C2 Dual Compressor

Operators Manual

C2
Dual Compressor
Digital

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C2 Quick Reference

THINGS YOU NEED TO KNOW…

✓ ***BOLD*** MEANS HOLD!

Any functions that would produce possible unexpected level changes at the outputs are protected by a 'press and hold' action and printed in ***BOLD*** on the panel. These functions are:
Switching ***AUTO***matic time constants on and off;
Changing ***MODE*** between compressor and De-EQ;
Enabling the sidechain ***LISTEN*** function;
Ganging the two channels together in ***STEREO*** mode.

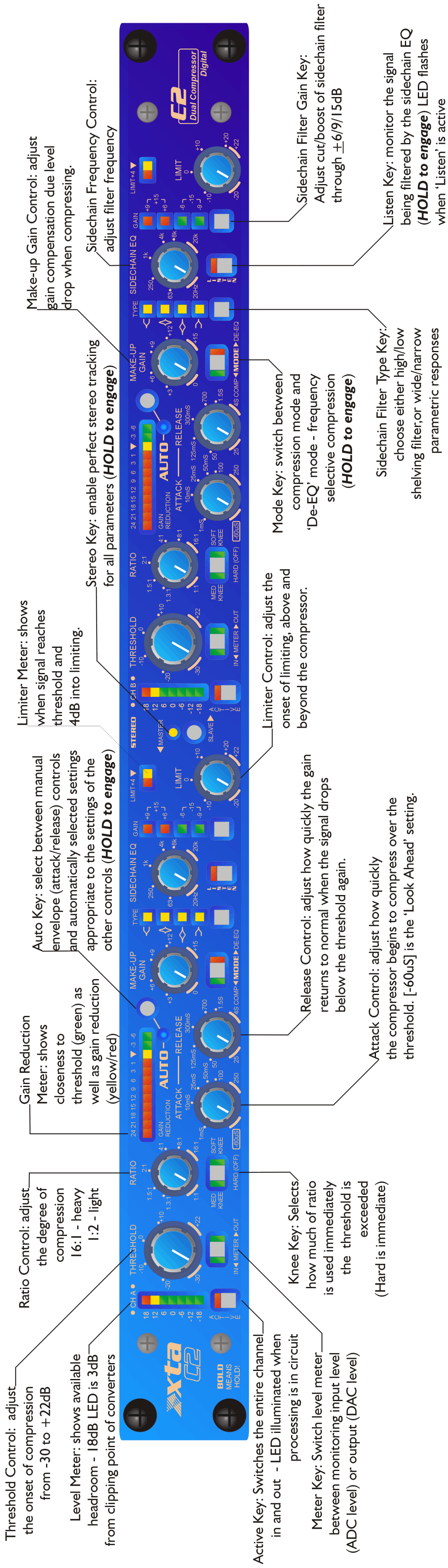
- ✓ The meters show level, in dB, from the clipping point of the unit – when the METER switch is set to 'IN' the meters show level, in dB, from clipping the input converter. Set to 'OUT' they show headroom available at the output – the DAC.

- ✓ When in De-EQ mode, the sidechain filter GAIN control will not function, as it is dynamically adjusted according to the threshold.

- ✓ The De-EQ mode LED always flashes when this mode is enabled.

- ✓ The Listen LED always flashes when Listen is enabled. Listen is automatically turned off if the unit is switched off and on

- ✓ When a channel is bypassed, all its metering and status LEDs will dim. All metering will continue to function.



Contents

An example of this equipment has been tested and found to comply with the following European and international Standards for Electromagnetic Compatibility and Electrical Safety:

Radiated Emissions (EU):	EN55013-1 (1996)
RF Immunity (EU):	EN55103-2 (1996) RF Immunity, ESD, Burst Transient, Surge, Dips &Dwells
Electrical Safety (EU):	EN60065 (1993)

Important Safety Information

Do not remove Covers.
No user serviceable parts inside, refer servicing to qualified service personnel.
This equipment must be earthed.

CAUTION
RISK OF ELECTRIC SHOCK
DO NOT OPEN
DO NOT EXPOSE TO RAIN OR MOISTURE

ATTENTION
RISQUE DE CHOC ELECTRIQUE
NE PAS ENLEVER
NE PAS EXPOSER A LA PLUIE NI A L'HUMITE

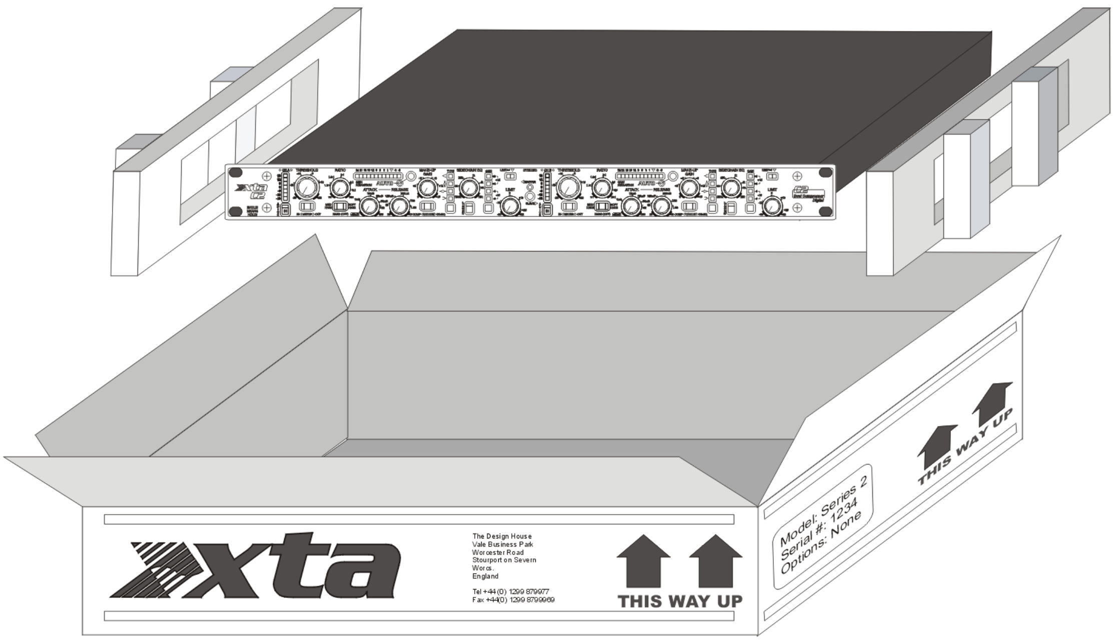
It should not be necessary to remove any protective earth or signal cable shield connections.
Do not defeat the purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wider blade and the third prong are provided for your safety. When the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
Only use this equipment with an appropriate mains cord.
In the USA the cord should comply with the requirements contained in the Standard for Cord Sets and Power Supply Cords, UL 817, be marked VW-1, and have an ampacity rating not less than the marked rating of the apparatus.

Thanks

Thank you for choosing the XTA *C2* Dual Compressor for your application. Please spend a little time reading through this manual, so that you obtain the best possible performance from the unit.
All XTA products are carefully designed and engineered for cutting-edge performance and world-class reliability. If you would like further information about this or any other XTA product, please contact us.
We look forward to hearing from you in the near future.

Unpacking the *C2*

After unpacking the unit, please check it carefully for any damage. If any is found, immediately notify the carrier concerned - you, the consignee, must instigate any claim. Please retain all packaging in case of future re-shipment.



Introduction

The *C2* is a powerful DSP based audio dynamics processor, ideally suited for live sound applications, where it combines the accessibility and immediacy of a pure analogue design with the quality and accuracy of a digital design in a compact 1U unit. To achieve this, the *C2* has an analogue control surface, following the 'one control – one function' philosophy and a pure digital signal path, with 24-bit conversion, 40-bit internal processing and a professional 48kHz sampling rate.

The *C2* is also available with optional AES/EBU digital inputs and outputs.

Features

- ♦ Look ahead attack available on the compressor for instantaneous control of transients
- ♦ Automatic time constants which are intelligently adjusted depending on the setting of other controls (namely the sidechain EQ frequency when in De-EQ mode)
- ♦ Sidechain EQ built-in for the compressor - switchable to lo/hi shelf and wide/narrow parametric, with variable stepped gain and high-resolution frequency adjustment.
- ♦ De-EQ mode - an extension of traditional De-Essing where the sidechain filter band not only affects the sensitivity of the compressor at certain frequencies, but also only compresses around those frequencies. De-Essing, and De-Popping, and any other band reduction in power as required.
- ♦ Digitally accurate metering with indication of closeness to compression threshold.
- ♦ Independent limiter section with metering.
- ♦ Stereo mode, where both channels are linked and track 100% accurately.
- ♦ AES/EBU Digital input and output interfaces are available as an option.
- ♦ Input and output balancing transformers are also available as an option.

