SiDD™

Seriously Intelligent Digital Dynamics

Operators Manual
Revision 1.02

XTA Electronics Ltd.
The Design House,
Vale Business Park,
Worcester Road,
Stourport-on-Severn,
Worcs. DY13 9BZ.
England
Tel: 01299 879977 (Intl. +44 1299 879977)
Fax: 01299 879969 (Intl. +44 1299 879969)
Web: http://www.xta.co.uk

Designed & written by Waring Hayes 03/2001
If you have any comments or suggestions about this manual, please contact XTA at the address above, or email manuals@xta.co.uk

SiDD Quick Reference

Editing Parameters: press '**A**'/'**COMP**'/'**LIMIT'** etc. for access to relevant section. 1st press accesses that module's threshold and most important parameters. To scroll through a module's parameters, use the BACK and NEXT keys. 2nd press accesses last viewed parameter. 3rd press will display module overview information.

Accessing Menus: press the MENU key. Use the BACK and NEXT keys to select the sub-menu required, and enter using the ENTER key. This applies to all levels of menu. ENTER always confirms selections. Double pressing MENU will jump to the last viewed selection.

The Menus and their Contents:

<u>Memory Sub-menu:</u> Used for storing and recalling settings either module by module, or for entire channels. Also used for storing and recalling complete memory sets using a PC card, and preview of memories and presets.

<u>Configuration Sub-menu:</u> Used for setting expander or gate action, stereo linking, control ganging, editing level, and naming channels.

<u>Security Sub-menu:</u> Used for locking the front panel of the unit with a four digit code.

<u>System Sub-menu:</u> Used to view the unit's status, and select various global options such as parametric EQ 'Q' or bandwidth units, LCD brightness, LCD contrast, and meter mode (meters on or off in bypass). Unit software updates from a PC card and unit cloning are accessible here.

<u>Ext. Interface Sub-menu:</u> Used to configure the remote control interface(s).

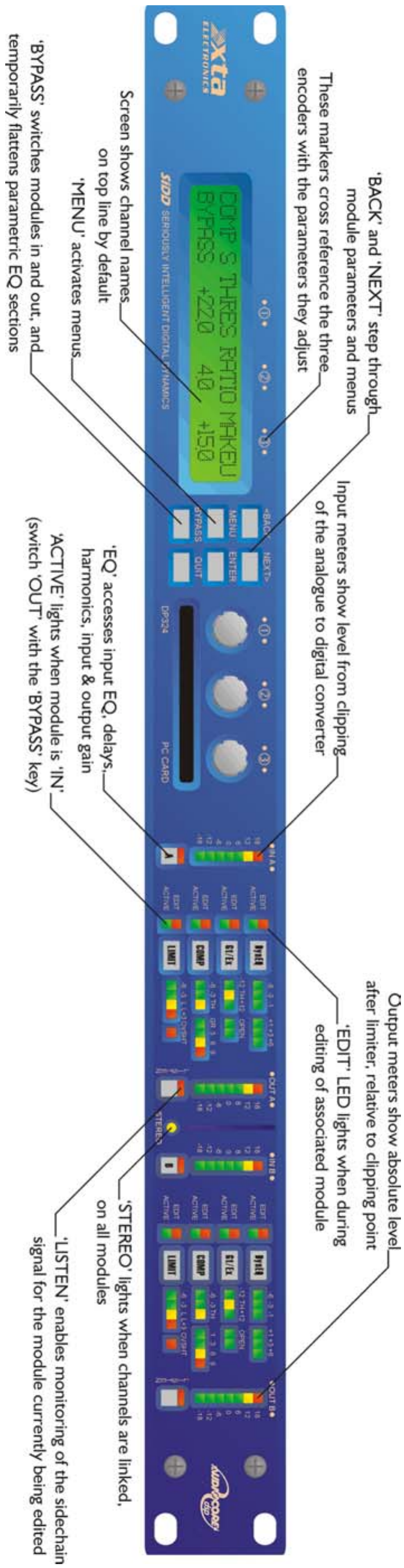
<u>AES/EBU Sub-menu:</u> Used to configure the digital audio interface(s), if fitted.

Notes:

The sidechain listen function only operates when a module is being edited.

The input meters show level, in dB, from clipping the input converter, prior to the input gain control.

The STEREO LED only illuminates when all modules are stereo ganged.



Contents

CE

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

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

Important Safety Information

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

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

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

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

Thanks

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

Unpacking *SIDD*

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

Introduction

SIDD is a powerful DSP based audio processor, ideally suited for live sound applications, where it combines the functions of a multitude of conventional products in a compact 1U unit. To achieve this, it has two inputs and two main outputs, together with two auxiliary outputs.

Each channel has a six band parametric equaliser and delay line, a separate variable gain 'A.D.T.' delay and a harmonics generator, as well as four fully featured dynamics modules, designed to cope with all possible processing applications.

Remote control is catered for in the form of MIDI, RS232 and RS485 ports, and multiple user memories are provided for the storage and recall of settings. Security lock-out is available for all controls.

SIDD is also available with optional AES/EBU digital inputs and outputs.
It has been designed for quick, intuitive adjustment through the use of multiple controls to provide an easy-to-use interface. Alternatively, it may be controlled externally by XTA's proprietary Windows™ software, along with up to thirty one other units.

Features

- Superb audio quality – carefully optimised double precision signal processing coupled with a 40-bit internal data path ensures a dynamic range in excess of 110dB. The high sampling rate means minimal filtering providing exceptional sonic purity.

- A highly accurate 6 band parametric equaliser on each input, providing +15 to –30dB of gain at centre frequencies between 20Hz and 20kHz, with a wide range of 'Q's available between 0.4 to 128. All parameters feature fine resolution with 1/36 octave frequency steps, 0.1dB gain increments, and 100 'Q' settings. Any parametric section can also be set to operate as a high or low shelving filter.

- A fully featured compressor with two bands of sidechain equalisation, and featuring variable knee and 'look-ahead' delay for absolute control.

- A noise gate / expander, again with dedicated sidechain processing – high and low pass filters plus two parametrics. The 'look ahead' design allows the gate to open before the signal arrives, completely eliminating clicks.

- A dynamic equaliser – compress or expand only the frequency band you choose, with unprecedented control and precision.

- A delay line with separate 'A.D.T' module that may be routed to the auxiliary outputs. This allows for 'staging' of performers 'front/back' and 'left/right' of the soundstage.

