

# StudioLive™ 24.4.2

## Owner's Manual



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## 0.1 Important Safety Instructions

 The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in this manual.

 The lightning flash with arrowhead symbol within an equilateral triangle is intended to alert the user to the presence of uninsulated "dangerous" voltage within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to humans.

 CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT REMOVE THE COVER. NO USER-SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED PERSONNEL.

 CAUTION: To reduce the risk of electric shock, do not expose this appliance to rain and moisture. The apparatus shall not be exposed to dripping or splashing liquids and no object filled with liquids, such as vases, shall be placed on the apparatus.

 CAUTION: These service instructions are for use by qualified service personnel only. To reduce the risk of electric shock, do not perform any servicing other than that contained in the operation instructions. Repairs must be performed by qualified service personnel.

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use this apparatus near water.
6. Clean only with dry a cloth.
7. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
8. Do not install near any heat sources, such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
9. Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades, with one wider than the other. A grounding-type plug has two blades and a third grounding prong. The wide blade and the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
10. Protect the power cord from being walked on or pinched, particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
11. Use only attachments/accessories specified by PreSonus.

12. Use only with the cart, stand, tripod, bracket, or table specified by the manufacturer or sold with this apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.



13. Unplug this apparatus during lightning storms or when unused for long periods of time.

14. Servicing is required when the apparatus has been damaged in any way, such as if a power-supply cord or plug is damaged; or liquid has been spilled, or objects have fallen, into the apparatus; or if the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped. All PreSonus products in the USA 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 USA should contact their local distributor. Your distributor's contact information is available at www.presonus.com.

15. The apparatus shall be connected to a Mains power outlet with a protective grounding/earthing connection.

16. Where the Mains plug or an appliance coupler is used as the disconnect device, the disconnect device shall remain readily operable.

### EU Directives on the Protection of the Environment and Other Euro Stuff

**RoHS** This product is compliant with the EU Directive 2002/95/EG for the Restriction of the use of Certain Hazardous Substances in Electrical and Electronic Equipment. No lead (Pb), cadmium (Cd), mercury (Hg), hexavalent chromium (Cr+6), PBB or PBDE is intentionally added to this device. Any traces of impurities of these substances contained in the parts are below the RoHS specified threshold levels.

**REACH** This product is compliant with the European Union Directive EC1907/2006 for the Registration, Evaluation, Authorization, and Restriction of chemicals (REACH) and contains none or less than 0.1% of the chemicals listed as hazardous chemicals in the REACH regulation.

**WEEE** As with the disposal of all old electrical and electronic equipment, this product is not to be treated as regular household waste. Instead it shall be handed over to the applicable collection point for the recycling of electrical and electronic equipment.



**CE** This product complies with the European Union Council Directives and Standards relating to electromagnetic compatibility (EMC Directive 89/336/EEC) and the Low Voltage Directive (73/23/EEC).

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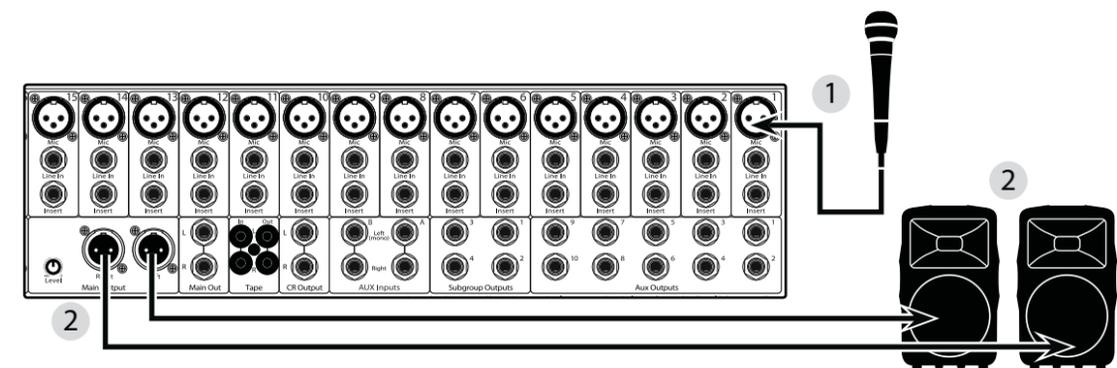
## 1.0 Quick Start

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 the StudioLive to adjust the level up or down until you have achieved the optimal volume.
- Do not allow your inputs to clip. Watch the level meters; when the LEDs near the Clip mark, the top LED will illuminate, indicating that the analog-to-digital converters are in danger of being overdriven. Overdriving the converters will cause digital distortion, which sounds terrible. The XMAX™ preamps in your StudioLive provide plenty of headroom; take advantage of it.
- Your P.A. and studio equipment should be powered on in the following order:
  - A. Sound sources (keyboards, direct boxes, microphones, etc.) connected to the StudioLive inputs
  - B. StudioLive 24.4.2
  - C. Computer (if applicable)
  - D. 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!

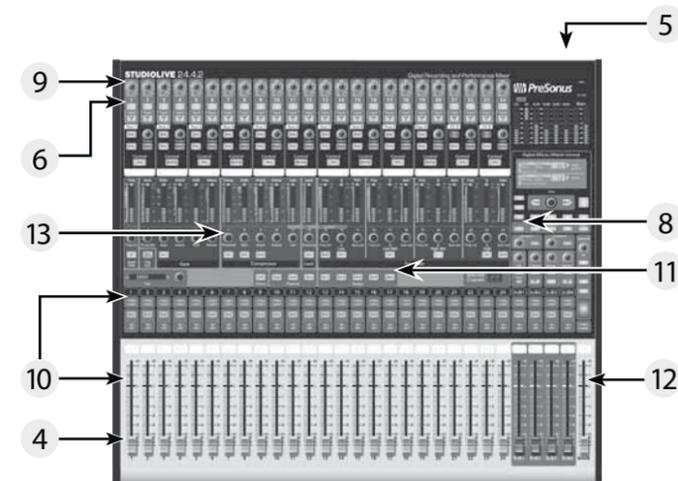


1. Grab a microphone and a mic cable and plug them into the StudioLive's Channel 1 mic input.
2. Connect the Main Outs (TRS or XLR) of your Studio Live to your power amplifier or powered monitors.

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 to the ∞ setting.  
Make sure that the Trim knob on Channel 1 is all the way counter-clockwise.
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.
8. Press the Input button in the Meter section.
9. Speak or sing into your microphone at about the same volume as during the upcoming performance.  
  
Turn the trim knob on Channel 1 clockwise while watching the first meter in the Fat Channel. Adjust the Channel 1 trim knob until a little more than half of the green LEDs are lighting up. The red LED at the top of the meter should **never** light up!
10. Press the Select button on Channel 1 and move the Channel 1 fader up until it reaches "U" (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 Section 8.7 on page 144.*



## 2.0 Overview

### 2.1 Introduction



Thank you for purchasing the PreSonus StudioLive™ 24.4.2. PreSonus Audio Electronics has designed the StudioLive utilizing high-grade components to ensure optimum performance that will last a lifetime. Loaded with 24 high-headroom, XMAX™ microphone preamplifiers; a built-in 32x26 FireWire recording and playback engine; Fat Channel processing with 4-band EQs, compressors, limiters, and expander/gates; DSP effects; 10 aux buses; 4 subgroups; extensive LED metering; mixer save/recall; channel-strip save/recall/copy/paste; talkback; and more, StudioLive breaks new boundaries for music performance and production. All you need is a compatible computer with a FireWire connection, a few microphones and cables, speakers, and your instruments, and you are ready to record in the studio or in front of a live audience!

We encourage you to contact us at 1+225-216-7887 (in the U.S. Central Time Zone) with any questions or comments you may have regarding the 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 a computer. This will help you avoid problems during installation and setup.

- Quick Start: Level Setting
- Overview
- Hookup
- Controls
- Scenes, Presets & System Menu
- Connecting to a Computer
- Software Universal Control, Capture, and Studio One Artist
- Tutorials
- Technical Information
- Troubleshooting & Warranty

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

The StudioLive™ 24.4.2 performance and recording digital mixer is a fully loaded professional digital mixer, combined with a complete 32x26 FireWire recording system. Racks of processing effects—including compressor, limiter, gate, four-band parametric EQ, reverb and delay—are available on every input channel, subgroup, aux, and main mix, delivering total control in a compact, rugged steel chassis. The StudioLive 24.4.2 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.

## 2.2 Summary of StudioLive 24.4.2 hardware features

- 24-bit/48 kHz sampling rate
- 24 Class A XMAX microphone preamplifiers
- 24 line-level inputs
- 10 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
- 32x26 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 fully 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
- Rugged steel chassis
- PreSonus Capture™ multitrack-recording software
- Compatible with Cubase, Digital Performer, Logic, Nuendo, Sonar, Studio One, and others
- PC and Mac® compatible

## 2.3 Summary of Capture™ software features

Included with your StudioLive is Capture, a digital-audio multitracking application designed to make recording quick and easy to set up and operate. Perfect for live recording and for mixing your audio in real time to a stereo audio file, Capture software was designed to interface perfectly with StudioLive 24.4.2, allowing instant setup and recording of performances.

In addition to recording the 24 channels of the StudioLive 24.4.2, Capture also allows you to record a single stereo track from the StudioLive's main output, one pair of subgroup outputs, or a pair of aux sends, giving you the option to record your main mix or create a separate recording mix. You can arm all 26 tracks to record with the click of a single button. Please consult the Capture user manual for complete instructions.

- 26x26 multitrack recording application
- Record with two mouse clicks
- Essential editing suite (copy, cut, paste, splice, resize)
- Peak LED-style meter bridge with clip indicators
- Marker placement and recall
- Export between marker
- Record stereo mix from StudioLive mixer
- Import/export .wav, .aiff, or OpenTL files

## 2.4 Summary of StudioOne™ Artist software features

All PreSonus audio interfaces include PreSonus Studio One Artist recording software, which comes with over 4 GB of plug-ins, loops, and samples, giving you everything you need for music recording and production. The Studio One Artist *Quick Start Guide* is located in Section 4.1 of this manual. You will find a complete user manual on the Studio One Artist installation DVD.

- Unlimited track count, inserts, and sends
- 20 high-quality native plug-ins: amp modeling (Ampire), delay (Analog Delay, Beat Delay), distortions (Redlight Dist), dynamics processing (Channel Strip, Compressor, Gate, Expander, Limiter, Tricomp), equalizer (Channel Strip, Pro EQ), modulation (Autofilter, Chorus, Flange, Phaser, X-Trem), reverb (MixVerb, Room Reverb), and utility (Binaural Pan, Mixtool, Phase Meter, Spectrum Meter, Tuner)
- Over 4 GB of loops, samples, and instruments, featuring: Presence virtual sample player, Impact virtual drum machine, SampleOne virtual sampler, Mojito virtual analog-modeled subtractive synthesizer
- Innovative and intuitive MIDI mapping
- Powerful drag-and-drop functionality for faster workflow
- Mac OS X® and Windows® compatible

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## 2.5 Summary of Virtual StudioLive software features

The Virtual StudioLive application is completely integrated with your StudioLive 24.4.2. VSL is a highly advanced editor/librarian and control panel. Because of the continuous bidirectional communication between your StudioLive and VSL, whatever you do on the StudioLive's control surface will be reflected in VSL and vice versa.

- Easy drag-and-drop workflow
- Drag presets directly to channels
- Drag parts of presets directly to components in the Fat Channel
- Adjust the Fat Channel gate, compressor, and EQ plus the graphic EQ and effects
- Quickly drop entire Scenes to the mixer for instant recall of all channel, effects, and graphic EQ settings
- Load effects quickly by simply dragging presets into the GUI
- Makes StudioLive as easy to use as Studio One
- Use the mouse to quickly assign channels to multiple buses, mute, solo, etc.
- Timestamped backups of the entire board
- Preset Management
  - Store and organize presets on your computer, then transfer them to your mixer for that night's gig
  - Create presets on the road, store them to the mixer, then drag them to your preset pool the next time you're connected to the computer
  - Reorder presets in mixer memory; easily order presets to best fit your workflow for the gig
  - Share presets with friends; drag presets out of the browser and email, IM, or disk-swap presets with other StudioLive owners
- Mixer Overview
  - See all of the most-used parameters on the mixer at once
  - See the state of all Fat Channel settings at once
  - See all aux mixes at once
  - See the current effects and parameter settings
  - See graphic equalizer settings

## 2.6 What is in the box

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

- PreSonus StudioLive 24.4.2 digital recording and performance mixer



- 6' (1.8 m) 6-pin-to-6-pin FireWire 400 cable
- 6' (1.8 m) 6-pin-to-9-pin FireWire 400-to-800 cable



- IEC power cord



- StudioLive Software Library containing
  - PreSonus StudioLive Artist program DVD plus gigabytes of third-party content
  - PreSonus Capture™ CD with demo Sessions
  - PreSonus Virtual StudioLive and Universal Control CD



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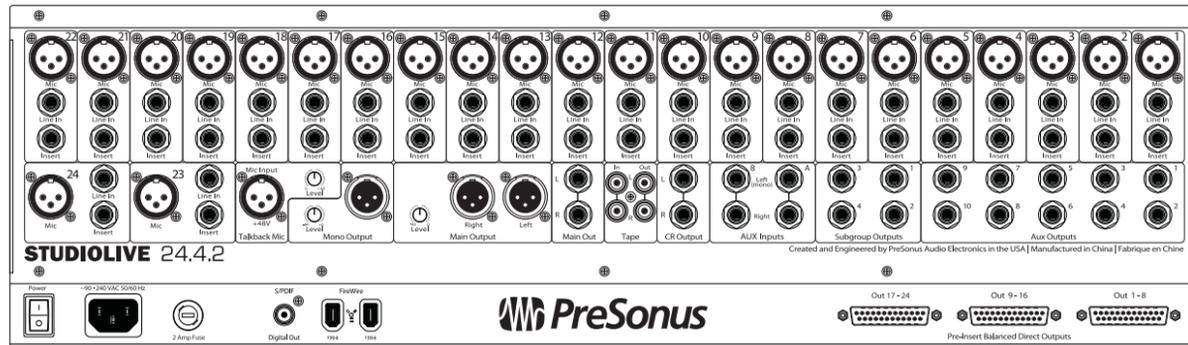
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## 3.0 Hookup

### 3.1 Rear-Panel Connections



**Microphone Inputs.** Your StudioLive is equipped with 24 PreSonus XMAX microphone preamplifiers for use with all types of microphones. (A 25th XMAX preamp serves the talkback section.) The 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.

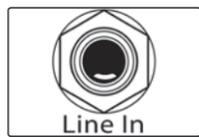


**48-volt Phantom Power.** The StudioLive provides 48V phantom power for the microphone input on each channel. This feature can be individually enabled for each channel using buttons on the top panel.

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

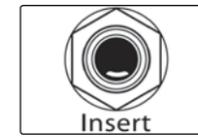
XLR connector wiring for phantom power:

- Pin 1 = GND
- Pin 2 = +48V
- Pin 3 = +48V

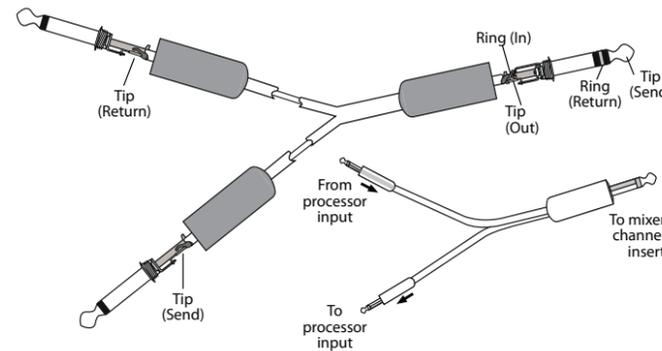


**Line-level Input.** Each channel of the StudioLive has a 1/4-inch, balanced TRS connection for line-level input. When these inputs are engaged, the microphone-preamp circuit is bypassed. Typical examples of line-level connections are synthesizer outputs, CD/DVD-player outputs, and (with exceptions) signal-processor inputs and outputs.

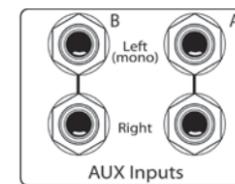
**WARNING:** Please note: As with any mixer, plugging in a microphone or a line-level input device, or turning phantom power on or off, will create a momentary spike in the audio output of your StudioLive. Because of this, it is highly recommended that you mute or turn down the channel trim before changing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



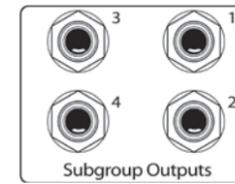
**Insert.** Each channel of the StudioLive has a direct-insert point. These unbalanced, 1/4-inch connectors can be used to connect external processors (such as compressors, EQs, de-essers, and filters) to your StudioLive's preamps and line inputs. The insert's send is after the channel's gain control but before the digital bus. The return goes straight to the digital bus. So if you insert a de-esser on your vocalist's channel, you will be sending an unprocessed, amplified signal to the de-esser; the processed signal returned to the StudioLive will then be routed to the digital bus, where it can be sent through the Fat Channel, Aux and FX buses, etc.



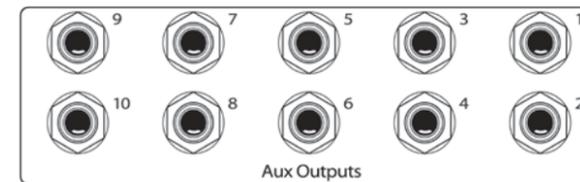
- Insert connector wiring:
- Tip = send (output to inserted device)
- Ring = return (input from inserted device)
- Sleeve = common ground



**Aux Inputs.** The StudioLive offers two auxiliary inputs, which are generally used as effects returns. In Section 8.5.2, 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 normaled to the right input, so if you are returning a mono signal to the mix, connect it to the left input, and the signal will be routed to both sides of the mix.



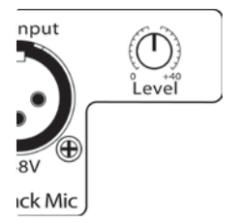
**Subgroup Outputs.** These are balanced mono outputs for each subgroup.



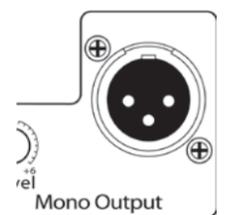
**Aux Outputs.** The StudioLive is equipped with six auxiliary outputs. In Section 8.5, we discuss in detail how to create aux mixes for monitoring and effects processing. Aux mixes are routed to these outputs.



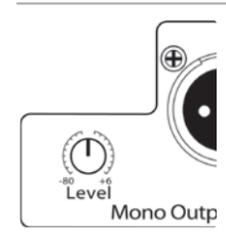
**Talkback Mic Input.** The StudioLive does not have an onboard talkback mic; an external mic must be used. Phantom power is always enabled on this microphone preamp, so either a dynamic or a condenser microphone can be used. This is the same high-quality XMAX preamp that is featured on Channels 1 through 24, and the talkback can be used as an extra input when using the StudioLive as an audio interface. See Section 6.4.2 for details.



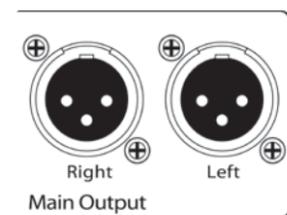
**Talkback Mic Level.** This is the trim control for your talkback microphone. It adjusts the gain of the talkback input.



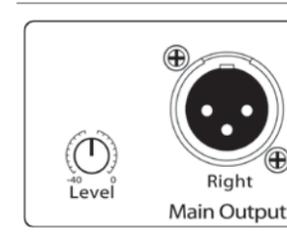
**Mono Output.** This balanced output carries a mono, summed version of the stereo signal from the main bus.



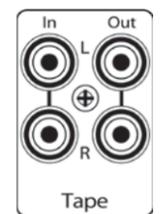
**Mono Output Level.** This knob controls the maximum level of the Mono Output signal. The signal can be attenuated to -80 dB and boosted up to +6 dB.



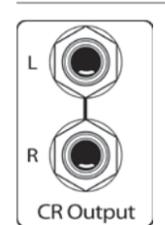
**Main Output.** The StudioLive features both XLR and TRS main outputs. These outputs are parallel to each other and to the mono output.



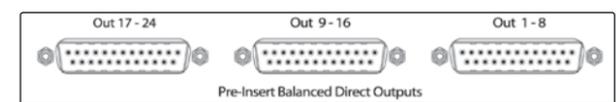
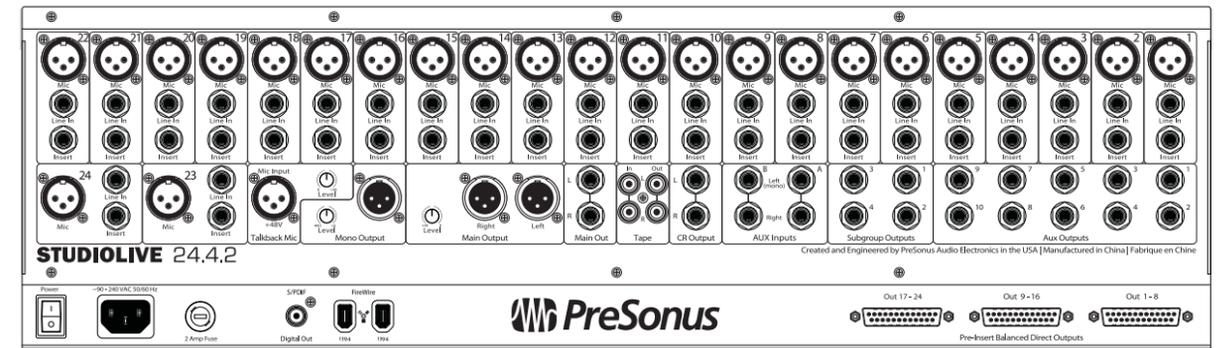
**Main Output Level.** This knob controls the maximum output level of the XLR and TRS main outputs. The signal can be attenuated to -40 dB and boosted up to 0 dB.



**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.



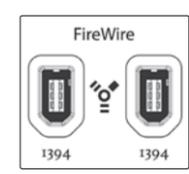
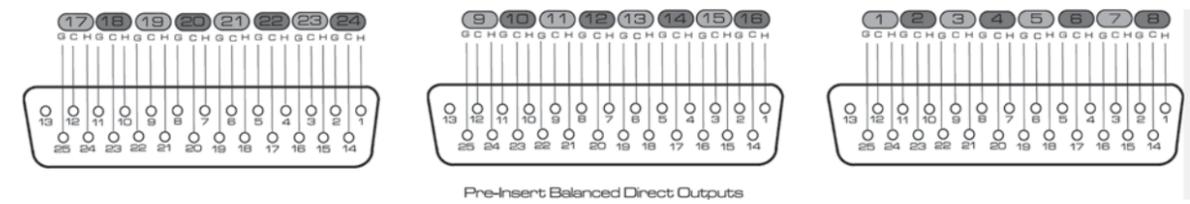
**CR Output.** These are the balanced control-room outputs. The level is controlled by the Monitor knob in the Monitor section on the top panel.



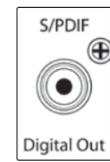
**Pre-Insert Balanced Direct Outputs.** These are the balanced, direct analog outputs for the 24 channels. The DB25 connectors divide the channels into three groups of eight. Balanced DB25 fan-out snakes can be obtained in various configurations at most recording and live-sound retailers.

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 cannot be patched to the direct outputs. (See Section 6.4.2 for more information.)

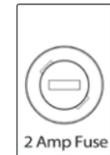
- DB25 pin-outs
- H = Hot
  - C = Cold
  - G = Ground



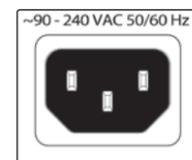
**FireWire Ports.** There are two standard 6-pin FireWire 400 ports on the back of the StudioLive. Either port can be used to connect your StudioLive to a FireWire port on your computer. If your computer has a 4-pin connector (commonly found on laptops), you will need to purchase a 4-to-6-pin adaptor 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 an additional StudioLive mixer. You can connect two StudioLive 24.4.2 mixers to create a standalone, large-format mixing console.



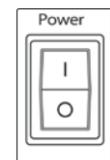
**S/PDIF Output.** By default, the S/PDIF output receives the same signal as the main outputs, so no activation is necessary. However, any buses that can be routed to the auxiliary FireWire returns can be routed to the S/PDIF output, either through the System menu in the Digital Effects | Master Control section, or in the StudioLive Control Panel. (See Sections 5.4 and 7.2.5 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.



**2 Amp Fuse.** This is the StudioLive's fuse housing. Your StudioLive uses a 5 mm x 20 mm, 250 VAC, fast-acting fuse.

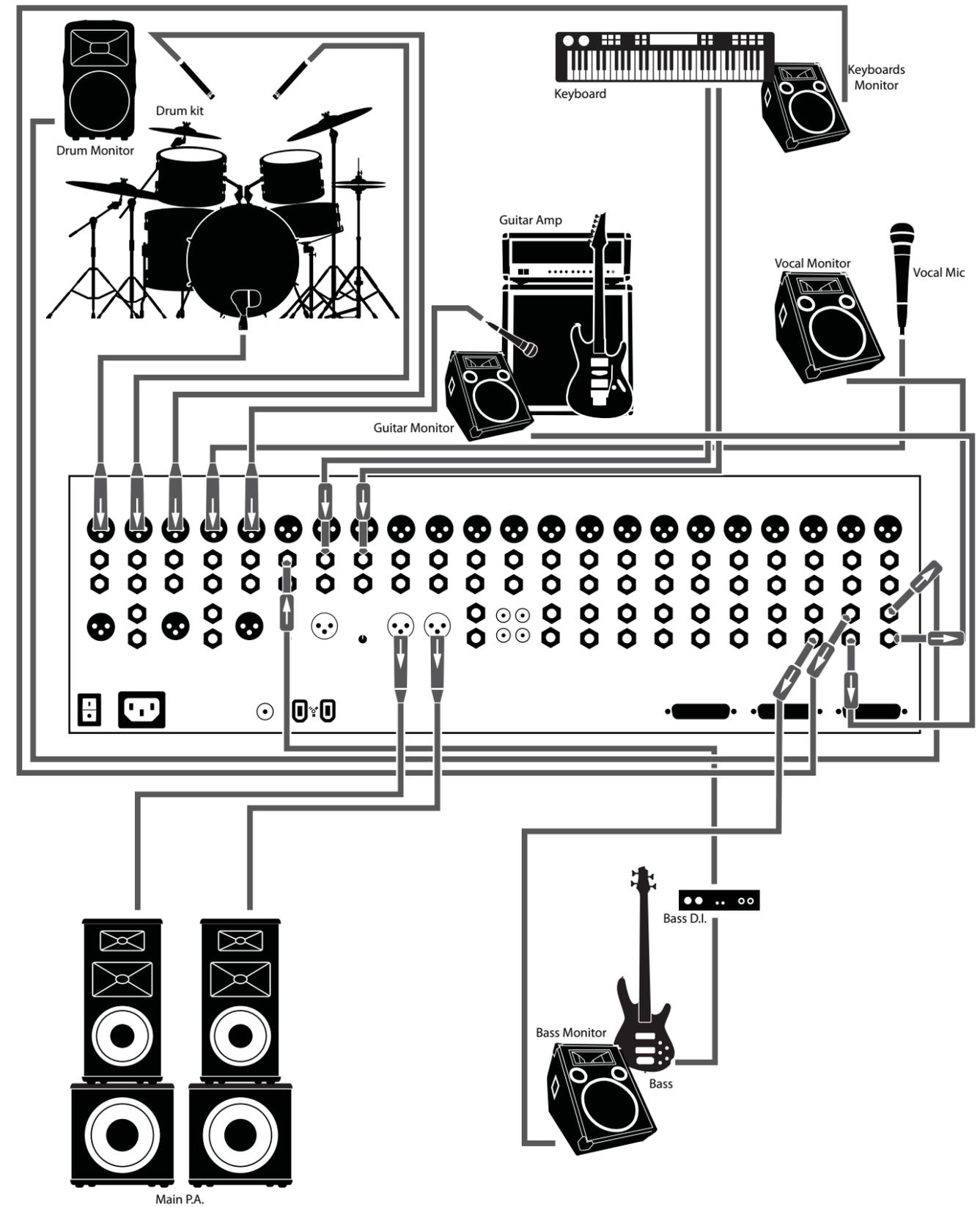


**Power-adaptor Input.** This is where you plug in the provided IEC power cable.

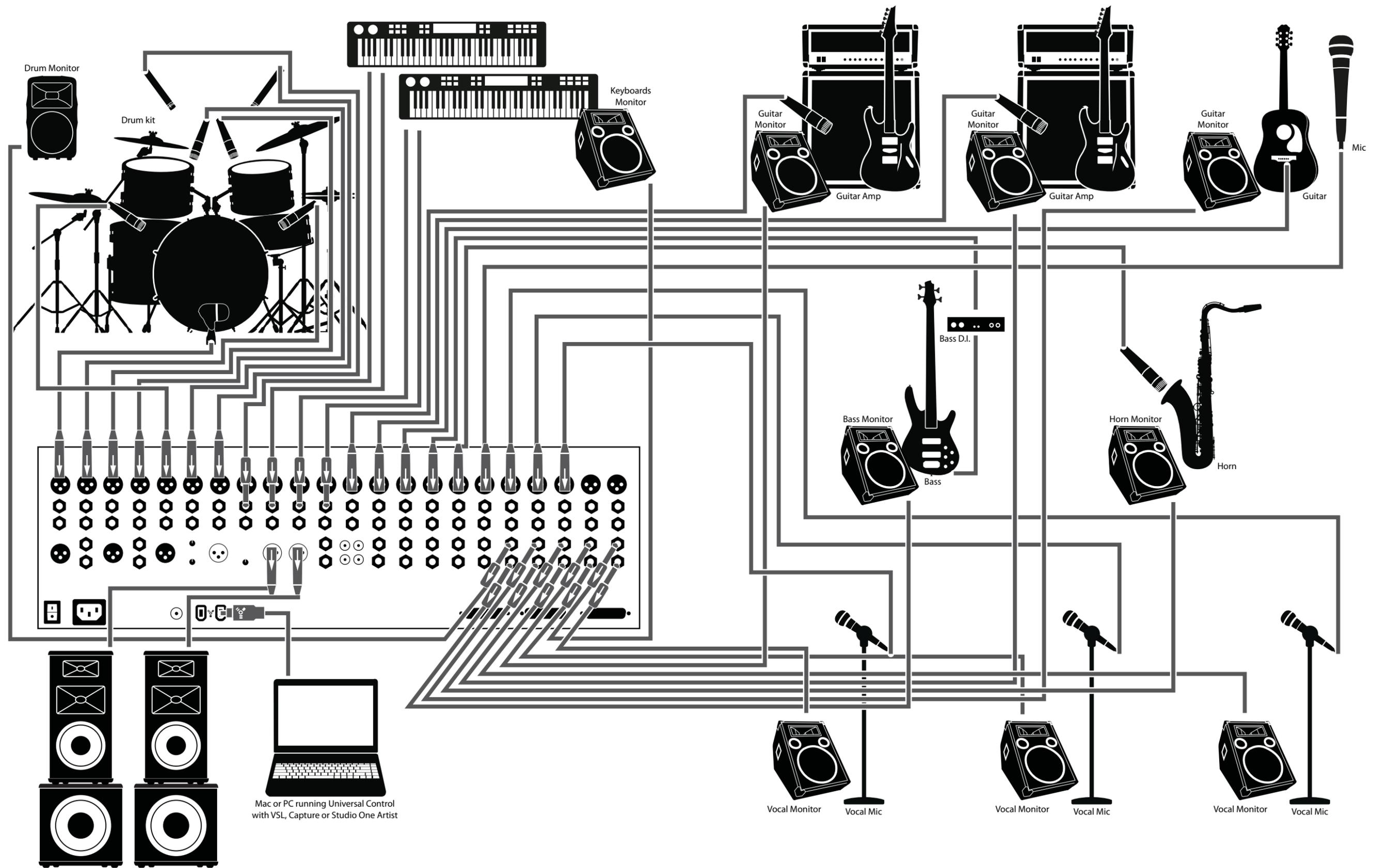


**Power Switch.** Push the top part of the switch ( | ) to turn on your StudioLive. Push the bottom part of the switch ( O ) to turn it off.

