AudioBox™ 1818VSL
Owner's Manual

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

PreSonus™

www.presonus.com

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 EC1907/206 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 (2006/95/EC) and the Low Voltage Directive (2004/108/EC).

AudioBox™ 1818VSL

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

Overview

Hookup

Connecting
to a
Computer

Software: Virtual StudioLive
and Studio One Artist

Tutorials

Technical
Information

Troubleshooting
and Warranty

iv

1.0   Overview

1.1   Introduction

Thank you for purchasing the PreSonus AudioBox™ 1818VSL. PreSonus Audio Electronics has designed the AudioBox 1818VSL utilizing high-grade components to ensure optimum performance that will last a lifetime. Loaded with 8 high-headroom, Class A, XMAX™ microphone preamplifiers; a built-in 18x18 USB 2.0 recording and playback engine; MIDI I/O; S/PDIF I/O; ADAT I/O; BNC word clock output; Fat Channel processing with 3-band EQs, compressors, limiters, and downward expanders; delay and reverb effects; and more, AudioBox 1818VSL breaks new boundaries for music performance and production. All you need is a computer with a USB 2.0 connection, a few microphones and cables, powered speakers, and your instruments, and you are ready to record in the studio or in front of a live audience!

We encourage you to contact us at 225-216-7887 (9 a.m. to 5 p.m. Central Time) with questions or comments regarding your PreSonus AudioBox 1818VSL. 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 and are confident that you will enjoy your AudioBox 1818VSL!

ABOUT THIS MANUAL: *We suggest that you use this manual to familiarize yourself with the features, applications, and correct connection procedures for your AudioBox 1818VSL before trying to connect it to your computer. This will help you avoid problems during installation and setup.*

Throughout this manual you will find Power User Tips that can quickly make you an AudioBox 1818VSL expert. In addition to the Power User Tips, you will find an assortment of audio tutorials at the back of this manual. These tutorials cover everything from microphone placement to equalizer and compression-setting suggestions.

1.2   Summary of AudioBox 1818VSL Hardware Features

- 24-bit/96 kHz sampling rate
- 8 Class A XMAX microphone preamplifiers
- 2 Instrument inputs
- 6 Line inputs
- MIDI I/O
- S/PDIF I/O
- ADAT I/O
- BNC word clock output
- High-definition analog-to-digital converters (114 dB dynamic range)
- 18x18 USB 2.0 audio interface
- Headphone output
- Rugged steel chassis
- Virtual StudioLive™ monitor-mixing software
- Studio One™ Artist
- Compatible with Cubase, Digital Performer, Logic, Nuendo, Sonar, Studio One, and other audio-recording software
- Mac OS X®- and Windows®-compatible

1.3   Summary of Studio One 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), distortion (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

1.4   Summary of Virtual StudioLive Software Features

Virtual StudioLive is a powerful monitor-mixing application that allows you to add professional dynamics processing, EQ, and effects to your monitor mixes.

- Easy drag-and-drop workflow

- Drag presets directly to channels

- Drag parts of presets directly to components in the Fat Channel

- StudioLive 16.0.2 Fat Channel with:

  - High-pass Filter

  - Downward Expander

  - Compressor

  - 3-band semi-parametric EQ

  - Limiter

- Two effects buses with 32-bit floating-point reverbs and delays

- Quickly drop entire Scenes to the mixer for instant recall of all channel and effects settings

- Load effects quickly by simply dragging and dropping

- Mixer overview:

  - See all of the most-used mixer parameters 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

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

1.5   **What is in the Box**

In addition to this manual, your AudioBox 1818VSL package contains the following:

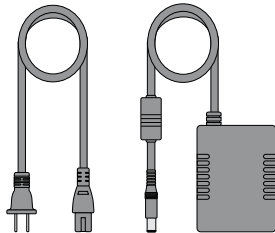PreSonus AudioBox 1818VSL USB 2.0 recording interface

6' (1.8m) USB cable

External power supply

Software library containing:

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

- PreSonus Virtual StudioLive and AudioBox driver CD

Connecting
to a
Computer

Software: Virtual StudioLive
and Studio One Artist

Tutorials

Technical
Information

Troubleshooting
and Warranty

2.0   Hookup

2.1   Front-Panel Connections

Microphone Inputs. Your AudioBox 1818VSL is equipped with PreSonus XMAX microphone preamplifiers for use with all types of microphones. The XMAX 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 background noise.

The eight analog inputs of the AudioBox 1818VSL feature combo jacks. This convenient connector accepts either a ¼-inch phone plug or an XLR plug.

48-volt Phantom Power. The AudioBox 1818VSL provides 48V phantom power for every microphone input, in banks of four. This feature can be individually enabled for each bank, using the buttons on the left side of the front 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

Instrument Inputs. The ¼-inch TS connectors on channels 1 and 2 are for use with instruments (guitar, bass, etc.). When an instrument is plugged into the instrument input, the mic preamp is bypassed, and the AudioBox 1818VSL becomes an instrument preamplifier.

Power User Tip: Active instruments are those that have an internal preamp or a line-level output. Active instruments should be plugged into a line input rather than into an instrument input. Plugging a line-level source into the instrument inputs on the front of the AudioBox 1818VSL not only risks damage to these inputs but also results in a very loud and often distorted audio signal.

⚠ *Please note: As with any audio input device, plugging in a microphone or an instrument, or turning phantom power on or off, will create a momentary spike in the audio output of your AudioBox 1818VSL. Because of this, we highly recommend that you 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.*

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

Line-level Inputs. Channels 3-8 of the AudioBox 1818VSL have a ¼-inch, balanced TRS connection for line-level input. When these inputs are engaged, the microphone-preamp circuit is bypassed. Typical examples of line-level connections are synthesizer outputs, CD/DVD-player outputs, and (with exceptions) signal-processor outputs.

Input Gain/Trim Control. These knobs provide the following gain structure:

- **XLR Microphone / TS ¼-inch Instrument inputs:** 80 dB of variable gain (-15/-30 dB to +65/50 dB)

- **TRS ¼-inch Line Level inputs:** 40 dB of variable gain (-20 dB to +20 dB)

Clip Indicator. All channels feature clip LEDs next to the trim controls. The red clip indicator LED will illuminate when the channel's input signal reaches 0 dBFS. At this level, your mic preamp/line trim signal will exhibit signs of clipping (distortion).

Power User Tip: Never run your input levels higher than the channel inputs can handle. If you overdrive the analog-to-digital converters, it will cause digital distortion (digital clipping), which sounds terrible. The XMAX™ preamps in your AudioBox 1818 VSL provide plenty of headroom; take advantage of it.

Phones. The Phones knob controls the volume of the headphone output on the front of the unit. The headphone amplifier is quite powerful, and the volume goes to 11, so *use the maximum setting with extreme caution.*

Main. The Main knob controls the output level for the Main Outputs on the back of the AudioBox 1818VSL, with a range of -80 dB to 0 dB.

USB Sync LED. This LED will illuminate blue when the AudioBox 1818VSL is properly powered and synced to a USB 2.0 connection. It will flash blue and red when the unit is properly powered but no USB connection is detected.

¼-inch Phones Jack. This is where you connect headphones to your AudioBox 1818VSL.

Main Output Meters. These meters display the signal level going to the Main and Line 1/2 Outputs on the back panel of the AudioBox 1818VSL. The level set by the Main Output knob on the front panel has no effect on the metered level.

Power User Tip: Because Outputs 1 and 2 share the same signal source, you can use these meters to monitor that output pair as well.

2.2   Rear-Panel Connections
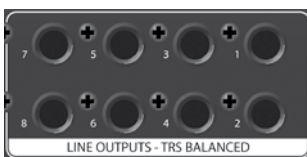
Line Outputs. The AudioBox 1818VSL has eight line outputs to route to external devices, such as headphone amps and DJ mixers. Outputs 1 and 2 share their playback streams with the Main Outs. Outputs 7 and 8 share their playback streams with the headphone output. Outputs 3 through 6 have independent playback streams.

Main Out. These are the main outputs for the AudioBox 1818VSL. The output level of the Main Outs is controlled by the Main volume knob on the front of the unit.
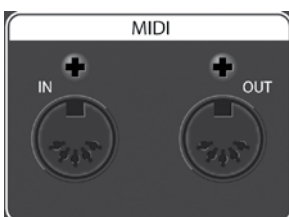
MIDI I/O. MIDI stands for "Musical Instrument Digital Interface." MIDI inputs and outputs allow connection to, and communication with, external MIDI equipment. One function of these ports is MIDI sequencing but the MIDI protocol can be used for much more than instruments and sequencing.

NOTE: MIDI is not audio but is frequently used to trigger or control an audio source (such as a plug-in or synthesizer). It's important to ensure that your MIDI data is correctly sent and received by the appropriate hardware or software devices. If the devices generate audio, you may also need to return the audio to an AudioBox 1818VSL input channel. Please consult the User's Manuals of your MIDI devices for help with MIDI setup and usage.

S/PDIF In and Out Jacks. The S/PDIF standard allows two channels of audio to be transmitted at rates up to 24 bit, 96 kHz. The S/PDIF I/O also allows the AudioBox 1818VSL to send and receive word clock to external digital devices.

Power User Tip: Remember to set "S/PDIF" as the Clock Source and set the sample rate to correspond to the external device in the AudioBox VSL Setup Tab (Windows) or in Audio MIDI Setup (OS X) when using the S/PDIF input for external sync.

ADAT–S/MUX Input and Output. These are ADAT–S/MUX connections for your external digital devices. When recording or playing back at 44.1 kHz or 48 kHz, the ADAT input and output will stream 8 channels. When recording or playing back at 88.2 kHz or 96 kHz, each connection will stream 4 channels. The ADAT–S/MUX I/O also allows the AudioBox to send and receive word clock to external digital devices.

Power User Tip: Remember to select "ADAT " as the Clock Source and set the sample rate to correspond the external device in the AudioBox VSL Setup Tab (Windows) or in Audio MIDI Setup (OS X) when using the ADAT input for external sync.
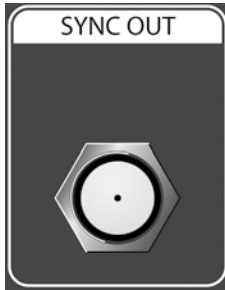
BNC Sync Output. This is a 75Ω word clock output that allows the AudioBox 1818VSL to transmit word clock information to other digital audio devices.

Power User Tip: A 75Ω BNC word clock cable is required to achieve proper sync.

USB 2.0 Port. This is where you connect the USB cable from your AudioBox 1818VSL to your computer. The AudioBox 1818VSL must be connected to a USB 2.0 port to work properly.

Power Input. This is where you connect the power supply for your AudioBox 1818VSL. The AudioBox 1818VSL cannot be bus powered.

Power Switch. This is the On/Off switch for your AudioBox 1818VSL.