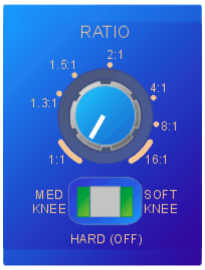
Front Panel Familiarisation

Level Meter: Dependant on the setting of the 'METER' switch, this will display the instantaneous available headroom, at either the input to the channel (analogue to digital converter) or the output from the channel, post limiter (digital to analogue converter).

Active Key: Switches the entire channel on/off – LED illuminated when processing is active. All metering and status LEDs for the channel dim when the channel is bypassed.

Threshold Control: Set the threshold at which compression will begin, from -30dBu to +22dBu (effectively off).

Meter Key: Select what the level meter shows – either input headroom or output headroom. Still active even when the channel is bypassed.



Ratio Control: Set the degree of compression to take place, once the signal exceeds the threshold, from 1:1 (no compression) to 16:1 (very heavy compression)

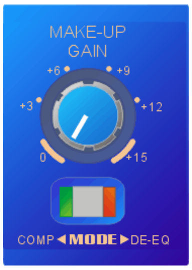
Knee Key: Cycles through 'Hard' knee (compression begins at the selected ratio as soon as the threshold is exceeded), 'Medium' (compression is less pronounced just above threshold), and 'Soft' knee (compression is least pronounced until well over threshold).

Gain Reduction Meter: Green LEDs show 6dB and 3dB *below* the threshold of compression, with the yellow LED illuminating as the threshold is reached. The remaining red LEDs indicate progressively more gain reduction (compression) up to 24dB.

***AUTO* Key:** Enable the automatic time constants – in compressor mode these are preselected to provide transparent compression in as many situations as possible. In De-EQ mode, time constants track the filter frequency.

Attack Control: Adjust how quickly compression starts after threshold is exceeded. The [-60uS] setting uses 'look-ahead' delay to preempt the signal going over the threshold and so prevents any overshoot.

Release Control: Adjust how quickly the gain reduction decreases once the signal drops below the threshold again.

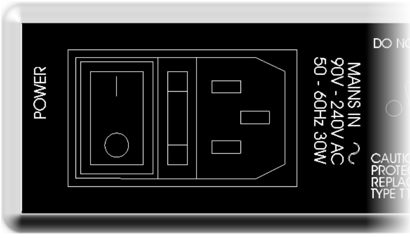


Make-up Gain Control: Apply extra gain to compensate for the compression gain reduction, helping to maintain the signal at its original level.
***MODE* Key:** Switch between normal compression mode, and De-EQ mode, where only the frequency band selected with the sidechain EQ is monitored and compressed.

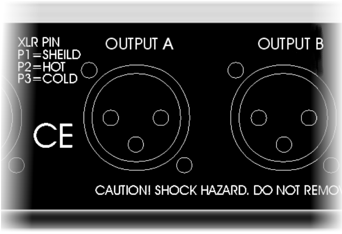
Sidechain EQ Type: Choose the shape of the sidechain filter moving between, from top to bottom, high shelf; narrow 'Q' parametric; 'wide 'Q' parametric; low shelf.
Sidechain EQ Frequency: Select the frequency range over which the sidechain filter will operate – this will be the centre frequency of the filter in parametric mode, or the corner frequency of the filter in shelf mode.
***LISTEN* Key:** Switches the output of the sidechain filter into the main signal path, so that the required range of frequencies may be more easily selected. LED flashes as a reminder that this is selected.
Sidechain EQ Gain Key: Adjust the gain of the sidechain filter to sensitise or desensitise the compressor. $\pm$6/9/15 of gain is available, as well as 0db (off).

Limiter Threshold Control: Set the threshold for the onset of limiting – effectively bypass the limiter by setting the threshold to +22dB.
Limiter Activity Meter: Yellow LED illuminates when the threshold is reached, with the red LED indicating 4dB into limiting.
***STEREO* Key:** Enabling the 'Stereo' mode will disable the right hand set of controls and force both channels to assume the parameter values of the left channel. The sidechains are also linked so that the two channels track perfectly, maintaining the stereo image.

Rear Panel Connections



Power Switch: turns the units mains supply off and on.
Mains Fuse: located in a finger-proof holder adjacent to the mains inlet. A spare fuse is also located in this holder.
Mains Inlet: connected via a standard IEC socket.



Audio In-Out: 3 pin XLR sockets are provided for each channel. All are fully balanced, pin 2 hot, 3 cold, 1 screen.

Always replace the fuse with the correct type and rating as shown on the rear panel legend.

Operating the *C2*

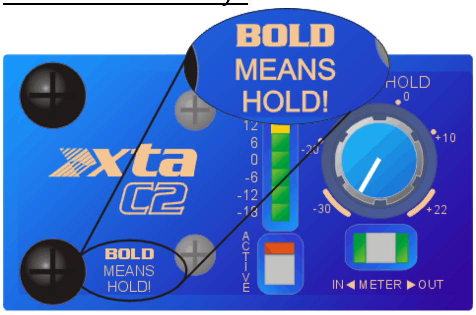
Operation of the *C2* is very straightforward, but there are a few points worth noting which, once understood, will make using the unit even easier.

Switching the unit on and start-up procedure

After plugging in the power and switching the power on suing the rear panel switch, confirmation is quickly given that all is well by various status LEDs illuminating almost immediately after power-up. These will include, as a minimum, METER (In/Out); *MODE* (COMP/De-EQ); and Sidechain EQ Type (Hi/Lo shelf etc.).

The gain reduction will fully illuminate and, after the bypass relays disengage, begin to 'count down' accompanied by the output level fading up to normal operating level. The entire process is complete when the input/output meters and gain reduction begin to operate normally. This whole start-up procedure only takes a few seconds.

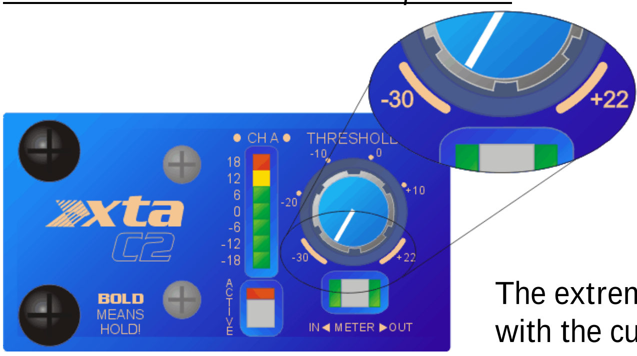
Press-and-hold Keys



The legending on the front panel alerts the user to the fact that several keys require a 'press and hold' action to initiate them. These keys relate to functions that could accidentally introduce large changes of level at the outputs, causing undesirable effects and possible damage. These keys have their function marked in *BOLD* (and a different font) to make it clear that they will only change state if the key is held in for a time.

The keys in question are…
LISTEN – switching the output to monitor the sidechain EQ signal (See page 15)
MODE – switching between compressor mode and De-EQ mode (See page 16)
AUTO – switching the automatic time constants on and off (See page 26)
STEREO – linking the two channels together to operate in stereo (See page 26)

Minimum and maximum control positions



To ensure that the *C2* is 100% accurate all of the time, and that what it says on the front panel is exactly what the unit is doing, it has been necessary to introduce 'end-stops' on the controls.

The extreme regions on each control marked with the curved line designate this entire region as relating to the parameter value shown. This is to compensate for the mechanical tolerances of the potentiometers.

Compression – What It Is and What It Can Do

What does a Compressor do?

A compressor is designed to do as the name suggests – compress. What it compresses is the dynamic range of the audio fed to it. This means that it is used to control the level of the audio in such a way as to limit its maximum value, but in a gradual fashion. As you might imagine, it would be possible to prevent the level from ever going over a maximum threshold in a very severe manner – generally this type of action as known as 'Limiting', and is the responsibility of the Limiter Module also in the *C2*. The method by which a compressor limits the maximum level is performed in a more gradual way so as to gently control the dynamic range, rather than dramatically 'cap' it, which is likely to introduce audible side-effects.

Why are Compressors necessary?

Consider again the term 'dynamic range' for a moment. The dynamic range of any instrument or device is a measure of the ratio of the maximum possible output level to the minimum level that can be reproduced. In the case of, for example, a compact disc player, the dynamic range exceeds 90dB. However, considering a typical analogue tape recorder, the higher noise floor limits the dynamic range to about 70dB.