### 3.2 Basic Hookup Diagram



3.3 Typical Band Hookup

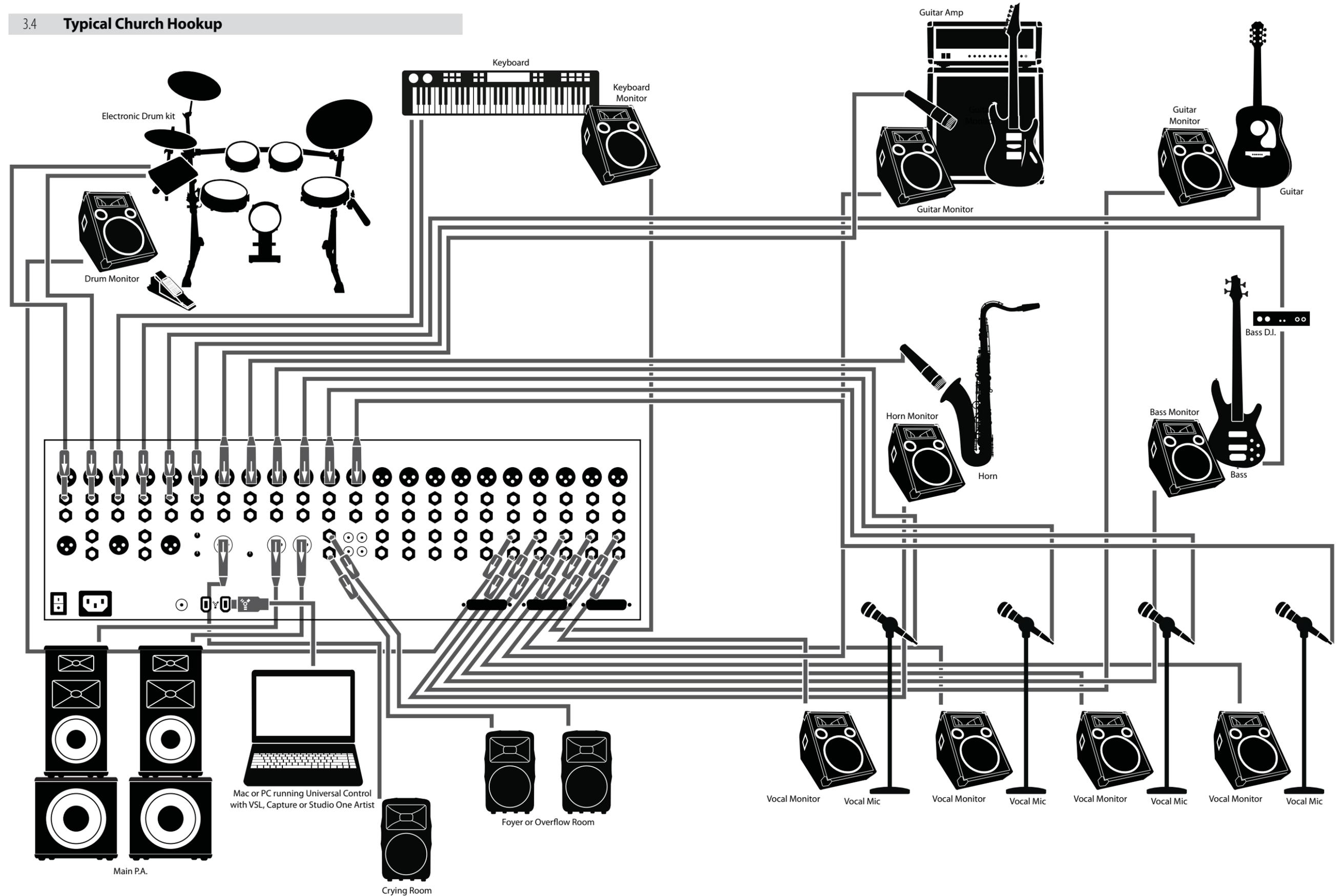


Mac or PC running Universal Control with VSL, Capture or Studio One Artist

Main P.A.

- Quick Start Level Setting
- Overview
- Hookup**
- Controls
- Scenes, Presets & System Menu
- Connecting to a Computer
- Software Universal Control, Capture, and Studio One Artist
- Tutorials
- Technical Information
- Troubleshooting & Warranty

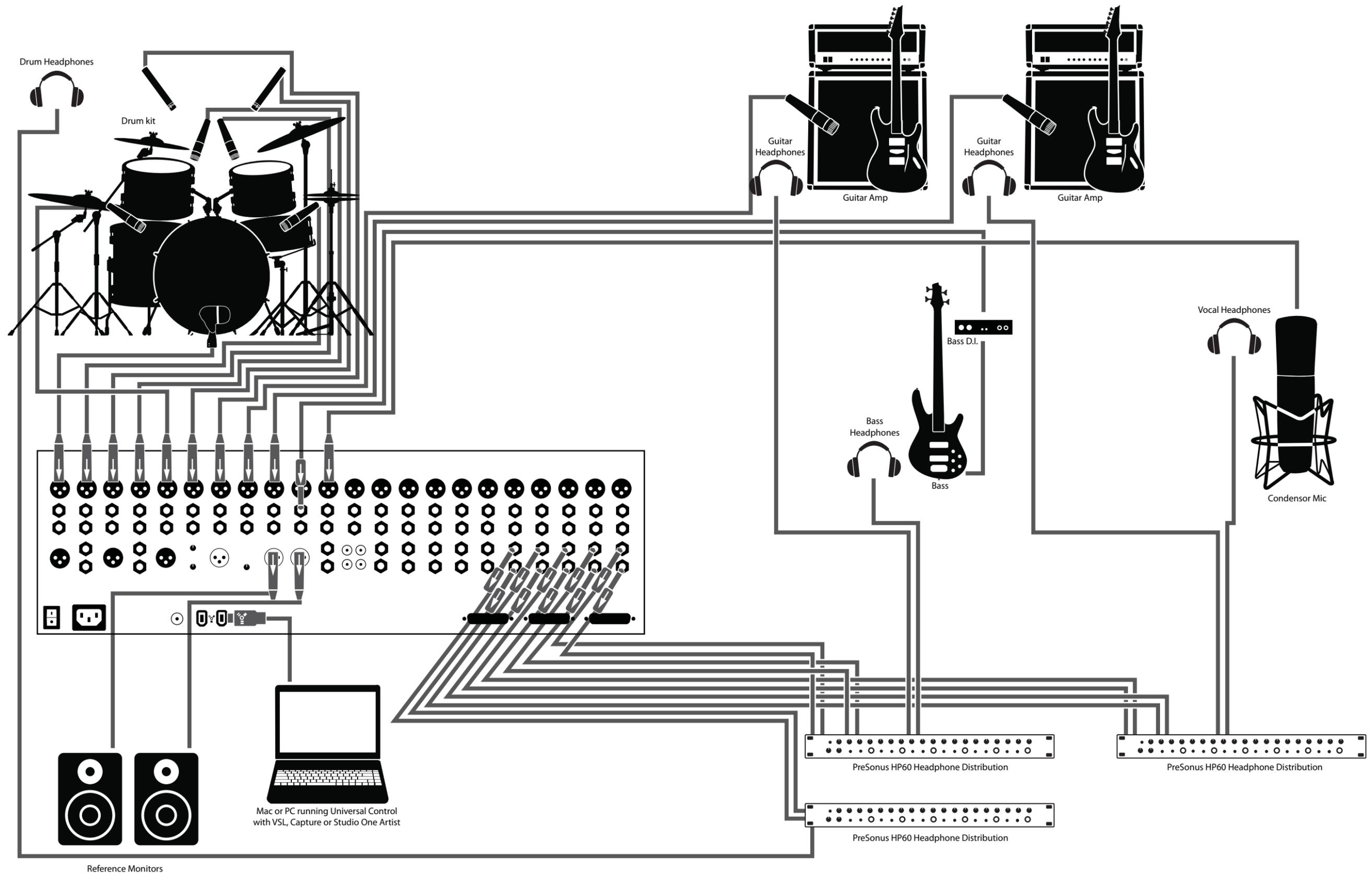
- Quick Start Level Setting
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### 3.5 Typical Recording Hookup

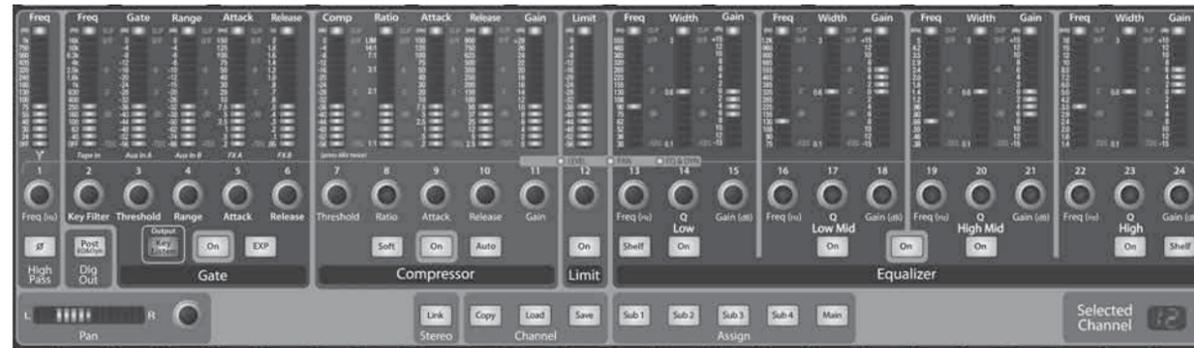


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- Controls: Level Setting
- Scenes, Presets & System Menu
- Connecting to a Computer
- Software: Universal Control, Capture, and Studio One Artist
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- Hookup**
- Controls: Level Setting
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## 4.0 Controls

### 4.1 The Fat Channel



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 24 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 10 analog aux sends and both internal effects buses
- Assign subgroup and main routing
- Meter inputs, post-dynamics-processing outputs, and gain reduction for all 24 channels
- Meter aux-send outputs
- Copy, save, and load mix scenes
- Recall your fader position for stored mixes

#### 4.1.1 Select Buttons, Meters and the Fat Channel



**Select Buttons.** All around the StudioLive, you will see Select buttons. There is a Select button on each of the 24 inputs, each of the 10 analog aux sends, both of the internal effects buses, each of the 4 subgroups, the 2 auxiliary inputs, and the Main output bus. Each of these buttons serves exactly the same purpose: to access the Fat Channel parameters for its channel or bus.



**Selected Channel Display.** In the lower right corner of the Fat Channel, you will find an LED readout. The currently selected channel will always be displayed here. (Numbers 1-24 indicate one of the 24 input channels is selected, S1-S4 indicates Subgroups 1-4, MA indicates the Main bus, A1-A10 indicates Aux 1-10, A11 and A12 indicate EFX A and EFX B, and F1 and F2 indicate Aux inputs A and B.)



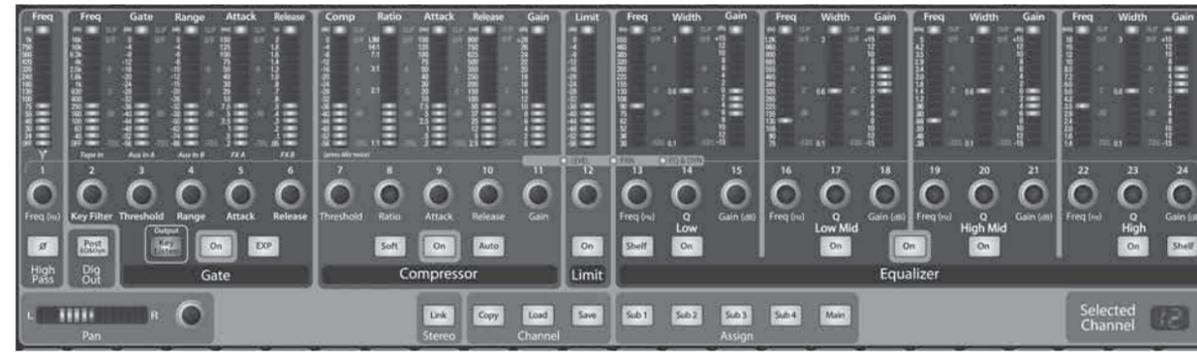
**Selected Channel Meters.** In addition, two meters—part of a set of seven meters located in the top right section of the mixer—are dedicated to displaying information about the currently selected channel. The meter on the far left of this section displays the pre-fader input level for the selected 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 24 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 high-pass filter is only available on the inputs, 10 auxes, and 2 internal FX buses, and the phase reverse is only available on the 24 inputs. In addition, other inputs and buses without Select buttons are available to route to the auxiliary FireWire returns. (See Section 6.4.2 for more details.)

#### 4.1.2 What You Can Process with the Fat Channel and FireWire Sends.

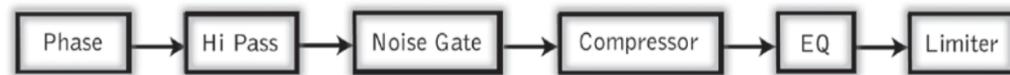
The following table provides a quick guide to the processing that is available for each bus in the StudioLive:

Bus	Phase Reverse	High-Pass Filter	Noise Gate	Compressor	EQ	Limiter	FireWire Send
Inputs (Ch 1-24)	✓	✓	✓	✓	✓	✓	✓
Subgroups			✓	✓	✓	✓	✓
Main Out L/R			✓	✓	✓	✓	✓
Aux Sends 1-10		✓	✓	✓	✓	✓	✓
Internal FX Sends A & B		✓	✓	✓	✓	✓	✓
External FX Returns A & B		✓	✓	✓	✓	✓	✓
Tape Input							✓
Talkback Mic							✓
Solo Bus							✓
Monitor Bus							✓



### 4.1.3 Fat Channel: Dynamics Processing and EQ

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



#### Phase Reverse Button

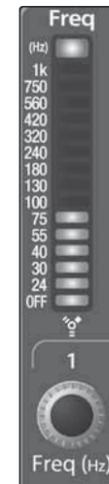


Reverses the Phase of the Selected Channel.

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/reinforcing each other.

*Phase reverse is only available on the 24 channels of the input bus.*

#### High Pass Filter



Adjusts 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 8.3.1 for more details.

The slope of the High Pass Filter is -6dB/octave.

*The high-pass filter is available on the 16 channels of the input bus, the 6 Auxes, both internal FX buses, and both FX returns.*

### Fat Channel Gate Section

#### 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.*

#### Gate Key Listen Button



Enables Key Listen in the Solo Bus.

This button engages and disengages the Key Listen function in the solo bus. It will illuminate to indicate that the Key Listen is active. When Key Listen is enabled, and the Selected channel is soloed, you will be able to monitor what is being removed by the gate, using the Control Room outputs.

#### Gate Expander Button



Turns the Noise Gate into an Expander.

The StudioLive allows you to choose between an expander and a noise gate for each channel or output. By default, the Expander button will be enabled.

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

#### Gate Key Filter

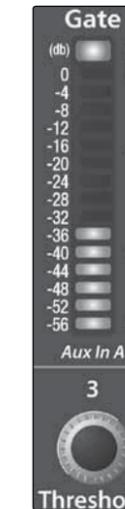


Sets and Displays the Frequency at Which the Gate Will Open.

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

*A properly set Key Filter on a gate can greatly improve the overall sound quality of a mix. For example, if you are miking a bass amp and are picking up a bit of the kick drum from that mic, setting the gate to open at the frequency of the kick drum, combined with a fast attack and release, can tighten the rhythm section. Increase the release time to loosen the feel.*

#### 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 -56 dB.

**Gate Range** Sets and Displays the Range of the Gate.



This encoder sets, and the meter displays, the amount of gain reduction that the gate will produce. The range goes from 0 to -86 dB.

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



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

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

**Gate Attack** Sets and Displays the Gate Attack Setting for the Selected Channel.



This encoder sets, and the meter displays, the rate at which the gate opens on the selected channel or output. You can set the attack time from 0.2 to 150 ms.

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

**Fat Channel Compressor Section**

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



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

The compressor is available for all input and output buses.

**Soft Knee / Toggle Button** Engages Soft-Knee Compression.



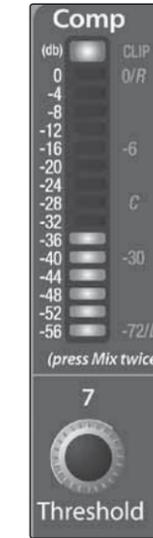
In normal operating mode, the compressor is set for hard-knee compression, meaning that the gain reduction applied to the signal occurs as soon as the signal exceeds the level set by the threshold. When the Soft Knee button is engaged, the onset of gain reduction occurs gradually after the signal has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

**Auto Mode Button** Enables Automatic Attack and Release Mode.



When Auto mode is active, the Attack and Release controls become inoperative, and a preprogrammed attack and release curve is used. In this mode, the attack is set to 10 ms, and the release is set to 150 ms. All other compressor parameters can still be adjusted manually.

**Compressor Threshold** Sets and Displays the Threshold of the Compressor for the Selected Channel or Output Bus.

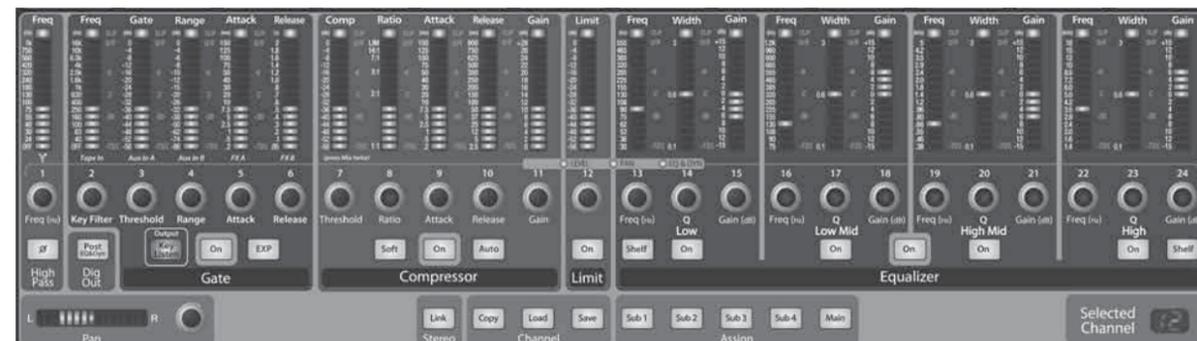


This encoder sets, and the meter displays, the compressor threshold for the selected channel or output bus. When the signal's amplitude (level) exceeds the threshold setting, the compressor engages. Turning the knob counterclockwise lowers the threshold so that compression begins at a lower amplitude.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

**Compression Ratio** Sets and Displays the Compression Ratio for the Selected Input Channel or Output Bus.

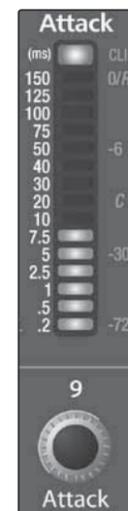


This encoder sets, and the meter displays, the compression ratio (or slope) for the selected channel or output bus. The ratio sets the compression slope, which is a function of the output level versus the input level. For example, if you have the ratio set to 2:1, any signal levels above the threshold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



**Compressor Attack**

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



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

**Compressor Release**

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



This encoder sets, and the meter displays, the release setting of the compressor for the selected channel or output bus. Release sets the length of time the compressor takes to return the gain reduction back to zero (no gain reduction) after crossing below the compression threshold. Release can be set from 40 to 1,000 milliseconds.

*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.

**Fat Channel Equalization Section**

**Equalizer On/Off Button**



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

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

The equalizer is available for all input and output buses.

**Low EQ On/Off Button**

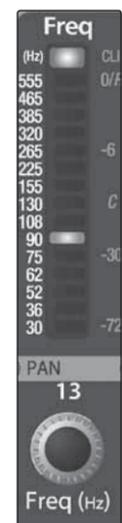


Activates Control for the Low Band EQ for the Selected Input or Output Bus.

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 EQ band is available for all input and output buses.

**Low EQ Frequency Control**



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

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

You can adjust the center frequency from 30 to 555 Hz.

**Low EQ Q Control**



Sets and Displays the Q of the Low Frequency Band.

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

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

**Low EQ Gain Control**



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

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

**Low Shelf EQ Button**

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

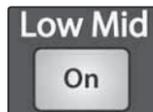


When the Shelf button is not engaged, the Low band is parametric. Enabling the Shelf button turns the Low band into a low-shelving EQ that alters, by a fixed amount, a band of low frequencies at and below a user-selected shelving frequency.

A Low Shelving EQ is like a bass-control knob on a stereo. In this mode, the Center Frequency control selects the shelving frequency.

**Low-Mid EQ On/Off Button**

Activates Controls for the Low-Mid 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 EQ is available for all input and output buses.

**Low-Mid EQ Frequency Control**

Sets and Displays the Center Frequency of the Low-Mid EQ.



This encoder sets, and the meter displays, the center frequency for the Low-Mid band. You can adjust the center frequency from 75 Hz to 1.2 kHz.

**Low-Mid EQ Q Control**

Sets and Displays the Q of the Low-Mid Frequency Band.



This encoder sets, and the meter displays, the Q for the Low-Mid band.

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

**Low-Mid EQ Gain Control**

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



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

**High-Mid EQ On/Off Button**

Activates Controls for the High-Mid 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 EQ is available for all input and output buses.

**High-Mid EQ Frequency Control**

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



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

**High-Mid EQ Q Control**

Sets and Displays the Q of the High Mid Frequency Band.



This encoder sets, and the meter displays, the Q for the High-Mid band. The Q is the ratio of the center frequency to the bandwidth. When the center frequency is constant, the bandwidth is inversely proportional to the Q, so as you raise the Q, you narrow the bandwidth.

**High-Mid EQ Gain Control**

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



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

**High EQ On/Off Button**

Activates Control for the High 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 EQ band is available for all input and output buses.

**High EQ Frequency Control**

Sets and Displays the Center Frequency of the High EQ.



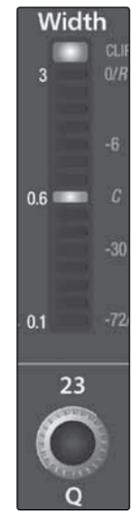
This encoder sets, and the meter displays, the center frequency of the High band. You can adjust the center frequency from 1.4 to 18 kHz.

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**High EQ Q Control**

Sets and Displays the Q of the High Frequency Band.



This encoder sets, and the meter displays, the Q for the High band. The Q is the ratio of the center frequency to the bandwidth. When the center frequency is constant, the bandwidth is inversely proportional to the Q, so as you raise the Q, you narrow the bandwidth.

**High Shelving EQ Button**

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



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

**High EQ Gain Control**

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



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

**Note:** See Also Graphic Equalizers, Section 5.3



**Fat Channel Limiter Section**

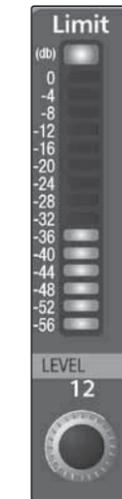
**Limiter On/Off**



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

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

**Limiter Threshold Control**



Sets and Displays the Threshold of the Limiter for the Selected Channel or Output Bus.

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

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

- |                    |                    |
|--------------------|--------------------|
| Channels 1 and 2   | Channels 21 and 22 |
| Channels 3 and 4   | Channels 23 and 24 |
| Channels 5 and 6   | Aux 1 and Aux 2    |
| Channels 7 and 8   | Aux 3 and Aux 4    |
| Channels 9 and 10  | Aux 5 and Aux 6    |
| Channels 11 and 12 | Aux 7 and Aux 8    |
| Channels 13 and 14 | Aux 9 and Aux 10   |
| Channels 15 and 16 | Subgroups 1 and 2  |
| Channels 17 and 18 | Subgroups 3 and 4  |
| Channels 19 and 20 |                    |

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

Note that this is a nondestructive paste; when the Link button is disengaged, the other channel's previous settings will be restored. 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.

### 4.1.5 Fat Channel: 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 ten 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 to the main outputs. This includes the 24 main inputs and the 2 auxiliary inputs. The internal effects returns can also be assigned to any or all of the subgroups and to 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 available when one of the 24 channel inputs is selected. The subgroups, main output, aux bus, aux inputs, 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 6.

### 4.1.6 Copying, Loading, and Storing Fat Channel Settings

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.



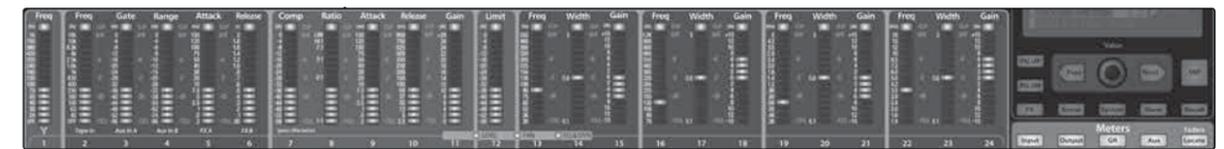
To paste the current channel's Fat Channel setting to another channel or bus, simply press that channel's Select button. It will stop flashing and illuminate. After you have selected every channel to which you want the settings pasted, press the Load button. The StudioLive will return to its normal state, 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 5.2.



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 5.2.

## 4.2 Metering



The StudioLive offers flexible metering at the touch of a button. The 24 meters in the Fat Channel section can monitor:

- All 24 inputs, post-gain and pre-dynamics, pre-EQ, and pre-fader
- All 24 inputs, post-dynamics, post-EQ, and post-fader
- The gain reduction for all 24 inputs
- The output volume of each of the 10 aux sends
- The output volume of both of the internal effects buses

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

### 4.2.1 StudioLive Metering Controls

The Meters section of the StudioLive is located below the System section. Each of these buttons are toggle switches; you turn them on and off by pressing them. The meter state can also be changed by pressing another button in the Meter section, or 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 the 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, but the changes will not be displayed 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



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

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

#### Output Metering Button



**Turns Post-Fader Output Metering On 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 Metering Button.** Turns Gain Reduction Metering On and Off.



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

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



Displays the output level of each of the aux sends.

Meters 1, 3, 5, 7, 9, 11, 13, 15, 17, and 19 display the output levels of Aux Sends 1-10, respectively.

Meters 21 and 23 display the output level of EFX A and EFX B.

**Fader Locate Button** 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.

### 4.3 Input Channel Strip

The StudioLive is equipped with all of the standard input controls of an analog mixer. In addition, the StudioLive provides the added flexibility of routing a playback stream from your audio-recording software to the mixer via the 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.

#### 4.3.1 Input Channel Controls

**Trim Control** Adjusts the Input Gain Level.



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

**It is very important to properly adjust this control in order to minimize noise and avoid overload distortion. Follow the Quick Start level-setting instructions on page 5 before operating a channel.**

**Phantom Power Button** Turns Phantom Power On/Off.



The StudioLive is equipped with individual phantom power for every microphone input. The 48V button will illuminate when phantom power is activated. For more information on phantom power and the microphones that require it, please review Section 8.1.



**FireWire Input Select Switch** Turns FireWire Playback Streaming On/Off.



The FireWire Input button routes a playback stream from your audio software to the StudioLive's channel inputs, where it is routed and processed the same way as analog input signals. For example, if you want a particular recorded track to play back on mixer channel 3, simply route that track in your audio software to StudioLive Output 3. This button can also be used to insert a plug-in effect into the mix. For more information on this feature, please review Section 6.4.2.

*Do not engage this button when your StudioLive is not connected and synced to a computer, as it will mute the analog input globally.*

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



As previously described in Section 4.1.1, the Select button routes its channel through the Fat Channel, allowing you to add dynamics processing, EQ, and panning; assign the output routing; and more.

**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 8.8 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.

**4.4 Aux Sends**

The StudioLive has ten analog aux buses and two internal effects buses. The aux buses are mono; however, two aux buses can be linked to create a stereo bus. Section 8.5 discusses how to use these aux buses for monitor and effects mixing.



**4.4.1 Analog Aux Send Controls**

**Solo Button**

**Turns Soloing On and Off.**

This button will solo the aux bus to the main outputs or to the monitor outputs. Only PFL Soloing is available for the aux buses

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 bus 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, so that 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, which makes the send occur after all Fat Channel processing but before the fader. Please review Section 5.4 for details.

**Output Level Control**

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



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

**Aux Send Select Button**

**Enables Fat Channel Viewing.**



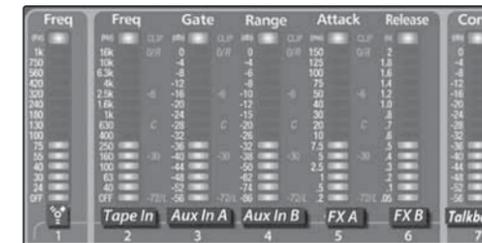
As previously described in section 4.1.1, the Select button routes its aux bus through the Fat Channel, allowing you to add dynamics processing and EQ

**Mix Button**

**Enables Aux Bus Mixing in the Fat Channel.**



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



Pressing the Mix button a second time will switch the first six meters and encoders to become the aux-send level controls for the Main FireWire Return, Analog Tape In, Aux Input A, Aux Input B, the output of FXA and FXB, and the Talkback mic. While in this mode, the Mix button will remain illuminated, and LEDs at the top, middle, and bottom of meters 7-24 will also illuminate.

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

**Mix|Pan Button**

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



On Aux Sends 2, 4, 6, 8, and 10 the Mix button becomes the Pan control when two auxes are linked. When this button is enabled, the 24 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.

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

For more information of stereo linking, please review Section 4.1.4.

## 4.4.2 Internal Aux 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-fader and pre-Fat Channel dynamics processing and EQ. Please review Section 5.4 for details.