2.3   **Typical Band Hookup Diagram**
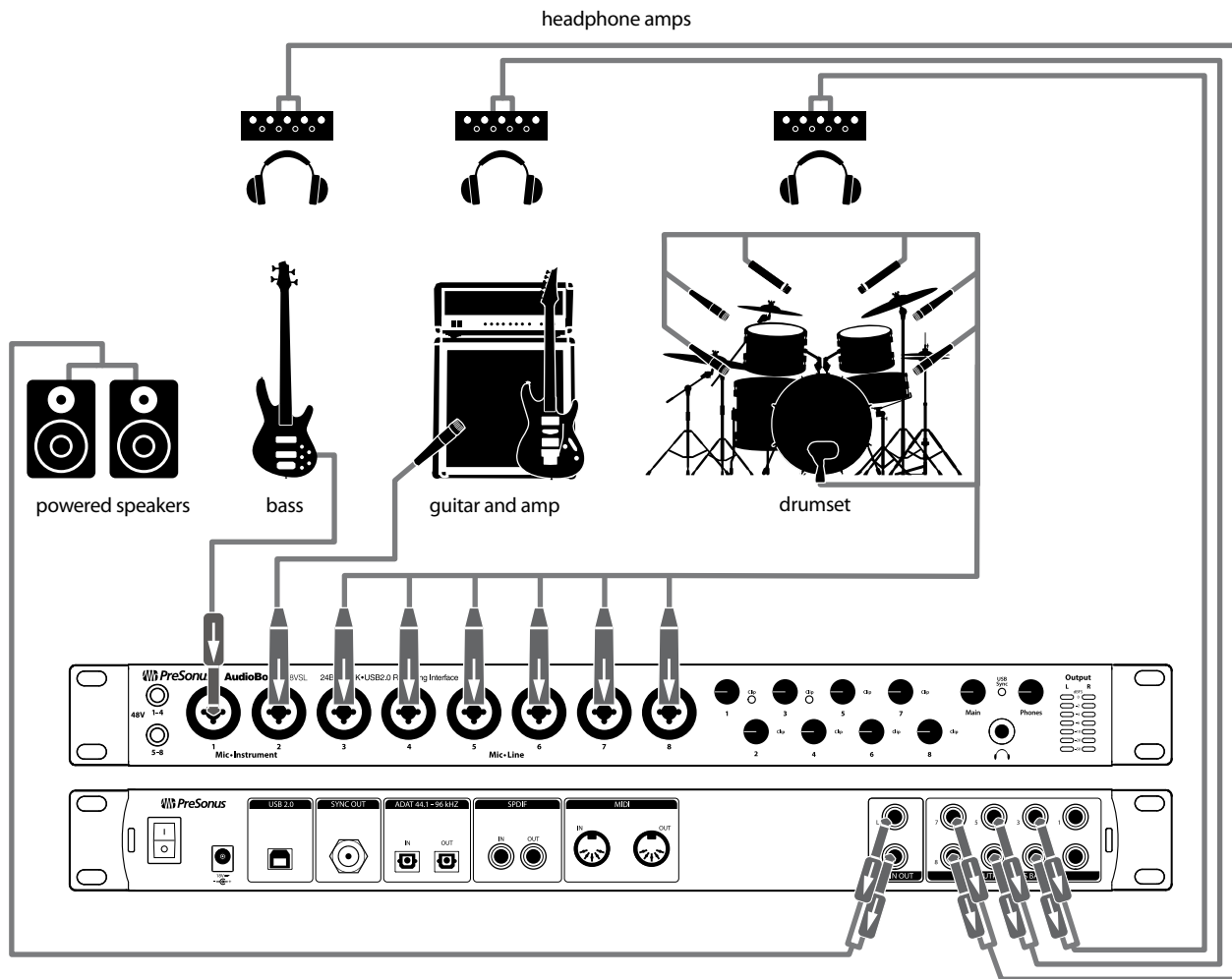
headphone amps

powered speakers | bass | guitar and amp | drumset

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

2.4   Speaker-Switching Hookup Diagram

studio monitors   studio monitors   monitor station
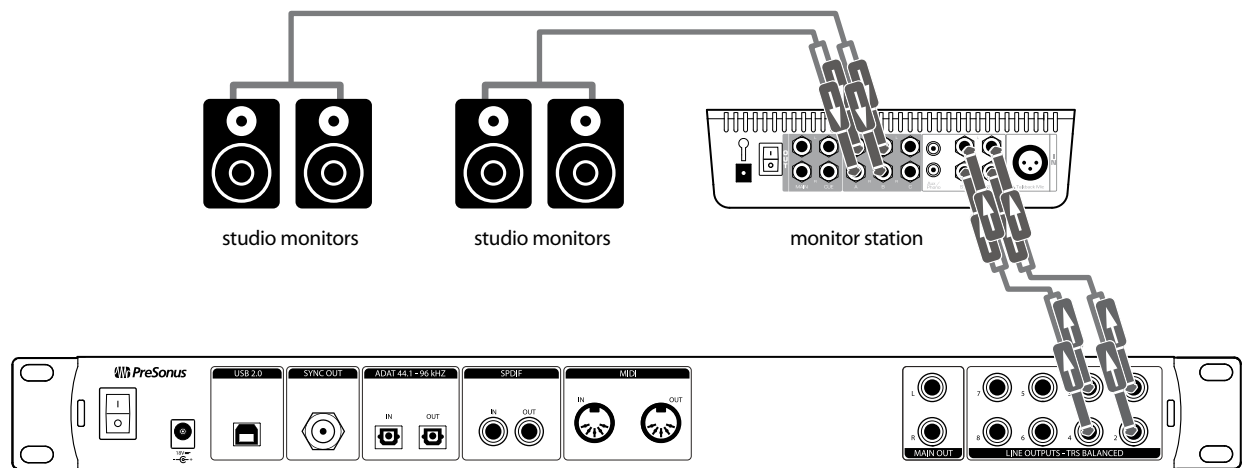
3.0   Connecting to a Computer

Your AudioBox 1818VSL is a rugged USB 2.0 audio interface with flexible monitoring control and professional audio tools.

3.1   System Requirements

Below are the minimum computer-system requirements for PreSonus Studio One Artist and Virtual StudioLive software.*

Macintosh

- **Operating Systems:**

  - Mac OS X 10.6.2 (Snow Leopard) or higher, including Mac OS X 10.7 (Lion)

- **Hardware:**

  - Minimum: Intel Core 1 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 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 for updates.

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

Overview

Hookup

Connecting
to a
Computer

Software: Virtual StudioLive
and Studio One Artist

Tutorials

Technical
Information

Troubleshooting
and Warranty

3.2   Installation for Windows

After you insert the Installation CD into your CD-ROM drive, the AudioBox 1818VSL installer will take you through each step of the installation process. Please read each message carefully to ensure the AudioBox 1818VSL driver is properly installed. It is especially important that you not connect your AudioBox 1818VSL too soon!

We made the AudioBox 1818VSL installer as simple and easy to follow as possible. Before beginning the installation setup, please close all applications, including antivirus software. Do not connect your AudioBox 1818VSL to your computer until prompted to do so by the installer. After the installation is successfully completed, don't forget to reenable your antivirus protection!

Windows Vista Users: If you see any Windows Security alerts, click "Install this driver software anyway."

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, the Found New Hardware wizard will launch. Follow the recommended steps. Your AudioBox 1818VSL is now synced to your computer and ready to use!

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

3.3   Installation for Mac OS X

After inserting the Installation CD into your disc drive, browse the disc, and run the AudioBox 1818VSL 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 AudioBox too soon.
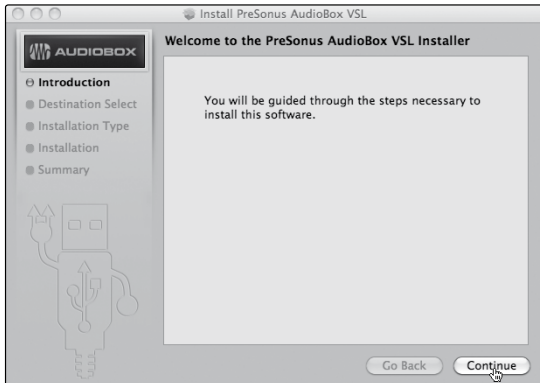
1. You will be directed to the Welcome screen. Click "Continue" and follow the onscreen instructions.
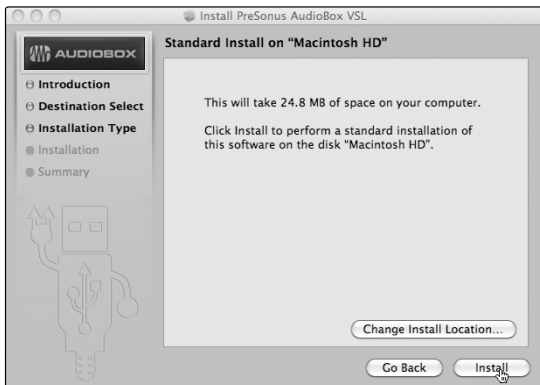
2. You will be directed to choose the hard drive onto which you want to install the drivers and Virtual StudioLive software. You must choose your system drive; storage drives and partitions cannot be used as hosts for the driver.

    When 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."
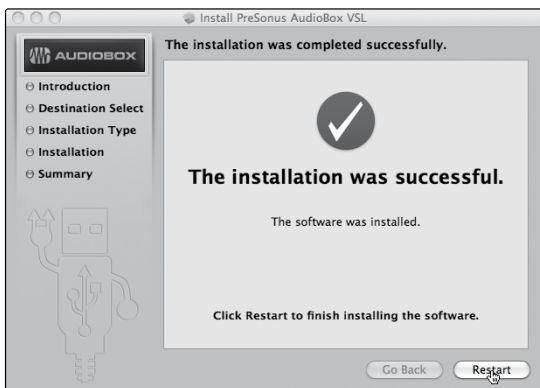
3. When installation is completed, your computer will restart. Once it has rebooted, connect your AudioBox 1818VSL to your Mac with a USB cable and power it on. You are now ready to use your AudioBox 1818VSL!

3.4   **Using the AudioBox 1818VSL with Popular Audio Applications**

Complete setup instructions for Studio One Artist and a brief tutorial on its features can be found in Section 4.2 of this manual. However, you can use your AudioBox 1818VSL 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 AudioBox 1818VSL driver as the audio-device driver for your software.

Below are basic driver-setup instructions for four 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 AudioBox 1818VSL** from the device menu.

6.   You will be asked if you'd like to relaunch Logic. Click "**try (re)launch**."

7.   Your **AudioBox 1818VSL** features custom I/O labels for faster workflow. 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 AudioBox 1818VSL. When you are done, close this window.

9.   You are now ready to use your AudioBox 1818VSL.

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 AudioBox 1818VSL** from the **ASIO Driver** dropdown list.

