Focusrite

GREEN RANGE

VOICEBOX

manual

VOICEBOX

Voicebox

The Voicebox is a combined microphone preamplifier, dynamics unit and equaliser. This makes it an excellent recording tool, since you no longer need to use a mixing desk when recording vocals: instead, you need just a microphone, recorder (such as a DAT machine) and a Voicebox to record direct to tape, monitoring using the return from the recorder. This ensures the highest quality signal onto tape, 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 Voicebox:

Microphone preamplifier

Expander

Compressor

De-esser

Equaliser

When you are getting to know the unit, use it on a track that you are familiar with (for example, run a favourite CD through the unit, by connecting the CD player's headphone output into the unit). With only one part of the unit activated, try all of the controls for that part in turn, to hear how they affect the sound. 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 and de-esser. 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 Voicebox.

Power Connections

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.

Power Supply

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

Most modules will operate on a range of voltages, and have 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.

Signal Connections

All the signal connections are via connectors mounted on the rear panel. Standard XLR connectors are used for all mic and line level signals, and are wired to the AES standards, which are:

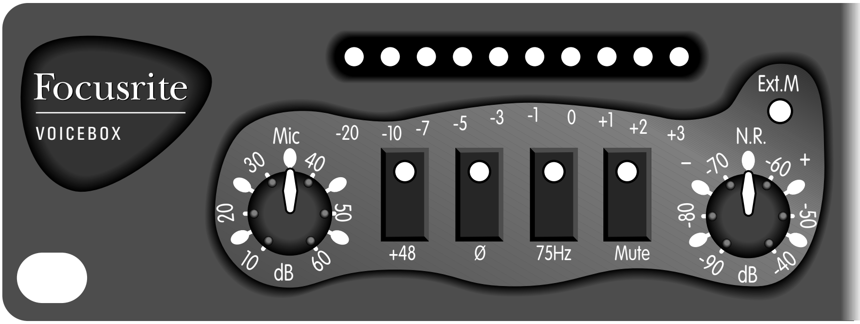
Pin 1	Screen	chassis
Pin 2	Live	audio 0°
Pin 3	Return	audio 180°

For all inputs and outputs, the screen (pin 1 of the XLR) is connected to the chassis earth point.

In the Green range, jack inputs are used only for external switching applications or as unbalanced guitar inputs on some units.

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.

Microphone Preamplifier



Controls

The gain control provides continously-variable gain from +10 dB to +60 dB.

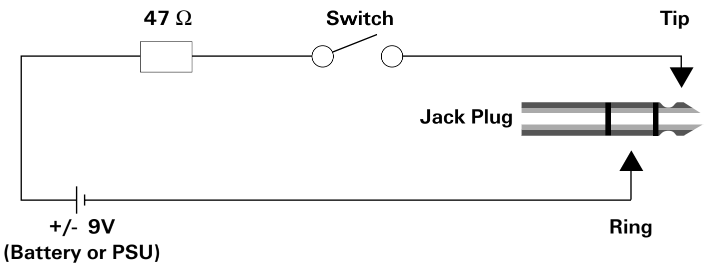
+48 is the phantom power button. When lit, it provides phantom power to the microphone connected to the channel.

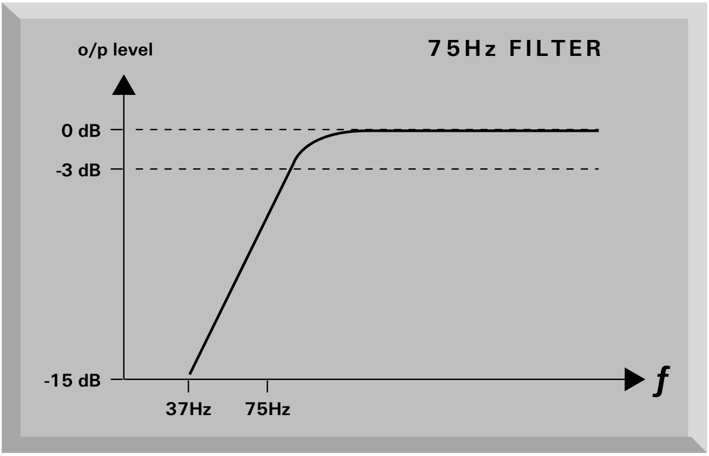
Ø is the phase button, and reverses the phase of the channel when lit.

