# Studio Channel

## Vacuum Tube Channel Strip

Owner's Manual
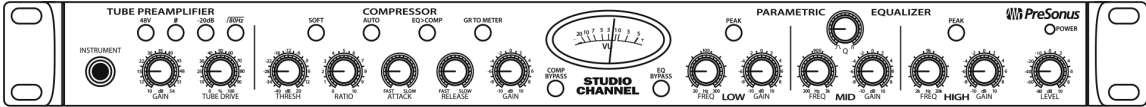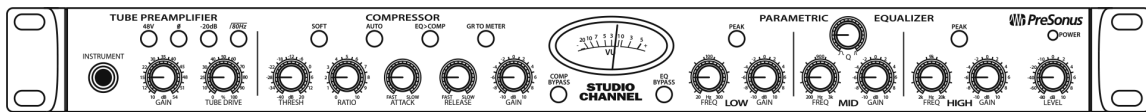
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# 1. Overview

## 1.1 Introduction



Thank you for purchasing the PreSonus Studio Channel. PreSonus Audio Electronics has designed the Studio Channel utilizing high-grade components to ensure optimum performance that will last a lifetime. The Studio Channel is a professional channel strip combining Class A tube preamplifier, VCA-based compressor and three-band parametric equalizer perfect for the professional and project studio. Great for all types of microphones, instruments, keyboards and synths, the Studio Channel has the sonic power and flexibility to achieve any tone you can dream of – luscious in your face vocals, crystal clear acoustic guitars, fat solid bass guitar, dynamic acoustic piano, cracking snare, punchy bass, huge MPC tracks,… you get the picture.

Please reach out to us on the social media platform of your choice with any questions or comments you may have regarding your PreSonus hardware or software. We're committed to constant product improvement and value your suggestions highly! The best way to achieve our goal of constant product improvement is by listening to the *real* experts - you. We appreciate your support!

We suggest you use this manual to familiarize yourself with the features, applications and correct connection procedure for your Studio Channel before trying to connect it to your recording system. This will hopefully alleviate any unforeseen issues that you may encounter during installation and set up. Please pay close attention when connecting your Studio Channel to your system.  Bad cables and improper grounding are the most common causes of problems encountered in recording and live P. A. environments.  We recommend checking your cables, connections and grounding if you experience any noise or sonic performance problems.

Thank you, once again, for buying our product, and we hope you enjoy your Studio Channel!

## 1.2 Product Registration

Be sure to visit my.presonus.com, create or log into your user account, and register your Studio Channel.This will allow you to submit a support ticket should you need any help!



PreSonus is committed to delivering the best experience for our customers. My PreSonus is a one-stop portal for all our registered customers' needs. From your my PreSonus account, you can view all your PreSonus hardware and software registrations; contact support; track orders and more.

To register your Studio Channel, go to My.PreSonus.com and follow the onscreen instructions.

**OR...**



Download the MyPreSonus app from the Apple App Store or Google Play.

## 1.3   Features

The Studio Channel has everything needed to deliver big tone. The preamplifier stage of the Studio Channel is based on the award winning PreSonus BlueTube preamplifier with a high output 12AX7 vacuum tube operating on voltages double than all other preamplifiers in its class.  It delivers high headroom and big tone, featuring dual control Gain and Tube Drive to create a wide range of sounds. The Studio Channel also comes loaded with a fully variable ultra-fast and smooth VCA-based compressor for musicality and ultra fast attack and parametric EQ that delivers sweet sounding gain/cut for ultra smooth highs, deep solid lows, and clear midrange.  The Studio Channel is like a Swiss Army pocket knife giving you all the tools you need to tailor your tone to enhance your sound.

### Summary of features

- Class A vacuum tube microphone / instrument preamplifier
- Tube gain and tube drive
- Variable VCA compressor (threshold, ratio, attack, release, make up gain, auto, soft)
- Three-band parametric EQ
- Precision analog VU meter for gain reduction and output
- EQ Pre/Post compressor switch
- 80Hz high pass filter
- 20dB pad
- Phase invert
- Rugged metal chassis
- High headroom
- Ultra low noise design

## 1.4 What is in the Box

Your Studio Channel package contains the following:

**Studio Channel**
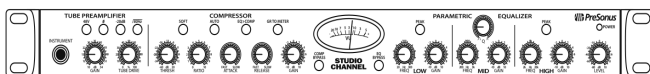
**12VAC 1A Power Supply**
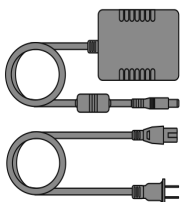
**Registration Card**

**Health and Safety Guide**
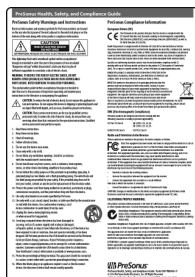
# 2.   Controls and Connections

## 2.1    Front panel layout

### 2.1.1   Tube Microphone Preamp
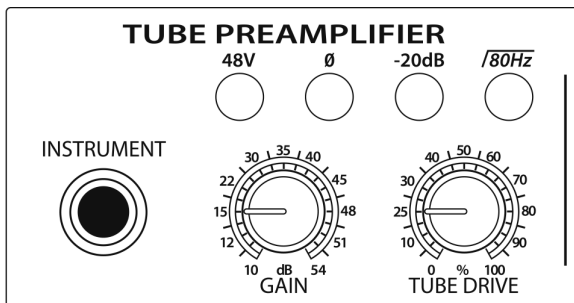


**Instrument Input.** The ¼" TS connector on the front panel is for use with a passive instrument (guitar, bass, etc.). When an instrument is plugged into the instrument input, the microphone preamp is bypassed, and the Studio Channel becomes an active instrument preamplifier.

**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 Studio Channel 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 front panel jack.)**

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

    - **XLR Microphone and TS Instrument/Hi-Z Inputs.** 44 dB of variable gain (+10 dB to +54 dB)

- **Tube Drive.** The Tube Drive control increases the amount of signal routed through the 12AX7 vacuum tube. (The gain controls overall volume, whereas the drive controls volume routed through the tube.) The effect achieved ranges from subtle to extreme, depending on the setting being used:

    - **"Warming Up the Sound".**  This effect is achieved by adding in a small amount of Tube Drive (30% or less).  It is especially desirable for vocals and electric bass.  The resulting sound is richer and sweeter.

    - **"Overdriven Tube Sound".**  This effect is achieved by adding in 30-100% of Tube Drive.  The more Tube Drive you add, the more overdriven the sound will be.  This sound is extremely useful in creating distorted guitar and that authentic "blues harp" harmonica sound.

- **48 Volt Phantom Power.**  This button enables phantom power to the XLR input.

XLR connector wiring for Phantom Power:

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

- **Phase Reverse.** Reverses the polarity of the signal. Use the phase reverse when recording with more than one open microphones to combat phase cancellation between microphones.

- **-20 dB Pad**. This button attenuates the input signal by 20 dB.  The pad can be used to keep a hot signal from overdriving the microphone preamp.

- **80Hz Roll-off.** The 80Hz button is a low-end roll-off filter. When pushed in, the 80Hz button causes all frequencies below 80Hz to be attenuated (dropped) by 12dB. This filter can be handy in live and studio applications. For example, the 80Hz filter can help to reduce the "boominess" or "muddiness" of a vocal and improve the overall clarity.

## 2.1.2   Compressor
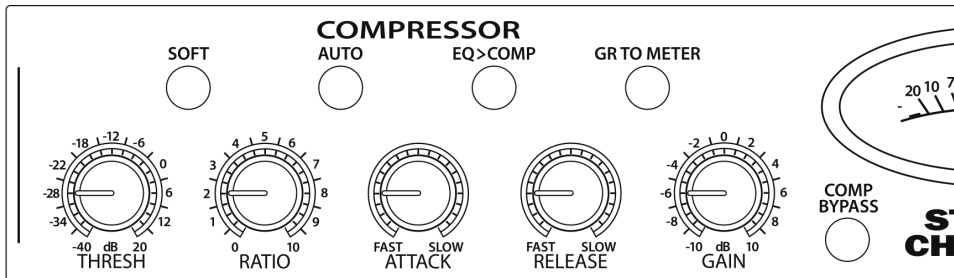


- **Threshold.** Sets the level at which compression begins. When the signal is above the Threshold setting, it becomes 'eligible' for compression. Basically, as you turn the Threshold knob counter-clockwise, the input signal is compressed (if you have a ratio setting of greater than 1:1).  The threshold can be set from -40 to +20 dB.