- A 'brickwall' limiter with sidechain EQ and 'look ahead' ability for pre-emptive adaptation to signal peaks, and so complete prevention of overshoot.

- A harmonics generator to introduce controlled levels of second and third harmonic distortion, for simulation of 'valve' sound characteristics.

- Three velocity-sensitive encoders provide a familiar and intuitive control format with all filter information displayed simultaneously on a 2 x 24 character backlit LCD screen.

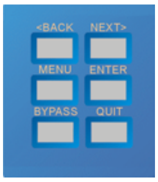
- The comprehensive standard specification also includes 256 memories, PC Card storage and remote control via MIDI, RS232 or RS485 ports, with security lock-out.

- AES/EBU digital input and output interfaces are available as an option.

- Input and output balancing transformers are also available as an option.

Front Panel Familiarisation

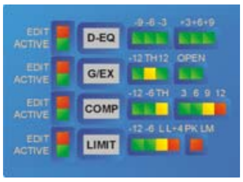
LCD Screen : shows menu options, channel information, and various parameters as they are adjusted.



Next Key : moves forwards through the list of available parameters.
Back Key : moves back through the list of available parameters.
Menu Key : activates the main menu. Pressing a second time selects the last menu edited. Selection of different menus is accomplished using the Back and Next keys, or by turning the encoder 3.
Enter Key : enters the chosen menu and confirms choices.
Bypass Key : bypasses the currently selected processing module.
Quit Key : exits the menu.

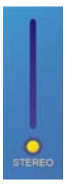
Encoders : three velocity sensitive controls allow the relative parameter displayed on the LCD screen to be adjusted.
PC Card : PCMCIA card slot allows back-up of memory sets, unit cloning data and facilitates the download of software updates and preset files.

Input Meter and EQ Edit Key: displays available headroom below clipping, before the input gain control. The red LED illuminates 3dB below the clipping point of the input circuitry. EQ, delays, harmonics and gains are all accessible with this key.



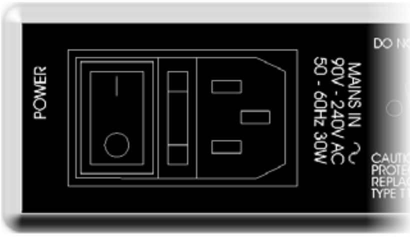
Module Edit Keys: pressing the required module key will display the main parameters available for editing. The associated EDIT LED will illuminate. Use the 'BYPASS' key to switch the module in or out of the signal path whilst edit is active. 'In' is shown by the 'ACTIVE' LED being illuminated.

Output Meter: display headroom below clipping for all modules combined output.
Listen Key: routes the appropriate sidechain signal to the outputs, dependant on the module being edited.



Stereo LED: Illuminates when all modules are linked for stereo operation.

Rear Panel Connections



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

External: RS232 standard via a 9 pin D-type connector, for connection to a PC. Data is converted to RS485 standard and relayed to slave units via the RS485 sockets.
Midi In: 5 pin DIN socket. See page 60 for more information.

RS485 In-Out: XLR sockets. Used for transmission of remote control data over long distance or multiple unit applications. See page 60 for more information.

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

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

Operating *SIDD*

Note about operation with SIDD™ Windows™ software.

The following operating information covers control of SIDD via the front panel controls only. Please consult the manual supplied with this software for information regarding computer control.

Preliminary Set-up

The procedure below should be followed when first installing *SIDD*.

Start with a clear audio path! By default, all processing modules are switched out of circuit, and all EQ is set to 0dB. The result of this is that the signal going into the unit should be at exactly the same level as that coming out. It's best to introduce modules one at a time, if a lot of processing is being applied. This way, the effects of each module on the final signal can be carefully gauged.

Use the edit keys on each module, together with the BACK and NEXT keys to select the parameters required for editing. Note that the default function mode is 'Expert' – that is, all parameters will be displayed. See page 50 for further details of how to adjust the function mode.

Note that when in a menu, ENTER is always used to confirm selections. The current selection is marked with an asterisk '*'.

Note that if no action is taken in menu mode, the unit will return to normal 'default' mode. Repeat the above directions to return to menu mode.

Audio Function Screens – Input EQ etc.

Pressing the key below the input meter once will display the first of eight possible screens associated with editing input EQ, gain and delay parameters, and the harmonics generator. The first press will always display the input/output gain adjustment. A second press will display the last selected parameter screen. A third press will show an overall channel summary, which will be explained below.

Gain Screen

Input gain is adjustable in 0.1dB increments over a ±15dB range. It is post converter, and so does not affect the reading displayed on the input meter. The output gain has the same range and increment size, but will affect the reading on the output meter, dependant on the 'Output Mode' setting.

For further details on this topic, see the section entitled "Channel Routing" on page 48.

Main Delay Screen

The main delay is adjustable in 21uS steps up to a maximum of 300mS.

ADT Screen

The ADT delay is adjustable in 21uS steps up to a maximum of 100mS. A separate gain control is provided for this delay tap, offering adjustment from 'Off', through –60dB up to 0dB in 0.5dB steps. When the 'Output Mode' is set to 'ADT', the 'BYPASS' key becomes active, allowing the contribution of the secondary delay to be removed from the signal path (effectively setting the gain to 'Off' temporarily). Gain range is increased to -60 through +15dB maximum in 'SPLIT' mode.

Harmonics Screen

The harmonics generator allows both the second and third harmonic distortion content of the signal to be precisely controlled. The addition of second harmonic distortion can be used to simulate the affect of passing the signal through a 'valve stage', increasing perceived warmth. Both second and third harmonic content are variable from 'Off' through –60dB to 0dB in 0.5dB steps. Additionally, the 'BYPASS' key may be used to temporarily switch the harmonics off.

High Pass Filter Screen

The high pass filter may be varied over a frequency range from below10Hz up to 16kHz, in 1/36th Octave steps. Using encoder 2, the response may be switched between 12dB/Octave Bessel and 12dB/Octave Butterworth. Note that whilst the 'BYPASS' key may be used to switch the high pass filter out of circuit, the BYPASS text in the bottom left corner of the screen does not appear. Instead, 'Flat' appears in place of 'HighPass' on the top right of the screen.