If two pieces of equipment are connected together (to record the CD for example), the unit with the lowest dynamic range sets the dynamic range for the entire system. In the example above, the dynamic range available is limited to 70dB, losing 20dB.

This is true in many live sound situations as well, where the dynamic range from one component of the system will prove to be the limiting factor in the whole set-up. Typically this will be the power amplifier or speaker system. Running the system at such a level so that the average volume is adequate will not allow signal peaks to be handled correctly. Given a musical peak that is 6dB above the average signal level, if the system is forced to try and reproduce a level that exceeds its dynamic range, distortion will almost always occur.

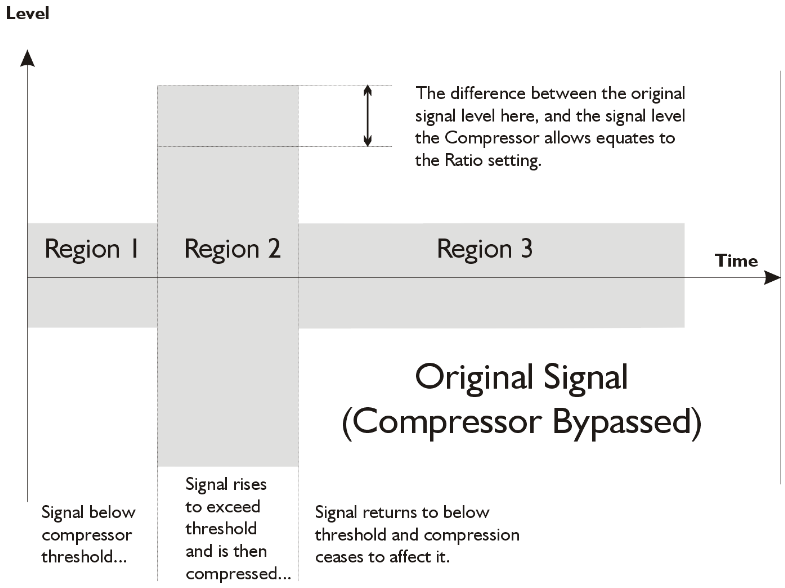
So how can this scenario be avoided, or at least improved upon?

In an ideal world, all devices used to record and reproduce sound would have boundless dynamic ranges and so present no compatibility problems with each other. As this is not the case, some method must be employed to limit the dynamic range of signals in situations where they will prove troublesome or incompatible. This could be achieved just by 'turning down' the offending level until it reaches a point where the maximum possible output will not exceed the dynamic range of the system as a whole.. Unfortunately, this type of action rarely produces satisfactory results, as the average level will also be attenuated, meaning the perceived volume will drop.

What is required is a device that can monitor the signal passed to it, and allow the *average level* to pass untouched. At a *threshold level* the device will begin to turn down the level by a certain amount. This *amount* can be varied, to offer more or less attenuation of signals above the threshold. What would also be useful would be some way of controlling the *rate* at which the level is turned down (slower rates will not sound as unnatural as the effect of ‘reaching for the volume control and giving it a sharp twist’). Additionally, the ability to change the *rate of recovery* of the original gain over a period is as important.

<u>How does a Compressor work?</u>

This is, in essence, exactly what a compressor does. Consider the diagram below, which shows a signal level versus time . It represents a burst of audio, which is initially below the threshold of the compressor - in region 1. As this is in the ‘safe’ region, it is unaffected by the compressor. It can be thought of as the average signal level and remains unattenuated. However, in region 2, the signal has risen (very sharply, hence the sudden jump) to a level above the threshold.

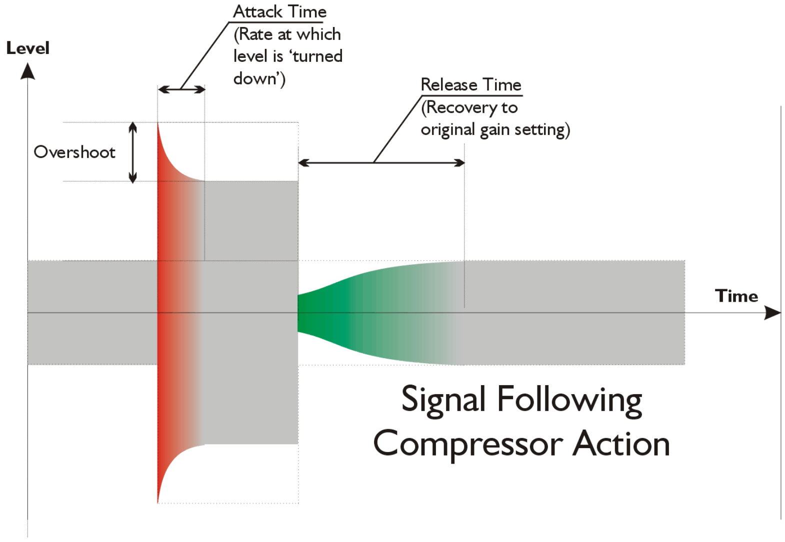


This signal is potentially too high and must be reduced in level to a more acceptable value. The difference between the original signal and the compressed signal is set by the Ratio. So, for example, consider the input jumping from 0dB to +10dB, with the threshold set to 0dB.

If the output rose by 5dB, then the ratio would be 10/5 or 2:1. Comparing the level displayed on the input meter with the level on the output meter (assuming all other dynamics modules are bypassed) will demonstrate the level of Gain Reduction being applied to any signals that exceed the threshold. This amount is shown on the Gain Reduction meter for the compressor.

Overleaf is the same signal after passing through the compressor. Important in region 2 is the red shaded area – this represents the time to fully apply the gain reduction set with the ration control, or the Attack time.. The faster the attack time, the less signal will ‘escape’ through the compressor before it reacts and applies gain reduction. The setting of this control will be discussed in the next section. It’s value can have a profound effect on the transparency of the compressor – that is, how natural it sounds and how noticeable its action appears.

The last region of interest is region 3 where the signal returns once again to below the threshold. The green shaded area represents the recovery time for the compressor to stop applying gain reduction and effectively ‘recognise’ that the signal is to pass through unaffected. This represents the Release time and again must be carefully set so as to prevent audible artefacts.



Note the level difference marked as ‘Overshoot’. This phenomenon cannot be avoided because the compressor has no way of knowing when the signal will reach the threshold, and so cannot ‘prepare in advance’ for this happening.

Consequently, there will always be a brief period before it can react, even at the minimum attack setting (fastest rate of gain reduction application).

<u>What makes the XTA Compressor different</u>?

Being a totally digital dynamics processor, the *C2* has several advantages over conventional compressor designs. By inserting a short delay into the sidechain signal path, the compressor is able to ‘look ahead’ and anticipate peaks before they occur. This gives it the ability to prevent the overshoot that is unavoidable with analogue compressors. In this way, the compressor behaves in a more predictable and controllable manner, with no compromises in performance.

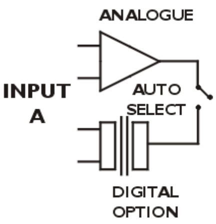
It is quite common practice to insert some EQ into the sidechain of a compressor to make it ‘frequency conscious’. This would normally involve patching an external equaliser through the external sidechain loop (or external key), tying up a valuable resource in the process. the *C2* ’s compressor has a band of fully featured parametric equalisation built into the sidechain. Each band can be swept from 20Hz to 20kHz and is selectable as either a parametric filter, or a shelving filter. Additionally, the filters can be bypassed, so that, in conjunction with the ‘Key Listen’ facility, its effect on the sidechain can be quickly and accurately determined.

Modelling all the dynamics processing digitally allows the time constants set by the user to be one hundred per cent accurate all the time, with no drift due to atmospheric changes. The ability to link the two channels together also permits completely accurate stereo tracking.

C2 Configurations

There are two distinct modes of operation of the *C2* - normal compressor mode (augmented by a sidechain equaliser section), and the De-EQ mode. To explain the operation of these two modes and clarify the differences between them, they will be broken down into building blocks, which are connected differently in the two modes. The building blocks and their operation is explained below.

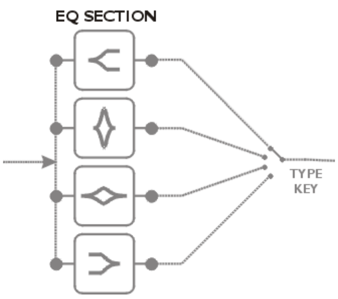
Input Section:



The input section is the same for both modes – the audio signal is either converted to digital, ready for processing, or is fed directly from a digital AES stream. Note that the digital input is a factory fitted option only.

