



for Windows 95/98/ME/2000

Operators Manual Software Version 2.00



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Thanks

Thank you for choosing the XTA  for Windows™ software.

Please spend a little time reading this manual, so that you obtain the best possible performance from your purchase. 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.



Introduction

This manual covers the installation and use of XTA's proprietary software package for the DP324 processor, or SiDD. This software is a remote control application running under Microsoft Windows™, and is currently compatible with DP324's only.

The software is designed to allow all these products to be connected to a computer through a variety of interfaces (RS232/485 or Midi) and will permit full remote control of up to 32 devices.

Please note that SiDD processors may co-exist on the same network as other XTA products, such as the DP224/6 Loudspeaker Management Systems, or DP200 Digital Equaliser/Processors. Our other PC control package, Audiocore, may be used to control these devices. The only caveat is that the total number of devices connected must not exceed 32. Audiocore will ignore connected SiDDs, and vice versa.

For the latest information about XTA's software and product range, visit the website at

www.xta.co.uk.



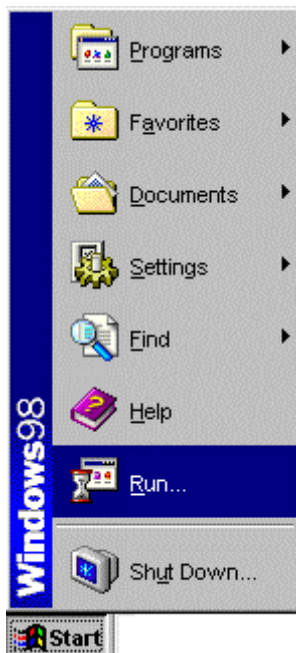
Installation

Please note that this software will **not** run under Windows™ 3.1 or 3.11. It is designed for Windows™ 95/98/NT4.

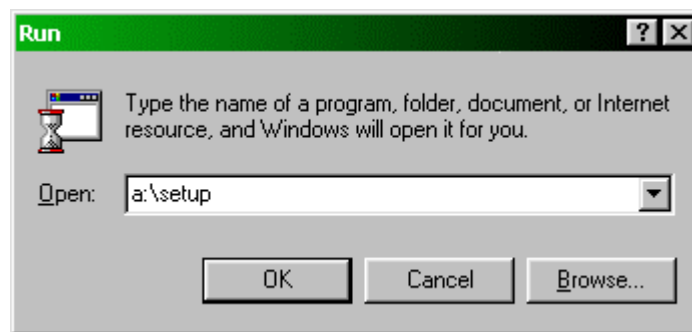
Follow these steps to install your copy of *SiDD*™ for Windows™.

Insert the disk supplied into floppy disk drive A on your computer.

It is recommended that you close all other applications before attempting to install this software. This is true during all software installation, and is not peculiar to this package.

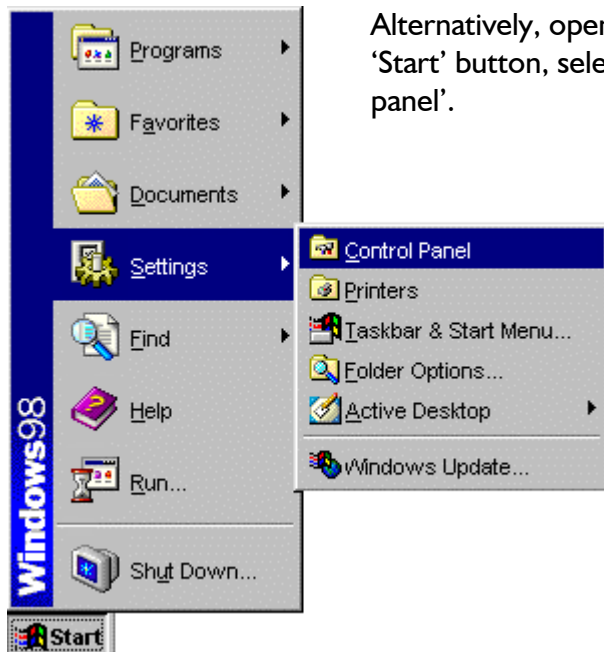


Click on the 'Start' button on the toolbar, and select 'Run' with another click.. Type 'a:\setup' in the text box as shown, and then click 'OK'.

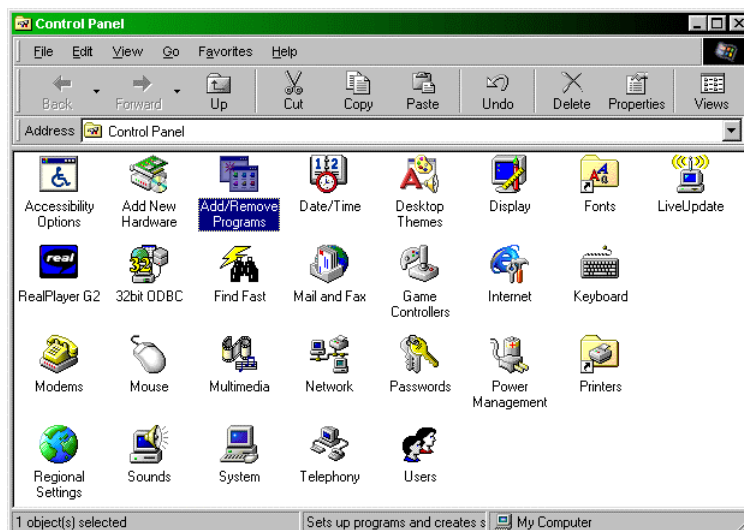


After a couple of seconds the installation will begin. Follow the instructions that appear on the screen.

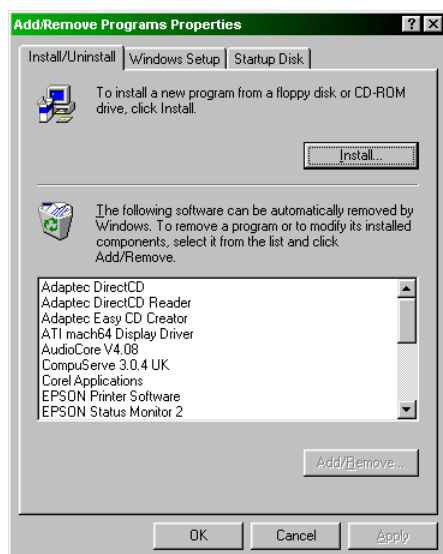




Alternatively, open the control panel by clicking on the 'Start' button, selecting 'Settings' and then 'Control panel'.



From this window, select the 'Add/Remove Programs' icon, and double click on it, opening the window shown below.



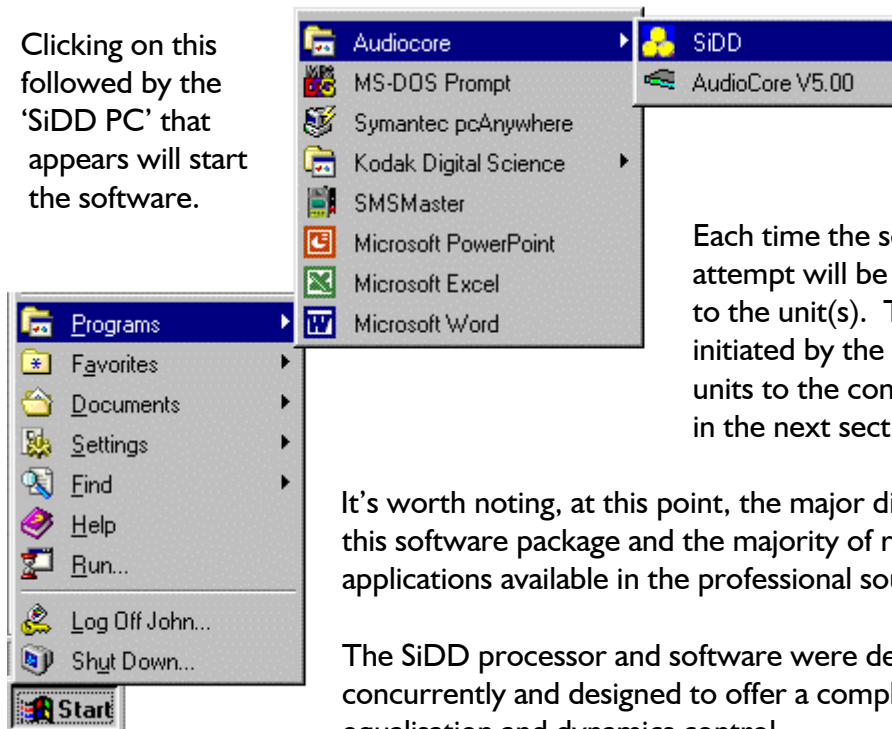
Click on the button marked 'Install...' which will start the software installation wizard. Follow the instructions to install the software.



Starting the Software

The installation procedure will have created an entry in the 'Programs' list off the 'Start' menu. This will be called 'AudioCore'.

Clicking on this followed by the 'SiDD PC' that appears will start the software.



Each time the software starts no attempt will be made to connect to the unit(s). This should be initiated by the user. Connecting units to the computer is covered in the next section.

It's worth noting, at this point, the major difference between this software package and the majority of remote control applications available in the professional sound market.

The SiDD processor and software were developed concurrently and designed to offer a complete solution to equalisation and dynamics control.

To this end, neither the software, nor the unit(s) assumes responsibility as a master when the two are working together. This means that any changes via the PC will immediately update the relevant unit, **and any changes to the unit via the front panel will be immediately reflected on the PC**. Changes made to a unit when the software is 'Off-line' or the PC is not connected will be automatically uploaded when the computer is next connected. In this way, the computer and units always remain synchronised, and there is no possibility of settings stored in one overwriting settings in the other.



Connecting Units to the Computer

Three types of remote interface may be used to connect units to the computer. These are outlined below.

Using Midi...

To output Midi information from the computer, it must be fitted with suitable Midi device and driver software appropriate to that device. A Midiman 'Portman PC/P' can be used to convert the parallel port to a Midi port, if there are not enough spare Com (serial) ports available, or if the computer is a laptop. If this device is used, be sure to select the driver with the 'w/LEDs' extension in the 'Setup' menu, and ensure that the 'Enable Device' box is checked. If the Midi device is internal, such as a Soundblaster sound card, an adapter cable is required to 'break-out' the Midi connections from the 25-way D-connector on the card. Note that the maximum recommended cable run for Midi is 15 metres.

Using RS232...

RS232 uses a serial port connection from the computer (normally a 9-pin D-connector for Com1 ports) and allows remote control of one unit directly. In the case of the DP226, additional units may be cascaded from the RS485 connectors on the rear of the first unit. This may be a pair of XLR connectors, or RJ45 (telephone jack) connectors. Typical set-ups are included in the section following this.

Using RS485...

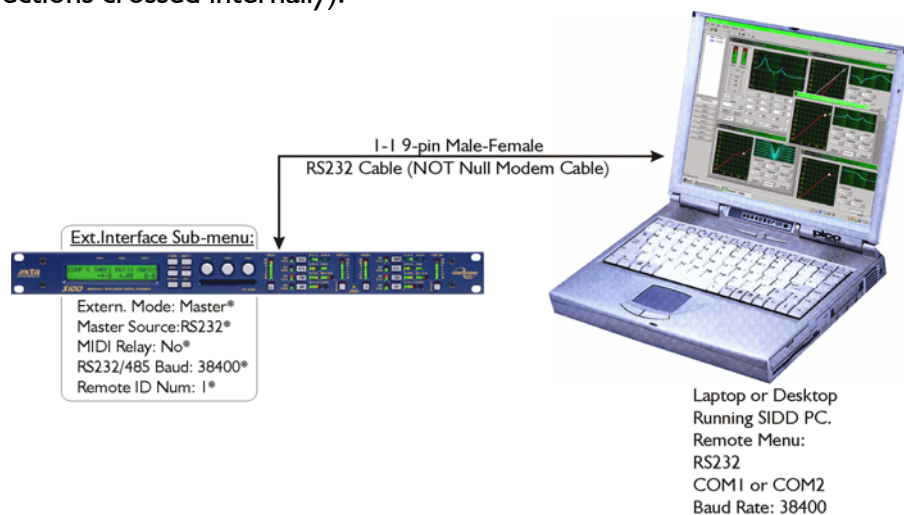
To output RS485 from a computer requires either a converter to connect to the serial (RS232) port, available from XTA, or a plug in card along the same lines as a Midi or Sound-card. The main advantage that RS485 has over RS232 is that the RS485 system is a 'multi-drop' system as opposed to a 'point-to-point' system, so if one unit fails, all other units will continue to operate as no unit is dependant on the one electrically previous to it in the remote control chain for the relay of data. Additionally, the electrical specification of RS485 permits cable runs of up to 1 km before any regeneration is required. The basic specification of RS232 only caters for cable runs up to 25m.



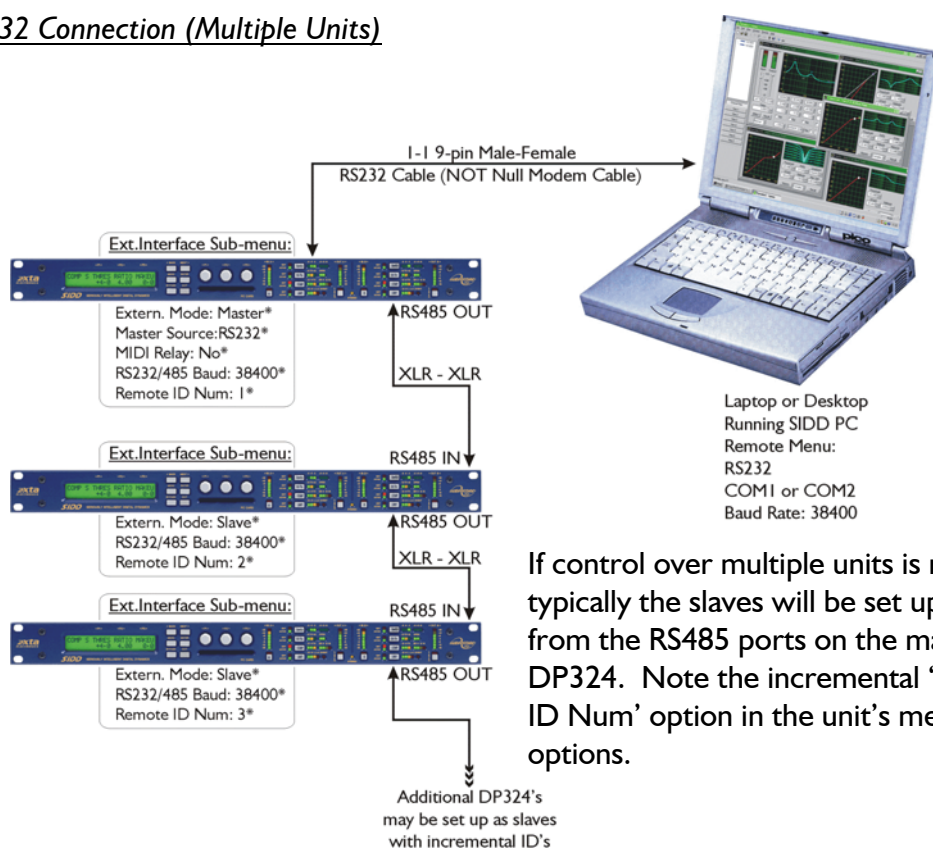
Typical Interface Set-ups

RS232 Connection (Single Unit)

A typical interface set-up might involve running an RS232 link from laptop or a desktop computer to one *SIDD* set up as a master unit. The diagram below shows this method of connection, the required menu options are also given. Note that the RS232 cable must be a I-I connection type, NOT a null modem cable (which has connections crossed internally).



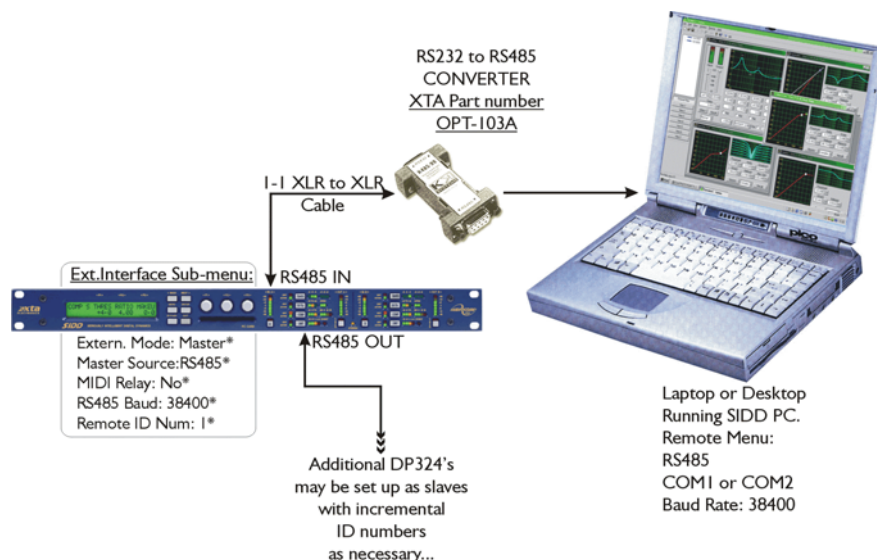
RS232 Connection (Multiple Units)



If control over multiple units is required, typically the slaves will be set up to run from the RS485 ports on the master DP324. Note the incremental 'Remote ID Num' option in the unit's menu options.

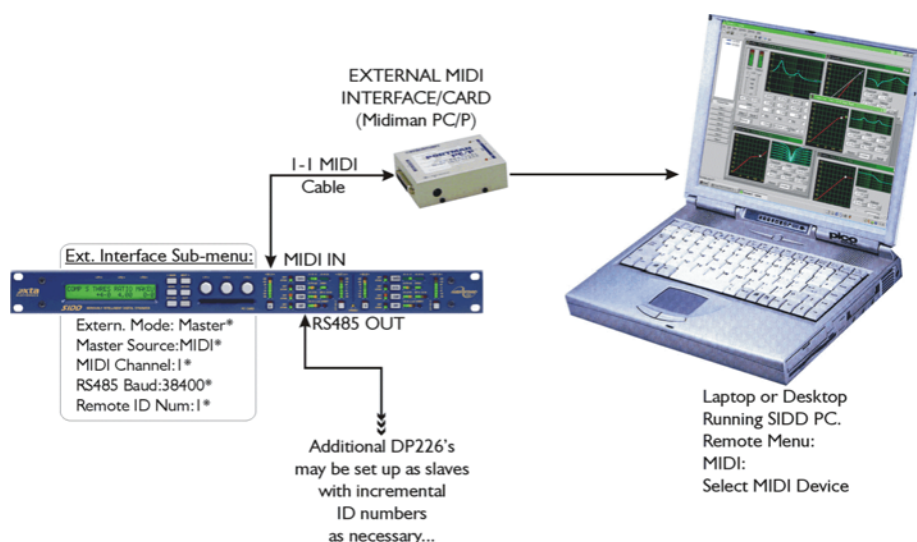
RS485 Connection

To use RS485 communication directly from a computer, a master DP324 must be configured to receive RS485. You must have a suitable RS485 port on your computer, or a converter connected to the serial port in use. This configuration is shown below, along with the required menu options.



Midi Connection

To use MIDI communications, the DP324 must be configured to receive as a master via its MIDI port. You must have a MIDI card or interface connected to your computer. The setup is shown below.



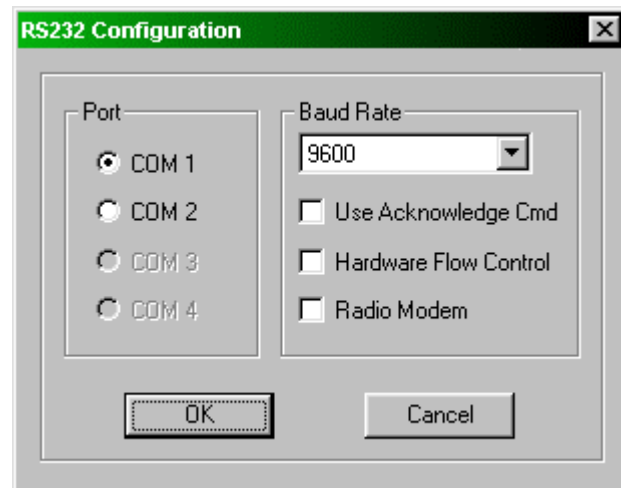
Note that two-way communication is NOT possible via the MIDI interface – only download to the unit is possible, and no settings are returned to the connected computer.



Setting Up Communications and Going On-line

Configuring the Computer and Software

After starting the software, choose Remote ► RS232/RS485 as appropriate from the menu bar. A window will appear containing the configuration settings required for the interface.



In the case of RS232, this is as shown here, with selectable options for the 'Com' (serial) port to use, the speed of this port and options for improving the robustness of the interface, if the port supports them. 'Use Acknowledge Cmd' ensures that every message sent from the software reaches its destination correctly, but at the expense of operational speed. It is recommended to enable this option when long cable runs are used in noisy environments.

'Hardware Flow Control' uses additional physical lines to control the traffic on the port, and is only available with the RS232 interface. If the cable being used to connect the units to the computer does not physically have these additional lines connected, do not enable this option as communications will not work correctly.

Choose the RS485 selection if you plan to connect to the first unit via it's RS485 XLR socket. The window is virtually the same as that shown above, but without the ability to select 'Hardware Flow Control' or 'Radio Modem'.

Important note about interaction between SiDD and other connected units.




Remember that SiDD's and other XTA units can co-exist on the same network, so long as the total number of units connected does not exceed 32. Although Audiocore will ignore any connected SiDDs, they must be set to ID numbers not in use by any other unit. If they share an ID number with another unit, neither unit will appear on the network in either Audiocore or SiDD's software.

