



DOCUMENTAZIONE

Moonwalk Service

c/o Claudio Giulio Prencipe

Noleggio Audio e Luci | Installazioni per Eventi
via Camogli 11/3 - Rivoli frazione Cascine Vica, 10098 (TO)

+39/3477822465

moonwalkservice@gmail.com

www.noleggio-audio-luci.it | www.moonwalkgroup.com

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StudioLive™

16.4.2



16 Channel Digital Recording and Performance Mixer

User Manual Version 1.10

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StudioLive

User's Manual

Version 1.10

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PreSonus Audio Electronics, Inc.
7257 Florida Blvd.
Baton Rouge, LA 70806
www.presonus.com

Safe Operation Guidelines

To avoid damage to your StudioLive and your other audio equipment please review and adhere to the following safety guidelines:

- Follow the safety guidelines in the manual.
- Do not block the ventilation openings.
- Do not drop your StudioLive.
- Do not install near a heat source (radiators, heat registers, amplifier heat sinks, etc.).
- Do not expose your StudioLive to liquids. Do not place containers filled with liquids near your StudioLive.
- Do not allow dust particles to collect in excess on your StudioLive. Keeping the unit covered when not in use is highly recommended and will extend the life of your product.
- Protect the power cord from being walked on, wheeled over, or pinched. If your IEC cord becomes damaged, purchase a new one.
- Unplug your StudioLive when not in use for long periods of time and during electrical storms, hurricanes, tornadoes, and other extreme weather.
- Use only the attachments/accessories recommended or manufactured by PreSonus for your StudioLive.
- All domestic PreSonus products should be serviced at the PreSonus factory in Baton Rouge, Louisiana. If your StudioLive requires a repair contact techsupport@presonus.com to arrange for a return authorization number. Customers outside the U.S. should contact their local distributor. Your distributor's contact information is available at www.presonus.com.

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1.1 Introduction



Thank you for purchasing the PreSonus StudioLive™ 16.4.2. PreSonus Audio Electronics has designed the StudioLive™ utilizing high-grade components to ensure optimum performance that will last a lifetime. Loaded with 16 high-headroom, XMAX microphone preamplifiers; a built-in 32x18 FireWire recording and playback engine; Fat Channel processing with 4-band EQs, compressors, limiters, and expander/gates; DSP effects; 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 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 at (225) 216-7887 with any questions or comments you may have regarding your PreSonus StudioLive. 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.

We suggest that you use this manual to familiarize yourself with the features, applications, and correct connection procedures for your StudioLive before trying to connect it to your computer. This will help you avoid problems during installation and setup.

1.2 Features

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

The StudioLive™ performance and recording digital mixer is a fully-loaded professional digital mixer combined with a complete 32x18 FireWire recording system. Racks of processing effects including compressor, limiter, gate, four-band parametric EQ, reverb and delay are available on every channel, subgroup, aux, and main mix delivering total control in a compact rugged steel chassis. StudioLive™ includes CAPTURE™, a fully integrated live recording software by PreSonus, allowing you to record every performance and rehearsal with a few clicks of your mouse.

Intuitive, flexible and powerful, StudioLive revolutionizes music production opening endless creative possibilities.

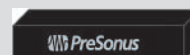
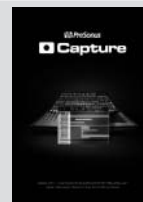
Summary of features

- 24-bit/48 kHz sampling rate
- 16 Class A XMAX microphone preamplifiers
- 16 line-level inputs
- 6 auxiliary buses
- 4 subgroups
- High-definition analog to digital converters (118 dB dynamic range)
- Unlimited-headroom, 32-bit floating-point, digital mixing and effects processing
- 32x18 FireWire digital recording interface with two FireWire 400 (IEEE 1394) ports
- Scene automation with load/save/recall of all settings
- Fat Channel with:
 - High-pass filter
 - Compressor
 - Limiter
 - Expander/Gate
 - 4-band semi-parametric EQ
 - Pan
 - Load/save
- 2 master DSP effects (reverb and delay with Load and Save)
- 100 mm long-throw faders
- Military-grade quick-touch buttons
- Fast-acting LED meters
- Talkback communication system
- Compact, 19" rack-mountable, rugged steel chassis
- PreSonus Capture™ multitrack-recording software
- Compatible with Logic, Cubase, Nuendo, Sonar, Digital Performer and others
- PC and Mac compatible

1.3 What is in the Box

In addition to this manual, your StudioLive package contains the following:

- PreSonus StudioLive digital recording and performance mixer
- 6' 6-pin-to-6-pin FireWire 400 cable
- IEC power cord
- Software installation DVD containing:
 - PreSonus StudioLive drivers
 - PreSonus Capture
 - PreSonus Capture Demo Session
- PreSonus Capture Reference Manual
- StudioLive Rack Ear Installation Kit



1.4 Getting Started: Mixing

Before you begin, there are a few general rules of thumb that we recommend you follow:

- 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 your StudioLive to adjust the level up or down until you have achieved your 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
 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!

1. Grab a microphone and a mic cable and plug it into the StudioLive’s Channel 1 mic input.
2. Connect the Main Outs (TRS or XLR) of your Studio Live to your power amplifier or powered monitors.
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.
5. Plug your StudioLive into a power outlet and turn it on.
6. If your microphone requires phantom power, engage the 48V button on Channel 1 of your StudioLive
7. Turn on your amplifier or powered monitors.



Input

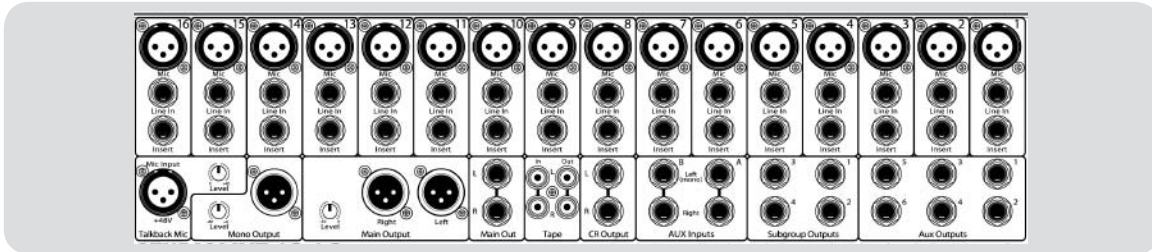
1.4

Getting Started: Mixing

8. Press the Input button in the Meter section.
9. While speaking or singing into your microphone, turn the trim knob on Channel 1 until the signal reaches the desired level, but don't raise it too much or the input will clip. Watch the level meter!
10. Press the Select button on Channel 1 and bring the fader to unity gain.
11. Press the Main button in the Assign section of the Fat Channel so that it is illuminated. This routes the channel to the main output bus.
12. Bring up the Main fader until you can comfortably listen to your microphone through your speakers.
13. With Channel 1 selected, you can use the Fat Channel to add dynamics processing and EQ.

For more detailed level-setting procedures and techniques, refer to sections 5.7 and 5.8.3

2.1 The Patch Bay



- Microphone Inputs.** Your StudioLive is equipped with 16 custom-designed PreSonus XMAX microphone preamplifiers for use with all types of microphones (including dynamics, condensers, and ribbons). The award winning PreSonus 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 has 48V phantom power available for each channel individually via buttons on the top panel.
- XLR Connector Wiring for Phantom Power:**
 - Pin 1 = GND
 - Pin 2 = +48V
 - Pin 3 = +48V
- +22 dBu Headroom.** The StudioLive microphone preamplifier has +22 dBu headroom. This gives you wide dynamic range and excellent transient-response characteristics.
- Line-Level Input.** Each channel of the StudioLive has a 1/4-inch, balanced TRS connection for line-level input. When these inputs are engaged, the microphone-preamp circuit is bypassed.

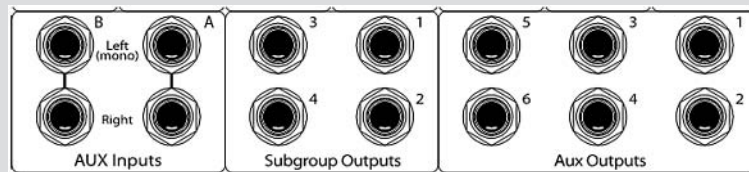
Please Note: As with any mixer, plugging in a microphone or a line-level input device as well as turning phantom power on or off will create a momentary spike in the audio output of your StudioLive. Because of this, it is highly recommended that you mute or turn down the trim of any channel before changing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

- Insert.** Each channel of the StudioLive has a direct-insert point. These unbalanced 1/4" connectors can be used to connect external processors such as compressors, EQs, de-essers, and filters with your StudioLive's preamps and line 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. In other words, 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.

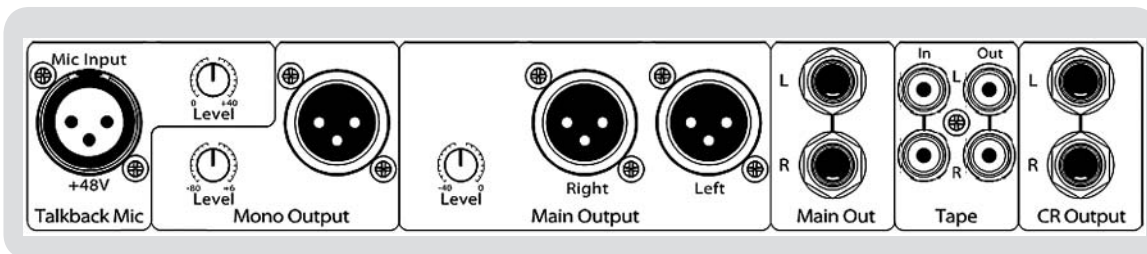
2.1

The Patch Bay

- Insert connector wiring:
 - Tip = send (output to inserted device)
 - Ring = return (input from inserted device)
 - Sleeve = common ground



- **Aux Inputs.** The StudioLive is equipped with two auxiliary inputs (effects returns). In Section 5.5.2, 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. If you are returning a mono signal to the mix, simply connect it to the Left input; the signal will be routed to both sides of the stereo mix.
- **Subgroup Outputs.** These are balanced mono outputs for each subgroup.
- **Aux Outputs.** The StudioLive is equipped with six auxiliary outputs. In Section 5.5, we discuss in detail how to create aux mixes for monitoring and effects processing. Aux mixes are routed to these outputs.



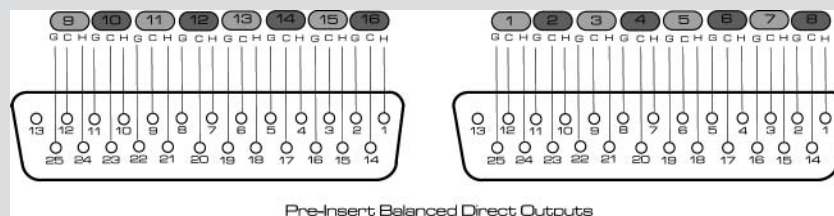
- **Talkback Mic Input.** The StudioLive does not have an onboard talkback mic; an external mic must be used. Phantom power is always enabled on this microphone preamp, so either a dynamic or a condenser microphone can be used. This is the same high-quality XMAX preamp that is featured on Channels 1-16 and can be used as an extra input when using the StudioLive as an audio interface. See Section 3.3.1 for more details.
- **Talkback Mic Level.** This is the gain control for your talkback microphone.
- **Mono Output.** This balanced output carries a mono, summed version of the stereo signal from the main bus.

2.1 The Patch Bay

- **Mono Output Level.** This knob controls the maximum level of the Mono Output signal. The signal can be attenuated to -40 dB and boosted up to +10 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 both the XLR and TRS main outputs. The signal can be attenuated to -40 dB and boosted up to +10 dB
- **Tape In/Out.** The StudioLive is equipped with stereo RCA (coaxial) 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 2Track In knob on the top panel. The main bus is routed post-fader to the tape output.
- **CR Output.** 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 16 channels. The DB25 connectors divide the channels into two 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 firewire returns (See Section 3.3.1 for more information) cannot be patched to the direct outputs.

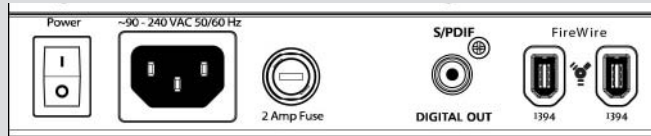


DB25 Pinout

H = Hot
C = Cold
G = Ground

2.1

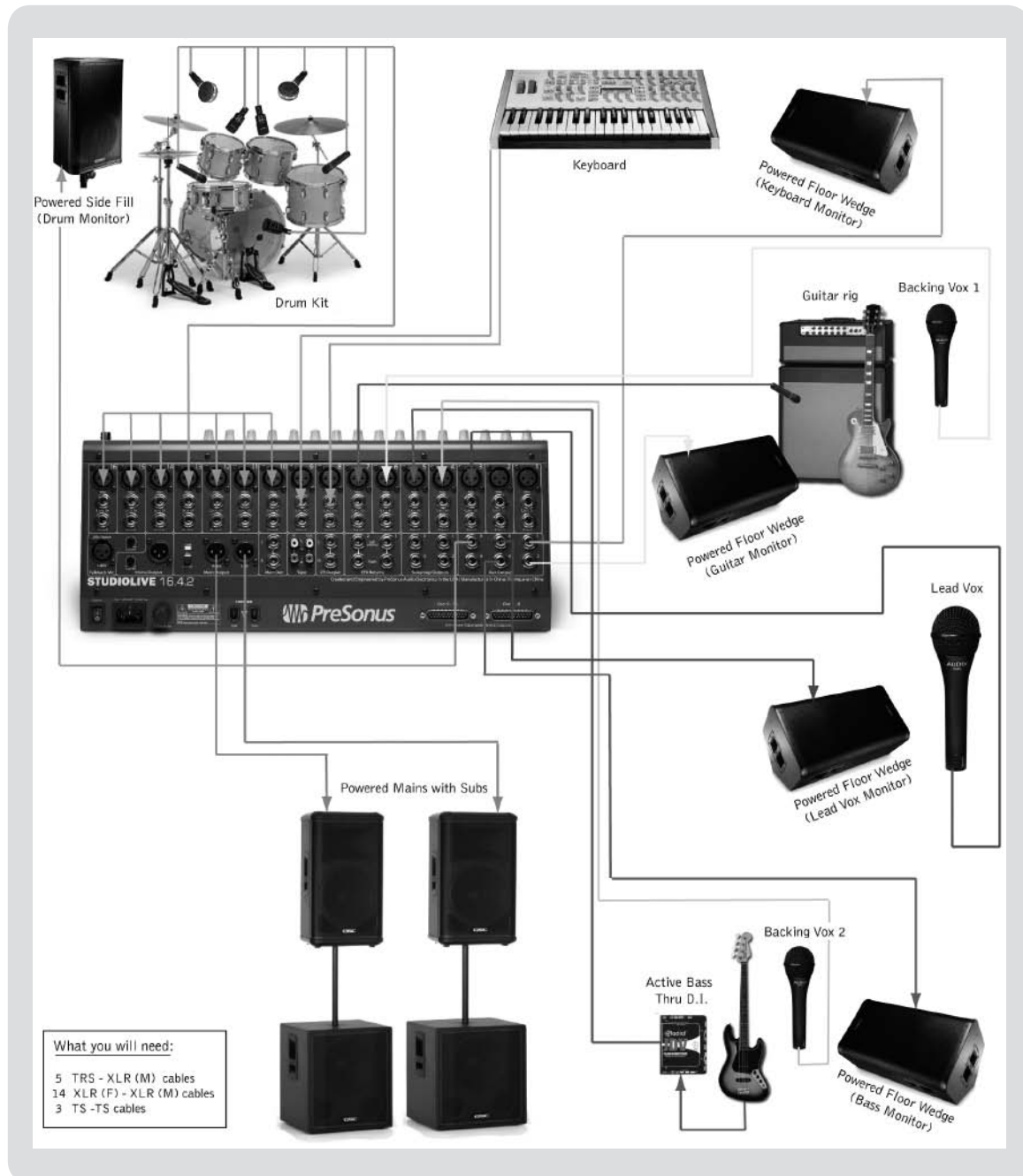
The Patch Bay



- **FireWire Ports.** There are two standard 6-pin FireWire 400 ports on the back of the StudioLive. Either port can be used to connect your StudioLive to a FireWire port on your computer. If your computer has a 4-pin connector (commonly found on laptops), you will need to purchase a 4-to-6-pin adaptor or cable. These adaptors and cables can be found at your local computer supply store. Use the second FireWire port to connect additional FireWire devices (such as external hard drives) to your computer or to daisy chain additional StudioLive mixers. You can connect two StudioLive mixers to a computer for recording, or you can daisy-chain up to eight StudioLive mixers to create a standalone large-format mixing console.
- **2 Amp Fuse.** This is the StudioLive's fuse housing. Your StudioLive uses a 5 mm x 20 mm, 250 VAC, fast-acting fuse.
- **S/PDIF Output.** By default, the S/PDIF output receives the same signal as the main outputs, so no activation is necessary. However, any of buses that can be routed to the auxiliary FireWire returns, can be routed to the S/PDIF output, either through the System menu in the Digital Effects | Master Control section or in the StudioLive Control Panel (see Sections 2.10.4 and 3.3.3 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.
- **Power-Adapter Input.** This is where you plug in the provided IEC power cable.

2.2

Hookup Diagram: On Stage with the StudioLive



2.3 The Fat Channel



- **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.

The heart of the StudioLive is the revolutionary Fat Channel. 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 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:

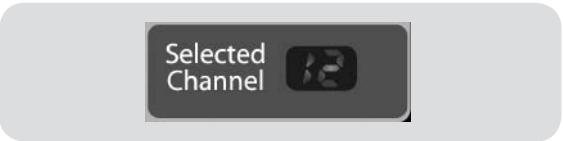
- Add dynamics processing and EQ to every input and output
- Create sends and effects mixes for all 6 analog aux sends and both internal effects buses
- Assign subgroup and main routing
- Meter inputs, post-dynamic outputs, and gain reduction for all 16 channels
- Meter aux send outputs
- Copy, save, and load mix scenes
- Recall your fader position for stored mixes

2.3.1 The Select Button

All around the StudioLive, you will see Select buttons. There is a Select button on each of the 16 inputs, each of the 6 analog aux sends, both of the internal effects buses, each of the 4 subgroups, the two auxiliary inputs, and last, but certainly not least, the main output bus. Each of these buttons serves exactly the same purpose: to access the Fat Channel parameters for its channel or bus.



In the lower right corner of the Fat Channel, you will find an LED read out. The currently selected channel will always be displayed here. (Numbers 1-16 indicate one of the 16 inputs channels is selected, S1-S4 indicates Subgroups 1-4, MA indicates the Main bus, A1-A6 indicates Aux 1-6, A7 and A8 indicate EFX A and EFX B, and F1 and F2 indicate Aux inputs A and B.)



In addition, two meters—part of a set of seven 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 for the selected channel. It is important to mention that these meters are only active when one of the 16 input channels or an aux bus is selected.



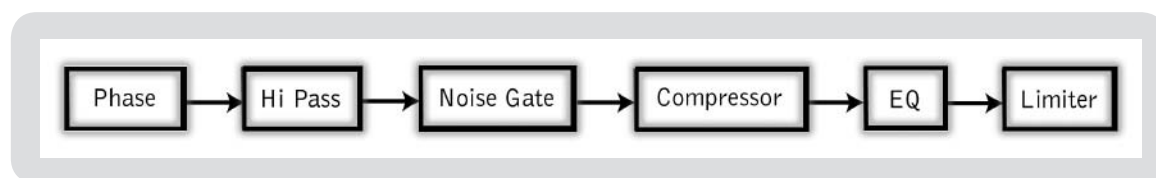
It should be noted that while the Noise Gate, Compressor, EQ, and Limiter are available on every input and bus, the phase reverse and hi-pass filter are only available on the 16 inputs. In addition, other inputs and buses without Select buttons are available to route to the auxiliary FireWire returns (see section 3.3.1 for more details). The following table provides a quick guide to the processing that is available for each bus in the StudioLive:

2.3.1 The Select Button

Bus	Phase Reverse	Hi-Pass Filter	Noise Gate	Compressor	EQ	Limiter	FireWire send
Inputs (Ch 1-16)	✓	✓	✓	✓	✓	✓	✓
Subgroups			✓	✓	✓	✓	✓
Main Out L/R			✓	✓	✓	✓	✓
Aux Sends 1-6			✓	✓	✓	✓	✓
Internal FX Sends 1-2			✓	✓	✓	✓	✓
External FX Returns 1-2			✓	✓	✓	✓	✓
Tape Input							✓
Talkback Mic							✓
Solo Bus							✓
Monitor Bus							✓

2.3.2 Dynamics Processing and EQ

The main function of the Fat Channel is to provide dynamics processing and filtering for every input and output on your StudioLive. The rotary encoders work in conjunction with the meters directly above them to adjust the dynamics processing and EQ. The Fat Channel's processing section consists of five parts: Hi Pass filter, Noise Gate, Compressor, Limiter, and semi-parametric EQ. Each can be turned on or off and controlled separately. The signal flows as follows:



Phase Reverse button



Reverses the Phase of the Selected Channel.

Push this button to invert the phase of the selected channel's signal (that is, to alter the phase by 180°). The button will illuminate, indicating that the phase reverse is active. The Phase Reverse button can be used to correct audio signals that are out of phase and cancelling each other out.

Phase reverse is only available on the 16 channels of the input bus.

High Pass filter



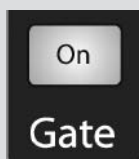
Engages the High Pass Filter.

The High Pass filter section consists of an encoder and a meter. You will notice that there is a frequency range to the left of the meter. The high-pass 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.

Remember, that all frequencies below a high-pass filter's threshold are attenuated. See section 5.3.1 for more details.

The high-pass filter is only available on the 16 channels of the input bus.

Gate On/Off button



Turns the Gate On and Off for the Selected Channel.

