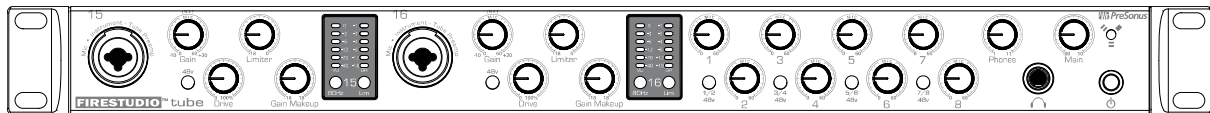


FIRESTUDIO TUBE

24-bit/96k Recording Interface
w/ Two Class A Vacuum Tube "SuperChannels"



User's Manual
Version 1.0



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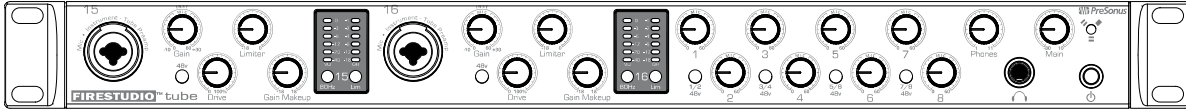
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OVERVIEW

1.1 INTRODUCTION



Thank you for purchasing the PreSonus FireStudio Tube. PreSonus Audio Electronics has designed the FireStudio Tube utilizing high-grade components to ensure optimum performance that will last a lifetime. Loaded with 24-bit 96kHz converters, two tube microphone preamplifiers with built-in limiter, eight PreSonus X-MAX microphone preamplifiers, six line level inputs and Cubase LE 48-track music production software, the FireStudio Tube is ready to go out of the box for professional quality computer recording. All you need is a computer with FireWire connection, a few microphones and cables along with your instruments and you are ready to record!

We encourage you to contact us at 225-216-7887 with any questions or comments you may have regarding your PreSonus FireStudio Tube. PreSonus Audio Electronics is committed to constant product improvement, and we value your suggestions highly. We believe the best way to achieve our goal of constant product improvement is by listening to the real *experts*, our valued customers. We appreciate the support you have shown us through the purchase of this product.

We suggest you use this manual to familiarize yourself with the features, applications and correct connection procedure for your FireStudio Tube before trying to connect it to your computer. This will hopefully alleviate any unforeseen issues that you may encounter during installation and set up.

Thank you, once again, for buying our product, and we hope you enjoy your FireStudio Tube!

OVERVIEW

1.2 FEATURES

The FireStudio Tube is a powerful and affordable computer recording system complete with everything needed for real-world recording applications. The FireStudio Tube comes complete with two “SuperChannels” featuring class A Vacuum tube microphone and instrument preamps with built-in analog limiter, eight high-quality PreSonus X-MAX microphone preamps, MIDI I/O, and rock solid drivers, as well as a plethora of music recording and production software.

The FireStudio Tube includes PreSonus ProPak Software Suite with Cubase LE 4 and over 2 GB of plug-ins, drum loops and samples – giving you everything you need for professional music recording and production.

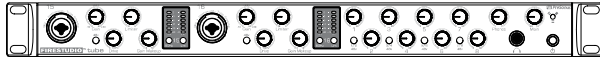
Summary of features

- High-speed FireWire (IEEE 1394)
- 24-bit / 96 kHz sampling rate
- 16 inputs/6 outputs simultaneous record/playback channels
- 2 SuperChannels (tube microphone/instrument preamplifier with analog limiter)
- 8 XMAX microphone preamplifiers
- 6 balanced TRS outputs
- MIDI I/O
- 16x10 FireControl DSP mixer
- JetPLL jitter control for improved imaging and clarity
- High performance A/D D/A converters
- Cubase LE 4 Music Production Software included
- PreSonus ProPak Software Suite (over 2 GB of plug-ins, loops and samples)
- OS X and Windows compatible

OVERVIEW

1.3 WHAT IS IN THE BOX

Your FireStudio Tube package contains the following:



- FireStudio Tube Recording Interface



- 6' 6-pin to 6-pin FireWire cable



- 6' Standard IEC Power Cable



- MIDI Breakout Cable



- Software installation discs:
 - PreSonus FireStudio Tube Installation Drivers
 - Cubase LE 4
 - ProPak Software Suite



- PreSonus Warranty Card

OVERVIEW

1.4 SYSTEM REQUIREMENTS

Below are the minimum computer system requirements for your FireStudio Tube.

Macintosh

- OS: Mac OS X 10.4.x or later
- Computer: Apple Macintosh series with FireWire 400 port
- CPU/Clock: PowerPC G4/1 GHz or higher (Intel/Dual 1 GHz recommended)
- Memory (RAM): 512 MB (1 GB or more recommended)

Windows

- OS: 32-bit Microsoft Windows Vista or XP (with at least SP1)
- Computer: Windows compatible computer with FireWire 400 port.
- CPU/Clock: Pentium or AMD with 1 GHz or higher (Dual 1 GHz recommended)
- Memory (RAM): 512 MB (1 GB or more recommended)

NOTE: The speed of your processor, amount of RAM and size & speed of your hard drives will greatly affect the overall performance of your recording system. A more powerful system (faster processor with more RAM) allows for lower latency (signal delay) and better overall performance.

OPERATION

2.1 QUICK START UP

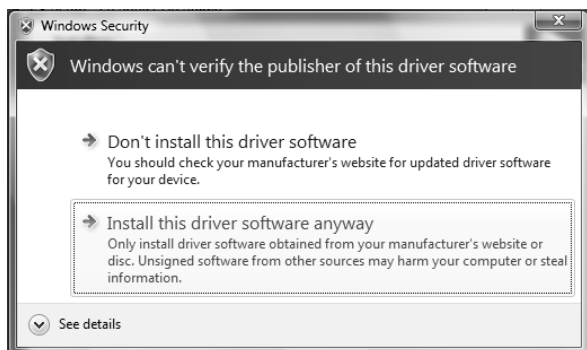
2.1.1 Installation in Microsoft Windows

After inserting the Installation CD into your CD-ROM drive, the FireStudio Tube installer will take you through each step of the installation process. Please read each message carefully – ensuring especially you do not connect your FireStudio Tube early.



- 1) The FireStudio Tube Installer was written to be as simple and easy-to-follow as possible. Please read each message carefully to ensure the FireStudio Tube is properly installed.

Before continuing the FireStudio Tube Installation Setup, please close all multimedia applications and disconnect your FireStudio Tube from your computer.

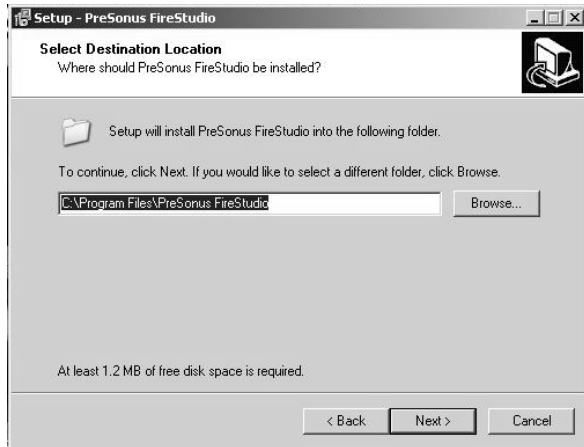


- 2) If you see any Windows Security alerts, click "Install this driver software anyway" (Vista) or "Continue anyway" (XP).