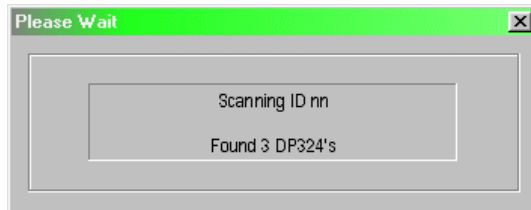


Going 'On-line'

If the remote interfaces have been configured correctly at both the units(s) end and the computer/software end, it will now be possible to go 'on-line'.

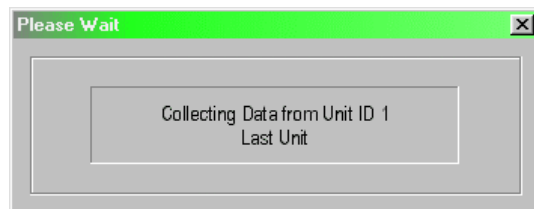
This is achieved simply by pressing the On/Off line button  on the tool bar.

The network will be scanned for any connected units – this is accompanied by a message window as below. The ID number will increment from 1 to 32, and the entire scanning process typically takes about 4 seconds with a baud rate of 38400 bits/S and only one unit connected. The selected communications speed will obviously affect this time - set to 2400 baud, this period is extended to typically 16

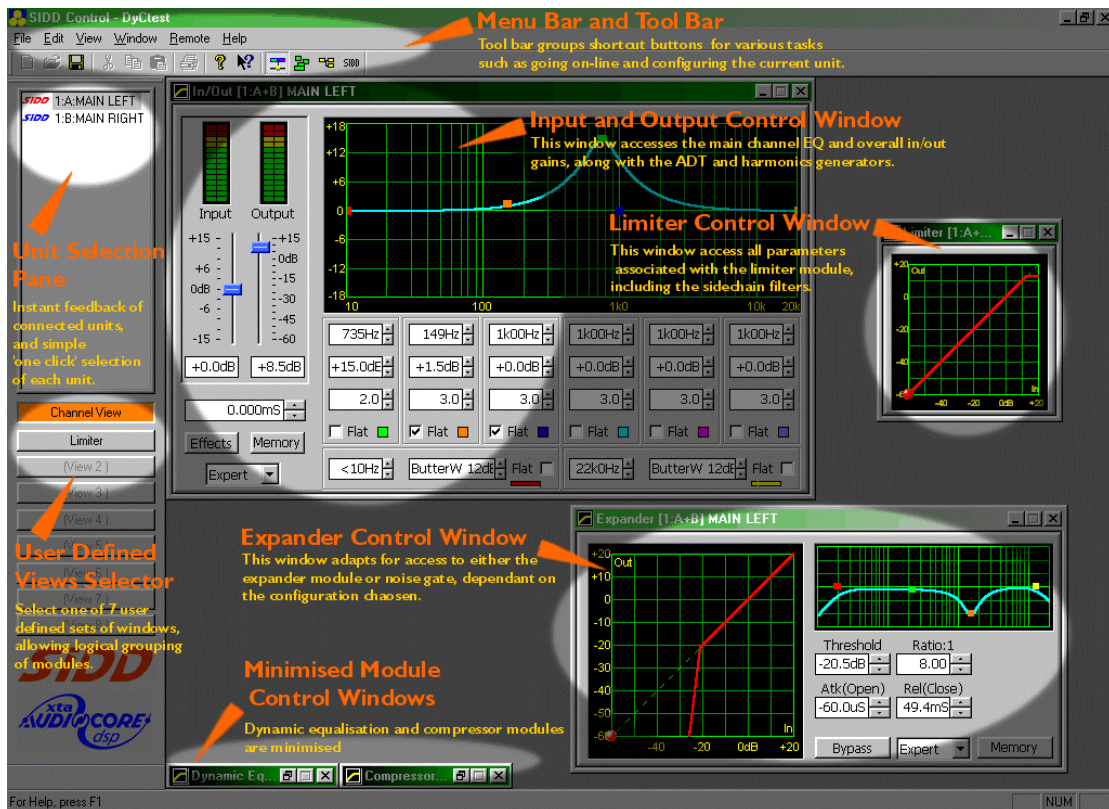


seconds. As units are located, the 'Found' figure will increment. Immediately following the scan, all the units will be interrogated for their settings, and the data collected. This is accompanied by the message...

Once this process is completed (again the slower the baud rate, the longer this will take), all the units found will be listed, channel by channel in the unit selection window. This is shown overleaf.



Breakdown of the Main Screen.



Starting at the top there is the menu bar, and the tool bar. The tool bar consists of...



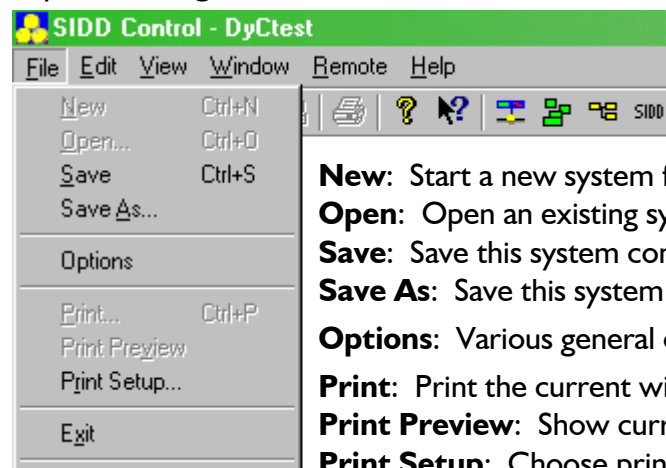
- Open file: Open a different file of previously saved information.
- New file: Start a new file from scratch.
- Save file: Save the current information with the chosen filename.
- Cut settings: Delete settings, copying them to the clipboard.
- Copy settings: Keep current settings, but copies them to the clipboard.
- Paste settings: Overwrites settings with those currently on the clipboard.
- Print information: Print out details of current settings.
- Help: Starts main help system, allowing subjects to be searched and printed.
- "What's This?" help: Instant pop-up help on any screen control.
- On/off-line: Initiate search, synchronisation and connection to chain of units.
- Modify views: Change which windows are displayed in user selectable views.
- Modify configurations: Change expander/gate operation and stereo linkage.
- Audiocore: Switch to Audiocore software (if running – greyed out if not)¹
- Test button: Reserved for future use and factory test functions.

¹ Audiocore version 5.11 or above
SiDD PC Software Version 2.00



Menu Bar and Tool Bar Summary Guides

The main menu items are summarised below, along with references where greater explanation is given elsewhere in this manual.



New: Start a new system from scratch

Open: Open an existing system configuration

Save: Save this system configuration

Save As: Save this system configuration with another name

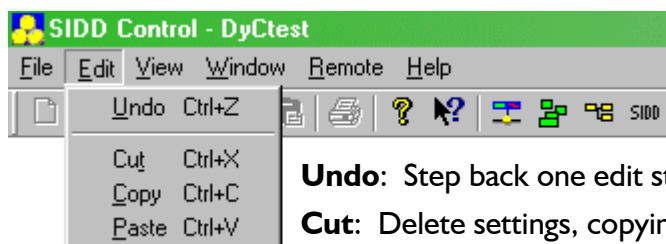
Options: Various general configuration options

Print: Print the current window

Print Preview: Show current window printed page layout

Print Setup: Choose printer and format of print

Exit: Close the software

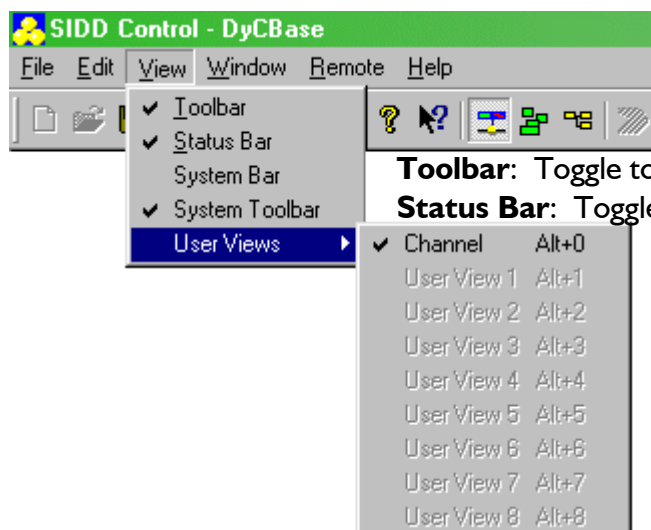


Undo: Step back one edit stage.

Cut: Delete settings, copying them to the clipboard.

Copy: Keep current settings, but copy them to the clipboard.

Paste: Overwrites settings with those on the clipboard.



Toolbar: Toggle toolbar below menus.

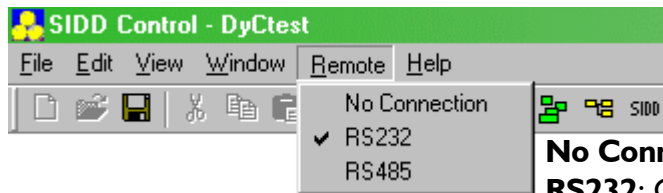
Status Bar: Toggle status bar at bottom of screen.

System Bar: Toggle main channel and views toolbar.

System Toolbar: Toggle main channel and views dropdown menu version of system bar.

User Views: Select view in same way as from System Bar or System Toolbar.






No Connection: Go 'off-line'

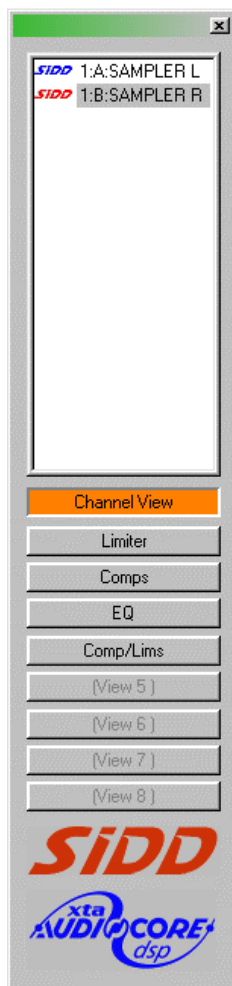
RS232: Choose & configure RS232 settings.

RS485: Choose & configure RS232 settings.



Note that in addition to pressing 'F1' on your keyboard for help, the 'What's This?' help system invoked using  provides additional hints and tips. In addition, right clicking over many sections of any window will invoke context sensitive help relating to that item.

System Bar and System Toolbar



Navigating around the various units connected and selecting which edit windows to see at any time is accomplished using the System Bar shown here to the left, or the System Toolbar, shown below.



These both do the same thing, allowing you to select from the connected units, channel by channel, and the module editing windows, as set up using the Modify Views option. For more information on modifying the user views see page 16.



Function Modes – Expert/Normal/Basic...

To simplify editing in high-pressure situations, or to hide certain parameters from user tampering, SiDD has three levels of editing complexity, known as the function modes. These are designated 'Expert', 'Normal', and 'Basic'.

The differences between each mode are explained in the tables below. The most comprehensive mode – 'Expert' – gives access to all the editable parameters of a module. 'Normal' mode dispenses with the less used features. 'Basic' mode offers only a single parameter screen showing only the most important parameter(s).

Input EQ. etc.		
Expert Mode	Normal Mode	Basic Mode
Input/Output Gain	Input/Output Gain	Input/Output Gain
Main Delay	Main Delay	Main Delay
ADT/Split Delay	ADT/Split Delay	
Harmonics		
High Pass Filter	High Pass Filter	High Pass Filter
Low Pass Filter	Low Pass Filter	Low Pass Filter
Parametric Filters x 6	Parametric Filters	Parametric Filters

Dynamic EQ		
Expert Mode	Normal Mode	Basic Mode
Threshold	Threshold	Threshold
Ratio	Ratio	
Attack		
Release		
Dynamic Filter	Dynamic Filter	Dynamic Filter
Operating Mode		

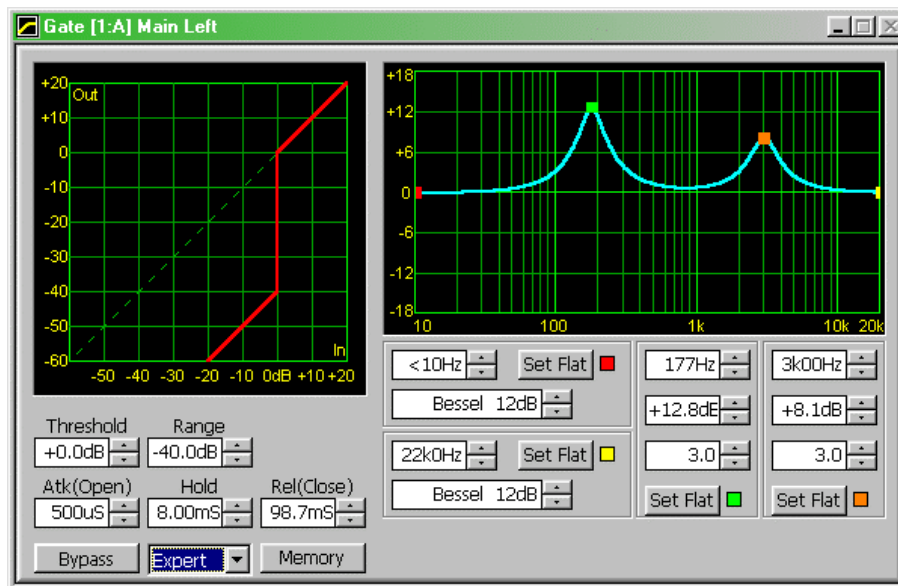
Expander Gate (Gate Mode)		
Expert Mode	Normal Mode	Basic Mode
Threshold	Threshold	Threshold
Ratio (Range)	Ratio (Range)	Ratio (Range)
Attack	Attack	
(Hold)	(Hold)	
Release	Release	
Sidechain HPF	Sidechain HPF	
Sidechain LPF	Sidechain LPF	
Sidechain PEQ x 2		



Compressor		
Expert Mode	Normal Mode	Basic Mode
Threshold	Threshold	Threshold
Ratio	Ratio	Ratio
Make-up Gain	Make-up Gain	Make-up Gain
Attack	Attack	
Release	Release	
Knee	Knee	
Sidechain PEQ x 2		

Limiter		
Expert Mode	Normal Mode	Basic Mode
Threshold	Threshold	Threshold
Overshoot	Overshoot	
Attack	Attack	
Release	Release	
Sidechain PEQ x 2		

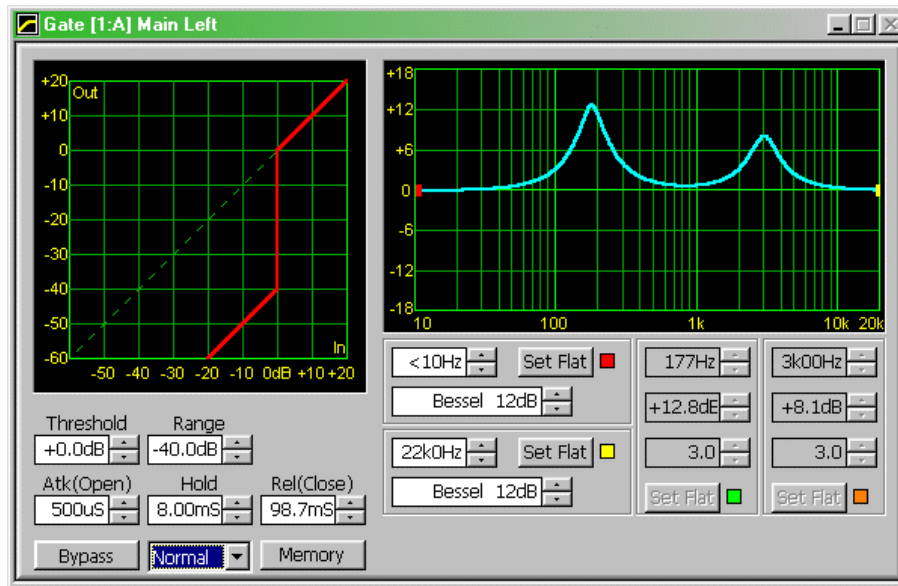
The function mode of any module is selectable via a drop-down list at the bottom of each editing window. Examining the noise gate window, the 'Expert' mode is shown below.



All parameters are accessible as detailed in the table above.

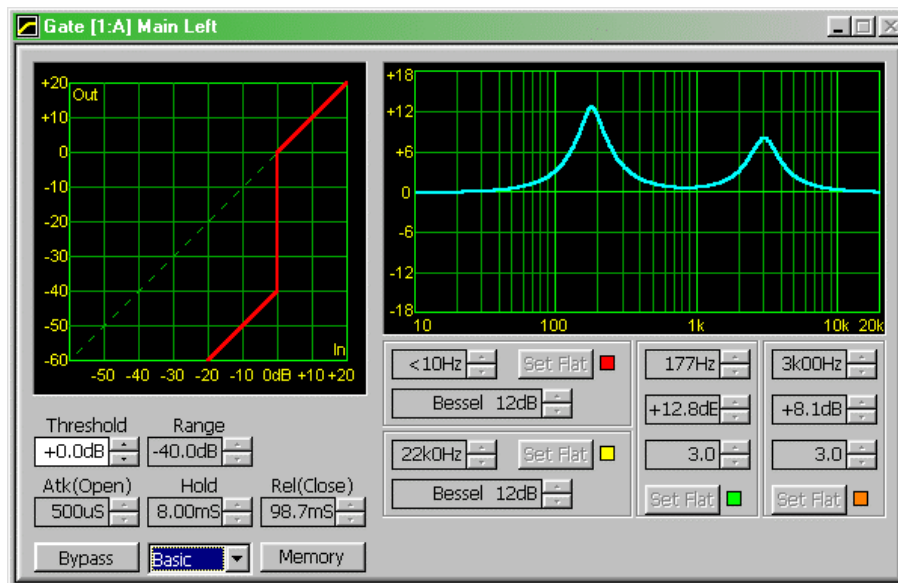
Selecting 'Normal' will grey out some of the parameters as show overleaf.





Note that the coloured nodes are missing on the sidechain frequency response graph, indicating that these parameters cannot be edited.


Selecting 'Basic' limits the available parameters to just the threshold, as shown.



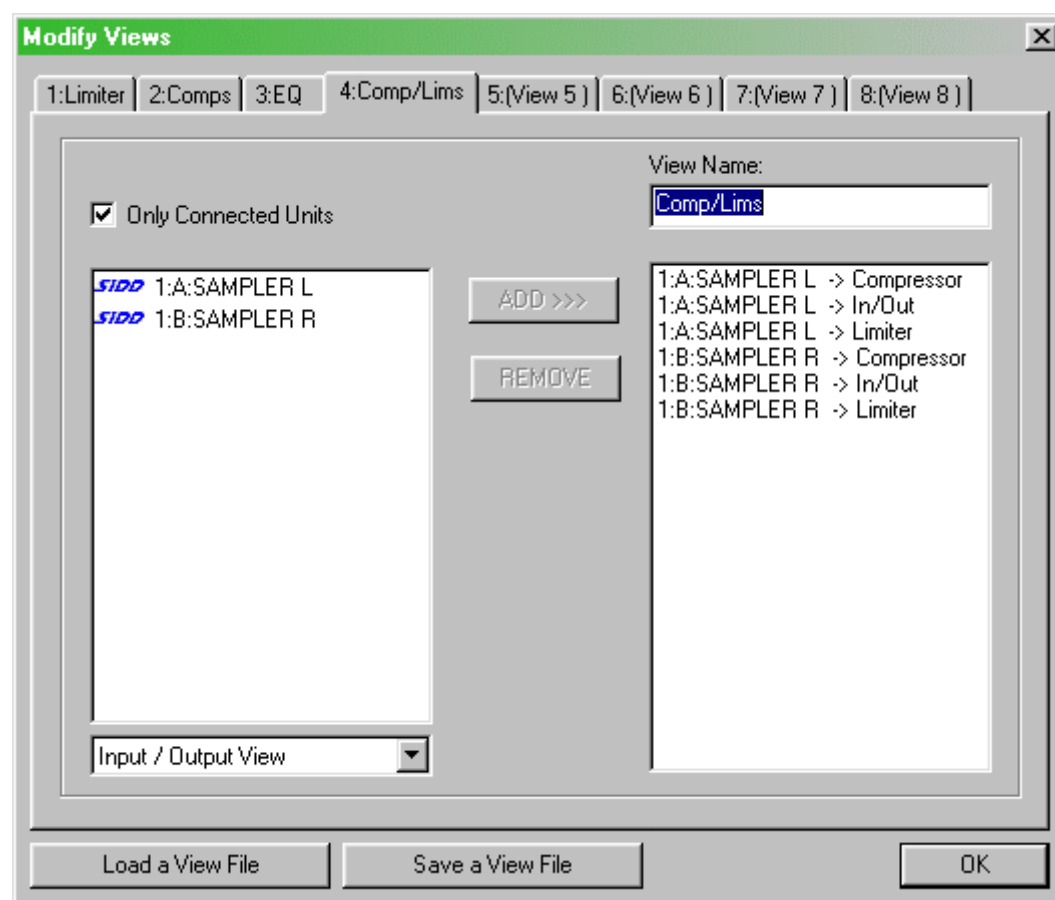
These changes are immediately reflected on the unit as well, with the current function mode for any module shown during editing, on the module summary information screen.



Modifying the User Views

Pressing the 'Modify Views' button on the tool bar -  - will display the window shown below. There are eight tabs corresponding to the eight buttons on the System Bar (in addition to the fixed 'Channel View' button with show all the edit winows for a particular channel and cannot be edited.) Selecting a tab allows the channel of any unit to be accessed using the left hand pane. Below this is the type of window to be included in the view.

So, the process consists of selecting the channel, the window and then pressing the 'ADD>>>' button. This will include the chosen window in the right hand pane. To remove a window from the view, first select in from the list of those already included in the right hand pane, and press the 'REMOVE' button.