Low Pass Filter Screen

The low pass filter may be varied over a frequency range from 35Hz up to 22kHz, in 1/36th Octave steps. Using encoder 2, the response may be switched between 12dB/Octave Bessel and 12dB/Octave Butterworth. Note that whilst the 'BYPASS' key may be used to switch the high pass filter out of circuit, the BYPASS text in the bottom left corner of the screen does not appear. Instead, 'Flat' appears in place of 'LowPass' on the top right of the screen.

Note that, to avoid loosing all signal through a channel, *SIDD* will not allow the high pass filter to be set to a frequency above that of the low pass filter, and vice versa. For example, this means that it the high pass was set to 250Hz, the low pass would not extend down to 35Hz, but would stop at 250Hz. This action is irrespective of the bypass settings of the high and low pass.

Parametric Filter Screens

Each parametric section can be positioned at a frequency from 20Hz to 20kHz and features a wide range of 'Q's to produce response curves ranging from broad to notch. The gain control ranges from +15dB to -30dB in 0.1dB steps. Frequency steps are 1/36th Octave resolution for precise control. Since all filtering is achieved in DSP all settings are re-settable with absolute accuracy and in ganged mode parameters track identically. Very narrow band notch filters (maximum 'Q' of 128) can be achieved and unlike analogue filters these tight 'Q' filters are entirely stable. The maximum notch depth is -30dB.

To switch a filter from parametric mode to shelving mode, set the gain of the filter to 0dB and then turn the 'Q' control anti-clockwise until 'HSF' or 'LSF' is displayed. Now set the frequency and gain as required. To set the filter back to a parametric section, just ensure that the gain is at 0dB, otherwise the 'Q' control will have no effect.

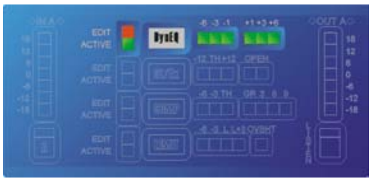
Note: To show parametric filters in bandwidth (BW) rather than 'Q', go into the 'System sub-menu', select 'filter Q or BW', select BW.

Input Channel Summary Screen

Whilst no adjustments can be made in this screen, it displays all the salient information about the input section.

The text 'CHANNEL' means that this information is about the input channel processing. (Similarly it would show 'LIMITER' when the same screen is selected during limiter editing).
Beside the 'CHANNEL' is 'A' – this denotes the input being edited – either 'A', 'B' or 'S' when the channels are stereo ganged. For information about how to stereo gang modules see the section entitled "Stereo Ganging of Modules" on page 44.
Lastly on the top line, the chosen name for the channel (in this case 'BACKLINE L'). This may be selected from a predefined list, or user defined from scratch. For details of how to select and/or edit the name, see the section entitled "Channel Names" on page 49.
The bottom line shows, on the left hand side, the name of the last recalled memory for this module ('38: VALVE SND'). Recall and storage of memories is dealt with in detail starting on page 32.
Finally, 'Expert' denotes the function mode of this module. The function mode is the level of editing complexity – 'Basic', 'Normal' or 'Expert'. This may be selected as described in the section entitled "Function Modes" on page 50.

Audio Function Screens – Dynamic EQ Module



Pressing the key marked 'DynEQ' once will display the first of five possible screens associated with editing the dynamic equaliser module . The first press will always display the threshold adjustment. A second press will display the last selected parameter screen. A third press will show an overall channel summary, which will be explained below.

Threshold/Ratio Screen

The parameters displayed on this first screen set the threshold point for the dynamic equalisation to begin. The filter will be applied to act as either a boost or a cut above or below this threshold. The threshold may be adjusted over the range of –30dB to +22dB, in 0.5dB steps. The action is set in the 'Operating Mode' screen, explained later in this section. The degree by which this action is accelerated is controlled by the Ratio parameter. The maximum ratio available will depend on the operating mode. For 'Cut Below' and 'Boost Above' and 'Boost Above' modes, the ratio has a maximum value of 2:1; for 'Cut Above' the maximum ratio is 4:1. The 'BYPASS' key is also active, bypassing the entire module.

Attack/Release Screen

The envelope parameters are displayed next – attack and release times for the action of the dynamic equalisation. Attack time covers the range 30uS up to 2.0 seconds. The release time covers the same range. Note that *SIDD* will not allow release times to be less than attack times, due to the likely 'parameter abuse' involved and subsequent poor performance.

Dynamic Filter Section

The dynamic filter itself is next. Remember that this filter should be thought of as being two identical filters – one in the sidechain listening to the main signal, and one in the main signal path, dynamically altering its contribution, based on the operating mode and envelope settings. The frequency and 'Q' parameters are the same as those for any filter block, and the 'MaxG' parameter sets the ceiling gain of the filter. If, for example, the 'MaxG' was set to +6dB and the operating mode was 'Boost Above', then no matter how much above the threshold the main signal went, the filter could only apply a maximum of 6dB of boost over its selected frequency range. It is variable over a range from 6dB up to 15dB for 'Boost' modes and 18dB for cut modes.

The filter behaviour may be changed from a parametric response to a high or low shelving filter. However, this adjustment may only be made when the Ratio is set to 1:1. Turning the 'Q' control (on rotary 2) fully anticlockwise will select high/low shelving responses or even full bandwidth, whereupon the dynamic EQ module acts as a normal compressor or expander, dependant on the operating mode set.

Note: In this module above all others, using the sidechain listen facility is crucial to the correct implementation of dynamic equalisation. See the section entitled "Context Aware Sidechain Monitoring" on page 29.

Operating Mode

The operating mode may only be changed when the Ratio is set to 1:1. Four modes are available. These are:

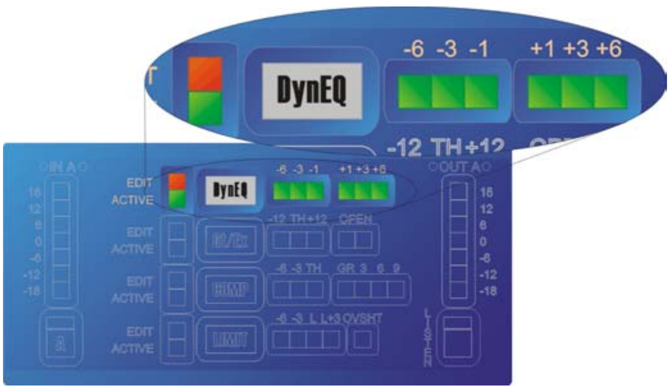
'Cut Above' – when the signal rises above the threshold, the band of frequencies selected by the D-EQ filter will be progressively reduced in level.
'Boost Above' - when the signal rises above the threshold, the band of frequencies selected by the D-EQ filter will be progressively increased in level.
'Cut Below' – when the signal drops below the threshold, the band of frequencies selected by the D-EQ filter will be progressively reduced in level.
'Boost Below' - when the signal drops below the threshold, the band of frequencies selected by the D-EQ filter will be progressively increased in level.