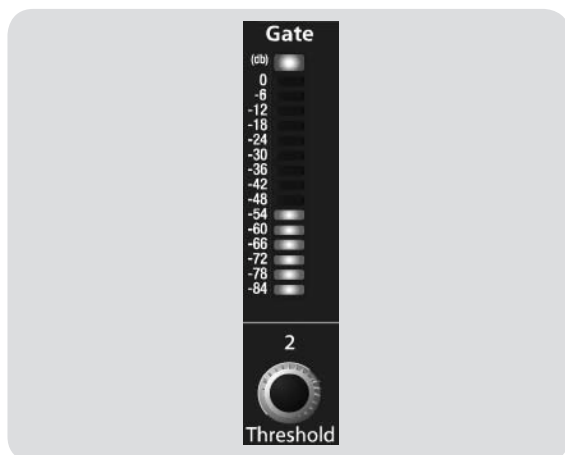
This button engages and disengages the gate for the selected channel. It will illuminate to indicate that the gate has been enabled.

The gate is available for all input and output buses.

2.3.2

Dynamics Processing and EQ

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 threshold determines the level at which the gate will open. Essentially, all signals above the threshold setting are passed through unaffected. - You can set the threshold from 0 to -84 dB.

Gate Release



Sets and Displays the Rate 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 rate can be set from 2 to 0.05 seconds.

Remember, 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" in 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 button



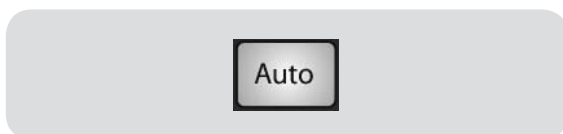
Turns the Compressor On and Off for the Selected Channel or Output Bus.

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 input and output buses.

2.3.2 Dynamics Processing and EQ

Turns the Auto Button On



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.

Soft Knee/Hard 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 onset of gain reduction occurs gradually after the signal has exceeded the threshold.

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 threshold of the compressor for the selected channel or output bus. When the signal's amplitude (level) exceeds the threshold setting, the compressor is engaged. Turning the knob counterclockwise lowers the threshold, so compression begins at a lower amplitude. The threshold can be set from -56 to 0 dB.

Compression Ratio



Sets and Displays the Compression Ratio Setting 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.

2.3.2 Dynamics Processing and EQ

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 nonharmonic 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

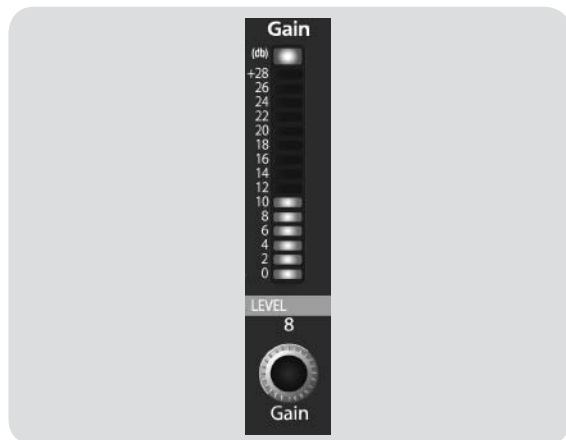


Sets and Displays the Compressor Release Setting for the Selected Input Channel or Output Bus.

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 40 to 1000 milliseconds.

Remember, 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.

2.3.2 Dynamics Processing and EQ

EQ Low Band On/Off Button



Activates Control for the Low Band EQ for the Selected Input or Output Bus.

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

The Low Band EQ is available for all input and output buses.

Low Band EQ Frequency Control



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

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 Band 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

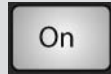


Turns on the Low Shelf EQ for the Selected Input or Output Bus.

When the Shelf button is not engaged, the Low band is a semi-parametric EQ. 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. It's like a bass-control knob on a stereo. In this mode, the Center Frequency control selects the shelving frequency.

2.3.2 Dynamics Processing and EQ

Low Mid EQ Button



Activates Controls for the Low Mid Band EQ for the Selected Input or Output Bus.

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

The Low Mid Band EQ is available for all input and output buses.

Low Mid Band EQ Frequency Control



Sets and Displays the Center Frequency of the Low Mid Band 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.

Hi Q Button



Enables a Narrow Bandwidth for the Low Mid Band EQ on the Selected Input or Output.

Q is the ratio of the EQ band's center frequency to its bandwidth. With a constant center frequency, higher Q values indicate a narrower bandwidth, so Q is often equated with bandwidth. By default, the Q is set to a value of 0.55. When the Hi Q button is engaged, the Q setting will be increased to 2.0, thus narrowing the bandwidth to provide more precise control.

Low Mid Band EQ Gain Control

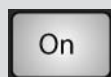


Sets and Displays the Gain Attenuation or Boost of the Center Frequency for the Low Mid Band.

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.

2.3.2 Dynamics Processing and EQ

High Mid EQ Button



Activates Controls for the High Mid Band EQ for the Selected Input or Output Bus.

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

The High Mid Band EQ is available for all input and output buses.

High Mid Band EQ Frequency Control



Sets and Displays the Center Frequency of the High Mid Band EQ.

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

Hi Q Button



Enables a Narrow Bandwidth for the High Mid Band EQ on the Selected Input or Output.

Q is the ratio of the EQ band's center frequency to its bandwidth. With a constant center frequency, higher Q values indicate a narrower bandwidth. By default, the High Mid EQ is set to a low Q = 0.55. When the Hi Q button is engaged, the bandwidth will narrow to 2.0 to provide more exact control.

High Mid Band EQ Gain Control

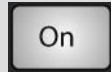


Sets and Displays the Gain Attenuation or Boost at the Center Frequency.

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.

2.3.2 Dynamics Processing and EQ

High Band EQ On/Off Button



Activates Control for the High Band EQ for the Selected Input or Output Bus.

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

The High Band EQ is available for all input and output buses.

High Band EQ Frequency Control



Sets and Displays the Center Frequency of the High Band 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 Band EQ Gain Control



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

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

High Shelf EQ Button

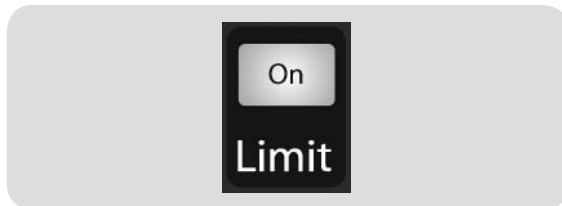


Turns on the High Shelf EQ for the Selected Input or Output Bus.

When the Shelf button is not engaged, the High band is a semi-parametric EQ. Enabling the Shelf button turns the High band into a high shelving EQ that alters, by a fixed amount, a band of low frequencies at and above a user-selected shelving frequency. It's like a treble-control knob on a stereo. In this mode, the Center Frequency control selects the shelving frequency. .

2.3.2 Dynamics Processing and EQ

Limiter On/Off Button



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

When the limiter is engaged the button will illuminate. The threshold for the limiter is set to 0 dBFS. The Ratio is ∞ :1.

2.3.3 Panning and Stereo Link

The Pan Control for each Input or 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 stereo pair, the LED display will automatically change to stereo pan.



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

- Channels 1 & 2
- Channels 3 & 4
- Channels 5 & 6
- Channels 7 & 8
- Channels 9 & 10
- Channels 11 & 12
- Channels 13 & 14
- Channels 15 & 16
- Aux 1 & Aux 2
- Aux 3 & Aux 4
- Aux 5 & Aux 6
- Subgroups 1 & 2
- Subgroups 3 & 4

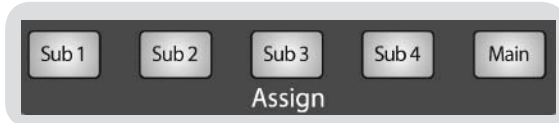
A stereo link can be enabled when either channel in the pair is selected. When the stereo link button is illuminated all the dynamics settings, subgroup assignments, and main assignments are pasted onto the other channel in the pair. Note that this is a nondestructive paste. When the Link button is disengaged, the other channel's settings will be restored to its previous setting. For instance, if Channel 8 is selected when the Stereo Link button is engaged, all of Channel 8's settings will be copied onto Channel 7. If Channel 7 is selected when the Stereo Link button is engaged, Channel 7's settings will be copied onto Channel 8. Because the settings are copied nondestructively, it is possible to A/B dynamics settings with the touch of two buttons. Whichever channel is selected when the link button is engaged will be the Link Master. When either channel in the stereo link is selected, both channels' Select buttons will illuminate, but the Link Master's ID number will be displayed in the Selected Channel LED read-out in the Fat Channel.



2.3.4 Output Assignments

Output assignments are set within the Fat Channel. It should be noted that the StudioLive will prevent you from creating a feedback loop. Subgroups can only be assigned to the main outs and the six aux sends 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 the main outputs. This includes the 16 main inputs and the 2 auxiliary inputs. The internal effects returns can also be assigned to any or all of the subgroups and the main outputs.



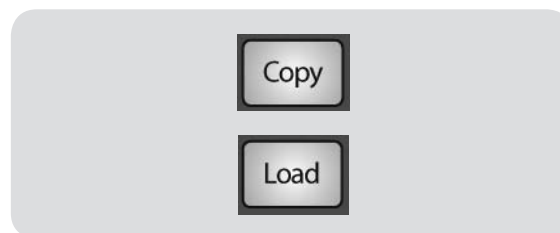
The Fat Channel also gives you the option of sending just the unprocessed audio to your computer or including the Fat Channel settings in the recorded signal. When the Dig Out button is enabled, the signal being sent to the FireWire bus is post-EQ and post-dynamics processing; the button will illuminate to indicate this signal flow. When the button is disabled, the signal being sent to the FireWire bus is pre-Fat Channel. The Dig Out button is only optional when one of the 16 inputs is selected. The Subgroups, Main Output, Aux bus, Aux Ins and Internal Effects Returns automatically send their signals post Fat Channel dynamics and EQ. All FireWire 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 Section 3.



2.3.5 Copy, Save, Load

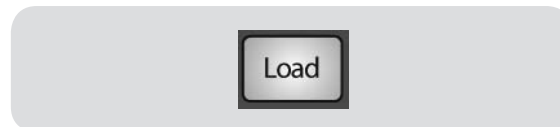
Every setting in the Fat Channel can be copied from one to channel to another and can be saved and stored as a user preset to be recalled later.

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.

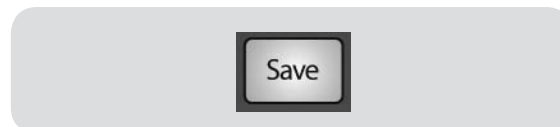


To paste the current channel's Fat Channel setting to another channel or bus, simply press that channel's Select button. It will stop flashing and illuminate. After you have selected every channel to which you want the settings pasted, press the Load button. The StudioLive will return to its normal state, indicating that the Fat Channel settings have been successfully pasted.

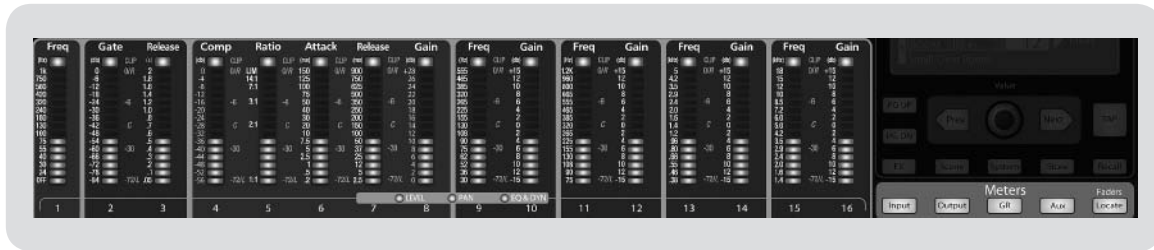
The Load button can also be used to recall saved settings and presets. For complete recall instructions, see Section 2.10.3



You can save your Fat Channel settings for future use. To store the currently selected channel's Fat Channel settings, press the Save button. For complete channel-preset storage instructions, see Section 2.10.3.



2.4 Metering



The StudioLive offers flexible metering at the touch of a button. The 16 meters in the Fat Channel section can monitor:

- All 16 inputs, post-gain and pre-dynamics, pre-EQ, and pre-fader
- All 16 inputs, post-dynamics, post-EQ, and post-fader
- The gain reduction for all 16 inputs
- The output volume of each of the 6 Aux Sends

Last, but certainly not least, the meters can be used to recall the fader settings for a saved scene.

2.4.1 Metering Controls

Below the System section of the StudioLive, you will find the Meters section. Each of these buttons are toggle switches, meaning you can turn them on or off by simply pressing them and then pressing them again. The meter state can also be changed by pressing another button in the Meter section, any Select button on the StudioLive, or a Mix or Mix/Pan button in the Aux section.

It is important to mention that the meters simply overlay on the previously selected Fat Channel state. For instance, if you have Channel 16 selected and then press the Output button in the Meter section, the knobs and buttons in the Fat Channel section will still be active, and any changes made will be applied even though they will not be reflected in the Meter section. The advantage of this is that you can make adjustments in the Fat Channel, press a Meter button, and monitor your entire mix, then press the same button to return to setting up your Fat Channel selection.

Input Metering Button

A rectangular button with the word "Input" in the center.

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

A rectangular button with the word "Output" in the center.

Turns Post-Fader Output Metering On and 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 Meter Button

A rectangular button with the letters "GR" in the center.

Turns Gain Reduction Metering On and Off.

Displays the gain reduction of the input bus. Meters are one to one (Meter 1 shows the gain reduction of Channel 1, etc.).

Aux Metering Button

A rectangular button with the word "Aux" in the center.

Turns Aux Bus Master Out Metering On and Off.

Displays the output level of each of the aux sends.

- Meters 1, 3, 5, 7, 9, and 11 display the output levels of Aux Sends 1-6, respectively.
- Meters 13 and 15 display the output level of EFX A and EFX B.

Fader Locate Button

A rectangular button with the words "Faders" and "Locate" stacked vertically in the center.

Turns Fader-Recall Metering On and Off.

Displays the fader position of the stored scene. When recalling a fader position, adjust the fader until only the center LED is visible in its meter.

2.5 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 FireWire bus, just as if it were an analog input. This lets you incorporate digital audio tracks into the main mix as well as inserting plug-in effects and software instruments from your audio program.

2.5.1 Input Channel Controls

Trim Control



Adjusts the Input Gain Level.

The Trim control adjusts the gain of the channel's analog input.

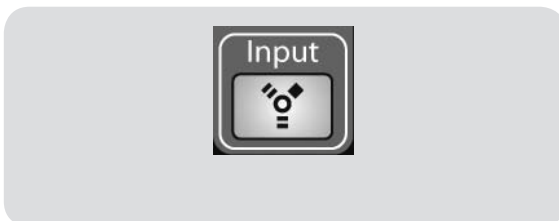
Phantom Power Button



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. For more information on phantom power and the microphones that require it, please review Section 5.1

FireWire Input Select Switch



Turns FireWire Playback Streaming On/Off.

The FireWire Input button routes a playback stream from your recording 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 back on mixer channel 3, simply route that track in your recording 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 section (3.3.2).

Do not engage this button when your StudioLive is not connected and synced to a computer.

2.5.1 Input Channel Controls

Input-Channel Select Button



Enables Fat Channel Processing and Routing.

As previously described in section 2.3.1, 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.

Solo Button



Turns Soloing On and Off.

This button will solo its channel to the main outputs or to the monitor outputs, depending on whether PFL or SIP is selected in the Solo bus section. Please review Section 2.10 for details.

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

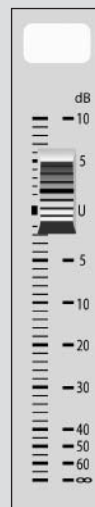
Mute Button



Turns Muting On and Off.

This button mutes its channel. It will illuminate red when the channel is muted. When a mute button is engaged, its channel will be muted in all of its assigned outputs (Subgroups, Mains, and Aux Sends).

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.

2.6 The Aux Bus

The StudioLive has six analog aux sends and two internal effects sends. The aux sends are mono; however, two aux sends can be linked to create a stereo bus. Section 4.5 discusses how to use these aux buses for monitor and effects mixing.

2.6.1 Analog Aux Sends

Solo Button



Turns Soloing On and Off.

This button will solo the aux bus to the main outputs or to the monitor outputs, depending on whether PFL or SIP is selected in the Solo bus section. Please review Section 5.8 for details.

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

Post-Fader Send Button



Enables and Disables Post-Fader Send.

When this button is enabled, the send levels to that aux send will be post-Fat Channel and post-fader, and therefore are affected by fader settings. If the button is disabled, by default, the aux buses use the Pre 1 setting, and all channels on your StudioLive are sent to the aux bus after the phase reverse switch, high pass filter, and noise gate but before the compressor, EQ, limiter, and fader. However, in the System Menu you can select the Pre 2 setting, to move the send after all Fat Channel processing but before the fader. Please review Section 2.10.4 for details.

Output Level Control



Adjusts the Master Output Level of the Aux Send.

This knob adjusts the master output level of the aux send.

2.6.1 Analog Aux Sends

Aux Send Select Button



Enables Fat Channel Viewing.

As previously described in section 2.3.1, the Select button routes its aux send through the Fat Channel, allowing you to add dynamics processing and EQ.

Mix Button



Enables Aux Bus Mixing and Metering in the Fat Channel.

When this button is enabled, the 16 encoders in the Fat Channel become the aux-send level controls for each of their respective input channels. The meters will display the send level of each of the input channels.

Mix/Pan Button

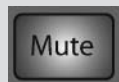


Enables Pan Control and Metering in the Fat Channel (Stereo Send Mode Only).

On Aux Sends 2, 4, and 6, the Mix button becomes the Pan control when two sends are linked. When this button is enabled, the 16 encoders in the Fat Channel become the pan controls for each of their respective input channels. The meters will display the pan setting of each of the input channels. For more information of stereo linking, please review Section 2.3.3

2.6.2 Internal Effects Send Controls

Mute Button



Mutes and Unmutes the Internal Effects Bus.

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

Post Fader Send Button



Enables and Disables Post Fader Send.

By default, this button is enabled, and the send levels to the effects buses are post-EQ, post-dynamics processing, and post-fader, and therefore are affected by fader settings. If the button is disabled, by default, the EFX sends use the Pre 2 setting, in which the sends are post-EQ and post-dynamics but pre-fader. However, in the System Menu, you can switch the EFX sends to the Pre 1 setting, in which the sends are pre-Fat Channel dynamics processing and EQ and pre-fader. Please review Section 2.10.4 for details.

Output Level Control



Adjusts the Master Level of the Effects Send.

This knob adjusts the master level of the internal effects send.

Internal Effects Send Select Button



Enables Fat Channel Viewing.

As described in section 2.3.1, the Select button routes its aux send through the Fat Channel allowing you to add dynamics processing and EQ, as well as routing the aux send to outputs.

Mix Button



Enables Aux Bus Mixing and Metering in the Fat Channel.

When this button is enabled, the 16 encoders in the Fat Channel become the aux-send level controls for each of their respective input channels. The meters will display the send level of each of the input channels.

2.7 Subgroups

As explained in Section 5.4, subgroup mixing is very useful for grouping channels for master level control of a submix (e.g., drums) and for providing a fader level for effects.

Subgroup Channel Select Button



Enables Fat Channel Viewing.

As previously described in section 2.3.1, the Select button routes its channel through the Fat Channel, allowing you to add dynamics processing, EQ, panning, etc.

Solo Button



Turns Soloing On or Off.

This button will solo its channel either to the main outputs or the monitor outputs depending on whether PFL or SIP is selected in the Solo bus section. Please review Section 4.8 for more details.

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

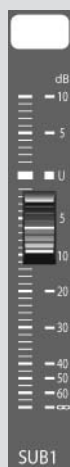
Mute Button



Turns Muting On or Off.

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

Channel Fader



Controls the Level of the Subgroup Bus.

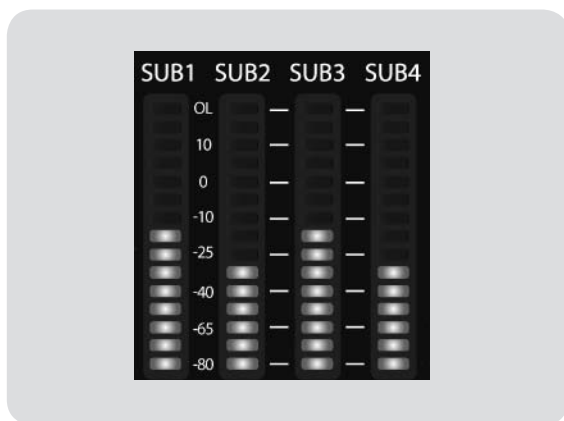
The fader controls the overall volume of the signal on its subgroup output.

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.

2.7 Subgroups

Subgroup Meters



Displays 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.

2.8

Main Output Bus

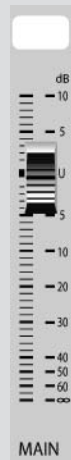
Main Select Button



Enables Fat Channel Viewing.

As previously described in section 2.3.1, the Select button routes its channel through the Fat Channel, allowing you to add dynamics processing, EQ, panning, etc.

Main Fader



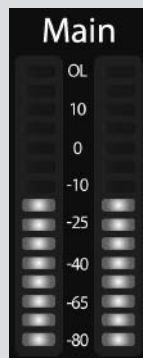
Controls the Level of the Main Output.

The fader controls the overall level of the main stereo output.

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.

Main Bus Meters



Displays 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.

2.9 Master Section



The Master section of the StudioLive controls the monitoring, talkback output, solo bus, and aux and tape inputs.

2.9.1 **Aux Inputs A and B**

The StudioLive features an internal effects processor that allows you to load two different effects at the same time, but it also allows you to patch return processed signals from an external effects processor, using the Aux bus. Section 5.5.2 goes into the connections and procedures in greater detail.

Aux Input Select Button



Enables Fat Channel Viewing.

As previously described in section 2.3.1, the Select button routes its channel through the Fat Channel, allowing you to add dynamics processing, EQ, and panning, as well as enable output assignments.

Aux Input Level Control



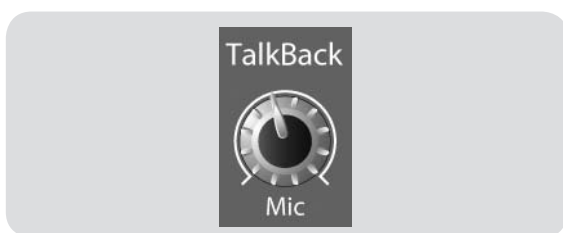
Controls the Signal Level of the Aux Input.

