G2 Dual Gate/Expander

Operators Manual

G2
Dual Gate/Expander
Digital

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G2 Quick Reference

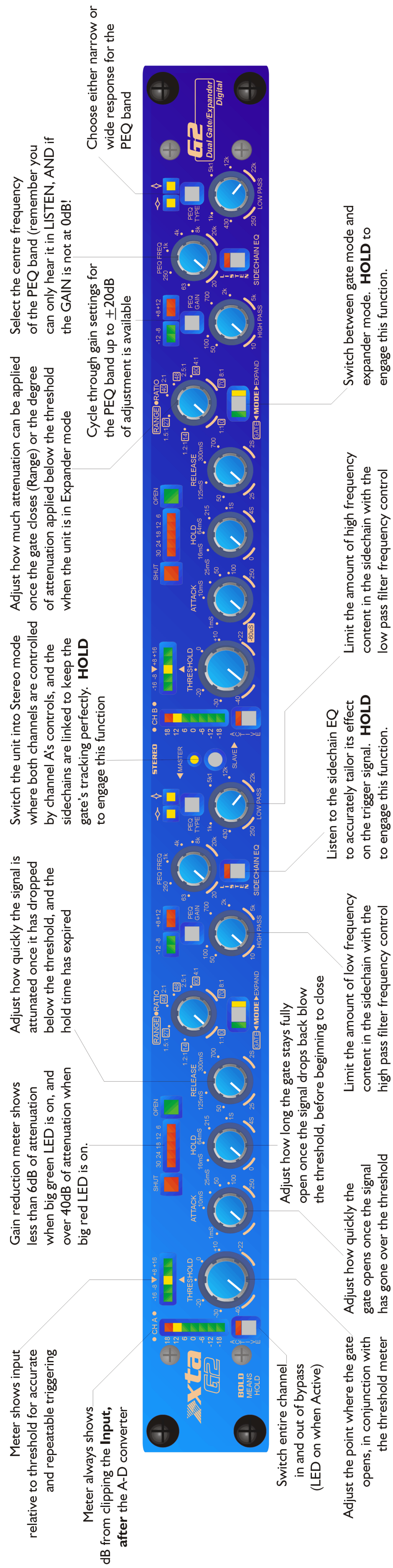
THINGS YOU NEED TO KNOW…

✓ ***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:

Changing ***MODE*** between Gate and Expander;
Enabling the sidechain ***LISTEN*** function;
Ganging the two channels together in ***STEREO*** mode.

- ✓ The main meters show level, in dB, from the clipping point of the input of the unit just after the analogue to digital converter stage. As the unit is unable to add any gain to the signal, there is no need for an output meter.
- ✓ When in Expander mode, the Hold knob has no effect.
- ✓ Remember that you won't hear the effect of the parametric sidechain band, unless there is cut or boost applied – use the GAIN switch to cycle through the levels.
- ✓ The Listen LED always flashes when Listen is enabled. Listen is automatically turned off if the unit is switched off and on.
- ✓ Remember that the Range/Ratio control applies to both modes of operation – the values in the boxes are for Gate mode, and are in dB, the other values are ratios for Expander mode.



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

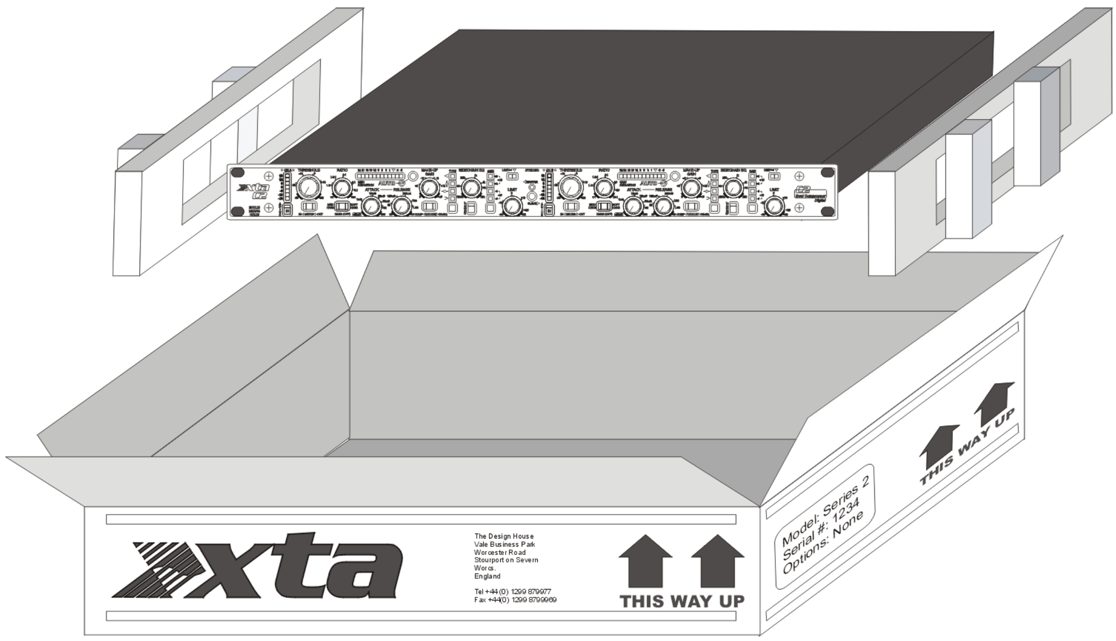
It should not be necessary to remove any protective earth or signal cable shield connections.
Do not defeat the purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wider blade 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 *G2* Dual Gate/Expander 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 *G2*

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 *G2* 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 *G2* 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 *G2* is also available with optional AES/EBU digital inputs and outputs.

