Daking Audio

FET III Stereo Compressor/Limiter Manual

VERSION 1.4

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3/26/2010

Safety Considerations

1. Read, follow and keep these instructions.
2. Heed all warnings.
3. Do not use this equipment in or near water. Do not place liquids on or near the device because the device might be damaged during a spill.
4. Clean only with a soft dry cloth.
5. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
6. Use only Daking supplied power supplies to prevent damage to your device or create safety hazards.
7. Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
8. Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding-type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
9. Protect the power cord and all connecting cables from being walked on or pinched particularly at plugs, receptacles, and the point where they exit from the device.
10. Only use attachments/accessories specified by the manufacturer.
11. Unplug this device when unused for long periods of time.
12. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as when a power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.
13. Do not overload wall outlets and extension cords as this can result in a risk of fire or electric shock.

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1 Daking Audio FET III Stereo Compressor/Limiter Manual

1.1 About Daking Audio

Congratulations! You've purchased a FET III Stereo Compressor/Limiter, a very high end piece of gear! The FET III uses all discrete transistor Class A circuits, THAT Corporation differential amps and printed circuit board mounted switches. Signal capacitors are precision polypropylene or ultra low-leakage electrolytic types. Our boards are assembled on a mil-spec assembly line. The chassis are stainless steel for maximum RF and hum rejection and a long lasting finish. Every unit is hand finished, tested, burned in, and tested again in a second facility.

Also, we just couldn't stand to use plastic knobs, so we designed our own anodized, engraved aluminum knobs that give a much more precise and quality feel. We designed our gear to be gear you'd own for life, not some passing fancy you'd leave in the dust once you figured what the good stuff sounds like. This IS the good stuff.

-Geoff Daking

1.2 Quick Start Guide

1.2.1 Don't read the manual!

Most of you will already know how to use a compressor perfectly well and might be even a little offended at the idea of reading the instruction manual. So don't read it. This manual is not for you.

This manual *is* for someone that knows enough to buy the very best (Daking of course!), but doesn't have a lot of experience using recording gear.

You might be a bass player who just got a DAW and wants more control over your dynamics in your home studio. You might be a student that just got a check from Mom & Dad and wants to go buy something nice for yourself.

You might be the guy standing ankle-deep in a pool of salt water, trying to yank the grounding pin off your mixer's power cord so you can plug it into your 2-prong ungrounded outdoor outlet.

This manual is especially for you!

Whenever you see the Duh! Guy, you can be assured that most professionals will already know this stuff. Be sure to explain this stuff to your friends in a snotty and condescending tone, so you too can be part of the tradition of know-it-all engineers and recordists!

1.2.2 Basic Set Up

Your compressor can be used in a variety of different ways and patched into the signal chain of many different set-ups. The FET III can be used while you are tracking a microphone, while you are mixing tracks down, and in mastering scenarios.

The FET III can be used in both dual mono and in stereo modes.

1. Patch the outputs of your line-level audio devices (like a mic preamps or an insert send on your console) to the Line Input jacks on the rear of the compressor.

2. Patch the Line Outputs on the rear of the compressor to the input of your line-level audio device (like the insert return on your console)

3. Attach 6-Pin DIN cable on the power supply to the back of the compressor

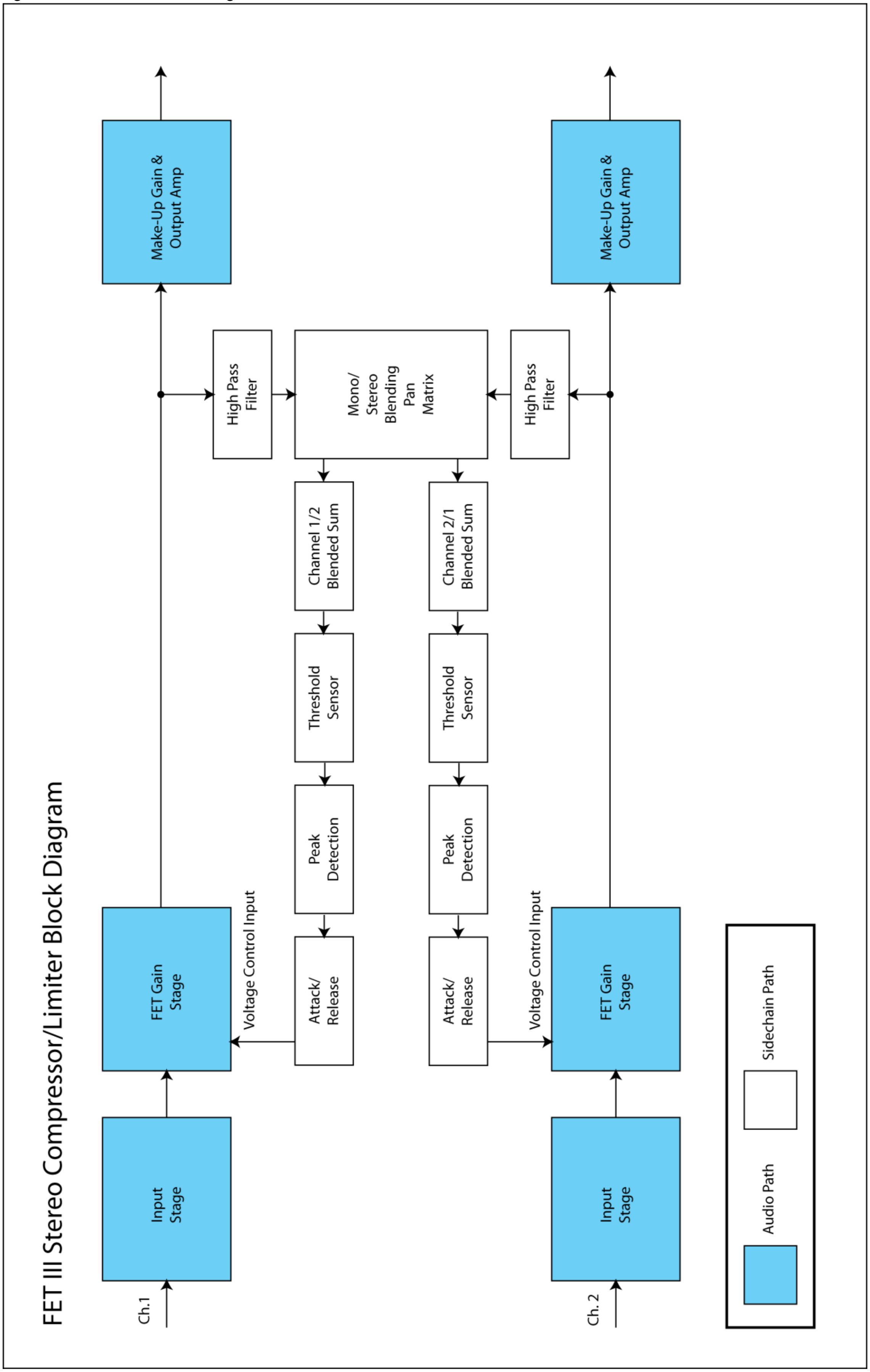
4. Plug the power supply into a grounded outlet, preferably with AC line filtering, surge suppression and voltage regulation.