This knob controls the overall volume of the of the Aux Input signal.

2.9.2 Talkback System

The StudioLive features a Talkback microphone input on the back panel. This can be routed to each of the aux outputs as well as to the mains. It is important to note that the aux outputs are in stereo pairs in this section. For example, if you are using Aux 3 and 4 as mono monitor sends to the bass player's floor wedge and to the keyboard player's in-ear monitors, the talkback signal will be sent to both monitors—so don't say anything you wouldn't want both to hear!

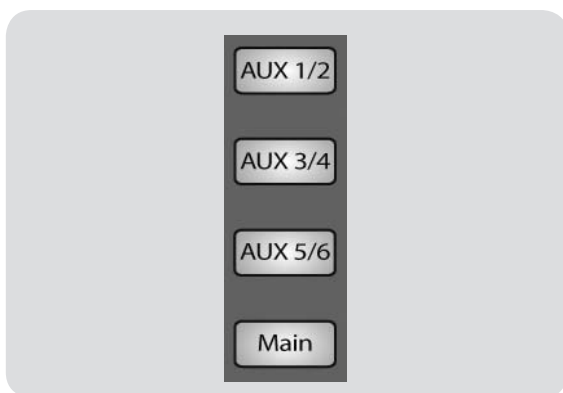
Talkback Mic Level



Sets the Input Gain of the Talkback Mic.

This knob adjusts the gain of the external talkback mic.

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.

Talkback Button



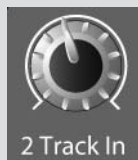
Turns the Talkback Mic On and Off.

This latching button turns the talkback mic on and off. It will illuminate to indicate that the talkback mic is active.

2.9.3 **2 Track In**

The StudioLive allows you to patch in an analog input or a playback stream from your recording application. This enables you to compare audio sources during mixdown or patch in intermission music between bands without using any of your 16 input channels. This is a summing bus, so both the external tape input and the two-track FireWire 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 FireWire return.

Tape Input On/Off Button



Sends the Tape Input to the Mains.

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.

FireWire Source On/Off Button



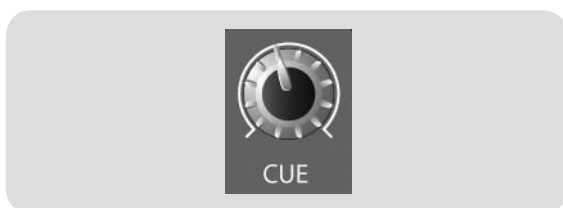
Sends the Master Out from the Audio Application to the Mains.

This button routes the main left/right FireWire returns (Outs 17-18) to the main outputs. It will illuminate to indicate that the main FireWire return is being sent to the main outputs of the StudioLive. To disable the routing, simply press the button again.

2.9.4 Solo Bus

The StudioLive features a solo bus for flexible monitoring. In this way, you can fine-tune a vocal compressor in your control-room monitors or headphones while the rest of the band is warming up, or audition an aux mix for more accurate adjustments. For more information on using the Solo bus, please review Section 5.8.

Cue Mix Volume Control



Adjusts the Overall Volume of the Solo Bus.

This knob adjusts the overall level for the Solo bus.

PFL/AFL Toggle Button



Enables PFL Solo Mode.

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. The Aux bus soloing is always PFL with or without this mode engaged.

SIP (Solo In Place) On/Off Button



Enables Solo In Place Mode

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 into 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.

It is important to note that when SIP is engaged, channel mutes will only apply to the Subgroup & Main assignments. Unlike independent muting, SIP does not mute input channels in Aux Send mixes.

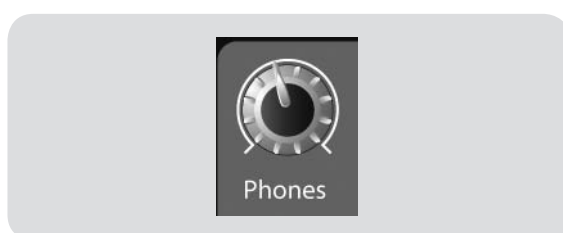
2.9.5 Monitor Bus

The StudioLive features a headphone output as well as control-room outputs, giving you the ability to monitor multiple sources on the StudioLive. The Monitor bus on the StudioLive allows you to monitor the main outputs, Solo bus, main FireWire return from your computer, and the stereo analog tape input. Because the Monitor Bus is a summing amp, you can even monitor the World Series on your headphones while still running sound at a show.

Headphone Output Level Control

Adjusts the Overall Level of the Headphone Output.

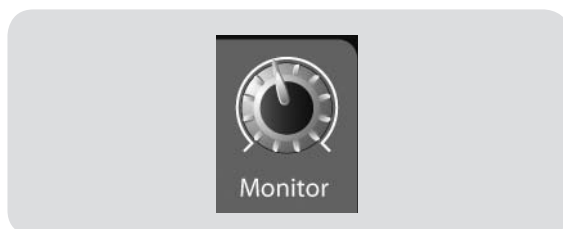
This knob adjusts the overall level for the headphone output.



Control-Room Monitor Level Control

Adjusts the Overall Level of the Control Room Monitor Outputs.

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



Solo Bus Monitor Button

Assigns the Solo Bus to the Monitor Outputs.

The Solo Bus Monitor button patches any soloed channel, subgroup, or aux send 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.).



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.



2.9.5 **Monitor Bus**

FireWire Monitor Button



Assigns the Main Left/Right FireWire Return to the Monitor Bus.

The FireWire Monitor button patches the main left/right FireWire return (that is, the signal from your audio software's master outputs) to the monitor bus.

Main Mix Monitor Button



Assigns the Main Mix to the Monitor Bus.

The Main Mix Monitor button patches the same signal that is being sent out of the main outs to the Monitor bus. This signal is always pre-fader.

2.10

Digital Effects | Master Control

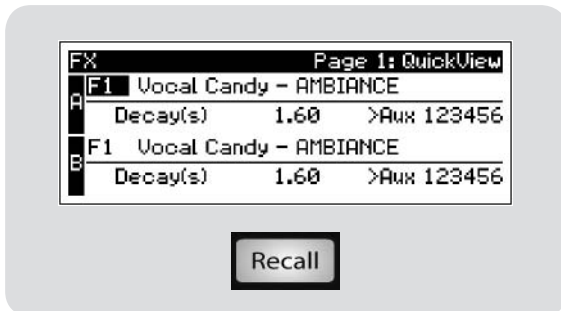
From the Digital Effects | Master Control section, you can select and change the parameters of the two internal effects processors, as well as store and recall every setting on your StudioLive at the touch of a button. Because almost all of the StudioLive's features are controlled from the mixing surface (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.

2.10.1 The FX Menu

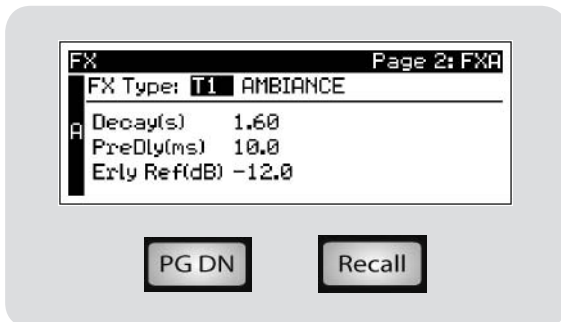
The StudioLive features two internal effects processors. Each processor can access the StudioLive's selection of high-quality reverbs and delays. As described in Section 2.6.2, each of these effects can be routed to any of the subgroups, the aux bus, or the main outputs. To access the effects library and adjust each parameter, press the FX button in the Master Control section.



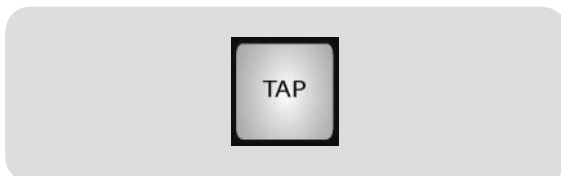
The first page of the FX menu is the QuickView screen. It displays both of the effects assigned to the internal effects buses, as well as the main parameter for each and to which Aux buses the effect is being routed. Effect A is assigned to EFX A bus, and Effect B is assigned to EFX B bus. Use the Next and Prev buttons to navigate through the screen. To change a parameter, use the Value encoder directly beneath the LCD screen. The color will invert for each parameter when it is selected for modification. The Next button will scroll through this screen in the following order: FX A library selection, FX A main parameter, FX B library selection, FX B main parameter. When choosing your effects 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 page of the FX menu. Pages 2 and 3 of the FX menu display the rest of the parameters for FX A and FX B, respectively. These parameters will change depending on what 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. If you would like to change the effect from, for example, a Large Hall reverb to a Room Ambiance reverb, press the Recall button to load the effect, and then change the various parameters to taste. Press the Page Up button again to move to the full parameter view of FX B.



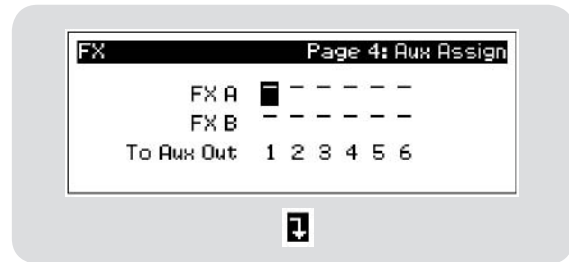
When a delay algorithm is chosen, you will notice two parameters: Time (ms) and Time X. Time is the delay time, discussed in Section 5.6.2, which is the time between the source sound and the echo. This can either be adjusted manually with the Value encoder, or you can use the Tap button to enter a delay time in tempo with the music being played. 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



2.10.1 The FX Menu

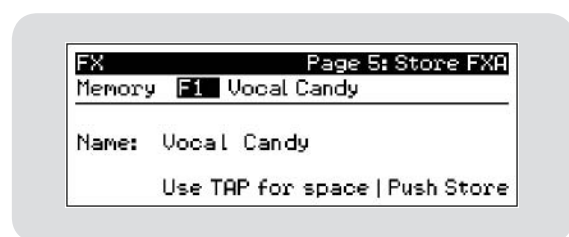
time. You do not have to select the Time parameter in order to use the Tap button. However, you have to be on the Extended Parameters page (page 2 for FX A, page 3 for FX B) or you must select the Time parameter for the desired effect on the first screen of the FX menu. The Tap button does not function when any other page is active.

Page 4 of the Effects menu is the Aux Assign screen. This screen allows you to route the internal effects buses to any or all of your aux sends. Use the Next and Prev buttons to navigate through the screen. When you reach the aux bus to which you want to route the effect, turn the Value encoder clockwise. A down arrow will appear, indicating the effect has been routed to an aux bus, in addition to any routing set in the Fat Channel. The Aux bus routing will be displayed for each effect on the QuickView screen.



Because only one mix can be created for each effect, you may wish to consider dedicating one of your internal EFX buses to your monitor mixes and the other one to your main mix. In cases where it is necessary for the performer to hear the same effect as the audience (for example, a vocalist needing to hear a tap delay so that they can interact with it in their performance), you would obviously want to route the same effects chain to their monitor that you have routed to the mains. In this instance, you may also wish to use the subgroup-mixing technique mentioned in Section 4.4.2 to control the amount of the effects being sent to the audience, while using the Output knob for the internal EFX A and B buses to control the amount of the effects being sent to the monitor(s).

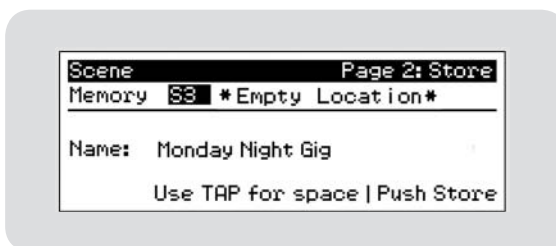
Pages 5 and 6 allow you to store your changes to the same or to a new location and customize the name of your creation for FX A and for FX B respectively. Use the Value Encoder to change the library location to which you will store your new effects preset, unless you wish to overwrite the currently selected preset location. Press the Next button to navigate to the first letter of the preset name. Turn the Value Encoder clockwise or counter-clockwise to change the letter. The StudioLive allows you to customize the name with upper- and lowercase letters, as well as a selection of numerals and punctuation marks. You can press the Tap button to quickly insert a space. Once you are satisfied with your changes, press the Store button, which will illuminate while the effects preset is being written to the StudioLive's internal memory. Once the preset is saved, the Store button will return to its unlit status.



2.10.2 Creating & Recalling a Scene

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 FireWire playback stream). 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 applications. For example, in the studio, saving and recalling a scene allows you to move on 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 sound check and recall the mix when that band goes onstage. You also can save custom mixes for each venue that a band plays repeatedly.

To save a scene, press the Scene button and Page Down to the second screen. The memory locations will be selected. Use the Value Encoder to scroll to a free location in the Scene library. 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, as well as a selection of numerals and punctuation marks. Press the Tap button to quickly insert a space. Continue this process until you are satisfied with your changes, 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.

Located at position S1 is a scene named Zero Out (Board Reset). This scene cannot be overwritten and returns your StudioLive to its factory settings. All you have to do it lower the faders and return all trim knobs and output volume knobs (Solo bus, 2 Track In, Phones, and Monitor) to their lowest position. Your StudioLive will be zeroed out as follows:

INPUTS & BUSES	SETTINGS							MODE
	ASSIGN	PAN	SOLO	MUTE	SELECT	LINK	SENDS	
INPUT CHANNELS	MAINS	C	OFF	OFF	CH1	OFF	NULL	
SUBGROUPS	NULL	C	OFF	OFF	NULL	OFF		
AUXES 1-6	NULL	C	OFF		NULL	OFF		PRE
FX A / FX B	MAINS	C		OFF	NULL			BOTH SET TO F1
AUX IN A & B	NULL	C		OFF	NULL			
TALKBACK	NULL							
MAIN BUS		C			NULL			
SOLO BUS								AFL
MONITOR BUS								MAIN
TAPE IN								NULL
METERS								NULL
MASTER CONTROL								FX

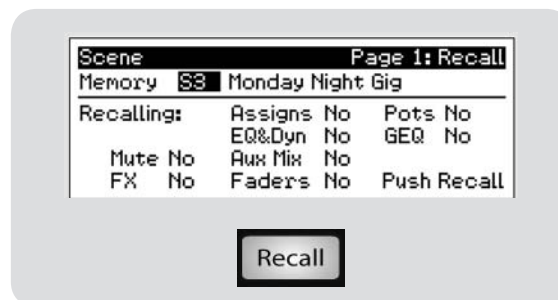
2.10.2 Creating & Recalling a Scene

The Fat Channel will be restored to the same setting for every input and output on your StudioLive. Each of the dynamics as well as the four bands of the EQ will be turned off. Their parameters will be set as follows:

FAT CHANNEL PARAMETERS											
HIPASS	NOISE GATE		LIMIT	COMPRESSOR		EQ	LOW	L.MID	H.MID	HIGH	
OFF	THR	N/A	OFF	THR	0dB	P/S	shelf	n/a	n/a	shelf	
	REL	.05 s		RAT	2:1	H/L	n/a	low	n/a	low	
				ATK	20 ms	FREQ	130Hz	320Hz	1.4kHz	5kHz	
				REL	150ms	GAIN	0dB	0dB	0dB	0dB	
				GAIN	0dB						
				KNEE	hard						

Before beginning any new mixing situation, it is always recommended to 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.

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

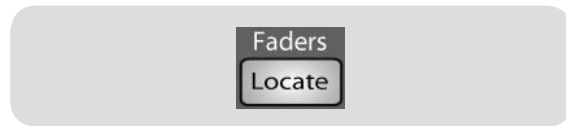


- Mute: All channel and bus mutes (Channels 1-16, Subgroups 1-4, FXA, FXB)
- FX: All parameters for the internal effects assigned to FXA and FXB
- Assigns: All output and bus routing (Mains, Subgroups, Stereo Linking, Solo and Monitor bus assignments, and Internal Effects to Aux Bus routing)
- EQ & Dyn: All Fat Channel dynamics processing and filter parameters (Phase, High Pass, Noise Gate, Compressor, EQ, and Limiter) for every channel and bus
- Aux Mix: All aux mixes (analog Aux Mixes 1-6 as well as FXA and FXB mixes)
- Faders: All fader positions
- Pots: All knob positions (Subgroups 1-4, and Main Output faders and Aux 1-6 Outputs, FXA and EFXB Levels, Cue, 2 Track In, Phones, Monitor, and Aux Input A and Aux Input B knobs)
- GEQ: Graphic EQ settings

2.10.2 Creating & Recalling a Scene

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 the center LED is illuminated in each meter. The subgroup and main meters at the top right of the StudioLive will display the recalled positions for their respective faders. As long as you remain in Fader Locate mode, the faders on your StudioLive will not be active. The level of the audio passing through them will be set according to the stored fader position, not the current location. Once you have recalled each of the faders to their Scene location, press the Locate button again. This will take you out of Fader Locate mode and reactivate your faders so that they control the level of their channels and buses.



If you enable knob positions as a part of your scene recall, all of the digital knobs (Aux 1-6 Outputs, FXA and EFXB Levels, Cue, 2 Track In, Phones, Monitor, 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 and recall a scene, 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.

A Quick Note About the AutoStore Feature

It is not necessary to create a scene for your StudioLive in order to remember its settings when you power it down. Periodically, you will notice the Store button flash once, indicating that the current settings on your StudioLive have been saved to internal memory. The StudioLive will take a snapshot of the current position of every parameter on the mixer whenever no changes have been made for ten seconds. If you wish to ensure that your StudioLive will hold its current settings before you power down, simply wait for a few seconds until the Store button flashes. This is your indication that it is now safe to turn off your StudioLive. The next time you turn it on, all of your settings will be recalled automatically.

Please note: AutoStore does not change the settings for a recalled scene that has been altered. 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.

2.10.3 Saving and Loading Channel Presets

The StudioLive comes with a suite of channel-strip presets created by professional PreSonus users and endorsed artists and engineers. 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.

To load a preset to any channel on the StudioLive, first press the Select button for the desired channel. From the Fat Channel, press the Load button. You will notice that LCD now displays the Channel Preset Load menu.



The Channel Preset Load menu always displays the selected channel into which the preset will be loaded. 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.

Channel Preset Page 1: Load
Memory P1 DRM: Punchy Kick
Load To: Channel Strip 1
Push Recall

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. You will notice that the LCD will display the Channel Preset Save menu.



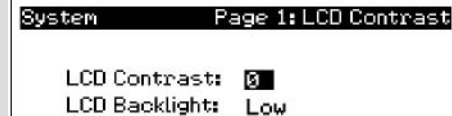
To begin, use the Value encoder to scroll to an empty position in the Channel Preset library. Press the Next button to navigate to the category location. Create the category in which your preset would fit (DRM, VOX, GTR, etc.). Press the Next button again to navigate to the first letter of the preset name. 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. 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.

Channel Preset Page 2: Save
Memory P1 DRM: Punchy Kick
Name: DRM: Punchy Kick
Use TAP for space | Push Store

2.10.4 The System Menu

In most digital consoles 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 very little to do with mixing a show.

Press the System button to access the System menu. 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.



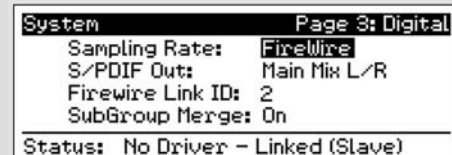
```
System Page 1: LCD Contrast
LCD Contrast: 0
LCD Backlight: Low
```

Press the Page Down button to access the Aux Pre Position page. By default all six Aux buses are set to Pre 1. This routes all 16 channels to each aux bus before the fader, limiter, EQ, and compressor and after the phase-reverse switch, high-pass filter, and noise gate. The two internal effects buses are set to Pre 2, which routes each of the 16 channels after all Fat Channel dynamics and EQ but pre-fader. From this menu you can choose between these two Pre positions for each Aux mix. If you have two aux buses stereo linked, you only need to set the Pre position on the Link Master. Remember, the Link Master is the aux that was selected when the link was created. To find out which Aux is the Link Master, simply select either of the Aux buses in the pair. The Link Master's ID will be shown in the Select Channel LED read-out



```
System Page 2: Aux Pre
Aux 1:Pre1 2:Pre1 3:Pre1 4:Pre1
Aux 5:Pre1 6:Pre1 A:Pre1 B:Pre1
HP>Gate <Pre1> Comp>EQ>Lim <Pre2>
```

Press the Page Down button to access the Digital Information page. From here you can change the Sample Rate, view your FireWire connectivity, assign which stereo mix you would like to send to the S/PDIF output, assign which StudioLive in a cascaded chain is the Master, and merge or unmerge the Subgroups in a StudioLive chain. When the StudioLive is connected and synced to a computer via FireWire, the Status will read “Driver On” and the Sampling Rate will read “FireWire.” Neither parameter can be adjusted from the StudioLive while it is synced to a computer so you will need use the Universal Control Device Window to make your S/PDIF assignments and change your sample rate (See Section 3.3.3 for more details). When two or more StudioLives are cascaded together, the Status will read “Linked” and you will be able to see which StudioLive is the Master unit. For more information on using multiple StudioLives, please review Section 4.



```
System Page 3: Digital
Sampling Rate: FireWire
S/PDIF Out: Main Mix L/R
Firewire Link ID: 2
SubGroup Merge: On
Status: No Driver - Linked (Slave)
```

Please Note: 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 and while it is connecting to a computer. 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. Because of this, it very important that the sample rate be selected and locked in prior to beginning any recording or performance.

2.10.4 The System Menu

Press the Page Down button again to access the Gate Type menu. The StudioLive lets you choose between an Expander and a Noise Gate for each pair of Channels. 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. Because of this, the global default setting for the StudioLive is the expander. Any changes made to this page are stored with custom scenes. When you recall as scene, the gate type that you have chosen for each stereo pair of channels will also be recalled. For more information on expanders and noise gates, please review Section 4.2.2)

System		Page 4: Gate Type	
Ch 1,2:	Expander	Ch 9,10:	Expander
Ch 3,4:	Expander	Ch 11,12:	Expander
Ch 5,6:	Expander	Ch 13,14:	Expander
Ch 7,8:	Expander	Ch 15,16:	Expander

Your StudioLive features a lockout mode that allows you to create a password and lock the controls your StudioLive. 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 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 Sections 3.3.6 for more information on this feature.