Features

- Look ahead attack for true 'clickless' operation and instantaneous opening of the gate
- Fully adjustable envelope controls, including hold time (in Gate mode) with large, easy to see, metering
- Sidechain EQ built-in including low and high pass filters, and an additional parametric section with variable stepped gain and high-resolution frequency adjustment
- Expander mode – for more subtle gating applications, with gain reduction metering and fully variable ratio control
- Stereo link mode locks the sidechains together for perfect stereo tracking as well as parameter matching
- Threshold metering to allow accurate and repeatable triggering of the gate
- 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: this meter displays the instantaneous available headroom available at the input, just after the analogue to digital converter.
Active Key: Switches the entire channel on/off – LED illuminated when processing is active. Note that the Listen mode may still be active when the channel is bypassed.
Threshold Control: Set the threshold at which the gate will open, from -40dBu to +22dBu (effectively off).
Threshold Meter: This meter display the closeness to the threshold, ranging itself dependant on the setting of the Threshold control and the input level. The yellow LED shows the point at which the gate will open.

Attack Control: Set how fast the gate will open (the fade-in time) – the '60uS' setting uses look ahead delay to open the gate before the main signal arrives, so preserving the entire waveform.
Hold Control: Set how long the gate stays fully open after the signal has dropped below the threshold, and before entering the release phase.
Release Control: Once the signal has dropped below the threshold and the hold time has expired, this control determines how quickly it is attenuated (the fade-out time).
Gate and Gain Reduction Metering: The large green LED is only ever illuminated when there is less than 6dB of attenuation on the channel (so effectively the gate is fully 'open'). As the attenuation increases, this LED goes off, and the red meter begins to illuminate right to left. This shows increasing gain reduction, until the final large red LED illuminates when there is more than 40dB of attenuation.

Range/Ratio Control: In Gate mode, this control sets the maximum attenuation allowed (the Range) when the gate is fully closed. Note that if the Range is set to less 40dB, the large RED led may never illuminate. In Expand mode, this control determines the degree of attenuation as the signal drops below the threshold.
***MODE* Key:** Switch the channel between Gate mode and Expand mode. HOLD this key to change modes. Note that the control markings for Gate mode range are boxed.

Sidechain PEQ Gain Key: Adjust the gain of the sidechain filter to sensitise or desensitise the gate to certain frequencies. ±8/12/20(!) of gain is available, as well as 0db (off).

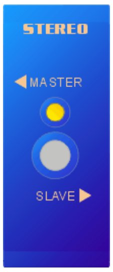
Sidechain PEQ Type: Choose the shape of the sidechain filter between, narrow 'Q' parametric and 'wide 'Q' parametric.

Sidechain PEQ Frequency: Select the frequency range over which the sidechain filter will operate.

High Pass Frequency Control: Cut low frequency content out of the sidechain signal up to 5kHz.

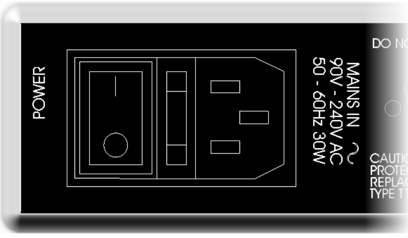
Low Pass Frequency Control: Cut high frequency content out of the sidechain signal down to 250Hz.

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. HOLD to engage this function.

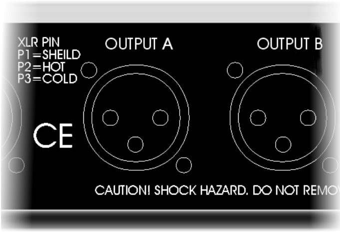


***STEREO* Key:** Enabling the 'Stereo' mode will disable the right hand set of controls and force both channels to assume the parameter values of the left channel. The sidechains are also linked so that the two channels track perfectly, maintaining the stereo image.

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 *G2*

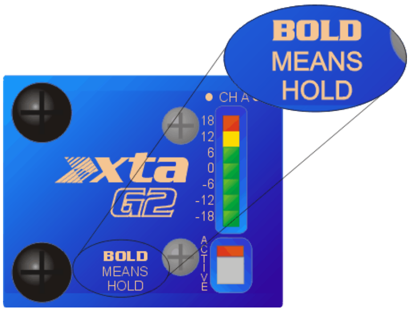
Operation of the *G2* is very straightforward, but there are a few points worth noting which, once understood, will make using the unit even easier.

Switching the unit on and start-up procedure

After plugging in the power and switching the power on using 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, *MODE* (Gate/Expand); and Sidechain EQ Type (Wide/Narrow ‘Q’).

The gain reduction will animate 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…
MODE – switching between gate mode and expand mode (See page 25)
LISTEN – switching the output to monitor the sidechain EQ signal (See page 19)
STEREO – linking the two channels together to operate in stereo (See page 25)

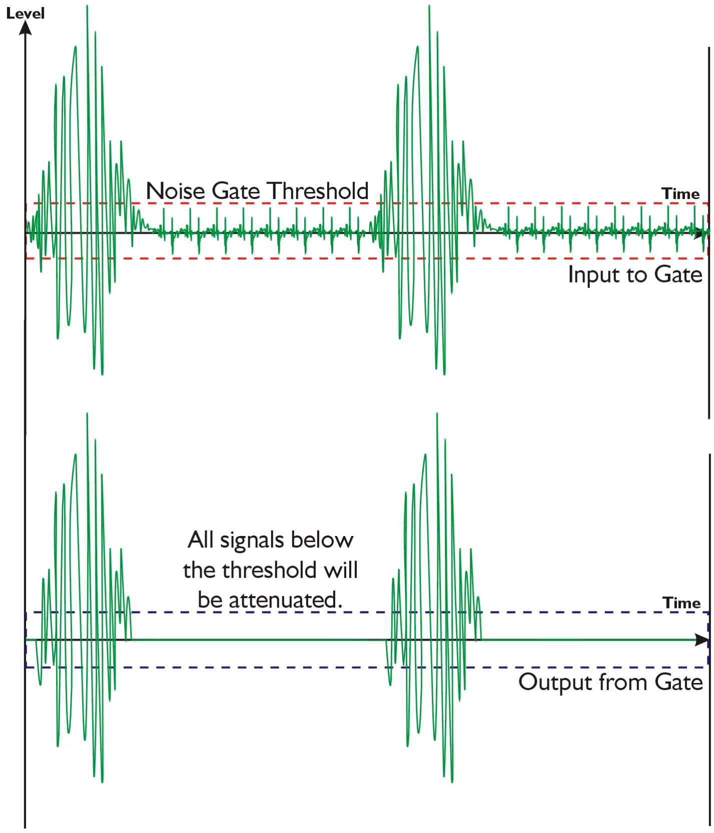
Minimum and maximum control positions

To ensure that the *G2* 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.

