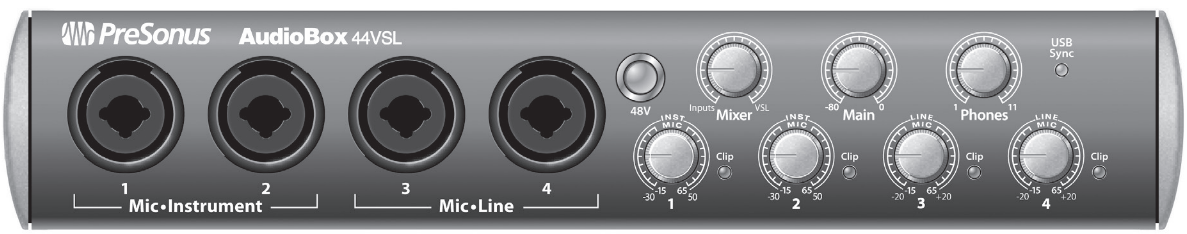
AudioBox™ 22/44VSL

Owner's Manual



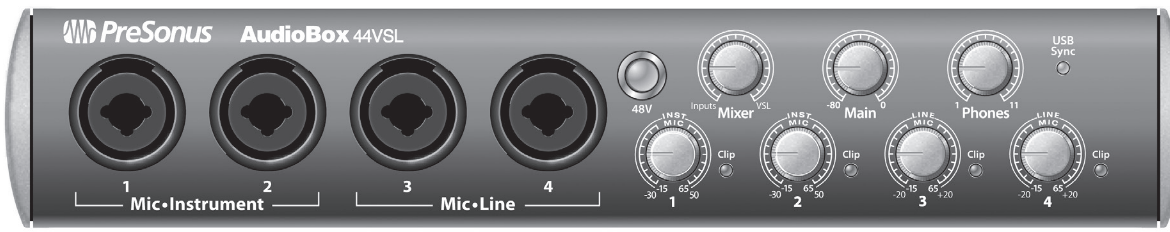
PreSonus®
www.presonus.com

AudioBox™ 22/44VSL

Overview
Hookup
Connecting to a Computer
Software: Virtual StudioLive SL Remote, and Studio One Artist
Tutorials
Technical Information
Troubleshooting and Warranty

1.0 Overview

1.1 Introduction



Thank you for purchasing the PreSonus AudioBox™ 22/44VSL. PreSonus Audio Electronics has designed the AudioBox 22/44VSL utilizing high-grade components to ensure optimum performance that will last a lifetime. Loaded with 2 high-headroom, Class A, XMAX™ microphone preamplifiers; a built-in 2x2/4x4 USB 2.0 recording and playback engine; MIDI I/O; and more, AudioBox 22/44VSL 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!

We encourage you to contact us at 1-225-216-7887 between 9 a.m. and 5 p.m. Central Time (GMT -06:00 CST) with questions or comments regarding your PreSonus AudioBox 22/44VSL. 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 22/44VSL!

ABOUT THIS MANUAL: *We suggest that you use this manual to familiarize yourself with the features, applications, and correct connection procedures for your AudioBox 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 22/44VSL 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 22/44VSL Hardware Features

- 24-bit/96 kHz sampling rate
- 2 /4 Class A XMAX microphone preamplifiers
- 2 Instrument inputs
- 2 Balanced Line inputs (AudioBox 44VSL Only)
- MIDI I/O
- High-definition analog-to-digital converters (108 dB dynamic range)
- 2x2/4x4 USB 2.0 audio interface
- Headphone output
- Analog monitor mixing with playback/input mix control
- Rugged steel chassis
- Studio One™ Artist
- Compatible with Cubase, Digital Performer, Logic, Nuendo, Pro Tools 9+, Sonar, Studio One, and others
- 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.

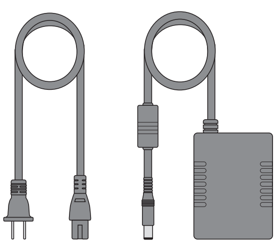
- Unlimited track count, inserts, and sends
- 20 high-quality native plug-ins: amp modeling (Ampire), delay (Analog Delay, Beat Delay), distortion (Redlight Dist), dynamics processing (Channel Strip, Compressor, Gate, Expander, Limiter, Tricomp), equalizer (Channel Strip, Pro EQ), modulation (Autofilter, 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 What is in the Box

In addition to this manual, your AudioBox 22/44VSL package contains the following:


PreSonus AudioBox 22VSL or 44VSL USB 2.0 recording interface

6' (1.8m) 6-pin-to-6-pin USB cable



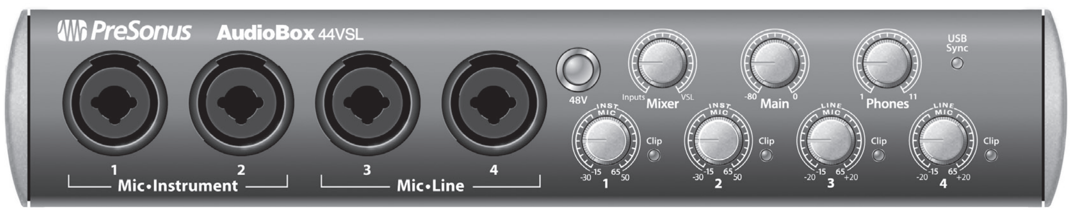
External Power Supply (AudioBox 44VSL only)

Available for download from your My PreSonus account:

- PreSonus Studio One Artist program DVD plus gigabytes of third-party content
- ASIO driver for Windows

2.0 Hookup

2.1 Front-Panel Connections



Microphone Inputs. Your AudioBox 22/44VSL 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 unwanted background noise.

Both channels of the AudioBox 22VSL (and the first two channels of the AudioBox 44VSL) have Mic/Instrument combo jacks. This convenient connector accepts either a ¼-inch phone plug or an XLR plug.

- **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 22/44VSL becomes an instrument preamplifier.
- **Line-level Inputs (AudioBox 44VSL Only).** Channels 3 and 4 of the AudioBox 44VSL 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.

***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 22/44VSL 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 22/44VSL. 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.*

48-volt Phantom Power. The AudioBox 22/44VSL provides 48V phantom power for the microphone input on each channel. Press the 48V button to enable phantom power for all microphone inputs.

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

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

- **XLR Microphone / TS 1/4" Instrument inputs:** 80 dB of variable gain (-15/-30 dB to +65/50 dB)
- **TRS ¼-inch Line Level inputs (AudioBox 44VSL only):** 40 dB of variable gain (-20 dB to +20 dB)

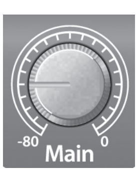
Clip Indicator. All channels feature clip LEDs next to the trim controls. The red clip indicator LED will illuminate when the channel's input signal reaches 0 dBFS. At this level, your mic preamp/line trim signal will exhibit signs of clipping (distortion).

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 22/44VSL provide plenty of headroom; take advantage of it.*

Mixer. The Mixer knob allows you to blend your input signals with the playback streams from your computer. This allows you to monitor your input signal with zero latency. If the knob is positioned at 12 o'clock, the input signal and the playback stream will be equally balanced. Turning the knob to the left will increase the level of the input signal relative to the playback stream; turning to the right will increase the level of the playback stream relative to the input signal.

Please note: *When creating monitor mixes using Virtual StudioLive software, it is important that you turn the Mixer knob all the way to the VSL position. Monitoring both the input signal and the playback stream will create a doubling effect that will make monitoring difficult.*

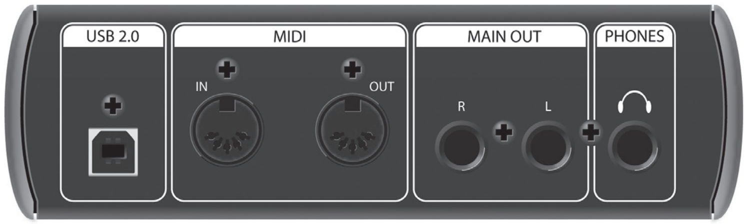
Phones. The Phones knob controls the volume of the headphone output on the front of the unit. The headphone amplifier is quite powerful, and the volume goes to 11, so *use the maximum setting with extreme caution.*

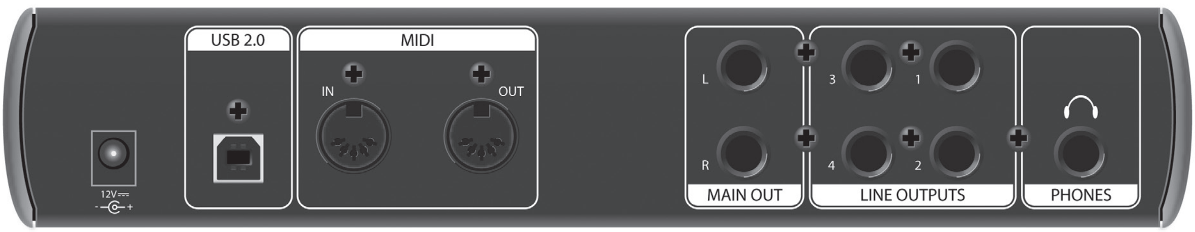


Main. The Main knob controls the output level for the Main Outputs on the back of the AudioBox, with a range of -80 dB to 0 dB.