Dynamic Equaliser Summary Screen

Whilst no adjustments can be made in this screen, it displays all the salient information about the D-EQ module.

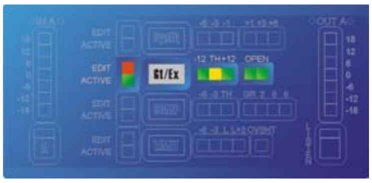
The text 'DYNAM EQ' means that this information is about the dynamic equaliser module. (Similarly it would show 'LIMITER' when the same screen is selected during limiter editing).
Beside the 'DYNAM EQ' is 'A' – this denotes the input being edited – either 'A', 'B' or 'S' when the channels are stereo ganged. For information about how to stereo gang modules see the section entitled "Stereo Ganging of Modules" on page 44.
There may be a lower case 'x' displayed beside the channel identifier. This means that the sidechains of the two dynamic equalisers are linked. This allows their effect to track perfectly when processing a stereo signal. Linking the sidechains is covered in the section entitled "Function Linking" on page 45.
Lastly on the top line, the chosen name for the channel (in this case 'BACKLINE L'). This may be selected from a predefined list, or user defined from scratch. For details of how to select and/or edit the name, see the section entitled "Channel Names" on page 49.
The bottom line shows, on the left hand side, the name of the last recalled memory for this module ('4: HF VOX BOOST'). Recall and storage of memories is dealt with in detail starting on page 32.
Finally, 'Expert' denotes the function mode of this module. The function mode is the level of editing complexity – 'Basic', 'Normal' or 'Expert'. This may be selected as described in the section entitled "Function Modes" on page 50.

What does the metering show?



For the dynamic EQ module, the three green LEDs marked "-6, -3, -1" will show the action of dynamic EQ when cut is being applied. Remember that the amount of cut (or boost) is dependant not only on the 'MaxG" parameter (which puts a limit on how much may be applied), but also on the ratio and the actual signal level relative to the threshold.
Similarly, the three LEDs marked "+1, +3, +6" will show dynamically applied boost, when the operating mode is set to "Boost Below" or "Boost Above". Under normal conditions, when the signal has not reached the threshold (from whatever side will cause dynamic EQ action) there will be no LEDs illuminated.

Audio Function Screens – Noise Gate / Expander Module



Pressing the key marked ‘Gt/Ex’ once will display the first of seven possible screens associated with editing the gate or expander module . The first press will always display the threshold adjustment. A second press will display the last selected parameter screen. A third press will show an overall channel summary, which will be explained below.

The screens for the expander will display different parameters in a few cases, compared to those for the gate. Each channel may be independently set up to operate as either an expander or a gate. For details of how to change the behaviour from expander to gate, see the section entitled “Expander / Gate Option” on page 42.

Threshold/Ratio Screen

The first two parameters displayed are dependant on the function mode as mentioned above. Both the expander and the gate have a Threshold point, which may be set anywhere in the range – 30dB to +22dB, in 0.5dB steps. The gate offers Range as its other parameter – this is the amount of attenuation applied by the gate when the signal drops below the threshold. So, for example, if a signal was 2dB below the threshold (which would have caused the gate to close) and the Range was 30dB, it would appear at the output at a level of 32dB below the threshold. It is variable from 10dB down to 70dB.

The expander displays its Ratio in place of the gate Range – the ratio determines how severely the signal is attenuated once it drops below the threshold. For example, a ratio of 2:1 means that if the input drops 10dB below the threshold, the output will be attenuated to effectively 20dB below. In reality, a ratio of 8:1 or above is severe enough for the expander to behave in a very similar fashion to the gate. The ratio may be adjusted from 1.02:1 down to 8:1.

Attack/Release Screen

The attack and release times for the expander or gate are displayed next. The attack time is defined at the time for the expander or gate to reach its full unattenuated level again, once the signal has gone back over the threshold. It is variable from –60uS (look ahead delay action) up to 500mS. The release time is defined as the time taken for the signal to reach its fully attenuated state, once the signal drops below the threshold. It is variable from 0.5uS to 2.5 seconds. Note that *SIDD* will not allow release times to be less than twice the attack times, due to the likely 'parameter abuse' involved and subsequent poor performance.

For further details about the negative attack time setting, and concept of look-ahead delay, see the section entitled "Look Ahead Delay – Pre-emptive Action" on page 65.

Sidechain High Pass Filter

The first of four filters available in the sidechain of the expander/gate is displayed next. The high pass filter acts identically to the one in the main signal path, accessible under the input EQ sections. The high pass filter may be varied over a frequency range from below10Hz up to 16kHz, in 1/36th Octave steps. Using encoder 3, the response may be switched between 12dB/Octave Bessel to 12dB/Octave Butterworth.

Note that the 'BYPASS' key is still active on this screen, but will switch the gate/expander out of circuit not the filter. The BYPASS text in the bottom left corner of the screen doesn't appear, but the 'ACTIVE' LED associated with the gate/expander edit key will extinguish.

Sidechain Low Pass Filter

The low pass sidechain filter may be varied over a frequency range from 35Hz up to 22kHz, in 1/36th Octave steps. Using encoder 2, the response may be switched between 12dB/Octave Bessel and 12dB/Octave Butterworth.

Note that, to avoid losing all signal through the sidechain, *SIDD* will not allow the high pass filter to be set to a frequency above that of the low pass filter, and vice versa. For example, this means that it the high pass was set to 250Hz, the low pass would not extend down to 35Hz, but would stop at 250Hz. This action is irrespective of the bypass settings of the high and low pass. If you are finding it impossible to get the gate to open, it might be that the sidechain filters are removing a large percentage of the available energy from the signal. Use the listen facility to check this. See the section entitled "Context Aware Sidechain Monitoring" on page 29.

