StudioLive™ AI-Series Mixers

Digital Mixing System with Active Integration™

Owner's Manual

PreSonus®

www.presonus.com

Table of Contents

1   Overview

1.1   Introduction

Thank you for purchasing your PreSonus® StudioLive™ AI-series Performance and Recording Digital Mixer. PreSonus Audio Electronics has designed the StudioLive utilizing high-grade components to ensure optimum performance that will last a lifetime. Loaded with 32/24/16 high-headroom, XMAX™ microphone preamplifiers; a built-in 48x34/40x26/32x18 96 kHz FireWire s800 recording and playback interface; Fat Channel processing with 4-band EQs, compressors, limiters, and expander/gates; reverb and delay effects; 14/10/6 aux buses; 4 subgroups; extensive LED metering; mixer save/recall; channel-strip save/recall/copy/paste; talkback; and more, StudioLive breaks new boundaries for music performance and production. All you need is a compatible computer with a FireWire connection, a few microphones and cables, speakers, and your instruments, and you are ready to record in the studio or in front of a live audience!

We encourage you to contact us with questions or comments regarding this product. PreSonus Audio Electronics is committed to constant product improvement, and we value your suggestions highly. We believe the best way to achieve our goal of constant product improvement is by listening to the real experts: our valued customers. We appreciate the support you have shown us through the purchase of this product.

For technical support, *please see Section 7.1: Troubleshooting.*

1.2   About This Manual

We suggest that you use this manual to familiarize yourself with the features, applications, and connection procedures for your StudioLive before trying to connect it to your computer. This will help you avoid problems during installation and setup. This manual covers hardware functions for all three StudioLive AI-series mixers. When functional differences are called out, the 32.4.2AI will be mentioned first, followed by the 24.4.2AI and then the 16.4.2AI. *Note: All illustrated examples use images of the 32.4.2AI.*

A separate manual, also included with your StudioLive mixer, covers the StudioLive AI Software Library, as well as instructions for connecting and using your StudioLive with a computer.

Throughout this manual you will find **Power User Tips**. These tips provide mixing tricks, some of which are unique to the StudioLive, as well as explanations of various audio terms. In addition, you will find an assortment of audio tutorials at the back of this manual. These tutorials cover everything from microphone placement to equalizer and compression-setting suggestions and are included to help you get the most from your StudioLive mixer.

Thank you, once again, for buying our product. We are confident that you will enjoy your StudioLive!

1.3   Summary of StudioLive AI-series Mixer Hardware Features

- 24-bit resolution, up to 96 kHz sampling rate
- 32/24/16 Class A XMAX microphone preamplifiers with individual phantom power
- 32/24/16 line-level inputs
- 32/24/16 analog inserts
- 32/24/16 direct outputs
- 2 balanced stereo auxiliary inputs
- Analog stereo tape I/O
- 4 subgroups
- 14/10/6 auxiliary buses
- High-definition AD/DA converters (118 dB dynamic range)
- 32-bit floating point digital mixing and effects processing
- 48x34/40x26/32x18 digital recording interface with two FireWire s800 (IEEE 1394b) ports
- S/PDIF output
- Optional Thunderbolt and Dante I/O cards
- Scene automation with load/save/recall of all settings
- Fat Channel with: high-pass filter, gate, compressor, limiter, 4-band parametric EQ
- Alt EQ & Dynamics button lets you A/B two different Fat Channel settings
- USB 2.0 control port supports included USB wireless LAN adapter
- 4 master DSP effects with Load and Save
    - 2 with reverb
    - 2 with delay
- 6 mute groups with All On/All Off (32.4.2AI only)
- 6 user-assignable Quick Scene Recall buttons (32.4.2 only)
- 31-band graphic EQ on each analog aux and the main bus
- 100 mm faders
- Fast-acting LED meters

- Talkback-communication system

- Windows® and Mac® compatible

- Powerful StudioLive Software Library includes:

    - Virtual StudioLive-AI (VSL-AI) advanced editor/librarian/control software

    - StudioLive Remote-AI (SL Remote-AI) remote control app for iPad® (free from Apple App Store)

    - QMix™-AI remote aux-mix app for iPhone®/ iPod touch® (free from Apple App Store)

    - Capture™ integrated multitrack-recording software

    - Studio One® Artist digital audio workstation with more than 6 GB of plug-ins, loops, and sounds

1.4   What is in the Box

In addition to a Visual Quick Start Guide, your StudioLive package contains the following:

- PreSonus StudioLive AI-series digital recording and performance mixer
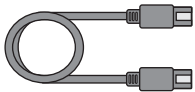
- 6-foot (1.8 meter), 9-pin-to-9-pin FireWire s800 cable
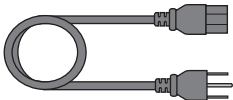
- IEC power cord

- USB wireless LAN adapter

- StudioLive Software Library Download instructions for:

    - PreSonus Studio One Artist program plus gigabytes of third-party content

    - PreSonus Capture with demo sessions

    - Universal Control AI with VSL-AI

    - SL Remote AI for iPad®

    - QMix™ AI for iPhone®/iPod® touch

2   Getting Started

Before you begin, here are a few general rules of thumb:

- Always turn the Main fader and both the Monitor and Phones knobs in the Monitor section down before making connections.

- Before plugging or unplugging a microphone while other channels are active, mute the channel to which you are connecting.

- Your faders should be set on or near the "U" mark whenever possible. The "U" indicates unity gain, meaning the signal is neither boosted nor attenuated. If the main output of your StudioLive is too high or too low when your faders are at or near unity, you can use the output-level knob on the rear panel of the StudioLive to adjust the level up or down until you have achieved the optimal volume.

- Do not allow your inputs to clip. Watch the level meters; when the LEDs near the Clip mark, the top LED will illuminate, indicating that the analog-to-digital converters are in danger of being overdriven. Overdriving the converters will cause digital distortion, which sounds terrible. The XMAX™ preamps in your StudioLive provide plenty of headroom; take advantage of it.

Your P.A. and studio equipment should be powered on in the following order:

1. Sound sources (keyboards, direct boxes, microphones, etc.) connected to the StudioLive inputs

2. StudioLive AI mixer

3. Computer (if applicable)

4. Power amplifiers or powered monitors

When it's time to power down, your system should be turned off in the reverse order. Now that you know what not to do, let's get some audio going!

2.1   Level Setting Procedure
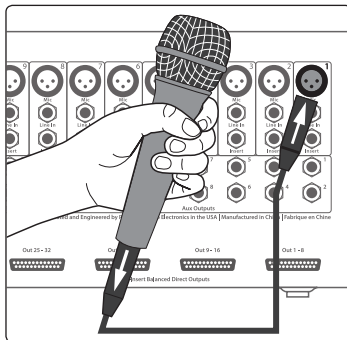
1. Grab a microphone and a mic cable and plug them into the StudioLive's Channel 1 mic input.
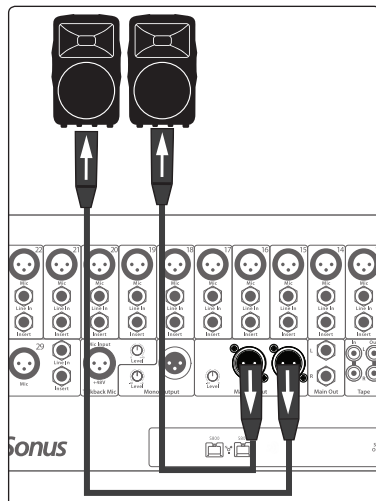
2. Connect the Main outs (TRS or XLR) of your StudioLive to your power amplifier or powered monitors.

4
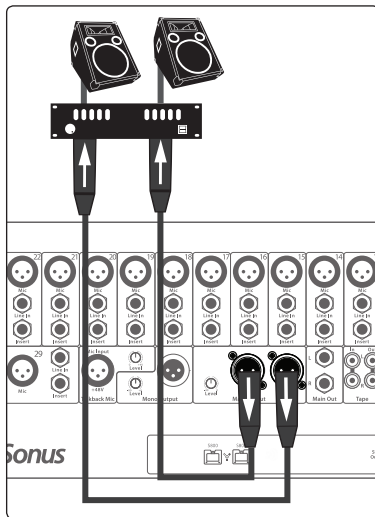
3. If you're using passive speakers, connect them to your power amplifier using speaker cable.
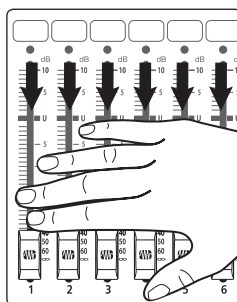
4. Bring down all the faders on your StudioLive to the ∞ setting.
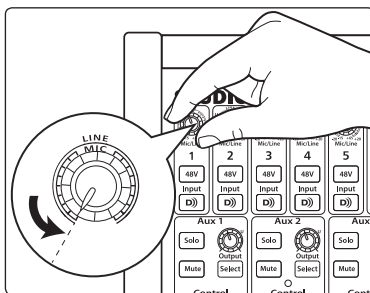
5. Make sure that the Mic/Line knob on Channel 1 is all the way counter-clockwise.
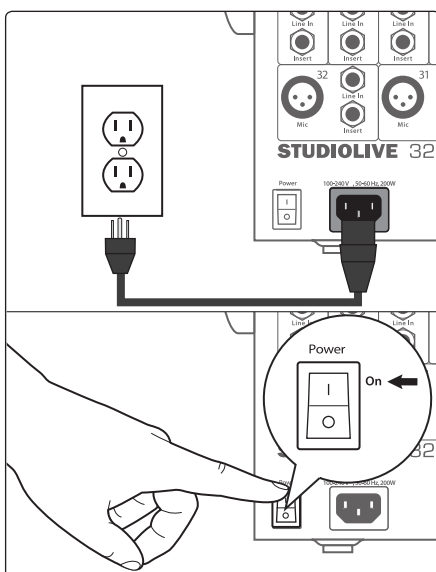
6. Plug your StudioLive into a power outlet and turn it on.
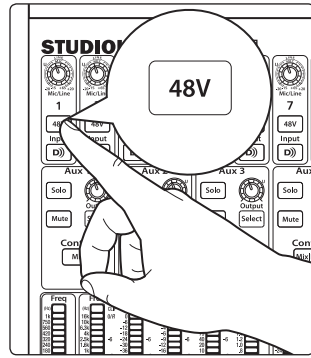
7. If your microphone requires phantom power, engage the 48V button on Channel 1 of your StudioLive.
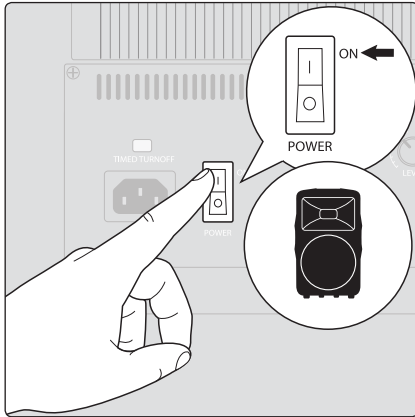
8. Turn on your amplifier or powered monitors.
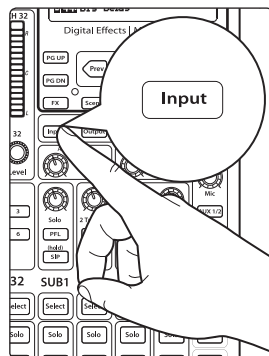
9. Press the Input button in the Meters section.

10. Speak or sing into your microphone at approximately the same volume you expect during the performance.
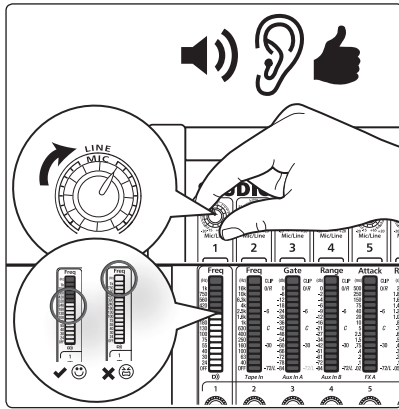
11. Turn the trim knob on Channel 1 clockwise while watching the first meter in the Fat Channel. Adjust the Channel 1 trim knob until a little more than half of the green LEDs are lit. The red LED at the top of the meter should never light up.
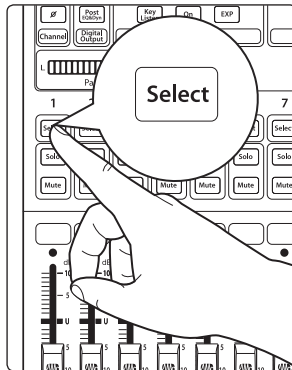
12. Press the Select button on Channel 1.
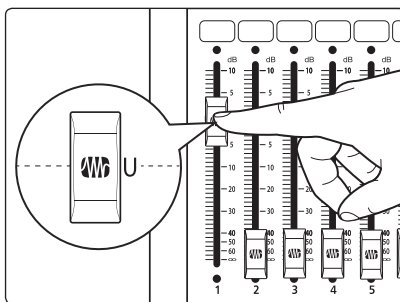
13. Raise the Channel 1 fader until it reaches "U" (unity gain).
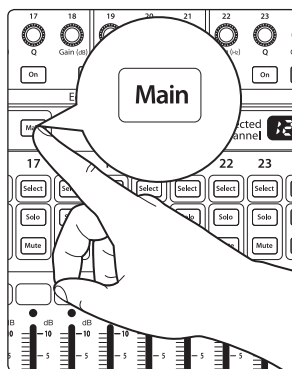
14. Press the Main button in the Assign section of the Fat Channel so that it illuminates. This routes the channel to the main output bus.
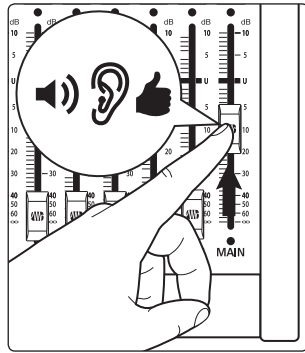
15. Bring up the Main fader until you can comfortably listen to your microphone through your speakers.
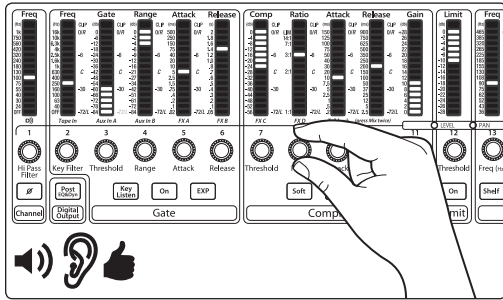
16. With Channel 1 selected, you can use the Fat Channel to add dynamics processing and EQ.

3 Hookup

3.1 Rear-Panel Connections

Microphone Inputs. Your StudioLive is equipped with 32/24/16 PreSonus XMAX microphone preamplifiers for use with all types of microphones. The XMAX preamplifier has a Class A input buffer, followed by a dual-servo gain stage. This arrangement results in ultra-low noise and wide gain control, allowing you to boost signals without increasing unwanted background noise.

48-volt Phantom Power. The StudioLive provides 48V phantom power for the microphone input on each channel. This feature can be individually enabled for each channel using buttons on the top panel.

⚠ **WARNING:** Phantom power is required for condenser microphones but can severely damage some dynamic mics, especially ribbon mics. Therefore, switch phantom power off for all channels where it is not required.

Power User Tip: Dynamic microphones and ribbon microphones are generally lower-output devices and require no external power source. The most important thing to note about ribbon microphones is that they very rarely require phantom power. In fact, unless a ribbon microphone calls specifically for phantom power, sending phantom power to it can cause severe damage. Condenser microphones are generally more sensitive than dynamic and ribbon microphones and typically require external +48V phantom power. Always review your microphone's documentation to ascertain the manufacturer's recommended operating practices.

XLR connector wiring for phantom power:
Pin 1 = GND   **Pin 2** = +48V   **Pin 3** = +48V

Line-level Inputs. Each channel of the StudioLive has a ¼-inch, balanced TRS connection for line-level input. When these inputs are engaged, the microphone-preamp circuit is bypassed. Typical examples of line-level connections are synthesizer outputs, CD/DVD-player outputs, and (with exceptions) signal-processor inputs and outputs.

⚠ **Note:** *As with any mixer, plugging in a microphone or a line-level input device, or turning phantom power on or off, will create a momentary spike in the audio output of your StudioLive. Because of this, it is highly recommended that you mute or turn down the channel trim before changing connections or turning phantom power on or off. This simple step will add years to life of your audio equipment.*

Insert. Each input channel on the StudioLive has a direct-insert point. These unbalanced, ¼-inch connectors can be used to connect external processors (such as compressors, EQs, de-essers, and filters) to your StudioLive's channel inputs. The insert's send is after the channel's gain control but before the digital bus. The return goes straight to the digital bus. So if you insert a de-esser on your vocalist's channel, you will be sending an unprocessed, amplified signal to the de-esser; the processed signal returned to the StudioLive will then be routed to the digital bus, where it can be sent through the Fat Channel, Aux and FX buses, etc.
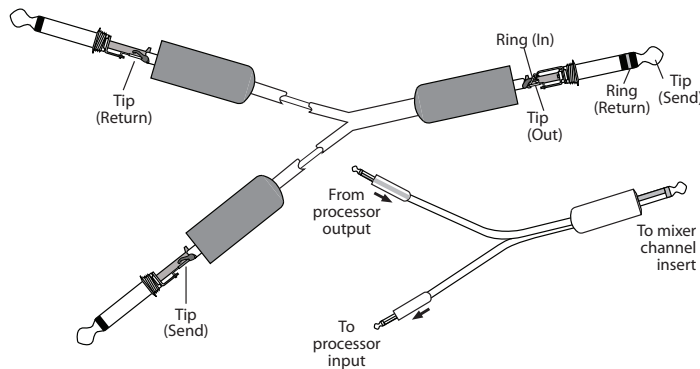
Insert-connector wiring:

Tip = send (output to inserted device)
Ring = return (input from inserted device)
Sleeve = common ground
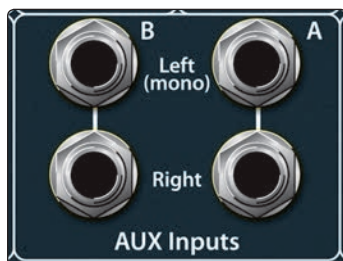
Aux Inputs. The StudioLive offers two auxiliary inputs. While these line inputs are generally used as effects returns, they can also be used for any line-level source (synthesizers, amp modelers, etc.). In *Section 4.4.7*, we discuss using an aux bus to send several channels to an external effects processor; the aux inputs can be used to return the processed signal to the mixer. Each input is balanced stereo. The left input is normalled to the right input, so if you are returning a mono signal to the mix, connect it to the left input, and the signal will be routed to both sides of the mix.

Subgroup Outputs. These are balanced mono outputs for each subgroup.
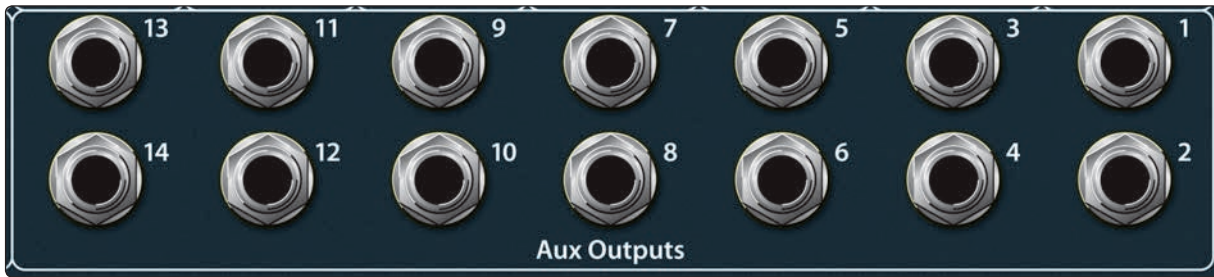
Aux Outputs. The StudioLive is equipped with 14/10/6 auxiliary outputs. Aux mixes are routed to these outputs. In *Section 4.4.5 and 4.4.6*, we discuss in detail how to create aux mixes for monitoring and effects processing.

Talkback Mic Input. The StudioLive does not have an onboard talkback mic; an external mic must be used. Phantom power is always enabled on the Talkback microphone preamp, so either a dynamic or a condenser microphone can be used. However, we recommend reviewing the documentation that came with your dynamic microphone to verify that phantom power will not harm it.

Power User Tip: The Talkback microphone preamp is the same high-quality XMAX preamp that is featured on StudioLive channel mic inputs. The Talkback input can also be selected as a recording input. See "Auxiliary Digital Sends" in the StudioLive AI-Series Software Library Manual for details.

Talkback Mic Level. This is the trim control for your talkback microphone. It adjusts the gain of the talkback input.

Mono Output. This balanced output carries a mono, summed version of the stereo signal from the Main bus.

Mono Output Level. This knob controls the maximum level of the Mono Output signal. The signal can be attenuated to -80 dB and boosted up to +6 dB.

Main Output. The StudioLive features both XLR and TRS main outputs. These outputs are parallel to each other and to the mono output.

Main Output Level. This knob controls the maximum output level of the XLR and TRS main outputs. The signal can be attenuated to -40 dB and boosted up to 0 dB.

Power User Tip: All StudioLive main outputs (XLR stereo, TRS stereo, and XLR mono) are active all of the time. Therefore, you can send your main mix to five speakers at the same time. This can be especially useful when you need to send a mix to another room or add another set of speakers to accommodate a larger venue.

Tape In/Out. The StudioLive is equipped with stereo RCA inputs and outputs that can be used to connect a tape deck, CD player, or other consumer device. The tape-input level is controlled by the 2 Track In knob on the top panel. The Main bus is routed post-fader to the tape output.

CR Outputs. These are the balanced control-room outputs. The level is controlled by the Monitor knob in the Monitor section on the top panel.

Pre-Insert Balanced Direct Outputs. These are the balanced, direct analog outputs for the 32/24/16 channels. The DB25 connectors divide the channels into groups of eight. Balanced DB25 fan-out snakes can be obtained in various configurations at most recording and live-sound retailers. Common fan-outs are DB25 to (8) XLRM and DB25 to (8) TRS.

These outputs are post-gain, pre-insert, and pre-A/D converter. Only the microphone preamps and line-level inputs are available through the direct outputs. The digital returns cannot be patched to the direct outputs.
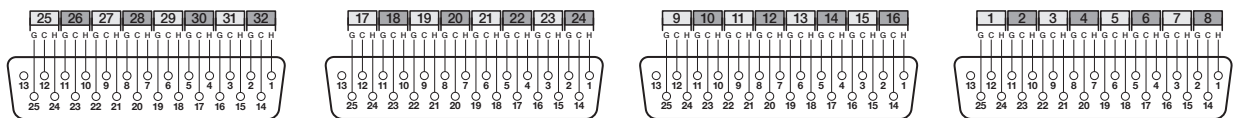
Pre-Insert Balanced Direct Outputs

DB25 pin-outs: **H** = Hot   **C** = Cold   **G** = Ground
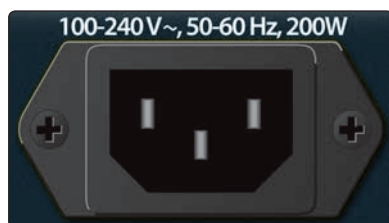
Power-cable Input. This is where you plug in the provided IEC power cable.

Power Switch. Push the top part of the switch (|) to turn on your StudioLive. Push the bottom part of the switch (O) to turn it off.

3.1.1   Installed Option Card

Each StudioLive AI-series mixer includes an installed option card that provides FireWire s800 connectivity, Ethernet for remote control, and S/PDIF output. For more information about user-installable StudioLive AI-series mixer option cards and their availability, please visit www.presonus.com.
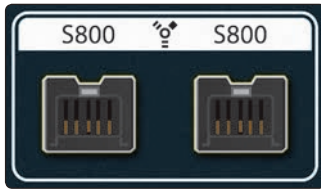
FireWire s800 Ports. There are two standard 9-pin FireWire s800 ports on the back of the StudioLive. Either port can be used to connect the StudioLive to a FireWire s800 port on your computer. Use the second FireWire s800 port to connect additional s800 devices (such as external hard drives) to your computer.

Please note: While connecting your StudioLive AI-series mixer to a FireWire 400 connection on your computer and daisy-chaining a FireWire 400 device to the secondary s800 port are supported, this will reduce your StudioLive bus speed to 400 Mbps, and you may experience reduced performance. To experience the full power of your StudioLive AI-series mixer, it is highly recommended that you connect to a true s800 connection or to a Thunderbolt connection using an adapter.
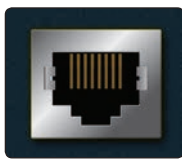
Ethernet. This standard RJ45 connection has been included to hardwire your StudioLive to a LAN network. By adding a wireless router to the network, you will be able to wirelessly remote control your StudioLive from virtually anywhere. This is especially useful when your wireless control devices (iPad, iPhone, or iPod touch) must be used in a different room from where your StudioLive is installed, or if there is quite a bit of wireless interference in your venue. For more information on connecting your StudioLive to a LAN network, *please review "Networking your StudioLive AI Mixer" in the StudioLive AI-Series Software Library Manual*.

S/PDIF Output. By default, the S/PDIF output receives the same signal as the main outputs, so no activation is necessary. However, any bus can be routed to the S/PDIF output, either through the System menu in the Digital Effects | Master Control section, or in VSL-AI. (*See Section 5.5 in this manual* or *"VSL-AI: Setup Tab" in the StudioLive AI-Series Software Library Manual* for more information.) Because the StudioLive cannot be synced externally, you will need to use it as the master clock and set your S/PDIF-equipped device to receive word clock externally via S/PDIF. Please consult the documentation for your external digital device for instructions.

3.2   Front Panel Connections

USB. The USB port is for use with the included USB Wi-Fi module. This module will allow you to connect your StudioLive AI-series mixer to a wireless network and remote control it using VSL-AI, SL Remote-AI, and QMix-AI. More information about wireless networking can be found in *"Networking your StudioLive AI Mixer" in the StudioLive AI-Series Software Library Manual.*

This 12V BNC connection is provided to connect a third-party console lamp. Do not use a bulb that is larger than 6W.

3.3   Typical Band Setup Diagrams

electric guitar amp   bass/DI   drum kit

wireless in-ear   wireless in-ear   drum monitor   sidefill
(keys)   (lead vocals)

floor wedges
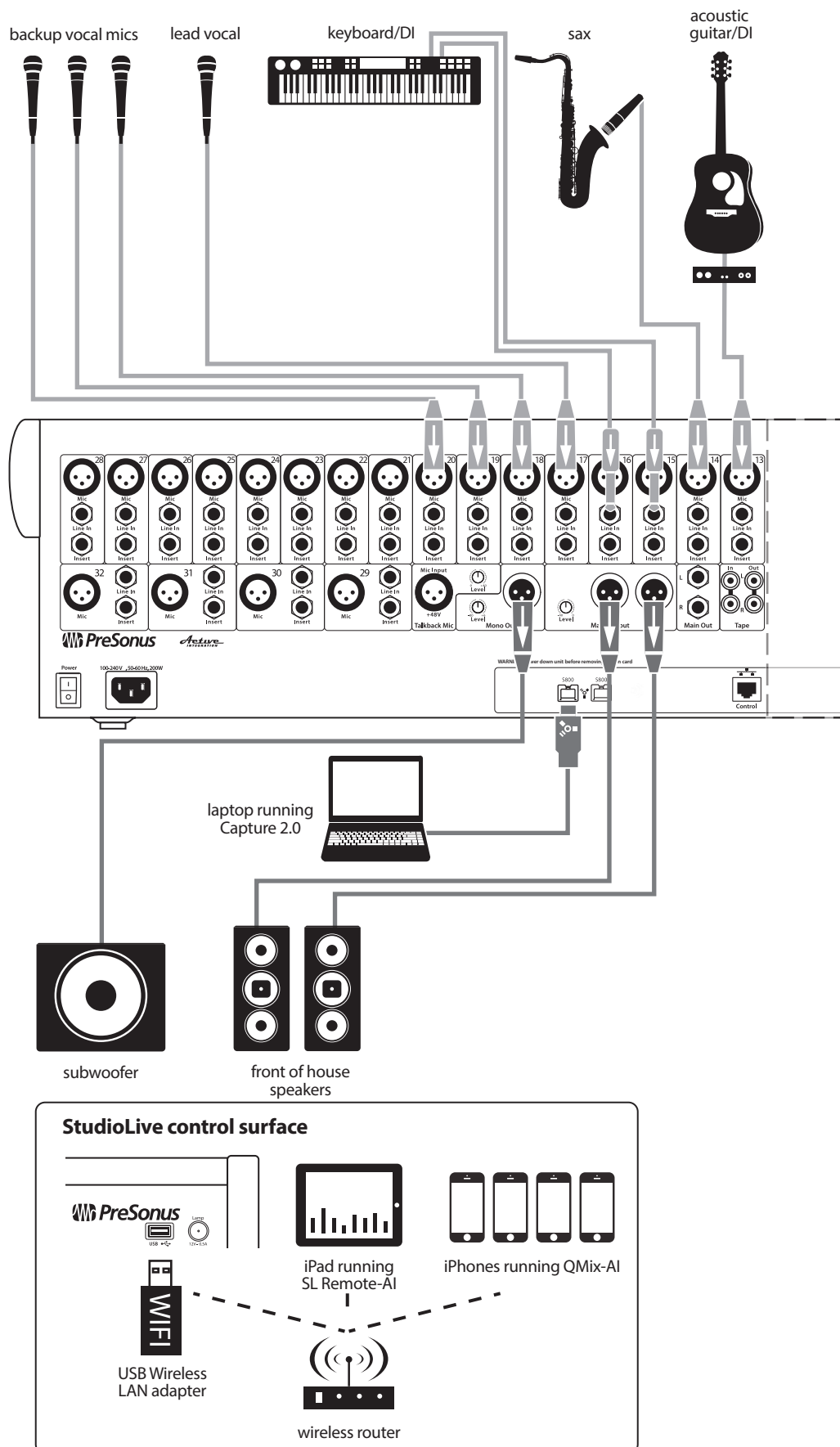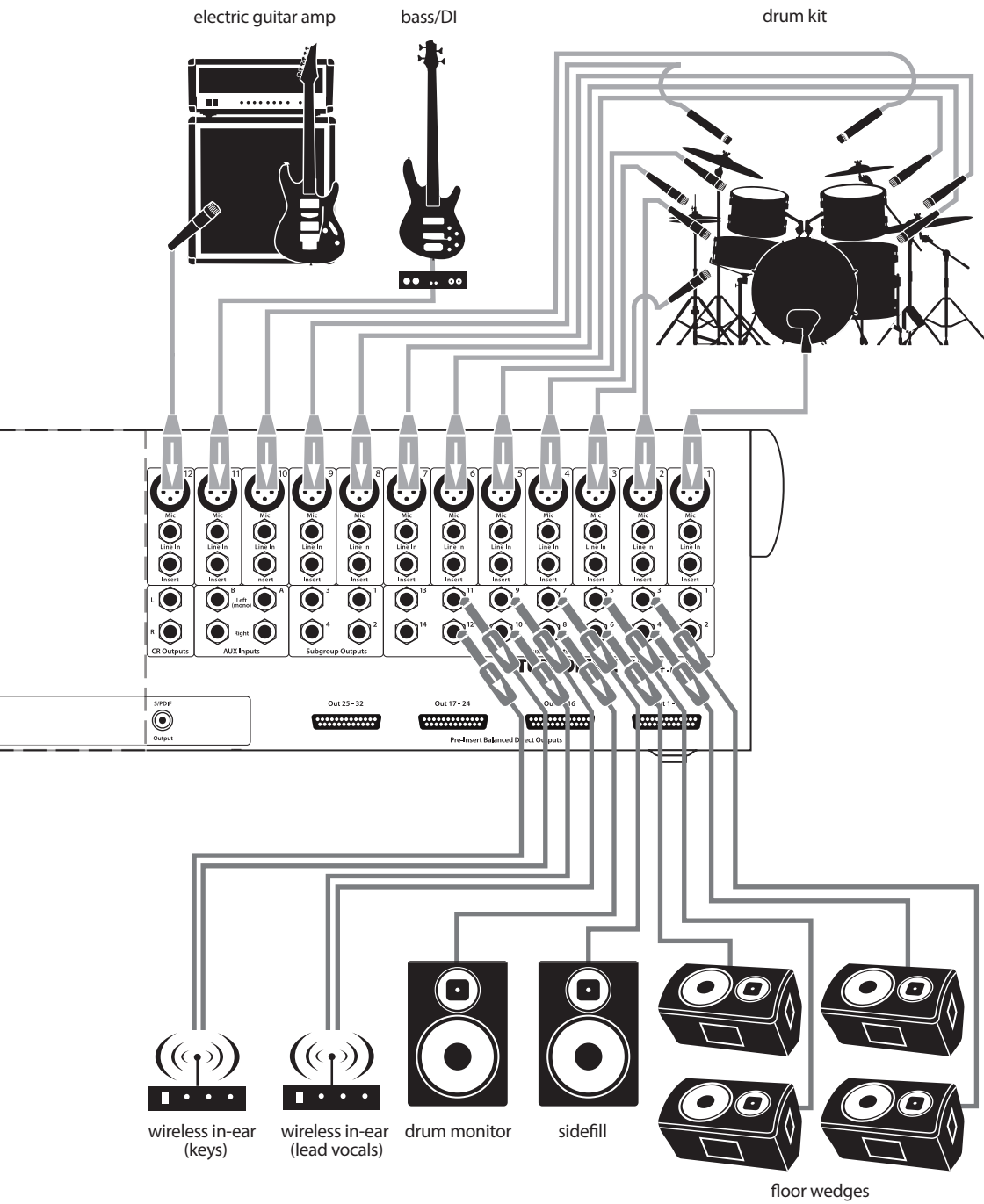
3.4   Typical Church Setup Diagrams

hanging choir mics

podium mic

backup vocal mics

lead vocal

keyboard/DI

acoustic guitar/DI

piano

subwoofers

dvd player

cry room

laptop running Capture 2.0

front of house speakers

electric guitar
amp (rhythm)

electric guitar
amp (lead)

bass/DI

drum kit

iPads running
SL Remote-AI

iPhones running QMix-AI

wireless
router

wireless in-ear
(keys)

wireless in-ear
(lead vocals)

wireless in-ear
(bass)

wireless in-ear
(lead guitar)

floor wedges

4   Controls

4.1   The Fat Channel

The revolutionary Fat Channel is the heart of the StudioLive. The Fat Channel makes dynamics, routing, and panning for every input and output on the StudioLive available at the touch of a Select button. The 32/24/16 multipurpose knobs and meters located in the Fat Channel control nearly every adjustment you will need to make on your StudioLive. From the Fat Channel, you can:

- Add dynamics processing and EQ to every input and output and A/B them

- Create sends and effects mixes for every analog aux bus and internal effects buses

- Assign subgroup and main routing

- Meter inputs, post-dynamics-processing outputs, and gain reduction for every input channel

- Meter aux-send outputs

- Copy, save, and load mix scenes

- Recall your fader position for stored mixes

NOTE: Fat Channel Processing is disabled on all output buses when HD Mode (88.2 and 96 kHz sample rates) is active.