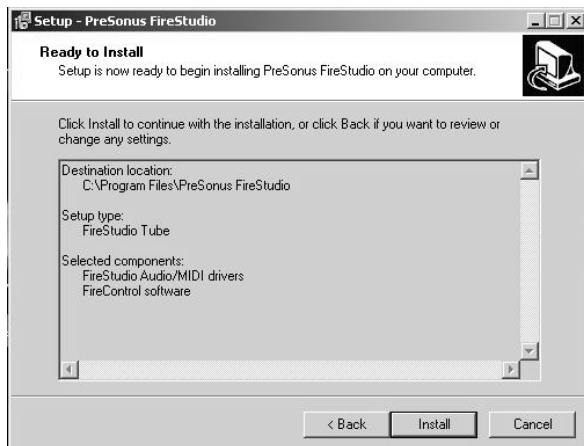


- 3) This installer can be used for the entire FireStudio family of interface products. Select "FireStudio Tube" from the menu on the left of the installer window

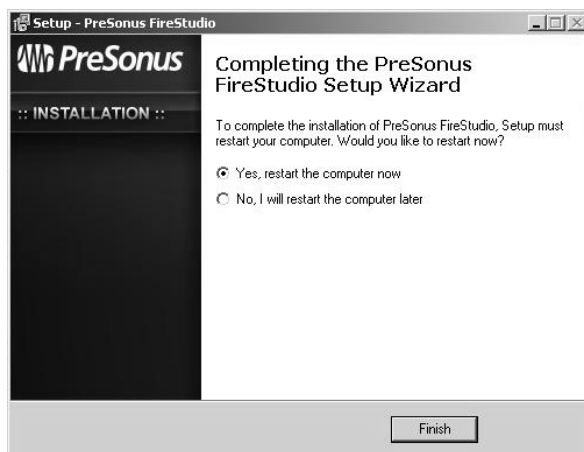
OPERATION



- 4) The installer will now display the directory path into which the FireControl application will be installed. Click "Next"



- 5) The installer is now ready to begin. Click "Install" to copy the correct components to your computer.



- 6) The installer will now prompt you to reboot your computer to complete the installation. Click "Finish" to automatically restart your PC. Once your computer has rebooted, connect the FireStudio Tube. If a Found New Hardware wizard appears, follow the recommended steps. When the sync light remains solid blue, your FireStudio Tube is synced to your computer and ready to use!

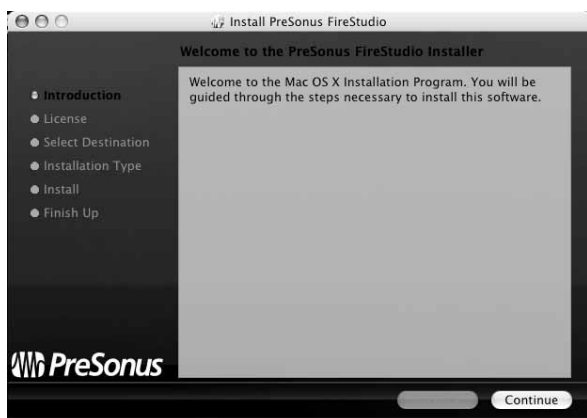
OPERATION

2.1.2 Installation in Mac OS X

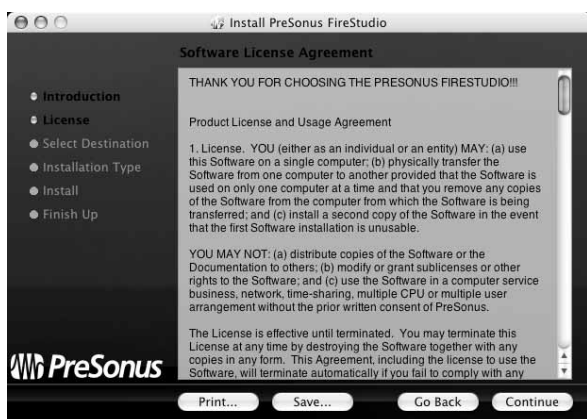
After inserting the Installation CD into your disc drive, browse the disc and run the FireStudio Tube installer, which will take you through each step of the installation process.



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

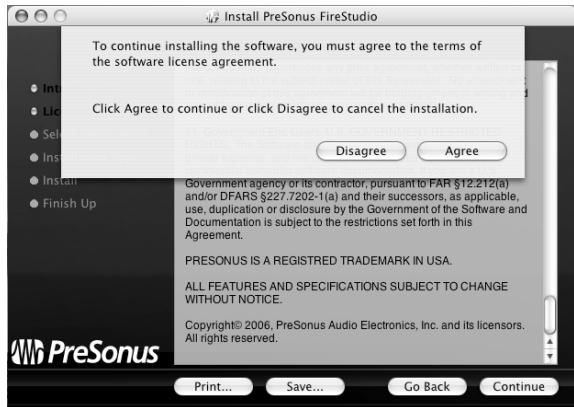


- 2) You will be directed to the welcome screen.
Click "Continue"

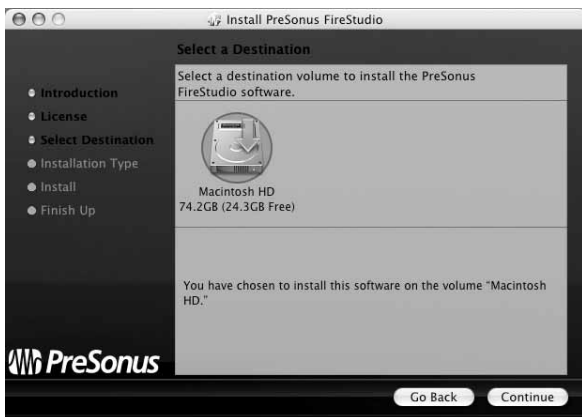


- 3) Take a moment to review the license agreement and click "Continue".

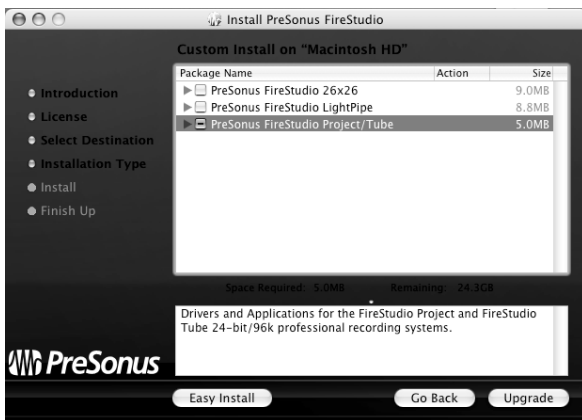
OPERATION



4) Click "Agree" to continue installation.



5) You will be directed to choose the hard drive onto which you want to install the drivers and FireControl Software. You must choose your system drive. Storage drives and partitions cannot be used as hosts for the driver.



6) The FireStudio Tube installer contains the installations for the entire FireStudio family of interfaces. Select "FireStudio Tube" from the list. A " - " will appear in the box to the left.

Click "Install" or "Upgrade" .

OPERATION



- 7) The FireStudio Tube installer requires that your user password be entered as a security measure.

Enter your password and click "OK"



- 8) You will be notified that the FireStudio Tube installation requires that your Mac be restarted to finish the installation process.

Click "Continue Installation"



- 9) Click "Restart" to reboot your Mac automatically.

After rebooting your computer, connect your FireStudio Tube to your computer via the included FireWire cable. When the sync light remains solid blue, your FireStudio Tube is synced to your computer and ready to use!

OPERATION

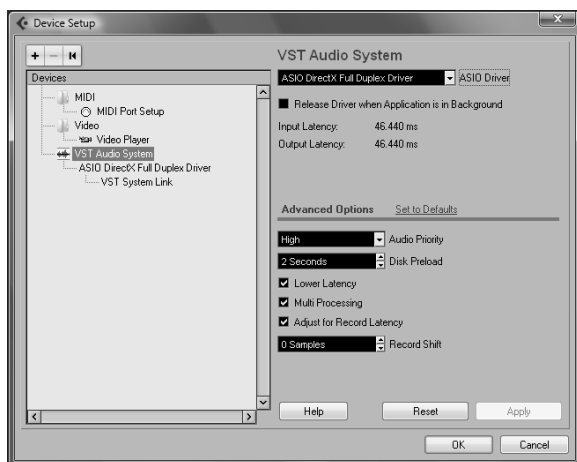
2.1.3 Cubase LE 4 – Device Setup

Once you have installed the FireStudio Tube drivers and connected your FireStudio Tube, you can use the Cubase LE 4 Music Creation and Production System software included with your FireStudio Tube to begin recording, mixing and producing your music. Insert the Cubase LE 4 installation CD into your computer and run the installer.

