AudioBox™ USB

Owner's Manual



PreSonus™
www.presonus.com

Important Safety Instructions

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

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

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

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

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

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



13. Unplug this apparatus during lightning storms or when unused for long periods of time.
14. Servicing is required when the apparatus has been damaged in any way, such as if a power-supply cord or plug is damaged; or liquid has been spilled, or objects have fallen, into the apparatus; or if the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped. All PreSonus products in the USA should be serviced at the PreSonus factory in Baton Rouge, Louisiana. If your PreSonus product requires a repair, contact support@presonus.com to arrange for a return-authorization number. Customers outside the USA should contact their local distributor. Your distributor's contact information is available at www.presonus.com.
15. The apparatus shall be connected to a Mains power outlet with a protective grounding/earthing connection.
16. Where the Mains plug or an appliance coupler is used as the disconnect device, the disconnect device shall remain readily operable.

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

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

REACh This product is compliant with the European Union Directive EC1907/206 for the Registration, Evaluation, Authorization, and Restriction of chemicals (REACh) and contains none or less than 0.1% of the chemicals listed as hazardous chemicals in the REACh regulation.



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

CE This product complies with the European Union Council Directives and Standards relating to electromagnetic compatibility EMC Directive (2006/95/EC) and the Low Voltage Directive (2004/108/EC).

AudioBox™ USB

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1.0 Overview

1.1 Introduction



Thank you for purchasing the PreSonus AudioBox™ USB. PreSonus Audio Electronics has designed the AudioBox USB utilizing high-grade components to ensure optimum performance that will last a lifetime. Loaded with 2 high-headroom, Class A, microphone preamplifiers; a built-in 2x2 USB 1.1 recording and playback engine; MIDI I/O; and more, AudioBox USB breaks new boundaries for music performance and production. All you need is a computer with a USB connection, a few microphones and cables, powered speakers, and your instruments, and you are ready to record in the studio or in front of a live audience!

We encourage you to contact us with questions or comments regarding this product. You can reach us by email at support@presonus.com or call us at 1-225-216-7887 between 9 a.m. and 5 p.m. Central Time (GMT -06:00 CST). 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 USB!

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 USB 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 USB Hardware Features

- 24-bit/48 kHz sampling rate
- 2 Class A microphone preamplifiers
- 2 Instrument inputs
- MIDI I/O
- 2x2 USB 1.1 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 more than 6 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)
- More than 6 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

Your AudioBox USB package contains the following:



PreSonus AudioBox USB recording interface



6' (1.8m) USB cable



Software library containing

- PreSonus Studio One Artist program DVD plus gigabytes of third-party content
- AudioBox driver and manual CD

2.0 Hookup

2.1 Front-Panel Connections





Microphone/Instrument Inputs. Both channels of the AudioBox USB have Mic/Instrument combo jacks. This convenient connector accepts either a ¼-inch phone plug or an XLR plug.

- **Microphone Inputs.** The XLR inputs on channels 1 and 2 are equipped with two custom PreSonus microphone preamplifiers. These preamps work great with all types of microphones including dynamic, ribbon, and condenser microphones.

Power User Tip: *Dynamic microphones and ribbon microphones (which are a special type of dynamic mic) are generally lower-output devices that, with few exceptions, require no external power source. Sending phantom power to a ribbon mic that doesn't require it can cause severe damage to the mic – usually beyond repair. Condenser microphones are generally more sensitive than dynamic and ribbon microphones and typically require external +48V phantom power. Always review your microphone's documentation and follow its recommended operating practices.*

- **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 USB becomes an instrument preamplifier.

Power User Tip*: Active instruments are those that have an internal preamp or a line-level output. Active instruments should be plugged into a line input rather than into an instrument input. Plugging a line-level source into the instrument inputs on the front of the AudioBox USB 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 USB. 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 USB 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 some 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 Instrument inputs:** 40 dB of variable gain (-10 to +30 dB)
- **TS ¼-inch Instrument Level inputs:** 60 dB of variable gain (0 dB to +60 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.*



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 Studio One or another DAW, it is important that you turn the Mixer knob all the way to the Playback 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 rear 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 10 dB.



Power LED. This LED will illuminate red when the AudioBox is properly powered and synced to a USB connection.

2.2 Rear-Panel Connections





¼-inch Phones (headphone) Jack. This is where you connect headphones to your AudioBox USB.



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

Power User Tip: *You must connect speakers or headphones directly to your AudioBox USB in order to monitor playback from your computer.*



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.