4.1.1   Select Buttons, Meters, and the Fat Channel

Select Buttons. All around the StudioLive, you will see Select buttons. There is a Select button on each of the input channels, each of the analog aux buses, all of the internal FX (effects) buses, each of the subgroups, both auxiliary inputs, and the main output bus. Each of these buttons serves exactly the same purpose: to access the available Fat Channel parameters for its channel or bus.

It should be noted that while the noise gate, compressor, EQ, and limiter are available on every input channel and bus, the high-pass filter is only available on the input channels, aux buses, and internal FX buses, and Polarity Invert is only available on the input channels.

Selected Channel Display. In the lower right corner of the Fat Channel, you will find an LED readout. The currently selected channel will always be displayed here.

- Input Channels: Channel number

- Subgroup: S1-S4

- Aux Outputs: A1-A9, AA-AE

- FX Buses A-D: FA-FD

- Auxiliary Input A and B: RA and RB

- Main Bus: MA
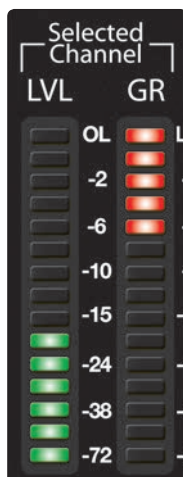
Selected Channel Meters. In addition, two meters—part of a set of eight meters located in the top right section of the mixer—are dedicated to displaying information about the currently selected channel. The meter on the far left of this section displays the pre-fader input level for the selected channel. The meter to the right of it displays the gain reduction applied to the selected channel. It is important to mention that these meters are only active when one of the input channels or an aux bus is selected.

Channel Info Page. When a channel or output bus is selected, the Channel Info page will open in the LCD. This is the default screen for your StudioLive. From this page you can customize the name for each channel or bus and view helpful information about each channel and bus on your StudioLive mixer. ***See Section 5.1 for more information.***

4.1.2   Fat Channel Processing Guide

The following table provides a quick guide to the processing that is available for each bus in the StudioLive. Note that all of these inputs and buses can be recorded. For more information on digital sends, ***please see "Digital Sends and Returns" in the StudioLive AI-Series Software Library Manual.***

Bus	Phase Reverse	High-Pass Filter	Noise Gate	Compressor	EQ	Limiter	Digital Send
Inputs Channels	✔	✔	✔	✔	✔	✔	✔
Subgroups			✔	✔	✔	✔	✔
Main Out L/R			✔	✔	✔	✔	✔
Aux Buses		✔	✔	✔	✔	✔	✔
FX Buses		✔	✔	✔	✔	✔	✔
Auxiliary Stereo Inputs							✔
Tape Input							✔
Talkback Mic							✔
Solo Bus							✔

4.1.3   Fat Channel: Dynamics Processing and EQ

The main function of the Fat Channel is to provide dynamics processing and filtering for every input and output on the StudioLive. The rotary encoders work in conjunction with the meters directly above them to adjust and display the dynamics processing and EQ parameters. The Fat Channel's processing section consists of five parts: high-pass filter, noise gate, compressor, limiter, and parametric EQ. Each can be turned on or off and controlled separately. The signal flows as follows:

Ø → Hi-Pass → Noise Gate → Compressor → EQ → Limiter
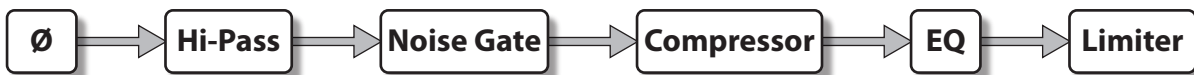
Polarity Invert. Push this button to invert the polarity of the selected channel's signal (that is, to alter the polarity by 180°). The button will illuminate, indicating that Polarity Invert is active. The Polarity Invert button can be used to correct audio signals that are out of phase and are canceling/reinforcing each other.

Power User Tip: *When recording with more than one open microphone, use the polarity invert to combat phase cancellation between microphones.*

Polarity Invert is only available on the 32/24/16 channels of the input bus.
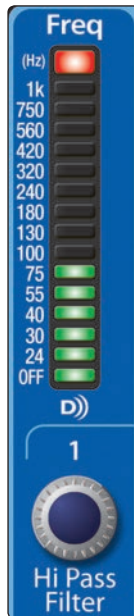
High Pass Filter. Sets the High-Pass Filter Frequency Threshold for the Selected Channel or Output Bus.

The High Pass Filter section consists of an encoder and a meter. Frequency range is indicated to the left of the meter. The filter's threshold can be set from 24 Hz to 1 kHz. When the meter is set to its lowest point, the filter is off. The high-pass filter's slope is -12 dB/8va.

Power User Tip: A high-pass filter attenuates all frequencies below the set threshold. Use the Fat Channel high-pass filter to remove unwanted low-frequencies from your source signal, rather than trying to EQ them out.

The high-pass filter is available on the 32/24/16 channels of the input bus and on every analog aux and internal FX bus.

Gate On/Off Button. Turns the Gate On and Off for the Selected Channel.

This button engages and disengages the gate for the selected channel. It will illuminate to indicate that the gate has been enabled.

The gate is available for all 32/24/16 input channels and every output bus.

Gate Expander Button (32.4.2AI and 24.4.2AI only). Turns the Noise Gate into an Expander.

The StudioLive allows you to choose between an expander and a noise gate for each channel or output. By default, the Expander button will be enabled. StudioLive 16.4.2AI users can choose between an Expander or Noise Gate from the Channel Info page for the selected Channel or Bus.

Power User Tip: In practice, expanders and noise gates are used almost identically. The main difference is that an expander is smoother and more gradual, so that it is easier to set the attack and release times correctly.

Gate Key Listen Button (32.4.2AI and 24.4.2AI only). Enables Key Listen in the Solo Bus.

This button engages and disengages the Key Listen function in the solo bus. It will illuminate to indicate that the Key Listen is active. When Key Listen is enabled, and the selected channel is soloed, you can use the Control Room outputs to monitor what the gate key filter is removing.

Press and hold this button to set a sidechain to the key filter. *See Section 4.1.4* for details. 16.4.2AI users can access the Key Listen and Sidechaining functions from VSL-AI and SL Remote-AI.
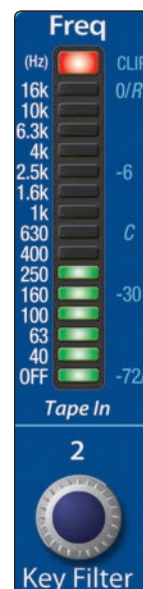
Gate Key Filter (32.4.2AI and 24.4.2AI only). Sets and Displays the Frequency at Which the Gate Will Open.

This encoder sets, and the meter displays, the frequency at which the gate will open. Setting a specific frequency, in addition to a specific decibel level, provides more sonic shaping.

The key filter can be triggered by the selected channel or bus's signal or by sidechaining a channel and using its signal as the source. *See Section 4.1.4* for more information on using a sidechain with the gate. 16.4.2AI users can access the variable Key Filter control from VSL-AI and SL Remote-AI.

Power User Tip: A properly set key filter on a gate can greatly improve the overall sound quality of a mix. For example, if you are inserting a gate on a snare-drum mic, you may get enough bleed from the kick drum to open the gate. This is where a key filter can come in handy. By setting the key filter to remove some of those low frequencies, the gate won't be as apt to open for the kick drum.
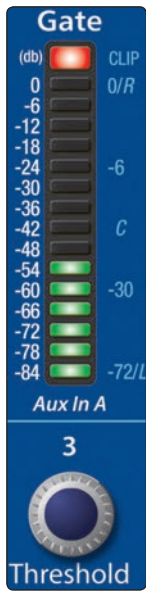
Gate Threshold. Sets and Displays the Threshold of the Gate for the Selected Channel.

This encoder sets, and the meter displays, the gate threshold for the selected channel. The gate threshold sets the level at which the gate opens. Essentially, all signals above the threshold setting are passed through unaffected, whereas signals below the threshold setting are reduced in level by the amount set by the range control. If the threshold is set fully counterclockwise, the gate is turned off (always open), allowing all signals to pass through unaffected. You can set the threshold from 0 to -84 dB.
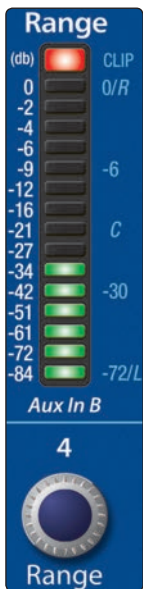
Gate Range (32.4.2AI and 24.4.2AI only). Sets and Displays the Range of the Gate.

This encoder sets, and the meter displays, the amount of gain reduction that the gate will produce. The range can be set from 0 to -84 dB.

Range control is not available when using the expander. 16.4.2AI users can access the variable Gate Range control from VSL-AI and SL Remote-AI.

Gate Attack (32.4.2AI and 24.4.2AI only). Sets and Displays the Gate Attack Setting for the Selected Channel.

This encoder sets, and the meter displays, the rate at which the gate opens on the selected channel or output. You can set the attack time from 0.02 to 500 ms. 16.4.2AI users can access the variable attack time from VSL-AI and SL Remote-AI.

Power User Tip: A fast attack rate is crucial for percussive instruments. Slow-rising signals such as vocals and bass guitar require a slower attack; with these signals, a faster attack can cause an audible click. All gates have the ability to click when opening but a properly set gate will never click.

Gate Release. Sets and Displays the Rate at Which the Gate Closes on the Selected Channel.

This encoder sets, and the meter displays, the rate at which the gate for the selected channel closes. The release time can be set from 0.05 to 2 seconds.

Power User Tip: Gate-release times should typically be set so that the natural decay of the instrument or vocal being gated is not affected. Shorter release times help to clean up the noise in a signal but may cause "chattering" with percussive instruments. Longer release times usually eliminate chattering and should be set by listening carefully for the most natural release of the signal.

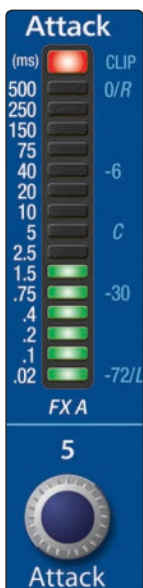**Compressor On/Off.** Turns the Compressor On and Off for the Selected Channel or Output Bus.

This button engages or disengages the compressor for the selected channel or output bus. It will illuminate to indicate that the compressor has been enabled.

The compressor is available for all 32/24/16 input channels and every output bus.

Soft Knee Toggle Button. Engages Soft-Knee Compression.

In normal operating mode, the compressor is set for hard-knee compression, meaning that the gain reduction applied to the signal occurs as soon as the signal exceeds the level set by the threshold. When the Soft Knee button is engaged, the ratio increases gradually as the signal reaches the threshold.

Auto Mode Button. Enables Automatic Attack and Release Mode.

When Auto mode is active, the Attack and Release controls become inoperative, and a preprogrammed attack and release curve is used. In this mode, the attack is set to 10 ms, and the release is set to 150 ms. All other compressor parameters can still be adjusted manually.
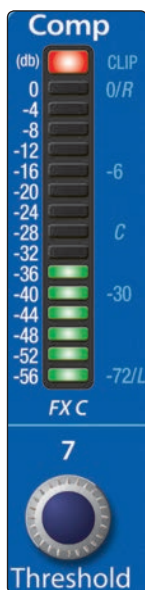
Compressor Threshold. Sets and Displays the Threshold of the Compressor for the Selected Channel or Output Bus.

This encoder sets, and the meter displays, the compressor threshold for the selected channel or output bus. When the signal's amplitude (level) exceeds the threshold setting, the compressor engages. Turning the knob counterclockwise lowers the threshold so that compression begins at a lower amplitude. The threshold can be set from -56 to 0 dB.

Ratio. Sets and Displays the Compression Ratio for the Selected Input Channel or Output Bus.
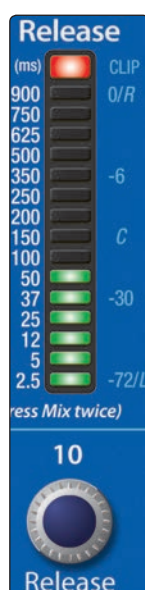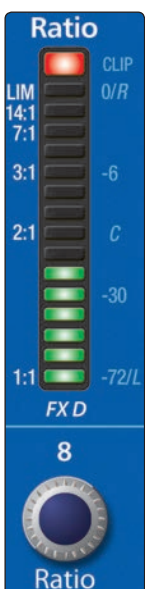
This encoder sets, and the meter displays, the compression ratio (or slope) for the selected channel or output bus. The ratio sets the compression slope, which is a function of the output level versus the input level. For example, if you have the ratio set to 2:1, any signal levels above the threshold setting will be compressed at a ratio of 2:1. This means that for every 2 dB of level increase above the threshold, the compressor's output will only increase 1 dB. The ratio can be set from 1:1 to 14:1.

Compressor Attack. Sets and Displays the Compressor Attack Setting for the Selected Input Channel or Output Bus.

This encoder sets, and the meter displays, the compressor's attack setting for the selected channel or output bus. Attack sets the speed at which the compressor acts on the input signal. A slow attack time (fully clockwise) allows the beginning component of a signal (commonly referred to as the initial transient) to pass through, uncompressed, whereas a fast attack time (fully counterclockwise) triggers compression immediately when a signal exceeds the threshold. You can set the attack from 0.2 to 150 milliseconds.

Compressor Release. Sets and Displays the Compressor Release Setting for the Selected Input Channel or Output Bus.

This encoder sets, and the meter displays, the release setting of the compressor for the selected channel or output bus. Release sets the length of time the compressor takes to return the gain reduction back to zero (no gain reduction) after crossing below the compression threshold. Release can be set from 2.5 to 900 milliseconds.

Power User Tip: Very short release times can produce a choppy or "jittery" sound, especially when compressing instruments that have a lot of low-frequency components, such as bass guitar. Very long release times can result in an overcompressed, or "squashed," sound. All ranges of release can be useful, however, and you should experiment to become familiar with different sonic possibilities.
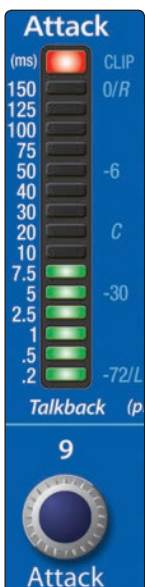
Compressor Makeup Gain. Sets and Displays the Amount of Makeup Gain for the Compressor on the Selected Input Channel or Output Bus.

This encoder sets, and the meter displays, the makeup-gain setting of the compressor for the selected channel or output bus. When compressing a signal, gain reduction usually results in an overall attenuation of level. The gain control allows you to restore this loss in level and readjust the volume to the precompression level (if desired). You can adjust Makeup Gain from 0 dB (no gain adjustment) to +28 dB.

Equalizer On/Off Button (32.4.2AI and 24.4.2AI only). Globally switches all EQ Bands On or Off for the Selected Input Channel or Output Bus.

This button engages or disengages the equalizer for the selected channel or output bus. It will illuminate to indicate that the equalizer has been enabled.

The equalizer is available for all 32/24 input channels and every output bus.

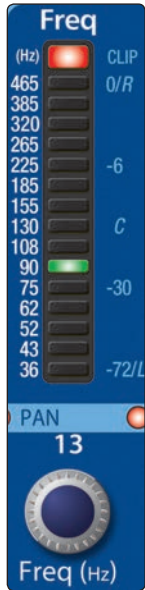**Low EQ On/Off Button.** Turns the Low Band EQ On/Off for the Selected Input or Output Bus.

This button activates the equalizer's Low band for the selected channel or bus. The button will illuminate to indicate control is active.

The Low EQ band is available for all 32/24/16 input channels and every output bus.

Low EQ Frequency Control. Sets and Displays the Center Frequency of the Low EQ Band.

This encoder sets, and the meter displays, the center frequency of the equalizer's Low band. The center frequency is the middle of the passband (the mean) between the lower and upper cutoff frequencies that define the limits of the band.

You can adjust the center frequency from 36 to 465 Hz.
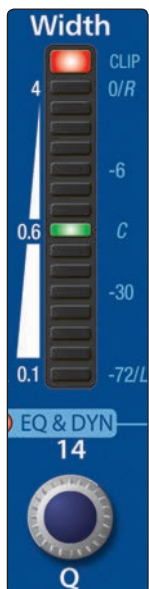
Low EQ Q Control (32.4.2AI and 24.4.2AI only). Sets and Displays the Q of the Low Frequency Band.

This encoder sets, and the meter displays, the Q for the Low band. The Q is the ratio of the center frequency to the bandwidth. When the center frequency is constant, the bandwidth is inversely proportional to the Q, so as you raise the Q, you narrow the bandwidth. 16.4.2AI users can access the variable Low EQ Q Control from VSL-AI and SL Remote-AI.

Low EQ Gain Control. Sets and Displays the Gain Attenuation or Boost of the Center Frequency.

This encoder sets, and the meter displays, the gain cut or boost at the center frequency for the Low band. The level of the center frequency can be set between -15 and +15 dB.

Low Shelf EQ Button. Turns on the Low Shelving EQ for the Selected Input or Output Bus.

When the Shelf button is not engaged, the Low band is parametric. Enabling the Shelf button turns the Low band into a low-shelving EQ that alters, by a fixed amount, a band of low frequencies at and below a user-selected shelving frequency.

Power User Tip: A low shelving EQ is like a bass-control knob on a stereo. In this mode, the Center Frequency control selects the shelving frequency.

Low-Mid EQ On/Off Button. Turns the Low-Mid EQ On/Off for the Selected Input or Output Bus.

This button activates the equalizer's Low-Mid band for the selected input or output. The button will illuminate to indicate control is active.

The Low-Mid EQ is available for all 32/24/16 input channels and every output bus.
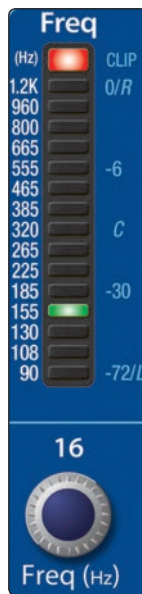
Low-Mid EQ Frequency Control. Sets and Displays the Center Frequency of the Low-Mid EQ.

This encoder sets, and the meter displays, the center frequency for the Low-Mid band. You can adjust the center frequency from 90 Hz to 1.2 kHz.

Low-Mid EQ Gain Control. Sets and Displays the Gain Attenuation or Boost of the Center Frequency for the Low-Mid Band.

This encoder sets, and the meter displays, the Gain cut or boost at the center frequency of the Low-Mid band. The level of the center frequency can be set between -15 and +15 dB.

Low-Mid EQ Q Control (32.4.2AI and 24.4.2AI only). Sets and Displays the Q of the Low Mid Frequency Band.

This encoder sets, and the meter displays, the Q for the Low-Mid band. The Q is the ratio of the center frequency to the bandwidth. When the center frequency is constant, the bandwidth is inversely proportional to the Q, so as you raise the Q, you narrow the bandwidth. 16.4.2AI users can access the variable Low-Mid EQ Q Control from VSL-AI and SL Remote-AI.

Note: While the StudioLive 16.4.2AI doesn't have a variable Q control for any of its bands from its control surface, it does offer a Hi Q button for the Low-Mid Frequency Band. This button will narrow the Q of the Low-Mid Band to 2.0 to provide more exact control. By default, the Low-Mid Q is set to 0.55. As previously mentioned, fully variable Q control for all bands is available from SL Remote-AI and VSL-AI.

High-Mid EQ On/Off Button. Turns the High-Mid EQ On/Off for the Selected Input or Output Bus.

This button activates the High-Mid band for the selected input or output. The button will illuminate to indicate that the control is active.

The High-Mid EQ is available for all 32/24/16 input channels and every output bus.

High-Mid EQ Frequency Control. Sets and Displays the Center Frequency of the High-Mid EQ.

This encoder sets, and the meter displays, the center frequency of the High Mid band. You can adjust the center frequency from 380 Hz to 5 kHz.

High-Mid EQ Q Control (32.4.2AI and 24.4.2AI only). Sets and Displays the Q of the High Mid Frequency Band.

This encoder sets, and the meter displays, the Q for the High-Mid band. The Q is the ratio of the center frequency to the bandwidth. When the center frequency is constant, the bandwidth is inversely proportional to the Q, so as you raise the Q, you narrow the bandwidth. 16.4.2AI users can access the variable High-Mid EQ Q Control from VSL-AI and SL Remote-AI.

Please Note: While the StudioLive 16.4.2AI doesn't have a variable Q control for any of its bands from its control surface, it does offer a Hi Q button for the High-Mid Frequency Band. This button will narrow the Q of the High-Mid Band to 2.0 to provide more exact control. By default, the High-Mid Q is set to 0.55. As previously mentioned, fully variable Q control for all bands is available from SL Remote-AI and VSL-AI.

High-Mid EQ Gain Control. Sets and Displays the Gain Attenuation or Boost at the Center Frequency.

This encoder sets, and the meter displays, the gain cut or boost at the center frequency of the High-Mid band. The level of the center frequency can be set between -15 and +15 dB.

High EQ On/Off Button. Turns the High EQ On/Off for the Selected Input or Output Bus.

This button activates the High band for the selected channel or bus. The button will illuminate to indicate control is active.

The High EQ band is available for all 32/24/16 input channels and every output bus.

High EQ Frequency Control. Sets and Displays the Center Frequency of the High EQ.

This encoder sets, and the meter displays, the center frequency of the High band. You can adjust the center frequency from 1.4 to 18 kHz.

High EQ Q Control (32.4.2AI and 24.4.2AI only). Sets and Displays the Q of the High Frequency Band.

This encoder sets, and the meter displays, the Q for the High band.

The Q is the ratio of the center frequency to the bandwidth. When the center frequency is constant, the bandwidth is inversely proportional to the Q, so as you raise the Q, you narrow the bandwidth. 16.4.2AI users can access the variable High EQ Q Control from VSL-AI and SL Remote-AI.

High EQ Gain Control. Sets and Displays the Gain Attenuation or Boost at the Center Frequency of the High Frequency Band.

This encoder sets, and the meter displays, the gain cut or boost at the center frequency of the High EQ band. The level of the center frequency can be set between -15 and +15 dB.

High Shelf EQ Button. Turns on the High Shelving EQ for the Selected Input or Output Bus.

When the Shelf button is not engaged, the High band is a parametric EQ. Enabling the Shelf button turns the High band into a high shelving EQ that alters, by a fixed amount, a band of high frequencies at and above a user-selected shelving frequency.

Power User Tip: *A high shelving EQ is like a treble-control knob on a stereo. In this mode, the Center Frequency control selects the shelving frequency.*

Limiter On/Off. Turns on the Limiter for the Selected Input Channel or Output Bus.

When the limiter is engaged the button will illuminate. The ratio is ∞:1.

The Limiter is available for all 32/24/16 input channels and every output bus.

Limiter Threshold (32.4.2AI and 24.4.2AI only). Sets and Displays the Threshold of the Limiter for the Selected Channel or Output Bus.

This encoder sets, and the meter displays, the threshold of the limiter for the selected channel or output bus. When the signal's amplitude (level) exceeds the threshold setting, the limiter is engaged. Turning the knob counterclockwise lowers the threshold, so limiting begins at a lower amplitude. The threshold can be set from -28 to 0 dB. 16.4.2AI users can access the variable Limiter Threshold control from VSL-AI and SL Remote-AI.

### 4.1.4 Sidechaining (32.4.2AI and 24.4.2AI only)

As previously mentioned, the key filter can be sidechained to another channel. This allows you to select a different channel as the trigger source for your StudioLive Gate's Key Filter. Sidechaining has many uses. You can use sidechained key filter to tighten up a rhythm section by sidechaining the kick drum channel to the bass channel and setting the gate to open at the frequency of the kick drum. This, combined with a fast attack and release, will make your rhythm section more cohesive. Increase the release time to loosen the feel.

Another great use for a sidechain is as an effect in electronica production. Try sidechaining a drum loop to a sustained source, like pads or strings. By doing this, every time a drum hit triggers the key filter, your sustained source will be heard. Between hits, this source will be silenced. Playing with the attack and release will transform this effect from a rhythmic pulse all the way to a chopped-up stutter.

This tutorial will guide you through the first use case. Please note, that while sidechaining the kick drum to the bass channel can tighten up a good rhythm section and make them sound even better, it will not correct timing issues and will actually exaggerate them if your bass player and drummer aren't in the pocket.

For this purposes of this Tutorial, we will be sidechaining the kick drum on Channel 1 to the bass on Channel 15.

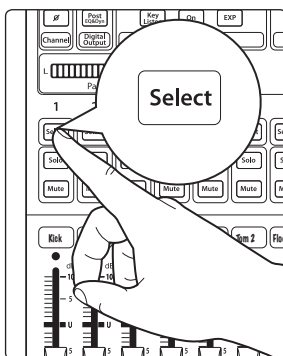1. Press the bass channel's Select button.
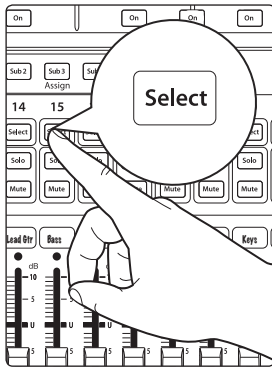
2. In the Fat Channel's Gate section, press and hold the Key Listen button. All the channel Select buttons on your StudioLive will flash except the bass channel.
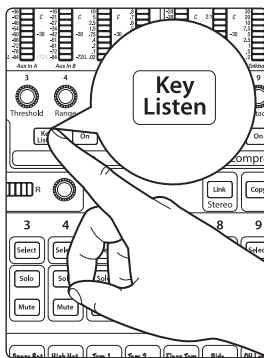
3. Press the kick drum channel's Select button. The bass channel's Select button will illuminate and the kick drum channel's Select button will continue to flash. This will be your indication that the channel has a sidechain. Every time you select the bass channel, the kick drum channel's Select button will flash.

    Adjust the key filter frequency to match the kick drum.

    Power User Tip: The Smaart Spectra™ tools in VSL-AI can make this process quick and easy. ***See "Enabling Smaart Analysis" in the StudioLive AI-Series Software Library Manual*** *for more information.*

4. To break the sidechain, simply repeat Step 2 and press the bass channel's Select button.

Power User Tip: While the 16.4.2AI does not offer Sidechaining from its control surface, this feature is available from both VSL-AI and SL Remote-AI

27

4.1.5   A/B Fat Channel Settings

StudioLive AI-series mixers let you create two complete Fat Channel settings for each channel and bus and compare the two using the Alt EQ & Dyn button. In this way, you can experiment with a new sound without having to struggle to re-create your old standby, and after several minutes of careful adjustment, you can verify that a new Fat Channel setting is better than it was before you started tweaking.
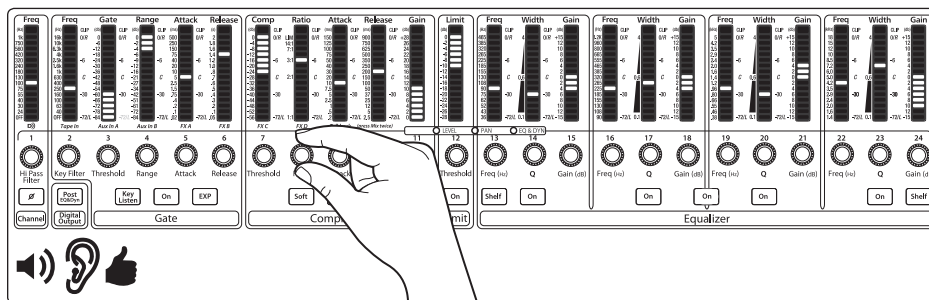
To A/B Fat Channel settings:

1. Select a channel and create a Fat Channel setup.
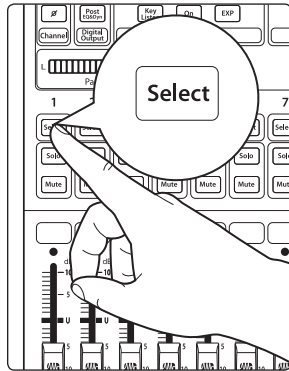
2. Press the Alt EQ & Dyn button. The button will light up to alert you that you are using the Alt Fat Channel layer.
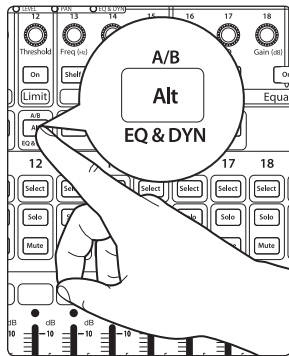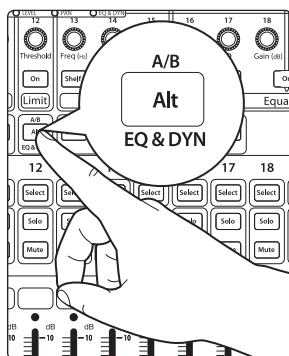
3. Dial in a new Fat Channel setting.

4. Press the Alt EQ & Dyn button again to listen to your original Fat Channel setting.

    Power User Tip: Alt EQ&Dyn can also be used to create two distinct Fat Channel settings for the same channel. So the next time the guitar player tells you they switch to a hollowbody mid-show, you can be ready.

4.1.6   Fat Channel Panning, Stereo Link, and Link Master
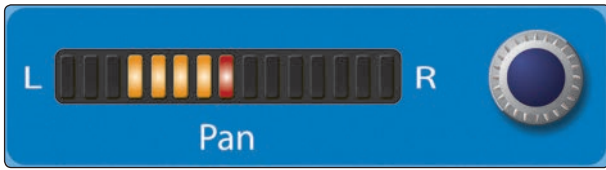
Panning for each input and output bus is set on the Fat Channel. The LED display shows the Pan setting, and the encoder to the right of the display controls panning for the selected input or output bus. When two channels are linked as a stereo pair, the LED display will automatically change to a stereo pan.


Stereo linking is done within the Fat Channel. Input channels, aux buses, and subgroups can be linked to create a stereo pair. The stereo pairs are predefined and cannot be changed. They are as follows:

Channels 1 and 2	Channels 27 and 28 (32.4.2AI only)
Channels 3 and 4	Channels 29 and 30 (32.4.2AI only)
Channels 5 and 6	Channels 31 and 32 (32.4.2AI only)
Channels 7 and 8	Aux 1 and Aux 2
Channels 9 and 10	Aux 3 and Aux 4
Channels 11 and 12	Aux 5 and Aux 6
Channels 13 and 14	Aux 7 and Aux 8 (32.4.2AI and 24.4.2AI only)
Channels 15 and 16	Aux 9 and Aux 10 (32.4.2AI and 24.4.2AI only)
Channels 17 and 18 (32.4.2AI and 24.4.2AI only)	Aux 11 and Aux 12 (32.4.2AI only)
Channels 19 and 20 (32.4.2AI and 24.4.2AI only)	Aux 13 and Aux 14 (32.4.2AI only)
Channels 21 and 22 (32.4.2AI and 24.4.2AI only)	Subgroups 1 and 2
Channels 23 and 24 (32.4.2AI and 24.4.2AI only)	Subgroups 3 and 4
Channels 25 and 26 (32.4.2AI only)	

A stereo link can be enabled when either channel in the pair is selected. When the Stereo Link button is illuminated, all dynamics settings (except Polarity Invert), subgroup assignments, and main assignments are destructively pasted to the other channel in the pair.