Then, launch Cubase LE 4 and follow these steps to begin recording with your FireStudio Tube:

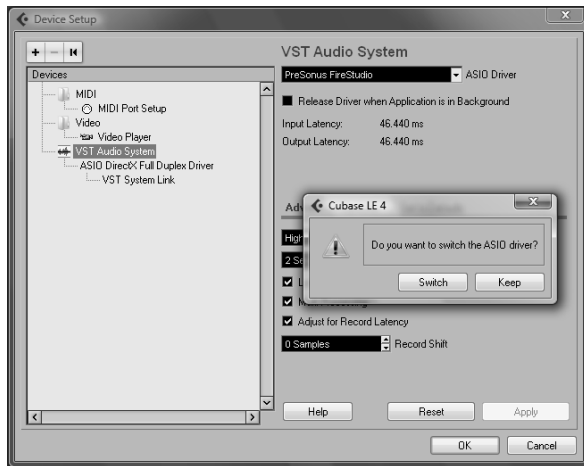


1) Select 'Device Setup...' from the *Devices* menu.

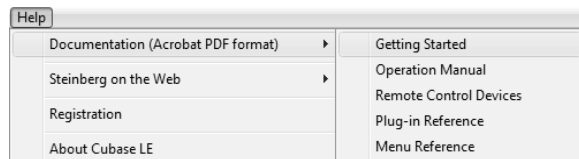


2) Select 'VST Audio System' from the *Devices* column in Device Setup.

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- 3) Select "PreSonus FireStudio" from the *ASIO* Driver dropdown list. A pop-up window will appear asking if you'd like to switch the ASIO driver. Click 'Switch' to begin using the FireStudio Tube Driver.



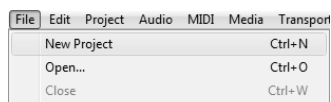
For more help on using Cubase LE 4, please read the *Getting Started* and *Operation Manual* help documents located in Help > Documentation.

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2.1.4 Cubase LE 4 – Creating a Project

To begin recording in Cubase LE, you first need to create a project. Once you have the project environment, you can begin adding audio and MIDI tracks to record and edit on. Follow these simple steps to begin recording your first audio in Cubase LE:

- 1) Plug an XLR microphone or instrument into FireStudio Tube channel one and turn on 48V phantom power if needed for your microphone (typically only for condenser microphones.)

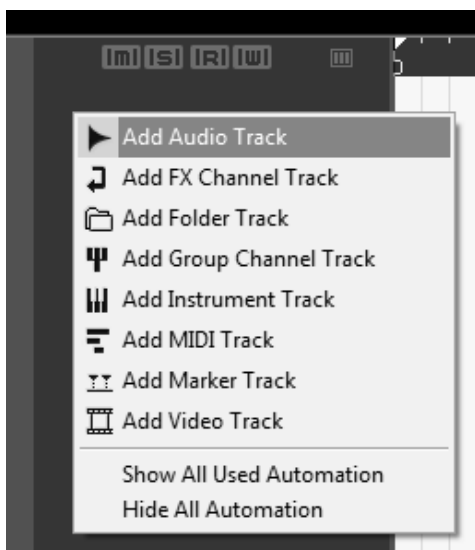


- 2) Enter the *File* menu and select 'New Project'.

You can also use the keyboard shortcut Ctrl+N.

- 3) Select 'Empty' when prompted for a Template.

Click 'OK'.



- 4) Right-click in the track space within the new project and select 'Add Audio Track' from the dropdown menu which appears.

You can also add a track by entering the Project menu and selecting 'Add Track'.

- 5) When the Add Audio Track dialogue appears, choose the number of Audio Tracks you wish to add, whether they will be Mono or Stereo and then click 'OK'.

If you have two audio signals making up a stereo pair (i.e. keyboard, FX processor, cd player, etc.), you should use a Stereo Configuration.

Otherwise, if you have a single, mono audio source (i.e. microphone or instrument), you will want to choose a Mono Configuration.

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- 6) Click the Record Enable button to arm the new track for recording.
- 7) Click the Monitor button to hear the input through the FireStudio Tube's 1/2 and Main CR outputs.
- 8) Select FireStudio Tube input 'Stereo In > Left'.

When choosing a mono from a stereo pair, the "left" channel is typically the odd-numbered channel, and the "right" channel is typically the even-numbered channel (i.e. Left is 1; Right is 2).

Press <F4> on your keyboard to enter the VST Connections menu for more details on your specific input and output routing.

- 9) Turn up the gain control knob on the front panel of the FireStudio Tube for channel 1 while speaking/singing into the microphone or playing your instrument. You should see the input meter in Cubase LE react to the input. Adjust the gain so the input level is near its maximum **without clipping**.
- 10) Connect a set of headphones to the FireStudio Tube headphone output. You may also wish to connect monitors to the FireStudio Tube's Main Outs.
- 11) You are now ready to record.

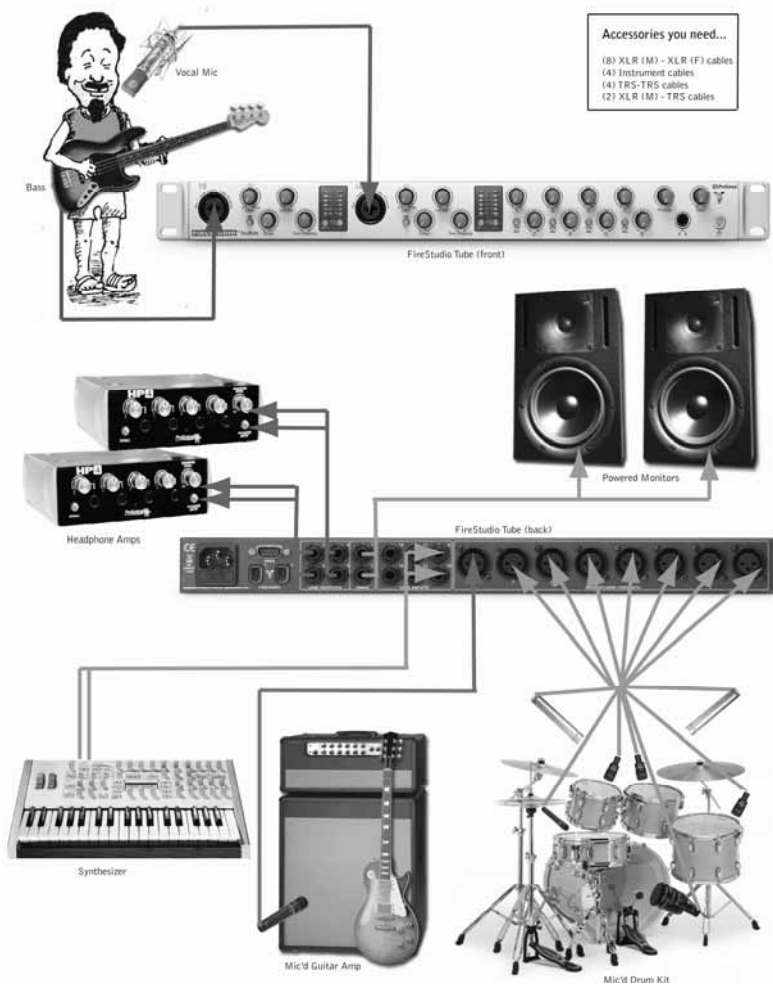
Cubase LE help is available by pressing <F1> on your keyboard or online by visiting www.steinberg.net.