System		Page 5: Lockout	
Lockable:	Faders:	Yes	
	Master:	Yes	
	Aux Mixes:	Yes	
	Scene Recall:	No	
Status: Unlocked			

StudioLive 16.4.2 firmware version 1.10 and higher features a 31-band 1/3 octave graphic EQ that can be inserted on the Main output bus. When Page 6 of the System 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 20Hz to 20 kHz.

System		Page 6: Main GEQ	
Show Band:	1		
Freq (Hz):	20	Gain (dB):	-15.9
EQ Status:	On		
Push Recall to Flatten			

The frequency for each band is fixed. Bands 1-16 are controlled by encoders 1-16 respectively. When channel 16 is selected in the Show Band field in the System Menu, the meters will flip. Bands 16-31 will then be controlled by encoders 1-16 respectively. You will notice that all meters have one LED illuminated to display the current gain position for each band, 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 16 of the 31 bands at one time.

The StudioLive Graphic EQ features an innovative design that sets it apart from traditional graphic EQs. Traditionally, a 31-band graphic EQ uses 31 second order shelving filters with fixed frequencies in order to simulate a curve set by the user via 31 front panel sliders. A well designed graphic EQ creates an output frequency responses that corresponds as closely as possible to the curve displayed graphically by the sliders.

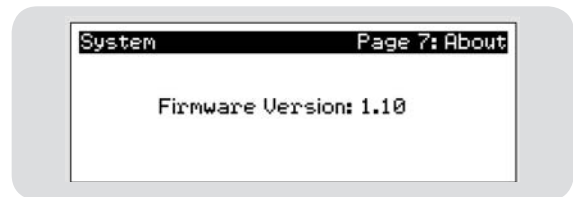
2.10.4 The System Menu

In an analog EQ, this is achieved by carefully choosing the bandwidth of the filter and deciding how or if it varies with the gain and how the filters are summed or cascaded. In general, narrower bandwidth signifies a higher quality EQ. But in traditional graphic EQ designs, the center frequency of each band is fixed.

PreSonus took a different approach with the StudioLive. The StudioLive Graphic EQ is a pool of shelving filters from which coefficients like cutoff frequency, bandwidth, and gain are extracted through a process of curve fitting. The curve entered by the user is first oversampled. The system then works with an internal curve made up of 128 bands to find coefficients for the first shelving filter that, when subtracted from the user's curve, will produce the flattest possible response: 0 dB. The resulting response is then used to find coefficients for the second shelving filter through the same optimization process. Coefficients for all available shelving filters are found through a recursive process.

Unlike conventional designs, the frequency and bandwidth of the “bands” depends on the curve entered by the user. This allows for much tighter matching of that curve. Because of this innovative design, the accuracy of the StudioLive EQ might feel “wrong” at first. The curve fitting process is capable of very steep transitions and, unlike conventional analog graphic EQs, what you see is what you get. With a carefully drawn, smooth curve the StudioLive EQ will have almost no frequency ripple. The StudioLive EQ is also capable of very sharp notches. To widen a notch or a bump on a particular band, just bring the adjacent bands closer to that band's gain and the bandwidth will change accordingly.

Press the Page Down button to view the firmware version currently loaded on your StudioLive. You normally will only view this screen when performing a system update, in order to confirm that the new firmware was successfully loaded.



3.1 System Requirements

Below are the minimum computer-system requirements for your StudioLive and Capture software.**

Macintosh

- Operating Systems:
 - Mac OS X 10.4.11
 - Mac OS X 10.5.2 or higher
- Hardware:
 - Minimum: PowerPC G4 1.25 GHz or Intel Core Solo 1.5 GHz processor, 1 GB RAM
 - Recommended: PowerPC G5, Intel Core Duo, or Intel Xeon processor, 2 GB or more RAM

Windows

- Operating Systems:
 - Windows XP (SP2 or greater) 32-bit and 64-bit
 - Windows Vista 32-bit and 64-bit
- Hardware:
 - Minimum: Intel Pentium 4 1.6 GHz processor , AMD Athlon 64 3000+ (Turion) , 1 GB RAM
 - Recommended: Intel Pentium 4 2.8 GHz EM64T or better, AMD Athlon 64 XP 3000+ or better, 2 GB or more RAM

NOTE: The speed of your processor, amount of RAM, and capacity and size and speed of your hard drives will greatly affect the overall performance of your recording system. A more powerful system (faster processor with more RAM) allows lower latency (signal delay) and better overall performance. Windows 64-bit system compatibility requires driver version 3.4 or later.

** Subject to change check www.presonus.com for updates

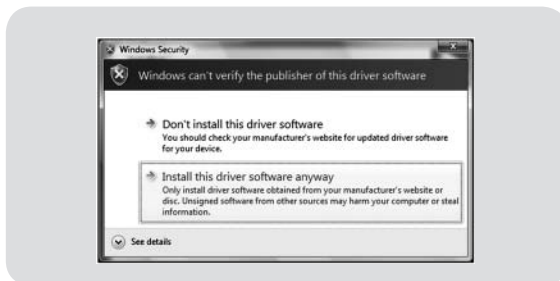
3.2.1 Installation Windows

After you insert the Installation CD into your CD-ROM drive, the StudioLive installer will take you through each step of the installation process. Please read each message carefully—ensuring especially that you do not connect your StudioLive too soon.



- 1) The StudioLive Installer was made as simple and easy to follow as possible. Please read each message carefully to ensure the StudioLive driver is properly installed.

Before beginning the StudioLive Installation Setup, please close all applications, including anti-virus software and disconnect your StudioLive from your computer. After the installation is successfully completed, don't forget to re-enable your anti-virus protection!

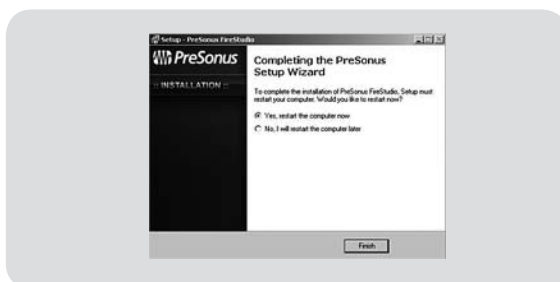


- 2) If you see any Windows Security alerts, click “Install this driver software anyway” (Vista) or “Continue anyway” (XP).



- 3) You installation disc contains the installers for the StudioLive™ audio drivers, Capture™, and a Capture™ demo session that will help you to quickly familiarize yourself with both the functions of your StudioLive and Capture. Each installer must be run separately.

Select “StudioLive” from the menu on the left of the installer window.



- 4) Follow the onscreen instructions to complete the installation. When the installer has finished, it will prompt you to reboot your computer.

Click “Finish” to automatically restart your PC. Once your computer has rebooted, connect the StudioLive. When the Found New Hardware wizard launches, follow the Recommended steps. Your StudioLive is synced to your computer and ready to use!

3.2.1 Installation in Mac OS X

Your Installation disc contains the installers for your StudioLive drivers, Capture, and a Capture demo session. Each needs to be installed separately. After inserting the Installation CD into your disc drive, browse the disc and run the StudioLive installer, which will take you through each step of the installation process. Please read each message carefully—ensuring especially that you do not connect your StudioLive too soon.



- 1) To begin installing the StudioLive driver on your Mac, double-click on the StudioLive logo.



- 2) You will be directed to the Welcome screen.

Click "Continue" and follow the onscreen instructions.



- 3) You will be directed to choose the hard drive onto which you want to install the drivers and Universal Control Panel. You must choose your system drive. Storage drives and partitions cannot be used as hosts for the driver.

Once the installation is completed, you will find the Universal Control program in your Applications folder. It is recommended that you place this in your Dock.

Click "Install"



- 4) When the Installation has completed, connect your StudioLive to your Mac with a FireWire cable and power it on. You are now ready to use your StudioLive!

3.3 Using Your StudioLive as an Audio Interface

Your StudioLive features a 32x18 FireWire interface. Any input and or bus with a Select button, as well as the Solo bus, tape input, and talkback mic, can be recorded with any audio-recording application that supports CoreAudio or ASIO. There are many such applications on the market from which to choose. Please consult the documentation that came with your audio application for specific instructions on how to select the StudioLive driver as the audio device driver for your software. It is important to note that your StudioLive uses the same driver as the PreSonus FireStudio family of interfaces, so its driver will be displayed as “PreSonus FireStudio” in all driver-selection menus.

Below are basic driver-setup instructions for three popular audio applications:

Apple Logic Pro/ Express 7+:	<ol style="list-style-type: none"> 1. Launch Logic Pro/Express. 2. Go to Logic Preferences Audio. 3. Click on the Devices Tab. 4. On the CoreAudio tab, check Enabled. 5. Select PreSonus FireStudio from the device menu. 6. You will be asked if you'd like to relaunch Logic. Click “try (re)launch.” 7. Your StudioLive features custom I/O labels for faster work flow. To enable these labels for use in Logic, go to Options Audio I/O Labels. 8. The second column in the pop-up window will be named “Provided by Driver.” Activate each of these labels for your StudioLive. When you are done, close this window. 9. You are now ready to use your StudioLive.
Steinberg Cubase 4	<ol style="list-style-type: none"> 1. Launch Cubase 4. 2. Go to Devices Device Setup. 3. Select “VST Audio System” from the Devices column in the Device Setup. 4. Select PreSonus FireStudio from the ASIO Driver dropdown list. 5. Click “Switch” to begin using the StudioLive Driver. 6. Once you have successfully changed the driver, go to Devices VST Connections to enable your input and output buses.
Cakewalk Sonar 6+	<ol style="list-style-type: none"> 1. Launch Sonar. 2. Go to Options Audio... and click on the Advanced tab. 3. Change the Driver Mode to “ASIO.” 4. Click the “OK” button. 5. Restart Sonar. 6. Go to Options Audio... and click on the Drivers tab. 7. Highlight all input and output drivers beginning with “PreSonus FireStudio.” 8. Go to Options Audio... and click on the General tab. 9. Set the Playback Timing Master to “PreSonus FireStudio ... DAW Out 1.” 10. Set the Recording Timing Master to “PreSonus FireStudio ... Mic/Inst 1.”

3.3.1 FireWire Sends and Returns

When using the StudioLive as an audio interface, it is important to understand the terms “FireWire send” and “FireWire return.” Because the audio interface in the StudioLive is completely integrated with the other functions of the mixer, the FireWire I/O is designed to work as an independent bus. You can route (send) signals from other buses to it, and its output, or return, signal is hard-coded to designated mixer channels. Your StudioLive has 32 available sends and 18 total returns.

FireWire sends 1-16 are hardcoded to be sent pre-fader from the 16 input channels of the StudioLive. These sends can be pre- or post-Fat Channel EQ and dynamics. To record the EQ and dynamics on any channel, simply enable the Dig Out button. It will illuminate, indicating that the Fat Channel signal path is being routed to the FireWire send. If this mode is not enabled, the signal sent via FireWire will be post-trim and post analog insert (if applicable). As previously discussed in Section 2.3.4, the Dig Out button is not operable for the Subgroups, Main Outputs, Aux bus, Internal FX Returns, and Aux Inputs. The Subgroups, Main Outputs, and Aux Inputs are also sent post-fader to their FireWire sends.



FireWire sends 17-32 are routed in 8 stereo pairs from the Universal Control Device Window. Any combination of eight of the following inputs and buses can be selected:

- Main Mix Left / Right
- Subgroup 1 & 2
- Subgroup 3 & 4
- Aux Send 1 & 2
- Aux Send 3 & 4
- Aux Send 5 & 6
- Aux Send A & B (EFXA & EFXB Send)
- Aux Return A (Aux Input A)
- Aux Return B (Aux Input B)
- FX Return A (EFXA Return)
- FX Return B (EFXB Return)
- 2 Track Left / Right
- Talkback Left / Right
- Solo Left / Right

3.3.1 FireWire Sends and Returns

Note that either channel of the stereo pair can still be accessed on a mono track in your host application; only the routing must be assigned as a stereo pair. For instance, if you have a backing-vocal group routed to Subgroup 1, and a guitar group routed to Subgroup 2, you would create two mono tracks in your recording application. The track whose input is assigned to Subgroup 1 will record the backing-vocal group, and the track whose input is assigned to Subgroup 2 will record your guitar group. If, however, you have a stereo drum group assigned to Subgroups 3-4, you can create a stereo track in your recording application and assign its input to Subgroup 3 and 4. For more information on subgroup mixing, please see Section 5.4.

The obvious exception to this principle is the Talkback Mic input. The talkback-microphone input has the same high quality XMAX preamp as the channel inputs, and the Universal Control Device Window allows you to assign it to one of the stereo FireWire sends. In this instance, the same signal will be printed on both sides of the send. In your recording software, you can create a mono track and assign it to either side of the stereo FireWire send to which it is routed. Both FireWire sends receive the same mono signal from the talkback-mic preamp.

Please note that you have access to both the Internal Effects Send group as well as the Return. If you record an Internal Effects Send (Aux Send A & B), you will be recording the unprocessed or dry mix of the channels being sent to that internal effects bus. To record the processed mix, you will need to choose the Internal Effects Returns (FX Return A or FX Return B). This is the direct output from the internal effect, so it will be 100% wet. Because of this, you may wish to record both the dry and the wet mix so that you can blend them to taste within your session.

Section 3.3.3 discusses how to select the sources for these eight stereo FireWire sends as well as the other features in the Universal Control application.

Each of the StudioLive's 16 inputs are hardcoded to receive their respective FireWire returns. Outs 1-16 in your recording application route these playback streams to their respective channels on the StudioLive (that is, the software's Output 1 always goes to StudioLive Channel 1 input and so on). Once you route a track in your recording application to play back through one of these outputs, it will always be accessible on its channel by simply pressing the FireWire Input button.



To provide you with the most flexible mixing environment, the main mix output for any application should be assigned to Outputs 17 and 18. These FireWire returns are hard-coded



to the FireWire input buttons in the 2 Track In and Monitor sections of the StudioLive. In this way, you can monitor the main output from your recording application without using two channels on your StudioLive, leaving the other 16 outputs available to be routed to the Fat Channel or for inserting a plug-in on a live instrument (see the next section for more details).

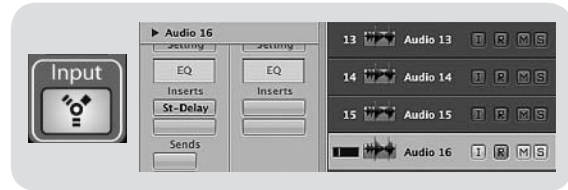
3.3.2 Using Plug-In Effects as Inserts

The FireWire streaming on your StudioLive is continuously bidirectional. This means that the StudioLive is always sending signals from the 16 direct FireWire sends on each input channel, as well as from the auxiliary inputs and buses assigned to the second bank of 16 FireWire sends. At the same time, it is receiving signals back from the 18 FireWire returns. Because the FireWire returns always come back to their respective channels on the StudioLive, you can quickly insert a plug-in from your recording application into any channel strip and monitor it in real time.

In this example, we will insert a Mono Delay plug-in from Apple's Logic™ 8 application onto Channel 16 of the StudioLive. To begin, create a mono audio track. Assign its input to Channel 16 and its output to Output 16. (Note: because Logic does not offer mono output buses, in this case we are routing this stream to channels 15-16 and panning the channel all the way to the right so that it will only be sent to output 16. Several other applications use this type of bus routing, as well. Please consult your software's user manual for specific instructions.)



Once you have the routing set up in Logic, record-arm the track and insert the desired plug-in. Press the FireWire Return Input button on your StudioLive. You can now monitor your analog signal with your inserted effect as you record.



A Quick Note on Printing Fat Channel Dynamics and EQ:

Using the Fat Channel dynamics processing and EQ during post-production is obviously desirable. The question is how to print these changes to your recording. An easy way to do this is through the subgroup FireWire returns. Begin by routing a playback stream to an individual channel on the StudioLive from within your recording application. With the FireWire return selected as the signal source for that channel on your StudioLive, press its Select button and assign it to a subgroup bus, ensuring that only that channel is routed to the subgroup. You can then process the playback stream through the Fat Channel for both the individual channel and for the subgroup. In your recording software, assign the subgroup as an input source for a new track and record it. The unprocessed track can then either be removed from the session or used in tandem with the processed track as an effect. The flexible routing and mixing capability of the StudioLive provides you with an arsenal of tools previously unavailable to "in the box" mixes. Don't be afraid to test the limits of modern mixing techniques and to create a few of your own.

Note: The Dig Out button does not function when you have a FireWire Return selected as an input source. If Dig Out is enabled, any Aux, Subgroup, and the Main bus to which it is routed will not receive the post Fat Channel signal. The unprocessed signal will still be routed to these buses, as well as to the FireWire bus, but the Fat Channel dynamics and EQ will not. If you wish to add Fat Channel dynamics to a FireWire return, you must disable Dig Out.

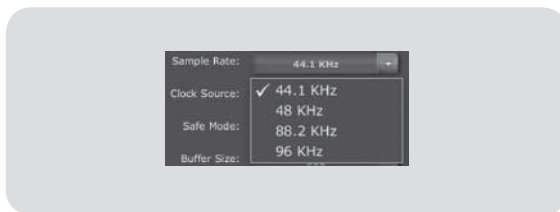


3.3.3 Universal Control



The StudioLive 16.4.2 is compatible with the Universal Control application. As with all FireStudio family interfaces, Universal Control includes two windows: the Launcher window and the Device window. For the StudioLive, the Device window is the Auxiliary Inputs Router. As discussed in the Section 3.3.1, the Auxiliary Inputs Router allows you the freedom to choose which additional buses to route to the eight stereo auxiliary FireWire sends as well as create a password for your StudioLive so that you can lock out unauthorized users (See Section 3.3.6 for details). From the Launcher window, you can set basic parameters such as buffer size and sample rate. In addition, you can use the Launcher window to configure your WDM outputs (PC only, see Section 3.3.4)

Sample Rate Selector



Changes StudioLive Sample Rate.

You will set the sample rate to 44.1 or 48 kHz from the Launcher Window when your StudioLive is connected to a computer. While higher Sample Rates are included in this menu, these are only for use with FireStudio family products that support them. When using a StudioLive alone or with another FireStudio family interface, the maximum Sample Rate that can be selected is 48kHz.

A higher sample rate will increase the fidelity of the recording. It will also increase the file size and the amount of system resources necessary to process the audio.

Please Note: 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 or when the mixer is connecting to a computer. 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. Because of this, it very important that the sample rate be selected and locked in prior to beginning any recording or performance.

3.3.3 Universal Control

Buffer Size Selector (PC only)



Changes the StudioLive Buffer Size.

You can set the buffer size from 64 to 4,096 samples. The buffer size (or latency) determines the roundtrip time it takes audio data to be converted from analog to digital and back to analog. As a general rule, the higher the buffer size, the better the system performance, but the less playable virtual instruments and the like become. In general, 512 samples (11 to 12 milliseconds) will provide you with a large enough buffer for optimum system performance, but low enough to be unobtrusive. You should set your buffer size and sample rate prior to launching your host application.

On Macintosh, the buffer size is set from inside your host application

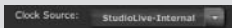
Operation Mode



Changes How the StudioLive Driver's Buffer Size is Set

- **Normal Mode.** Input and Output buffers are both identical to the Buffer Size setting.
- **Safe Mode 1-3.** Increases the output buffer size for added stability.

Clocksource



Changes the Clocksource for Chained FireStudio family devices

The StudioLive cannot slave to an external clock, however, if you are cascading additional FireStudio family products that do feature digital inputs, this becomes selectable.

The clock source setting will determine where the device chain is receiving word clock information. This keeps the device chain in sync with other digital devices and with each other. The menu options are determined by the available digital inputs in the chain.

In general, you will want the StudioLive to be your Master Clock so you will set the clock to StudioLive-Internal.

Device Window Button



Opens the Device Window

Click on this button to open the Auxiliary Inputs Router.

To give your StudioLive a custom name, double click on the default label (StudioLive) to open a text field. When you have finished entering your custom name, hit the Enter key.

3.3.3 Universal Control

Settings Menu

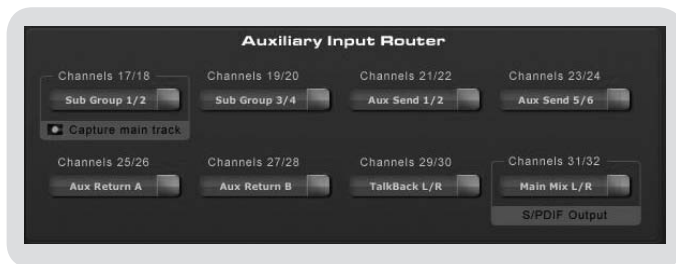


Changes the Window Views

The Settings menu in the Launcher window provides you with several options for viewing the Launcher and Device Windows and to set up WDM playback (PC only).

- **WDM Setup (PC Only).** Opens the WDM Channel Mapper window. See Section 3.3.4 for more details.
- **Launcher Floats.** Allows the Launcher window to be in the foreground while other applications are active.
- **Show All Devices.** Opens the Device Window for every Studio Control compatible device in the chain.
- **Close All Devices.** Closes all Device Windows.

As discussed in the previous section, the StudioLive allows you to route any 8 of 14 buses and inputs, in addition to its 16 input channels. This is done via the Device Window's Auxiliary Input Router. To route FireWire Sends 17-32, first decide which buses and inputs you would like to record in addition to your input channels. Once you've determined your input pairs, simply route them to a stereo pair of auxiliary inputs of your choice, using the pull-down menu beneath each Auxiliary FireWire Send pair in the Universal Control Auxiliary Input Router. Remember, all of these buses and inputs are automatically set to send their signals post Fat Channel dynamics and EQ (where applicable). The inputs and buses selected in the Auxiliary Input Router will be displayed in your recording application with the name of their routing.