Get compressin'!

1.3 The Audio Path and the Sidechain Path

One important concept to understand about compression is the difference between the audio path and the sidechain path. The audio path is the audio that is going into the compressor, being processed by the compressor, and then coming out of the compressor. The sidechain path is the audio that is used to determine when and how the compressor will behave. The simplest compressors operate by using the audio input for both the audio path *and* the sidechain path. More versatile processors allow you to alter the sidechain audio, use a different audio source or blend two sources together. The FET III allows you to filter the sidechain audio and to blend the audio from channels 1 and 2 together in the chain. This allows much more precise control of how the compressor works both in mono and stereo modes.

Figure 1 FET III Block Diagram



1.4 Front Panel

1.4.1 In General: Rotary Switches and Potentiometers

Your compressor is equipped with both rotary switches (Ratio Control) and with precision potentiometers. Switches make a big click when you turn them and potentiometers offer a smooth continuously variable feel as you rotate them.

One of the big advantages of switches is that you can reproduce the same settings over and over again. The big advantage of continuously variable potentiometers is that you can get all of the setting in-between the positions on the switch. The FET III uses switches on the Ratio control so that it is easy to use the exact same ratio on both the left and right side of the compressor.

1.4.2 The Threshold Knob (-10 dB to +10dB)

The Threshold knob controls at what level in decibels the compressor starts to work. When a signal goes above the Threshold, the compressor starts to attenuate (reduce the level of) the signal. Signals below the threshold get left alone. If the knob is set to "0," then all the sounds above 0 dB will be reduced in gain and the sounds below will be untouched.

If you want more compression, then you should lower the Threshold. If you want less compression then you should raise the Threshold. See Figure 2.

1.4.3 Ratio Knob (1.5:1, 2:1, 3:1, 5:1, 10:1, 20:1)

The Ratio knob controls how much attenuation (gain reduction) happens to the signal above the Threshold. The Ratio is a comparison of the input to the output, when the signal is above threshold.