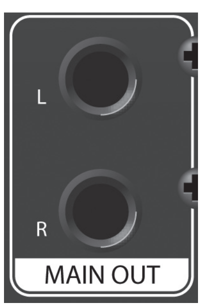
Power (22VSL)/USB Sync (44VSL) LED. This LED will illuminate blue when the AudioBox is properly powered and synced to a USB 2.0 connection. AudioBox 44VSL users: this LED will flash blue & red when the unit is properly powered but no USB connection is detected.

2.2 Rear-Panel Connections

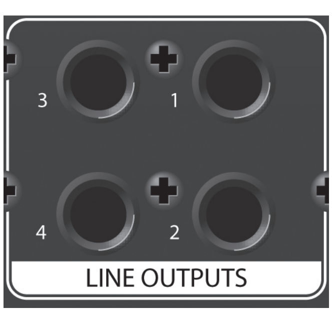




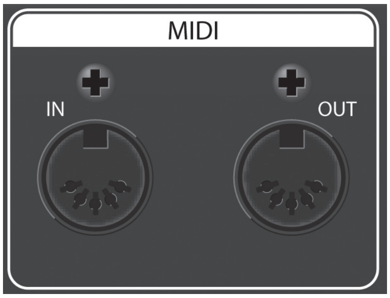
¼-inch Phones (headphone) Jack. This is where you connect headphones to your AudioBox 22/44VSL.



Main Out. These are the main outputs for the AudioBox 22/44VSL. The output level of the Main Outs is controlled by the Main volume knob on the front of the unit.

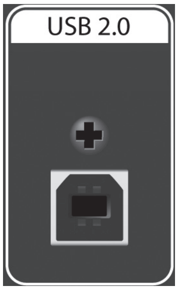


Line Outputs (AudioBox 44VSL Only). The AudioBox 44VSL has four line outputs to route to external devices, such as headphone amps and DJ mixers. Outputs 1 and 2 share their playback streams with the Main Outs and the headphone output. Outputs 3 and 4 have independent playback streams.

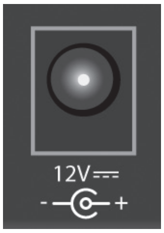


MIDI I/O. MIDI stands for "Musical Instrument Digital Interface." MIDI inputs and outputs allow connection to, and communication with, external MIDI equipment. One function of these ports is MIDI sequencing but the MIDI protocol can be used for much more than instruments and sequencing.

> ***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 22/44VSL input channel. Please consult the User's Manuals of your MIDI devices for help with MIDI setup and usage.*

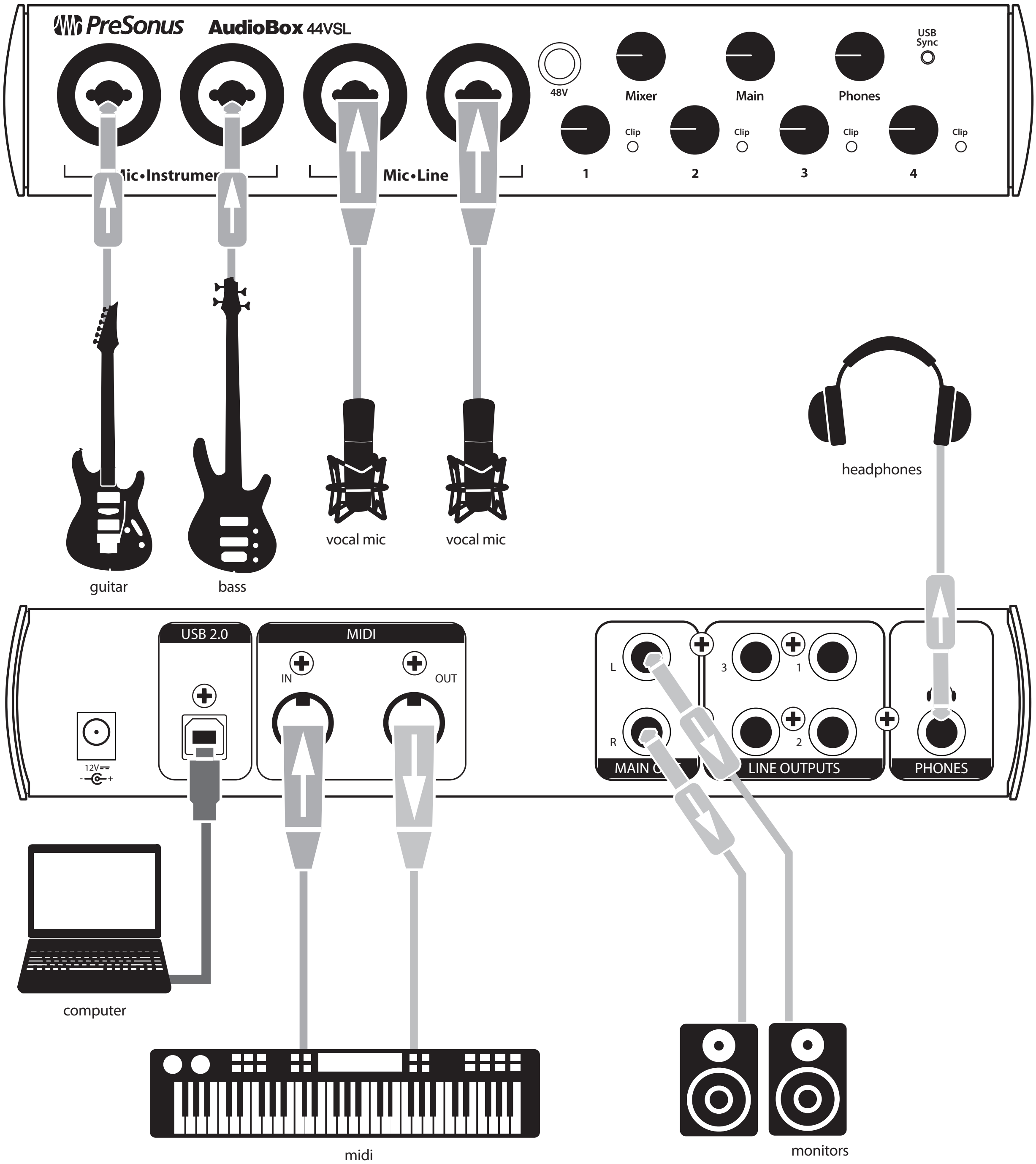


USB 2.0 Port. This is where you connect the USB cable from your AudioBox 22/44VSL to your computer. The AudioBox 22VSL is bus-powered via this connection and does not require an external power supply.

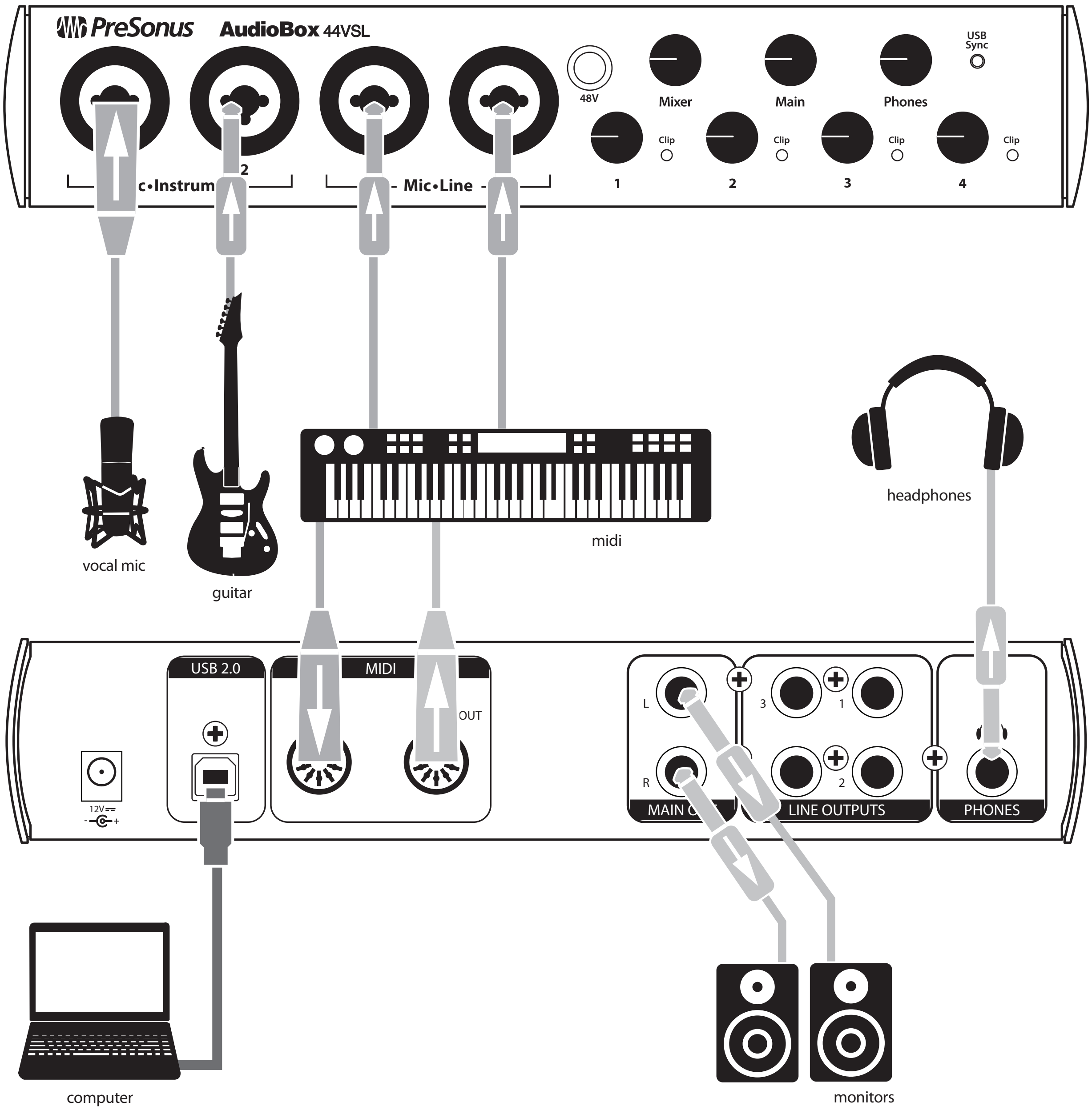


Power Input (AudioBox 44VSL Only). This is where you connect the power supply for your AudioBox 44VSL. The AudioBox 44VSL cannot be bus powered.