For instance, the routed pairs in the above picture would translate as follows in your host recording application:

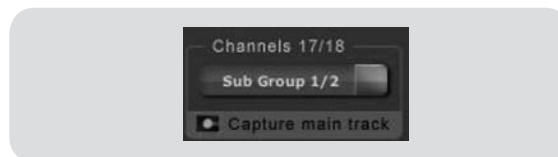
3.3.3 Universal Control

PHYSICAL	SOFTWARE
Subgroup 1	Auxiliary In 17
Subgroup 2	Auxiliary In 18
Subgroup 3	Auxiliary In 19
Subgroup 4	Auxiliary In 20
Aux Send 1	Auxiliary In 21
Aux Send 2	Auxiliary In 22
Aux Send 5	Auxiliary in 23
Aux Send 6	Auxiliary In 24
Aux Return A (Aux Input A)L	Auxiliary In 25
Aux Return A (Aux Input A)R	Auxiliary In 26
Aux Return B (Aux Input B)L	Auxiliary In 27
Aux Return B (Aux Input B)R	Auxiliary In 28
Talkback	Auxiliary In 29
Talkback	Auxiliary In 30
Main Mix Left	Auxiliary In 31
Main Mix Right	Auxiliary In 32

The Auxiliary Input Router also allows you to designate a specific bus to the S/PDIF output as well as to select which stereo pair will be recorded on the Main Track in Capture. When the StudioLive is not connected to a computer, the S/PDIF output is routable through the System Menu (see Section 2.10.4). However, when your StudioLive is synced to your computer, you can route any of the 14 routable buses to the S/PDIF output by routing them to FireWire sends 31 and 32. In addition to being available for recording via Aux In 31 and Aux In 32, these sends are normalled to the S/PDIF output.

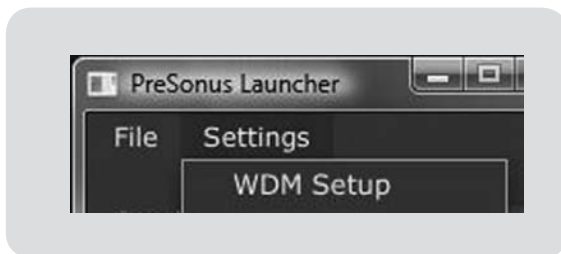


Capture enables you to record a stereo mix track in addition to the 16 Channel inputs on the StudioLive. Auxiliary FireWire Inputs 17 and 18 are the source for this Main Track in Capture, and it is user definable through the Auxiliary Input Router. Whichever source is routed to Channels 17/18 can be recorded as a stereo track in Capture. For more details on the Main Track and Capture, please consult the Capture User Manual that came with your StudioLive.



3.3.4 Advanced WDM Features (PC only)

Just like the award-winning FireStudio family of interfaces, the StudioLive features some advanced WDM features to enhance its use as an audio interface. In the upper left hand corner of the Universal Control Launcher window, you will find the Settings menu, from here you can configure your WDM input and output streams. *Mac users: please skip to the next section for important information about using your StudioLive as your main Core Audio device.*



- 1) In the Launch window, go to Settings | WDM setup

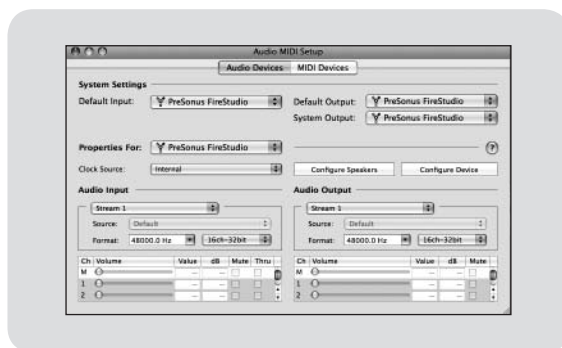


- 2) The WDM Channel Mapper window will open. By default WDM Outputs 1 and 2 are routed to StudioLive Channels 1 and 2. As previously mention, it is recommended that they be set, instead, to Channels 17 and 18, so as not to use up input channels on your StudioLive just for WDM playback. To begin, check the box next to Output. Click on WDM 1 (Front Left) and drag it down to place it next to StudioLive 2TrackIn Left. Click on WDM 2 (Front Right) and drag it down to place it next to StudioLive 2TrackIn Right.

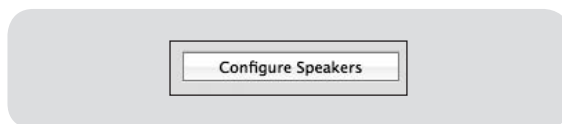
Your stereo WDM playback will not be routed directly to the Main FireWire Return buttons in the 2TrackIn and Monitor bus in the StudioLive Master Section.

3.3.5 Configuring Your StudioLive for Core (Mac only)

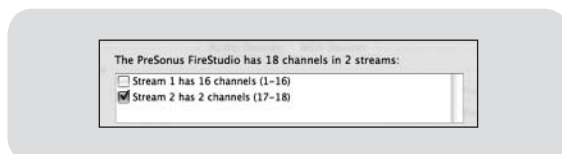
Your StudioLive can be used as the main Core Audio device for your Mac, and you can route the main outputs of your Mac to any of its available output streams. By default these streams are routed to Channels 1 and 2 on your StudioLive. The obvious limitation of this is that you will use up two input channels on your StudioLive just for Core Audio playback. Channels 17 and 18 are available through the FireWire Return buttons inside the 2 Track In and Monitor Sections on your StudioLive. It is recommended that you use these channels instead for Core Audio playback. The following instructions will help you configure your StudioLive to be used as the main audio playback device for your Mac.



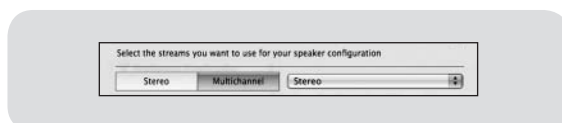
- 1) Go to MacHD | Applications | Utilities and launch Audio MIDI setup.



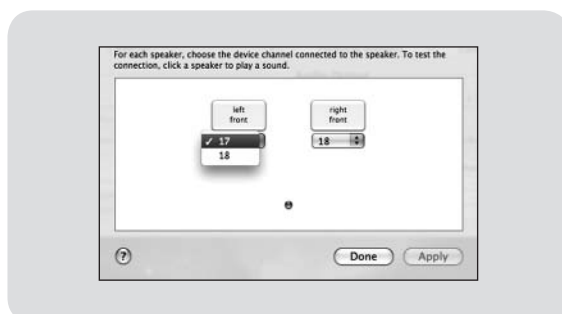
- 2) Click on the Configure Speakers button.



- 3) Deselect Stream 1 and select Stream 2 instead.



- 4) In the same window, click on Multichannel and select your Speaker Configuration from the pull down menu (i.e. Stereo).



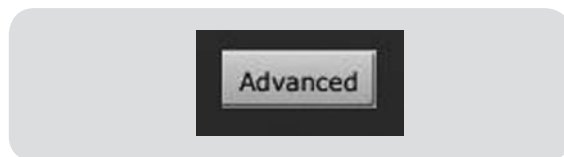
- 5) Choose Channel 17 for the left speaker and Channel 18 for the right speaker.

When you are finished, click Apply and then Done.

Your StudioLive will now monitor the Main Output for your Mac through the FireWire return buttons in the 2 Track In and Monitor Sections.

3.3.6 Enabling and Using Lockout Mode

As mentioned in Section 2.10.4, your StudioLive features a lock-out mode which allows you to temporarily disable nearly every feature on the StudioLive. It is important to mention that analog portions (i.e., input trim knobs, faders, and cue, tape input, and monitor levels) can all still be adjusted. Because of this, it is recommended that you take a quick glance over your input trims and output levels prior to resuming your mix after you unlock your StudioLive. If you have set your fader position to be locked, you will be able to recall your pre-lockout fader positions using the Locate button in the meter section. Until you connect your StudioLive to a computer, the mixer cannot be locked, so don't worry about accidentally locking yourself out.



With your StudioLive connected and synced to your computer, launch the StudioLive Device window in Universal Control, click on the Advanced button. The Lockout window will launch.



To set your custom password, click on the box next to "Device is Lockable." At this point, a cursor will appear in the password box. Enter a five-digit code using any number between 1 and 9 and click the Set button. Your password will no longer be displayed. Should you need to change your password, simply click on the box next to "Device is Lockable." Your old password will be deleted, and you will be able to enter a new password. Once you have set your password, the StudioLive can be locked with or without being synced to a computer.

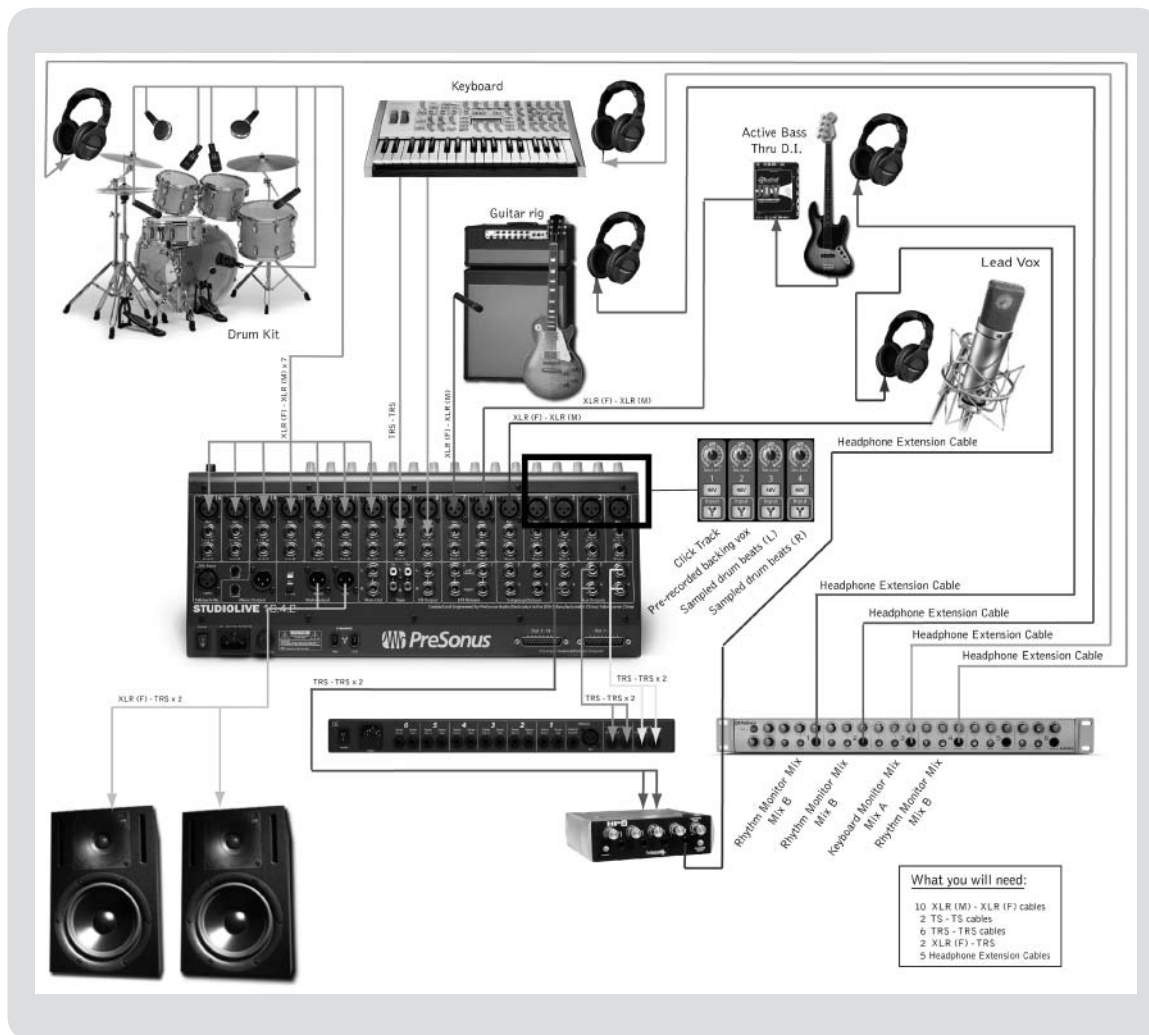


To lock your StudioLive, press the System button in the Digital Effects | Master Control section and navigate to Page 4: Lockout. Your StudioLive cannot be locked unless this page is active. If you have multiple StudioLive mixers connected, you only need to lock the Master unit to lock all the mixers in the chain.

At its lowest level of security, Lockout mode allows you to freeze the current Fat Channel and Effects settings. Faders, Aux mixing, and Scene Recall can still be enabled. Each of these mix functions can be added to Lockout mode as you see fit. To lock your StudioLive, press the Select buttons that correspond to the five-digit password you have set. In this example, the password is 12345, so you would press the Select buttons for Channels 1, 2, 3, 4, and 5, in that order. The Panel Status will switch to Locked, indicating that Lockout mode is now active.

To unlock your StudioLive, navigate back to Page 4: Lockout in the System menu if this page is not currently active. Press the same sequence of Select buttons. The Panel Status will change to Unlocked, and you will be able to resume your mix.

3.4 Hook-up Diagram: In The Studio with StudioLive



This diagram features a PreSonus HP60 headphone amplifier receiving two different mixes from Aux 1 & 2 and Aux 3 & 4, while a third monitor mix is sent from Aux 5 & 6 to a PreSonus HP4 headphone amplifier. Channels 1-4 on the StudioLive are switched to FireWire Playback so that a click track, a prerecorded vocal track, and a stereo track of sampled drum beats can be included in each musician's monitor mix.

It should be noted that in this diagram, the control-room monitors are shown as connected to the main outputs. This is only one way of connecting them. You can also use a pair of subgroup outputs or the Control Room Monitor outputs. Each has benefits and drawbacks. We suggest you experiment to find the best setup for your mixing style.

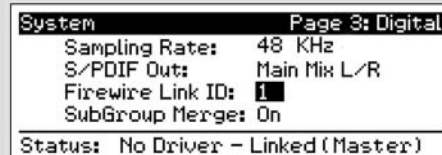
4.1 Getting Started

Firmware version 1.10 and above enables you to use multiple StudioLives cascaded together by a FireWire connection. Multiple units can be connected together with or without a computer connected. Standalone, up to four units can be connected together. Two units can be connected to a computer for 32 channels of recording and channel playback. This section will guide you through the syncing process and explain some of the features.

4.1.1 Configuring Multiple Units

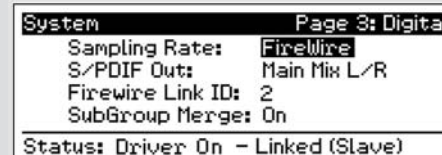
Firmware version 1.10 or higher is required in order to cascade multiple units. The latest firmware is available at www.presonus.com/technical-support. StudioLive firmware is user updatable and it is recommended that you check the PreSonus website periodically for driver and firmware updates. The driver must be installed on your Mac or PC and your StudioLive must be connected and synced before you can run the firmware updater. Once all of your units have been updated, connect a FireWire cable from the computer to the first StudioLive. Using a standard 6-pin FireWire cable, connect the remaining FireWire port on the first StudioLive to either FireWire port on the second StudioLive. If you are combining multiple units to create a standalone system (without a computer), simply connect a FireWire cable from the first unit to the second unit, another FireWire cable from the second unit to the third unit and so on.

Press the System button on the first unit and page down to Page 3: Digital. Use the Next button to move to the Firewire Link ID field. Use the value encoder to set the ID to 1. Any unit in the chain can be designated as the Master. Since the entire chain will function as one large console, sending all channels to the Master unit's Main output, Aux Sends, and Subgroups (optional), we suggest that the unit farthest on the right be designated as the Master. The unit with the lowest Firewire Link ID greater than zero is automatically set as the Master.



System	Page 3: Digital
Sampling Rate:	48 KHz
S/PDIF Out:	Main Mix L/R
Firewire Link ID:	1
SubGroup Merge:	On
Status: No Driver - Linked (Master)	

Press the System button on each of the cascaded units and page down to Page 3: Digital. Use the Next button to move to Firewire Link ID field and give each unit a unique ID. Any value larger than that of the Master can be used for each of the cascaded units. The unit with the lowest ID is the Master, its channels will be the last bank of 16. The unit with the next lowest ID will be the second to last and so on.



System	Page 3: Digital
Sampling Rate:	FireWire
S/PDIF Out:	Main Mix L/R
Firewire Link ID:	2
SubGroup Merge:	On
Status: Driver On - Linked (Slave)	

If you will be recording with two StudioLive 16.4.2 mixers, take a moment to look at the serial numbers on the bottom of each mixer. The mixer with the lowest serial number will be designated as Inputs and Outputs 1-16 in your recording application. Because of this, it is recommended that the mixer with the lowest serial number be set as the Slave and the unit with the highest value serial number be set as the Master.

4.1.2 **Aux Mixing With Cascaded Mixers**

Every channel in the mixer chain can be sent to the six Aux outputs on the Master unit. When you press the Mix button on any of the six auxes, you will notice that the Mix button for the corresponding aux will illuminate on every mixer in the chain. For example, if you press the Mix button on Aux 1 on the Master unit, the Mix button on Aux 1 on each of the slaved units will also illuminate. Creating an aux mix with multiple mixers works exactly the same way as with one mixer. Each of the encoder beneath the meters in the Fat Channel control the amount of send level for each of their corresponding channels on that mixer to the enabled aux.

For example, if you have three mixers chained together, Channels 1-16 will reside on Slave 2, Channels 17-32 will reside on Slave 1, and Channels 33-48 will reside on the Master unit. Let's say that you want to create an aux mix on Aux 3. To begin, press the Aux 3 mix button. The Fat Channel meters and encoders on all three mixers will be ready for you to create an Aux 3 mix. you will use the Fat Channel meters and encoders on Slave 2 to set the Aux 3 send levels for Channels 1-16 and the meters and encoders on Slave 1 to set the Aux 3 send levels for Channels 17-32. Use the Fat Channel meters and encoders on the Master unit to set the Aux 3 send levels for Channels 33-48. The resulting mix is then routed from the Aux 3 output on the Master unit. 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 Fat Channel to dial in your dynamics and EQ settings.

Note that the Aux outputs on the slaved mixers are still active, but they only have access to the local channels. So in this example, the Aux 3 mix for Channels 1-16 on Slave 2 is routed locally to the Aux 3 output and merged with the overall Aux 3 mix on the Master unit at the same time. The same is true of Channels 17-32 on Slave 1. Only the Master mixer receives the merged signals from all cascaded mixers in addition to its own local channels.

Because Aux mixes are sent to the Master unit before the Fat Channel (for obvious reasons), if you wish to create a stereo Aux mix, you must link the Aux channels on each of the mixers in the chain in order to control panning for the channels on the slaved mixers.

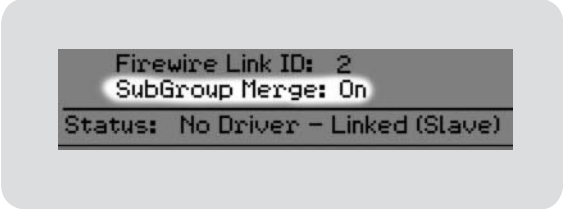
4.1.3 Internal Effects Buses

Unlike the 6 Aux buses, the two Internal Effects buses on each mixer are independent. Using the same example as the previous section, Channels 1-16 can only be routed to EFXA and EFXB on Slave 2, Channels 17-32 are routed to Slave 1's EFXA and EFXB buses, and Channels 33-48 are processed using the Master unit's two internal effects buses. The advantage is that you get twice the effects buses as you have mixers in the chain!

Of course, if you'd like to send all channels to the same effect, you can simply load the same effect on every mixer. But with some careful patching, you can take advantage of the extra effects buses at your 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.

4.1.4 Subgroups: To Merge Or Not To Merge

On Page 3: Digital in the System menu, you will find the SubGroup Merge field. The Subgroups on each of the mixers in the chain can either be merged together or local only. This is determined on each mixer for the unit with the next ID number in the chain and therefore the previous bank of channels (i.e. if SubGroup Merge is enabled on the Master, Slave 1's Subgroups will be merged with the Master's Subgroups; if SubGroup Merge is enabled on Slave 1, Slave 2's Subgroups will be merged with Slave 1's Subgroup). The Subgroup faders on the mixer from which SubGroup Merge was enabled will control the entire group and dynamics processing can be added to the entire mix using that unit's Fat Channel.



```
Firewire Link ID: 2
SubGroup Merge: On
Status: No Driver - Linked (Slave)
```

If Subgroup Merge is turned Off, each of the four subgroups on each mixer are locally controlled. Using our three mixer example, Channels 1-16 can only be assigned to the Subgroup on Slave 2, Channels 17-32 can only be assigned to the Subgroups on Slave 1, and Channels 33-48 can only be assigned to the Subgroups on the Master unit. Because the Subgroups on each of the mixers in the chain can still be routed to the Main output on the master unit, unmerging the Subgroups offers advantages for Effects group mixing (See Section 5.4.2 for details), but this can also be useful in any situation where you want 16 or fewer channels assigned to the same Subgroup, just do a little careful patching!

4.1.5 **Scene Store and Recall**

Channel Strip presets, Effects presets and mix Scenes are stored and recalled locally on each mixer. However, if you are creating a mix Scene, the name that you create on any of the mixers will be transmitted to the entire chain. For example, if you store a scene named “3 Mixer Show” on the Master unit, the name is broadcasted to the slaved units so when you press the Scene and Store buttons on either Slave, the name you just created on the Master unit is already entered; all you have to do is scroll to an empty location and press the Store button again.

4.1.6 **Copy and Load**

Channel strip settings from any channel or bus on any mixer in the chain can be copied to any other channel or bus on any other mixer in the chain. For example, if you select Channel 4 on Slave 2 and press the Copy button, every Select button on every mixer in the chain will begin to flash. To paste the Channel 4 settings to any other channel simply press that channel's Select button, it will stop flashing and will illuminate. Press the Load button to paste the Fat Channel setting.

4.1.7 **Universal Control & Multiple Mixers**

Because of FireWire limitations, you can only record additional buses from the Master unit when two mixers are connected to a computer. While the Launcher Window will show a Device Window Button for each of the two StudioLives connected, only the Master Unit's Auxiliary Input Router will open when either button is clicked.

4.2

Local Versus Merged Buses And Inputs

As discussed earlier, the two Internal effects buses are local only, meaning they are controlled from each mixer separately and they can only accept signals from the channels on their respective mixers.