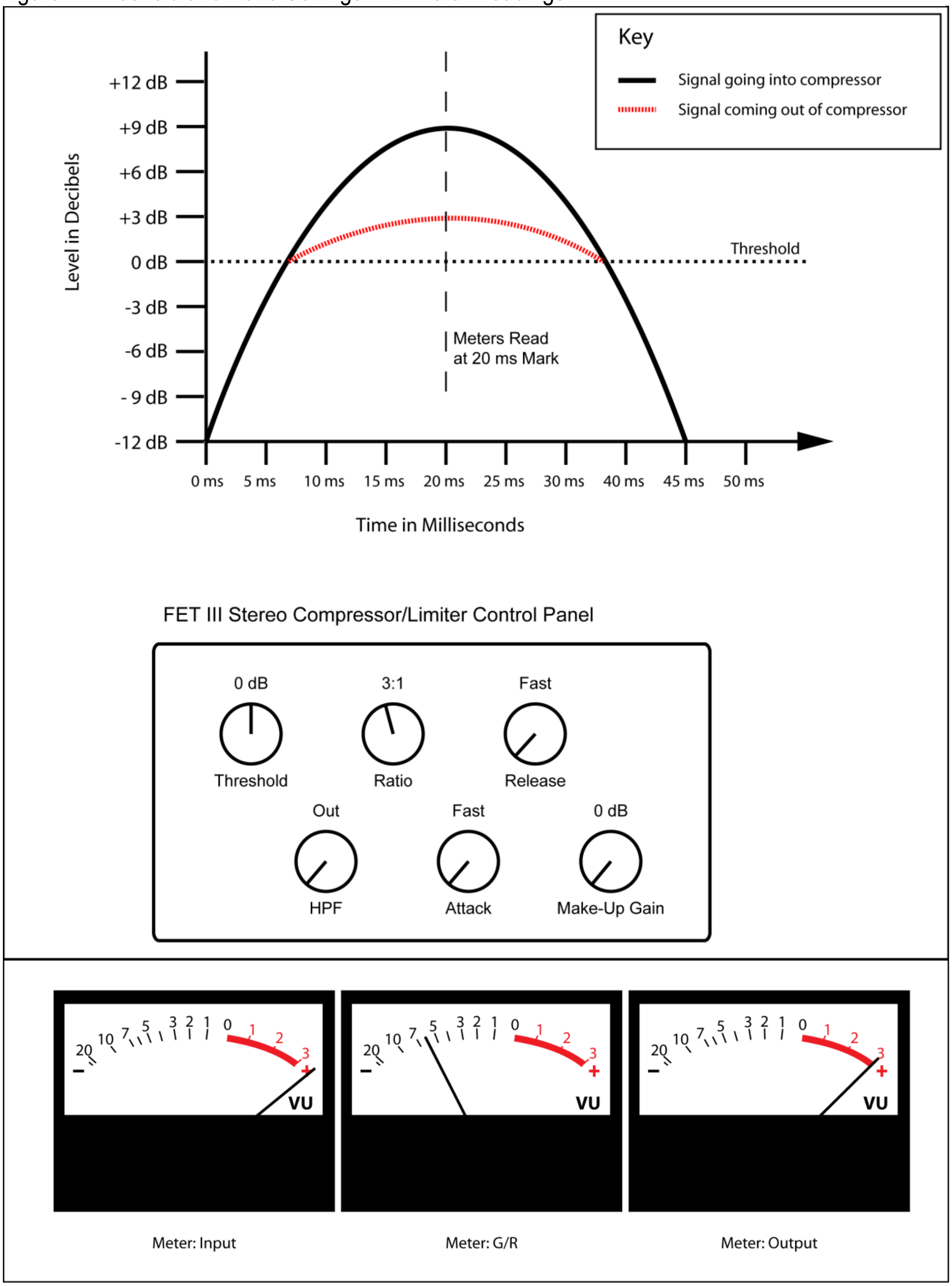
For instance, imagine you have set a 3:1 ratio and set the Threshold to 0 dB. Note in Figure 2 below, the audio input signal rises +9 dB above 0 dB Threshold. As you can see the output signal is only +3 dB above Threshold. The increase of 3 dB input above Threshold yields only 1 dB above the Threshold output.

Higher ratios are easier to hear because they are making a bigger change from the original. So if you are trying to be subtle, stick with generally lower ratios. Sometimes of course you will need to use heavy compression to achieve the results that you are after.

Compressors with ratios of 10:1 or higher are called limiters. Limiters are often used after preamps to protect recorder inputs from sudden overloads, like intermittent screaming or someone dropping a mic during a live recording. The limiter helps prevent the recording from being ruined!

Limiters can also be used to set a volume limit for a playback system to protect speakers from damage.

Figure 2 Threshold and Ratio Settings with Meter Readings



1.4.4 Attack Knob (from 250 µs to 64 ms)

Unfortunately, the Attack knob has nothing to do with using the power supply as a weapon or stopping the guitar player from playing too much!

The Attack knob controls how fast the compressor turns on after a signal crosses the Threshold. If the Attack is set slow, the compressor will react slowly to transients above threshold. If the Attack is set fast, the compressor will react the instant the transient exceeds the Threshold. Fast settings are to the left and slower settings are to the right. See Figure 3 below.

If you set the compressor's Attack too fast the transient will be attenuated too much and the power of the instruments will be reduced. When drums sound wimpy or flat, it is often because the Attack settings are too fast. If you set the Attack to slow, you might miss the transients altogether and not compress the audio enough.

If you have a lot quick explosive peaks, speed up the attack to prevent overloads of downstream devices. If you don't have many fast peaks, use slower settings to level out small dynamic changes in a vocal performance or bass part.

Your Daking compressor is fast enough to control even the most intense transients from drums and percussion. The fastest attack time on the FET III is 250 µs (1/4 of a millisecond) while the slowest attack time is 64 ms.