2.3 Basic Hookup Diagram



2.4 MIDI Hookup Diagram



3.0 Connecting to a Computer

Your AudioBox 22/44VSL 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 10.8 or higher
- **Hardware:**
  - Intel Core Duo processor (Intel Core 2 Duo or Intel Core i3 or better recommended)
  - 2 GB RAM (4 GB or more recommended)

Windows

- **Operating Systems (32- or 64-bit)**:
  - Windows® 7x 64/x86 SP1 or Windows 8/8.1 x64/x86 or Windows 10 x64/x86
- **Hardware**:
  - Intel® Core™ 2 processor, 4 GB RAM (Intel® Core™ i3 processor or better recommended)
  - 4 GB RAM (8 GB or more recommended)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.*

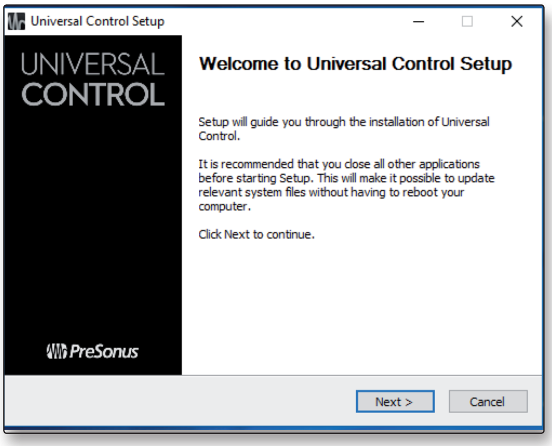
*Subject to change. Check www.presonus.com for updates.

3.2 Installation for Windows

All PreSonus interface products connect to the Universal Control application. The Universal Control installer will install both the ASIO and WDM drivers for your AudioBox 22/44VSL and the Univeral Control control panel application.

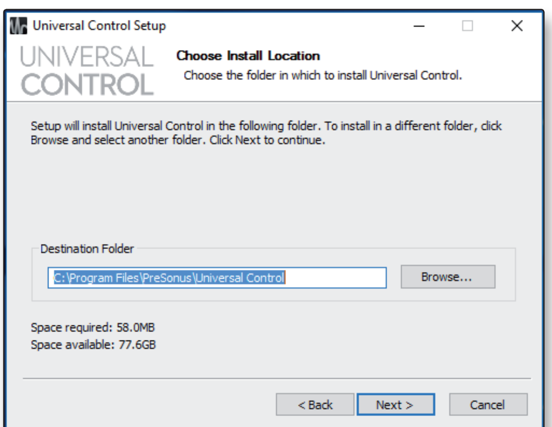
It is recommended that you quit all applications before you start the installation.

Should you see any Windows Security alerts during the installation process, you will need to select "Allow Access."



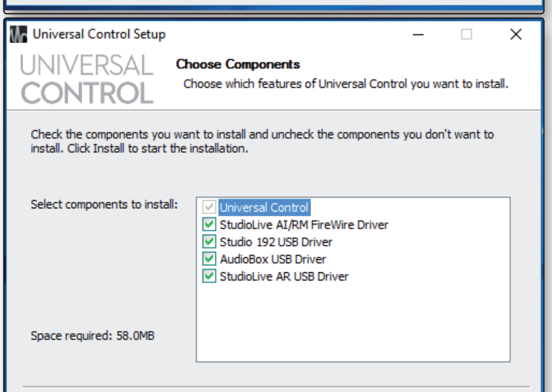
1. The installer will open to the Welcome screen.

   Click "Next."



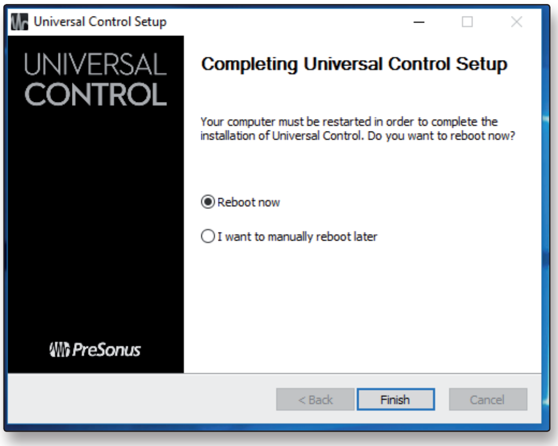
2. You will be asked if you where you would like to install the Universal Control application. For nearly every system, you will want to choose the default location.

   Click "Next."



3. You will be givin the option to select for which PreSonus interface products you would install drivers. Unless you are using other PreSonus interface products, check the box next to "AudioBox USB Driver."

   Click "Install Driver."



4. Once the driver has been installed successfully, you will be alerted. In order for the driver to operate properly, you must restart your computer.

   Click "Finish" to complete the installation and reboot your computer.

   Once your computer has restarted,
   your AudioBox 22/44VSL is now ready to use!

3.2.1 Universal Control AI (Windows only

Sample Rate. Changes the sample rate.

You can set the sample rate to 44.1, 48, 88.2, or 96 kHz. A higher sample rate will increase the fidelity of the recording but will increase the file size and the amount of system resources necessary to process the audio.

Safe Mode. Adjusts the Input Buffer Size.

These modes allow you to adjust the input buffer size to optimize performance for your computer.

When adjusting the safe mode, the block size will be adjusted automatically to provide the best performance.

Block Size. Sets the buffer size.

From this menu, you can set the buffer size from 64 to 8192 samples. Lowering the buffer size will lower latency; however, this will also increase performance demands on your computer. In general, you will want to set the buffer size as low as your system can safely support. If you begin to hear pops, clicks, or distortion in your audio path, try raising the buffer size.

Note: When adjusting the block size, the Safe Mode will automatically change to provide the best performance

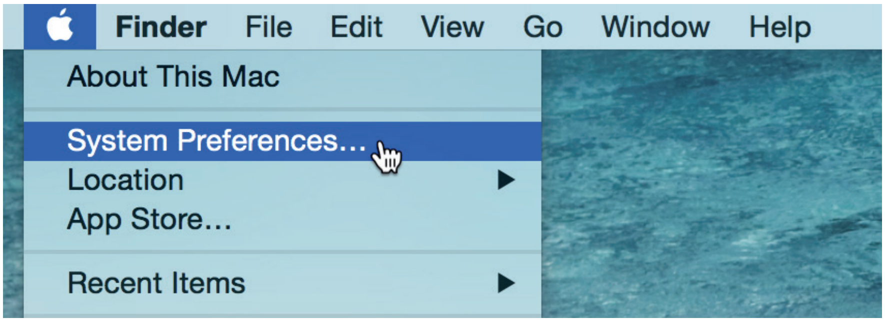
3.3 Installation for Mac OS X

The AudioBox 22/44VSL is a class compliant core audio device. No installation is necessary. Simply connect your interface to your computer.

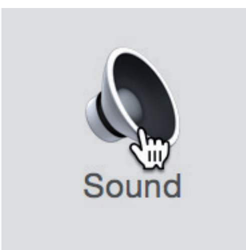
3.3.1 Using the AudioBox for System Sound (OS X)

You can configure your AudioBox 22/44VSL as the audio interface for computer system audio (for iTune playback, Skype, etc.) from the System Preferences menu.

1. Open System Preferences



2. Open your System Sound Preferences



3. Select your AudioBox 22/44VSL from the Outputs list. If you would like to use your AudioBox for System input, select it from the inputs tab as well.

3.4 Using the AudioBox 22/44VSL with Popular Audio Applications

Complete setup instructions for Studio One Artist and a brief tutorial on its features can be found in Section 5.2 of this manual. However, you can use your AudioBox 22/44VSL 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 22/44VSL 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 22(44)VSL** from the device menu.

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

7. Your **AudioBox 22/44VSL** 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 22/44VSL. When you are done, close this window.

9. You are now ready to use your AudioBox 22/44VSL.

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 22(44)VSL** from the **ASIO Driver** dropdown list.