Each view can be given a more meaningful name by typing directly in the 'View Name' text box above the included windows list. Pressing 'OK' will update the user views and their respective buttons and drop-down list in the case of the System Toolbar.

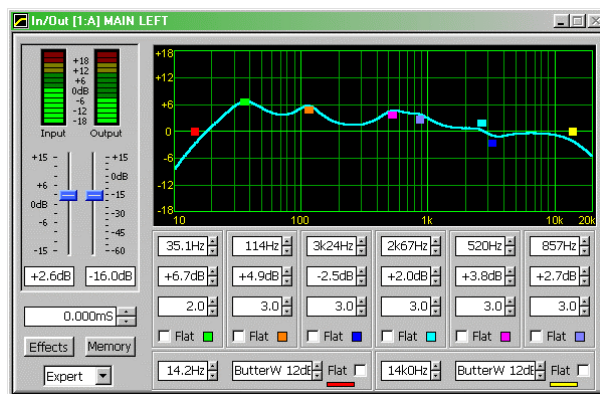
Note that the set of views can be saved by pressing the 'Save a View File' button. This permits the views for as many systems as required to be recalled and saved as required.



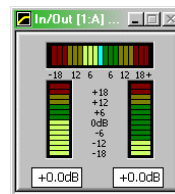
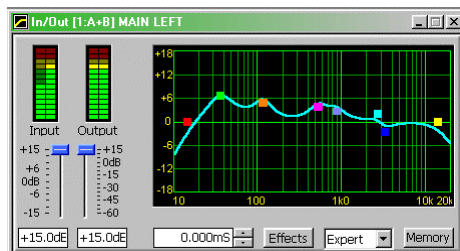
Changing the Size of the Edit Windows

Before examining the edit windows in detail, it's worth knowing that each window can be displayed in three different sizes. This reduces clutter on the screen and makes it easier to monitor several different modules or units at the same time. Using the in/out window as an example, it may be displayed as below...

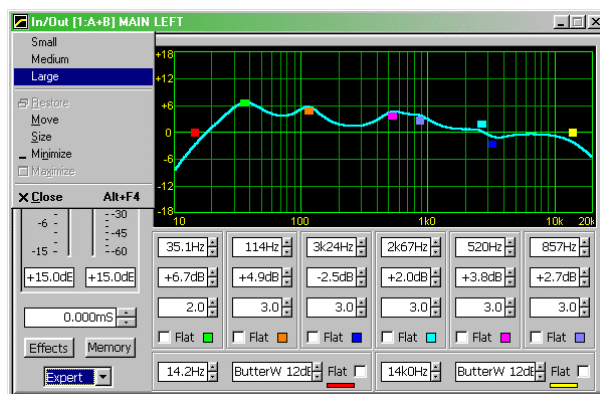
'Large'...all parameters are directly editable;



'Medium'...all parameters still available, but filters adjustable only by curve dragging;



'Small'...only metering available (with threshold adjustment for module if appropriate).



The size is selected by clicking on the icon in the top left hand corner of the window, as shown below.

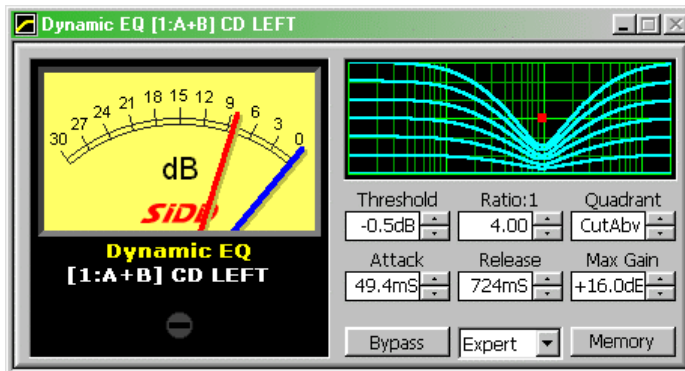
The basic rule is that the 'Medium' size window retains full parameter control at the expense of losing relevant filter details, and 'Small' retains appropriate metering information, and still allows the threshold of dynamics modules to be

adjusted by dragging the appropriate node on the transfer function graph.



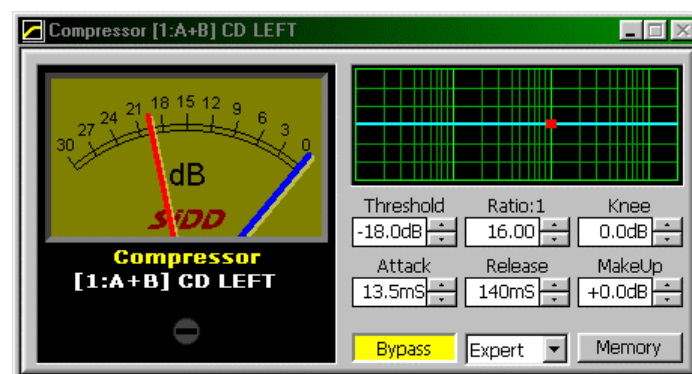
Analogue Metering Mode

The metering mode for the dynamics modules may be swapped between the 'bouncing ball' view with adjustable threshold, and an analogue gain reduction meter mode, as shown below. This example shows the dynamic EQ module's metering (in stereo), with the module active.



Double click on the 'bouncing ball' view will toggle between this view, and the VU meter mode.

Note that the gain reduction will be shown, even if the module is bypassed. The bypassed state is represented by the meter 'backlight' going out, as in the compressor window example below.



This meter mode is ideal for setting up a selection of views forming a monitoring panel.



Modifying the Unit's Configuration

Pressing the 'Configure Unit' button on the toolbar -



- allows various aspects of the currently selected unit's configuration to be adjusted.

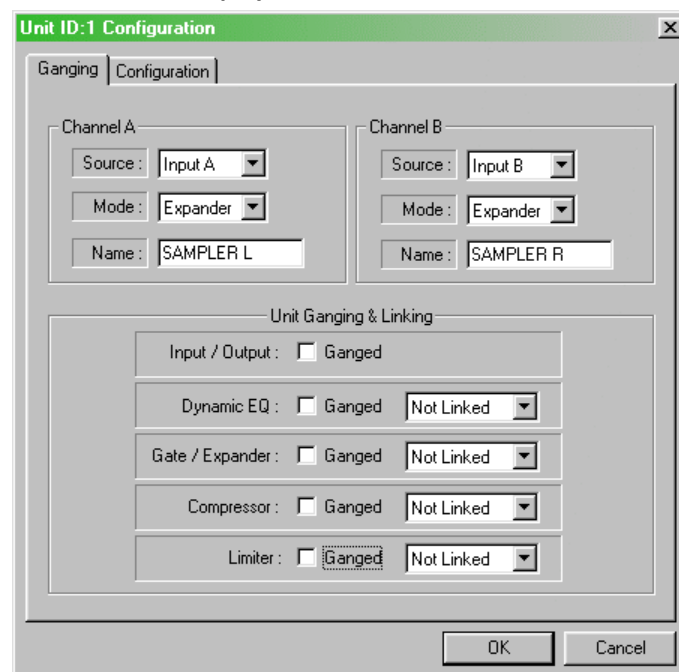
These include the ability to:

- ◆ Change the name of the channel, as displayed in the software and on the unit;
- ◆ Choose either Expander behaviour or Gate behaviour per channel;
- ◆ Choose the signal source for the output;
- ◆ Gang modules on both channels on the same unit together for parameter tracking;
- ◆ Link module sidechains on both channels on the same unit for stereo tracking...

...and, with the preset configurations available, the ability to:

- ◆ Choose ADT or SPLIT format for the unit;
- ◆ Position the gain control pre or post-limiter;
- ◆ Route the sidechain listen signal to the main or auxiliary outputs;

The window displayed is shown below.



There are several important points to be made about the configuration parameters available here.



Source: This selects the complete input, which will be processed by the channel. By default, each input feeds its complementary output – A to A, B to B. However, the inputs can be swapped if required, or summed (A+B option). Additionally, both outputs can be fed from the same input allowing, for example, a four output delay to be realised.²

Mode: Independent selection of the expander/gate module for each channel allows the behaviour to be changed between a downward expander or a noise gate.

Name: The channel's name can be easily changed here. This name appears in the System Bar (and System Toolbar) lists, and all the associated edit windows. It also immediately updates the unit with the channel name.

Unit Ganging & Linking: The operation of each module can be adjusted so that it shares parameters with the same module in the other channel for the unit, and/or shares the sidechain signal. Please read the following section, which explains these concepts in detail.

What is the Difference Between Ganging and Linking?

These two terms are easily confused and justify a section explaining what each means before dealing with the options themselves.

On a traditional dynamics processor, if it has at least two channels, there is often the ability to 'slave' one channel to the other. Under these circumstances the left hand channel normally becomes the 'master'. The parameter controls for the 'slave' channel become redundant with both channels assuming the parameter settings of the master channel.

Stopping at this point – this is what the 'Ganging' on *SIDD* accomplishes. The parameters are linked from one channel to the other, so that any adjustment of one module will set the same parameter on the other channel's module to an identical value. Adjustments are absolute – that is no offset will be maintained between the same parameter on each module. If, for any reason, an offset is required gang the modules, set the one channel, then ungang and introduce the offset value.

Parameters will only assume the same value once they have been adjusted in the stereo gang mode. For this reason, it is best to enable the stereo ganging prior to any editing to ensure all parameters are the same.

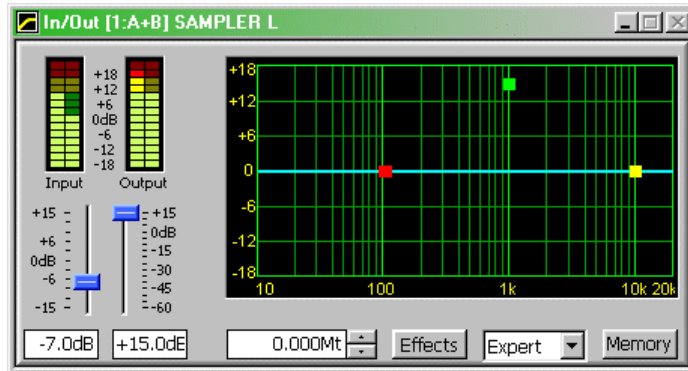
Considering the traditional dynamics processor again – pressing the 'Stereo' or 'Slave' button not only links the controls on the front panel, but connects the sidechains of the two channels internally. The reason for this is to maintain the same degree of processing on both channels when treating a stereo signal. So, for example, if in the case of a stereo limiter arrangement, one channel begins to limit by 3dB, the same

² Set source to 'A' for both channels and select 'SPLIT' mode.



amount of limiting will be applied to the other channel. This prevents shifts in the stereo image that would occur if the level dropped in one channel but not the other.

On the unit being edited, stereo ganging is also indicated on each modules' window title bar. The title changes to read...

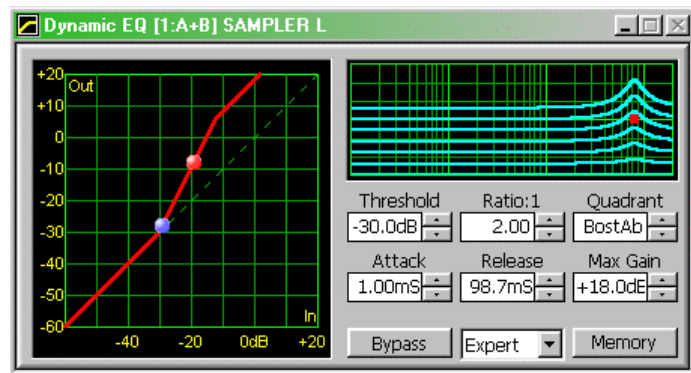


[1:A+B] meaning ID number 1 and channels ganged [A+B].

Note that, in the case of the input/output window, the meters are also displayed in stereo.

On all other windows (i.e. dynamic processors), the 'bouncing ball' for both channels will be displayed together, like this.

On the unit itself, the summary information screen – the channel ID changes from 'A' or 'B' to 'S' for stereo.



Linkage is only indicated in this configuration window, but is also shown on the relevant module's summary information screen, by an "x" with a bar over it, like this.



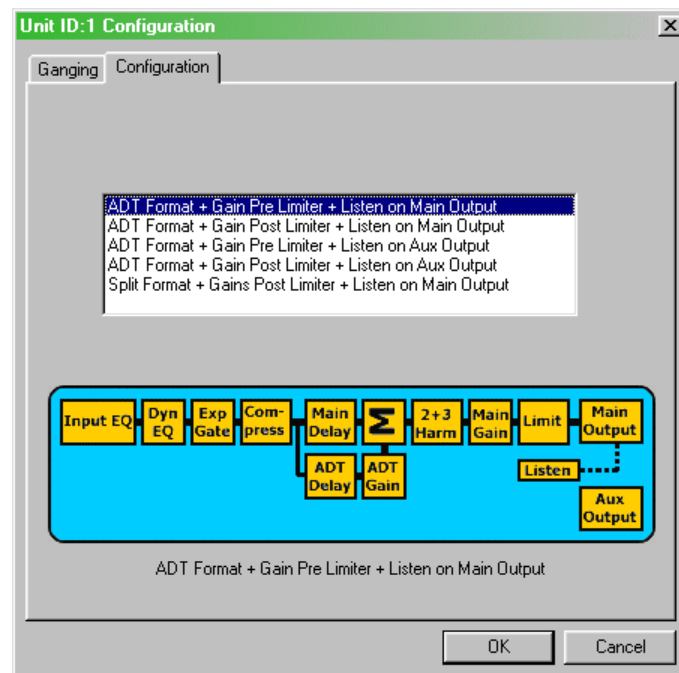
Summarising the difference between the two functions:



(Stereo) Ganging Links Parameters
(Function) Linking Links Sidechains.

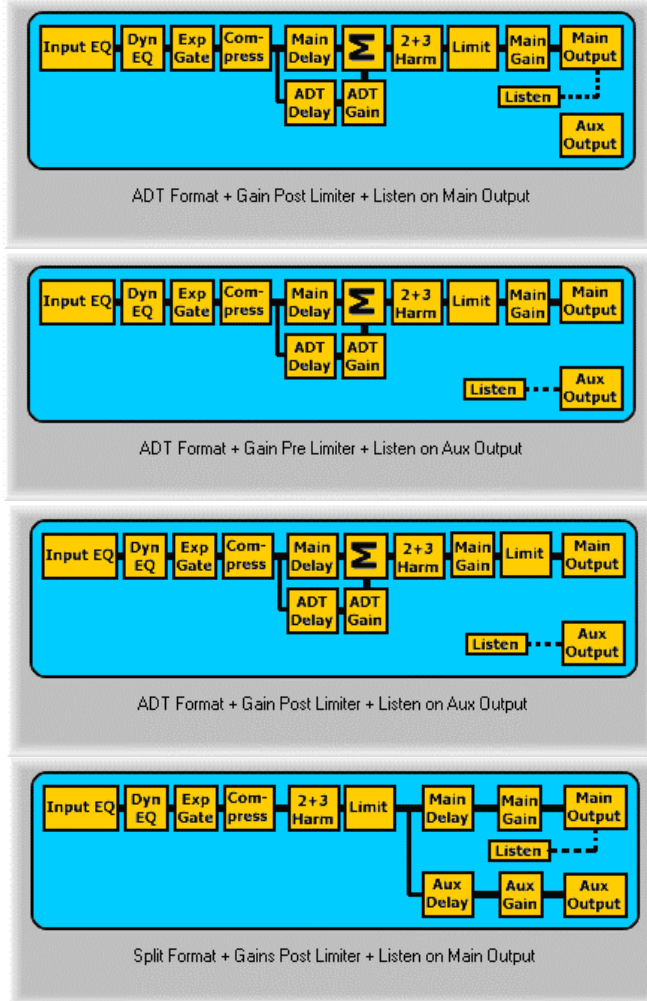


Choosing the Configuration

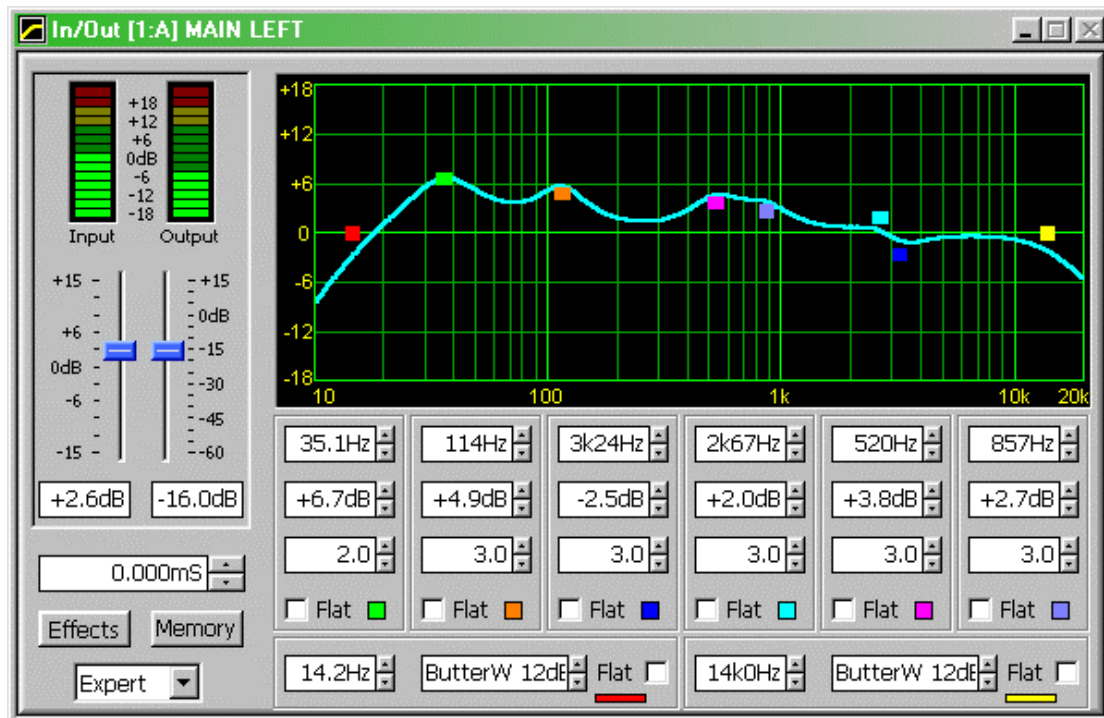


SiDD offers five configurations which allow various modal permutations.

These are listed in the pane as shown, and each has an accompanying routing diagram to clarify how the signal path changes.



Editing Parameters on the In/Out Window



This window accesses all the parameters accessible under the channel button on SiDD – the 'A' or 'B' button. These include the input and output gains, the six parametric EQ bands, the high and low pass filters, the delay line and the harmonics generator.

Input/Output Gains

The input and output gains for the channel (or channels if A & B are ganged – see page 21 for details on how to achieve this) are adjustable via the two faders shown above. In stereo, there will be a pair of meters shown for input level and output level.

Note that the input metering is *pre-processing* – adjusting the input gain fader will *not* affect the input headroom, so the input meter will not change. Most likely the output meter reading will rise, unless there is a large amount of compression or limiting being applied to the signal. If the input meter is reading too high, reduce the signal level fed to the unit accordingly.

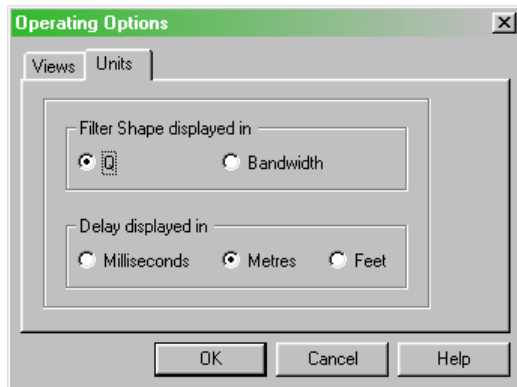
Other points to note about the operation of this section are:

- ♦ Clicking on the gain slider track, rather than the knob will change the gain in 1dB steps.
- ♦ Clicking on the text box below the gain slider allows direct numerical entry. Press ENTER to accept the new value. To reject a value typed in accidentally, delete all text from the box and click on any other box. The gain will revert to that shown by the slider.



Main Delay

The main delay time may be adjusted using the nudge buttons beside the readout directly below the input and output faders. Clicking in the box below the gain slider allows direct numerical entry. Press ENTER to accept the new value. The units used for the readout may be changed from time (milliseconds) to distance (metres or feet). This facility is accessed by selecting **File** ► **Options** from the menu bar, displaying the window below.



Select the 'Units' tab as shown and then choose the desired readout units from the lower pane.

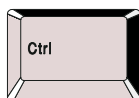
Parametric Filters, High and Low Pass Filters.

The EQ curve is fully interactive – just click and hold a coloured node corresponding to the band to be adjusted, and drag it – the frequency and level will adjust in real time as it is moved. To adjust the 'Q' (or bandwidth) of a filter using the mouse, select the node and, holding the left mouse button down, move the scroll wheel.³

The keyboard may also be used to adjust filter settings, not just on it's own, but also to enhance the feature available during curve dragging.



Holding down 'Shift' while dragging locks the frequency of the filter, whilst allowing the gain to be adjusted.



Holding down 'Ctrl' while dragging locks the gain of the filter, whilst allowing the frequency to be adjusted.

If your mouse does not have a scroll wheel, then the page up and down buttons may be used to adjust the 'Q' during curve dragging.



Whilst a filter is selected, (left mouse button down) pressing 'Page Up' will increase the filter 'Q' by one step.



Whilst a filter is selected, (left mouse button down) pressing 'Page Down' will decrease the filter 'Q' by one step.

³ An 'intellipoint' type mouse is required for this operation.



Selecting File ► Options and then the 'Units' tab allows the units displayed for parametric filter bandwidth to be changed from 'Q' to actual bandwidth, in octaves.

For accurate adjustments of the filters, use the nudge buttons positions beside each parameter readout. Selecting the associated 'Flat' checkbox by a filter will set its gain to zero, but leave the 'Q' and frequency unaffected – the coloured node will remain in place.