Noise Gate Know-how

Most engineers know the principle reason for using a noise gate – to shut off the output of some device that, in the absence of the desired output signal, produces (or passes on) an undesirable level of background noise. The behaviour of the noise gate is controlled by adjusting, principally, its threshold. This sets the point at which the level of the desirable signal is deemed quiet enough to turn it off, thereby cutting off all the background noise as well.



Considering the diagram shown here, the desired signal would be the high level burst and, in between, low level noise. An example of this scenario might be the signal from a bass drum microphone – a burst of signal, but perhaps with unwanted pickup from lighting dimmers producing a background buzz.

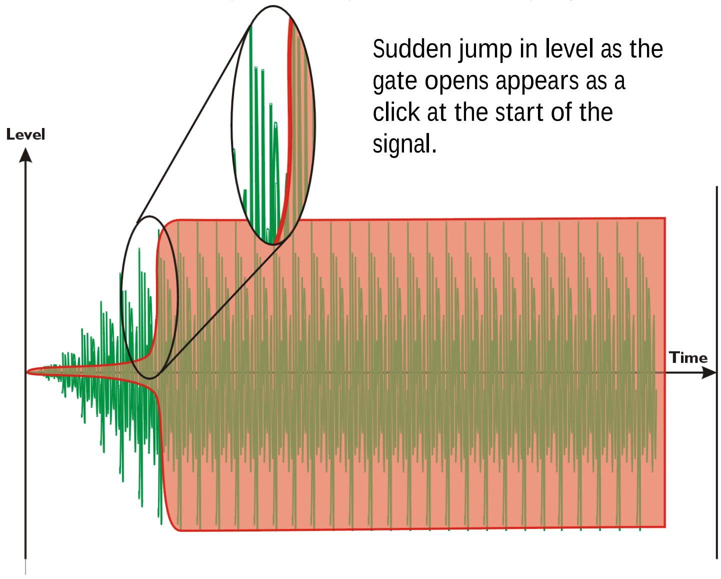
Setting the threshold just above the level of the buzz would keep the gate closed during periods were the bass drum was not being played, only allowing signal through when the output would be sufficient to mask the noise anyway.

The output graph above highlights several other points that are particularly important with noise gates.

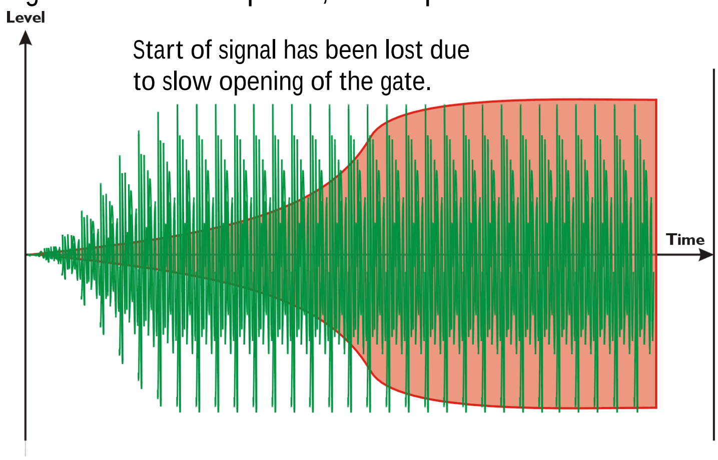
Attack and Release Times.

As the gate opens, there will be a finite delay before the signal crosses the threshold, cutting off part of the start of the desired signal's envelope. Similarly, as the signal drops below the threshold, the gate will close, cutting off some of the end of the envelope. The diagram above is not truly representative of how the signal approaching the threshold would be affected.

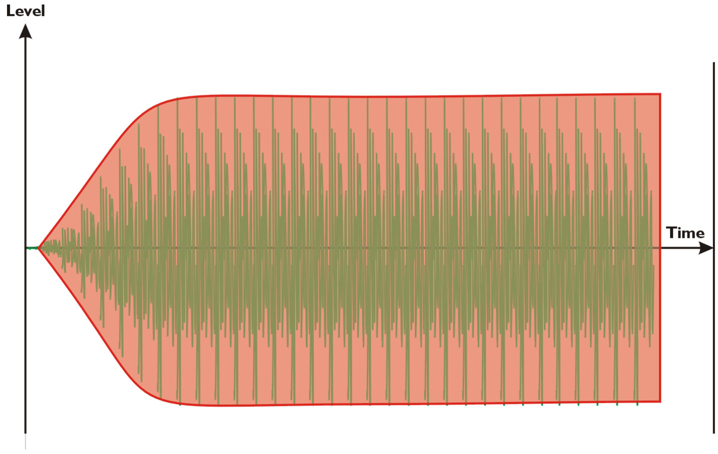
Controlling the speed at which the gate opens (attack time) and closes (release time) is crucial to providing transparent operation. Setting the attack time too fast on a signal that naturally has a relatively slow attack will produce a click. Such a case often occurs when gating the human voice. The attack time for typical speech or singing is in the order of several milliseconds – setting the gate to faster than this will make the voice 'appear out of nowhere' with an audible click. This is due to the sudden change in level from the gate being closed to fully open. The fact that this is a large transient change will make it sound like a click. Changing the 'distance' between the gate being closed and open will remove this click, but also defeat the purpose of the gate in the first place – if the range is set high enough to prevent the click, it's probably not gating the signal enough anyway.



As mentioned earlier, if the attack is set too slow, the gate will not open in time, and part of the signal will be lost. For most musical instruments, the essential part of the signal is the attack phase, as this part identifies the instrument to the human ear. Hearing only the sustain or release of a sound makes it difficult to discern what type of instrument is being played, and sounds very unnatural.



Setting a sympathetic attack time that is similar to that of the required signal will produce the most natural and transparent results. As the diagram below demonstrates, the gate opens at a similar rate to the signal, allowing the full attack portion to be heard.



The same rule applies to the release of the gate – but for slightly different reasons.

Too slow a release, and some of the background noise will be heard as the gate closes; too fast, and the signal will be prematurely cut off, sounding unnatural.