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2.2 SAMPLE HOOK UP DIAGRAM

With the FireStudio Tube, you can simultaneously record and play back up to 16 channels. Since it is loaded with ten preamplifiers, you can plug in ten microphones to the FireStudio Tube along with six line level devices to record a full band. This makes recording extremely easy. All you need are a few microphones, some cables to connect them, a musician (or two or three or more) and the creative energy to bring it all together.

This is a typical rock band setup. Your needs may vary depending upon the number and variety of sources you are recording. Feel free to adapt the sample setup below to your precise needs.



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2.3 MICROPHONES

The FireStudio Tube works with many microphones including dynamic, ribbon and condenser microphones.

2.3.1 Condenser

Condenser microphones tend to generate a high-quality audio signal and are one of the most popular mic choices for today's studio recording applications. Because of their design technology, condenser microphones require a power source, which can be provided from a small battery, external power supply or from microphone inputs as *phantom power*. The FireStudio Tube sends phantom power over XLR inputs only.

2.3.2 Dynamic

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

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

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 have very high quality sound reproduction qualities – especially higher frequencies sounds. However, they are very fragile and typically cannot handle high SPL's.

The most important thing to note about Ribbon microphones is that nearly all ***Ribbon Microphones do NOT require phantom power.*** In fact, unless a Ribbon microphone specifically calls for phantom power, sending phantom power to a ribbon microphone will destroy it – usually beyond repair.

2.3.3 USB and other types

There are a vast number of microphone types available, and as technology increases, it is very likely more will surface. One type of microphone to emerge recently is the USB microphone. Most USB microphones have their own built-in preamp and use drivers separate from the FireStudio Tube drivers.

If you are using a new or non-standard type of microphone (i.e., USB, headset, laser, MEMS, etc.), please consult your microphone's user's manual for power requirement or compatibility information.

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

OPERATION

2.4 A BRIEF TUTORIAL ON DYNAMICS PROCESSING

The SuperChannels on the front panel of the FireStudio Tube feature a hard limiter. This section is an excerpt from brief tutorial on dynamics processing written by PreSonus President and Chief Technical Officer, Jim Odom. It is included to help you get the most out of your FireStudio Tube. This tutorial will take you through the basics of dynamics processing as well as explain the various types of dynamics processors, including a hard limiter.

2.4.1 Common Questions Regarding Dynamics

What is dynamic range?

Dynamic range can be defined as the distance between the loudest possible level to the lowest possible level. For example, if a processor states that the maximum input level before distortion is +24dBu and the output noise floor is -92dBu, then the processor has a total dynamic range of $24 + 92 = 116$ dB. The average dynamic range of an orchestral performance can range from -50dBu to +10dBu on average. This equates to a 60 dB dynamic range. 60 dB may not appear to be a large dynamic range but do the math and you'll discover that +10dBu is 1000 times louder than -50dBu! Rock music on the other hand has a much smaller dynamic range, typically -10dBu to +10dBu, 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 un-compressed 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 -30dBu to +10dBu. The passages that are +10dBu and higher will be heard over the mix, no problem. However, the passages that are at -30dBu 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 +5dBu. At this level, the dynamic range of the vocal is from 0dBu to +10dBu. 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 discussion 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 of each instrument.

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, 'overcompression' 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.

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, oftentimes very subtle, to be properly heard in a mix.

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Why do you need noise gates?

Consider the compressed vocal example above and you now have a 20dB dynamic range for the vocal channel. Problems arise when there is noise or instruments in the background of the vocal mic that became more audible after the lower end of the dynamic range was raised. (air conditioner, loud drummer, etc.) You might attempt to mute the vocal between phrases in an attempt to remove the unwanted signals; however this would probably end disastrous. 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 -10dBu, such that the gate would 'close' out the unwanted signals between the phrases.

If you have ever mixed live you know well the problem cymbals can add to your job by bleeding through your 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 toms so that the cymbals no longer ring through the tom mics will give you an enormous boost in cleaning up the overall mix.

2.4.2 Types of Dynamics Processing

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

Compression / Limiting

Punch, apparent loudness, presence...just three of many terms used to describe the effects of compression/limiting. Compression and limiting are forms of dynamic range (volume) 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 reproduction 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 the output level of an audio signal by a fixed ratio relative to the input. 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 two for signals greater than the compression threshold. Ratios above 10:1 are considered hard limiting.

Limiting refers to the point at which the signal is restrained from going any louder at the output. The level of input signal at which the output is reduced is determined by the compression threshold. As the compression threshold is lowered, more and more of the input signal is compressed (assuming a nominal input signal level). Care must be taken not to 'over compress' a signal. Too much compression destroys the acoustic dynamic response of a performance. ('Over compression', however, is used by some engineers as an effect, and with killer results!)

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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 level is increased, the kick drum stays 'lost in the mud'. Add a touch of compression and 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 (the very loudest portion of the signal) can be far outside the average level of the vocal signal. It is extremely difficult to ride the level with a console fader. A compressor/limiter automatically controls gain without altering the subtleties of the performance.
- A solo guitar can seem to be masked by the rhythm guitars. Compression can make your 'lead' soar above the track without shoving the fader through the roof .
- Bass guitar can be difficult to record. A consistent level with good attack can be achieved with proper compression. Your bass doesn't have to be washed out in the low end of the mix. Let the compressor/limiter give your bass the punch it needs to drive the bottom of the mix.

Expansion

There are two basic types of expansion: dynamic and downward. Expansion increases the dynamic range or level 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 rising 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 5dB below the expansion threshold, the expander will reduce it to 10dB below the threshold.)

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

Noise Gating

Noise gating is the process of removing unwanted sounds from a signal by attenuating all signals below a set threshold. As described above, the 'gate' works independent 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.

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2.4.3 Vocabulary of Dynamics Processors

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

Compressors

- *Threshold.* The Compressor threshold sets the level at which compression begins. When the signal is above the threshold setting, it becomes 'eligible' for compression. Basically, as you turn the threshold knob counterclockwise, more of the input signal becomes compressed. (If you have a ratio setting of greater than 1:1.)
- *Ratio.* Ratio sets the compression slope. This is defined as the output level versus the input level. For example, if you have the ratio set to 2:1, any signal levels above the threshold setting will be compressed at a compression ratio of 2:1. This simply means that for every 1 dB of level increase into the compressor, the output will only increase ½ dB, thus producing a compression gain reduction of 0.5 dB per 1 dB. As you increase the ratio, the compressor gradually becomes a limiter.

A limiter is defined as a processor that limits the level of a signal to the setting of the threshold. For example, if you have the threshold knob set at 0dB, and the ratio turned fully clockwise, the compressor becomes a limiter at 0dB. This means that the signal will be limited to an output of 0dB regardless of the level of the input signal.

- *Attack.* Attack sets the speed at which the compressor 'acts' on the input signal. A slow attack time (fully clockwise) allows the beginning envelope of a signal (commonly referred to as the initial transient) to pass through the compressor uncompressed, whereas a fast attack time (fully counterclockwise) immediately subjects the signal to the ratio and threshold settings of the compressor.
- *Release.* Release sets the length of time the compressor takes to return the gain reduction back to zero (no gain reduction) after crossing below the compression threshold. Very short release times can produce a very choppy or 'jittery' sound, especially in low frequency instruments such as a bass guitar. Very long release times can result in an over compressed sound, 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 sound 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 pre-programmed attack and release curve is used.
- *Gain Makeup.* 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.)

OPERATION

- *Compressor Sidechain.* The sidechain jack interrupts the signal that the compressor is using to determine the amount of gain reduction to 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. This signal can then be processed by an equalizer for example to reduce sibilance (de-essing) in a vocal track. The signal is then returned to the unit via the connector. The signal returned to the sidechain could be that of a narrator or vocalist. In this application, the audio that is passing through the compressor will automatically 'duck' when the narrator speaks or vocalist sings.