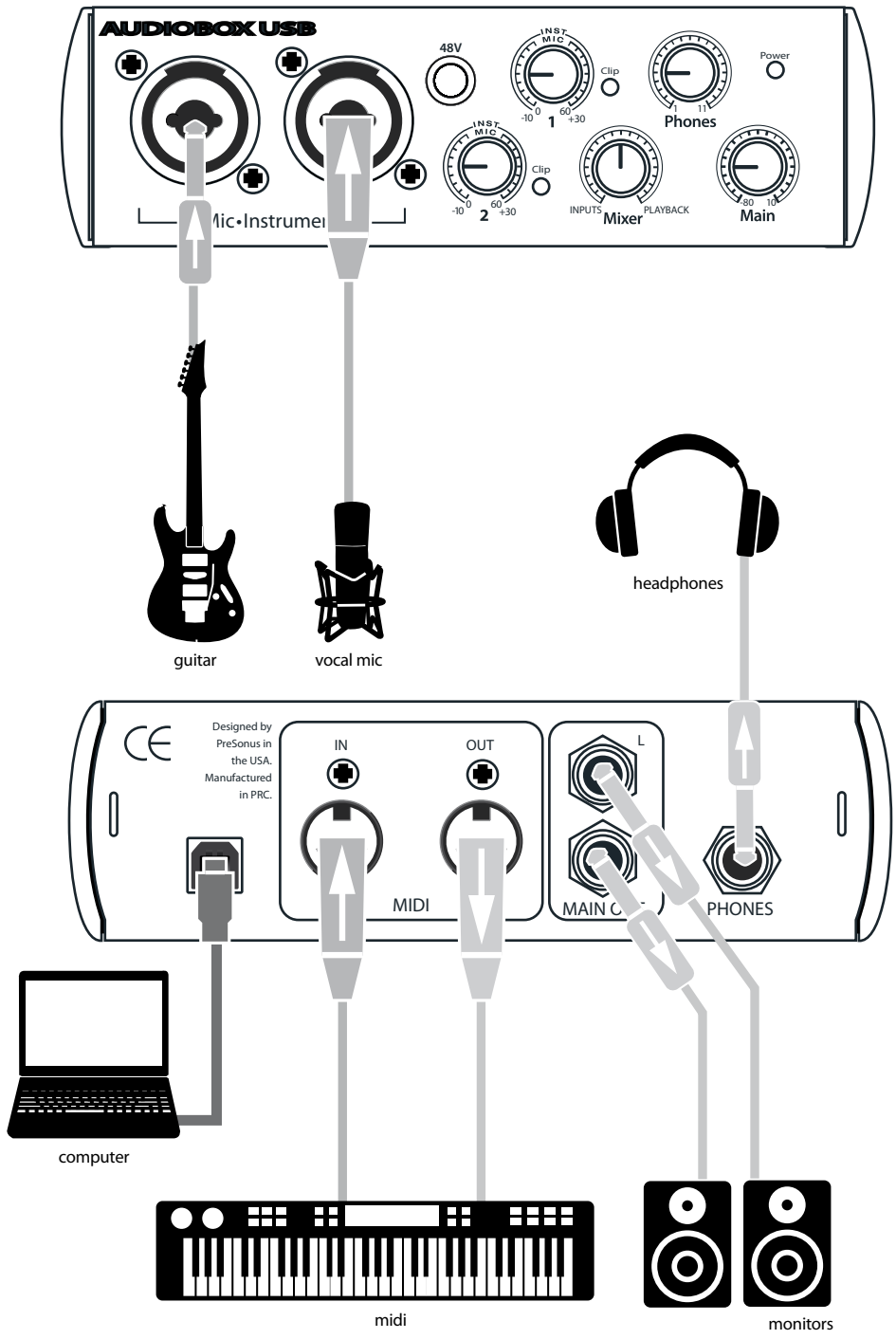
> ***NOTE:*** *MIDI is not audio but is frequently used to trigger or control an audio source (such as a plug-in or synthesizer). It's important to ensure that your MIDI data is correctly sent and received by the appropriate hardware or software devices. If the devices generate audio, you may also need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



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

Your AudioBox USB is compatible with all types of USB ports (1.1, 2.0, and 3.0).

2.3 Basic Hookup Diagram



3.0 Connecting to a Computer

Your AudioBox USB is a rugged USB 1.1 audio interface with zero-latency analog monitoring control and professional audio tools.