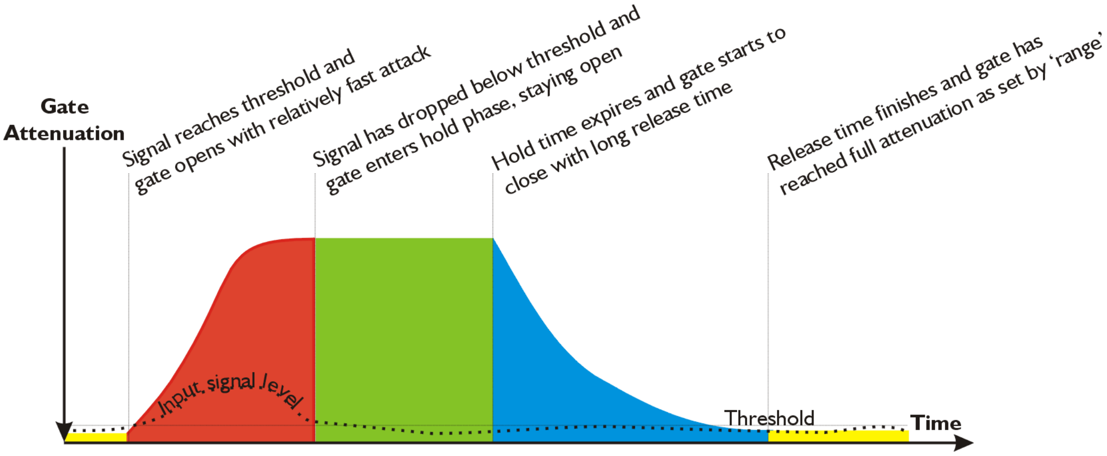
Setting the Range Correctly.

The range control sets how much attenuation is applied to a signal once it drops below the threshold and closes the gate. It is quite often overlooked when setting up a gate, the temptation being to set the range at maximum and leave it. When the gate is fully closed, no signal will be heard when the range is set like this. In some situations, this much attenuation can sound quite unnatural, especially if the threshold is set high to gate a loud signal, due to the stark contrast between the gate fully closed and fully open.

It's also worth remembering that if the background noise is quite severe, removing it all will only highlight the noise more when the gate opens and the noise becomes apparent again. It's often better to set the range to around 25-30dB of attenuation so that it can still be heard very slightly.

Using the Hold Time.

When the gate is being used with transient signals that require fast attack but relatively slow decays, it can be beneficial to introduce a hold time into the gates envelope. This has the effect of prolonging the time the gate stays open, after the signal drops below the threshold. The diagram overleaf explains this concept.



Adding some hold time to the envelope can help with tricky situations where the gate might open and close erratically, such as when there is a lot of background noise. This problem, coupled with a long release, can cause sporadic bursts of noise to open the gate in situations when it would be better to hold it open slightly longer.

Expanders and Expansion – the subtler approach

What does an Expander do?

An expander can be thought of as a complementary device to a compressor. Whilst a compressor is designed to compress the dynamic range of a signal, an expander is designed to expand it. The threshold at which the expander begins to operate sets the point *below* which the signal begins to be attenuated by a certain amount. The fact that signals are only ever attenuated defines the behaviour as 'downward expansion' in nature. This distinction is important because if the expander was to process the dynamic range at a certain threshold, attenuating below the threshold ***and*** amplifying above, even with a low ratio of perhaps 3:1, a limited dynamic range signal might well end up with a dynamic range well in excess of 120dB, which would be impossible to utilise, even with the best equipment.

Why are Expanders necessary?

The main use for an expander is to improve the dynamic range of an instrument or processing device. The dynamic range of any device is limited by two factors – it's minimum output, and it's maximum. It would seem reasonable to assume that the minimum output would be exactly zero, and that this level should be the same for all devices. This is definitely not the case. Certainly, when the majority of instruments and devices are turned off (assuming they are powered in some way) they will all share a similar minimum output level. However, once on, it is the *noise floor* of the electronics within that determines the minimum output. A signal that is below the noise floor of any piece of equipment will not be resolved by that equipment.

The maximum output level is largely determined by the power supply capabilities of the device. Many manufacturers (especially Far Eastern) tend to use power supplies with lower voltage rails, resulting in a poor drive capabilities and limited maximum output level.

As the expander functions below the threshold, it is in the area of minimum output level that improvements can be made. Thinking laterally, the noise floor need not just refer to the electronically generated variety. Noise can be thought of as any "irregular fluctuations accompanying a transmitted signal but not relevant to it"[1].

This may now include other sources of noise such as:

- Hum due to poor power supply regulation or induced by an earth loop
- Background interference from lighting dimmers
- Stage noise picked up from a microphone or transmitted up a microphone stand
- Handling noise from a microphone
- Overspill when 'miking-up' a drum kit

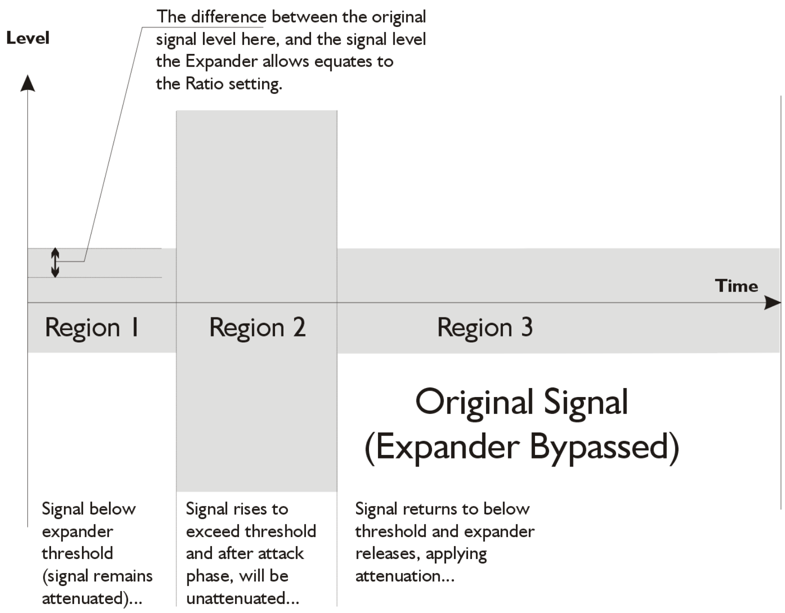
[1] Concise Oxford English Dictionary 2000 Definition 3.

