Focusrite

GREEN RANGE

CHANNEL STRIP

manual

GREEN 5: CHANNEL STRIP

Green 5 Channel Strip

Introduction

The Channel Strip is a combined preamplifier, equaliser and dynamics unit.

This makes it an excellent recording tool, since you no longer need to use a mixing desk when recording: instead, you need just a microphone, recorder (such as a DAT machine) and a Channel Strip to record direct, monitoring using the return from the recorder. This ensures the highest quality of recorded signal, since it removes unwanted elements from the signal chain and so reduces the amount of noise added to the signal.

There are five separate parts to the Channel Strip:

- Preamplifier.
- Filters.
- Equaliser.
- Gate/Expander.
- Compressor.

Sections can be switched out of the audio path when not in use. This ensures the maximum quality, since unused components are not present in the audio path when not required. When no dynamics are used, the VCA is relay bypassed.

Getting to Know the Unit

The easiest way to get to know the unit, particularly if you are not familiar with all its separate parts, is to try each control in turn, so that you can hear its effect. This section gives a checklist for working through the unit's controls in a logical sequence.

When you are getting to know the unit, use it on a track that you are familiar with (for example, you could run a favourite CD through the unit); working with a familiar track makes interpretation of the results easier.

Note, however, that tracks are already compressed for CD, so you may find it hard to hear the results of using the compressor. If this is the case, try using samples instead (if you have access to them), or record your own track uncompressed and then play it back through the Channel Strip.

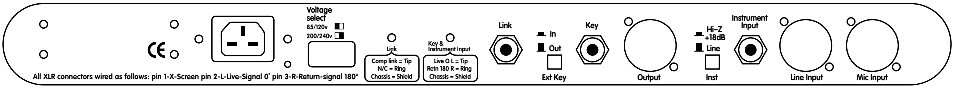
1. Before you start to use the unit, ensure that it is installed and set up correctly (see the Setup section). For a description of how to select the input socket that is used for the audio signal, see the section Selecting the Input Signal.

2. Ensure that all the buttons on the front panel are out (not lit). Adjust the signal level into the unit as described in the section Setting the Gain.
3. Set the other controls on the front panel to the following starting positions (reading from left to right):
   - Filters off (low filter control fully anti-clockwise, high filter control fully clockwise).
   - Gain (dB) controls in the centre (click) position for Low Eq, Low Mid Eq, High Mid Eq, and High Eq.
   - Gate threshold at maximum (fully clockwise).
   - Gate release in its central position.
   - Compressor threshold at maximum (fully clockwise).
   - Ratio on full (fully clockwise).
   - Compressor release in its central position.
   - Make up control at minimum (fully anti-clockwise).
4. To hear the effect of the filters, press the Filters button (so that it lights), and adjust the low filter control – notice how it affects the bass. Then set the low filter off again, and adjust the high filter control to see its effect. Pressing the Filters button in and out lets you monitor the effect of the filters. When you have finished with the Filters, switch off the Filters button.
5. Before listening to the effect of the equaliser, turn down the Output Fader control (since further adjustments may raise the volume, and you don't want to damage any other units). Press the Eq button (so that it lights), and set the Low Eq gain (dB) control to +18 (fully clockwise). Now adjust the Low Eq Freq control, and hear how this boosts the low end of the signal. Pressing the Eq button in and out lets you monitor the effect. Set the gain control back to its centre (click) position, then set the High Eq gain control to +18 and adjust its Freq control. If you find that the unit is overloading due to increased signal level from the Eq, the level of the Input Trim control can be reduced to lower the internal signal level of the unit. When you have finished, return its gain control to its centre position.
6. Now do the same thing with the Low Mid Eq and High Mid Eq – with one's gain in the centre position, set the other's gain to +18 and adjust its frequency control. Again, pressing the Eq button in and out lets you monitor the effect. Notice how these controls affect less frequencies than the Low and High Eq. Experiment with the Fine button for each, and see how the effect becomes subtler when Fine is lit. When you have finished, switch off the Eq button.

7. To monitor the effect of the gate, press the Gate In button (so that it lights), and adjust the left-hand threshold control, which is the threshold control for the gate. Adjusting the threshold clockwise gates more signal – on quiet passages, you should be able to set the threshold so that very quiet parts are gated (removed) but louder parts are allowed through. Pressing the To Meter button (so that it lights) causes the meter's red LEDs to monitor the gate – when the left LED is lit, the gate is shut, when the right LED is lit, the gate is open. Adjust the gate's release control – as you slow the release, you can see the gate closing more slowly on the meter.
8. To monitor the effect of the expander, press the Expand button (so that it lights), and adjust the same threshold and release controls as before. Notice how the expander has a less dramatic effect than the gate – instead of switching the signal on and off, it fades the signal in and out. When you have finished with the expander, switch off the Expand button and the Gate In button.
9. Press the Comp In button (so that it lights) to bring the compressor into the circuit, and adjust the right-hand threshold control. This is the threshold control for the compressor – as you turn it anti-clockwise, more of the signal is affected, so it becomes easier to hear the compression. If the To Meter button is lit, the meter also reacts using the yellow LEDs to show compression, lighting more to the left as more compression is applied.
10. Set the threshold at -24 (fully anti-clockwise), and adjust the make up control to restore the signal volume, so that pressing the Comp In button in and out gives a similar sort of signal level.
11. Turn down the make up control (since further adjustments may reduce the compression and so raise the volume, and you don't want to damage any other units). Adjust the ratio control – as you reduce the ratio, so the signal is less compressed and the effect becomes less noticeable.
12. Set the ratio control where you can easily hear the effect of compression, and adjust the release control. The effect of release is particularly easy to hear on drums (a looped snare is good, if you have access to one). Also try pressing the Auto Rel button to hear the effect this has.
13. Set the release back to its central position, and ensure that the low filter control is fully anti-clockwise, and the high filter control fully clockwise. Press the S/C Filters button (so it lights), and adjust the low filter control – notice how bass becomes less affected by compression as you increase the low filter. Then set the low filter off again, and adjust the high filter control to hear its effect. Pressing the To S/C filters button in and out lets you monitor the effect of the filters.

Connections

All of the unit's connections are on the rear panel.



Power Input

There is an IEC mains lead supplied in the package which should have the correct moulded plug for your country. The wiring colour code used in all Focusrite products is:

For units shipped to the USA, Canada, Taiwan and Japan
Live – Black **Neutral** – White **Earth** – Green

For units shipped to any other country
Live – Brown **Neutral** – Blue **Earth** – Green and Yellow

The chassis is connected directly to the mains safety earth. We do not provide an earth lifting switch, since such a switch can allow for a dangerous wiring arrangement.

Warning: For safety reasons, it is absolutely IMPERATIVE that the mains safety earth is connected.