### Output Level Control

**Adjusts the Master Level of the Effects Send.**



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

### Internal Effects Send Select Button

**Enables Fat Channel Viewing.**



As described in section 4.1.1, the Select button routes its Effects bus through the Fat Channel, allowing you to add dynamics processing and EQ and routing the aux send to outputs.

### Mix Button

**Enables Aux Bus Mixing and Metering in the Fat Channel.**

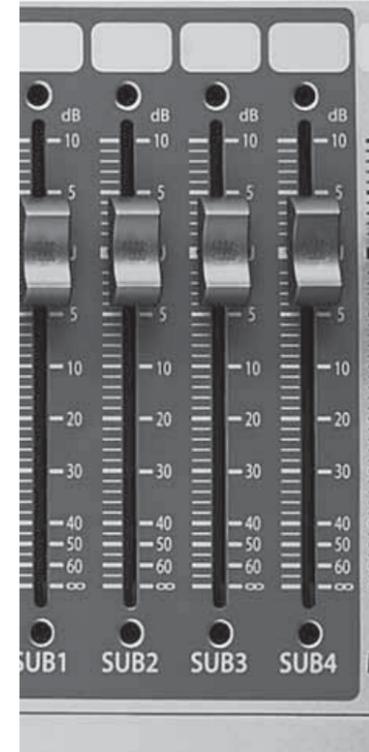


When this button is enabled, the 24 encoders in the Fat Channel become the fx-send level controls for each of their respective input channels. The meters will display the send level of each of the input channels.

## 4.5 Subgroups

As explained in Section 8.4, 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.

### 4.5.1 Subgroup Controls



#### Subgroup Channel Select Button



**Enables Fat Channel Adjustments of Subgroup**

As previously described in Section 4.1.1, the Select button routes its channel through the Fat Channel, allowing you to add dynamics processing, EQ, panning, etc.

#### Solo Button



**Turns Soloing On or Off for the Subgroup Bus.**

This button will solo it subgroup either to the monitor outputs post-fader. PFL and SIP are not available for the Subgroups.

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

#### Mute Button



**Turns Muting On or Off for the Subgroup Bus.**

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

#### Subgroup Fader



**Controls the Level of the Subgroup Bus.**

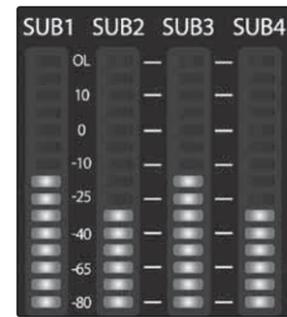
The fader controls the overall volume of the signal on its subgroup output.

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.

**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.

**4.6 Main Output Bus**

**Main Select Button**

Enables Fat Channel Viewing.



As previously described in Section 4.1.1, the Select button routes its channel through the Fat Channel, allowing you to add dynamics processing, EQ, panning, etc.

**Main Fader**

Controls the Level of the Main Output.



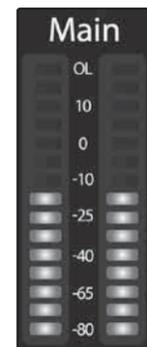
The fader controls the overall level of the main stereo output.

*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. Spit works pretty well, too.

**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.



**4.7 Master Section**



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

**4.7.1 Aux Inputs A and B**

The StudioLive features an internal effects processor that allows you to load two different effects at the same time. You also can patch processed return signals from an external effects processor using Aux Inputs A and B. Section 8.5 goes into the connections and procedures in greater detail.

**Aux Input Select Button**

Enables Fat Channel Viewing.



As previously described in section 4.1.1, the Select button routes its channel through the Fat Channel, allowing you to 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.

**4.7.2 Talkback System**

The StudioLive features a Talkback microphone input on the back panel. This can be routed to the aux outputs and to the mains. It is important to note that the aux outputs are grouped in this section. For example, if you are using Aux 3 as mono monitor send to the bass player's floor wedge and Auxes 5 and 6 as the stereo send 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**

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



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

**Output Selector Buttons**

**Assigns the Talkback Mic to the Aux and/or Main Outs.**



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

**Talkback Button**

**Turns the Talkback Mic On and Off.**



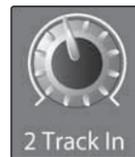
This latching button turns the talkback mic on and off. It will illuminate to indicate that the talkback mic is active.

**4.7.3 2 Track In**

The 2 Track input allows you to patch in an analog input or a playback stream from your audio application. This enables you to compare audio sources during mixdown or patch in intermission music between bands without using any of your 24 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 Main Outputs.**



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

**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 (Outputs 25 and 26) 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. For more information on using the main FireWire returns, please consult Sections 6.4.2, 6.4.4, and 6.4.5.

**4.7.4 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 8.8.

**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, regardless of whether this mode is 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.

**When SIP is engaged, channel mutes will only apply to the Subgroup and Main assignments. SIP does not mute input channels in Aux Send mixes.**

### 4.7.5 Monitor Bus

The StudioLive features a headphone output and 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 running sound at a show.

#### 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.)

#### 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.



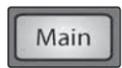
This knob adjusts the overall level of the control-room monitor outputs.

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



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

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



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

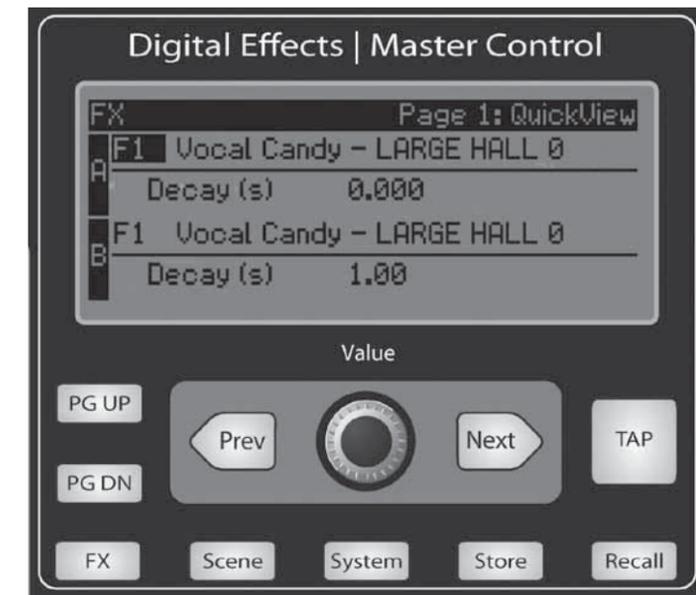
#### FireWire Monitor Button Assigns the Main Left/Right FireWire Return to the Monitor Bus.



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. The level for this input is controlled first by the level set from the computer application producing the audio (e.g., Studio One) and then by a knob in the 2 Track In section.

### 4.8 Digital Effects | Master Control

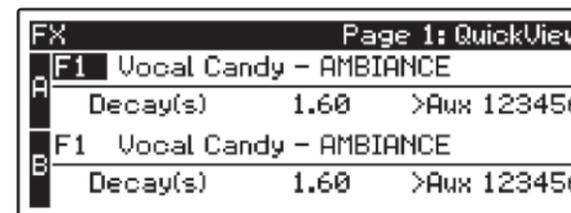
From the Digital Effects | Master Control section, you can select and change the parameters of the two internal effects processors, and you can store and recall every setting on your StudioLive. (See Section 5, "Scenes, Presets, and System Menu.") Because almost all of the StudioLive's features are controlled 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.



#### 4.8.1 The Digital FX (Effects) 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 8.4, 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 make adjustments to effect parameters, press the FX button in the Master Control section.

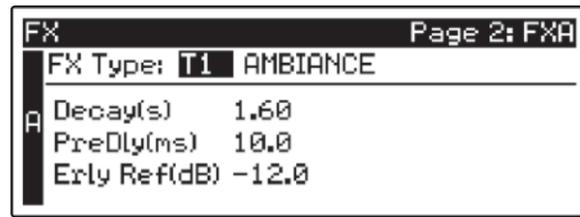


The first page of the FX menu is the QuickView screen. It displays both of the effects assigned to the internal effects buses, the main parameter for each, and to which Aux buses the effect is being routed. Effect A is assigned to EFX A bus, and Effect B is assigned to EFX B bus. Use the Next and Prev 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.

For example, if you would like to change the effect from 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 you choose a delay algorithm, you will notice two parameters: **Time (ms)** and **Time X**.

Time is the delay time, discussed in Section 8.6.2, which is the time between the source sound and the echo. This can 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 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 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.

Pages 4 and 5 allow you to store your changes for FX A and FX B, respectively, to the same location or to a new location, and to customize the name of your creation. 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 uppercase and lowercase letters and a selection of numerals and punctuation marks. 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 written to the StudioLive's internal memory. Once the preset is saved, the Store button will return to its unlit status.



## 4.8.2 Digital 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
F1	AMBIENCE	Natural
F2		Lively
F3	SMALL ROOM	Closet
F4		Studio A
F5		Studio B
F6		Bedroom
F7	BRIGHT ROOM	Kitchen
F8		Tile Floors
F9		Tile Bathroom
F10		Concrete Floors
F11	SMALL HALL	Radio Booth
F12		Small Club
F13		Big Club
F14	BRIGHT HALL	Brick House
F15		Linoleum Room
F16		Tile Room
F17	WARM HALL	Log Cabin
F18		Wood Floors
F19		Brick Club
F20		High Ceilings
F21	GATED HALL	GateVerb Short
F22		GateVerb Med
F23		GateVerb Long
F24		GateVerb Extreme

	TYPE	NAME
F25	LARGE HALL	Church
F26		Cathedral
F27		Gymnasium
F28		Arena
F29	PLATE	PlateVerb Shimmer
F30		PlateVerb Thick
F31		PlateVerb Drums
F32		PlateVerb Vox
F34	MONO DELAY	M: Short Slap
F35		M: Long Slap
F36		M: Triplet
F37		M: Triplet
F38	FILTER DELAY	Analog Slap
F39		Analog Trip
F40		Analog 8 <sup>th</sup>
F41	STEREO DELAY	Slap Quick
F42		Long Slap
F43		Spacey
F44		Long Tail
F45		ST: Triplet
F46	PING-PONG DELAY	Ping-pong Slap
F47		Ping-pong Delay
F48		Ping-pong Spacey
F49		Ping-pong Trip
F50		Ping-Pong Purple Rain
F50-99	USER-CREATED PRESETS	

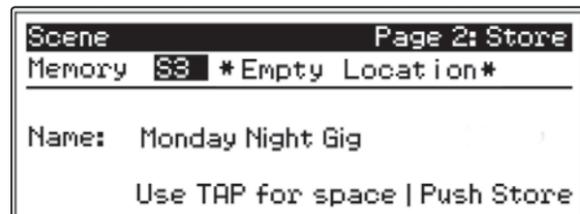
## 5.0 Scenes, Presets, and the System Menu

### 5.1 Creating and 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. For example, in the studio, saving and recalling a Scene allows you to move to another song or project and come back to the current mix later. For live shows with multiple bands, you can set up custom mixes for each band at 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 and 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.

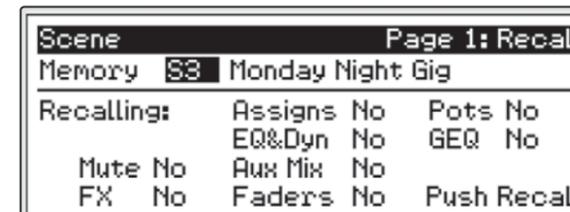
Located at position S1 is a scene named Zero Out (Board Reset). This scene cannot be overwritten and returns your StudioLive to its default factory setting. All you have to do is lower the faders and return all trim knobs and output volume knobs (Solo bus, 2 Track In, Aux Input A&B, Aux Output 1-10, FXA&B Output, Phones, and Monitor) to their lowest position. Your StudioLive will be zeroed out as follows:

SETTINGS								
INPUTS & BUSES	ASSIGN	PAN	SOLO	MUTE	SELECT	LINK	SENDS	MODE
INPUT CHANNELS	MAINS	C	OFF	OFF	CH1	OFF	NULL	
AUXES 1-10	N/A	C	OFF		NULL	OFF		PRE 1
FX A & B	NULL	C		OFF	NULL			POST
MAIN BUS		C			NULL			
MONITOR BUS								MAIN
								NULL
METER MODE								NULL

The Fat Channel will be restored to the same setting for every input and output on your StudioLive. Each of the dynamics processors and 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	KEY	N/A		OFF	THR	0dB	P/S	PEAK	n/a	n/a	PEAK
	THR	.05 s	THR	0 dB	RAT	2:1	Q	0.6	0.6	0.6	0.6
	RNG	N/A			ATK	20 ms	FREQ	130Hz	320Hz	1.4kHz	5kHz
	ATK	5 ms			REL	150ms	GAIN	0dB	0dB	0dB	0dB
	REL	0.7 ms			GAIN	0dB					
	MOD	EXP			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 stored parameters (Fat Channel settings, channel muting and soloing, aux mixes, and internal effects parameters) except fader and knob positions.

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 1-24, 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, FireWire Returns to Inputs, Solo and Monitor bus assignments, and Master Control Assignments)
- EQ and 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 1-10, FXA and FXB, and Pre1/Pre2 positions)
- Faders: All fader positions
- Pots: All digital knob positions (Aux 1-10 Outputs, FXA and FXB Levels, Cue, 2 Track In, Phones, Monitor, and Aux Input A and Aux Input B knobs)
- GEQ: Graphic EQ settings

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 to recall the stored position. The subgroup and main meters at the top right of the StudioLive will display the recalled positions for their respective faders. 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 of the physical fader. 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 1-10 Outputs, FXA and FXB 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 a scene and recall it later, it is highly recommended that you use the recall sheet provided in the back of this manual to note the stored position of these controls.

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

### A Quick Note About the AutoStore Feature

It is not necessary to create a Scene for your StudioLive in order to preserve 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: If you make changes to a stored Scene, AutoStore will not save these changes as a permanent part of the Scene. Any changes made to a Scene in the StudioLive's library must be saved using the Scene menu, as described in the first part of this section.**

## 5.2 Saving and Loading Channel Presets

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

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 onto 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.



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.

### 5.2.1 Channel Presets Library

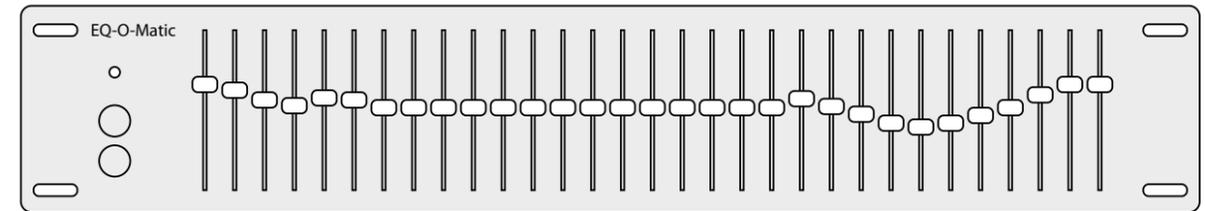
Your StudioLive comes with 50 Fat Channel presets custom designed by professional PreSonus users. These presets can be altered, renamed and overwritten; however, there are 50 additional empty storage locations for you to build your own custom library of channel strip settings.

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

### 5.3 Graphic Equalizers

The StudioLive 24.4.2 features 4 stereo (dual mono), 31-band, 1/3 octave graphic EQs that can be inserted on the Main output bus, the Subgroups, or any of the Auxes. When the GEQ button is enabled, the meters and encoders of the Fat Channel become the controls for the Graphic EQ. As you touch a knob, you will notice that its band number, frequency, and gain are displayed in the System Menu. The 31 bands range from 20 Hz to 20 kHz.

If these were analog graphic equalizers, each would look like this:



The frequency for each band is fixed. Bands 5-28 are controlled by encoders 1-24, respectively. When Band 4 or 29 is selected in the Show Band field in the GEQ menu, the meters will flip. You will notice that all meters have one LED illuminated to display the current gain position for each band, and the meter for the selected band in the Show Band field is inverted, meaning that all LEDs will be illuminated except the LED displaying the current gain position for that band. The band does not have to be selected in the Show Band field for its encoder to be active. All encoders are active so you can make changes to 24 of the 31 bands at one time.

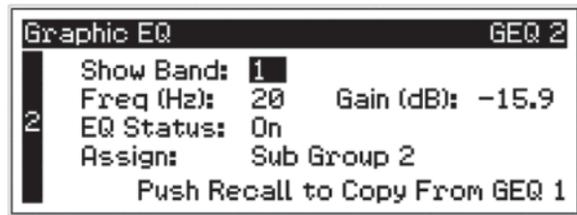
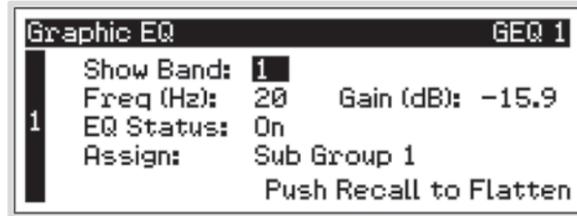
The StudioLive Graphic EQ features an innovative design that sets it apart from traditional graphic EQs. Traditionally, a 31-band graphic EQ uses 31 second-order shelving filters with fixed frequencies in order to simulate a curve set by the user via 31 front-panel sliders. A well designed graphic EQ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

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. Making drastic adjustments too quickly, may create an audio artifact as the new curve is redrawn and crosses over an audio peak. If you are hearing audio artifacts while setting up your Graphic EQ, try creating a smoother curve.

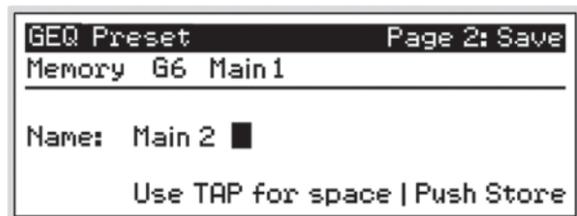


When you press the GEQ button on your StudioLive, the Graphic EQ menu will be displayed on your LCD screen. There are 4 dual-mono (8 channels) Graphic EQs in your StudioLive 24.4.2. These Graphic EQs are assignable in stereo pairs. Each GEQ can be assigned to the Mains, Subgroups 1-4, or Aux 1-10. The bus assignment can only be made from an odd-numbered GEQ. The even-numbered GEQ will automatically be assigned to the other half of the stereo pair. For example, if you assign GEQ 1 to Subgroup 1, GEQ 2 will automatically be assigned to Subgroup 2.

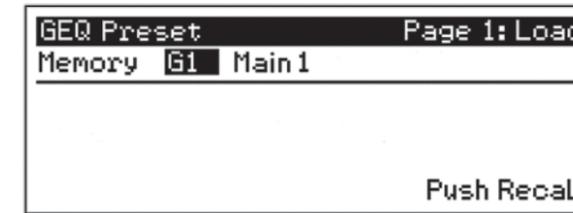
Each of the eight channels of Graphic EQ can be controlled individually or odd-even pairs can be controlled as a stereo unit. To copy the left side (odd) GEQ settings to the right side (even), simply page down to the right side and press the Recall button. Pressing the Recall button while using an odd numbered GEQ will flatten it.

Use the Page Up and Page Down buttons to navigate between Graphic EQs. Use the Next and Prev buttons to navigate through each GEQ screen and the Value knob to change a parameter.

Like all other parameters on your StudioLive, your Graphic EQ settings can be stored and recalled. If you have created a GEQ setting that you would like to save to the GEQ Preset library, press the Save button in the Fat Channel while that GEQ is active. You will notice that the LCD will display the GEQ Save menu.



To begin, use the Value encoder to scroll to an empty position in the GEQ Preset library. 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 GEQ preset is being written to the StudioLive's internal memory. Once the GEQ preset is saved, the Store button will return to its unlit state.



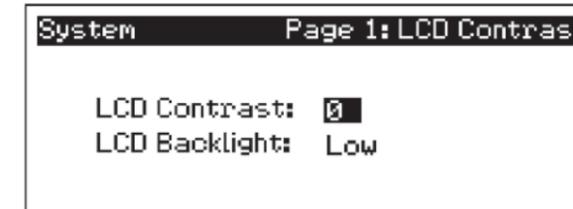
To load a preset to any channel on the StudioLive, first press the GEQ button and Page down to the desired GEQ. From the Fat Channel, press the Load button. You will notice that LCD now displays the GEQ Load menu. Use the Value encoder to locate the preset you would like to use.

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

## 5.4 System Menu

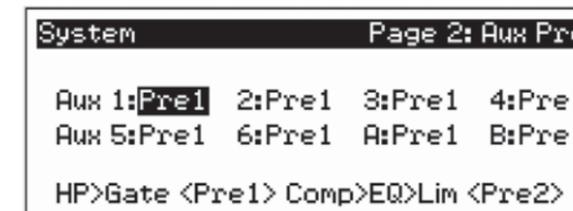


In most digital-console manuals, the phrase "System menu" inspires a sense of dread. However, with the StudioLive, you have nothing to fear. The System menu on the StudioLive serves just a few simple functions, most of which have very little to do with mixing a show.



### LCD Contrast and LCD Backlight.

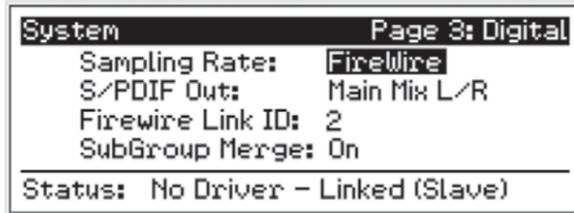
Press the System button to access the System menu. The first page allows you to adjust the LCD Contrast and LCD Backlight for optimal viewing in your working environment. These settings are retained when the unit is powered down.



### Aux Pre Position.

Press the Page Down button to access the Aux Pre Position page. By default, all 10 Aux buses are set to Pre 1. This places the send of all 24 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 2 internal effects buses are set to Pre 2, which routes each of the 24 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 readout.

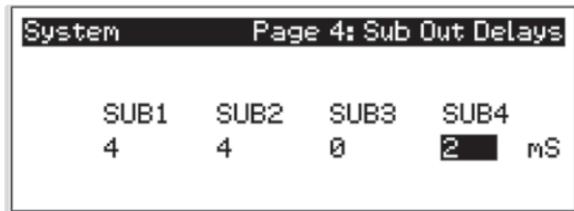


**Digital Information.**

Press the Page Down button to access the Digital Information page. From here, you can change the Sample Rate, view your FireWire connectivity, route a stereo mix 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, in which case you need to use the Universal Control Device Window to make your S/PDIF assignments and change your sample rate. (See Section 7.1.1 for more details.) When two 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 8.9.

*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.*



**Sub Out Delay**

On Page 4 of your System menu, you will find the Sub Out Delays. Your StudioLive provides you with the ability to delay the audio from each of your subgroup outputs by 2 to 500 ms, in increments of 2 ms. When you correctly set the delay time for these outputs, you ensure that the sound from each speaker in the P.A. system arrives at the listening position at the same time.

The delay should be set for the speakers to which your listener will be closest.

Let's say you are using your StudioLive in a large theater with a balcony, and you have three pairs of speakers: a pair in front of the stage, a pair in the rear of the auditorium, and a pair at the front of the balcony. You will need to delay the rear speakers so that the listeners closest to them will hear the audio from both the rear and stage speakers at the same time. This is also true for the audience in the balcony. While the balcony speakers will provide the loudest source of audio, the balcony audience will still be able to hear the stage speakers, so a delay must be

set for the balcony speakers. You will need to calculate approximately 0.885 ms of delay time per foot. So if your balcony speakers in the above example are 61 feet from your stage speakers, you will need to set their delay time to 54 ms.

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



**Lockout Mode**

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 processing and the like.

Right out of the box, the StudioLive cannot be locked, so don't worry about hitting the wrong button. To enable the Lockout feature, you must first connect and sync your StudioLive to a computer. Please review Section 7.2.6 for more information on this feature.



**Firmware Version**

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.

## 6.0 Connecting to a Computer

The StudioLive 24.4.2 is more than just a mixer. It's also a very powerful computer interface that allows you to record any mixer input or bus with a Select button — and also control the mixer from a separate laptop computer via Virtual StudioLive.

### 6.1 System Requirements

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

#### Macintosh

- **Operating Systems:**
  - Mac OS X 10.5.2 or higher (including Mac OS X 10.6)
- **Hardware:**
  - Minimum: PowerPC G4 1.25 GHz or Intel Core Solo 1.5 GHz processor, 1 GB RAM
  - Recommended: 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
  - Windows 7 32-bit and 64-bit
- **Hardware:**
  - Minimum: Intel Pentium 4 1.6 GHz processor or AMD Athlon 64 3000+ (Turion) , 1 GB RAM
  - Recommended: Intel Pentium 4 2.8 GHz EM64T or better or AMD Athlon 64 XP 3000+ or better, 2 GB or more RAM

NOTE: The speed of your processor, amount of RAM, and capacity, size, and speed of your hard drives will greatly affect the overall performance of your recording system. A faster processor and more RAM can reduce signal latency (delay) and improve overall performance.

\*\*Subject to change check [www.presonus.com](http://www.presonus.com) for updates.

### 6.2 Installation for 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.

We made the StudioLive installer 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 antivirus software, and disconnect the StudioLive from your computer. After the installation is successfully completed, don't forget to reenale your antivirus protection!



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

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 now synced to your computer and ready to use!

## 6.3 Installation for Mac OS X

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, and be especially careful that you do not connect your StudioLive too soon.



1. You will be directed to the Welcome screen. Click "Continue" and follow the onscreen instructions.



2. 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."



3. When the installation is completed, connect your StudioLive to your Mac with a FireWire cable and power it on. You are now ready to use your StudioLive!

## 6.4 Using the StudioLive as an Audio Interface

Your StudioLive features a 32x26 FireWire interface. Any input and bus with a Select button, plus the Solo bus, tape input, and talkback mic, can be recorded with any audio-recording application that supports Core Audio or ASIO. 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.

### 6.4.1 Using the StudioLive with Popular Audio Applications

Complete setup instructions for PreSonus Studio One Artist and a brief tutorial on its features are located in Section 7.0 of this manual. Below are basic driver-setup instructions for three popular audio applications.

#### Apple Logic Pro/Express 7+:

1. Launch Logic Pro/Express.
2. Go to Logic | Preferences | Audio.
3. Click on the Devices Tab.
4. On the Core Audio 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+

1. Launch Cubase.
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+

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."

### 6.4.2 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 the FireWire bus, and its output (return) signal is hard-coded to designated mixer channels. Your StudioLive has 32 available sends and 26 returns.

FireWire sends 1 through 24 are hard-coded to be sent pre-fader from the 24 input channels of the StudioLive. These sends can be pre- or post-Fat Channel EQ and dynamics. To record the EQ and dynamics processing 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 discussed in Section 4.5.1, 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 25-32 are routed in 4 stereo pairs from the Universal Control Device window. Any combination of four 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 7 & 8
- Aux Send 9 & 10

- 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

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 Subgroups 3 and 4. For more information on subgroup mixing, please see Section 8.4.

The obvious exception to this principle is the Talkback Mic input. The Universal Control Device window allows you to assign the talkback output 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 effects returns. If you record an internal effects send (Aux Sends A and B), you will be recording the unprocessed ("dry") mix of the channels being sent to that internal effects bus. To record the processed mix, choose the internal effects returns (FX Return A or FX Return B). This is the direct output from the internal effects, so it will be 100% wet. You may wish to record both the dry and the wet mix so that you can blend them to taste within your session.

Section 7.2.5 discusses how to select the sources for these 4 stereo FireWire sends.

Each of the StudioLive's 24 inputs are hardcoded to receive their respective FireWire returns. Outputs 1 through 24 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 through one of these outputs, it will always be accessible on its channel by simply pressing the FireWire input button.

To provide the most flexible mixing environment, the main mix output for any application should be assigned to Outputs 25 and 26. 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 24 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).



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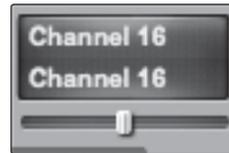
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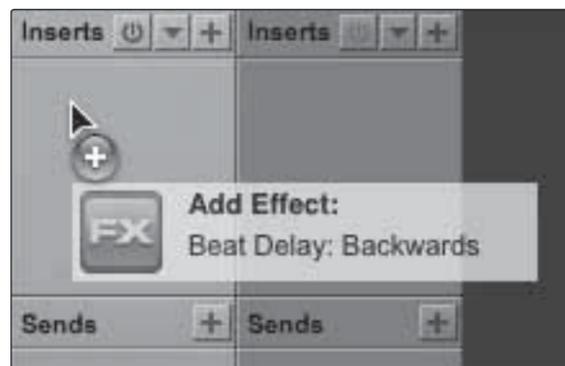
### 6.4.3 Using Plug-In Effects As Inserts

FireWire streaming on your StudioLive is continuously bidirectional. This means that the StudioLive is always sending signals from the direct FireWire sends on all 24 input channel, as well as from the auxiliary inputs and buses assigned to the second bank of eight FireWire sends. At the same time, the StudioLive is receiving signals back from the 26 FireWire returns. Because the FireWire returns always come back to their respective StudioLive channels, 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 the Beat Delay plug-in from PreSonus Studio One onto Channel 16 of the StudioLive.

1. To begin, create a mono audio track in Studio One. Assign its input to Channel 16 and its output to Output 16. (Several DAW applications, including Apple Logic™, do not offer mono output buses. If this is the case, you must route the output stream to Channels 15-16 and pan the channel all the way to the right so that it will only be sent to output 16. Please consult your software's user manual for specific instructions.)



2. Once you have the routing set up in Studio One, drag-and-drop the Beat Delay plug-in onto your track and record-enable it. Software monitoring will be enabled automatically.



3. Press the FireWire Return Input button on Channel 16 of your StudioLive. You can now monitor your analog signal with your inserted effect as you record from any of the output buses on the StudioLive.

#### A Quick Note About Printing Fat Channel Dynamics and EQ

You probably will want to use the Fat Channel dynamics processing and EQ during post-production. 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, press the channel's Select button and assign the channel 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 the return 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 processing will not.*

*If you wish to add Fat Channel processing to a FireWire return, you must disable Dig Out.*

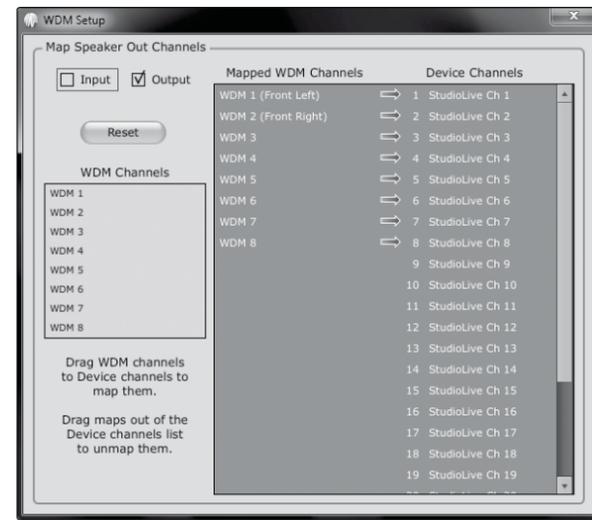
### 6.4.4 Advanced WDM Features (PC only)

Like the FireStudio-family interfaces, the StudioLive features 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. For more information on Universal Control, please consult the next section.