75 Hz is a filter that removes very low frequencies from the signal. Use it to remove rumble or bass lift (a proximity effect of microphones, giving a bass boost as the singer gets closer to the microphone).–See diagram opposite.

The **Mute** switch mutes the channel (that is, it reduces the volume to zero). This can be useful during a live performance (for example, between songs). You can mute the channel manually, using the switch on the front panel, or you can mute it remotely by connecting a suitable switch to the external jack socket on the rear panel. The diagram below shows the circuit for such a switch.

When you trigger the external switch, the LED on the front panel lights. Note that the muting circuit is optically isolated, to prevent interference from external signals.





Setting the Gain

Use the meter and gain 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 gain control to modify the gain, and set the gain 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 20 dB of usable headroom).

Setting the Phase

When recording a single source using more than one microphone, it is possible for the signals from the microphones to be out of phase, which affects the quality of the recording since signals that are out of phase tend to sound “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. Set the monitoring to mono.

3. Use the phase switch to reverse the phase on one of the signals. When the two signals are in phase, the signal sounds bigger.

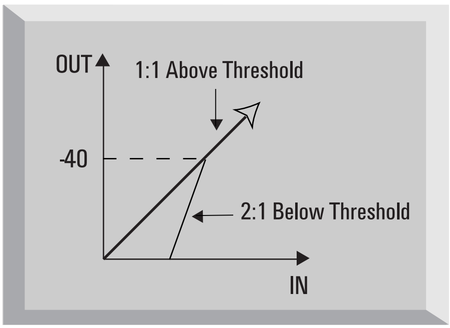
NR (Noise Reducing Expander)

NR controls a downward expander, which reduces noise. Although the unit is very quiet, it is possible to increase the noise floor when using the EQ and compressor together (by using the make-up gain, and by boosting high frequencies). The expander compresses all signals that fall below its threshold, and so improves the signal-to-noise ratio during quiet passages. (If you are not familiar with compression, see the Compressor section.)

The expander does not affect musical quality. However, if you overuse it, you may notice its effect on some passages where the signal dies away unnaturally quickly - in this case, reduce the amount of expansion by turning the control clockwise.

The numbers on the control represent the worst case noise buildup during quiet passages (for example, with the input to register around 0VU on the meter, then -40 on the expander control represents 40 dB below that). Increasing the expansion, by turning the control counter-clockwise, reduces this worst case level.

Off position: Set the control fully clockwise, at -40.



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.

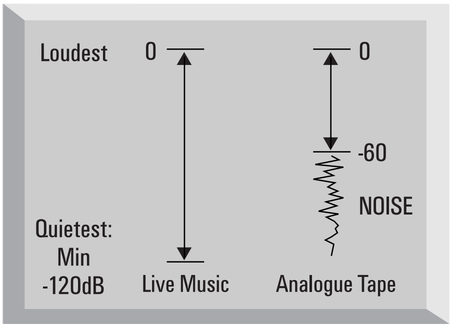
Note that compression tends to even out a performance since it stops the vocal getting very loud or very quiet in the mix.

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 120 dB.

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 96 dB. It is essential that you do not exceed this limit.

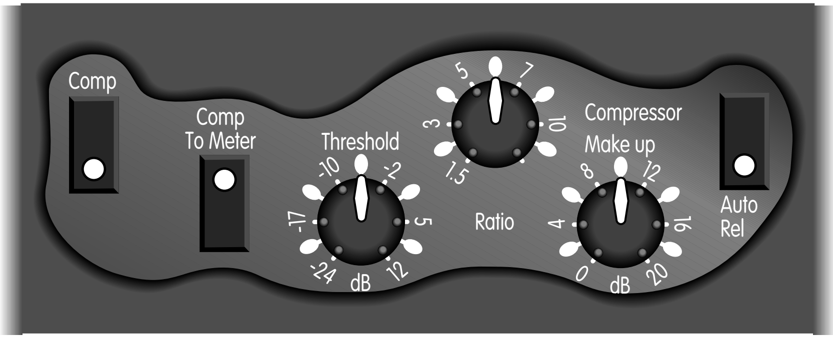
Analogue tape has a dynamic range in the order of 60 dB (though noise reduction can add between 15 and 30 dB). It is not always necessary to limit dynamic range when recording onto analogue tape, as the tape saturates naturally when recording loud signals, which in some cases can be a desired effect

FM radio has a dynamic range of 40 to 50 dB

AM radio has a dynamic range of 20 to 30 dB

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

The Controls



The Comp button switches the compressor in or out.