5. Click "**Switch**" to begin using the AudioBox 22/44VSL 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 22(44)VSL."**

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**."

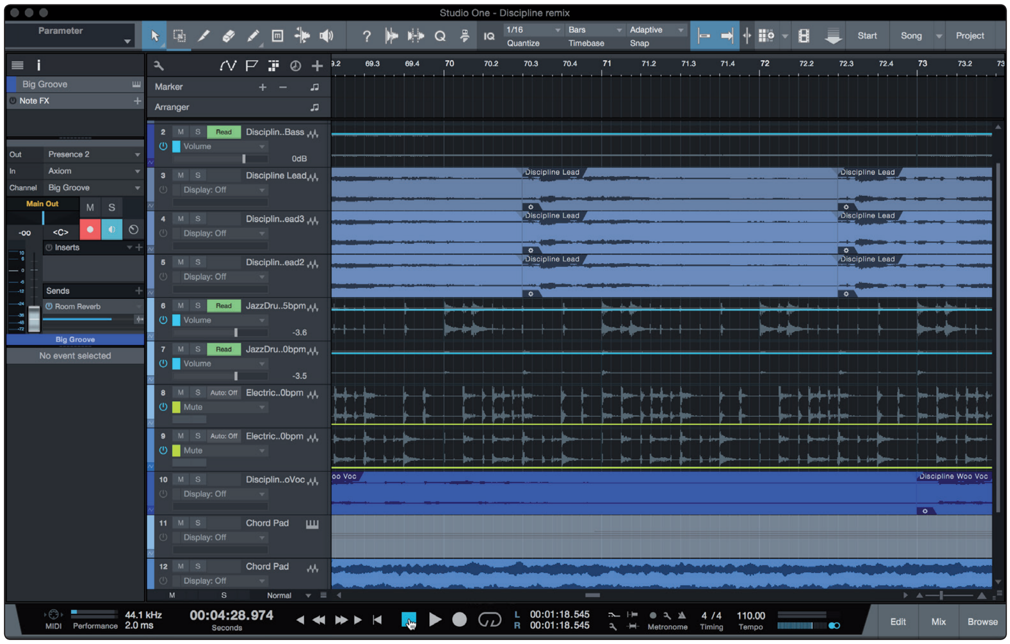
Ableton Live 5+

1. Launch **Ableton Live**
2. Go to **Options | Preferences | Audio**
3. Choose **Driver Type: Asio | Audio Device: ASIO PreSonus AudioBox 22(44)VSL**
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.

ProTools 10+

1. Launch **ProTools**
2. Go to **Setup | Hardware** and select **AudioBox 1818VSL** from the Peripherals list. Click **OK**.
3. Go to **Setup | Playback Engine** and select **AudioBox 1818VSL** from the menu at the top of the window. Click **OK.**4.0 Software: Virtual StudioLive and Studio One Artist

4 Studio One Artist Quick Start



All PreSonus professional recording products come 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.

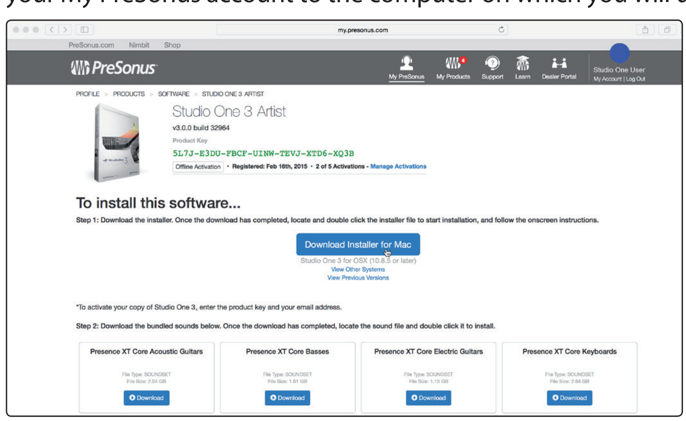
Power User Tip: *As a valued PreSonus customer, you are eligible for a discount upgrade to Studio One Professional. For more details on the Studio One upgrade program for PreSonus customers, please visit http://studioone.presonus.com/.*

4.1 Installation and Authorization

Once you have installed the drivers for your audio interface and connected it 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, log into your My PreSonus account and register your interface. Your product key for Studio One Artist will automatically be registered to your My PreSonus account with your hardware registration.

Downloading and running the Studio One installer.

To install Studio One Artist, download the Studio One Artist installer from your My PreSonus account to the computer on which you will use it.



- **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 Mac hard drive.

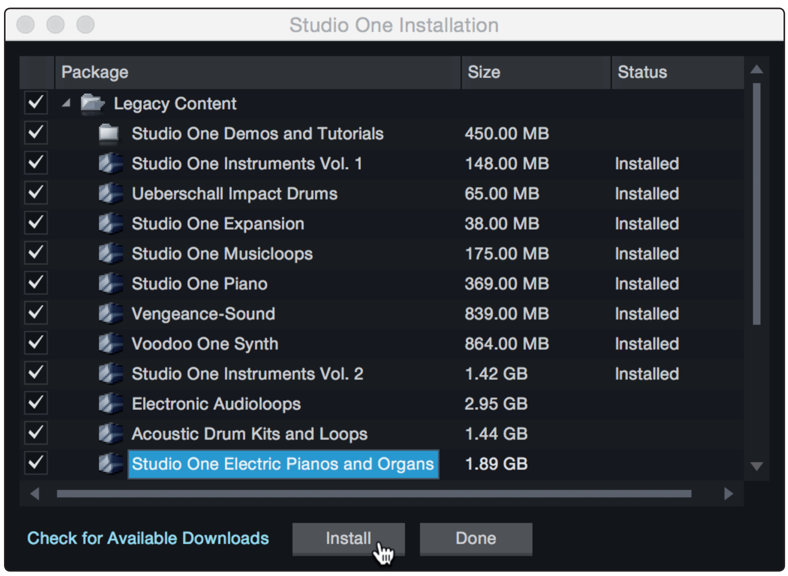
Authorizing Studio One

When Studio One is launched for the first time on your computer, it will communicate with your My PreSonus account and verify your registration. To ensure a seamless authorization process, make sure to download your installer to the computer on which you will be using it and be sure that your computer is connected to the Internet when you launch the application for the first time.

Installing bundled content for Studio One Artist.

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

The first time you launch Studio One Artist, you will be prompted to install its companion content. Select the content you wish to add and click "Install." The content will automatically begin to download and install from your My PreSonus user account.



Power User Tip: *You may be prompted to enter your My PreSonus user account information. Clicking "Remember Credentials" will allow you to have immediate access to any content you purchase from the PreSonus Marketplace.*

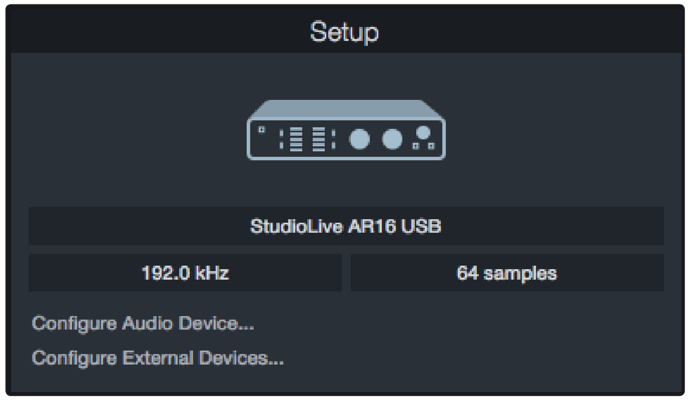
4.2 Setting Up Studio One

Studio One Artist was designed to work with PreSonus interfaces and provides unique interoperability and simplified setup. 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.

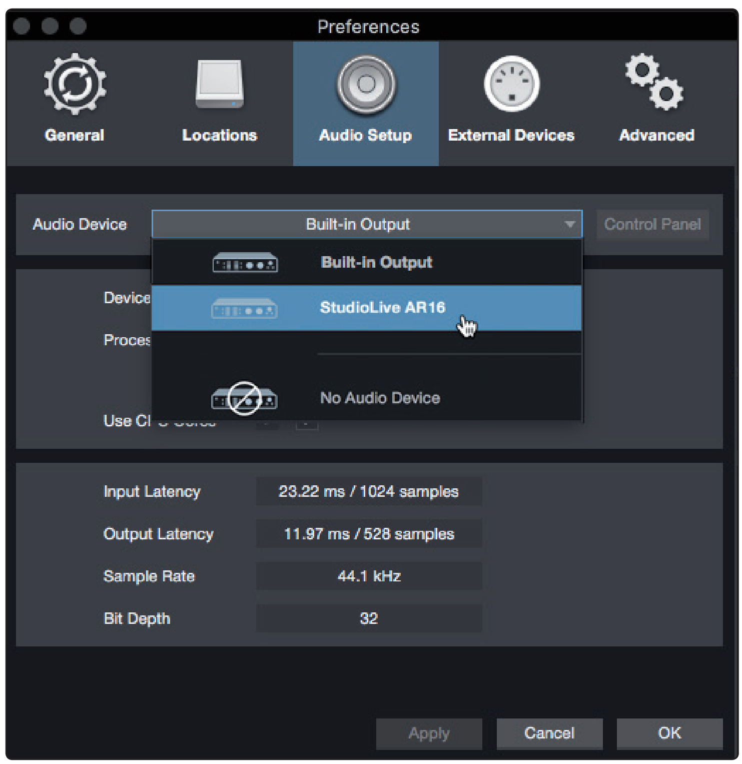
Complete information on all aspects of Studio One Artist is available in the Reference Manual PDF located within Studio One. The information in this tutorial covers only the basic aspects of Studio One Artist and is intended to get you set up and recording as quickly as possible.