3.1 System Requirements

Below are the minimum computer-system requirements for PreSonus Studio One Artist and AudioBox USB.*

Mac

- **Operating Systems:**
  - Mac OS X 10.6 or higher
- **Hardware:**
  - Minimum: Intel® Core™ Duo 1.5 GHz processor, 2 GB RAM
  - Recommended: Intel Core 2 Duo or Intel iCore processor or better, 4 GB or more RAM

Windows

- **Operating Systems (32- or 64-bit)**:
  - Windows 7, Vista
- **Hardware**:
  - Minimum: Intel Core Duo or AMD Athlon™ X2 processor, 2 GB RAM
  - Recommended: Intel Core 2 Duo or AMD Athlon X4 or better, 4 GB or more RAM

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

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

3.2 Installation for Windows

The AudioBox installer requires that your AudioBox USB be connected to your computer. Connect your AudioBox USB to an available USB port and insert the Installation CD into your disc drive. The AudioBox installer will auto-launch and take you through each step of the installation process. Please read each message carefully.

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

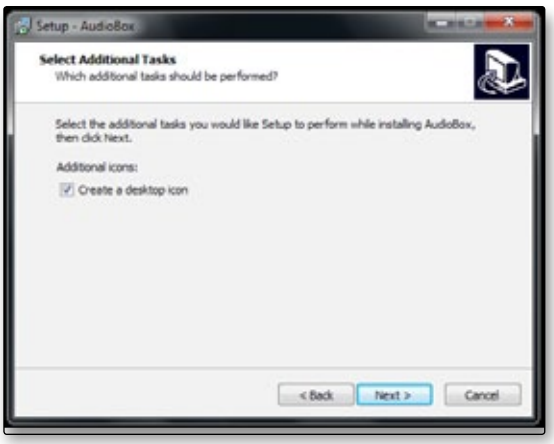


Windows Vista Users: If you see any Windows Security alerts, click "Install this driver software anyway."



1. The installer will open to the Welcome screen.

   Click "Next."



2. You will be asked if you would like the installer to automatically create an icon on your desktop for the AudioBox control panel.

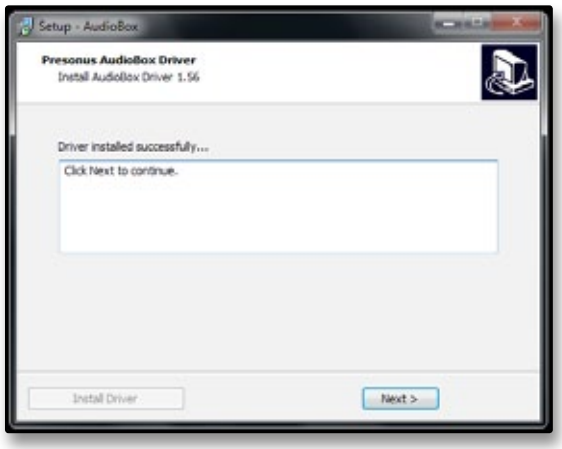
   This control panel allows you to adjust the buffer size and sample rate of your AudioBox USB. Creating a desktop icon will provide you with quick access to these controls.

   Click "Next."



3. The installer will now scan your computer, you will be prompted to begin the installation. If your AudioBox is not detected by the installer, you will prompted to connect it.

   Click "Install Driver."



4. Once the driver has been installed successfully, you will be alerted.

   Click "Next" to finish the installation.



5. In order for the driver to operate properly, you must restart your computer.

   Click "Finish" to automatically reboot your computer.

6. Once your computer has restarted, your AudioBox USB is now ready to use!

3.3 Installation for Mac OS X

Your AudioBox USB is a class-compliant Core Audio device. No installation is necessary on Mac computers. Simply connect your AudioBox USB to your Mac to begin using it.

3.4 Using the AudioBox USB with Popular Audio Applications

Complete setup instructions for Studio One Artist and a brief tutorial on its features can be found in Section 4 of this manual. However, you can use your AudioBox USB 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 USB 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 USB** from the device menu.
6. You will be asked if you'd like to relaunch Logic. Click "**try (re)launch**."
7. Your **AudioBox USB** 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 USB. When you are done, close this window.
9. You are now ready to use your AudioBox USB.

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 USB** from the **ASIO Driver** drop-down list.
5. Click "**Switch**" to begin using the AudioBox USB 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 USB."**
8. Go to **Options | Audio**... and click on the **General** tab.
9. Set the **Playback Timing Master** to "**PreSonus AudioBox USB... DAW Out 1**."
10. Set the **Recording Timing Master** to "**PreSonus AudioBox USB... 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 USB**
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 USB's inputs and outputs for each track created in Live.

3.5 Windows Control Panel

Your AudioBox USB features a control panel on Windows systems that allows you to adjust the buffer size and sample rate.

On Mac systems, these functions are accessible from within your DAW application.