**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.

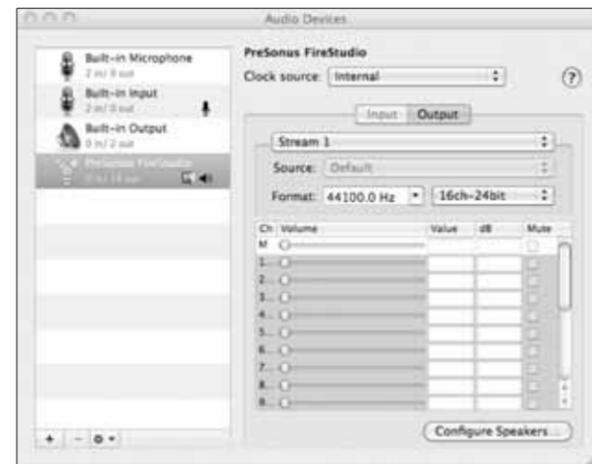


2. The WDM Channel Mapper window will open. By default, WDM Outputs 1 and 2 are routed to StudioLive Channels 1 and 2. As previously mentioned, we recommend that they be set, instead, to Channels 25 and 26, so as not to use input channels on your StudioLive for WDM playback. To begin, check the box next to Output. Click on WDM 1 (front left) and drag it down to place it next to StudioLive 2 Track In Left. Click on WDM 2 (front right) and drag it down to place it next to StudioLive 2 Track In Right.

Your stereo WDM playback will now be routed directly to the main FireWire return buttons in the 2 Track In and Monitor buses in the StudioLive Master Section.

### 6.4.5 Configuring the StudioLive For Core Audio (Mac Only)

Your StudioLive can be used as the main Core Audio device for a 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 25 and 26 are available through the FireWire Return buttons inside the 2 Track In and Monitor sections of the StudioLive, and we recommend that you use these channels 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.



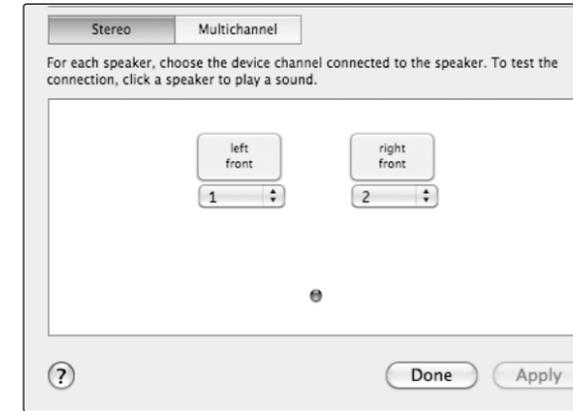
Go to MacHD | Applications | Utilities and launch Audio MIDI setup.

Configure Speakers...

Click on the Configure Speakers button.



3. Deselect Stream 1 and select Stream 2 instead.



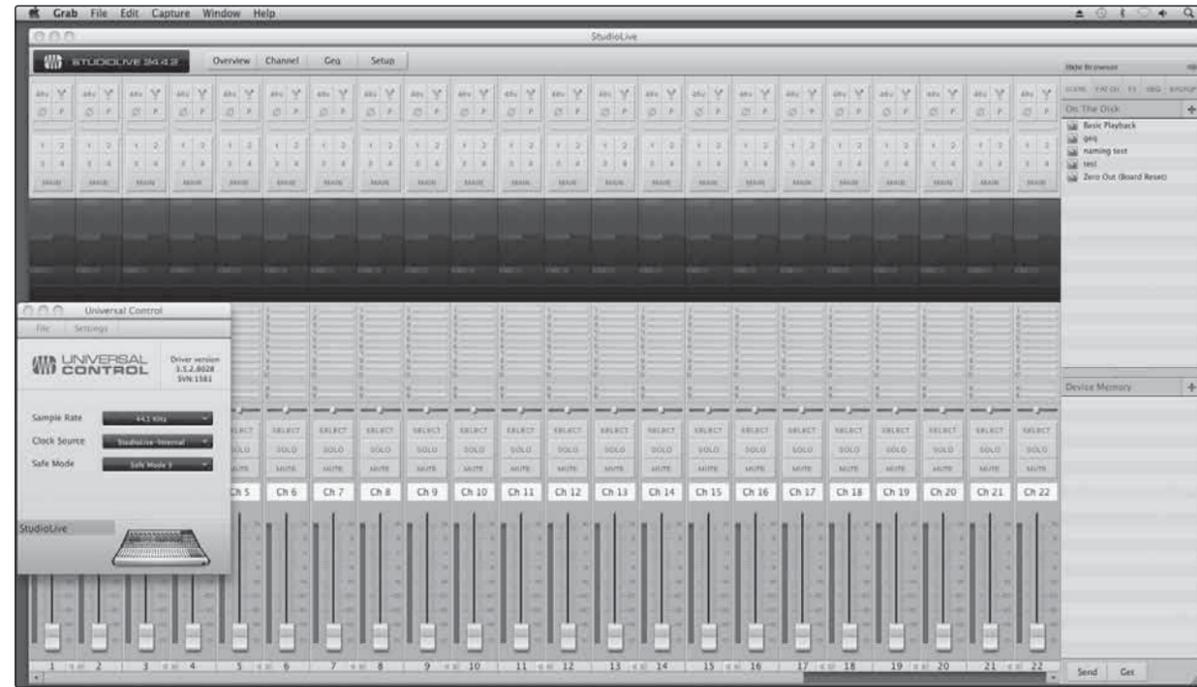
4. Choose Channel 25 for the left speaker and Channel 26 for the right speaker.

5. When you are finished, click Apply and then Done.

## 7.0 Software: Universal Control With VSL, Capture, and Studio One Artist

You StudioLive 24.4.2 comes with a powerful software bundle that provides you everything you need to record live shows or in the studio; produce and mix an album; and even back up, restore, and organize all your user Scenes and Presets on your mixer.

### 7.1 Universal Control



The StudioLive 24.4.2 is compatible with the Universal Control application. As with all FireStudio- and StudioLive-family interfaces, Universal Control includes two windows: the Launcher window and the Device window. For the StudioLive, the Device window is the Virtual StudioLive (VSL) application. VSL provides bidirectional control of channel, subgroup, and main-bus levels; Fat Channel parameters; aux mixes; effects; and the Graphic EQs. It also provides a visual overview of your StudioLive settings so that you can see, adjust, and organize them. VSL includes a librarian, allowing you to easily manage your presets and Scenes.

Controlling your StudioLive with VSL is as easy as drag-and-drop. Load Fat Channel presets and scenes by simply dragging them onto the channel or mixer overview. You can load Fat Channel presets as a complete channel strip or as individual gate, compressor, or EQ presets. Through VSL, you

can back up all of the Scenes and presets stored on your StudioLive. These stored settings can be loaded from disk or sent to, and stored internally on, the StudioLive. You can even drag presets out of the browser and email, IM, or disk-swap them with other StudioLive owners.

Since the control is bidirectional, fader moves and parameter changes made on the StudioLive are reflected in VSL, as well. So, for example, you can set up the StudioLive the way you want it and then save your Scene or other presets in VSL.

As discussed in Section 6.4.2 VSL includes an auxiliary-inputs router to allow you the freedom to choose which additional buses to route to the eight stereo auxiliary FireWire sends. It also lets you create a password for your StudioLive so that you can lock out unauthorized users. (See Section 7.2.6 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 6.4.4).

*Note that the Meter Style and Meter Decay options in the Launcher window are not active when only a StudioLive is connected to a computer.*

### 7.1.1 Universal Control: Launch Window

#### Sample Rate Selector



#### Changes StudioLive Sample Rate.

Set the sample rate to 44.1 or 48 kHz from the Launcher window when your StudioLive is connected to a computer.

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.

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.

### Clocksource

Changes the Clock Source for Chained FireStudio-family Devices.



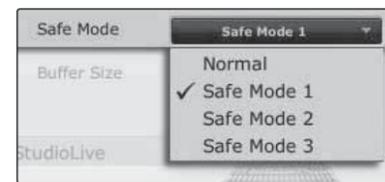
The StudioLive cannot slave to an external clock. However, if you are cascading it with FireStudio-family products that do feature digital inputs, the clock source becomes selectable.

The clock source setting will determine where the device chain is receiving word-clock information. This keeps the chained devices 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, in which case set the clock to StudioLive-Internal.

### 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.

### 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 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 the Mac, the buffer size is set from inside your host application.

### Device Window Button

Opens the Device Window.

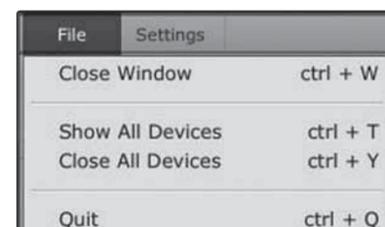


Click on this button to open the VSL Virtual StudioLive application.

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.

### File Menu

Opens and Closes Launch and Device Windows.



From the **File** menu of the Launch Window, you can open and close both windows, as well as quit the Universal Control application

**Close Window.** Closes just the Launch window.

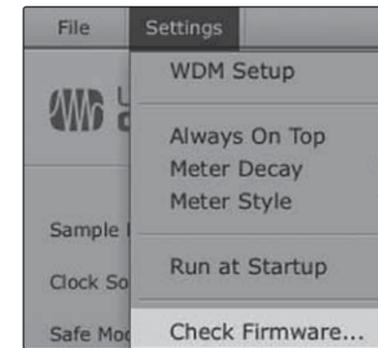
**Show All Devices.** Opens the Device window for all of the connected FireStudio-family interfaces.

**Close All Devices.** Closes the Device window for all of the connected FireStudio-family interfaces.

**Quit.** Quits the Universal Control application.

### Settings: Check Firmware

Scans Your StudioLive and Updates the Firmware.

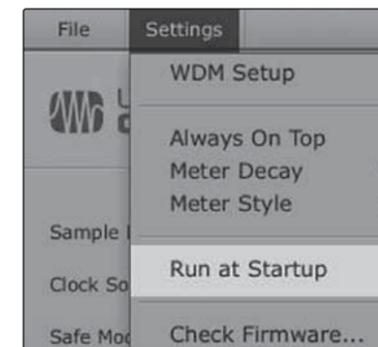


A firmware updater is built into the Universal Control application. Periodically, a driver update will require that the firmware on your StudioLive be updated. Whenever you install an update for the Universal Control or add a new FireStudio-family product to your system, you should use this feature to ensure that the firmware is up to date. If the firmware needs to be updated, the update application will launch automatically.

The firmware updater can also be used to reset your StudioLive 24.4.2 to factory default. To reset your StudioLive, hold down the CTRL and ALT [OPTION] keys while clicking on "Check Firmware." Once the firmware updater has finished, select "Reset" rather than "Restore."

### Settings: Run at Startup

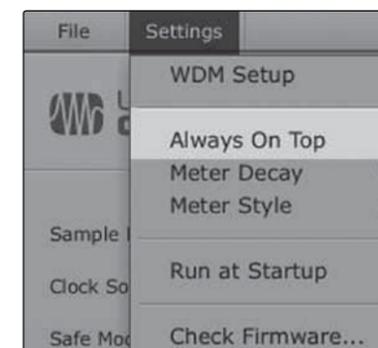
Launches the Universal Control Application Automatically on Startup (Windows Only).



When this is enabled, the Universal Control application will automatically launch each time you boot your Windows PC.

### Settings: Always on Top

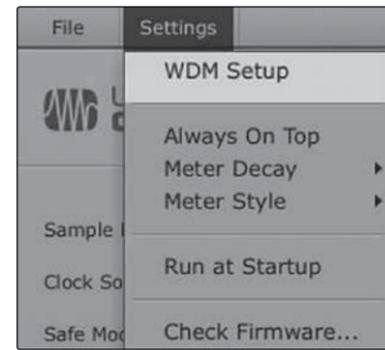
Allows the Launch Window to Stay in View When Other Applications Are in Use.



When this is enabled, the Launch window will remain in the foreground when other applications are active, rather than being in the background behind the current application's window.

**Settings: WDM Setup**

Opens the WDM Channel Mapper.



The StudioLive features advanced WDM features. Please see Section 6.4.4 for further details.

**7.2 VSL: Virtual Studio Live**

VSL is a highly advanced editor/librarian and control application that is completely integrated with the StudioLive 24.4.2. Because of the continuous bidirectional communication between your StudioLive and VSL, whatever you do on the StudioLive's control surface will be reflected in VSL and vice versa.

VSL for StudioLive 24.4.2 requires a minimum screen resolution of 1024x768. For vertical resolutions set below 914, VSL will dynamically change the channel faders to knobs.



**7.2.1 VSL: Browser**

When you first launch VSL, notice the Browser window along the right side of the screen. The Browser in VSL functions similarly to the Browser in Studio One. From the Browser, you can see all of the Scenes, Fat Channel presets, FX presets, and graphic EQ settings that are saved on your StudioLive and on your computer. You can also create new settings and can back up your entire library from this window. Simply drag-and-drop a Scene or preset to load it on your StudioLive.

**Get Button**

Transfers All Scenes, Fat Channel, FX, and Graphic EQ Presets Stored on the StudioLive to VSL.

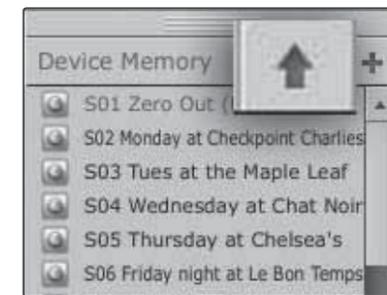


When you first launch VSL, you will need to create a link between your StudioLive's internal memory and your computer. To do this, click on the Get button.

A dialog will open, prompting you to click the transfer button. Any settings that are temporarily stored in the local cache (i.e., settings that are currently visible in the Device Memory section of the Browser window) will be overwritten.

**Add to Disk Button**

Transfers all Scenes, Fat Channel, FX, and Graphic EQ Presets from Temporary Memory to the Permanent Cache.



VSL allows you to back up your Scenes, Fat Channel, FX, and graphic EQ presets and permanently store them on your computer. Each type of preset can be added separately. In this way, you can back up only what you want, when you want.

To move a Scene or preset from temporary memory into permanent memory, simply select one preset or Scene and click the Add to Disk button. To select multiple presets, hold the Shift key while making your selections.

**Browser Tab Buttons**



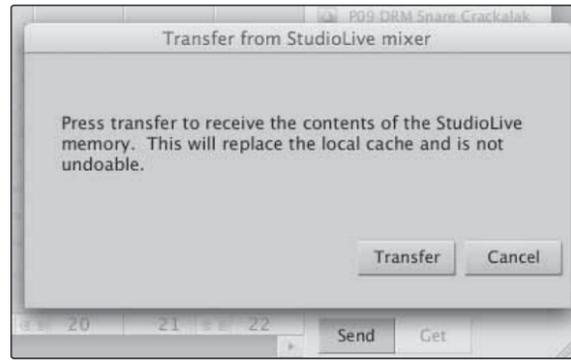
Displays the Different Preset Categories on Your StudioLive and on Your Computer.

All of your Scenes and presets are contained in dedicated folders in VSL. To view a specific set of presets, simply click on its tab.

- **SCENE.** Displays stored Scenes.
- **FAT CH.** Displays stored Fat Channel presets.
- **FX.** Displays stored effects presets.
- **GEQ.** Displays graphic EQ presets.
- **BACKUP** Displays any backup logs that have been created in VSL.

**Send Button**

**Transfers Designated Scenes, Fat Channel, FX, and Graphic EQ Presets from VSL to StudioLive Memory.**



VSL makes reorganizing all the Scenes and presets stored on your StudioLive as easy as dragging-and-dropping a file. To load your StudioLive with new Scenes and presets, simply drag any Scene or preset from the On the Disk section of the Browser to any position in the Disk Memory section of the Browser.

A dialog will open asking you to verify that you would like to overwrite the Scene or preset at the new position. This will not immediately overwrite what is stored internally on your StudioLive; it will merely overwrite what is stored in the VSL cache memory.

Once you have organized the files you wish to transfer to your StudioLive, press the Send button. When the transfer is complete, you can disconnect your StudioLive from your computer and take your chosen Scenes and presets with you.

**Add New Button**

**Creates a New Scene or Preset.**



In the On the Disk and Device Memory sections of the Browser, you will see the Add New button. Clicking this button will immediately create a new Scene or preset. If you want this new preset to be temporarily stored in VSL and immediately sent to the StudioLive's internal memory, add the new preset in the Device Memory section. If you would prefer to have this new preset stored in permanent memory on your computer, use the Add New button in the On the Disk section. In either case, the new preset can be sent to your StudioLive at any time.

**Remove Button**

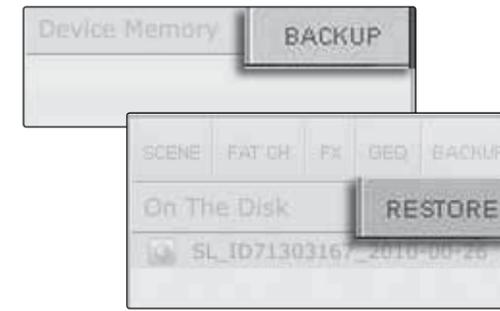
**Deletes a Stored Scene or Preset.**



In the On the Disk section of the Browser, notice the Remove button. Clicking this button will delete the currently selected stored Scenes or presets.

**Backup Tab**

**Creates and Restores Backups of Your StudioLive.**



The Backup tab allows you to create complete time-stamped snapshots of your StudioLive. This can be especially useful when completing a project that may need to be revisited in the future. To create a backup, simply click on the Backup button.

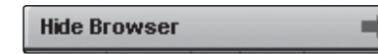
To restore any backup file, select it in the On the Disk portion of the Browser and click the Restore button. You will be warned that any Scene or preset currently loaded in the Device Memory section of the Browser will be overwritten. Once your backup is restored, you can click the Send button to transfer your Scenes and presets back to the StudioLive.

**Show/Hide Browser**

**Displays/Hides the Browser Window.**



The Browser can be hidden from view to provide more real-estate for your mix. Simply click on the "Hide Browser" button to close the Browser.



To reopen the Browser, click on the "Show Browser" button in the upper right corner of your VSL window.

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7.2.2 VSL: Overview Tab



At the top of the VSL window, you will see four tabs: Overview, Channel, GEQ, and Setup. The Overview tab provides you with a complete graphical representation of your StudioLive. As you adjust parameters on the StudioLive, you will notice that the VSL overview is also updated. If you use your mouse to adjust a parameter in VSL, the StudioLive will be updated remotely. It is important to remember that every button, knob, slider, and fader on the VSL corresponds directly to a button, knob, slider, or fader on your StudioLive.

The graphic above identifies each controller labeled with its corresponding control on the StudioLive and can be used as a map to quickly learn how to navigate the Overview Tab.

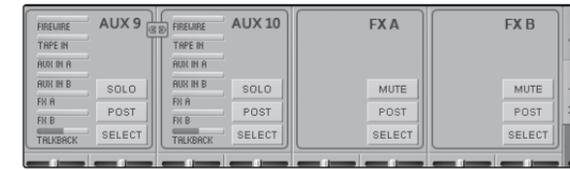
Aux Send View



VSL provides two views for your StudioLive Aux Outputs and FX Buses. The Aux Send View gives you an overview of the send levels of every channel on your StudioLive to every Aux Output and FX Bus.

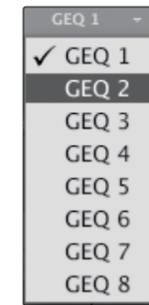
To quickly set up an aux mix, use your mouse to set the send level for Channel 1. Right-click on the Channel 1 send level and sweep your mouse across the other channel sends for that Aux. The set send level will be copied to every other channel for that Aux.

Masters View



The Masters View displays the controls for each Aux Output and FX bus just as they are on your StudioLive. The Masters View also displays the send levels for the Aux Flip Mode Channels (Main FireWire Return, Tape Input, Aux In A & B, FXA, FXB, and Talkback). From within the Masters View, you can also Select each Aux and FX bus to edit its Fat Channel insert.

GEQ in Focus



In general, most Graphic EQ settings will be created prior to any live show and not adjusted after that. However, sometimes minor adjustments must be made later. VSL makes this quick and easy. You will have noticed that there is a GEQ above the Master section on the Overview tab. You can focus any GEQ here by one of three ways:

- **Select a GEQ on the GEQ Tab.** For example, if you leave GEQ 5/6 in focus on the GEQ tab and switch back over to the Overview tab, you will still be able to adjust it using the Focus GEQ above the Master Section.
- **Select the GEQ from the GEQ Focus Menu.** In the upper left corner of the GEQ in Focus, you will find a pull-down menu. From this menu, you can select any of the eight GEQs to pull into focus on the Overview tab.
- **Click on a Focus GEQ button.** Once a Graphic EQ has been inserted on an Aux, Subgroup, or Main, a Focus GEQ button will be displayed above that bus. (Note: For Auxes, Focus GEQ buttons are displayed in the Masters View only.) Clicking on this button will pull that bus's inserted Graphic EQ into focus on the Overview Tab.

### 7.2.3 VSL: Channel Tab

The Channel Tab provides a detailed overview of the Fat Channel parameters for the selected Channel. The selected Channel will always be shown above the Gate section. It is important to remember that you have continuous bidirectional control. If you wish to grab a point in the EQ with your mouse, for example, you will be changing the parameters both in VSL and on your StudioLive.

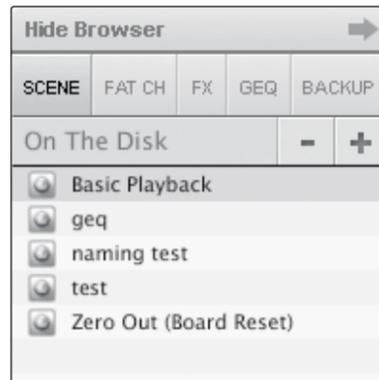
To access the Fat Channel for Aux Ins A and B, you will need to press its Select button on the StudioLive. You can then load presets and make adjustments using VSL from within the Channel tab. Auxes 1-10, FXA, and FXB can be selected using the Masters View.



### 7.2.4 Loading Scenes and Presets from VSL

As stated in the previous section, the Browser window in VSL functions much in the same way as the Browser in Studio One. This means that to load a Scene or preset from the Browser window, you simply select it and drag it over the mixer or channel you wish to load it on. Scenes and presets can be dragged from either the On Disk or the Device Memory section of the Browser and dropped onto the Overview or the Channel tab.

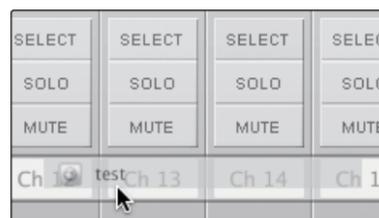
#### Loading a Scene



To load a new Scene on your StudioLive, select it from the Browser window and drag it over the mixer in either the Overview or the Channel tab. The window will grey out indicating that a new Scene is about to be loaded. Please note: only the parameters that have been enabled for recall on the StudioLive will be recalled on the mixer.

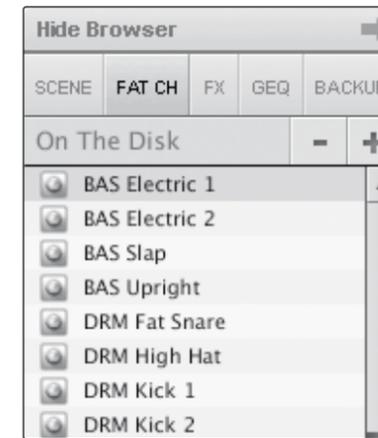
See Section 5.1 for more details.

#### Loading Scribble Strip Labels



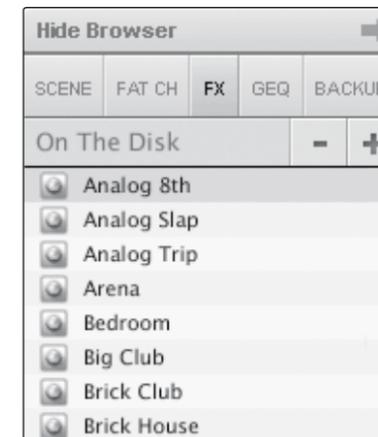
Scribble Strip labels are stored with your Scenes. You can load just the labels by selecting the Scene from the Browser window and dragging it over the row of Scribble Strips above the Channel Faders. The Scribble Strips will grey out indicating that the Scribble Strip labels from the Scene are about to be loaded.

#### Loading an Entire Fat Channel Preset



To load every component in a Fat Channel preset (Gate, Compressor, EQ), select it from the Browser window and drag it over any part of the desired channel. If you drag it over any of the component quick views, it will load only that component (e.g., if you drag a preset over the Gate Quick View, only the Gate will be loaded).

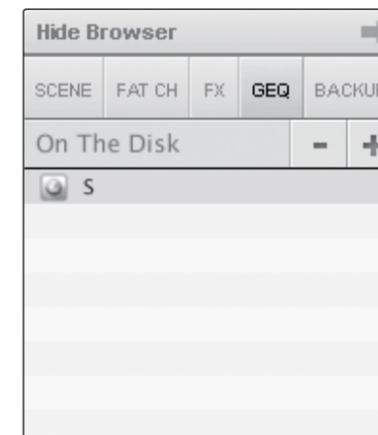
#### Loading an FX Preset



To load an FX preset, select it from the Browser window and drag it over any part of the desired FX bus in the Master section of the Overview tab. Once it is loaded, you can use the FX Type menu to change the effect and create new presets.

*Note: At this time, VSL does not transfer the name of the preset to the StudioLive. All FX presets loaded from within VSL will be labeled "Natural" in the FX menu on your StudioLive.*

#### Loading a GEQ Preset

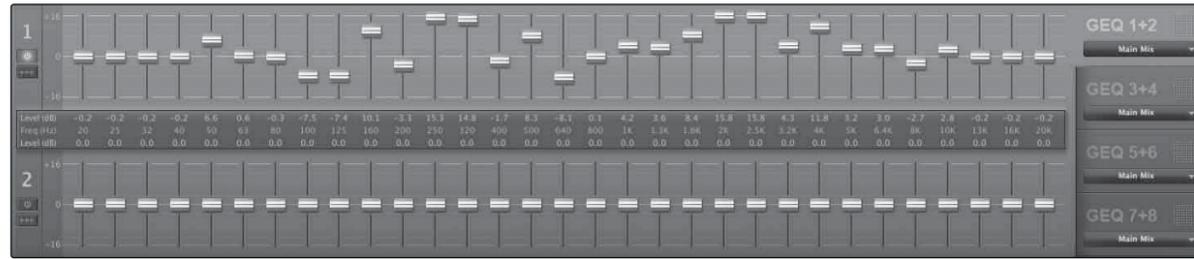


To load a graphic EQ preset, select it from the Browser window and drag it over any part of the focused graphic EQ. Graphic EQ presets can be loaded on the Overtab or the Geq tab. Once it is loaded, you can use the sliders in VSL or the encoders on the StudioLive to make adjustments. Note that you must be in the GEQ menu page for the Graphic EQ you wish to control in order to use the encoders on your StudioLive to control each graphic EQ in VSL. See next section for details.

7.2.5 VSL: GEQ Tab

The StudioLive 24.4.2 features 4 dual-mono Graphic EQs. Assignable in stereo pairs, these Graphic EQs can be inserted on any of the 10 Aux Buses, the 4 Subgroups, or the Main Bus. In addition to allowing you to control each Graphic EQ individually, VSL automatically links each dual-mono Graphic EQ to create a true stereo Graphic EQ if the pair is inserted onto a stereo bus, allowing you to control both sides at once (e.g., if you insert GEQ 1/2 onto Subgroups 3 and 4 and stereo-link the Subgroups, GEQ 1/2 will function as a stereo Graphic EQ rather than a dual-mono Graphic EQ).

*Note: The GEQ Menu on your StudioLive does not have to be active to make changes to a Graphic EQ from VSL. If you wish to use the Fat Channel encoders to control the Graphic EQs in VSL, then you must open the GEQ Menu on your StudioLive by pressing the GEQ button.*

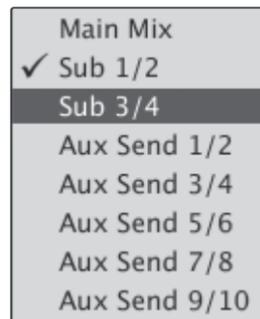


Selecting a GEQ to Edit



Along the right side of the GEQ tab, you will see another set of tabs, one for each dual-mono GEQ. To bring a GEQ in focus so that you can edit it, simply click on its tab.

GEQ Bus Assignment



To Insert a Graphic EQ on a bus, click on the pull-down menu below the GEQ's number and select the pair of buses onto which you'd like to insert it.

Enabling a GEQ



By default, all Graphic EQs on your StudioLive 24.4.2 are disabled. To enable them remotely, via VSL, simply click on the Enable button to the left of the sliders.

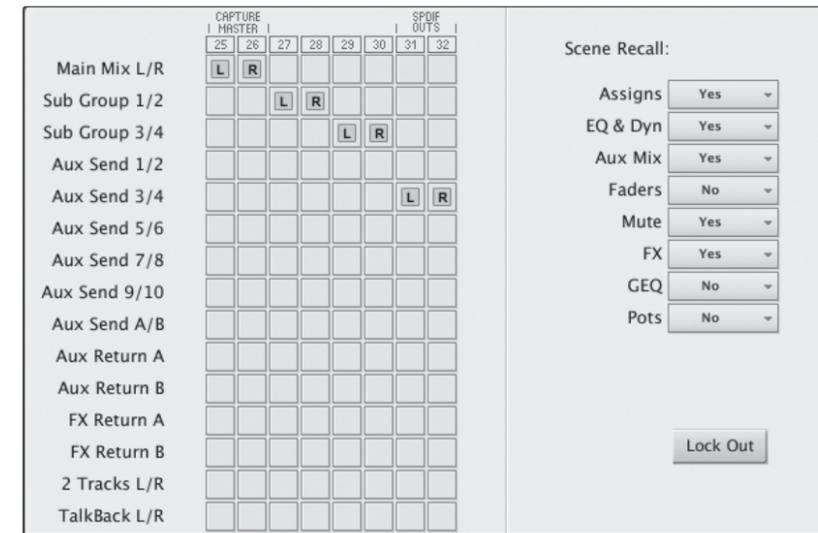
Flattening a GEQ Curve



To zero out all curve settings on any GEQ, click on the Flatten GEQ button directly below the Enable button. This will flatten all band gains to 0 dB so that they neither boost nor attenuate any of the 31 bands.