The signal will be split at this point – one path forms the main signal path, the other forms the sidechain (or control) path that is used to modify the main one – in this application to compress it.

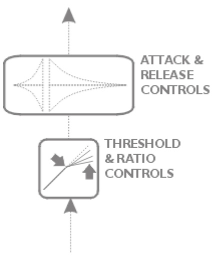
Sidechain EQ Sections:



The signal used for controlling the main path may be equalised prior to its use as a control signal – this enables it to be made more or less sensitive to certain frequencies, changing the way the compressor reacts the audio. The degree of sensitivity may also be also be adjusted (the Sidechain EQ Gain control) when in compressor mode. More on this later.

This signal (suitably equalised) is then passed into the control section where it is used to determine the degree of compression of the main signal and how this is applied.

Control Sections:



This control signal is compared against the **threshold** setting to see if it is to start affecting the main path. Included in this section is the **ratio** control which determines how severely the main signal is compressed as it continues over the threshold, and the **attack** and **release** times which determine how quickly the compressor reacts and 'leaves go' as the threshold is crossed.

This signal is now used to directly affect the gain (in compressor mode) of the main signal, and conceptually does so using the …

Gain Control Section:

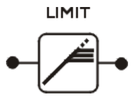
This element is simple in its operation – the signal to be controlled passes straight through, and the signal fed in from the side can 'turn down' the main signal, but is never heard itself. To hear this control signal, it is necessary to monitor the sidechain path.

The sidechain *LISTEN* signal may be monitored at the output, using the *LISTEN* key on the front panel. If the signal has been heavily compressed, chances are the overall level will have dropped considerably. At this point it might be necessary to introduce some additional gain to compensate for this loss.

Make-up Gain Section:

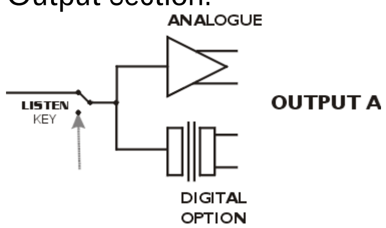
The introduction of extra (static) gain is best applied by ear, but the use of the metering will aid the process. The intention is to make the input and output levels of the unit match. Frequent comparisons using the 'ACTIVE' key to bypass the unit, and watching the gain reduction meter, which will show the instantaneous amount of compression being applied, are the keys to matching the levels.

Limiter Module:



Separate to the compressor is a fully independent limiter module. This follows the compressor and is post-make-up gain at all times. It determines the absolute maximum output level from the unit, set with the **threshold** control. However, the *LISTEN* signal does **not** get routed through the limiter, so be careful when engaging the sidechain monitoring.

Output Section:



As mentioned in the limiter section, the *LISTEN* key is located just before the output stage, and routes the output of the sidechain (equalised or not) to the main output, so that it can be accurately tailored to achieve the desired response. The output is then converted back to analogue or transmitted as a digital signal to an independent output. Note that the digital output is a factory fitted option – please see Appendix 1 for further information about this.

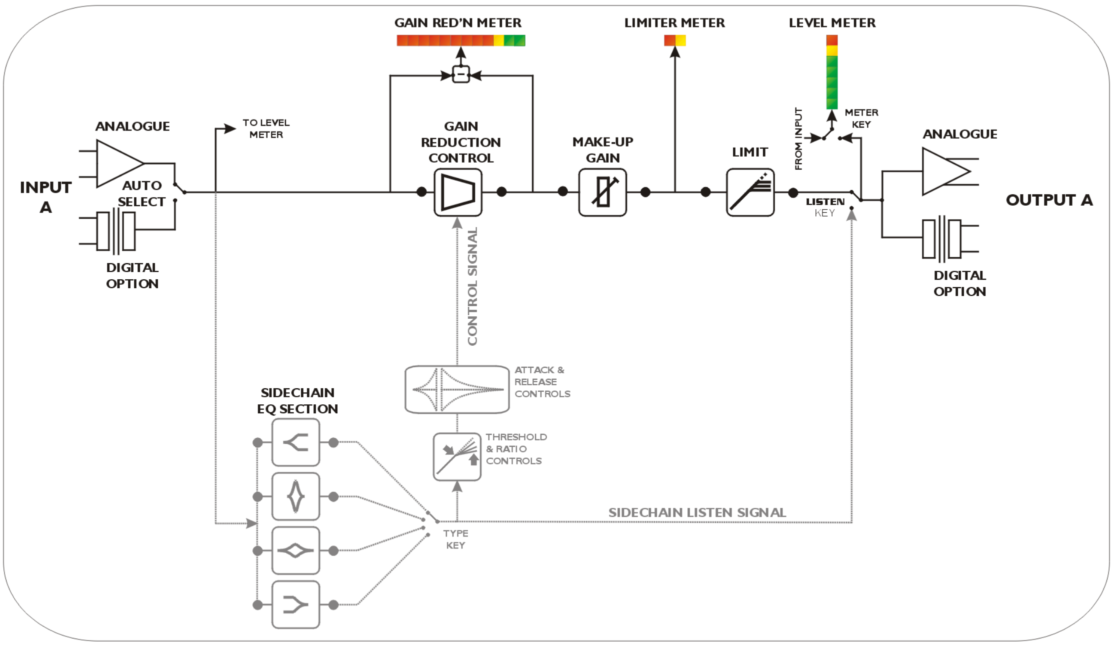
Now that the main building blocks have been explained, their interactivity and connections can be outlined with reference to the two modes of operation.

Compressor MODE

Below is the block diagram showing how the compressor works.

Following the input section, the signal is split into the main path (audio) and the sidechain (control). The main path consists of the gain control element, followed by the make-up gain section, then through the limiter and directly to the output.

The sidechain may be equalised by passing it though the EQ section, and this equalised signal may be monitored by engaging the *LISTEN* function. In this mode, the sidechain control signal may be equalised to make it more or less sensitive to certain frequencies. This will in turn affect the threshold of the compressor, making it compress to a greater or lesser degree depending on the frequency content of the incoming signal.



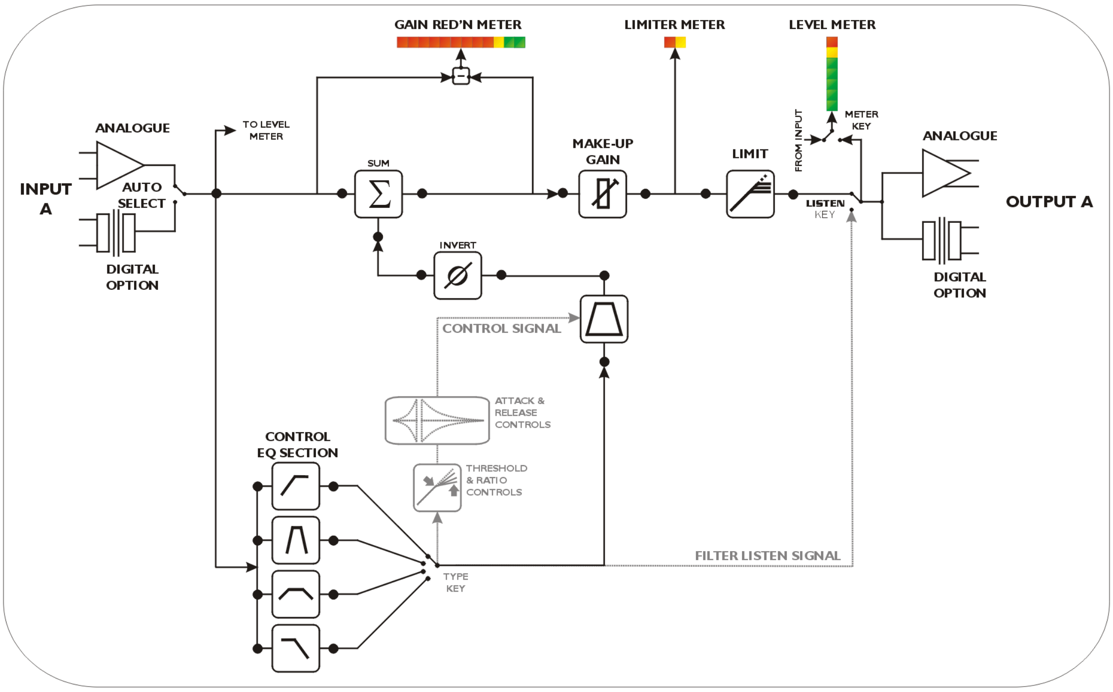
However, no matter how the sidechain is equalised, the compression is applied to the entire main signal – and all frequencies are reduced in gain by the same amount. This is known as '**broadband compression**'. The introduction and use of the sidechain EQ adds the facility to make this '**frequency conscious**'.