Voltage Select

The Channel Strip will operate on a range of voltages, and has a two-position switch on the rear panel that should be set to the correct voltage:

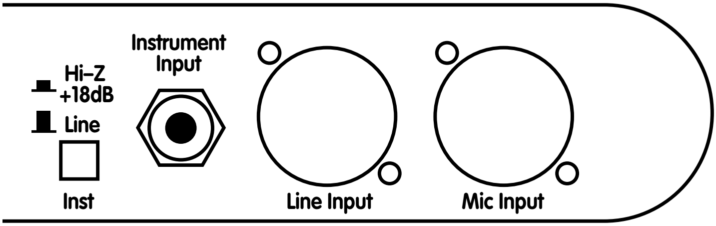
115V	Set to this position if the module is to be used with voltages in the range 85V to 120V
230V	Set to this position if the module is to be used with voltages in the range 200V to 250V

To comply with the safety codes in some countries, modules may be supplied without a voltage selector. In this case, the module is preset to the local supply voltage, which is clearly marked on the rear of the module. Check that the voltage is set correctly.

The module will work correctly from either 50Hz or 60Hz power supplies, and draw approximately 35VA from the mains supply at highest load.

Audio Inputs and Outputs

All the audio signal connections are via connectors mounted on the rear panel.



Mic	XLR input for microphones only.
Line	XLR input for line level signals (such as the output from a tape machine) .
Instrument	1/4″ stereo jack input for instruments (from guitars with coil pickups to keyboards) – inserting a mono jack plug in this socket automatically gives unbalanced input. Connecting anything into this input disconnects the line input XLR.
Inst switch	Modifies the impedance of the Instrument socket, to suit the instrument connected. If connecting an instrument with pickups, switch the Inst button on the back panel to the Hi-Z setting, which prevents the unit from loading the pickups, so giving full frequency response. Selecting Hi-Z illuminates the Inst light on the front panel. If connecting a line-level instrument (such as a keyboard) switch the Inst button on the back panel to the Line setting, or use the Line input.

All XLR connections are wired to the AES standards, with the screen (pin 1) connected to the chassis earth point:

Pin 1	Screen	chassis
Pin 2	Live	audio 0°
Pin 3	Return	audio 180°

You can make either balanced or unbalanced connections to the inputs and output.

When the screen and earth wiring of the module is completed correctly, all modules which are marked with the European Community CE marking comply fully with the CE EMC regulations.

Link Socket

This is a stereo jack socket that you can use to connect the compressor sections of two Channel Strips.

When you connect two units (using a normal stereo jack to jack lead), the compressors behave as if both units were receiving the same volume signal. Since there is no master/slave relationship between the units, whichever unit is currently receiving the louder signal determines the assumed signal volume in both units.

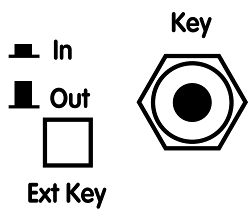
For example, units A and B are linked: the signal going into unit A is at 0dB, and the signal going into unit B is 5dB. In this case, both units react as if they were receiving a signal of 5dB. If, subsequently, the signal going into unit A stays at 0dB, but the signal going into unit B drops to -10dB, then both units react as if they were receiving a signal of 0dB. Note that the compression applied to the signal depends on each unit's setting: if unit A's threshold is set to 10dB, and unit B's threshold is set to -5dB, then unit A will perform no compression on the signals, while unit B will compress both. For normal stereo compression, both units should be set identically. For more details about compression, see the Compressor section.

Key Socket and Ext Key Switch

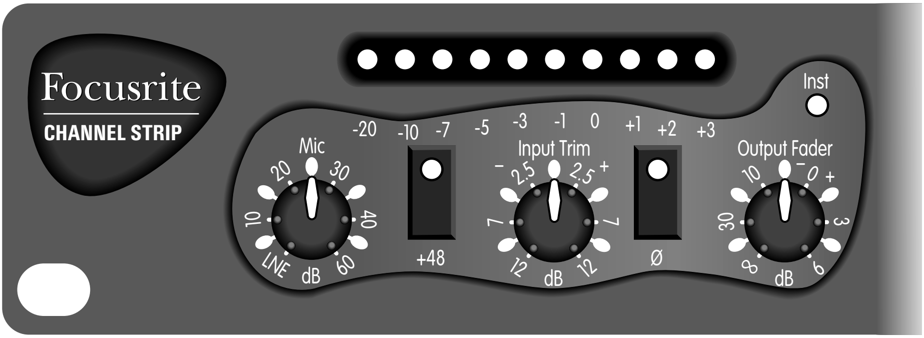
The key socket is a stereo jack socket where you connect an input signal to control the gate/expander. Press in the Ext Key switch and the front panel Key switch to feed the signal to the gate/expander. Note that when the Ext Key switch is in, the Ext light on the front panel is illuminated. For more details of using the key socket, see the description of the Key switch in the Gate/Expander section.

You can connect either balanced or unbalanced inputs to the key socket. You can connect an unbalanced input in one of two ways: by using a mono jack plug, or by using a stereo jack plug with the ring connected to the shield.

For a description of making balanced inputs, see the Audio Inputs and Outputs section.



Preamplifier



Controls

The Mic and Input Trim rotary controls set the gain on the input signal (see the section Setting the Gain).

+48 is the phantom power button. When lit, it provides phantom power to the microphone connected to the Mic socket.

ø is the phase button, and reverses the phase of the unit when lit (see the section Setting the Phase).

The Output Fader controls the output level from the unit, and is used to balance the output level to match the input level of the next piece of equipment.

The Inst light indicates that the Inst switch on the back panel is switched on.

Selecting the Input Signal

You can select which of the three input sockets on the back panel is used as the input signal to the unit:

Input	Used when...
Inst	Mic rotary control clicked to Line position, instrument connected to the Inst input on the back panel.
Line	Mic rotary control clicked to Line position.
Mic	Mic rotary control not clicked to Line position.

Setting the Gain

Depending on the input you are using, you use either the Mic rotary control or the Input Trim rotary control to set the gain:

Input	Rotary control
Mic	Mic
Inst or Line	Input Trim

Use the meter and the correct rotary control to match the incoming level and gain to the internal operating level. With an input signal coming into the channel, watch the meter as you use the rotary control to modify the gain, and set the control so that the meter registers between -3 VU and 0 VU. This sets the level above the noise level of the unit, and leaves room for any sudden increase in performance level (it gives about 20dB of usable headroom).

Note that when using the Mic rotary control, the Input Trim is not disabled, since you could use it to make fine adjustments if required. However, you normally switch off the Input Trim when using the Mic rotary control.

Off position: To switch off the Input Trim (for example, when using the mic input), set the control in its centre (click) position. To switch off the Mic rotary control, click it to the LINE position.

Setting the Phase

When recording a single source using more than one signal, it is possible for the signals to be out of phase, which affects the quality of the recording by sounding “thin”.