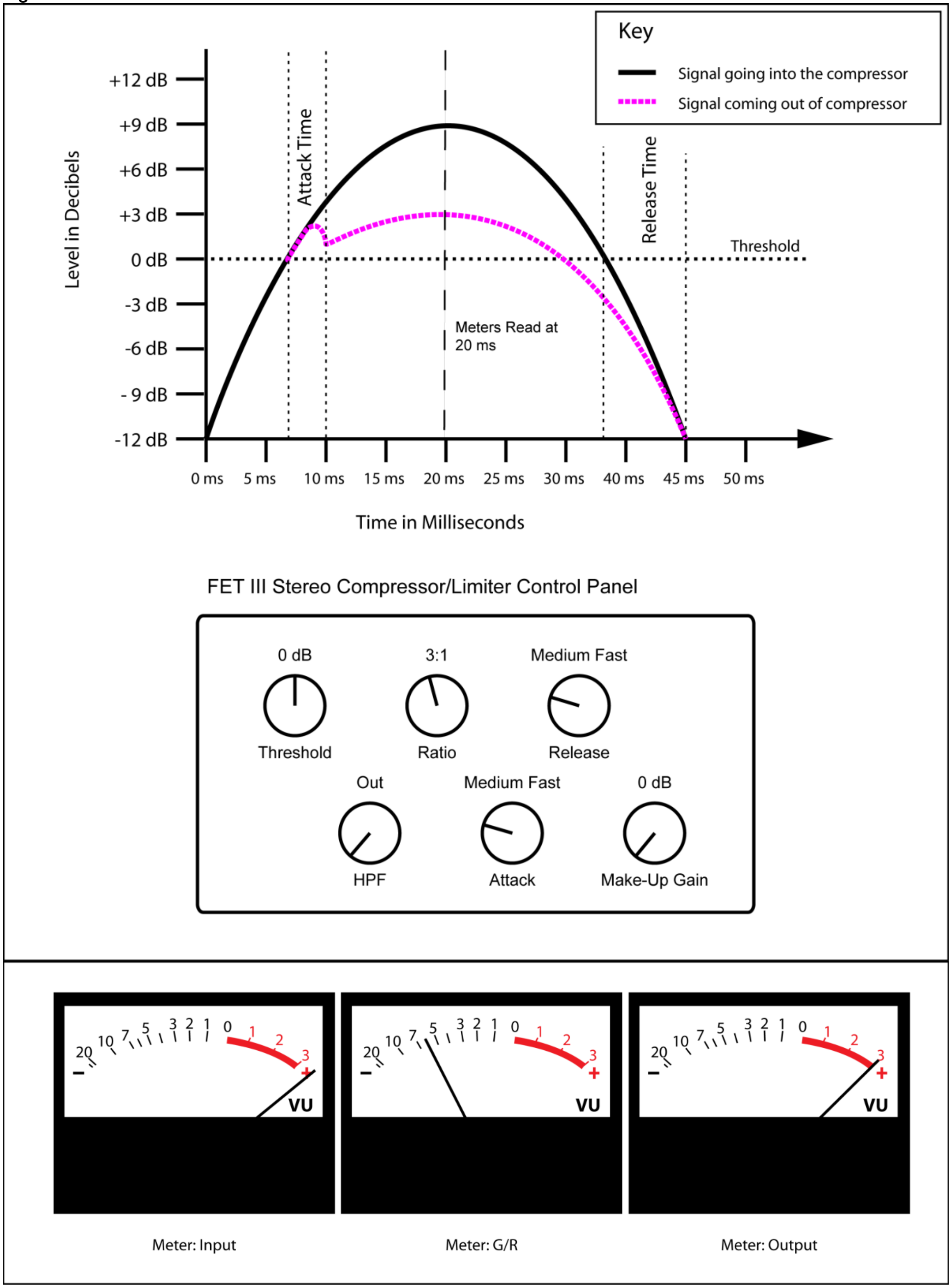
1.4.5 Release Knob (from 500 ms to Dual Time Constant Auto)

There are NO hostages!

The Release knob controls how quickly the compressor stops compressing after the signal drops below the Threshold. See Fig 3 below. The Release control can be used to lengthen sustain times and to blend audio signals together. When Release times are faster, the compression may be less obvious and be more transparent. If Release times are set improperly the compressor may sound as if it is causing the audio to swell, pump or breathe.

The Release time and behavior of a compressor is very important to the signature sound of the unit. The FET III has release times that range from about a half second to a dual time constant release based on the Audio & Design (Recording) Limited's famous F760X 'Compex' limiter. When the FET III is in full Auto mode, the release starts out pretty fast, but then slows down. The total release time can be as long as 7-8 seconds, but dual time constant makes it sound very natural. See Figure 3.

Figure 3 Attack and Release Times



1.4.6 HPF (High Pass Filter: from Off to 200 Hz)

The High Pass Filter only affects the sidechain path and not the audio path. This means that it cuts low frequencies below the Hertz setting on the HPF by 12 dB/Octave in the audio signal that goes through the detection path. See Figure 1. Removing the low frequencies from the detection circuit causes the compressor to not compress when low frequency sounds dominant the audio signal. This allows the compressor to avoid attenuating low sounds like kick drum and bass and still control the dynamics of the midrange and high frequencies.

This feature is especially useful when you are using the FET III for stereo buss compression. As you turn up the HPF knob, you will notice that it has the effect of boosting the bass frequencies, while the mids and upper frequencies get more controlled.

1.4.7 Make-up Gain Knob (from +0dB to +11dB)

The Make-up Gain control restores the overall level and compensates for the attenuation of the audio signal through the process of compression.

1.4.8 VU Meter

Your compressor is equipped with a wonderful device, called a VU meter. A VU (Volume Unit) meter measures loudness more like the way people hear loudness: as an average of the levels over time (in this case, the last 300 ms.) Typically 0 on a VU is equivalent to -18 dBu full scale, or 18 dB lower than digital audio systems' 0 dB. It is also +4dBu, the standard operating level of professional audio equipment. Most digital recorders and DAWs use PPM or Peak Program Meters to show the loudest part of a sound wave. This is really important for digital because anything that goes above 0 dB full scale is going to distort and sound horrible.

The meter can also be used to determine gain reduction in addition to input and output levels. When the compressor is working, the needle will move to the left to show how much the compressor is attenuating (reducing) the gain.