5.   Click "**Switch**" to begin using the AudioBox 1818VSL 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." *(Using WDM, rather than ASIO, for pro-audio applications is not recommended.)*

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 AudioBox 1818VSL."**

8. Go to **Options | Audio**... and click on the **General** tab.

9. Set the **Playback Timing Master** to "**PreSonus AudioBox... DAW Out 1**."

10. Set the **Recording Timing Master** to "**PreSonus AudioBox... Mic/Inst 1**."

Ableton Live 5+

1. Launch **Ableton Live**

2. Go to **Options | Preferences | Audio**

3. Choose **Driver Type: Asio | Audio Device: ASIO PreSonus AudioBox 1818VSL**

4. Go to **Input Config: Enable** and select the desired Input channels.

5. Go to **Output Config: Enable** and select the desired Output channels.

6. You may now select the AudioBox 22/44VSL's inputs and outputs for each track created in Live.

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

4.0   Software: Virtual StudioLive and Studio One Artist

Your AudioBox 1818VSL comes with a powerful software bundle that provides everything you need to record, produce and mix an album. This software package, including Virtual StudioLive™ (VSL) and Studio One™ Artist, is designed to help you get the most from your studio.

4.1   Virtual StudioLive

Virtual StudioLive is a powerful monitoring software that essentially turns your AudioBox 1818 into a mini-StudioLive™ 16.0.2. VSL provides control of channel, effects bus, and Main bus levels; Fat Channel parameters; and effects. VSL also includes a librarian, allowing you to easily manage your presets and mix Scenes.

Controlling 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, and EQ presets. You can even drag presets out of the browser and exchange them with other AudioBox 22/44/1818VSL owners.

4.1.1   Virtual StudioLive: Setup

Setup

The VSL Setup tab lets you set basic driver parameters such as buffer size and sample rate and enables you to adjust the user preferences in VSL. To access the Setup options, click on the Setup Tab button at the top of the VSL window.
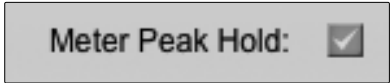
Meter Decay   **Sets the Response Time for the Meters in the Device Window.**

VSL gives you the option to set the response time for for the channel meters. By default, this is set to Normal. Enable Slow Meter decay to more accurately meter the peaks and falls of the signal. Enable Fast Metering to monitor your signal in real time.

Meter Peak Hold   **Sets the Clip Indicator to Remain Illuminated.**

When this option is enabled, the clip indicator on any given channel meter in VSL will remain illuminated until manually cleared, even if the signal is no longer clipping. The meters will also display the last peak value until a new peak is established. To clear the clip indicator, click on it with your mouse.

Meter Post Fader   **Sets the Channel Metering to Post Fader Signal**

When this option is enabled, the channel meters in VSL will display the signal level after it has been boosted or attenuated by the channel fader. By default, all metering in VSL is pre-fader.

Run at Startup (Window Only)

Launches the Virtual StudioLive Application Automatically on Startup.

When this is enabled, VSL will automatically launch each time you boot your Windows.

On the Mac, you can set this up first by dragging VSL to your dock, then by right-clicking it and selecting Options>Open at Login.

Sample Rate Selector (OS X Only)

Changes Sample Rate.

Sets the sample rate to 44.1, 48, 88.2, or 96 kHz.

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.
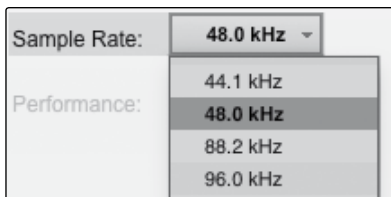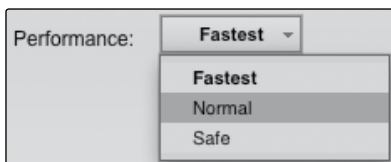
Changing the sample rate will produce a momentary popping noise. Because of this, it is recommended that you turn the Main and Phones volumes down on your AudioBox prior to changing the sample rate.

On Windows systems, the sample rate is set from inside your host audio application.

Performance

Changes VSL's Input Monitoring Buffer Size.

VSL offers three Performance modes for input monitoring: Fastest, Normal, and Safe. These modes allow you to adjust the VSL's buffer to either lower the input latency or increase it depending on your computer's speed.

- *Fastest:* Sets VSL's input buffer to 1 millisecond.

- *Normal:* Sets VSL's input buffer to 2 milliseconds.

- *Safe:* Sets VSL's input buffer to 4 milliseconds.

For most machines, Normal mode will provide excellent stability with nominal latency.

Buffer Size Selector (Windows only)

Changes the Buffer Size.
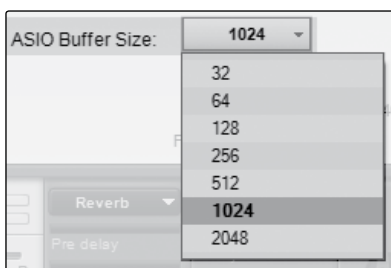
You can set the buffer size from 32 to 2048 samples. The buffer size determines the roundtrip time it takes audio data to be converted from analog to digital and back to analog. As a general rule, the higher the buffer size, the better the system performance, but the less playable virtual instruments 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.

Power User Tip: The buffer size for monitoring in VSL is set by the Performance Mode. The buffer size you set in VSL (Windows) or in your audio application (Mac OS X), applies only to the driver's performance in your ASIO or Core Audio application.
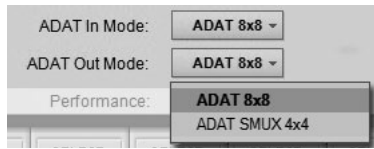
Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

Overview

Hookup

Connecting
to a
Computer

Software: Virtual StudioLive
and Studio One Artist

Tutorials

Technical
Information

Troubleshooting
and Warranty

ADAT Mode (Window Only)   **Sets ADAT I/0 to ADAT or S/MUX Mode**

When recording or playing back at 44.1 kHz or 48 kHz, the ADAT I/O on the AudioBox 1818VSL will provide 8 audio channels over the optical inputs and outputs. When recording or playing back at 88.2 kHz or 96 kHz, these connections will provide 4 audio channels.

In order to record or playback at 88.2 kHz or 96 kHz over ADAT, you must set the correct Input and Output ADAT Modes. AudioBox VSL provides two modes:

ADAT 8x8: Channels 1-8 (44.1/48 kHz)

ADAT SMUX 4x4: Channels 1-4 (88.2/96 kHz)

If you are recording at 88.2 kHz or 96 kHz, you must set the ADAT In Mode to "ADAT SMUX 4x4" in addition to setting the correct sample rate.

If you are outputting audio through the optical outputs on your AudioBox 1818VSL at 88.2 kHz or 96 kHz, you must set the ADAT Out Mode to "ADAT SMUX 4x4" .

By default, these Modes are set to ADAT 8x8. As long as you are using your AudioBox 1818VSL at 44.1 kHz or 48 kHz, you do not need to make any changes.

Power User Tip: Failure to set the correct ADAT Mode for the sample rate being used will result in audio artifacts. Setting ADAT mode is not necessary when using Analog or S/PDIF I/O at higher sample rates.

ADAT Mode (OS X Only)   **Sets ADAT I/0 to ADAT or S/MUX Mode**

When recording or playing back at 44.1 kHz or 48 kHz, the ADAT I/O on the AudioBox 1818VSL will provide 8 audio channels over the optical inputs and outputs. When recording or playing back at 88.2 kHz or 96 kHz, these connections will provide 4 audio channels.

In order to record or playback at 88.2 kHz or 96 kHz over ADAT, you must set the correct Input and Output ADAT Modes. These settings are available in OS X's Audio MIDI Setup Utility.

By default, these Modes are set to 18ch -24bit. As long as you are using your AudioBox 1818VSL at 44.1 kHz or 48 kHz, you do not need to make any changes.

To record or playback at 88.2 kHz or 96 kHz, open the Audio MIDI Setup Utility on your Mac (Applications/Utilities/Audio MIDI Setup).
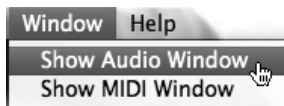
Once Audio MIDI Setup opens, click on the Audio window. If the Audio window doesn't open when Audio MIDI Setup first opens, go to Window and select "Show Audio window."

Select "AudioBox 1818VSL" from the audio device list on the left. To the left of the Sample Rate selection menu, you will find the ADAT Mode menu:

18ch -24bit: Channels 1-8 (44.1/48 kHz)

14ch - 24bit: Channels 1-4 (88.2/96 kHz)
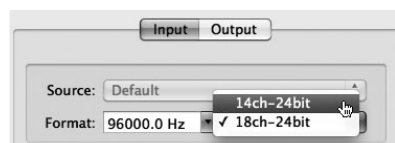
If you are recording at 88.2 kHz or 96 kHz, you must set the ADAT Input Mode to "14ch -24bit" in addition to setting the correct sample rate.
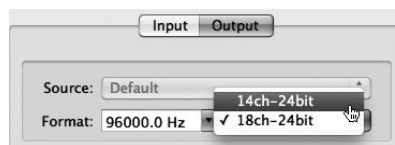
If you are outputting audio through the optical outputs on your AudioBox 1818VSL at 88.2 kHz or 96 kHz, you must set the ADAT Output Mode to "14ch - 24bit".

Power User Tip: Failure to set the correct ADAT Mode for the sample rate being used will result in audio artifacts. Setting ADAT mode is not necessary when using Analog or S/PDIF I/O at higher sample rates.

**Clock Source Select
(Windows Only)**
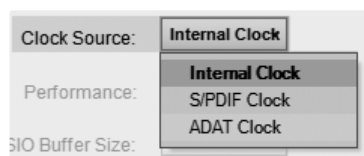
Changes the Clock Source for the AudioBox 1818VSL.

The clock-source setting determines the port from which the AudioBox 1818VSL is receiving word clock information. This keeps the AudioBox in sync with other digital devices. You can choose between Internal, S/PDIF, or ADAT.

In general, you will want the AudioBox 1818VSL to be your master clock, in which case you should set the clock to "Internal" This setting also means that your AudioBox is generating word clock and sending this information out of its digital outputs.

If you want the AudioBox to receive sync from an external device, choose the digital input to which the external device is connected (S/PDIF or ADAT). The AudioBox 1818VSL's sync light will flash from blue to red. When the AudioBox is in sync, the light will be blue.

Power User Tip: When slaved to an external clock, the AudioBox 1818VSL will not automatically change its sample rate to match the external clock. As a result, it may fail to sync to the clock source. If your AudioBox is not syncing to an external source, make sure that both your master device and the AudioBox 1818VSL are set to the same sample rate.

**Setting Clock Source
(OS X)**

Using Audio MIDI Setup to Set the Clock Source for the AudioBox 1818VSL.

The clock-source setting determines the port from which the AudioBox 1818VSL is receiving word clock information. This keeps the AudioBox in sync with other digital devices. You can choose between Internal, S/PDIF, or ADAT.

In general, you will want the AudioBox 1818VSL to be your master clock, in which case you should set the clock to "Internal." This setting also means that your AudioBox is generating word clock and sending this information out of its digital outputs.

If you want the AudioBox to receive sync from an external device, open the Audio MIDI Setup Utility on your Mac (Applications/Utilities/Audio MIDI Setup).

Once Audio MIDI Setup opens, click on the Audio window. If the Audio window doesn't open when Audio MIDI Setup first opens, go to Window and select "Show Audio window."