An expander can be used to improve the noise floor under all these circumstances, by attenuating signals below a set threshold, thereby improving the clarity of the signal fed to it.

One other area where expander can prove useful is in the control of feedback in a live situation. In the case of a vocal microphone, leaving it 'open' all the time is likely to cause problems not only with the pick-up of extraneous noise, but allowing feedback paths to build up. Whilst the vocalist is actually using the microphone, their voice effectively stops all other sounds from entering. By attenuating the output from the microphone during pauses, the build up of feedback can be avoided.

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

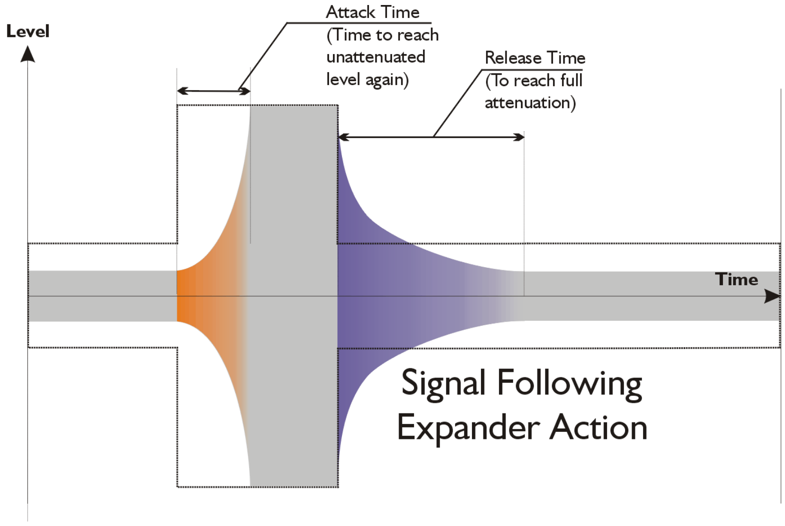
Consider again the signal burst shown below. Regions 1 and 3 are set to below the threshold of the expander, and will consequently be attenuated by an amount equating to the Ratio parameter. In region 2, the signal has risen to a level above the threshold, and so the expander will stop attenuating, and allow the signal to pass straight through. The difference between the attenuated signal when below the threshold and the original signal level (i.e. the degree of attenuation) is set by the Ratio.



Continuing with the compressor analogy again – the ratio can be determined by considering the degree of attenuation required. If the signal drops to 6dB below the threshold, and the ratio is set to 2:1, then the actual output will be 12dB down.

The expander is equipped with a set of envelope controls, which allow adjustment of how it reacts to the signal as it crosses the threshold.

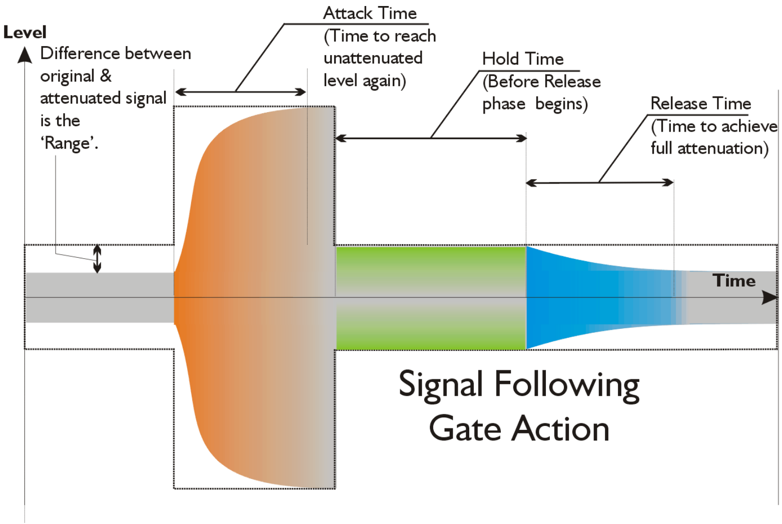
Examining the purple shaded area in region 3, this represents the release time of the expander- that is the time for it to achieve full attenuation by the amount set with the Ratio control, once the signal has dropped below the threshold. The orange shaded area in region 1represents the attack time of the expander. This is the time period for the expander to return to the unattenuated level again, the signal having risen above the threshold. As with a compressor, the setting of the envelope controls is as important as choosing the correct threshold.



Expanders are susceptible to the same abuse as compressors as far as adjustment of the envelope controls is concerned. If the attack time is too fast, there will be an audible ‘click’ as the attenuated signal rises above the threshold and normal (unattenuated) gain is reapplied.

What is the difference between an Expander and a Noise Gate?

In the same way as a limiter can be thought of as a compressor with an infinitely high ratio, a noise gate is an expander with an infinite ratio. As the expander and noise gate share similar controls and ultimately work in the same way, consider the burst diagram below, which demonstrates the action of a noise gate.



As the signal in region 1, is below the threshold set for the gate to be 'open', the output will be attenuated. The amount of attenuation applied is not dependant on a 'Ratio' this time, but on the 'Range' control setting.

This determines the attenuation in dB's, which is applied to all signals below the threshold irrespective of how far below the threshold they are. Whilst the expander gradually applies attenuation to a greater degree as the signal falls further below the threshold, the gate will 'close' below the threshold applying the number of dB's of attenuation set by the 'Range' control. If, for example, the threshold is set to 0dB, and the level drops to –10, if the 'Range' is set to 20dB, then the output will be -30dB: -10 with 20dB of attenuation.

The orange region represents the 'Attack' phase of the noise gate, as the signal rises above the threshold and the gate 'opens'. The green shaded section is the 'Hold Time' parameter. This has been explained in the earlier section 'Using the Hold Time' in the noise gate section on page 13.