The Comp 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 meter acts as a peak meter displaying the amount of compression. It displays the 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 3 dB 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).

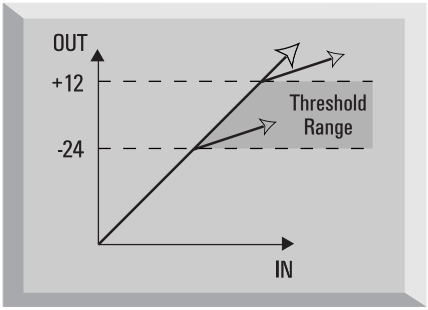
The Ratio and Threshold controls set the amount of compression applied to the signal (see below).

The Make-up gain sets the output volume of the compressed signal (see below).

The Auto Rel button sets an automatic release time for the compressor (see below).

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:

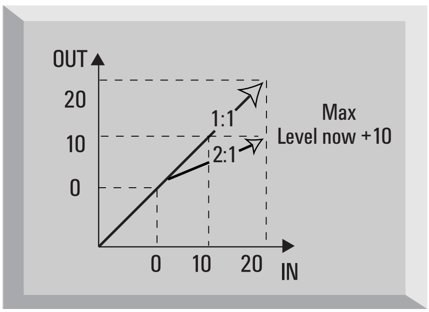


Setting the Threshold

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 2 dB change in the input level, the output level changes by 1 dB, as shown in the following diagram:



Setting the Ratio

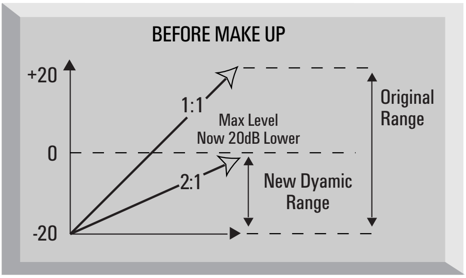
As you increase the ratio, the sound becomes tighter and the effect of the compression becomes more noticeable. A lower ratio has a softer slope, which preserves more of the original dynamic range, since an increase in input level still results in a significant increase in output level.

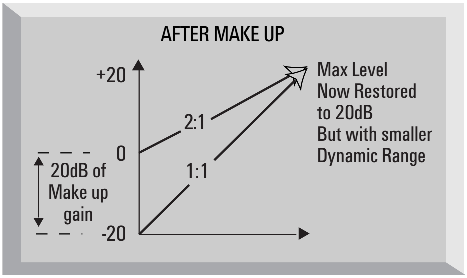
Auto Rel

Auto Rel modifies how quickly the compressor releases after compression (that is, how quickly it stops compressing when the signal returns below the threshold). The release on the Voicebox has been optimised for vocal use, which under certain circumstances may not sound natural. Auto Rel 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 20 dB, 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.

If you make heavy use of make-up gain, you can introduce hiss into the signal - if so, counteract it by using the NR control (as described earlier).

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

NR off (-40)

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.

If the compression sounds unnatural to you, try using Auto Rel

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

5. If you can hear noise buildup during quiet passages, adjust the NR control to increase the amount of expansion until the noise is reduced sufficiently

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

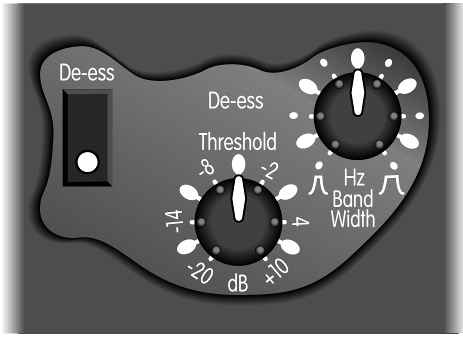
Compress more softly:

When the attack is an important characteristic of the sound

De-esser

The de-esser lets you remove excessive sibilance from a vocal performance, by selecting the frequency range that contains the sibilants and heavily compressing that frequency range.