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

Overview

Hookup

Connecting
to a
Computer

Software: Virtual StudioLive
and Studio One Artist

Tutorials

Technical
Information

Troubleshooting
and Warranty
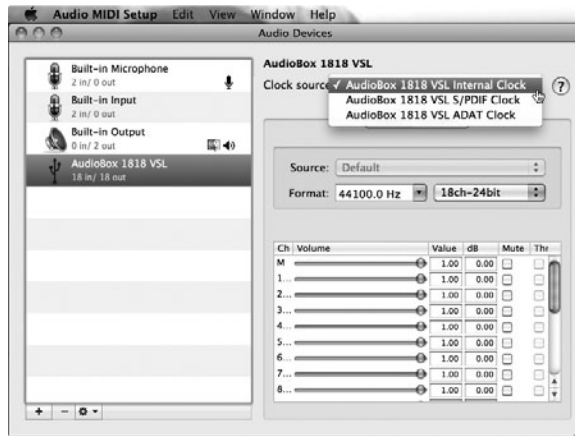
Select "AudioBox 1818VSL" from the audio device list on the left. Using the Clock Source pull-down menu, choose the digital input to which the external device is connected (S/PDIF or ADAT). The AudioBox's sync light will flash from blue to red. When the AudioBox is in sync, the light will be blue.

Power User Tip: When slaved to an external clock, the AudioBox will not automatically change its sample rate to match the external clock. As a result, it may fail to sync to the clock source. If your AudioBox is not syncing to an external source, make sure that both your master device and the AudioBox are set to the same sample rate.

4.1.2   Virtual StudioLive: 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 your Scenes, Fat Channel presets, and FX presets that are saved on your computer. You can also create new settings from this window. Simply drag-and-drop a Scene or preset to load it on your AudioBox 1818VSL.

Browser Tab Buttons   **Displays the Different Preset Categories.**

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. A Scene is like a snapshot of your mix. It stores all Fat Channel parameters for every input and bus, as well as fader positions, the aux and effects mixes, FX bus settings, and channel mutes and solos. Creating a Scene requires simply dialing in a mix that you would like to use at a later date and saving it.

- **FAT CH.** Displays stored Fat Channel presets. VSL comes equipped with Fat Channel presets that have been custom-designed by professional audio engineers. These presets can be altered, renamed, and overwritten; you can build your own nearly limitless custom library of channel-strip settings.

- **FX.** Displays stored effects presets. VSL comes loaded with 50 custom reverb and delay presets designed by PreSonus. In addition to these presets, you can build your own nearly limitless custom effects library. The factory presets can be altered, renamed, and overwritten.

Add New Button   **Creates a New Scene or Preset.**

Just below the Browser tabs, you will see the Add New button. Clicking this button will immediately create a new Scene or preset.

Overview

Hookup

Connecting
to a
Computer

Software: Virtual StudioLive
and Studio One Artist

Tutorials

Technical
Information

Troubleshooting
and Warranty

Remove Button   **Deletes a Stored Scene or Preset.**

Right next to the Add New button, you will find
the Remove button. Clicking this button will delete
the currently selected, stored Scenes or presets.

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.

4.1.3   Virtual StudioLive: Overview and Fat Channel Tab

VSL provides you with two ways to look at your mix and to
dial in FX settings: The Overview and Fat Channel Tabs.

The Overview Tab provides you with a complete graphical
representation of your Virtual StudioLive mix environment.

The Fat Channel Tab opens a detailed overview of the Fat Channel
parameters for the currently selected channel. The Fat Channel
Tab can be opened either from the tab button at the top of the VSL
window or by simply double-clicking on any of the microviews.

From either tab you can:

- Adjust levels for all analog and digital inputs, all eight playback
  streams, both effects buses, the main output level of the
  entire mix, and the output levels of each monitor mix.

- Adjust the panning for the analog and digital
  inputs and playback streams

- Adjust the send level for both the analog and digital
  inputs and playback streams to each FX bus

- Adjust the send level for each channel to
  outputs 3 and 4, 5 and 6, and 7 and 8

- Adjust the parameters for the FX preset on each FX bus

- Solo and mute each analog input and playback stream

- Meter the inputs and outputs

From either the Overview or the Fat Channel tab, you can
access the show/hide channel buttons to help keep your
mix organized. You can hide the following channel groups
from your mix: ADAT, DAW Returns, and Outputs.

Show/Hide ADAT Inputs Button   **Displays/Removes ADAT Input Channels from the Mix.**

Adat

When this button is enabled, the ADAT input channels will be displayed in the mixer. If you disable this button, you will no longer see the ADAT channels in the mixer; however, any changes or routing that have been made will still be active, and these channels will still be heard in your mixes.
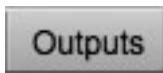
Show/Hide DAW Returns Button   **Displays/Removes DAW Playback Streams from the Mix.**

Daw

When this button is enabled, DAW Playback Streams 1 through 8 will be displayed in the mixer. If you disable this button, you will no longer see these channels in the mixer; however, any changes or routing that have been made will still be active, and these channels will still be heard in your mixes.

Please Note: From VSL, you only have access to DAW Returns 1-8. The ADAT and S/PDIF output streams can only be accessed from your DAW.

Show/Hide Outputs Button   **Displays/Removes Output Channels from the Mix.**

Outputs

When this button is enabled, the stereo output channels will be displayed in the mixer. If you disable this button, you will no longer see the stereo outputs in the mixer; however, any changes or routing that have been made will still be active, and you will still be able to hear your monitor mixes through all eight outputs.

This button will not remove the Main Output from the mixer. This output is always visible.

Please Note: From VSL, you can create and send mixes to all eight Line Outputs on your AudioBox 1818VSL. Outputs 1 and 2 share a mix with the Main Outputs, so the mix you create for your Main Outputs will be heard through both pairs of outputs. Outputs 7 and 8 share a mix with the headphone output, so creating a mix for Outputs 7 and 8 will send an identical mix stream to the headphone outputs on the front panel of the AudioBox 1818VSL.

4.1.4   VSL Controls: Definitions and Uses

VSL is a powerful application that allows you to create monitor mixes using all of the processing features of a StudioLive 16.0.2 digital mixer. These monitor mixes are not recorded in your host application. It is vital to remember that lowering the channel fader in VSL will not lower the signal in your host application, so it is possible to clip the recording without clipping the monitor mix. You must set the level for the recording using the trim knobs on the face of the AudioBox 1818VSL.

A quick note on playback streams: The channels labeled "DAW" in VSL carry a playback stream from your host application (DAW). Traditionally, if you wanted to route a track in your DAW to a physical output on your interface, you assigned this output in your host application. Because the Virtual StudioLive application provides much more flexible routing, you can now route this same track to one output or every output, by itself or as part of a mix.

VSL On/Off	**Enables / Disables VSL Mixing.**
1818VSL ⏻	When the VSL On/Off button is disengaged, your AudioBox 1818VSL functions as a simple 18x18 audio interface with no monitor mix or DSP capability. If you plan on mixing exclusively in your DAW or do not need to create monitor mixes using your computer, disabling VSL can greatly simplify your work environment.
	When VSL is turned on, you will have access to all its mixing capability on your computer.
Select Button	**Accesses the Fat Channel.**
SELECT	Above each fader in VSL, you will see Select buttons. There is a Select button on each of the analog and digital inputs, each DAW return, on both effects buses, and on the Main output bus. Each of these buttons serves exactly the same purpose: to access the Fat Channel parameters for its channel or bus.
Solo Button	**Turns Soloing On and Off.**
SOLO	This button will solo its channel to the outputs so that only the soloed channel will be audible. 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 and Off.**
MUTE	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 outputs.
Monitor Sends and Pan Controls	**Sets the Level and Panning of Each Channel in the Monitor Mixes.**
	Above each channel are channel sends and pans for each output pair. A channel send is just like a fader on a mixer. It sets the volume level of its channel in the monitor mix. The pan control sets the channel's relative position in the left/right stereo mix. When a pair of channels is stereo linked, the pan control sets the spread of the channels in the left/right stereo mix.
	The top control for each output pair is the send level, and the lower control is the pan control.
FX Sends Controls	**Sets the Level of Each Channel in the FX Mixes.**
A ▬▬▬ B ▬▬▬	The FXA and FXB Send controls sets the level of each channel in the FX Buses. The higher the level, the wetter the processed signal will be.
Channel Pan Control	**Sets the Panning of Each Channel in the Main Mix.**
L ▮ R	The pan control sets the channel's relative positions in the stereo main mix. When a pair of channels is stereo linked, the pan control sets the spread of the channels in the left/right mix.

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

Overview

Hookup

Connecting
to a
Computer

Software: Virtual StudioLive
and Studio One Artist

Tutorials

Technical
Information

Troubleshooting
and Warranty

Fader

Controls the Overall Level of the Channel or Bus.

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

Meter

Displays the Pre-fader Level of Each Channel.

Each channel features a fader for accurate level monitoring. Each meter displays the pre-fader, pre-Fat Channel signal. If the Post button is engaged, the meter will display the post-Fat Channel, pre-fader signal.

In this way, you will always have a visual cue to how hot your recording signal is in your monitor mix.

Stereo Link Button

Joins the Channel into a Stereo Pair.

The Link button joins two channels into a stereo pair. When channels are linked, the volume of both will be controlled with either fader. A stereo link can be enabled when either channel in the pair is selected. When the Stereo Link button is enabled, all dynamics settings, bus assignments, etc., are pasted to the other channel in the pair.

Phase Reverse Button

Reverses the Phase of the Input Channel.

Push this button to invert the phase of the selected channel's signal (that is, to change the phase 180°). The button will illuminate, indicating that 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 accessible from the Main VSL Page and from the Gate Zoom.

Phase Reverse is only available on the analog input channels.

Post Button

Engages Post Fat Channel Recording.

Each of the analog inputs can be recorded pre- or post-Fat Channel EQ and dynamics processing. To record the EQ and dynamics processing on any channel, simply enable the Post button. It will illuminate, indicating that the Fat Channel signal path is being routed to your DAW's recording input. If this mode is not enabled, the signal recorded in your DAW will not include any of the Fat Channel processing (Phase Reverse, Gate, Compressor, EQ, Limiter).

It should be noted that the effects buses cannot be recorded. You can add reverb and delay to your recorded files later, using DAW plug-ins.

The Post button is only available on the Overview Tab and only for the analog and digital input channels.

High-Pass Filter Freq
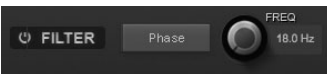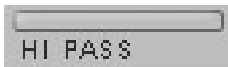
Adjust the High-pass Filter's Cutoff Frequency.

The high-pass filter's cutoff frequency can be set from 18 Hz to 1.3 kHz.

Remember that all frequencies below a high-pass filter's cutoff frequency are attenuated. See Section 5.3.1 for details.

The slope of the high-pass filter is -6 dB/octave.

The high-pass filter is accessible from either the Overview or Fat Channel tabs and is available on all input channels and playback streams.

Gate On/Off Button

Turns the Gate On and Off for the Selected Channel or Bus.

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 Threshold

Sets the Threshold of the Gate for the Selected Channel or Bus.

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

The gate can be controlled either via the Threshold knob or from the Gate Graph.