Parametric or Shelving Filter Behaviour

Any parametric filter may be changed to a shelving type response by setting the 'Q' control one nudge past the widest 'Q' value. This will select 'LoShelf', followed by 'HiShelf'. Note that the gain of the filter must be set to 0dB to access these responses types. If the filter is 'flattened' using the associated tick box, the gain will be automatically reset to 0dB should the response be changed from parametric to shelf.

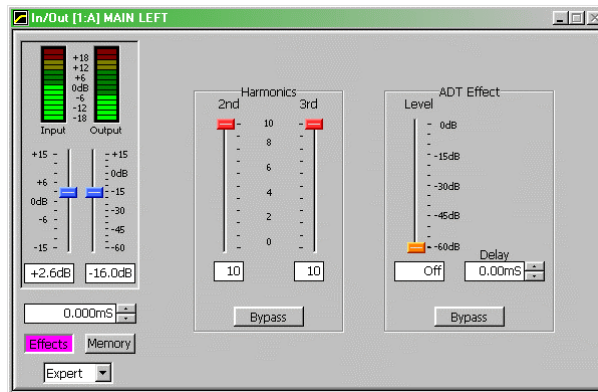


Holding down a frequency nudge button will start moving the filter in 1/36 octave steps (the highest resolution), and accelerate to 1/6 octave, and finally 1/2 octave steps. Leave go near the desired frequency and nudge in single clicks.



Harmonics Generator and ADT/Aux Delay Controls.

Pressing the button marked 'Effects' immediately below the main delay text box accesses additional controls for the harmonics generator and secondary delay line.

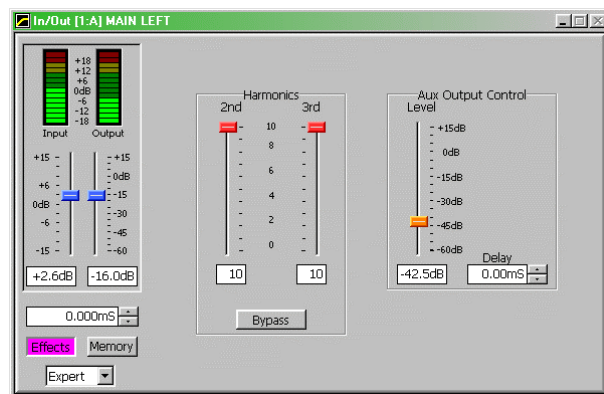


The second and third harmonics content may be individually adjusted using the two faders shown in red. The entire harmonics generator may be bypassed using the 'Bypass' button.

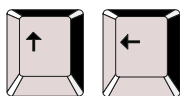
The right hand pane will be titled either 'ADT Effect' or 'Aux Output Control' depending on the chosen configuration settings. (To change

configuration settings, see page 21 for details.) The ADT level is adjustable from -60dB up to 0dB; for the auxiliary delay level, the range extends +15dB above 0dB. A bypass facility is only available in ADT mode.

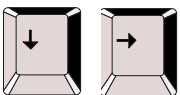
Note that if the level fader is pulled right down to minimum it will turn the effect off completely.



The keyboard may also be used to navigate around parts of this screen.



Adjust **gain up** in 0.1dB steps, **aux/ADT level up** in 0.5dB steps, or **harmonics up** 1 step.



Adjust **gain down** in 0.1dB steps, **aux/ADT level down** in 0.5dB steps, or **harmonics down** 1 step.



Adjust **gain up** in 1dB steps, **aux/ADT level up** in 3dB steps, or **harmonics up** 1 step.



Adjust **gain down** in 1dB steps, **aux/ADT level down** in 3dB steps, or **harmonics down** 1 step.

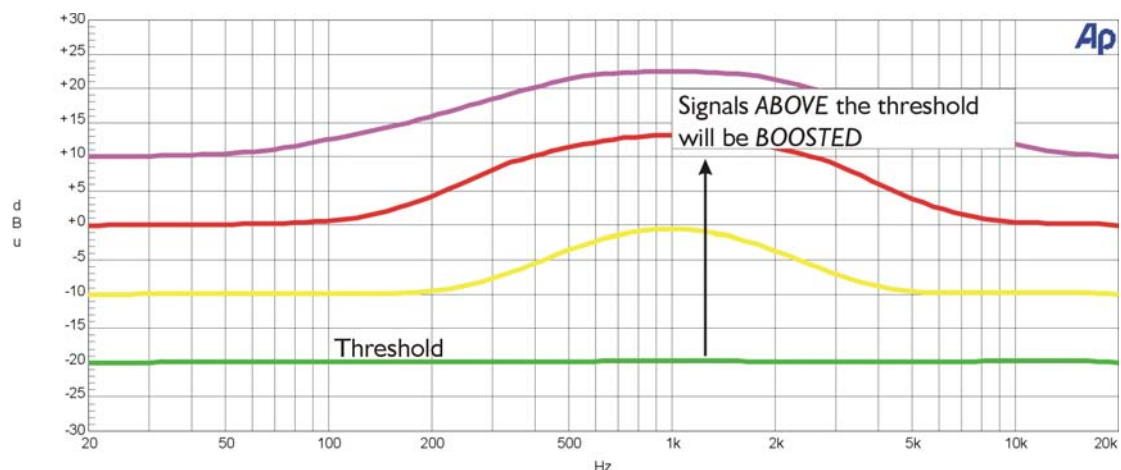


What is Dynamic EQ?

Dynamic EQ is essentially a compressor or expander that can be set to respond and act upon only a certain range of frequencies. Its behaviour is dependant on the operating mode chosen – two of these are relatively traditional, whilst two modes offer the possibility to turn the normal action of compressors and expanders on their head to allow innovative adaptive control of the program material. The four operating modes are explained in detail below.

Quadrant/Mode I: “Boost Above”

This is the other less than traditional mode of operation, offering upward expansion, where the signal is boosted once it reaches the threshold. The example below shows that 1kHz filter again, this time with the threshold at +10dB. As can be seen, as the signal rises above the threshold it is progressively boosted around the 1kHz region.



Uses of “Boost Above” mode.

This mode is more useful than it might first appear – the ability to add EQ only at higher signal levels allows some very effective emphasis of certain parts of the spectrum to be added, without the side effect of a permanent audible peak.

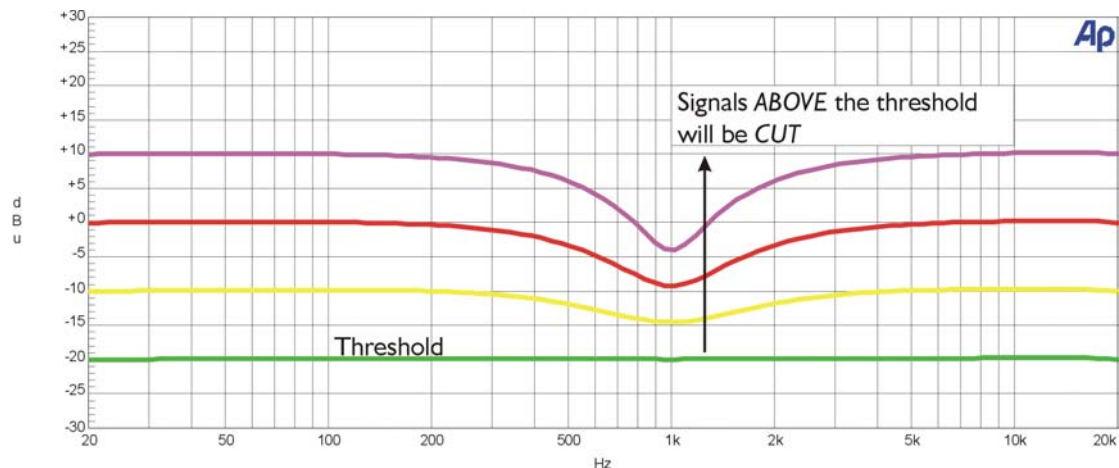
Adding some ‘top end sparkle’ – try picking out high-hats and cymbals with a filter at 12kHz, ‘Q’ of 1 Octave, and fast attack and release, typically 5mS and 25mS. This gives a significant boost to the top end, without bringing up noise in the absence of any high frequency content.

Similarly, ‘punch’ can be re-introduced to a lacklustre bass drum by setting the filter to about 80Hz and slowing the attack to 49mS and the release to 100mS. As the bass drum causes the EQ only to be applied on peaks, there is no additional muddiness added to the bottom end of the spectrum.



Quadrant/Mode II: “Cut Above”

This is one of the more traditional modes of operation. 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 , with a ‘Q’ of 1.0 .



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 “Cut Above” 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.

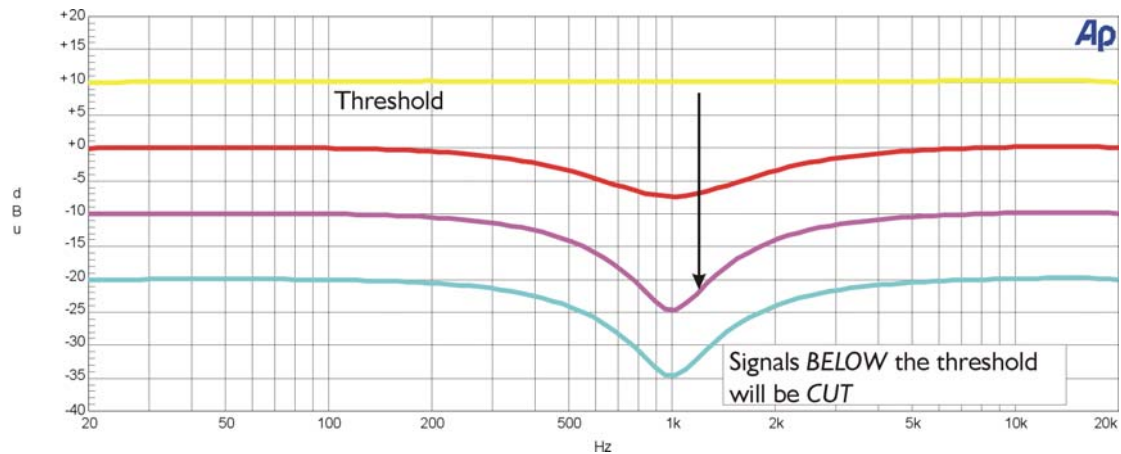
The difference with dynamic EQ is that only the band selected is compressed. This means that it becomes possible to compress the low frequency content of material without affecting the high frequencies at all. The result is increased volume and perceived level without sacrificing clarity. Any instance 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 $8\text{--}9\text{kHz}$, and a relatively narrow ‘Q’ of 3.2 , and a maximum gain of 12dB , attack 1ms , release 100ms .



Quadrant/Mode III: “Cut Below”

Having selected the frequency band to work with, the dynamic eq will listen to this band and act upon it by cutting any frequencies present in it that drop below the predetermined threshold. Consider the example below where the threshold is set to +10dB, and the selected frequency band is centred around 1kHz, with a ‘Q’ of 1.0.



Signals above the threshold will pass flat, but as the level decreases, those frequencies centred around 1kHz will be cut or expanded. The amount of gain reduction applied depends on how much the signal drops the threshold and the ratio set – a 2:1 ratio would mean that for every drop of 1dB below the threshold, the band centred around 1kHz would drop by 2dB.

Uses of “Cut Below” mode.

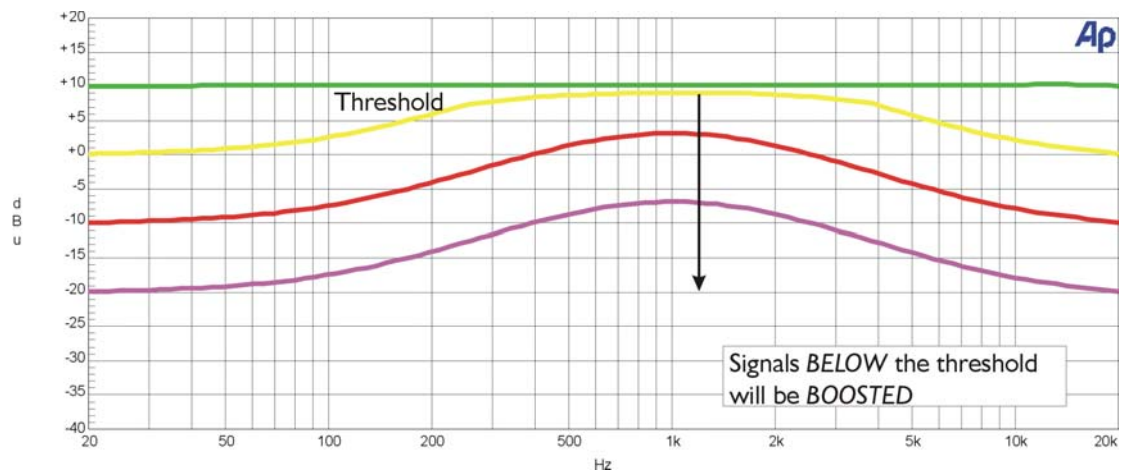
Reducing the level of high frequency noise can be effectively implemented in this mode. Particularly effective on percussive material, unwanted tape noise and interference can be usefully removed without affecting the signal at normal levels.

Try the filter set to a high shelf mode at 4kHz, and a maximum gain of 12dB, attack 25mS, release 100mS. The threshold setting is more crucial in this mode than usual, with the trade-off being effective removal of noise against possible intrusive dulling of the program material.



Quadrant/Mode IV: “Boost Below”

This mode operates in a slightly unconventional manner insofar as behaving as an ‘upwards expander’, as opposed to the more traditional ‘downwards expander’. What this means is that as the signal drops below the threshold, the selected band of frequencies will be progressively boosted in relation to the rest of the spectrum, offering a perceived ‘lift’ in the band. Consider the example below where the threshold is set to +10dB, and the selected frequency band is centred around 1kHz, with a ‘Q’ of 1.0.



Signals above the threshold pass unaltered but, as the signal drops below the threshold, frequencies around the 1 kHz region will be progressively boosted (or expanded). How much boost is applied will depend on the ratio set and how far below the threshold the signal actually is.

Uses of “Boost Below” mode.

One of the best uses of this mode is in the area of voice levelling and clarification. Placing the filter at about 700Hz (lower to nearer 600Hz for men, up to 800Hz for women/children) with a wide ‘Q’ – typically 0.7, a ratio of 2:1, a maximum gain of 12dB, attack 10mS and release 100mS. This will ensure that quiet talkers will have their vocal range boosted, without bringing up system noise or microphone handling noise/room rumble.

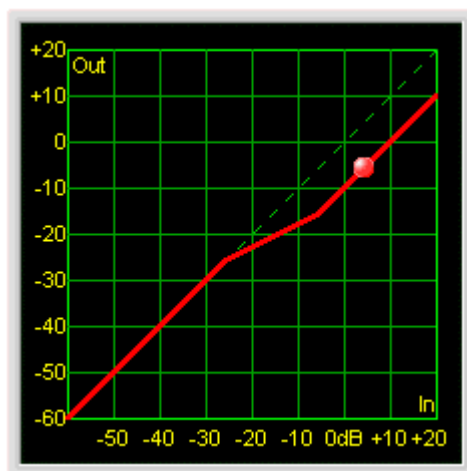


How Do the Dynamic EQ Controls Interact?

The best way to explain how the dynamic EQ controls interact with each other is to adjust the parameters in the edit window and examine what happens to the transfer function graph. In this way the behaviour of the process can be broken down into simple stages.

What does the edit window show?

In its 'large' form, the edit window displays two graphs and a set of controls for all available parameters of the dynamic EQ module. The left hand graph is known as the input/output transfer function and represents how the output of the dynamic EQ module varies with different input levels.



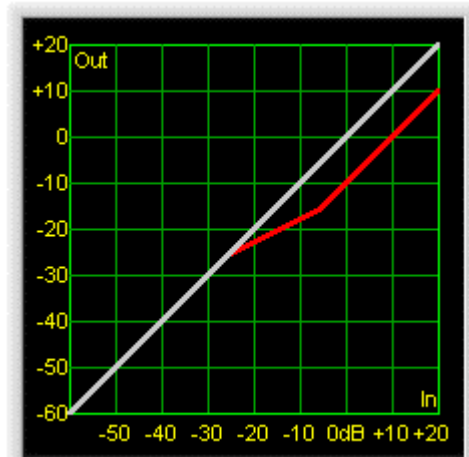
The scale along the bottom of the graph shows increasing input level, from -60dB up to $+20\text{dB}$. Correspondingly up the left hand side, the output level scale runs from -60dB to $+20\text{dB}$ as well.

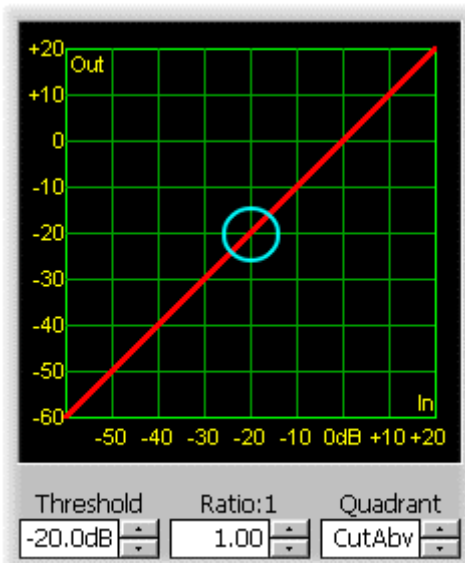
If the dynamic EQ was bypassed, there would be no difference between the signal going in and that coming out, so the graph would show a straight diagonal line running bottom left to top right. (Following the dotted green line shown.)

Pressing the 'Bypass' button will demonstrate this – the red line is overlaid by a grey line as shown here...

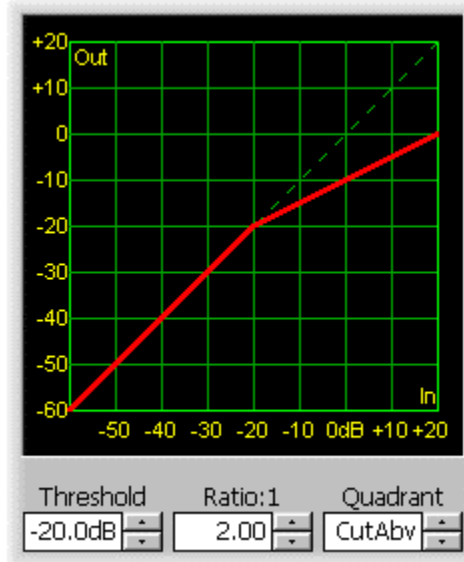
The most important relationship is between the ratio and the threshold. Set the threshold to -20dB and the ratio to 1:1 initially. Select the Quadrant to be 'CutAbv' (cut above). Make sure the 'Bypass' button is not pressed.

The graph will still be displaying a straight line, as if the module was bypassed. Effectively, the module is bypassed – if the ratio is 1:1 then no matter where the threshold is set, the gain of the output will be 1 x the gain of the input. Confirm this by adjusting the threshold – no change to the graph. Remember that the threshold can be adjusted by clicking on the graph at the threshold point and dragging up and down. When the mouse is over the threshold point, the pointer changes to a cross – pressing and dragging will then show a double-ended arrow.



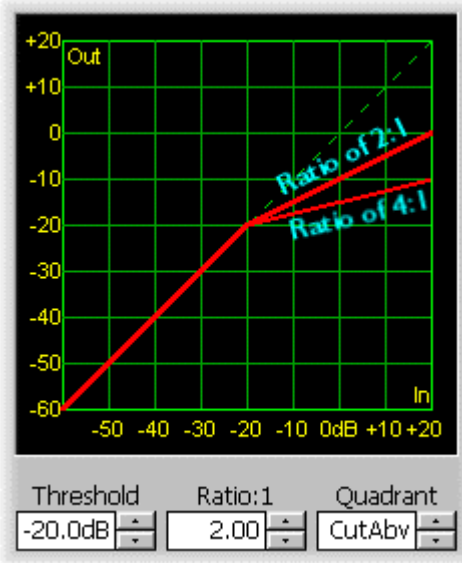


Setting the threshold back to -20dB (centred in the cyan circle as shown), adjust the ratio until it reaches 2:1. The graph will begin to bend at the threshold point, moving away from the 1:1 position. It should now look like this:



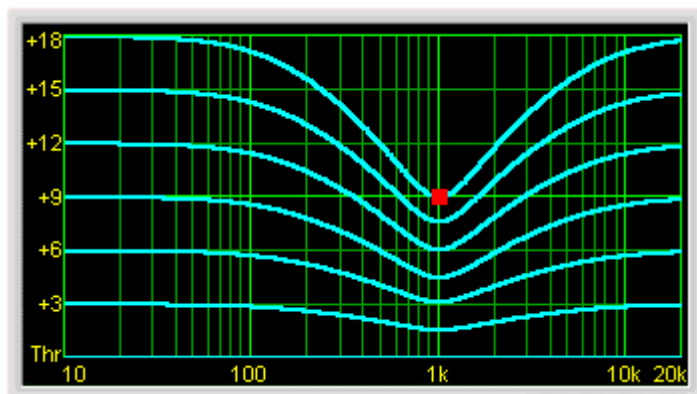
If the signal rises above the threshold now, it will be reduced in level by half (or a ratio of 2:1). So, if the input rises by 10dB (for example from -20dB to -10dB) the output only rises by 5dB (correspondingly from -20dB to -15dB). Hence the slope of the graph. As the ratio is increased, the angle of the line above the threshold point will become ever more horizontal.

Try increasing the threshold to 4:1 and note the change of slope.



As the ratio was adjusted, the other graph (on the right hand side of the edit window) will have been changing as well. This other graph represents the change of frequency response that the dynamic EQ filter will impose on the output. Until now, the transfer function graph could just have easily been referring to a normal compressor, set to a ratio of 2:1. Introducing the dynamic filter into the equation will now be simpler to explain.