For example, when recording a snare drum with two microphones (one on the top of the snare, the other on the bottom) they will be out of phase. Use the phase switch to reverse the phase on one of the microphones (but not both) – it normally doesn’t matter which microphone you reverse. However, if the source is being picked up by another microphone (for example, by an ambient microphone) then you need to ensure that you do not put your two close microphones out of phase with the ambient microphone.

If you think two signals are out of phase, listen for phase as follows:

1. On your monitor system, pan one signal to the left and the other to the right.
2. Use the phase switch to reverse the phase on one of the signals. When the two signals are in phase, the signal sounds bigger.

Equaliser

An equaliser is a sophisticated tone control, that boosts or cuts selected frequency bands, and so alters frequency response. By modifying different frequency bands in this way, you modify the tone quality of instruments and vocals, which can fix problems with the original sound, or help a track stand out in the mix, for example. For more details, see the section How to Use Equalisation.

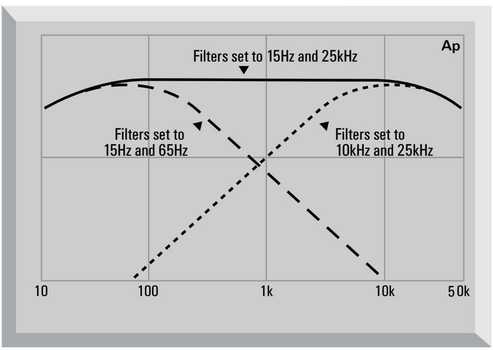
There are three types of equalisation on the Channel Strip:

- Low and High filters.
- Low and High Eq (shelving).
- Low Mid and High Mid Eq (parametrics).

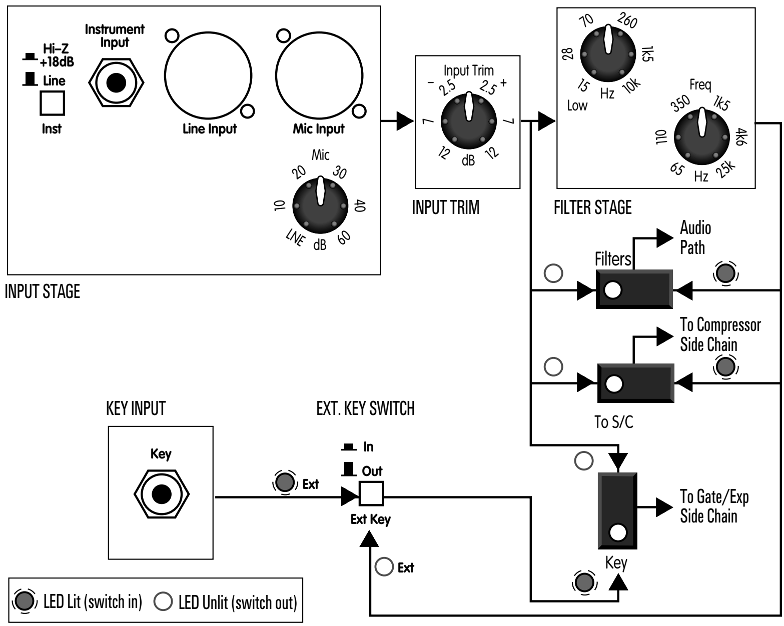
Low and High Filters

The Low filter allows everything above a certain frequency to pass through, hence it removes very low frequency signals. Setting the Low filter to a certain frequency attenuates (reduces the volume of) all frequencies below that. Similarly, the High filter removes high frequency signals, so setting the High filter to a certain frequency attenuates all frequencies above that.

You cannot change the amount of attenuation of a filter – setting one of these filters always attenuates the affected frequencies by a preset amount of 12dB per octave. For example, setting the High filter at 12kHz attenuates the signal at 24kHz by 12dB and the signal at 48kHz by 24dB (for any frequency, the octave above it is at double the frequency).

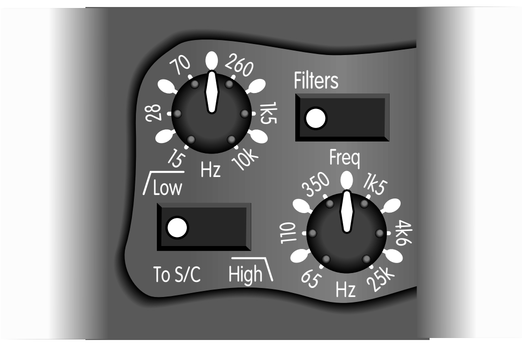


Note that the filters can be used to affect the audio signal, as described below, or they can be used to affect the action of the compressor or the gate/expander on the unit, as described in the Compressor and Gate/Expander sections. Any combination is possible, but only with the same filter frequency settings (see flow diagram opposite).



How filters fit into the circuit

Controls



- The Low and High rotary controls set the frequencies affected by the filter. For the Low filter, all frequencies below the specified frequency are attenuated. For the High filter, all frequencies above the specified frequency are attenuated.
- The Filters button switches the filters in and out of the audio signal.
- The To S/C button switches the filters in and out of the compressor. For more details, see the Compressor section.
- The Key button (located with the gate/expander controls) can assign the filters to the gate or expander. For more details, see the Gate/Expander section.

Off position: For practical purposes, the off position for the Low filter is 15Hz, and for the High filter is 25kHz. To switch both filters out of the audio signal, use the Filters button.

Using the Low and High Filters on Audio

The Low and High filters are usually used to correct problems with a signal rather than being used in a creative way (for example, to create a special effect).

Use the Low filter to remove:

- Unwanted rumble.
- Bass lift (a proximity effect of microphones, giving a bass boost as the singer gets closer to the microphone). This is most apparent with unidirectional microphones.
- Hum from noisy sources.

Use the High filter to remove:

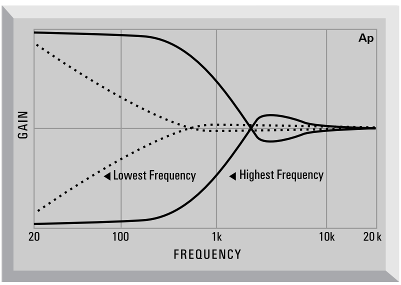
- Unwanted hiss.
- High-frequency bleed-through from other sources when recording bassy signals (such as bass drum).

One possible creative use of the filters gives an effect similar to a telephone. To achieve this effect, use the filters to dramatically reduce the frequencies allowed through the unit (for example, set the Low filter at 100Hz and the High filter at 3kHz).