The VU meter can be switched via the 3-way meter switch to provide metering for your input signal, your output signal or the gain reduction that the compressor is providing.

1.4.9 3-Way Meter Switch

The meter switch allows you to change which signal you are evaluating with the VU meter. You can choose input, gain reduction, or output. The gain reduction setting shows how much the signal is being attenuated by moving to the left. This meter becomes important when using compression and make up gain to compare the before (input) and after (output) of your audio.

1.4.10 Bypass Switch

This switch lets you choose between having the compressor turned on and the compressor being bypassed. Bypassing is a good way to hear the before and after of what you have done with the compressor and to hear how good your settings are.

1.4.11 Link: Dual Mono to Stereo

Your compressor can be used as two single mono channel compressors or linked together into a variable stereo pair. The knob controls a mono to stereo pan matrix that blends the left and right channel sidechain signals together. This allows the compressors of both channels to compress at the same time and in the same way.

One important concept of the Link feature is that the audio feeding the sidechain is the affected audio, NOT the audio that you hear. See Figure 1.

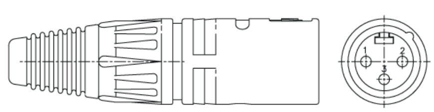
1.5 Back Panel

1.5.1 In General: XLR Connectors

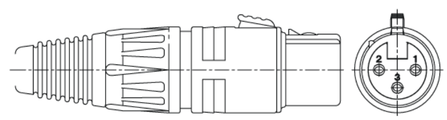
XLR connectors are more expensive, more reliable and offer a stronger connection than ¼" TRS connectors. They also have the option of a locking latch that helps to keep the cable from being pulled out accidentally. If worse comes to worse, you can connect two XLR cables together to make a longer run. The XLR connection is strong enough that you can swing a hand-held microphone around your head like a cowboy for quite a long time before the mic flies off and knocks someone's teeth out.

XLR males are used for Outputs and XLR female are used for Inputs. Makes sense, right? Many people confuse the male and female XLR parts, because the female plug fits into the male plug to join together. The male XLR has 3 pins (male pins…) inside the plug and the female XLR has three holes inside the plug (female holes…). Check out the diagrams below:

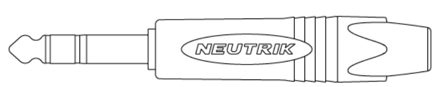
Neutrik Male XLR Plug



Neutrik Female XLR Plug



Neutrik 1/4" TRS Plug



1.5.2 Line Input (XLR or ¼" TRS)

The line input accepts a line-level signal, not a mic level or instrument level signal. You connect to the input of your compressor with an XLR cable at +4bBu.

Your compressor is not expecting a microphone-level signal and definitely not a speaker-level signal! If you want to compress a mic signal you need to run the mic into a mic pre-amp first (a Daking Mic Pre is a good choice!), which boosts the gain of the signal from mic level to line level. Then you can send the output of the mic pre into the compressor. Plugging a mic directly into the compressor just won't work. It's kind of like putting a PB & J into a VHS machine: you can do it, but it's a bad idea.

Plugging a speaker level ¼ input into your Daking Compressor may blow it up. This signal is far too hot to work properly with your compressor. If you need to control dynamic range of a speaker, then compress it as a line level, before the signal goes to the amplifier.

1.5.3 Line Output (XLR)

The output signal from your compressor comes from here. If you are connecting your compressor to a patch bay on a console instead of a line XLR input on another piece of gear, you will want to purchase a female XLR to male ¼" TRS adaptor cable to make this easy. The output signal is line level, <u>not</u> mic level, so patching it into a mic pre-amp afterwards is unnecessary and probably will just cause problems.

1.5.4 +48V DC with a 6 pin DIN connector

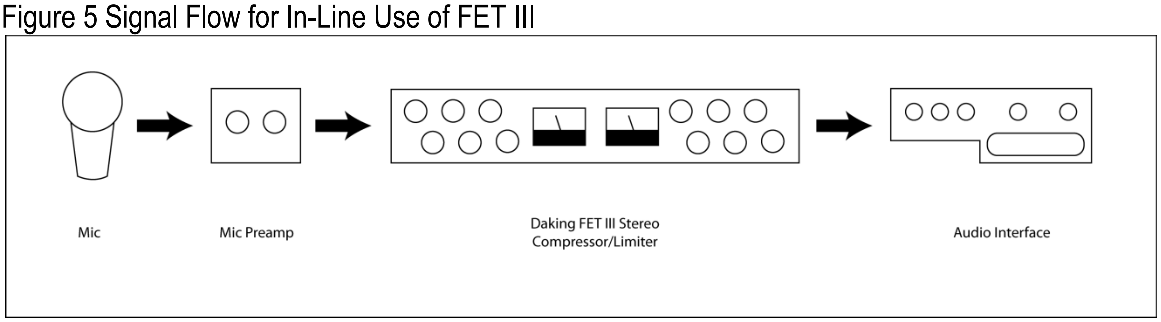
Your FET III Compressor uses an +48V DC external power supply with a DIN connector that is very similar to a laptop power supply. External power supplies offer many advantages over internal power supplies like reducing hum from 50 or 60 cycle power sources and improving the safety of the equipment you are using. Even with the improved safety of the external power supply we still recommend that you avoid recording in the bathtub or swimming pools.