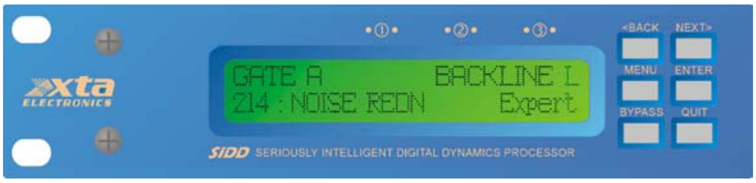
Sidechain Parametric Filters

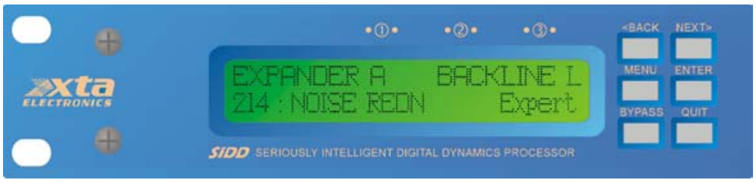
In addition to the high and low pass filters, there are two bands of fully adjustable parametric EQ available in the sidechain. These are designated 'SIDE 1' and 'SIDE 2' in the bottom left corner of the screen. Each parametric section can be positioned at a frequency from 20Hz to 20kHz and features a wide range of 'Q's to produce response curves ranging from broad to notch. The gain control ranges from +15dB to -30dB in 0.1dB steps. Frequency steps are 1/36th Octave resolution for precise control.

To switch a filter from parametric mode to shelving mode, set the gain of the filter to 0dB and then turn the 'Q' control anti-clockwise until 'HSF' or 'LSF' is displayed. Now set the frequency and gain as required. To set the filter back to a parametric section, just ensure that the gain is at 0dB, otherwise the 'Q' control will have no effect.

Note: To show parametric filters in bandwidth (BW) rather than 'Q', go into the 'System sub-menu', select 'filter Q or BW', select BW.

Gate/Expander Summary Screen





Whilst no adjustments can be made in this screen, it displays all the salient information about the gate/expander module. The information shown will be dependant on the gate/expander option.

The text 'GATE' means that this information is about the gate module. (Similarly it would show 'EXPANDER' as shown when the gate/expander option is set to expander).

Beside the 'GATE' is 'A' – this denotes the input being edited – either 'A', 'B' or 'S' when the channels are stereo ganged. For information about how to stereo gang modules see the section entitled "Stereo Ganging of Modules" on page 44.

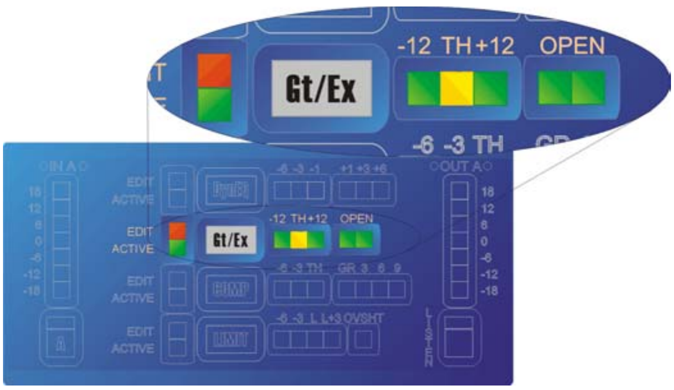
There may be a lower case 'x' displayed beside the channel identifier. This means that the sidechains of the two dynamic equalisers are linked. This allows their effect to track perfectly when processing a stereo signal. Linking the sidechains is covered in the section entitled "Function Linking" on page 45.

Lastly on the top line, the chosen name for the channel (in this case 'BACKLINE L'). This may be selected from a predefined list, or user defined from scratch. For details of how to select and/or edit the name, see the section entitled "Channel Names" on page 49.

The bottom line shows, on the left hand side, the name of the last recalled memory for this module ('Z14: NOISE REDN'). Recall and storage of memories is dealt with in detail starting on page 32.

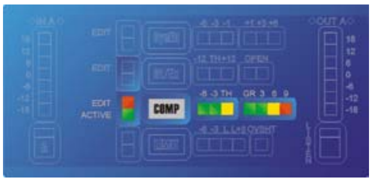
Finally, 'Expert' denotes the function mode of this module. The function mode is the level of editing complexity – 'Basic', 'Normal' or 'Expert'. This may be selected as described in the section entitled "Function Modes" on page 50.

What does the metering show?



In gate or expander mode the 3 LEDs (green/yellow/green) show level around the threshold – 12dB below, at threshold, and 12dB above or more from left to right. In gate mode the two LEDs marked 'OPEN' illuminate together when the gate opens, and remain illuminated until the release phase has begun (i.e. during any Hold time that has been set.) In expander mode, they can light individually and show gain reduction of 3dB and 6dB, right to left respectively.

Audio Function Screens – Compressor Module



Pressing the key marked 'COMP' once will display the first of five possible screens associated with editing the compressor module . The first press will always display the threshold adjustment. A second press will display the last selected parameter screen. A third press will show an overall channel summary, which will be explained below.

Threshold/Ratio/Make-up Gain Screen

The threshold of the compressor is variable from –32dB up to +22dB in 0.5dB steps. The ratio may be adjusted from 1.02:1 up to a maximum of 16:1. The application of large amounts of compression normally necessitates some form of gain compensation, or make-up gain. Up to 15dB of make-up gain may be applied, in 0.5dB steps, using encoder 3. Note that the make-up gain is independent of both the input and output gain parameters. This means that bypassing the compressor may produce a significant change in level if heavy compression (i.e. a high ratio) has been chosen, and the likely use of substantial make-up gain.

Attack/Release/Knee Screen

The attack and release controls determine how quickly the compressor reacts to the input signal, and how quickly it restores the original level, after the signal has dropped below the threshold again. The attack time is variable from –60uS (look ahead delay action) up to 152mS. The release time is variable from 49mS to 3.2 seconds. Note that *SIDD* will not allow release times to be less than twice the attack times, due to the likely 'parameter abuse' involved and subsequent poor performance. The 'Knee' parameter allows the onset of compression to be softened – that is then compression will begin at a lower ratio, below the threshold set, and reach full ratio at the threshold. The knee can be moved up to 12dB below the threshold – 0dB is 'hard', 12dB is 'soft.

For further details about the negative attack time setting, and concept of look-ahead delay, see the section entitled "Look Ahead Delay – Pre-emptive Action" on page 65.