Run at Startup

Launches the AudioBox USB Control Panel Automatically on Startup.



When this is enabled, the AudioBox USB Control Panel will automatically launch each time you boot your Windows.

Sample Rate Selector

Changes Sample Rate.



Sets the sample rate to 44.1 or 48 kHz.

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

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

ASIO Buffer Size (Windows Only)

Changes the Buffer Size.



You can set the buffer size from 32 to 2,048 samples. The buffer size determines the roundtrip time it takes audio data to be converted from analog to digital and back to analog. As a general rule, the higher the buffer size, the better the system performance, but the less playable virtual instruments and the like become. In general, 512 samples (11 to 12 milliseconds) will provide you with a large enough buffer for optimum system performance but small enough to be unobtrusive. You should set your buffer size prior to launching your host application.

4.0 Studio One Artist Quick Start



Your AudioBox USB comes with Studio One Artist recording and production software. Whether you are about to record your first album or your fiftieth, Studio One Artist provides you with all of the tools necessary to capture and mix a great performance. As a valued PreSonus customer, you are also eligible for a discount upgrade to Studio One Producer or 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 AudioBox USB drivers and connected your AudioBox USB to your computer, you can use the included PreSonus Studio One Artist music-production software to begin recording, mixing, and producing your music. To install Studio One Artist, insert your installation disc into your computer's DVD drive. Follow the onscreen instructions to complete the installation process.

Running the Studio One Installer



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

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

Creating a User Account



1. After installing Studio One Artist, launch the program, and the Activate Studio One menu will appear. Click on the Activate or Upgrade link to begin.



2. If you are a new Studio One user, you will need to create a user account. Click on the "Create New Account" link if your computer is connected to the Internet.
   - If your computer is not connected to the Internet, skip to "Activating Studio One Artist Offline."
   - If you already have a PreSonus user account, you can skip to "Activating Studio One Artist On Line."



3. Fill out the user registration form. You will be asked to create a username and password. This information will be used to access your PreSonus user account on the PreSonus Web site. With this account, you can manage registration for all of your PreSonus hardware and software products. You will be notified of, and will have access to, important information and updates related to your PreSonus products, ensuring you get the best performance possible from them.

   Your username and password will also provide you with access to the PreSonus user forums to chat with other PreSonus users as well as PreSonus employees.



4. Once your user account has been successfully created, you will be alerted that the activation email has been sent to the email address that you provided.

   Don't forget to activate your PreSonus user account the next time you check your email!

Activating Studio One Artist Online



Now that you have created a user account, you can activate your copy of Studio One Artist. Click on the Activate Online link and enter your previously created account username, password, and the product key you received with the Studio One Artist installation disc.

Your Studio One Product Key is located on the Studio One Artist disc wallet.

Click on the Activate button to finish the activation process.

Activating Studio One Artist Offline

1. If your computer is not connected to the Internet, visit http://www.presonus.com/registration/ on an Internet-connected computer to create your account.

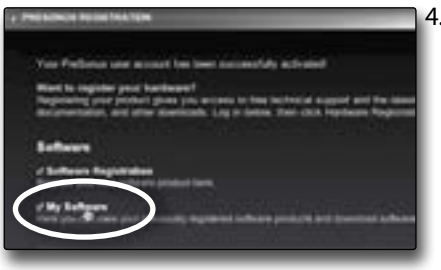


2. Once you have created your user account, log in and click on the Software Registration link.



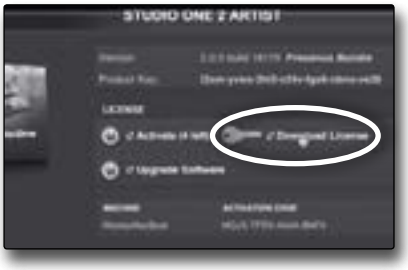
3. Enter the Product Key located on the Studio One Artist disc wallet.

   Click "Submit."



4. When product key is registered, you see a list of links to download Studio One Artist and its bundled content. Click the Registration Home link to go back to the main page.

   Click on the "My Software" link.



5. Click on the "Download License" link.

   A file called "Studio One 2 Artist.license" will be saved onto your computer. You will need to copy this onto a piece of removable media and transfer it onto the computer on which you want to run Studio One Artist.



6. Launch Studio One Artist.

   From the Activate Studio One Menu, click on the "Activate Offline" link.



7. Drag the "Studio One 2 Artist.license" file onto the Offline Activation window to complete the activation.

Installing Bundled Content for Studio One Artist



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

Upon completing the Studio One Artist installation and activation process, the Studio One Content Installer will appear.