1.6 Signal Flow: Patching Into and Out of Your FET III Compressor

1.6.1 Mic Pre to FET III to Audio Interface or Mixer

Nowadays most people are recording digitally into a Digital Audio Workstation (DAW) like ProTools, Cubase or Sonar. Most of the audio interfaces that are available don't have high quality microphone preamps, so many recordists purchase outboard preamps, like Daking's Mic Pre One, to ensure professional quality sound. The mic preamp and compressor are connected inline directly into the audio interface.

Figure 5 Signal Flow for In-Line Use of FET III



The job of the mic pre amp is to raise the level of a mic level signal to line level so that it can be manipulated or recorded. You can't plug a mic into the FET III and hope to get anything useful out of it.

Basic Cables Needed:
(2) Microphone Type Cables (Female XLR to Male XLR)
(1) Female XLR to Male ¼" TRS Balanced Cable

Here are the steps:

1. After making sure that phantom power isn't turned on and that your studio monitors are muted, patch a microphone to the mic preamp with a mic cable (XLR Female to XLR Male).
2. Patch out of the mic preamp to an input jack on the FET III, with another mic cable (XLR Female to XLR Male).
3. Patch out of the FET III into the line input on your interface or your mixer. Use a XLR Female to ¼" TRS Male cable. Make sure that you aren't going into another mic preamp! If you do, you will likely end up with a distorted signal.
4. Turn on phantom power (if needed) and set levels.

1.6.2 Connecting Via a Single Insert Jack

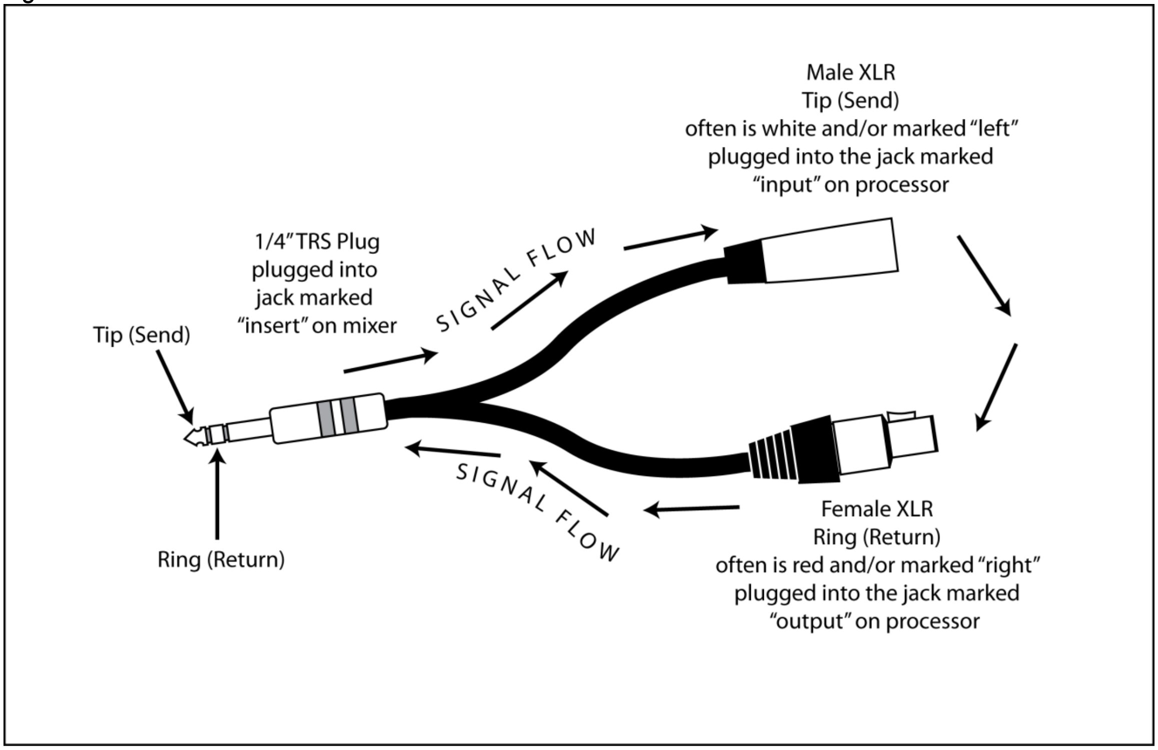
On many live and hybrid mixers, inserts are patched via a single insert jack using a special insert cable. The insert cable for the FET III (see Figure 3 below), often called a 'Y' cable, consists of a ¼" TRS plug on one side and 2 XLR plugs on the other side, one male and one female.

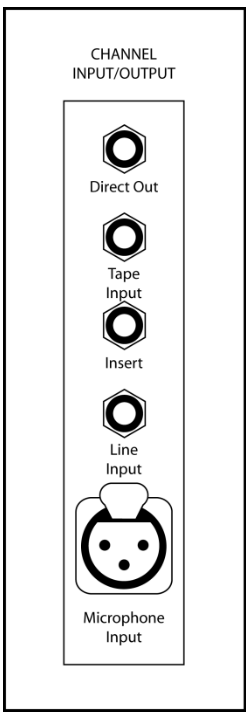
The tip of the TRS plug is wired to the 2 pin of the Male XLR plug (usually the white or left plug if marked) and the ring of the TRS is wired to the 2 pin of the Female XLR connector (usually the red or right plug if marked.) The sleeve of the TRS plug is wired to both pins 1 and 3.