7.2.6 VSL: Setup Tab

As discussed in Section 6.4.2, the StudioLive allows you to route any 8 of 23 buses and inputs, in addition to its 24 input channels. This is done via the Auxiliary Inputs Router on the Setup tab. To route FireWire Sends 25 through 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 patch them to a stereo pair of auxiliary inputs of your choice. Remember, all of these buses and inputs are automatically set to send their signals post-Fat Channel dynamics and post-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.

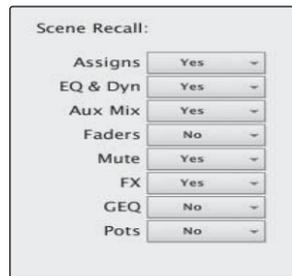


For instance, the routed pairs in the picture would translate as the following chart in your host recording application:

	PHYSICAL	SOFTWARE
Main Mix Left		Auxiliary In 25
Main Mix Right		Auxiliary In 26
Subgroup 1		Auxiliary In 27
Subgroup 2		Auxiliary In 28
Subgroup 3		Auxiliary In 29
Subgroup 4		Auxiliary In 30
Aux Send 3		Auxiliary In 31
Aux Send 4		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 Auxiliary Stereo Track in Capture. When the StudioLive is not connected to a computer, the S/PDIF output is routable through the System Menu (see Section 5.4). 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.

### Scene Recalling Menu



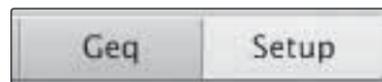
As discussed in Section 5.1, your StudioLive allows you to decide which group of parameters you would like to recall with a Scene. The Scene Recalling Menu on the Setup tab corresponds directly with the Scene Recalling Menu on your StudioLive.

#### 7.2.7 Enabling Lockout Mode

Your StudioLive features a Lockout mode that allows you to temporarily disable nearly every feature on the StudioLive, although analog features (e.g., input-trim knobs, faders, and cue, tape-input, and monitor levels) can still be adjusted.

Because of this, after unlocking your StudioLive and before resuming mixing, you should take a quick glance at your input trims and output levels. If you have locked your fader position, 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.



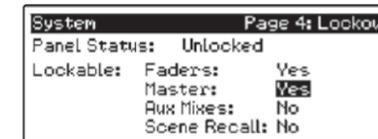
1. With your StudioLive connected and synced to your computer, launch VSL and click on the Setup tab.



2. Click on the Lock Out button.



3. 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 5-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.



4. 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 two StudioLive mixers linked, you only need to lock the Master unit to lock both 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. To lock your StudioLive, press the Select buttons that correspond to the 5-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.*

#### 7.2.8 Remotely Controlling StudioLive with VSL

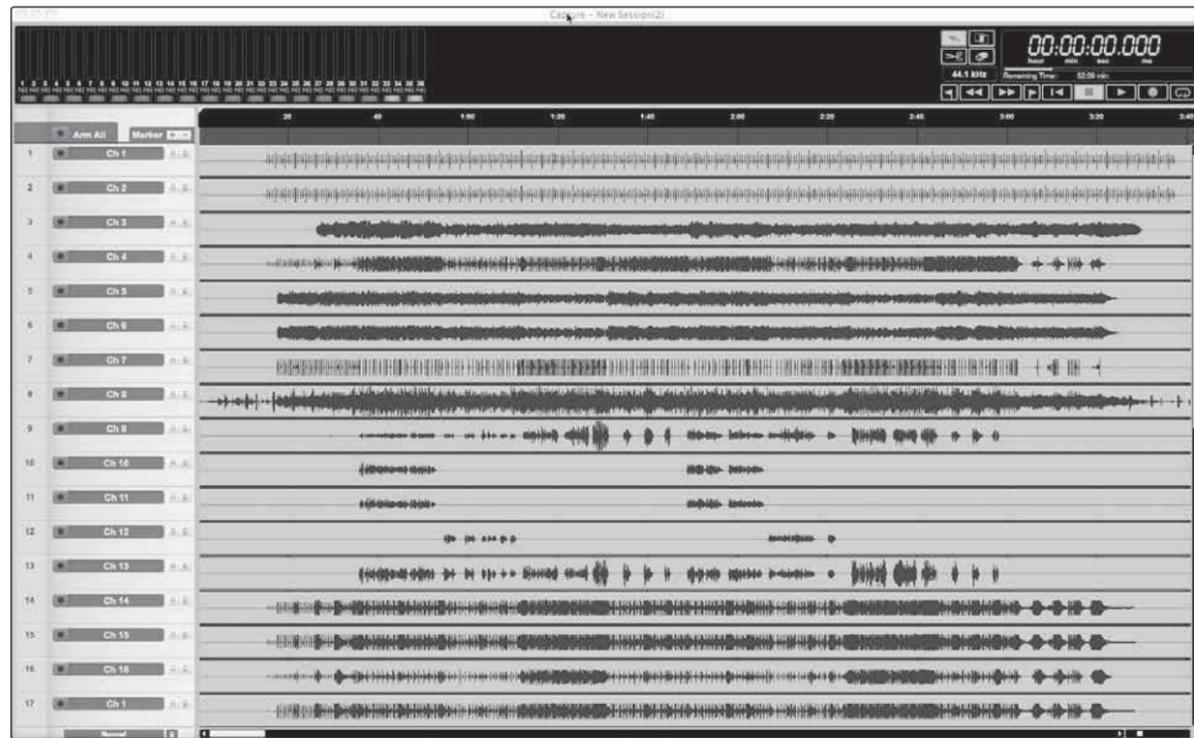
VSL allows you to control almost every parameter in your StudioLive remotely from your desktop or laptop computer. Your StudioLive will reflect every change made in VSL, including fader movements. This means that if you move Channel 12's fader in VSL, you will hear the level set in VSL in your mix rather than the level represented by the current physical position of the Channel 12 fader on your StudioLive.

To sync the faders on your StudioLive with the faders in VSL, simply press the Locate button on your StudioLive. The Fat Channel meters on your StudioLive will reflect the current position of each fader in VSL. While in Locate Mode, the faders on your StudioLive will not be controlling the level of each channel, so you will not hear any level changes while you recall each fader position. Once you exit Locate Mode, your StudioLive faders will reactivate.

Additionally, you can use a wireless device (such as a laptop, iPad, or iPhone) to control VSL. It is important to note that your StudioLive must be connected and synced to a computer with a FireWire 400 connection. Once you establish a wireless network between this computer and your wireless device, you can use a VPN application to remotely control VSL.

Please contact your computer's and/or wireless device's supporting documentation or manufacturer to learn more about VPN and wireless-network creation.

## 7.3 Capture



### 7.3.1 What is Capture?

Capture is an audio-recording application created exclusively for use with PreSonus StudioLive-series mixers. To record from the StudioLive mixer into Capture, you will need to install the StudioLive driver and connect the mixer to a FireWire port on your computer. Once the StudioLive mixer has been connected to your computer, and Capture has been installed, simply launch Capture, and you are ready to record.

This section contains general information, including the basic layout and workflow of Capture and is a great place to start for new audio-recording software users.

#### Versions

There is one version of Capture, which runs under the Windows and Mac OS X operating systems. This manual explains both the Windows and the Mac versions.

#### Tips

Several shortcuts and alternative methods or functions are mentioned in this section. These tips are intended to improve your workflow and will be shown as follows:

#### Really Helpful Information

Also note that if you float the mouse over most tools, buttons, or windows in Capture for a few seconds, a Tooltip will appear that names the function that the tool, button, or window serves.

### Key Commands

Many operations in Capture have associated key commands (keyboard shortcuts) that can be used instead of navigating menus with the mouse. Several of these key commands use modifier keys, some of which differ depending on the operating system (Windows or Mac OS X).

Key commands with modifier keys are shown with the Windows modifier key first, as follows: [Win modifier key]/[Mac modifier key]+[key].

For example, [Ctrl]/[Cmd]+[C] means “press [Ctrl]+C in Windows, or press [Cmd]+C in Mac OS X.”

Where there is no difference between the Windows and Mac version of a key command, only one key command will be displayed; for example: [F3].

### Menu Functions

Many functions can be reached via hierarchical menus. Where menu navigation is referenced, it will appear as follows: Menu/Menu Entry/.../Function.

### Non-Destructive Editing and Undo/Redo

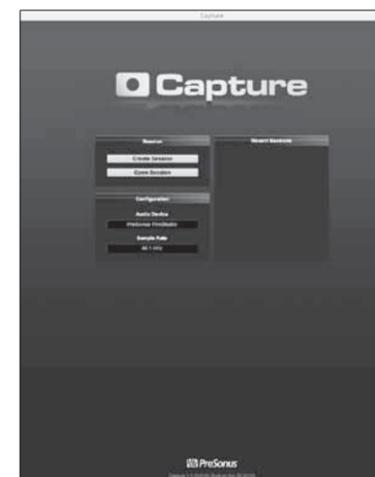
Almost every user action in Capture can be undone and redone. There is no limit to how far back actions can be undone and how far forward undone actions can be redone.

Because of this, feel free to explore without worrying you will permanently alter anything. In fact, just pressing buttons might be the quickest way to learn.

### High Quality Audio

To match your StudioLive's 32-bit floating-point DSP, Capture features a 32-bit floating-point audio engine. An audio engine is the “number cruncher” that does the math required to record and play back digital audio. To put it simply, Capture will process your audio as accurately as current technology allows and maintains the highest sound quality possible. All you have to do is make great music!

### 7.3.2 Start Page



When Capture is launched, by default you will be taken to the Start Page.

#### Create and Open Session

At the upper left of the Start Page are two buttons: Create Session and Open Session. A Session is a document in which you can record and edit audio. Click on the Create Session button to create a new Session or click on the Open Session button to browse for and open an existing Session.

#### Recent Files

To the right of the Start Page menu section, the Recent Files list includes links to the most recently opened documents. Click on any of these links to quickly open the Session.

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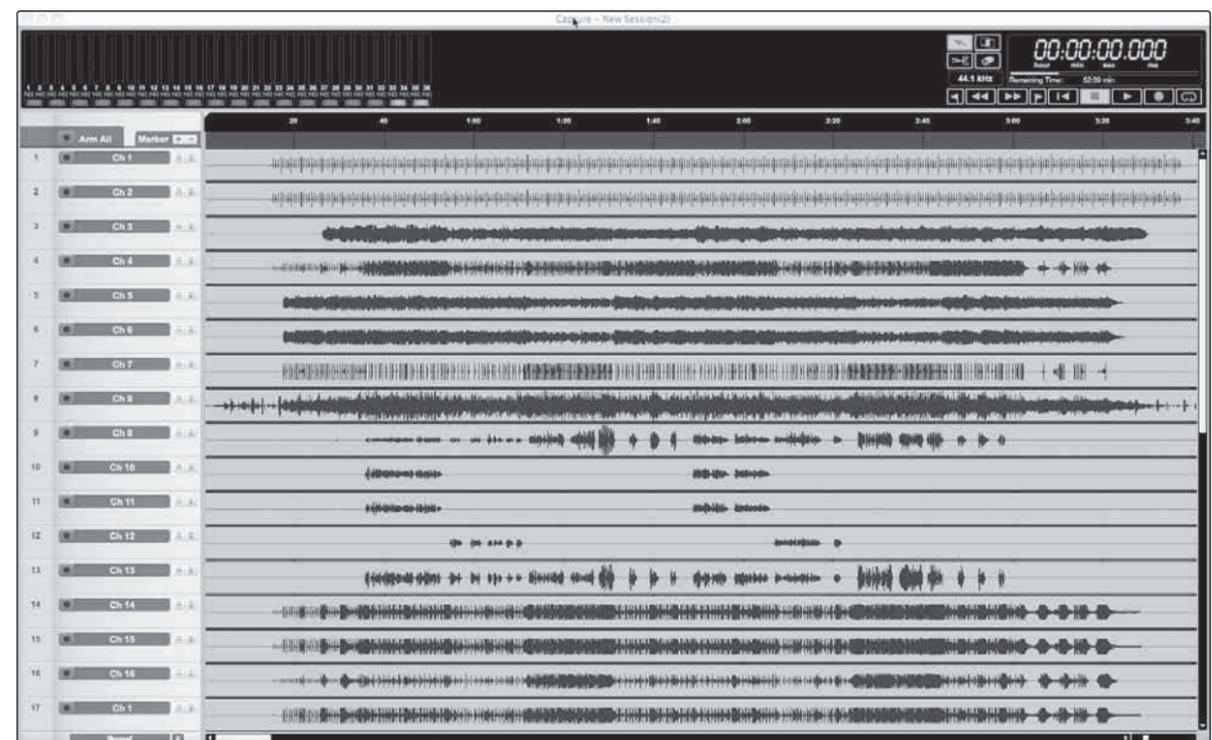
### Audio Device and Sample Rate

Capture was specifically created for use with the PreSonus StudioLive series mixer and will only work with that device. It is not possible to configure any other audio device for use with Capture. The Audio Device window will either display "PreSonus FireStudio" or "No Audio Device"; the former indicates that your StudioLive mixer is connected to the computer correctly and is ready to use with Capture. The Sample Rate window displays the currently configured sample rate, as set on the StudioLive mixer.

### About Capture

To view information about Capture, including your version number and acknowledgements, choose About Capture from the Help file menu.

## 7.3.3 The Session Page



Capture features a single-window user interface so you don't need to manage multiple windows and views. When a new Session is created or an existing Session is opened, you will be taken to the Session page. This page contains all of the necessary tools to record and edit multitrack audio.

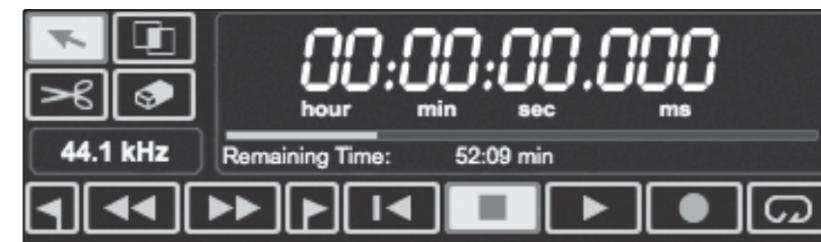
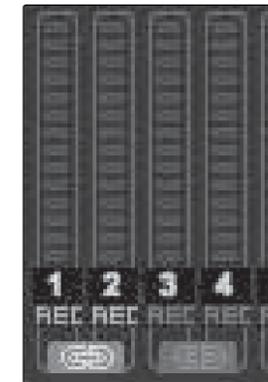
The following are diagrams and descriptions of the Session interface.



### Transport

The Transport is at the top of the Session page and contains:

- **Meter Bridge:** Peak LED-style meters with clip indicators for each input into Capture from your StudioLive mixer, plus an additional pair of meters for the routable Master track. (See Section 7.2.5 for details on how to route a bus to this track.)
- **Link Button:** Between each meter you will find a link button. When this button is active, Capture will record that track as a stereo interleave file. By default, the routable bus pair (25 and 26) is link-enabled.
- **Record Arm Button:** Below each meter, you will find the Record Arm button for that track. When this button is active, Capture will record audio from that input.



### Mouse Tools

- **Arrow Tool:** The default tool for access to most functions.
- **Range Tool:** Select the range of an Audio Event for editing.
- **Splice Tool:** Splice Audio Events.
- **Eraser Tool:** Erase Audio Events.
- **Sample Rate Display:** Displays the current Capture/StudioLive Sample Rate.

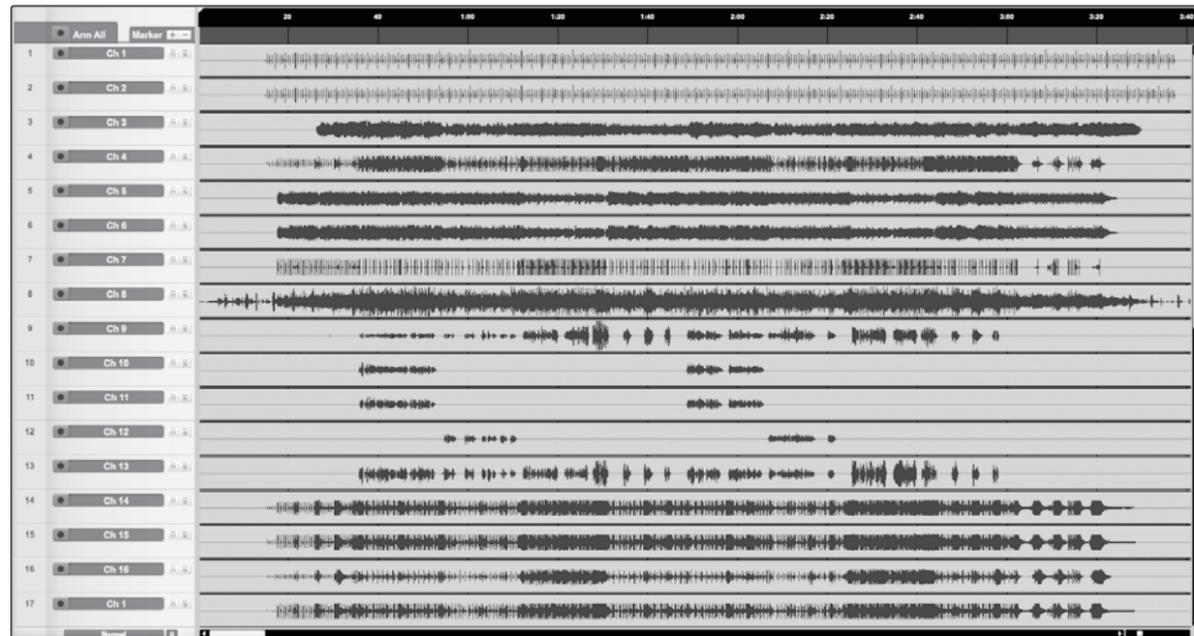
### Transport Controls

- **Previous Marker:** Jump the playback cursor to the previous marker.
- **Rewind:** Rewind for the duration this button is held down.
- **Fast Forward:** Fast Forward for the duration this button is held down.
- **Next Marker:** Jump the playback cursor to the next marker.
- **Back to Beginning:** Return to the beginning of the Session.
- **Stop:** Stop playback.
- **Play:** Start playback at the current playback-cursor position.
- **Record:** Start recording at the current playback-cursor position.

- **Loop:** Engage/Disengage Loop Mode.
- **Time Display:** Displays the time at the current playback-cursor position.
- **Remaining Time:** Displays the remaining time that can be recorded, based on the size of the available storage left on the hard drive to which you are recording.

### 7.3.4 The Edit Window

The Edit window is the main view of the Session page, which provides an overview of the Session for editing.



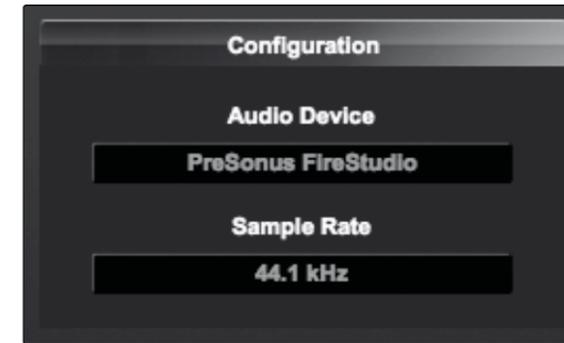
The Edit Window contains the following:

- **Timeline Ruler:** Displays time increments in seconds.
- **Marker Lane:** Displays user-inserted markers.
- **Arm All:** Click this button to arm all inputs for recording.
- **Marker ±:** Click on the plus (+) button to add a marker at the current playback cursor position. Select a marker and click on the minus (-) button to remove the marker.
- **Track Column:** Contains a dedicated mono audio track for each audio input into Capture from the StudioLive mixer, including a Main stereo track.
- **Vertical Zoom:** Increases or decreases the size of the tracks and audio files.
- **Horizontal Zoom:** Zooms the session in or out.

### 7.3.5 Getting Started in Capture

Tracking is the first stage of audio production. The following chapter discusses aspects of recording tracks in Capture, including Session creation and setup, Audio Tracks and Events, and more.

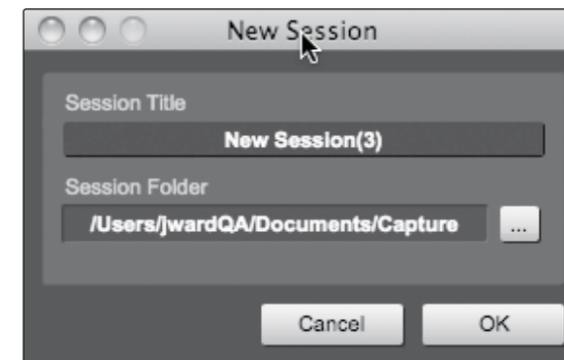
#### Audio Device



Before recording anything in Capture, please take a moment to be sure your StudioLive mixer is properly connected to the computer via FireWire and that Capture displays the StudioLive as your audio device on the Start page.

When the StudioLive mixer is not connected to the computer successfully, Capture will display "No Audio Device" in the Audio Device window.

#### Creating a New Session



A Session is the document type in which all recording takes place in Capture. To create a new Session, do one of the following:

1. From the Start page, click on the New Session button.
2. Navigate to File/New Session.
3. Press [Ctrl]/[Cmd]+N on the keyboard.

#### Session Setup

When a new Session is created, a setup menu with the following options will pop up:

##### Session Title

This will be the title of your Session. This name will be the Session file name (Title.Session), as well as the name of the new folder that contains all data related to your Session.

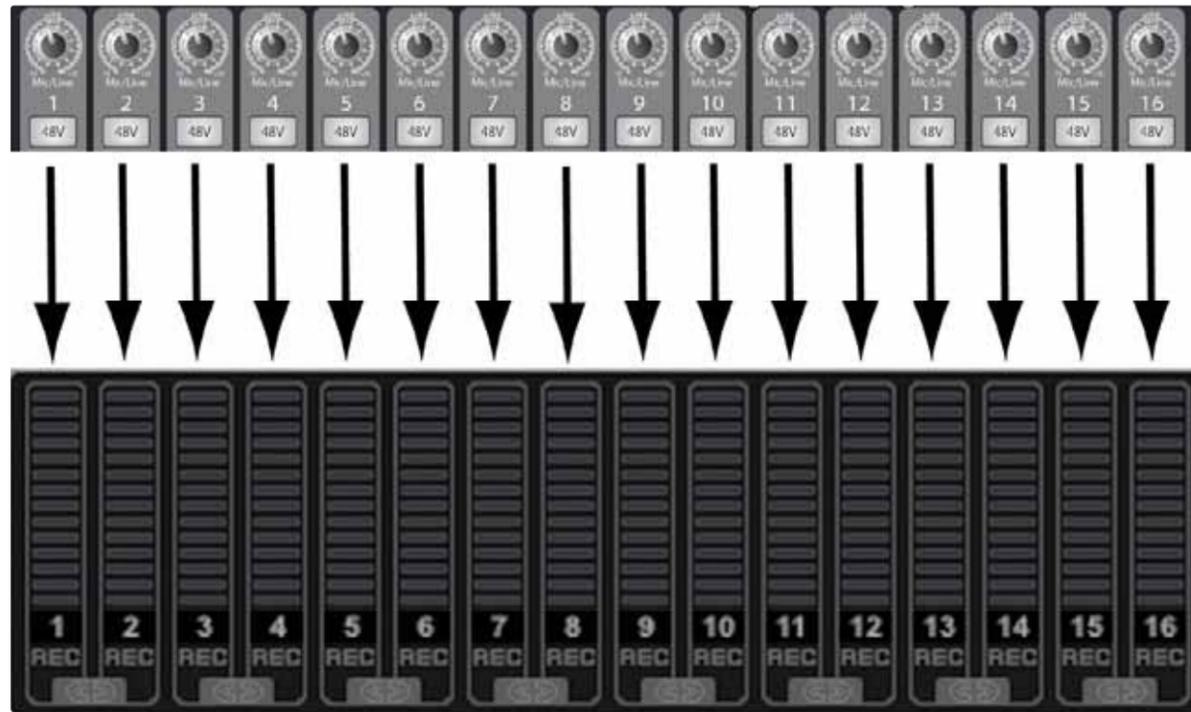
##### Session Location

This is where new Sessions and all related data will be saved. The Session location can be chosen each time you create a Session. By default, the new Session location will be the Capture folder in your Documents folder. You can choose a different file location by clicking on the Browse button and browsing to a new location. The last known save location will appear as the default the next time you create a new Session.

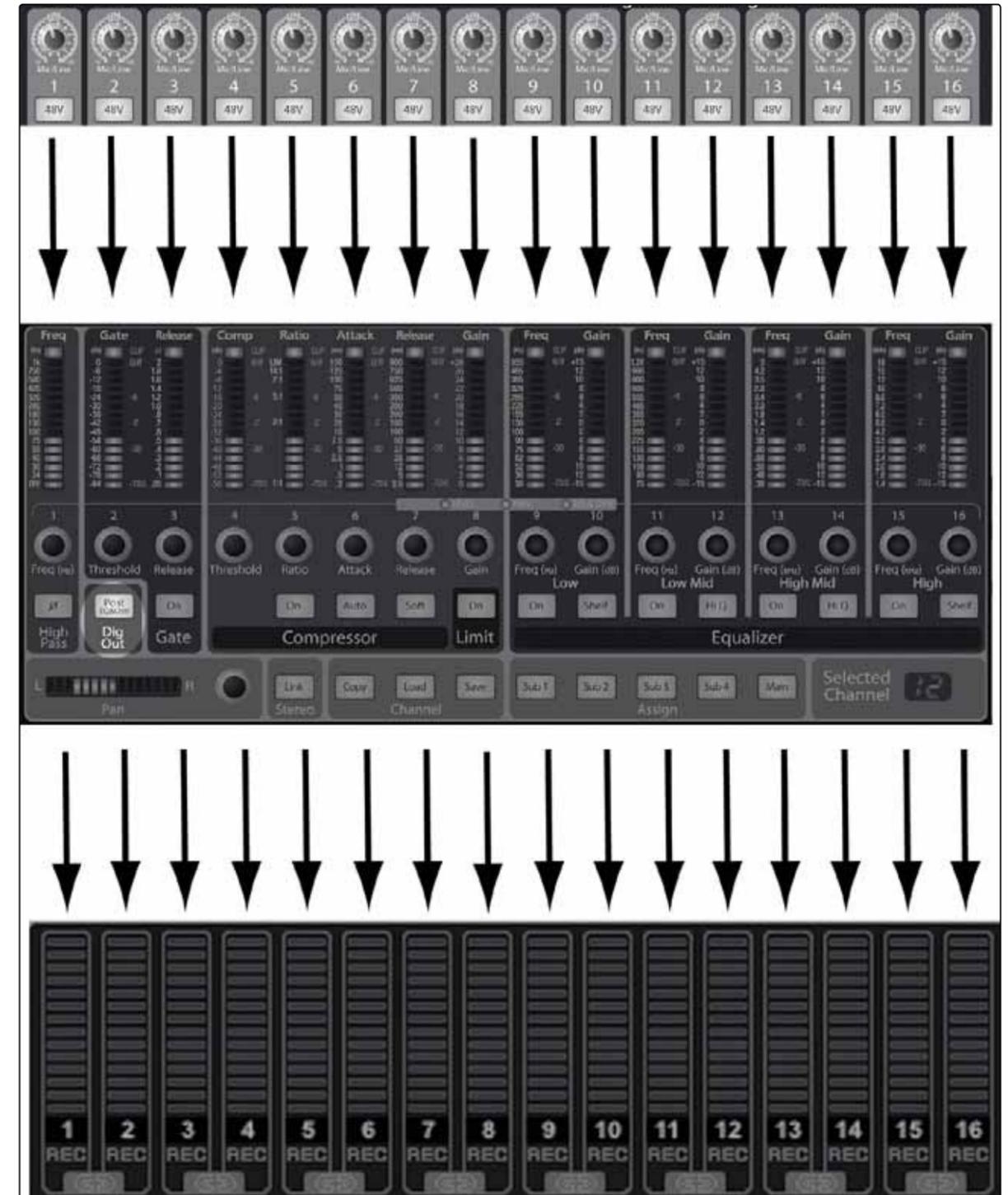
### Capture Audio I/O Setup

There is no need for audio inputs and outputs to be set up in Capture, as it automatically detects which type of StudioLive mixer (16.4.2 or 24.4.2) is connected and self-configures. Each input from your StudioLive mixer is represented with a mono input track in the Track column of the Session, and each input track has a corresponding level meter in the meter bridge.

The following diagram illustrates the one-to-one relationship of the StudioLive and Capture input channels. The input signal into each channel on the StudioLive mixer is automatically routed to each respective input channel in Capture.



It is possible to process each StudioLive mixer input channel with the Fat Channel before the input signal is routed to Capture. To do this, you will need to engage the Post button in the Dig Out section of the Fat Channel for each channel in your StudioLive mixer. This routes the signal post-EQ and post-dynamics processing. The following diagram illustrates the signal flow in this scenario (below).



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### Capture Main Source Selection

The Auxiliary Stereo Track in Capture is a stereo channel intended to be used to record a stereo mix from your StudioLive. This mix can be recorded while recording the 24 individual input channels, so a mix is available immediately after a live show, or you can use this channel to route your final studio mixes through the StudioLive.

As discussed in Section 7.2.6, VSL features an Auxiliary Inputs Router that allows you select which buses you want to record in addition to your input channels. Auxiliary Inputs 25 and 26 are dedicated for recording a stereo mix in Capture. However, in addition to the Main L/R mix, you can selection from any of the following outputs, buses, and inputs: Sub 1/2, Sub 3/4, Aux 1/2, Aux 3/4, Aux 5/6, EFX Send A, EFX Send B, Aux Return A, Aux Return B, 2 Track, Talkback, EFX Return A, EFX Return B, and the Solo Bus.

## 7.3.6 Recording in Capture

### Record-Enabling an Audio Track

To record to an audio track, the track must be record-enabled, or *armed*. Capture provides two Record Enable buttons for each input: one on the track and on the Meter Bridge.

To record-enable an audio track, click on either the Record Enable button on the track or on the Meter Bridge. Both buttons will turn entirely red in color when either button is enabled, and the track's corresponding meter in the meter bridge should begin to move up and down if there is live audio on that channel in the StudioLive mixer.

Click on the Arm All button at the top of the Track Column to record-enable every track at once.

Once an audio track is record enabled, you are ready to record audio to that track. Refer to the Activating Recording section for more on this topic.

### Monitoring Live Audio Input

All monitoring of live audio input should happen on the StudioLive mixer itself as normal. There is no separate monitoring capability within Capture.

### Setting Input Levels

Setting the proper input levels is critical to making a good live mix and recording. The basic idea is to set the input gain on your StudioLive mixer as high as possible without overloading the input. Overloading the input will cause clipping (digital distortion), which is particularly unpleasant and will ruin the recording. This damage cannot be undone in software. There is a clip indicator for each input on the StudioLive mixer for this purpose.

If an input channel is not clipping on your StudioLive mixer, you can be sure that there it will not clip in Capture. Refer to the Section 8.7 for a Level Setting Tutorial.