DE-Eq MODE

Below is the block diagram showing how the De-EQ mode works.

Following the input section, the signal is split into two paths, which are eventually recombined to form the main signal path again.

One signal passes through the equalisation sections that, in this mode, operate slightly differently to the sidechain EQ for the compressor mode. In compressor mode, the EQ can be used to cut or boost a band of frequencies, but in De-EQ mode, these filters are only used to 'home in' on a band, to make the signal more sensitive in this area.



This filtered signal is then used to control the threshold at which it is recombined with the other unfiltered signal. Note that the gain control element is in this part of the signal path with the control signal adjusting its level.

The crucial part is following the gain control element. The filtered signal is now *inverted* so that, when it is re-combined with the other signal, it will cancel out some of the overall level around the selected band, so compressing just those frequencies.

Sidechain Equalisation – How and When to Use it

When would Sidechain EQ be useful?

When it is necessary to adjust the sensitivity of the compressor, tailoring the sidechain signal with equalisation is the only way to achieve this without affecting the main signal path.

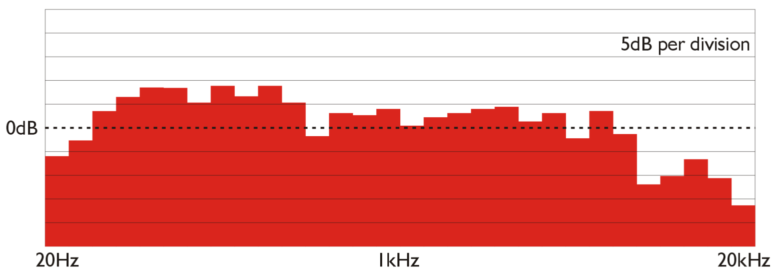
De-essing and de-popping.

For example, a de-esser is really a compressor that has had its sidechain made more sensitive to sibilance (sounds such as 's' and 't'), which occurs in the range 6kHz to 9kHz. A de-popper will be a compressor sensitised to respond to low frequency plosives (sounds such as 'p' and 'b') in the range 80Hz to 150Hz.
In reality, the attack and release times of the compressors as implemented for de-essing and de-popping will be set appropriately to prevent the compression having too great an effect on the overall signal.

Both of these examples illustrate using the sidechain EQ to make the sidechain more sensitive to certain ranges of frequencies. It can be very useful to be able to de-sensitise the sidechain to certain frequency bands to prevent excessive activity at inappropriate times.

Maximising loudness without dulling the mix.

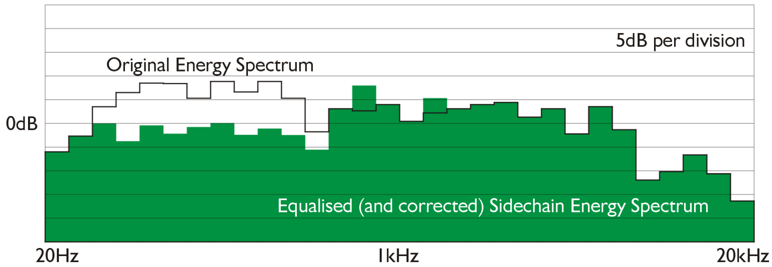
Loudness, as opposed to volume, is dependant on the signal density as much as the level and so compressing the program can increase the perceived loudness without actually requiring more headroom. For example, consider the situation where bass-heavy program material is to be compressed to maximise loudness in a live situation. The spectrum might look like this.



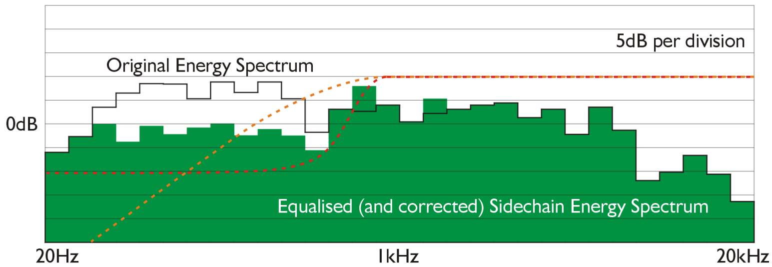
If the threshold was set to 0dB (the dotted line) it can be seen that the compressor will start to act first when the low frequency part of the spectrum crosses the threshold.

This will have the effect of pulling down the level across the *entire* spectrum (including the high frequencies) and so causing the familiar 'dulling' that can easily occur with full range compression. The solution to this problem is to introduce some complementary sidechain EQ that will remove some of the offending bass from the control signal, making the compression more 'even-handed' again. The required effect on the spectrum would be similar to that shown below.

Having used EQ to tame the bottom end in the sidechain, the compressor will now be much less sensitive to low frequencies and so won't start applying gain reduction in response to only bass.

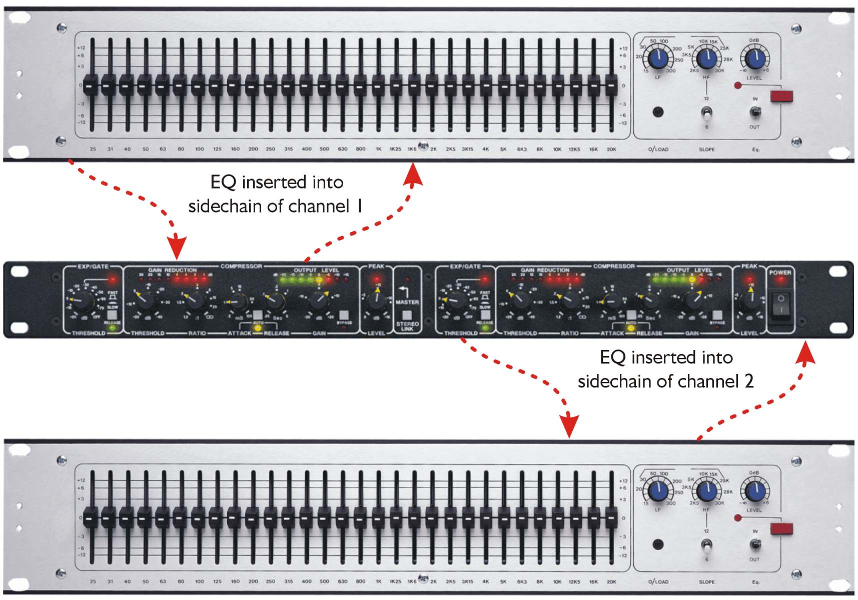


Traditional sidechain EQ is normally only featured in noise gates, and is limited to high and low pass filters. In this situation, whilst a high pass filter could have been used to remove some of the low frequency weight present, a wide shelving cut band would be better, offering much more accurate control of the amount of bass removed. The curve imposed by a high pass filter is shown in orange, and that of a shelving filter in red. This highlights the advantage to incorporating more flexible EQ into the sidechain than that normally offered. The high and low pass filters may be used to perform quite severe tailoring of the frequency response, but for accurate highlighting of a small band of frequencies, parametric filters are much better.



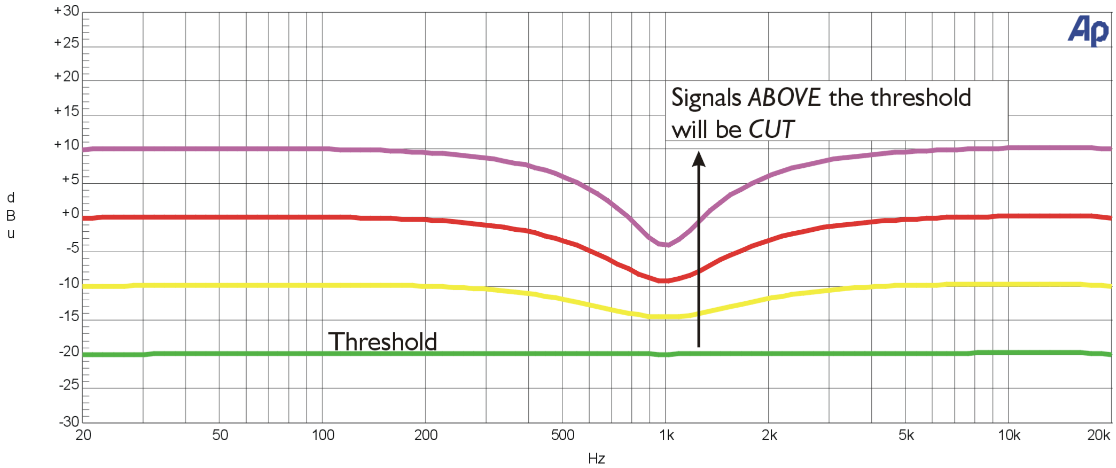
__How would sidechain EQ normally be implemented?__

Typically, sidechain EQ would be patched into the in/out external path as offered by the dynamics processor. The *C2* obviates the need for this external processing for any of the dynamics modules.