Expanders

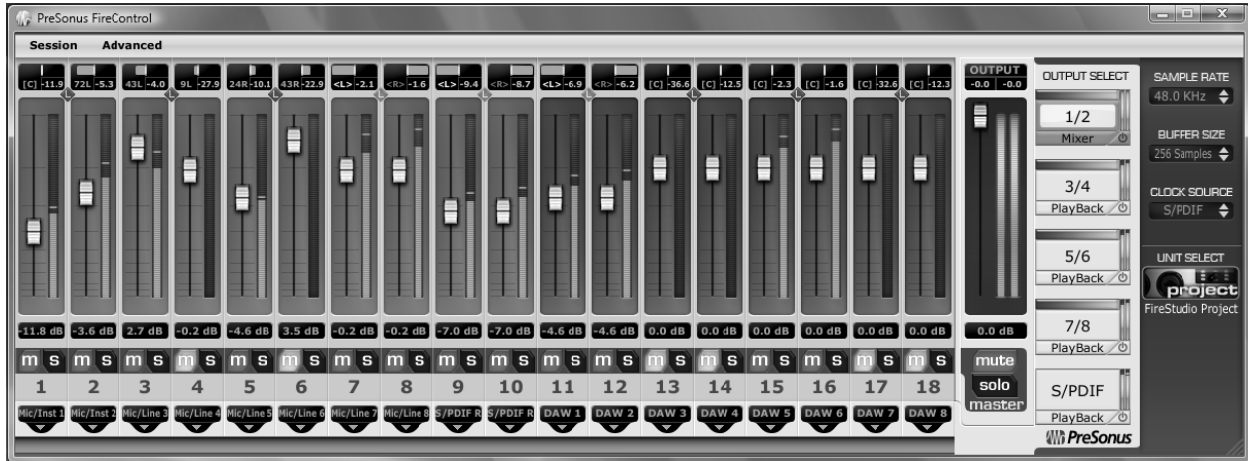
- *Downward Expansion.* Downward expansion is the most common expansion used in pro audio and recording. This type of expansion applies noise reduction to all signals below a set threshold level.
- *Ratio.* The expansion ratio sets the amount of noise reduction applied to a signal once the signal has dropped below the expansion threshold. For example, a 2:1 expansion ratio attenuates a signal 2dB for every 1dB it drops below the threshold. Ratio's 4:1 and higher act much like a noise gate 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 counter-clockwise, the gate is turned off (always open), allowing all signals to pass through unaffected.
- *Attack.* The gate attack time sets the rate at which the gate opens. A fast attack rate is crucial for percussive instruments, whereas signals such as vocals and bass guitar require a slower attack. Too fast of an attack can, on these slow rising signals, cause an artifact in the signal producing an audible 'click'. All gates have the ability to 'click' when opening, however a properly set gate will never click. Below is an example of a gate opening very fast on a pure sine wave. Notice the almost vertical edge of the waveform when the gate opens. This is what produces the high frequency artifact known as 'click'.
- *Hold.* Hold time is used to keep the gate open for a fixed period of time following the signal going below the gate threshold. This can be really useful for effects such as 'gated snare' where the gate remains open after the snare hit for the duration of the hold time then abruptly closes.
- *Release.* The gate release time determines the rate at which the gate closes. Release times should typically be set so that the natural decay of the instrument or vocal being gated is not affected. Shorter release times help to clean up the noise in a signal but may cause 'chattering' in percussive instruments. Longer release times usually eliminate 'chattering' and should be set by listening carefully for the most natural release of the signal.
- *Range.* The gate range is the amount of gain reduction that the gate closes down to. Therefore, if the range is set at 0dB, there will be no change in the signal as it crosses the threshold. If the range is set to -60dB, the signal will be gated (reduced) by 60 dB, etc.

CONTROLS & CONNECTIONS

3.1 PRESONUS FIRECONTROL SOFTWARE

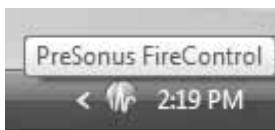


The FireControl is a powerful, flexible and easy to use solution for creating multiple mixes from your inputs and DAW playback streams with zero latency. It allows you to create a zero-latency mix with up to 18 of your input and output signals for each of your FireStudio Tube's output pairs.

The FireControl also acts as the FireStudio Tube's control panel with the ability to change various hardware and software settings of the FireStudio Tube hardware and drivers.

The Mixer section of the FireControl was designed to look and 'feel' like a hardware mixer, so most of the features may already be familiar to you. The following section describes in detail the FireControl layout and the functions of the various buttons found in the FireControl.

3.1.1 FireControl Application Icon



Once you have successfully installed your FireStudio Tube, the FireControl will be available from the Notification Area of your taskbar (typically at the bottom right corner of your screen near the clock).

The FireControl icon is red when your FireStudio Tube is disconnected or not installed and blue when the FireStudio Tube is connected and properly installed.

Open the FireControl by double-clicking the blue FireControl icon or by right-clicking the icon and selecting "Open PreSonus FireControl".

Right-click the FireControl icon and select "Quit" to completely exit the FireControl application and remove it from your Taskbar.

Mac OS X users will find the FireControl application in the Applications folder.

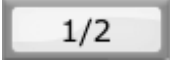




CONTROLS & CONNECTIONS


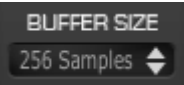
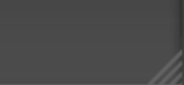
3.1.2 FireControl Master Section

The Master Section is always visible and gives you control over the FireStudio Tube's outputs' source and level as well as the Sample Rate and Buffer Size of the FireStudio drivers.

Output Select Section



<ul style="list-style-type: none">• Output Select Buttons 	<p>Changes the Output Mixer displayed</p> <p>The FireControl has five output mixes – one for each of the FireStudio Tube's five outputs.</p>
<ul style="list-style-type: none">• Mixer Toggle Buttons 	<p>Enables or disables the associated Output's mixer</p> <p>When set to 'PlayBack', the output pair's DAW playback outputs are routed directly to it.</p>
<ul style="list-style-type: none">• Post Fader Output Meters 	<p>Displays the output level of each stereo pair</p> <p>These meter post fader levels, which means they represent the actual audio output levels of your FireStudio Tube.</p>

Hardware Settings Section

<ul style="list-style-type: none">• Sample Rate Selector 	<p>Changes FireStudio Tube Sample Rate</p> <p>A Higher Sample Rate increases frequency range and sonic accuracy but also increases recorded audio file sizes.</p>
<ul style="list-style-type: none">• Buffer Size Selector 	<p>Changes the FireStudio Tube Buffer Size <i>This selector is not present in Mac OS X.</i></p> <p>A large buffer increases latency but is more stable.</p>
<ul style="list-style-type: none">• Resizing Corner 	<p>Resizes the width of the FireControl</p> <p>Click and drag the bottom right corner of the FireControl to resize its width on your screen and increase / decrease the number of channels visible in the Mixer.</p>

CONTROLS & CONNECTIONS

Master Output Section



<ul style="list-style-type: none">• Master Output Fader 	<p>Changes the selected output's audio level</p> <p>Move the Master Fader up and down to increase and decrease the level of the selected output's audio.</p> <p>The dB value of boost or attenuation is displayed below the Channel Fader from -infinity to 0.0 dB.</p> <p>Two virtual LED meters to the right of the Master Fader display the pre fader levels of the audio signal.</p> <p>The peak value in dBFS of the channel's audio is displayed directly above the Master Output's virtual LED meter.</p> <p>NOTE: If the peak values display "CLIP", then your Mixer channels and/or your DAW playback levels should be reduced. The clip indicators clear when the mouse pointer hovers over them.</p>
<ul style="list-style-type: none">• Master Mute and Solo Clear 	<p>Clears all active channel Mutes or Solos</p> <p>The Master Mute and Solo clear buttons are only active when a channel's mute or solo button is active.</p>

3.1.3 FireControl Mixer

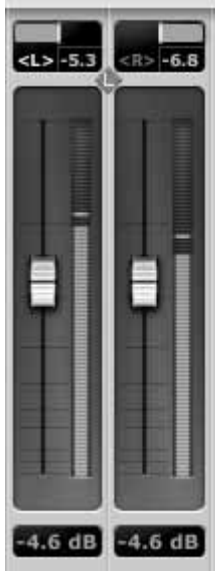



The FireControl Mixer allows you to create a customized mix of up to 22 of the FireStudio Tube's hardware inputs and software playback outputs and adjust their levels independently for each of the FireStudio Tube's stereo output pairs with zero latency. The Mixer section is only visible if the selected output pair's Mixer Toggle is on.