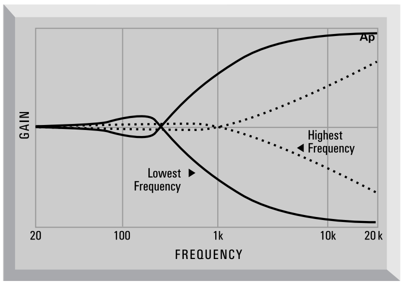
Low and High Eq

The Low and High Eq affect all frequencies below or above a given frequency, letting you attenuate (reduce the volume of) the signal, or add gain to it. Thus, the Low Eq lifts or cuts the low-frequency end of the frequency spectrum, and the High Eq lifts or cuts the high-frequency end – using them together you can, for example, boost the low end and cut the high end of the frequency spectrum, so appearing to tilt the frequency response towards the low end.

LOW SHELF

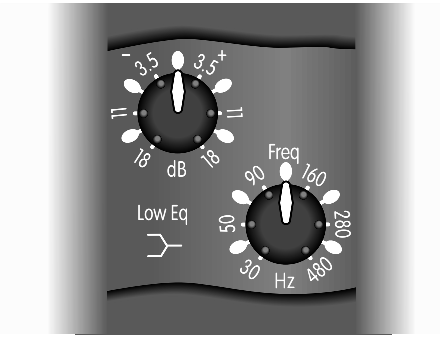


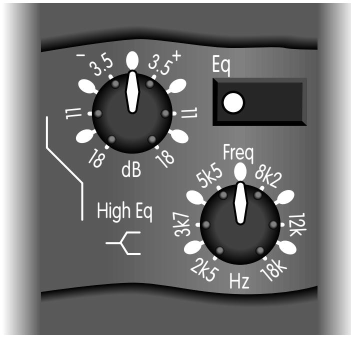
HIGH SHELF



Controls

- Freq sets the frequency below which (for the Low Eq) or above which (for the HighEq) all frequencies are affected.
- dB sets the amount of attenuation or gain applied to the affected frequencies.





Off position: Set the gain control in the middle (click) position, so that it is neither boosting nor cutting frequencies.

Using the Low and High Eq

Low and High Eq are often used to correct problems with a signal:

- To compensate for a lack of something in the original recording (for example, if you had to roll off a lot of bass during recording because you were getting a lot of bass lift).
- To replace something you lost in the recording format (particularly top end).
- To reduce something excessive (such as a very bassy sound).

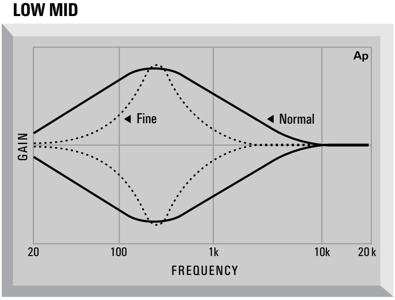
Use the Low Eq to boost subsonic information, thus giving a bassy sound, or to attenuate a sound that is too bassy.

Use the High Eq to boost ambience and reverb in a room, or to attenuate an over-bright sound.

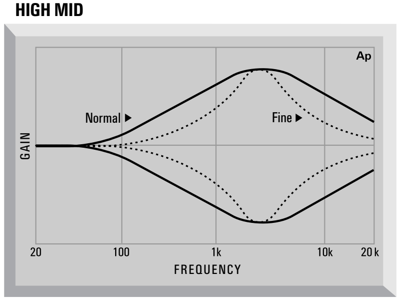
Low Mid Eq and High Mid Eq

The Low Mid Eq and High Mid Eq give you control over the entire frequency spectrum. Unlike the High and Low Eq, the Low Mid Eq and High Mid Eq affect only a given frequency, plus some frequencies to either side.

LOW MID



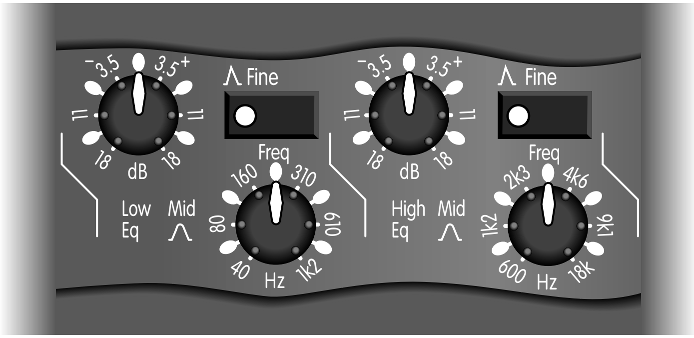
HIGH MID



Controls

- Freq sets the centre point of the affected frequencies.
- dB sets the amount of attenuation or gain applied to the affected frequencies.
- The Fine button sets the range of frequencies affected. Note that the Fine setting for the Low Mid Eq is narrower than the Fine setting for the High Mid Eq, for a tight bass sound.

Off position: Set the gain control in the middle position, so that it is neither boosting nor cutting frequencies.



Using the Low Mid and High Mid Eq

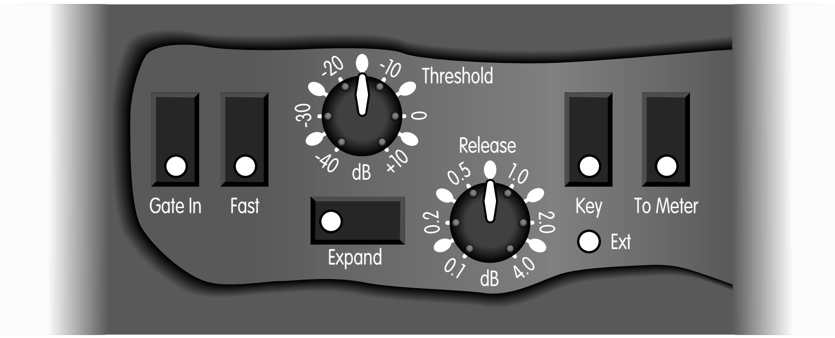
Use them to colour the sound and create a presence. For example, you can take the sound of a vocal and improve its clarity in the mix. To isolate the frequency you want to boost or cut:

1. Add some gain to the signal, so that you can hear the effect easily.
2. Ensure that Fine is not switched in, so that it is easier to hear the area you are affecting.
3. Modify the frequency until you find the area you want to work on.
4. Modify the gain to control the amount of the selected frequency that is added to or subtracted from the signal.
5. If you want to affect a narrower band of frequencies (for example, to give a tight bass sound), use the Fine switch.

Gate/Expander

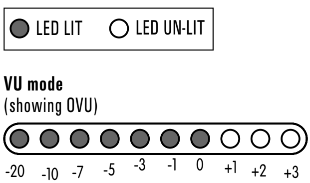
A gate and an expander both serve to keep out unwanted noise. A gate does not allow any signal through until the volume of the signal passes a certain threshold, while an expander compresses the volume of any signal below a certain threshold. (If you are not familiar with compression, see the Compressor section.) Which you should use depends on the characteristic of what you are recording, since a gate is essentially on or off, whereas the expander comes in and out more gradually.