### Activating Recording in Capture



Once you have the desired tracks record-enabled, the next step is to record. To activate recording, click on the Record button in the Transport. The Record button will turn red, and the Play button will turn green in the Transport; the Playback Cursor will start to scroll from left to right from its current position; and a new Audio Event will be recorded to any record enabled tracks.

Recording will continue until you manually stop it by clicking on the Stop button in the Transport, or press [Spacebar] on the keyboard.

Capture utilizes the Broadcast Wave file format for recording. This is the only format utilized, as it is the most widely used format, and it contains timestamps that mark when recordings start within a Song.

When recorded Broadcast Wave audio files get bigger than 4 GB, the RF64 file format is automatically used as the standard file format.

The recommended file system for the recording partition on your computer is NTFS on Windows and HFS+ on Mac OS X.

After recording, the next step in production is usually editing. In the editing process, recorded events are manipulated in many ways to achieve a desired sound.

## 7.3.7 Editing in Capture

All audio that exists within the timeline of your Session will be visually represented by Audio Events, which contain waveform representations of the recorded audio. Audio Events can be moved and split, among other possibilities.

### Mouse Tools



Mouse Tools allow direct interaction with Audio Events using the mouse. It is helpful to remember that actions done using the Mouse Tools can be undone at any time, so you can feel free to explore their functions without permanently altering anything. The following Mouse Tools and related functions are available.

### Arrow Tool



This is the default selected tool. Click on the Arrow Tool button or press [NumPad 1] on the keyboard to select the Arrow Tool. The following describes various possible functions using the Arrow Tool.

#### Move an Audio Event

To move an event using the Arrow Tool, click anywhere on the event and drag left, right, up, or down. Dragging the event left or right will move the event backward and forward in time, relative to the Timeline zoom.

Dragging the event up or down will move the Audio Event to the adjacent Audio Track.

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### Size an Audio Event

Events can be thought of as windows into audio files, where what you see is what you will hear. Sizing is a fundamental technique wherein Audio Events are made shorter or longer, so that only a portion of the audio they contain is seen, and therefore heard.

To size any Audio Event using the Arrow Tool, float the mouse to the left or right edge of the event to reveal the Sizing Tool. When this tool appears, click and drag left or right to size the event. Audio Events can be sized and resized nondestructively any number of times.



The pictures above illustrate an Audio Event before and after it is sized.

### Select Multiple Audio Events

Multiple Audio Events can be selected at once in order to edit them all at once with a single action. To select multiple Audio Events with the Arrow Tool, do one of the following:

- Click outside of the range of an Audio Event, and then drag over any other event(s); a gray box will be drawn while you drag over the target selection area. Release the click once the box is drawn over all of the events you wish to select, and these events will now all be selected and can be edited at once.
- Click on any event, and then while holding [Ctrl]/[Cmd] on the keyboard click on any other event(s) to select them. This allows you to select multiple events that are not close in proximity to each other. All selected events can now be edited at once.

### Range Tool



The Range Tool is used to select a range, or area, within Audio Events. Click on the Range Tool button or press [NumPad 3] on the keyboard to select the Range Tool.

To select a range within events, using the Range Tool, click and drag over the area to be selected; a gray box will be drawn over the target selection area. Release the click once the box is drawn over the range of the events you wish to select. The range of the events you have selected is now treated as a single consolidated event.

- A common use of the Range Tool is to quickly select and delete a range of audio within events, rather than using the Split Tool

### Splice Tool



to make two splits, then selecting and deleting the section with the Arrow Tool. This process is shown in the pictures below.

- Once a range has been selected, the Range Tool will automatically become the Arrow Tool when the mouse cursor is floated over the selected range. This makes quickly selecting and editing any range of an event very simple.
- The Range Tool can be temporarily toggled when the Arrow Tool is selected by pressing and holding [Ctrl]/[Cmd] on the keyboard.

Using the Splice Tool, single Audio Events can be split into multiple Audio Events. Click on the Split Tool button, or press [NumPad 4] on the keyboard to select the Split Tool.

Click on any Audio Event with the Split Tool to split the event at that position.

Notice that the split will occur at the leftmost edge of the Split tool icon (the scissors). By splitting a single event, you essentially create two events that can be edited independently. If multiple events are selected across multiple tracks, the Split Tool will affect all of the selected events in the same way.

### Eraser Tool



The Eraser Tool is used to delete an Audio Event. Click on the Eraser Tool button or press [NumPad 5] on the keyboard to select the Eraser Tool.

- To delete any event using the Eraser Tool, simply click on the event. The Eraser Tool is unaffected by the current selection and will only affect the event that is directly clicked on.

### Common Editing Actions

As with most other software applications, Capture supports basic Cut, Copy, and Paste actions. Once a selection of events, or a range of events, has been made, these actions can be performed as follows:

- **Cut:** Press [Ctrl]/[Cmd]+X on the keyboard to cut the current selection.
- **Copy:** Press [Ctrl]/[Cmd]+C on the keyboard to copy the current selection.
- **Paste:** Once a selection is cut or copied, press [Ctrl]/[Cmd]+V on the keyboard to paste the selection. The selection will be pasted to the current Playback Cursor position, or to the beginning of the track from which the selection originated if the Playback Cursor is not currently set anywhere.

## Suggestions

### Listen While Editing

In audio production, the editing process can be particularly unforgiving. Small inaccuracies when splitting, moving, or performing other actions on recorded audio can lead to unwanted results.

As simple as it may seem, the act of listening while editing is often overlooked. For instance, when sizing the edges of a vocal part to remove unwanted sounds between words, it is tempting to make the edits based on the visual representation of the waveform. While this may work sometimes, it is a much better idea to listen as you size the events to be sure you are not removing any critical part of the vocals.

Listening to your edits as you make them will save time and frustration in nearly every case.

## 7.3.8 Capture Session Navigation

The ability to quickly navigate to certain areas of your Session is highly useful during the mixing process. The following describes the ways in which you can quickly navigate throughout your Session.

### Scrolling

To scroll left and right through time in your Session, click and drag the horizontal scroll bar near the bottom of the Edit window.

### Timeline Zooming

You can zoom in and out of your Session with respect to time in the Edit window, so that you can look at the entire length of the Session or just view a small region of time to make accurate edits. To zoom, do one of the following:

Click and drag left or right on the Zoom scroll bar in the lower right-hand corner of the Edit window to zoom in or out.

Press [W] on the keyboard to zoom out slightly. Press [E] on the keyboard to zoom in slightly.

Click inside the Timeline ruler at any point in time and drag down or up to zoom in or out.

## Using Markers

In Capture, the Marker Lane is used to place markers at desired places in the Timeline, after which navigation to the markers is easy. Markers are also helpful when exporting your Session to individual mixes or to work in another recording application.

The Marker Lane is always visible, just below the Timeline.



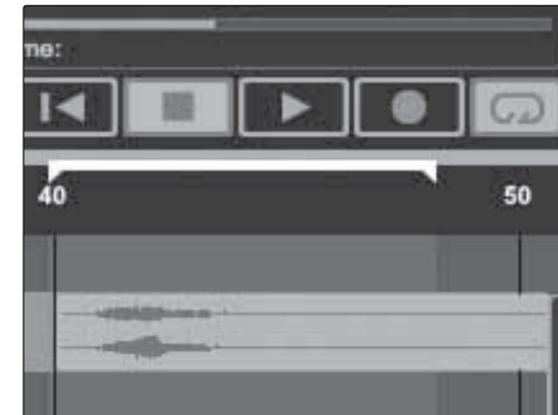
## Inserting Markers

To insert a new Marker into the Marker Lane, with playback running or stopped, click on the Add Marker button, or press [Insert] on the keyboard. Each new marker will be numbered sequentially by default (#1, #2, #3...). To rename a marker, double-click on the marker in the Marker Lane, type in a new name, and then press [Enter] on the keyboard.

## Navigating Markers

It is possible to quickly jump the Playback Cursor between markers in the Marker Lane. Click on the Previous Marker button in the Transport, or press [B] on the keyboard, to jump to the previous marker. Click on the Next Marker button in the Transport, or press [N] on the keyboard, to jump to the next marker. Jumping to markers during playback allows quick comparisons between different sections of your Session.

## Looping



To Loop the playback of any range of your Capture session, do one of the following:

- Select an Event with the Arrow Tool, or select a Range with the Range Tool, that you wish to loop. Press [P] on the keyboard to set the loop range to your selection. The range will be indicated by a bar drawn in the timeline, with a left and right flag on either end.
- Mouse over the gray line above the time line. You will notice that your cursor changes to a pencil tool and you can draw your loop range.

Once your loop range is set, you can click on the Loop button in the Transport, or press [L] or [/] on the keyboard to engage Loop mode. Start playback before the end of the Loop range, and when the end of the range is reached, playback will jump back to the beginning of the Loop range automatically for as long as Loop is engaged.

Engaging Record in the Transport will disable Loop Mode and begin recording at the current playback cursor position. It is not possible to engage Loop while recording.

## 7.3.9 Mixing in Capture

Mixing is the part of the production process where all recorded tracks are combined to mono or stereo. The goal is to balance relative volume, frequency, and dynamic content in order to achieve a desired cohesive sound.

### Mixing with Capture and StudioLive

No mixing is done in Capture directly. Instead, the individual outputs of each audio track in Capture are hardwired to the FireWire returns for

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each corresponding input channel on the StudioLive mixer. Thus, with the FireWire returns engaged for each input channel on your StudioLive mixer, the output of your multitrack Capture Session is routed to the StudioLive mixer, where it can be mixed just like any other audio input.

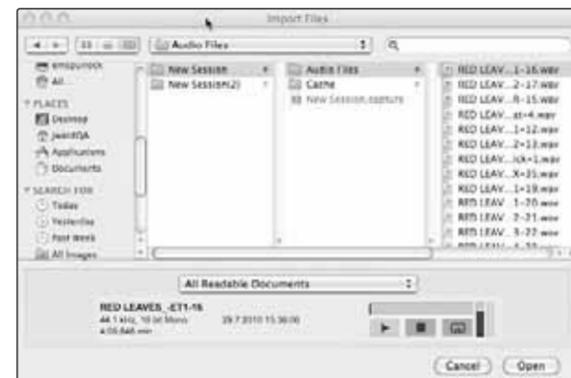
### Creating a Mix

The process of creating a mix with Capture and the StudioLive mixer is quite simple. You play back your recorded tracks through the StudioLive and record the main output of the StudioLive to the Auxiliary Stereo Track in Capture. The process works as follows:

1. In VSL, select "Main L/R" for Auxiliary Inputs 25 and 26.
2. Be sure Record Arm is disabled for all input Audio Tracks in Capture, and the FireWire returns for all input channels on the StudioLive mixer are engaged.
3. In Capture, select the Auxiliary Stereo Track (25 and 26) and enable Record Arm.
4. Bring the playback cursor position to the beginning of the Session by clicking the Back to Beginning button in the Transport.
5. Activate recording by clicking on the Record button in the Transport; the playback cursor should begin to move from left to right, and audio should be playing through each input on the StudioLive mixer. The Auxiliary Stereo Track in Capture should be recording a new Audio Event, which is the main mix from the StudioLive mixer.

The new Audio Event that has been recorded from the StudioLive mixer will be stored as a stereo WAV file at the sample rate to which the StudioLive mixer is set.

### Importing Audio Files



It is possible to import WAV and AIFF audio files into your Session. To import an audio file, select Import Audio File from the Session menu to open the Import File menu. Browse to the desired file and click on it to select it. Then click on Open to import the file into your Session.

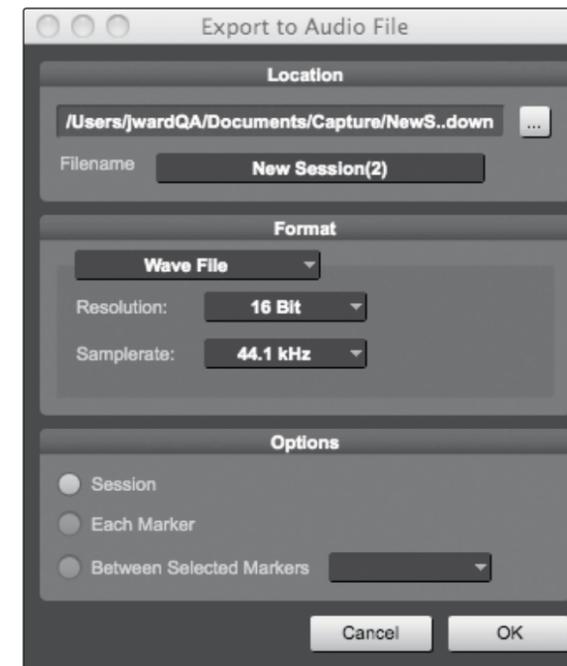
An Audio Event for the imported file will be created and placed on the currently selected track in your Session, at the current playback cursor position.

### Preview Player



When browsing for files in the Import File menu, it is possible to listen to the audio files as you are browsing in the Preview Player. Click the Play button in the Preview Player to play the currently selected audio file. Click the Stop button to stop playback. Click the Loop button to loop the playback of the currently selected audio file.

### Export Audio Files



To export audio from your Session in Capture, navigate to Session/Export to Audio File, or press [Ctrl]/[Cmd]+E on the keyboard to open the Export to Audio File menu.

#### Location

The top section of the Export to Audio File menu is where you will select a location and name for the audio file. Click on the Location button to choose a file location. Double-click on the filename, type in a new name, and then press Enter to choose a name for the file.

#### Format

Select the format of the mixdown audio file in the middle section of the Export to Audio File menu. Choose from Wave (WAV) or AIFF and then choose the desired Resolution and Sample Rate.

If you wish to put your audio on a standard audio CD, the format should be a Wave file with 16-bit resolution and a 44.1 kHz sample rate.

#### Options

The bottom section of the Export to Audio File menu has several options that will affect how the file(s) is created.

Choose from Export Session, Each Marker, or Between Selected Markers.

- The Export Session option will export the entire range of your Session, which will be to the furthest point in time that any Audio Event on a track extends.
- The Export Each Marker option will export separate files for each range between the markers in the Marker Lane.
- The Export Between Selected Markers option will export an audio file(s) between the range of any two selected markers in the Marker Lane.

### Export Your Final Mix to an Audio File

The most common medium to which recordings are physically published is the audio CD. In order to put your final mix on an audio CD, the mix file must be a 16-bit, 44.1 kHz WAV file. Once you have recorded your final mix to the Auxiliary Stereo track in Capture, it is recommended you export the audio from the Auxiliary Stereo track to an audio file for this purpose.

Once the correct audio file for your mix has been exported, you may use any CD-burning application to put that mix on a recordable audio CD.

### Mixing a Capture Session in Studio One

Many users will want to use Capture strictly as a track-recording tool and mix the recorded tracks later in a DAW application. PreSonus has included a copy of Studio One Artist with your StudioLive for just this purpose. Both Studio One Artist and Studio One Pro can open Capture Session files. All markers, edits, track names, etc will be preserved, no further effort is required; simply launch Studio One and open your Capture session. (See Section 7.4 for more details.)

### Mixing a Capture Session in a Different Recording Application

For users who wish to mix their Capture Session in a DAW other than Studio One, there are several ways in which to import your Capture session, as described in the following.

#### Save a Capture Session as Open TL

Many audio-recording programs provide support for opening a document type called Open TL (Open Track List). An Open TL file provides a reference for all of the audio tracks and Audio Events, with their corresponding positions, in your Capture Session, so that another audio program can reconstruct your Session automatically. This makes it possible for you to open your Capture Session and work as you normally do in your favorite recording application.

To save an Open TL version of your Session, select Save Session As in the File menu of Capture. Name the file as usual, and choose Open TL (\*.tl) in the Save As Type selection box.

Click on Save to save your Session as an Open TL document and confirm the command by clicking Yes in the confirmation window. This file will be placed by default in your Session folder and will not overwrite the original Capture Session file. Note that the Open TL format will not save your markers.

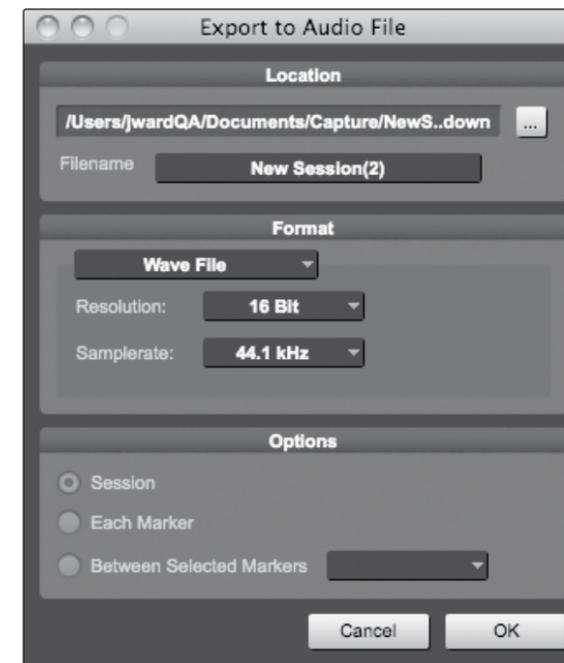
### Import Capture Session Audio Files Directly

Some audio-editing applications (e.g., digital audio workstations, or DAWs) do not support the Open TL format, which makes the options somewhat limited when importing sessions created in other programs. In this case, you must manually import the Audio Events from your Capture Session into the audio-editing application, using whatever means possible in that program. Refer to the documentation of your software for more information on this process.

Note that in some cases, you may have more than one Audio Event on a given track, as opposed to a single continuous Audio Event or have audio events that do not have the same start position. In either case, it is highly recommended that you first export the audio for each track before attempting to import into an audio-editing program, as described in the Export Audio Files section of this section.

You will want to be sure to have the Export Tracks option selected in the Export Audio File menu as shown, so that the result of the process will be a single continuous audio file for each track in the session.

Once you have single continuous audio files for each track, directly importing the files into an audio editor is much easier. All that is required to reconstruct your Capture Session is to align each file at the same start time.



7.3.10 Capture Key Commands

Key Commands	Key Command
<b>Action</b>	
<b>File</b>	
New Session	Cntrl/Cntrl/Cmd+N
Open Session	Cntrl/Cntrl/Cmd+O
Close Session	Cntrl/Cmd+W
Save Session	Cntrl/Cmd+S
Save Session As	Cntrl/Cmd+Shift+S
Quit	Cntrl/Cmd+Q
<b>Edit</b>	
Undo	Cntrl/Cmd+Z
Redo	Cntrl/Cmd+Y
Cut	Cntrl/Cmd+X
Copy	Cntrl/Cmd+C
Paste	Cntrl/Cmd+V
Delete	Del
Select All	Cntrl/Cmd+A
Deselect All	Cntrl/Cmd+D
Zoom In	Cntrl/Cmd + or E
Zoom Out	Cntrl/Cmd - or W
Zoom Full	F
<b>Session</b>	
Import File	Command+I
Export to Audio File	Command+E
Insert Marker	Ins, I
Insert Named Marker	Shift+Ins, Shift+I
Locate Next Marker	N
Locate Previous Marker	B
Recall Marker 1-9	NumPad+1-9, Command+1-9
Mute Track	M
Solo Track	S
Arm Track	A
<b>Tools</b>	
Select Arrow Tool	1
Select Range Tool	2
Select Split Tool	3
Select Eraser Tool	4
<b>Transport</b>	
Toggle Start/Stop	Space Bar
Start	Enter
Record	NumPad *
Stop	NumPad 0
Return to Zero	NumPad .

Key Commands ...continued	Key Command
<b>Action</b>	
<b>Navigation</b>	
Focus Next	Tab
Focus Previous	Shift+Tab
Left	Left Arrow
Extend Selection Left	Shift+Left Arrow
Extend Selection Left Add	Cntrl/Cmd+Shift+Left Arrow
Skip Left	Cntrl/Cmd+Left Arrow
Right	Right Arrow
Extend Selection Right	Shift+Right Arrow
Extend Selection Right Add	Cntrl/Cmd+Shift+Right Arrow
Skip Right	Cntrl/Cmd+Right Arrow
Up	Up Arrow
Extend Selection Up	Shift+Up Arrow
Extend Selection Up Add	Cntrl/Cmd+Shift+Up Arrow
Skip Up	Cntrl/Cmd+Up Arrow
Down	Down Arrow
Extend Selection Down	Shift+Down Arrow
Extend Selection Down Add	Cntrl/Cmd+Shift+Down Arrow
Skip Down	Cntrl/Cmd+Down Arrow
Page Up	Page Up
Extend Selection Page Up	Shift+Page Up
Extend Selection Page Up Add	Cntrl/Cmd+Shift+Page Up
Skip Page Up	Cntrl/Cmd+Page Up
Page Down	Page Down
Extend Selection Page Down	Shift+Page Down
Extend Selection Page Down Add	Cntrl/Cmd+Shift+Page Down
Skip Page Down	Cntrl/Cmd+Page Down
Start	Home
Extend Selection Start	Shift+Home
Extend Selection Start Add	Cntrl/Cmd+Shift+Home
Skip Start	Cntrl/Cmd+Home
End	End
Extend Selection End	Shift+End
Extend Selection End Add	Cntrl/Cmd+Shift+End
Skip End	Cntrl/Cmd+End



### 7.3.11 Capture Applications and Scenarios

When recording live, most people usually make a single, continuous multitrack recording. This involves record-enabling Audio Tracks in Capture, triggering recording, and continuing to record until the performance is done. The following describes aspects of the live-recording workflow using the StudioLive mixer and Capture.

#### StudioLive Mixing and Recording a Live Performance

The StudioLive mixer is capable of mixing your live show with stunning flexibility and ease, while Capture records the show to your computer. This is possible because each input channel on the StudioLive mixer has a hardwired output that continuously streams audio over FireWire to your computer, and Capture is designed to receive and record this stream.

Once the StudioLive mixer has been set up to mix a live performance, you will need to decide, individually for each StudioLive channel, whether to add pre- or post-Fat Channel dynamics processing and EQ to the signal before it is sent to the computer. To switch between pre/post processing, press the Post button (Dig Out subsection) in the Fat Channel section for each input channel on the StudioLive mixer. If the Post button is illuminated, the channel's signal will be sent to your computer after it is processed in the Fat Channel; if the Post button is not illuminated, the signal will be sent without Fat Channel EQ and dynamics processing.

After setting the input channels on the StudioLive mixer for pre/post processing, the next step is to launch Capture and create a new Session. Once in the new Session, record-enable each Audio Track you would like to use, or just record-enable all of the Audio Tracks by clicking on the Arm All button at the top of the Track column. If audio is streaming in from the StudioLive and the Audio Tracks are record-enabled, you should see each track's input level displayed in the track's level meter in the meter bridge.

You should set input levels for each input channel on the StudioLive mixer while preparing to mix the live performance. If the hardware inputs on the StudioLive mixer are not clipping, you do not need to worry about clipping in Capture.

Once the desired tracks are record-enabled, and you have verified input levels in Capture's meter bridge, you are ready to begin recording.

#### Worry-Free Live Recording

Many people are concerned when a computer is involved while mixing a live show. While Capture has been engineered to be remarkably stable, there are many variables on any given computer that are not controlled by Capture.

This is why the StudioLive mixer has been designed to be completely independent of any computer to which it is connected. The FireWire connection to the computer simply allows the computer to receive audio outputs that are completely independent from all other channels in the StudioLive mixer. Even if the computer

blows up during the live show, your live mix will be completely unaffected. You can mix and record a live performance using the StudioLive mixer and Capture with complete confidence.

#### Insert Markers While Recording

It is highly recommended that markers be inserted during the recording of a live performance, as the markers will make the rest of the production process much easier. Each of the examples that follow begins with the same scenario:

You are mixing and recording a live show using the StudioLive mixer and Capture. You start recording before the start of the actual performance, and you will not stop recording until the end of the performance. During the performance, you insert a marker during the break between each song of the ten-song set. At the end of the set, you stop recording and save the Session. You are then ready to shut down your system. (Remember to shut down in the proper sequence: first turn the master level down on the StudioLive mixer, then turn off power amps and speakers, shut down the computer, and power down the StudioLive, in that order.)

##### Basic Example

The band you recorded the previous night comes over the next day to check out the recordings. You launch Capture, open the Session, and begin playback through the StudioLive mixer. After half a minute of tense silence while the band wonders what went wrong, you realize that the band did not start the actual performance until 20 minutes after you started recording because the lead singer took so long to get on stage. (It is, of course, traditional to blame the lead singer!)

You click on the Next Marker button in the Capture Transport to quickly advance to the next marker, which is the first song of the set, and the band perks up. The band thinks they played terribly during the first song, and they want to skip to the next song to see if they did any better. You click on the Next Marker button again, and the next song starts immediately. The band listens to a bit of each song, and you skip through the whole set within a few minutes as if you were skipping through finished tracks on a CD.

The band is so impressed with the quality of the recording and your production skills that they hire you on the spot to mix the show and turn it into a live-performance album they can release to their adoring fans.

##### Advanced Example

The day after the gig, you prepare to mix the previous night's live recording in your personal studio. You realize that each song will probably be mixed in a different way. Wouldn't it be convenient if you could break up the original, continuous recording of the entire set of ten songs so you could work on one song at a time? With Capture, you can use the markers you inserted between each song to automatically generate all the multitrack files for each individual song.

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To do this, first open the original Session. If you have not already done so, rename each marker to the desired name for each song. Then, select Export to Audio File from the Session menu. In the Export to Audio File menu's Options section, select Export Tracks and Each Marker, and set the Format to the desired file-format settings. Then click OK.

Capture will then automatically generate separate audio files for each track in each song. The names and files will be based on the name you specified in the Export to Audio File menu, the names of the markers, and the track names in your Session (Export Name-Track Name-Marker Name). You can then import the files for each song into a new Capture Session, or into a different audio recording application, to be mixed with the StudioLive mixer.

A few days after the original performance, you have finished producing a ten-song live album for the band you recorded, and they could not be more impressed.

The studio-recording process often appears very different from the live-recording process. However, the StudioLive mixer and Capture are equally at home in the studio and in concert. The following describes aspects of the studio-recording workflow using the StudioLive mixer and Capture.

### Overdubbing

In a studio-recording session, as opposed to recording live, you might wish to record on only a few tracks, or on a single track at a time, and then record more later. This process of adding to your recorded tracks later is called "overdubbing." Using Capture and the StudioLive mixer, you can play recorded audio on any tracks through the StudioLive mixer and simultaneously record new audio to tracks in Capture.

To do this, it is a good idea to first establish a good mix to listen to while recording. Just as when creating monitor mixes for a live performance with the StudioLive mixer, you can create a number of mixes to listen to. For instance, you might create a stereo mix of the playback from Capture on Auxes 1 and 2 for a guitarist who is about to overdub a lead guitar part. To accomplish this, do the following:

For all input channels on the StudioLive mixer on which you wish to play tracks from Capture, engage the FireWire return by pressing each channel's FireWire Input Select switch (near the top of the channel strip) so that it illuminates. This will route the output of the tracks in Capture to their respective input channels on the StudioLive mixer.

Start playback in Capture. The output from each track in Capture will play through their respective input channels on the StudioLive mixer (Capture Track 1 on StudioLive Channel 1, etc.).

At this point, link Aux 1 and 2 on the StudioLive mixer and create a mix for the linked Aux1/2 by adjusting the level for each input channel that has playback coming from Capture. This is a stereo aux mix, so you can also access the pan for each input channel in the mix by pressing the Mix/Pan button for Aux 2. (For details on how to set aux mixes, see Section 2.7 of the StudioLive manual.)

Now you can plug the guitar or guitar-amp mic into an input channel on the StudioLive mixer and set the input level as you normally would. Then add the guitar-input channel to the Aux 1/2 mix that the guitarist will listen to so he can hear himself.

If you record-enable the Audio Track that corresponds to the guitar input channel in Capture, you should see that track's level meter showing live input in the meter bridge.

With the Aux 1/2 mix set so that the guitarist can hear his live guitar-playing and the tracks from Capture, you are ready to record the lead guitar. You can position the playback cursor to any point in the Timeline so that recording starts there or just start recording from the beginning of the Session. Once you have the lead-guitar part recorded, you might find a few things you would like to correct in the part. This is where punching, as described in the next section, comes in handy.

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## 7.4 Studio One Artist Quick Start



Your StudioLive 24.4.2 comes with Studio One Artist recording and production software. Whether you are about to record your first album or your fiftieth, Studio One Artist provides you with all of the tools necessary to capture and mix a great performance. As a valued PreSonus customer, you are also eligible for an upgrade discount for Studio One Pro when you are ready to master your work, create a digital version for the Web, or incorporate third-party VST plug-ins into your recording process. For more details on the Studio One Pro upgrade program for PreSonus customers, please visit [www.presonus.com](http://www.presonus.com).

Studio One Artist was designed for easy integration with both the StudioLive and Capture. Sessions recorded in Capture can be opened directly in Studio One; no exporting or importing is required. Any markers, edits, and so on that are made in Capture will be preserved when the Capture session is opened in Studio One.

### 7.4.1 Installation and Authorization

Once you have installed the StudioLive drivers and connected your StudioLive to your computer, you can use the PreSonus Studio One Artist music-production software included with your StudioLive to begin recording, mixing, and producing your music. To install Studio One Artist, insert your installation disc into your computer's DVD drive. Follow the onscreen instructions to complete the installation process.

### Running the Studio One Installer



To install Studio One Artist, insert your Studio One Artist installation DVD into your computer's DVD drive.

- **Windows Users:** Launch the Studio One Artist installer and follow the onscreen instructions.
- **Mac Users:** Drag the Studio One Artist application into the Applications folder on your Macintosh HD.

### Creating a User Account



- After installing Studio One Artist, launch the program, and the Activate Studio One menu will appear. If you are a new Studio One user, you will need to create a user account. Follow the Create Account link if your computer is connected to the Internet. Once you have created your account, continue activating Studio One Artist online.
- If your computer is not connected to the Internet, visit the Studio One product page at [www.presonus.com](http://www.presonus.com) on an Internet-connected computer to create your account. After you have created your account, skip to "Activating Studio One Artist Offline."

### Activating Studio One Artist Online

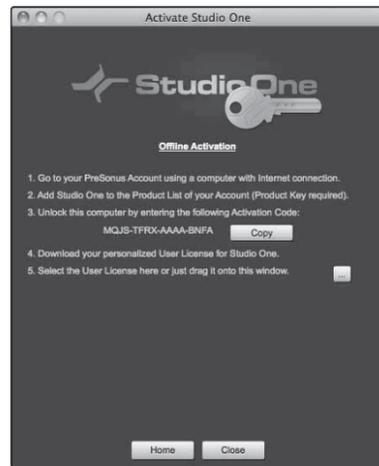


- Now that you have created a user account, you can activate your copy of Studio One Artist. Launch Studio One Artist, and the Activate Studio One menu will appear.
- Click on the Activate Online link and enter your previously created account username, password, and the product key you received with the Studio One Artist installation disc. Click on the Activate button to finish the activation process.