NOTE: The Mixer is common to all of the FireStudio Tube's outputs, so if you add or remove a channel from the Mixer, every output with the Mixer Toggle on will be affected by the change.

These are the individual elements of the Mixer and their functions:

<ul style="list-style-type: none">• Channel Pan Sliders 	<p>Moves the channel's audio left/right in the stereo field</p> <p>Double-clicking the slider returns the pan to center.</p>
<ul style="list-style-type: none">• Link Buttons 	<p>Links the adjacent channels as a stereo pair</p>

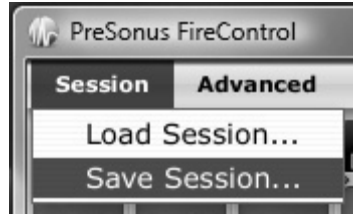
CONTROLS & CONNECTIONS

	<p>When channels are linked, changing either channel's level changes both, and pan states are hard left/right.</p>
<ul style="list-style-type: none">• Channel Faders  <p>The image shows two vertical channel faders. At the top of each fader, there are pan controls labeled '<L> -5.3' and '<R> -6.8'. Below the pan controls are two vertical sliders. At the bottom of each fader, there is a virtual LED meter and a dB readout, both showing '-4.6 dB'.</p>	<p>Changes the channel's audio level</p> <p>Move the Channel Fader up and down to increase and decrease the channel's audio level.</p> <p>The dB value of boost or attenuation is displayed below the Channel Fader from -infinity to +6 dB.</p> <p>A virtual LED meter to the right of each fader displays the post fader level of the channel's audio.</p> <p>The peak value in dBFS of the channel's audio is displayed directly above each channel's virtual LED meter.</p>
<ul style="list-style-type: none">• Channel Mutes and Solos  <p>The image shows two square buttons. The left button has a lowercase 'm' and the right button has a lowercase 's'.</p>	<p>Adds the channel to the Mute or Solo bus</p> <p>Muting a channel silences the channel's audio. Soloing a channel mutes all other non-soloed channels.</p>
<ul style="list-style-type: none">• Channel Scroll Bar  <p>The image shows a horizontal scroll bar with a central slider and arrowheads at both ends.</p>	<p>Changes the visible channels if the FireStudio is resized</p> <p>When the FireControl is resized to show less than 18 channels, click in the scroll bar area to slide the channels left and right and view the hidden channels.</p>
<ul style="list-style-type: none">• Input Selectors  <p>The image shows four buttons with dropdown menus. The buttons are labeled 'Mic/Line 7', 'Mic/Line 8', 'DAW 1', and 'DAW 2'.</p>	<p>Changes the source of the FireControl Mixer channels</p> <p>Choosing "No Input" will hide the channel. "Inputs" are your hardware inputs (Mic/Inst/Line/SPDIF). "PlayBack" are your software DAW PlayBack outputs.</p>

CONTROLS & CONNECTIONS

3.1.4 FireControl Menus

- **Session.** Allows you to Save and Load the FireControl Mixer and Output settings



- **Load Session...** Loads a saved FireControl session file
- **Save Session...** Saves the FireControl Mixer and Output settings for loading later
- **Advanced.** Gives additional options specific to Windows functionality.



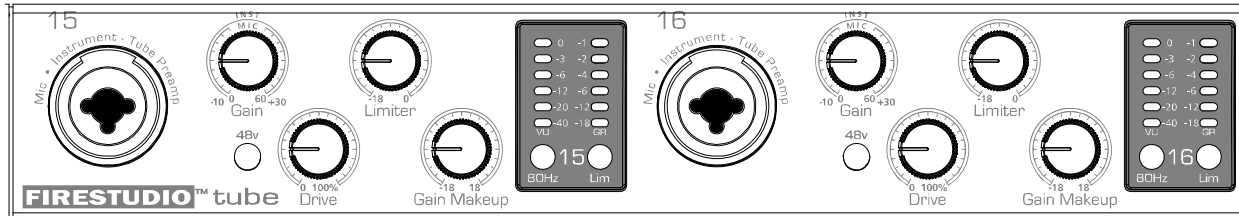
- **Sample Rate Lock Mode.** Sets which types of applications can change the Sample Rate.
 - **Unlock.** Any WDM or ASIO application can change the Sample Rate.
 - **Lock WDM/Windows.** Only ASIO applications can change the Sample Rate.
 - **Lock WDM and ASIO.** Only the FireControl can change the Sample Rate.
- **Enable WDM Audio.** Unchecking this disables WDM audio in Windows. Disabling WDM Audio is useful when the FireStudio Tube is dedicated for DAW use only.



- **Operation Mode.** Changes how the FireStudio drivers output buffer size is set.
 - **Normal Mode.** Input and Output buffers are both identical to the Buffer Size.
 - **Safe Mode 1-3.** Increases the output buffer size for added stability.

CONTROLS & CONNECTIONS

3.2 FRONT PANEL LAYOUT



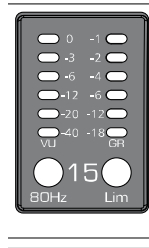
- **SuperChannels.** Your FireStudio Tube is equipped with two “SuperChannels”. These channels are Class A Vacuum Tube microphone and instrument preamplifiers and full-featured analog limiters. Each channel has its own controls.
 - **Neutrik Combo Connectors.** Both SuperChannels have a Mic/Line connector using the Neutrik Combo connector. This revolutionary style connector lets you use either ¼” phone or XLR connectors in the same female input. Channels 15 and 16 of FireStudio Tube are ¼” instrument and microphone XLR inputs.
 - **48 Volt Phantom Power.** Each SuperChannel has 48V Phantom power available individually via its push button switches on the front panel.
 - [XLR connector wiring for Phantom Power](#)
 - Pin 1 = GND
 - Pin 2 = +48V
 - Pin 3 = +48V
 - **Input Gain/Trim Control.** This knob provides the following gain structure:
 - 54dB of variable gain (+4dB to +58dB)
 - **Drive.** This knob controls the amount of signal routed to the 12AX7 vacuum tube. This feature lets you control the amount of tube saturation. Greater levels of tube saturation give the signal greater warmth and a richer sound. This works equally well on microphones and instruments.
 - **Limiter.** This knob sets the threshold of the analog limiter from -18dBfs to 0dBfs (-8dBu to +10dBu). The threshold is the point at which the limiter will engage. No more of the input signal will be allowed to pass through once this threshold is reached. Thus reducing the gain. Turning the knob clockwise increases the threshold -- allowing more of the input signal through. Turning the knob counter-clockwise decreases the threshold -- allowing less of the input signal through.
 - **Gain Makeup.** This knob adjusts the gain after the limiter threshold. This is used to compensate for gain reduction, raising the gain of the output signal without raising the limiter threshold. This is especially useful when using the limiter as an effect to over compress the signal. By using the Gain Makeup knob, you can set the threshold extremely low and still maintain a strong signal to record.

CONTROLS & CONNECTIONS

- **Instrument Inputs (Channels 15 and 16).** The ¼" TS connector on channels 15 and 16 are for use with an instrument (guitar, bass, etc.). When an instrument is plugged into the instrument input, the microphone preamp is bypassed, and the FireStudio Tube becomes an active instrument preamplifier.