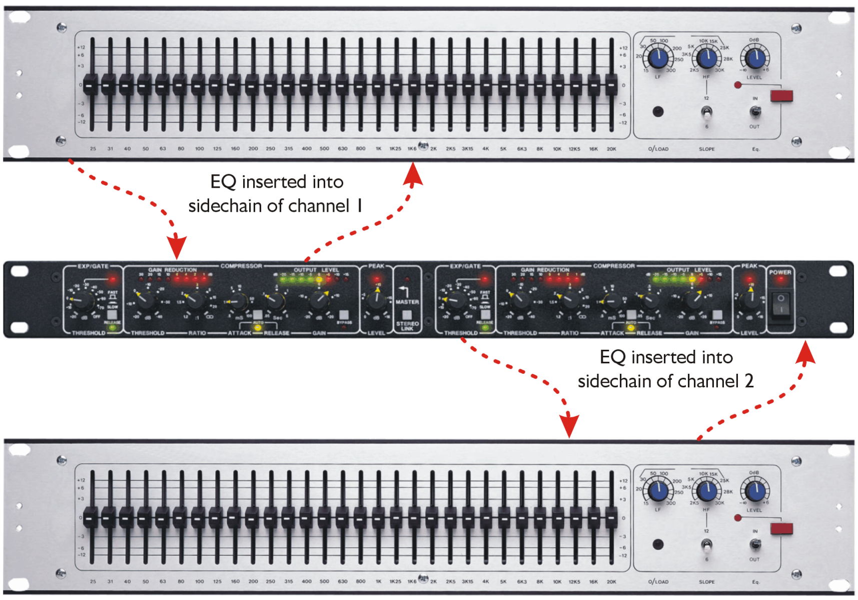
Sidechain Equalisation – How and When to Use it

When would Sidechain EQ be useful?

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

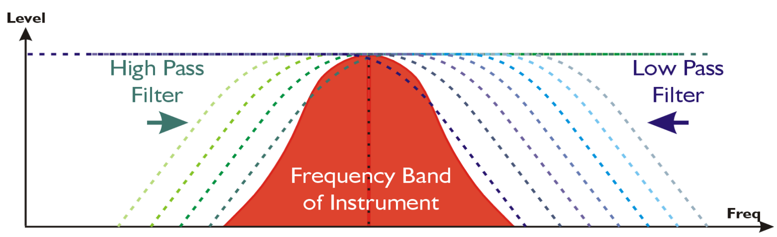
How would sidechain EQ normally be implemented?

Typically, sidechain EQ would be patched into the in/out external path as offered by the dynamics processor. The *G2* obviates the need for this external processing by including both standard high and low pass filters, and a full frequency band of parametric EQ in the side chain for each channel.



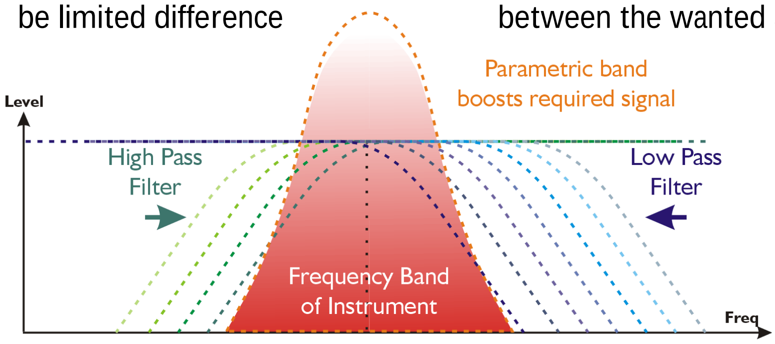
Picking out instruments for gating.

Sidechain EQ is most often associated with homing in on a band of frequencies to improve the performance of a noise gate. Being able to exclude extraneous program material will make the triggering of a gate much more reliable and easier to set up.



The *G2*'s noise gate (and expander) offers not only the high and low pass filters, but an additional band of fully parametric EQ per channel.

This allows the high and low pass filters to be used for the removal of extraneous program material as normal, but also the parametric to 'highlight' any particular band of frequencies. This can prove very useful in situations where there may be limited difference between the wanted and unwanted signals.



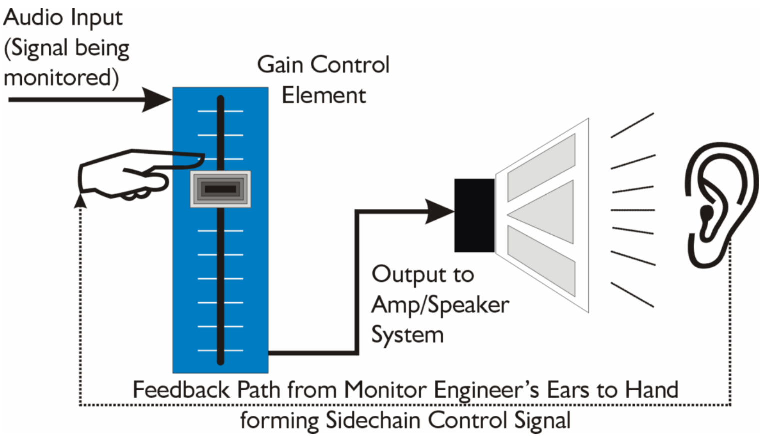
Being able to accentuate the wanted band at the same time as attenuating the unwanted material, especially with the added possibility of highlighting an one area for triggering could prove invaluable.

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[2]. 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.

