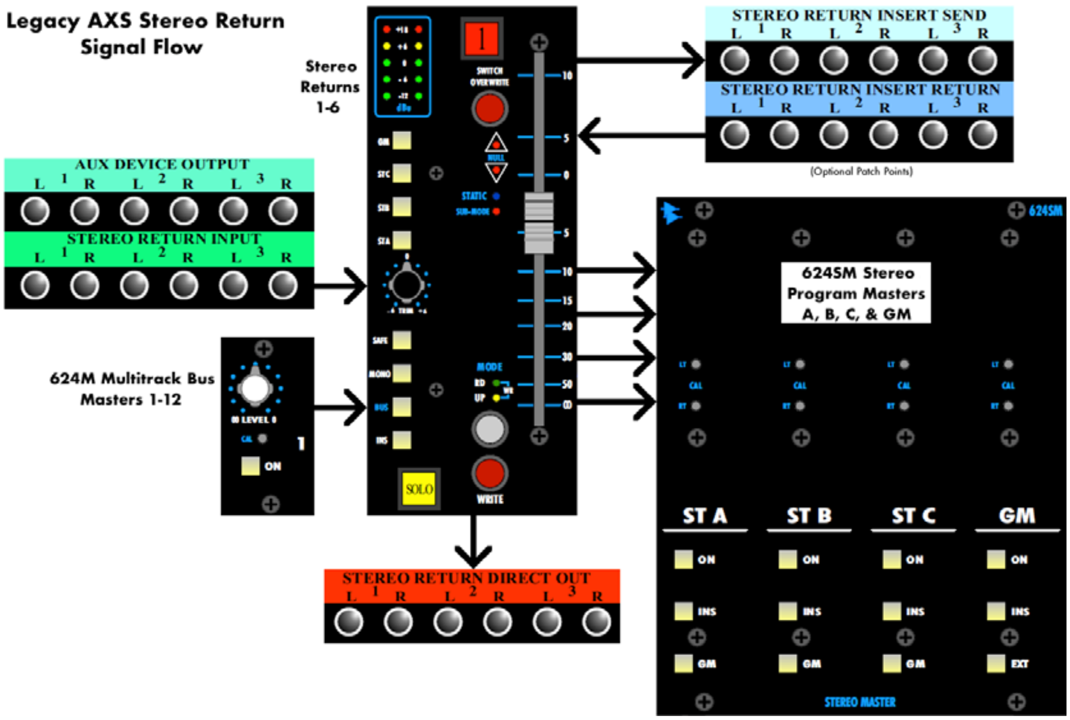
- Balanced, low-impedance, line-level inputs
- Alternate input from associated pair of Multitrack Buses
- Input TRIM: +/- 6dB
- Balanced stereo INSERT (controlled by automation)
- Mono input summing
- Stereo Direct Output patch points
- 100mm long-throw resistive automated fader: +10dB to -∞dB range
- MUTE (controlled by automation)
- SOLO with Solo SAFE
- 5-segment LED meter



(Automation controls not labeled)

Stereo Return Signal Flow

The diagram below shows the basic Stereo Return signal flow from input to the stereo Program Masters/Grand Master and Direct Output. In interest of space, patch points for only 3 of the 6 Stereo Returns are shown.



The Stereo Returns have two input options:

- Line-Level: STEREO RETURN INPUT patch points
- Multitrack Bus: Output of the associated Multitrack Bus Master in pairs

The default line inputs to the Stereo Returns are the STEREO RETURN INPUT patch points in the System patch bay. These patch points are fed from the AUX DEVICE OUTPUT patch points via half-normal. Outputs of external devices can be connected to the AUX DEVICE OUTPUT patch points via a multi-pin connector on the rear of the System patch bay. This will allow a normalled regular connection through the patch bay.

If desired, the output of the associated pair Multitrack Bus Masters can be routed to the Stereo Return input by engaging the **BUS** button. This will replace the patch bay input to the return with the bus output. Only the first 12 Multitrack Buses can be directly routed to the Stereo Returns. The associated Multitrack Buses are as follows:

Multitrack Buses		Stereo Return
1-2	=	1
3-4	=	2
5-6	=	3
7-8	=	4
9-10	=	5
11-12	=	6

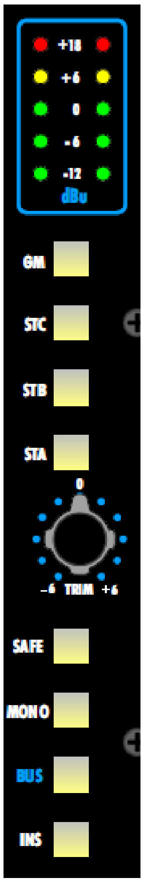
The Left and Right inputs to the Stereo Returns can be summed to mono by engaging the **MONO** button. This mono summing happens pre-Insert Send.

A balanced stereo Insert with an in/out switch is provided for each Stereo Return. Engage the **INS** button to route the Insert Return to the Stereo Return signal path. Stereo Return Insert patch points are available on the System patch bay.

In addition to Stereo Program Bus and Grand Master output assignments, the Stereo Returns have Direct Outputs that are available in the System patch bay (STEREO RETURN DIRECT OUT).

Stereo Return Controls

The controls for the Stereo Returns function as follows:



GM (Grand Master): Routes the Stereo Return outputs to the Grand Master Program Bus
- Illuminates when engaged

STC (Stereo C): Routes the Stereo Return outputs to the Stereo C Program Bus
- Illuminates when engaged

STB (Stereo B): Routes the Stereo Return outputs to the Stereo B Program Bus
- Illuminates when engaged

STA (Stereo A): Routes the Stereo Return outputs to the Stereo A Program Bus
- Illuminates when engaged

TRIM: Applies +6dB of gain or -6dB of attenuation to the input signal
- Located pre-Insert Send

SAFE: Activates the Solo Safe mode for the Stereo Return
- The SAFE button protects the Stereo Return from being muted when the Solo-In-Place function is active and another return or channel is soloed
- Illuminates when engaged

MONO: Sums the inputs of the Stereo Return to mono
- Sums pre-Insert
- Illuminates when engaged

BUS: Routes the associated pair of Multitrack Buses to the input of the Stereo Return
- Replaces the STEREO RETURN INPUT patch point as the Stereo Return source
- Illuminates when engaged

INS (Insert): Engages the Insert Return in the Stereo Return signal path
- Insert Send is fed post-TRIM
- Insert Send is always active
- Insert Return is located pre-fader
- The Insert Return is active only when the **INS** button is engaged
- Controlled by automation
- Illuminates when engaged

SOLO: Activates the selected solo function for the Stereo Return. The following solo functions may be selected via the Master Section:
- Pre-fader Listen (PFL): Non-destructive, stereo
- After Fader Listen (AFL): Non-destructive, stereo
- Solo-In-Place (SIP): Destructive, panned, post-fader
- Illuminates in yellow when engaged

NOTE: AFL is the default solo mode.

MUTE: Cuts the Stereo Return output signal
- The MUTE button is the on/off switch for the Stereo Return
- Can be engaged by Solo-In-Place solos

- Illuminates in red when engaged

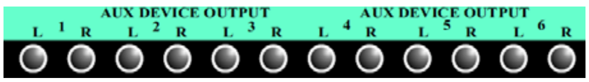
Stereo Fader (not shown): Controls the output level of the Stereo Return output
- Full-size 100mm resistive fader
- +10dB of gain
- -∞dB of attenuation
- Unity gain when set to 0dB
- Controlled by automation

Meter: Displays Stereo Return output level
- Stereo 5-segment LED meter

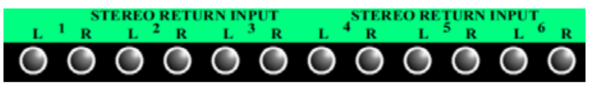
To make stereo Program Bus and/or Grand Master assignments from the Stereo Return, engage the buttons for the desired Program Bus(es) (**STA**, **STB**, and/or **STC**) and/or the Grand Master (**GM**). The engaged button(s) will illuminate indicating the assignment has been made to that bus.

Stereo Return Patch Points

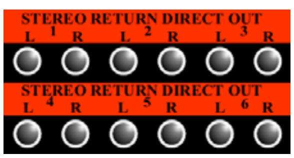
The following section outlines Stereo Return patch points.



AUX DEVICE OUTPUT: Stereo output connections for external devices
- Balanced, line-level, low-impedance
  - Half-normalled to the STEREO RETURN INPUT
  - Outputs split if patched

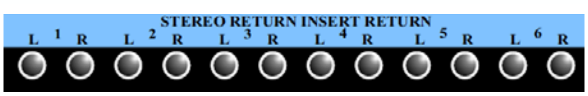


STEREO RETURN INPUT:
- Balanced, line-level, low-impedance
- Half-normalled from the AUX DEVICE OUTPUT
  - Breaking input: Inserted patch cord will replace normalled connection



STEREO RETURN DIRECT OUTPUT: Stereo Return Direct Output
- Outputs split if patched

STEREO RETURN INSERT SEND: Stereo Return Insert Send
- Pre-fader
  - Always active
  - Outputs split if patched



STEREO RETURN INSERT RETURN: Stereo Return Insert Return
- Patched signal is inserted in the Stereo Return signal flow when the **INS** button is engaged
- Pre-fader

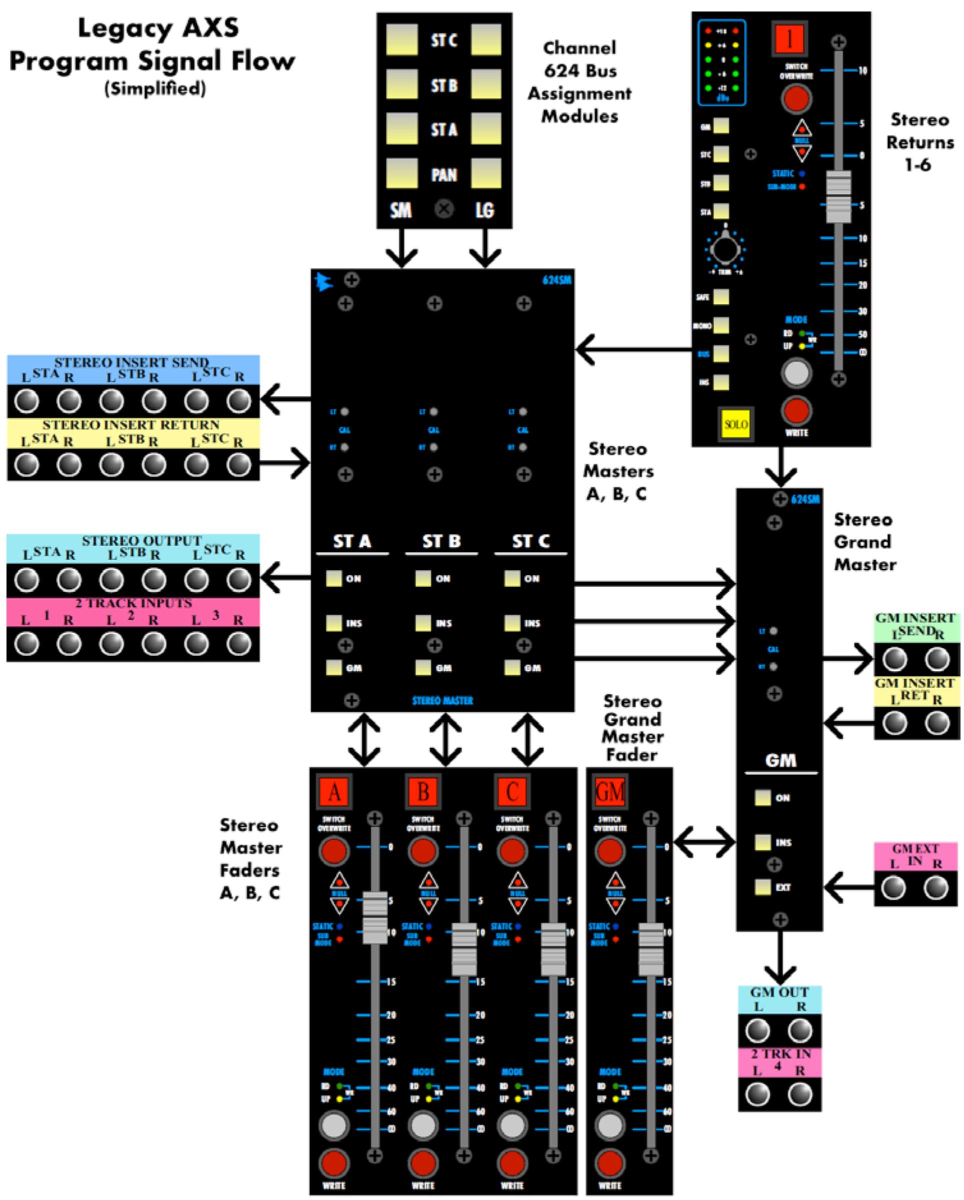
Stereo Program Bus Masters and Master Faders

The Legacy AXS is equipped with three stereo Program Buses (STA, STB, and STC) that are accessible from channels and the six Stereo Returns. The output of each stereo Program Bus "Active Combining Amplifier" (ACA) feeds its own Program Master. Each Program Master is equipped with an on/off switch, balanced insert with switch, 100mm automated resistive Master Fader, output trim pots, associated patch points, and assignment to the Grand Master.

A stereo Grand Master is provided for summing the A, B, and C Program Bus outputs and the outputs of the Stereo Returns into a single master stereo program output. Like the Program Masters the Grand Master is equipped with an on/off switch, balanced insert with switch, 100mm automated resistive fader, output trim pots, and associated patch points, but also inputs for an external stereo source.

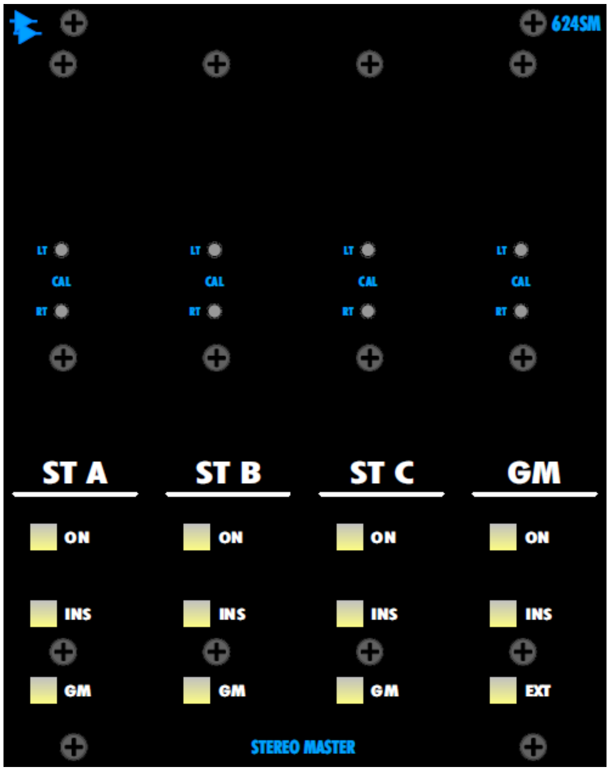
In combination, the three stereo program masters and stereo Grand Master provide a host of flexible routing possibilities parallel processing, mix versions, segues, and other mixing techniques.

Stereo Program Bus Master and Master Fader Signal Flow



The diagram on the previous page shows the basic Stereo Program Bus signal flow from channel Program Bus assignment, through the stereo Masters and Grand Master, to the 2Track Inputs.

624SM Stereo Program Master Module



The 624SM Stereo Program Master Module contains the controls for the three stereo Program Buses as well as the stereo Grand Master.

Each of the four stereo Program Masters is equipped as follows:

ON: ON/Off switch for the Program Master output
- Illuminates when engaged

INS (Insert): Routes the Program Bus Insert Return to the Program Master
- Illuminates when engaged

GM (Grand Master): Routes the output of the Program Master to the Grand Master Program Bus
- Illuminates when engaged

CAL LT & RT (Calibrate Left & Right): Left-Right calibration trim pots for the Bus Master output

EXT (External Input): Routes the stereo signals at the GM EXT IN patch points directly to the Grand Master summing bus.
- Grand Master only
- Illuminates when engaged

The three stereo Program Buses may be fed simultaneously from the following sources:
- Small Fader path output
- Large Fader path output
- Stereo Returns 1-6

Stereo Program Bus assignments from channels are made on the 624 Bus Assignment Module. Stereo Program Bus assignments are made direct.

Stereo Grand Master

A stereo Grand Master Bus and Master Fader are provided to facilitate a wide range of operations. The Program Masters, Stereo Returns, and an External Input can be routed to the stereo Grand Master Bus for summing to a master stereo program output.

The following stereo sources can be routed to the Grand Master Program Bus:
- Stereo Program Master A
- Stereo Program Master B
- Stereo Program Master C
- Stereo Returns 1-6
- External Stereo Source (on patch)

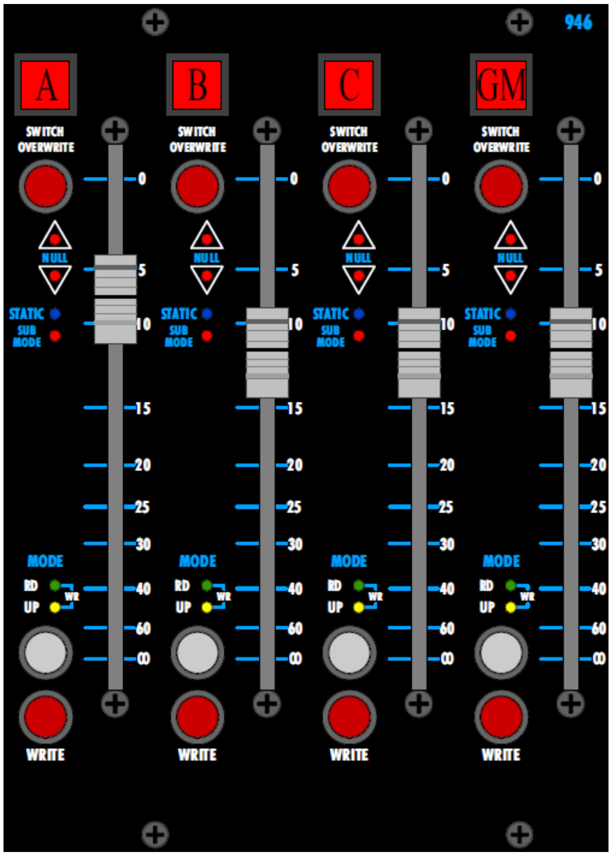
Routing to the Grand Master from the stereo Program Masters or Stereo Returns is accomplished by engaging the **GM** button.

An external stereo line-level source can be routed to the Grand Master by patching the source into the GM EXT IN patch points and engaging the **EXT** button in the Grand Master controls. The will route the external source directly to the Grand Master summing bus.

2500C Stereo Bus Compressor

The Legacy AXS is equipped with a 2500C Stereo Bus Compressor. This compressor is available only on patch, but can be routed to STA, STB, STC, or GM via insert. See the user guide in the appendix.

946 Stereo Program Master Faders



The 946 Stereo Master Fader Module contains the three stereo Program Master Faders (A, B, & C) and one Grand Master Fader (GM).

Master Faders: Controls the output level of the stereo Program Masters.