The equaliser and compressor sections of the unit do not affect the gating/expansion applied to the signal, since the original input signal is passed to the gate/expander. (The only exception to this is the filters section, which can affect the gate/expander – see the section Using the Low and High Filters with the Gate/Expander.)



Controls

- The Gate In button switches the gate/expander in or out.
- The To Meter button modifies what the meter displays, and how it displays it.
- When the switch is out, the meter acts as a VU meter displaying the level of the input signal. It displays the signal in bar mode (that is, the meter lights all the LEDs up to the signal level, thus creating a bar effect).



Key Button

When the Key button is out, the unmodified input audio signal to the unit is used to control the gate/expander. When the Key button is in, a different signal is used to control the gate/expander – which signal is used depends on the setting of the Ext Key switch on the back of the unit:

Ext Key switch	Control signal
In	The signal entering through the Key socket on the rear of the unit
Out	The input audio signal to the unit, passed through the filters.

Note that the gating/expansion is actually applied to the unmodified input audio signal – the signals above are only used to control how the gate/expander works.

Using the Low and High Filters with the Gate/Expander

If you are not inputting a control signal via the Key socket on the back of the unit, then pressing the Key button passes the audio signal through the Low and High filters before it reaches the gate/expander. This can help you to set the gate to reduce spill on a microphone. For example, when recording a cymbal on a drum kit, there will be spill from the other parts of the kit. Since a cymbal contains little or no bass, you can use the Low filter to remove all the bass frequencies from the signal that controls the gate – therefore, if the filters are set correctly, the bass frequencies never reach the gate, so it does not open until the cymbal is hit.

Note that using the Low filter in this way is different from using it to remove the bass information from the audio signal, as all frequencies are passed when the gate is open:

- Modifications made to the audio signal in the equaliser section are not passed to the gate/expander, which works on the unmodified signal unless you use the Key button.
- Bass information (such as overtones) is still present in the audio signal that passes through the gate/expander.

In some situations, you may also want to switch the filters into the audio path. For example, if the cymbal microphone is also picking up spill from the bass guitar, switching the filters into the audio path would cause the Low filter to remove this.

Compressor

A compressor acts like an automatic volume control, turning down the volume of a signal if it gets too loud. It reduces changes in volume, automatically reducing the gain when the signal gets louder than a certain threshold, so that the dynamic range of a compressed signal is lower than the dynamic range of the input signal. To understand a compressor, you must understand dynamic range – if you do not, you should read the following section about dynamic range.

Note that compression tends to even out a performance. Example; it can stop the vocal getting very loud or very quiet in the mix.

For reasons of quality, the circuitry that determines the compression on the signal is in a separate side chain to the gate expander. This means that the equaliser and gate/expander sections of the unit do not affect the compression applied to the signal, since it is the original input signal that is passed to the compressor. (The only exception to this is the filters section, which can affect the compressor – see the section Using the Low and High Filters with the Compressor.)

Dynamic Range

The dynamic range of a signal is the difference in volume between the quietest and loudest parts: for music, the dynamic range can be as wide as 120dB.

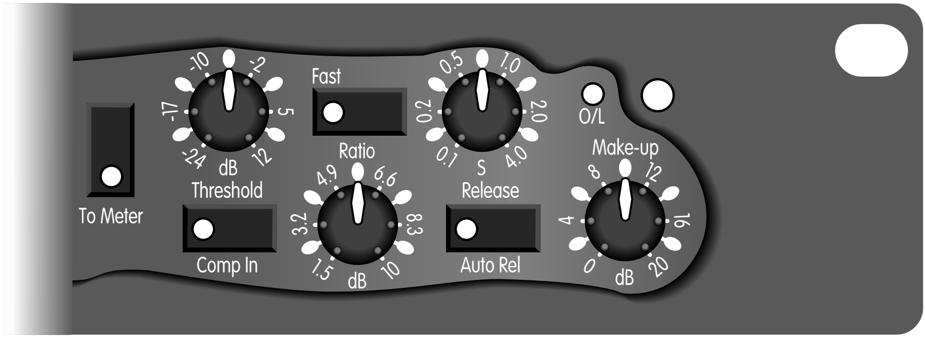
Signals with wide dynamic range demand greater attention from the listener, and require listening conditions with low background noise. Consequently, in areas with high background noise, such as a restaurant, it is hard to listen to signals with a wide dynamic range – only the loud parts are heard, with the quiet parts being lost in the ambient noise. Compressing the signal reduces the dynamic range and so makes it easier to hear in such situations.

Similarly, the dynamic range of the signal can exceed that of the medium used to carry it:

- 16-bit digital recordings (such as DAT) have a theoretical maximum dynamic range of 96dB. It is essential that you do not exceed this limit.
- Analogue tape has a dynamic range in the order of 60dB (though noise reduction can add between 15 and 30dB). It is not always necessary to limit dynamic range when recording onto analogue tape, as the tape saturates naturally when recording loud signals, which in some cases can be a desired effect.
- FM radio has a dynamic range of 40 to 50dB.
- AM radio has a dynamic range of 20 to 30dB.

In all of these cases, you can use a compressor to restrict the dynamic range of the signal to that of the medium.

Controls

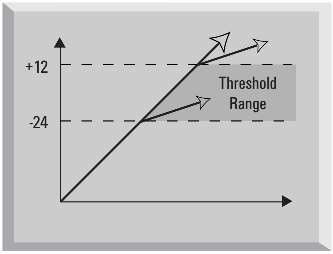


- The Comp In button switches the compressor in or out.
- The To Meter button modifies what the meter displays, and how it displays it. When the switch is out, the meter acts as a VU meter displaying the level of the input signal. It displays the signal in bar mode (that is, the meter lights all the LEDs up to the signal level, thus creating a bar effect).
- When the switch is in, the yellow LEDs in the meter act as a peak meter, displaying the amount of compression in dot mode (that is, the meter lights only the LED for the signal level). Since compression reduces the volume of the signal, the meter drops as compression is applied: for example, a 3dB drop shows as -3 VU on the meter. The change to peak action means that it is easy to see when transients occur (which will be compressed more heavily).
- Note that the To Meter button also causes the meter to show the amount of gating or expansion applied to the signal, using the red LEDs. For more details, see the Gate/Expander section.
- The threshold and ratio controls set the amount of compression applied to the signal (see overleaf).
- The Fast button and release control set the duration of the compression (see Attack and Release, below).
- The Auto Rel button sets an automatic release time for the compressor (see overleaf).
- The Make-Up gain sets the output volume of the compressed signal (see below).
- The To S/C button (located with the filter controls) can assign the filters to the compressor (see the section Using the High and Low Filters with the Compressor, further on).

Threshold

The threshold determines when the compressor starts to compress the signal.