4.1.7   Fat Channel: Subgroup and Main Output Assignments
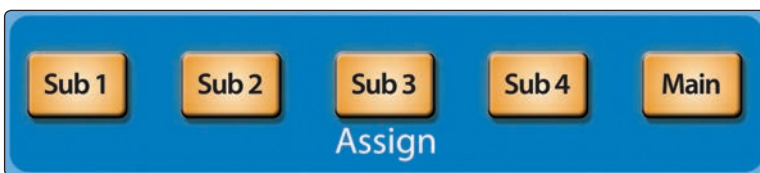

Output assignments are set within the Fat Channel. It should be noted that the StudioLive prevents you from creating a feedback loop.

Subgroups can only be assigned to the main outs, and the aux buses cannot be assigned to a subgroup or to the main outputs.

Any channel on the input bus can be assigned to any or all of the subgroup outputs, as well as to the main outputs. This includes the 32/24/16 channels in the input bus and the 2 auxiliary inputs. The internal effects bus returns can also be assigned to any or all of the subgroups and to the main outputs.
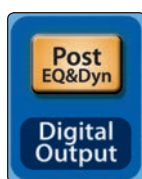
StudioLive AI mixers provide optional Fat Channel processing on the input channel digital sends. When the Post EQ&Dyn button is enabled, the signal sent to the digital bus is post-EQ and post-dynamics processing; the button will illuminate to indicate this signal flow. When the button is disabled, the signal sent to the digital bus is pre-Fat Channel. The subgroups, main output, and aux buses always send their signals post-Fat Channel dynamics and EQ. All digital sends are pre-fader except for the subgroups and the main outputs. For more information on using your StudioLive as an audio interface, *please consult "Connecting to a Computer" in the StudioLive AI-Series Software Library Manual.*
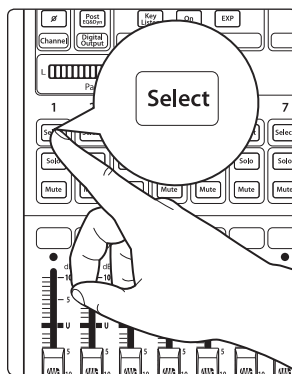
4.1.8   Copying Fat Channel Settings

In addition to being able to create and save custom Fat Channel presets, every setting in the Fat Channel can be copied from one channel or bus to any other channel or bus.
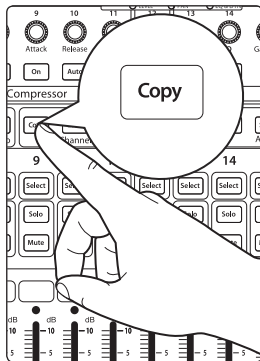

1. Press the Copy button to copy the settings on the selected channel or bus. Every Select button on the StudioLive except the button for the currently selected channel will begin to flash. The Select button for the selected channel will not illuminate. You can copy a Fat Channel setting from any channel or bus to any combination of channels and buses.

    The Load button will also start to flash.

2. To paste the current channel's Fat Channel setting to another channel or bus, simply press that channel's Select button. It will stop flashing and will illuminate.
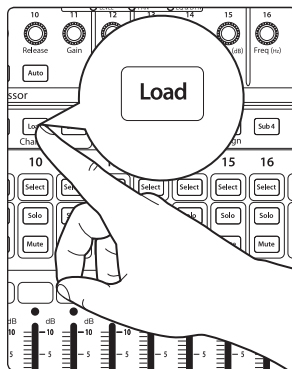

3. After you have selected every channel to which you want the settings pasted, press the Load button. The StudioLive will return to its normal state, indicating that the Fat Channel settings have been successfully pasted.

4.1.9   Loading Fat Channel Presets

The StudioLive comes with a suite of channel-strip presets created by professional users of PreSonus products. These presets provide a great jumping-off point to create a mix quickly and easily. The StudioLive also allows you to create your own library of presets.

1. To load a preset to any channel on the StudioLive, first press the Select button for the desired channel.
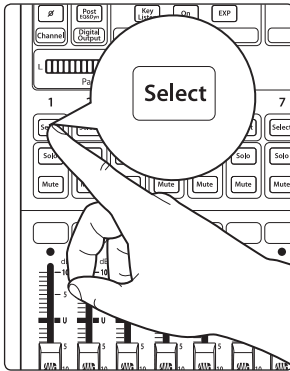
2. From the Fat Channel, press the Load button. You will notice that the LCD now displays the Channel Preset Load menu.
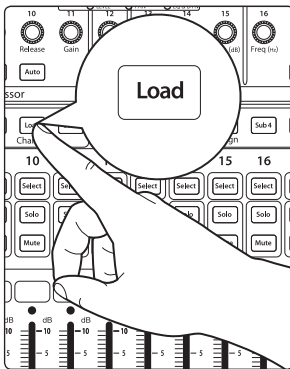
3. The Channel Preset Load menu always displays the selected channel onto which the preset will be loaded. Use the Value encoder to scroll through the preset library.

   You can also press the Next button to scroll through preset categories. You can choose from: DRM – Drum, BAS – Bass, GTR – Guitar, KEY – Keyboards, HRN – Horns, PRC – Percussion, VOX – Vocal, SYS – System, MIS – Miscellaneous.

   Once you have selected the category, press the Next button and use the Value encoder to scroll through the presets within that category.

4. Once you have made your selection, press the Recall button. If at any time you would like to cancel this operation, simply press the Load button again.

   Power User Tip: *Load will stay active until you press the button again to disable it, even if you select another channel. Because of this, you can quickly add a preset to every channel and give yourself a jumping-off point to dial in your mix.*
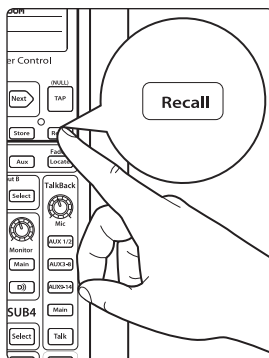
4.1.10 Saving Fat Channel Presets
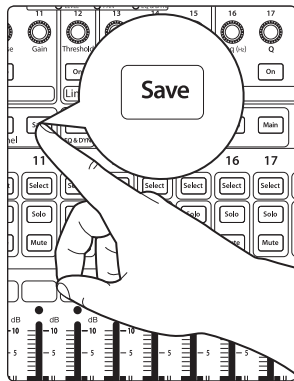
If you have created a channel-strip setting in the Fat Channel that you would like to save to the Channel Preset library, press the Fat Channel's Save button.

1. You will notice that the LCD will display the Channel Preset Save menu. Use the Value encoder to scroll to an empty position in the Channel Preset library.
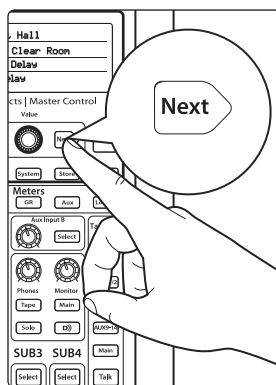
2. Press the Next button to navigate to the category location.

3. Use the Value encoder to scroll to the category in which your preset would fit (DRM, VOX, GTR, etc.).
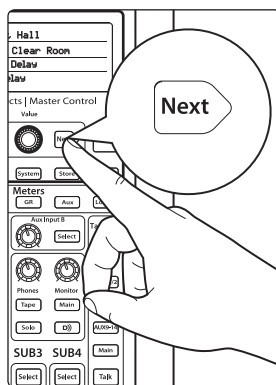
4. Press the Next button again to navigate to the first letter of the preset name.
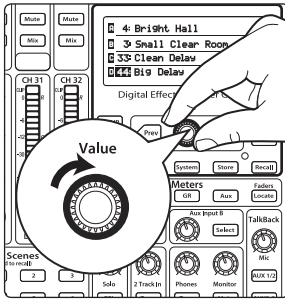
5. Turn the Value encoder clockwise or counter-clockwise to change the letter. The StudioLive allows you to customize the name with uppercase and lowercase letters, as well as a selection of numerals and punctuation marks. You can insert a space by simply pressing the Tap button.
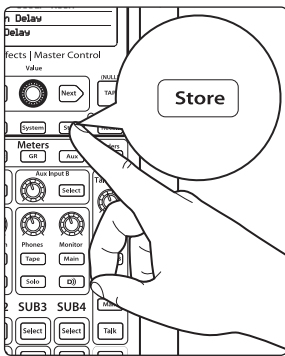
6. Once you are satisfied with your changes, press the Store button. It will illuminate while the Channel preset is being written to the StudioLive's internal memory. Once the Channel preset is saved, the Store button will return to its unlit state.

4.1.11 Channel Presets Library

Your StudioLive comes with 50 Fat Channel presets custom designed by professional PreSonus users. These presets can be altered, renamed, and overwritten; however, there are 49 additional empty storage locations for you to build your own custom library of channel-strip settings.

|				
01	DRM: Kick 1	**27**	GTR: Acoustic Strumming
02	DRM: Kick 2	**28**	GTR: Acoustic Fingerstyle
03	DRM: Kick Funk 1	**29**	KEY: Piano Bright
04	DRM: Kick Funk 2	**30**	KEY: Piano Warm
05	DRM: Kick Hip-Hop	**31**	KEY: Piano Jazz
06	DRM: Kick Jazz	**32**	KEY: Piano Electric
07	DRM: Snare 1	**33**	KEY: Piano Electric 2
08	DRM: Fat Snare	**34**	KEY: Vibes
09	DRM: Snare Crackalak	**35**	HRN: Trumpet
10	DRM: Snare Snappy	**36**	HRN: Trombone
11	DRM: Toms Mid	**37**	HRN: Sax
12	DRM: Toms Low	**38**	HRN: Sax Solo
13	DRM: Toms High	**39**	PRC: Congas
14	DRM: Overhead Rock	**40**	PRC: Bongos
15	DRM: Overhead Jazz	**41**	PRC: Cowbell
16	DRM: High Hat	**42**	PRC: Tambourine
17	BAS: Electric 1	**43**	VOX: Male 1
18	BAS: Electric 2	**44**	VOX: Male 2
19	BAS: Slap	**45**	VOX: Male 3
20	BAS: Upright	**46**	VOX: Female 1
21	GTR: Rock 1	**47**	VOX: Female 2
22	GTR: Rock 2	**48**	VOX: Female 3
23	GTR: Funk	**49**	VOX: Speech 1
24	GTR: Metal	**50**	VOX: Speech 2
25	GTR: Jazz	**51 - 99**	EMPTY LOCATION
26	GTR: Acoustic		

4.2   Metering
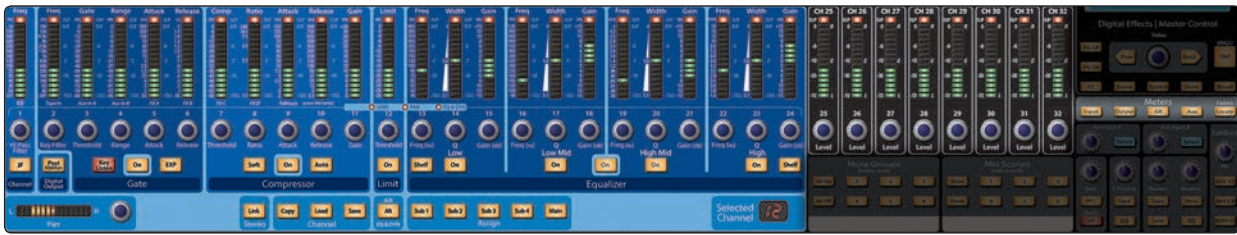
The StudioLive offers flexible metering at the touch of a button.
The meters in the Fat Channel section can monitor:

• The post-gain, pre-dynamics/EQ, and pre-fader signal of each input

• The post-gain, post-dynamics/EQ, and post-fader signal of each input

• The amount of gain reduction being applied to each input

• The output volume of each of the aux buses

• The output volume of all of the internal effects buses

• Finally, the meters can be used to recall the fader settings for a saved scene.

When functioning as a meter bridge, the Fat Channel
meters are scaled from -72 to 0 dBFS.

4.2.1   StudioLive Fat Channel Metering Controls

The Meters section of the StudioLive is located below the Master Control section. Each of these buttons are toggle switches; turn them on and off by pressing them. The meter state can also be changed by pressing another button in the Meters section, or any Select button on the StudioLive, or a Mix or Mix/Pan button in the Aux section.

Your StudioLive mixer defaults to peak metering. Peak hold metering is also available and can be enabled in the System menu. *See Section 5.5* for details.

Power User Tip: *It is important to mention that the meters simply overlay the selected Fat Channel state. For instance, if you have Channel 16 selected and then press the Output button in the Meters section, the knobs and buttons in the Fat Channel section will still be active. The advantage of this is that you can make adjustments in the Fat Channel while monitoring your entire mix.*

Input Metering Button. Turns PFL Input Metering On and Off.

Switches the meters to display the pre-dynamics, pre-fader level of the input bus. Meters are one to one (Meter 1 shows the level of Channel 1, etc.).

Output Metering Button. Turns Post-Fader Output Metering On/Off.

Switches the meters to display the post-dynamics, post-fader level of the Input bus. Meters are one to one (Meter 1 shows the level of Channel 1, etc.).

**Gain Reduction Metering Button.** Turns Gain Reduction Metering On and Off.

Displays the gain reduction of the input bus. Meters have a one-to-one relationship with channels (that is, Meter 1 shows the gain reduction of Channel 1 and so on).

Aux Metering Button. Turns Aux Bus Master Out Metering On and Off.

Displays the output level of each of the aux and effects buses.

Each odd-numbered meter displays the output level for the aux bus directly above it (Meter 1 shows the output level for Aux 1, Meter 3 displays the output level of Aux 2, etc.).
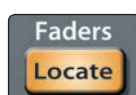
**Fader Locate Button.** Turns Fader-Recall Metering On/Off.

Displays the fader position of the stored scene. *Please review Section 5.3.5*.

4.2.2   Subgroup and Main Meters
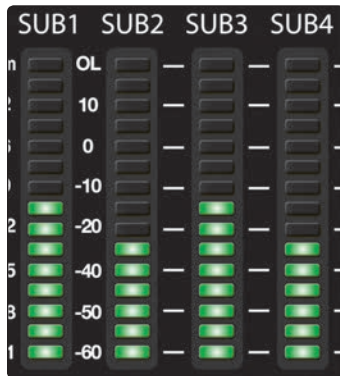
Subgroup Meters. Display the Level of the Subgroups.

In the upper-right corner of the StudioLive are the subgroup meters, which display the levels of the subgroup outputs. These meters are scaled from -60 to +18 dBu.
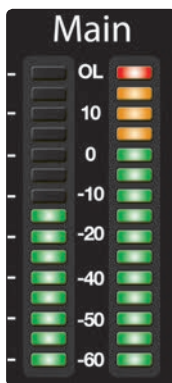
Main Bus Meters. Display the Level of the Main Output.

In the upper right corner of the StudioLive are the Main meters, which display the output levels of the main stereo bus. These meters are scaled from -60 to +24 dBu.

Power User Tip: Because of the different full-scale reference between the subgroups and the Main Output meters, you may notice a slight metering variation when a single channel is patched to both a subgroup and the Main bus.

4.2.3   A Quick Note About dBu and dBFS

StudioLive-series mixers meter the channels and the main and subgroup buses using two different reference scales. The channel levels are measured in dBFS and the output levels are measured in dBu. To understand how this impacts the metering on your StudioLive, you must first be aware of the difference between dBu and dBFS and how both are measured.

dB is an abbreviation for "decibel." One decibel is equal to one-tenth of a Bel. When a measurement is stated in dB, it is describing the ratio between two levels: the level being measured and the level being referenced. Without a reference point, the absolute dB value cannot be determined.

dBFS, or "decibels relative to full scale," is used to measure amplitude levels in digital systems that have a maximum peak level (the point at which the A/D converter will clip). When measuring in dBFS, 0 dBFS cannot be exceeded. 0 dBFS can equal +10 to +24 dBu, depending on the device's maximum peak level. For StudioLive-series mixers, 0 dBFS equals +18 dBu in the subgroups and +24 dBu in the main output meters.

dBu measures decibels relative to 0.775 volts with an open or unloaded circuit. (The "u" in "dBu" stands for unloaded.) While 0.775 volts may seem rather arbitrary, it is the voltage level that delivers 1 mW in a 600Ω resistor, which is the standard reference impedance in a telephone audio circuit. (The Bel was named after Alexander Graham Bell). Because the StudioLive subgroup and main outputs are electrical, rather than digital, it is more advantageous to meter them relative to voltage.

The Subgroup and Main meters display signals from -60 to +18 dBu (-78 to 0 dBFS) and -60 to +24 dBu (-72 to 0 dBFS). The StudioLive channel meters display signals from -72 to 0 dBFS. This means that the Subgroup meters will display 6 dBu of signal before your channel meters will register a signal.

If you have multiple channels set to unity and routed to the Subgroup and Main buses, you will see even more signal in your output meters. Every time you double the channel count, you add 3 dB more signal to the bus. So if you have 16 inputs set to unity and routed to a subgroup and to the mains, you will see 12 dB more signal level than you would if only one channel were routed. This means your

subgroup fader meter could be registering -58 dBu, and your Main meters will display -52 dBu of signal, but your channel meters still won't display anything.

By now you're probably wondering just how loud -78 dBu is. Imagine walking into the vocal booth at your favorite professional recording studio and closing the door. The ambient noise of a very quiet room, like a professional vocal booth, is around -80 dBu. Humans perceive a noise level doubling about every 10 dBu. So -78 dBu is pretty darn quiet.

4.3   Input Channel Strip

The StudioLive is equipped with all of the standard input controls of an analog mixer. In addition, the StudioLive provides the added flexibility of routing a playback stream from your audio-recording software to the mixer via the digital bus, just as if it were an analog input. This lets you incorporate digital audio tracks into the main mix, as well as insert plug-in effects and software instruments from your audio program.

4.3.1   Input Channel Controls

Trim Control. Adjusts the Input Gain Level.

The Trim control adjusts the gain of the channel's analog input. It is very important to properly adjust this control in order to minimize noise and avoid overload distortion. **Follow the Quick Start level-setting instructions in Section 2.1** before operating a channel.

Phantom Power Button. Turns Phantom Power On/Off.

The StudioLive is equipped with individual phantom power for every microphone input. The 48V button will illuminate when phantom power is activated.

The 48 volts supplied by way of the XLR input provides power for condenser microphones and other devices requiring continuous phantom power. This power is supplied at a constant level to prevent any signal degradation.

Digital Return Button. Turns Digital Playback Streaming On/Off.

The Digital Return button routes a playback stream from your audio software to the StudioLive's channel inputs, where it is routed and processed the same way as analog input signals. For example, if you want a particular recorded track to play on mixer Channel 3, simply route that track in your audio software to StudioLive Output 3. This button can also be used to insert a plug-in effect into the mix. For more information on this feature, *please review "Using Plug-in Effects as Inserts" in the StudioLive AI-Series Software Library Manual*.

Engaging a digital return will mute the analog input on that channel globally to the mix bus; however, the signal will still be routed to the digital send so that it can be recorded or processed by a plug-in. For more information *see "Digital Returns" in the StudioLive AI-Series Software Library Manual*.

Input-Channel Select Button. Enables Fat Channel Processing and Routing.

As previously described in *Section 4.1.1*, the Select button routes its channel through the Fat Channel, allowing you to add dynamics processing, EQ, and panning; assign the output routing; and more. This will also open the Channel Info page for the selected channel in the LCD.

Solo Button. Turns Soloing On/Off.

This button will solo its channel to the main outputs or to the monitor outputs, depending on whether PFL (Pre-Fader Listening) or SIP (Solo In Place) is selected in the Solo bus section. *Please review Section 4.9* for details.

When a Solo button is enabled, that channel or bus will automatically be selected, and its Select button will illuminate.

Mute Button. Turns Muting On/Off.

This button mutes its channel. It will illuminate red when the channel is muted. Where a channel will be muted is determined by the Global Mute setting in the System menu. By default, Global Mute is set to "Yes". While in this mode, engaging a channel mute button will mute the channel in all of its assigned outputs (subgroups, mains, aux and FX buses). Disabling Global Mute will allow the channel to continue to be heard in the Aux bus mixes while muting in all other bus mixes. See Section 5.5 for more information.

Channel Fader. Controls the Overall Level of the Channel.

Each input channel features a 100 mm long-throw fader for accurate level adjustment. Unity gain (0 dB) is denoted by a "U."

The white area above the fader can be used as a scribble strip. Use only oil pencils; other types of pens or pencils cannot be wiped off.

To clean the scribble strip, use a lightly damp cloth to remove the writing.

4.4   Aux and FX Buses

The aux bus provides outputs to create auxiliary mixes that are separate from the main and subgroup mixes. The StudioLive is equipped with 18/14/10 aux buses: Auxes 1 through 14/10/6, which have physical output jacks, and internal effects buses FX A, B, C, and D.

Aux buses can be used for many applications, the two most common of which are creating monitor mixes and inserting external effects processors into the mix. As with the other buses, the StudioLive allows you to add global dynamics processing and EQ to these aux buses.

The analog aux buses are mono; however, two aux buses can be linked to create a stereo bus.

4.4.1   Analog Aux Bus Controls
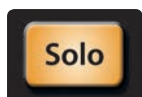
Solo Button. Turns Soloing On/Off.

This button will solo the aux bus to the monitor outputs.

When a Solo button is enabled, that channel or bus will automatically be selected, and its Select button will illuminate.

Aux Mute Button. Turns Muting On/Off.

This button will mute or unmute its aux bus. It will illuminate red when the aux is muted.

Output Level Control. Adjusts the Master Level of the Aux Output.

This knob controls the overall output level of the aux mix.

Aux Bus Select Button. Enables Fat Channel Viewing.

As previously described in *Section 4.1.1*, the Select button routes its aux bus through the Fat Channel, allowing you to add dynamics processing and EQ. This will also open the Channel Info page for the selected aux bus in the LCD.

4.4.2   Internal FX Bus Controls

FX Level Control. Adjusts the Master Level of the Effects-Send Mix.

This knob controls the overall output level of the internal effects mix.

Internal Effects Bus Select Button. Enables Fat Channel Viewing.

As described in *Section 4.1.1*, the Select button routes its effects bus through the Fat Channel, allowing you to add dynamics processing and EQ. From the Fat Channel, you can also route each effects bus to the subgroups or mains. By default, FXA, FXB, FXC, and FXD are routed to the Main bus. This will also open the Parameter Detail page for the selected FX bus in the LCD. *See Section 5.1* for more information.

NOTE: FXB and FXD are disabled when HD Mode (88.2 and 96 kHz) is active.

Mute Button. Mutes/Unmutes the Internal Effects Bus.

This button will mute or unmute its internal effects (FX) bus. It will illuminate red when the bus is muted.

4.4.3   Aux and FX Bus Channel Sends

In addition to setting the dynamics for each channel and bus and metering each channel and output, the Fat Channel also allows you to create aux mixes and quickly view the send level for each channel.

The Mix and Mix|Pan buttons for each aux and FX bus are used for just this purpose. Each of these buttons allows you to view and set the send level for each channel to that aux or FX mix.

Mix Button. Enables Aux Bus Mixing in the Fat Channel.

When the Mix button is pressed once, the 32/24/16 encoder knobs in the Fat Channel become the aux-send level controls for each of their respective input channels. The meters will display the send amount for each of the input channels.

D))	Tape In	Aux In A	Aux In B	FX A	FX B	FX C	FX D	Talkback	(press Mix twice)
1	2	3	4	5	6	7	8	9	10

Aux Flip Mode. Enables Aux Send Level Control for Main Digital return, Tape In, Aux Ins A and B, FXA, FXB, FXC, FXD, and Talkback.

Pressing the Mix button a second time will switch the first nine meters and encoders to become the aux-send level controls for the Main Digital Return, Analog Tape In, Aux Input A, Aux Input B, the output of FXA, FXB, FXC, FXD, and the Talkback mic.

While in this mode, the Mix button will remain illuminated, and an LED at the top and bottom of the remaining meters will also illuminate.

Press the Mix button a third time to disengage Aux Mix mode.

Mix|Pan Button. Enables Pan Control and Metering in the Fat Channel (Stereo Send Mode Only).

When two auxes are stereo-linked, and the Mix button is enabled on the even-numbered aux bus (2, 4, 6, etc.), the Fat Channel encoders become the pan controls for their respective input channels. The meters will display the pan setting of each of the input channels. When the Mix button is enabled on the odd-numbered aux bus (1, 3, 5, etc.), the Fat Channel encoders are send controls for their respective channels.

Press the Mix|Pan button a second time to disengage Aux Mix Pan mode.

For more information on stereo linking, *please review Section 4.1.6.*

Pan control is not available for the additional buses accessed by Aux Flip Mode.

4.4.4   Pre/Post Channel Sends
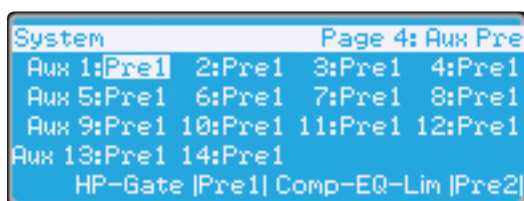
On pages 4 and 5 of the System menu, you can change the send position for every channel to each Aux and FX bus. To access these controls, press the System button and press the Page Down button to navigate to the Aux Pre Position pages.

By default, all aux buses are set to Pre 1. This places the send of every input channel to each aux bus before the fader, limiter, EQ, and compressor, but after the Polarity Invert switch, high-pass filter, and gate.

The four internal effects buses are set to Pre 2 by default, which routes each of the input channels after all Fat Channel dynamics and EQ but pre-fader.

From this menu, you can choose between three send positions for each Aux and FX mix:

Pre 1: Sends each channel to the Aux bus after the polarity invert, high-pass filter, and gate.

Pre 2: Sends each channel to the aux bus after all Fat Channel processing (polarity invert, high-pass filter, gate, compressor, EQ, and limiter) but before the fader.

Post: Sends each channel to the Aux bus after all Fat Channel processing (polarity invert, high-pass filter, gate, compressor, EQ, and limiter), and after the fader.

Power User Tip: *Use the Pre 2 position for headphone and in-ear mixes to give your performers a polished "studio" sound. This setting should be avoided for floor wedges, as compression can cause feedback problems.*

4.4.5   Creating Monitor Mixes

Creating custom monitor mixes is critical. If musicians can't hear themselves or their bandmates, their performance will suffer. A monitor mix can be mono or stereo. Most often, an individual live monitor mix is mono and is sent to a floor-wedge or sidefill monitor. (The obvious exception is in-ear monitor systems.) A studio monitor mix is usually stereo because it is sent to a headphone amplifier that requires both a left- and a right-channel input. In both cases, the function of the aux bus is the same.

As an example, let's create a mono monitor mix on Aux 1.
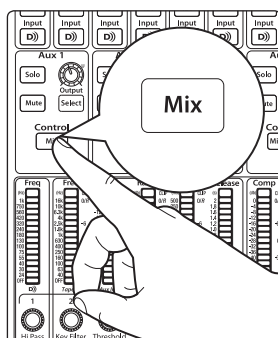
1. To begin, press the Mix button in the Aux 1 section. The Fat Channel meters will display each input channel's send level to Aux 1. The encoders below each meter control the channel's level in the Aux 1 mix. Use these encoders the same way that you use faders to set the output level to your main mix. Ask your musicians what they would like in their monitor mix and use their requests as a starting point.
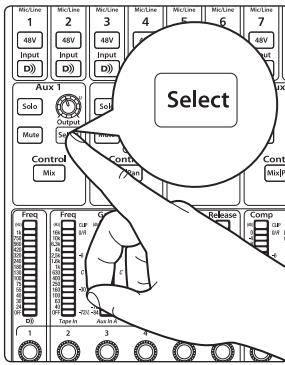
2. By pressing the Select button for Aux 1, you can add dynamics processing and EQ to the overall monitor mix. These are especially useful for eliminating feedback in a monitor. Keep in mind that an equalizer can also be used to increase the presence of an instrument by boosting that particular frequency range without necessarily boosting the volume in the mix. This is great for getting the lead guitar to cut through in the guitarist's monitor mix and to provide that extra rumble in the bassist's mix.
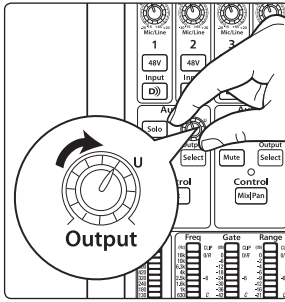
3. Use the Aux 1 Output knob to control the level of the entire aux mix.
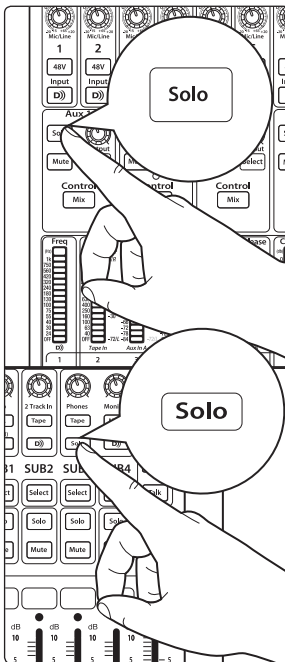
4. You can listen to the aux mixes you are creating, using your headphones or your control-room monitor, by simply soloing the aux and selecting Solo as the source in the Monitor section.

4.4.6   Creating Internal FX Mixes

There are at least two main advantages to creating an effects mix, rather than inserting an effect in a channel. First, several channels can be sent to a single processor. In addition to greatly simplifying the number of parameters you have to control, this can create a cohesive sound in your mix. The second advantage of creating an effects mix is that you can vary the level sent from each channel to the processor, rather than patching the output directly into the effect. This allows you to add a lot or a little of an effect to any given channel.

The StudioLive features four internal effects buses. These are used much in the same way the aux buses are used to create monitor mixes.
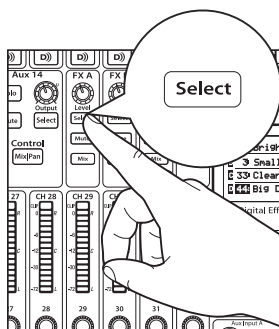
1. To begin, press the FXA Select button and decide to which outputs you'd like to route your effects mix.
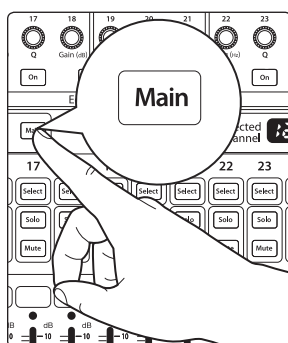
2. To patch your effects mix to any of the subgroups or to the Main outputs, press the desired output's button in the Assign section of the Fat Channel.
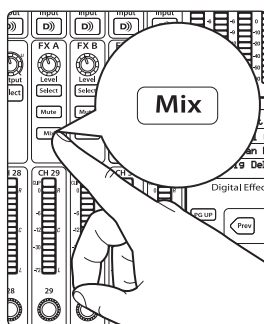
3. Next, press the Mix button in the FXA section. The Fat Channel meters will display the send level of each of the input channels to FXA. The encoders below each meter control the channel's level in FXA's mix. Use these encoders the same way that you use the faders to set the output level to your main mix. The higher a channel's level is in the effects mix, the more processed ("wetter") it will sound.
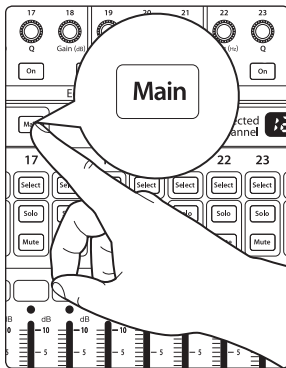
    Let's say that you are using reverb to liven up a relatively dead room. You might send a little bit of each input to the reverb, but you probably will not want much of the drums and bass to be processed, as too much reverb could reduce their impact and leave your mix without a sturdy foundation. So rather than turning the aux-send level for the kick drum channel all the way up, turn it so that the meter reads between 20% and 30% saturation. This way, only a small portion of the kick-drum input will be affected by the reverb.

4. By pressing the FXA Select button again, you can add dynamics processing and EQ to the overall monitor mix. These are great for adding sustain, removing too much ring, etc.
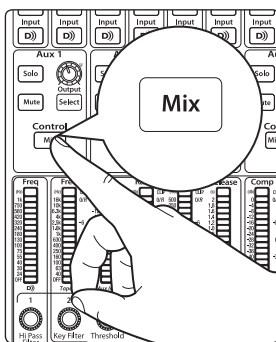
5.   Use the FXA Level knob to increase or decrease the overall effects-mix send level.

6.   To send FXA's mix to a monitor mix, press that aux bus's Mix button twice and use Encoder 5 to dial in the right level. (Encoders 6-8 control FXB-FXD's send levels.)

For information on changing the effects preset, type, or parameters, *see Section 5.2*.

4.4.7   Using an External Effects Processor

This section will detail how to use an external effects processor with your StudioLive. In this example, we will use Aux 1 to feed an external effects processor.

1.   To begin, connect the Aux 1 output to the input of your external effects processor and connect the outputs of your processor to Aux In A of your StudioLive, as illustrated.