- Stereo fader
- Full-size 100mm resistive fader
- Cut only: 0dB of gain
- -∞dB of attenuation
- Unity gain when set to 0dB

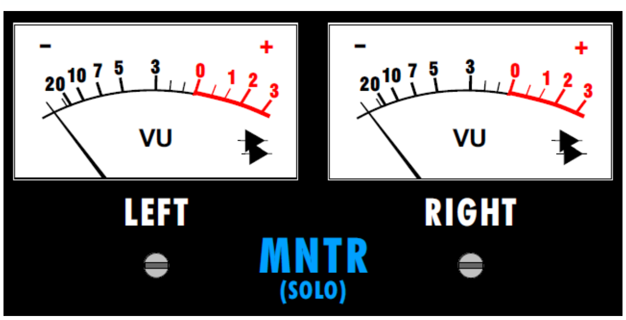
The A, B, C Master Faders are the master output level control for their respective stereo Program Masters and outputs

The GM fader is the Grand Master Fader. The Grand Master is the master output level control for the Grand Master output.

Master Faders are controlled by automation.

Main Stereo Meters

A pair of large, high-quality VU meters are installed in the top bay of the master bucket for indicating the stereo Program, Playback, and Solo levels.



The source for the Main Stereo Meters is selected using the Control Room Monitor source selectors (**MNTR**). Accordingly the Main Meters can be fed from the following sources:

- Stereo Program Masters (PGM MNTR):
  - A, B, or C
  - Grand Master
  - Aux 7/8, 9/10, & 11/12
- Solo Bus
- External Playback Sources (PLAY MNTR):
  - 2-Track 1-4
  - 6-Track 1-2 (Left and Right)

While recording and mixing the Main Meters display the level of the stereo Program Master of the Program Master that's being used for mixing and monitoring. When a PFL or AFL **SOLO** is engaged, the control room Monitor Source is replaced with the output of the Solo Bus and the Solo Bus is displayed on the LEFT and RIGHT Main Meters (SOLO).

The Main Meters have two controls in the METER SELECT area of the Master Section.

The METER SELECT controls for the Main Meters function as follows:

<u>**MAIN VU 0=+10**</u>: Inserts a -10dB pad before the Main Meter inputs

- Use to meter high level signals
- Illuminates when engaged

<u>**MAIN VU PEAK**</u>: Changes the Main Meter ballistics from VU to Peak

- A fixed peak hold circuit feeds the meter
- Illuminates when engaged

Stereo Program Master Patch Points

The following section outlines Stereo Program Master patch points.

<u>STEREO INSERT SEND</u>: Insert Send from stereo Program Masters A, B, C

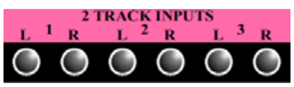
- Always active
- Pre-fader

<u>STEREO INSERT RETURN</u>: Insert Return to stereo Program Masters A, B, C

- Patched signal is inserted in the Program Master signal flow when the **INS** button is engaged
- Pre-fader

<u>STEREO OUTPUT</u>: Main outputs for stereo Program Masters A, B, C

- Half-normalled to 2 TRACK INPUTS 1, 2, 3
- Outputs split if patched



<u>2 TRACK INPUTS 1-3</u>: Input to 2-track recorders 1-3

- Half-normalled from STEREO OUTPUT A, B, C
- Breaking input: Inserted patch cord will replace normalled connection

<u>GM EXT IN (Grand Master External Input)</u>: Stereo external inputs to the Grand Master summing bus

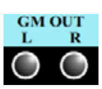
- The **EXT** button on the Grand Master must be engaged for the external source to be routed to the Grand Master summing bus

<u>GM INSERT SEND (Grand Master)</u>: Insert Send from the stereo Grand Master

- Always active
- Pre-fader

<u>GM INSERT RETURN (Grand Master)</u>: Insert Return to the stereo Grand Master

- Patched signal is inserted in the Program Master signal flow when the **INS** button is engaged
- Pre-fader



<u>GM OUTPUT (Grand Master Out)</u>: Main outputs for the stereo Grand Master

- Half-normalled to 2 TRACK INPUT 4
- Outputs split if patched

<u>2 TRACK INPUT 4</u>: Input to 2-track recorder 4

- Half-normalled from STEREO OUTPUT A, B, C
- Breaking input: Inserted patch cord will replace normalled connection

<u>2500 COMPRESSOR</u>: 2500C stereo Bus Compressor patch points

- <u>IN</u>: Left/Right compressor input
- <u>S/C (side chain)</u>: Left/Right input to the compressor detection path
- <u>OUT</u>: Left/Right compressor output

NOTE: The 2500C Stereo Bus Compressor must be patched to be used.

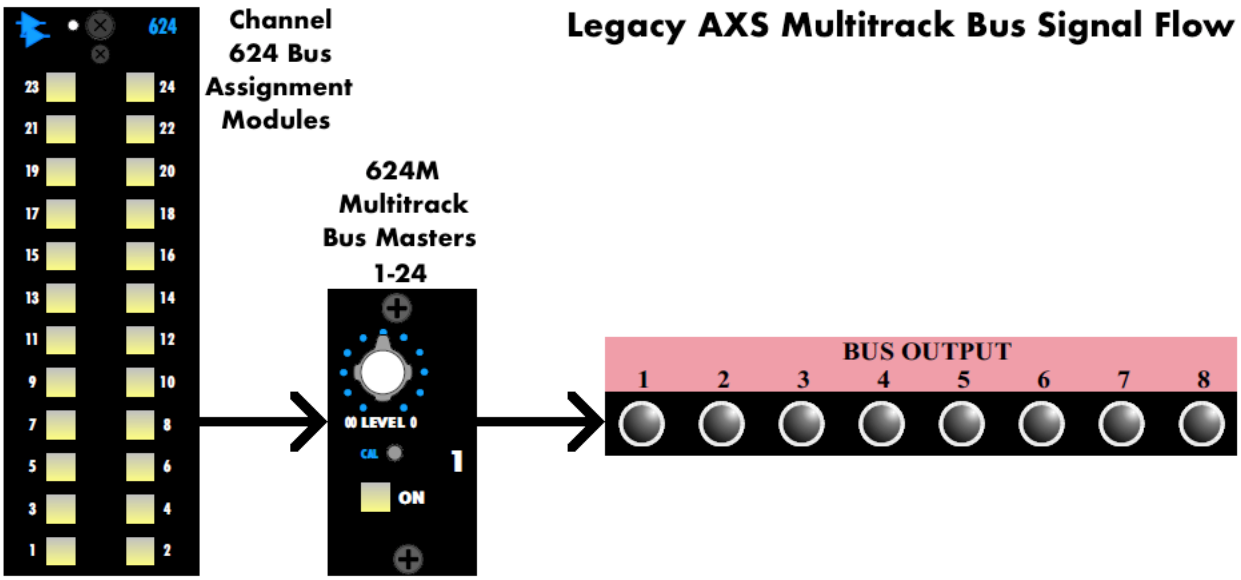
Multitrack Buses

There are twenty-four (24) Multitrack Buses. Multitrack Bus assignments are made using the 624 Bus Assignment module in each channel.

Multitrack Buses may be fed from either the Small Fader path or Large Fader path, but not both simultaneously. The default source for the Multitrack Buses is the Large Fader path. To feed the Multitrack Buses from the Small Fader path, press the **24** button in the Small Fader controls on the 968 Input Module.

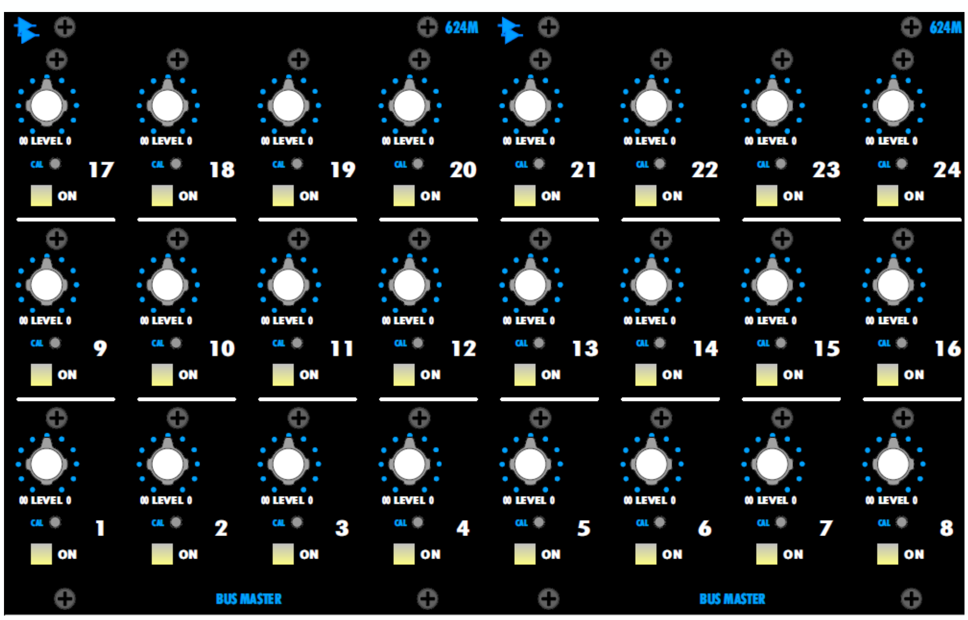
Multitrack Signal Flow

The diagram below shows the multitrack signal flow. Only the first eight Bus Outputs are shown.

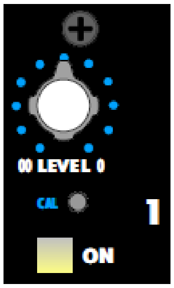


624M Multitrack Bus Masters

Each of the 624 assignments feeds its respective Multitrack Bus and the corresponding 624M Multitrack Bus Master in the master bucket.



Each Multitrack Bus Master has a gain control and trim control. These controls feed the BUS OUTPUT patch points 1-24 in the System patch bay.
Controls for the 624M Multitrack Bus Masters are as follows:



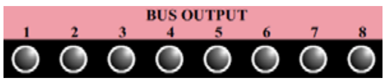
ON (on/off switch): On/off switch for the Multitrack Bus outputs
- Illuminates when engaged

LEVEL: Sets the level for the Multitrack Bus output
- -∞ to 0dB range
- 0dB is unity gain (fully clockwise)
- -∞dB is full attenuation (fully counter-clockwise)

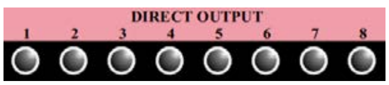
CAL (Calibration): Calibration trim pot
- Allows for precise adjustment of the bus output for calibration

The output of the Multitrack Bus Masters feed the BUS OUTPUT patch points in the System patch bay. These patch points are not normalled to the MULTITRACK IN patch points.

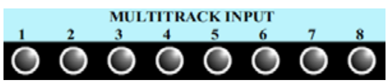
Multitrack Patch Points



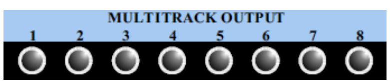
BUS OUTPUT: Multitrack Bus outputs
□ Not normalled in the standard configuration



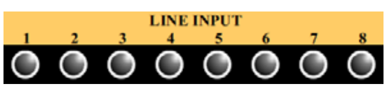
DIRECT OUTPUT: Channel Direct Output (as assigned)
□ Fed post-fader from the Large Fader by default
□ Half-normalled to the MULTITRACK INPUT
- Outputs split if patched



MULTITRACK INPUT: Connection to multitrack recorder inputs
□ Half-normalled from DIRECT OUTPUT
- Breaking input: Inserted patch cord will replace normalled connection



MULTITRACK OUTPUT: Multitrack recorder returns
- Half-normal to LINE INPUT
- Outputs split if patched



LINE INPUT: Main line-level inputs
- Balanced, low-impedance, line-level input
- Half-normal from MULTITRACK OUTPUT
  - Breaking input: Inserted patch cord will replace normalled connection

Auxiliary Buses and Masters

The Legacy AXS console provides a powerful Auxiliary Send system with a variety of options for headphone cue sends, effects sends, and other applications.

There are twelve (12) Auxiliary Buses and Masters. While all the Aux Buses are technically mono, like the channel Aux Sends, the Auxiliary Masters are organized as follows:

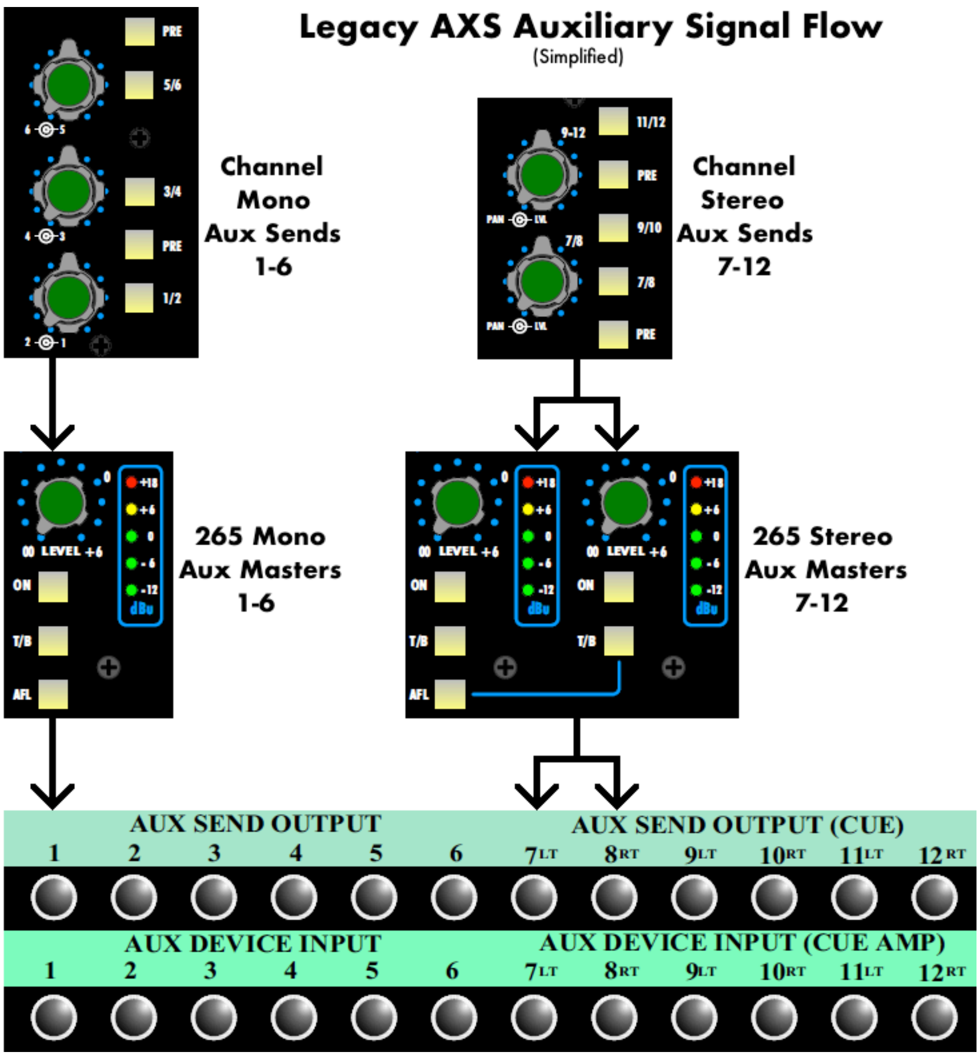
- Six mono Aux Masters: 1-6
- Three stereo Aux Masters: 7/8, 9/10 and 11/12

Mono Aux Sends 1-6 are primarily used as effects sends during recording and mixing.

Stereo Aux Sends 7/8, 9/10, and 11/12 are typically used as headphone cue sends during recording, and stereo effects sends during mixing.

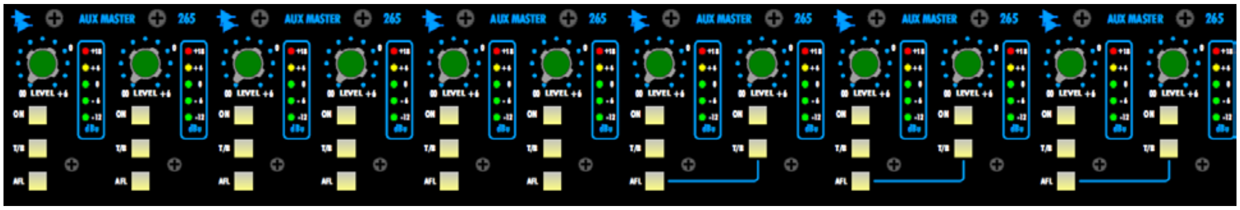
Auxiliary Signal Flow

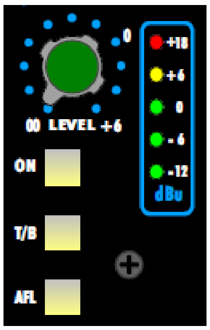
The diagram below shows the basic Auxiliary signal flow.



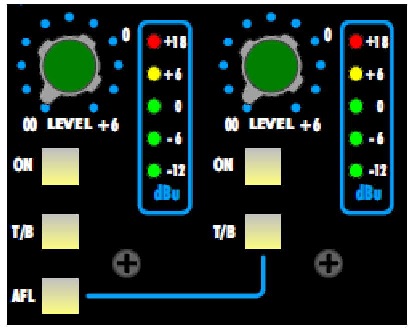
Auxiliary Masters 1-12

Auxiliary Bus Masters 1-12 are located above the Stereo Returns in the 265 Aux Master module. Each Aux Master is the output control for the corresponding Auxiliary Bus.





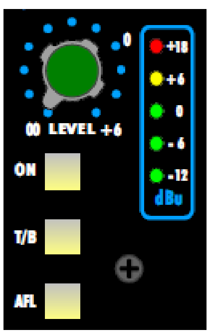
Aux Bus Masters 1-6 provide output level control for mono Aux Sends & Buses 1-6.



Aux Bus Masters 7-12 provide output level control for stereo Aux Sends 7/8, 9/10, and 11/12.

The mono and stereo Aux Masters are identical with the exception of the AFL solo. Mono Aux Masters 1-6 are equipped with an individual AFL solo button for mono operation. Aux Masters 7/8, 9/10, 11/12 are equipped with a ganged AFL solo function for stereo operation.