The following inputs and bus are local only:

- **Talkback Microphone:** 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. While the talkback mics on the slaves are active on their respective units, they cannot be patched into the Master unit. Because of this, you should connect your talkback mic to the Master unit and control it from there.
- **Analog Tape Input:** As described in Section 2.9.3, the Analog Tape Input on the StudioLive is patched directly to the Main Output using a button in the Master Section. While the Tape Input on every mixer in the chain is active, this button only patches the input to the Main bus on that unit. For example, the Tape In button on Slave 1 patches it to the Main bus on Slave 1 only. Because of this, we recommend that you only use the Tape Input on the Master unit.
- **Main FireWire Returns:** When two mixers are connected to a computer, the StudioLive becomes a 32x34 audio interface. As discussed in Section 3.3.1, just like with a single mixer connected to a computer, the 32 FireWire inputs on two chained units are hard-coded to receive their respective FireWire returns. Outputs 1-32 in your recording application will route these playback streams to their respective channels on the two StudioLives. FireWire Returns 33 and 34 (17 and 18 with one mixer connected) are hard-coded to the FireWire input buttons in the 2Track In and Monitor Sections on both the Master and the Slave mixer. If the 2TrackIn FireWire Return button is enabled on both mixers, the duplicated signals will both be routed to the Master unit's main output. As with the Analog Tape Input, it is recommended that only the Master unit's 2TrackIn is used.
- **FireWire Sends:** As also discussed in Section 3.3.1, a single StudioLive has 32 available FireWire sends, 16 from its channels and 16 from a selection of buses and aux inputs. When two StudioLive mixers are connected to a computer, each StudioLive has 16 available FireWire sends for a total of 32. As mentioned earlier, FireWire sends 1-16 output from the StudioLive with the lowest serial number. FireWire sends 17-32 output from the StudioLive with the highest serial number. These sends cannot be changed, nor can they be merged. For the best workflow, it is recommended that the unit with the lowest serial number be placed to the left and be set as the Slave.
- **Monitor Bus:** You must use the Monitor section on the Master unit to monitor all channels in the chain (by listening to the Solo bus or the the Main bus), the Tape Input for the Master Main bus, and the Main FireWire Return. The Monitor bus on the slaved units can only receive signal from the mixer it is on.

The Solo bus is a merged bus; any channel on any mixer can be soloed and monitored from the Master unit. When either PFL or SIP are engaged on any mixer, every mixer is placed into the same mode at the same time.

5.1 Microphones

The StudioLive works with most types of microphones, including dynamic, ribbon, and condenser microphones.

5.1.1 Condenser

Condenser microphones generally capture sound with excellent fidelity and are one of the most popular microphone choices for studio recording and, increasingly, for live performance as well. Condenser microphones require a power source, which can be provided by a small battery, an external power supply, or phantom power, which is usually provided by a mixer, preamplifier, or direct (DI) box. Phantom power is sent over the same mic cable that carries the audio signal; the term derives from the fact that there is no visible power cord, and the voltage is not perceptible in the audio path. The StudioLive sends 48 VDC phantom power from the XLR inputs only.

5.1.2 Dynamic

Dynamic microphones are possibly the most widely used microphone type, especially in live shows. They are relatively inexpensive, resistant to physical damage, and typically handle high sound-pressure levels (SPL) very well. Unlike condenser microphones, dynamic microphones typically do not require a power source. In most cases, phantom power has no effect on a dynamic microphone's audio quality or sensitivity.

Dynamic microphones, especially ribbon microphones, tend to generate low output voltages, so they typically need more preamp gain than condenser microphones.

Ribbon

Ribbon microphones are a special type of dynamic microphone and get their name from the thin metal ribbon used in their design. Ribbon microphones capture sound with very high fidelity—especially higher frequencies. However, they often are very fragile (many newer models are less so) and typically cannot handle high SPLs.

The most important thing to note about ribbon microphones is that nearly all ribbon microphones do not require phantom power. In fact, unless a ribbon microphone specifically calls for phantom power, sending phantom power to a ribbon microphone can severely damage it—usually beyond repair.

5.1.3 **USB and other Types**

There are many microphone types available, and as technology increases, it is very likely more will be developed. One type of microphone to emerge recently is the USB microphone. These may be dynamic or condenser mics, but many of them have built-in preamps and need drivers to work with computers. Because a USB microphone is, in effect, an audio interface, we recommended that you not use them with the StudioLive, as the likelihood of conflicting drivers is great.

If you are using a new or nonstandard type of microphone (e.g., USB, headset, laser, MEMS), please consult your microphone's user's manual for power requirement and compatibility information.

Regardless of the microphone type you are using, we recommend reading your microphone's user's manual thoroughly before engaging phantom power and if other usage questions arise.

5.1.4 **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 form. 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 and have been noted for your convenience.

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 applications as well.



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 (select the “fuller”-sounding position). To use this technique in a live application, omit the condenser microphone.



Acoustic Guitar

Place a small-diaphragm condenser microphone pointed 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 photo 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 same technique can be used in live applications.



5.1.4 Microphone Placement

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 up. 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 also be used in live applications.



5.2 **A Brief Tutorial on Dynamics Processing**

The heart of the StudioLive is the Fat Channel dynamics section. What follows is an excerpt from brief tutorial on dynamics processing written by PreSonus president and founder, Jim Odom. It is included to help you get the most out of your StudioLive. This tutorial will take you through the basics of dynamics processing and will explain the various types of dynamics processors.

5.2.1 **Common Questions Regarding Dynamics Processing**

What is dynamic range?

Dynamic range can be defined as the distance between the loudest possible level to the lowest possible level. For example, if a processor states that the maximum input level before distortion is +24 dBu, and the output noise floor is -92 dBu, then the processor has a total dynamic range of $24 + 92 = 116$ dB.

The average dynamic range of an orchestral performance can range from -50 dBu to +10 dBu, on average. This equates to a 60 dB dynamic range. Although 60 dB may not appear to be a large dynamic range, do the math, and you'll discover that +10 dBu is 1,000 times louder than -50 dBu!

Rock music, on the other hand, has a much smaller dynamic range: typically -10 dBu to +10 dBu, or 20 dB. This makes mixing the various signals of a rock performance together a much more tedious task.

Why do we need compression?

Consider the previous discussion: You are mixing a rock performance with an average dynamic range of 20 dB. You wish to add an uncompressed vocal to the mix. The average dynamic range of an uncompressed vocal is around 40 dB. In other words, a vocal performance can go from -30 dBu to +10 dBu. The passages that are +10 dBu and higher will be heard over the mix. However, the passages that are at -30 dBu and below will never be heard over the roar of the rest of the mix. A compressor can be used in this situation to reduce (compress) the dynamic range of the vocal to around 10 dB. The vocal can now be placed at around +5 dBu. At this level, the dynamic range of the vocal is from 0 dBu to +10 dBu. The lower level phrases will now be well above the lower level of the mix, and louder phrases will not overpower the mix, allowing the vocal to “sit in the track.”

The same points can be made about any instrument in the mix. Each instrument has its place, and a good compressor can assist the engineer in the overall blend.

5.2.1

Common Questions Regarding Dynamics

Does every instrument need compression?

This question may lead many folks to say “absolutely not, overcompression is horrible.” That statement can be qualified by defining overcompression. The term itself must have been derived from the fact that you can hear the compressor working. A well-designed and properly adjusted compressor should not be audible! Therefore, the overcompressed sound is likely to be an improper adjustment on a particular instrument—unless, of course, it is done intentionally for effect.

Why do the best consoles in the world put compressors on every channel? The answer is simply that most instruments need some form of compression, often very subtle, to be properly heard in a mix.

Why do you need noise gates?

Consider the compressed vocal example above; you now have a 20 dB dynamic range for the vocal channel. Problems arise when noise or instruments (air conditioner, loud drummer, etc.) in the background of the vocal mic become more audible after the lower end of the dynamic range is raised. You might attempt to mute the vocal between phrases in an attempt to remove the unwanted sounds; however this would probably end disastrously. A better method is to use a noise gate. The noise-gate threshold could be set at the bottom of the dynamic range of the vocal, say -10 dBu, such that the gate would shut out the unwanted signals between the phrases.

If you have ever mixed live sound, you know the problems cymbals can create by bleeding through the tom mics. As soon as you add some highs to get some snap out of the tom, the cymbals come crashing through, placing the horn drivers into a small orbit. Gating those tom mics so that the cymbals no longer ring through them will give you an enormous boost in cleaning up the overall mix.

Dynamics processing is the process of altering the dynamic range of a signal, thereby enhancing the ability of a live sound system or recording device to handle the signal without distortion or noise and aiding in placing the signal in the overall mix.

Compression/Limiting

Punch, apparent loudness, presence—these are just three of the many terms used to describe the effects of compression/limiting.

Compression and limiting are forms of dynamic-range (gain) control. Audio signals have very wide peak-to-average signal-level ratios (sometimes referred to as dynamic range, which is the difference between the loudest level and the softest level). The peak signal can cause overload in the audio-recording or sound-reinforcement chain, resulting in signal distortion.

A compressor/limiter is a type of amplifier in which gain is dependent on the signal level passing through it. You can set the maximum level a compressor/limiter allows to pass through, thereby causing automatic gain reduction above some predetermined signal level, or *threshold*. *Compression* refers, basically, to the ability to reduce, by a fixed ratio, the amount by which a signal's output level can increase relative to the input level. It is useful for lowering the dynamic range of an instrument or vocal, making it easier to record without distorting the recorder. It also assists in the mixing process by reducing the amount of level changes needed for a particular instrument.

Take, for example, a vocalist who moves around in front of the microphone while performing, making the output level vary up and down unnaturally. A compressor can be applied to the signal to help correct this recording problem by reducing the louder passages enough to be compatible with the overall performance.

How severely the compressor reduces the signal is determined by the compression ratio and compression threshold. A ratio of 2:1 or less is considered *mild compression*, reducing the output by a factor of two for signals that exceed the compression threshold. Ratios above 10:1 are considered *hard limiting*.

As the compression threshold is lowered, more of the input signal is compressed (assuming a nominal input-signal level). Care must be taken not to overcompress a signal, as too much compression destroys the acoustic dynamic response of a performance. (That said, overcompression is used by some engineers as an effect, and with killer results!)

Limiting refers to the processing that prevents the signal from getting any louder (that is, it prevents any increase in the signal's amplitude) at the output.

Compressor/limiters are commonly used for many audio applications. For example:

- A kick drum can get lost in a wall of electric guitars. No matter how much the level is increased, the kick drum stays lost in the “mud.” A touch of compression can tighten up that kick-drum sound, allowing it to punch through without having to crank the level way up.

5.2.2 Types of Dynamics Processing

- A vocal performance usually has a wide dynamic range. Transients (normally the loudest portions of the signal) can be far outside the average level of the vocal signal. Because the level can change continuously and dramatically, it is extremely difficult to ride the level with a console fader. A compressor/limiter automatically controls gain without altering the subtleties of the performance.
- A solo guitar can seem to be masked by the rhythm guitars. Compression can make your lead soar above the track without shoving the fader through the roof.
- Bass guitar can be difficult to record. A consistent level with good attack can be achieved with proper compression. Your bass doesn't have to be washed out in the low end of the mix. Let the compressor/limiter give your bass the punch it needs to drive the bottom of the mix.

Expansion

There are two basic types of expansion: dynamic and downward. Expansion increases the dynamic range of a signal after the signal crosses the expansion threshold. *Dynamic expansion* is basically the opposite of compression. In fact, broadcasters use dynamic expansion to “undo” compression before transmitting the audio signal. This is commonly referred to as *companding* or *COMP*ression followed by *exp*ANDING.

By far the most common use of expansion is *downward expansion*. In contrast to compression, which decreases the level of a signal after it rises above the compression threshold, expansion decreases the level of a signal after the signal goes below the expansion threshold. The amount of level reduction is determined by the expansion ratio. For example, a 2:1 expansion ratio reduces the level of a signal by a factor of two. (e.g., if a level drops 5 dB below the expansion threshold, the expander will reduce it to 10 dB below the threshold.)

Commonly used for noise reduction, expansion is very effective as a simple noise gate. The major difference between expansion and noise gating is that expansion is dependent on the signal level after the level crosses the threshold, whereas a noise gate works independent of a signal's level beyond the threshold.

Noise Gating

Noise gating is the process of removing unwanted sounds from a signal by attenuating all signals below a set threshold. As described above, the gate works independently of the audio signal after being ‘triggered’ by the signal crossing the gate threshold. The gate will remain open as long as the signal is above the threshold. How fast the gate opens to let the ‘good’ signal through is determined by the *attack time*. How long the gate stays open after the signal has gone below the threshold is determined by the *hold time*. How fast the gate closes is determined by the *release*. How much the gate attenuates the unwanted signal while closed is determined by the range.

In scientific research, it is widely accepted that if one needs to master a subject quickly, one must learn the associated vocabulary and industry terms. The same is true with audio recording and production. Most manuals and text books assume a prior level of knowledge. We hope this section will help you to get the greatest benefit from your independent study in the world of music production.

Compressors

- **Threshold.** The compressor threshold sets the level at which compression begins. When the signal is above the threshold setting, it becomes eligible for compression. Basically, as you turn the threshold knob counterclockwise, more of the input signal becomes compressed (assuming you have a ratio setting greater than 1:1).
- **Ratio.** Ratio sets the compression slope. The ratio is the relationship between the output level and the input level. In other words, the ratio sets the compression slope. For example, if you have the ratio set to 2:1, any signal levels above the threshold setting will be compressed at a compression ratio of 2:1. This simply means that for every 1 dB of level increase into the compressor, the output will only increase 0.5 dB, thus producing a compression gain reduction of 0.5 dB/dB. As you increase the ratio, the compressor gradually becomes a limiter.
- **Limiter.** A limiter is a compressor that is set to prevent any increase in the level of a signal above the threshold. For example, if you have the threshold knob set at 0 dB, and the ratio turned fully clockwise, the compressor becomes a limiter at 0 dB, so that the output signal cannot exceed 0 dB regardless of the level of the input signal.
- **Attack.** Attack sets the speed at which the compressor acts on the input signal. A slow attack time (fully clockwise) allows the beginning envelope of a signal (commonly referred to as the *initial transient*) to pass through the compressor unprocessed, whereas a fast attack time (fully counterclockwise) immediately subjects the signal to the ratio and threshold settings of the compressor.
- **Release.** Release sets the length of time the compressor takes to return the gain reduction back to zero (no gain reduction) after the signal level drops below the compression threshold. Very short release times can produce a very choppy or “jittery” sound, especially in low-frequency instruments such as bass guitar. Very long release times can result in an overcompressed sound; this is sometimes referred to as “squashing” the sound. All ranges of release can be useful at different times, however, and you should experiment to become familiar with the different sonic possibilities.
- **Hard/Soft Knee.** With hard-knee compression, the gain reduction applied to the signal occurs as soon as the signal exceeds the level set by the threshold. With soft-knee compression, the onset of gain reduction occurs gradually after the signal has exceeded the threshold, producing a more musical response (to some folks).

5.2.3 Vocabulary of Dynamics Processors

- **Auto.** Places a compressor in automatic attack and release mode. The attack and release knobs become inoperative and a preprogrammed attack and release curve is used.
- **Makeup Gain.** When compressing a signal, gain reduction usually results in an overall reduction of level. The gain control allows you to restore the loss in level due to compression. (Like readjusting the volume.)
- **Compressor Sidechain.** The sidechain jack interrupts the signal that the compressor is using to determine the amount of gain reduction it should apply. When no connector is inserted into this jack, the input signal goes directly to the compressor's control circuitry. When a connector is inserted into this jack, the signal path is broken. The control signal can then be processed by an equalizer, for example, to reduce sibilance (de-essing) in a vocal track. The control signal is then returned to the unit via the connector. One common application for a sidechain is when using a compressor to reduce the level of music or other background sound whenever a narrator speaks or vocalist sings, allowing the voice to be clearly heard. In this application, the vocal signal is routed to the sidechain input, while the music is routed through the main compression circuitry. Now the compressor will automatically duck—that is, reduce the level of—the music whenever the narrator speaks or the vocalist sings.

Expanders

- **Downward Expansion.** Downward expansion is the most common expansion used in pro audio and recording. This type of expansion reduces the level of a signal when the signal falls below a set threshold level. This is most common used for noise reduction.
- **Ratio.** The expansion ratio sets the amount of reduction applied to a signal once the signal has dropped below the expansion threshold. For example, a 2:1 expansion ratio attenuates a signal 2 dB for every 1 dB it drops below the threshold. Ratios of 4:1 and higher act much like a noise gate but without the ability to tailor the attack, hold, and release times.

Noise Gates

- **Threshold.** 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.
- **Attack.** The gate attack time sets the rate at which the gate opens. A fast attack rate is crucial for percussive instruments, whereas signals such as vocals and bass guitar require a slower attack. Too fast of an attack can, on these slow-rising signals, cause an artifact in the signal producing an audible click. All gates have the ability to click when opening but a properly set gate will never click.

- **Hold.** Hold time is used to keep the gate open for a fixed period after the signal drops below the gate threshold. This can be really useful for effects such as gated snare, where the gate remains open after the snare hit for the duration of the hold time, then abruptly closes.
- **Release.** The gate release time determines the rate at which the gate closes. 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” in percussive instruments. Longer release times usually eliminate “chattering” and should be set by listening carefully for the most natural release of the signal.
- **Range.** The gate range is the amount of gain reduction that the gate produces. Therefore, if the range is set at 0 dB, there will be no change in the signal as it crosses the threshold. If the range is set to -60 dB, the signal will be gated (reduced) by 60 dB, etc.

5.2.4 General Compressing 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 your 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 “brick wall,” 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

5.2.4

General Compressing Setting Suggestions

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. It is very contemporary.

Threshold	Ratio	Attack	Release
0 dB	1.9:1	1 ms	0.001 ms

5.3 Equalizers

The StudioLive is equipped with a 4-band semi-parametric equalizer on every input and output bus. Here's a brief explanation of how an EQ functions, as well as some charts to help you navigate the frequency ranges of various instruments so you can quickly choose the best EQ settings for your recordings and live mixes and to help you get the most out of your StudioLive.

5.3.1 What is an EQ?

An equalizer is a filter that allows you to adjust the level of a frequency, or range of frequencies, of an audio signal. In its simplest form, an EQ will let you turn the treble and bass up or down, allowing you to adjust the coloration of, let's say, your car stereo or an iPod. In recording, equalization is a sophisticated art. Good equalization is critical to a good mix.

When used correctly, an equalizer can provide the impression of nearness or distance, “fatten” or “thin” a sound, and help blend or provide separation between similar sounds in a mix allowing them to both shine through the mix.

The Parametric EQ

The Parametric EQ and semi-Parametric EQ are the most common equalizers found in recording and live situations because they offer continuous control over all parameters. A parametric EQ offers continuous control over the audio signal's frequency content, which is divided into several bands of frequencies (most commonly three to seven bands). A fully parametric EQ offers control over the bandwidth (basically, the range of frequencies affected), the center frequency of the band, and the level (boost/cut) of the designated frequency band. It may also offer separate control over the Q, which is the ratio of the center frequency to the bandwidth. A semi-parametric EQ, like those in the StudioLive, provides control over most of these parameters but usually the bandwidth is fixed.

The Q

Although a semi-parametric EQ such as those in the StudioLive does not provide a continuous bandwidth control, it offers an indirect way of switching between wide and narrow bandwidth: the Hi Q switch. Q is the ratio of center frequency to bandwidth, and if the center frequency is fixed, then bandwidth is inversely proportional to Q—meaning that as you raise the Q, you narrow the bandwidth. (In fully parametric EQs, you have continuous bandwidth control and/or continuous Q control, rather than a high/low Q switch.) This allows you to attenuate or boost a very narrow or wide range of frequencies.

A narrow bandwidth (high Q) has obvious attenuation benefits for removing unpleasant tones. Let's say the snare drum in your mix has an annoying ring to it. With a very narrow bandwidth, you can isolate this one frequency (usually around 1 kHz) and remove, or reject, it. This type of narrow band-reject filter is also known as a *notch* filter. By notching out the offending frequency, you can remove the problem without removing the instrument from the mix. A narrow bandwidth is also useful in boosting pleasant tones of an instrument such as the attack. Take for instance, a kick drum. A kick

5.3.1 What is an EQ?

drum resonates somewhere between 60 to 125 Hz, but the attack of the kick drum is much higher at 2 to 5 kHz. By setting a narrow bandwidth and boosting the attack a bit, you can achieve a punchier kick drum without overpowering the rest of the mix.

A broad bandwidth accentuates or attenuates a larger band of frequencies. The broad and narrow bandwidths (high and low Q) are usually used in conjunction with one another to achieve the desired effect. Let's look at our kick drum again. We have a kick drum that has a great, big low-end sound centered around 100 Hz and an attack hitting almost dead-on at 4 kHz. In this example, you would use a broad bandwidth in the low frequency band, centered at 100 Hz, and a narrow bandwidth boosted at 4 kHz. In this way you are accentuating the best and downplaying everything else this particular kick drum has to offer.

Shelving EQ

A shelving EQ attenuates or boost frequencies above or below a specified cutoff point. Shelving equalizers come in two different varieties: high-pass and low-pass.

Low-pass shelving filters pass all frequencies below the specified cutoff frequency while attenuating all the frequencies above it. A high-pass filter does the opposite: passing all frequencies above the specified cut-off frequency while attenuating everything below.

Graphic EQ

The StudioLive 16.4.2 features a 31 band Graphic EQ that can be inserted on the Main bus. A graphic EQ is a multi-band equalizer that uses sliders to adjust the amplitude for each frequency band. It gets its name from the positions of the sliders "graphing" the resulting frequency response curve. As explained in Section 2.10.4, the encoders in the Fat Channel are used to make amplitude adjustments, and the meter LED's display the "slider" positions. The center frequency and bandwidth are fixed for each band, the level (amplitude) for each band is the only adjustable parameter.

Graphic EQ's are generally used to fine-tune the overall mix it to 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(s). 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 (16Hz to 60 Hz): These very low 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 (60Hz to 250Hz): 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.

5.3.1 What is an EQ?

- Low-Mids (250Hz to 2 kHz): In general, you will want to emphasize the lower portion of this range and deemphasize the upper portion. Boosting the range from 250Hz to 500Hz will accent ambience in the studio and will add clarity to bass and lower frequency instruments. The range between 500Hz and 2 kHz can make mid-range instruments (guitar, snare, saxophone, etc) “honky” and too much boost between 1 kHz and 2 kHz will make your mix sound thin or “tinny”.
- High-Mids (2 kHz to 4 kHz): The attack in percussive and rhythm instruments occurs in this range. High-mids are also responsible for the projection of mid-range instruments.
- Presence (4 kHz to 6 kHz): This frequency range is responsible for the clarity of your 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.