- **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 level above the Threshold setting will be compressed at a compression ratio of 2:1. This simply means that that for every 1dB of level increase into the compressor; the output will only increase ½ dB, thus producing an attenuation of 0.5 dB. The Ration can be set from 1:1 to 10:1.

- **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.** Sets the length of time the compressor takes to return the Gain reduction back to zero (no gain reduction). 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 overly compressed signal (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.

- **Gain Make Up.** 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 and readjust the volume to the pre-compression level (if desired).  You can adjust the Gain Make Up from -10 to +10 dB.

- **Soft.**The Soft button selects Soft Knee and Hard Knee compression curves. When this button is pushed in, Soft knee compression curves are used, otherwise hard knee compression curves are used. 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.

- **Auto.** The Auto button places the compressor in automatic attack and release mode. The Attack and Release knobs become inoperative and a preprogrammed Attack and Release curve is used instead.

- **EQ>Compressor.** This button places the EQ before the Compressor in the signal path. By default the Studio Channel's signal path is as follows:



- **GR>Meter.** This button changes the VU metering to read the Gain Reduction of the Compressor rather than the output level of the Studio Channel.

- **Compressor Bypass.** This button bypasses the compressor.

## 2.1.3   Parametric EQ
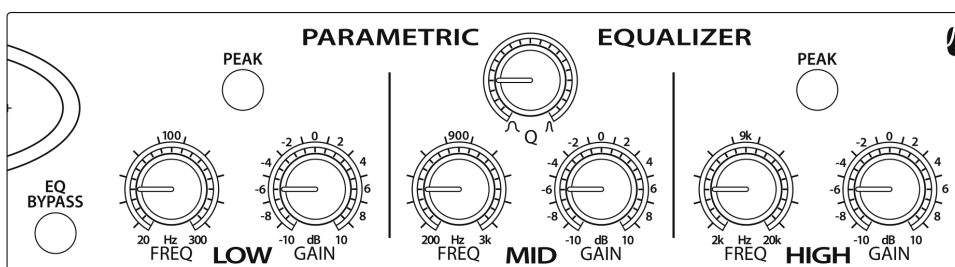


- **EQ Bypass.** Bypasses the EQ.  When both the Compressor and the EQ are bypassed the Studio Channel acts as Tube Preamplifier only.**Low Band Frequency.** Sets the Center Frequency of the "Q" or frequency range width of the Low Band EQ.  You can adjust the center frequency from 20 to 300 Hz.

- **Low Band Peak Button.** By default, the Low EQ is set as a shelving EQ.  When the peak button is enabled, the Low Band EQ becomes a standard peak equalizer with a fixed Q of 0.7.

- **Low Band Gain.** Sets the Gain attenuation or boost of the Q.  The level of the center frequency can be set between -10 and +10 dB.

- **Mid Band Frequency.**  Sets the Center Frequency of the "Q" or frequency range width of the Mid Band EQ.  You can adjust the center frequency from 200 to 3 kHz.

- **Mid Band Gain.**  Sets the Gain attenuation or boost of the Q.  The level of the center frequency can be set between -10 and +10 dB.

- **Mid Band Q.**  Sets the frequency range for the Mid Band EQ.  The Frequency knob sets the center of this range.

- **High Band Frequency.**  Sets the Center Frequency of the "Q" or frequency range width of the High Band EQ.  You can adjust the center frequency from 2 to 20 kHz.

- **High Band Peak Button.**  By default, the High EQ is set as a shelving EQ.  When the peak button is enabled, the High Band EQ becomes a standard peak equalizer with a fixed Q of 0.7.

- **High Band Gain.** Sets the Gain attenuation or boost of the Q.  The level of the center frequency can be set between -10 and +10 dB.

## 2.1.4 Master and VU Meter
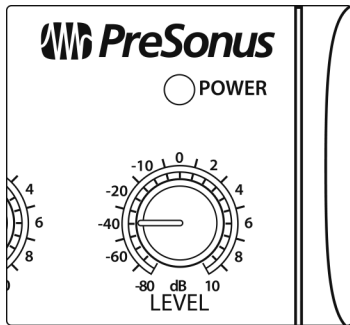


- **Power Button.** This button turns your Studio Channel On and Off.  When your Studio Channel is powered on, this button will illuminate.
- **Master Level.** Adjust the overall output volume of the Studio Channel
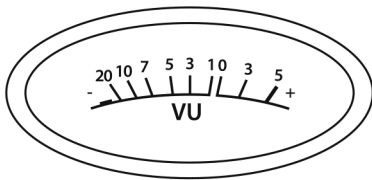


- **VU Meter.** The analog VU meters the output level of your Studio Channel.  When the GR>Meter button in the compressor section is enabled, the VU meter will show the amount of Gain Reduction.