The controls for all Aux Masters function as follows:



LEVEL: Output level control.
- Feeds the AUX SEND OUTPUT 1-12
- -∞ to +6dB range

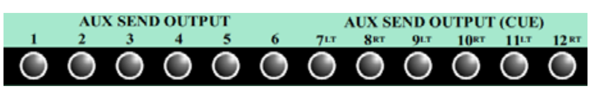
ON (on/off): On/off switch for the Aux Bus outputs
- Illuminates when engaged

T/B (Talkback): Routes the Talk Back output to the Aux Master
- Talkback output will be added to the Aux Master output when the **AUX** Talkback button is pressed in the Master Section.
- Illuminates when engaged

AFL (After Fader Listen): Activates the AFL solo function for the Aux Master
- AFL is the only solo function available for the Aux Masters
- Each mono Aux Master (1-6) has an individual **ALF** button
- Each pair of stereo Aux Masters (7/8, 9/10, 11/12) share a common AFL button for stereo operation
- Illuminates when engaged

Meter: Displays Aux Master output level
- 5-segment LED meter

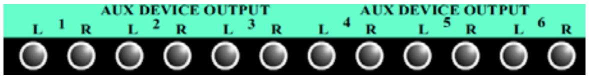
Auxiliary Master Patch Points



AUX SEND OUTPUT: Auxiliary bus outputs
- 1-6 are mono aux outputs
- 7-12 are stereo (LT/RT) aux/cue outputs
  - Half-normalled to AUX DEVICE INPUT
  - Outputs split if patched

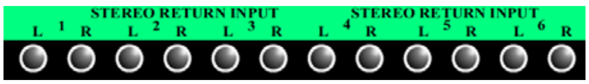
AUX DEVICE INPUT: Connection to external effects devices, cue systems, and/or other devices

- 1-6 are fed from mono sends
  - 7-12 are fed from stereo (LT/RT) aux/cue sends
  - Half-normalled from the AUX SEND OUTPUT
  - Breaking input: Inserted patch cord will replace normalled connection



AUX DEVICE OUTPUT: Stereo output connections for external devices

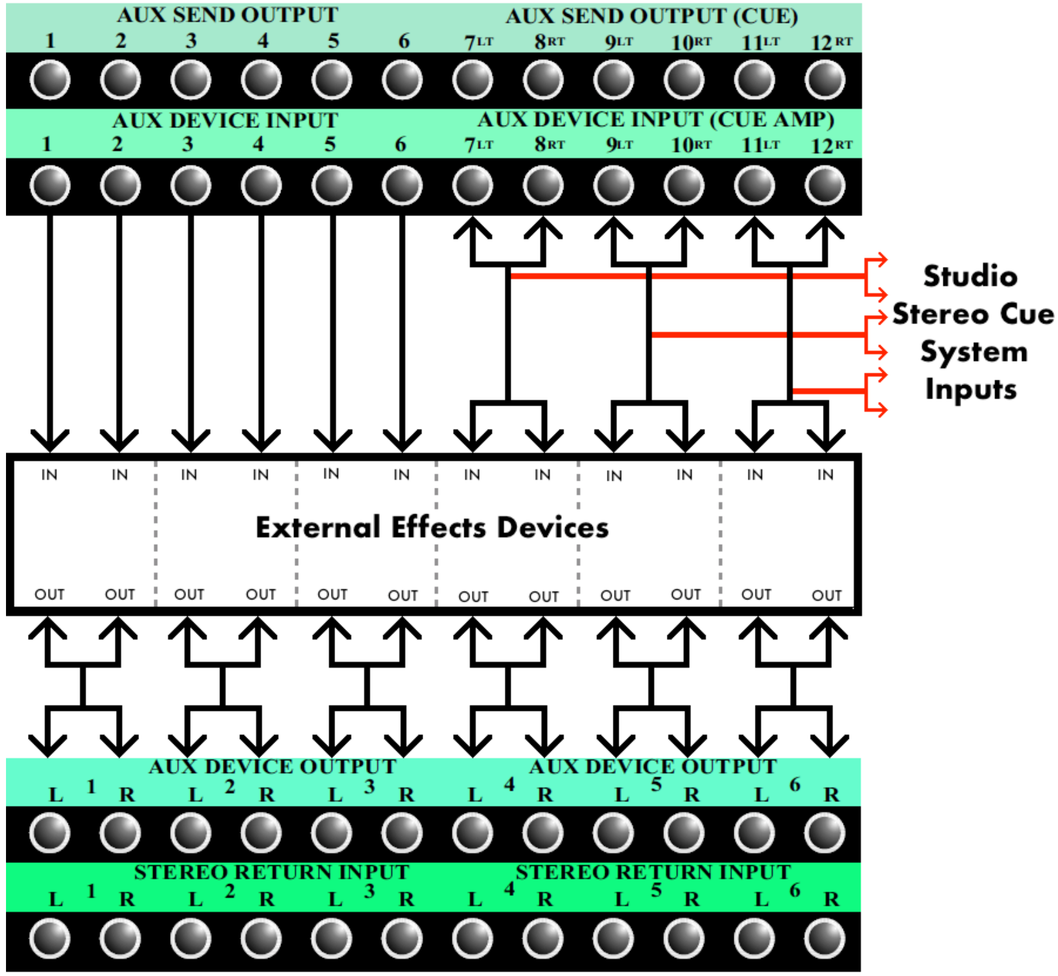
- Half-normalled to the STEREO RETURN INPUT
  - Outputs split if patched



STEREO RETURN INPUT:

- Balanced, line-level, low-impedance
- Half-normalled from the AUX DEVICE OUTPUT
  - Breaking input: Inserted patch cord will replace normalled connection

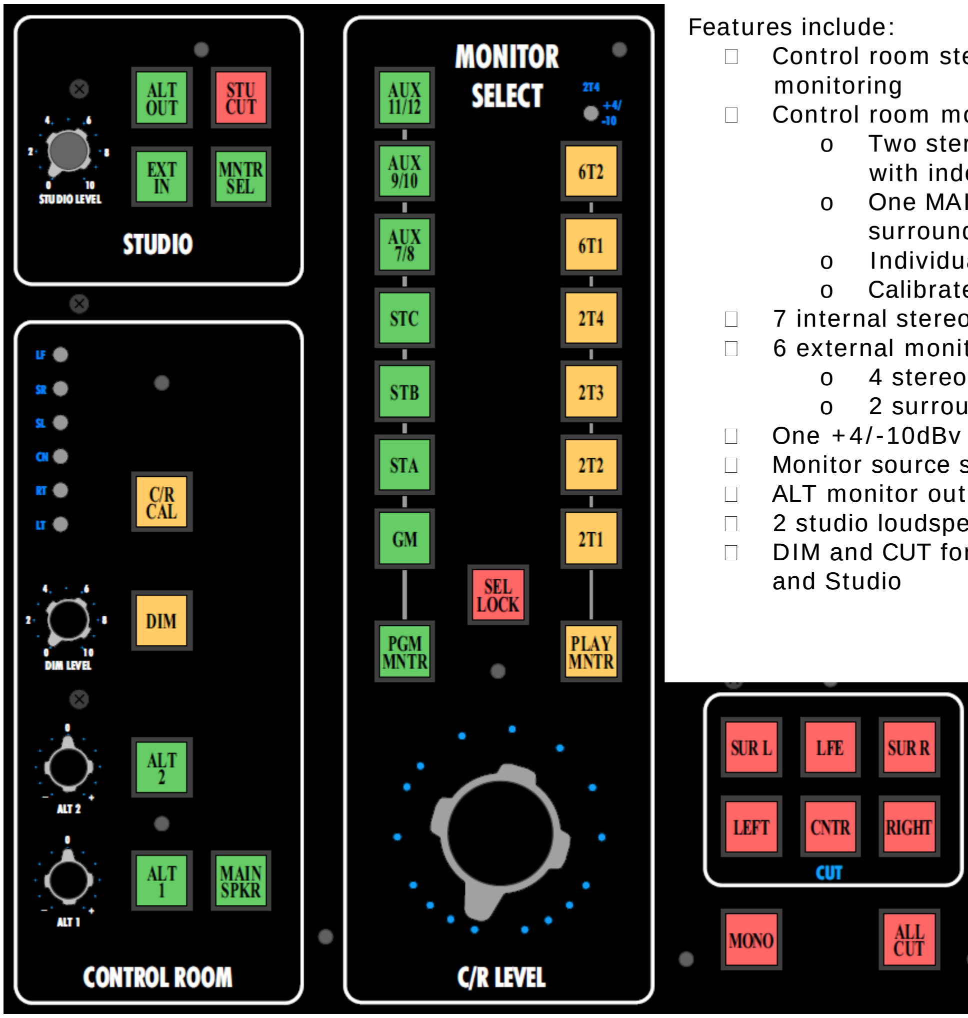
To create normalled connections to headphone cue systems and effects devices, through external effects processors, and back to the program buses for mixing, the Legacy AXS provides AUX DEVICE INPUT and AUX DEVICE OUTPUT patch points to. The AUX DEVICE INPUT (CUE AMP) patch points are half-normalled from the AUX SEND OUTPUT (CUE) patch points. The AUX DEVICE OUTPUT patch points are half-normalled to the STEREO RETURN INPUT patch points. The diagram below illustrates an example of how theses patch points can be interfaced to provide normalled connections to external effects devices and cues systems.



AUX SEND OUTPUT
AUX SEND OUTPUT (CUE)
1 2 3 4 5 6 7LT 8RT 9LT 10RT 11LT 12RT
AUX DEVICE INPUT
AUX DEVICE INPUT (CUE AMP)
1 2 3 4 5 6 7LT 8RT 9LT 10RT 11LT 12RT
Studio
Stereo Cue
System
Inputs
IN
External Effects Devices
OUT
AUX DEVICE OUTPUT
AUX DEVICE OUTPUT
L 1 R L 2 R L 3 R L 4 R L 5 R L 6 R
STEREO RETURN INPUT
STEREO RETURN INPUT
L 1 R L 2 R L 3 R L 4 R L 5 R L 6 R

Monitor Control

The Legacy AXS console provides comprehensive control room and studio monitor control facilities.



Features include:

- Control room stereo and surround monitoring
- Control room monitor systems:
  - o Two stereo ALT systems with independent level
  - o One MAIN stereo/5.1 surround system
  - o Individual monitor cuts
  - o Calibrated level control
- 7 internal stereo monitor sources
- 6 external monitor sources:
  - o 4 stereo
  - o 2 surround
- One +4/-10dBv external source
- Monitor source selection lock
- ALT monitor output trims
- 2 studio loudspeaker systems
- DIM and CUT for Control Room and Studio

Control Room Monitoring

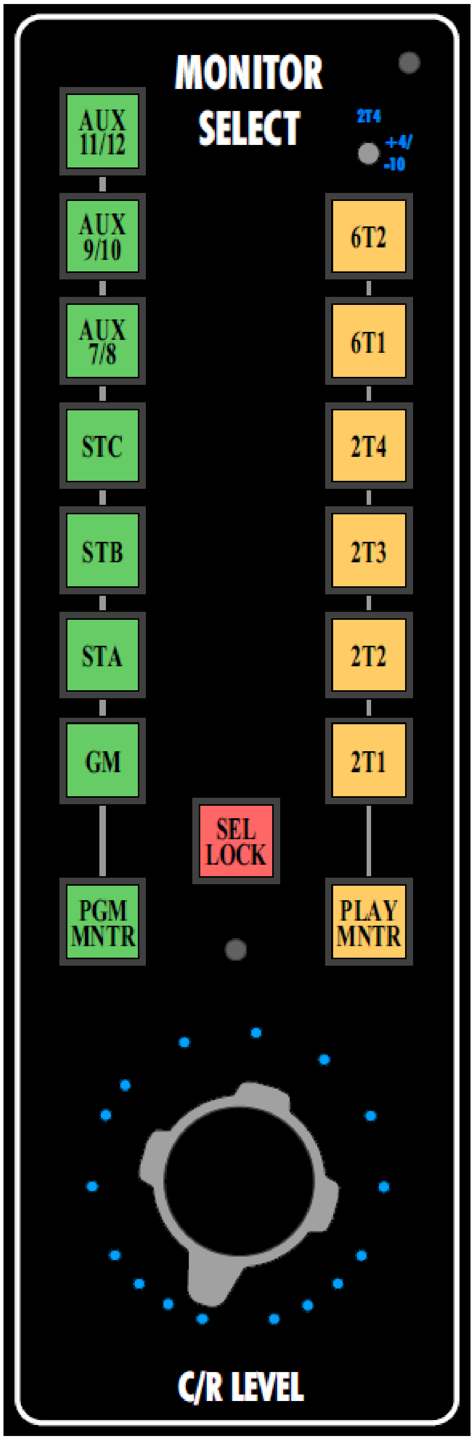
Support is provided for three (3) pairs of control room monitors, MAIN, ALT 1, and ALT 2. The source for the monitors may be selected between seven (7) internal (PGM MNTR) and six (6) external (PLAY) sources. The MAIN monitors support 5.1 surround playback and stereo monitoring.

Stereo Program (PGM MNTR) sources include stereo Program Buses A, B, C, Grand Master, and Aux 7/8, 9/10, 11/12. External sources (PLAY MNTR) include four (4) 2-track recorders/stereo sources and two (2) 5.1 surround external inputs. 2-track #4 can accept a -10dBv unbalanced input.

The following features may be applied:

- Speaker system selection
- Control room level controls
- ALL CUT
- DIM with level
- Left and Right monitor mute
- Mono summing

Control Room Monitor Source Selection (MONITOR SELECT)



The MONITOR SELECT section contains selectors for seven internal console (**PGM MNTR**) or six external (**PLAY MNTR**) sources for the control room monitor systems. This section also contains the main Control Room monitor level control (**C/R LEVEL**).

Internal monitor sources are selected by engaging the **PGM MNTR** (Program Monitor) button and the selector for the desired internal source.

The following internal sources may be selected as the Control Room monitor source:
GM: Output of the stereo Program Grand Master
STA: Output of stereo A Program Master
STB: Output of stereo B Program Master
STC: Output of stereo C Program Master
Aux 7/8: Output of Aux Masters 7/8
Aux 9/10: Output of Aux Masters 9/10
AUX 11/12: Output of Aux Masters 11/12

NOTE: All internal monitor sources are stereo.

External monitor sources are selected by engaging the **PLAY MNTR** (Playback Monitor) button and the selector for the desired external source.

The following external sources may be selected as the Control Room monitor source:
2T1: Output of 2-track recorder/stereo source #1
2T2: Output of 2-track recorder/stereo source #2
2T3: Output of 2-track recorder/stereo source #3
2T4: Output of 2-track recorder/stereo source #4
6T1: Output of surround source #1
6T2: Output of surround source #2

2T4 +4/-10: Switches the input for **2T4** between -10dBv to +4dBu level

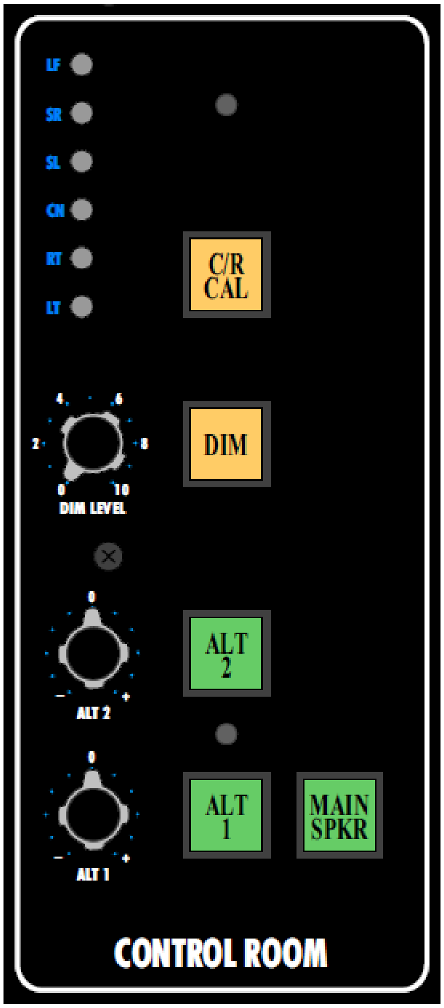
SEL LOCK (Selection Lock): Protects the control room monitor source from changes

- Illuminates when engaged

C/R LEVEL (Control Room Level): Level control for the selected control room monitor system

- Primary level control for the MAIN and ALT control room monitors
- -∞dB to 0dB range
- Fed from the selected MONITOR SELECT source

Control Room Speaker Selection and Controls



The Legacy AXS console provides support for three (3) control room monitoring systems, with a main C/R LEVEL control and independent level trims for ALT1 and ALT2. The Left and Right MAIN monitors are used primarily for stereo monitoring. However, the MAIN monitors support a 5.1 monitor system for playback of surround audio. The Left and Right MAIN monitors are used for surround monitoring as well.

The CONTROL ROOM section of the Master Section contains the primary controls for the three control room monitor systems:

MAIN SPKR (Main Speakers): Activates the MAIN monitor system

- Illuminates when engaged

ALT 1 (Alternate 1): Activates alternate monitor system #1

- Level trim for system level matching
- Illuminates when engaged

ALT 2 (Alternate 2): Activates alternate monitor system #2

- Level trim for system level matching
- Illuminates when engaged

DIM: Activates the Control Room monitor DIM function

- Attenuates the output to the active control room monitor system by the amount set with the DIM LEVEL
- Illuminates when engaged

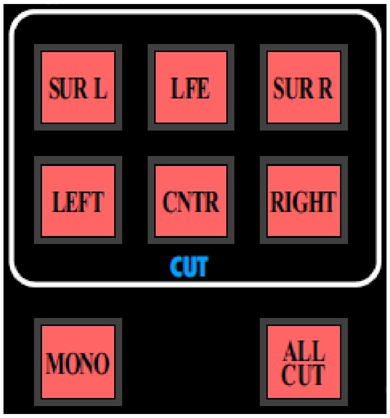
DIM LEVEL: Sets the amount of attenuation applied when the DIM button is engaged

- -∞dB to 0dB range

LT, RT, CN, SL, SR, LF: 5.1 trim pots for setting the **C/R CAL** the control room monitor level

C/R CAL (Control Room Calibrate): Sets the control room monitors to a preset level

- Bypasses the **C/R LEVEL** pot
- Trim pots for 5.1 monitor system
- Useful for film and other sound-for-picture work
- Illuminates when engaged



Individual **CUT** switches are included for each monitor output. These **CUT** switches support the MAIN, ALT 1, and ALT 2 monitor system and apply to the currently selected monitor system.

CUT LEFT: Mutes the Left loudspeaker

- Illuminates when engaged

CUT RIGHT: Mutes the Right loudspeaker

- Illuminates when engaged

CUT CNTR: Mutes the Center loudspeaker

- Illuminates when engaged

CUT SUR L: Mutes the Left Surround loudspeaker

- Illuminates when engaged

CUT SUR R: Mutes the Right Surround loudspeaker

- Illuminates when engaged

MONO: Sums the Left and Right speaker feeds to mono

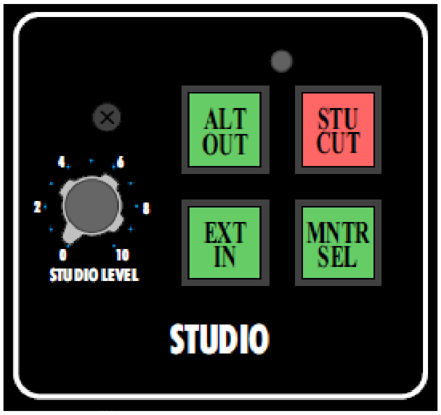
- Illuminates when engaged

ALL CUT: Cuts the feed to all control room speaker feeds

- Illuminates when engaged

Studio Monitoring

Studio monitor feeds are provided for two (2) studio monitor systems.



There is one studio monitor output that normally feeds the main studio monitor system. The output can be switched to feed an alternate studio monitor system (ALT OUT) instead of the main system.