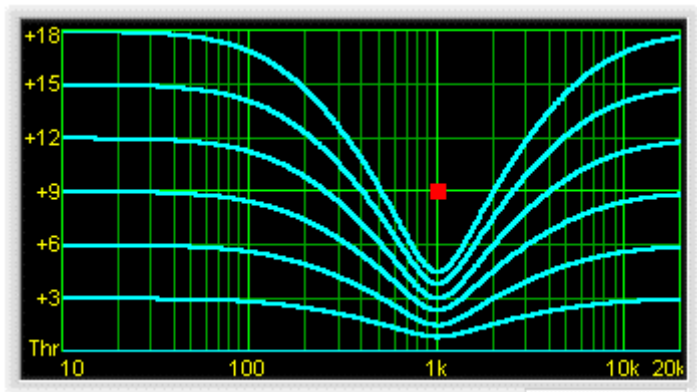
This graph shows how the output will vary, not only with frequency, but at a selection of input levels. These are represented by the six cyan lines, at 3dB intervals **above the threshold**.

As the lines are above the threshold, adjusting it will

not affect this graph. What will have an effect is the ratio. The graph above represents a ratio of 2:1. At the centre frequency of the filter, (1 kHz in this case) it is simple to correlate what this graph shows with the transfer function graph.

Remembering the ratio of 2:1 if the signal at 1 kHz is, for example, 6dB above the threshold (the second line up from the bottom), the output level will be half this – 3dB. Similarly, 12dB in will come out at 6dB and so on.

Increasing the ratio will have the effect of bunching the lines closer together – ever increasing signal levels above the threshold will be cut by more and more – examine the graph below where the ratio has been increased to 4:1.



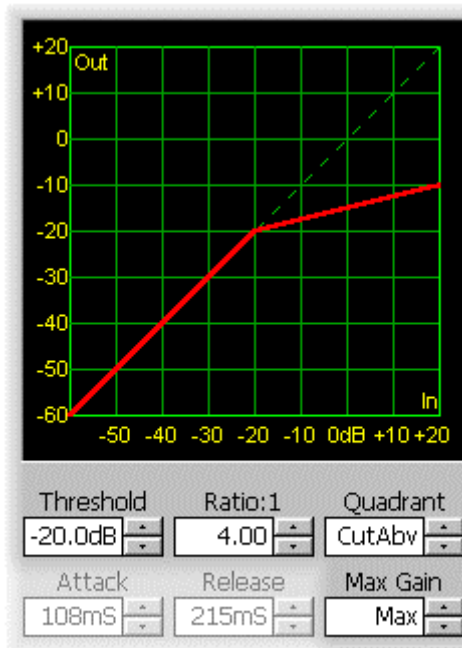
Now, an input of 12dB will only come out at $\frac{1}{4}$ of the input level (ratio is 4:1 remember), so at 3dB.

The added element of the filter means that this response will vary with frequency, hence the shape of the curve(s). The most

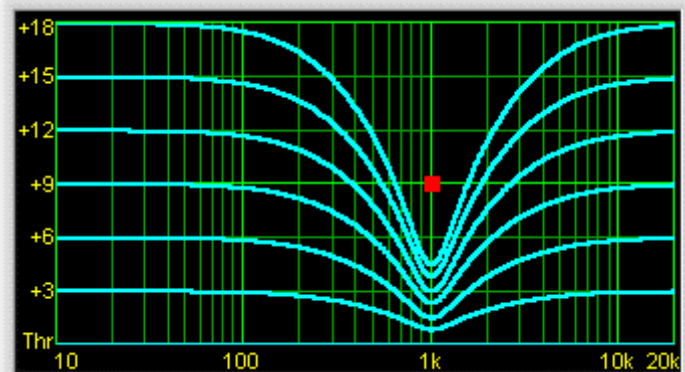
cut is applied at 1 kHz, with the amount tailing off on either side of this frequency. Try adjusting the 'Q' of the filter – this will 'tighten' or 'loosen' the band of frequencies affected by the compression characteristic.

The other parameter that will affect this graph (and the in-out transfer function) is the value of the 'Max Gain'. This determines the absolute allowable gain reduction (or boost for the 'Boost Above' and 'Boost Below' quadrants) that can be imposed by the dynamic EQ. If the ratio is thought of as setting how pronounced the effect will be, then the 'Max Gain' sets the ceiling on how much can be applied. This can be useful when operating with very high or very low thresholds and high ratios – sudden changes in level might well produce far too much cut or boost, causing clipping or unacceptable loss of level, both of which could be very obtrusive.

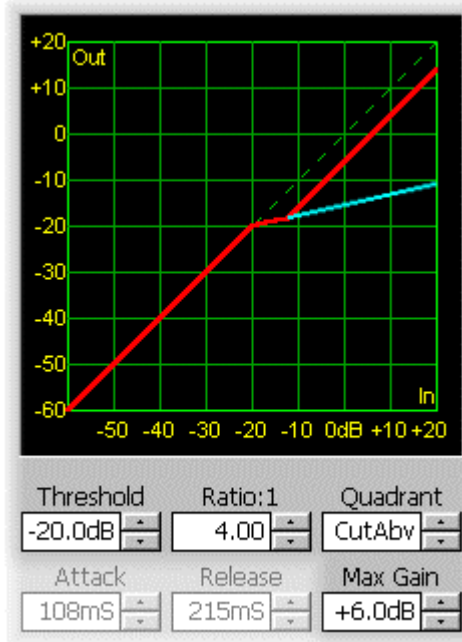




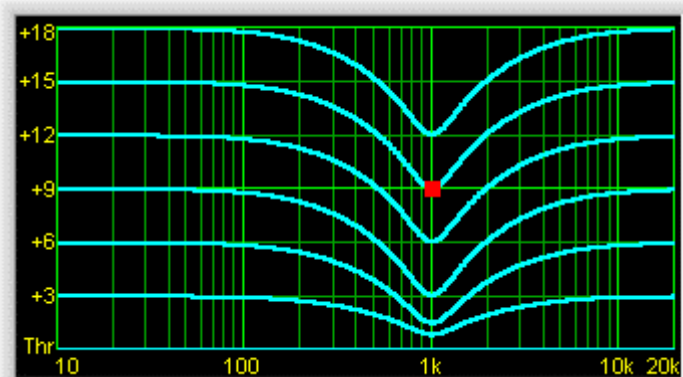
With a ratio of 4:1 and the threshold as before, at -20dB , leaving the 'Max Gain' at 'Max' means that the gain reduction that can be applied will not 'bottom out', and the in-out transfer function will look as expected. Similarly, the change of gain with frequency will follow the laws set by the threshold and ratio parameters, as shown.



Reducing the 'Max Gain' parameter to 6dB means that even if the signal goes high enough above the threshold to impose more than 6dB of gain reduction, only 6dB will be applied. In this example, anything over 8dB above the threshold will be capped with 6dB of gain reduction; in at 8dB with a ratio of 2:1 would come out at 2dB , giving $(8 - 2 =) 6\text{dB}$ of gain reduction. Over 8dB in, and more than 6dB of gain reduction would normally be applied. The effect of this can be seen in the graphs.



The cyan line shows the way the process would behave, had all the gain been made available. The red line shows what will actually happen – normal until 8dB over the threshold and then capped at 6dB . As this is fixed, all that will be happening is a constant drop of 6dB , hence the line running parallel to the 'straight-through' case (dotted).



This is also visible on the frequency response graph. All the lines 8dB above the threshold become equally spaced as only a maximum 6dB

of gain reduction can be applied.

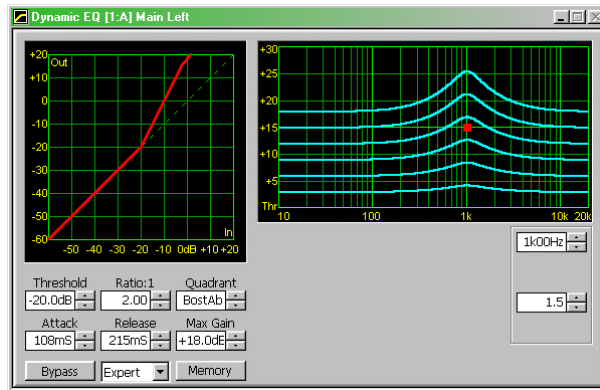


Comparing the Four Quadrants...



Remember that the Quadrant can only be changed if the ratio is set back to 1:1. Similarly, the filter response can only be changed from parametric to shelving (of full range) if the ratio is set to 1:1.

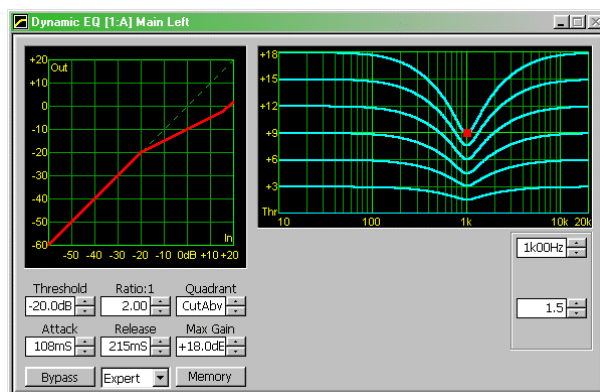
Quadrant/Mode I: “Boost Above”



Above the threshold, the gain would be increased at a ratio of 2:1, so the in/out transfer function shows the curve going over the 1:1 green dotted line.

The frequency related curves show progressively more boost about the filter centre frequency as signal level increases over the threshold, which is at the bottom of the graph.

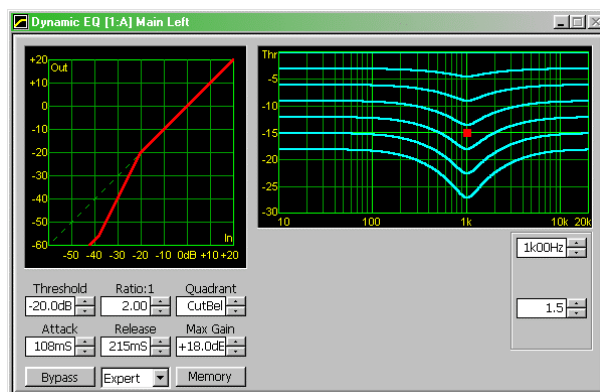
Quadrant/Mode II: “Cut Above”



Above the threshold, the gain would be decreased at a ratio of 2:1, so the in/out transfer function shows the curve going under the 1:1 green dotted line.

The frequency related curves show progressively more cut about the filter centre frequency as signal level increases over the threshold, which is at the bottom of the graph.

Quadrant/Mode III: “Cut Below”

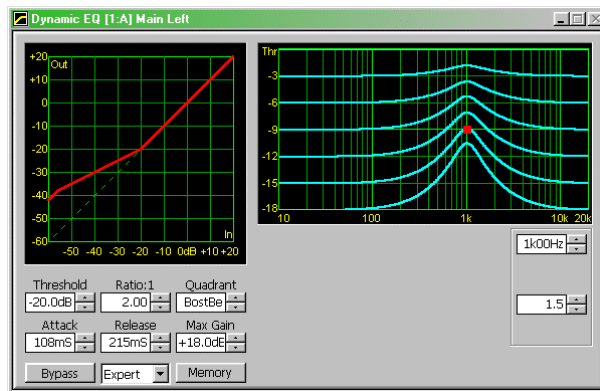


Below the threshold, the gain would be decreased at a ratio of 2:1, so the in/out transfer function shows the curve going under the 1:1 green dotted line.

The frequency related curves show progressively more cut about the filter centre frequency as signal level drops below the threshold, which is at the top of the graph.



Quadrant/Mode IV: "Boost Below"



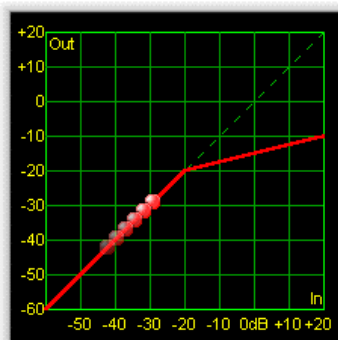
Below the threshold, the gain would be increased at a ratio of 2:1, so the in/out transfer function shows the curve going under the 1:1 green dotted line.

The frequency related curves show progressively more boost about the filter centre frequency as signal level drops below the threshold, which is at the top of the graph.

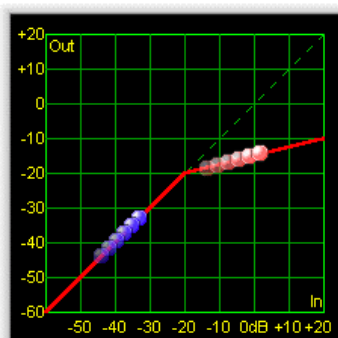
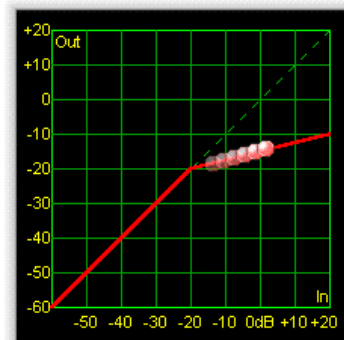
What does the bouncing ball show?

The bouncing ball (not shown on the above series of graphs) represents the real-time input level to the dynamic EQ module. The bouncing ball is only shown when the PC is connected to a unit (or units) and on-line. Please see page 11 for details of how to accomplish this.

Assuming the PC is connected, and on-line, the ball will move along the in-out curve in real-time, giving a clear indication of the level in relation to the threshold, and the settings of the envelope controls (attack time, hold time where appropriate, and release time).



A red ball represents the left channel on any unit (channel A), and the right channel (channel B) with a blue ball. Whilst the input signal to any particular module is below the threshold, the ball will be noticeably darker than when above the threshold. Once the ball crosses the threshold, gain reduction is being applied (or gain boost in two modes of dynamic EQ) and the ball will lighten.



In the case of a stereo configuration, one graph will be shown with both balls appearing on the same curve. In this example, channel A is above the threshold, and channel B is below.

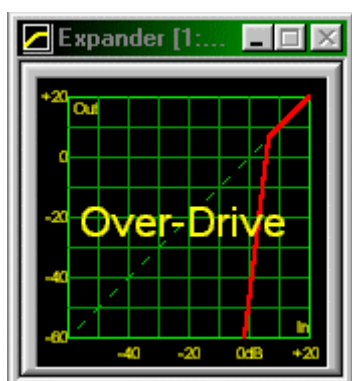


Indication of Clipping The Input to a Module

Before exploring editing any further, it's worth noting that whilst SiDD possesses very high headroom due to the 40-bit accumulators used for all calculations (as opposed to more usual 24 or 32 bit), it is still possible to reach an overload state internally.

Even if the input to the unit appears to be relatively low according to the input meter, if input gain has been added and there is a lot of boosted EQ, the signal level within the unit can be running at a very high level. This will not always be apparent, especially in the case of the dynamic EQ module where a large amount of signal amplification can be introduced, but only on peaks (which by their nature are higher than the usual signal level).

If the input to a module is too high, the in/out transfer function graph will flash the message 'Overdrive' as shown below.



Reduce the level from the offending module to until the warning disappears, or audible distortion will occur.

Note that this message refers to the **input** of the module, so the level needs to be reduced '**upstream**' from the module displaying the message.

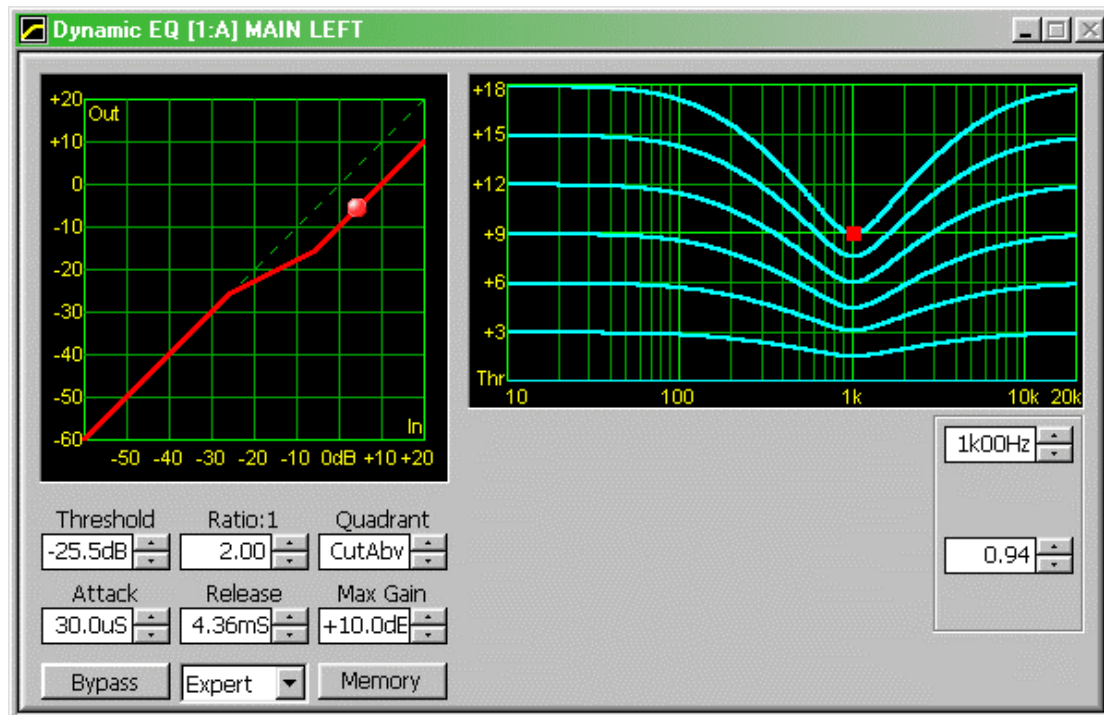
The table below outlines possible scenarios.

Overload Condition	Possible Cause	Possible Action
Input meter clipping (in the red)	Actual input level to the unit is too high	Reduce the direct feed to the unit itself (adjusting the input gain control will not work)
Dynamic EQ is being overdriven	Too much input EQ, or input gain too high. Also, ADT level may be too high ⁴ or harmonics generator level(s)	Reduce EQ boost, input gain, ADT level, or harmonics level
Gate or Expander is being overdriven	Dynamic EQ is applying too much boost.	Set DEQ threshold higher or reduce the 'Max Gain' parameter if the threshold must remain fixed
Compressor is being overdriven	Either Input section is running too hot, or DEQ is running too hot	See DEQ action and Gate/Expander action
Limiter is being overdriven	Input section is running too hot, DEQ is running too hot, or Compressor makeup gain too high.	See DEQ action and Gate/Expander action. Lower compressor threshold, or turn down makeup gain if threshold must remain fixed.
Output meter clipping (in the red)	Any previous condition, or output gain too high	Check all above actions or turn down output gain

⁴ Only applicable in ADT mode



Editing Parameters on the Dynamic Eq Window



Having explained in detail how the dynamic EQ module works, adjusting the parameters is very straightforward. The threshold may be adjusted by either nudging the value using the arrows, or dragging it up and down directly on the in-out transfer function curve. Remember that to change the Quadrant (Operating mode), the ratio must first be set to 1:1.

Attack time covers the range 30uS up to 2.0 seconds. The release time covers the same range. Note that the software will not allow release times to be less than attack times, due to the likely 'parameter abuse' involved and subsequent poor performance.

Adjusting the filter can again be accomplished by either nudging the frequency with the button or directly dragging the centre frequency, denoted by the red square, along the frequency response curve. Note that, unlike the EQ or sidechain frequency response curves, this square will be fixed in its vertical position, as the gain is the parameter being adjusted dynamically.

To adjust the 'Q' (or bandwidth) of the filter, use the nudge buttons beside the readout or, using the mouse, select the node and, holding the left mouse button down, move the scroll wheel.⁵



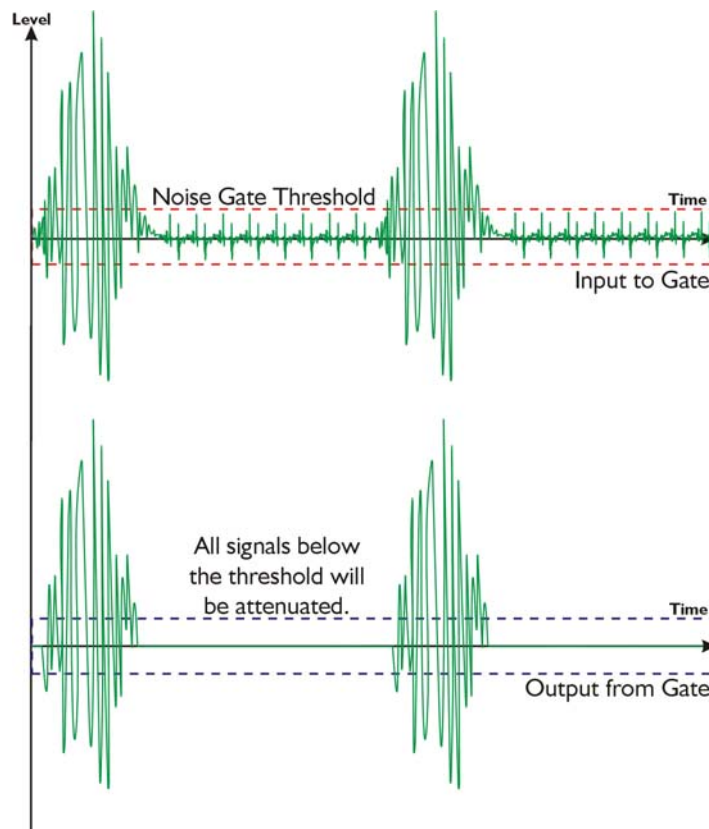
Holding down 'Shift' while moving the scroll wheel locks the frequency of the filter, making adjustment of the 'Q' easier.

⁵ An 'intellipoint' type mouse is required for this operation.



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 where the bass drum was not being played, only allowing signal through when the output would be sufficient to mask the noise anyway.