Sidechain Parametric Filters

There are two bands of fully adjustable parametric EQ available in the sidechain. These are designated 'SIDE 1' and 'SIDE 2' in the bottom left corner of the screen. Each parametric section can be positioned at a frequency from 20Hz to 20kHz and features a wide range of 'Q's to produce response curves ranging from broad to notch. The gain control ranges from +15dB to -30dB in 0.1dB steps. Frequency steps are 1/36th Octave resolution for precise control.

To switch a filter from parametric mode to shelving mode, set the gain of the filter to 0dB and then turn the 'Q' control anti-clockwise until 'HSF' or 'LSF' is displayed. Now set the frequency and gain as required. To set the filter back to a parametric section, just ensure that the gain is at 0dB, otherwise the 'Q' control will have no effect.

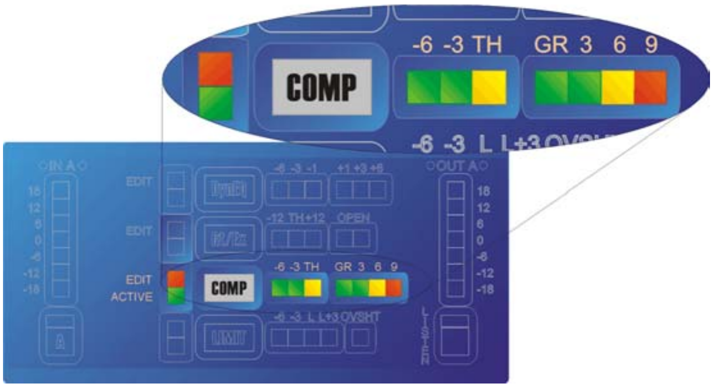
Note: To show parametric filters in bandwidth (BW) rather than 'Q', go into the 'System sub-menu', select 'filter Q or BW', select BW.

Compressor Summary Screen

Whilst no adjustments can be made in this screen, it displays all the salient information about the compressor module.

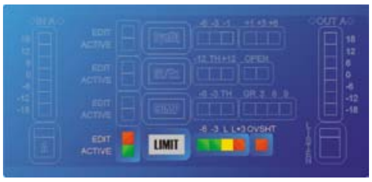
The text 'COMPRESS' means that this information is about the dynamic equaliser module. (Similarly it would show 'LIMITER' when the same screen is selected during compressor editing).
Beside the 'COMPRESS' is 'A' – this denotes the input being edited – either 'A', 'B' or 'S' when the channels are stereo ganged. For information about how to stereo gang modules see the section entitled "Stereo Ganging of Modules" on page 44.
There may be a lower case 'x' displayed beside the channel identifier. This means that the sidechains of the two compressors are linked. This allows their effect to track perfectly when processing a stereo signal. Linking the sidechains is covered in the section entitled "Function Linking" on page 45.
Lastly on the top line, the chosen name for the channel (in this case 'BACKLINE L'). This may be selected from a predefined list, or user defined from scratch. For details of how to select and/or edit the name, see the section entitled "Channel Names" on page 49.
The bottom line shows, on the left hand side, the name of the last recalled memory for this module ('12: KICK DRUM'). Recall and storage of memories is dealt with in detail starting on page 32.
Finally, 'Expert' denotes the function mode of this module. The function mode is the level of editing complexity – 'Basic', 'Normal' or 'Expert'. This may be selected as described in the section entitled "Function Modes" on page 50.

What does the metering show?



The group of three LEDs (green, green, yellow) show how close the sidechain signal is to the threshold – 6dB below, 3dB below, and at the threshold, left to right. Coupled with these are the group of four LEDs marked "GR, 3, 6, 9". These show the application of gain reduction as the signal goes over the threshold – the first LED shows 1dB of gain reduction.

Audio Function Screens – Limiter Module



Pressing the key marked 'LIMIT' once will display the first of five possible screens associated with editing the limiter module . The first press will always display the threshold adjustment. A second press will display the last selected parameter screen. A third press will show an overall channel summary, which will be explained below.

Threshold/Overshoot Screen

The limiter threshold may be adjusted from –32dB to +22dB in 0.5dB steps. Linked to the setting of the limiter threshold is the Overshoot parameter. This is not linked to the audio signal path, except for monitoring purposes, and sets the threshold of the 'OVSHT' LED for that channel. For example, the threshold of the limiter is set to 0dB, and the overshoot threshold is set to +6.0dB, the 'OVSHT' LED will light if the limiter output exceeds the threshold by 6dB or more. This might happen if the limiter attack time is sufficiently slow so as to allow fast transients through. The range of monitoring extends from 0dB (i.e. exactly the same as the limiter threshold) to 12dB above it, in 0.5dB steps.

Attack/Release Screen

The attack and release controls determine how quickly the limiter reacts to the input signal, and how quickly it restores the original level, after the signal has dropped below the threshold again. The attack time is variable from –60uS (look ahead delay action) up to 152mS. The release time is variable from 49mS to 3.2 seconds. Note that *SIDD* will not allow release times to be less than twice the attack times, due to the likely 'parameter abuse' involved and subsequent poor performance. The 'Knee' parameter allows the onset of compression to be

For further details about the negative attack time setting, and concept of look-ahead delay, see the section entitled "Look Ahead Delay – Pre-emptive Action" on page 65.

Sidechain Parametric Filters

There are two bands of fully adjustable parametric EQ available in the limiter sidechain. These are designated 'SIDE 1' and 'SIDE 2' in the bottom left corner of the screen. Each parametric section can be positioned at a frequency from 20Hz to 20kHz and features a wide range of 'Q's to produce response curves ranging from broad to notch. The gain control ranges from +15dB to -30dB in 0.1dB steps. Frequency steps are 1/36th Octave resolution for precise control.

To switch a filter from parametric mode to shelving mode, set the gain of the filter to 0dB and then turn the 'Q' control anti-clockwise until 'HSF' or 'LSF' is displayed. Now set the frequency and gain as required. To set the filter back to a parametric section, just ensure that the gain is at 0dB, otherwise the 'Q' control will have no effect.

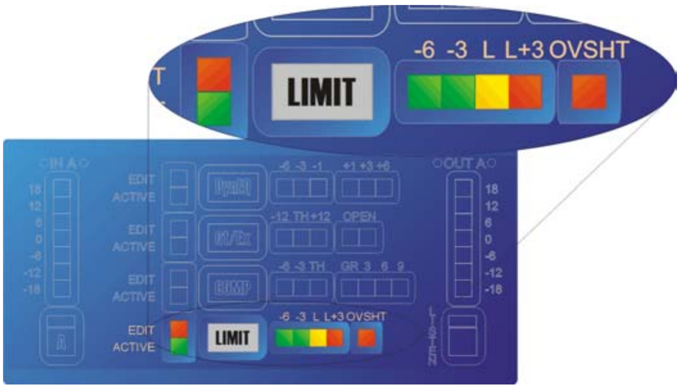
Note: To show parametric filters in bandwidth (BW) rather than 'Q', go into the 'System sub-menu', select 'filter Q or BW', select BW.