4.2.1 Configuring Audio Devices

1. 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.



2. If you do not see your device listed on the Start page when you launch Studio One, click on the Configure Audio Devices link in the Setup area to open the Options window.



In the Options window, click on the Audio Setup tab and select your device driver from the pull-down.

4.2.2 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 modules. Please consult the Reference Manual located within Studio One 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 any required drivers for these devices before beginning this section. Please consult the documentation that came with your MIDI hardware for complete installation instructions.

If you do not have any MIDI devices, please skip to Section 5.4.

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 modules will be set up as an Instrument.

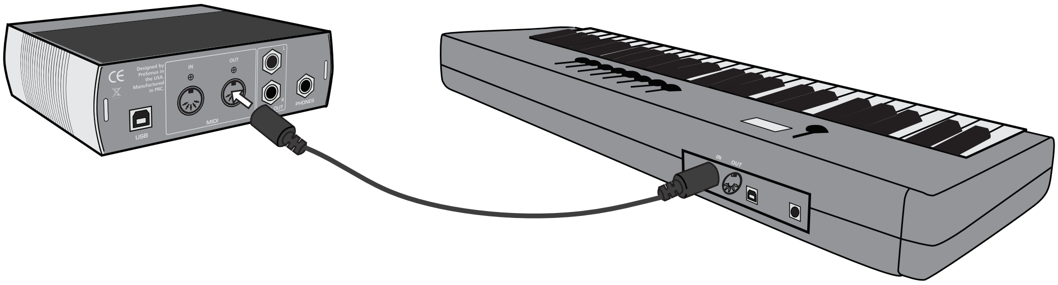
You can set up your external MIDI devices from the Setup area in the Start page. Before setting up a new Song for recording, take a moment to configure external devices.

Make sure you have connected the MIDI Out of your external MIDI controller to a MIDI In on your PreSonus audio interface (if available) or other MIDI interface. If you are using a USB MIDI controller, connect it to your computer and power it on.

Setting up an External MIDI Sound Module from the Start Page

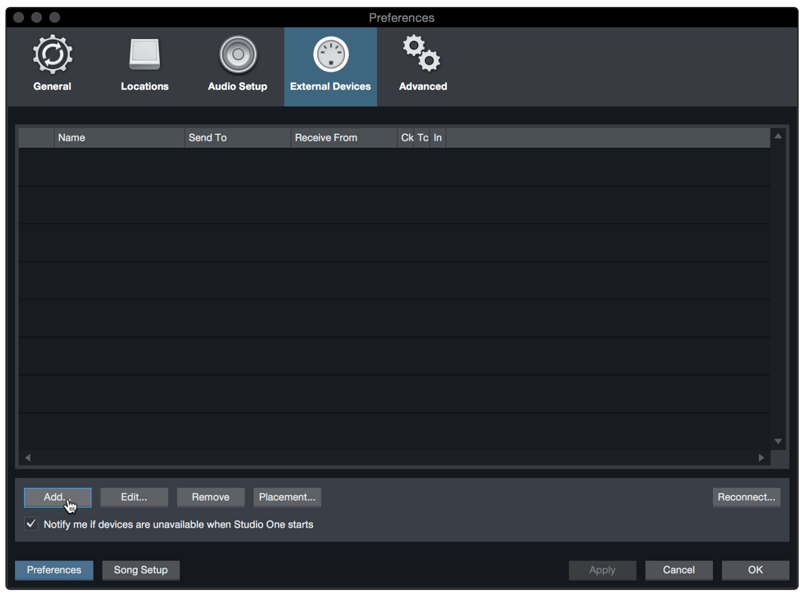
MIDI instrument controllers (keyboards, MIDI guitars, etc.) send musical information in the form of MIDI data to tone modules and virtual instruments, which respond by generating sound, as instructed. Tone modules can be standalone sound devices or can be integrated into a MIDI instrument, such as a 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 AudioBox 22/44VSL or other MIDI interface.



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

3. Click the Add button. This will launch the Add Device window.



4. From the menu on the left, select your MIDI controller from the list of manufacturers and models. If you do not see your MIDI controller listed, select New Keyboard. At this point, you can customize the name of your keyboard by entering the manufacturer and device names.

5. 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 choose, select all 16.

6. Studio One allows you to filter out specific control functions. If you would like Studio One to ignore Aftertouch, Pitch Bend, Program Change, or All CC messages, enable filtering for any or all of these messages.

7. In the Receive From drop-down menu, select the MIDI interface input from which Studio One Artist will receive MIDI data (that is, the MIDI port to which your keyboard is connected).

Power User Tip: *In the Send To drop-down menu, select the MIDI interface output from which your Studio One Artist will send MIDI data to your keyboard. If your keyboard controller doesn't need to receive MIDI data from Studio One, you can leave this unselected.*

8. 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.

9. Click OK.

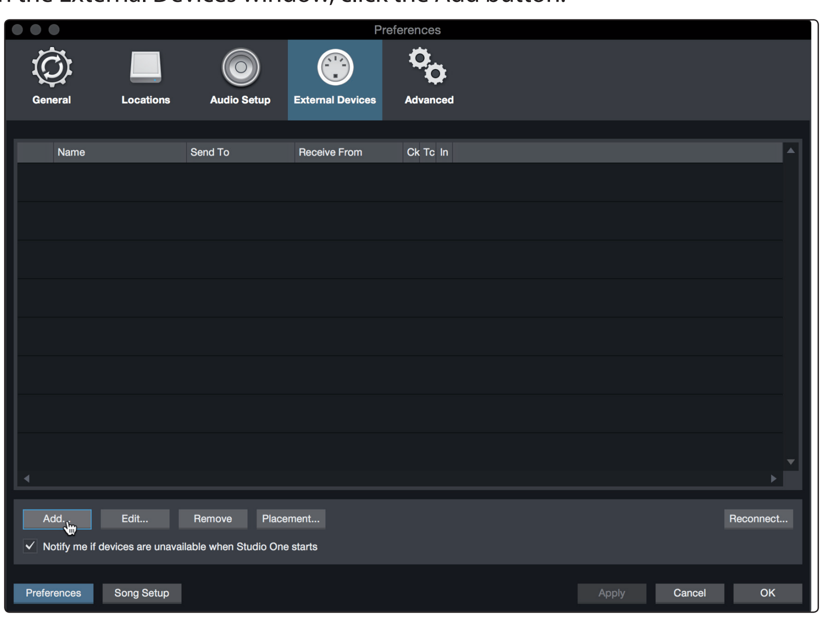
If you have a sound module that you'd like to connect, leave the External Devices window open and proceed to the next part of this section. If not, you can close the window and skip to the next section.

Setting up an external MIDI sound module from the Start page.

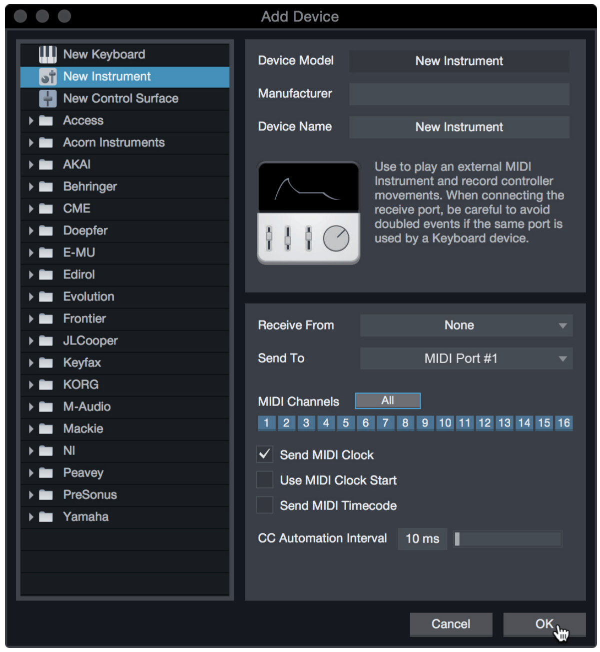
MIDI instrument controllers (keyboards, MIDI guitars, etc.) send musical information in the form of MIDI data to tone modules and virtual instruments, which respond by generating sound, as instructed. Tone modules can be standalone sound devices or can be integrated into a MIDI instrument, such as a 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.

Make sure you have connected the MIDI In of your external sound module to the MIDI Out of your MIDI interface.

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



2. Select your device in the menu on the left. If your device is not listed, select New Instrument. At this point you can customize the name of your keyboard by entering the manufacturer and device names.



3. Specify which MIDI channels will be used to communicate with this sound module. 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.