By setting a threshold, you do not compress all of the input signal – instead, you compress the signal only when it is louder than the threshold, as shown in the following diagram:

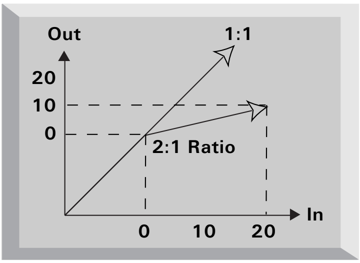


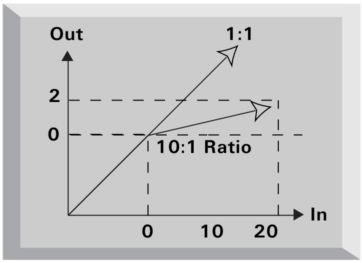
Setting the Threshold

By setting a threshold, you determine that quieter passages maintain their natural dynamic range, and only loud passages (that go above the threshold) are compressed.

Ratio

The ratio control determines how much compression is applied to the signal. The ratio (such as 2:1) refers to the ratio of change in input level to the change in output level. So, a ratio of 2:1 means that for every 2dB increase in the input level above the threshold, the output level changes by 1dB, as shown in the following diagram:





Setting the Ratio

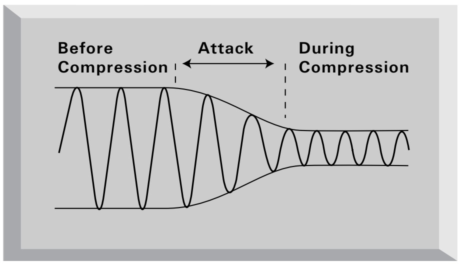
As you increase the ratio, the sound becomes tighter and the effect of the compression becomes more noticeable. A lower ratio has a softer slope, which preserves more of the original dynamic range, since an increase in input level still results in a significant increase in output level.

Attack and Release

Attack and release determine how quickly the compressor switches on and off at the threshold. With instant attack, full compression would be applied to the signal as soon as it got louder than the threshold. Similarly, with instant release, the compressor would switch off as soon as the signal got quieter than the threshold. Varying the attack and release rates can help give more natural-sounding compression, since the optimum rates vary with the instrument being recorded, and with the performance. For example, when recording a snare drum, a fast attack and release are needed – a slow release over-compresses the signal, with beats after the first dulled slightly because the compressor is still on.

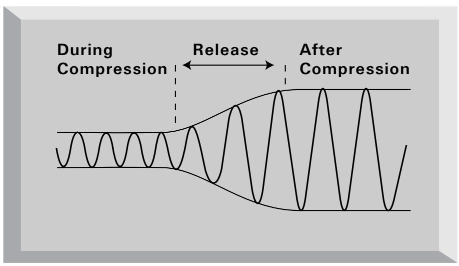
Setting the Attack Rate

Using the unit's normal attack rate, the compressor gradually comes to full compression, instead of compressing immediately. Transient response is less affected, so maintaining the presence of each note. Using the unit's fast attack rate, the compressor quickly comes to full compression, so keeping transients under control.



Attack does not need to be fast when recording onto analogue tape. The fastest transients are lost by saturation of the tape and become inaudible, and longer duration peaks can be controlled by the compressor, giving a more natural sound. When recording onto a digital medium such as DAT or hard disk, you may need it to use the fast attack rate so that transients do not overload the medium (since this can lead to undesirable digital distortion).

Setting the Release Rate

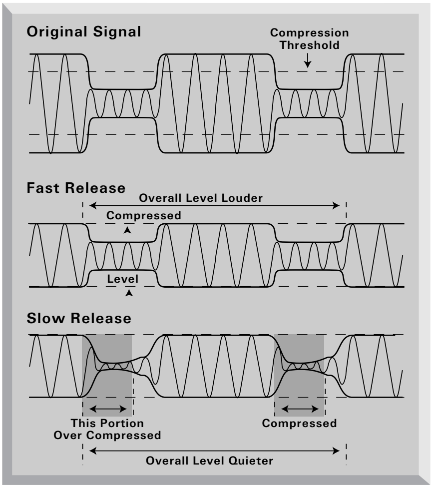


By slowing the release rate, the compressor recovers more slowly from compression, so it does not turn off immediately when the signal returns below the threshold.

The release rate is probably the most important variable when recording rock music, since it controls loudness. Loudness is determined by the maintenance of high mean levels: compression increases the proportion of high-level signal content, and as the diagram shows, the faster the unit releases, the more low-level signal is brought to a higher level. Therefore, the faster the release rate, the higher the perceived loudness of the recording.

Auto Rel

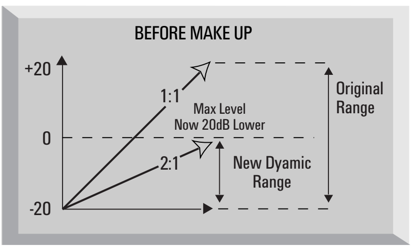
Auto Rel modifies how quickly the compressor releases after compression (that is, how quickly it stops compressing when the signal returns below the threshold). It reacts to the dynamic range of the input, so the higher the signal is above the threshold, and the longer it stays there, then the slower the release is. This means that fast signals that aren't compressed hard have a fast release time, while longer signals release more slowly, which makes the compression in context with the signal.

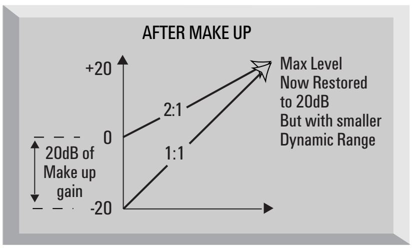


Make-Up Gain

Compressing a signal makes it quieter. After you have set the compression on the signal, use the make-up gain to restore the signal to its original or desired volume.

For example, in the diagram the compressor reduces the signal by 20dB, which reduces the dynamic range of the input and in so doing makes the signal quieter. Using the make-up gain restores the volume of the compressed signal.





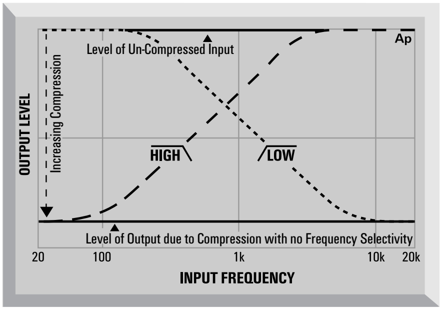
Using the Low and High Filters with the Compressor

If you use the To S/C switch in the equaliser section, the filters determine the frequencies that are sent to the compressor's side chain, and so affect how the signal is compressed. The low filter prevents low frequencies entering the side chain, and so stops the compressor reacting to those frequencies. Similarly, the high filter prevents high frequencies entering the side chain.