2.   Turn the Output knob in the Aux 1 section to Unity (U).

3. Press the Aux In A Select button.

4. In the Fat Channel, assign Aux In A to the main outputs.

5. Press the Mix button for Aux 1. The Fat Channel meters will display each input channel's send level to Aux 1. The encoders below each meter control the channel's level in Aux 1's mix. You will use these encoders to set the send levels for each channel to the effects processor, the same way you used them to create a monitor mix. As with creating an internal FX mix, the higher a channel's level is in the FX mix, the more processed ("wetter") it will sound.
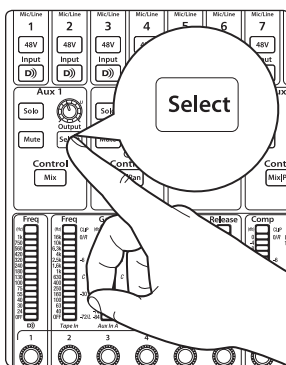
6. Once you have determined your effects mix, you can press the Select button for Aux 1 to add dynamics processing and EQ to the aux mix before it is sent to the external effects processor.
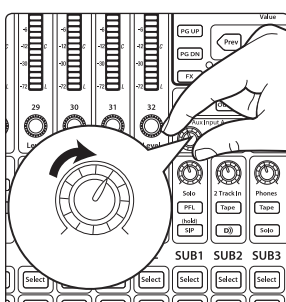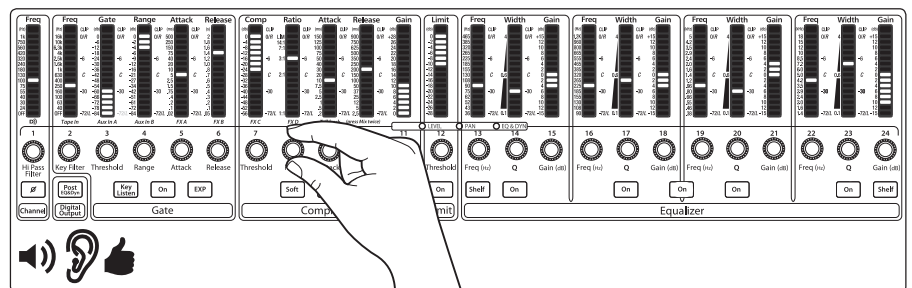
7. The Level knob for Aux Input A controls the level of the aux mix relative to the level of your main mix.

4.5   Subgroups

A subgroup allows you to combine multiple channels into a single bus so that the overall level for the entire group is controlled by a single fader. The StudioLive also allows you to apply the Fat Channel's noise gate, limiter, compression, and EQ to the group as a whole, in addition to the processing available for each channel. Subgroups can be soloed and muted.

You will find many uses for subgroups that will make mixing more convenient and will provide better control of your mix. At the end of this section, we explore two different ways in which subgroups can help you to create a more efficient mixing environment and a more successful live mix. But first let's go over the subgroup controls.

4.5.1   Subgroup Controls

Subgroup Channel Select Button. Enables Fat Channel Adjustments for a Subgroup.

As previously described in *Section 4.1.1*, the Select button routes its subgroup through the Fat Channel, allowing you to add dynamics processing, EQ, panning, etc., or to patch a subgroup to the Main bus. This will also open the Channel Info page for the selected subgroup in the LCD.

Solo Button. Turns Soloing On/Off for the Subgroup Bus.

This button will solo its subgroup to the monitor outputs, post-fader. PFL and SIP are not available for the subgroups.

When a Solo button is enabled, that channel or bus will automatically be selected, and its Select button will illuminate.

Mute Button. Turns Muting On/Off for the Subgroup Bus.

This button mutes its subgroup. It will illuminate red when the channel is muted.

Subgroup Fader. Controls the Level of the Subgroup Bus.

The fader controls the subgroup output's overall signal level.

The white area above the fader can be used as a scribble strip. Use only oil pencils. Other types of pens or pencils cannot be wiped off.

To clean the scribble strip, use a lightly damp cloth to remove the writing.

4.5.2   Creating Instrument Subgroups

Grouping individual instruments that create a section in your mix has obvious advantages: The entire group can be muted or soloed, brought up or down in a mix, and faded in or out for a more polished intro or outro. Some of the most common submix groups are drums, backing vocals, horn sections, and string sections. Drums are a classic application for subgroup mixing. We will use a drum group in this particular example but these principles can be applied when grouping any type of instrument section in a live mix.

A drum group is especially useful when every piece in the drum kit has a microphone on it. In this example, our drums will be connected to the StudioLive as follows:

Channel 1: Kick
Channel 2: Snare Top
Channel 3: Snare Bottom
Channel 4: Floor Tom
Channel 5: Tom 1
Channel 6: Tom 2
Channel 7: Overhead Left
Channel 8: Overhead Right
Channel 9: Hi-Hat

We will create a stereo subgroup by linking Subgroups 1 and 2.

1. The first step in creating a subgroup is to get a good mix of the instruments you are grouping—in this case, the drums. Beginning with Channel 1 (Kick), raise the fader and, with the drummer's assistance, set the input trim, EQ, and dynamics for each drum separately.

2. As you select each channel and dial in its Fat Channel settings, assign each channel to be routed to Subgroup 1 and unassign it from the Main bus. When you are done with each channel, lower the fader before moving on. When you get to Channels 7 and 8 (Overheads), you may want to consider stereo linking them.

3. After you have gone through the entire kit and are satisfied with each channel's level, EQ, and dynamics, have the drummer play the entire kit, and set the relative volume and panning for each mic in the mix. Press the Select button above Subgroup 1.
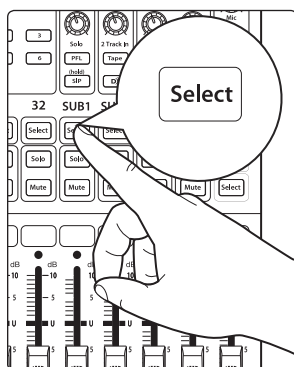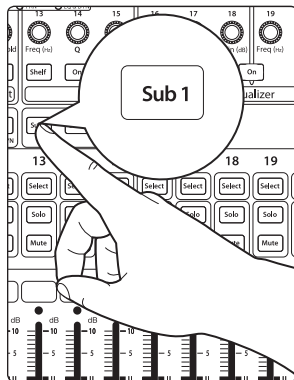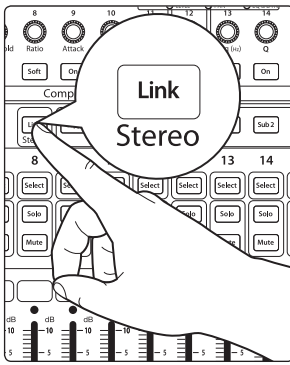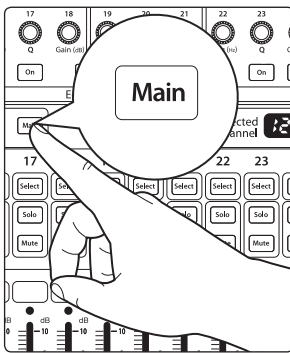
4. In the Fat Channel's Stereo section (to the right of the Pan display), enable Link.
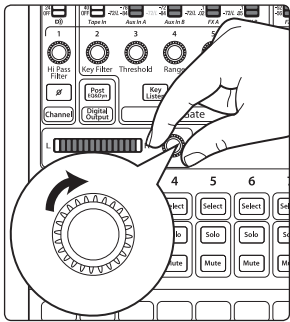
5. Turn the Pan knob all the way clockwise to set the stereo pan to hard left and right. Now Subgroups 1 and 2 are linked, with Sub 1 panned hard left and Sub 2 panned hard right. The channel panning is preserved.

6. Now assign Subgroup 1/2 to the Main outputs. You can now use the Fat Channel section to add dynamics processing and EQ to the stereo drum group. Subgroup 1's fader controls the level for the left side of your drum mix, and Subgroup 2's fader controls the right side.

4.5.3   Creating Effects Groups

This is perhaps one of the most creative ways in which a subgroup can be used. By assigning an effects mix to a subgroup, the front-of-house engineer can become, in essence, a member of the band. This is especially useful when employing specialty or signature effects. For instance, a typical vocal mix for an electronic band dramatically trails off in a wash of reverb, whereas a reggae band usually has delay on vocals. The StudioLive allows you to assign any of the onboard effects buses to a subgroup.

There are several advantages to assigning an effect like delay or reverb to a subgroup rather than simply using the FX Bus level control:

*   You can quickly add or subtract the effect by grabbing a fader.

*   When using a subgroup to control the effects level, you are controlling the level of the effects return, whereas the FX Level controls the overall send-mix level.

*   The effect can be muted or soloed.

*   The performers on stage can have a different amount of the effects in their monitor mix than the audience hears in the main mix, enabling you to reduce the possibility of feedback, while providing the performers with the tools they need for their best performance.
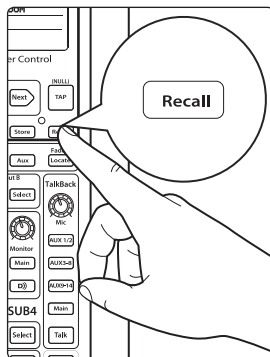
Let's take the example of the reggae band and assign the delay on FX C to Subgroup 3.
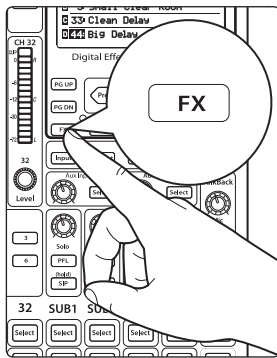
1. Press the FX button in the Master Control section to access the Effects menu.
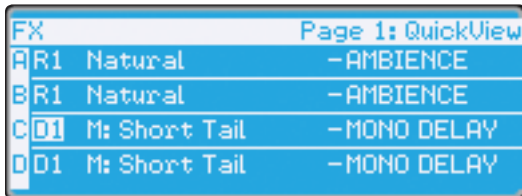
2. In the FX C preset load field, use the Value encoder to scroll through the effects library until you find a suitable delay.

3. Press Recall to load it.

4. Press the FXC Select button to jump to the Parameter Detail page. This will display every parameter available for the preset that was loaded, and you can adjust its parameters to taste. (See Section 5.1 for complete operating instructions.)
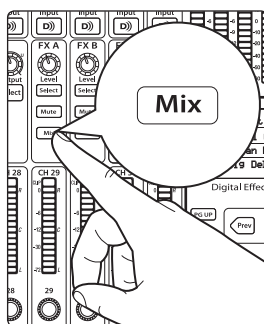
5. Next, decide which channels should be sent to the effects bus. In dub and reggae music, the vocals are most often sent to a delay, so let's send our two vocalists on Channels 10 and 11 to that delay. To assign the vocals, select the Mix button on FX C.
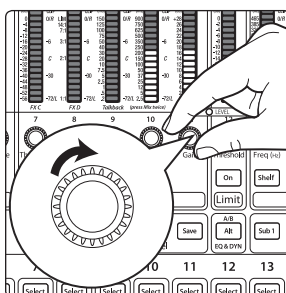
6. Using the meter section, locate the send encoders for Channels 10 and 11 and turn them to a little more than 50%.
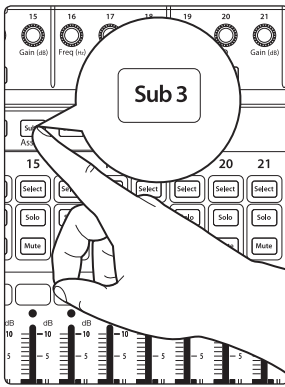
7. Press the Select button for FX C and assign this bus to Subgroup 3 and unassign it from the Main bus. If you like you can also add some dynamics processing and EQ at this point.
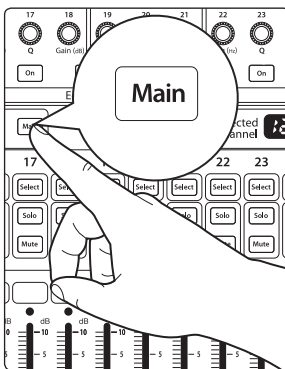
8. Press the Select button for Subgroup 3 and assign the group to the Main output. (Because a delay can increase the signal's volume quite dramatically, you may want to experiment with the delay at its most intense setting, with FX C's output turned up, and use the limiter for Subgroup 3 to keep the level under control.)

    The vocal-delay level is now controlled by the Subgroup 3 fader, and you can use it to season the reggae band's performance.

    The Tap button allows you to go one step further and set the tempo of the delay to match the tempo of the song.

4.6   Main Output Bus

Main Select Button. Enables Fat Channel Viewing.

As previously described in *Section 4.1.1*, the Select button routes its channel through the Fat Channel, allowing you to add dynamics processing, EQ, panning, etc. This will also open the Channel Info page for the Main bus in the LCD.

Main Fader. Controls the Level of the Main Output.

The fader controls the overall level of the main stereo and mono outputs.

The white area above the fader can be used as a scribble strip. Use only oil pencils. Other types of pens or pencils cannot be wiped off.

To clean the scribble strip, use a lightly damp cloth to remove the writing. Spit works pretty well, too.

4.7   Mute Groups (StudioLive 32.4.2AI only)

The StudioLive 32.4.2AI features six mute groups. A mute group allows you to mute and unmute multiple channels and buses with the press of a single button. With the six mute groups on the StudioLive 32.4.2AI, you could, for example, assign the drum mics to Mute Group 1, the instrumentalists to Mute Group 2, the background vocalists to Mute Group 3, all the aux buses to Mute Group 4, all four FX buses to Mute Group 5, and every channel on the mixer to Mute Group 6. Then, during the acoustic jam, you can mute all of the drum mics with one button. When the lead singer is introducing the band and saying "Hello, New Orleans!" you can mute all effects assigned to his vocal. When the band goes on break, you can mute all channels at once. And when it's time to break down the stage and start unplugging things, you can just lower the main fader and mute all stage monitors with one button.

Power User Tip: While the 24.4.2AI and 16.4.2AI do not have Mute Groups available on the control surface, the same features can be accessed from within VSL-AI. See the "VSL-AI: Overview Tab" section in the StudioLive AI Software Library Reference Manual for details.
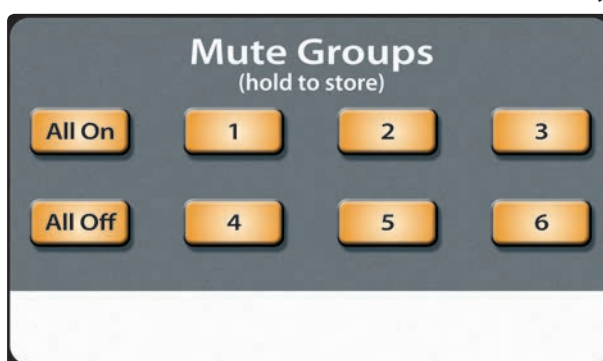
All On Group. Mutes all Channels and Buses.

The All On group is a preconfigured mute group that includes every channel and bus with a Mute button. Pressing this button activates every mute on the StudioLive mixer. Press it again to unmute them.

If you unmute a channel while the All On group is active, the All On button will blink to alert you that the mute group has been altered.

All Off Button. Clears all Mutes.

When the All Off button is pressed, any channel or bus that has been muted will be unmuted.

**Mute Group 1-6 buttons.** Engages/Disengages assigned Mute Groups.

When any of the Mute Group buttons is pressed, the assigned group of channels or buses will be muted/unmuted. To create a mute group, press and hold any Mute Group button.

If you unmute a channel or bus in an active mute group, that Mute Group button will blink to alert you that the mute group has been altered. Press and hold the Mute Group button to resave your group without that channel or bus, or reactivate the channel or bus mute to return the mute group to the saved setting.

Power User Tip: Both the mute groups and the All On group only add mutes to your mix and remove the same mutes they added. Mute groups will not clear mutes that were active prior to the mute group being engaged. (i.e., if a mute is engaged when its mute group or the All On group are enabled, it will still be engaged when you disable the mute group or the All On group.) The exception to this rule is the All Off button. This button will clear any mute that is currently enabled and will deactivate any mute group that is active, including the All On group.

4.7.1   Creating a Mute Group (StudioLive 32.4.2AI only)

Creating a mute group is quick and easy. In this example, we will be creating a mute group for Channels 1-10, using Mute Group 1.
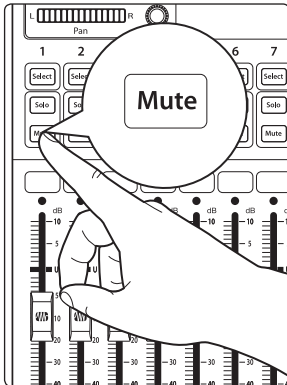
1. Mute Channels 1-10.
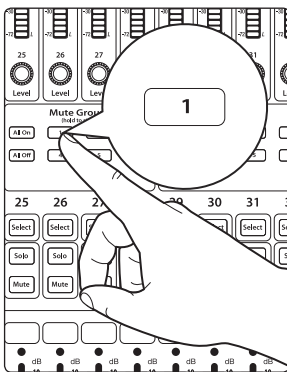
2. Press and hold the Mute Group 1 button. It will flash for one second indicating that it is storing the group. When the group has been stored, it will illuminate.

4.8   Master Section

The Master section of the StudioLive controls Aux Inputs A and B, the Talkback controls, and the 2 Track Input.

4.8.1   Aux Inputs A and B

The StudioLive AI mixers feature an internal effects processor that allows you to load four different effects at the same time. You also can patch processed return signals from an external effects processor using Aux Inputs A and B.

Power User Tip: *Aux Inputs A and B can also be used as stereo line-input channels for line-level devices like keyboards, synths, and amp modelers. Because Aux Inputs A and B can be sent to any aux mix, subgroups, or the Main mix, they should be used as extra input channels whenever you need them.*

Aux Input Select Button. Enables Fat Channel Viewing.

As previously described in **Section 4.1.1**, the Select button routes its channel through the Fat Channel, allowing you to adjust panning and to enable output assignments. This will also open the Channel Info page for the selected aux input.

Aux Input Level Control. Controls the Signal Level of the Aux Input.

This knob controls the overall level of the aux-input signal.

4.8.2   Talkback System

The StudioLive features a Talkback microphone input on the back panel. This can be routed to the aux outputs and to the mains. It is important to note that the aux outputs are grouped in the Talkback section. For example, if you are using Aux 1 as a mono monitor send to the bass player's floor wedge and Aux 2 as the monitor send to the keyboard player, the talkback signal will be sent to both monitors—so don't say anything you wouldn't want both to hear!

*Power User Tip: While the talkback microphone is patched to aux outputs in groups, you do have independent level control over this input in each of the aux mixes by using Aux Flip mode. Because of this, you can adjust the talkback signal up or down, or even off, on a per-aux basis. For more information about Aux Flip mode, **please see Section 4.4.3**.*

Talkback Mic Level. Controls the Overall Level of the Talkback Mic.

This knob controls the overall volume of the external Talkback input. The trim for the Talkback input is located on the back panel next to the input. *See Section 3.1* for details.

Output Selector Buttons. Assigns the Talkback Mic to the Aux and/or Main Outs.

These buttons assign the talkback mic to a specified output. These buttons are toggled on/off and will illuminate indicating that the Talkback output is active. The talkback mic can be assigned to every output at the same time.

*Power User Tip: While the talkback is assigned and unassigned to multiple Aux buses at once, each Aux mix provides its own individual level control for the talkback via Aux Flip mode. **See Section 4.4.3 for details.***

Talkback Button. Turns the Talkback Mic On/Off.

This latching button turns the talkback mic on and off. It will illuminate to indicate that the talkback mic is active. When illuminated, the talkback-mic signal will be patched to all assigned outputs.

4.8.3   2 Track In

The 2 Track input allows you to patch in an analog input or a playback stream from your audio application. This enables you to compare audio sources during mixdown or patch in intermission music between bands without using any of your input channels. This is a summing bus, so both the external tape input and the two-track digital return can be engaged at the same time.

2 Track In Volume Control. Adjusts the Level for the Tape Input.

This knob adjusts the level for both the tape input and the main digital return.

Tape Input On/Off Button. Patches the Tape Input to the Main Outputs.

This button patches the Tape input to the main outputs. It will illuminate to indicate that the Tape input is being sent to the mains. To disable the routing, simply press the button again.

Main Digital Return On/Off Button. Patches the Main Digital Return to the Main Outputs.

This button routes the main left/right digital returns (StudioLive AI Outputs 33-34/25-26/17-18) to the main outputs. It will illuminate to indicate that the main digital return is being sent to the main outputs of the StudioLive. To disable the routing, simply press the button again. For more information, *please consult "Digital Sends and Returns" in the StudioLive AI-Series Software Library Manual*.

4.9   Solo Bus

The StudioLive features an independent Solo bus. This feature is extremely useful in setting levels for monitor mixes, dialing in dynamics processing on each channel, and fixing issues during a live show without interrupting the main mix.

The Solo bus has three different modes: AFL (default), PFL, and SIP.

- AFL (After-Fader Listen). AFL sends the channel or subgroup signal to the Solo bus post-fader so that you can control the level of the soloed signal with the fader. This is the StudioLive's default setting.

- PFL (Pre-Fader Listen). PFL sends the channel or subgroup signal to the Solo bus before it reaches the fader so the fader does not affect the soloed signal.

- SIP (Solo In Place). This is also known as "destructive solo." When channels are soloed in this mode, every channel that isn't soloed will be muted, and only the soloed channels will be sent to their assigned outputs. While useful in dialing in dynamics during soundcheck, this mode is dangerous during a live show. We recommend that this mode be turned off when mixing live events.

4.9.1   Solo Bus Controls

Cue Mix Level Control. Adjusts the Overall Level of the Solo Bus.

This knob adjusts the overall level for the Solo bus.

PFL/AFL Toggle Button. Enables PFL Soloing.

The default setting for the Solo bus is After-Fader Listen (AFL); by pressing PFL, Pre-Fader Listening is enabled. In either mode, pressing Solo on any channel or bus routes that channel to the Solo bus and has no effect on the main or subgroup mixes.

PFL soloing is not available for the subgroups.

Aux bus soloing is always PFL, regardless of whether this mode is engaged.

SIP (Solo In Place) On/Off Button. Enables Solo In Place.

SIP (Solo In Place), or "destructive soloing," mutes every unsoloed channel on the StudioLive. If one of the muted channels is routed to the mains or a subgroup, it will be muted in those outputs. This also applies to soloed channels: The output routing is still active. Note that while you can manually unmute a channel, this mode should be used with extreme caution during a live performance. Only the input channels can be placed in destructive soloing. The subgroups and aux buses are omitted from SIP mode.

To enable SIP, press and hold the button until it illuminates red. This ensures that you cannot enter into destructive Solo mode by accident.

Power User Tip: *When SIP is engaged, channel mutes will only apply to the subgroup and main bus assignments. SIP does not mute input channels in aux-bus mixes. Because of this, you can use SIP to dial in a mix in the mains without disturbing the musicians' last-minute rehearsal on stage.*

Destructive soloing is also a great way to tune each channel's dynamics individually in live-mixing situations or do surgical editing in the studio. SIP mode mutes every channel and bus that is not soloed in the Main bus (that is, if Channel 3 is soloed, you will only hear Channel 3 in your mains). This makes a great fine-tuning tool but it can quickly destroy a live mix. We highly recommend that you drop out of this mode once the show has started.

4.9.2   Solo Modes
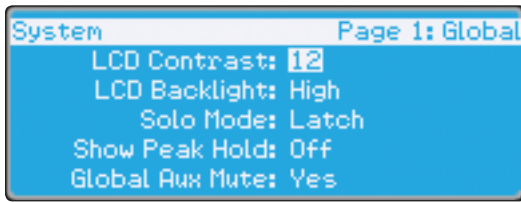
From the first page of the System menu, you can choose between three Solo modes. To access these modes, press the System button and navigate to Page 1: Global. Press the Next button to navigate to the Solo Mode field and use the Value encoder to scroll through the three modes:

- **Latch:** This is the default Solo mode. When Latch Solo mode is active, you can solo multiple channels and buses at once.

- **Radio:** When Radio Solo mode is active, you can solo only one channel or bus at a time.

- **CR:** When CR Solo mode is active, soloing any channel will automatically patch the Solo bus to the Monitor bus and disable any other buses that are currently engaged in the Solo bus. While in CR Solo mode, you can solo multiple channels and buses at once, but you can only patch one input or bus to the monitor bus at a time. Latch Solo and Radio Solo modes allow the Monitor bus to function as a summing amp. ***See Section 4.10 for details.***

4.9.3   Solo Clear
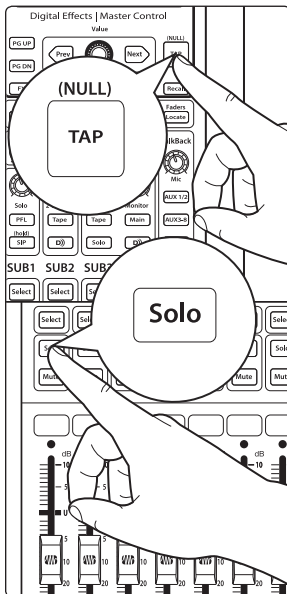
To clear all solos on your StudioLive, press and hold the Tap Tempo button while simultaneously pressing any Solo button the mixer. All Solo buttons will be disengaged.

4.9.4   Using the Solo Bus for Monitoring

When mixing live, or when recording multiple musicians at once, it is often necessary to quickly listen to just one instrument or group. The Solo and Monitor buses can be used together for this purpose. It is important to note that if you wish to monitor with speakers, rather than with headphones, it is necessary to connect the speakers to the Control Room outputs on the back of your StudioLive, rather than to one of the main output pairs.
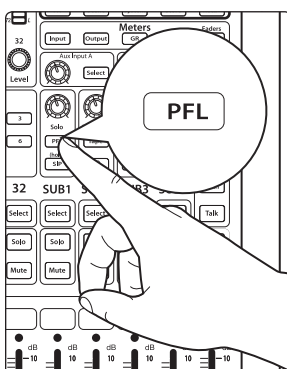
1. First decide whether you want to listen to your soloed channels before or after the fader setting. If you'd like to monitor before the fader level, press the PFL button in your Solo bus section.
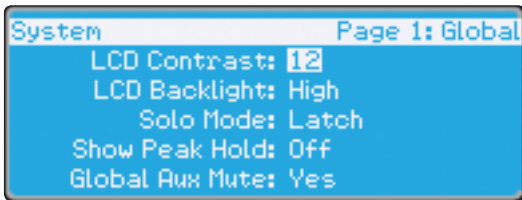
2. By default, your StudioLive mixer is set to Latch Solo mode. For the purposes of this tutorial, this is the setting you will need to use. *See Section 4.9.2* for more information about solo modes.
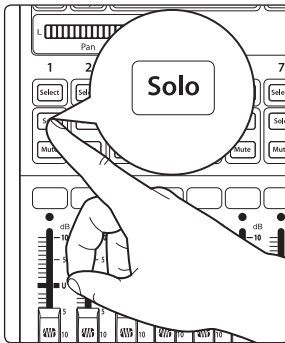
3. Next, press the Solo buttons on the channels, auxes, and subgroups you want to monitor.
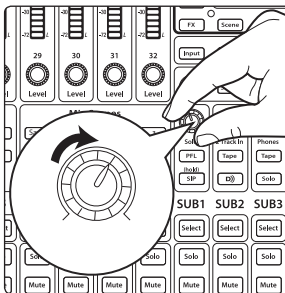
4. Turn the Solo Level knob in the Solo section to about 12 o'clock.
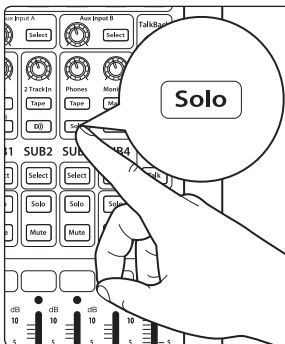
5. Finally, select the Solo button in your monitor bus and dial in a comfortable listening volume for your headphones or monitors. You can increase the overall volume of the Solo bus using the Solo Level knob in the Solo section.

Power User Tip: This feature can also be used to listen to a monitor mix that is routed to an aux send. Let's say a vocalist on stage complains that there is too much bass in his monitor but you are confident that no bass is being sent to that particular aux send. You could be mistaken but most likely an open microphone on stage is picking up the bass signal. To determine the cause, solo only the aux send in question and, again, select the Solo button for the monitors/headphones. You can now listen to exactly the same mix as your troubled vocalist and can fix his monitor mix quickly. This application is also useful in heading off a feedback problem.

4.9.5   Using Solo in Place (SIP) to Set Up a Mix

We started this manual with a quick and easy way to set up the input levels for your StudioLive, ensuring that you have the highest possible input level without clipping your analog-to-digital converters. The next step is to set up your mix by dialing in the dynamics, EQ, and fader settings for each channel. Enter Solo In Place (SIP). As previously mentioned, Solo In Place is a great way to dial in your mix without disturbing your musicians' last minute rehearsal or subjecting your audience to that impromptu jam session on stage. Radio Solo mode is especially useful for this purpose.

1. Radio Solo mode is especially useful for this purpose because it allows you to quickly solo just one channel at a time. To begin, press the System button and change Solo mode to "Radio."
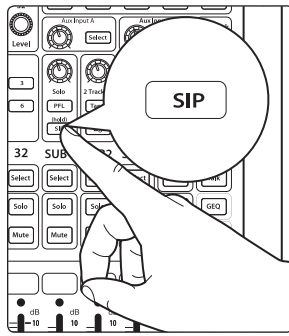
2. Press and hold the SIP button in the Solo section until it illuminates.

3. Raise all your channel faders and your main fader to unity gain.
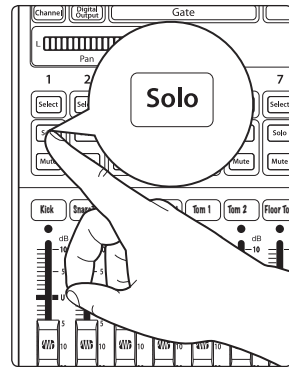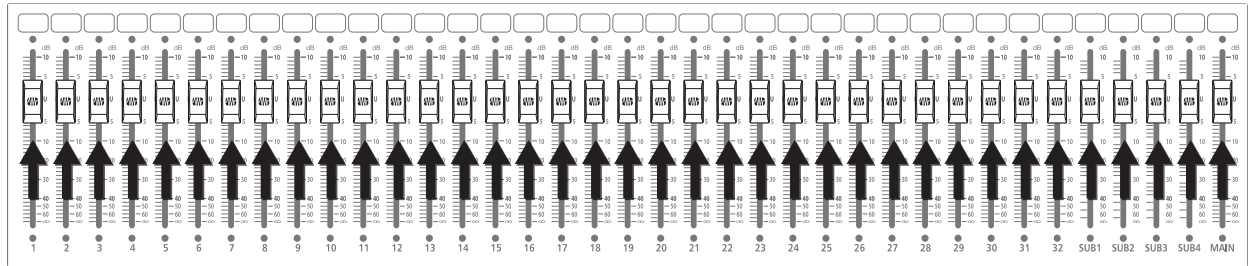
4. Most engineers start with the drums and work from the bottom up, so press the Solo button on your kick-drum mic channel. Notice that all the other channels on your StudioLive have been muted, and the kick-drum channel is selected.

5. The Fat Channel will display the dynamics processing, EQ, output routing, and pan settings for the kick drum.

6. Using the encoders and meters in the Fat Channel, set up the compressor and EQ for this channel.
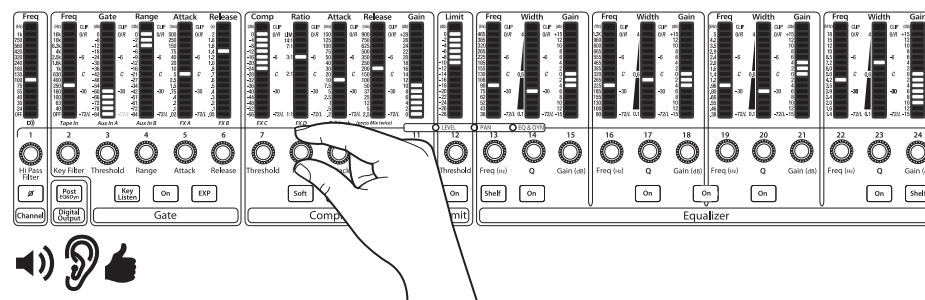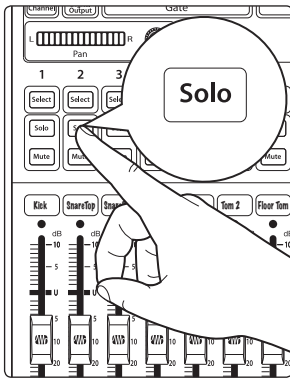
7. Once you are satisfied, press the Solo button on the snare-mic channel and repeat steps 4 through 5. In this way, continue with each drum mic and then move on to the other instruments that are connected to the StudioLive. When you have finished with all instruments, press the SIP button again and set up your fader mix.

Power User Tip: *While Radio Solo mode is especially useful for setting levels as described above, it is not ideal for soloing during a live show. Because of this, once you have set your levels, go back to the System menu and select either Latch Solo or CR Solo mode.*

4.10   Monitor Bus

The StudioLive features a headphone output and control-room outputs, giving you the ability to monitor multiple sources. The Monitor bus on the StudioLive allows you to monitor the main outputs, Solo bus, main digital return from your computer, and the stereo tape input.