Compressor On/Off

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

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

25

Auto Mode Button

Enables Automatic Response 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.
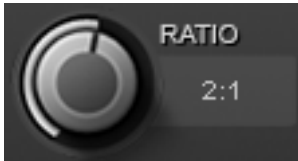
Compressor Threshold

Sets the Threshold of the Compressor for the Selected Channel or 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.

Compressor Ratio

Sets the Compression Ratio for the Selected Channel or Bus.

Compression 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, then 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 the Compressor Attack for the Selected Channel or Bus.

Compressor 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 the Compression Release for the Selected Channel or Bus.

Compression 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.

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

Compressor Makeup Gain
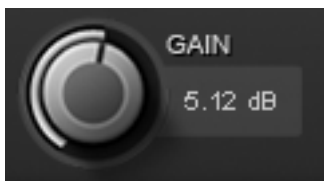
Sets the Amount of Makeup Gain for the Compressor on the Selected Channel or Bus.

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. You can adjust Makeup Gain from 0 dB (no gain adjustment) to +28 dB.

Owner's Manual

Software: Virtual StudioLive and Studio One Artist   4
Virtual StudioLive   4.1

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

Limiter On/Off	**Turns the Limiter for the Selected Channel or Bus On or Off.**
	When the limiter is engaged, this button will illuminate. The threshold for the limiter is set to 0 dBFS. The ratio is ∞:1.
	The limiter is available for all input and output buses.

Low EQ On/Off Button	**Activates Control for the Low Band EQ for the Selected Channel or 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 the Center Frequency for the Low EQ Band.**
	The center frequency is the middle of the passband (the mean) between the lower and upper cutoff frequencies that define the limits of the band.
	You can adjust the center frequency from 36 to 465 Hz.

Low EQ Gain Control	**Sets the Gain Attenuation or Boost of the Center Frequency.**
	This control sets 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 semi-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 consumer audio device. In this mode, the Center Frequency control selects the shelving frequency.

Mid EQ On/Off Button	**Activates Controls for the Mid EQ for the Selected Input or Bus.**
	This button activates the controls for the equalizer's Mid band for the selected input or output. The button will illuminate to indicate control is active.
	The Mid EQ band is available for all input and output buses.

Mid EQ Frequency Control	**Sets the Center Frequency of the Mid EQ.**
	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 380 Hz to 5 kHz.

Mid EQ Gain Control	**Sets the Gain Attenuation or Boost of the Center Frequency of the Mid Band.**
	This control sets the Gain cut or boost at the center frequency of the Mid band. The level of the center frequency can be set between -15 and +15 dB.

Mid Hi Q Button	**Enables a Narrow Bandwidth for the Mid Band EQ on the Selected Input or Bus.**
	Q is the ratio of the EQ band's center frequency to its bandwidth. With a constant center frequency, higher Q values indicate a narrower bandwidth, so Q is often equated with bandwidth. By default, the Q is set to a value of 0.55. When the Hi Q button is engaged, the Q setting will be increased to 2.0, thus narrowing the bandwidth to provide more precise control.

High EQ On/Off Button	**Activates Control for the High EQ for the Selected Input or Bus.**
	This button activates control of the High EQ 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 the Center Frequency of the High EQ.**
	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 1.4 to 18 kHz.

High EQ Gain Control	**Sets the Gain Attenuation or Boost at the Center Frequency of the High EQ Band.**
	This control sets the gain cut or boost at the center frequency of the High EQ band. The level of the center frequency can be set between -15 and +15 dB.

High Shelving EQ Button	**Turns on the High Shelving EQ for the Selected Input or Bus.**
	When the Shelf button is not engaged, the High EQ band is semi-parametric. Enabling the Shelf button turns the High band into a high shelving EQ that alters, by a fixed amount, a band of high frequencies at and above a user-selected shelving frequency.
	A high shelving EQ is like a treble-control knob on a consumer audio device. In this mode, the Center Frequency control selects the shelving frequency.

4.1.5   Loading Scenes and Presets from VSL

To load a Scene or preset from the VSL Browser window, you simply select it and drag it over the mixer or channel on which you wish to load it. 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 Fat Channel tab.

Loading a Scene

To load a new Scene, select it from the Browser window and drag it over the VSL mixer in either the Overview or the Fat Channel tab. The window will gray out, indicating that a new Scene is 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 Microviews, it will load only that component; for example, if you drag a preset over the Gate Microview, only the Gate will be loaded.

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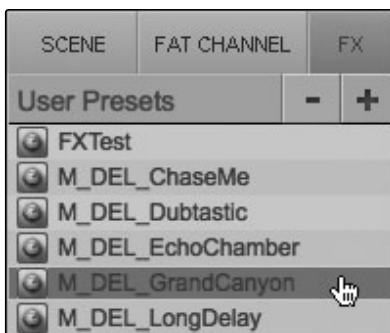
To load an FX preset, select it from the Browser window, and drag it over any part of the desired FX bus in the Master section of the Overview tab. Once the preset is loaded, you can use the FX Type menu to change the effect and create new presets.

Overview

Hookup

Connecting
to a
Computer

Software: Virtual StudioLive
and Studio One Artist

Tutorials

Technical
Information

Troubleshooting
and Warranty

Overview

Hookup

Connecting
to a
Computer

Software: Virtual StudioLive
and Studio One Artist

Tutorials

Technical
Information

Troubleshooting
and Warranty

4.1.6   Using VSL to Create a Monitor Mix

Virtual StudioLive allows you create a monitor mix with the same sound quality and flexibility you'd get from a StudioLive 16.0.2 digital mixer. By adding reverb, delay, and Fat Channel dynamics processing and EQ to your monitor mix, you can enhance the quality of the performer's recording experience and, by extension, the quality of their performance.

A monitor mix is a dedicated mix that is usually routed to a performer's headphones. This mix is separate from the mix that is recorded. It allows you, the engineer, to leave your recording mix intact, while providing the artist with the mix that they need to achieve their best performance, saving hours of retakes and frustration.

STEP 1   Creating a Mix for the Main and Headphone Outputs
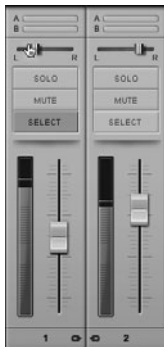
Using VSL to create a mix for your Main and Headphone outputs is like using a mixer. Raise the fader for each channel to raise its volume in the mix and use the pan control to adjust the source signal's position in the stereo field. To adjust the overall mix level, raise or lower the Main fader in VSL or use the Main knob on the front of your AudioBox 1818VSL.

Power User Tip: Raising a channel's volume is not always the best way to make it louder in the mix. By adjusting the pan position or adding some EQ, you can raise the perceived loudness of the channel without touching the fader.

STEP 2   Creating a Mix for Outputs 3/4, 5/6, and 7/8

The AudioBox 1818VSL also allows you to create a separate mix for Outputs 3/4, 5/6, and 7/8. This done by using *channel sends* that enable you to set the levels for each channel that is routed to a specific output bus. For any given channel, you can create one mix for your Main and Line 1 and 2 outputs and create entirely separate mixes for Outputs 3/4, 5/6, and 7/8 and the headphone outputs.

It is important to mention that each of these outputs are stereo-linked, so you will have one send level for each channel but a given channel's level will be the same at both outputs in the pair. The top slider for each pair sets the level for each channel, while the lower slider is the pan control for each channel.

STEP 3   Adding Fat Channel Dynamics

The AudioBox 1818VSL provides a Fat Channel to process each input and DAW channel, the two effects buses, and the analog outputs in VSL. Each Fat Channel consists of a Downward Expander, a Compressor, a 3-band semi-parametric EQ and a Limiter. For more information on how these dynamics processors effect audio, please review Sections 5.2 and 5.3.

The Overview tab in VSL provides Microviews of the Gate, Compressor, and Limiter. To make changes to one of these processors, double-click on its Microview. This will open the Fat Channel tab and focus on the chosen dynamics processor.

From the Fat Channel tab, you will see a zoomed in view of all the Fat Channel controls.
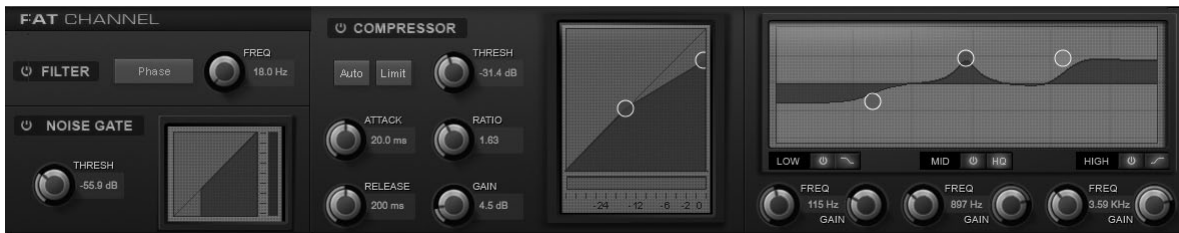
STEP 4: Creating an Effects Mix

Once you have your levels set and your Fat Channel processing dialed in for each input, you can add reverb and delay to your monitor mixes.

To begin, you must first assign an effects preset to a bus. Click on the FX tab and select one of the presets. To use that preset, simply drag-and-drop it onto one of the FX buses (A or B).
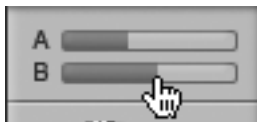
An effects mix is created by using a channel send. The A and B channel sends control the volume settings for each channel to FX buses A and B, respectively.

To control the overall amount of the effects you will hear in your mix, use the FX A and FX B faders. These will raise and lower the 100% wet effects mix.

It is important to mention that both FX bus A and FX bus B must be patched to an output pair in order to hear them in the mix. On each of the output buses, you will see buttons labeled FXA and FXB. These buttons route the output of the FX buses to the physical outputs on your interface. By default, both FX buses are routed to all outputs.