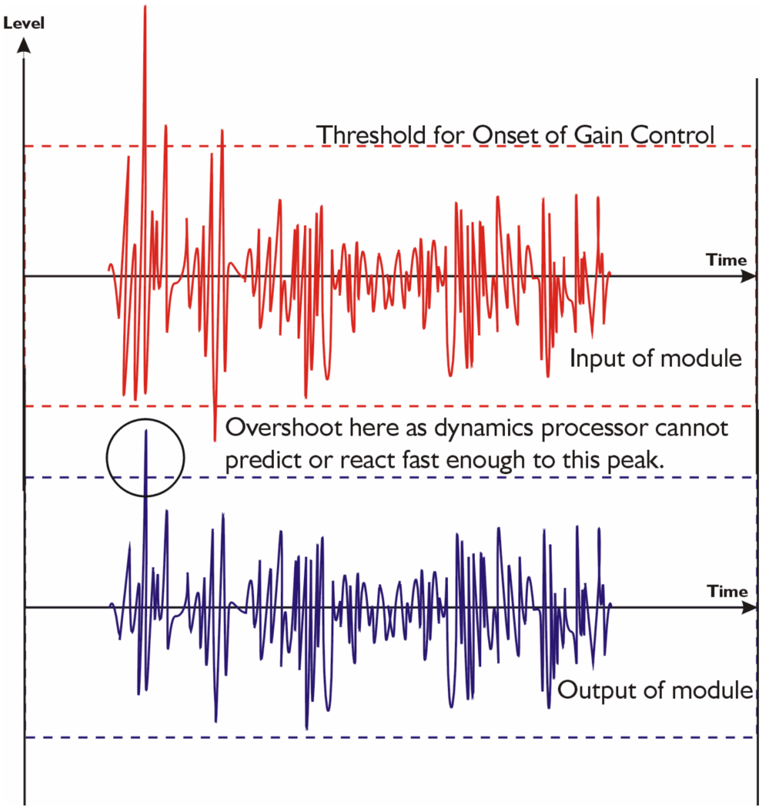


There is a hidden sidechain in operation even in this case. The main signal path is fed through the monitor desk and the gain controlled by adjusting the fader. The sidechain is formed by the feedback path between the engineer's ears checking the level and his brain instructing his hand to turn the fader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

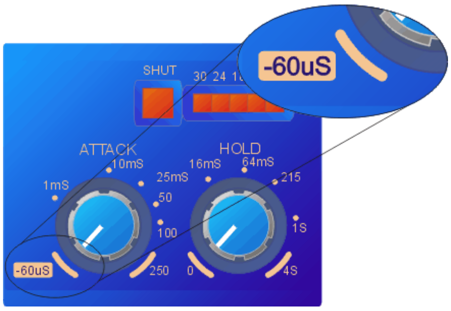
[2] XTA would like to point out that whilst the *G2* might 'sound' male, not all engineers are necessarily male. Some might well be female, or at least have long hair.



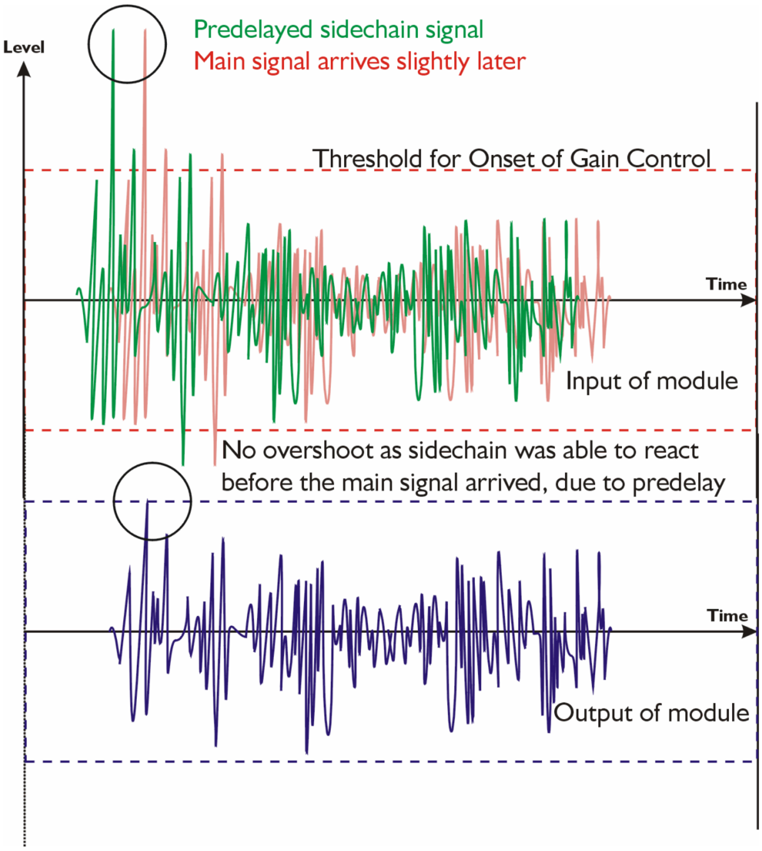
The red waveform represents the input to the dynamics module, with the dotted line showing the threshold for gain control to occur. There are several peaks towards the start of this signal that are above the threshold, and so the dynamics processing should react to these as appropriate. (In this case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

The blue waveform shows 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 *G2*'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.



Setting the attack time control to its minimum setting will force the *G2* 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.

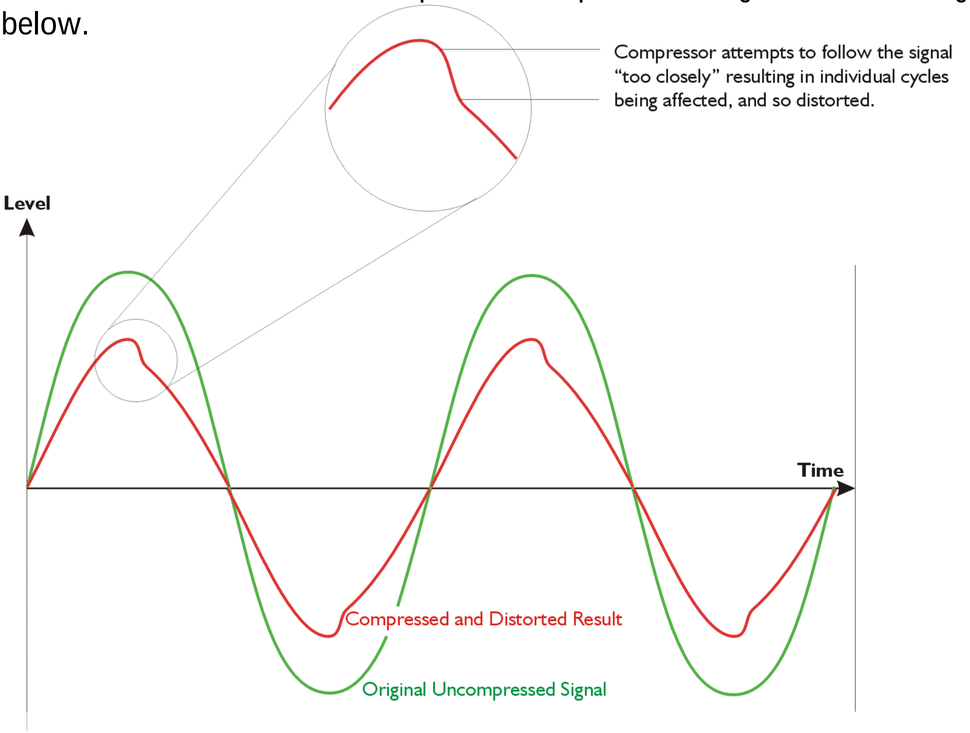
As the main signal arrives slightly after the sidechain, the output from the *G2* does not suffer from the overshoot problem.