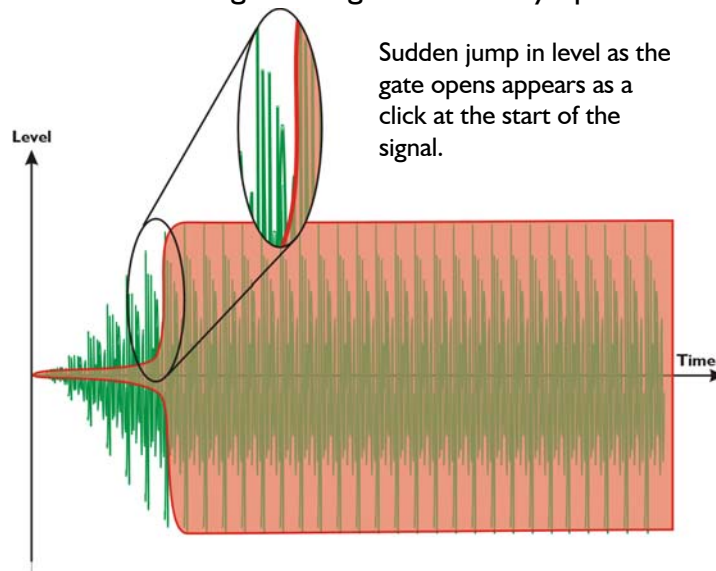
The output graph above highlights several other points that are particularly important with noise gates.

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.

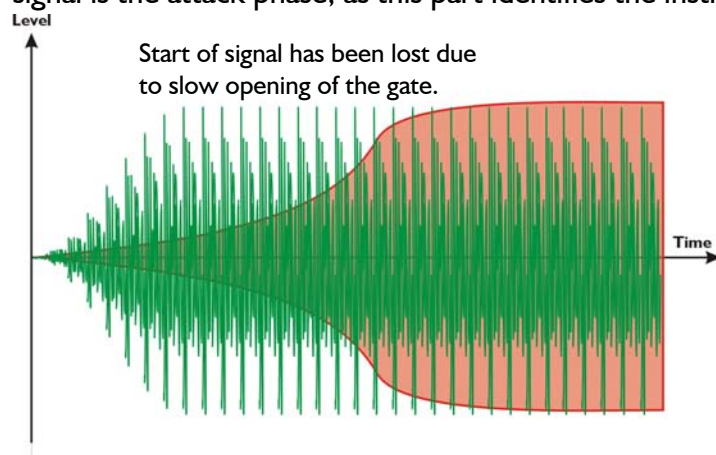


Controlling the speed at which the gate opens (attack time) and closes (release time) is crucial to providing transparent operation. Setting the attack time too fast on a signal that naturally has a relatively slow attack will produce a click. Such a case often occurs when gating the human voice. The 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.

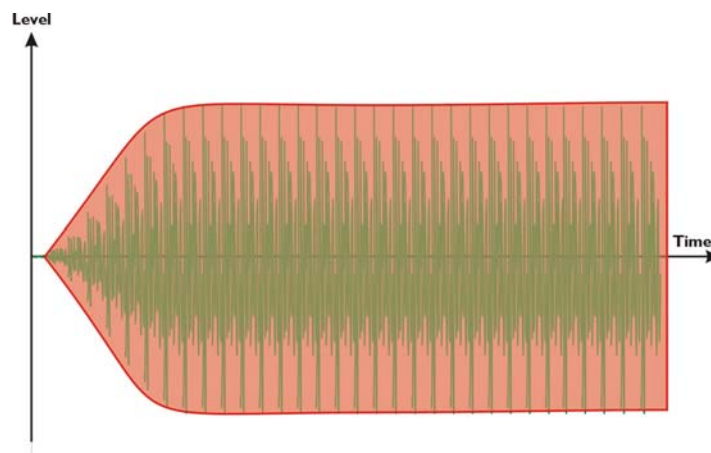
As mentioned earlier, if the attack is set too slow, the gate will not open in time, and part of the signal will be lost. For most musical instruments, the essential part of the signal is the attack phase, as this part identifies the instrument to the human ear.



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.

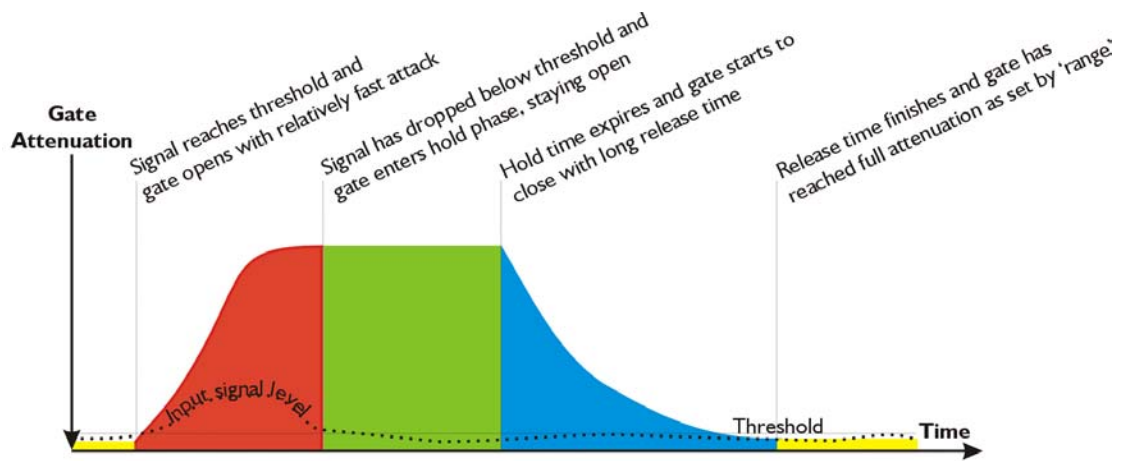
The range control sets how much attenuation is applied to a signal once it drops below the threshold and closes the gate. It is quite often overlooked when setting up a gate, the temptation being to set the range at maximum and leave it. When the gate is fully closed, no signal will be heard when the 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.

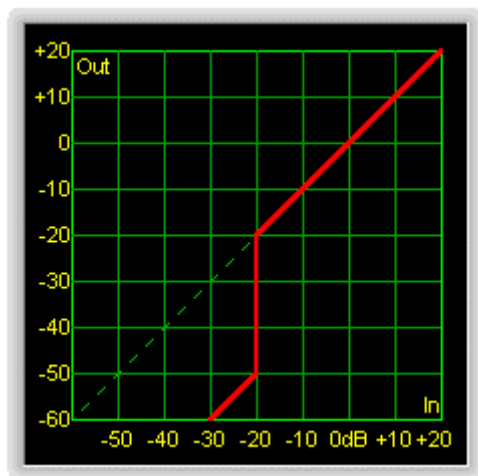


How Do the Noise Gate Controls Interact?

As with the dynamic EQ module, the best way to explain how the controls interact with each other is to adjust the parameters in the edit window and examine what happens to the transfer function graph.

What does the edit window show?

In its 'large' form, the edit window displays two graphs and a set of controls for all available parameters of the noise gate. The left hand graph is known as the input/output transfer function and represents how the output of the gate module varies with different input levels.



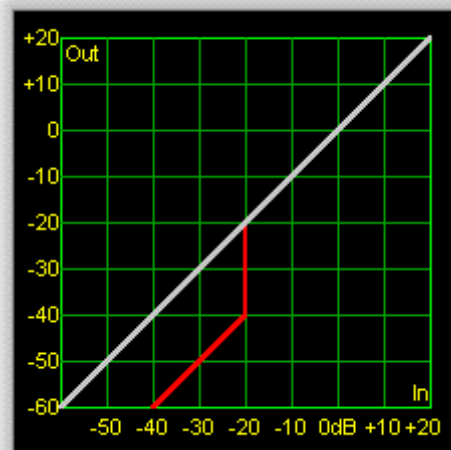
The scale along the bottom of the graph shows increasing input level, from -60dB up to $+20\text{dB}$. Correspondingly, up the left hand side, the output level scale runs from -60dB to $+20\text{dB}$ as well.

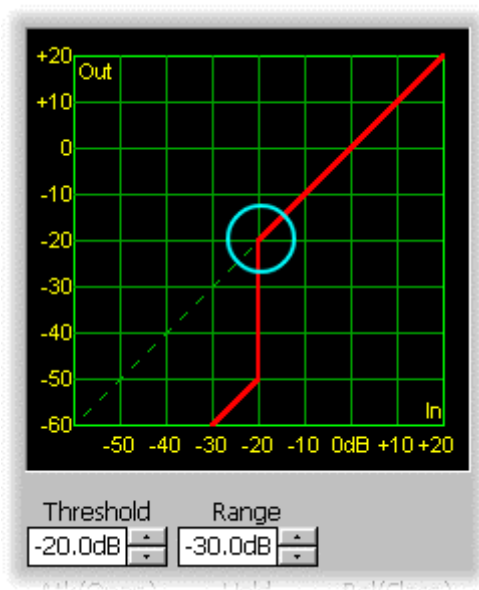
If the noise gate was bypassed, there would be no difference between the signal going in and that coming out, so the graph would show a straight diagonal line running bottom left to top right. (Following the dotted green line shown.)

Pressing the 'Bypass' button will demonstrate this – the red line is overlaid by a grey line as shown here...

The most important relationship is between the threshold and the range. Set the threshold to -20dB and the range to 0dB initially. Make sure the 'Bypass' button is not pressed.

The graph will still be displaying a straight line, as if the module was bypassed. Effectively, the module is bypassed – if the range is 0dB then no matter what level the input is, the gate cannot apply any attenuation below the threshold. Confirm this by adjusting the threshold – no change to the graph. Remember that the threshold can be adjusted by clicking on the graph at the threshold point and dragging up and down. When the mouse is over the threshold point, the pointer changes to a cross – pressing and dragging will then show a double-ended arrow.



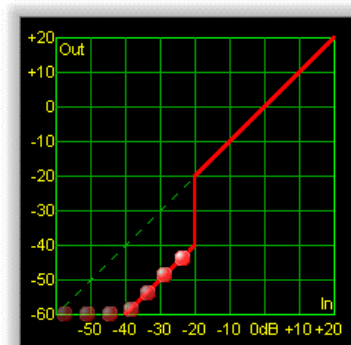


Setting the threshold back to -20dB (centred in the cyan circle as shown), adjust the range until it reaches -30dB . The graph will begin to bend at the threshold point, moving dropping vertically by 30dB and then continuing in a diagonal line as shown here.

What this shows now is that all signals below the threshold will be dropped by 30dB in level. At the threshold, the gain will jump back to 0dB and the output will be at the same level as the input – hence the straight diagonal line above the threshold, following the path as if the gate was bypassed.

What does the bouncing ball show?

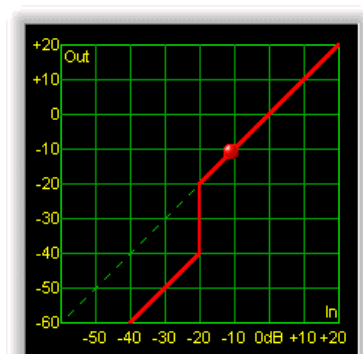
The bouncing ball (not shown on the above series of graphs) represents the real-time input level to the gate module. The bouncing ball is only shown when the PC is connected to a unit (or units) and on-line. Please see page 11 for details of how to accomplish this.



Assuming the PC is connected, and on-line, the ball will move along the in-out curve in real-time, giving a clear indication of the level in relation to the threshold, and the settings of the envelope controls (attack time, hold time where appropriate, and release time).

Note that the ball will move along the bottom axis of the graph even when the output level should force it off the bottom to follow the lower linear portion.

As the ball rises up the vertical part of the curve, it is crossing the threshold, and the gate will open. The ball will continue along the upper linear part of the curve with 0dB of attenuation, as shown. (It will have darkened slightly as it is now over the threshold.)



Sidechain Equalisation – How and When to Use it

Each of SiDD's dynamics modules, with the exception of the dynamic EQ, offers the facility to add EQ to the sidechain signal controlling the gain through the module. This obviates the need for a host of external processors that would have to be pressed into service for this task, using up vital EQ resources that could otherwise be used for normal EQ tasks.

When would Sidechain EQ be useful?

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

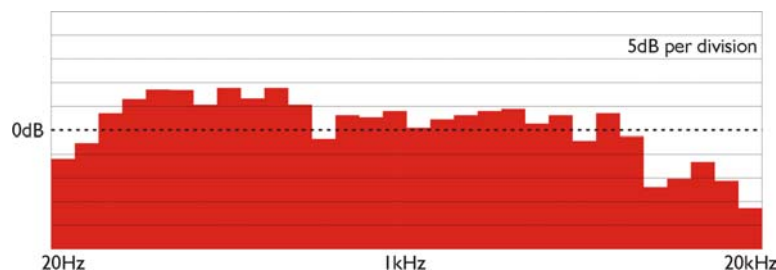
De-essing and de-popping.

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.

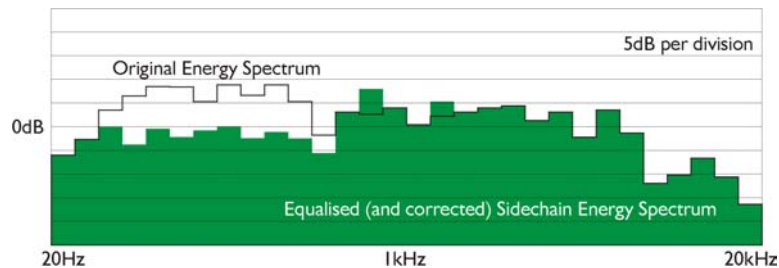


If the threshold was set to 0dB (the dotted line) it can be seen that the compressor will start to act first when the low frequency part of the spectrum crosses the threshold.



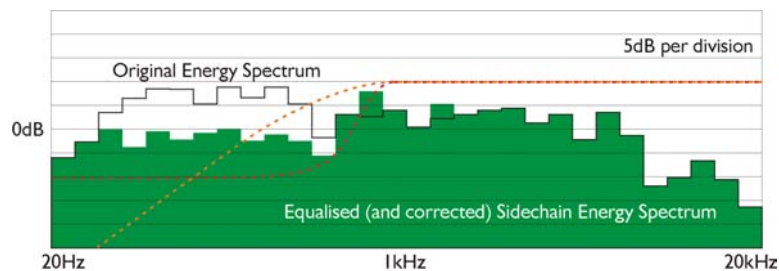
This will have the effect of pulling down the level across the *entire* spectrum (including the high frequencies) and so causing the familiar ‘dulling’ that can easily occur with full range compression. The solution to this problem is to introduce some complementary sidechain EQ that will remove some 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.

The noise gate (or expander) available in SiDD has two bands of fully parametric equalisation, along with traditional high and low pass filters. The parametric filters may be transformed into shelving filters if required. The compressor and limiter both have two bands of fully parametric EQ in their sidechains.

Suppressing feedback without adding notch filters.

The two parametric bands (as opposed to the high and low pass filters) allow accurate selection of the two main feedback nodes on a monitor mix. Traditionally the main feedback nodes would be picked out with a couple of notch filters placed in the main signal path. This is not ideal for several reasons.

Firstly, whilst this method is used by common feedback eliminations systems, it does produce audible artefacts. The introduction of a multitude of notches that are present regardless of feedback conditions cannot be anything other than detrimental to the overall sound quality.

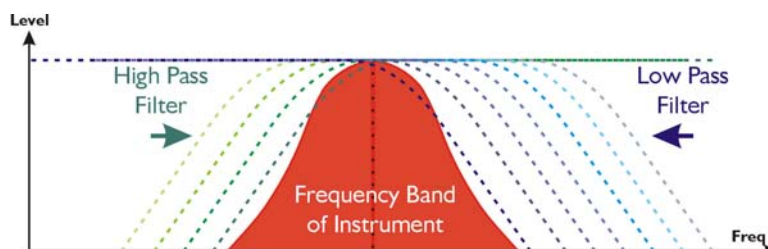


Secondly, the inclusion of another processor in the signal path, with its analogue to digital and digital to analogue conversion will only increase system noise and degrade performance - consider the situation with one of these systems in place on each monitor!

By sensitising the sidechain to the two main feedback nodes, SiDD can be made to compress (and so attenuate) the main signal, *but only when feedback occurs*. In this way, the main signal is completely unaffected until feedback is present. Whilst this will not stop the feedback completely, it will reduce it to a level where it will no longer build up. The most important factor in feedback suppression is the avoidance of 'system runaway' where feedback continues to build to the point of clipping the amplifier and possibly destroying the speaker. This method of suppression will prevent just that situation and not introduce any audible artefacts in the process.

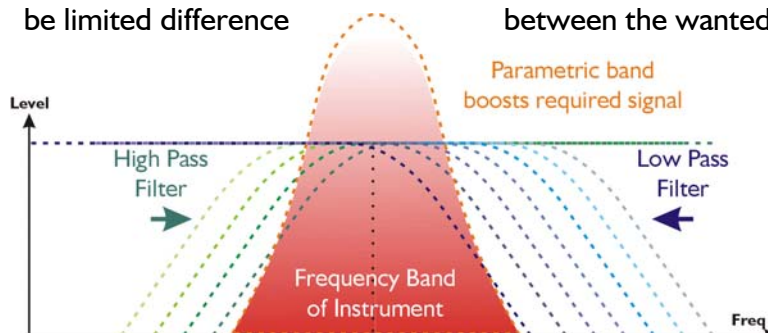
Picking out instruments for gating.

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.



SiDD's noise gate (and expander) offers not only the high and low pass filters, but two additional bands of fully parametric EQ.

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

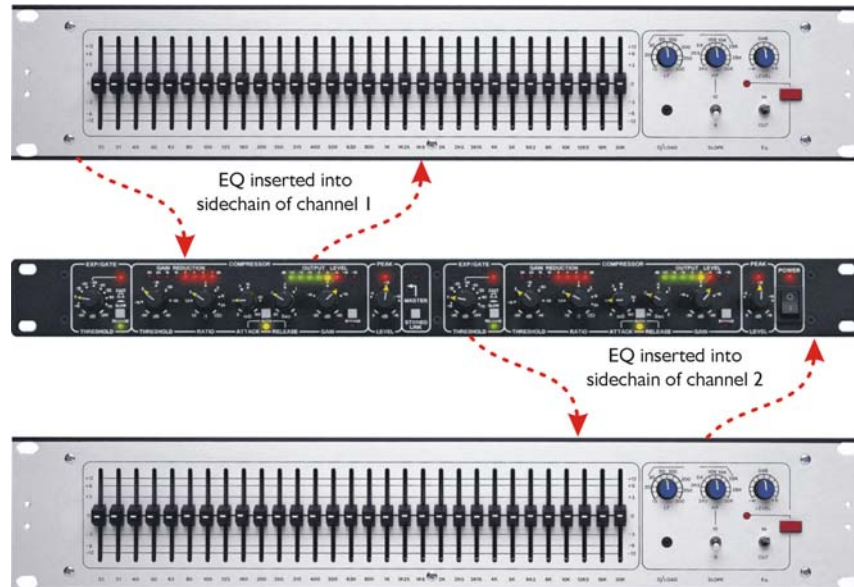


Being able to accentuate the wanted band at the same time as attenuating the unwanted material, especially with the added possibility of highlighting more than one area for triggering (there are two parametric sidechain filters!) could prove invaluable.



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. SiDD obviates the need for this external processing for any of the dynamics modules.



In practical terms, the internal sidechain filters are saving a potential six external units worth of devices!⁶ This isn't even including the introduction of delay lines to allow for look ahead delay.

The concept of look-ahead delay is explained in the following section.

⁶ One EQ unit per processing module, excluding the dynamic EQ, and two channels worth.
SiDD PC Software Version 2.00

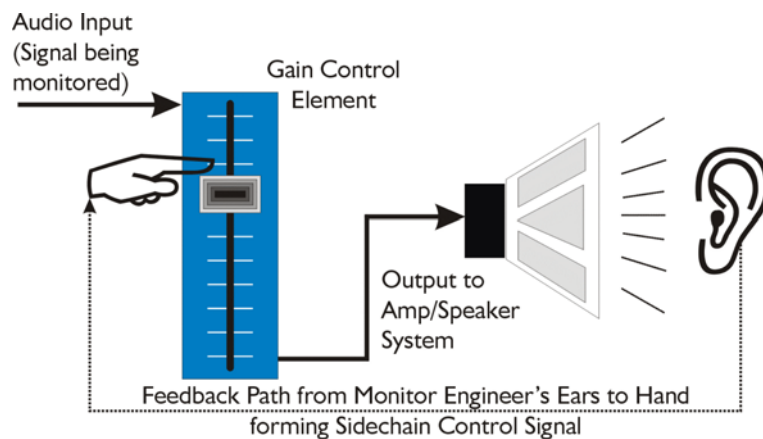


Look Ahead Delay – Pre-emptive Action

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 perform⁷. 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

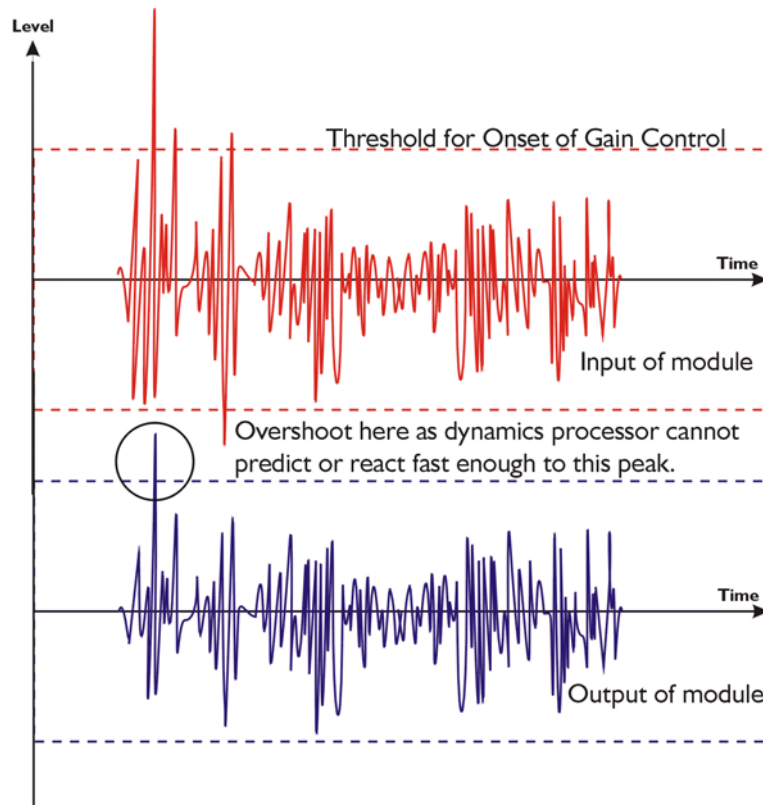
brain instructing his hand to turn the fader down if the volume goes over the threshold he has chosen.