Note: *The Monitor bus is a summing amp in the default Latch mode and in Radio mode. However, when CR Solo mode is active, soloing a channel or bus will assign the Solo bus to the Monitor bus and will disable all other assignments. While in this mode, the Monitor bus will not function as a summing amp, and only one input or bus can be patched at a time. Soloing a channel will override all other assignments (e.g., if you are monitoring the Tape input when you solo a channel, the Solo bus will be patched and the Tape input will be unpatched).* ***See Section 4.9.2 for details.***

Tape Input Monitor Button. Assigns the Tape-Input Signal to the Monitor Bus.

The Tape monitor button routes the signal from the tape inputs (tape returns) to the monitor bus. The level for this input is controlled by the knob in the 2 Track In section.

Main Mix Monitor Button. Assigns the Main Mix to the Monitor Bus.

The Main Mix Monitor button routes the same signal that is being sent from the main outputs to the Monitor bus. This signal is always pre-fader.

Digital Return Monitor Button. Assigns the Main Left/Right Digital Return to the Monitor Bus.

The Digital Return Monitor button patches the main left/right digital return (StudioLive Outputs 33-34/25-26/17-18) to the Monitor bus. The level for this input is controlled first by the level set from the computer application producing the audio (e.g., Studio One) and then by a knob in the 2 Track In section. For more information about the main digital returns, ***please review "Digital Sends and Returns" in the StudioLive AI-Series Software Library Manual.***
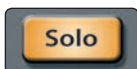
Solo Bus Monitor Button. Assigns the Solo Bus to the Monitor Bus.

The Solo Bus Monitor button patches any soloed channel, subgroup, or aux bus to the Monitor bus. This can be useful in any number of ways. For example:

•   Auditioning an aux-send monitor mix

•   Dialing in the dynamics processing and EQ on a subgroup

•   Creating a better blend for instrumental sections (horns, strings, etc.)

Power User Tip: *Because the Monitor bus is a summing amp in Latch Solo and Radio Solo modes, you can listen to the Solo and Main buses at the same time. In this way you can use the Solo level control to bring up a channel's level in just your monitor mix. This is especially useful when trying to determine the source of an odd frequency or tone, mid-show.*

Headphone Output Level Control. Adjusts the Overall Level of the Headphone Output.

This knob adjusts the overall level for the Headphone output.

The Headphone output is located on the front of the mixer, below the main fader.
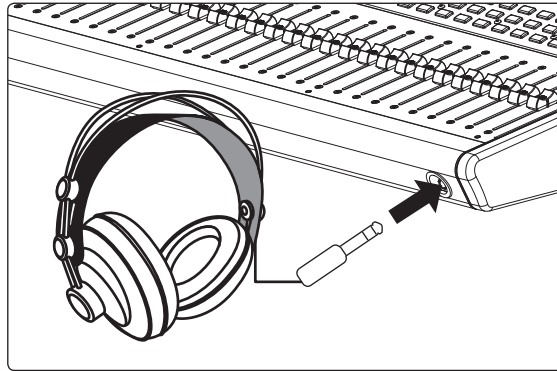
Control-Room Monitor Level Control. Adjusts the Overall Level of the Control-Room Monitor Outputs on the Rear Panel.

This knob adjusts the overall level of the control-room monitor outputs.

5   Digital Effects | Master Control

From the Digital Effects | Master Control section, you can select and change the parameters of the four internal effects processors, and you can store and recall every setting on the StudioLive. Because almost all of the StudioLive's features are controlled from the mixing surface (rather than using menus and submenus), you will mainly use this section to adjust the internal effects processors and to save and recall presets and scenes.

Power User Tip: With all menus, the StudioLive remembers which page you were on when you navigated away to another menu. When no menu buttons are selected (FX, Scene, System, or GEQ), the Channel Info page for the currently selected channel or bus will be displayed. To quickly jump to page 1 of any menu, simply press and hold the Tap Tempo button.

5.1   Channel Info Page

As previous mentioned in *Section 4.1.1*, the Channel Info page will launch whenever a channel or output bus is selected. This is the default menu for your StudioLive. Pressing the FX, Scene, System, or GEQ buttons will open their respective menus. When no other menu button is enabled, the Channel Info page will be visible.

Please Note: When an FX bus is selected, the Parameter Detail page for that FX bus will launch.

5.1.1   Customizing Channel and Bus Names

To rename your channel:
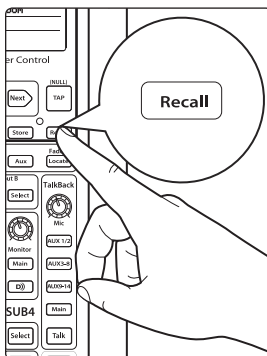
1. **Press the Recall button** to create a custom name.

2. **Turn the Value encoder** clockwise or counter-clockwise to change the letter. The StudioLive allows you to customize the name with uppercase and lowercase letters and offers a selection of numerals and punctuation marks. You can insert a space by simply pressing the Tap button.
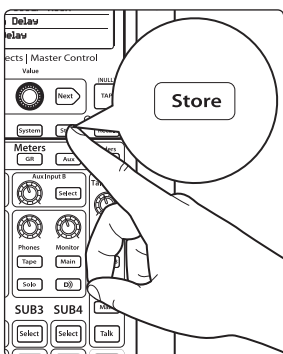
3. Once you are satisfied with your changes, **press the Store button**. It will illuminate while the name is being written to the StudioLive's internal memory. Once the Channel preset is saved, the Store button will return to its unlit state.

You can also use VSL-AI or SL Remote-AI to create custom names. For more information, *please review "Universal Control AI and VSL-AI" and "StudioLive Remote AI for iPad" in the StudioLive AI-Series Software Library Manual.*

NOTE: There is no Channel Info page for the FX buses. When one of the FX buses is selected, the FX menu will open to that FX bus's parameter detail page.

5.2   The Digital FX (Effects) Menu

The StudioLive AI mixers feature four internal effects processors. The processors for FX A and FX B are dedicated to reverb and access the StudioLive's reverb library. The processors for FX C and FX D are dedicated to delay effects and access the StudioLive's library of delay presets. These effects buses can be routed to any of the subgroups, the aux buses, or the main outputs.

NOTE: FX B and FX D are disabled when HD Mode (88.2 and 96 kHz) is active.

To access the effects library and make adjustments to effects parameters, press the FX button in the Master Control section.
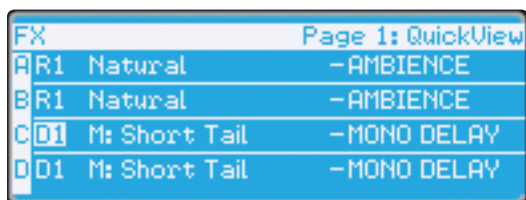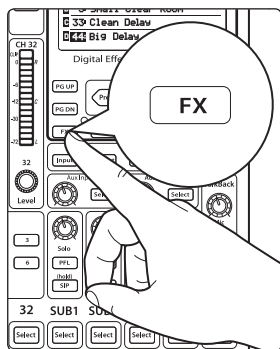
The first page of the FX menu is the QuickView screen. It displays the effects presets loaded on each of the internal effects buses and the main parameter for each. Effect A is assigned to FX A bus, Effect B is assigned to FX B bus, Effect C is assigned to FX C, and Effect D is assigned to FX D.
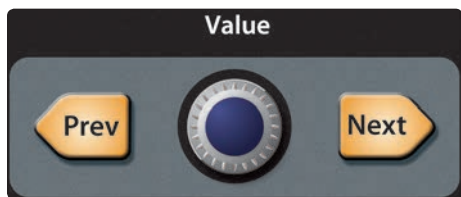
Use the Next and Prev buttons to navigate through the screen. To change a preset, use the Value encoder to scroll through the library.
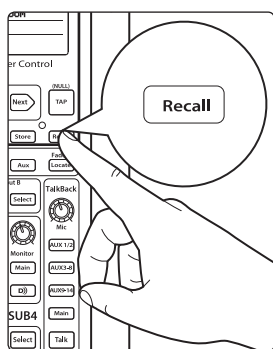
When you have arrived at your selection, press the Recall button to load it.

Press the Page Down button to move to the next pages of the FX menu. Pages 2-5 of the FX menu display the Parameter Detail pages for FX A, FX B, FX C, and FX D, respectively. The Parameter Detail page displays all the parameters for the presets loaded onto the corresponding effects bus. These parameters will change depending on the type of effect you have chosen. Again, use the Next and Prev buttons to navigate through the screen and use the Value encoder to change the selected parameter.

Power User Tip: *You can jump to an FX bus's Parameter Detail page by simply pressing its Select button; for example, if you press FXC's Select button, the FX page will open to FX C's Parameter Detail page.*

5.2.1 Creating FX Presets

Page 1 of the FX menu provides access to your library of effects presets. Pages 2-5 provide access to the 13 FX types. An FX preset is made by adjusting the default parameters of an FX type, so one FX type can be the foundation for many different presets.

The StudioLive contains a library of 50 custom reverb and delay presets designed by PreSonus. In addition to these presets, there are 148 available locations for your custom effects library. The factory presets can be altered, renamed, and overwritten.

Create an FX preset using a factory preset as a jumping-off point, or start from scratch with an FX type of your choosing. This section describes the latter approach.

1. Navigate to page 2 of the FX menu or press FX A's Select button.
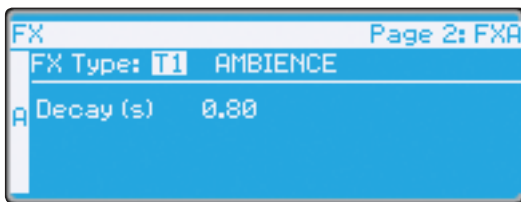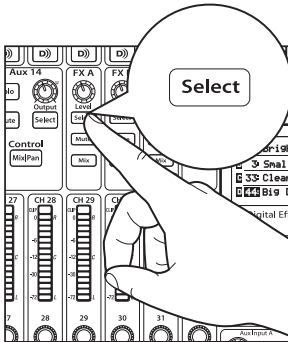
2. Using the Value encoder, navigate through the FX Type library until you find the FX type you'd like to use.
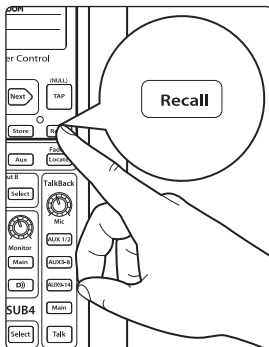
3. Press the Recall button to load the FX type and its default parameters.

4. Use the Next button and the Value encoder to dial in your FX preset to taste.

5. Page 6 allows you to store your changes to presets loaded on any of the FX buses to the same location, or to a new location, and to customize the name of your creation.
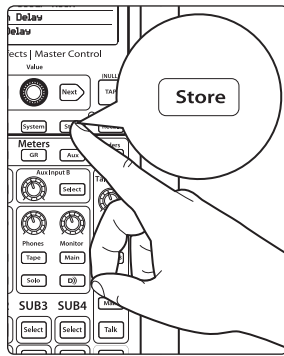
6. To jump to this page, simply press the Store button while you have a field in the desired effect selected, either in the QuickView page, or in the Parameter Detail page.
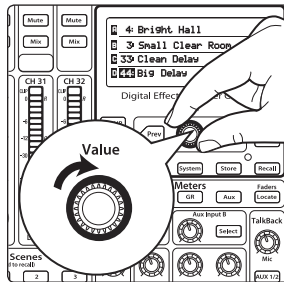
7. Use the Value encoder to change the library location to which you will store your new effects preset, unless you wish to overwrite the currently selected preset.
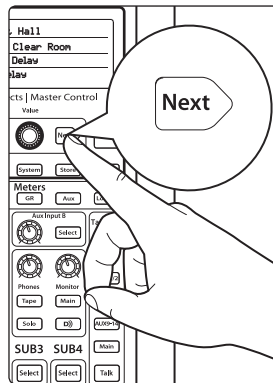
8. Press the Next button to navigate to the first letter of the preset name.
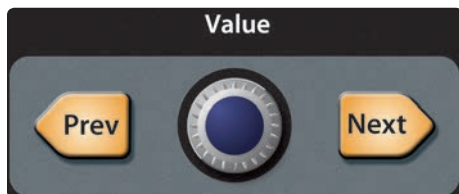
9. Turn the Value encoder clockwise or counter-clockwise to change the letter. The StudioLive allows you to customize the name with uppercase and lowercase letters and a selection of numerals and punctuation marks. Press the Tap button to quickly insert a space.
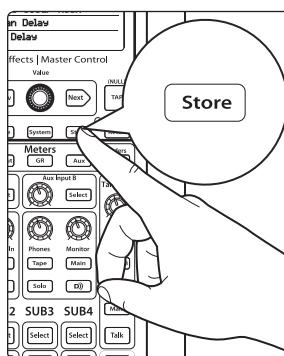
10. Once you are satisfied with your changes, press the Store button, which will illuminate while the effects preset is written to the StudioLive's internal memory. Once the preset is saved, the Store button will return to its unlit status.
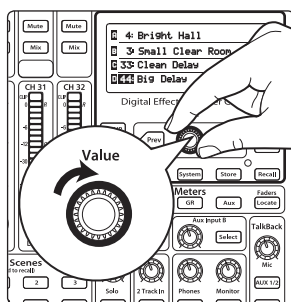
11. If you'd like to store the effects currently loaded on multiple FX buses, simply navigate to the FX Bus field and use the Value encoder to select another bus.

5.2.2   Reverb and its Parameters

Reverberation—or reverb, as it is more commonly known—is perhaps the most widely used effect. Natural reverb is created by sound waves reflecting off of a surface or many surfaces. For example, when you walk across the wooden stage in a large hall, thousands of reflections are generated almost instantaneously as the sound waves bounce off the floor, walls, and ceilings. These are known as early reflections, and their pattern provides psycho-acoustic indications as to the nature of the space that you are in, even if you can't see it. As each reflection is then reflected off of more surfaces, the complexity of the sound increases, while the reverb slowly decays.

The reason for the widespread use of reverb in audio recording is fairly self-evident: human beings don't live in a vacuum. Because our brains receive cues about the nature of the space around us based partially on audio reflections, a sense of space makes an audio recording sound more natural and, therefore, more pleasing.

Note: Reverb types and presets can only be loaded on FXA and FXB.

The following parameters are available for the nine reverb types the StudioLive offers:

Decay. Decay is the time (in seconds) required for the reflections (reverberation) to die away. In most modern music production, reverb decay times of between one and three seconds are prevalent. A reverb setting with strong early reflections and a quick decay are a great way to create a stereo effect from a mono source.

Predelay. Predelay is the time (in milliseconds) between the end of the initial sound and the moment when the first reflections become audible. Imagine you're back on that stage in a large music hall. This time you stand on the very edge of the stage and shout "Hello world!" toward the center of the hall. There will be a brief pause before you hear the first noticeable reflections of your voice, because the sound waves can travel much further before encountering a surface and bouncing back. (There are closer surfaces, of course—notably the floor and the ceiling just in front of the stage—but only a small part of the direct sound will go there, so those reflections will be much less noticeable.) Adjusting the predelay parameter on a reverb allows you to change the apparent size of the room without having to change the overall decay time. This will give your mix a little more transparency by leaving some space between the original sound and its reverb.

Note: Predelay control is not available on every reverb type.

Early Reflections. Early reflections are those that reach the listener a few milliseconds after the direct signal arrives. Your brain uses them to identify the size of the room you're in. If you are trying to simulate a specific type of room, this control will be extremely important. This control allows you to set the level (in decibels) of the early reflections. The louder the early reflections, the smaller the room will seem.

Note: Early Reflections control is not available on every reverb type.

5.2.3   Delay and its Parameters

A delay essentially creates an echo, although you can often use delays to create more complex time-based effects. The source signal is delayed so that it is heard later than it actually occurred.

Note: Delay types and presets can only be loaded on FXC and FXD.

The following parameters are available for the four delay types the StudioLive offers:

Time. This is the time (in milliseconds) between the source signal and its echo. The simplest delay effect is a single repeat. A short delay between 30 and 100 ms can be used to create slap-back echo, while longer delay times produce a more distant echo. Delay times that are too short to hear as distinct echoes can be used to create thickening effects. Whether these echoes are timed with the tempo is a matter of stylistic choice.

This is the parameter that is controlled by the Tap Tempo button. Using the Tap button on the StudioLive, you can speed up or slow down these repeats or, more commonly, time the repeats to occur with the tempo of the music.

Power User Tip: While you have to select the Time parameter in order to use the Tap button, you only have to do this the first time you use the Tap button for that effect. Once the Tap button has been used to control the Time parameter on FX buses C or D, it will always control the time of that particular delay, no matter what page you are currently viewing. To assign the Tap button to control another delay, simply navigate to that delay's Time parameter and use the button to enter the desired delay time.

Time X. Time X is the value of the beat you are using as a reference for the tempo. The basic unit of measure is a quarter note, so for example, if the beats you are tapping represent quarter notes in the music, you would set Time X to 1.00. If they are eighth notes, you would set Time X to 0.50; half notes would be 2.00, and so on. In this way, you can precisely synchronize or syncopate the delay echoes to the music in real time.

Note: The Stereo Delay offers two Time X controls. With the Ping Pong delay, the Pong X parameter serves the same purpose.

Variable Feedback. Variable feedback, or regeneration, produces multiple decaying repeats. Increasing the feedback value increases the number of echoes, as well as the resonance that is created as one echo disappears into another.

F_Frequency. Sets the center frequency in Hz for the Filter Delay.

F_Gain. Sets the boost at the center frequency for the Filter Delay.

F_Q. Sets the Q for the Filter Delay. The Q is the ratio of the center frequency to the bandwidth. When the center frequency is constant, the bandwidth is inversely proportional to the Q, so as you raise the Q, you narrow the bandwidth.

5.2.4   Reverb Effects Preset Library

REVERB EFFECTS
POS.	TYPE	NAME	POS.	TYPE	NAME
R1	AMBIENCE	Natural	R17	WARM HALL	Log Cabin
R2		Lively	R18		Wood Floors
R3	SMALL ROOM	Closet	R19		Brick Club
R4		Studio A	R20		High Ceilings
R5		Studio B	R21	GATED HALL	GateVerb Short
R6		Bedroom	R22		GateVerb Med
R7	BRIGHT ROOM	Kitchen	R23		GateVerb Long
R8		Tile Floors	R24		GateVerb Extreme
R9		Tile Bathroom	R25	LARGE HALL	Church
R10		Concrete Floors	R26		Cathedral
R11	SMALL HALL	Radio Booth	R27		Gymnasium
R12		Small Club	R28		Arena
R13		Big Club	R29	PLATE	PlateVerb Shimmer
R14	BRIGHT HALL	Brick House	R30		PlateVerb Thick
R15		Linoleum Room	R31		PlateVerb Drums
R16		Tile Room	R32		PlateVerb Vox
			R33-99	USER-CREATED PRESETS	

5.2.5   Delay Effects Preset Library

DELAY EFFECTS
POS.	TYPE	NAME	POS.	TYPE	NAME
D1	MONO DELAY	M: Short Tail	D10		Long Slap
D2		M: Short Slap	D11		Spacey
D3		M: Long Slap	D12		Long Tail
D4		M: Triplet	D13		ST: Triplet
D5		M: Triplet	D14	PING-PONG DELAY	Ping-pong Slap
D6	FILTER DELAY	Analog Slap	D15		Ping-pong Delay
D7		Analog Trip	D16		Ping-pong Spacey
D8		Analog 8th	D17		Ping-pong Trip
D9	STEREO DELAY	Slap Quick	D18		Ping-Pong Purple Rain
			D19-99	USER-CREATED PRESETS	

5.2.6 Digital Effects Types

The StudioLive contains 13 different effects types with which you can create custom presets or redesign the included library of presets.

NAME	POS	PARAM (L1)	PARAM (L2)	PARAM (L2)	PARAM (L2)	PARAM (L2)	PARAM (L2)	PARAM (L2)
Ambience	T1	Reverb	**Decay (s) Default:** 0.69 **Range:** 0.29 – 1.09					
Small Room	T2	Reverb	**Decay (s) Default:** 0.79 **Range:** 0.39 ~ 0.59	**Predelay (ms) Default:** 12.0 **Range:** 1.00 ~ 40.0	**Early Reflec (dB) Default:** -15.0 **Range:** -25.0 ~ -8.00			
Bright Room	T3	Reverb	**Decay (s) Default:** 1.00 **Range:** 0.50 ~ 1.79	**Predelay (ms) Default:** 12.0 **Range:** 1.00 ~ 40.0	**Early Reflec (dB) Default:** -16.0 **Range:** -22.0 ~ -10.0			
Small Hall	T4	Reverb	**Decay (s) Default:** 1.39 **Range:** 0.59 ~ 2.19	**Predelay (ms) Default:** 20.0 **Range:** 1.00 ~ 50.0	**Early Reflec (dB) Default:** -22 **Range:** -35.0 ~ -15.0			
Bright Hall	T5	Reverb	**Decay (s) Default:** 1.59 **Range:** 0.79 ~ 2.39	**Predelay (ms) Default:** 24.0 **Range:** 1.00 ~ 60.0	**Early Reflec (dB) Default:** -22.0 **Range:** -35.0 ~ -15.0			
Warm Hall	T6	Reverb	**Decay (s) Default:** 1.59 **Range:** 0.79 ~ 2.50	**Predelay (ms) Default:** 50.0 **Range:** 10.0 ~ 100.0	**Early Reflec (dB) Default:** -30.0 **Range:** -40.0 ~ -20.0			
Gated Hall	T7	Reverb	**Decay (s) Default:** 1.00 **Range:** 0.59 ~ 1.79	**Predelay (ms) Default:** 40 **Range:** 5.00 ~ 80.0				
Large Hall	T8	Reverb	**Decay (s) Default:** 2.39 **Range:** 1.39 ~ 5.00	**Predelay (ms) Default:** 35.0 **Range:** 40.0 ~ 90.0	**Early Reflec (dB) Default:** -30.0 **Range:** -40.0 ~ -20.0			
Plate	T9	Reverb	**Decay (s) Default:** 1.39 **Range:** 0.50 ~ 4.00	**Predelay (ms) Default:** 10.0 **Range:** 1.00 ~ 40.0				
Mono Delay	T10	Delay	**Time (ms) Default:** 645 **Range:** 5.00 ~ 1.28k	**Time X Default:** 1.00 **Range:** 0.25 ~ 2.00	**Feedback Default:** 0.25 **Range:** 0.000 ~ 0.94			
Filter Delay	T11	Delay	**Time (ms) Default:** 645 **Range:** 5.00 ~ 1.28k	**Time X Default:** 1.00 **Range:** 0.25 ~ 2.00	**Feedback Default:** 0.25 **Range:** 0.000 ~ 0.94	**F_Freq (Hz) Default:** 800 **Range:** 100 ~ 3.00k	**F_Gain Default:** 12.0 **Range:** 0.000 ~ 24.0	**F_Q Default:** 0.69 **Range:** 0.19 ~ 2.00
Stereo Delay	T12	Delay	**Time (ms) Default:** 645 **Range:** 5.00 ~ 1.28k	**Time1 X Default:** 1.00 Range: 0.25 ~ 2.00	**Time2 X Default:** 1.00 Range: 0.25 ~ 2.00	**Feedback1 Default:** 0.25 **Range:** 0.000 ~ 0.94	**Feedback2 Default:** 0.25 **Range:** 0.000 ~ 0.94	**L-R Spread Default:** 0.50 **Range:** 0.000 ~ 1.00
Ping Pong	T13	Delay	**Time (ms) Default:** 645 **Range:** 5.00 ~ 1.28k	**Pong X Default:** 1.00 **Range:** 0.25 ~ 2.00	**Pong X Default:** 1.00 **Range:** 0.25 ~ 2.00	**Feedback Default:** 0.25 **Range:** 0.000 ~ 0.94	**L-R Spread Default:** 0.50 **Range:** 0.000 ~ 1.00	

5.3   Scenes

The StudioLive allows you to create and store a library of scenes. A scene is like a snapshot of your mix. It stores each Fat Channel parameter for every input and bus, as well as each fader's position, the aux and effects mixes, channel mutes and solos, and the input selection (analog input or digital playback stream).

5.3.1   S1: Zero Out (Board Reset)

Located at position S1 is a scene named Zero Out (Board Reset). This scene cannot be overwritten and returns each parameter to its default setting. All you have to do is lower the faders and return all trim knobs and output volume knobs (Solo bus, 2 Track In, Aux Input A and B, Aux Outputs, FX Outputs, Phones, and Monitor) to their lowest positions. The StudioLive will be zeroed out as follows:

INPUTS AND BUSES	SETTINGS							
	OUTPUT LEVEL	ASSIGN	PAN	SOLO	MUTE	STEREO LINK	AUX/FX SENDS	MODE
INPUT CHANNELS	NULL	MAINS	C	OFF	OFF	OFF	NULL	N/A
AUX BUSES	NULL	N/A	N/A	OFF	N/A	OFF	N/A	PRE 1
FX BUSES	NULL	MAINS	N/A	N/A	OFF	N/A	N/A	POST/R1/D1
AUX IN A/B	NULL	MAINS	N/A	N/A	N/A	N/A	NULL	N/A
SOLO BUS	NULL	N/A	N/A	N/A	N/A	N/A	N/A	AFL
MONITOR BUS	NULL	N/A	N/A	N/A	N/A	N/A	N/A	MAIN
TAPE IN	NULL	N/A	N/A	N/A	N/A	N/A	NULL	NULL

Each GEQ will be flattened and returned to its default Off state. The Fat Channel will be restored to the same setting for every input and output on the StudioLive. Each of the dynamics processors and the four EQ bands will be turned off.

Their parameters will be set as follows:

STUDIOLIVE 32.4.2AI AND 24.4.2AI FAT CHANNEL PARAMETERS												
HIPASS	NOISE GATE		LIMITER		COMPRESSOR		EQ		LOW	L.MID	H.MID	HIGH
OFF	KEY	N/A		OFF	THR	0 dB	P/S		PEAK	N/A	N/A	PEAK
	THR	0.05 s	THR	0 dB	RAT	2:1	Q		0.6	0.6	0.6	0.6
	RNG	N/A			ATK	20 ms	FREQ		130 Hz	320 Hz	1.4 kHz	5 kHz
	ATK	5 ms			REL	150 ms	GAIN		0 dB	0 dB	0 dB	0 dB
	REL	0.7 ms			GAIN	0 dB						
					KNEE	hard						

STUDIOLIVE 16.4.2AI FAT CHANNEL PARAMETERS											
HIPASS	NOISE GATE		LIMITER	COMPRESSOR		EQ	LOW	L.MID	H.MID	HIGH
OFF	THR	0.05 s	OFF	THR	0 dB	P/S	PEAK	N/A	N/A	PEAK
	REL	0.7 ms		RAT	2:1	Q	0.55	0.55	0.55	0.55
				ATK	20 ms	FREQ	130 Hz	320 Hz	1.4 kHz	5 kHz
				REL	150 ms	GAIN	0 dB	0 dB	0 dB	0 dB
				GAIN	0 dB					

Power User Tip: *Before beginning any new mixing situation, we recommend you recall the Zero Out (Board Reset) scene. This is the easiest way to ensure that there are no lingering parameter settings that could cause you some trouble in your new mix.*

5.3.2   Nulling Parameters

To return any Fat Channel parameter to its Zero Out scene setting, simply press and hold the Tap Tempo button and turn its encoder. For example, if you want to null the current setting for the Compressor Threshold on Channel 1:
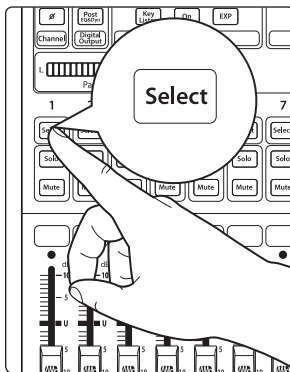
1.   Select Channel 1.
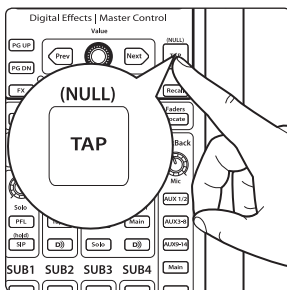
2.   Press and hold the Tap Tempo button.
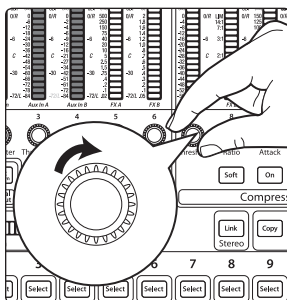
3.   Simultaneously turn the Compressor Threshold in the Fat Channel. The compressor threshold for Channel 1 will return to its Zero Out setting (0 dB).

5.3.3   Creating a Scene

Creating a scene requires simply dialing in a mix that you would like to use at a later date and saving it. This has obvious benefits for both studio and live sound. For example, in the studio, saving and recalling a scene allows you to move to another song or project and come back to the current mix later. For live shows with multiple bands, you can set up custom mixes for each band at soundcheck and recall the mix when that band goes onstage.

You also can save custom mixes for each venue that a band plays repeatedly.
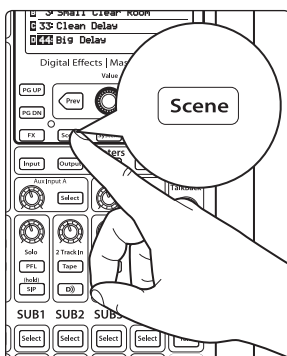
1.   To save a scene, press the Scene button and either page down to the second screen or press Store to automatically jump to this menu.
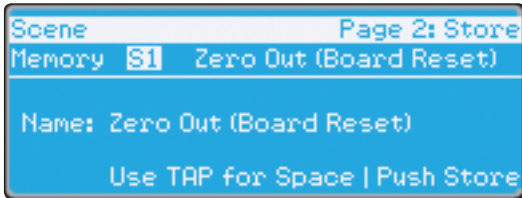
2. The memory locations will be selected. Use the Value encoder to scroll to a free location in the Scene library.
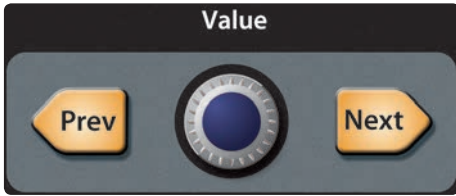
3. Now name your scene: Press the Next button to navigate to the first letter of the preset name and turn the Value encoder clockwise or counter-clockwise to change the letter.

The StudioLive allows you to customize the name with uppercase and lowercase letters and a selection of numerals and punctuation marks. Press the Tap button to quickly insert a space.
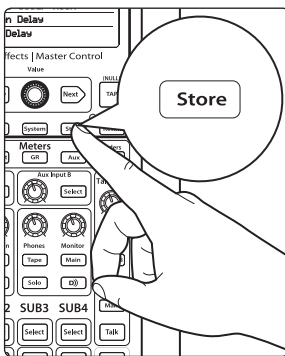
4. Continue this process until you are satisfied with your changes and then press the Store button. It will illuminate while the scene is being written to the StudioLive's internal memory. Once the scene is saved, the Store button will return to its unlit status.

5.3.4   Scene Recall
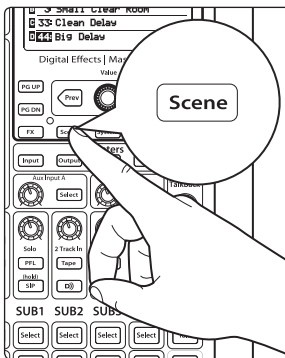
1. To recall a scene, press the Scene button and use the Value encoder to scroll through the Scene library.
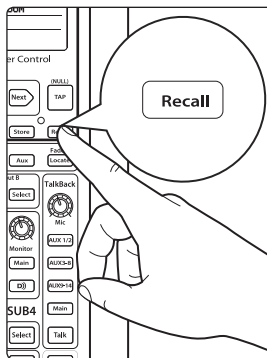
2. When you have found the scene you wish to recall, press the Recall button. By default, the StudioLive will recall all stored parameters (Fat Channel settings, channel muting and soloing, aux mixes, and internal effects parameters) except fader, knob, and graphic EQ positions.

3. If you do not wish to recall a certain set of parameters, simply use the Next and Prev buttons to navigate through the screen. When the parameter that you wish to disable is selected, turn the Value encoder counter-clockwise to move it to the No (off) position. Once you have disabled the parameters you do not wish to recall, press the Recall button.

The StudioLive's recallable parameters are grouped as follows:

Name: All Channel and Bus names as displayed in the Channel Info page, VSL-AI, SL Remote-AI, and QMix-AI.

Mute: All mute states. This includes input channels, subgroups, and FX buses.

FX: All parameters for the internal effects assigned to FXA, B, C, and D.

Assigns: All output and bus routing. This includes:

*   Channel and bus routing to mains and subgroups
*   Stereo linking for all channels and buses
*   Digital returns to inputs
*   Channel and bus solo states
*   Monitor-bus assignments
*   2 Track In assignments
*   Talkback assignments and talkback on/off

EQ and Dyn: All Fat Channel dynamics processing, filter parameters, and pan positions for every channel and bus.

Aux Mix: All aux mixes parameters including:

*   Channel sends to aux mixes
*   Channel sends to FXA - FXD
*   Pre1/Pre2/Post position for each aux and FX bus

Faders: All fader positions.

GEQ: Settings for all 16/12/8 graphic equalizers.