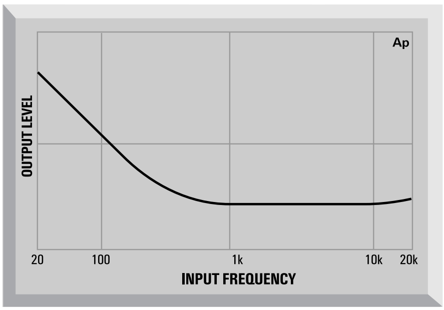
For example, one of the problems in compressing a mixed programme is that gain reduction tends to be controlled by one dominant instrument or sound.

For more natural compression, you need to attenuate (reduce the volume of) the sound of the dominant instrument, but this is probably not acceptable since it would affect the mix. Therefore, you can use the filters to remove the frequencies affected by the dominant instrument, so the signal is compressed as if the dominant instrument had been attenuated. When using the filters in this way, use the Filters switch to switch the filters into the audio signal, so that you can hear what you are doing, then when you have the compression you desire, switch them out of the audio signal again.

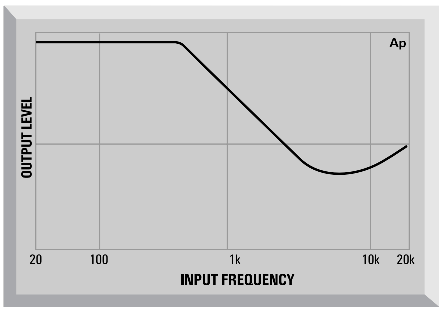
This technique can be very useful when compressing bass-heavy dance music. By attenuating the bass using the low filter, the bass in the original retains more of its dynamics.



Filtered Compression



Bass Dance Mix

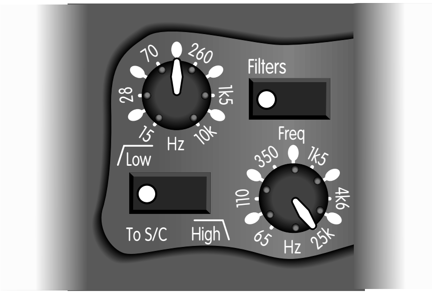


De-Essing

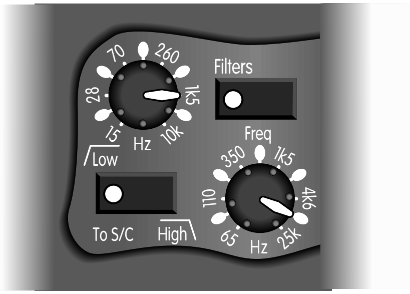
Another application of this technique is in de-essing. When de-essing, you want sibilants in a voice to be heavily compressed. To achieve this, set a high level of compression, but use the filters to remove frequencies that are not affected by sibilance (as a starting point, set the low filter at 2kHz and the high filter at 10kHz, and adjust accordingly). This causes only the sibilants to be heavily compressed.

Note that, because of the range of the filters, it is possible to set them so that there is no compression at all. Since only the frequencies between the two filter settings are compressed, be careful not to set the frequency of the high filter below that of the low filter: if you do, no frequencies reach the compressor side chain, so no compression is applied to the signal.

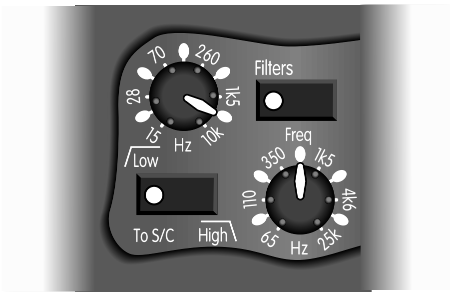
BASS DANCE MIX



DE-ESSING



NO COMPRESSION



When to use Equalisation

You can apply equalisation when recording or while mixing down a finished track. You usually record flat (without equalisation) and then apply equalisation at mixdown, unless you are assigning several instruments to a single track, in which case you need to get the equalisation correct during recording.

Use equalisation to:

- Remove unwanted noise, such as rumble, bass lift and hum (by using the Low filter).
- Reduce hiss (by using the High filter).
- Replace missing bass or treble (by using the Low or High Eq).
- Reduce excessive bass or treble (by using the Low or High Eq).
- Boost room ambience (by using the High Eq).
- Improve tone quality (by using all the controls).
- Help a track stand out in the mix (by using the Low Mid and High Mid Eq).
- Reduce noise and leakage (by using the Low and High filters). Note that you can also use gating or expansion to do this.
- Boost lows and highs when recording loud bands (by using the Low and High Eq).

How to use Equalisation

1. If recording using a microphone, ensure that the microphone placement is correct. Listen to the sound from the microphones with no equalisation applied, and modify the microphone placement until you get the sound you want.
2. Set the operating level.
3. Consider what you don't want (for example, you don't want too much bass on an analogue tape machine). This varies with the recording format and varies with the input signal. If necessary, use the low and high filters to remove parts of the signal.
4. Listen to the ambience and room sound that comes back off tape, and check that it has the frequency response that you are looking for (for example, the room may be a bit dull in the top end, or there may be too much bass). If the frequency response is not correct, use the Low and High Eq to correct it.
5. Create a sound and bring out the character of the instruments by using the Low Mid and High Mid Eq. The section on Bringing an Instrument Forward in the Mix has a table showing important frequencies for different instruments.
6. Set the final output level using the output fader.

Bringing an Instrument Forward in the Mix

You can use equalisation to bring an instrument forward in the mix (that is, to make it easier to hear when mixed with the other instruments). However, when doing this, beware of using equalisation simply to make the instrument louder by boosting the fundamental frequency (the musical note).

An instrument's sound is made up of a fundamental frequency and harmonics, even when playing a single note, and it is the harmonics that give the note its unique character. If you use the equaliser to boost the fundamental frequency, you simply make the instrument louder, and don't bring it out of the mix. Boosting the harmonic frequencies, on the other hand, boosts the instrument's tone quality, and so makes it stand out in the mix. The table shows useful frequencies for a number of common instruments.

Instrument	Tone Quality	Useful Frequencies
Voice	Presence Sibilance Boominess Fullness	5kHz 7.5 - 10kHz 200 - 240Hz 120Hz
Electric guitar	Fullness Bite	240Hz 2.5kHz
Bass guitar	Attack or pluck Bottom String noise	700 - 1000Hz 60 - 80Hz 2.5kHz
Bass drum	Slap Bottom	2.5kHz 60 - 80Hz
Snare drum	Fatness Crispness	240Hz 5kHz
Hi hat and cymbals	Shimmer Clank or bell	7.5 - 10kHz 200Hz
Tom toms	Attack Fullness	5kHz 240Hz
Floor toms	Attack Fullness	5kHz 80 - 120Hz

When to use Compression

Compress hard:

- To stop dropping in and out (particularly vocals – compress quite hard so they sit above the mix).
- When recording bass (for example) – there is a lot of bass energy that can easily get out of control.
- When recording snare.
- Anything you want to maintain a continuous presence in the mix.

Compress more softly:

- When the attack is an important characteristic of the sound.