Why is the De-EQ mode so useful?

Having selected the frequency band to work with, the dynamic eq will listen to this band and act upon it by cutting(compressing) any frequencies present in it that go above the predetermined threshold. Consider the example below where the threshold is set to –20dB, and the selected frequency band is centred around 1kHz.



Signals below the threshold will pass unaltered, but as increasing signal is applied, those frequencies centred around 1kHz will be cut or compressed. The ratio in the above example is set at 2:1 so, as with any compressor, the amount of gain reduction applied depends on how much the signal exceeds the threshold. The red line represents a signal at 0dB, which is 20dB above the threshold. At 1kHz, therefore, the signal has been compressed to –10dB or 2:1.

Uses of "De-EQ" mode.

Traditional use of 'frequency conscious' compression is to control or 'tame' a certain band of frequencies within the program material. Insertion of EQ into the sidechain will make the compressor respond to the required band, but it will cause broadband compression of the signal, so any peaks will cause the entire signal to be compressed. This produces the familiar problem of dulling the material if it is bass-heavy, or causing unnecessary dips and changes in ambience when attempting to remove sibilance.

The difference with dynamic EQ is that only the band selected is compressed. This means that it becomes possible to compress the low frequency content of material without affecting the high frequencies at all. The result is increased volume and perceived level with out sacrificing clarity. Any instance where the desired result is to control a band of frequencies, such as de-essing, or de-popping, without affecting the surrounding frequency ranges is an ideal use for this mode.

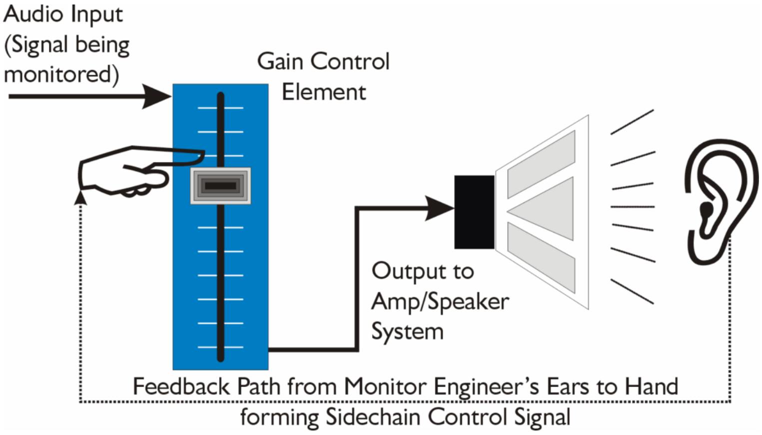
Try de-essing with the filter centred at 8-9kHz, and a narrow bandwidth setting, attack 25mS, release 100mS.

Look Ahead Delay – Pre-emptive Action

One of the most significant advantages of digital signal processing over analogue is the ability to delay the audio signal precisely and without extensive complex hardware. The entire domain of digital signal processing is based around the combination of delaying, multiplying, and accumulating numbers (representing samples of audio) to implement all the filters and dynamics processing we have come to expect today.

In the case of dynamics processing, being able to delay a signal allows the processor module to delay the main signal in relation to the sidechain (the signal being monitored relative to the threshold), so that it can compensate for peaks prior to the arrival of the main signal.

Consider the situation of a monitor engineer listening to a band perform[1]. Having no access to dynamics processors, he has had to resort to manually 'riding the faders' in an attempt to keep control of the levels. Should the level of one of the channels on his desk reach an unacceptably high level, he will turn it down appropriately.

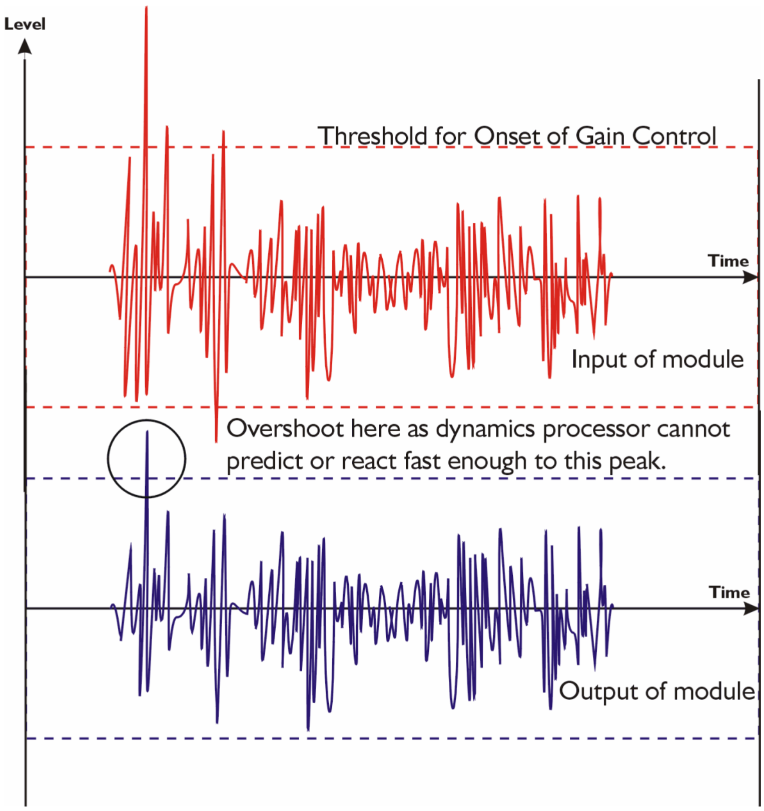


There is a hidden sidechain in operation even in this case. The main signal path is fed through the monitor desk and the gain controlled by adjusting the fader. The sidechain is formed by the feedback path between the engineer's ears checking the level and his brain instructing his hand to turn the fader down if the volume goes over the threshold he has chosen.

In this case, the delay between the signal actually going over the threshold, the engineer registering the situation, and then turning the signal down will be in the order of several hundred milliseconds at best. This will only be true if he is not distracted – in reality, it may be several seconds before any gain reduction is imposed on the signal to bring it under control.

For an analogue dynamics processor, the situation is much better. Controlling the gain electronically, and not relying on a human sidechain feedback mechanism, it can react much more quickly.

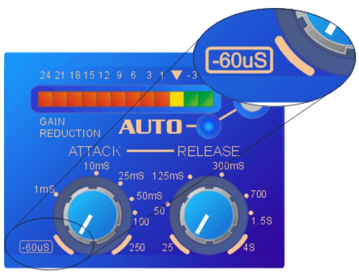
[1] XTA would like to point out that whilst the *C2* might 'sound' male, not all engineers are necessarily male. Some might well be female, or at least have long hair.



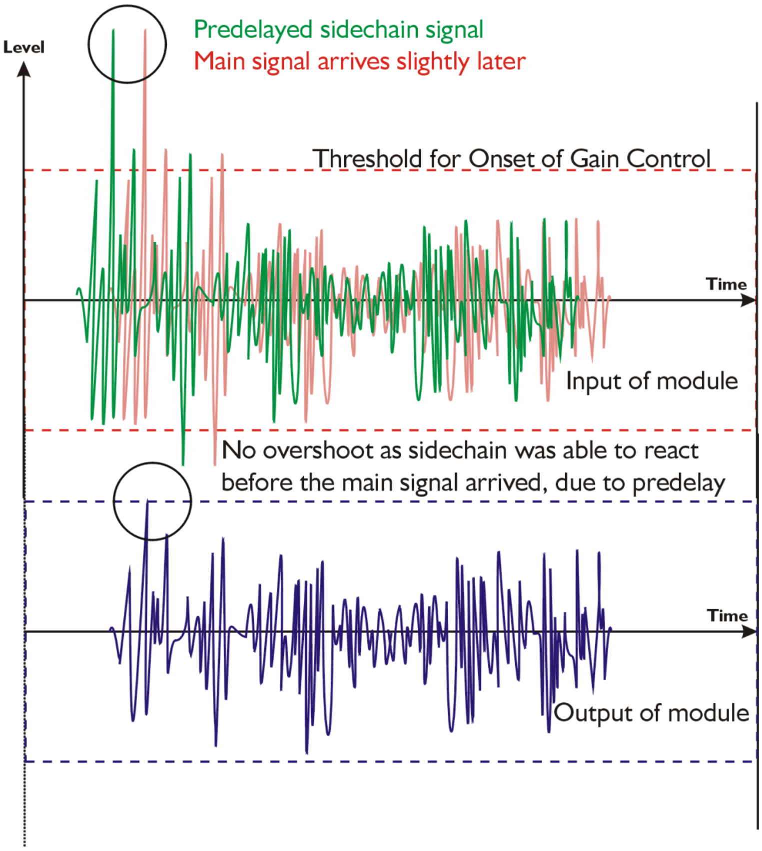
The red waveform represents the input to the dynamics module, with the dotted line showing the threshold for gain control to occur. There are several peaks towards the start of this signal that are above the threshold, and so the dynamics processing should react to these as appropriate. (In this case reduce the gain).