## 2.2 Back panel layout



- **Microphone Pre-Amplifier.** Your Studio Channel is equipped with a custom designed PreSonus microphone preamplifier for use with all types of microphones including dynamic, condenser and ribbon.

- **Line Level Input.** The Studio Channel also features a balance line level input to connect synthesizers, amp modelers, etc.

- **Main Outputs.** The Studio Channel features both a balanced TRS and an XLR output for greater ease of connectivity.

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

- **Power Supply Connector.** This is the connection for your external power supply.

# 3. Operation

## 3.1 Microphones

The Studio Channel works with all standard microphones including dynamic, ribbon and condenser microphones.

### 3.1.1 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 Studio Channel sends phantom power over XLR inputs only.

### 3.1.2 Dynamic

Dynamic microphones are possibly the most widely used microphone type – especially in live shows and when recording loud source signals such as guitar amplifiers and kick drums. They are usually less expensive than condenser and ribbon microphones, 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. In the vast majority of cases, phantom power will have no effect on a dynamic microphone's audio quality or sensitivity and will not damage the microphone. You should consult your microphone's documentation to confirm.

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. PLEASE NOTE: unless a Ribbon microphone calls specificallyfor phantom power, sending phantom power to a ribbon microphone will severely damage it – usually beyond repair.

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.

## 3.2    A Brief Tutorial on Dynamics Processing

The compressor in the Studio Channel is a fully variable, ultra-fast and smooth VCA-based circuit with all of the controls needed for a wide range of compression settings: ratio, threshold, attack, release and gain make up. The VCA-based compressor is known for musicality as well as ultra fast attack with the ability to tame the fastest transients. From slap bass, to snare drum, to vocals, the Studio Channel's compressor stage allows you to record as hot as needed without clipping giving you total dynamic control. What follows 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 Studio Channel. This tutorial will take you through the basics of dynamics processing as well as explain the various types of dynamics processors.

### 3.2.1   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 = 116dB.

The average dynamic range of an orchestral performance can range from -50dBu to +10dBu on average. This equates to a 60dB dynamic range. 60dB 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 20dB. 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 20dB. You wish to add an un-compressed vocal to the mix.  The average dynamic range of an uncompressed vocal is around 40dB. 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. 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 10dB. 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.

## 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 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 -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.

## 3.2.2   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 a factor of 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!)

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 'undue' 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.

### 3.2.3 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 1dB of level increase into the compressor, the output will only increase 1/2dB, thus producing a compression gain reduction of 0.5dB/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 counter-clockwise) 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.)
- **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**.  he 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.

- **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 60dB, etc.

## 3.2.4   General Compression Setting Suggestions

The following are compression presets used in the PreSonus BlueMax.  We have included them as a jumping off point for setting up compression on your Studio Channel.

## Vocals

**Soft.** This is an easy compression with a low ratio setting for ballads allowing a wider dynamic range.  Good for live use.  This setting lets the vocal sit "in the track".

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

**Medium.**  This setting has more limiting than the 'Soft' compression setting above for a narrower dynamic rage.  It moves the vocal more up front in the mix.

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

**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 | 0.002mS | 38mS |

## Percussion

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

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

**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.  More 'boom' less 'room'.

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

# 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 | 45.7mS | 189mS |

**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 |
|-----------|-------|--------|---------|
| -6.3dB | 3.4:1 | 188mS | 400mS |

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

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

# 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 had to be heard with the right hand.

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

**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 |
|-----------|-------|--------|---------|
| -11.9dB | 1.8:1 | 0.002mS | 85mS |

**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 |
|-----------|-------|--------|---------|
| 3.3dB | 2.5:1 | 1.8mS | 50mS |

## Stereo Mix

**Stereo Limiter.** Just as the name implies, this is a hard limiter or 'brick wall' setting ideal for controlling the level to a 2 track mixdown deck or stereo output.

| Threshold | Ratio | Attack | Release |
|-----------|-------|--------|---------|
| 5.5dB | 7.1:1 | 0.001mS | 98mS |

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

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

## Effects

**Squeeze.** This is dynamic compression for solo work, especially electric guitar.  It gives you that glassy 'tele/strat' sound.  It is a true classis.