Pots: All digital knob positions:

*   Aux output levels
*   FXA- FXD levels
*   Solo level
*   2 Track In
*   Phones level
*   Monitor level
*   Talkback level (level, not trim control)
*   Aux Input A and Aux Input B level

Power User Tip: *If you enable knob positions (recalling group "pots") as a part of your scene recall, all of the digital knobs (Aux outputs, FXA-FXD levels, Solo, 2 Track In, Phones, Monitor, Talkback, Aux Input A and Aux Input B knobs) will remain at their stored position until they are moved manually. Once a knob is turned, its value will jump to the value of its current physical position. Therefore, if you intend to save a scene and recall the "pots" group later, it is highly recommended that you use the recall sheet provided in the back of this manual to note the stored position of these controls.*

It is important to note that the recall groups have no effect on what parameters are stored with a scene. All storable parameters are saved with a scene regardless of what recall groups are enabled.

Note: *The following StudioLive global parameters are not recallable:*

- Input Trim controls
- Output Trim controls
- Subgroup Delay settings
- LCD Contrast
- LCD Brightness
- Link ID
- Sample Rate

5.3.5   Fader Locate

If you enable fader positions as a part of your scene recall, the StudioLive will automatically put the meters in Fader Locate mode after you press the Recall button. The Fader Locate button will illuminate, and the meter section of the Fat Channel will display the recalled fader position.
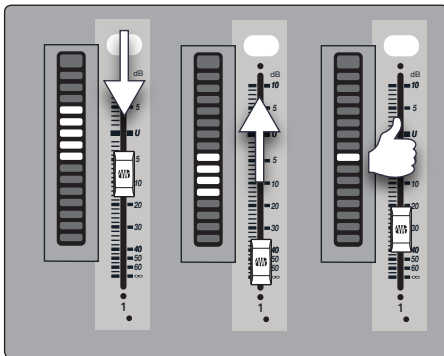
Move the faders up or down until only one LED is illuminated in each meter to recall the stored position. The subgroup and main meters at the top right of the StudioLive will display the recalled positions for their respective faders.

Power User Tip: *While in Fader Locate mode, the faders on your StudioLive will not be active until their current position matches their stored fader position. Once the stored position has been recalled on each fader, it will resume level control.*

5.3.6   Quick Scenes (StudioLive 32.4.2AI only)

The Quick Scene buttons in the Fat Channel allow you to create a scene without storing it to permanent memory and giving it a name. This is especially useful when mixing multiple bands that you might not ever mix again. You can also use this to have important stored scenes at your fingertips, which is great, for example, when mixing for live theater.

Power User Tip: *While the 24.4.2AI and 16.4.2AI do not have these controls available on their surfaces, this feature is available from within VSL-AI. See "VSL-AI: Overview Tab" in the StudioLive AI Software Library Reference Manual for more details.*

Creating a Quick Scene

To create a Quick Scene, dial in a mix you'd like to have access to later, or recall a scene that you want to have instantly available during a show.

1. Press the Scene Store button. All the Quick Scene buttons will start to flash.
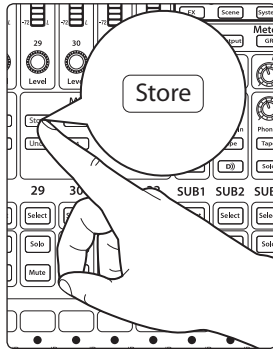
2. Press any of the six flashing buttons to store the scene to the button.
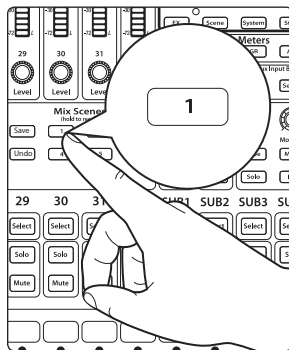
Recalling a Quick Scene

To recall a Quick Scene, press and hold a Quick Scene button. The scene stored to that position will be recalled according to the Recalling Filters. *See Section 5.3* for details.
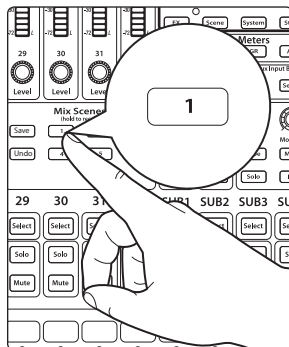
When you recall a Quick Scene, your StudioLive automatically stores a snapshot of the current state of the mixer. When you press the Undo button, your StudioLive will revert to this snapshot.

Note: *Until you recall a Quick Scene for the first time, the Undo button will recall the StudioLive's factory-default settings.*
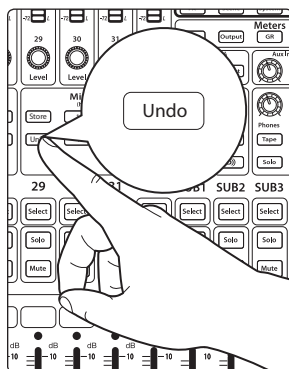
5.3.7 AutoStore

It is not necessary to create a scene for your StudioLive in order to preserve its settings when you power it down. The StudioLive takes continuous snapshots of the current position of every parameter on the mixer and stores them to memory every 3 seconds. This ensures that the next time you turn your StudioLive on, all of your settings will be restored to the same setting they were in when you powered down.

Note: *If you make changes to a stored scene, AutoStore will not save these changes as a permanent part of the scene. Any changes made to a scene in the StudioLive's library must be saved using the Scene menu, as described in the first part of this section.*

5.4 Graphic Equalizers

The StudioLive features a 31-band graphic EQ for each of the analog aux buses and a stereo, 31-band graphic EQ for the Main bus. A graphic EQ is a multiband equalizer that uses physical or virtual sliders to adjust the amplitude for each frequency band. It gets its name from the positions of the sliders, which graphically display the resulting frequency-response curve. While the GEQ menu is active, the encoders in the Fat Channel are used to make amplitude adjustments, and the meter LEDs display the "slider" positions. The center frequency and bandwidth are fixed; the level (amplitude) for each band is the only adjustable parameter.

Graphic EQs are generally used to fine-tune the overall mix for a particular room. For instance, if you are mixing in a "dead" room, you may want to boost high frequencies and roll off some of the lows. If you are mixing in a "live" room, you might need to lower the high-midrange and highest frequencies. In general, you should not make drastic amplitude adjustments to any particular frequency band. Instead, make smaller, incremental adjustments over a wider spectrum to round out your final mix. To assist you with these adjustments, here is an overview of which frequencies affect different sound characteristics:

Sub-Bass (16 Hz to 60 Hz). The lowest of these bass frequencies are felt, rather than heard, as with freeway rumbling or an earthquake. These frequencies give your mix a sense of power, even when they only occur occasionally. However, overemphasizing frequencies in this range will result in a muddy mix.

Bass (60 Hz to 250 Hz). Because this range contains the fundamental notes of the rhythm section, any EQ changes will affect the balance of your mix, making it fat or thin. Too much emphasis will make for a boomy mix.

Low Mids (250 Hz to 2 kHz). In general, you will want to emphasize the lower portion of this range and de-emphasize the upper portion. Boosting the range from 250 Hz to 500 Hz will accent ambience in the studio and will add clarity to bass and lower frequency instruments. The range between 500 Hz and 2 kHz can make midrange instruments (guitar, snare, saxophone, etc.) "honky," and too much boost between 1 kHz and 2 kHz can make your mix sound thin or "tinny."

High Mids (2 kHz to 4 kHz). The attack portion of percussive and rhythm instruments occurs in this range. High mids are also responsible for the projection of midrange instruments.

Presence (4 kHz to 6 kHz). This frequency range is partly responsible for the clarity of a mix and provides a measure of control over the perception of distance. If you boost this frequency range, the mix will be perceived as closer to the listener. Attenuating around 5 kHz will make the mix sound further away but also more transparent.

Brilliance (6 kHz to 16 kHz). While this range controls the brilliance and clarity of your mix, boosting it too much can cause some clipping so keep an eye on your main meter.

5.4.1   The GEQ Menu

Each GEQ on your StudioLive is assigned to a specific bus: Main Left, Main Right, and Auxes 1 through 14/10/6. The bus assignment cannot be changed. When the GEQ menu is active, the meters and encoders of the Fat Channel become the controls for the graphic EQ. As you touch a knob, you will notice that its band number, frequency, and gain are displayed in the System menu. The 31 bands range from 20 Hz to 20 kHz.

The frequency for each band is fixed:

StudioLive 32.4.2AI: Bands 1 through 31 are controlled by encoders 1-through 31, respectively.

StudioLive 24.4.2AI: Bands 5 through 28 are controlled by encoders 1-24, respectively. When Band 4 or 29 is selected in the Show Band field in the GEQ menu, the meters will flip.

StudioLive 16.4.2AI: Bands 9 through 24 are controlled by encoders 1-16, respectively. When Band 8 or 25 is selected in the Show Band field in the GEQ menu, the meters will flip. Notice that all meters have one LED illuminated to display the current gain position for each band, and the meter for the selected band in the Show Band field is inverted, meaning that all LEDs will be illuminated except the LED displaying the current gain position for that band. The band does not have to be selected in the Show Band field for its encoder to be active. All encoders are active so you can make changes to all 31 bands at one time.

To open the GEQ menu, press the GEQ button.
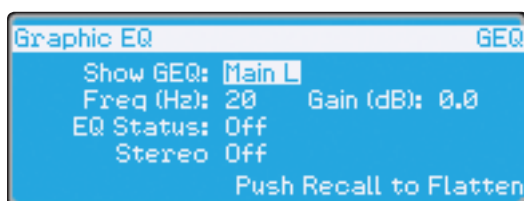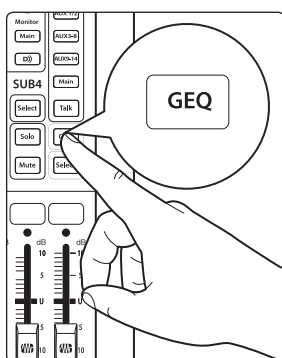
Navigate to the Show GEQ field and use the Value knob to scroll through the different graphic EQs. Use the Next and Prev buttons to navigate through each GEQ screen and use the Value knob to change a parameter.
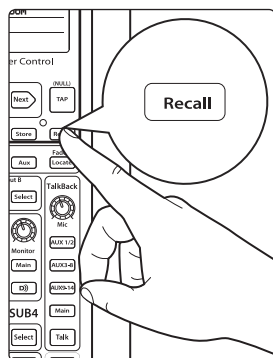
To flatten a GEQ curve, press the Recall button.

5.4.2   Saving and Loading Graphic EQ Presets
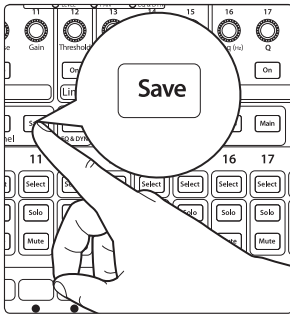
Like all other parameters on the StudioLive, graphic EQ settings can be stored and recalled. If you have created a graphic EQ setting that you would like to save to the GEQ Preset library, press the Save button in the Fat Channel while that graphic EQ is active.

Notice that the LCD will display the GEQ Save menu. To continue:
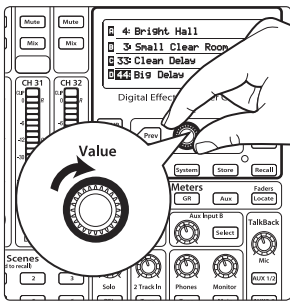
1. Use the Value encoder to scroll to an empty position in the GEQ Preset library.
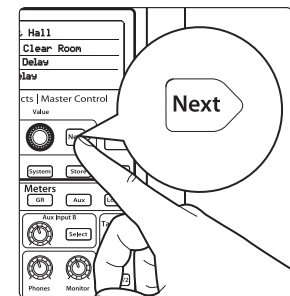
2. Press the Next button again to navigate to the first letter of the preset name.
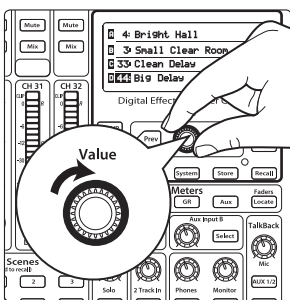
3. Turn the Value encoder clockwise or counter-clockwise to change the letter. The StudioLive allows you to customize the name with uppercase and lowercase letters, as well as a selection of numerals and punctuation marks. You can insert a space by simply pressing the Tap button.
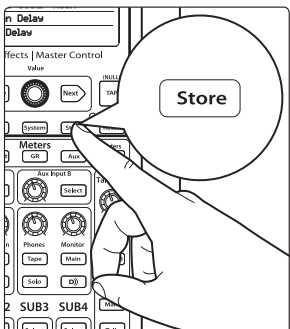
4. Once you are satisfied with your changes, press the Store button. It will illuminate while the GEQ preset is being written to the StudioLive's internal memory. Once the GEQ preset is saved, the Store button will return to its unlit state.
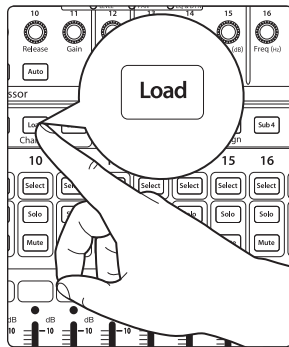
To load a preset to any channel on the StudioLive, first press the GEQ button and page down to the desired GEQ. From the Fat Channel, press the Load button.
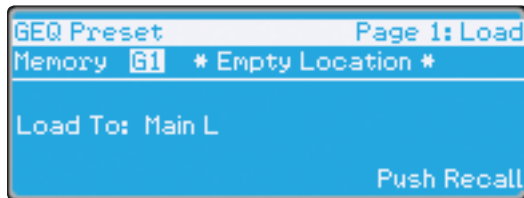
You will notice that the LCD now displays the GEQ Load menu. Use the Value encoder to locate the preset you would like to use.
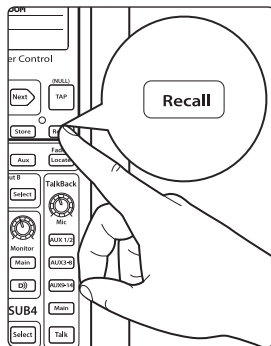
Once you have made your selection, press the Recall button. If at any time you would like to cancel this operation, simply press the Load button again.

5.5   System Menu

In most digital-console manuals, the phrase "System menu" inspires a sense of dread. However, with the StudioLive, you have nothing to fear. The System menu on the StudioLive serves just a few simple functions, most of which have little to do with mixing a show.

Like the Scene and FX Menus, the StudioLive remembers on which page of the System menu you were when you navigated away. To quickly jump to page 1, simply press and hold the Tap Tempo button.

Press the System button to access the System menu.
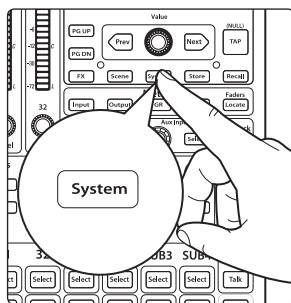
Global

The first page allows you to adjust the LCD Contrast and LCD Backlight for optimal viewing in your working environment. These settings are retained when the unit is powered down.

From the Global page, you can also change the Solo mode. For more information on this feature, *please review Section 4.9*.

As previously mentioned, your StudioLive offers both Peak and Peak Hold metering. To view the Peak Hold, simply turn this preference On. The Global page also allows you to control where your channels are silenced when their mute buttons are engaged. By default, channels are set to mute globally. Setting Global Aux Mute to "No" will allow you to mute your channels only in the Subgroup and Main buses. Aux and FX buses will follow the send position setting (i.e. Post fader sends will mute in the bus, Pre1/Pre2 sends will not).

Network
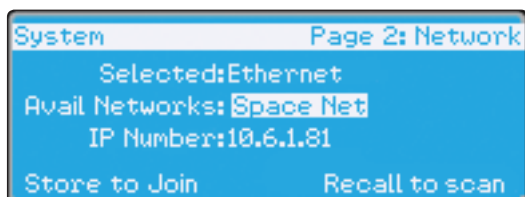
Press the Pg Dn button to access the Network page. The StudioLive AI-series mixers ship with a USB wireless adapter that allows you to connect to any Wi-Fi network. To connect to your wireless network, press the Recall button to scan for available networks then use the Value encoder to scroll through the available network names. When you have located your network, press the Store button to join. This will open the Join Network page where you can navigate to the password field and follow the same steps you used to create a preset or scene name *(see Section 4.1.10)* to enter the password. When you are done, press the Store button. This will store the network information in your StudioLive and make the network the default to which your StudioLive will try to connect.

You do not need to enter the network name if you are hardwiring your StudioLive to the wireless network using an Ethernet cable.

For more information on wireless and wired networks and using a laptop, iPad, or iPhone/iPod touch to remote control a StudioLive as well as a complete illustrated tutorial on connecting your StudioLive mixer to a LAN network, *please review "Networking your StudioLive AI Mixer" in the StudioLive AI-Series Software Library Manual*.

Note: *Only the wireless adapter that was included with your StudioLive AI is supported. This adapter must be connected to the USB port on the top of your StudioLive before you power on your mixer for wireless remote control to function.*

iOS Devices

```
System              Page 3: iOS Setup
Device: My iPad
Permissions:
              Mix:FOH
              Other:none
```
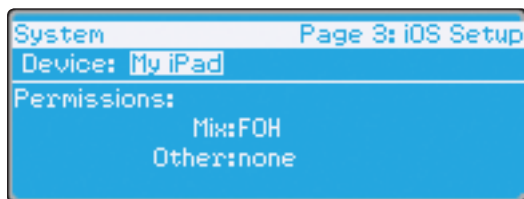
Your StudioLive AI mixer can be wirelessly remote-controlled with an iPad or iPhone/iPod touch. While this allows you to move about the venue freely, it can also put the full power of the StudioLive in multiple hands — some more adept than others. Therefore, your StudioLive enables you to limit each iOS device's access to the mixer features by setting permissions.

Once an iOS device is connected to your wireless network and has launched SL Remote-AI or QMix-AI, the device will be displayed in the Devices list on the iOS Setup page in the System menu. Each device will be listed using its device name, so you can easily identify which device is which. This name can be changed in iTunes or in the General>About settings on the iOS device.

Once you have connected and configured an iOS device, the same permissions will be set for that device every time you connect it. *Complete information about SL Remote-AI and QMix-AI can be found in the StudioLive AI-Series Software Library Manual*.

To set device permissions, use the Value Encoder to scroll through the list of available devices. When setting permissions for SL Remote-AI users, you will choose between giving full access to all SL Remote-AI functions or providing limited access to just a few aux-mix functions. In most cases, one iPad will be configured to control front-of-house (FOH), and the others will be configured to control aux mixes.

When setting permissions for QMix-AI users, you will choose between providing full access to all aux mixes and providing access to only a single aux mix. You can also limit the user to just the Wheel of Me functions.

To remove all control from any iOS device user, simply select "None" from the Mix field.

Aux Pre

```
System                Page 4: Aux Pre
Aux 1:Pre1  2:Pre1  3:Pre1   4:Pre1
Aux 5:Pre1  6:Pre1  7:Pre1   8:Pre1
Aux 9:Pre1 10:Pre1 11:Pre1  12:Pre1
Aux 13:Pre1 14:Pre1
      HP-Gate |Pre1| Comp-EQ-Lim |Pre2|
```
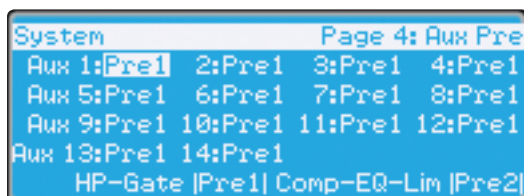
Pages 4 and 5 allow you to select the send position for every channel to a particular bus. From Page 4: Aux Pre, you can change the send position for Aux Buses 1-14/10/6. By default, All Aux Buses are set to Pre 1, which puts the send position for each channel after the polarity invert, high-pass filter and gate but before the compressor, EQ, and limiter. For more information on the three send positions, *please review Section 4.4.4*.

FX Pre

```
System                 Page 5: FX Pre

FX A:Post  B:Post   C:Post   D:Post


      HP-Gate |Pre1| Comp-EQ-Lim |Pre2|
```
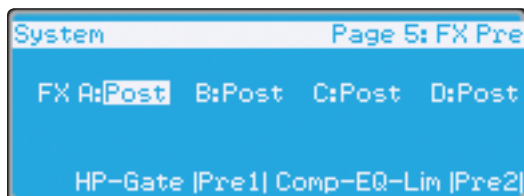
From Page 5: FX Pre, you can change the send position for FX Buses A through D.

By default FX Buses A through D are set to Post, which puts the send position for each channel after the fader. For more information on the three send positions, *please review Section 4.4.4*.

Digital Information

```
System                 Page 6: Digital
     Sampling Rate: 48 kHz
     S/PDIF Out: Main Mix L/R
     Link ID: Master
     Subgroup Merge: No
Status: No Driver - Unlinked
```
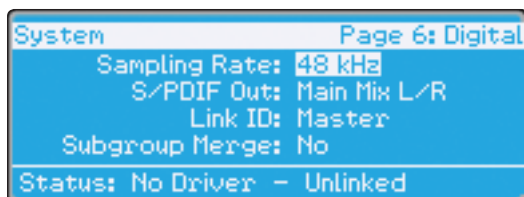
Press the Page Down button to access the Digital Information page. From here, you can change the sample rate, view your computer connectivity, configure cascaded mixers, and route a stereo mix to the S/PDIF output. Any bus mix, Aux In A, Aux In B, or the Tape Input can be routed to the S/PDIF output.

Power User Tip: To ensure the safety of the audio equipment connected to it, the StudioLive will mute all post-converter outputs for two seconds when the sample rate is changed. This includes the main and the control-room outputs, as well as the

aux and subgroup outputs. While this offers a good measure of protection to your sound system, it could put the brakes on a live show. Therefore, it very important that the sample rate be selected and locked prior to beginning any recording or performance.

When either 88.2 or 96 kHz is selected, your StudioLive AI mixer will enter HD Mode. While in HD Mode, the following functions will be disabled:

- **Output Bus Fat Channel Processing.** Fat Channel processing on the Main, Aux, Subgroup, and FX buses.

- **FX B and FX D Buses and Processors.** Both the bus and processor will be disabled.

- **Cascading.** Mixers in HD Mode cannot be cascaded.

When the StudioLive is connected and synced to a computer, the status will read "Driver On." (**See "Connecting to a Computer" in the StudioLive AI-Series Software Library Manual** for details on using your StudioLive as an audio interface.)

When two StudioLives are cascaded together, the status will read "Linked," and you willl be able to see which StudioLive is the Master unit. For more information on using multiple StudioLive mixers, please review **Section 5.6.**
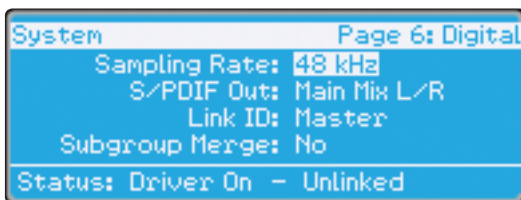
Sub Out Delays
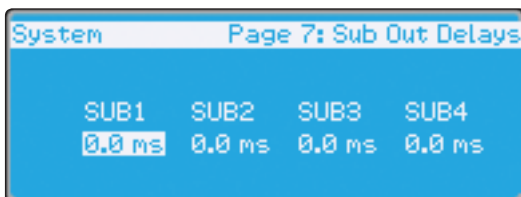
On page 7 of the System menu, you will find the Sub Out Delays. The StudioLive provides the ability to delay the audio from each subgroup output by 0.5 to 300 ms. You can adjust the delay in 0.5 ms increments.

When you correctly set the delay time for these outputs, you ensure that the sound from each speaker in the P.A. system arrives at the listening position at the same time. The delay should be set for the speakers to which your listener will be closest.

Let's say you are using a StudioLive in a large theater with a balcony, and you have three pairs of speakers: a pair in front of the stage, a pair in the rear of the auditorium, and a pair at the front of the balcony. You will need to delay the rear speakers so that the listeners closest to them will hear the audio from both the rear and stage speakers at the same time. This is also true for the audience in the balcony. While the balcony speakers will provide the loudest source of audio, the balcony audience will still be able to hear the stage speakers, so a delay must be set for the balcony speakers. You will need to calculate approximately 1.1 ms of delay time per foot. So if the balcony speakers in the above example are 61 feet from the stage speakers, set their delay time to 67 ms.

Speaker delay can also be used to correct off-axis phasing issues in small clubs. Because of space restrictions, you can't always place your main speakers for the best possible sound reproduction. Sometimes your left-side speaker will need to be closer to the audience to make room for a pillar or a staircase or the bathroom, so an audience member standing in the center of the room will not have the best listening experience. By using a pair of subgroups as the source for your mains, you can factor in that 2 to 4 ms delay the left speaker needs so that unbalanced speaker placement is no longer an issue.

Power User Tip: *Virtual StudioLive-AI features the Smaart® System Delay wizard, which is designed to calculate and set this delay automatically by analyzing your main and delay systems with a measurement microphone. For more information on this powerful feature, as well as additional information on configuring a delay system,* ***please review "Smaart System Delay Wizard" in the StudioLive AI-Series Software Library Manual.***

Lockout Mode

The StudioLive features a Lockout mode that allows you to create a password and lock the controls. This is especially useful in situations where several people will be running sound but only one or two are knowledgeable enough to set up dynamics processing and the like.

Right out of the box, the StudioLive cannot be locked, so don't worry about hitting the wrong button. To enable the Lockout feature, you must first connect and sync your StudioLive to a computer. *Please review "Enabling Lockout Mode" in the StudioLive AI-Series Software Library Manual* for more information on this feature.

Firmware Version
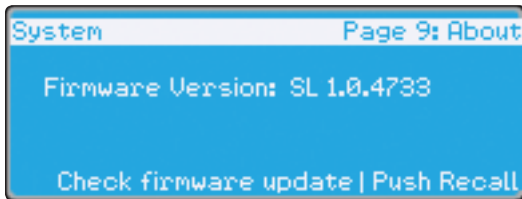
The last page of the System menu displays the firmware version currently loaded on your StudioLive. From this page, you can update your mixer directly from a USB thumb drive. Please *see Section 5.5.1* for complete firmware update instructions.

Power User Tip: StudioLive mixers running firmware versions 1.0.5203 and later can be updated by simply connecting to a network with Internet access. Navigate to the last page of the System Menu and press Recall to update your firmware directly from the Internet.

5.5.1 Firmware Updates

Your StudioLive mixer can be updated by simply connecting a USB thumb drive loaded with the firmware files. Mixers running firmware version 1.0.5203 or later can directly download firmware updates over any network with Internet access. Firmware updates are available through your MyPreSonus account on the PreSonus website. You will need to register your mixer to download firmware files, VSL-AI, and Capture.

1. Log into your MyPreSonus account.
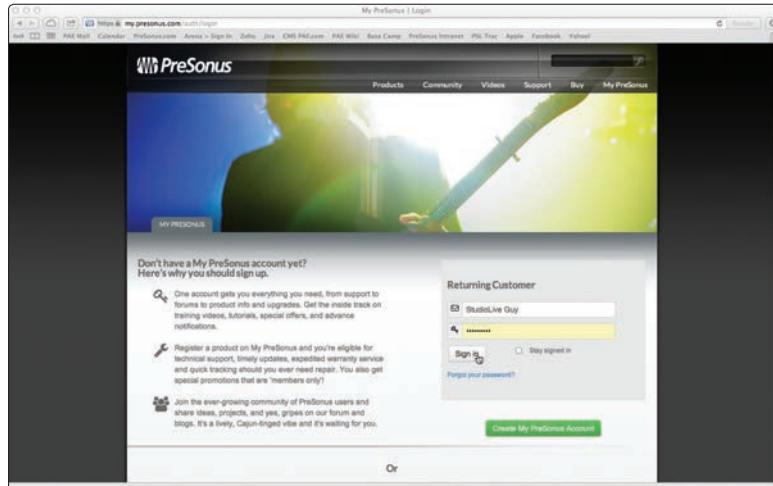
2. Click on your mixer registration in the My Hardware section and download the firmware files for your mixer.
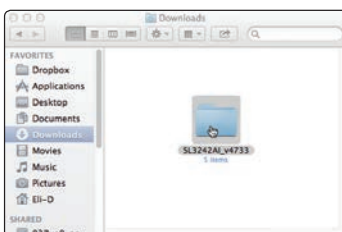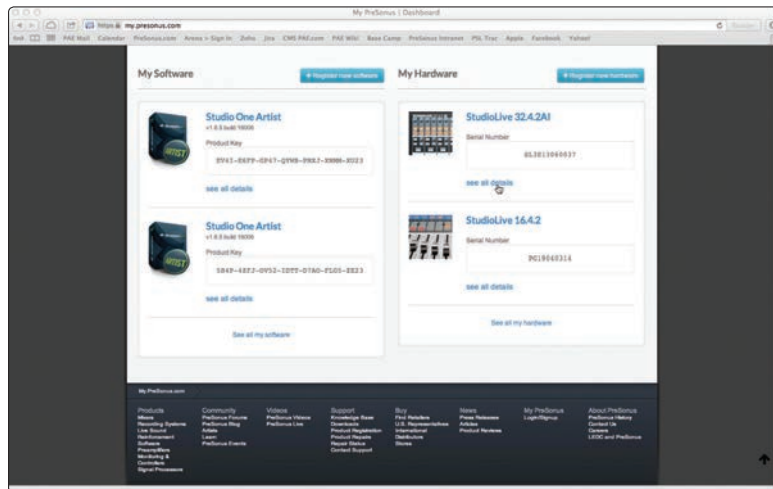
3. Open your downloads folder and locate the StudioLive firmware folder. The folder will be named with your mixer's model number and the new firmware version (e.g., SL3242AI_v4733).
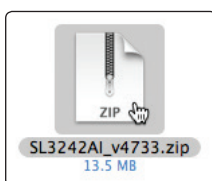
4. Depending on your system preferences, the firmware folder may not automatically uncompressed from the .zip file. To expand it, simply double-click it.
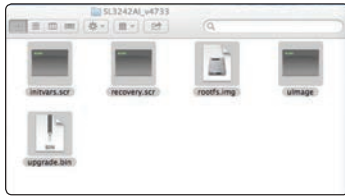
5.   Open the firmware folder; you will find five files inside:

• Initvars.scr

• Recovery.scr

• Rootfs.img

• uImage

• upgrade.bin

You will need all five files to upgrade your mixer.
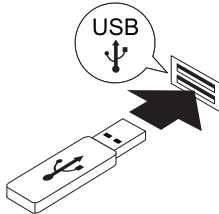
6.   Connect a FAT32-formatted USB thumb drive to your computer.

Power User Tip: Most small USB drives (16 GB or less) are already formatted correctly, so no formatting is required. If your computer does not detect your thumb drive, verify that it is formatted correctly.

7.   Select all five firmware files and either copy/paste or drag them to the root of your thumb drive. Once they have copied over, eject your drive and disconnect it from your computer.
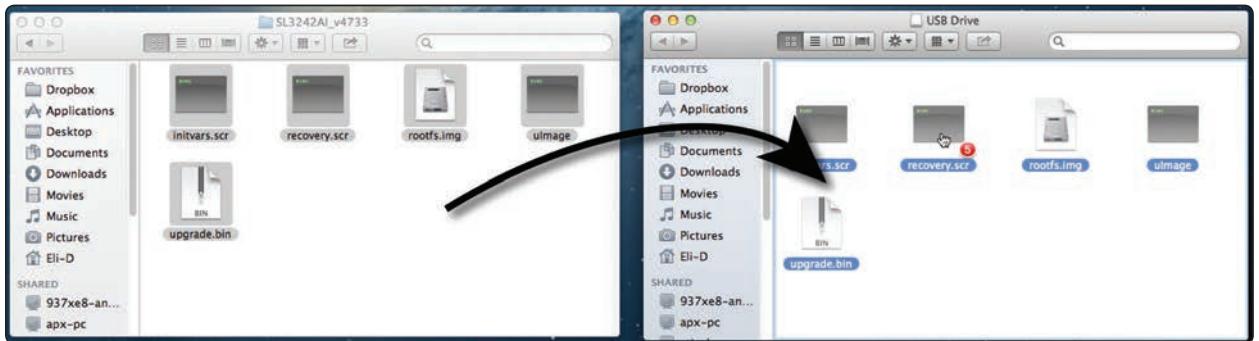
8.   Power on your StudioLive AI mixer.
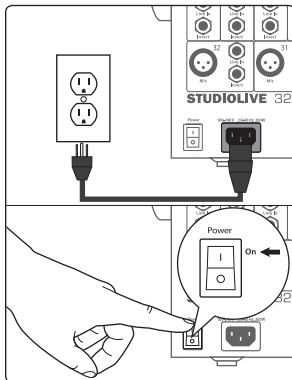
9.   Remove the Wi-Fi LAN adapter (if connected).
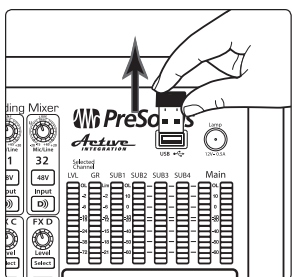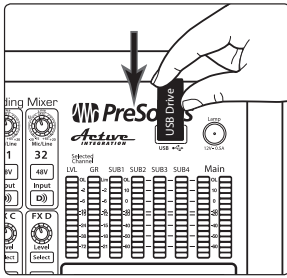
10. Plug in the FAT32-formatted USB thumb drive.