The blue waveform show the output of the dynamics module. The circled peak demonstrates that the processor has missed the first peak above the threshold (as it is very fast and short), but has 'caught up' shortly afterwards, keeping all other peaks under control. As it is unable to predict what is coming, this will always be a failing with analogue dynamics processing.

The *C2*'s ability to predelay the sidechain allows it to predict what will be appearing in the main signal path and react before the signal arrives, thus preventing the overshoot seen above.



Making sure that the *AUTO* system is off (blue LED is not illuminated), setting the attack time control to its minimum setting will force the *C2* to use predelay in the sidechain. The effect of this is explained in the next section.



The predelayed sidechain is shown in green, with the main signal in red.

As the main signal arrives slightly after the sidechain, the output from the *C2* does not suffer from the overshoot problem.

Remember that this delay is only in the order of 10 to 60uS,and is a **predelay** – the sidechain is moved **back** in time in relation to the main signal. Inserting a delay into the **main** signal path of an analogue dynamics processor will achieve similar results, but with the penalty of delaying the main signal by the amount of look ahead delay introduced.

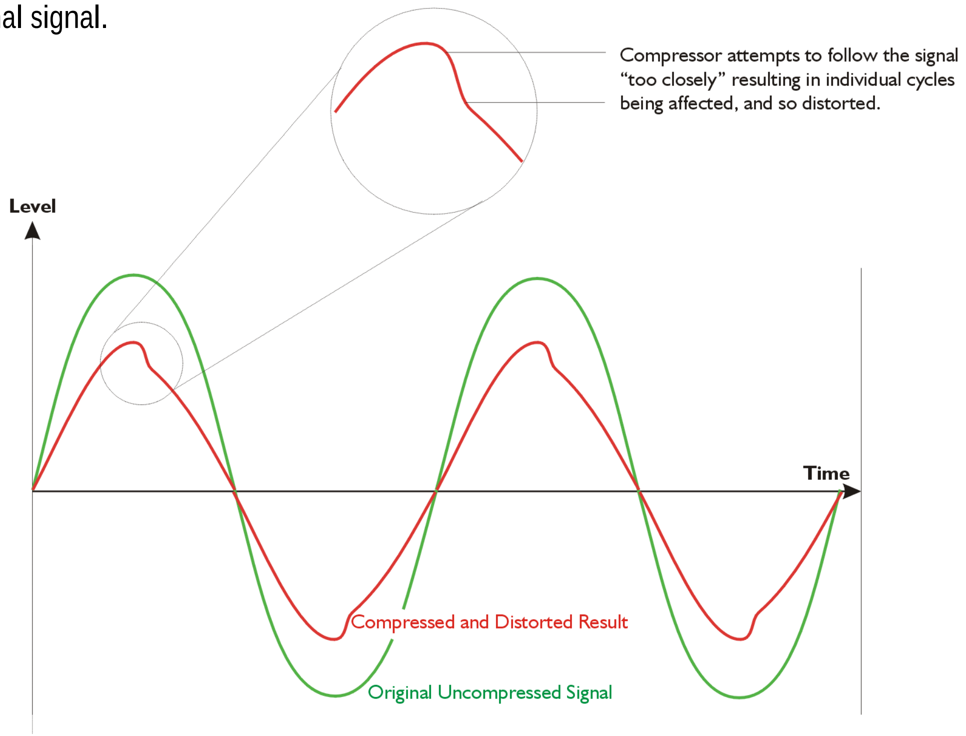
Note that the *C2* does not have to use this look ahead delay – in many cases, it is preferable to allow the overshoot. For example, compressing percussive instruments where the overshoot retains a degree of the original high frequency energy, stopping the sound from becoming lifeless and dull.

Times when it is useful include:

- Preventing the limiter from ever overshooting when mastering for digital media;
- Ensuring maximum level into a desk using the compressor to level signals without clipping the input to the channel.

Setting the Attack and Release times

As with all compressors, using too fast attack and release times on low frequency program (such as a bass guitar) will cause the compressor to respond to individual cycles of the signal, rather than the overall envelope. This will result in obvious distortion, which might be described as sounding like clicking superimposed on the original signal.



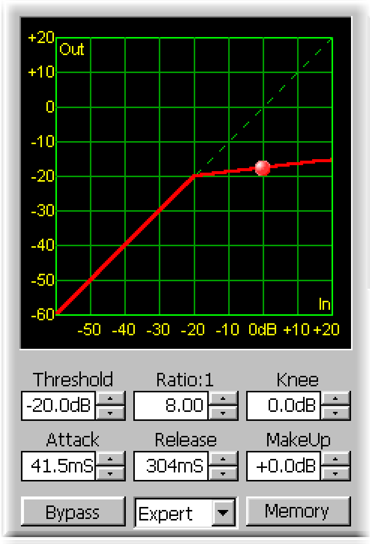
The compressor release time has been deliberately restricted to a minimum of 25mS to prevent excessive distortion on low frequency signals, even with fast attack times and high ratios. None the less, it is still possible to introduce some distortion if care is not taken with the settings. The best way to ensure that the signal is not being excessively distorted is to make good use of the 'ACTIVE' button, constantly comparing the original signal with the compressed version.

The Compressor 'Knee' Control.

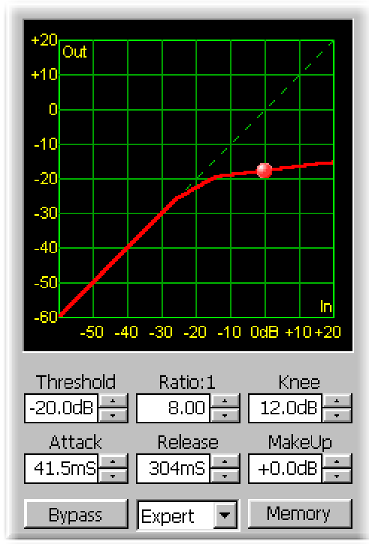
One of the most misunderstood parts of a compressor is the parameter usually labelled the 'Knee'. This may be a fully variable control, or a switchable parameter, normally with 'Hard' and 'Soft' settings. The knee control permits a softening of the compressor action, which can prove to be especially useful at high compression ratios.

Without the inclusion of a knee control, at the threshold of gain reduction, a sudden transition occurs between unity gain and the ratio by which the compressor attenuates. When using high compression ratios, the use of a hard knee can result in a very unnatural sound.

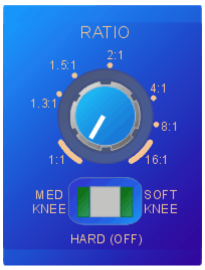
Consider a compressor with a relatively high ratio of 8:1. A typical in-out transfer function would be as below[2]. The very sharp introduction of gain reduction is obvious at the threshold point (-20dB).



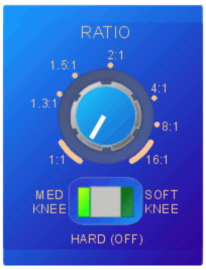
The 'Knee ' parameter is set to 0dB in this example – this corresponds to a 'hard' setting. Increasing the 'Knee' to its maximum of 12dB, spreads the onset of the compression over a wider area, (6dB above and 6dB below the threshold), reducing the severity of the compressor. This shows on the graph as a flattening at the threshold, rather than a sharp bend.



The *C2* offers three preset knee settings:



Hard Knee: Compression begins immediately the threshold is exceeded, at the full ratio setting.