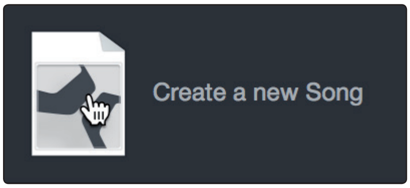
4. In the Send To menu, select the MIDI interface output from which Studio One Artist will send MIDI data to your sound module. Click OK and close the External Devices window. You are now ready to start recording in Studio One Artist.

The rest of this Quick Start Guide will go over how to set up a Song and will discuss some general workflow tips for navigating through the Studio One Artist environment.

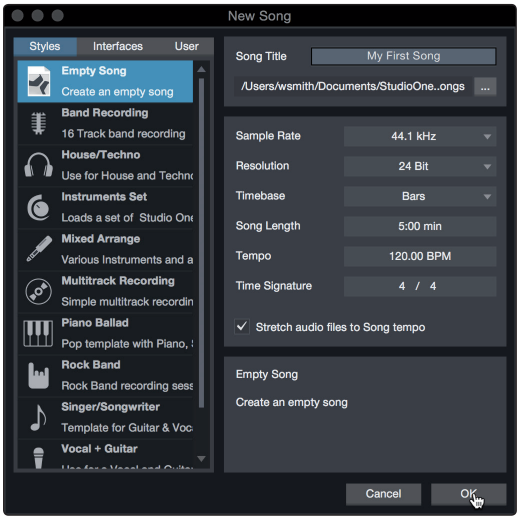
4.3 Creating a New Song

Now that you've configured your audio and 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 New Song window, name your Song and choose the directory in which you'd like it saved. From the Interfaces tab, you can select custom templates for your StudioLive AR-series mixer that will set all configuration and I/O settings for you. The rest of section will describe creating a Song from an empty session.

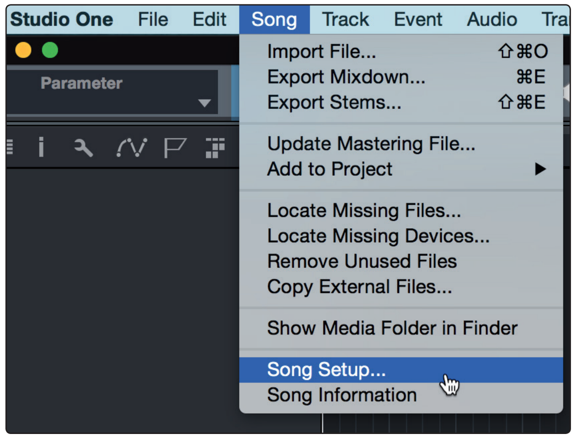


3. Select Empty Song from the Templates list. At this point, you should give your Song a name and select your preferred sample rate and bit depth for recording and playback. You can also set the length of your Song and the type of time format you would like the timeline to follow (notation bars, seconds, samples, or frames). Click the OK button when you are finished.

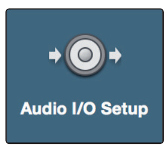
Power User Tip: *If you plan to import loops into your Song, make sure that the Stretch Audio Files to Song Tempo option is selected. This will automatically import loops at the correct BPM.*

4.3.1 Configuring Your I/O

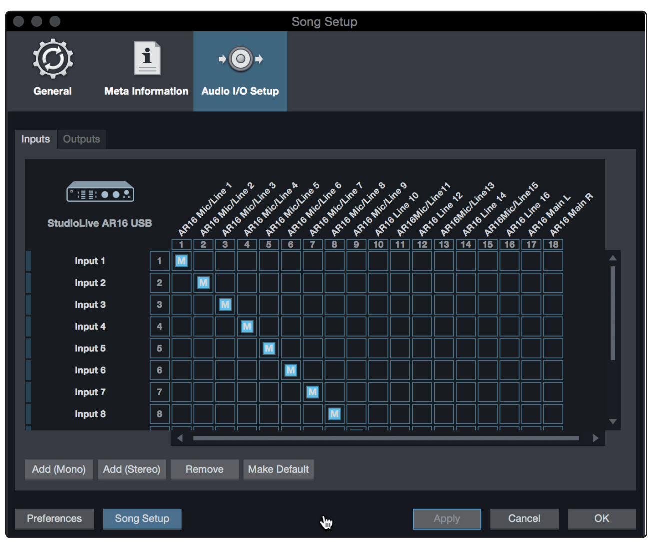
1. Click on Song | Song Setup to set your sample rate and resolution and configure your audio I/O.



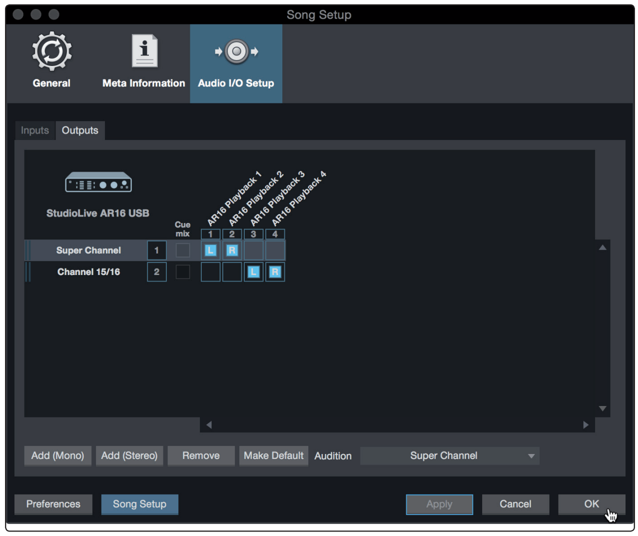
2. Click on the Audio I/O Setup tab.



3. From the Inputs tab, you can enable any or all of the inputs on your StudioLive AR mixer that you'd like to have available. We recommend that you create a mono input for each of the inputs on your interface. If you plan on recording in stereo, you should also create a few stereo inputs. You can give each input a custom name by simply clicking on the default name. Press the TAB key to edit the next name.



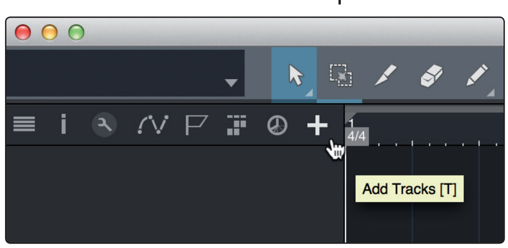
4. Click on the Outputs tabs to enable any or all of the outputs on your StudioLive AR Mixer. In the lower right corner, you will see the Audition select menu. This allows you to choose the output from which you will audition audio files prior to importing them into Studio One Artist. In general, you will want this to be the main output bus. You can give each output a custom name by simply clicking on the default name. Press the TAB key to edit the next name.



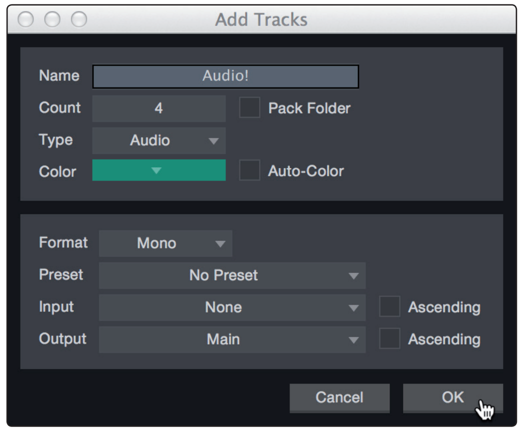
Power User Tip: *If you would like this I/O configuration to be the same every time you open Studio One, click the Make Default button.*

4.3.2 Creating Audio and MIDI Tracks

1. In the upper left corner of the Arrange window, you will notice several buttons. The button furthest to the right is the Add Tracks button. Click on this button to open the Add Tracks window.

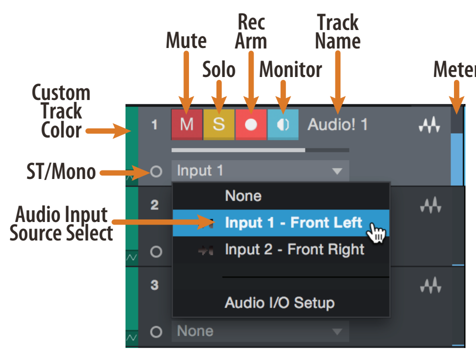


2. In the Add Tracks window, you can customize the track name and color, add a preset rack of effects, and set the physical source for the input and output of your audio tracks. Most important, you can select the number and type of tracks you'd like to create.



- **Audio.** Use this track type to record and playback audio files.
- **Instrument.** Use this track to record and playback MIDI data to control external MIDI devices or Virtual Instrument plug-ins.
- **Automation.** This track type lets you create automated parameter controls for your session.
- **Folder.** This track helps you to manage your session as well as to quickly edit multiple tracks at once.