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

4.2   Studio One Artist Quick Start
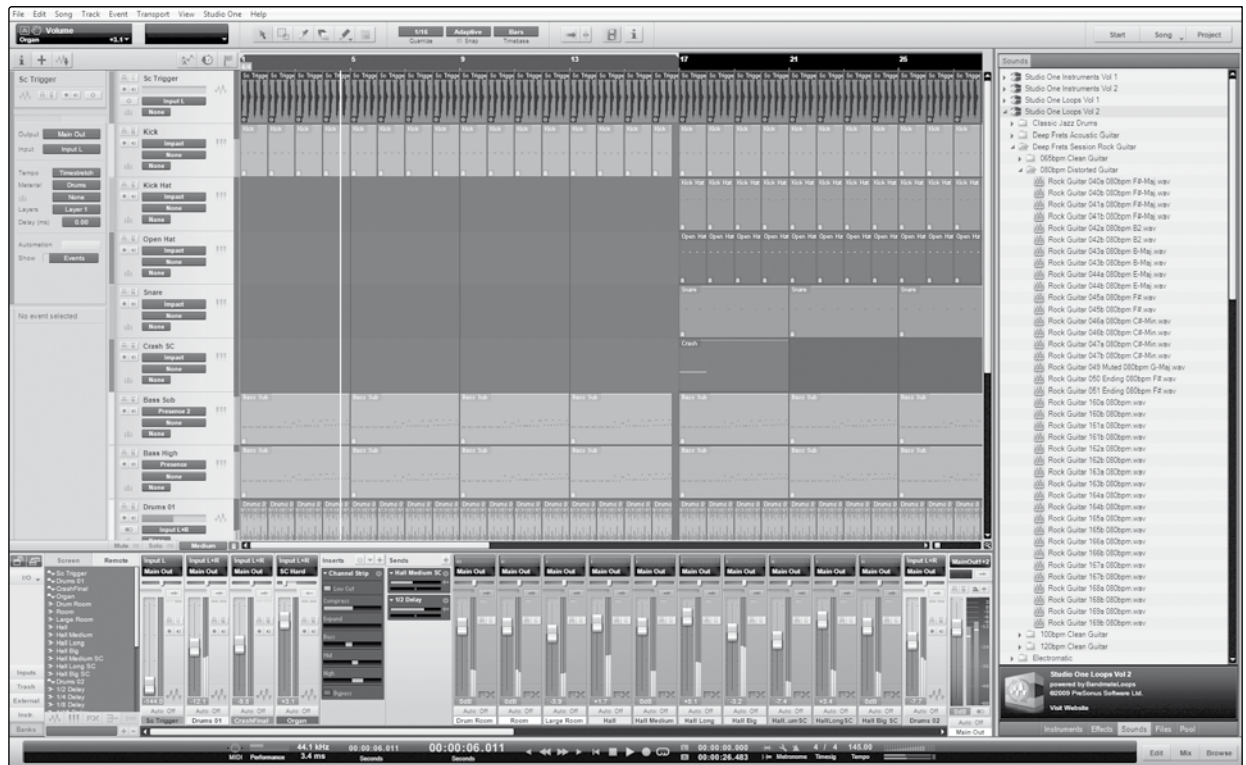
Your AudioBox 1818VSL comes with Studio One Artist recording and production software. Whether you are about to record your first album or your fiftieth, Studio One Artist provides you with all of the tools necessary to capture and mix a great performance. As a valued PreSonus customer, you are also eligible 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.

4.2.1   Installation and Authorization

Once you have installed the AudioBox 1818VSL drivers and connected your AudioBox 1818VSL to your computer, you can use the included PreSonus Studio One Artist music-production software 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.
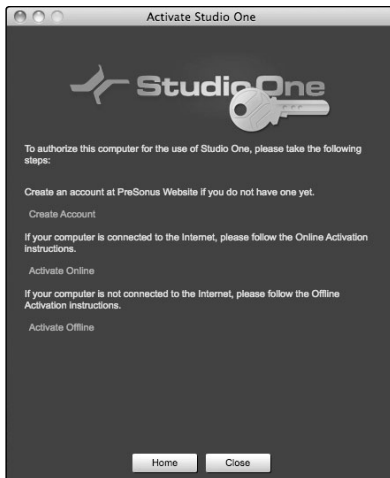

Running the Studio One Installer

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 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
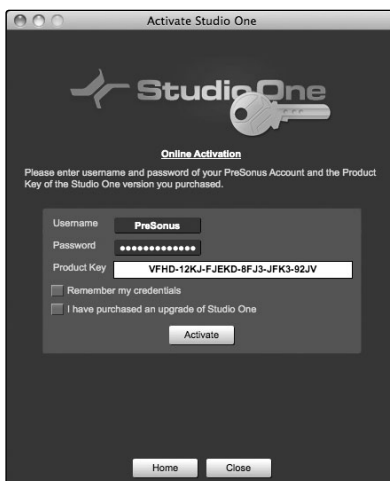
- Now that you have created a user account, you can activate your copy of Studio One Artist. Launch Studio One Artist, and the Activate Studio One menu will appear.

- 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.

Activating Studio One Artist Offline
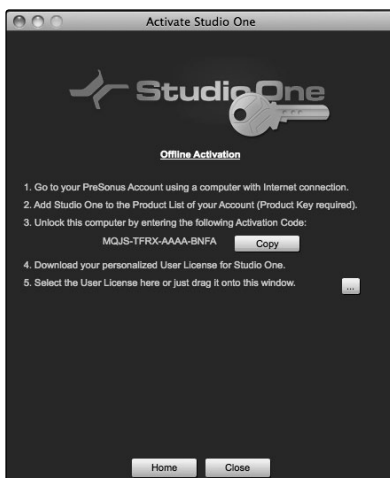
- Once you have created a user account, launch Studio One Artist. From the Activate Studio One Menu, click on the Activate Offline link. Follow the instructions to log in to your previously created user account, register the product, and obtain a license file.

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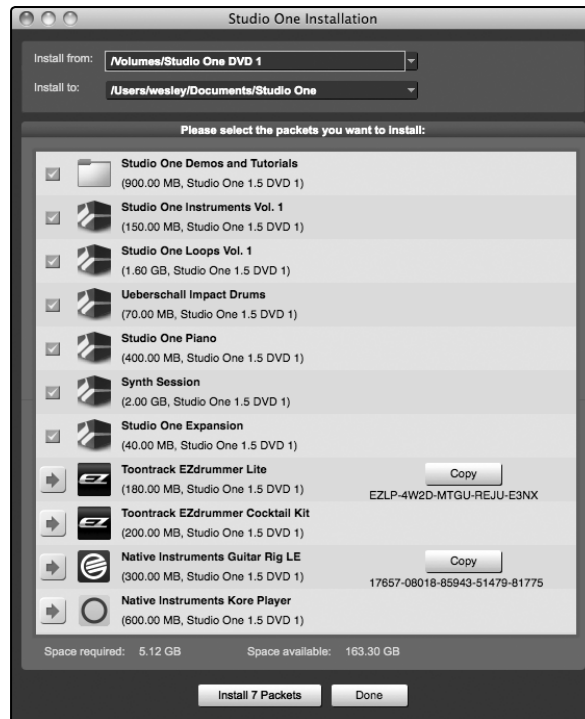
Studio One Artist comes bundled with an array of demo and tutorial material, instruments, loops, samples, and other third-party content. The Studio One Artist bundle includes all that you need to begin producing music.

- 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.

Installing Third-Party Content

To Install any of the third-party content, click on the arrow button next to its name in the content list (Toontrack EZdrummer Lite, Toontrack EZdrummer Cocktail Kit, Native Instruments Guitar Rig LE, Native Instruments Kore Player, and Native Instruments Kore Sounds Vol. 1).

Notice that serial numbers are listed next to Toontrack EZdrummer Lite and Native Instruments Guitar Rig LE. You will need these numbers to authorize these plug-ins the first time you open them.

When you have finished installing content, click the "Done" button.

4.2.2   Enabling the Audio Driver

Studio One Artist was designed with PreSonus interfaces in mind, so AudioBox 1818VSL 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
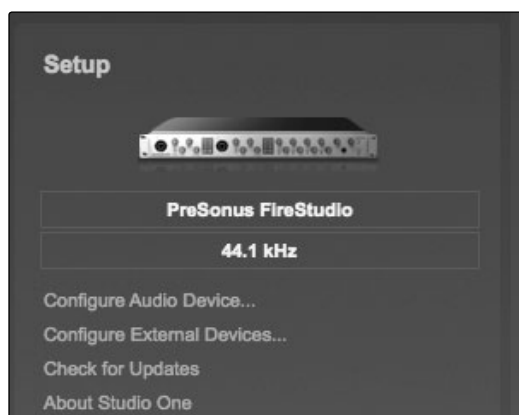
Shows Active Audio Driver and Sample Rate and Provides Quick Links to Configure Audio and MIDI.

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.

If you do not see "PreSonus AudioBox" 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 AudioBox." Click the Apply button and then OK.

After you have verified that the PreSonus AudioBox 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 4.3.4.

4.2.3   Configuring 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 guide 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.

If you are using a third-party MIDI interface or USB MIDI-controller keyboard, you must install the drivers for these devices before beginning this section. Please consult the documentation that came with your MIDI hardware for complete installation instructions.

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

Overview

Hookup

Connecting
to a
Computer

Software: Virtual StudioLive
and Studio One Artist

Tutorials

Technical
Information

Troubleshooting
and Warranty

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 controller and tone-generation functions 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 setting up a new Song for recording, take a moment to configure external devices.

2. Connect the MIDI Out of your external MIDI controller to a MIDI In on your AudioBox 1818VSL or other MIDI interface. If you are using a USB MIDI controller, connect it to your computer and power it on.

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

4. Click the Add button.

5. The Add Device window will launch.

6. 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.

7. 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.

8. 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, 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.

✓ Default Instrument Input

9.   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 that 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 and virtual instruments, 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.

2.   In the External Devices window, click the Add button.

Overview

Hookup

Connecting
to a
Computer

Software: Virtual StudioLive
and Studio One Artist

Tutorials

Technical
Information

Troubleshooting
and Warranty
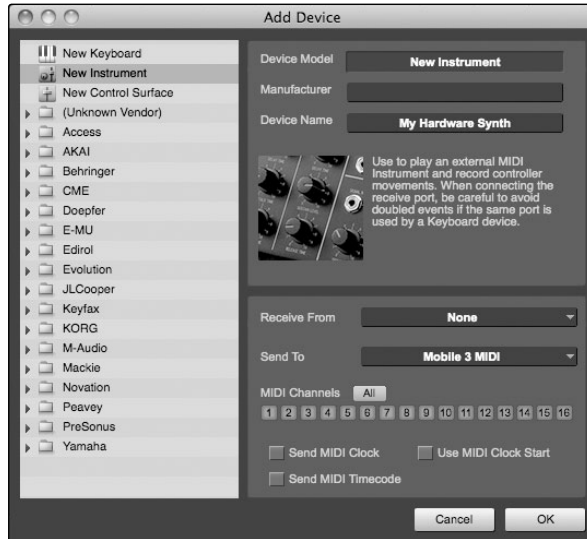
3. The Add Device window will launch.

4. 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.

5. 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.

6. 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 enviroment.

4.2.4   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.
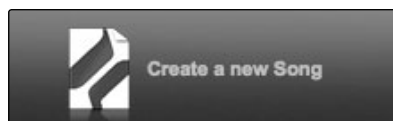
1. From the Start page, select "Create a new Song."
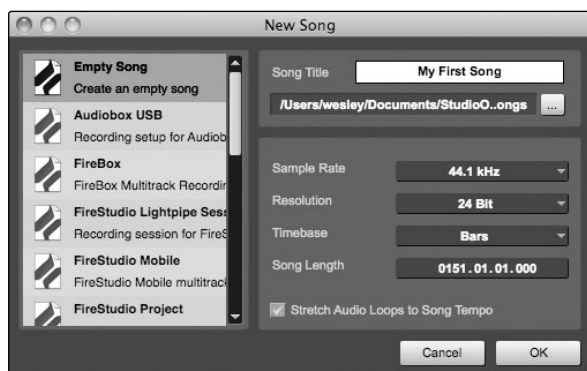
2. In the browser window, name your Song and choose the directory in which you'd like it saved. You'll notice a list of templates on the left. The AudioBox 1818VSL template will create a Song with a track for each of the AudioBox inputs. Every track is armed for recording, and no further setup is required. Select this template and click "OK."