Controls



The **De-ess button** switches the de-esser in or out.

The **Threshold control** sets the threshold in the same way as for the compressor (for more details see the Compressor section).

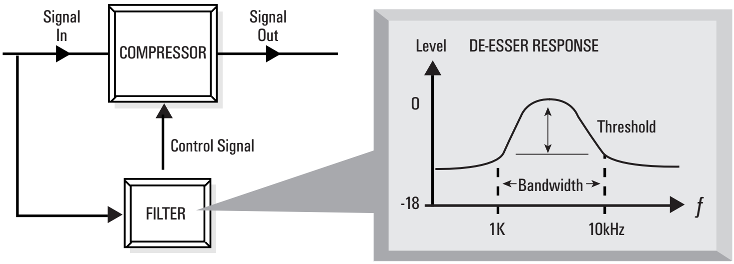
The **Bandwidth control** determines the frequency range that is compressed.

How to Use the De-esser

1. Set the Threshold to minimum (-20)

2. With a signal going through the Voicebox, adjust the Bandwidth control until the sibilants are heavily affected

3. Increase the Threshold until the sibilants are compressed in a natural-sounding way



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 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 two types of equalisation on the Voicebox:

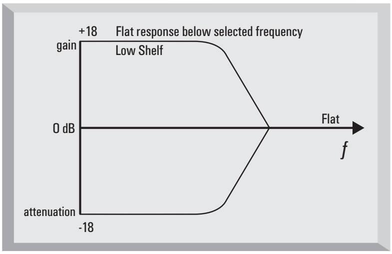
Low and High Eq

Mid Eq

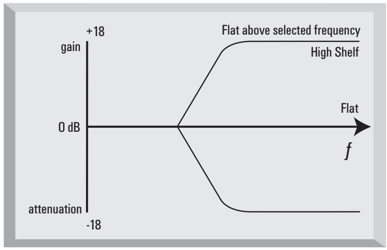
Low and High Eq

The Low and High Eq affect all frequencies below or above a given frequency, letting you attenuate reduce the volume of, or add gain to the signal. 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



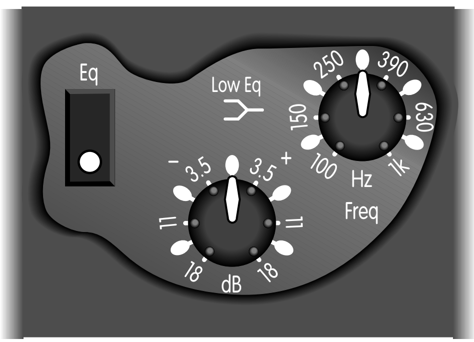
HIGH SHELF

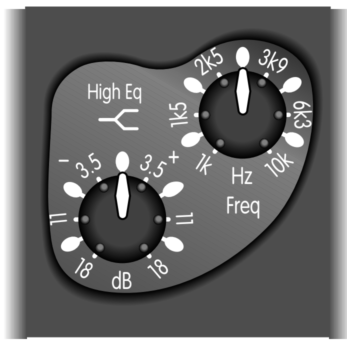


Controls

You can set the frequency and the amount of attenuation or gain applied beyond that frequency.

Off position: To turn off a shelving filter, set the gain control in the middle 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)

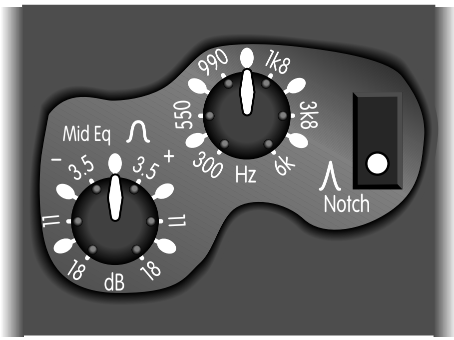
Use the Low Eq to boost subsonic information, thus giving a bassy sound, or to attenuate a sound that is too bassy.