The studio monitor systems can be fed from the following sources:

MNTR SEL (Monitor Select): The selected control room monitor source is routed to the studio loudspeakers

- Illuminates when engaged

STU CUT (Studio Cut): Cuts all studio speaker feeds

- Illuminates when engaged

EXT IN (External Input): Route the signal from the EXT STUDIO MNTR IN patch points to the studio monitor feed

- Illuminates when engaged

ALT OUT (Alternate Output): (Alternate Output): Activates the studio monitor alternate outputs

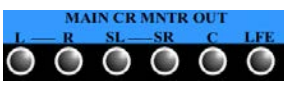
- Activates a second set of studio monitors
- Illuminates when engaged

STUDIO LEVEL: Level control for the studio monitor outputs

- Sets the level for the main studio and alternate monitor system outputs
- Fed from the studio monitor source selectors

Monitor System Patch Points

Control Room Monitor Patch Points



MAIN CR MNTR OUT (Main Control Room Monitor Outputs): Outputs to the MAIN control room monitor system

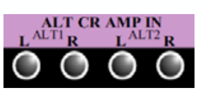
- Half-normalled to MAIN CR MNTR AMP INPUT
  - Outputs split if patched

MAIN CR MNTR AMP INPUT (Main Control Room Monitor Amplifier Inputs): Inputs to the main control room monitor amplifiers

- Half-normalled from MAIN CR MNTR OUT
  - Breaking input: Inserted patch cord will replace the feed to the main monitor amps with the patch cord signal

ALT CR MNTR OUT (Alternate Control Room Monitor Outputs 1 & 2): Outputs to the alternate control room monitor systems 1 & 2

- Half-normalled to ALT CR MNTR AMP INPUT 1 & 2
  - Outputs split if patched



ALT CR MNTR AMP INPUT (Alternate Control Room Monitor Amplifier Inputs 1 & 2): Inputs to the alternate control room monitor amplifiers 1 & 2

- Half-normalled from ALT CR MNTR OUT 1 & 2
  - Breaking input: Inserted patch cord will replace the feed to the alternate monitor amps with the patch cord signal

CAUTION: Exercise great care when patching into any of the Amplifier Input (AMP IN) patch points. There is no level control between these patch points and the inputs to the monitor system amplifiers. Patching unattenuated signals into these patch points can result in very loud output from the Control Room monitor system.

Studio Loudspeaker Patch Points

EXT STUDIO MNTR INPUT (External Studio Monitor Input): External input to the studio monitor source selector

- The **EXT IN** button in the STUDIO monitor controls must be engaged for the external source to be routed to the studio loudspeaker outputs

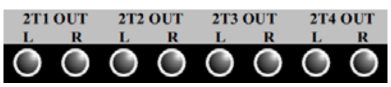
STUDIO OUT: Outputs to the main and alternate studio monitor systems

- L-R feed are the outputs for the main studio loudspeakers
- ALT L-R feed are the outputs for the alternate studio loudspeakers
  - L-R points are half-normalled to STUDIO AMP INPUT
  - ALT L-R points are half-normalled to ALT STUDIO AMP INPUT
  - Outputs split if patched

STUDIO AMP IN: Inputs to the main and alternate studio monitor amplifiers

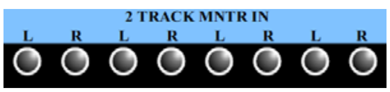
- L-R feed are the inputs for the main studio amps
- ALT L-R are the inputs for the alternate studio loudspeakers
  - L-R points are half-normalled from STUDIO OUT
  - ALT L-R points are half-normalled from ALT STUDIO OUT
  - Patching to these points will replace the feed to the studio monitor amps with the patch cord signal

External Playback Source Patch Points



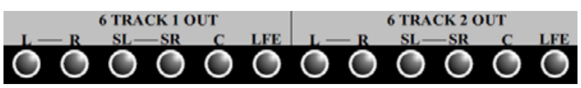
2T1-2T4 OUT (2-Track Outputs 1-4): Returns from the external 2-track record/playback device outputs

- Half-normalled to 2 TRACK MNTR IN 1-4



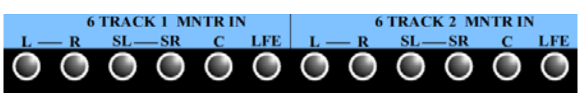
2 TRACK MNTR IN (2-Track Monitor Inputs):

- Feeds the 2T1, 2T2, 2T3, 2T4 monitor playback selectors
- Half-normalled from 2T1 – 2T4 OUT
  - Patching to these points will replace the feed to the 2 track monitor selectors with the patch cord signal



6T1-6T2 OUT (6-Track Outputs 1-2): Returns from the external 5.1 surround playback source outputs

- Left, Right, Left Surround, Right Surround, Center, and LFE
- Half-normalled to 6 TRACK MNTR IN 1-2



6 TRACK MNTR IN (2-Track Monitor Inputs):

- Feeds the 6T1 & 6T2 monitor playback selectors
  - Half-normalled from 6 Track 1-2 OUT
  - Patching to these points will replace the feed to the 6-track monitor selectors with the patch cord signal

Solo Modes and Controls

Solo Modes

The Legacy AXS provides five solo modes:
- After-Fader Listen (AFL)
- Pre-Fader Listen (PFL)
- Pre-Fader Listen 2(PFL 2)
- Solo-In-Place (SIP)
- Mix-Over-Solo

After-Fader Listen (AFL) is the default Solo mode (no buttons engaged).

Solo modes are selected using the SOLO controls in the Master Section. The selected solo mode is activated when a channel or Stereo Return SOLO button is engaged.

<u>After-Fader Listen (AFL)</u>:
- The stereo Solo Bus is fed post pan-pot (stereo)
- Non-destructive
- Control room monitor source is replaced with the Solo Bus
- Solo level control

<u>Pre-Fader Listen (PFL)</u>:
- The stereo Solo Bus is fed post-insert pre-fader (mono)
- Non-destructive
- Control room monitor source is replaced with the Solo Bus
- Solo level control

<u>Pre-Fader Listen 2 (PFL 2)</u>:
- The stereo Solo Bus is fed post-preamp/COMP and pre-EQ (mono)
- PFL 2 allows monitoring of the Direct Output PRE signal
- Non-destructive
- Control room monitor source is replaced with the Solo Bus
- Solo level control

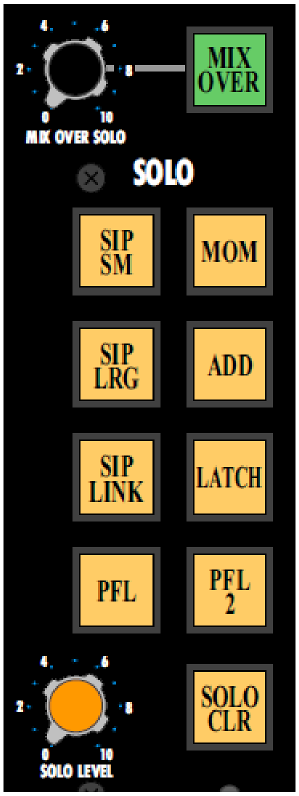
<u>Solo-In-Place (SIP)</u>:
- Destructive (all other channels/returns will mute when a SOLO button is engaged)
- Monitored via the assigned Program Buses
- Post-fader and pan-pot (stereo)
- Solo-In-Place for the Large Fader and Small Fader may be assigned individually
- Solo-In-Place for Large Fader and Small Fader may operate independently or be linked

<u>Mix-Over-Solo:</u>
- Mixes the Solo Bus with the active control room monitor source in the control room monitors
- Level control adjusts the balance between the Solo Bus and the control room monitor source
- AFL and PFL Solo modes only (does not work with Solo-In-Place)
- A channel or return **SOLO** button must be engaged to activate Mix-Over-Solo

Auxiliary Master Solos

The output of the 265 Auxiliary Masters can be soloed by engaging the **AFL** button on the master. After-Fader Listen is the only solo mode available to the Aux Masters.

Solo Controls



SOLO controls operate as follows:

MIX OVER: Activates the "Mix Over Solo" function
- Enables mixing of the Solo Bus and the selected control room monitor source
- Illuminates when engaged

MIX OVER SOLO: Level control for the "Mix Over Solo" function
- Adjusts the balance between the Solo Bus and the control room monitor source when the Mix-Over-Solo function is activated

SIP SM (Solo-In-Place Small): Activates the Solo-In-Place mode for Small Fader path solos
- Illuminates when engaged

SIP LRG (Solo-In-Place Large): Activates the Solo-In-Place mode for Large Fader path solos
- Illuminates when engaged

SIP LINK (Solo-In-Place Link): Links Large Fader and Small Fader SIP solos
- Solo-In-Place must be the active solo mode
- Engaging a Small Fader solo will cause unsoloed/unsafed Large and Small faders to mute and vice versa
- Without SIP LINK active, mutes will only be applied to the fader path where a SIP solo is active
- Illuminates when engaged

PFL (Pre-Fader Listen): Activates the Pre-Fader Listen Solo mode of for all solos
- Fed post-Insert and pre-fader
- Illuminates when engaged

PFL 2 (Pre-Fader Listen 2): Activates the Pre-Fader 2 Listen Solo mode of for all solos
- Fed post-preamp/COMP and pre-EQ
- Allows monitoring of the DIR PRE feed point
- Illuminates when engaged

SOLO CLR (Solo Clear): Disengages all active channel **SOLO**s
- Momentary (does not stay engaged)
- Does not clear AUX MASTER AFL solos
- Flashes to indicate any active SOLO or AFL

SOLO LEVEL: Sets the level of the Solo Bus feed to the Control Room monitors
- Sets AFL Solo level
- Sets PFL Solo level

MOM (Momentary): Activates the momentary solo mode on channels and Stereo Returns
- The solo function will be active only while a **SOLO** button is held down
- One or more **SOLO** buttons may be engaged at once
- Illuminates when engaged

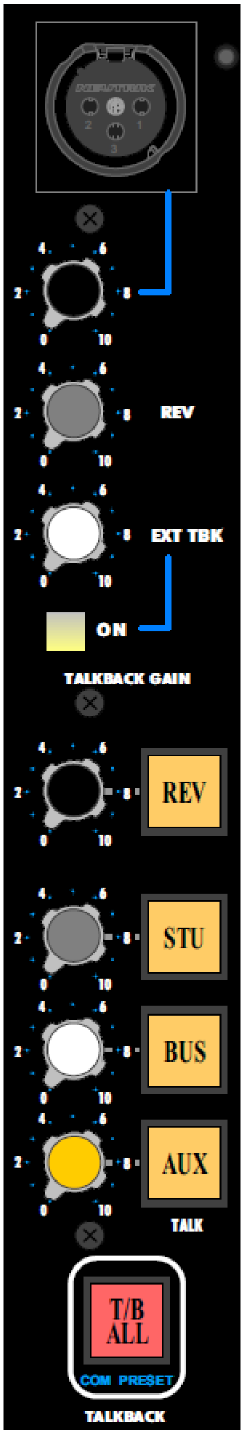
ADD (Additive): Activates the additive solo mode on channels and Stereo Returns
- Multiple **SOLO** buttons may be engaged at once
- Any pressed **SOLO** buttons will remain engaged when pressed and released
- Individual **SOLO** buttons may be disengaged by pressing a them second time
- Illuminates when engaged

LATCH: Activates the Latch solo mode on channels and Stereo Returns

- Only one **SOLO** button may be engaged at once
- The **SOLO** button will remain engaged when pressed and released
- The active **SOLO** button will be disengaged by pressing a it second time
- Pressing a second **SOLO** button will disengage the active **SOLO** and engage the second one
- Illuminates when engaged

Talkback

The Legacy AXS console provides a comprehensive Talkback system with flexible routing and options. The control room Talkback facility has two inputs: a microphone mounted on a “gooseneck” in the console Master Section and an EXT TBK microphone input that appears on the patch bay. These microphones provide control room communication to a variety of destinations.



The Legacy AXS console facilitates two (2) talkback signal flows:

<u>Talkback Mic/EXT TBK</u>: Control room talkback to the following:

- Aux Masters (cue sends)
- Studio loudspeakers
- Multitrack and Program Buses

<u>Reverse Talkback</u>: An external microphone to the control room monitors

- Typically used for studio to control room communication

An external mono signal can be routed to the Talkback output for slating and other purposes.

Talkback Controls

Talkback controls operate as follows:

<u>Talkback Mic Input:</u> Input to the Talkback Mic preamp

- 3-pin female XLR

<u>Talkback Mic (GAIN)</u><u>:</u> Talkback Mic preamp level control

- Sets initial Talkback Mic level
- Black knob located under Talkback Mic XLR

<u>REV (GAIN)</u><u>:</u> Reverse talkback mic preamp level control

- Sets initial Reverse Talkback mic level

<u>EXT TBK (GAIN)</u><u>:</u> External Talkback mic preamp level control

- Sets initial External Talkback mic level

<u>EXT TBK ON</u><u>:</u> On/off switch for the External Talkback source

- Routes the T/B IN MAIN patch point signal to the Talkback output
- Illuminates when engaged

<u>REV (Reverse TALK)</u><u>:</u> Routes the output of the Reverse Talkback mic preamp to the control room monitors

- Engages **DIM** when pressed
- Illuminates while pressed

<u>REV</u> <u>(level):</u> Sets the level of the reverse talkback mic to the control room monitors

<u>STU (Studio TALK)</u><u>:</u> Routes the output of the Talkback Mic preamp to the studio loudspeakers

- Can be activated from the **T/B ALL** button
- Illuminates while pressed

<u>STU</u> <u>(Studio level):</u> Sets the level of the Talkback output to the studio loudspeakers

<u>BUS (TALK)</u><u>:</u> Routes the output of the Talkback Mic preamp to stereo Program Buses and Multitrack Buses

- Can be activated from the **T/B ALL** button
- Illuminates while pressed

<u>BUS</u> <u>(level):</u> Sets the level of the Talkback output to the stereo Program Buses and Multitrack Buses

AUX (TALK): Routes the output of the Talkback Mic preamp to Auxiliary Masters Useful for talking to headphone cues sends

- The **T/B** button must be engaged on the Aux Master for Talkback to be passed to its output
- Can be activated from the **T/B ALL** button
- Illuminates while pressed

AUX (level): Sets the level of the Talkback output to the Auxiliary Masters

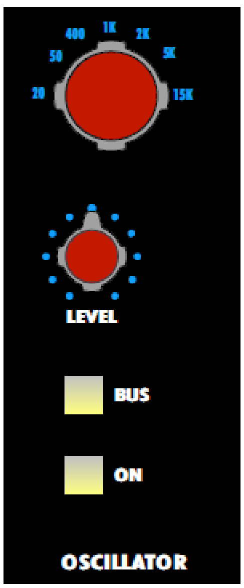
T/B ALL (Talkback All): Activates the Talkback to all destinations.

Talkback Patch Points

T/B IN (Talkback Input MAIN and REV): Inputs for external sources routed to Talkback

- MAIN: Line input an external mono source
  - o The **EXT TBK** button must be engaged for the external source to be routed to the Talkback outputs when Talkback is activated
- REV (Reverse Talkback): Input for the reverse talkback mic

Oscillator



A sine wave Oscillator is provided to generate test tones for calibration and trouble-shooting. The Oscillator output is routed to a pair of patch points when the Oscillator is **ON**. The output can be routed to the Program and Multitrack Buses by engaging the **BUS** button.

Oscillator Controls

ON: On/off switch for the Oscillator output

- The oscillator signal will be present at the OSC OUT patch points
- Activates the Control Room DIM function.
- Illuminates when engaged

BUS: Routes the Oscillator to Program and Multitrack Bus Master outputs

- Illuminates when engaged

LEVEL: Sets the Oscillator output level

Frequency Selector: Selects the Oscillator frequency

- 20Hz, 50Hz, 400Hz, 1kHz, 2kHz, 5kHz, 15kHz

CAUTION: Exercise great care when activating and routing the Oscillator. Control Room monitors should be dimmed, cut, or turned down to a low level. Cue systems and studio loudspeakers should be CUT.

Oscillator Patch Points

OSC OUT (Oscillator Output): Output of the console sine-wave generator

- Signal is present when the Oscillator **ON** button is engaged
- The same signal is present at both points

VU Meters and Peak Indicators

The Legacy AXS console provides the following displays for monitoring levels:

- Main stereo VU meters
- Channel VU meters
- Channel peak indicators
- 200 Module LED meters
- Stereo Return LED meters
- Auxiliary Master LED meters

The following 200 Series modules have built-in LED output meters:

- 205 Direct Input
- 212 Microphone Preamp

The following 200 Series modules have built-in LED gain reduction meters:

- 225 Compressor/Limiter
- 235 Noise Gate/Expander

VU Meter Bridge

The Meter Bridge is populated with the VU meters. The Meter Bridge provides the ability to monitor the levels of the following signal paths:

- Channel fader paths (Small or Large)
- Direct Out
- Multitrack Bus 1-24
- Stereo Program Buses A, B, C
- Grand Master Program Bus
- Auxiliary Masters 7/8, 9/10, 11/12
- Stereo Solo Bus

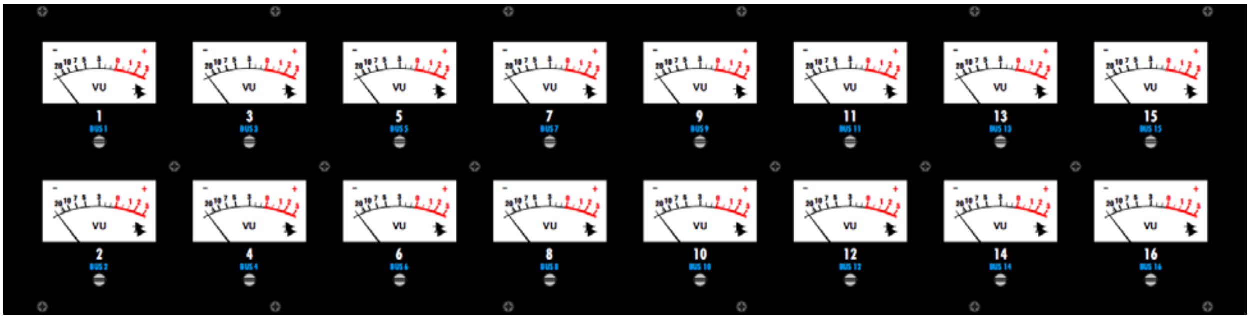
The Meter Bridge has two sections:

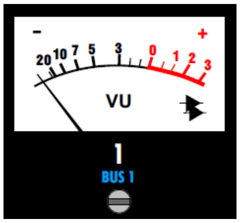
- Channel VU Meters
- Program VU Meters

Channel VU Meters and Peak Indicators

A VU meter is provided for each channel, along with a LED Peak Indicator for each audio path (Small Fader and Large Fader).

The channel VU meters are organized in banks of sixteen (16), one for each installed channel bucket. The meters are stacked in pairs with 1 over 2, 3 over 4, and so on.





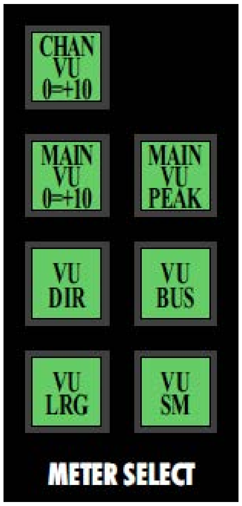
VU (Volume Unit)**:** Channel level indicator

The channel VU meter can be fed from the following points:

- Small Fader path input (pre-EQ & Insert)
- Large Fader path input (pre-EQ & Insert)
- Channel Direct Output
- Associated Multitrack Bus 1-24 outputs (meters 1-24)

- Stereo Program Buses A, B, C, and Grand Master (meters 25-32)

The feed to the channel VU meter is determined by the selection made using the VU SELECT controls in the Master Section. These selections are global and apply to all channel VU Meters.