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.

⁷ XTA would like to point out that whilst SiDD might sound male, not all engineers are necessarily male. Some might well be female, or at least have long hair.





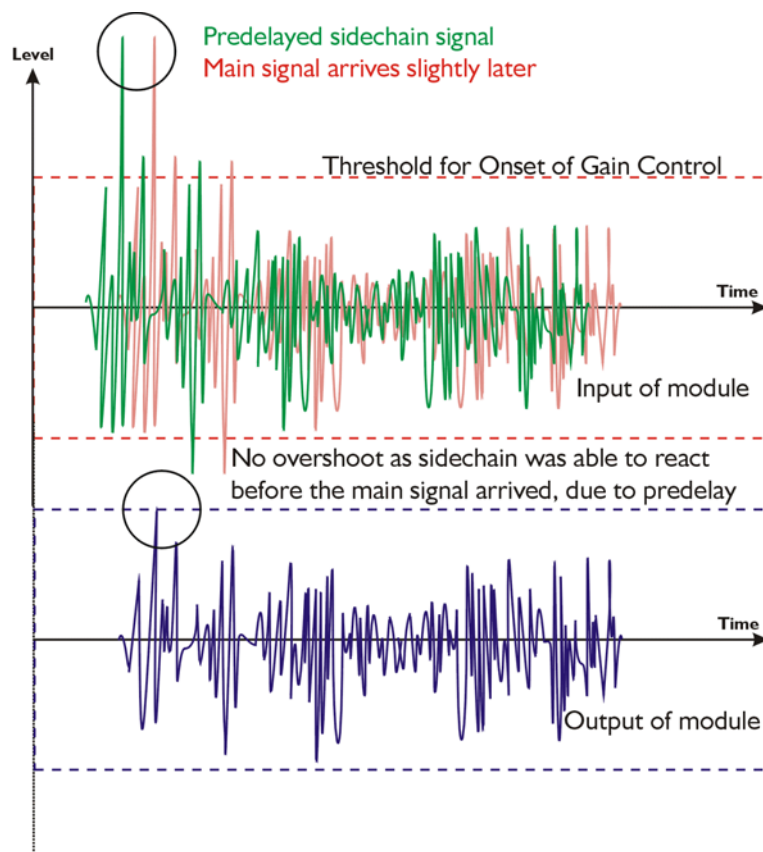
The red waveform represents the input to the dynamics module, with the dotted line showing the threshold for gain control to occur. There are several peaks towards the start of this signal that are above the threshold, and so the dynamics processing should react to these as appropriate. (In this case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

SIDD'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.





SIDD's predelayed sidechain is shown in green, with the main signal in red.

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

Remember that this delay is only in the order of 10 to 60 μ s, 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.

SIDD 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;
- Opening the noise gate in advance to prevent clicks;
- Ensuring maximum level into a desk using the compressor to level signals without clipping the input to the channel.



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 present in the *Advance Series One*. 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 which exceeds its dynamic range, distortion will almost always occur.

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

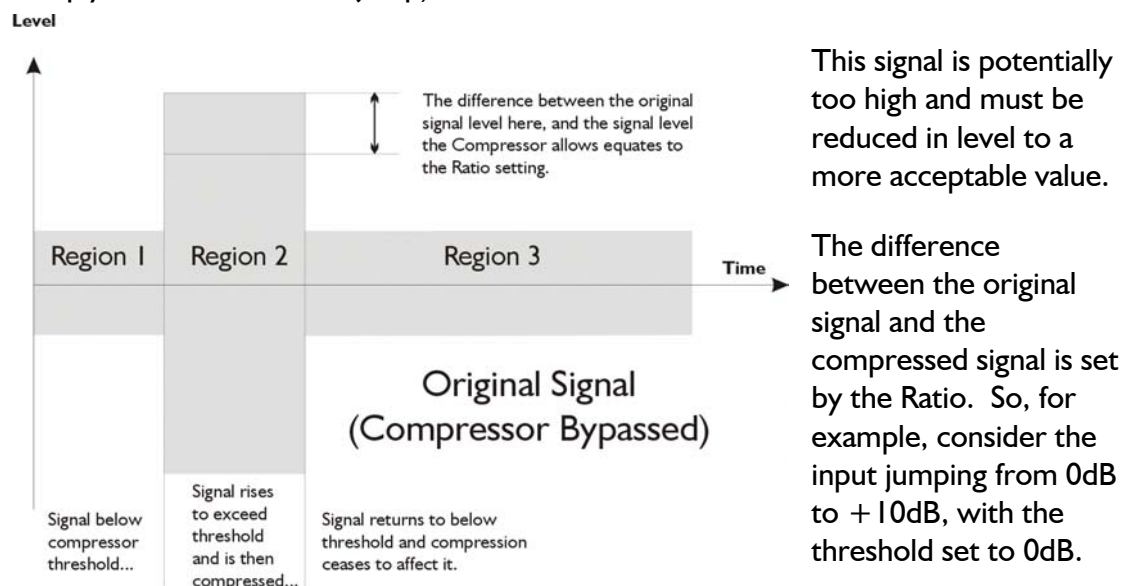
In an ideal world, all devices used to record and reproduce sound would have boundless dynamic ranges and so present no compatibility problems with each other. As this is not the case, some method must be employed to limit the dynamic range of signals in situations where they will prove troublesome or 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.

How does a Compressor work?

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. However, in region 2, the signal has risen (very sharply, hence the sudden jump) to a level above the threshold.

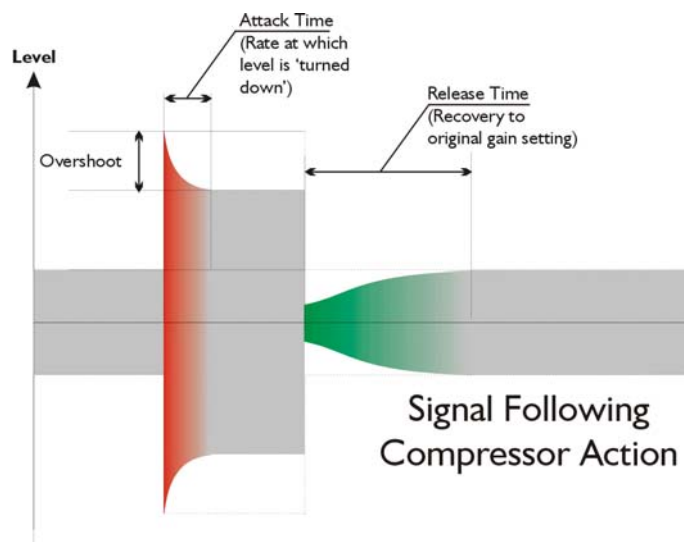


If the output rose by 5dB, then the ratio would be 10/5 or 2:1. Comparing the level displayed on the input meter with the level on the output meter (assuming all other dynamics modules are bypassed) will demonstrate the level of Gain Reduction being applied to any signals that exceed the threshold. This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

What makes the XTA Compressor different?

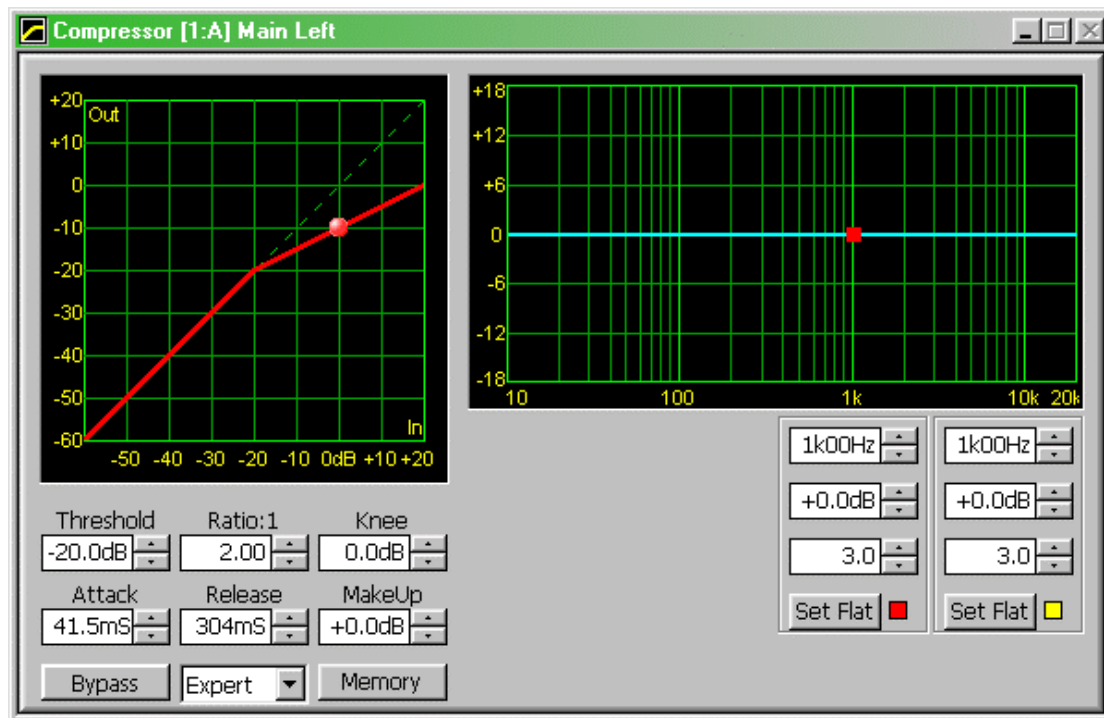
Being a totally digital dynamics processor, SiDD has several advantages over conventional compressor designs. By inserting a short delay into the sidechain signal path, the compressor is able to 'look ahead' and anticipate peaks before they occur. This gives it the ability to prevent the overshoot that 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. SiDD's compressor has two bands of fully featured parametric equalisers built into the sidechain. Each band can be swept from below 20Hz to 20kHz and is selectable as either a parametric filter, or a shelving filter. Additionally, the filters can be bypassed individually, so that, in conjunction with the 'Key Listen' facility, their 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.



Editing & Interaction of the Compressor Parameters



The compressor window follows the same layout rules as the other dynamic module windows – input vs. output transfer function on the left, and the sidechain equalisation on the right.

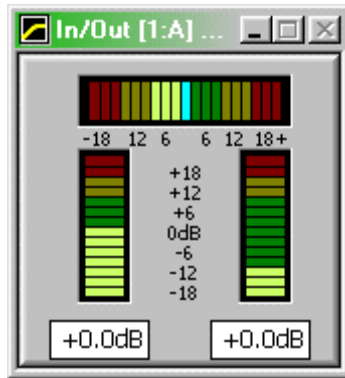
In the example above, with the threshold is set to -20dB and the ratio at 2:1, the straight 'bypass' line begins to bend at the -20dB point on the graph. The higher the ratio, the steeper the bend on the graph. It can be seen that with the ratio of 2:1, for every 1dB the input exceeds the threshold, the output will only rise by 0.5dB. So, as the input is at 0dB (the position of the ball), the output is sitting at -10dB , half of the 20dB above the threshold.

Make-up Gain.

Whilst the job of the compressor is to keep excessive levels in check, it is often desirable to add some gain after the dynamic gain reduction imposed by compression. This can restore the average level to its original, whilst still controlling the peak level. The gain reduction meter may be used in conjunction with the input and output meters to gauge roughly how much makeup gain will be necessary.

Consider the example above where the compressor is applying 10dB of gain reduction. Assuming the other modules are bypassed and all EQ is flat, the metering would be displayed as overleaf...



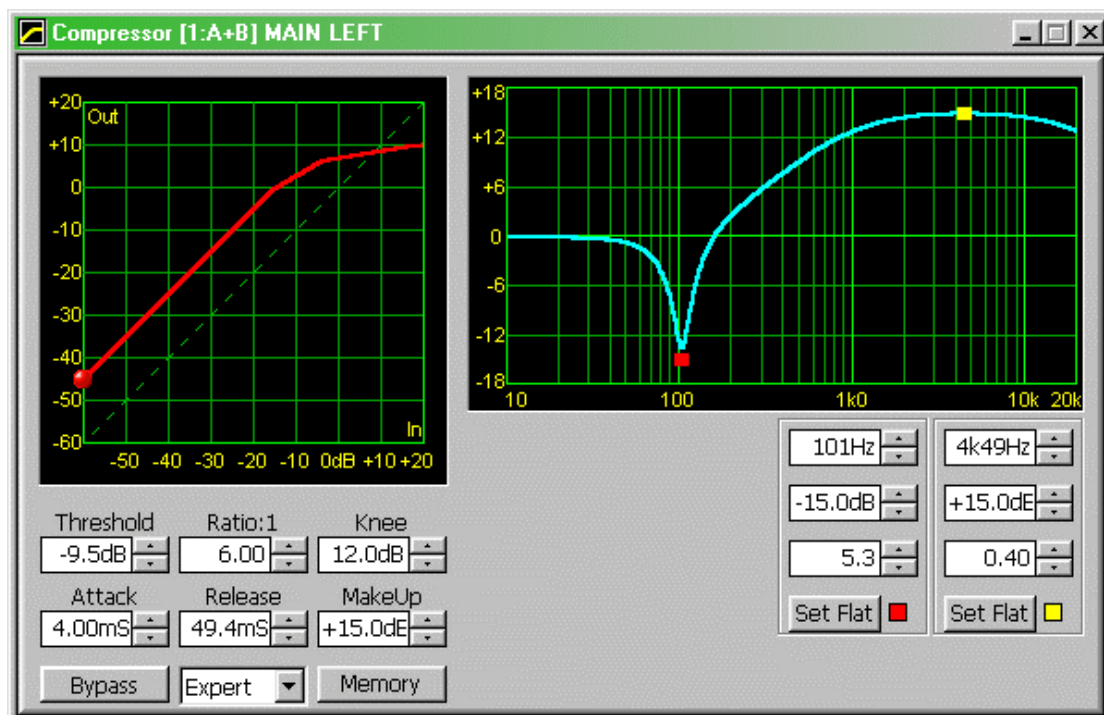
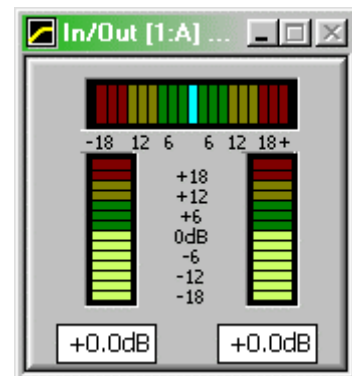


The input is at 0dB, and the output reads 10dB lower. The top meter, showing gain reduction left of the central cyan line, shows 10dB of difference.

Using the 'MakeUp' gain facility included in the compressor, 10dB of gain may be applied to redress the balance and restore the original level. This will have the effect of setting the meters back to 0dB and the horizontal gain reduction meter back to the middle position.

Note the effect of applying the make-up gain on the in-out transfer function. The entire line moves away from the 'bypass' position, vertically offset.

Note: This offset will be in addition to any standard output gain that has been added. Standard output gain can be used as make-up gain, but will not be removed when the compressor is bypassed, making comparisons between the processed/unprocessed signal difficult.

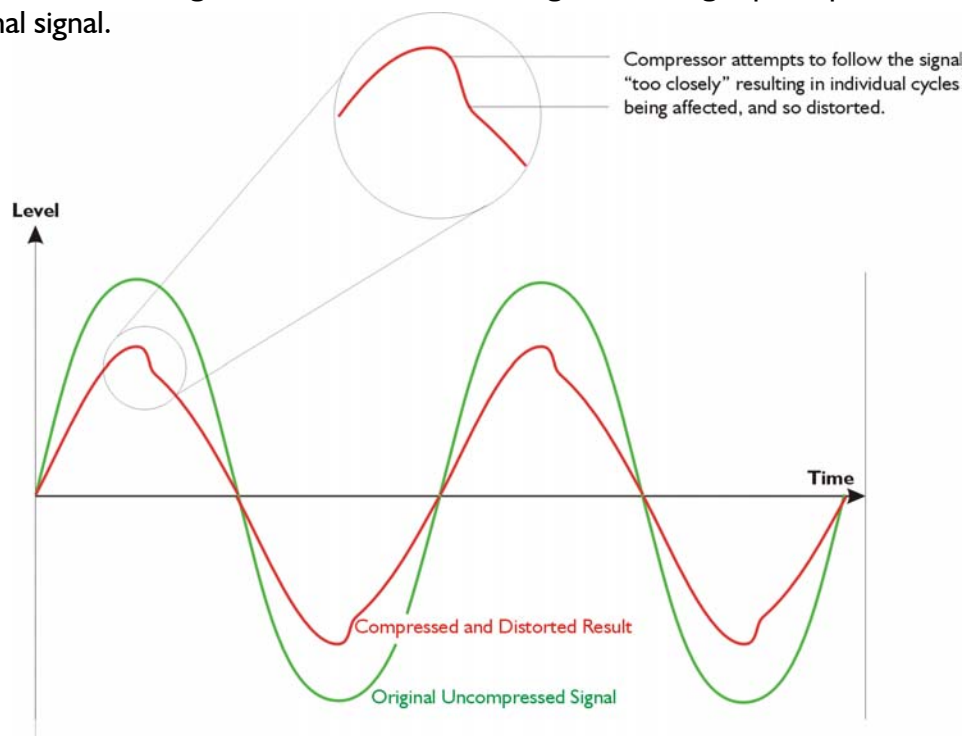


Please note that this is only a guide of how to apply make-up gain – the actual amount necessary for the *perceived* level to appear the same will depend just as much on the attack and release settings as the instantaneous gain reduction at any given moment. Again, the best method of setting the make-up gain is to use the 'BYPASS' key and compare input with output. The human ear is a much better gauge than a digital meter.



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 49mS 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 'BYPASS' button, constantly comparing the original signal with the compressed version.

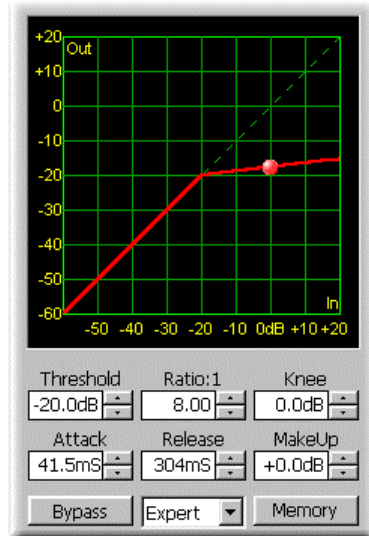
The Compressor 'Knee' Control.

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.

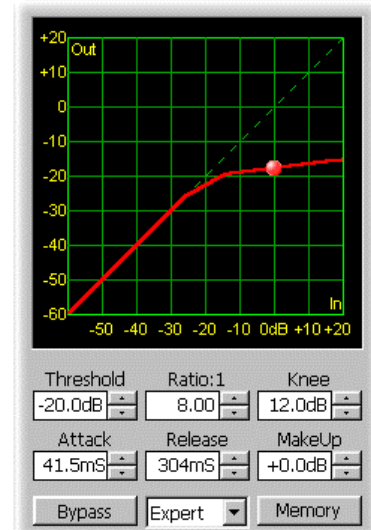
Without the inclusion of a knee control, at the threshold of gain reduction, a sudden transition occurs between unity gain and the ratio by which the compressor attenuates. When using high compression ratios, the use of a hard knee can result in a very unnatural sound.



Consider a compressor with a relatively high ratio of 8:1. The in-out transfer function as seen in the software would be as below. 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 Expander Module – The Complimentary Process

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 – its minimum output, and its 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.

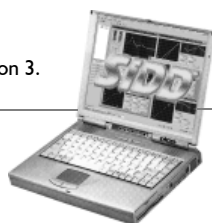
The maximum output level is largely determined by the power supply capabilities of the device. Many manufacturers (especially Far Eastern) tend to use power supplies with lower voltage rails, resulting in a poor drive capabilities and limited maximum output level.

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”⁸.

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

⁸ Concise Oxford English Dictionary 2000 Definition 3.

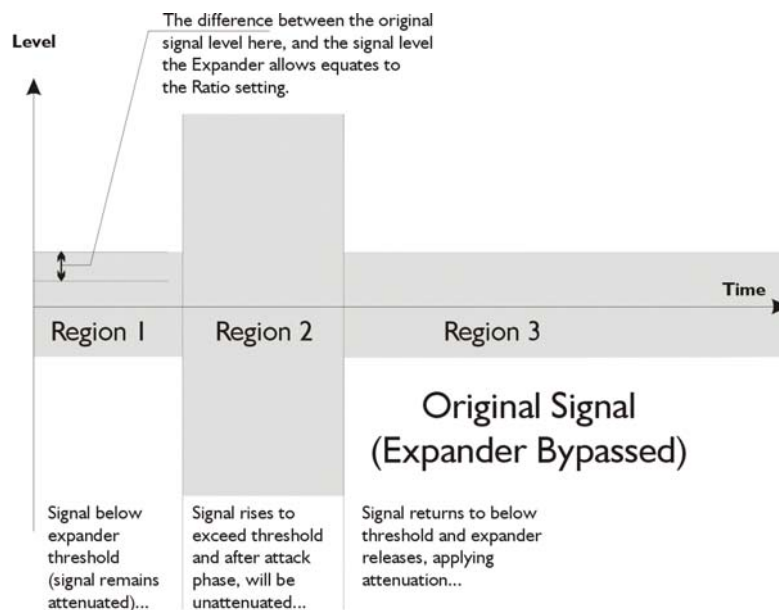


An expander can be used to improve the noise floor under all these circumstances, by attenuating signals below a set threshold, thereby improving the clarity of the signal fed to it.