1. Click on "Install content from DVD or local folder."



2. At the top of the installation menu, select the source from which the content will be installed, as well as the location where you wish to install the content. The source of the content will be the same DVD from which you installed Studio One Artist. By default, Studio One Artist will point to your DVD drive as the content source. Listed in the installation menu are separate entries for each available item. Click the check box next to each item you wish to install, then click on the Install Packets button at the bottom left of the menu to install the selected content.

   When finished installing content, click on the Done button to exit the menu.

Power User Tip: *Studio One Artist content can be installed at any time by accessing the Studio One 2>Studio One Installation... menu. If you choose not to install any portion of the content, you can install it at a later time.*

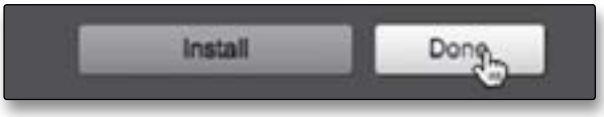
Installing Third-Party Content

To Install any of the third-party content, click on the arrow button next to its name in the content list (Celemony Melodyne Trial, Native Instruments Komplete Player).



Notice that next to Native Instruments Komplete there is serial number listed. You will need this number to authorize this plug-in the first time you open it.

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



4.2 Enabling the Audio Driver

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

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

Start Page: Setup Area

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



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

Selecting a Different Audio Driver from the Start Page.

If you do not see "PreSonus AudioBox USB" on the Start page when you launch Studio One, click on the Configure Audio Devices link in the Setup area to open the Audio Setup Options window.

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

After you have verified that the PreSonus AudioBox driver has been detected, please continue to the next section to set up your external MIDI devices. If you do not have any MIDI devices to connect at this time, please skip to Section 4.4.

4.3 Configuring MIDI Devices

From the External Devices window in Studio One Artist, you can configure your MIDI keyboard controller, sound modules, and control surfaces. This section will guide you through setting up your MIDI keyboard controller and sound module. Please consult the Reference Manual located on your Studio One Artist installation DVD for complete setup instructions for other MIDI devices.

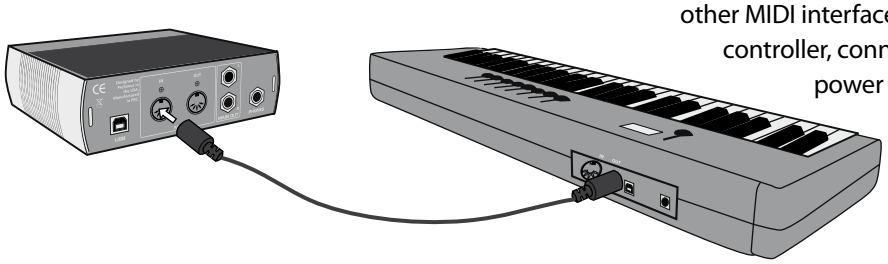
If you are using a third-party MIDI interface or USB MIDI-controller keyboard, you must install the drivers for these devices before beginning this section. Please consult the documentation that came with your MIDI hardware for complete installation instructions.

Setting Up an External MIDI Keyboard Controller from the Start Page

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

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.

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





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.



4. The Add Device window will launch.

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

6. 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.



7. In the Receive From drop-down menu, select the MIDI-interface input from which Studio One Artist will receive MIDI data. In this case, the AudioBox USB.

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



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.

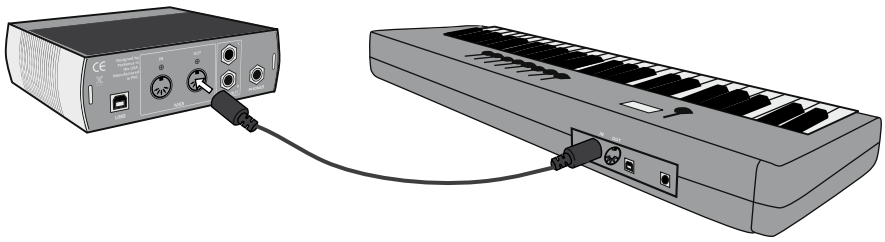
Click OK.

If you have a sound module that you'd like to connect, leave the External Devices window open and proceed to the next part of this section. If not, you can close this window and skip to Section 4.4.

Setting up an External MIDI Sound Module from the Start Page

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

1. Connect the MIDI In of your external sound module to the MIDI Out of your AudioBox USB or other MIDI interface.





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



3. The Add Device window will launch.

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

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

6. In the Send To drop-down menu, select the MIDI-interface output from which Studio One Artist will send MIDI data to your sound module—in this case, via the AudioBox USB.