The tip of the TRS plug is the "send" and the ring of the TRS plug is the "return." A really good way of remember this stuff is:

Red, Right, Returning and Ring all start with "R."

Figure 6 Insert Cable for Use with FET III





On the familiar RCA plugs used on home theater equipment, the red plug is always the right side. 'Y' cables often use the same color scheme because usually the cable was intended to split a stereo signal into two mono signals: left and right. When you use this 'Y' cable as an insert cable the tip side is the send side and the ring side is the return side. This cable is BOTH an output and an input!

Steps for Patching with an Insert Jack and Cable:

1. Patch the TRS side of the Insert or 'Y' cable into the jack marked "insert" on the patch bay of your mixer. If you had signal playing through the channel at the time, this cable will break the connection, and you shouldn't hear anything. If you touch the tips of the other two plugs together, you will get signal back.

2. Plug the connector marked "tip" or "send" or simply in white into the input connector.

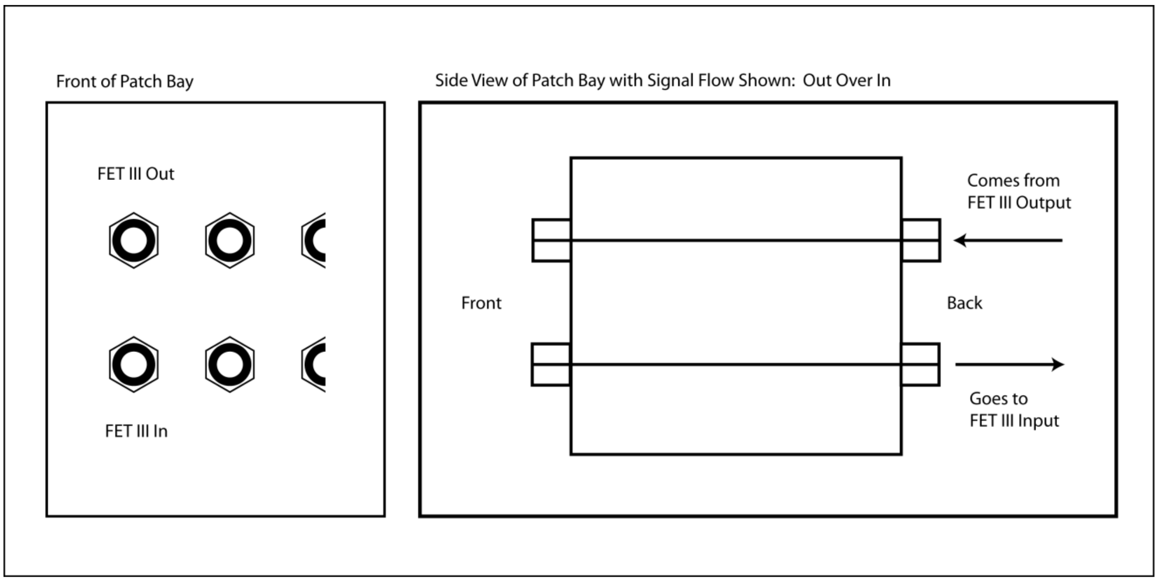
3. Plug the connector marked "ring" or "return" or that is in red into the output connector.

1.6.3 Via a Patch Bay

In most professional setups, processors are racked and then installed as part of a patch bay system. This makes it easier to make connections, because you don't need to go behind the rack to plug and unplug cables.

Patch bays are usually made up of modules with 2 jacks in the front, one over the other, and two jacks (or solder terminals, or DB25, etc) on the back. For the sake of this manual, we'll assume you're using a ¼" TRS patch bay with jacks on the front and back.

You do not want to normal the inputs and outputs from the compressor to each other. This will create a feedback loop. Converting a half-normal module to a non-normaled module is sometimes as simple as rotating the module in the patch bay. See the manual for your patch bay.



Cables Needed:
(1) XLR Male to ¼" TRS Male
(2) XLR Female to ¼" TRS Male

Steps:

1. Patch from the bottom back jack on the patch bay module to the input on the FET III using the ¼" TRS to XLR Male cable.

2. Patch to top jack of the patch bay module from the output of the FET III using the XLR Female to ¼" TRS cable.

3. Now you can patch into the compressor from the front of the patch bay. Simply run a patch cable from your source and into the bottom

jack on the module and then run from the top jack on the module to wherever the signal needs to go.

1.6.4 Stereo Setup as a Bus Compressor for Mixing or Mastering

The purpose of the Stereo Link mode for bus compression is to make sure that both compressors compress in unison. This prevents the stereo image from shifting to the left or right when one side of the stereo field is compressed and the other side is not because of a big sound on only one side of the stereo mix.

Stereo bus compression should used in a subtle way because aggressive compression in stereo tends to create undesirable artifacts. Bus compression is most often used to unify a mix into a cohesive whole and to control dynamics of an entire mix allowing loudness to be maximized.