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.

How does a Expander work?

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.

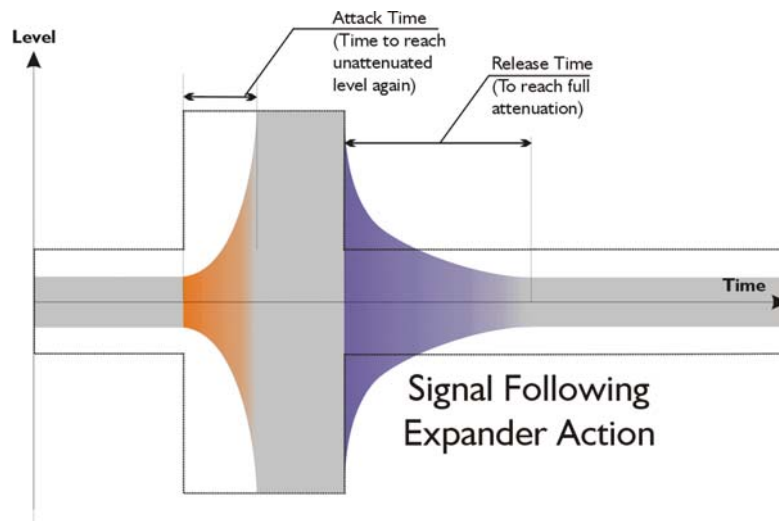


Continuing with the compressor analogy again – the ratio can be determined by considering the degree of attenuation required. If the signal drops to 6dB below the threshold, and the ratio is set to 2:1, then the actual output will be 12dB down.

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 1 represents 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 the compressor, the setting of the envelope controls is as important as choosing the correct threshold.

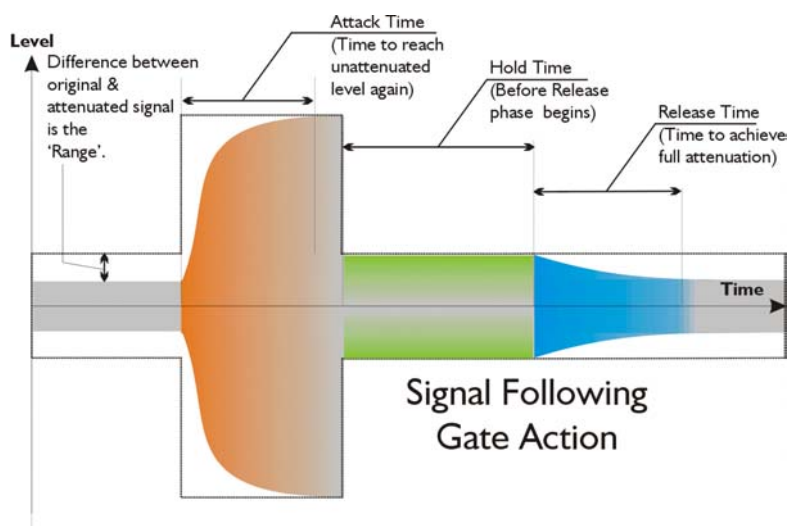


Expanders are susceptible to the same abuse as compressors as far as adjustment of the envelope controls is concerned. If the attack time is too fast, there will be an audible 'click' as the attenuated signal rises above the threshold and normal (unattenuated) gain is reapplied.



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.



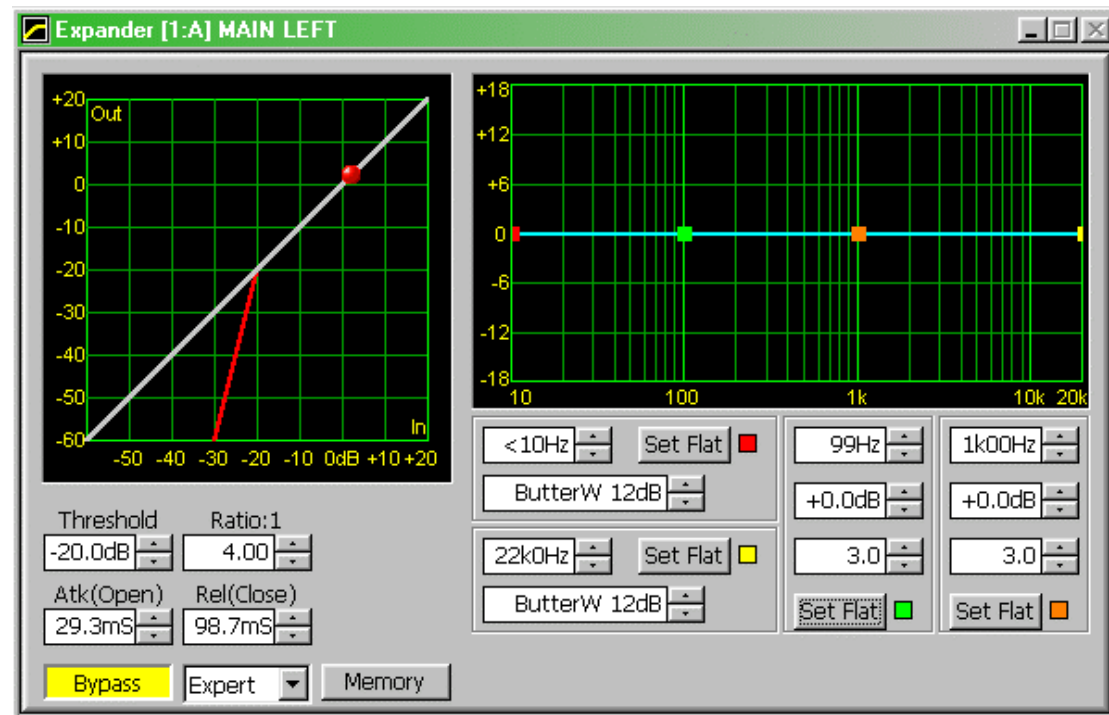
As the signal in region 1, is below the threshold set for the gate to be 'open', the output will be attenuated. The amount of attenuation applied is not dependant on a 'Ratio' this time, but on the 'Range' control setting.

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 44.



Editing & Interaction of the Expander Parameters



At the transfer function above shows, the expander and noise gate share similar responses – above the threshold, the signal passes unaltered (hence following the grey 'bypassed' line), and below the threshold is attenuated with a severity determined by the 'Ratio' parameter.

The higher the setting of the ratio, the steeper the line below the threshold, and the more severe the attenuation. In the above example, with a ratio of 4:1, for every 1dB the signal drops below the threshold, it will be attenuated by 4dB. At very high ratios, the expander will behave in a very similar manner to the noise gate – if the ratio is 16:1 then a drop of only 2dB below the threshold will attenuate the signal by $2 \times 16 = 32\text{dB}$.

Setting the Attack and Release times.

As with the noise gate, the speed at which the expander returns to zero attenuation above the threshold (attack time) and attenuates below the threshold (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. Please read the section about noise gate attack and release settings starting on page 42.

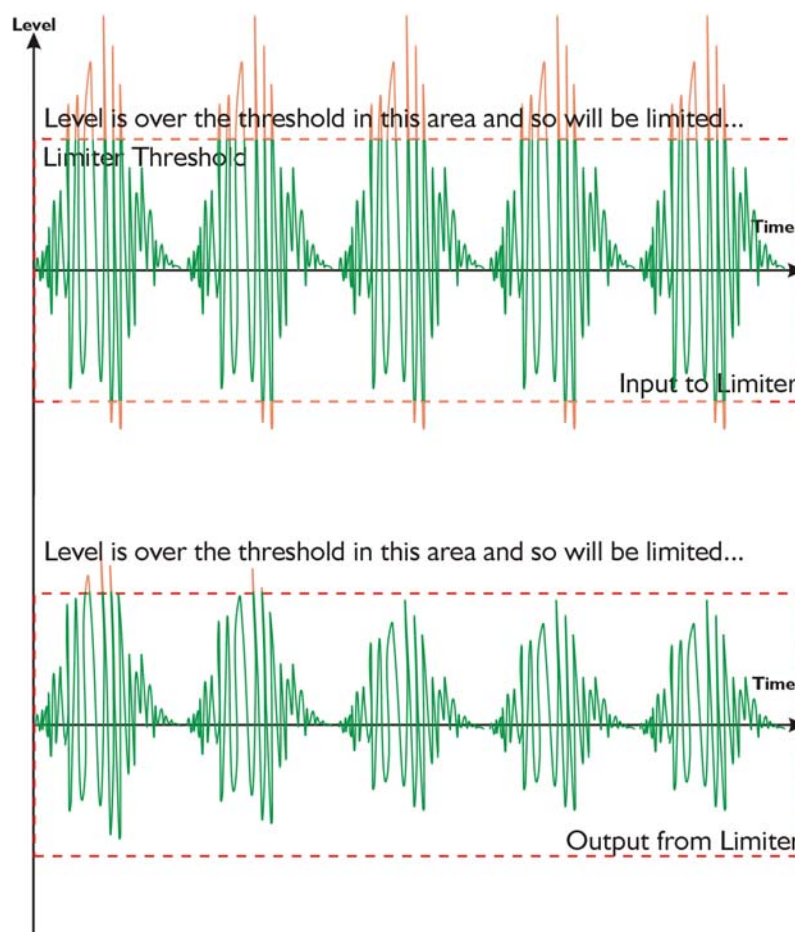


Limiters – Use (and Abuse)

Limiting is probably the most widely misused dynamics process. The primary use of a limiter is in the area of protection, where the output can be set to never exceed an absolute maximum threshold. The reduction of signal level introduced in the process of ensuring this threshold is not exceeded should be controlled in such a manner so as to introduce as little distortion as possible.

Main Limiting Methods.

The diagram below shows the onset of limiting on a repetitive waveform that would normally be over the limiter threshold.



Due to the slow setting of the attack time ($> 20\text{ms}$), the limiter takes a relatively long period to react to the first transient that exceeds the threshold.

However, due to a long release time (100's ms), it does not recover too quickly from the gain reduction when the signal drops below the threshold again and so keeps further excursions in check.

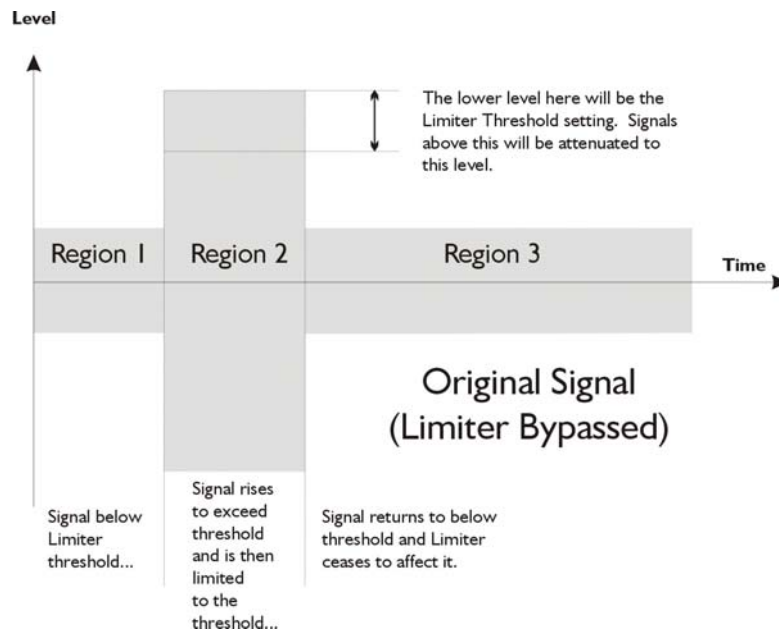
This type of limiting is known as *program limiting* and is designed to control the *average signal*

level.



Another limiting method is known as *peak limiting* and is characterised by the limiter having very a fast attack time ($< 10\text{ms}$) and relatively fast release time (generally about 4 times the attack). This method is designed to prevent any excursion above the threshold (and so into the over-range region), and to recover from the gain reduction as fast as is musically possible.

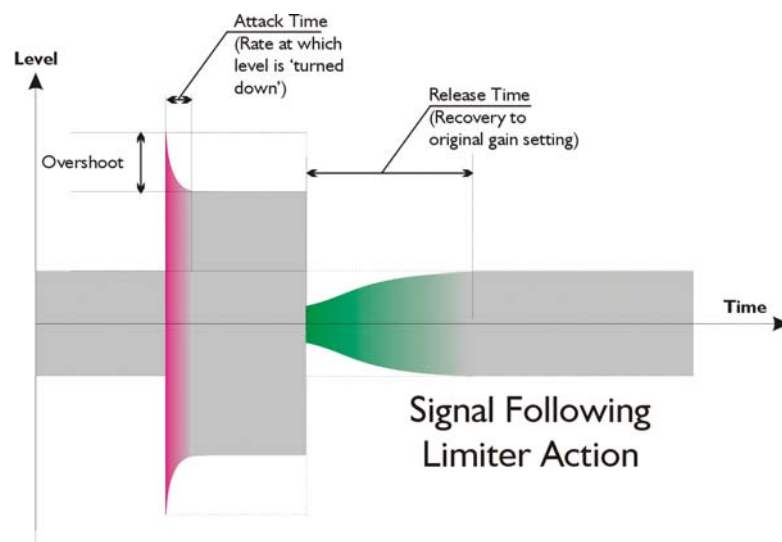
Consider the familiar signal burst diagram below, where region 2 represents the program material that has gone over the limiter threshold.



In extreme cases, the limiter could be required to react instantaneously to the over-range condition, and prevent *any* overshoot. This sort of limiting is often utilised when mastering digitally – above the 0dBfs point there is no margin of ‘give’, unlike analogue tape mastering.

‘Brickwall’ limiting, as this is called, has to delay the main signal so that the limiter can actually pre-empt the excursion and react before it actually happens.

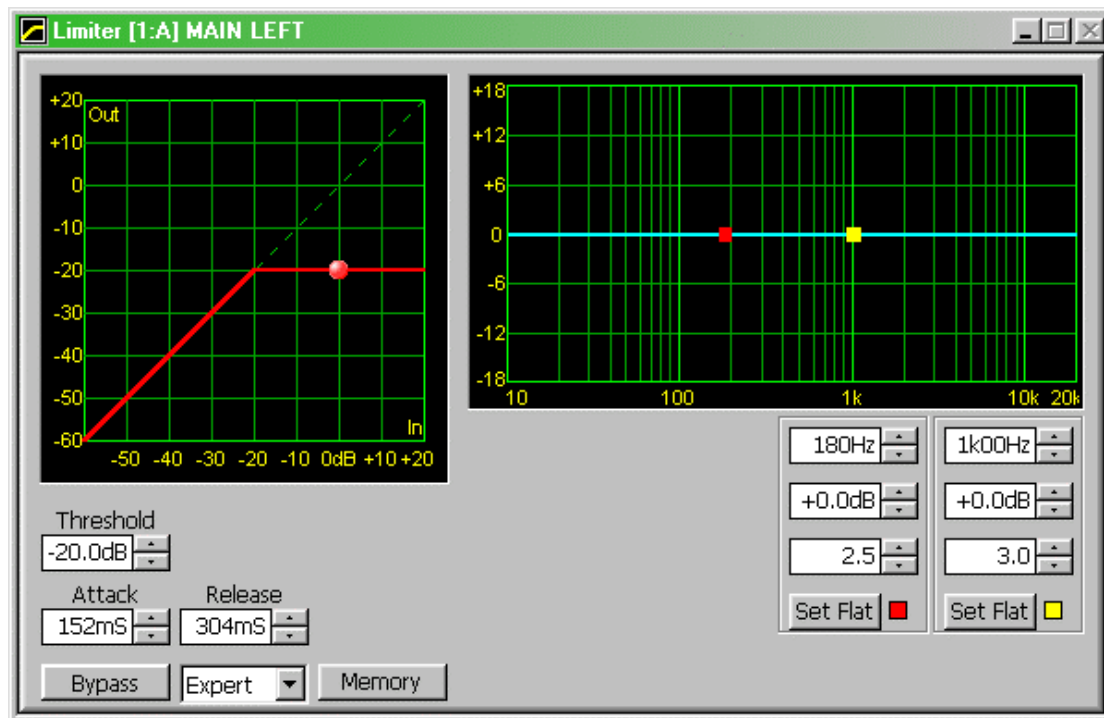
Engineering this normally involves inserting a delay line into the main signal path. SiDD does not need to do this, as it is possible to select a *negative* attack time, which effectively moves the sidechain signal *back in time relative to the main signal*, rather than delaying the main one.



Please read the section entitled ‘Look Ahead Delay – Pre-emptive Action’ on page 52 for a fuller explanation of this facility.



Editing the Limiter

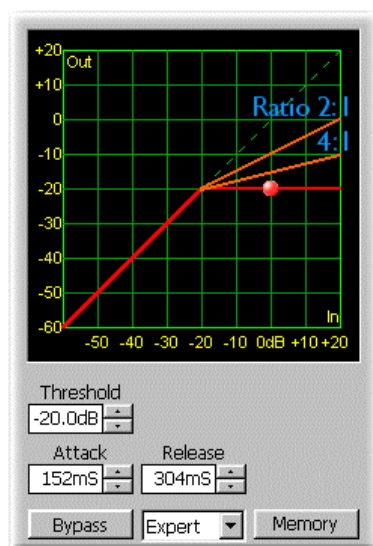


The limiter has limited⁹ controls available for adjustment. Aside from the threshold, there are the attack and release settings and access to a pair of parametric sidechain filters.



Remember: A parametric filter may be changed to a shelving type response by setting the 'Q' control one nudge past the widest 'Q' value. This will select 'LoShelf', followed by 'HiShelf'.

Note that the gain of the filter must be set to 0dB to access these responses types.



As this is a limiter, above the threshold, the 'curve' is completely flat, implying that for any input level over, in this case, -20dB the output will remain at -20dB. Compare the transfer function of the limiter with the compressor, threshold -20dB, and ratios of 2:1 and 4:1.

Once the ball has reached the threshold and starts to move along the horizontal part of the curve, limiting is taking place. The amount of limiting can be easily calculated by measuring how far to the right of the threshold the ball is at any time – in this example the signal is 20dB into limiting (threshold at -20dB and input at 0dB).

⁹ Limiter...limited...hehe
SiDD PC Software Version 2.00

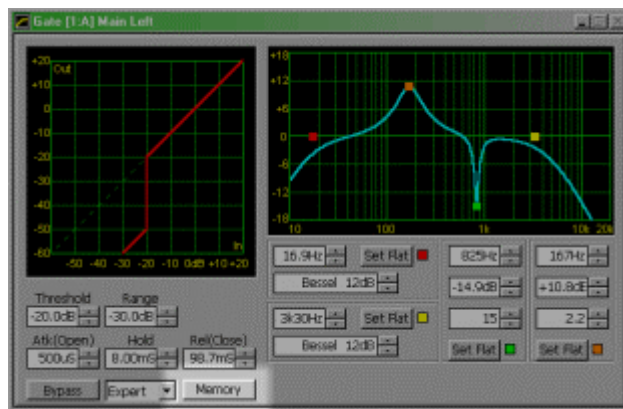


Memory Handling and Features

Before describing how to store and recall settings, it's worth explaining how memories are organised and handled by the software itself.

The memories in SiDD are **completely independent** to those in the software – it is not possible to store and recall memories kept in an individual unit's memory. Instead, the software handles all memories independently, offering virtually unlimited storage possibilities and the ability to accumulate files of favourite memories.

Accessing Memories

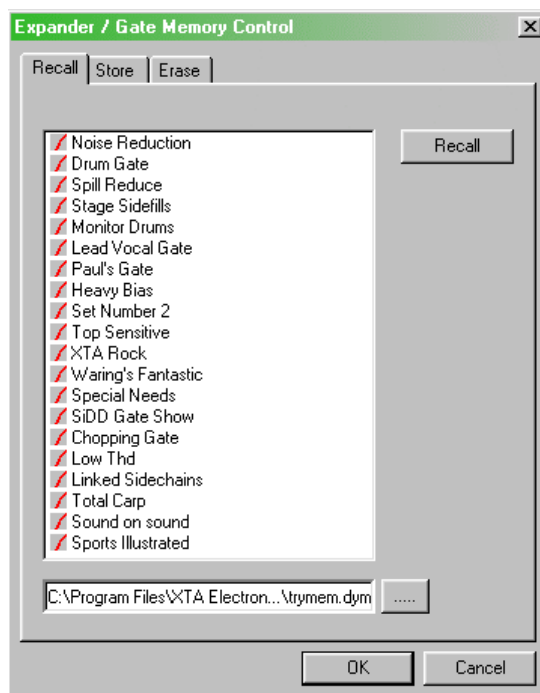


Memories are accessed on the corresponding module's editing window – in this example the noise gate – which will display the memory access window with the available memories for this type of module.

From this window, it's possible to recall, store, or erase a memory from the list. Remember that the memories in the unit will NOT appear in this list, and no settings stored by pressing the 'Store' button will be sent to the unit.

Remember to press the Recall button, and not just the 'OK' button.

The ellipsis ('...') button at the bottom of the pane allows a different file to be selected, containing an alternate selection of memories. Note that this file contains **all** the memory types – selecting a different set will update the memories available in the in/out, dynamic eq, compressor and limiter views.



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