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

4. 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.

5. 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.

6. When the Song window launches, launch the Audio Setup window by going to Studio One | Options… (Windows) or Studio One | Preferences (Mac), and click on the Audio Setup button.
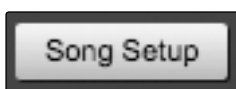
7. Click on the Song Setup button to open the Song Setup window, then click on the Audio I/O Setup icon.
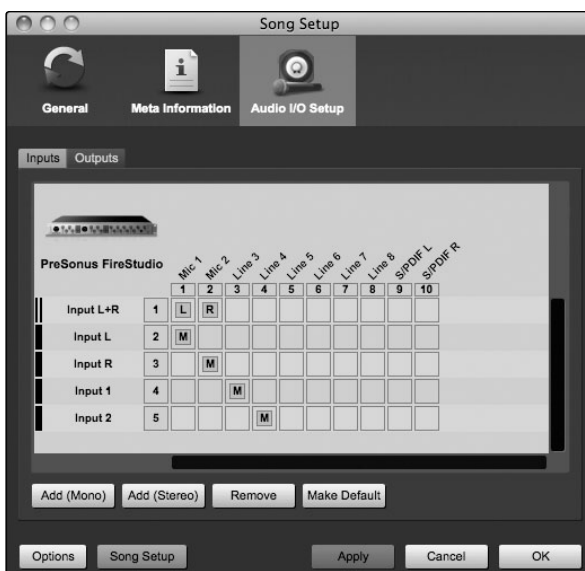
8. Click on the Inputs tab in the Audio I/O Setup window, and you will see all of the available inputs on your AudioBox 1818VSL. At this time, you can add the number and type of inputs you plan to use.

9. We recommend that you create a mono input for each of the inputs on your AudioBox 1818VSL. 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.

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

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

11. Click on the Outputs tab, and you will see all of the available outputs on your AudioBox 1818VSL. At this time, you can add the number of output buses to which you would like to have access and can give them names.

12. We recommend that you create stereo outputs for the Main outputs on your AudioBox 1818VSL and for Outputs 3/4, 5/6, 7/8. 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.

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

14. 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!

4.2.5   Creating Audio Tracks

1. 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.
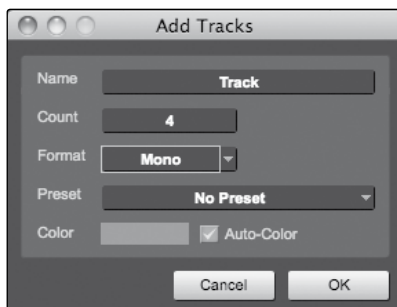
2. In the Add Tracks window, you can select the number and type of tracks you'd like to create (Mono Audio, Stereo Audio, Instrument, or Automation) and can customize the track name and color.

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

3. 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.

4. 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.

5. To begin recording, create an audio track, assign it to Input 1, and connect a microphone to the first channel on the AudioBox 1818VSL. Select Record Enable on your track in Studio One Artist. Turn up the Channel 1 level 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).

6. Connect a set of headphones to the AudioBox headphone output. You may also wish to connect monitors to the AudioBox's Main 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.

4.2.6   Creating MIDI Tracks
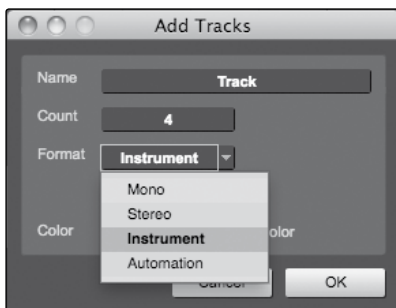
1. Click on the Add Tracks button. When the Add Tracks window launches, select Instrument as the track format and click the OK button.
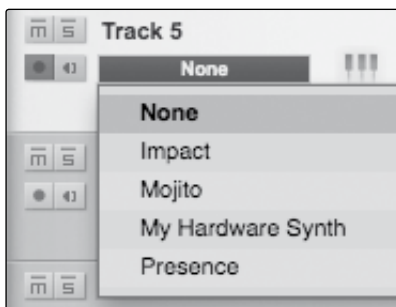
2. To assign your MIDI input, click on the MIDI Inputs list and select your external sound module.

   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.

3. To the left of the Add Track button, you will find the Inspector button. Click it to display more parameters for the selected track.
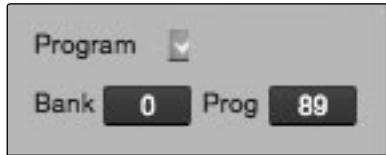
4. At the bottom of the Inspector menu, you will see your Bank and Program selections. From here, you can remotely change the patch on your sound module.

MIDI data does not contain audio signals. To hear your sound module, you must connect the audio output of the sound module to an AudioBox input, then connect the AudioBox'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.

4.2.7   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
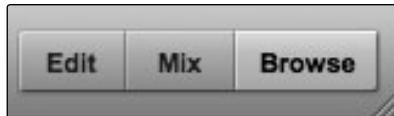
In the lower right corner of the Arrange window are three buttons. The Edit button opens or closes the audio editor or the MIDI piano-roll editor, depending on which type of track is selected. The Mix button opens and closes the mixer window.

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 the plug-in or one of its presets in the effects browser, then drag the selection over the track to which you would like to add it.

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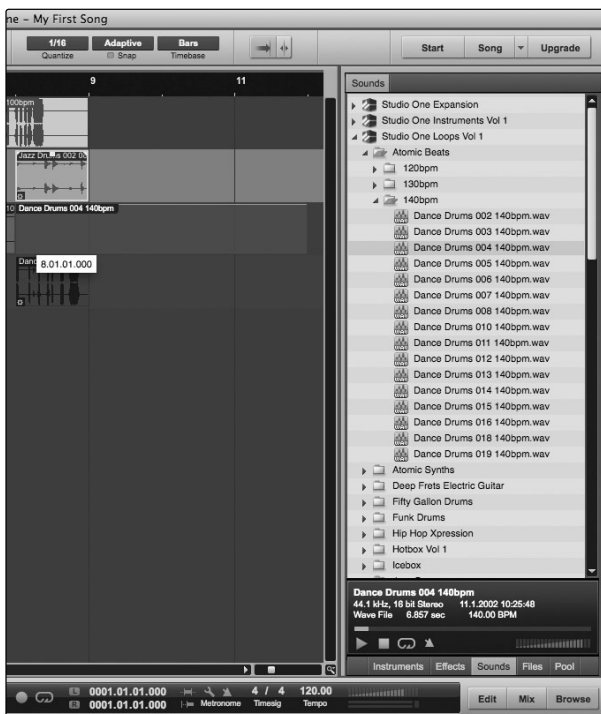
Audio and MIDI files can be quickly located, auditioned, and imported into your Song by dragging them from the file browser into the Arrange view. If you drag the file to an empty space, a new track will be created with that file placed at the position to which you dragged it. If you drag the file to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

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

5.0   Tutorials

5.1   Microphone Types

The AudioBox 1818VSL works with most types of microphones, including dynamic, ribbon, and condenser microphones.

5.1.1   Condenser

Condenser microphones generally capture sound with excellent fidelity and are among 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 AudioBox 1818VSL sends 48 VDC phantom power from the XLR inputs only.

5.1.2   Dynamic

Dynamic microphones are possibly the most widely used microphone type, especially in live shows. They are relatively inexpensive, resistant to physical damage, and typically handle high sound-pressure levels (SPL) very well. Unlike condenser microphones, most dynamic microphones do not require a power source.

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.

5.1.3   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 AudioBox 1818VSL, as the likelihood of conflicting drivers is great.

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

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

5.1.4   Microphone Placement

The following are a few recording applications to help you get started with your AudioBox 1818VSL. 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.

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials|

Technical Information

Troubleshooting and Warranty

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 AudioBox 1818VSL microphone input. Connect the line output from the passive direct box to the other microphone input on your AudioBox. Be sure to keep the trim level for this input very low so as not to clip the converters. 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.

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorialsl

Technical Information

Troubleshooting and Warranty

5.2   A Brief Tutorial on Dynamics Processing

Each channel in Virtual StudioLive (VSL) for AudioBox 1818VSL includes a Fat Channel dynamics-processing section. What follows is an excerpt from a brief tutorial on dynamics processing written by PreSonus president and founder Jim Odom. It is included to help you get the most out of VSL and the Fat Channel. This tutorial will take you through the basics of dynamics processing and will explain the various types of dynamics processors.

5.2.1   Common Questions Regarding Dynamics Processing

What is dynamic range?

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

5.2.2   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.

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials|

Technical Information

Troubleshooting and Warranty

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.

Limiter. A limiter is a compressor that is set to prevent any increase in the level of a signal above the threshold. For example, if you have the threshold knob set at 0 dB, and the ratio turned fully clockwise, the compressor becomes a limiter at 0 dB, so that the output signal cannot exceed 0 dB regardless of the level of the input signal.

Attack. Attack sets the speed at which the compressor acts on the input signal. A slow attack time 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 COMPression followed by expANDING.

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.

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 the will cause the gate to open or close.

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials|

Technical Information

Troubleshooting and Warranty

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 in VSL.

5.2.3   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

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

5.3   Equalizers

The Fat Channels in VSL are equipped with a 3-band semi-parametric equalizer for every analog input and playback stream. 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.

5.3.1   What is an EQ?

An equalizer is a filter that allows you to adjust the level of a frequency, or range of frequencies, of an audio signal. In its simplest form, an EQ will let you turn the treble and bass up or down, allowing you to adjust the coloration of, let's say, your car stereo or 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 and 16.0.2 and the AudioBox 1818VSL, 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