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

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

4.4 Creating a New Song

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



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



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

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

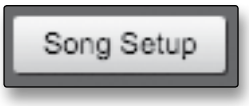


4. To begin a new Song, select Empty Song from the Templates list. At this point, you should give your Song a name and select your preferred sample rate and bit depth. You can also determine the length of your Song and the type of time format you would like to use (Notation Bars, Seconds, Samples, or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

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



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



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



8. Go to Song>Song Setup menu and click on "Audio I/O Setup."

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

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



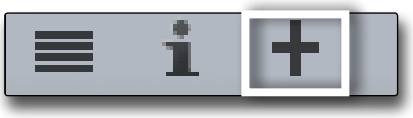
11. Click on the Outputs tab, and you will see all of the available outputs on your AudioBox USB. We recommend that you create stereo outputs for the Main outputs on your AudioBox USB.

12. In the lower right corner, you will see the Audition select menu. This allows you to choose the output from which you will audition audio files prior to importing them into Studio One Artist. In general, you will want this to be the main output bus.

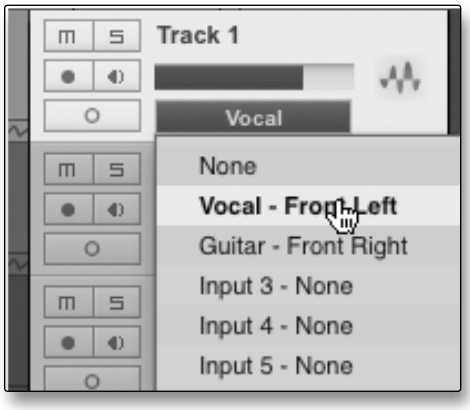
13. If you would like this output configuration available every time you launch Studio One Artist, click the Make Default button.

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

4.5 Creating Audio Tracks







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

2. In the Add Tracks window, you can select the number and type of tracks you'd like to create (Mono Audio, Stereo Audio, Folder, Instrument, or Automation) and can customize the track name and color as well as add a preset rack of effects to the track.

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

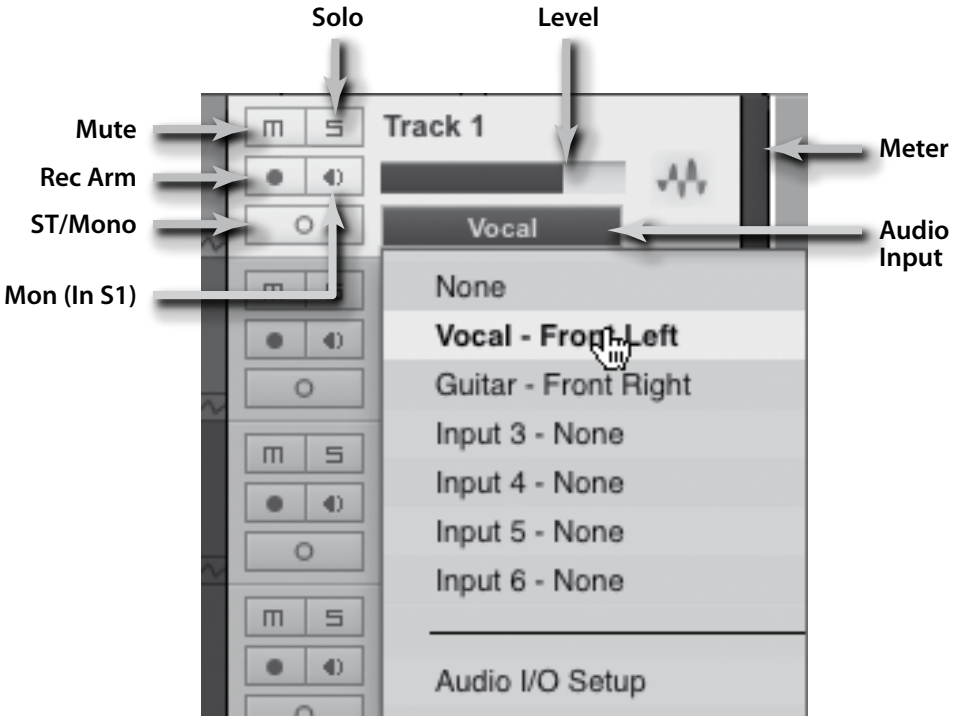
4. If you would like to add a track for each of the available inputs and have the routing automatically assigned, simply go to Track | Add Tracks for All Inputs.

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

6. Connect a set of headphones to the AudioBox headphone output. If you wish to listen to your AudioBox USB with studio monitors, make sure to connect them to the AudioBox's Main outputs.