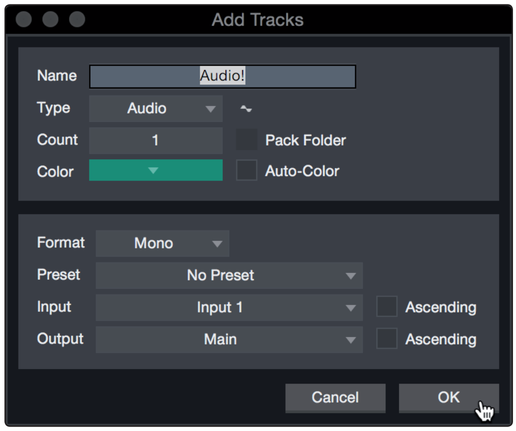
Power User Tip: *If you would like to add an audio track for each of the available inputs, simply go to Track | Add Tracks for All Inputs.*



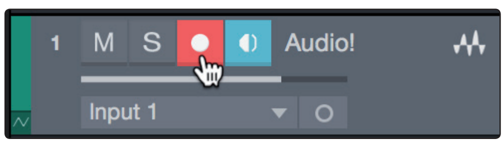
Note: *MIDI tracks are nearly identical to Audio tracks. The Input Source list for MIDI tracks lists available external MIDI devices as well as any virtual instruments that have been added to the Song.*

4.3.3 Recording an Audio Track

1. To begin recording, create an audio track from the Add Tracks window, set its input to Input 1 on your StudioLive AR-series mixer, and connect a microphone to the same input.



2. Select Record Enable on the track. Turn up the Input 1 level on your audio interface 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).



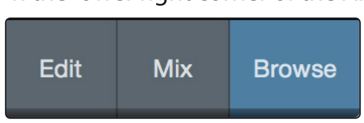
You are now ready to start recording. For complete instructions, please consult the Studio One Reference manual located in Help | Studio One Reference Manual.

4.3.4 Adding Virtual Instruments and Effects

You can add plug-ins and instruments to your Song by dragging-and-dropping them 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 preset without ever scrolling through a menu.

Opening the browser.

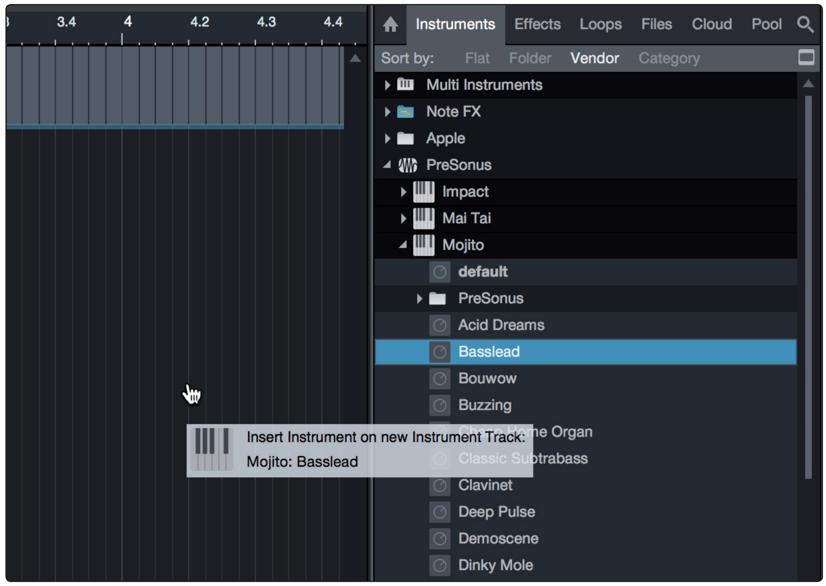
In the lower right corner of the Arrange window are three buttons:



- The Edit button opens and closes the audio and MIDI editors.
- The Mix button opens and closes the Mixer window.
- The Browse button opens the browser, 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.

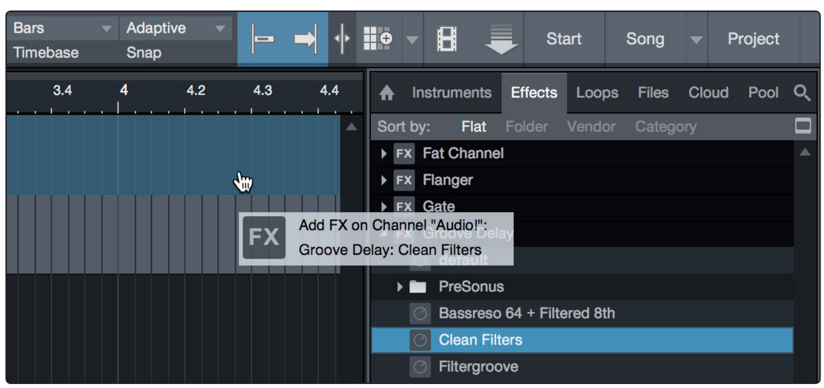
Drag-and-drop virtual instruments.

To add a virtual instrument to you session, open the browser and click on the Instrument button. 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.



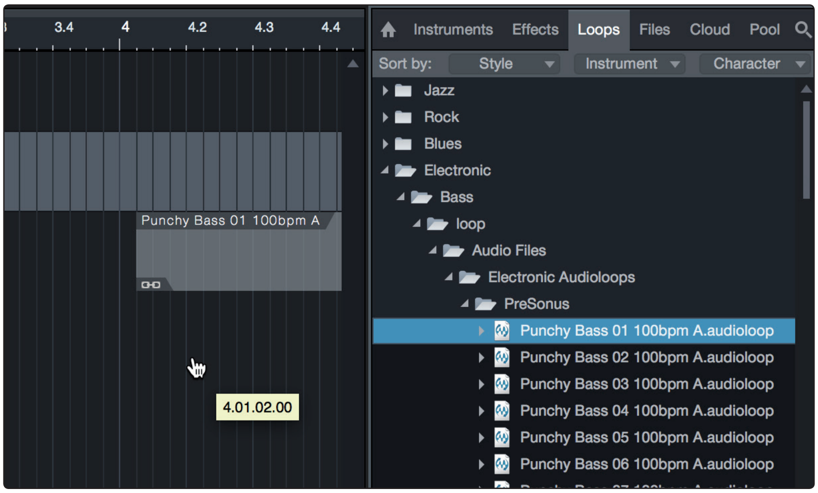
Drag-and-drop effects.

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



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 of the track.



5.0 Tutorials

5.1 Microphone Types

The AudioBox 22VSL and AudioBox 44VSL work 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 22/44VSL 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 22/44VSL, 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 22/44VSL. These are by no means the only ways to record these instruments. Microphone selection and placement is an art. For more information, visit your library or local bookstore, as there are many books and magazines about recording techniques. The Internet is also a great source of recording information, as are instructional videos. Some of these microphone-placement suggestions can be used in live applications, as well as for studio recording.

Grand Piano

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

Electric Guitar

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

Acoustic Guitar

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

Bass Guitar (Direct and Speaker)

Plug the electric bass guitar into a passive direct box. Connect the instrument output from the passive direct box to a bass amplifier. Place a dynamic microphone an inch or two away from the speaker and connect it to a AudioBox 22/44VSL 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.

5.2 A Brief Tutorial on Dynamics Processing

Studio One Artist software included with the AudioBox 22/44VSL comes with a variety of dynamics processing plugins. 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 Studio One Artist and its plugis. 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 lowest possible level. For example, if a processor states that the maximum input level before distortion is +24 dBu, and the output noise floor is -92 dBu, then the processor has a total dynamic range of 24 + 92 = 116 dB.

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

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

Why do we need compression?

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

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

Does every instrument need compression?

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

Why do the best consoles in the world put compressors on every channel?

The answer is simply that most instruments need some form of compression, often very subtle, to be properly heard in a mix.

Why do we need noise gates?

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

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

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

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

Overview
Hookup
Connecting to a Computer
Software: Virtual StudioLive SL Remote, and Studio One Artist
Tutorials
Technical Information
Troubleshooting and Warranty

performing, making the output level vary up and down unnaturally. A compressor can be applied to the signal to help correct this recording problem by reducing the louder passages enough to be compatible with the overall performance.

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

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

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

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

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

A vocal performance usually has a wide dynamic range. Transients (normally the loudest portions of the signal) can be far outside the average level of the vocal signal. Because the level can change continuously and dramatically, it is extremely difficult to ride the level with a console fader. A compressor/limiter automatically controls gain without altering the subtleties of the performance.

A solo guitar can seem to be masked by the rhythm guitars. Compression can make your lead soar above the track without shoving the fader through the roof.

Bass guitar can be difficult to record. A consistent level with good attack can be achieved with proper compression. Your bass doesn't have to be washed out in the low end of the mix. Let the compressor/limiter give your bass the punch it needs to drive the bottom of the mix.

Compressors — Terminology

Threshold. The compressor threshold sets the level at which compression begins. When the signal is above the threshold setting, it becomes eligible for compression. Basically, as you turn the threshold knob counterclockwise, more of the input signal becomes compressed (assuming you have a ratio setting greater than 1:1).

Ratio. The ratio is the relationship between the output level and the input level. In other words, the ratio sets the compression slope. For example, if you have the ratio set to 2:1, any signal levels above the threshold setting will be compressed such that for every 1 dB of level increase into the compressor, the output will only increase 0.5 dB. This produces a compression gain reduction of 0.5 dB/dB. As you increase the ratio, the compressor gradually becomes a limiter.

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.

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.

5.2.3 Compression Settings: Some Starting Points

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.

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

5.3 Equalizers

Studio One Artist includes several EQ plugins. This section is 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 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 22/44VSL, 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."