Limiter Summary Screen

Whilst no adjustments can be made in this screen, it displays all the salient information about the limiter module.

The text 'LIMITER' means that this information is about the dynamic equaliser module.
Beside the 'LIMITER' is 'A' – this denotes the input being edited – either 'A', 'B' or 'S' when the channels are stereo ganged. For information about how to stereo gang modules see the section entitled "Stereo Ganging of Modules" on page 44.
There may be a lower case 'x' displayed beside the channel identifier. This means that the sidechains of the two limiters are linked. This allows their effect to track perfectly when processing a stereo signal. Linking the sidechains is covered in the section entitled "Function Linking" on page 45.
Lastly on the top line, the chosen name for the channel (in this case 'BACKLINE L'). This may be selected from a predefined list, or user defined from scratch. For details of how to select and/or edit the name, see the section entitled "Channel Names" on page 49.
The bottom line shows, on the left hand side, the name of the last recalled memory for this module ('234: PEAK STOP'). Recall and storage of memories is dealt with in detail starting on page 32.
Finally, 'Expert' denotes the function mode of this module. The function mode is the level of editing complexity – 'Basic', 'Normal' or 'Expert'. This may be selected as described in the section entitled "Function Modes" on page 50.

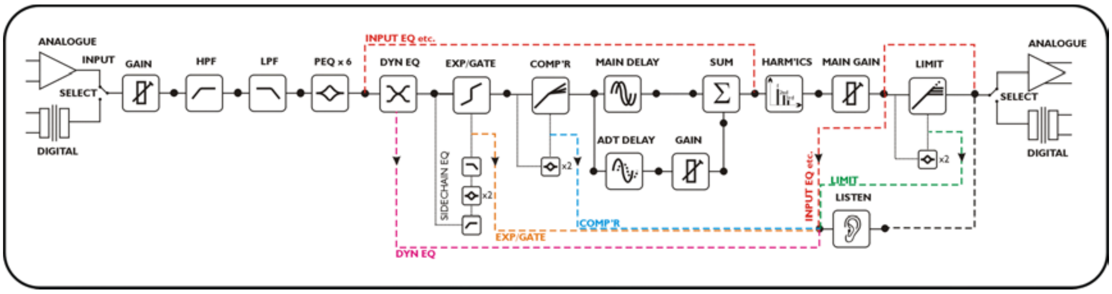
What does the metering show?



The left hand metering block shows how close the sidechain signal is to the threshold of limiting. The green LEDs show 6dB and 3dB from the threshold, the yellow LED (marked 'L') on the limit threshold, and the RED (marked 'L+3') 3dB into limiting.
As described in the section above, "Threshold/Overshoot Screen", the 'OVSHT' LED lights when the output of the limiter has gone over the threshold by an amount determined by the 'OVSHT' parameter.

Context Aware Sidechain Monitoring

As with traditional dynamics processors, it is possible to listen to the 'key' or sidechain signal during the setup/editing of any part of *SIDD*. Rather than add unnecessary complexity to the front panel with individual keys for sidechain monitoring of each module, both channels have one key dedicated to the listen facility. During editing, pressing the 'LISTEN' key will route the sidechain from the module being edited to the output, according to the diagram below. (For safety's sake, the output is dropped by 6dB in case excessive boost has been applied using the EQ.)



The signal for each sidechain path is derived as follows.

Dynamic EQ: The output is taken from the sidechain filter of the dynamic EQ module. This is a single parametric filter (which may be set to operate as a shelving filter). Note that the 'MaxG' parameter associated with the dynamic EQ module is not analogous to the gain of a parametric band, and so has no effect on the level appearing when 'LISTEN' is invoked. The input EQ (but not delay, ADT, or harmonics) will also be included in the sidechain signal.

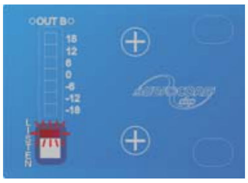
Expander/Gate: The output is taken from the end of the sidechain filtering associated with the expander/gate, which encompasses high pass, low pass, and two parametrics. The effect of any dynamic EQ will also be included in the signal path and so will affect the threshold setting of the expander/gate, in addition to the input EQ as before.

Compressor: The output is taken from the second of the two parametric filters in the compressor sidechain. It will include the effect of the dynamic EQ, input EQ *and* the expander/gate, so if there does not appear to be any signal present, the gate may be closed.

Limiter: The output is taken from the second of the two parametrics included in the limiter sidechain. It will include the combined contributions of the input EQ, the dynamic EQ, the expander/gate and the compressor. Additionally, as can be seen from the diagram, this in the only sidechain monitoring point that includes the delays and the harmonics.

Input EQ etc.: The output is effectively taken post limiter, but with effect of the dynamic EQ, expander/gate, compressor, limiter, and delays removed. This is shown on the diagram by the additional red dashed lines bypassing the dynamics processing, and the delays.

Pressing 'LISTEN' when no module is being edited will display "NO FUNCTION SELECTED!" and nothing will happen. Selecting a module for editing and then pressing 'LISTEN' will cause the 'LISTEN' LED to flash in time with the corresponding modules 'EDIT' LED.