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

**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. It is very contemporary.

| Threshold | Ratio | Attack | Release |
|-----------|-------|--------|---------|
| 0dB | 1.9:1 | 1mS | 0.001mS |

## 3.3    Equalizers

The equalizer in the Studio Channel is like a Swiss Army pocket knife, with the ability to carve out unwanted frequencies as well as tailor your tone to enhance your sound. The custom-designed amplifiers in the EQ deliver sweet-sounding gain/cut for ultra smooth highs, deep solid lows and clear midrange. The mid-band of the EQ section has variable Q for controlling the width around the center point of the selected frequency. High and low frequency bands give you the option for selecting peak/dip or shelving frequencies. What follows is a brief explanation of how an EQ functions as well as some charts to help you navigate the frequency range of various instruments to assist you in quickly setting the best EQ for your recordings and to help you get the most out of your Studio Channel.

An equalizer is a device that allows you to adjust the volume of a frequency or range or frequencies within 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 an iPod.  In recording, equalization is a high art.  Good equalization is critical to a good mix.  When used correctly, an equalizer can provide the impression of nearness or distance.  It can also provide separation between similar sounds in a mix allowing them to both shine through the mix.

## The Parametric EQ

The Parametric EQ and semi-Parametric EQ are the most common equalizers found in most recording and live situations because they offer continuous control over all parameters.  The Parametric EQ divides the audio signal's frequency range in bands, usually three to seven, and provides the same set of control parameters for each band: gain, frequency width (Q), and the center frequency of that Q.

## The Q

The Q is by far the most useful tool a Parametric EQ offers.  By adjusting the width of the Q you can attenuate or boost a very narrow or broad range of frequencies.  Each has its benefits, let's look at the narrow Q first.

A narrow Q has obvious attenuation benefits for removing unpleasant tones.  Let's say the snare drum in your mix has an annoying ring to it.  With a narrow Q, you can single this one frequency out (usually around 1 kHz) and remove it.  This 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 Q 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 60-125 Hz, but the attack of the kick drum is much higher at around 2-5 kHz.  By setting a narrow Q and boosting the attack a bit, you can achieve a punchier kick drum without overpowering the rest of the mix.

A broad Q accentuates or attenuates a larger band of frequencies.  The broad Q and narrow 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 somewhere centered around 100 Hz and an attack hitting in almost dead on at 4 kHz.  In this example, you would use a broad Q boosted in the low frequency band centered at 100Hz and a narrow Q 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.

Included at the end of this section are some general frequency principles to guide you through the wonderful world of equalization, but these are far from set in stone.  So how do you find the best and worst each instrument has to offer and adjust them accordingly? Here's a quick guide:

Each instrument resonates most in a specific bandwidth, so, for instance, if you are working on your kick

drum mic, you would start with the lowest bandwidth and 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' somewhere 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 when each instrument's EQ is set so that it shines through the mix.  Here are some things to keep in mind as you venture on into equalization:

- You can only do so much.  Not every instrument can 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 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, move on.

- Never be afraid of taking a risk.  The best EQ tricks were found by mad scientists of sound.  "Playing" applies to engineers as well as musicians.

### 3.3.3 To Boost or Not To Boost.....

With every instrument there are frequencies that can be attenuated or boosted to add clarity or fullness to it in the mix. And of course, there are frequencies that can make an instrument shrill, muddy, or just downright annoying. The following two charts offer some suggestions of frequency ranges that should be accentuated or downplayed for the most common instruments. Remember, these are just suggestions; these 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 |
| | 80Hz 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 kHz | 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



**BOOST**

| Frequency | |
|---|---|
| 50 Hz | • harder bass to low frequency instruments (kick, tom, bass) • guitar and snare fullness • piano and horns warmth |
| 100 Hz | |
| 200 Hz | • bass line clarity (especially for softer playback levels) |
| 400 Hz | |
| 800 Hz | • bass line clarity and punch |
| 1.5 kHz | |
| 3 kHz | • vocal presence • kick & tom attack • more finger sound on bass • piano and acoustic guitar attack • brighter rock guitar |