5.3.2 **How to Find the Best and Leave the Rest**

Included at the end of this section are some general frequency principles to guide you through the wonderful world of equalization, but these are far from set in stone. So how do you find the best and worst each instrument has to offer and adjust them accordingly? Here's a quick guide:

First, solo just the instrument with which you are working. Most engineers start building their mix with the drums and work from the bottom up (kick, snare, toms, hi-hat, overheads). Each instrument resonates the most in a specific frequency band, so if you are working on your kick-drum mic, start with the lowest band of the EQ. Tune in the best-sounding low end and move on to the attack. It is not uncommon to hear an annoying ringing or a “twang” mixed in with your amazing-sounding low end and perfect attack, so your next task will be to find that offending frequency and notch it out. Once you are satisfied with your kick drum, mute it, and move on to the next instrument.

Taking your time with equalization is well worth the effort. Your mix will have better separation and more clarity when each instrument's EQ is set so that it shines through the mix. Some final advice:

- You can only do so much. Not every instrument can or should have a full, rich low end and a sharp attack. If every instrument is EQ'd to have the same effect, it will lose its identity in the mix. Your goal is not individual perfection, it is perfection in unity.
- Step away from the mix. Your ears get fatigued just like the rest of you. If you are working particularly hard on one instrument, your ears will be quite literally numbed to that frequency range.
- Your memory is not what you think it is. Comparing a flat EQ and the curve that you've created allows you to see exactly what you've done. So be honest with yourself. Sometimes that EQ setting you've been working on for 15 minutes is not the right choice, so move on.
- Never be afraid of taking a risk. The best EQ tricks were found by mad scientists of sound. “Playing” applies to engineers as well as musicians.

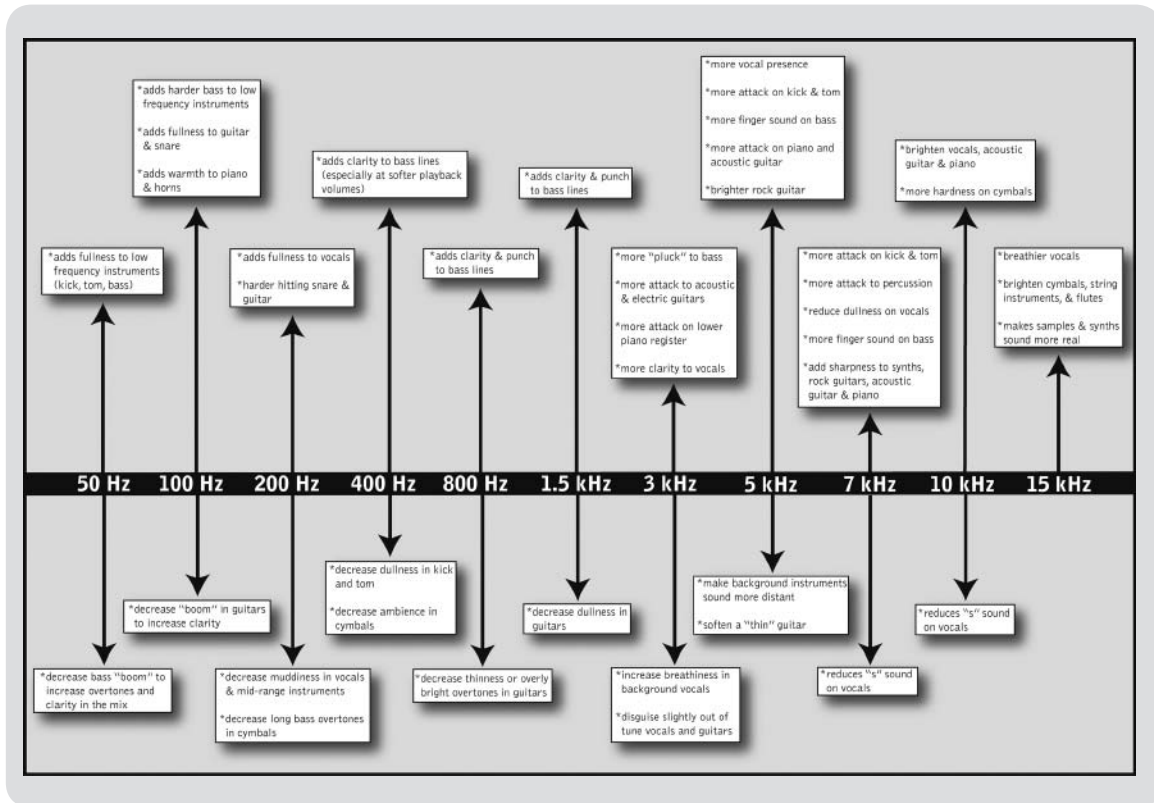
With every instrument there are frequencies that can be attenuated or boosted to add clarity or fullness to it in the mix. And of course, the wrong frequencies can make an instrument shrill, muddy, or just downright annoying. The following two charts suggest frequency ranges that should be accentuated or downplayed for the most common instruments. Remember, these are just suggestions; these frequencies may need to be adjusted up or down depending on the instrument, room, and microphone.

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

5.3.4 To Boost or Not To Boost

Table 2



5.3.4 General EQ Setting Suggestions

Included with your StudioLive is a library of Channel Strip presets. Section 3.8.3 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 4.2.4, 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 PEAK/SHELF	LOW FREQ (Hz)	LOW GAIN	LOW MID ON/OFF	LOW MID HI/LOW Q	LOW MID FREQ (Hz)	LOW MID GAIN
ON	PEAK	130	-2	ON	LOW	465	-2

HI MID ON/OFF	HI MID LO/HI	HI MID FREQ (kHz)	HI MID GAIN	HI ON/OFF	HI PEAK/SHELF	HI FREQ (kHz)	HI GAIN
ON	LO	2.4	+2	ON	PEAK	6.0	+8

Rock Female Vocals

LOW ON/OFF	LOW PEAK/SHELF	LOW FREQ (Hz)	LOW GAIN	LOW MID ON/OFF	LOW MID HI/LOW Q	LOW MID FREQ (Hz)	LOW MID GAIN
ON	SHELF	155	+4	ON	LOW	465	+6

HI MID ON/OFF	HI MID LO/HI	HI MID FREQ (kHz)	HI MID GAIN	HI ON/OFF	HI PEAK/SHELF	HI FREQ (kHz)	HI GAIN
ON	LO	1.4	+6	ON	PEAK	4.2	+2

Pop Male Vocals

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

HI MID ON/OFF	HI MID LO/HI	HI MID FREQ (kHz)	HI MID GAIN	HI ON/OFF	HI PEAK/SHELF	HI FREQ (kHz)	HI GAIN
ON	LO	2.0	+2	ON	PEAK	7.2	+4

Rock Male Vocals

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

HI MID ON/OFF	HI MID LO/HI	HI MID FREQ (kHz)	HI MID GAIN	HI ON/OFF	HI PEAK/SHELF	HI FREQ (kHz)	HI GAIN
ON	HI	2.4	-2	ON	SHELF	7.2	+4

5.3.4 General EQ Setting Suggestions

Percussion

Snare

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

HI MID ON/OFF	HI MID LO/HI	HI MID FREQ (kHz)	HI MID GAIN	HI ON/OFF	HI PEAK/SHELF	HI FREQ (kHz)	HI GAIN
ON	LO	1.6	+4	ON	SHELF	4.2	+4

Left/Right (Stereo) Overheads

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

HI MID ON/OFF	HI MID LO/HI	HI MID FREQ (kHz)	HI MID GAIN	HI ON/OFF	HI PEAK/SHELF	HI FREQ (kHz)	HI GAIN
ON	LO	2.9	+2	ON	SHELF	8	4

Kick Drum

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

HI MID ON/OFF	HI MID LO/HI	HI MID FREQ (kHz)	HI MID GAIN	HI ON/OFF	HI PEAK/SHELF	HI FREQ (kHz)	HI GAIN
ON	LO	1.6	0	ON	SHELF	6.0	+4

5.3.4 General EQ Setting Suggestions

Fretted Instruments

Electric Bass

LOW ON/OFF	LOW PEAK/SHELF	LOW FREQ (Hz)	LOW GAIN	LOW MID ON/OFF	LOW MID HI/LOW Q	LOW MID FREQ (Hz)	LOW MID GAIN
ON	SHELF	36	-8	ON	HI	130	+4

HI MID ON/OFF	HI MID LO/HI	HI MID FREQ (kHz)	HI MID GAIN	HI ON/OFF	HI PEAK/SHELF	HI FREQ (kHz)	HI GAIN
ON	LO	2.0	+4	ON	SHELF	4.2	0

Acoustic Guitar

LOW ON/OFF	LOW PEAK/SHELF	LOW FREQ (Hz)	LOW GAIN	LOW MID ON/OFF	LOW MID HI/LOW Q	LOW MID FREQ (Hz)	LOW MID GAIN
ON	PEAK	155	+4	ON	LOW	665	+2

HI MID ON/OFF	HI MID LO/HI	HI MID FREQ (kHz)	HI MID GAIN	HI ON/OFF	HI PEAK/SHELF	HI FREQ (kHz)	HI GAIN
ON	LO	2.0	0	ON	SHELF	6.0	+4

Distorted Electric Guitar

LOW ON/OFF	LOW PEAK/SHELF	LOW FREQ (Hz)	LOW GAIN	LOW MID ON/OFF	LOW MID HI/LOW Q	LOW MID FREQ (Hz)	LOW MID GAIN
ON	PEAK	320	+6	ON	LOW	960	0

HI MID ON/OFF	HI MID LO/HI	HI MID FREQ (kHz)	HI MID GAIN	HI ON/OFF	HI PEAK/SHELF	HI FREQ (kHz)	HI GAIN
ON	HI	3.5	+4	ON	SHELF	12	0

Keyboards

Piano

LOW ON/OFF	LOW PEAK/SHELF	LOW FREQ (Hz)	LOW GAIN	LOW MID ON/OFF	LOW MID HI/LOW Q	LOW MID FREQ (Hz)	LOW MID GAIN
ON	SHELF	108	-2	ON	LO	665	+2

HI MID ON/OFF	HI MID LO/HI	HI MID FREQ (kHz)	HI MID GAIN	HI ON/OFF	HI PEAK/SHELF	HI FREQ (kHz)	HI GAIN
ON	LO	2.9	+2	ON	PEAK	7.2	+4

5.4. Subgroup Mixing

A subgroup allows you to combine multiple channels in a single bus so that the overall level for the entire group is controlled by a single fader. In addition to level control, the StudioLive 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 also be soloed and muted.

You will find many uses for subgroups, nearly all of them make mixing more convenient and provide better control of your mix. In this section, we will explore two different ways in which subgroups can help you to create a more efficient mixing environment and a more successful live mix.

5.4.1 Instrument groups

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 be using a drum group in this particular example but these principles can be applied to group 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.

The first step in creating a subgroup is to get a good mix of the instruments you are grouping together, in this case, the drums. With the drummer's assistance, set the input trim, EQ, and dynamics for each drum separately. As you select and solo each channel, assign each channel to be routed to Subgroup 1.

After you have gone through the entire kit and are satisfied with each channel's level, EQ, and dynamics, unsolo your final channel. Have the drummer play the entire kit and set the relative



5.4.1 Instrument Groups

volume and panning for each mic in the mix. Choose the Select button above Subgroup 1. In the Fat Channel's Stereo section (to the right of the Pan display), enable Link, and 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.

Now assign Subgroup 1 to the Main outputs; since Subgroup 2 is linked to Sub 1, it is automatically assigned to the Main outs as well. 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.

5.4.2 Effects Group

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 effect, a member of the band. This is especially useful when employing specialty or signature effects. What would an electronic band sound like if the vocal mix didn't dramatically trail off in a wash of reverb—or a reggae band without delay on theirs? The StudioLive allows you to assign either or both of the onboard effects buses to a subgroup. So let's take the example of the reggae band.

In this example we will assign the delay on EFX A (effects bus A) to Subgroup 3. Press the FX button in the Master Control section to access the Effects menu. In the EFX A parameters, use the Value Encoder to scroll through the effects library until you find a suitable delay, and adjust its parameters according to taste. (See Section 2.10.1 for complete operation instructions.) Next, decide which channels should be sent to the effects bus. In dub and reggae music, the vocals are most commonly 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 EFX A. Using your meter section, locate the send encoders for channels 10 and 11 and turn them to a little more than 50%. Press the Select button for EFX A and assign this bus to Subgroup 3 and to the main outputs. If you like you can also add some dynamics processing and EQ at this point as well.

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 EFX A's output turned up, and use the limiter for Subgroup 3 to keep the level under control.) The level of the vocal delay is now controlled by the

Subgroup 3 fader, and you can use it to season your 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.



There are several advantages to assigning an effect like delay or reverb to a subgroup rather than simply leaving it on the effects aux bus:

- You can quickly add or subtract the effect by grabbing a fader.
- The effect can be muted or soloed.
- The performers on stage can have a different amount of the effect 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 the best performance possible.

5.5 Aux Bus Mixing

The Aux Bus provides outputs to create auxiliary mixes that are separate from the main and subgroup mixes. The StudioLive is equipped with 8 aux buses: Aux 1-6, which have physical output jacks, and EFX A and B, which are the internal effects buses. 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 subgroup buses, the StudioLive allows you to add global dynamics processing and EQ to these aux buses in addition to the individual channel processing.

5.5.1 Monitor Mixing

In both the live and studio environments, creating custom monitor mixes for your musicians is critical. If a musician can't hear themselves or their band mates, their performance will suffer. A monitor mix can be either mono or stereo. Most often, an individual live monitor mix is mono and is sent to a floor-wedge monitor. (The obvious exception is in-ear monitor systems.) A studio monitor mix is usually stereo and is sent to a headphone amplifier, so it requires both a left and a right channel input. In either case, the function of the aux bus does not change.

As an example, let's create a mono monitor mix on Aux 1. To begin, press the Mix button in the Aux 1 section. The meter section of the StudioLive will now display the amount of send to this aux bus from each of the 16 channels. Keep in mind that the aux mix is completely independent of every other output (main bus, subgroups, direct out, etc.). The encoders below each meter control the channel send level to Aux 1. Use them the same way you use the 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 to create the best mix for them.

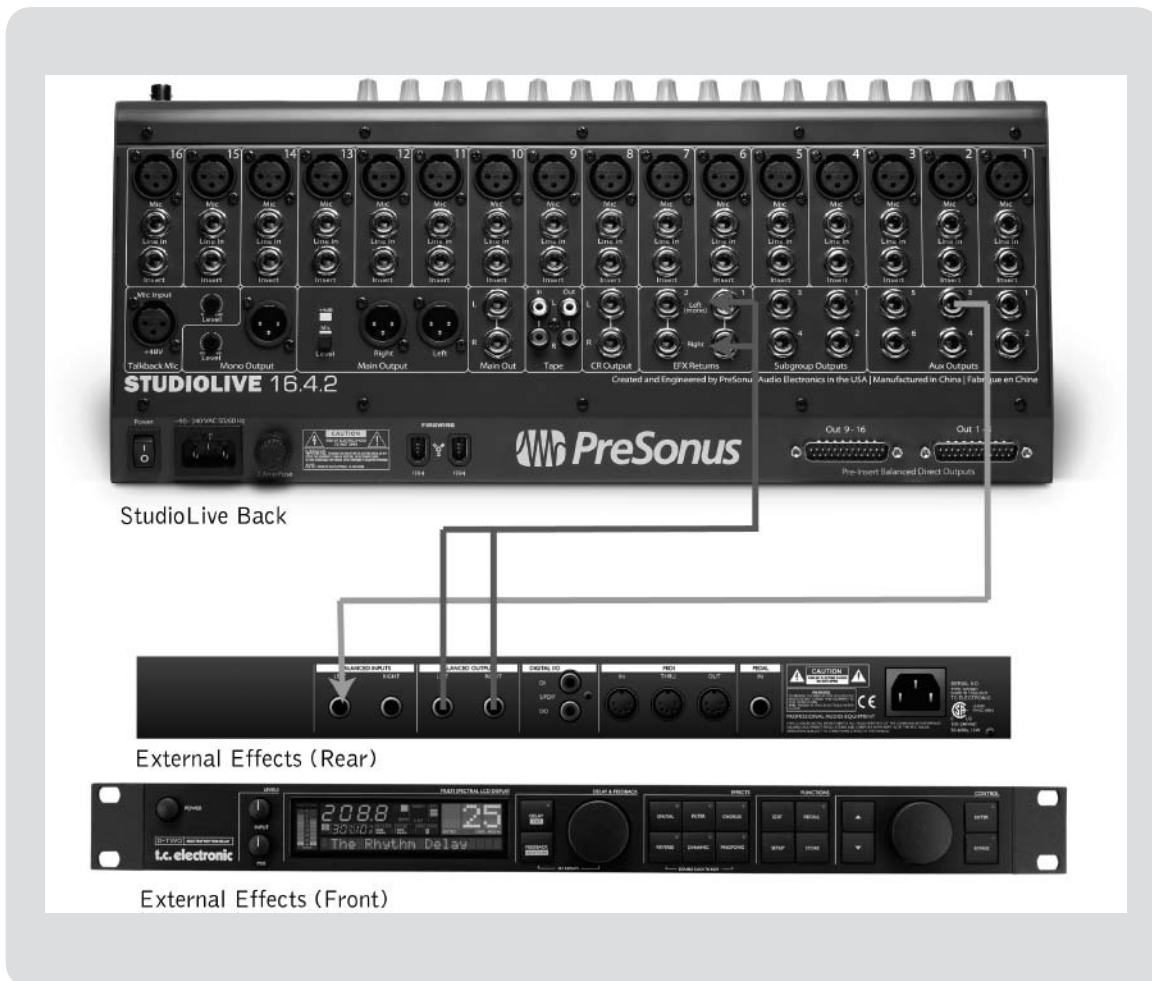
By pressing the Select button for Aux 1, you can add in some dynamic processing and EQ to the overall monitor mix. These are especially useful in eliminating feedback in a monitor. Keep in mind, an EQ can also be used to increase the presence of an instrument by boosting that particular frequency range 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. You can listen to the mix you are creating in any of the auxes in either your headphones or your control-room monitor by simply soloing the aux and selecting Solo as the source in the Monitor section.

4.5.2 Effects Processing

There are at least two advantages to using an aux bus for an effects processor rather than using a channel insert: several channels can be sent to a single processor, and you can vary the level sent from each channel to the processor, allowing you to create an effects mix.

The StudioLive features two internal effects buses. These are used much in the same way the aux buses are used to create monitor mixes, as described in the previous section. This section will detail how to utilize an external effects processor with your StudioLive mixes.

In this example, we will use Aux 3 to feed an external effects processor. To begin, connect your external effects processor to your StudioLive as detailed in the hookup diagram below:

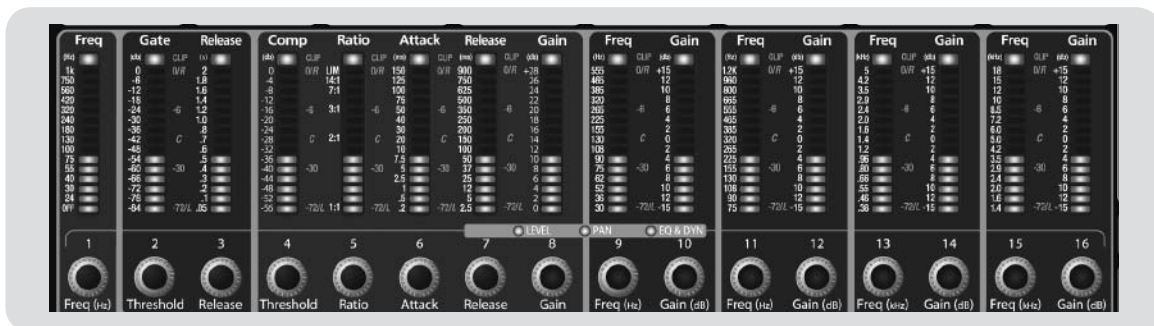


5.5.2 **Effects Processing**

Turn the Output knob in the Aux 1 section to 12 o'clock and press the Select button. In the Fat Channel, assign Aux 1 to the Main outputs.



Press the Mix button for Aux 3. The meter section of the StudioLive will now show the output levels of each of the 16 channels. The encoders below each meter control the output level of each channel into Aux 3. You will use these encoders to set the send levels from the aux bus to the effects processor, the same way you used them to create a monitor mix, only in this case you will decide which channels will be sent to the external multi-effects processor and how much will be sent.



Let's say that you are inserting an external 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 to the 7 or 8 o'clock position so that only a small portion of the kick drum input will be affected by the reverb.

Once you have determined your effects mix, you can press the Select button for Aux 3 to add dynamics processing and EQ to the aux mix before it is sent to the external effects processor. The effects processor's output is patched to Aux Input A, so you can use the Select button for Aux Input A to add Fat Channel dynamics processing and EQ to the reverb-enhanced signal. The knob for Aux Input A controls the level of the aux mix relative to the level of your main mix.

5.6 Digital Effects

The StudioLive includes a stereo effects processor that features the two most common type of effects is use today for live sound: reverb and delay.

.6.1 Reverb

Reverberation or *reverb*, as it is more commonly known, is perhaps the most widely used effect. Natural reverb is created by sound waves being reflected 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 as the nature of the space around us based partially on audio reflections, a sense of space in an audio recording sounds more natural, and therefore more pleasing to the listener.

There are several parameters that can be adjusted in a reverb effect:

Decay. Decay is the time required for the reflections (reverberation) to die away. In most modern music production, reverb decay times 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.

Pre-delay. Pre-delay is the time between the end of the initial sound and the beginning of the first reflections being 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 pre-delay 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.

HF and LF decay. The types of surfaces in a space also affect the sound. Carpet and soft furnishings will absorb more high-frequency waves, thereby reducing the high-frequency decay time, while hard surfaces such as tile or stone reflect sound extremely well, resulting in a "brighter" ambience. Similarly, the high-frequency (HF) and low-frequency (LF) decay time allow you to adjust the "brightness" or "darkness" of the reverb, enabling you to better emulate these environmental factors.