Medium Knee: Compression begins 6dB below the threshold at half the set ratio, and reaches full ratio 6dB above.

Soft Knee: Compression begins 12dB below the threshold at half the set ratio, and reaches full ratio 12dB above.

[2] Software screenshots are from SiDD PC control software, available for the SiDD dynamics processor

Using the AUTO system

The automatic system on the *C2* is responsible for taking care of the setting of the attack and release times in either compression mode or de-eq mode.

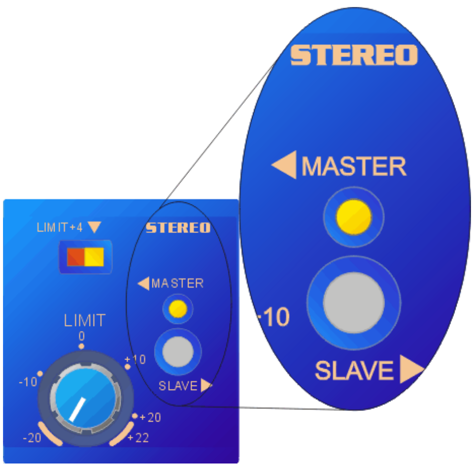
By adjusting these times dependant on the use of the use of the sidechain EQ, the compression can be made as transparent as possible.

The *AUTO* system is further supplemented by the *C2*'s ability to prevent combinations of attack and release times that would cause distortion when large amounts of gain reduction are being applied to the signal.

Even with the automatic time constants off, in compression mode, the release time cannot be set to less than double the attack time, whilst in de-eq mode, the release time cannot be set to less than the attack time. Whilst this operation is not made apparent by the unit's front panel controls, it does go a long way to guaranteeing that the compression will never sound terrible!

Enabling the *AUTO* system by *HOLDING* the *AUTO* key will illuminate the blue LED and render the attack and release controls inactive.

STEREO Linking



Pressing and *HOLDING* the *STEREO* key will illuminate the LED and gang the two channels of the unit together.

This has the effect of disabling the controls for channel B, as both channels will assume the settings on channel A. Additionally, the sidechains will be linked so that, if one channel compresses more than the other, the same amount of compression will be applied to both channels, so avoiding any shifts in the stereo image.

Metering remains independent between the two channels. Note that as soon as the stereo linking is turned off, channel B will assume the parameters as set by its controls. This might have a dramatic effect on the level of the channel, so be careful!

Operating Notes

Operating Level

With any audio signal processing equipment it is necessary to ensure adequate signal level is used through the device, to avoid sacrificing noise performance. It is suggested that the operating level chosen should give adequate level to just light the -12dB LED on the headroom meter with maximum program level being used. Since the meter is deliberately set to show clipping 3dB early, this still provides 9dB of headroom before clipping occurs. With equalisation in use it may be necessary to further reduce the input level, as gain within the unit may cause digital clipping, indicated by the top red LED's lighting independently of the rest of the meter.

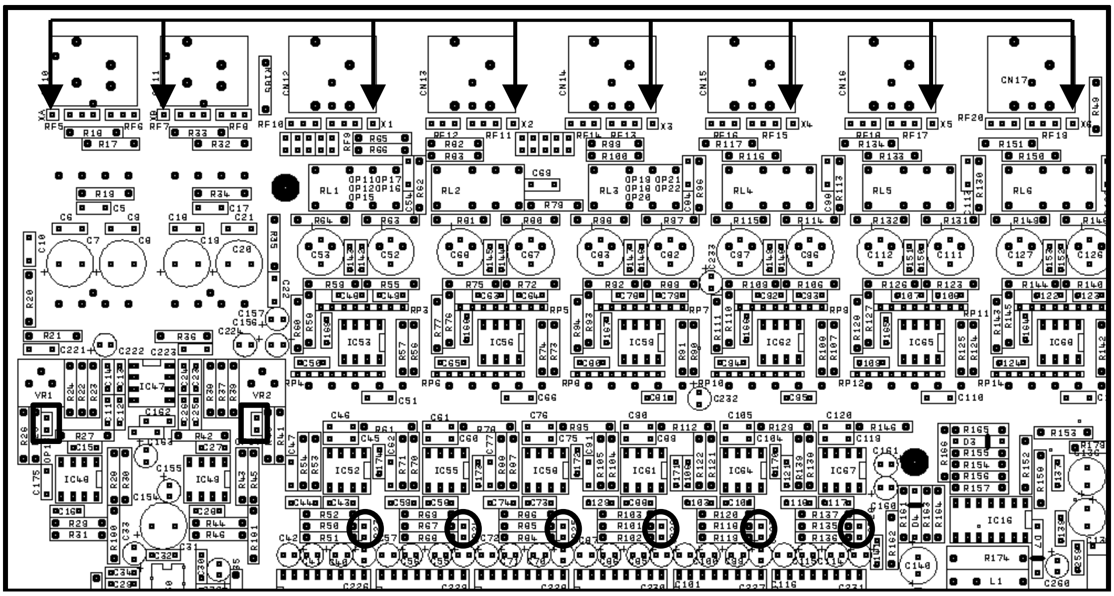
It should be noted that the figure quoted for the maximum input level options is the clipping point for that option (not a safe operating level). Always ensure that this clipping point is no lower than that for the following equipment in the signal chain, and allow extra margin if equalisation sections are boosted.

Grounding

The Screen (shield) pins on all audio connectors are normally connected directly to the ground pin of the IEC mains inlet. The chassis is also directly connected to this pin. Never operate this unit without the mains safety ground connected. Signal ground (0V) is in turn connected to the chassis ground.

To avoid ground loops, cable shields should be connected to ground at one end only. The normal convention is that the shield is only connected at the output XLR.

Provision is also made for separately isolating each input and output shield pin permanently within the *C2* by breaking the appropriate PCB track, where marked with a box and an arrow next to each XLR connector using a small drill bit or cutter. See the following diagram for details.



XLR pin 1 Isolation points (arrowed) and 10dB pads (circled)

Specifications

Inputs: 2 electronically balanced◆
Impedance: > 10k ohms.
CMRR : >65dB 50Hz - 10kHz.

Outputs: 2 electronically balanced◆
Source Imp: < 60ohms
Min. Load: 600ohm
Max. Level: +20dBm into 600 ohm

Frequency Resp.: ±½dB 20Hz-20kHz
Dyn Range: >110dB 20Hz-20k unwtd
Distortion: < .02%@1kHz, +18dBm

Compressor/De-EQ
Threshold: -30dBu to +22dBu
Ratio: 1:1 to 16:1
Attack: -60uS to 250mS
Release: 25mS to 4 S
Knee: Hard/Med/Soft characteristic.
Make-up Gain: 0 to +15dB

Sidechain EQ Section
Type: Selectable high/low shelf or Narrow(0.83 Oct.)/wide (1.25 Oct.) parametric response.
Centre/corner Freq: 20Hz to 20k
Gain: ±6,9 or 15dB and 'Off'
Sidechain monitor available.

Limiters
Threshold: -10dBu to +22dBu
Attack: 1mS (preset)
Release: 300mS (preset)

Input/Output meter: 7 point, -18dBu to +18dBu.
Gain Reduction meter: 12 point, 6dB below threshold to 24dB red'n.

Connectors
Inputs: 3 pin female XLR
Outputs: 3 pin male XLR.
Power: 3 pin IEC

Power: 60 to 250V ±15% @ 50/60Hz.
Consumption: < 20 watts.
Weight : 3.5kg. Net (4.8kg. Shipping)
Size: 1.75"(1U) x 19" x 11.8" (44 x 482 x 300mm) excluding connectors.

Options ◆ = Transformers available.
Optional Interfaces AES/EBU Digital Input/Output

Due to continuing product improvement the above specifications are subject to change.

Warranty

This product is warranted against defects in components and workmanship only, for a period of one year from the date of shipment to the end user. During the warranty period, XTA will, at it's discretion, either repair or replace products which prove to be defective, provided that the product is returned, shipping prepaid, to an authorised XTA service facility.
Defects caused by unauthorised modifications, misuse, negligence, act of God or accident, or any use of this product that is not in accordance with the instructions provided by XTA, are not covered by this warranty.
This warranty is exclusive and no other warranty is expressed or implied. XTA is not liable for consequential damages.

Options and Accessories

Part Number	Part Description
ITX-100	*C2* Transformer balanced inputs (factory fitted only)
OTX-100	*C2* Transformer balanced outputs (factory fitted only)
AES-C2	AES/EBU Digital inputs/outputs (factory fitted only)