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

Mid Eq

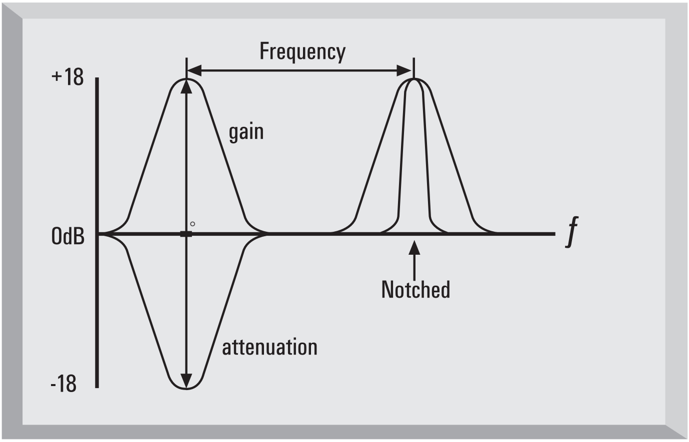
The Mid Eq gives you control over the vocal frequency spectrum. Unlike the High and Low Eq, the Mid Eq affects only a band of frequencies around a selected frequency. The width of the band of frequencies affected is determined by the Notch control.

Controls



The Mid Eq lets you set every parameter: the gain or attenuation, the frequency affected, and bandwidth. (see following diagram)

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



Using the Mid Eq

Use the Mid Eq 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. You can also use the Mid Eq with the Notch switch to notch out a frequency. For example, if the sibilants in the vocal sound are very pronounced, you may want to notch them out instead of making heavy use of the De-esser, to give a more natural sound.

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 Notch 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 only a very narrow band of frequencies, use the Notch switch - if you use the Notch switch, it is likely that you will have to fine-tune the frequency

How to Use Equalisation

1. 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 high-pass filter 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 shelving filters to correct it

5. Create a sound and bring out the character of the voice by using the Eq. The table below shows the important frequency ranges for the voice:

Presence	5 kHz
Sibilance	7.5 - 10 kHz
Boominess	200 - 240 Hz
Fullness	120 Hz

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 Eq)

Reduce hiss (by using the High Eq)

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)

How to use the Voicebox

1. Set the input signal using the meter and mic pre level control.

2. Compress the signal to the required dynamic range for the recording format.

3. Use the Make-up gain to compensate for the reduction in level.

4. Remove sibilants by using the De-esser, or notch filter, or both.

5. Improve the quality of the tone by using the EQ.

6. Set the output level to match the input level of the next item in the chain (usually a tape machine).

7. If there is noise buildup (because you have made heavy use of the compressor and EQ) use the NR control to remove it.

Stereo Linking

Two Voiceboxes can be liked together for stereo operation of the compressor and de-esser functions.

Stereo linking requires the connections of a stereo 1/4″ jack between the two units, inserted in the rear panel socket marked 'link'.

Non-Operation

If none of the LEDs light, check the mains supply:

1. If 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.

Changing a Fuse

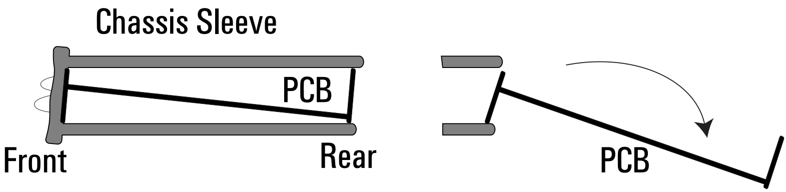
We strongly recommend that you do NOT attempt to change fuses unless you are absolutely certain that you know exactly what you are doing. If you are in any doubt whatsoever, contact your dealer or the factory before you open the module.

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 four screws that secure the back panel (there are two at each end).

3. Carefully slide out the inside of the unit with a downward motion. (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 four screws in the back panel.

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Ireland	CTI Control Techniques Ireland	Jim Dunne	Fumbally Court, Fumbally Lane, Dublin 8	00 3531 454 5400	00 3531 454 5726
Israel	Sonotronics Electronic EquipmentLtd	Sonny Shmueli	No.2 Y. BIN. NUN St. BNEI-BRAK 51261 Israel	00 972 3 5705223	00 972 3 6199297
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Korea	Best Logic Sound Co	Hi Ahn	RM #320, Nakwon Bldg., 284–6 Nakwon-Dong, Chongno-Ku, Seoul, Korea	00 82 2 741 7385 /7386	00 82 2 741 7387
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