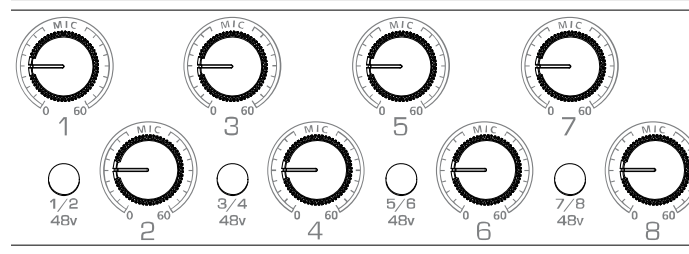
NOTE: Active instruments are those that have an internal preamp or a line level output. Active instruments should be plugged into a line input rather than into an instrument input. Plugging a line level source into the instrument inputs on the front of the FireStudio Tube not only risks damage to these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

(In other words, don't plug a line level source into the combo jacks of channel 15 or 16.)

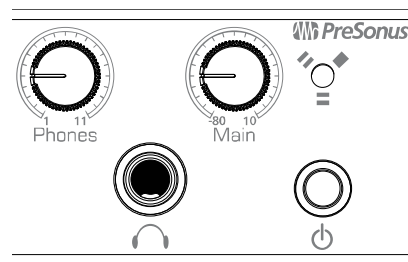


- **LED Meters.** Each SuperChannel features two 6-LED level indicators. The indicator on the left meters the input signal. The meter on the right meters the amount of gain reduction through the limiter
 - **VU indicator.** The VU indicator meters the input signal. The first three green LEDs will light up when your input signal from either the XLR (Mic) or ¼" (Instrument) reaches -40dBfs, -20dBfs, and -12dBfs respectively. The two yellow LEDs will light up when the channel's input signal reaches -6dBfs and -3dBfs respectively. The red clip indicator LED will illuminate when the channel's input signal reaches 0dBFS. At this level, your Mic/Instrument trim signal will exhibit signs of clipping such as distortion. It is highly recommended you do not allow your converters to clip (the red clip indicators to light up) as the sound quality will not be desirable.
 - **Gain Reduction indicator.** This meter shows the amount of gain reduction implemented by the limiter. As you turn the Limiter knob to the right, the amount of input gain being reduced will lessen.
- **80Hz Rumble Filter.** A rumble filter is available to eliminate low frequency noise. This lets you greatly reduce background noise from things such as air conditioners or wind noise with the flick of a switch without affecting the desired frequencies. The button will illuminate when the 80Hz filter is engage.
- **Limiter.** This button is used to engage the Limiter. The button will illuminate when the limiter is engaged.

CONTROLS & CONNECTIONS



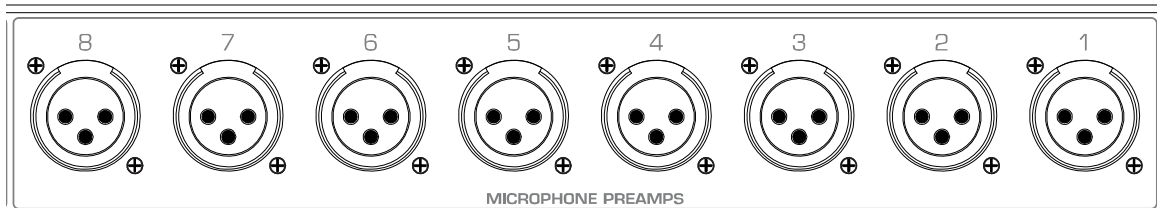
- **Input Gain/Trim Control.** These knobs control the gain for the microphone preamps located on the rear of the unit and provide the following gain structure:
 - **XLR Microphone Inputs.** 54 dB of variable gain (-4dB to +50dB)
 - **48 Volt Phantom Power.** The FireStudio Tube has 48V Phantom power available in groups of two via push button switches on the front panel. From left to right, each button activates Phantom power for channels 1 & 2, 3 & 4, 5 & 6 and 7 & 8.
 - XLR connector wiring for Phantom Power
 - Pin 1 = GND
 - Pin 2 = +48V
 - Pin 3 = +48V



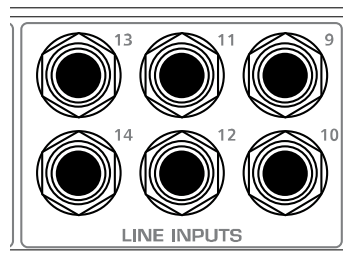
- **Phones.** The Phones knob controls the amount of volume going to the headphone output on the front of the unit. Notice the volume indicator goes to 11 ... use this setting with extreme caution.
- **Main.** The Main knob controls the output level for the Main Outputs on the back of the FireStudio Tube with a range of -80db to +10dB.
- **1/4" Headphone Jack.** This is where you connect your headphones to the FireStudio Tube.
- **Power Button.** This button turns your FireStudio Tube on and off.
- **Red-Blue Power/Sync Light.** This light is a clock source (sync) indicator. It lets you know if your unit is receiving word clock correctly. Word clock is the manner by which digital devices sync frame rates. Proper word clock sync prevents digital devices from having pops, clicks or distortion in the audio signal due to mismatched digital audio transmission.
 - **Blue.** FireStudio Tube is correctly synced via FireWire
 - **Flashing Red and Blue.** Sync invalid or not present

CONTROLS & CONNECTIONS

3.3 BACK PANEL LAYOUT



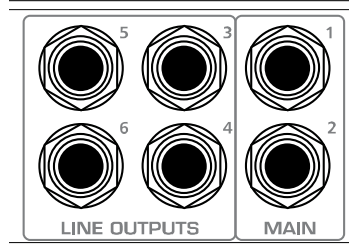
- **Microphone Pre-Amplifier.** Your FireStudio Tube is equipped with eight custom designed PreSonus X-MAX microphone preamplifiers for use with all types of microphones including Dynamics, Condensers and Ribbons as well as instruments and line level signals. The award winning PreSonus preamplifier design is a Class A input buffer followed by a dual servo gain stage. This arrangement results in ultra low noise and wide gain control allowing the FireStudio Tube user to boost desirable signal without increasing unwanted background noise.
 - **+22dBu Headroom.** The FireStudio Tube microphone preamplifier has +22dBu headroom. This feature gives you wide dynamic range and excellent transient response characteristics.



- **Line Inputs.** Channels 9-14 of your FireStudio Tube are line level inputs. They are designed to accept up to a maximum input level of +20dBu. Use these inputs to connect your line level devices such as synthesizers, amp modeling hardware, vocoders, etc. In general, line level devices have their own volume control. As these inputs are for line level signal only, there is no trim control for them on the FireStudio Tube. If you need to raise or lower the volume, the adjustment should be made from your line level device.

A NOTE ABOUT VINTAGE DEVICES: Some older and vintage line level devices may not output at a hot enough signal. If you find this to be the case, you should connect them to a direct box and plug them into the microphone inputs instead.

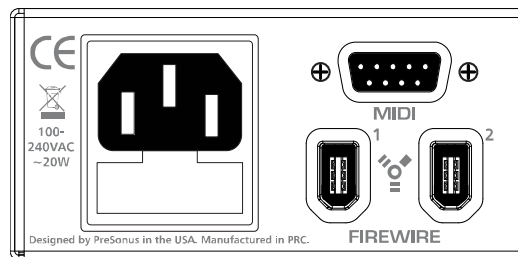
CONTROLS & CONNECTIONS



- **General Purpose Line Outputs (TRS Balanced).** These are general purpose line-level outputs. Line Outputs 1 & 2 are typically your computer's audio playback. All six Line Outputs can be accessed by your computer and can be used for separate mixes, surround sound, additional speakers, external effects processors, etc.

NOTE: By default, the General Purpose Outputs are DAW Playback 1-6 streams from your computer. However, the FireControl has final control over their audio sources.

- **Main Output (TRS Balanced).** Outputs 1 & 2 should be used as your Main Outputs left and right. Unless otherwise specified, most DAW applications will default configure to use device outputs 1 & 2 as the main output for the application.