The METER SELECT controls for the channel VU meters function as follows:

VU LRG (Large Fader): Routes the Large Fader path input to the VU meters
- Illuminates when engaged

VU SM (Small Fader): Routes the Small Fader path input to the VU meters
- Illuminates when engaged

VU DIR (Direct Output): Routes the Direct Output to the VU meters
- Illuminates when engaged

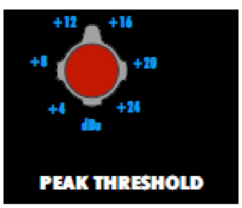
VU BUS (Bus Outputs): Routes the output of the Multitrack Buses 1-24 and the Program Bus STA, STB, STC, and GM to the VU meters
- Multitrack Buses 1-24 on channel VU meters 1-24
- STA, STB, STC, and GM on channel VU meters 25-32
- Illuminates when engaged

CHAN VU 0=+10: Inserts a -10dB pad before the channel VU Meter input
- Use to meter high level signals
- Illuminates when engaged

Channel Peak Indicators

Each fader path is equipped with a LED Peak Indictor. The peak reference level is set on a global basis by the PEAK THRESHOLD selector in the Master Section.

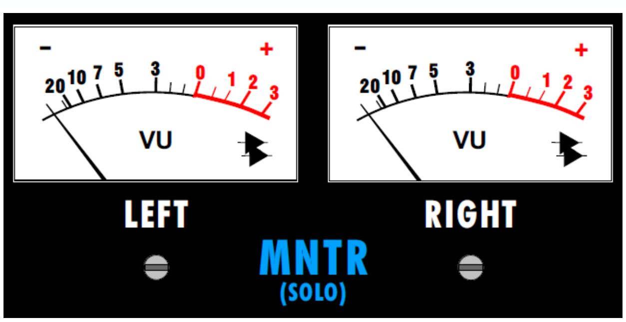
PK (Peak): LED peak indicator for Small and Large Fader paths
- A red LED illuminates when the selected Peak Threshold is reached in the audio path



PEAK THRESHOLD: Sets the peak reference level for the channel Peak Indicators console-wide
- +4dBu to +24dBu range
- Located in the Master Section

Main Stereo Meters

A pair of large, high-quality VU meters are installed in the top bay of the master bucket for indicating the stereo Program, Playback, and Solo levels.



The source for the Main Stereo Meters is selected using the Control Room Monitor source selectors (**MNTR**). Accordingly the Main Meters can be fed from the following sources:
- Stereo Program Masters (PGM MNTR):
  - A, B, or C
  - Grand Master
- Aux Masters 7/8. 9/10, & 11/12
- Solo Bus
- External Playback Sources (PLAY MNTR):
  - 2-Track 1-4
  - 6-Track 1-2 (Left and Right)

While recording and mixing the Main Meters display the level of the stereo Program Master of the Program Master that's being used for mixing and monitoring. When a PFL or AFL **SOLO** is engaged, the control room Monitor Source is replaced with the output of the Solo Bus and the Solo Bus is displayed on the LEFT and RIGHT Main Meters (SOLO).

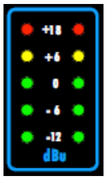
The Main Meters have two controls in the METER SELECT area of the Master Section.

The METER SELECT controls for the Main Meters function as follows:

MAIN VU 0=+10: Inserts a -10dB pad before the Main Meter inputs.
- Use to meter high level signals
- Illuminates when engaged

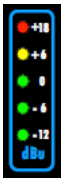
MAIN VU PEAK: Changes the Main Meter ballistics from VU to Peak
- A fixed peak hold circuit feeds the meter
- Illuminates when engaged

Stereo Return Meters



Each Stereo Return is equipped with a stereo 5-segment meter that indicates output level.

Auxiliary Master Meters



Each Auxiliary Masters is equipped with a 5-segment meter that indicates output level.

Power Supply Voltage Indicators

LED voltage indicators provide a quick visual indication of the status of the console power supply units and are most often used in maintenance and trouble-shooting applications. Green indicates the correct voltage level. **Red indicates there is a possible voltage rail failure. Turn the console off and investigate immediately.**
- +16V
- -16V
- +25V
- -25V
- +24V
- +5V
- +48V
- +12V
- +12V
- +7.5V

Patch Bay

All API console patch bays provide extended access throughout the audio signal flow. In the standard configuration 12 patch points are provided for every channel and options for additional points are available. A comprehensive set of patch points that support buses, masters, monitors, and central facilities is also provided. Each patch bay uses TT (tiny telephone) jacks and is 32 points wide. It fits in a standard 19" rack space, but the rack must be heavy-duty.

Normals

In conventional professional patch bays, rows of outputs are located above rows of inputs. To facilitate signal flow between the rows without the need for patching, the output rows are connected internally to the input rows via contacts in the patch points. These connections are called "normalled" patch points or simply "normals." There are two types of "normals:" Half-normal and full-normal.

Half-normal: The connection between output and input is made within the patch bay and does not require patching. Inserting a plug in the output does not break the connection to its normalled input. Inserting a plug in the input or return patch point will break the connection from its normalled output. The connected device input will receive the signal from the inserted plug. In other words: outputs "split" the signal, Inputs "break" the normal.

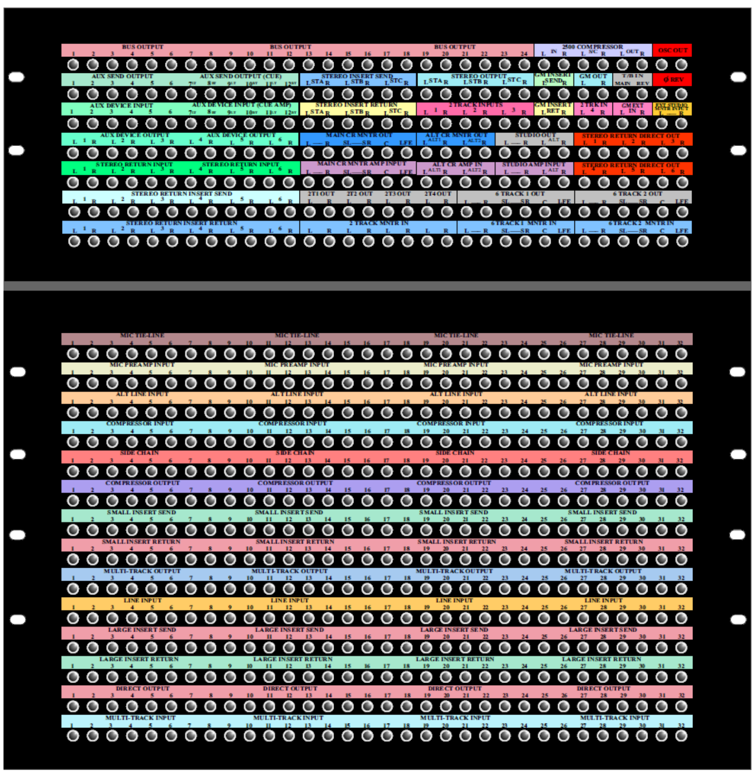
Full-normal: Both patch points break the normalled connection between points when a plug is inserted. Input AND output patch points "break."

Most Legacy AXS patch bay normals are half-normals.

> *NOTE: The MIC TIE-LINE patch points are the only patch points that are fully-normalled (both points will "break" the normal when a patch cord is inserted).*

Patch Bay Overview

The patch bays for 32 thru 64-channel frames can be mounted onboard the console or remotely. The patch bays for console frames larger than 64 channels must be mounted externally.



The patch bay has two sections that support different sections of the console and studio installation:

- Channel Bay
- System Bay

Patch bays are equipped with rear-mounted multi-pin connectors to provide multiple interfacing options.

Channel Patch Bay

The Legacy AXS has 14 patch points associated with channel signal flow. These points are located in the Channel patch bay.

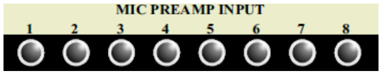
The 14 channel patch points are as follows:

- MIC TIE-LINE
- MIC PREAMP INPUT (upper 200 slot mic preamp in)
- ALT LINE INPUT
- COMPRESSOR INPUT (lower 200 slot)
- SIDE CHAIN (lower 200 slot dynamic processor key input)
- COMPRESSOR OUTPUT (lower 200 slot)
- SMALL INSERT SEND (Small Fader path)
- SMALL INSERT RETURN (Small Fader path)
- MULTITRACK OUTPUT
- LINE INPUT (Large Fader path)
- LARGE INSERT SEND (Large Fader path)
- LARGE INSERT RETURN (Large Fader path)
- DIRECT OUTPUT
- MULTITRACK INPUT

The following section outlines the channel patch points in the order they appear in the Channel patch bay. Depending on console configuration, there will be patch points for 32, 48, 64, or 80 channels in the bay. In the interest of space, patch points for only 8 channels are shown below.

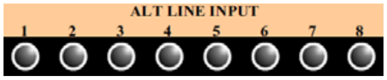
MIC TIE-LINE: Studio microphone outputs

- Full-normal to MIC PREAMP INPUT
- Breaking output: Inserted patch cord will replace studio tie-line connection



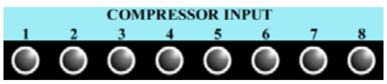
MIC PREAMP INPUT: 212 Mic Preamp inputs
- Balanced, low-impedance, mic-level input
- Full-normal from MIC TIE-LINE patch points
    - Breaking input: Inserted patch cord will replace normalled connection
    - Not used with 205 Direct Input Module

CAUTION: *The MIC PREAMP INPUT patch point will carry 48-Volt phantom power when the* ***48V*** *button is engaged on the associated 212 Mic Preamp. Care should be used and* ***48V*** *should be always be disengaged when patching to these points.*



- ALT LINE INPUT: Secondary line-level inputs
- Balanced, low-impedance, line-level input
- Alternate input to the Small Fader path
    - Half-normal from owner specified source (patch optional)
    - Breaking input: Inserted patch cord will replace normal input

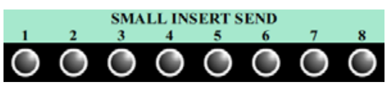
Patch points for alternate external sources (such as virtual instruments, extra DAW returns, etc.) normalled to the ALT LINE INPUT patch points are not provided in the standard configuration. However, the ALT LINE INPUTs are available via a multi-pin connector on the back of the patch bay to allow normal interfacing to these points.



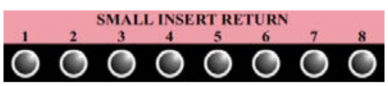
COMPRESSOR INPUT: Lower 200 slot input
- Balanced, low-impedance, line-level input
- Breaking input: Inserted patch cord will replace normalled connection

- SIDE CHAIN (input): Lower 200 slot detection path input
- Balanced, low-impedance, line-level input
- Full-normal from COMPRESSOR INPUT
    - Breaking input: Inserted patch cord will replace normalled connection

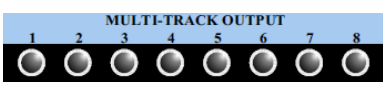
COMPRESSOR OUTPUT: Lower 200 slot output
- Balanced, low-impedance, line-level output
- Outputs split if patched



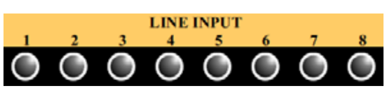
SMALL INSERT SEND: Insert Send from Small Fader path
- Always active
- Pre-fader, post-EQ



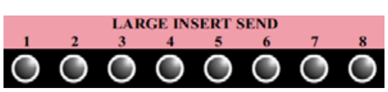
SMALL INSERT RETURN: Insert Return to the Small Fader path
- Patched signal is inserted in the Small Fader path when the **INS** button is engaged
    - Pre-fader, post-EQ



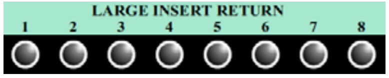
MULTITRACK OUTPUT: Multitrack recorder returns
- Half-normal to LINE INPUT
- Outputs split if patched



LINE INPUT: Main line-level inputs
- Balanced, low-impedance, line-level input
- Half-normal from MULTITRACK OUTPUT
    - Breaking input: Inserted patch cord will replace normalled connection



LARGE INSERT SEND: Insert Send from Large Fader path
- Always active
- Pre-fader, post-EQ



LARGE INSERT RETURN: Insert Return to the Large Fader path

- Patched signal is inserted in the Large Fader path when the **INS** button is engaged
  - Pre-fader, post-EQ

DIRECT OUTPUT: Channel Direct Output (as assigned)

- Fed post-fader from the Large Fader by default
- Half-normalled to the MULTITRACK INPUT
  - Outputs split if patched

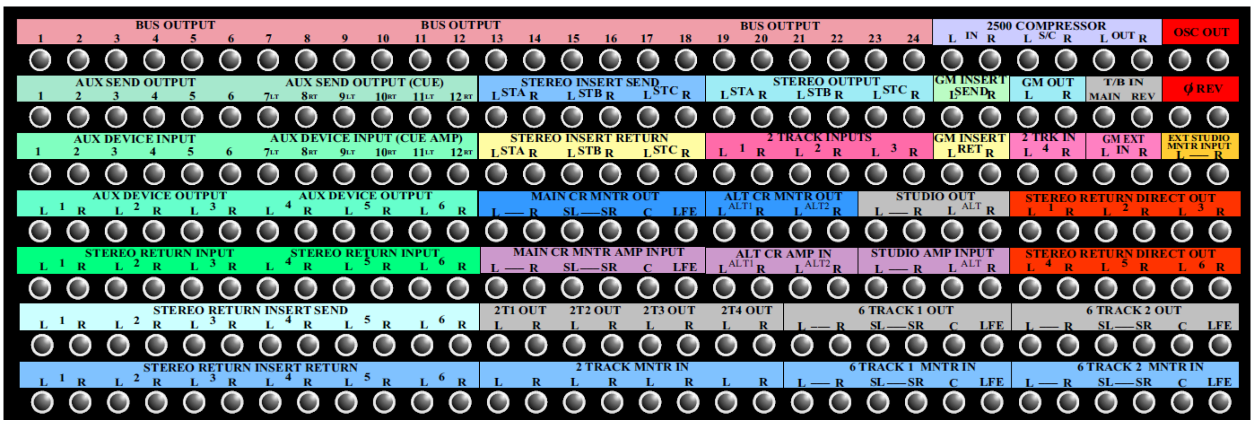
MULTITRACK INPUT: Connection to multitrack recorder inputs

- Half-normalled from the DIRECT OUTPUT
- Breaking input: Inserted patch cord will replace normalled connection

NOTE: The multitrack recorder may alternately be fed from the Multitrack Buses by patching the BUS OUTPUT patch points to the MULTITRACK INPUT patch points.

System Patch Bay

The master and system patch points are located in the System patch bay.

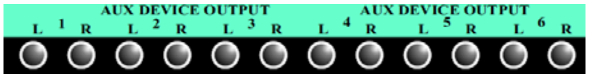


The System patch bay contains patch points that support the following:

- Stereo Returns: Inputs, inserts, and Direct Outputs
- Stereo Program Masters: Insert and output
- Stereo Grand Master: External input, insert, and output
- 2500C Stereo Bus Compressor: Inputs, side chain inputs, and outputs
- Multitrack Bus outputs
- Auxiliary Master outputs
- Auxiliary Device inputs and outputs
- 2-Track recorder inputs
- 2-Track record/playback device outputs
- Monitor return inputs
- Control room and studio monitor outputs
- Control room and studio amplifier inputs
- Talkback inputs
- Oscillator outputs
- Ø phase reverse (polarity inverter)

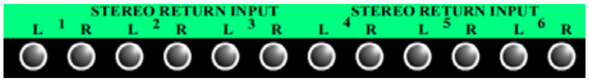
Stereo Return Patch Points

The following section outlines Stereo Return patch points.



AUX DEVICE OUTPUT: Stereo output connections for external devices

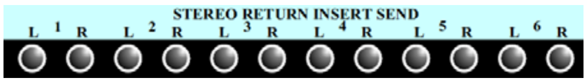
- Half-normalled to the STEREO RETURN INPUT
- Outputs split if patched



STEREO RETURN INPUT:

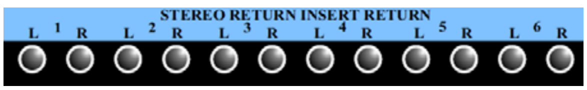
- Balanced, line-level, low-impedance
- Half-normalled from the AUX DEVICE OUTPUT
- Breaking input: Inserted patch cord will replace normalled connection

STEREO RETURN DIRECT OUTPUT: Stereo Return Direct Output



STEREO RETURN INSERT SEND: Stereo Return Insert Send

- Pre-fader
- Always active
- Outputs split if patched



STEREO RETURN INSERT RETURN: Stereo Return Insert Return

- Patched signal is inserted in the Stereo Return signal flow when the **INS** button is engaged
- Pre-fader

Stereo Program Master Patch Points

The following section outlines Stereo Program Master patch points.

STEREO INSERT SEND: Insert Send from stereo Program Masters A, B, C

- Always active
- Pre-fader

STEREO INSERT RETURN: Insert Return to stereo Program Masters A, B, C

- Patched signal is inserted in the Program Master signal flow when the **INS** button is engaged
- Pre-fader

STEREO OUTPUT: Main outputs for stereo Program Masters A, B, C

- Half-normalled to 2 TRACK INPUTS 1, 2, 3
- Outputs split if patched

2 TRACK INPUTS 1-3: Input to 2-track recorders 1-3

- Half-normalled from STEREO OUTPUT A, B, C
- Breaking input: Inserted patch cord will replace normalled Connection

GM EXT IN (Grand Master External Input): Stereo external inputs to the Grand Master summing bus

- The **EXT** button on the Grand Master must be engaged for the external source to be routed to the Grand Master summing bus

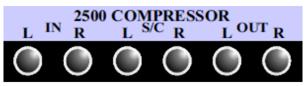
GM INSERT SEND (Grand Master): Insert Send from the stereo Grand Master
- Always active
- Pre-fader

GM INSERT RETURN (Grand Master): Insert Return to the stereo Grand Master
- Patched signal is inserted in the Program Master signal flow when the **INS** button is engaged
  - Pre-fader

GM OUTPUT (Grand Master): Main outputs for the stereo Grand Master
- Half-normalled to 2 TRACK INPUT 4
- Outputs split if patched

2 TRACK INPUT 4: Input to 2-track recorder 4
- Half-normalled from STEREO OUTPUT A, B, C
- Breaking input: Inserted patch cord will replace normalled connection

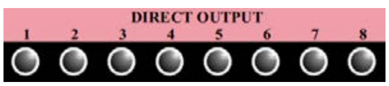
2500C Stereo Bus Compressor Patch Points



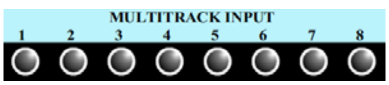
2500 COMPRESSOR: 2500C stereo Bus Compressor patch points
- IN: Left/Right compressor input
- S/C (side chain): Left/Right input to the compressor detection path
- OUT: Left/Right compressor output

NOTE: The 2500C Stereo Bus Compressor must be patched to be used.