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.

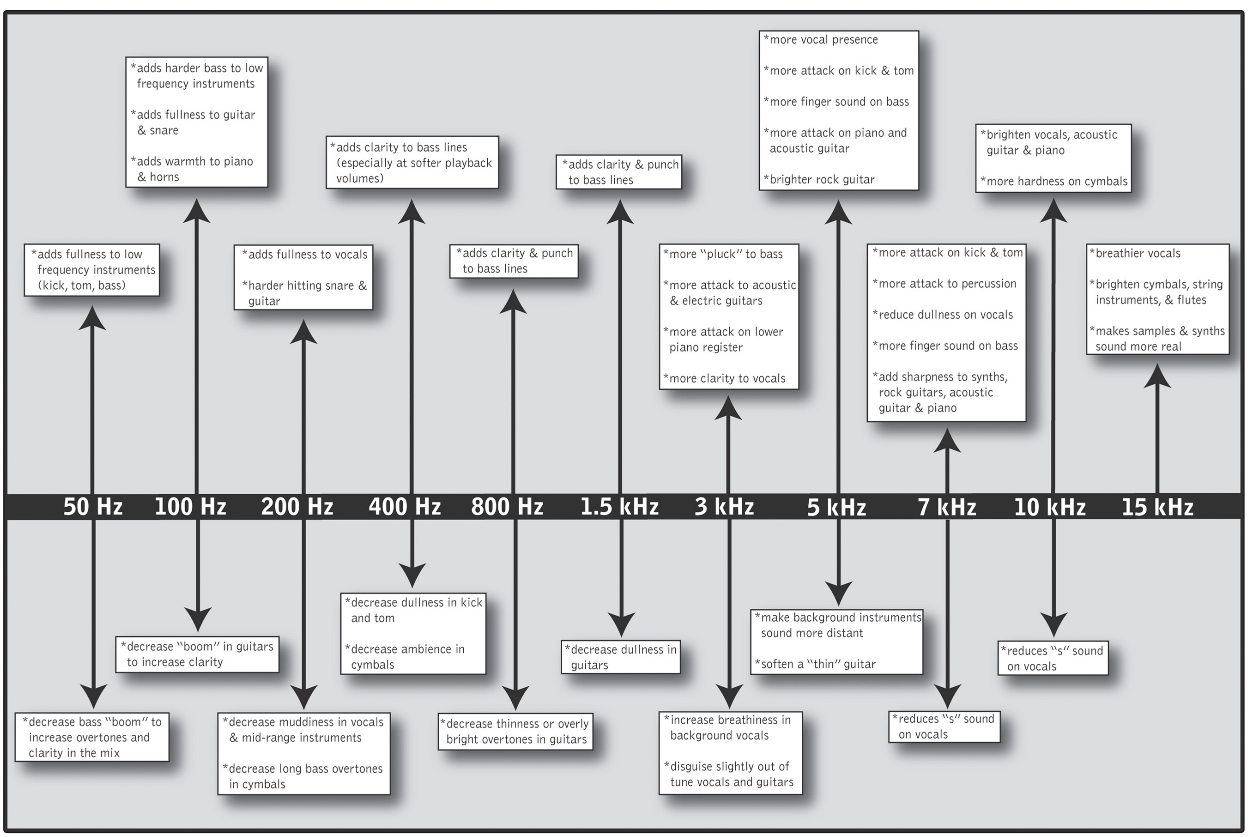
- **Never be afraid of taking a risk.** The best EQ tricks were found by mad scientists of sound. With every instrument, there are frequencies that can be attenuated or boosted to add clarity or fullness. Altering the wrong frequencies can make an instrument shrill, muddy, or just downright annoying. 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

Table 2



5.3.3 General EQ Suggestions

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

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

5.4 Digital Effects

Studio One Artist also includes an assortment of time-based and modulation effects. The following is a brief description of how each type works.

5.4.1 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.

5.4.2 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.

5.4.3 Modulation Effects

Chorus

As its name indicates, a Chorus effect creates copies of a single source signal to artificially create the impression that there is more than instrument playing, voice singing, etc. This ensemble effect is created using a series of short, continuously varying delays that produce slight pitch-shifts and add fullness to a sound.

LFO Speed and Width: The copies are delayed using a low-frequency oscillator. Some chorus effects allow you to adjust the speed and width of the waveform being applied to modulate the source signal.

Depth: The depth control affects how much the total delay time changes over time. As the delay time changes, you can hear slight frequency modulations.

Phase

Phase shifting creates a copy of the source signal and shifts the copy in time relative to the original signal, creating from 0 to 360 degrees of phase difference throughout the frequency spectrum. The shifted signal is blended with the source signal so that you can hear the copy moving in and out of phase with the original. This creates a characteristic "swoosh."

Flange

Flanging is a type of phase shifting. It is created by splitting an audio signal into two identical signals; applying a constantly varying, short delay to one signal; and mixing it with the unaltered signal. This results in a swept, "swooshy" effect. The effect was originally created by mixing the outputs of two synchronized tape decks playing the same material. By pressing a finger against the flange (top) of one tape reel, the speed of one machine was slowed slightly, creating phase shifts.

6.0 Technical Information

6.1 AudioBox 22VSL Specifications

Microphone Preamp

Type	Combo, XLR, female, balanced
Mic Preamp EIN	-129 dB, 20 kHz BW, max gain, Rs=40Ω, A-wtd
Frequency Response	20 Hz - 20 kHz, +/- 0.3 dB, unity gain
THD+N	0.005%, 0 dBu, 1 kHz, unity gain, 20 kHz BW, A-wtd
S/N Ratio	94 dB, 0 dBu, 1 kHz, unity gain, 20 kHz BW, unwtd
Gain Control Range	-15 dB to +65 dB
Input Max Headroom	+10 dBu, < 0.5% THD
Input Impedance	1.7 kΩ
Phantom Power	+48 VDC, 10 mA total

Instrument Input (channels 1 and 2 only)

Connector Type	Combo, ¼"TS, female, unbalanced
Frequency Response	20 Hz - 20 kHz, +/- 0.3 dB, unity gain
THD+N	0.006%, 0 dBu, 1 kHz, unity gain, 20 kHz BW, A-wtd
S/N Ratio	89 dB, 0 dBu, 1 kHz, unity gain, 20 kHz BW, unwtd
Gain Control Range	-30 dB to +50 dB
Input Max Headroom	+15 dBu, < 0.5% THD
Input Impedance	1 MΩ

Headphone Output:

Connector Type	¼"TRS, female, stereo
Maximum Power	30 mW/ch @ 60Ω load
Frequency Response	20 Hz - 20 kHz, ±0.5 dB, max gain
THD+N	0.08%, 1 kHz, max gain, 20 kHz BW, A-wtd
S/N Ratio	90 dB, 1 kHz, max gain, 20 kHz BW, unwtd

Analog Outputs:

Connector Type	¼" TRS, female, impedance balanced
Frequency Response	20 Hz - 20 kHz, ± 0.2 dB
THD+N	0.002%, 1 kHz, 20 kHz BW, A-wtd
S/N Ratio	107 dB, 1 kHz, max gain, 20 kHz BW, unwtd
Output Level	+10 dBu
Output Impedance	51Ω

Power:

Power (AB22)	USB Bus Power, 5 VDC, 500 mA

Digital:

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
ADC Dynamic Range	114 dB, 48 kHz sample rate, A-wtd
DAC Dynamic Range	114 dB, 48 kHz sample rate, A-wtd
MIDI I/O	5-pin DIN connectors

6.2 AudioBox 44VSL 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, 32 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 (channels 3 and 4 only):

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

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 Output:

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:

Power	2.1 mm barrel connector, 12 VDC, 1A

Digital:

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
ADC Dynamic Range	114 dB, 48 kHz sample rate, A-wtd
DAC Dynamic Range	114 dB, 48 kHz sample rate, A-wtd
MIDI I/O	5-pin DIN connectors

Overview
Hookup
Connecting to a Computer
Software: Virtual StudioLive SL Remote, and Studio One Artist
Tutorials
Technical Information
Troubleshooting and Warranty

7.0 Troubleshooting & 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 and Studio One™ digital audio workstation 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 1-225-216-7887, Monday through Friday, between the hours of 9 a.m. and 5 p.m. Central Time (GMT -06:00 CST). PreSonus technical support is available via email during the same hours at support@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.

Input Phasing While Monitoring in VSL

If you are using a DAW at the same time as AudioBox VSL, make sure that the input channel in your DAW has software monitoring disabled. You will be monitoring through AudioBox VSL, not your DAW.

7.2 PreSonus AudioBox 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

Declaration of Conformity

Responsible Party: **PreSonus Audio Electronics**

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

Phone: **225-216-7887**

declares that **AudioBox™ 22VSL/AudioBox 44VSL** 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

Baton Rouge • USA • www.presonus.com

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™ 22/44VSL
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 les limites applicables aux appareils numériques de classe A/de classe B (selon le cas) prescrites dans le règlement sur le brouillage radioélectrique édicté par le ministère des communications du Canada.

PreSonus®

18011 Grand Bay Ct. • Baton Rouge,
Louisiana 70809 USA• 1-225-216-7887

www.presonus.com

Part# 820-AB0013-D