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

Overview

Hookup

Connecting
to a
Computer

Software: Virtual StudioLive
and Studio One Artist

Tutorials

Technical
Information

Troubleshooting
and Warranty

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

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

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

5.3.2   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

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorialsl

Technical Information

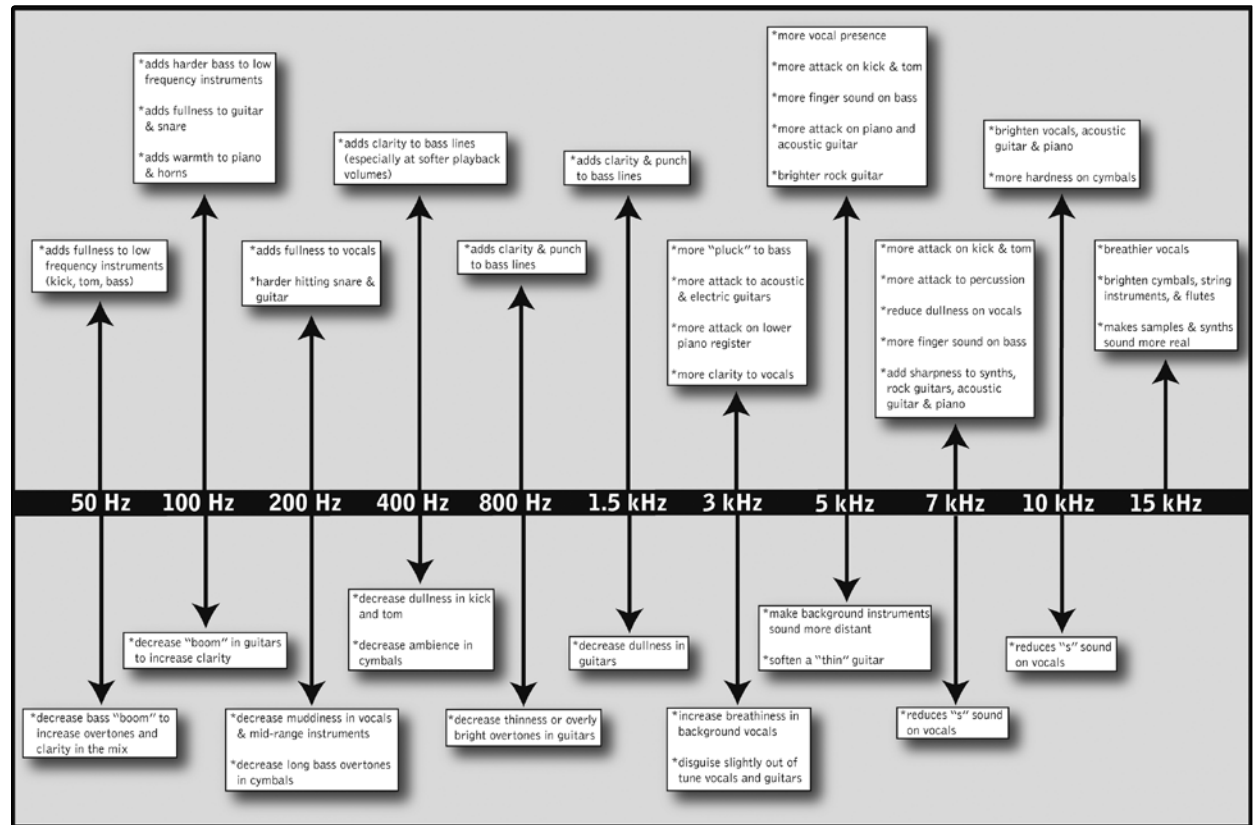Troubleshooting and Warranty

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

Table 2

5.3.3 General EQ Suggestions

Included with VSL is a library of Channel Strip presets. Section 4.1.5 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 5.2.3, 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 PEAK/SHELF	LOW FREQ (Hz)	LOW GAIN	LOW MID ON/OFF	LOW MID HI/LOW Q	LOW MID FREQ (Hz)	LOW MID GAIN
ON	PEAK	130	-2	ON	LOW	465	-2

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

Rock Female Vocals

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

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

Pop Male Vocals

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

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

Rock Male Vocals

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

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

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorialsl

Technical Information

Troubleshooting and Warranty

Percussion

Snare

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

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

Left/Right (Stereo) Overheads

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

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

Kick Drum

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

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

Fretted Instruments

Electric Bass

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

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

Acoustic Guitar

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

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

Distorted Electric Guitar

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

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

Keyboards

Piano

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

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

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorialsl

Technical Information

Troubleshooting and Warranty

5.4 Aux Bus Mixing

The Aux bus provides outputs to create auxiliary mixes that are separate from the main mixes. VSL for AudioBox 1818VSL is equipped with 4 stereo aux buses to create a separate mix for Outputs 3/4, 5/6, and 7/8, as well as two more buses for internal effects buses FX A and B. 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.

5.4.1 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.

As an example, let's create a mono monitor mix on Aux 1. To begin, click on the Overview tab in VSL. Below the Fat Channel Microviews, you will see the send level for this aux bus from each of the 16 channels. Keep in mind that the aux mix is completely independent of the Main output. The sliders on each channel control the channel send level to Aux 1. Use these sliders 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.

5.4.2 Effects Processing

VSL 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.

There are several advantages to using an aux bus for an effects processor. By using an aux bus to create an effects mix, several channels can be sent to a single processor. This allows you to use exactly the same reverb on every drum in a kit or for the entire band. A reverb is designed to emulate the psycho-acoustic properties of a space, and by using the same reverb for a band, you are putting the band "in the same room." This makes for a more cohesive sound.

In addition, by using an aux mix, you can vary the level sent from each channel to the processor, allowing you to create an effects mix. Let's say that you are using a reverb to liven up a relatively dead room. You might send a little bit of each input to the reverb, but you probably will not want much of the drums and bass to be processed, as too much reverb could reduce their impact and leave your mix without a sturdy foundation. So rather than turning the aux-send level for the kick drum channel all the way up, turn it so that the meter reads between 20% and 30% saturation. This way only a small portion of the kick drum input will be affected by the reverb.

The faders for FXA and FXB control the level of the aux mix relative to the level of your main mix.

5.5 Digital Effects

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

5.5.1 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.

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

5.5.2   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.

6.0   Technical Information

6.1   Specifications

Microphone Preamp

Connector Type	Combo, XLR, female, balanced
Mic Preamp EIN	-133 dB, 20 kHz BW, max gain, Rs=40Ω, A-wtd
Frequency Response	20 Hz - 22 kHz, +/- 0.25 dB, unity gain
THD+N	0.002%, +4 dBu, 1 kHz, unity gain, 22 kHz BW, A-wtd
S/N Ratio	97 dB, +4 dBu, 1 kHz, unity gain, 22 kHz BW, A-wtd
Gain Control Range	-15 dB to +65 dB
Input Max Headroom	+16 dBu, < 0.5% THD
Input Impedance	1.7 kΩ
Phantom Power	+48 VDC, 64 mA total unit

Instrument Input (channels 1 and 2 only)

Connector Type	Combo, ¼"TS, female, unbalanced
Frequency Response	20 Hz - 22 kHz, +/- 0.25 dB, unity gain
THD+N	0.003%, +4 dBu, 1 kHz, unity gain, 22 kHz BW, A-wtd
S/N Ratio	89 dB, +4 dBu, 1 kHz, unity gain, 22 kHz BW, A-wtd
Gain Control Range	-30 dB to +50 dB
Input Max Headroom	+21 dBu, < 0.5% THD
Input Impedance	1 MΩ

Line Inputs

Connector Type	Combo, ¼"TRS, female, balanced
Frequency Response	20 Hz - 22 kHz, +/- 0.25 dB, unity gain
THD+N	0.002%, +4 dBu, 1 kHz, unity gain, 22 kHz BW, A-wtd
S/N Ratio	98 dB, +4 dBu, 1 kHz, unity gain, 22 kHz BW, A-wtd
Gain Control Range	-20 dB to +20 dB
Input Max Headroom	+21 dBu, < 0.5% THD
Input Impedance	10 kΩ

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

HEADPHONE OUTPUT:

Connector Type	¼" TRS, female, stereo
Maximum Power	120 mW/ch @ 60Ω load
Frequency Response	20 Hz - 20 kHz, +/- 0.5 dB, max gain
THD+N	0.01%, 1 kHz, max gain, 20 kHz BW, A-wtd
S/N Ratio	96 dB, 1 kHz, max gain, 20 kHz BW, A-wtd

ANALOG OUTPUTS:

Connector Type	¼" TRS, female, impedance balanced
Frequency Response	20 Hz - 22 kHz, +/- 0.25 dB
THD+N	0.003%, 1 kHz, 22 kHz BW, A-wtd
S/N Ratio	109 dB, 1 kHz, 22 kHz BW, A-wtd
Output Level	+10 dBu, < 0.5% THD
Output Impedance	51Ω

POWER:

External Power Adapter	+18 VDC, 2.6 A, center-positive
Connector Type	Barrel: 5.5 mm OD, 2.1 mm ID, 9.5 mm length

DIGITAL:

ADC Dynamic Range	114 dB, 48 kHz, A-wtd
DAC Dynamic Range	114 dB, 48 kHz, A-wtd
Host Interface	USB 2.0 high-speed
Bit Depth	24-bit
Sample Rates	44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz
Maximum Latency	6 mS, analog input to analog output
ADAT Optical I/O	Toslink™ connectors, ADAT/SMUX formats
S/PDIF I/O	RCA female connectors
Word Clock Output	BNC connector, 4.5 Vpp @ 75Ω load
MIDI I/O	5-pin DIN connectors

7.0   Troubleshooting and Warranty

Overview

Hookup

Connecting to a Computer

Software: Virtual StudioLive and Studio One Artist

Tutorials

Technical Information

Troubleshooting and Warranty

7.1   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 AudioBox interface, Studio One™ digital audio workstation software, and AudioBox 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) 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.

AudioBox Will Not Connect to Computer

Verify that the USB cable is properly connected both to the AudioBox and to your computer. Disconnect unnecessary peripheral USB devices. Verify that your AudioBox is connected to a USB 2.0 connection.

AudioBox VSL Will Not Launch

AudioBox VSL will not launch if the AudioBox is not connected and synced to your computer. Verify that the sync light on the front of your AudioBox is solid blue and your USB cable is connected both to your computer and to your AudioBox.

VSL Has No Effect On Audio

Make sure that the VSL enable button is illuminated in AudioBox VSL.

No Effects in the Outputs

Verify that the FXA and FXB routing buttons on the desired bus are enabled. These buttons patch each of the effects bus to that output.

No Audio in Headphones

Verify that the Output 7/8 channels sends have been raised for each channels and that the fader for Output 7/8 in VSL is raised.

Distortion or Noise when using ADAT I/O

Verify that you have the correct ADAT mode set for the sample rate being used. If you are clocking the AudioBox 1818VSL externally, verify that the AudioBox is set to the same sample rate as your external clock.
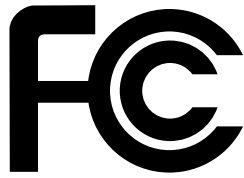
Overview

Hookup

Connecting
to a
Computer

Software: Virtual StudioLive
and Studio One Artist

Tutorials

Technical
Information

Troubleshooting
and Warranty

7.2   PreSonus Limited Warranty

PreSonus Audio Electronics, Inc., warrants this product to be free of defects in materials 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

Index

Declaration of Conformity

Responsible Party: **PreSonus Audio Electronics**

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

Phone: 1-**225-216-7887**

declares that **AudioBox™ 1818VSL** 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.

PreSonus

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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 tomotoes 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

AudioBox™ 1818VSL
Owner's Manual

EMC Statement:

NOTE: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.

- Increase the separation between the equipment and the receiver.

- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

- Consult the dealer or an experienced radio/TV technician for help.

CAUTION: Changes or modifications to this device not expressly approved by PreSonus Audio Electronics could void the user's authority to operate the equipment under FCC rules.

This apparatus does not exceed the Class A/Class B (whichever is applicable) limits for radio noise emissions from digital apparatus as set out in the radio interference regulations of the Canadian Department of Communications.

ATTENTION — Le présent appareil numérique n'émet pas de bruits radioélectriques dépassant las limites applicables aux appareils numériques de class A/de class B (selon le cas) prescrites dans le réglement sur le brouillage radioélectrique édicté par les ministere des communications du Canada.

ကြ။ PreSonus™