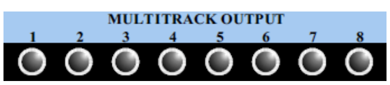
Multitrack Patch Points



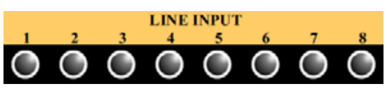
DIRECT OUTPUT: Channel Direct Output (as assigned)
- Fed post-fader from the Large Fader by default
- Half-normalled to the MULTITRACK INPUT
  - Outputs split if patched



MULTITRACK INPUT: Connection to multitrack recorder inputs
- Half-normalled from DIRECT OUTPUT
- Breaking input: Inserted patch cord will replace normalled connection



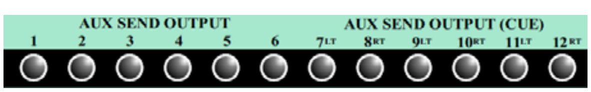
MULTITRACK OUTPUT: Multitrack recorder returns
- Half-normal to LINE INPUT
- Outputs split if patched



LINE INPUT: Main line-level inputs
- Balanced, low-impedance, line-level input
- Half-normal from MULTITRACK OUTPUT
  - Breaking input: Inserted patch cord will replace normalled connection

NOTE: The DIRECT OUTPUT, MULTITRACK INPUT/OUTPUT, and LINE IN patch points are associated with the channels and are not part of the Multitrack Bus system. Since BUS OUTPUTS are commonly patched to feed the multitrack recorder, the DIRECT OUTPUT, MULTITRACK INPUT/OUTPUT, and LINE IN patch points are shown here.

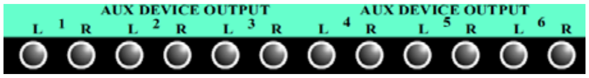
Auxiliary Master Patch Points



AUX SEND OUTPUT: Auxiliary bus outputs
- 1-6 are mono aux outputs
- 7-12 are stereo (LT/RT) aux/cue outputs
  - Half-normalled to AUX DEVICE INPUT
  - Outputs split if patched

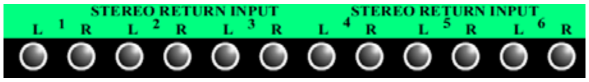
AUX DEVICE INPUT: Connection to external effects devices, cue systems, and/or other devices

- 1-6 are fed from mono sends
  - 7-12 are fed from stereo (LT/RT) aux/cue sends
  - Half-normalled from the AUX SEND OUTPUT
  - Breaking input: Inserted patch cord will replace normalled connection



AUX DEVICE OUTPUT: Stereo output connections for external devices

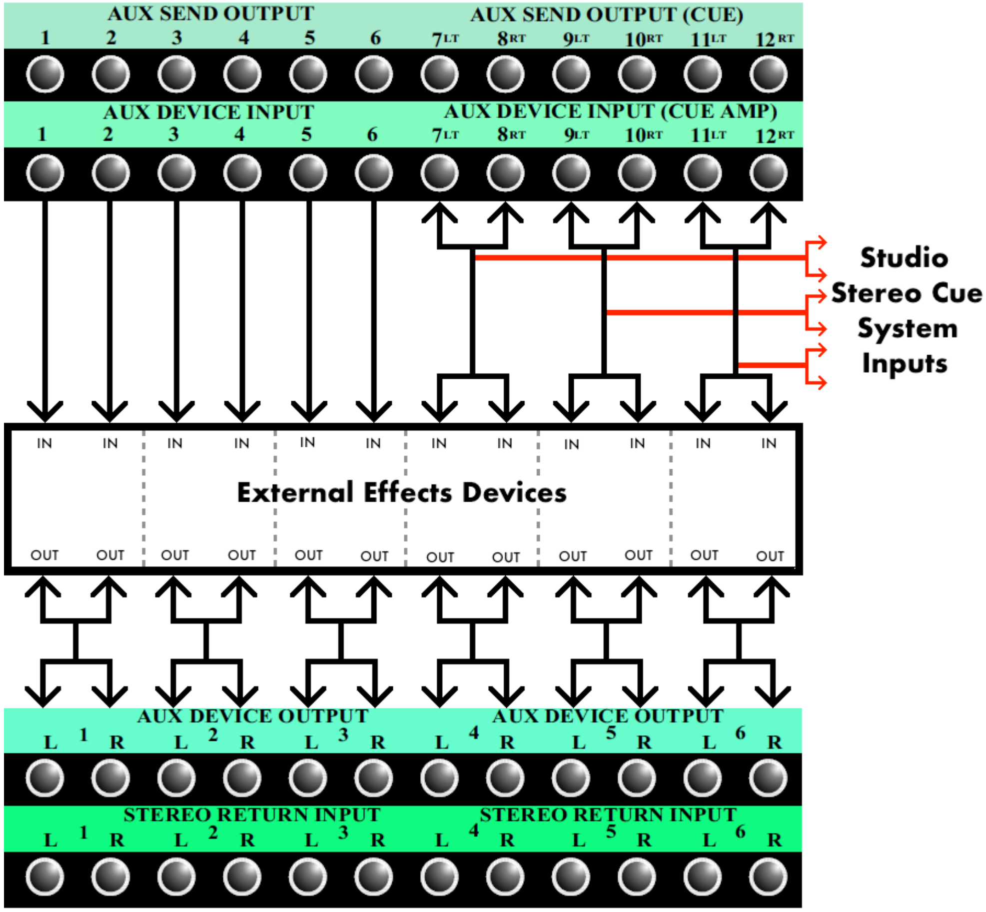
- Half-normalled to the STEREO RETURN INPUT
  - Outputs split if patched



STEREO RETURN INPUT:

- Balanced, line-level, low-impedance
- Half-normalled from the AUX DEVICE OUTPUT
  - Breaking input: Inserted patch cord will replace normalled connection

To create normalled connections to headphone cue systems and effects devices, through external effects processors, and back to the program buses for mixing, the Legacy AXS provides AUX DEVICE INPUT and AUX DEVICE OUTPUT patch points to. The AUX DEVICE INPUT (CUE AMP) patch points are half-normalled from the AUX SEND OUTPUT (CUE) patch points. The AUX DEVICE OUTPUT patch points are half-normalled to the STEREO RETURN INPUT patch points. The diagram below illustrates an example of how theses patch points can be interfaced to provide normalled connections to external effects devices and cues systems.


AUX SEND OUTPUT
AUX SEND OUTPUT (CUE)
1 2 3 4 5 6 7LT 8RT 9LT 10RT 11LT 12RT
AUX DEVICE INPUT
AUX DEVICE INPUT (CUE AMP)
1 2 3 4 5 6 7LT 8RT 9LT 10RT 11LT 12RT
Studio
Stereo Cue
System
Inputs
IN IN IN IN IN IN IN IN IN IN IN IN
External Effects Devices
OUT OUT OUT OUT OUT OUT OUT OUT OUT OUT OUT OUT
AUX DEVICE OUTPUT
AUX DEVICE OUTPUT
L 1 R L 2 R L 3 R L 4 R L 5 R L 6 R
STEREO RETURN INPUT
STEREO RETURN INPUT
L 1 R L 2 R L 3 R L 4 R L 5 R L 6 R

Monitor System Patch Points

Control Room Monitor Patch Points

<u>MAIN CR MNTR OUT (Main Control Room Monitor Outputs):</u> Outputs to the MAIN control room monitor system

- Half-normalled to MAIN CR MNTR AMP INPUT
  - Outputs split if patched

<u>MAIN CR MNTR AMP INPUT (Main Control Room Monitor Amplifier Inputs):</u> Inputs to the main control room monitor amplifiers

- Half-normalled from MAIN CR MNTR OUT
  - Breaking input: Inserted patch cord will replace the feed to the main monitor amps with the patch cord signal

<u>ALT CR MNTR OUT (Alternate Control Room Monitor Outputs 1 & 2):</u> Outputs to the alternate control room monitor systems 1 & 2

- Half-normalled to ALT CR MNTR AMP INPUT 1 & 2
  - Outputs split if patched

<u>ALT CR MNTR AMP INPUT (Alternate Control Room Monitor Amplifier Inputs 1 & 2):</u> Inputs to the alternate control room monitor amplifiers 1 & 2

- Half-normalled from ALT CR MNTR OUT 1 & 2
  - Breaking input: Inserted patch cord will replace the feed to the alternate monitor amps with the patch cord signal

CAUTION: Exercise great care when patching into any of the Amplifier Input (AMP IN) patch points. There is no level control between these patch points and the inputs to the monitor system amplifiers. Patching unattenuated signals into these patch points can result in very loud output from the Control Room monitor system.

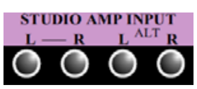
Studio Loudspeaker Patch Points

<u>EXT STUDIO MNTR INPUT (External Studio Monitor Input)</u>: External input to the studio monitor source selector

- The **EXT IN** button in the STUDIO monitor controls must be engaged for the external source to be routed to the studio loudspeaker outputs

<u>STUDIO OUT:</u> Outputs to the main and alternate studio monitor systems

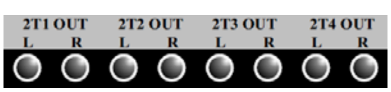
- L-R feed are the outputs for the main studio loudspeakers
- ALT L-R feed are the outputs for the alternate studio loudspeakers
  - L-R points are half-normalled to STUDIO AMP INPUT
  - ALT L-R points are half-normalled to ALT STUDIO AMP INPUT
  - Outputs split if patched



<u>STUDIO AMP IN:</u> Inputs to the main and alternate studio monitor amplifiers

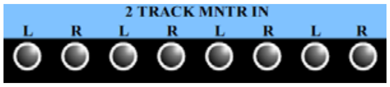
- L-R feed are the inputs for the main studio amps
- ALT L-R are the inputs for the alternate studio loudspeakers
  - L-R points are half-normalled from STUDIO OUT
  - ALT L-R points are half-normalled from ALT STUDIO OUT
  - Patching to these points will replace the feed to the studio monitor amps with the patch cord signal

External Playback Source Patch Points



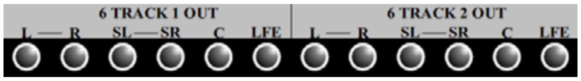
<u>2T1-2T4 OUT (2-Track Outputs 1-4)</u>: Returns from the external 2-track record/playback device outputs

- Half-normalled to 2 TRACK MNTR IN 1-4



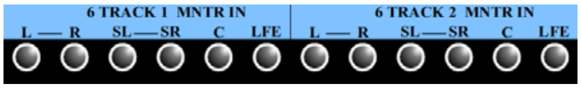
2 TRACK MNTR IN (2-Track Monitor Inputs):

- Feeds the 2T1, 2T2, 2T3, 2T4 monitor playback selectors
- Half-normalled from 2T1 – 2T4 OUT
  - Patching to these points will replace the feed to the 2 track monitor selectors with the patch cord signal



6T1-6T2 OUT (6-Track Outputs 1-2): Returns from the external 5.1 surround playback source outputs

- Left, Right, Left Surround, Right Surround, Center, and LFE
- Half-normalled to 6 TRACK MNTR IN 1-2



6 TRACK MNTR IN (2-Track Monitor Inputs):

- Feeds the 6T1 & 6T2 monitor playback selectors
- Half-normalled from 6 Track 1-2 OUT
- Patching to these points will replace the feed to the 6-track monitor selectors with the patch cord signal

Talkback Patch Points

T/B IN (Talkback Input MAIN and REV): Inputs for external sources routed to Talkback

- MAIN: Mic input an external mono source
  - The **EXT TBK** button must be engaged for the external source to be routed to the Talkback outputs when Talkback is activated
- REV (Reverse Talkback): Input for the reverse talkback mic

Oscillator Patch Points

OSC OUT (Oscillator Output): Output of the console sine-wave generator

- Signal is present when the Oscillator **ON** button is engaged
- The same signal is present at both points

Phase Reverse (polarity inverter)

Ø REV (Phase Reverse): Polarity inverting patch points

- The two points are wired out of polarity with each other
- A signal patched into one jack will be out of polarity (phase reversed) at the other point

API 2500C Stereo Bus Compressor

A BASIC OVERVIEW OF THE API 25OO COMPRESSOR

The 25OO has several unique features that will allow you to tailor the sound of the compression in many different directions. There are two features that can only be found on this unit, as they are either patented or patent-pending.

The patented "THRUST" feature has been used for many years in the famed ATI Paragon and Paragon II consoles as well as the Pro-6 Input Strip. This circuit places a filter in front of the RMS detector, with a slope of 10dB per decade, which is the inverse of the pink noise energy curve. In acoustics, the pink noise curve is used to equalize energy vs. frequency over the audio spectrum, as sound requires more low frequency energy than high frequency energy to sound correct to your ear. In Hi-Fi equipment, a "LOUDNESS" contour is used to equalize the music at lower levels so it sounds correct. Even with this curve, there is still a substantial amount of low frequency information compared to high frequency information in the audio signal path. When that signal is fed into an RMS detector, the detector will process the signal into a DC control voltage based on 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 switching the THRUST button in either the MED or LOUD positions, 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 much less compressed.

The LINK feature is also unique to the 25OO. First, there is a variable link control, ranging from IND (independent) to 50% through 100%. The variable linking allows combining of the left and right control voltages over a range, minimizing the interaction between channels, while still linking them to retain the stereo image. While engaged, the LINK control has a selectable HI-Pass, LOW-Pass and BAND Pass filter that can be inserted, but only into the LINK circuit. This feature can reducing peaks from cross- linking, reduce low frequencies from cross-linking or both. The value of this feature is shown when the signal contains a large amount of percussive instruments, spread around the stereo field. When all other compressors are linked, any peak in the left side will result in a gain reduction in the right side that will shift the stereo image, resulting in a less than desirable effect. By inserting the high or low pass filters, you can eliminate this undesired image shift while still linking the preferred frequency range.

The 25OO has two complete, identical compressor circuits, from the input through the RMS detector, the VCA's and to the Output section. When many compressors are linked to the "STEREO LINK" mode, only one channel is controlling the compression. When the 25OO is linked, the control voltages are summed together, but both channels are still detecting their own control voltages. This also eliminates changing of the tone of the compressor when linking is used. Many popular units actually change when linked!

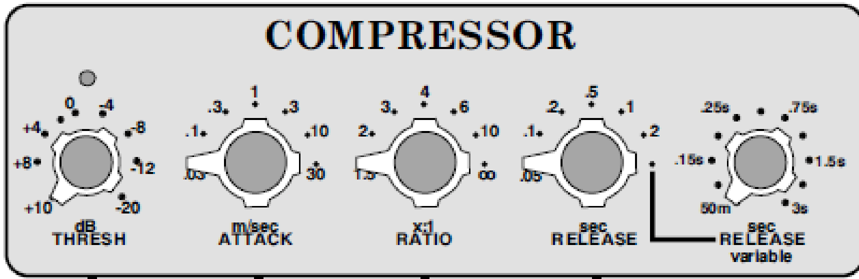
Additionally, each channel uses multiple VCA's to minimize noise and distortion found in single VCA compressors. The signal path from the input to the output is ALL DISCRETE, using the API 2510 and the API 2520.

The GAIN pot can be either switched IN for manual gain control, or it can be left OUT and the 25OO will bring the output level up and down automatically, keeping the signal at the same level regardless of where the THRESH or RATIO controls are set. This is the same as the API 525 Ceiling control.

Just to the right of the VU meters, there is a set screw adjustment. This adjustment is to rock or tilt the signal left or right when in the auto gain mode, allowing subtle corrections in the stereo image if it is off center when compressing. This control only affects the image when the 25OO is above the threshold, and does not do anything when the signal is not being compressed. Usually, this control should be in a vertical position.

One tip to get started: We try to design all of our processing modules with the understanding that there is never enough time to read a manual and learn all of the features in the 10 seconds that is allowed during a setup. If you simply place all of the knobs at the 12:OO position, the 25OO, and all of the 2OO series modules will have a useful, but conservative effect on the signal.

Compressor Section



THRESH:
This control sets the THRESHHOLD from +10 dBu to -20 dBu. Both channels are set independently with this control. Each channel has its own RMS detector and can operate as two single compressors or one stereo compressor. Even when using the LINK control, each channel ALWAYS has its own RMS detector for accuracy. This control is continuously variable. The THRESH control also effects the gain when in the AUTO gain make-up mode.

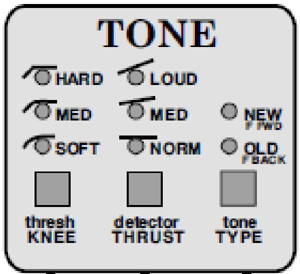
ATTACK:
This control sets the ATTACK time of each channel from 30 microseconds to 30 milliseconds. There are seven positions to choose from, 30u/sec, 100u/sec, 300u/sec, 1 m/sec, 3 m/sec, 10 m/sec and 30 m/sec. This rotary switch allows repeatability, while offering a wide range of settings.

RATIO:
This control sets the compression RATIO of each channel from 1.5:1 to INF:1 or above 20:1. There are seven positions to choose from, 1.5:1, 2:1, 3:1, 4:1, 6:1, 10:1 and INF:1. This rotary switch allows repeatability while offering a wide range of settings. The RATIO control also affects the gain when in the AUTO gain make-up mode.

RELEASE:
This control sets the RELEASE of each channel of the compressor, covering a wide range of release times including the last position, which switches it to the VARIALBLE RELEASE control to the right of it. There are seven positions to choose from, 50m/sec., 100m/sec, 200m/sec., 500/m/sec, 1 sec, 2 sec, and VARIABLE.

VARIABLE RELEASE:
This control sets the RELEASE time with a continuously variable pot covering a range from 50m/sec. to 3 seconds. This works when the RELEASE rotary switch control is fully clockwise. This allows for a continuously variable release, with the ability to match the "bounce" of a song with the release time.

Tone Section

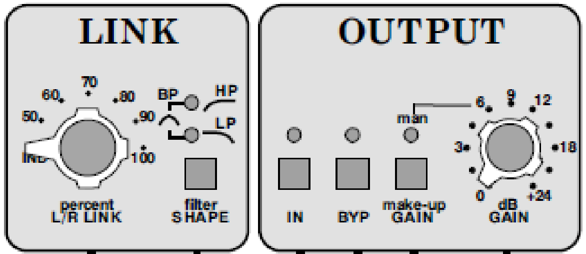


KNEE:
This control sets the KNEE, or how the point where the compressor begins to reduce the gain of the signal applied to the unit. When in the HARD position, the gain reduction begins at the set ratio and is a sharp transition into compression. The MED position has a slight "fade-in" up to the set ratio. The SOFT position has a gradual "fade-in" to the set ratio. The HARD position is very noticeable and the SOFT position is very subtle and similar to an "over-easy" type KNEE.

THRUST:
This control sets the THRUST, a patented circuit that inserts a hi-pass filter at the input of the RMS detector, limiting its response to lower frequencies. In the NORM mode, there is no filter and the 2500 compresses like most units on the market today. When MED is selected, there is a slight attenuation of the low frequencies and a slight boost of the high frequencies, with a flat midrange affecting the signal going into the RMS detector. This reduces the low frequencies from pumping the compressor as much and increases the sensitivity of the RMS detector to the higher frequencies, affecting the higher frequency peaks of the signal. When LOUD is selected, there is a gradual, linear filter, down 15 dB at 20 Hz and up 15dB at 20K Hz, equalizing the energy going into the RMS detector. This decreases the way the low frequencies pump the compressor and increases the way the higher frequencies are compressed. The overall difference is a noticeable increase of punch and low frequencies, but a uniformly compressed signal. It is the "little more punch" switch.