How to use Compression

1. Using an input signal with wide dynamic range, set the controls in the following starting positions:
   - Ratio in the middle (around 5:1).
   - Threshold at maximum (+12).
   - Auto Rel off.
   - Make-up gain at minimum.
   - Switch the meter to show gain change, so that you can monitor the effect of compression.
2. Reduce the threshold, and monitor the effect this has on the sound:
   - Listen to the reduction in the dynamic range.
   - Watch the meter to see the amount of gain reduction.

   As you reduce the threshold, increase the make-up gain to restore the level of the signal.
3. When you are close to the dynamic range you want, you need to adjust all the controls to achieve the quality of sound that you are looking for:
   - Adjust the ratio to compress softer or harder.
   - Adjust the threshold to compress sooner or later.

   The combination of ratio and threshold determines the maximum level of the loudest sections.
   - Adjust the release to even out compression between the loud and quiet sections. If the compression sounds unnatural to you, try using Auto Rel.

The release determines the level of the quietest signal. The combination of ratio, threshold and release determines the overall dynamic range.

- If transients are not being compressed quickly enough, use the Fast switch to give a faster attack.

4. Adjust the make-up gain to set the level of the output signal.

When to use Gating or Expansion

Use the gate or expander on a signal when there is a certain part of the signal that you want to cut out, while letting the rest through. For example, when recording a band live, microphones may pick up spill from other instruments, so you can use the gate or expander to get rid of the spill.

Whether you use the gate or the expander depends on the type of material you are recording. For example, you normally use the gate when recording drums, but when recording vocals you normally use the expander, since you might hear the gate cutting off the beginning and end of phrases as it opens and closes.

How to use Gating or Expansion

1. Set the release in the middle, and the threshold on minimum.
2. Select the Fast button when gating if it is important that the gate opens quickly (for example, when recording a snare drum). When using the expander, the Fast button is less noticeable.
3. Set the threshold so that the signal causes the gate/expander to open successfully. If the end of the signal is cut off by the gate/expander closing, lengthen the release time.
4. If necessary adjust the threshold again (particularly if you have made the release faster).

Non-Operation

If the unit does not appear to be operating correctly, perform these simple checks before assuming that it needs repair.

None of the LEDs light

Check the mains supply:

1. Is the module connected to the mains supply?
2. Is the socket switched off?
3. Is the voltage select switch on the back of the unit in the correct position?
4. If the supply is okay and the module turned on but no LEDs light, then a fuse has probably blown. See the section on changing a fuse.

LEDs light, but there is no audio out

Check that the unit is set as you expect:

1. Are you connected to the correct input on the back?
2. If using the Mic input, is the Mic rotary control clicked into the LINE position?
3. If using the Line or Inst input, is the Mic rotary control not clicked into the LINE position?
4. If using a microphone, does it require phantom power (+48)?
5. Is the Output Fader fully down?
6. Is the Filters button lit? If so, are the filters set correctly? (If the frequency of the Low filter is set above the frequency of the High filter, no audio is allowed through.)
7. If using the gate, is the threshold set too high, causing the gate to be permanently closed?
8. If the Key switch is illuminated, are the filters set correctly (as above)?
9. Is the Ext Key light illuminated? If so, is there a signal connected to the Ext socket? (If no signal is connected, the gate remains permanently shut.)

LEDs light, but the compressor does not appear to work

Check that the unit is set as you expect:

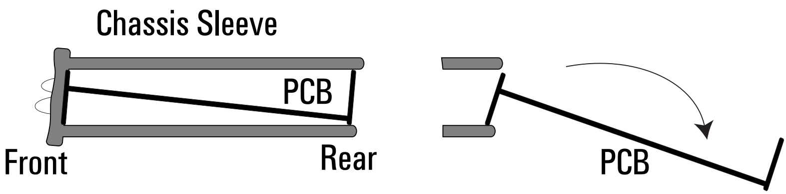
1. Is the To S/C button lit? If so, are the filters set correctly? (If the frequency of the Low filter is set above the frequency of the High filter, no audio is allowed through to the compressor side chain.)

Changing a Fuse

We strongly recommend that you do NOT attempt to change fuses unless you are absolutely certain that you know exactly what you are doing. If you are in any doubt whatsoever, contact your dealer or the factory before you open the module.

To change a fuse, if you are certain of your technical ability:

1. Disconnect the mains cable.
2. Viewing the module from the back, remove the screws that secure the back panel (there are two at each end).
3. Carefully slide out the inside of the unit (see diagram).



4. The fuse is in a holder close to the transformer. To remove the fuse, pull off the top of the fuse holder, which holds the fuse.
5. Replace the fuse with a 250 mA anti-surge type.
6. When you have replaced the fuse, slide the inside of the unit back into the outer cover.
7. Replace the screws in the back panel.

Technical Specifications

Preamplifier

	Gain	Impedance	Frequency Response	EIN
Line	0dB	10kΩ live to return	<10Hz to >200kHz +0/−1dB	−94dBu
Instrument	+18dB	1MΩ tip to shield	<10Hz to 90kHz +0/−1dB >165kHz +0/-3dB	−107dBu
Microphone	+10dB to +60dB	1.2kΩ live to return	<10Hz to 140kHz @+10dB gain, 50kHz @ +60dB gain +0/−3dB	−128dBu (+60dB gain)

THD dynamics out:	0.001% @ 20Hz to 20kHz
THD dynamics in:	0.006% @ 1kHz 0.01% @ 10kHz (distortion is predominantly second harmonic)
Clip level:	+26dBu (output fader fully clockwise).
Overload indication:	+20dBu (output fader fully clockwise).

Filters

Low:	15Hz – 10kHz 12dB/octave high pass.
High:	65Hz – 25kHz 12dB/octave low pass.

Equaliser

Low Eq:	Gain +/–18dB. Frequency 30Hz - 480Hz low shelving.
Low Mid Eq:	Gain +/–18dB. Frequency 40Hz - 1.2kHz parametric. Q @ +/–18dB gain: normal 0.7 fine 2.5.
High Mid Eq:	Gain +/–18dB. Frequency 600Hz - 18kHz parametric. Q @ +/–18dB gain: normal 0.5 fine 1.1.
High Eq:	Gain +/–18dB. Frequency 2.5kHz - 18kHz high shelving.

Gate

Threshold:	–40dBu - +10dBu.
Attack time:	6mS (Fast 70 µS).
Release:	100mS - 4S.
Expander ratio:	2:1.

Compressor

Threshold:	–24dB - +12dB.
Ratio:	1.5:1 - 10:1.
Attack:	8mS (Fast 1.5mS).
Release:	100mS - 4 S. Auto-Release is program dependent.
Make-up gain:	0dB - +20dB.

Signal Flow Diagram of Green 5 Channel Strip Overleaf

Signal Flow Diagram of Green 5 Channel Strip