- **Power Adaptor Input.** This is where you plug the provided IEC power cable into the FireStudio Tube.
- **FireWire Ports.** There are two standard 6-pin FireWire ports on the back of the FireStudio Tube. Either (and only one) should be used to connect your FireStudio Tube to a vacant FireWire port on your computer. If your computer has a 4-pin connector (commonly found on laptops), you will need a 4 to 6-pin connector to connect your FireStudio Tube to your computer.

Use the 'extra' FireWire port to connect additional FireWire devices in such as external Hard Drives.

- **MIDI Breakout Cable connection.** This is where you connect your MIDI breakout cable. MIDI stands for "Musical Instrument Digital Interface". However, the MIDI standard goes well beyond just instrumentation and sequencing. The MIDI inputs and outputs on the breakout cable allow connection and/or communication with external MIDI equipment. One function of this port is MIDI programming. These can also be used for MMC (MIDI Machine Control) and MTC (MIDI Time Code).

NOTE: MIDI is not audio but is frequently used to trigger or control an audio source (such as a VST plugin or synthesizer). When using MIDI, ensure your MIDI data is correctly sent and received by the appropriate hardware or software instruments. You may also need to return those devices' audio to the FireStudio Tube. Please consult the User's Manual of your MIDI devices for help with MIDI setup and usage.

TECHNICAL INFORMATION

4.1 TROUBLESHOOTING

Please note that many technical issues can arise when converting a standard computer into a DAW (Digital Audio Workstation). PreSonus will only provide support for issues that directly relate to the FireStudio Tube interface. It may be necessary to contact the manufacturer of the computer, operating system and/or software to obtain additional technical support. PreSonus does not provide support for issues in regards to operating systems, additional hardware or software. Please check our website, www.PreSonus.com regularly for software information and updates, firmware updates and technical support. Technical assistance may also be received by calling PreSonus at 225-216-7887 between the hours of 10 am and 5 PM Central Time.

Pops and Clicks

Pops and clicks in either your Input or Playback audio are typically caused by momentary losses of synchronization between the FireStudio Tube and its clock source. This can be due to a FireWire card not suited for FireWire audio devices or to a lack of CPU power or available memory. Try closing all unnecessary programs, increasing the Buffer Size in the FireControl and optimizing your operating system for audio.

Pops and clicks exclusive to the S/PDIF input or output are typically caused when the FireStudio Tube is not digitally synced to the external S/PDIF device. If this is the case, ensure your FireStudio Tube and external S/PDIF device are properly synced. *Please consult your S/PDIF device's User's Manual for more details on syncing it to your FireStudio Tube.*

Flashing Sync Light

The FireStudio Tube's Sync light flashes red and blue when it is not synced. Check the FireWire cable to ensure it hasn't become disconnected. If you are using an external S/PDIF device and your Clock Source is set to "S/PDIF", set the FireControl's Sample Rate to the same Sample Rate as that of your S/PDIF device. If you are not using an external S/PDIF device, open the FireControl and set the Clock Source to "Internal".

Audio Drop Outs

Audio drop outs can occur when the speed of your processor cannot buffer audio fast enough. Increase your Buffer Size in the FireControl and try the different Operation Modes from the Advanced menu.

Preamplifier

Q: I have a microphone plugged into channel one (or two) but I am not getting any signal.

Possible Solutions:

- 1) Check your mic cable. This should typically be an XLR connection.
- 2) Make sure the microphone does not require phantom power. If it does, press the 48V button.
- 3) Make sure nothing is plugged into the line input on the rear of the FireStudio Tube. The line inputs on channels 1 and 2 take precedence over the combo input on the front of the unit for channels 1 and 2. If a cable is plugged into the line input on channel, then the mic/instrument input on channel one will be inactive until the cable is removed from line input 1.

For help with Cubase LE 4, press F1 while running Cubase LE or visit: www.steinberg.net.

TECHNICAL INFORMATION

4.2 SPECIFICATIONS

Microphone Preamps 1-8

Type	XLR Female Balanced
Frequency Response (± 0.5 dB)	20 Hz to 50 kHz
Frequency Response (± 3.0 dB)	20 Hz to 150 kHz
Input Impedance (Balanced)	1600 Ω
THD+N (unwtd, 1 kHz @ +4 dBu Output, Unity Gain)	< 0.003%
EIN (unwtd, 55dB Gain, 150 Ω Input, 20Hz to 22 kHz)	-126 dBu
S/N Ratio (Unity Gain, unwtd, Ref. = +4 dBu, 20Hz to 22 kHz)	> 101 dB
Common Mode Rejection Ratio (1 kHz, 55 dB Gain)	> 55 dB
Gain Control Range (± 1 dB)	-12 dB to +37 dB
Maximum Input Level (Unity Gain, 1 kHz @ 0.5% THD+N)	+14 dBu
Phantom Power (± 2 VDC)	+48 VDC

Microphone Preamps 15 & 16 "SuperChannels"

Type	Neutrik Combination XLR Female Balanced
THD+N (unwtd, 1 kHz @ +4 dBu Output, Unity Gain)	< 0.2%
Gain Control Range (± 1 dB)	+4 dB to +58 dB
Limiter Gain Reduction	0 dBu to -18 dBu
Limiter Ratio	$\infty:1$
Signal Level LEDs	
(1) Red / Clip (± 0.5 dBu)	+10 dBu (0 dBFS)
(2) Yellow (± 0.5 dBu)	+7 dBu (-3 dBFS)
(3) Yellow (± 0.5 dBu)	+4 dBu (-6 dBFS)
(4) Green (± 0.5 dBu)	-2 dBu (-12 dBFS)
(5) Green (± 0.5 dBu)	-10 dBu (-20 dBFS)
(6) Green (± 0.5 dBu)	-30 dBu (-40 dBFS)
Gain Reduction LEDs	
(1) Green (± 0.5 dBu)	-1 dBu
(2) Green (± 0.5 dBu)	-2 dBu
(3) Green (± 0.5 dBu)	-4 dBu
(4) Green (± 0.5 dBu)	-6 dBu
(5) Green (± 0.5 dBu)	-12 dBu
(6) Green (± 0.5 dBu)	-18 dBu
Phantom Power (± 2 VDC)	+48 VDC

Instrument Input 15 & 16 "SuperChannels"

Type	Neutrik Combination 1/4" TRS Female Unbalanced
Input Impedance	1 M Ω

TECHNICAL INFORMATION

Line Inputs 9-14

Type ¼" TRS Female Balanced
Frequency Response (± 0.5 dB) 20 Hz to 50 kHz
Frequency Response (± 3.0 dB) 20 Hz to 150 kHz
Input Impedance (Balanced) 10 K Ω
THD+N (unwtd, 1 kHz @ +4 dBu Output, Unity Gain) < 0.003%
S/N Ratio (Unity Gain, unwtd, Ref. = +4 dBu, 20 Hz to 22 kHz) > 101 dB
Maximum Input Level (Unity Gain, 1 kHz @ 0.5% THD+N) +20 dBu

Line Outputs 1-6

Type ¼" TRS Balanced
Output Impedance 51 Ω

Headphone Outputs 7-8

Type ¼" TRS Active Stereo
Maximum Output 150 mW/Ch @ 60 Ω Load
Frequency Response (± 1.0 dB) 10 Hz – 70 kHz

Digital Audio

ADC Dynamic Range (Awt'd, 48 kHz Sample Rate) 114 dB
DAC Dynamic Range (Awt'd, 48 kHz Sample Rate) 114 dB
Jitter Specification < 20 pS
Bit Depth 24
Reference Level for 0dBFS +10 dBu
Internal Sample Frequency Selections (kHz) 44.1, 48, 88.2, 96
External Sample Frequency Input S/PDIF

Power

Input Connector Type IEC
Input Voltage Range 90 to 240 VAC
Power Requirements (Continuous) 24W

As a commitment to constant improvement, PreSonus Audio Electronics, Inc. reserves the right to change any specification stated herein at any time without notification.