   You are now ready to record. For complete instructions, please consult the Studio One Reference Manual, which is located on your Studio One Artist installation DVD.

Anatomy of an Audio Track



4.6 Creating MIDI Tracks



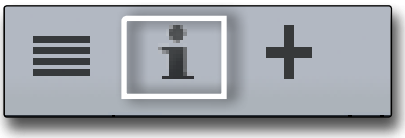
1. Click on the Add Tracks button. When the Add Tracks window launches, select Instrument as the track format and click the OK button.



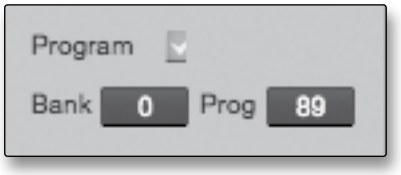
2. To assign your MIDI input, click on the MIDI Inputs list and select your external sound module.

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

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



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



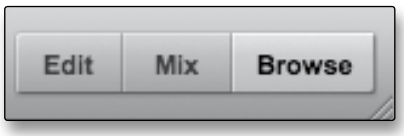
6. At the bottom of the Inspector menu, you will see your Bank and Program selections. From here, you can remotely change the patch on your sound module.

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

4.7 Adding Virtual Instruments and Plug-in Effects to Your Song

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

Opening the Browser



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

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

Drag-and-Drop Virtual Instruments



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

Drag-and-Drop Effects



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

5.0 Tutorials

5.1 Microphone Types

The AudioBox USB 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 USB 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 because a USB microphone is, in effect, an audio interface, we recommended that you not use them with the AudioBox USB, 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 USB. 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 USB 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 is included with the AudioBox USB. This powerful DAW includes an assortment of dynamics plug-ins. 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. 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 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. 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 Studio One Artist.

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 also includes several EQ plug-ins. Here's a brief explanation of how an EQ functions, as well as some charts to help you navigate the frequency ranges of various instruments so you can quickly choose the best EQ settings for your recordings and live mixes.

5.3.1 What is an EQ?

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

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

Parametric EQ

The parametric EQ and semi-parametric EQ are the most common equalizers found in recording and live situations because they offer continuous control over all parameters. A parametric EQ offers continuous control over the audio signal's frequency content, which is divided into several bands of frequencies (most commonly three to seven bands). A fully parametric EQ like those in the StudioLive 24.4.2 offers control over the bandwidth (basically, the range of frequencies affected), the center frequency of the band, and the level (boost/cut) of the designated frequency band. It also offers separate control over the Q, which is the ratio of the center frequency to the bandwidth. A semi-parametric EQ provides control over most of these parameters but the Q is fixed. Some devices, such as the StudioLive 16.4.2 and 16.0.2 and the AudioBox VSL-series interfaces, 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.

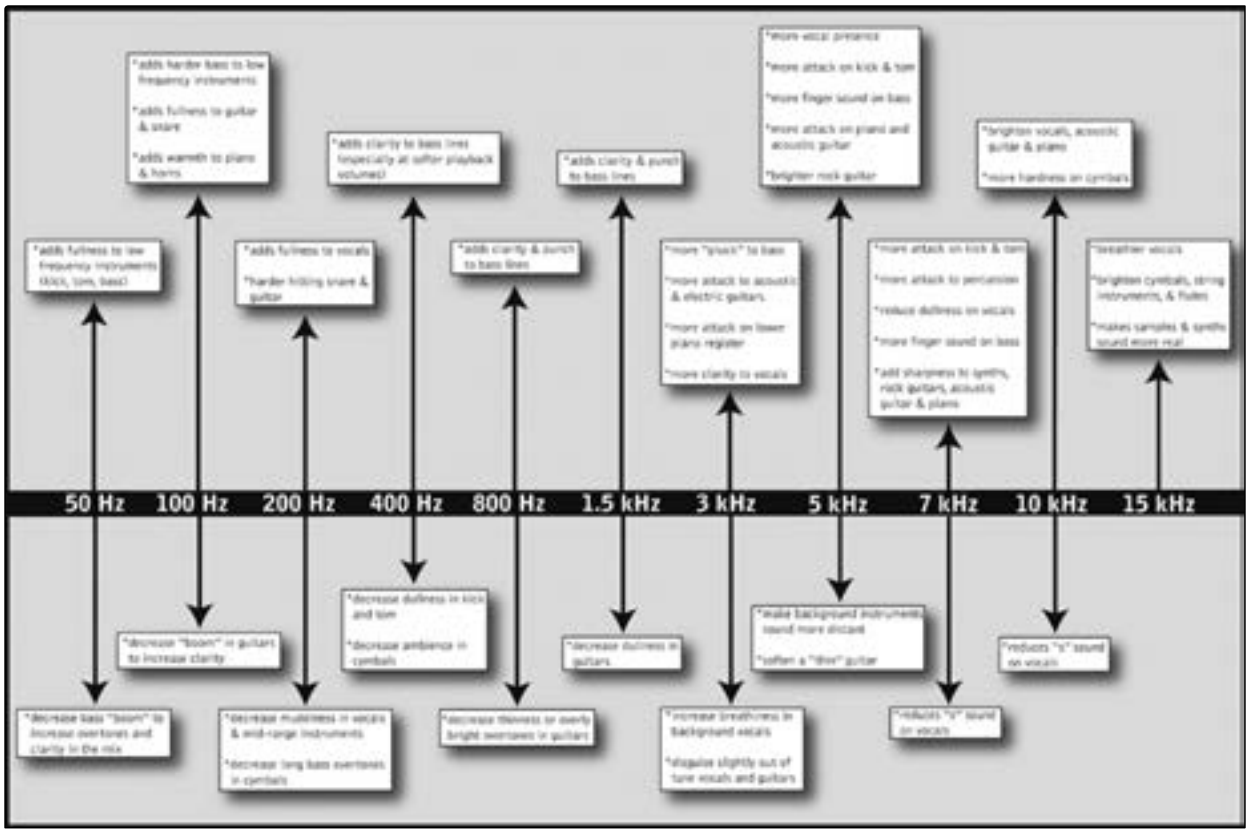
Additional advice:

- **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


50 Hz
100 Hz
200 Hz
400 Hz
800 Hz
1.5 kHz
3 kHz
5 kHz
7 kHz
10 kHz
15 kHz

5.3.3 General EQ Suggestions

Studio One Artist includes two different EQ plugins, Channel Strip and Pro EQ. Like every plugin in Studio One Artist, both EQ plugins come with a complete collection of factory presets and allow you to both customize factory presets and create your own. What follows are a few generic EQ settings for some of the most common 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 includes an array of time-based and modulation effects. What follows 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, slight frequency modulations can be heard.

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 USB Specifications

Microphone Preamp

Type	Combo, XLR, female, balanced
Mic Preamp EIN	-115 dB, 20 kHz BW, max gain, Rs=40Ω, A-wtd
Frequency Response	14 Hz - 70 kHz, +/- 0.3 dB, unity gain
THD+N	0.008%, 0 dBu, 1 kHz, unity gain, 20 kHz BW, A-wtd
S/N Ratio	>95 dB, 0 dBu, 1 kHz, unity gain, 20 kHz BW, unwtd
Gain Control Range	0 dB to +35 dB
Input Max Headroom	-3 dBu, < 0.5% THD
Input Impedance	1200Ω
Phantom Power	+48 VDC, 10 mA total

Instrument Input

Connector Type	Combo, ¼"TS, female, unbalanced
Gain Control Range	-30 dB to +50 dB
Input Impedance	0.5 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

Power:

Power	USB bus power

Digital:

Host Interface	USB 1.1
Bit Depth	24-bit
Sample Rates	44.1 kHz, 48 kHz,
Reference Level of 0dBFS	+4 dBu
ADC Dynamic Range	102 dB, 48 kHz sample rate, A-wtd
DAC Dynamic Range	110 dB, 48 kHz sample rate, A-wtd
MIDI I/O	5-pin DIN connectors

7.0 Troubleshooting and Warranty

7.1 Troubleshooting

Many technical issues can arise when using a standard computer as a digital audio workstation (DAW). PreSonus can only provide support for issues that directly relate to the AudioBox interface 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 connection.

Input Phasing While Monitoring using Studio One or Another DAW

Verify that the mixer knob on the front of the AudioBox is turned fully clockwise to the Playback position.

If you prefer to use the analog zero-latency monitoring on your AudioBox, make sure that the input channel in your DAW has software monitoring disabled. You will be monitoring through the analog bus on your AudioBox, not your DAW, using the Mixer knob on the front of your AudioBox to blend the playback from your computer with your analog input signal.

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

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Declaration of Conformity

Responsible Party: **PreSonus Audio Electronics**

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

Phone: **225-216-7887**

declares that **AudioBox™ USB** complies with Part 15 of the FCC rules.

Operation is subject to the following two conditions:

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

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

PreSonus

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

Jambalaya

Ingredients:

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

Cooking Instructions:

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

Serves 20

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AudioBox™ USB
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.

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