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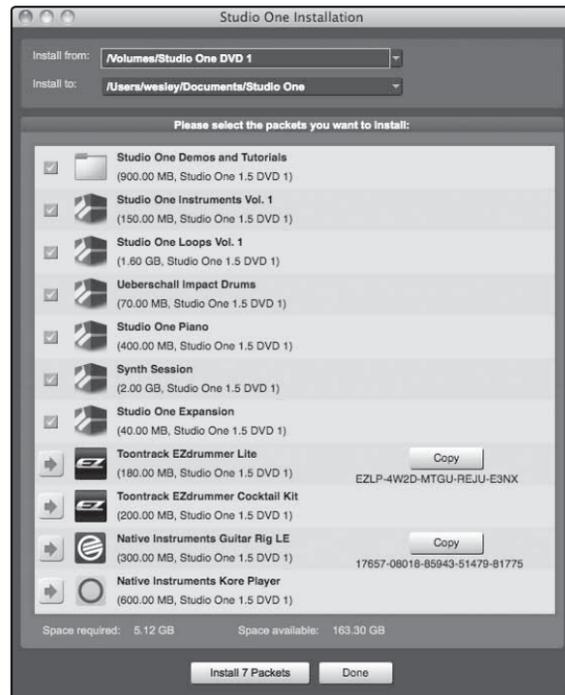
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### Activating Studio One Artist Offline



- Once you have created a user account, launch Studio One Artist. From the Activate Studio One Menu, click on the Activate Offline link. Follow the instructions to log in to your previously created user account, register the product, and obtain a license file.
- Next, copy the license file to the computer on which Studio One has been installed and locate the license file as instructed in the Activate Studio One menu. The activation process is now complete.

### Installing Bundled Content for Studio One Artist



Studio One Artist comes bundled with an array of demo and tutorial material, instruments, loops, samples, and other third-party content. The Studio One Artist bundle includes all that you need to begin producing music.

- Upon completing the Studio One Artist installation and activation process, the Studio One Content Installer will appear. If it does not appear, navigate to Help/Studio One Installation.
- At the top of the installation menu, select the source from which the content will be installed, as well as the location where you wish to install the content. The source of the content will be the same DVD from which you installed Studio One Artist. By default, Studio One Artist will point to your DVD drive as the content source. Listed in the installation menu are separate entries for each available item. Click the checkbox next to each item you wish to install, then click on the Install Packets button at the bottom left of the menu to install the selected content.
- When finished installing content, click on the Done button to exit the menu.

Studio One Artist content can be installed at any time by accessing the Help/Studio One Installation menu. If you choose not to install any portion of the content, you can install it at a later time.

### 7.4.2 Enabling the Audio Driver

Studio One and Studio One Artist were designed with PreSonus interfaces in mind. Your StudioLive is already integrated into Studio One Artist, so setup is quick and easy. When Studio One Artist is launched, by default you will be taken to the Start page. On this page, you will find document-management and device-configuration controls, as well as a customizable artist profile, a news feed, and links to demos and tutorials from PreSonus. If you have an Internet connection on your computer, these links will be updated as new tutorials become available on the PreSonus Web site.

Complete information on all aspects of Studio One Artist is available in the Reference Manual PDF located on the Studio One Artist installation disc. The information in this tutorial covers only the basic aspects of Studio One Artist and is intended to get you set up and recording as quickly as possible.

#### Start Page: Setup Area



Shows Active Audio Driver and Sample Rate and Provides Quick Links to Configure Audio and MIDI.

In the middle of the Start page, you will see the Setup area. Studio One Artist automatically scans your system for all available drivers and selects a driver. By default, it will choose a PreSonus driver if one is available.

#### Selecting a Different Audio Driver from the Start Page

Your StudioLive uses the same driver as the FireStudio family of interfaces. If you do not see "PreSonus FireStudio" on the Start page when you launch Studio One, click on the Configure Audio Devices link in the Setup area to open the Audio Setup Options window.

In the Audio Device menu, select "PreSonus FireStudio." Click the Apply button and then OK.

After you have verified that the PreSonus FireStudio Universal driver has been detected, please continue to the next section to set up your external MIDI devices. If you do not have any MIDI devices to connect at this time, please skip to Section 7.4.4.

### 7.4.3 Configuring Your MIDI Devices

From the External Devices window in Studio One Artist, you can configure your MIDI keyboard controller, sound modules, and control surfaces. This section will take you through setting up your MIDI keyboard controller and sound module. Please consult the Reference Manual located on your Studio One Artist installation DVD for complete setup instructions for other MIDI devices. Before beginning this section, you must first install the drivers for your MIDI interface or USB MIDI Controller keyboard. Please consult the documentation that came with your MIDI hardware for complete installation instructions.

### Setting Up an External MIDI Keyboard Controller from the Start Page

A MIDI keyboard controller is a hardware device that is generally used for playing and controlling other MIDI devices, virtual instruments, and software parameters. In Studio One Artist, these devices are referred to as Keyboards, and they must be configured before they are available for use. In some cases, your MIDI keyboard controller is also used as a tone generator. Studio One Artist views the two functions of these types of hardware as two different devices: a MIDI keyboard controller and a sound module. The MIDI controls (keyboard, knobs, faders, etc.) will be set up as a Keyboard. The sound module will be set up as an Instrument.



1. You can set up your external MIDI devices from the Setup area in the Start page. Before we set up a new Song for recording, let's take a moment to configure external devices.

Connect the MIDI Out of your external MIDI controller to a MIDI In on your MIDI interface. If you are using a USB MIDI controller, connect it to your computer and power it on.

Click on the Configure External Devices link in the Setup area on the Start page to launch the External Devices window.

Click the Add button.



2. The Add Device window will launch.

From the menu on the left, select New Keyboard. At this point, you can customize the name of your keyboard by entering the manufacturer and device names.

You must specify which MIDI channels will be used to communicate with this keyboard. For most purposes, you should select all MIDI channels. If you are unsure of which MIDI channels to select, select all 16.



3. In the Receive From drop-down menu, select the MIDI-interface input from which Studio One Artist will receive MIDI data.

In the Send To drop-down menu, select the MIDI interface output from which Studio One Artist will send MIDI data to your keyboard. If your keyboard controller does not need to receive MIDI data from Studio One (say, for moving motorized faders and the like), you can leave this unselected. If your keyboard does need to receive MIDI data, you must connect a MIDI cable from the MIDI Out of the MIDI interface to the MIDI In of the keyboard.



4. If this is the only keyboard that you will use to control your external synthesizers and virtual instruments, you should check the box next to Default Instrument Input. This will automatically assign your keyboard to control all MIDI devices in Studio One Artist.

Click OK.

If you have a sound module you'd like to connect, leave the External Devices window open, and proceed to the next part of this section. If not, you can close this window and skip to Section 7.4.4.

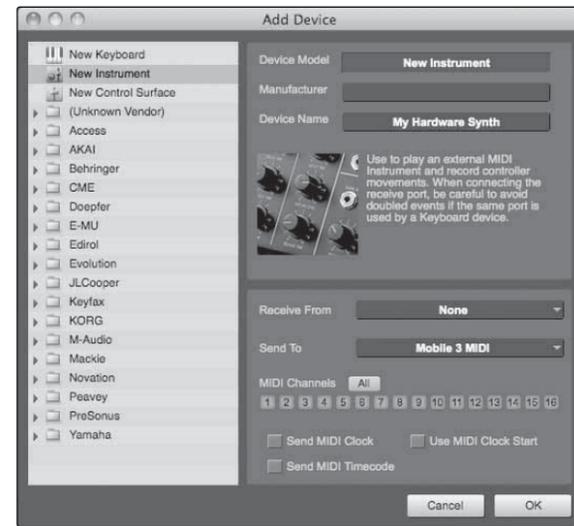
### Setting up an External MIDI Sound Module from the Start Page

MIDI instrument controllers (keyboards, MIDI guitars, etc.) send musical information in the form of MIDI data to tone modules, which respond by generating sound, as instructed. Tone modules can be standalone sound devices or can be integrated into a MIDI instrument, such as a keyboard synthesizer. Studio One Artist refers to all tone generators as Instruments. Once you have set up your MIDI keyboard controller, take a moment to configure your sound module.



1. Connect the MIDI In of your external sound module to the MIDI Out of your MIDI interface.

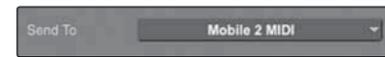
In the External Devices window, click the Add button.



2. The Add Device window will launch.

From the menu on the left, select “New Instrument.” At this point, you can customize the name of your sound module by entering the manufacturer and device names.

Specify which MIDI channels will be used to communicate with this keyboard. For most purposes, you should select all MIDI channels. If you are unsure of which MIDI channels to select, we suggest you select all 16.



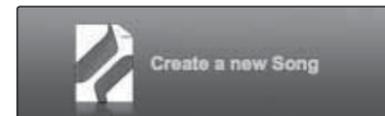
3. In the Send To drop-down menu, select the MIDI-interface output from which Studio One Artist will send MIDI data to your sound module.

In the Receive From drop-down menu, select the MIDI-interface output from which Studio One Artist will receive MIDI data from your sound module. If your sound module will not need to send information to Studio One, you can leave this unspecified.

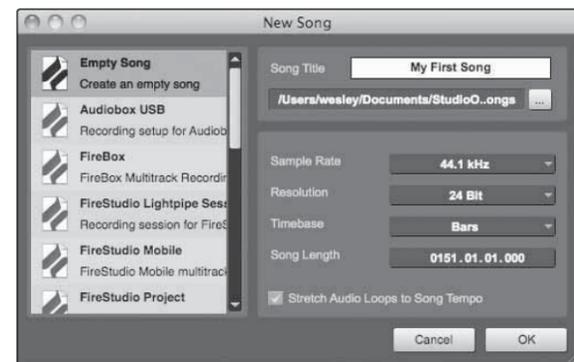
Click OK and close the External Devices window. You are now ready to start recording in Studio One Artist. The rest of this Quick Start Guide will go over how to set up a Song and will discuss some general workflow tips for navigating through the Studio One Artist environment.

## 7.4.4 Creating a New Song

Now that you’ve configured your MIDI devices, let’s create a new Song. We’ll start by setting up your default audio I/O.

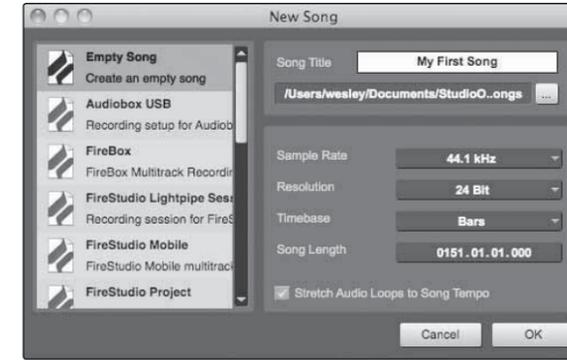


1. From the Start page, select “Create a new Song.”



2. In the browser window, name your Song and choose the directory in which you’d like it saved. You’ll notice a list of templates on the left. The StudioLive template will create a Song with a track for each of the 24 inputs. Every track is armed for recording, and no further setup is required. Select this template and click “OK.”

The rest of this section will guide you through creating a Song from an empty session.

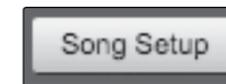


3. To begin a new Song, select Empty Song from the Templates list. At this point, you should give your Song a name and select your preferred sample rate and bit depth. You can also determine the length of your Song and the type of time format you would like to use (Notation Bars, Seconds, Samples, or Frames). Click the OK button when you are finished.

If you plan on importing loops into your Song, you may want to select Stretch Audio Loops to Song Tempo so that any loop of a known BPM (like those in the included content library) will import at the correct tempo.



4. When the Song window launches, launch the Audio Setup window by going to File | Options... (Windows) or Studio One | Preferences (Mac), and click on the Audio Setup button.



5. Click on the Song Setup button to open the Song Setup window, then click on the Audio I/O Setup icon.



6. Click on the Inputs tab in the Audio I/O Setup window, and you will see all of the available inputs on your StudioLive. At this time, you can add the number and type of inputs you plan to use.

We recommend that you create a mono input for each of the 24 inputs on your StudioLive. If you plan on recording in stereo, you should also create a stereo bus and assign it to the appropriate set of inputs. You can remove any bus by simply selecting it and clicking the Remove button. To customize the names of your buses, double-click on the default name to open a text box. When you have finished typing, hit Enter.

If you would like the same inputs to be available every time you launch Studio One Artist, click the “Make Default” button.



- Click on the Outputs tab, and you will see all of the available outputs on your StudioLive. At this time, you can add the number of output buses to which you would like to have access and can give them names.

We recommend that you create a mono output for each for the 24 FireWire returns on your StudioLive, as well as one stereo output assigned to FireWire returns 25-26 to use for your stereo main output. In the lower right corner, you will see the Audition select menu. This allows you to choose the output from which you will audition audio files prior to importing them into Studio One Artist. In general, you will want this to be the main output bus.

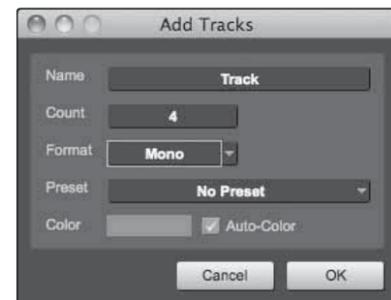
If you would like this output configuration available every time you launch Studio One Artist, click the Make Default button.

Now that you've configured your MIDI and audio I/O and created a new Song, let's go through some of the basics of Studio One Artist so you can start recording!

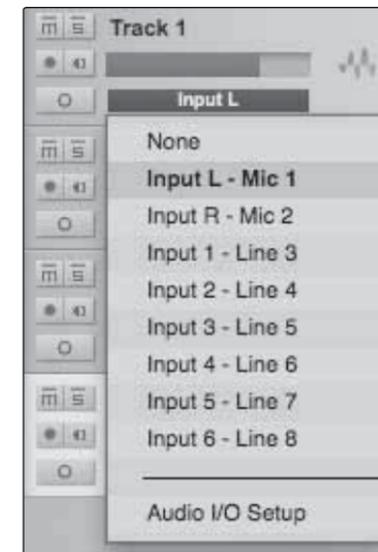
### 7.4.5 Creating Audio Tracks



- In the upper left corner of the Arrange window, you will notice several buttons. The middle button is the Add Tracks button. Click this button to bring up the Add Tracks window.



- In the Add Tracks window, you can select the number and type of tracks you'd like to create (Mono Audio, Stereo Audio, Instrument, or Automation) and can customize the track name and color.



- Once you have added your tracks, you can assign the input by simply clicking on the input to which a track is currently assigned. This will bring up the inputs list. You can also access the audio I/O setup from here.

If you would like to add a track for each of the available inputs and have the routing automatically assigned, simply go to Track | Add Tracks for All Inputs.

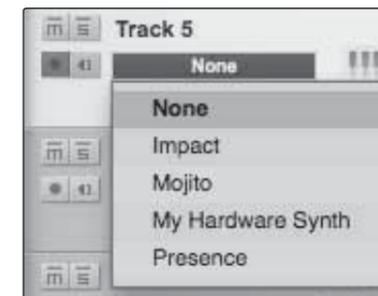
To begin recording, create an audio track, assign it to input 1, and connect a microphone to the first channel on the StudioLive. Select Record Enable on your track in Studio One Artist. Turn up the Channel 1 fader while speaking/singing into the microphone. You should see the input meter in Studio One Artist react to the input. Adjust the gain so the input level is near its maximum without clipping (distorting).

Connect a set of headphones to the StudioLive headphone output and select the Main button in the Monitor section on your StudioLive. You may also wish to connect monitors to the StudioLive's control-room outputs. You are now ready to record. For complete instructions, please consult the Studio One Reference Manual, which is located on your Studio One Artist installation DVD.

### 7.4.6 Creating MIDI Tracks



- Click on the Add Tracks button. When the Add Tracks window launches, select Instrument as the track format and click the OK button.



- To assign your MIDI input, click on the MIDI Inputs list and select your external sound module.

If you have added virtual inputs to your session, you will also see them as available inputs.

If you selected your MIDI keyboard controller as the default keyboard, it will already be selected. If not, choose your MIDI controller from the Output menu directly below.



- To the left of the Add Track button, you will find the Inspector button. Click it to display more parameters for the selected track.



- At the bottom of the Inspector menu, you will see your Bank and Program selections. From here, you can remotely change the patch on your sound module.

*MIDI data does not contain audio signals. To hear your sound module, you must connect the audio output of the sound module to a StudioLive input, then connect the StudioLive's audio outputs to a sound system. (You also can listen on headphones, using the headphone output.) When you are ready to mix your Song, you must convert the recorded MIDI data to an audio waveform by recording a new audio track.*

### 7.4.7 Adding Virtual Instruments and Plug-in Effects to Your Song

You can add plug-ins and instruments to your Song by dragging-and-dropping from the browser. You can also drag an effect or group of effects from one channel to another, drag in customized effects chains, and instantly load your favorite virtual-instrument patch without ever scrolling through a menu.

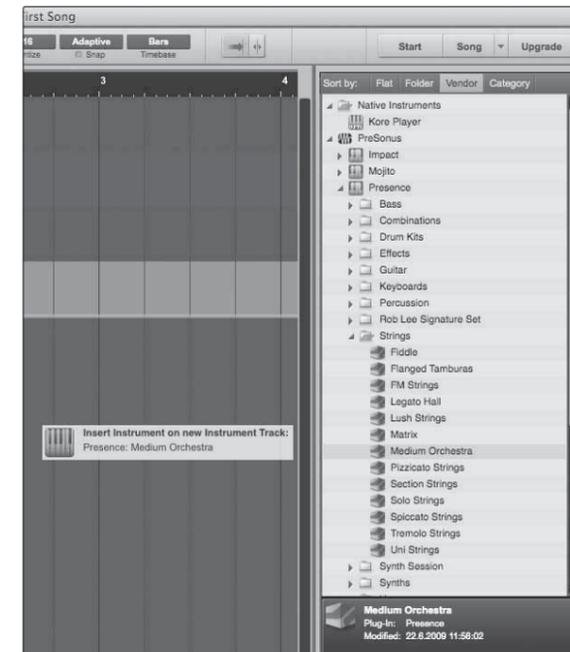
#### Opening the Browser



In the lower right corner of the Arrange window are three buttons. The Edit button opens or closes the audio editor or the MIDI piano-roll editor, depending on which type of track is selected. The Mix button opens and closes the mixer window.

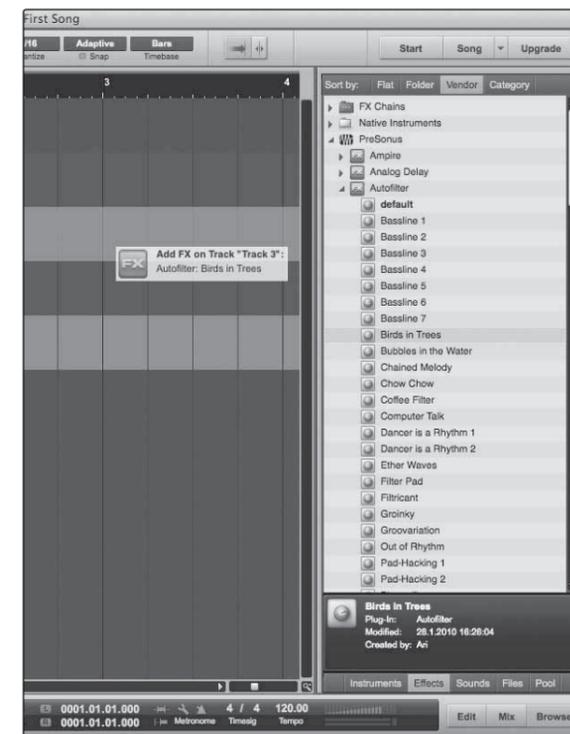
The Browse button opens the browser window, which displays all of the available virtual instruments, plug-in effects, audio files, and MIDI files, as well as the pool of audio files loaded into the current session.

#### Drag and Drop Virtual Instruments



To add a virtual instrument to your session, click the Browse and Instrument buttons to open the instrument browser. Select the instrument or one of its patches from the instrument browser and drag it into the Arrange view. Studio One Artist will automatically create a new track and load the instrument as the input.

#### Drag and Drop Effects

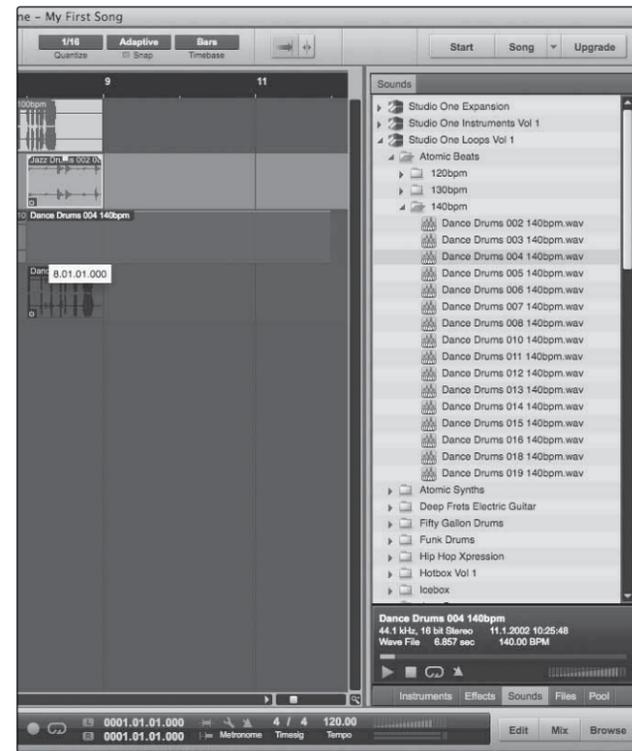


To add a plug-in effect to a track, click the Effects button and select it or one of its presets in the effects browser, then drag it over the track to which you would like to add it.

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### Drag-and-Drop Audio and MIDI Files



Audio and MIDI files can also be quickly located, auditioned, and imported into your Song by dragging them from the file browser into the Arrange view. If you drag the file to an empty space, a new track will be created with that file placed at the position to which you dragged it. If you drag the file to an existing track, the file will be placed as a new part on the track.

## 8.0 Tutorials

### 8.1 Microphone Types

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

#### 8.1.1 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.

#### 8.1.2 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 sound-pressure levels.

Most 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.

#### 8.1.3 USB Microphones and Other Types

Many microphone types are available, and as technology evolves, it is likely that 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.

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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.

### 8.1.4 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. For more information, visit your library or local bookstore, as there are many books and magazines about recording techniques. The Internet is also a great source of recording information, as are instructional videos. Some of these microphone-placement suggestions can be used in live applications, as well as for studio recording.

#### Grand Piano



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

#### Electric Guitar



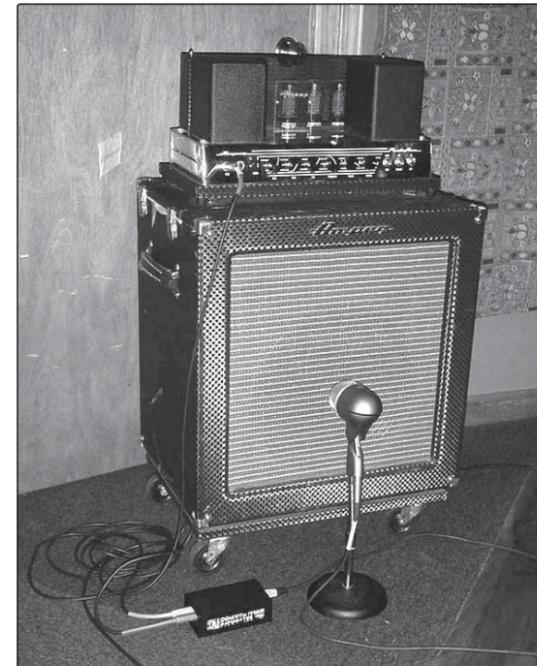
Place a dynamic microphone an inch or two away from the speaker of the guitar amplifier. Experiment with exact location. If you are recording an amp with multiple speakers, experiment with each one to see if one sounds better than the others. Place a condenser microphone approximately six feet away,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

#### Acoustic Guitar



Point a small-diaphragm condenser microphone at the 12th fret, approximately 8 inches away. Point a large-diaphragm condenser microphone at the bridge of the guitar, approximately 12 inches from the guitar. Experiment with distances and microphone placement. Another popular method is using an XY microphone placement with two small-diaphragm condenser microphones. (See drum-overheads 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 technique can also be used in live applications.

### Drum Overheads (XY example)



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

### Snare Drum (top and bottom)



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

## 8.2 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.

### 8.2.1 Common Questions Regarding Dynamics Processing

#### What is dynamic range?

Dynamic range can be defined as the ratio between the loudest possible audio level and 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.

#### 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 we need noise gates?

Consider the compressed vocal example discussed earlier; 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.

## 8.2.2 Types of Dynamic Processing

### 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, 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.

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.

### Compressors — Terminology

**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.** 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 such that for every 1 dB of level increase into the compressor, the output will only increase 0.5 dB. This produces a compression gain reduction of 0.5 dB/dB. As you increase the ratio, the compressor gradually becomes a limiter.

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**Limiters.** 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 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 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).

**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.

## 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.

### Expansion—Terminology

**Downward Expansion.** Downward expansion is the most common expansion used in live sound 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, which is heard as a 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 very 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.

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**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.

**Key Listen.** The key listen allows the user to listen to the signal that is being filtered by the gate.

**Frequency Key Filter.** Some gates offer a variable frequency control allowing the user to set a specific frequency band that will cause the gate to open or close.

**Noise Gating.** Noise gating is the process of removing unwanted sounds from a signal by attenuating all signals below a set threshold. As described, 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.

Noise gates were originally designed to help eliminate extraneous noise and unwanted artifacts from a recording, such as hiss, rumble, or transients from other instruments in the room. Since hiss and noise are not as loud as the instrument being recorded, a properly set gate will only allow the intended sound to pass through; the volume of everything else is lowered. Not only will this strip away unwanted artifacts like hiss, it will add definition and clarity to the desired sound. This is a very popular application for noise gates, especially with percussion instruments, as it will add punch or “tighten” the percussive sound and make it more pronounced.

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.

### 8.2.3 Compression Settings: Some Starting Points

#### Vocals

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

Threshold	Ratio	Attack	Release
-8.2 dB	1.8:1	0.002 ms	38 ms

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

Threshold	Ratio	Attack	Release
-3.3 dB	2.8:1	0.002 ms	38 ms

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

Threshold	Ratio	Attack	Release
-1.1 dB	3.8:1	0.002 ms	38 ms

#### Percussion

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

Threshold	Ratio	Attack	Release
-2.1 dB	3.5:1	78 ms	300 ms

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

Threshold	Ratio	Attack	Release
-13.7 dB	1.3:1	27 ms	128 ms

#### Fretted Instruments

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

Threshold	Ratio	Attack	Release
-4.4 dB	2.6:1	45.7 ms	189 ms

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

Threshold	Ratio	Attack	Release
-6.3 dB	3.4:1	188 ms	400 ms



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

Threshold	Ratio	Attack	Release
-0.1 dB	2.4:1	26 ms	193 ms

**Keyboards**

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

Threshold	Ratio	Attack	Release
-10.8 dB	1.9:1	108 ms	112 ms

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

Threshold	Ratio	Attack	Release
-11.9 dB	1.8:1	0.002 ms	85 ms

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

Threshold	Ratio	Attack	Release
3.3 dB	2.5:1	1.8 ms	50 ms

**Stereo Mix**

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

Threshold	Ratio	Attack	Release
5.5 dB	7.1:1	0.001 ms	98 ms

**Contour.** This setting fattens up the main mix.

Threshold	Ratio	Attack	Release
-13.4 dB	1.2:1	0.002 ms	182 ms

**Effects**

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

Threshold	Ratio	Attack	Release
-4.6 dB	2.4:1	7.2 ms	93 ms

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

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

8.2.4 Corresponding Frequencies for the Equal-Tempered Scale

The Noise Gate on the StudioLive 24.4.2 features a frequency Key Filter. Use the table below, and the tables in Section 8.3.2, to help find the best Key Filter settings. Please note: the Hertz (Hz) measurements below are rounded to the nearest Hertz, and the kilohertz (kHz) measurements are rounded to the nearest 10 Hz.

C-0	16 Hz	C-2	65 Hz	C-4	262 Hz	C-6	1.05 kHz
C#-0	17 Hz	C#-2	69 Hz	C#-4	277 Hz	C#-6	1.11 kHz
D-0	18 Hz	D-2	73 Hz	D-4	294 Hz	D-6	1.17 kHz
D#-0	19 Hz	D#-2	78 Hz	D#-4	311 Hz	D#-6	1.24 kHz
E-0	21 Hz	E-2	82 Hz	E-4	330 Hz	E-6	1.32 kHz
F-0	23 Hz	F-2	87 Hz	F-4	349 Hz	F-6	1.39 kHz
F#-0	24 Hz	F#-2	92 Hz	F#-4	370 Hz	F#-6	1.48 kHz
G-0	26 Hz	G-2	98 Hz	G-4	392 Hz	G-6	1.57 kHz
G#-0	28 Hz	G#-2	104 Hz	G#-4	415 Hz	G#-6	1.66 kHz
A-0	29 Hz	A-2	110 Hz	A-4	440 Hz	A-6	1.76 kHz
A#-0	31 Hz	A#-2	116 Hz	A#-4	466 Hz	A#-6	1.86 kHz
B-0	32 Hz	B-2	123 Hz	B-4	494 Hz	B-6	1.97 kHz
C-1	33 Hz	C-3	131 Hz	C-5	523 Hz	C-7	2.09 kHz
C#-1	35 Hz	C#-3	138 Hz	C#-5	554 Hz	C#-7	2.22 kHz
D-1	37 Hz	D-3	147 Hz	D-5	587 Hz	D-7	2.35 kHz
D#-1	39 Hz	D#-3	155 Hz	D#-5	622 Hz	D#-7	2.49 kHz
E-1	41 Hz	E-3	165 Hz	E-5	659 Hz	E-7	2.64 kHz
F-1	44 Hz	F-3	175 Hz	F-5	698 Hz	F-7	2.79 kHz
F#-1	46 Hz	F#-3	185 Hz	F#-5	740 Hz	F#-7	2.96 kHz
G-1	49 Hz	G-3	196 Hz	G-5	784 Hz	G-7	3.13 kHz
G#-1	52 Hz	G#-3	208 Hz	G#-5	830 Hz	G#-7	3.32 kHz
A-1	55 Hz	A-3	220 Hz	A-5	880 Hz	A-7	3.52 kHz
A#-1	58 Hz	A#-3	233 Hz	A#-5	932 Hz	A#-7	3.73 kHz
B-1	62 Hz	B-3	247 Hz	B-5	988 Hz	B-7	3.95 kHz

## 8.3 Equalizers

The StudioLive 24.4.2 is equipped with a 4-band 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.