11. It will take approximately five seconds for your mixer to detect your thumb drive. Count to five slowly before proceeding to the next step.
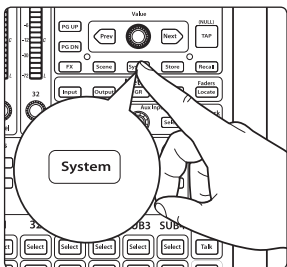
12. Press the System button.
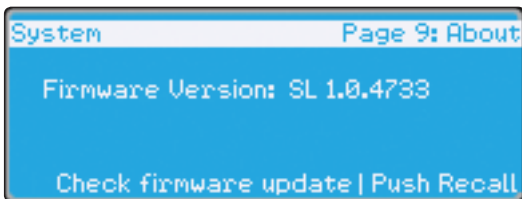
13. Press the Pg Dn button until you reach the last page of the System Menu (Page 9).
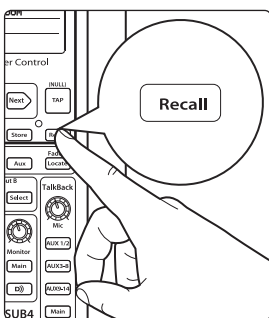
14. Press the Recall button.
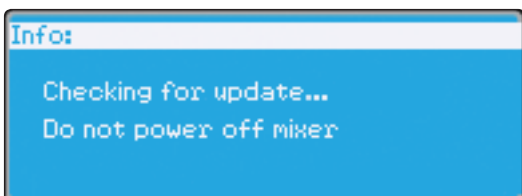
15. You will be instructed that the mixer is reading the firmware update files. Warning: Do not power off the mixer. The update will take up to 30 seconds.
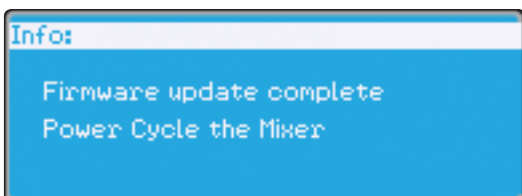
16. When the update is complete, you will be prompted to restart the mixer.
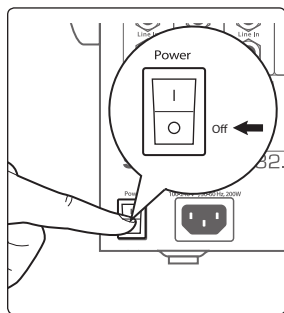
17. Power down your mixer.
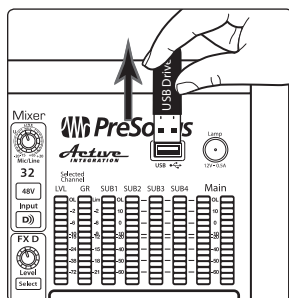
18. Remove the thumb drive.
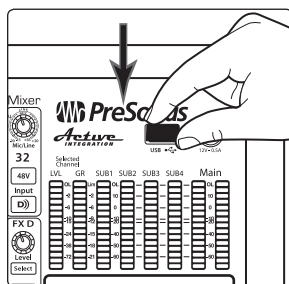
19. Insert your USB LAN adapter (if you want to connect wirelessly to your router) and power your mixer back on. Note: Be sure that your speakers are powered off, as there will be a small pop when the mixer reboots.

5.6   Cascading Mixers

Any two StudioLive AI-series mixers can be cascaded, using a FireWire 800 cable, to create a single large-format console with full recording capability, playback, and remote control. From a functional standpoint, this will create a single mixer with the following configurations:

Mixer Combination	Total Channels	Total Aux buses	Total FX buses	Total Subgroups	Recording Interface
2 x 16.4.2AI	40 (32 channels + 4 stereo aux inputs)	6	8 (local sends only)	4 or 8 (depending on subgroup merge state)	48 x 34
2 x 24.4.2AI	56 (48 channels + 4 stereo aux inputs)	10	8 (local sends only)	4 or 8 (depending on subgroup merge state)	64 x 50
2 x 32.4.2AI	72 (64 channels + 4 stereo aux inputs)	14	8 (local sends only)	4 or 8 (depending on subgroup merge state)	80 x 66
16.4.2AI (Slave) 24.4.2AI (Master)	48 (40 channels + 4 stereo aux inputs)	6 global sends 4 additional local sends (24.4.2AI only)	8 (local sends only)	4 or 8 (depending on subgroup merge state)	56 x 42
16.4.2AI (Slave) 32.4.2AI (Master)	56 (48 channels + 4 stereo aux inputs)	6 global sends 8 additional local sends (32.4.2AI only)	8 (local sends only)	4 or 8 (depending on subgroup merge state)	64 x 50
24.4.2AI (Slave) 32.4.2AI (Master)	64 (56 channels + 4 stereo aux inputs)	10 global sends 4 additional local sends (32.4.2AI only)	8 (local sends only)	4 or 8 (depending on subgroup merge state)	72 x 58

This section will guide you through the syncing process and explain how two StudioLive AI mixers cascade to function as a single larger mixer.

Power User Tip: When recording with your cascaded mixer system, you must be connected with a FireWire 800 connection, or Thunderbolt connection using an FireWire 800-to-Thunderbolt adapter. FireWire 400 does not provide the bandwidth necessary to support a cascaded system.

5.6.1   Configuring Multiple Units

1. To cascade two StudioLive AI mixers to create a standalone system, connect a FireWire 800 cable from the first unit to the second unit.
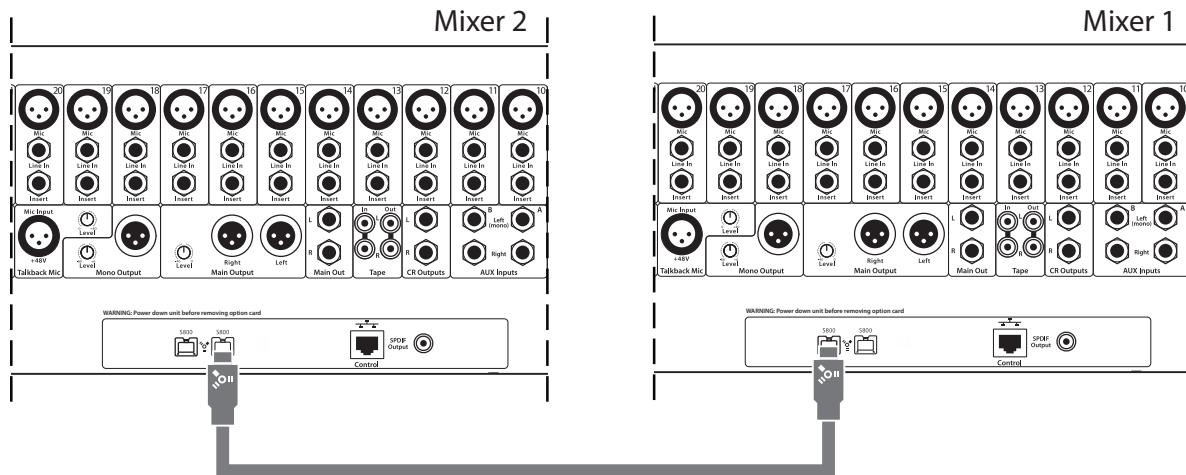
2. Press the System button on the unit on the left and page down to Page 6: Digital.
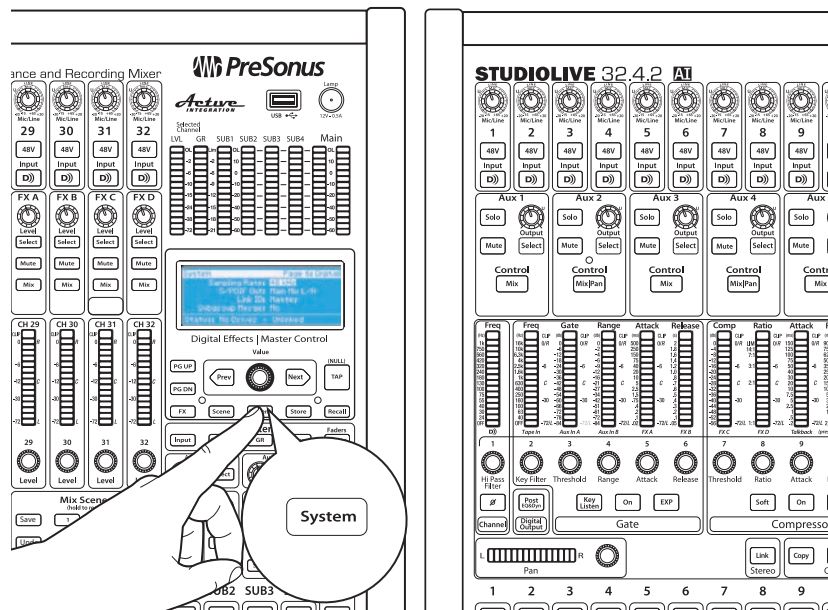
3. Use the Next button to move to the Link ID field.
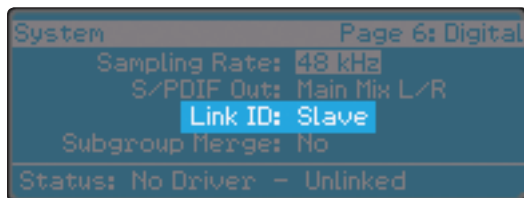
4.   Use the value encoder to set the ID to "Slave."

5.   Press the System button on the right unit and page down to Page 6: Digital.
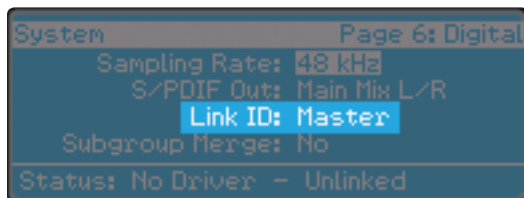
6.   Verify that the Link ID is set to "Master."

Power User Tip: Either unit in the chain can be designated as the Master. The unit designated "Slave" will merge all of its buses with the Master. In addition, when recording with a cascaded system, the Slave mixer's channels will be the first channels. For example, if you cascade two 16.4.2AI mixers, the Slave 16.4.2AI will provide channels 1-16 in your DAW, and the Master 16.4.2AI will provide channels 17-32. Because all channels will come from the Master unit's Main output, aux sends, and subgroups (optional), we suggest that the unit to the right be designated as the Master. If you are cascading mixers of different frame sizes, the smaller mixer must be designated as the Slave.

5.6.2   Aux Mixing with Cascaded Mixers

Every channel in the mixer chain can be sent to the aux outputs on the Master unit. When you press the Mix button on any of the auxes on either mixer, you will notice that the Mix button for that aux illuminates on both mixers in the chain. For example, if you press the Mix button for Aux 1 on the Master unit, the Mix button for Aux 1 on the Slave unit will also illuminate.

Creating an aux mix with two mixers works exactly the same way as with one mixer. Each encoder beneath the meters in the Fat Channel controls the send level to the enabled aux for the corresponding channel on that mixer.

To add Fat Channel processing to the aux mix, you must select the aux bus on the Master unit. The Fat Channel on the Slave unit will be disabled.

Example 1: Two StudioLive 24.4.2AIs

If you are cascading two StudioLive 24.4.2AIs, Channels 1 through 24 will reside on the Slave, and Channels 25 through 48 will reside on the Master unit.

Let's say that you want to create an aux mix on Aux 1:

1.   To begin, press the Aux 1 Mix button on either mixer.

2.   The Fat Channel meters and encoders on both mixers will be ready for you to create an Aux 1 mix.

3.   Use the Fat Channel meters and encoders on the Slave to set the Aux 1 send levels for Channels 1-24 and the meters and encoders on the Master unit to set the Aux 1 send levels for Channels 25-48.

4.   The resulting mix is then routed from the Aux 1 output on the Master unit.

5.   If you would like to add Fat Channel dynamics to the overall Aux 1 mix, simply press the Aux 1 Select button on the Master unit and use the Master unit's Fat Channel to dial in your dynamics and EQ settings.

Example 2: One StudioLive 16.4.2AI and one StudioLive 32.4.2AI

If you chain a StudioLive 16.4.2AI mixer to a StudioLive 32.4.2AI, Channels 1-16 will reside on the StudioLive 16.4.2AI (Slave), and Channels 17-48 will reside on the 32.4.2AI (Master).

Let's say that you want to create an aux mix on Aux 3:

1. To begin, press the Aux 3 Mix button on either mixer. The Fat Channel meters and encoders on both mixers will be ready for you to create an Aux 3 mix.

2. Use the Fat Channel meters and encoders on the 16.4.2AI to set the Aux 3 send levels for Channels 1-16 and the meters and encoders on the 32.4.2AI to set the Aux 3 send levels for Channels 17-48.

3. The resulting mix is then routed from the Aux 3 output on the 32.4.2AI unit.

4. If you would like to add Fat Channel dynamics to the overall Aux 3 mix, simply press the Aux 3 Select button on the Master unit and use the Master unit's Fat Channel to dial in your dynamics and EQ settings.

Please note: Because the Slave mixer merges its buses with those on the Master mixer, you will only be able to create an aux mix using all 48 channels on Auxes 1-6, as described in the above example. Aux mixes 7-14 can still be used to create mixes from from the local channels (Channels 17-48) on the 32.4.2AI.

5.6.3   Internal Effects Buses

Unlike the aux buses, the two internal effects buses on each mixer are independent. Using Example 1 from the previous section, Channels 1-24 can only be routed to FXA through FXD on the Slave mixer, and Channels 25-48 are processed using the Master mixer's four internal effects buses. The advantage is that you get twice the number effects buses!

Of course, if you'd like to send all channels to the same effect, you can simply load the same effect on both mixers. But with some careful patching, you can take advantage of the extra effects buses at you disposal. The internal effects buses on each mixer can be assigned to the Master unit's main output, or to a subgroup, as usual. Simply select the effects bus and press the desired assignment button in the Fat Channel.

5.6.4   Subgroups: To Merge or Not to Merge

On Page 6: Digital, in the System menu, you will find the Subgroup Merge field. This option allows you to merge the subgroups on the Slave mixer with those on the Master, giving you four total subgroups; or to leave the subgroups on the Slave mixer unmerged with those on the Master mixer, giving your eight total subgroups.

When Subgroup Merge is enabled, the subgroup faders on the Master mixer will control the entire group, and dynamics processing can be added to the entire mix using the Master's Fat Channel.

If Subgroup Merge is set to "No," each of the four subgroups on both mixers are locally controlled, so the channels on the Slave mixer can only be assigned to the subgroups on the Slave, and the channels on the Master mixer can only be assigned to subgroups on the Master. Because the subgroups on each of the mixers can still be routed to the Main output on the Master unit, unmerging the subgroups offers advantages in any situation where you want group channels and will use your subgroup as a master fader, as with a drum group, or if you want to use your subgroup faders as dedicated effects returns.

5.6.5 Presets, Scenes, and Fat Channel Copy

Presets

Channel strip and effects presets are stored and recalled locally on each mixer.

Scenes

Mix scenes for cascaded mixers must be saved and recalled on the Master mixer. This will store and recall the settings for the complete mix for your cascaded StudioLive AI system.

Power User Tip: Scenes can still be stored and recalled on the Slave mixer but this will only store and recall the local settings for Slave mix.

Fat Channel Copy

Channel-strip settings from any channel or bus on either mixer in the chain can be copied to any other channel or bus on the other mixer in the chain.

5.6.6 Master Buses and Inputs

The following inputs and buses can only be accessed from the Master mixer:

Talkback: The talkback microphone on the Master unit's Master section is the only talkback mic that can be routed to the aux mixes on the Master unit and to its main outputs. The Talkback microphone and its controls are disabled on the Slave mixer.

Monitor Bus: You must use the Monitor section on the Master unit to monitor (by listening to the Solo bus or the Main bus) all channels in the chain, the Tape input for the Master mixer, and the Main digital return. The Monitor bus on the Slave unit is disabled.

Solo Bus: You must use the Solo bus controls on the Master unit. The Solo bus controls (Level, PFL, SIP) on the Slave mixer are disabled.

Subgroups (Subgroup Merge On): When Subgroup Merge is set to On, the Subgroup controls (Level, Mute, Solo, Select) on the Slave mixer are disabled.

Aux Bus: The Aux Bus EQ and dynamics processing and the output level must be controlled from the Master mixer. While cascaded, the Fat Channel on the Slave mixer will go dark when an Aux bus is selected.

GEQ: The graphic EQ controls must be accessed from the Master mixer. The GEQ button is disabled on the Slave mixer while cascaded.

Mute Groups (32.4.2AI): The Mute Group controls on the Master mixer creates and recalls mute groups for the entire cascaded mixer system.

Quick Scenes (32.4.2AI): The Quick Scene controls on the Master mixer creates and recalls mix scenes for the entire cascaded mixer system.

6   Resources

6.1   Stereo Microphone Placement

The following are a few recording applications to help you get started with your StudioLive. These are by no means the only ways to record these instruments. Microphone selection and placement is an art. For more information, visit your library or local bookstore, as there are many books and magazines about recording techniques. The Internet is also a great source of recording information, as are instructional videos. Some of these microphone-placement suggestions can be used in live applications, as well as for studio recording.

Grand Piano
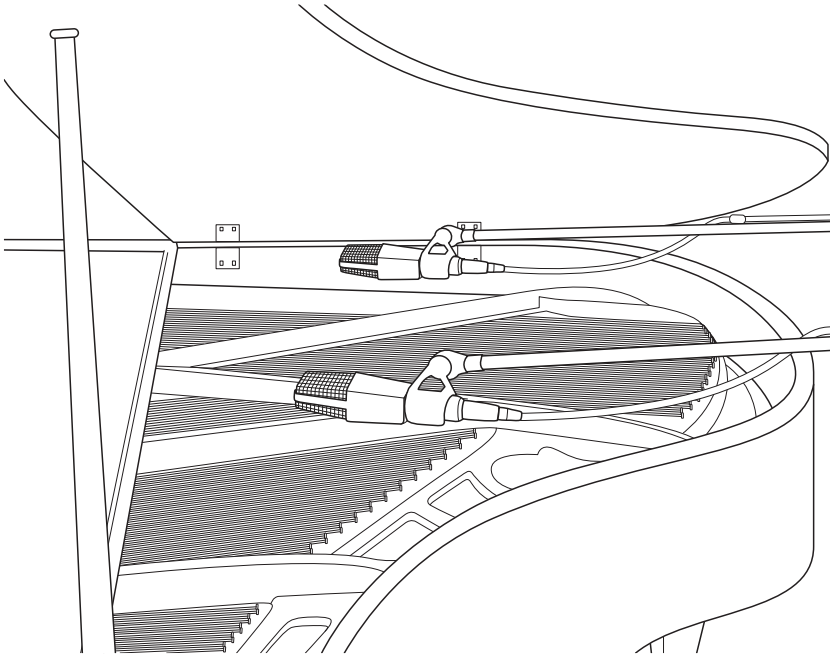
Place one microphone above the high strings and one microphone above the low strings. Experiment with distance (the farther back the more room you will capture). This technique can be used for live and studio applications.

Electric Guitar
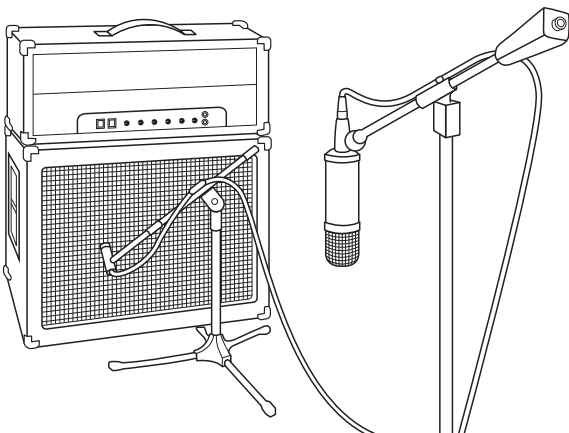
Place a dynamic microphone an inch or two away from the speaker of the guitar amplifier. Experiment with exact location. If you are recording an amp with multiple speakers, experiment with each one to see if one sounds better than the others. Place a condenser microphone approximately six feet away, pointed at the amp. Experiment with distance. Also experiment with inverting the phase of the room microphone to check for phase cancellation and reinforcement. (Select the "fuller"-sounding position.) To use this technique in a live application, omit the condenser microphone.

Acoustic Guitar

Point a small-diaphragm condenser microphone at the 12th fret, approximately 8 inches away. Point a large-diaphragm condenser microphone at the bridge of the guitar, approximately 12 inches from the guitar. Experiment with distances and microphone placement. Another popular method is using an XY microphone placement with two small-diaphragm condenser microphones. (See drum-overheads picture on the next page.)

Bass Guitar (Direct and Speaker)
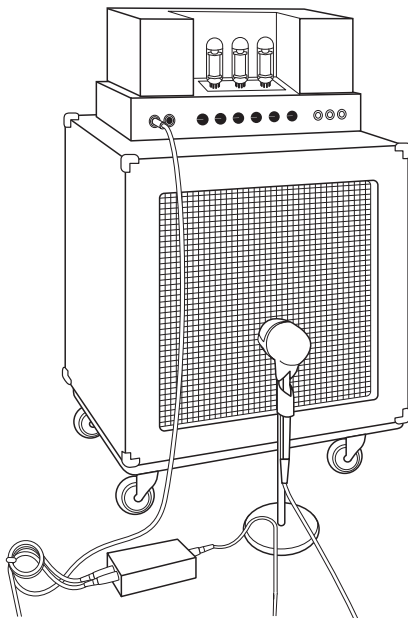
Plug the electric bass guitar into a passive direct box. Connect the instrument output from the passive direct box to a bass amplifier. Place a dynamic microphone an inch or two away from the speaker and connect it to a StudioLive microphone input. Connect the line output from the passive direct box to a line input on a different channel of the StudioLive. For recording, place these signals on separate tracks. During mixing, you can blend the direct and amplifier signal to taste. This technique can also be used in live applications.

Drum Overheads (XY example)
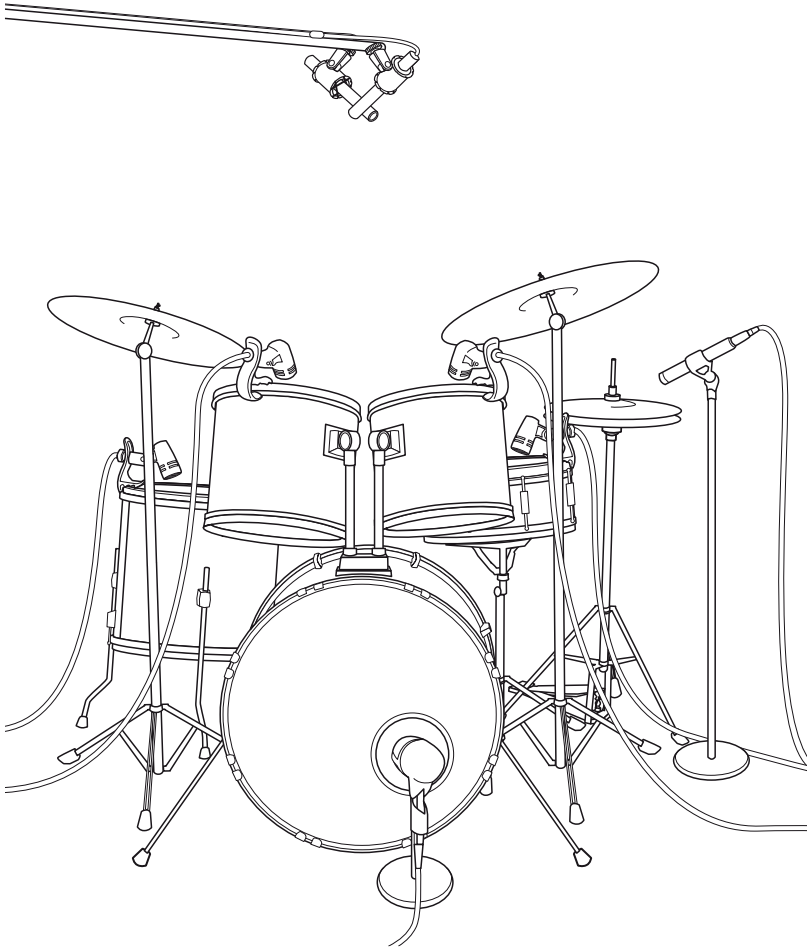
Place two small-diaphragm condenser microphones on an XY stereo-microphone holder (bar). Position the microphones so that each one is at a 45-degree angle, pointed down at the drum kit, approximately 7 or 8 feet above the floor or drum riser. Experiment with height. This technique can be used in live applications as well.

Snare Drum (top and bottom)
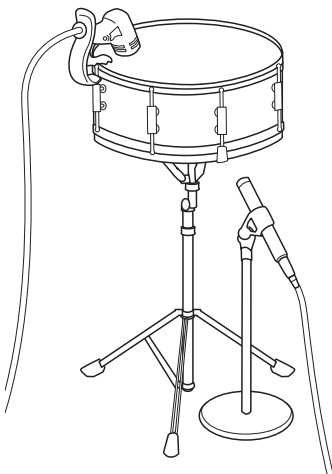
Point a dynamic microphone at the center of the snare, making sure it is placed so that the drummer will not hit it. Place a small-diaphragm condenser microphone under the drum, pointed at the snares. Experiment with the placement of both microphones. Also experiment with inverting the phase of the bottom microphone. This technique can be used in live applications.

6.2   Compression Setting Suggestions

The following are the compression presets that were used in the PreSonus BlueMax. We have included them as a jumping-off point for setting up compression on the StudioLive.

Vocals

Soft. This is an easy compression with a low ratio setting for ballads, allowing a wider dynamic range. It's good for live use. This setting helps the vocal "sit in the track."

THRESHOLD	RATIO	ATTACK	RELEASE
-8.2 dB	1.8:1	0.002 ms	38 ms

Medium. This setting has more limiting than the Soft compression setting, producing a narrower dynamic range. It moves the vocal more up front in the mix.

THRESHOLD	RATIO	ATTACK	RELEASE
-3.3 dB	2.8:1	0.002 ms	38 ms

Screamer. This setting is for loud vocals. It is a fairly hard compression setting for a vocalist who is on and off the microphone a lot. It puts the voice "in your face."

THRESHOLD	RATIO	ATTACK	RELEASE
-1.1 dB	3.8:1	0.002 ms	38 ms

Percussion

Snare/Kick. This setting allows the first transient through and compresses the rest of the signal, giving a hard "snap" up front and a longer release.

THRESHOLD	RATIO	ATTACK	RELEASE
-2.1 dB	3.5:1	78 ms	300 ms

Left/Right (Stereo) Overheads. The low ratio and threshold in this setting gives a "fat" contour to even out the sound from overhead drum mics. Low end is increased, and the overall sound is more present and less ambient. You get more "boom" and less "room."

THRESHOLD	RATIO	ATTACK	RELEASE
-13.7 dB	1.3:1	27 ms	128 ms

Fretted Instruments

Electric Bass. The fast attack and slow release in this setting will tighten up the electric bass and give you control for a more consistent level.

THRESHOLD	RATIO	ATTACK	RELEASE
-4.4 dB	2.6:1	45.7 ms	189 ms

Acoustic Guitar. This setting accentuates the attack of the acoustic guitar and helps maintain an even signal level, keeping the acoustic guitar from disappearing in the track.

THRESHOLD	RATIO	ATTACK	RELEASE
-6.3 dB	3.4:1	188 ms	400 ms

Electric Guitar. This is a setting for "crunch" electric rhythm guitar. A slow attack helps to get the electric rhythm guitar "up close and personal" and gives punch to your crunch.

THRESHOLD	RATIO	ATTACK	RELEASE
-0.1 dB	2.4:1	26 ms	193 ms

Keyboards

Piano. This is a special setting for an even level across the keyboard. It is designed to help even up the top and bottom of an acoustic piano. In other words, it helps the left hand to be heard along with the right hand.

THRESHOLD	RATIO	ATTACK	RELEASE
-10.8 dB	1.9:1	108 ms	112 ms

Synth. The fast attack and release on this setting can be used for synthesizer horn stabs or for bass lines played on a synthesizer.

THRESHOLD	RATIO	ATTACK	RELEASE
-11.9 dB	1.8:1	0.002 ms	85 ms

Orchestral. Use this setting for string pads and other types of synthesized orchestra parts. It will decrease the overall dynamic range for easier placement in the mix.

THRESHOLD	RATIO	ATTACK	RELEASE
3.3 dB	2.5:1	1.8 ms	50 ms

Stereo Mix

Stereo Limiter. Just as the name implies, this is a hard limiter, or "brickwall," setting—ideal for controlling the level to a two-track mixdown deck or stereo output.

THRESHOLD	RATIO	ATTACK	RELEASE
5.5 dB	7.1:1	0.001 ms	98 ms

Contour. This setting fattens up the main mix.

THRESHOLD	RATIO	ATTACK	RELEASE
-13.4 dB	1.2:1	0.002 ms	182 ms

Effects

Squeeze. This is dynamic compression for solo work, especially electric guitar. It gives you that glassy "Tele/Strat" sound. It is a true classic.

THRESHOLD	RATIO	ATTACK	RELEASE
-4.6 dB	2.4:1	7.2 ms	93 ms

Pump. This is a setting for making the compressor "pump" in a desirable way. This effect is good for snare drums to increase the length of the transient by bringing the signal up after the initial spike.

THRESHOLD	RATIO	ATTACK	RELEASE
0 dB	1.9:1	1 ms	0.001 ms

6.3   EQ Frequency Guides

Table 1

Instrument	What to Cut	Why to Cut	What to Boost	Why to Boost
Human Voice	7 kHz	Sibilance	8 kHz	Big sound
	2 kHz	Shrill	3 kHz and above	Clarity
	1 kHz	Nasal	200-400 Hz	Body
	80 Hz and below	Popping P's		
Piano	1-2 kHz	Tinny	5 kHz	More presence
	300 Hz	Boomy	100 Hz	Bottom end
Electric Guitar	1-2 kHz	Shrill	3 kHz	Clarity
	80 Hz and below	Muddy	125 Hz	Bottom end
Acoustic Guitar	2-3 kHz	Tinny	5 kHz and above	Sparkle
	200 Hz	Boomy	125 Hz	Full
Electric Bass	1 kHz	Thin	600 Hz	Growl
	125 Hz	Boomy	80 Hz and below	Bottom end
String Bass	600 Hz	Hollow	2-5 kHz	Sharp attack
	200 Hz	Boomy	125 Hz and below	Bottom end
Snare Drum	1 kHz	Annoying	2 kHz	Crisp
			150-200 Hz	Full
			80 Hz	Deep
Kick Drum	400 Hz	Muddy	2-5 kHz	Sharp attack
	80 Hz and below	Boomy	60-125 Hz	Bottom end
Toms	300 Hz	Boomy	2-5 kHz	Sharp attack
			80-200 Hz	Bottom end
Cymbals	1 kHz	Annoying	7-8 kHz	Sizzle
			8-12 kHz	Brilliance
			15 kHz	Air
Horns	1 kHz	Honky	8-12 kHz	Big sound
	120 Hz and below	Muddy	2 kHz	Clarity
String section	3 kHz	Shrill	2 kHz	Clarity
	120 Hz and below	Muddy	400-600 Hz	Lush and full

Table 2
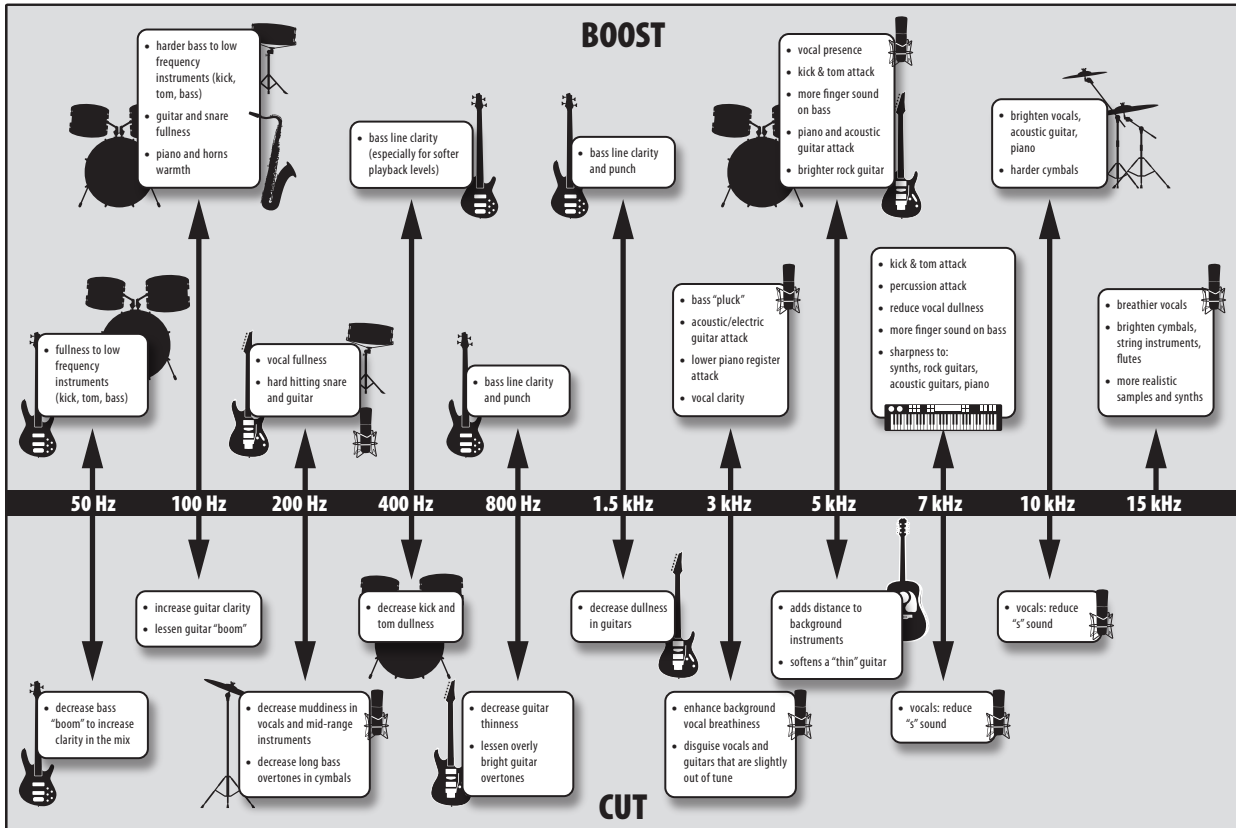
6.4   EQ Setting Suggestions

Included with your StudioLive is a library of Channel Strip presets. *Sections 4.1.9 and 4.1.10* discusses how to load these presets onto a channel or bus and how to create your own presets. For an idea of where to start, check out the following generic EQ settings for several different instruments. As with the compression settings in *Section 6.2*, the right EQ setting for any given instrument will depend upon the room and the tonality of the instrument.