TYPE:
This control sets the TYPE, or where the signal for the RMS detector comes from. In the NEW mode, the compressor works like most newer types of compressors, as in most of the VCA based units. This is called FEED-FORWARD compression, where the RMS detector sends a signal to the VCA that is an exact ratio of the desired compression, set by the RATIO control. When the OLD position is selected, the RMS detector gets the signal from the output of the VCA, and then feeds the VCA a signal based on a set ratio of that signal. This type of compression is called FEEDBACK compression and is how the older API 525, 1176 type and 660 type compressors worked. The NEW mode is much harder and the OLD mode is very smooth. When SOFT, LOUD, and OLD is selected you can hardly hear the compression.

Link and Output Section



L/R LINK:
This control sets the left to right LINK percentage. Most compressors only allow 100% linking between channels. The 2500 allows for linking starting at IND, which is 0% and 50% to 100% in six additional steps. At the same time, each channel is still controlled by its own detector, preventing loading and slaving from one side, which many times creates errors. This control mixed each channel's RMS detector controls together according to the switch position.

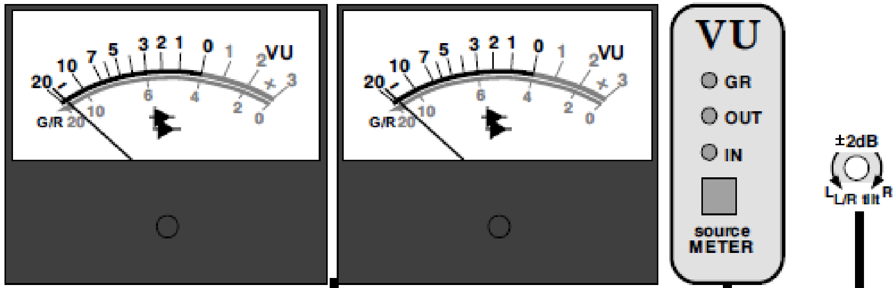
SHAPE:
This control adjusts the SHAPE of the LINK control voltage mixing. There are two filters, a high pass filter, eliminating the lows, a low pass filter, eliminating the highs and a combination of both filters creating a band pass filter. The value of this circuit allows the LINK control voltage to NOT include certain frequencies, such as the low frequency end of each channel, preventing the low frequencies from linking from the left side to the right side, etc. When the different combinations of the filters are used, sounds like sharp percussive instruments on one channel will not couple to the other channel and cause it to compress.

IN and BYPASS:
The IN switch is a soft IN/OUT button that defeats the compression action silently, but allows the signal to pass through the 2500.

The BYP switch is a relay hard-wire bypass that routes the signal directly from the input XLR to the output XLR, without going through any electronics. If the power fails to the 2500, the BYP automatically engages, preventing any loss of signal during the power fault.

GAIN:
When the GAIN switch is pressed IN, the GAIN pot controls the amount of make-up gain needed to maintain the desired overall gain after the compression action reduces the output level. When in the OUT position, the output make-up gain is automatically maintained regardless of the position of the THRESH or RATIO controls. If the input signal is below +4 dBu, rotating the THRESH control will increase the output level until the output level is around +4 dBu, and then as the signal gets compressed, it will maintain an even output level, allowing THRESH and RATIO adjustments without having to re-adjust the make-up gain. This is very useful during situations where an adjustment needs to be made without disturbing the output level to tape or air. This function is similar to the API 525 ceiling control.

Meter and Trim Section



VU METER:
The VU Meters displays the INPUT and OUTPUT levels in dBu, where +4dBu is O VU. This reference point can be adjusted internally to other reference levels.

The GAIN scale shows the amount of gain reduction during compression, with the 0 point being all the way to the right, allowing more resolution of the gain reduction scale. The range of the meter is from 0 (no gain reduction) to 20 db of gain reduction. The compressor can compress up to 30 dB of reduction.

VU:
The VU switch selects the INPUT, OUTPUT or GAIN REDUCTION to be displayed on the VU meters. It is a silent function and is isolated so the VU selection does not effect or load the signal in any way.

TILT:
The TILT adjustment is a screwdriver trim of the compression control voltage, allowing tilting of the compression left or right to equalize uneven signal compression that may offset the stereo image. This control adjusts about 2 dB in either direction, and does not affect the uncompressed signal, which is always unity in to out.

This control is useful in a situation where there is a slight difference in the stereo image when a signal is compressed, but not when it is below the compression threshold. Normally, this control should be straight up.

When in doubt start with all the controls straight up!

2500C Setup Sheet

ARTIST:

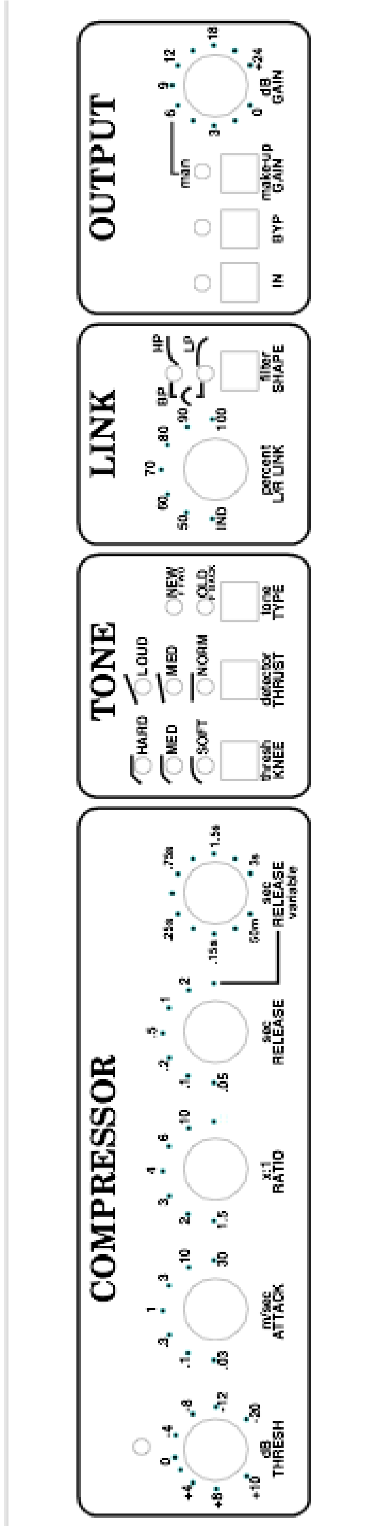
ENGINEER:

DATE:

NOTES:

DRAWN BY:

STUDIO:



ARTIST:

ENGINEER:

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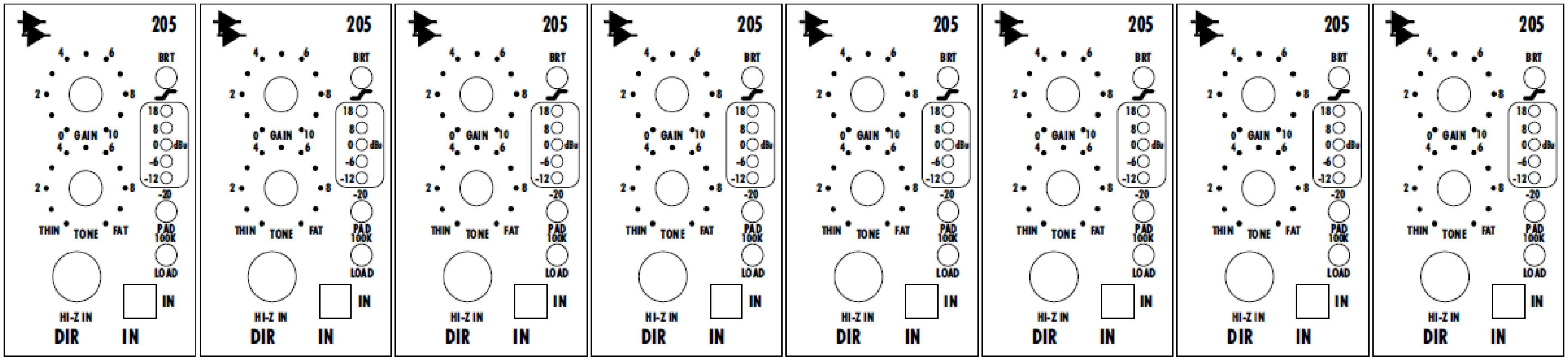

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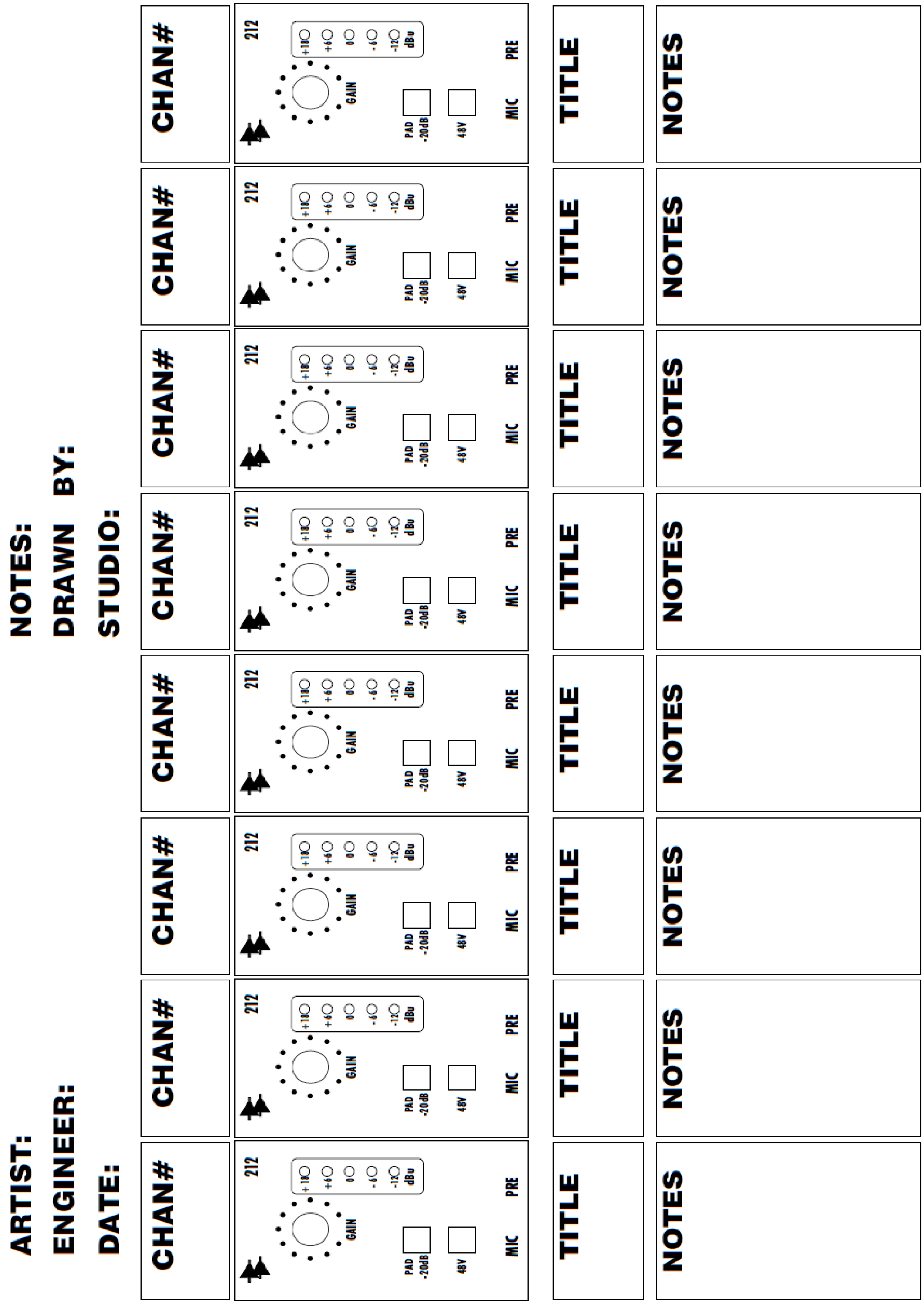
NOTES	NOTES	NOTES	NOTES	NOTES	NOTES	NOTES	NOTES

212 Setup Sheet



ARTIST:

ENGINEER:

DATE:

NOTES:

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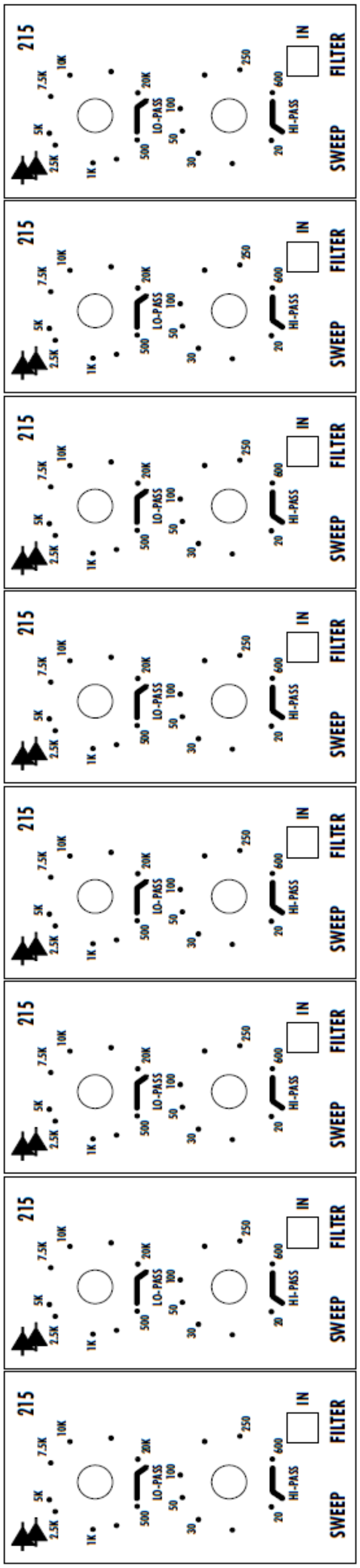

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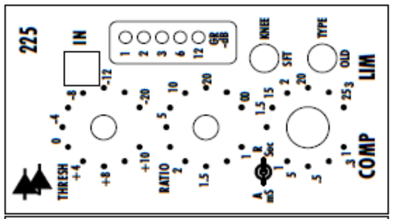
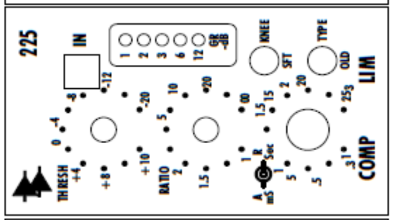
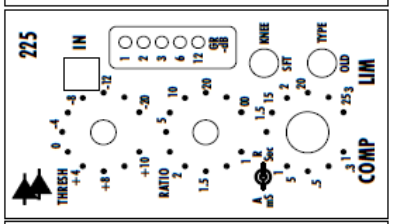
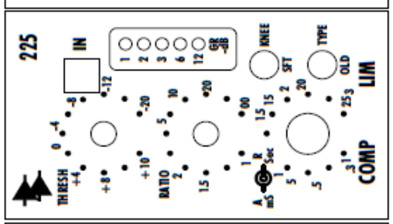
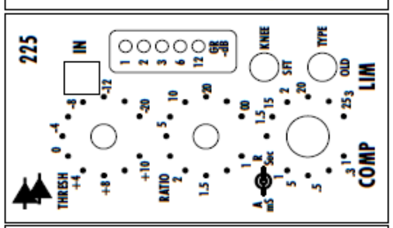
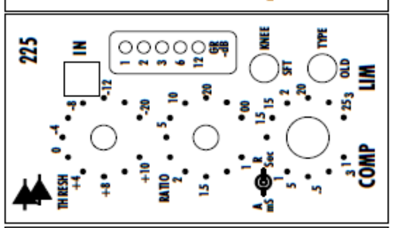
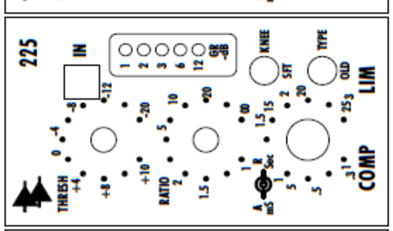
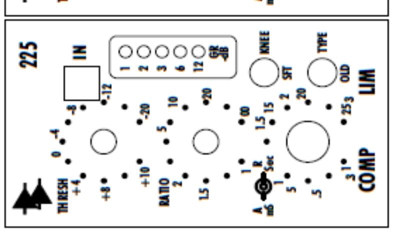
CHAN#		TITLE	NOTES
CHAN#		TITLE	NOTES
CHAN#		TITLE	NOTES
CHAN#		TITLE	NOTES
CHAN#		TITLE	NOTES
CHAN#		TITLE	NOTES
CHAN#		TITLE	NOTES
CHAN#		TITLE	NOTES

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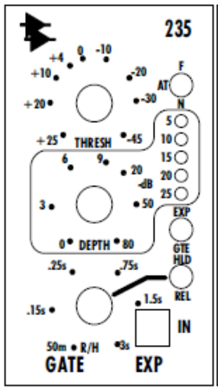	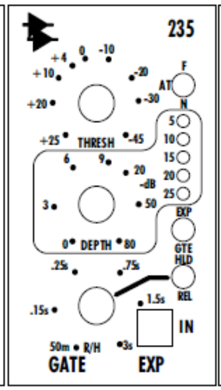	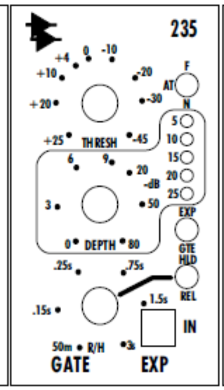	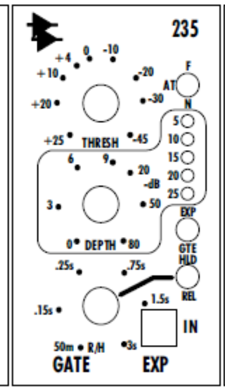	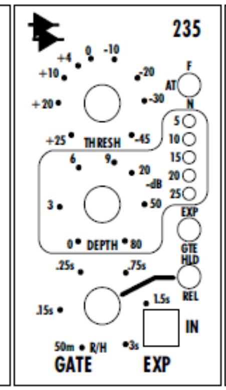	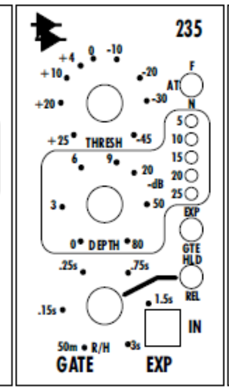	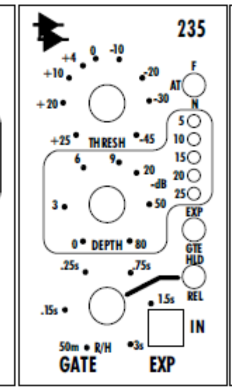	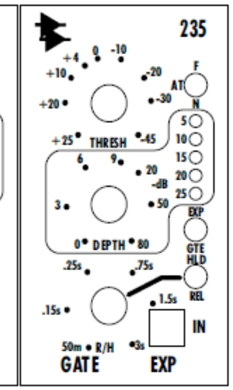
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550A Set Up Sheet