Remember that this delay is only in the order of 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 *G2* does not have to use this look ahead delay – in many cases, it is preferable to allow a slower attack. For example, gating a human voice it is often preferable to have an attack time in the order of 10mS to prevent the signal from clicking due to the sudden change in level, and to preserve the natural envelope of the voice.

Setting the Attack and Release times

As with all dynamics processors too fast attack and release times on low frequency program (such as a bass guitar) will cause the process 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. This is illustrated with the example of a compressor being used on a bass guitar signal below.



The gate 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 range settings. 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.

A general rule of thumb when setting the attack and release times would be to set them sympathetically with the original signal's envelope. Some basic settings are given below.

Type of signal	Attack	Release
Human Voice	10mS	200mS
Bass Guitar	5mS (if plucked)	Up to 500mS
Bass Drum	-60uS – 1mS	50mS
Snare Drum	-60uS	Up to 150mS
Strings	150mS	Up to 500mS
Brass	25mS	100mS

Gate and Expand MODE

Pressing and *HOLDING* the *MODE* key will change the operation of the channel between gate and expand modes. The major differences between these are explained below

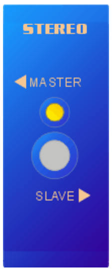
Gate mode

In Gate mode, once the signal has dropped below the threshold, it will be attenuated by a fixed amount, as set with the range control. How fast this attenuation is applied is controlled by the release control. The gate can be set to stay open for a fixed time after dropping below the threshold –this is the hold time. How fast the signal reaches zero attenuation again when it goes over the threshold is determined by the attack control.

Expander mode

In Expander mode, once the signal has dropped below the threshold, it will be gradually attenuated with a ratio as set with the ration control. For example, if the ratio is set to 4:1, for every dB the signal drops below the threshold, it will be attenuated to a ¼ of its normal value.
How fast this attenuation is applied is controlled by the release control. How fast the signal reaches zero attenuation again when it goes over the threshold is determined by the attack control. The hold control has **no affect** in expander mode.

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 opens so will the other, 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

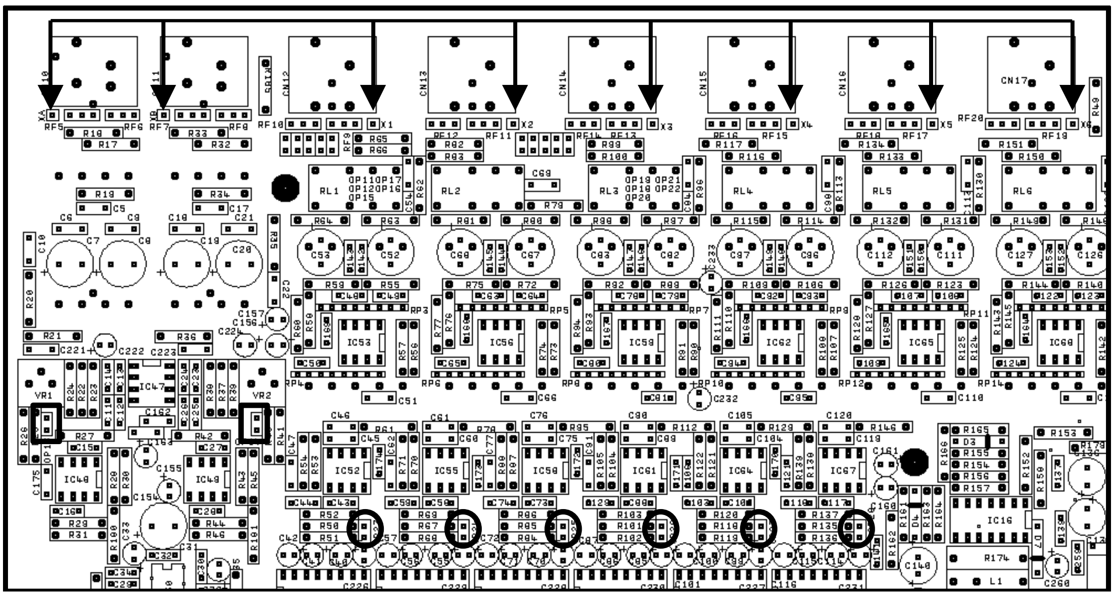
Operating Level

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. 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 *G2* 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

Gate/Expander
Threshold: -40dBu to +22dBu
Range: 0 to 70dB
Ratio: 1:1 to 8:1
Attack: -60uS to 250mS
Hold: 0 to 4 S
Release: 25mS to 2 S

Sidechain EQ Section
High Pass Filter
Type: 12dB/Oct. Butterworth
Corner Freq: 10Hz to 5kHz
Low Pass Filter
Type: 12dB/Oct. Butterworth
Corner Freq: 250Hz to 22kHz

Parametric Section
Type: Selectable narrow (0.8 Oct.)/wide (1.43 Oct.) band-pass.
Centre Freq: 20Hz to 20k
Gain: ±8,12 or 20dB and 'Off'
Sidechain monitor available.

Input meter: 7 point:
-18dBu to +18dBu.
Threshold meter: 5 point:
-16, -8, 0, +8, +16dBr to threshold setting
Gain Reduction meter: 7 point:
<6dB, 6, 12, 18, 24, 30, >40dB

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	*G2* Transformer balanced inputs (factory fitted only)
OTX-100	*G2* Transformer balanced outputs (factory fitted only)
AES-G2	AES/EBU Digital inputs/outputs (factory fitted only)