Vocals

Pop Female Vocals

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	OFF	130	0.6	-2	ON	465	0.6	-2

HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	2.4	0.4	+2	ON	OFF	6.0	0.3	+8

Rock Female Vocals

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	ON	155	N/A	+4	ON	465	0.4	+6

HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	1.4	0.6	+6	ON	OFF	4.2	0.5	+2

Pop Male Vocals

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	OFF	225	0.3	-2	ON	960	0.3	0

HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	2.0	0.6	+2	ON	OFF	7.2	0.5	+4

Rock Male Vocals

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	OFF	155	0.5	+2	ON	265	0.3	-6

HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	2.4	0.6	-2	ON	ON	7.2	0.6	+4

Percussion

Snare

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	OFF	130	0.6	-4	ON	665	0.5	+4

HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	1.6	0.3	+4	ON	ON	4.2	N/A	+4

Left/Right (Stereo) Overheads

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	OFF	108	0.6	-2	ON	385	0.6	-2

HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	2.9	0.3	0	ON	ON	8.0	N/A	+4

Kick Drum

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	OFF	108	0.4	+4	ON	265	2.0	-4

HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	1.6	0.6	0	ON	OFF	6.0	2.0	+4

Fretted Instruments

Electric Bass

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	ON	36	N/A	-8	ON	130	0.4	+4
HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	2.0	0.6	+4	ON	ON	4.2	N/A	+1

Acoustic Guitar

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	OFF	155	0.4	+4	ON	665	2.0	+2
HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	2.0	0.3	0	ON	ON	6.0	N/A	+4

Distorted Electric Guitar

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	OFF	320	0.5	+6	ON	960	0.4	0
HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	3.5	1.0	+4	ON	ON	12	N/A	0

Keyboards

Piano

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	ON	108	N/A	-2	ON	665	0.2	+2
HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	2.9	0.4	+2	ON	OFF	7.2	0.6	+4

6.5   Technical Specifications

Microphone Preamp

Type	XLR Female, balanced
Frequency Response to Direct Output (at unity gain)	20 Hz-40 kHz, ± 0.5 dBu
Frequency Response to Main Output (at unity gain)	20 Hz-20 kHz, ± 0.5 dBu
Input Impedance	1 kΩ
THD to Direct Output (1 kHz at unity gain)	<0.007%, +4 dBu, 20 Hz–20 kHz, unity gain, unwtd
THD to Main Output (1 kHz at unity gain)	<0.005%, +4 dBu, 20 Hz-20 kHz, unity gain, unwtd
EIN to Direct Output	-125 dB unwtd, -130 dB A-wtd
S/N Ratio to Direct Output (Ref = +4 dB, 20 kHz BW, unity gain, A-wtd)	105 dB
S/N Ratio to Main Output (Ref = +4 dB, 20 kHz BW, unity gain, A-wtd)	94 dB
Common Mode Rejection Ratio (1 kHz at unity gain)	65 dB
Gain Control Range (±1 dB)	-15 dB to +65 dB
Maximum Input Level (unity gain)	+16 dBu
Phantom Power (±2 VDC)	+48 VDC

Line Inputs

Type	¼"TRS Female, balanced mono
Frequency Response to Direct Outputs (at unity gain)	10 Hz-40 kHz, ± 0.5 dBu
Frequency Response to Main Outputs (at unity gain)	20 Hz-20 kHz, ± 0.5 dBu
Input Impedance	10 kΩ
THD to Direct Output (1 kHz at unity gain)	<0.007%, +4 dBu, 20 Hz-20 kHz, unity gain, unwtd
THD to Main Output (1 kHz at unity gain)	<0.005%, +4 dBu, 20 Hz-20 kHz, unity gain, unwtd
S/N Ratio to Direct Output (Ref = +4 dBu, 20 kHz BW, unity gain, A-wtd)	105 dB
S/N Ratio to Main Output (Ref = +4 dBu, 20 kHz BW, unity gain, A-wtd)	94 dB
Gain Control Range (±1 dB)	-20 dB to +20 dB
Maximum Input level (unity gain)	+22 dBu

Tape Inputs

Type	RCA Female, unbalanced (stereo pair)
Maximum Input Level	+10 dBu

Auxiliary Inputs

Type	¼"TRS Female, balanced (2 stereo pairs)
Maximum Input Level	+18 dBu

Main Outputs

Type	XLR Male, balanced (stereo pair); ¼"TRS Female, balanced (stereo pair); XLR Male, balanced (mono)
Rated Output Level	+24 dBu
Output Impedance	100Ω

Aux Outputs

Type	¼"TRS Female, balanced (mono)
Rated Output Level	+18 dBu
Output Impedance	100Ω

Subgroup Outputs

Type	¼" TRS Female, balanced (mono)
Rated Output Level	+18 dBu
Output Impedance	100Ω

Tape Outputs

Type	RCA Female, unbalanced (stereo pair)
Rated Output Level	+18 dBu
Output Impedance	100Ω

Control Room Outputs

Type	¼" TRS Female, balanced (stereo pair)
Rated Output Level	+18 dBu
Output Impedance	100Ω

System Crosstalk

Input to Output (Ref = +4 dBu, 20 Hz-20 kHz, unwtd)	-90 dBu
Adjacent Channels (Ref = +4 dBu, 20 Hz-20 kHz, unwtd)	-87 dBu

Noise Gate / Expander

Threshold Range (StudioLive 32.4.2AI and 24.4.2AI)	-56 dB to 0 dB
Attack Time	0.02s to 500 ms / 0.5 ms
Release Time	0.05s to 2s
Expander Attenuation Range	2:1 (fixed)
Gate Attenuation Range	-84 dB to 0 dB / ∞
Key Listen Filter (StudioLive 32.4.2AI and 24.4.2AI)	2nd-order resonant bypass; Q=0.7; OFF, 40 Hz - 16 kHz

Limiter

Threshold	-56 to 0 dB / -28 dBFS
Ratio	∞:1
Attack	20 ns
Hold	10 ms
Release	20 ms

Compressor

Threshold Range	-56 dB to 0 dB
Ratio	1:1 to 14:1
Attack Time	0.2 ms to 150 ms
Release Time	2.5 ms to 900 ms
Auto Attack and Release	Attack = 10 ms, Release = 150 ms
Curve Types	hard and soft knee

Parametric EQ

Type	2nd-order shelving filter
Low (Low-pass or Bandpass)	36 to 465 Hz, ± 15 dB
Low Mid	90 Hz to 1.2 kHz, ±15 dB
High Mid	380 Hz to 5 kHz, ±15 dB
High (High-pass or Bandpass)	1.4 kHz to 18 kHz, ±15 dB
Q	0.1 to 4 / Low Q= 0.55, Hi Q=2.0

Graphic EQ

31-Band 1/3rd Octave Controls	Curve-fitting algorithm
Gain/Attenuation	±15 dB

Digital Audio

ADC Dynamic Range (A-wtd, 48 kHz)	118 dB
DAC Dynamic Range (A-wtd, 48 kHz)	118 dB
FireWire	s800, 800 Mb/s
Internal Processing	32-bit, floating point
Sampling Rate	44.1, 48, 88.2, 96 kHz
A/D/A Bit Depth	24 bits
Reference Level for 0 dBFS	+18 dBu

Clock

Jitter	<20 ps RMS (20 Hz - 20 kHz)
Jitter Attenuation	>60 dB (1 ns in ≈ 1 ps out)

Power

Connector	IEC
Input-Voltage Range	100-230V, 50-60 Hz
Power Requirements (continuous)	200W

Physical

	32.4.2AI	**24.4.2AI**	**16.4.2AI**
Length	21.31" (54.13 cm)	21.31" (54.13 cm)	22.35" (58.6 cm)
Width (chassis only)	31.58" (80.21 cm)	25.19" (63.98 cm)	17.22" (43.74 cm)
Maximum Height	7.02" (17.82 cm)	7.02" (17.82 cm)	6.9" (17.53 cm)
Weight	50 lbs. (22.68 kg)	30 lbs. (13.6 kg)	23 lbs. (10.43 kg)

Global Warming

Recommended Ambient Operating Temperature	0˚ to 40˚ Celsius / 32˚ to 104˚ Fahrenheit
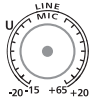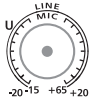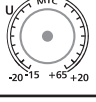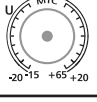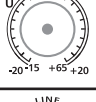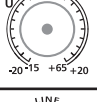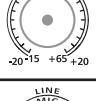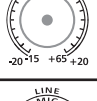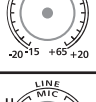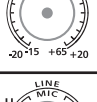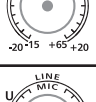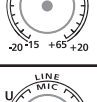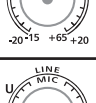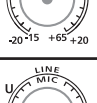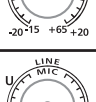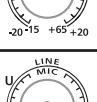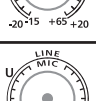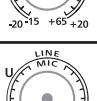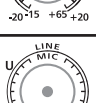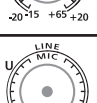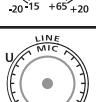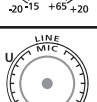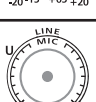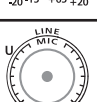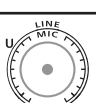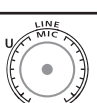
6.6   StudioLive AI Mixers Block Diagrams

We've finally made block diagrams too large for our printed manuals. Please visit the following pages on our website for the latest Block Diagrams for all three StudioLive AI-series mixers in Adobe PDF format:

http://www.presonus.com/products/StudioLive-AI-Series/downloads
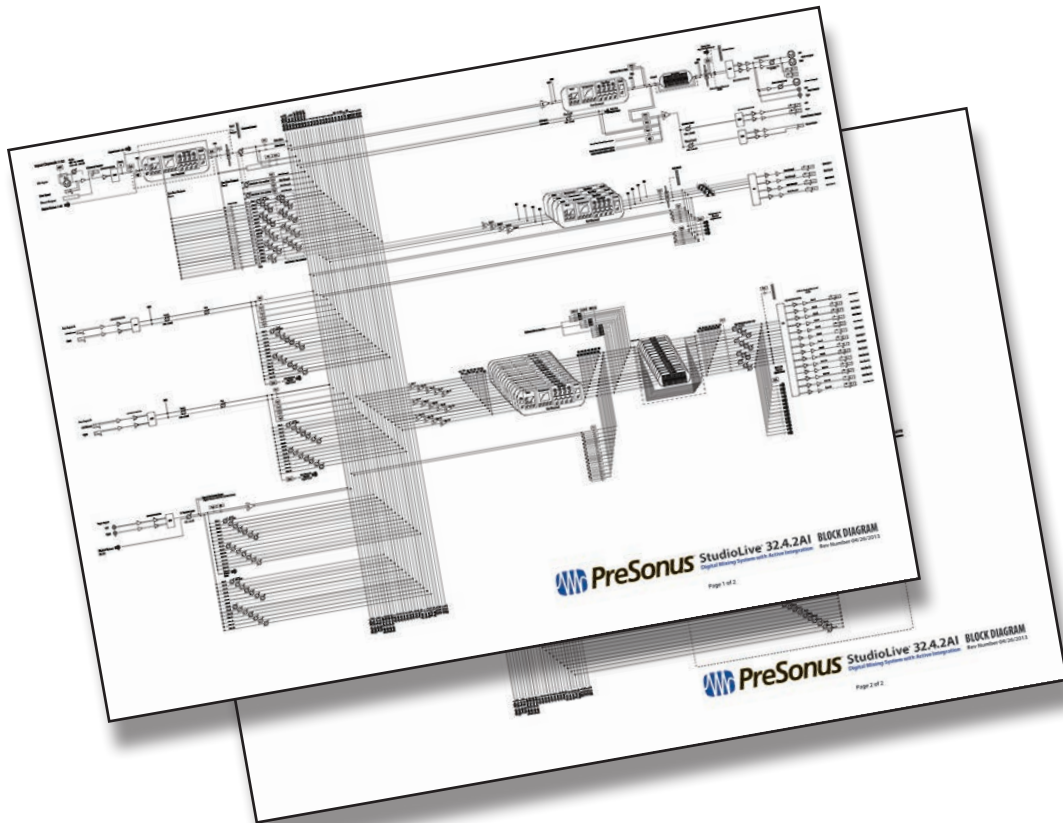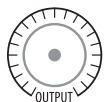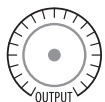
6.7   StudioLive 32.4.2AI Recall Sheet

TRACK	TRIM	PRODUCTION NOTES	TRACK	TRIM	PRODUCTION NOTES
1		Instrument_____ Mic used_____ Notes_____	17		Instrument_____ Mic used_____ Notes_____
2		Instrument_____ Mic used_____ Notes_____	18		Instrument_____ Mic used_____ Notes_____
3		Instrument_____ Mic used_____ Notes_____	19		Instrument_____ Mic used_____ Notes_____
4		Instrument_____ Mic used_____ Notes_____	20		Instrument_____ Mic used_____ Notes_____
5		Instrument_____ Mic used_____ Notes_____	21		Instrument_____ Mic used_____ Notes_____
6		Instrument_____ Mic used_____ Notes_____	22		Instrument_____ Mic used_____ Notes_____
7		Instrument_____ Mic used_____ Notes_____	23		Instrument_____ Mic used_____ Notes_____
8		Instrument_____ Mic used_____ Notes_____	24		Instrument_____ Mic used_____ Notes_____
9		Instrument_____ Mic used_____ Notes_____	25		Instrument_____ Mic used_____ Notes_____
10		Instrument_____ Mic used_____ Notes_____	26		Instrument_____ Mic used_____ Notes_____
11		Instrument_____ Mic used_____ Notes_____	27		Instrument_____ Mic used_____ Notes_____
12		Instrument_____ Mic used_____ Notes_____	28		Instrument_____ Mic used_____ Notes_____
13		Instrument_____ Mic used_____ Notes_____	29		Instrument_____ Mic used_____ Notes_____
14		Instrument_____ Mic used_____ Notes_____	30		Instrument_____ Mic used_____ Notes_____
15		Instrument_____ Mic used_____ Notes_____	31		Instrument_____ Mic used_____ Notes_____
16		Instrument_____ Mic used_____ Notes_____	32		Instrument_____ Mic used_____ Notes_____

Artist _____   **Scene** _____   **Date** _____

AUX INPUTS

AUX	LEVEL	AUX IN A SOURCE	AUX	LEVEL	AUX IN B SOURCE
A	(OUTPUT)	Notes_____	**B**	(OUTPUT)	Notes_____

AUX SENDS

AUX	LEVEL	AUX SEND	AUX	LEVEL	AUX SEND
1	(OUTPUT)	Notes_____ Monitor ☐ Send ☐	**8**	(OUTPUT)	Notes_____ Monitor ☐ Send ☐
2	(OUTPUT)	Notes_____ Monitor ☐ Send ☐	**9**	(OUTPUT)	Notes_____ Monitor ☐ Send ☐
3	(OUTPUT)	Notes_____ Monitor ☐ Send ☐	**10**	(OUTPUT)	Notes_____ Monitor ☐ Send ☐
4	(OUTPUT)	Notes_____ Monitor ☐ Send ☐	**11**	(OUTPUT)	Notes_____ Monitor ☐ Send ☐
5	(OUTPUT)	Notes_____ Monitor ☐ Send ☐	**12**	(OUTPUT)	Notes_____ Monitor ☐ Send ☐
6	(OUTPUT)	Notes_____ Monitor ☐ Send ☐	**13**	(OUTPUT)	Notes_____ Monitor ☐ Send ☐
7	(OUTPUT)	Notes_____ Monitor ☐ Send ☐	**14**	(OUTPUT)	Notes_____ Monitor ☐ Send ☐

SUBGROUP / MAIN ASSIGNMENT ROUTING MATRIX

	CH 1	CH 2	CH 3	CH 4	CH 5	CH 6	CH 7	CH 8	CH 9	CH 10	CH 11	CH 12	CH 13	CH 14	CH 15	CH 16	CH 17	CH 18	CH 19	CH 20	CH 21	CH 22	CH 23	CH 24	CH 25	CH 26	CH 27	CH 28	CH 29	CH 30	CH 31	CH 32
SUB 1																																
SUB 2																																
SUB 3																																
SUB 4																																
MAIN																																
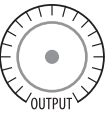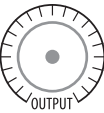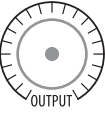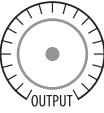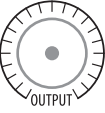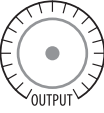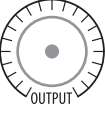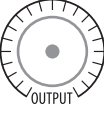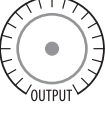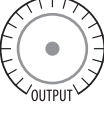
	FX**A**	FX**B**	FX**C**	FX**D**	AUX **A**	AUX **B**
SUB **1**						
SUB **2**						
SUB **3**						
SUB **4**						
MAIN						

6.8   StudioLive 24.4.2AI Recall Sheet

TRACK	TRIM	PRODUCTION NOTES	TRACK	TRIM	PRODUCTION NOTES
1		Instrument_____ Mic used_____ Notes_____ _____	13		Instrument_____ Mic used_____ Notes_____ _____
2		Instrument_____ Mic used_____ Notes_____ _____	14		Instrument_____ Mic used_____ Notes_____ _____
3		Instrument_____ Mic used_____ Notes_____ _____	15		Instrument_____ Mic used_____ Notes_____ _____
4		Instrument_____ Mic used_____ Notes_____ _____	16		Instrument_____ Mic used_____ Notes_____ _____
5		Instrument_____ Mic used_____ Notes_____ _____	17		Instrument_____ Mic used_____ Notes_____ _____
6		Instrument_____ Mic used_____ Notes_____ _____	18		Instrument_____ Mic used_____ Notes_____ _____
7		Instrument_____ Mic used_____ Notes_____ _____	19		Instrument_____ Mic used_____ Notes_____ _____
8		Instrument_____ Mic used_____ Notes_____ _____	20		Instrument_____ Mic used_____ Notes_____ _____
9		Instrument_____ Mic used_____ Notes_____ _____	21		Instrument_____ Mic used_____ Notes_____ _____
10		Instrument_____ Mic used_____ Notes_____ _____	22		Instrument_____ Mic used_____ Notes_____ _____
11		Instrument_____ Mic used_____ Notes_____ _____	23		Instrument_____ Mic used_____ Notes_____ _____
12		Instrument_____ Mic used_____ Notes_____ _____	24		Instrument_____ Mic used_____ Notes_____ _____

Artist   **Scene**   **Date**

AUX INPUTS

AUX	LEVEL	AUX IN A SOURCE	AUX	LEVEL	AUX IN B SOURCE
A	OUTPUT	Notes_____	B	OUTPUT	Notes_____

AUX SENDS

AUX	LEVEL	AUX SEND	AUX	LEVEL	AUX SEND
1	OUTPUT	Notes_____ Monitor ☐ Send ☐	6	OUTPUT	Notes_____ Monitor ☐ Send ☐
2	OUTPUT	Notes_____ Monitor ☐ Send ☐	7	OUTPUT	Notes_____ Monitor ☐ Send ☐
3	OUTPUT	Notes_____ Monitor ☐ Send ☐	8	OUTPUT	Notes_____ Monitor ☐ Send ☐
4	OUTPUT	Notes_____ Monitor ☐ Send ☐	9	OUTPUT	Notes_____ Monitor ☐ Send ☐
5	OUTPUT	Notes_____ Monitor ☐ Send ☐	10	OUTPUT	Notes_____ Monitor ☐ Send ☐

SUBGROUP / MAIN ASSIGNMENT ROUTING MATRIX

	CH 1	CH 2	CH 3	CH 4	CH 5	CH 6	CH 7	CH 8	CH 9	CH 10	CH 11	CH 12	CH 13	CH 14	CH 15	CH 16	CH 17	CH 18	CH 19	CH 20	CH 21	CH 22	CH 23	CH 24	EFX A	EFX B	AUX A	AUX B
SUB 1																												
SUB 2																												
SUB 3																												
SUB 4																												
MAIN																												

6.9   StudioLive 16.4.2AI Recall Sheet

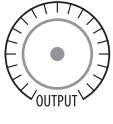| TRACK	TRIM	PRODUCTION NOTES	TRACK	TRIM	PRODUCTION NOTES
1		Instrument_____ Mic used_____ Notes_____ _____	9		Instrument_____ Mic used_____ Notes_____ _____
2		Instrument_____ Mic used_____ Notes_____ _____	10		Instrument_____ Mic used_____ Notes_____ _____
3		Instrument_____ Mic used_____ Notes_____ _____	11		Instrument_____ Mic used_____ Notes_____ _____
4		Instrument_____ Mic used_____ Notes_____ _____	12		Instrument_____ Mic used_____ Notes_____ _____
5		Instrument_____ Mic used_____ Notes_____ _____	13		Instrument_____ Mic used_____ Notes_____ _____
6		Instrument_____ Mic used_____ Notes_____ _____	14		Instrument_____ Mic used_____ Notes_____ _____
7		Instrument_____ Mic used_____ Notes_____ _____	15		Instrument_____ Mic used_____ Notes_____ _____
8		Instrument_____ Mic used_____ Notes_____ _____	16		Instrument_____ Mic used_____ Notes_____ _____

SUBGROUP / MAIN ASSIGNMENT ROUTING MATRIX

	CH 1	CH 2	CH 3	CH 4	CH 5	CH 6	CH 7	CH 8	CH 9	CH 10	CH 11	CH 12	CH 13	CH 14	CH 15	CH 16	EFX A	EFX B	AUX A	AUX B
SUB 1																				
SUB 2																				
SUB 3																				
SUB 4																				
MAIN																				

Artist _____ **Scene** _____ **Date** _____

AUX INPUTS			

AUX	LEVEL	AUX IN A SOURCE	AUX	LEVEL	AUX IN B SOURCE
A	OUTPUT	Notes_____	B	OUTPUT	Notes_____

AUX SENDS			

AUX	LEVEL	AUX SEND	AUX	LEVEL	AUX SEND
1	OUTPUT	Notes_____ Monitor ☐ Send ☐	4	OUTPUT	Notes_____ Monitor ☐ Send ☐
2	OUTPUT	Notes_____ Monitor ☐ Send ☐	5	OUTPUT	Notes_____ Monitor ☐ Send ☐
3	OUTPUT	Notes_____ Monitor ☐ Send ☐	6	OUTPUT	Notes_____ Monitor ☐ Send ☐

6.10   StudioLive 16.4.2AI Rack Ear Installation
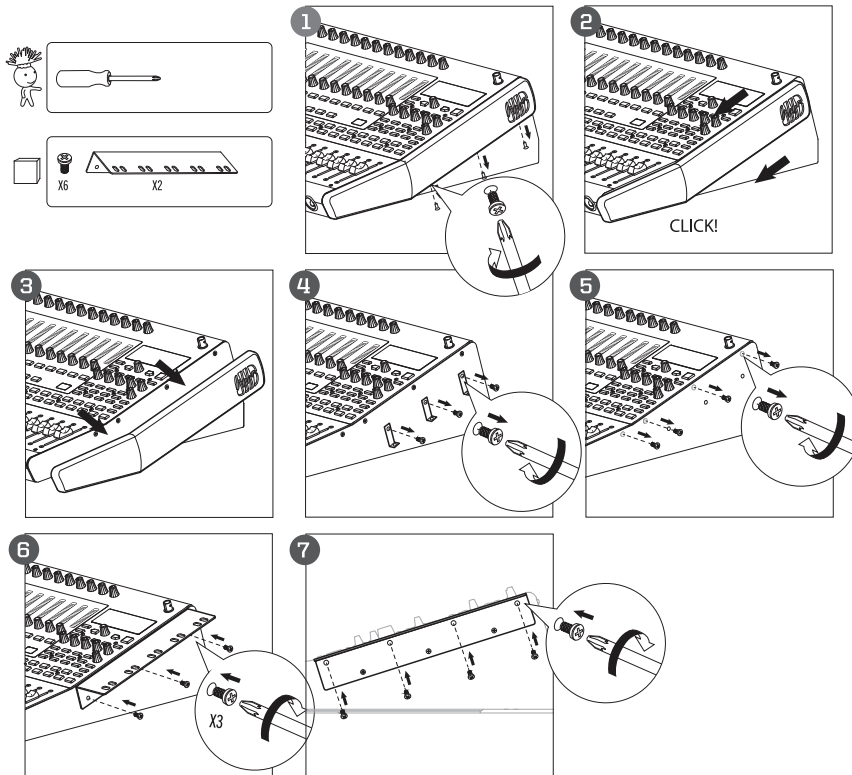
7   Troubleshooting and Warranty

7.1   Troubleshooting

Please check the PreSonus Web site (www.presonus.com) regularly for software information and updates, firmware updates, and support documentation, including frequently asked questions.

Online technical support is available to registered users through their My PreSonus account. Visit my.presonus.com to register.

PreSonus telephone technical support is available to customers in the USA on Monday through Friday from 9 a.m. to 5 p.m. Central Time by calling 1-225-216-7887. Customers outside of the USA should contact their national or regional distributor for telephone technical support. A list of international distributors is provided at www.presonus.com/buy/international_distributors.

No Output on a Channel

Press the Input button in the Metering section and verify that there is signal on that channel. If no signal is present, check the cable and the input source. Verify that the trim control is set at an appropriate level. Make sure the channel is not muted. Make sure that if your device requires phantom power, phantom power is engaged for the channel. If you are using an analog input, verify that the Digital return button for that channel is not engaged.

If signal is present, press the Select button for the channel in question. Make sure that the channel is assigned to the main output in the Assign section of the Fat Channel.

Fader Movements Have No Effect on Audio

Verify that your StudioLive is not locked by navigating to Page 8: Lockout, in the System menu. Verify that your StudioLive is not in Fader Locate mode. Select Output in the Metering section and verify that your fader movements are affecting the output signal. If so, make sure your channels are assigned to the main outputs.

No Internal Effects in the Main Bus

Press the Aux button in the Metering section and verify the output levels of the internal FX A-D aux buses. If the level is too low, use the Output knob to increase the master level for the effects mix. Press the Select button for each FX bus and make sure it is assigned to the main output in the Assign section of the Fat Channel.

No Output on the Solo Bus While Monitoring

Verify that both the Solo volume and the headphone, or monitor, volume is at a reasonable level for comfortable listening. Make sure that you only have Solo selected in the Monitor section of your StudioLive.

Buttons/Knobs Are Not Functioning

If your StudioLive is passing audio but you have no Fat Channel, fader, or menu control, verify that the StudioLive is not locked by navigating to Page 8: Lockout, in the System menu.

Can't Hear Main Mix in Headphones

Verify that the Main mix is enabled in the Monitor bus and that the Headphone output control is at a sufficient level.

Monitor Bus Controls Not Changing Routing

Verify that your monitors are connected to the Control Room outputs on the rear panel of your StudioLive, not the Main outputs.

Main Fader Doesn't Control Mix Level

Verify that your monitors are connected to the Main outputs on the rear panel of your StudioLive, not the Control Room outputs. Verify that StudioLive is not in Fader Locate mode.

7.2   PreSonus Limited Warranty

PreSonus Audio Electronics, Inc., warrants this product to be free of defects in material and workmanship for a period of one year from the date of original retail purchase. This warranty is enforceable only by the original retail purchaser. To be protected by this warranty, the purchaser must complete and return the enclosed warranty card within 14 days of purchase. During the warranty period PreSonus shall, at its sole and absolute option, repair or replace, free of charge, any product that proves to be defective on inspection by PreSonus or its authorized service representative. If you are located in the USA and need warranty repair, please submit an online technical support request at http://support.presonus.com to receive a return-authorization number and shipping information. If you are located outside of the USA, please contact the PreSonus distributor for your region for warranty repairs. All inquiries must be accompanied by a description of the problem. All authorized returns must be sent to the PreSonus repair facility postage prepaid, insured, and properly packaged. PreSonus reserves the right to update any unit returned for repair. PreSonus reserves the right to change or improve the design of the product at any time without prior notice. This warranty does not cover claims for damage due to abuse, neglect, alteration, or attempted repair by unauthorized personnel and is limited to failures arising during normal use that are due to defects in material or workmanship in the product. Any implied warranties, including implied warranties of merchantability and fitness for a particular purpose, are limited in duration to the length of this limited warranty. Some states do not allow limitations on how long an implied warranty lasts, so the above limitation may not apply to you. In no event will PreSonus be liable for incidental, consequential, or other damages resulting from the breach of any express or implied warranty, including, among other things, damage to property, damage based on inconvenience or on loss of use of the product, and, to the extent permitted by law, damages for personal injury. Some states do not allow the exclusion of limitation of incidental or consequential damages, so the above limitation or exclusion may not apply to you. This warranty gives you specific legal rights, and you may also have other rights, which vary from state to state. This warranty only applies to products sold and used in the United States of America. For warranty information in all other countries, please refer to your local distributor.

PreSonus Audio Electronics, Inc.

18011 Grand Bay Ct.

Baton Rouge, Louisiana 70809 USA

1-225-216-7887

www.presonus.com

Added bonus: PreSonus' previously Top Secret recipe for…

Chicken and Andouille Gumbo

Ingredients:

- 1 C All-Purpose flour
- ¾ C Vegetable Oil
- 1 large onion (diced)
- 1 small onion (quartered)
- 6 celery stalks (diced)
- 1 large green bell pepper (diced)
- 3 cloves garlic (2 minced, 1 whole)
- 1 lb link Andouille sausage
- 4 Chicken leg quarters
- 4 qt water
- 4 bay leaves
- 1 tsp thyme
- 1 tsp Old Bay seasoning
- 1-2 C frozen okra, sliced
- ¼ C fresh parsley, minced
- 6-8 eggs (optional)

Cooking Instructions:

1. In a large pot, combine whole chicken leg quarters, water, quartered onion, Old Bay, 2 bay leaves and 1 whole clove garlic. Cover and bring to a low boil. Simmer stock until chicken is falling off the bone. Remove the chicken and set aside. Discard the onion, bay leaves, and garlic, reserving the liquid.
2. In a heavy saucepan, heat 1 Tbsp of the oil on medium high heat and brown the andouille until it is cooked through. Set aside sausage for later.
3. In the same saucepan, add and heat remaining oil. Slowly add flour 1-2 Tbsp at a time, stirring continuously. Continue cooking and stirring the roux until it is a dark brown (it should look like melted dark chocolate). Be careful to not to get the oil too hot or the flour will burn and you'll have to start over.
4. Once roux has reached the correct color, add diced onion, celery, green pepper, and minced garlic. Cook until vegetables are very tender. Do not cover.
5. Slowly add 1 quart of chicken broth and bring to a low boil, stirring constantly.
6. Transfer roux mixture to a soup pot and bring to low boil. Do not cover, the roux will settle on the bottom of the pot and burn.
7. Add remaining chicken broth, bay leaves, and thyme. Simmer for 30 minutes.
8. While gumbo is simmering, debone and shred chicken and slice the andouille.
9. Add chicken and andouille to gumbo and return to a simmer. Simmer for 30-45 minutes.
10. Stir in frozen okra and parsley and bring to a rolling boil.
11. **Optional:** Crack one egg into a teacup and quickly pour into the boiling gumbo. Repeat with the other eggs being careful not to cluster them too closely. After all the eggs have risen back to the surface, reduce heat and simmer.
12. 12. Correct seasoning with salt and pepper (red, white and/or black) if necessary.
13. Serve over rice with potato salad.

Serves 12

StudioLive™ AI-Series Mixers

Digital Mixing System with Active Integration™

Owner's Manual