5.6.2 Digital Delay

A digital delay emulates an analog tape-echo unit. A tape-echo is a special type of tape recording. It uses one head to record a signal and up to four heads to replay it. The delay time is set by the tape speed and the space between the heads. A digital delay essentially performs exactly the same function as a tape-echo, but offers more refinements and adjustments than its mechanical counterpart. 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.

Delay time. Delay time is the time 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 distinct echoes can be used to create thickening effects. Whether these echoes are timed with the tempo is a matter of stylistic choice.

Variable feedback. The 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.

Note: 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.

5.7 Level Setting Procedure

Setting the proper levels is an important part of getting the right sound. The following steps will assist you in quickly setting your levels.

- 1) Turn each of the 16 trims to 0/-20.



- 2) Press the Input button in the Meter section.



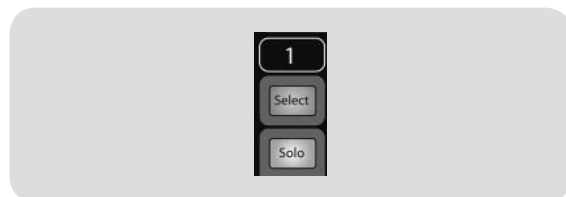
- 3) In the Solo bus section, select PFL and turn the Cue level to 12:00.



- 4) Select the Solo button in the Monitor section and adjust the volume for your headphones or control-room monitors.



- 5) Solo your first channel and turn the trim to the desired level on the meter. Be careful not get the signal above 0 dBFS. You do not want to clip the ADC converters.





When all else fails, just remember: button, button, knob, button, knob, button, knob

5.8 The Solo Bus

The StudioLive features an independent Solo bus. This feature is extremely useful in setting levels for monitor mixes, dialing in dynamics 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-Fade Listen).** AFL sends the channel or subgroup signal to the Solo bus post-fader so you can control the level of the soloed signal with the fader. This is the StudioLive's default setting.
- **PFL (Pre-Fade 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 channel(s) will be sent to their assigned outputs. While very useful in dialing in dynamics during sound check, this mode is very dangerous during a live show situation. It is highly recommended that this mode be turned off when mixing in strictly live applications.

5.8.2 Using the Solo Bus for Monitoring

When mixing live or when recording multiple musicians at once, it is often necessary to quickly listen in on 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 headphones, it is necessary to connect the speakers to the TRS Control Room outputs on the back of your StudioLive rather than to one of the main output pairs.

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. Next, press the Solo buttons on the channels and subgroups you want to monitor. Turn the Cue knob in the Solo section to about 12 o'clock. Finally, select the Solo button in your monitor bus and dial in a comfortable listening volume for you headphones or monitors. You can increase the overall volume of the Solo bus using the Cue knob in the Solo section.

This feature can also be used to listen in on a monitor mix that is being routed to an aux send. Let's say your vocalist on stage is complaining 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 fix his monitor mix quickly. This application is also useful in heading off a feedback problem.

5.8.2 Destructive Soloing

Destructive Soloing or Solo in Place (SIP) is a great way to tune each channel's dynamics individually in live mix situations or do some surgical editing in the studio. SIP mode mutes every Channel and Bus that is not soloed *in the main bus* (i.e. 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. It is highly recommended that you drop out of this mode once the show has started.

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

In Section 5.7, we discussed 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 discussed earlier in Section 4.3.2, most engineers start with the drums and work from the bottom up. To begin, bring all your faders down and raise your main fader to unity gain. Press and hold the SIP button in the Cue section until it illuminates and press the Solo button on your kick-drum mic channel. Notice that all the other channels on your StudioLive have been muted. Bring up the fader on the kick-drum channel and press the channel's Select button. The Fat Channel will now display the dynamics processing, EQ, output routing, and pan settings for the kick drum. Using the encoders and meters in the Fat Channel, set up the compressor and EQ for this channel. Once you are satisfied, bring the fader back down and press the Solo button again. Next, press the Solo button on the snare mic channel and repeat this procedure. In this way continue with each drum mic and then move on to the other instruments that are connected to your StudioLive. When you have finished with all the instruments, press the SIP button again and slowly bring up your faders to set up your mix.

6.1 Troubleshooting

Please note that many technical issues can arise when using a standard computer as a digital audio workstation (DAW). PreSonus can only provide support for issues that directly relate to the StudioLive interface and Capture audio-recording software. PreSonus does not provide support for computer hardware, operating systems, and non-PreSonus hardware and software, and it may be necessary to contact the manufacturer of these products for technical support. Please check our Web site, www.presonus.com, regularly for software information and updates, firmware updates, and support documentation for Frequently Asked Questions. Individual technical assistance may be received by calling PreSonus at 225-216-7887, Monday through Friday between the hours of 9 am and 6 pm CST. PreSonus Technical Support is available via email during the same hours at techsupport@presonus.com.

Pops and Clicks

Pops and clicks in your audio could be caused by momentary losses of synchronization between the StudioLive and its clock source. This can be due to a FireWire card that is not suited for FireWire audio devices or to a lack of CPU resources or available memory. Try closing all unnecessary programs, increasing the buffer size in the FireStudio Control Console, and optimizing your operating system for audio.

Other known causes of this issue are wireless networks and running other FireWire devices (FW400 or FW800) in parallel to your StudioLive. Because of this, it is recommended that PC users disable their wireless network from their System Control Panel and Mac users turn AirPort off while the StudioLive is connected to their computer. If other FireWire devices, like external hard drives, need to be connected to your computer at the same time, connect them to the auxiliary FireWire port on the back of your StudioLive or install a dedicate FireWire bus on your computer for either the StudioLive or your FireWire peripherals.

Recommended FireWire Chipsets

The StudioLive will work with a wide range of FireWire cards and configurations. However, due to the plethora of FireWire chipsets currently on the market, it is not possible to thoroughly test each one for compatibility. Most users will not need to alter their current PC configuration to use their StudioLive. The following FireWire chipsets, however, have been tested and approved for use with the StudioLive:

- Agere/LSI FW323-06
- TI TSB43AB23
- VIA VT6308
- VIA VT6306 (on some older motherboards this chipset will only support a limited number of playback channels: 32 channels at 44.1 or 48 kHz; 16 channels at 88.2 or 96 kHz)

6.1 Troubleshooting

At the time of the publication of this manual, PreSonus only recommends the following Express cards:

- ADS Pyro 1394a
- StarTech EC13942

Known Incompatible Hardware

- **ATI RADEON 9000/9001 IGP video chipset.** Symptoms are consistent click and pops during audio playback. This video chipset is only found in PC laptops and is entirely integrated as the computer's only video controller. PreSonus strongly recommends that users do not purchase a system with this chipset as there is currently no workaround for this incompatibility.
- **USB/FireWire and s400/s800 combo cards.** Symptoms usually include no audio recording/playback, even though the device will install and sync, as well as erratic audio performance. Rarely, combo cards will prevent the device from installing or achieving a stable sync. We recommend FireWire cards that have s400 FireWire connections only.
- **FireWire cards with NEC chipsets.** Symptoms include installation issues, erratic audio and extraneous static and noise.
- **Motherboards with nForce4 chipsets.** Symptoms include reduced or very poor performance, especially if using the onboard FireWire connection. Installing a PCIe (not PCI) FireWire 400-only card with an approved chipset is a known workaround but may not allow full performance.

StudioLive Control Panel will not launch

The StudioLive Control Panel will not launch if the StudioLive is not connected and synced to your computer. Verify that your FireWire cable is connected both to your computer and to your StudioLive. Navigate to Page 2: Digital in the StudioLive System menu and verify that the FireWire status is locked.

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. 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.

6.1 Troubleshooting

Fader Movements have no effect on audio

Verify that your StudioLive is not locked by navigating to Page 4: 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 EFX A and B 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 EFX 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 Cue 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.

6.2 Specifications

Microphone Preamp

Type	XLR Female, balanced
Frequency Response to Direct Output (at unity gain)	20-40 kHz, 0 / -0.5 dBu
Frequency Response to Main Output (at unity gain)	20-20 kHz, +/- 0.5 dBu
Input Impedance	1 k Ω
THD to Direct Output (1 kHz at unity gain).....	< 0.005%, +4 dBu, 20 -20 kHz, unity gain, unwt'd
THD to Main Output (1 kHz at unity gain)	<0.005%, +4 dBu, 20-20 kHz, unity gain, unwt'd
EIN to Direct Output.....	+125 dB unwt'd,, +130 dB A-wtd
S/N Ratio to Direct Output (Ref = +4 dB, 20 kHz BW, unity gain, A-wtd)	-97 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).....	-16 dB to +67 dB
Maximum Input Level (unity gain)	+16 dBu
Phantom Power (+/- 2 VDC)	+48 VDC

Line Inputs

Type	1/4" TRS Female, balanced mono
Frequency Response to Direct Outputs (at unity gain).....	10-40 kHz, 0 / -0.5 dBu
Frequency Response to Main Outputs (at unity gain)	20-20 kHz, +/- 0.5 dBu
Input Impedance	10 k Ω
THD to Direct Output (1 kHz at unity gain).....	<0.0007%, +4 dBu, 20-20 kHz, unity gain, unwt'd
THD to Main Output (1 kHz at unity gain)	<0.005%, +4 dBu, 20-20 kHz, unity gain, unwt'd
S/N Ratio to Direct Output (Ref = +4dBu, 20 kHz BW, unity gain, A-wtd)	-105 dB
S/N Ratio to Main Output (Ref = +4dBu, 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	+22 dBu

Auxiliary Inputs

Type	1/4" TRS Female, balanced (2 stereo pairs)
Maximum Input Level	+22 dBu

Main Outputs

Type	XLR Male, balanced (stereo pair); 1/4" TRS Female, balanced (stereo pair); XLR Male, balanced (mono)
Rated Output Level	+24 dBu
Output Impedance	10 k Ω

Aux Outputs

Type	1/4" TRS Female, balanced (mono)
Rated Output Level	+18 dBu
Output Impedance	51 Ω

6.2 Specifications

Subgroup Outputs

Type 1/4" TRS Female, balanced (mono)
 Rated Output Level +18 dBu
 Output Impedance 51 Ω

Tape Outputs

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

Control Room Outputs

Type 1/4" TRS Female, balanced (stereo pair)
 Rated Output Level +18 dBu
 Output Impedance 51 Ω

System Cross Talk

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

Noise Gate / Expander

Threshold Range -84 dB to 0 dB
 Attack Time 0.5 ms
 Release Time 0.05s to 2s
 Expander Attenuation Range 2:1
 Noise Gate Attenuation Range ∞

Compressor

Threshold Range -56 dB to 0 dB
 Ratio 1:1 to 14:1
 Attack Time 0.2 ms to 150 ms
 Release Time 40 ms to 1000 ms
 Auto Attack and Release Attack = 10 ms, Release = 150 ms
 Curve Types hard and soft knee

EQ

Type 2nd order shelving filter (Q = 0.55)
 Low (Lowpass 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 (Highpass or Bandpass) 1.4 kHz to 18 kHz, ± 15 dB

6.2 Specifications

Digital Audio

ADC Dynamic Range (A-wtd, 48 kHz).....	118 dB
DAC Dynamic Range (A-wtd, 48 kHz).....	118 dB
FireWire.....	S400, 400Mb/s
Internal Processing	32-bit, floating point
Sampling Rate.....	44.1, 48 kHz
A/D/A Bit Depth.....	24
Reference Level for 0 dBFS.....	-18 dBu

Clock

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

Power

Connector	IEC
Input-Voltage Range.....	90 to 240 VAC (factory-configured for country of destination)
Power Requirements (continuous).....	100 W

Physical

Length.....	22.35 inches (568.06 mm)
Width (chassis only)	17.22 inches (437.40 mm)
Width (with rack ears).....	19 inches (482.60 mm)
Maximum Height.....	6.90 inches (175.26 mm)
Weight.....	23 Lbs.

6.3 Channel Strip 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 50 additional empty storage locations for you to build your own custom library of channel strip settings.

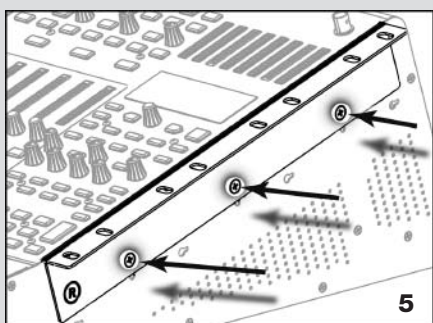
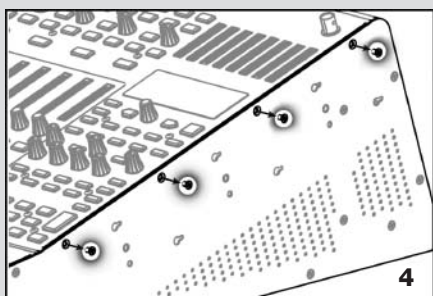
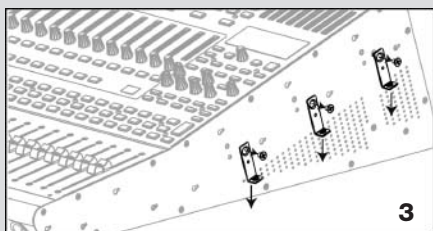
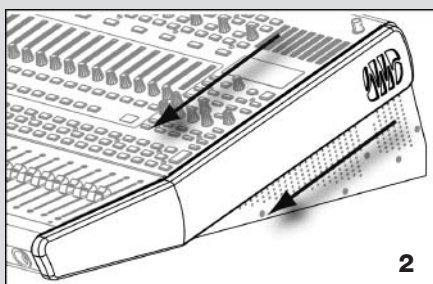
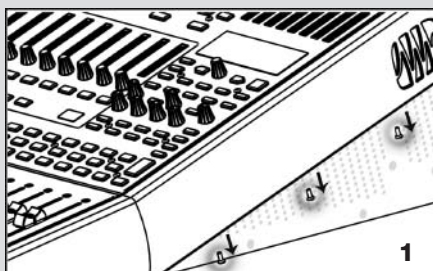
01	DRM: Kick 1	34	KEY: Vibes	67	EMPTY LOCATION
02	DRM: Kick 2	35	HRN: Trumpet	68	EMPTY LOCATION
03	DRM: Kick Funk 1	36	HRN: Trombone	69	EMPTY LOCATION
04	DRM: Kick Funk 2	37	HRN: Sax	70	EMPTY LOCATION
05	DRM: Kick Hip-Hop	38	HRN: Sax Solo	71	EMPTY LOCATION
06	DRM: Kick Jazz	39	PRC: Congas	72	EMPTY LOCATION
07	DRM: Snare 1	40	PRC: Bongos	73	EMPTY LOCATION
08	DRM: Fat Snare	41	PRC: Cowbell	74	EMPTY LOCATION
09	DRM: Snare Crackalak	42	PRC: Tambourine	75	EMPTY LOCATION
10	DRM: Snare Snappy	43	VOX: Male 1	76	EMPTY LOCATION
11	DRM: Toms Mid	44	VOX: Male 2	77	EMPTY LOCATION
12	DRM: Toms Low	45	VOX: Male 3	78	EMPTY LOCATION
13	DRM: Toms High	46	VOX: Female 1	79	EMPTY LOCATION
14	DRM: Overhead Rock	47	VOX: Female 2	80	EMPTY LOCATION
15	DRM: Overhead Jazz	48	VOX: Female 3	81	EMPTY LOCATION
16	DRM: High Hat	49	VOX: Speech 1	82	EMPTY LOCATION
17	BAS: Electric 1	50	VOX: Speech 2	83	EMPTY LOCATION
18	BAS: Electric 2	51	EMPTY LOCATION	84	EMPTY LOCATION
19	BAS: Slap	52	EMPTY LOCATION	85	EMPTY LOCATION
20	BAS: Upright	53	EMPTY LOCATION	86	EMPTY LOCATION
21	GTR: Rock 1	54	EMPTY LOCATION	87	EMPTY LOCATION
22	GTR: Rock 2	55	EMPTY LOCATION	88	EMPTY LOCATION
23	GTR: Funk	56	EMPTY LOCATION	89	EMPTY LOCATION
24	GTR: Metal	57	EMPTY LOCATION	90	EMPTY LOCATION
25	GTR: Jazz	58	EMPTY LOCATION	91	EMPTY LOCATION
26	GTR: Acoustic	59	EMPTY LOCATION	92	EMPTY LOCATION
27	GTR: Acoustic Strumming	60	EMPTY LOCATION	93	EMPTY LOCATION
28	GTR: Acoustic Fingerstyle	61	EMPTY LOCATION	94	EMPTY LOCATION
29	KEY: Piano Bright	62	EMPTY LOCATION	95	EMPTY LOCATION
30	KEY: Piano Warm	63	EMPTY LOCATION	96	EMPTY LOCATION
31	KEY: Piano Jazz	64	EMPTY LOCATION	97	EMPTY LOCATION
32	KEY: Piano Electric	65	EMPTY LOCATION	98	EMPTY LOCATION
33	KEY: Piano Electric 2	66	EMPTY LOCATION	99	EMPTY LOCATION

6.4 Effects Library

Your StudioLive contains a library of 50 custom Reverb and Delay presets designed by PreSonus. In addition to these presets, there are 50 free locations for you to build your own custom effects library. The Factory Presets can be altered, renamed, and overwritten.

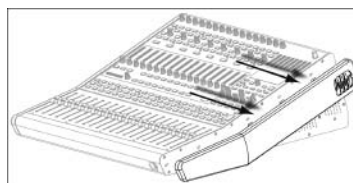
	TYPE	NAME	No	TYPE	NAME
F1	AMBIENCE	Natural	F51		
F2		Lively	F52		
F3	SMALL ROOM	Closet	F53		
F4		Studio A	F54		
F5		Studio B	F55		
F6		Bedroom	F56		
F7	BRIGHT ROOM	Kitchen	F57		
F8		Tile Floors	F58		
F9		Tile Bathroom	F59		
F10		Concrete Floors	F60		
F11	SMALL HALL	Radio Booth	F61		
F12		Small Club	F62		
F13		Big Club	F63		
F14	BRIGHT HALL	Brick House	F64		
F15		Linoleum Room	F65		
F16		Tile Room	F66		
F17	WARM HALL	Log Cabin	F67		
F18		Wood Floors	F68		
F19		Brick Club	F69		
F20		High Ceilings	F70		
F21	GATED HALL	GateVerb Short	F71		
F22		GateVerb Med	F72		
F23		GateVerb Long	F73		
F24		GateVerb Extreme	F74		
F25	LARGE HALL	Church	F75		
F26		Cathedral	F76		
F27		Gymnasium	F77		
F28		Arena	F78		
F29	PLATE	PlateVerb Shimmer	F79		
F30		PlateVerb Thick	F80		
F31		PlateVerb Drums	F81		
F32		PlateVerb Vox	F82		
F34	MONO DELAY	M: Short Slap	F83		
F35		M: Long Slap	F84		
F36		M: Triplet	F85		
F37		M: Triplet	F86		
F38	FILTER DELAY	Analog Slap	F87		
F39		Analog Trip	F88		
F40		Analog 8 th	F89		
F41	STEREO DELAY	Slap Quick	F90		
F42		Long Slap	F91		
F43		Spacey	F92		
F44		Long Tail	F93		
F45		ST: Triplet	F94		
F46	PING PONG DELAY	Ping Pong Quick	F95		
F47		Ping Pong Slap	F96		
F48		Ping Pong Spacey	F97		
F49		Ping Pong Trip	F98		
F50		Ping Pong Purple Rain	F99		

6.5 Rack Ear Installation Instructions



Step 1: Using a Philips screwdriver, remove the three screws beneath the decorative plastic side panel on the right side of your StudioLive. Set these screws aside. You will need them should you want to reinstall the side panel at some point in the future.

Step 2: Gently slide the plastic side panel toward the front of your StudioLive until you hear a click. Pull the side panel toward you to remove it.

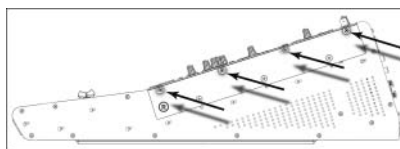


Step 3: Once the side panel is removed, you will see three metal clips. Remove the screw that attaches each of these clips to your StudioLive. Set aside these screws. You will need them to attach the rack ear to your StudioLive.

It is recommended that you attach the clips back to the decorative side panel using the screws you removed in Step 1 to store them. Both the clips and the screws are necessary to reattach the side panel.

Step 4: Remove the four screws at the top. These 5mm screws will be replaced with slightly longer ones. Set them aside and store them with the side panel.

Step 5: Using the screws you removed from the metal clips in Step 3, attach the Rack



Ear labeled 'R' to your StudioLive. In your Rack Ear Installation Kit, you will find four 6mm screws. Thread these through the four holes at the top of the Right Rack Ear and attach them to your StudioLive.

Repeat these steps to attach the Left Rack Ear to your StudioLive.

6.6

Session Data Recall Sheet

DATE: _____ ARTIST: _____ SCENE: _____

TRIM		PRODUCTION NOTES		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: _____	
LEVEL		SOURCE: _____		LEVEL		SOURCE: _____	
Aux In A		NOTES: _____ MONO (L) <input type="checkbox"/> STEREO (L+R) <input type="checkbox"/>		Aux In B		NOTES: _____ MONO (L) <input type="checkbox"/> STEREO (L+R) <input type="checkbox"/>	
LEVEL				LEVEL			
Aux 1		SEND TO: _____ MONITOR <input type="checkbox"/> EFFECTS <input type="checkbox"/>		Aux 4		SEND TO: _____ MONITOR <input type="checkbox"/> EFFECTS <input type="checkbox"/>	
Aux 2		SEND TO: _____ MONITOR <input type="checkbox"/> EFFECTS <input type="checkbox"/>		Aux 5		SEND TO: _____ MONITOR <input type="checkbox"/> EFFECTS <input type="checkbox"/>	
Aux 3		SEND TO: _____ MONITOR <input type="checkbox"/> EFFECTS <input type="checkbox"/>		Aux 6		SEND TO: _____ MONITOR <input type="checkbox"/> EFFECTS <input type="checkbox"/>	
EFX A		NOTES: _____		EFX B		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																				