- fullness to low frequency instruments (kick, tom, bass)
- vocal fullness
- hard hitting snare and guitar
- bass line clarity and punch
- bass "pluck"
- acoustic/electric guitar attack
- lower piano register attack
- vocal clarity
- kick & tom attack
- percussion attack
- reduce vocal dullness
- more finger sound on bass
- sharpness to: synths, rock guitars, acoustic guitars, piano
- brighten vocals, acoustic guitar, piano
- harder cymbals
- breathier vocals
- brighten cymbals, string instruments, flutes
- more realistic samples and synths

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

- increase guitar clarity
- lessen guitar "boom"
- decrease kick and tom dullness
- decrease dullness in guitars
- adds distance to background instruments
- softens a "thin" guitar
- vocals: reduce "s" sound
- decrease bass "boom" to increase clarity in the mix
- decrease muddiness in vocals and mid-range instruments
- decrease long bass overtones in cymbals
- decrease guitar thinness
- lessen overly bright guitar overtones
- enhance background vocal breathiness
- disguise vocals and guitars that are slightly out of tune
- vocals: reduce "s" sound

**CUT**

# 4. Technical Information

## 4.1 Specifications

| | |
|---|---|
| Channels | One |
| Dynamic Range | >115 dB |
| Frequency Response | 10 Hz to 50 kHz |
| Internal Operating Level | 0 dBu = 0 dB |
| **Microphone Preamp** | |
| Gain | -6 dB to +66 dB (+/- 10dB trim on line input) |
| Noise Floor | -96 dBu |
| EIN | -126 dB |
| THD + Noise (0% Tube Drive) | < 0.050% |
| THD + Noise (100% Tube Drive) | < 0.10% |
| Microphone Input Connector | XLR |
| Microphone Input Impedance | 1.3 kΩ |
| Instrument Input Connector | ¼" TS |
| Instrument Input Impedance | 1 MΩ |
| Line Input Connector | ¼" TRS |
| Line Input Impedance | 10 kΩ |
| Phantom Power (switchable) | +48V |
| Pad (switchable) | 0 dB / -20 dB |
| Phase (switchable) | 0°, 180° |
| Hi-pass filter (switchable) | -3 dB at 80 Hz |
| **Compressor** | |

| | |
|---|---|
| Threshold (variable) | -40 dBu to +2- dBu |
| Ratio (variable) | 1-10 (1:1 to 10:1) |
| Attack (variable) | 0.1 to 200 ms |
| Release (variable) | 0.05 ms to 3 S |
| Gain Make Up | -10 dB to +10 dB |
| Metering | Analog VU (Gain Reduction / Output) |
| **Equalizer** | |
| LF Band Controls: | |
| Frequency | Variable from 40 Hz to 500 kHz |
| Gain | Variable between ±10 dB |
| Q | 0.7 (on 'peak' setting) |
| MF Band Controls: | |
| Frequency | Variable from 200 Hz to 3 kHz |
| Gain | Variable from -10 dB to +10 dB |
| Q | Variable from 0.7 to 2.5 |
| HF Band Controls: | |
| Frequency | Variable from 2 kHz to 20 kHz |
| Gain | Variable between ±10 dB |
| Q | 0.7 (on 'peak' setting) |
| **Master** | |
| Output Fader | -80 dB to +10 dB |
| Output Connectors | XLR Balanced and ¼" TRS Balanced/Unbalanced |
| Output Impedance | 51 Ω |

| | |
|---|---|
| Metering (Output level) | Analog VU (-20 dBu to +6 dBu) |
| Metering (Gain Reduction) | Analog VU -20 dB to 0 dB |
| Output Headroom | +22 dBu |
| **Power 12 VDC, 1A, external power supply** | |
| **Physical** | |
| Size | 1U Rack |
| Dimensions | 19" x 1.75" x 5.5" |
| Weight | 5 lbs |

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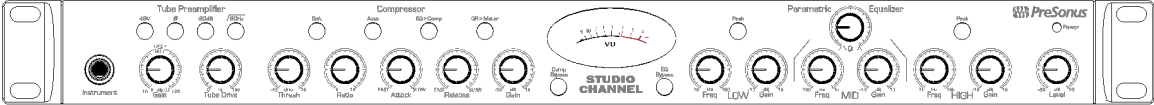
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# Studio Channel

## Vacuum Tube Channel Strip

# Owner's Manual



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