Steps:

1. Connect both sides of the FET III to the left and right sides of the program respectively.
2. Insert the compressors, or connect them inline as shown above.
3. Turn the Link knob all the way to Stereo Link.
4. Set both sides of the compressor to the same Ratio, Attack and Release, and then adjust the Threshold on both sides so that the meters show the same amount of Gain Reduction on both the left and right sides. You may find that Threshold controls may not look like they are in the same position even when they sound like they are balanced. This is normal because the audio on either side of the mix is not the same.
5. Switch your meter to monitor the Output Level and verify that both sides of the mix are still in balance. Make adjustments to the threshold controls if they are not in balance. Turn the Threshold down on the side that has too much level or turn the Threshold up on the side that has too little.
6. After you have balanced the left and right sides you can adjust the Output Gain to get the level that you desire.
7. Experiment with "in-between" settings of the Link knob and various permutations of the Attack and Release settings.

Tip for setting the compressor in Stereo Link:

1. Change the bus you plan on compressing into a mono pair, either by adjusting pan controls or engaging the "mono" button on your console. (Note: You can also send a test tone as two mono signals to both sides of the compressor)

2. Set the Ratio, Attack and Release to the same setting on both sides and then adjust the Threshold controls so that both sides are showing equal Gain Reduction and equal Output levels.
3. Return your bus back into stereo and both sides will be balanced equally.

1.7 Typical Uses of a compressor

1.7.1 Why do I need a compressor anyway?

Compressors help to control the dynamic range of audio signals. The dynamic range of an audio signal is the difference between the quietest part of a signal and the loudest part of the same signal over the course of the entire song or program. Compressors reduce this difference between loud and soft (signal dynamic range) to keep things more consistent from start to end. In a nutshell, a compressor allows you keep an element of a mix at the level you choose relative to all the other signals you are mixing. Without compression, the guitar solo might fall into the background or the drums might over take the vocals!

1.7.2 Keeping a vocal performance out front

Compression is THE way to get a vocal performance to stay out front and stand out in the mix. The compressor reduces the difference between the singer's loudest notes and softest notes to make it more consistent. This same idea works also with background vocals to keep them from rising too far out front. A great starting point for vocals is:

Parameter	Value
Threshold	0 dB
Ratio	5:1
Attack	1 ms
Release	Auto
Make-up Gain	+6 dB

If the vocalist is extremely aggressive you may need to speed up the attack time (how fast the compressor turns on after threshold is exceeded) to control the transients on plosive consonants like "P,", "T," "D," "B", "C," and "K."

If the vocalist is extremely dynamic you may need to increase the Ratio (increase the compression above the threshold) and/or reduce the Threshold (initiate compression at lower levels).

1.7.3 Keeping bass consistent

A compressor is a great tool to use to smooth out a bass guitar track and help to keep the bass audible during the louder parts of the mix. The bottom end of the band is very important! Bass guitar has a tremendous amount of energy and readily overloads tracks. A compressor helps reduce these major energy peaks to keep the bass under control. Try this setting:

Parameter	Value
Threshold	0 dB
Ratio	3:1
Attack	2
Release	.5
Make-up Gain	+6 dB

1.7.4 Fatten kick drums

A very common use of a compressor is to increase or decrease the sustain of a kick drum sound. Drums have a very fast and very intense transient followed by a quieter sustained sound which is the tone of the drum. A kick drum can be made to sound fuller, by decreasing the peak level (amplitude) of the attack of the drum and increasing the make up gain to achieve a lower but longer (sustain) of the tone of the drum.

Parameter	Value
Threshold	0
Ratio	5:1
Attack	Medium Fast to Fast
Release	.5
Make-up Gain	+6

Speed up the attack of the compressor to attenuate the transient more, slow the attack down to attenuate the transient less.

It's best to try to use the fastest release you can and still achieve the sustain that you are trying to achieve. A release time that is too long can diminish the intensity of the second or third attacks of a fast kick drum pattern, like a $1/16^{th}$ note pattern.

1.7.5 Limiting to preventing overload, clipping and distortion

Limiting uses a very high Ratio with a high Threshold. The idea is to control the just the loud peaks. Limiting can be useful in preventing digital distortion, overload of input in gear after the compressor, and tape overload (saturation) from too high an input. Using a very fast Attack time and a high Ratio will ensure that the fastest transients are captured.

A good starting point for limiting is:

Parameter	Value
Threshold	+10
Ratio	20:1
Attack	250 µs
Release	.5
Make-up Gain	0

1.7.6 Pointers and General Principles

Compression is used to reduce the dynamic range of an audio signal.

Use the Gain Reduction (GR) setting on the meter to see how much you are attenuating (reducing in gain) the audio signal. Boost the Make-Up Gain so that the input and outputs are similar levels.

To set the Attack and Release times turn the Ratio way up so it's easier to hear the compressor working. Set the Attack and Release correctly and then reduce the Ratio to a more reasonable level

1.8 Specifications

Method of Limiting: FET (Field Effect Transistor) used as a variable resistor

Stereo Linking Method: Continuously Variable Audio Summing

Power: 48volts dc @ 150 ma

Output Drive: Differentially balanced

+24dbv @ 1kHz 600 Ω

Ratios: 1.5:1, 2:1, 3:1, 5:1, 10:1, 20:1

Attack Times: 250 µs to 64 ms

Release Times: 500 ms to Auto Dual Time Constant

Frequency Response: ±1db 10Hz to 56kHz 3db down 63kHz

Noise: -82db 10Hz to 25kHz

Distortion: THD+N 0.033 @ 1kHz

Line Inputs: The XLR inputs are balanced with THAT Corporation differential amps. Up to +24dBu input.

Line Outputs: The XLR Outputs are balanced with THAT Corporation differential amps. Up to +24dBu output.