### 8.3.1 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 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.

#### 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 like those in the StudioLive 24.4.2 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 also offers separate control over the Q, which is the ratio of the center frequency to the bandwidth. A semi-parametric EQ provides control over most of these parameters but the Q is fixed. Some devices, such as the StudioLive 16.4.2, have quasi-parametric EQ, which is semi-parametric EQ with a simple, switchable Q setting (typically, High and Low Q).

#### Q

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, which allows you to attenuate or boost a very narrow or wide range of frequencies.

A narrow bandwidth (higher Q) has obvious 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 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 24.4.2 features 4 stereo, 31-band, graphic EQs that can be inserted on a variety of buses. (See Section 5.3 for details.) A graphic EQ is a multiband equalizer that uses sliders to adjust the amplitude for each frequency band. It gets its name from the positions of the sliders, which graphically display the resulting frequency-response curve. As explained in Section 5.3, the encoders in the Fat Channel are used to make amplitude adjustments, and the meter LEDs display the "slider" positions. The center frequency and bandwidth are fixed; the level (amplitude) for each band is the only adjustable parameter.

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

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

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

**Low Mids (250 Hz to 2 kHz).** In general, you will want to emphasize the lower portion of this range and deemphasize the upper portion. Boosting the range from 250 Hz to 500 Hz will accent ambience in the studio and will add clarity to bass and lower frequency instruments. The range between 500 Hz and 2 kHz can make midrange instruments (guitar, snare, saxophone, etc.) "honky," and too

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much boost between 1 kHz and 2 kHz can make your mix sound thin or “tinny.”

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

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

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

### 8.3.2 Equalization Settings: How to Find the Best and Leave the Rest

How do you find the best and worst each instrument has to offer and adjust their frequency content 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 primarily 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.

#### Additional advice:

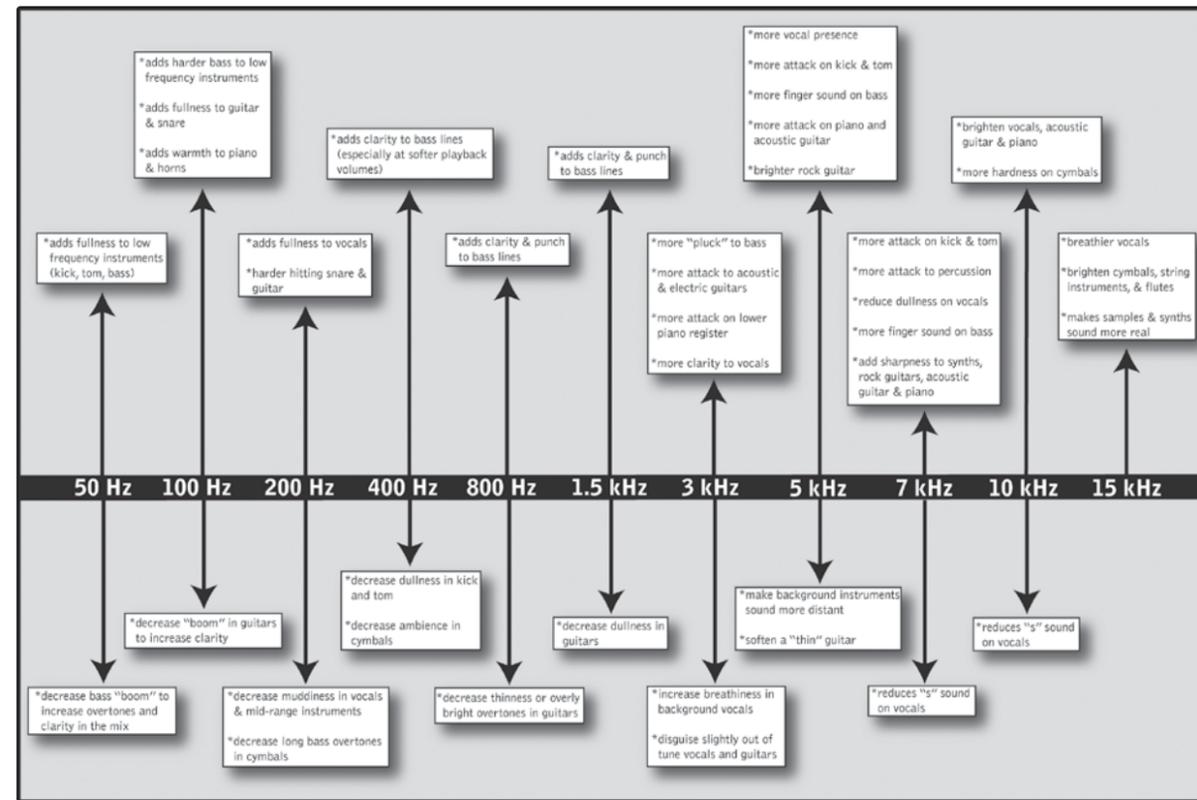
- **You can only do so much.** Not every instrument can or should have a full, rich low end and a sharp attack. If every instrument is EQ'd to have the same effect, it will lose its identity in the mix. Your goal is not individual perfection, it is perfection in unity.
- **Step away from the mix.** Your 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 and hear 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. With every instrument, there are frequencies that can be attenuated or boosted to add clarity or fullness. Altering 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. These are just suggestions; the 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

Table 2



8.3.3 General EQ Suggestions

Included with your StudioLive is a library of Channel Strip presets. Section 4.1.6 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 8.2.3, the right EQ setting for any given instrument will depend upon the room and the tonality of the instrument.

Vocals

Pop Female Vocals

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

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

Rock Female Vocals

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

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

Pop Male Vocals

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

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

Rock Male Vocals

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

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

**Percussion**

**Snare**

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

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

**Left/Right (Stereo) Overheads**

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

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

**Kick Drum**

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

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

**Fretted Instruments**

**Electric Bass**

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	ON	36	N/A	-8	ON	130	0.4	+4

HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	2.0	0.6	+4	ON	ON	4.2	N/A	+1

**Acoustic Guitar**

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	OFF	155	0.4	+4	ON	665	2.0	+2

HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	2.0	0.3	0	ON	ON	6.0	N/A	+4

**Distorted Electric Guitar**

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	OFF	320	0.5	+6	ON	960	0.4	0

HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	3.5	1.0	+4	ON	ON	12	N/A	0

**Keyboards**

**Piano**

LOW ON/OFF	LOW SHELF	LOW FREQ (Hz)	LOW Q	LOW GAIN	LOW MID ON/OFF	LOW MID FREQ (Hz)	LOW MID Q	LOW MID GAIN
ON	ON	108	N/A	-2	ON	665	0.2	+2

HIGH MID ON/OFF	HI MID FREQ (kHz)	HIGH MID Q	HIGH MID GAIN	HIGH ON/OFF	HIGH SHELF	HIGH FREQ (kHz)	HIGH Q	HIGH GAIN
ON	2.9	0.4	+2	ON	OFF	7.2	0.6	+4

## 8.4 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 that will make mixing more convenient and will provide better control of your mix. In this section, we explore two different ways in which subgroups can help you to create a more efficient mixing environment and a more successful live mix.

### 8.4.1 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 when grouping any type of instrument section in a live mix.

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

- Channel 1: Kick
- Channel 2: Snare Top
- Channel 3: Share 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

1. 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—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.



2. 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 volume and panning for each mic in the mix. Choose the Select button above Subgroup 1.

3. 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.
4. 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. The aux bus enables you to create and send auxiliary mixes that are separate from the main and subgroup mixes.

The StudioLive 24.4.2 is equipped with 12 aux buses: Auxes 1 through 10, 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.

### 8.4.2 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. For instance, a typical vocal mix for an electronic band dramatically trails off in a wash of reverb, whereas a reggae band usually has delay on vocals. The StudioLive allows you to assign either or both of the onboard effects buses to a subgroup. So let's take the example of the reggae band.

1. 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.
2. 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 to taste. (See Section 4.8.1 for complete operation instructions.)
3. 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.
4. Using the meter section, locate the send encoders for channels 10 and 11 and turn them to a little more than 50%.
5. 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.



6. 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 the reggae band's performance. The Tap button allows you to go one step further and set the tempo of the delay to match the tempo of the song.

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 their best performance.

## 8.5 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 12 aux buses: Aux 1 through 10, 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 individual channel processing.

### 8.5.1 Monitor Mixing

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

1. 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 display the amount of send to this aux bus from each of the 24 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 these encoders the same way that 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.
2. By pressing the Select button for Aux 1, you can add dynamics processing and EQ to the overall monitor mix. These are especially useful for eliminating feedback in a monitor. Keep in mind that an equalizer can also be used to increase the presence of an instrument by boosting that particular frequency range without necessarily boosting the volume in the mix. This is great for getting the lead guitar to cut through in the guitarist's monitor mix and to provide that extra rumble in the bassist's mix. You can listen to the aux mixes you are creating, using your headphones or your control-room monitor, by simply soloing the aux and selecting Solo as the source in the Monitor section.

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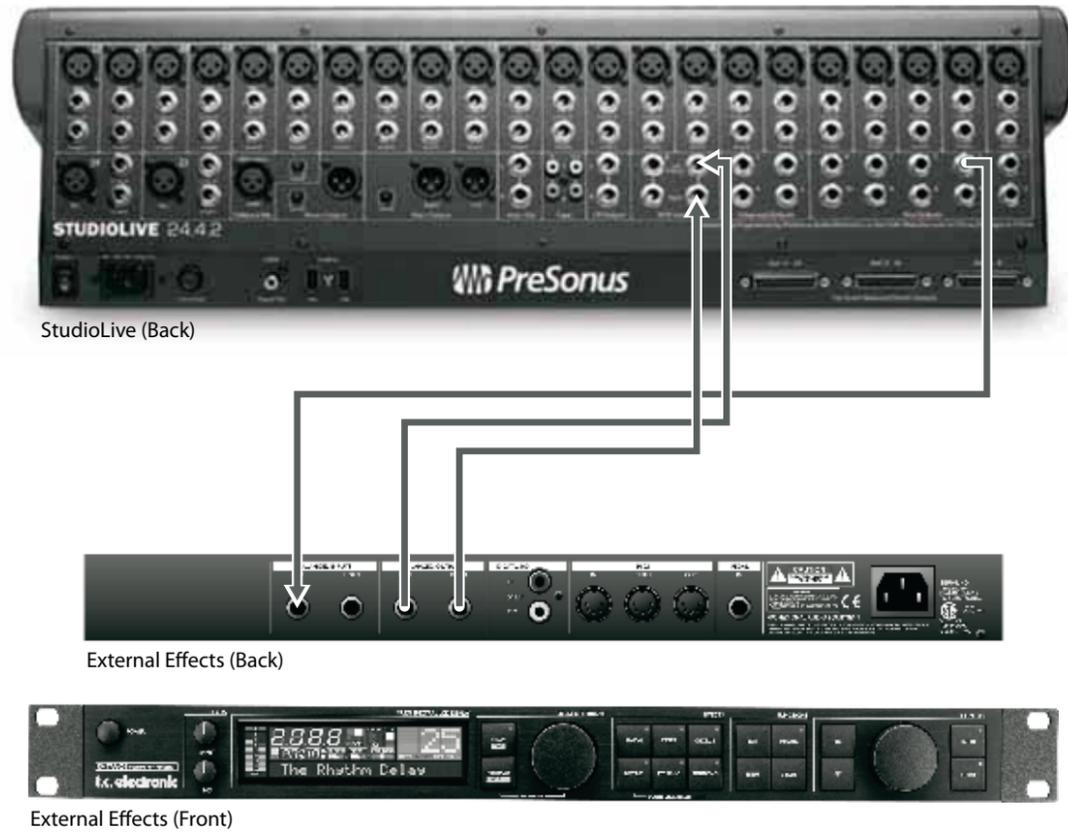
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### 8.5.2 Effects Processing

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 use an external effects processor with your StudioLive mixes.



1. 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 above.



2. 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.

3. Press the Mix button for Aux 3. The meter section of the StudioLive will now show the output levels of each of the 24 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 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.

## 8.6 Digital Effects

The StudioLive includes two stereo effects processors that feature the two most common type of effects is use for live sound: reverb and delay.

### 8.6.1 Reverb

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

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

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

### 8.6.2 Delay

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 as 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.** 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.*

## 8.7 Level Setting Procedure in Detail

**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 24 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 o'clock.



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



- 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 analog-to-digital converters!

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

## 8.8 The Solo Bus

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

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

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

**PFL (Pre-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 channels will be sent to their assigned outputs. While useful in dialing in dynamics during soundcheck, this mode is dangerous during a live show. We recommend that this mode be turned off when mixing live events.



### 8.8.1 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 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.

### 8.8.2 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 (that is, if Channel 3 is soloed, you will only hear Channel 3 in your Mains). This makes a great fine tuning tool but it can quickly destroy a live mix. We highly recommend that you drop out of this mode once the show has started.

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

In Section 8.7, 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).

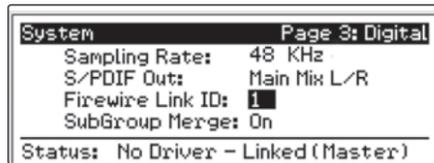
As discussed earlier in Section 8.7, most engineers start with the drums and work from the bottom up. To begin, bring all your faders down and raise your main fader to unity gain. Press and hold the SIP button in the Cue section until it illuminates and press the Solo button on your kick-drum mic channel. 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 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.

## 8.9 Cascading Two StudioLive 24.4.2s

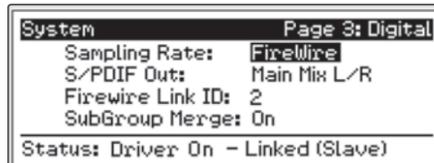
Your StudioLive 24.4.2 can be cascaded with another StudioLive 24.4.2, using a FireWire cable, to create a single 48-channel console. Because of the limitations of FireWire, you cannot connect a computer to your cascaded StudioLives; however, you can cascade one of several FireStudio family interfaces with a single StudioLive 24.4.2 to add more recording inputs. This tutorial will guide you through the syncing process and explain how two StudioLive 24.4.2 function as one 48-channel mixer.

### 8.9.1 Configuring Multiple Units

1. To cascade two StudioLive 24.4.2s to create a standalone system (without a computer), connect a FireWire cable from the first unit to the second unit.



2. 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. Either unit in the chain can be designated as the Master. Since both units 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.



3. Press the System button on the cascaded unit 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 the cascaded unit.

### 8.9.2 Aux Mixing with Cascaded Mixers

Every channel in the mixer chain can be sent to the ten aux outputs on the Master unit. When you press the Mix button on any of the ten auxes on either mixer, you will notice that the Mix button for the corresponding aux will illuminate on the other 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 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, Channels 1 through 24 reside on the Slave, and Channels 25 through 48 will reside 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 on either mixer. The Fat Channel meters and encoders on both mixers will be ready for you to create an Aux 3 mix. you will use the Fat Channel meters and encoders on the Slave to set the Aux 3 send levels for Channels 1-24 and the meters and encoders on the Master unit to set the Aux 3 send levels for Channels 25-48. 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 mixer are still active but only have access to the local channels. So in this example, the Aux 3 mix for Channels 1-24 on the Slave is routed locally to the Aux 3 output and merged with the overall Aux 3 mix on the Master unit at the same time. Only the Master mixer receives the merged signals from all cascaded mixers, in addition to its 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 both of the mixers in the chain in order to control panning for the channels on the slaved mixer.

### 8.9.3 Internal Effects Buses

Unlike the 10 Aux buses, the two Internal Effects buses on each mixer are independent. Using the same example as the previous section, Channels 1-24 can only be routed to EFXA and EFXB on the Slave, and Channels 25-48 are processed using the Master unit's two internal effects buses. The advantage is that you get twice the number effects buses!

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

### 8.9.4 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 or local only.

This is determined from the Master unit (i.e., if Subgroup Merge is enabled on the Master, the Slave's Subgroups will be merged with the Master's Subgroups). The Subgroup faders on the Master mixer will control the entire group, and dynamics processing can be added to the entire mix using the Master's Fat Channel.

If Subgroup Merge is turned Off, each of the four subgroups on both mixers are locally controlled. So Channels 1-24 can only be assigned to the Subgroups on the Slave and Channels 25-48 can only be assigned to the Subgroups on the Master unit. Because the Subgroups on each of the mixers can still be routed to the Main output on the master unit, unmerging the Subgroups offers advantages for Effects group mixing. (See Section 8.4.2 for details.) But this can also be useful in any situation where you want 24 or fewer channels assigned to the same Subgroup; just do a little careful patching!

### 8.9.5 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 either 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.

### 8.9.6 Copy and Load

Channel-strip settings from any channel or bus on either mixer in the chain can be copied to any other channel or bus on the other mixer in the chain. For example, if you select Channel 4 on the Slave and press the Copy button, every Select button on both mixers 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.

### 8.9.7 Local Versus Merged Buses and Inputs

As discussed earlier, the two internal effects buses on each mixer are controlled from the mixer to which they belong, 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 section on the slave is active through its outputs, it 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 4.7.3, the Analog Tape Input on the StudioLive is patched directly to the Main Output using a button in the Master Section. The Tape Inputs are active on both mixers in the chain, but the Tape In button only routes the associated Tape Input to the Main bus on that unit. For example, the Tape In button on the Slave patches it to the Main bus on the Slave only. Because of this, we recommend that you only use the Tape Input on the Master unit.
- **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 unit can only receive signals 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 either mixer, both mixers are placed into the same mode at the same time.

## 9.0 Technical Information

### 9.1 Specifications

#### Microphone Preamp

Type	XLR Female, balanced
Frequency Response to Direct Output (at unity gain)	20 Hz-40 kHz, 0 / -0.5 dBu
Frequency Response to Main Output (at unity gain)	20 Hz-20 kHz, ± 0.5 dBu
Input Impedance	1 kΩ
THD to Direct Output (1 kHz at unity gain)	< 0.005%, +4 dBu, 20 Hz-20 kHz, unity gain, unwt'd
THD to Main Output (1 kHz at unity gain)	<0.005%, +4 dBu, 20 Hz-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 Hz-40 kHz, 0 / -0.5 dBu
Frequency Response to Main Outputs (at unity gain)	20 Hz-20 kHz, 0/-0.5 dBu
Input Impedance	10 kΩ
THD to Direct Output (1 kHz at unity gain)	<0.0007%, +4 dBu, 20 Hz-20 kHz, unity gain, unwt'd
THD to Main Output (1 kHz at unity gain)	<0.005%, +4 dBu, 20 Hz-20 kHz, unity gain, unwt'd
S/N Ratio to Direct Output (Ref = +4 dBu, 20 kHz BW, unity gain, A-wtd)	-105 dB
S/N Ratio to Main Output (Ref = +4 dBu, 20 kHz BW, unity gain, A-wtd)	-94 dB
Gain Control Range (+/- 1 dB)	-20 dB to +20 dB
Maximum Input level (unity gain)	+22 dBu

#### Tape Inputs

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

#### Aux Outputs

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

#### Subgroup Outputs

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

#### Tape Outputs

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

#### Control Room Outputs

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

#### System Crosstalk

Input to Output (Ref = +4 dBu, 20 Hz-20 kHz, unwt'd)	-90 dBu
Adjacent Channels (Ref = +4 dBu, 20 Hz-20 kHz, unwt'd)	-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	2 <sup>nd</sup> - 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

### Digital Audio

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

### Clock

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

### Power

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

### Physical

Length	22.35 inches (568.06 mm)
Width (chassis only)	25.5 inches (647.7 mm)
Maximum Height	6.90 inches (175.26 mm)
Weight	51 Lbs. (23.133 kg)

### Global Warming

Recommended Ambient Operating Temperature	0° to 40° Celsius / 32° to 104° Fahrenheit
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## 9.2 Channel Presets

Your StudioLive comes with 50 Fat Channel presets that were custom-designed by professional PreSonus users. These presets can be altered, renamed and overwritten, and there are 50 additional empty storage locations where you can 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

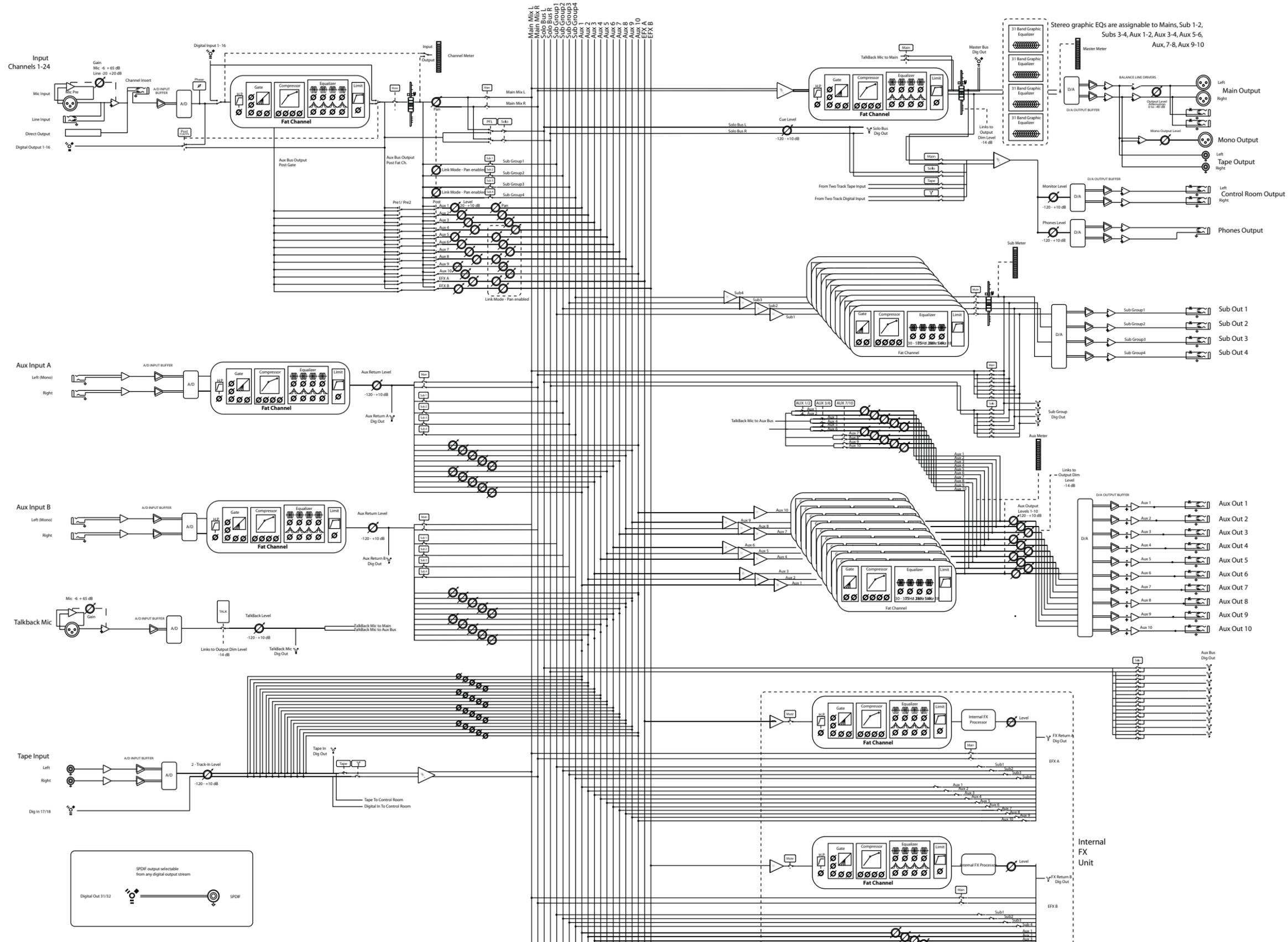
## 9.3 Digital Effects Library

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 where you can build your own custom effects library. The factory presets can be altered, renamed, and overwritten.

	TYPE	NAME
F1	AMBIENCE	Natural
F2		Lively
F3	SMALL ROOM	Closet
F4		Studio A
F5		Studio B
F6		Bedroom
F7	BRIGHT ROOM	Kitchen
F8		Tile Floors
F9		Tile Bathroom
F10		Concrete Floors
F11	SMALL HALL	Radio Booth
F12		Small Club
F13		Big Club
F14	BRIGHT HALL	Brick House
F15		Linoleum Room
F16		Tile Room
F17	WARM HALL	Log Cabin
F18		Wood Floors
F19		Brick Club
F20		High Ceilings
F21	GATED HALL	GateVerb Short
F22		GateVerb Med
F23		GateVerb Long
F24		GateVerb Extreme

	TYPE	NAME
F25	LARGE HALL	Church
F26		Cathedral
F27		Gymnasium
F28		Arena
F29	PLATE	PlateVerb Shimmer
F30		PlateVerb Thick
F31		PlateVerb Drums
F32		PlateVerb Vox
F34	MONO DELAY	M: Short Slap
F35		M: Long Slap
F36		M: Triplet
F37		M: Triplet
F38	FILTER DELAY	Analog Slap
F39		Analog Trip
F40		Analog 8 <sup>th</sup>
F41	STEREO DELAY	Slap Quick
F42		Long Slap
F43		Spacey
F44		Long Tail
F45		ST: Triplet
F46	PING-PONG DELAY	Ping-pong Slap
F47		Ping-pong Delay
F48		Ping-pong Spacey
F49		Ping-pong Trip
F50		Ping-pong Purple Haze
F50-89	USER-CREATED PRESETS	

9.4 Block Diagram



# StudioLive 24.4.2

# Old-Fashioned Analog Session Data

# Recall Sheet

(print out on US tabloid or Euro A3 paper or 2-page letter size or A4)

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Track	Trim	Production Notes
1		Instrument Mic used Notes
2		Instrument Mic used Notes
3		Instrument Mic used Notes
4		Instrument Mic used Notes
5		Instrument Mic used Notes
6		Instrument Mic used Notes
7		Instrument Mic used Notes
8		Instrument Mic used Notes
9		Instrument Mic used Notes
10		Instrument Mic used Notes
11		Instrument Mic used Notes
12		Instrument Mic used Notes

Track	Trim	Production Notes
13		Instrument Mic used Notes
14		Instrument Mic used Notes
15		Instrument Mic used Notes
16		Instrument Mic used Notes
17		Instrument Mic used Notes
18		Instrument Mic used Notes
19		Instrument Mic used Notes
20		Instrument Mic used Notes
21		Instrument Mic used Notes
22		Instrument Mic used Notes
23		Instrument Mic used Notes
24		Instrument Mic used Notes

Artist \_\_\_\_\_ Scene \_\_\_\_\_ Date \_\_\_\_\_

## AUX INPUTS

Level	AUX In A SOURCE	Level	AUX In B SOURCE
A	Notes _____	B	Notes _____

## AUX SENDS

Level	AUX SEND	Level	AUX SEND
1	Notes _____ Monitor <input type="checkbox"/> Send <input type="checkbox"/>	6	Notes _____ Monitor <input type="checkbox"/> Send <input type="checkbox"/>
2	Notes _____ Monitor <input type="checkbox"/> Send <input type="checkbox"/>	7	Notes _____ Monitor <input type="checkbox"/> Send <input type="checkbox"/>
3	Notes _____ Monitor <input type="checkbox"/> Send <input type="checkbox"/>	8	Notes _____ Monitor <input type="checkbox"/> Send <input type="checkbox"/>
4	Notes _____ Monitor <input type="checkbox"/> Send <input type="checkbox"/>	9	Notes _____ Monitor <input type="checkbox"/> Send <input type="checkbox"/>
5	Notes _____ Monitor <input type="checkbox"/> Send <input type="checkbox"/>	10	Notes _____ Monitor <input type="checkbox"/> Send <input type="checkbox"/>

## SUBGROUP / MAIN ASSIGNMENT ROUTING MATRIX

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

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## 10.0 Troubleshooting and Warranty

### 10.1 Troubleshooting

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, Capture™ audio-recording software, Studio One™ digital audio workstation software, Universal Control control-panel software, and VSL™ mixer-control 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](http://www.presonus.com)) regularly for software information and updates, firmware updates, and support documentation for frequently asked questions. You can get individual technical assistance by calling PreSonus at 225-216-7887, Monday through Friday, between the hours of 9 a.m. and 5 p.m. Central Time. PreSonus technical support is available via email during the same hours at [techsupport@presonus.com](mailto: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 Universal Control console, and optimizing your operating system for audio.

Other known causes of this issue are wireless networks and running other FireWire devices (FireWire 400 or FireWire 800) 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, such as 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.)

- As 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 clicks 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 3: 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.

#### Fader Movements Have No Effect on Audio

Verify that your StudioLive is not locked by navigating to Page 5: 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.

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### 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.

## Notes

## 10.2 PreSonus StudioLive 24.4.2 Limited Warranty

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

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

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## Declaration of Conformity

Responsible Party: **PreSonus Audio Electronics**

Address: **7257 Florida Blvd, Baton Rouge,  
LA 70806 USA**

Phone: **225.216.7887**

declares that **StudioLive™ 24.4.2** complies with  
Part 15 of the FCC rules.

Operation is subject to the following two conditions:

- (1) This device may not cause harmful interference, and;
- (2) This device must accept any interference received, including interference that may cause undesired operation

Note: No product support is available when you call the number above. Refer to your Certificate of Warranty in your Owner's Manual for PreSonus' Product Support telephone number.



Baton Rouge • USA • [www.presonus.com](http://www.presonus.com)

P/N FCD-SL00004

## Added bonus: PreSonus' previously Top Secret recipe for...

### Jambalaya

#### Ingredients:

- 5 lbs link andouille sausage
- 3 lbs boneless chicken
- 2 lbs ground beef
- 3 lbs onions (yellow or purple)
- 2 stalks of celery
- 1 lb bell peppers (green or red)
- 1 batch green onions
- 3 lbs rice
- Tony Chachere's Cajun Seasoning
- 1 bottle chicken stock concentrate (or 3 cubes chicken bullion)
- 1 can Rotel tomatoes with chilies, diced (regular hot)
- Tabasco sauce

#### Cooking Instructions:

1. In a 16 qt. pot or larger, slice link sausage and pan-fry until brown.
2. Add ground beef and brown.
3. Do not remove from pot Add diced onions, celery, and bell peppers, 1 can Rotel Original diced tomatoes w/chilies, 3 oz concentrate chicken stock, ½ teaspoon of Cajun seasoning, 1 teaspoon of Tabasco hot sauce (or more...maybe lots more).
4. Cook until onions are translucent.
5. Add chicken and cook until it turns white.
6. Add diced green onions, 1 tsp. salt, ½ gallon water and bring to a boil.
7. Add rice and bring to a boil. Cook on high for 8 minutes, covered, stirring every 2 minutes
8. Cook covered on low for 10 minutes, stirring only once.
9. Turn off and let sit for 30 minutes.
10. Serve and enjoy!

Serves 20

# The back page of the **StudioLive™ 24.4.2** Owner's Manual

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