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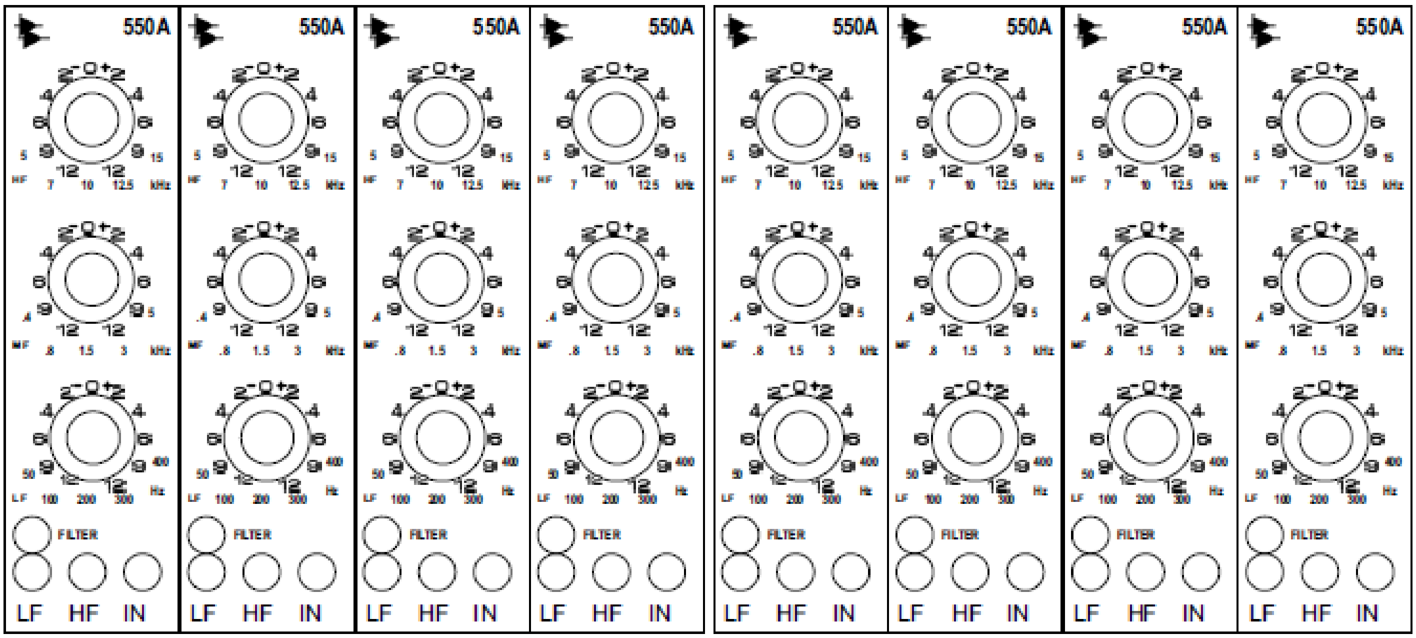

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550b Set Up Sheet

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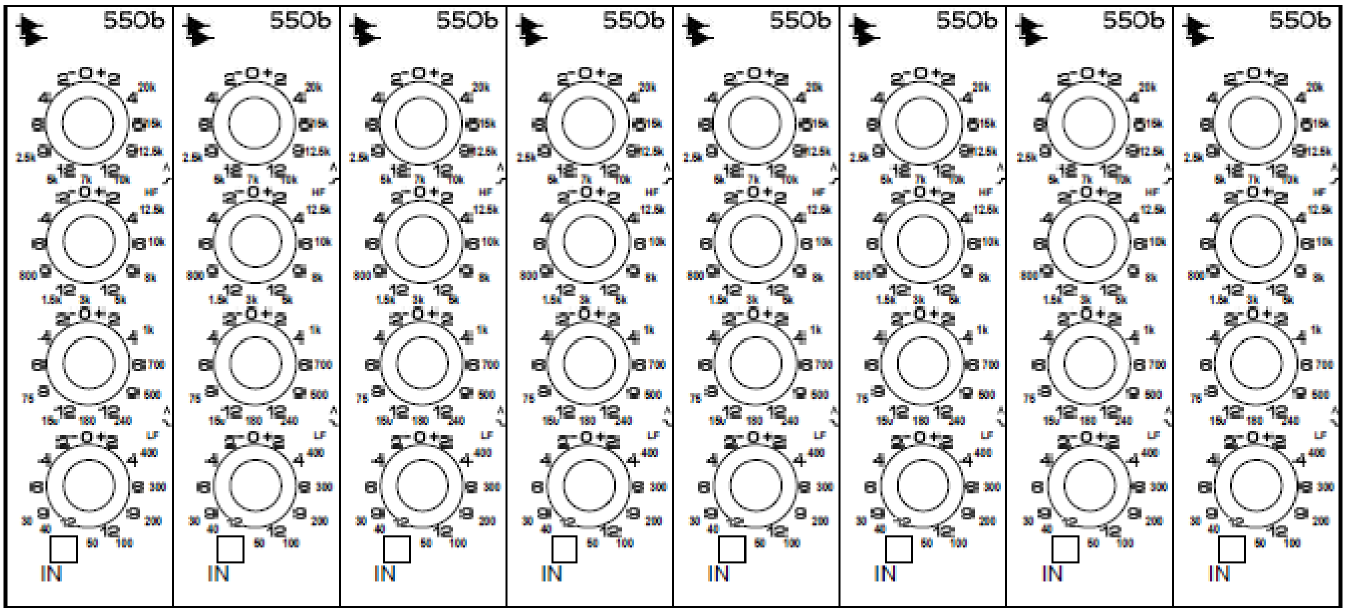

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560 Set Up Sheet

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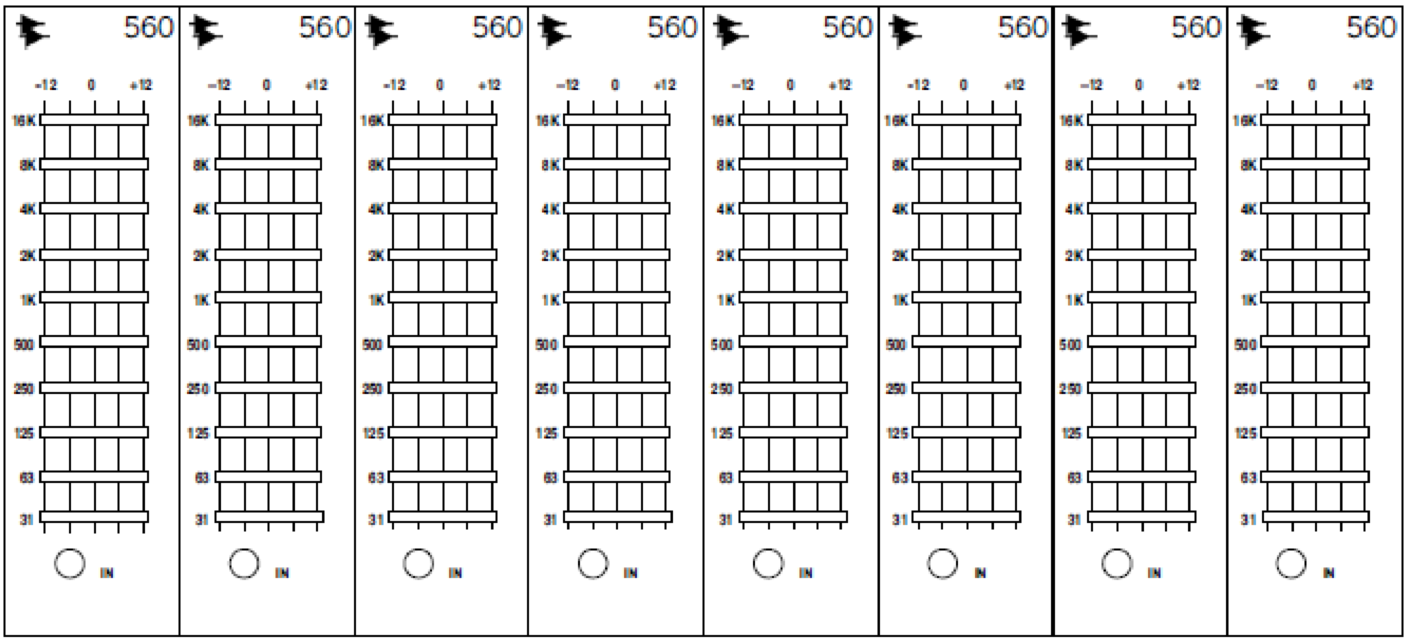

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624 Setup Sheet

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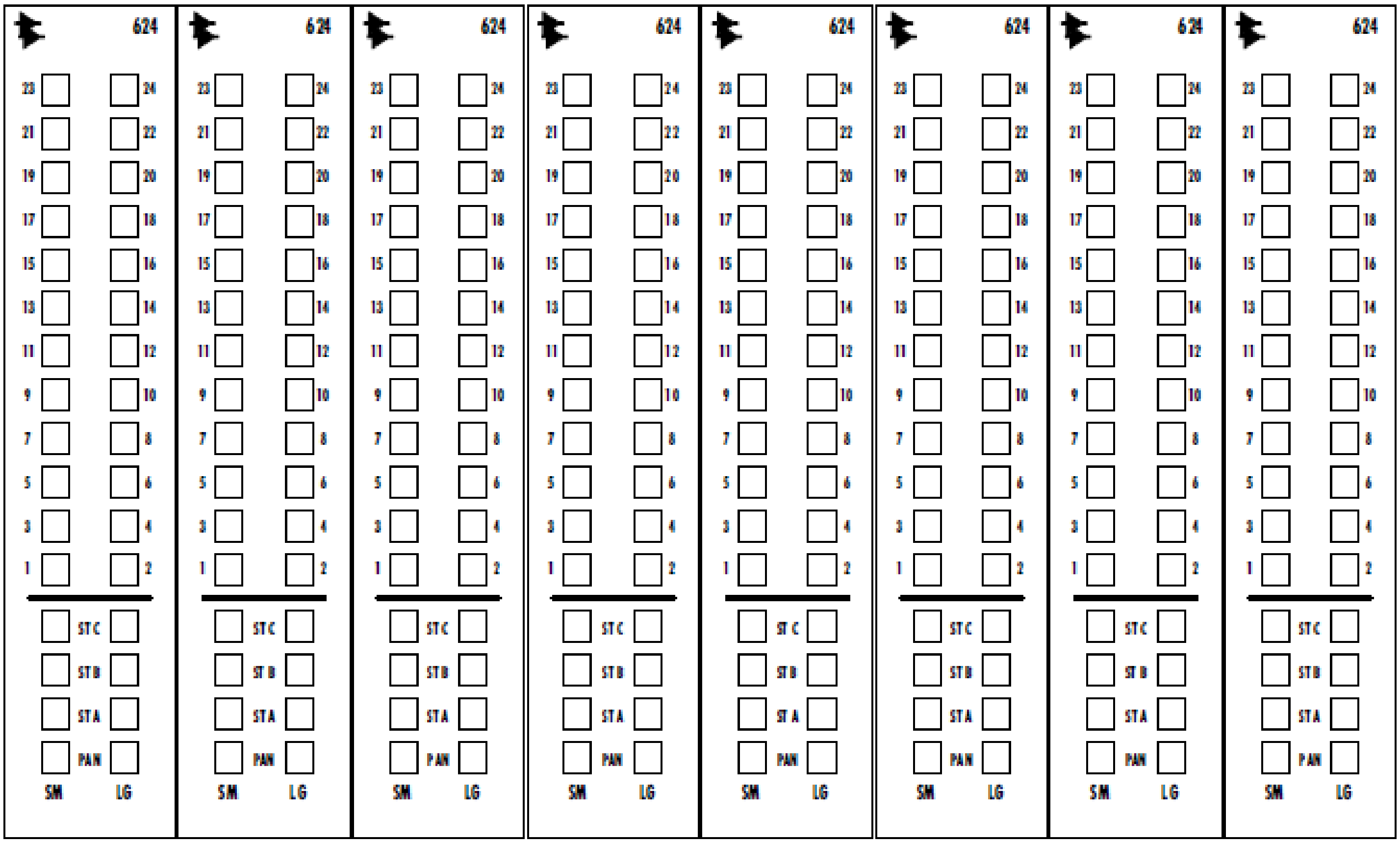

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968 Setup Sheet

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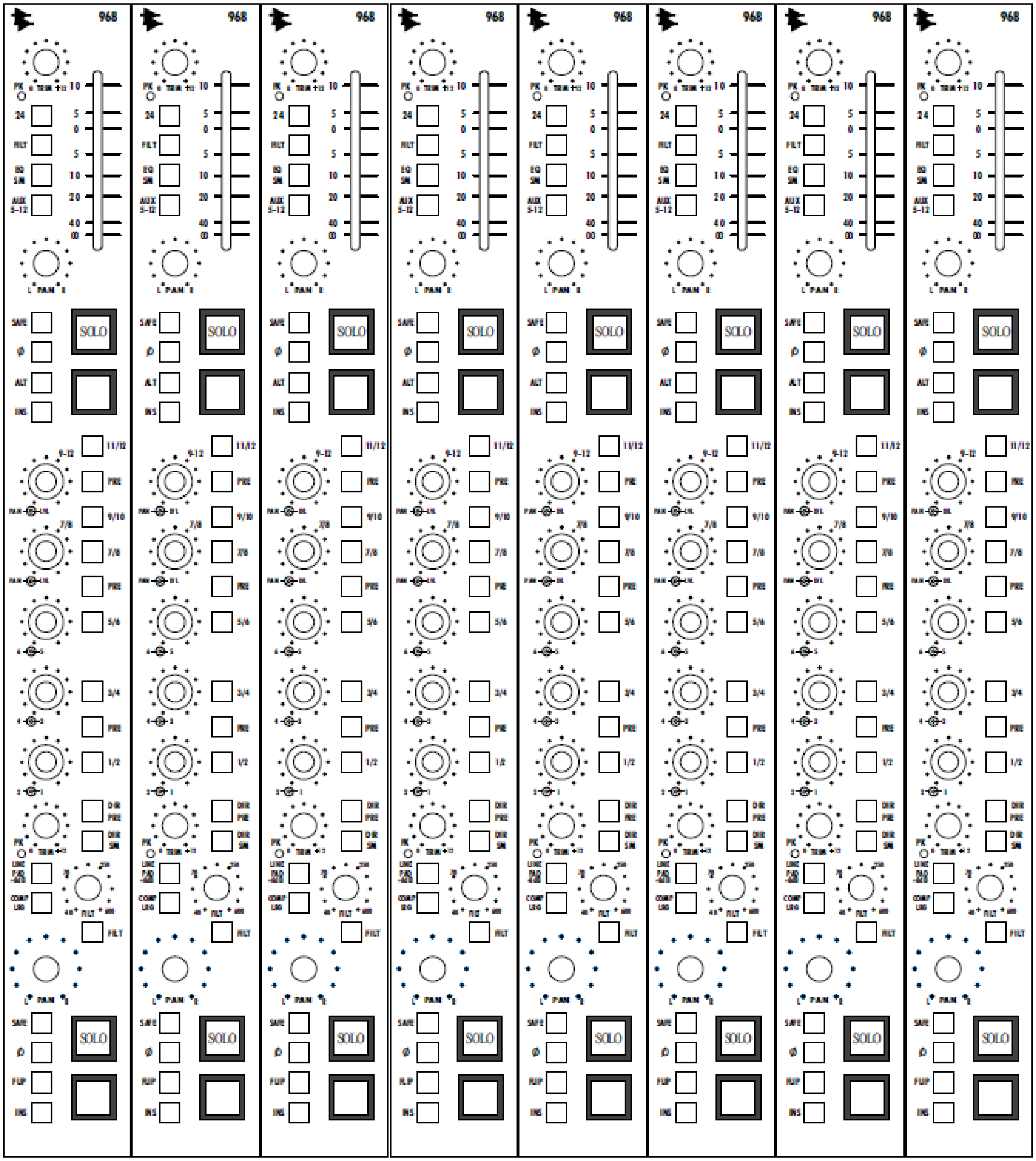

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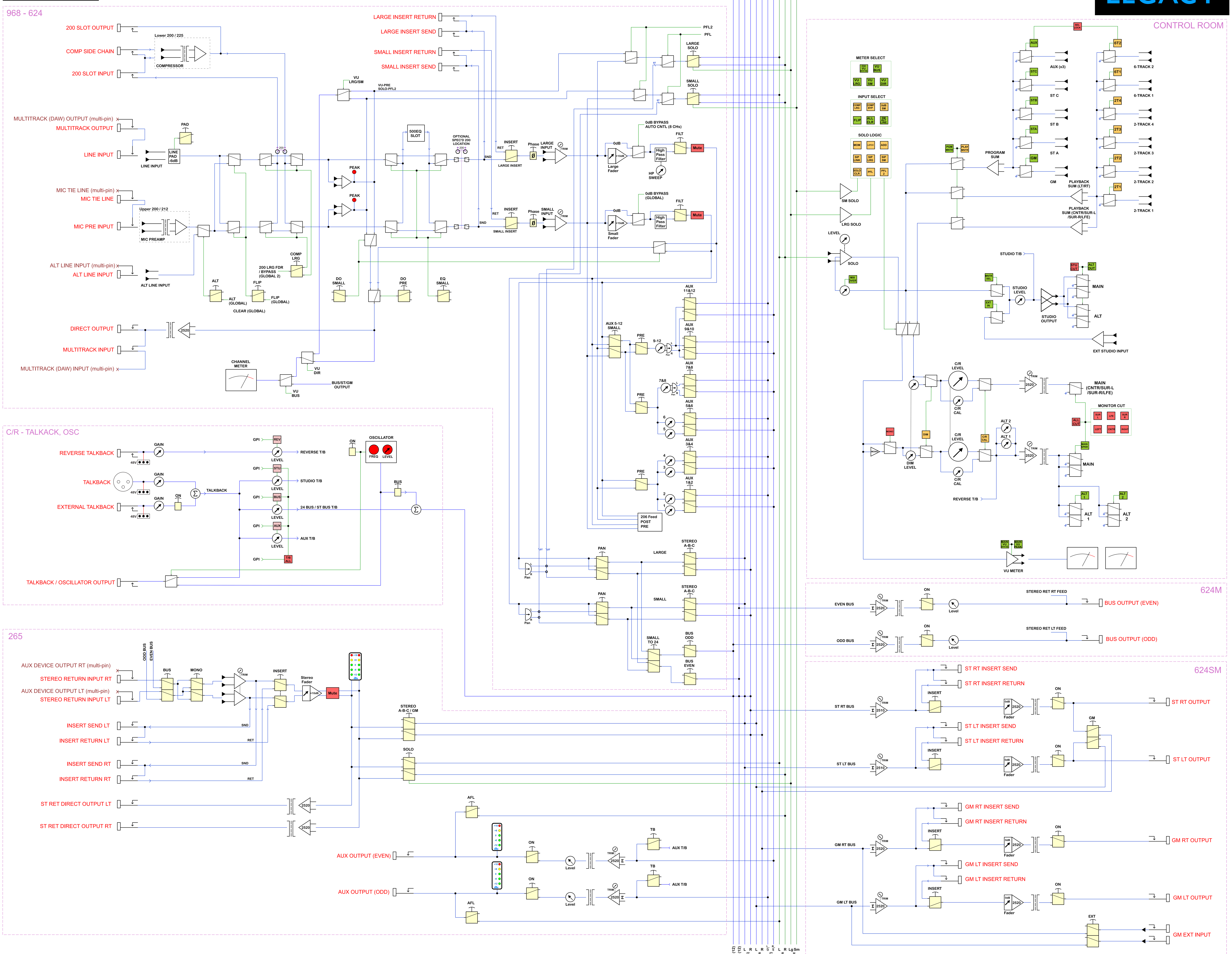
LEGACY AXS Block Diagram

Key:
- Audio
- Control
- Module



REV C, 2018-11-27

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