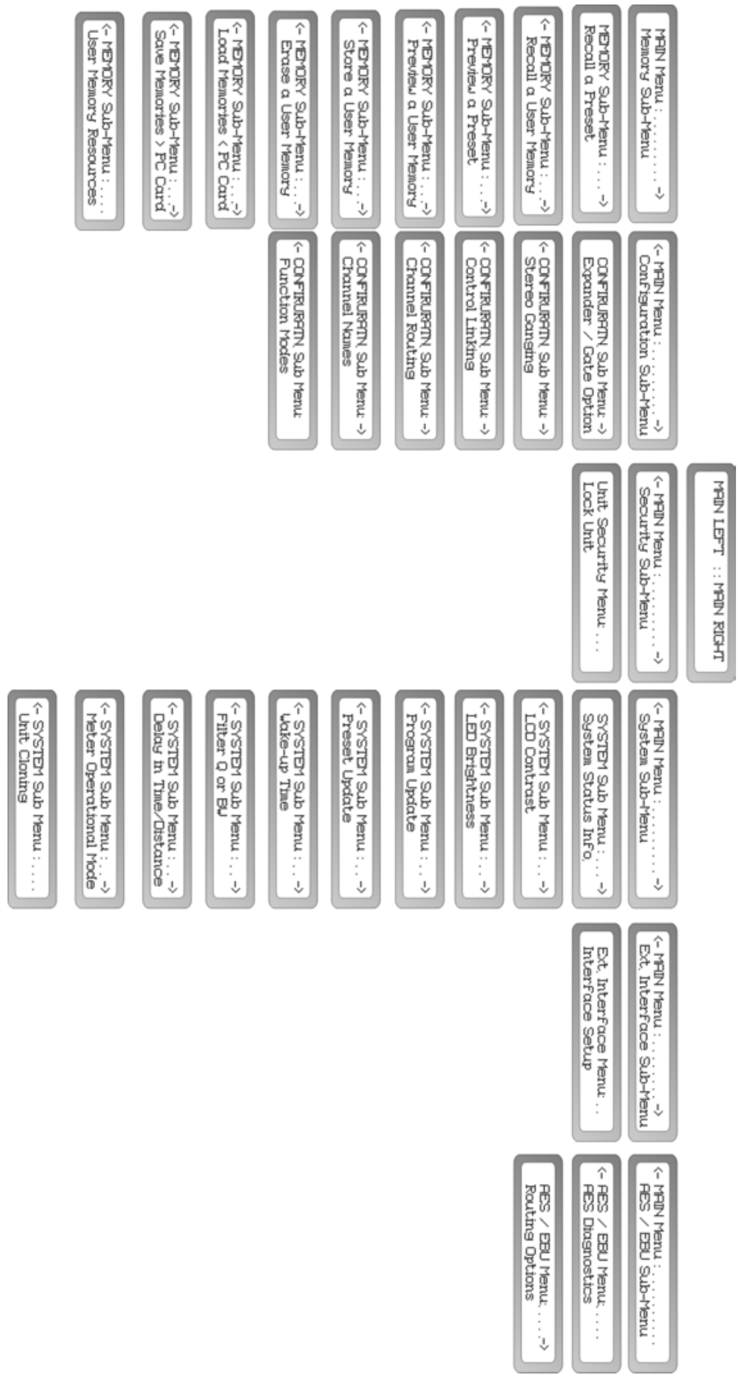


Listen may be cancelled by pressing it again, selecting another module to edit, or pressing 'QUIT' which will also quit the editing procedure.

Note: If pressing 'LISTEN' does not appear to have any effect, it is probably because of the 'Output Mode' setting. The 'Listen' routing can be set to appear on the auxiliary outputs instead of the main outputs. See the section entitled "Channel Routing" on page 48.

The Menu System

The menu system is arranged as shown in the tree diagram below.



Enter the Main Menu level by pressing the 'MENU' key once.
Navigate to the required sub-menu using the BACK and NEXT keys.
Enter the menu by pressing ENTER.
Navigate to the required function using the BACK and NEXT keys.
Select the function by pressing ENTER again.

Most functions require ENTER to be pressed to confirm selection, such as changing the Expander/Gate option; QUIT can always be pressed to cancel the current function and return to the default screen. The unit automatically exits from the menu system after approximately twenty seconds of inactivity.

Note: Pressing MENU twice from the default screen will jump to the last selected menu function for faster access.

Memory Sub-menu

The functions available in the Memory Sub-menu are as follows.

Recall a Preset	Choose an appropriate tool from the library of available presets as a starting point
Recall a User Memory	Choose settings previously created by a user for a particular purpose
Preview a Preset	Listen to available library settings before committing to a recall and overwriting current settings
Preview a User Memory	Listen to previously created settings before recalling and overwriting current settings
Store a User Memory	Commit a preferred configuration to memory for later use
Erase a User Memory	Clear a configuration to free up space if memory resources become low
Load Memories < PC Card	Backup currently stored configurations with new set from a PCMCIA card
Save Memories > PC Card	Copy all stored configurations onto a PCMCIA card for archiving purposes
User Memory Resources	Find out how many memories are free and what percentage of space is left

Memory Configurations and Presets – How Are They Organised?

In order to increase *SIDD*'s flexibility, each module may have its settings stored (or recalled) individually, or an entire channel may be stored (or recalled) at once. This will allow, for example, a preferred compressor setting to be recalled into the signal path after the input EQ has been set up, or the limiter introduced.

Prior to any access to memories, the type of memory must be selected be it a complete channel, or an individual module. From this point on, only memories of this type are displayed in the available list.

Presets are stored in non-volatile memory and are a valuable resource of useful settings for many applications. They are pre-loaded into the unit and cannot be overwritten, although they may be adapted if necessary to suit a particular application. The changed version may then be stored in a user memory.

There are 256 memory locations available within *SIDD*. These may contain any module's settings, or a complete channel's worth, although a complete channel will take up several memory slots resources. Remember, if the memory gets too full, the entire memory set can be copied onto a PCMCIA card for safe keeping, and the internal memory cleared and used again. See the section entitled "Saving Memories to a PC Card" on page 39.

Recalling a Preset or a User Memory

As the sequence of events for recalling presets and user memories is virtually identical, both will be dealt with in this section.

Selecting the memory sub-menu and then either

Recall a Preset or

Recall a User Memory

will prompt for the choice of what type of memory is to be recalled. As explained in the previous section this will be one of the following choices:

```
TYPE = [A]Complete Chan.
TYPE = [B]Complete Chan.
TYPE = [A]Input EQ etc.
TYPE = [B]Input EQ etc.
TYPE = [A]Dynamic EQ
TYPE = [B]Dynamic EQ
TYPE = [A]Expander
TYPE = [B]Gate
TYPE = [A]Compressor
TYPE = [B]Compressor
TYPE = [A]Limiter
TYPE = [B]Limiter
```

The Input EQ etc. selection refers to all the processing that may be edited under the input EQ section. This will include the delays, the EQ, the harmonics generator, and the input and output gains.

Note that the memory type for the selection of gate or expander will depend on the current configuration of the gate/expander module for that channel. If a channel is configured as a gate, then only gate memories will be offered. Each channel may be independently set up to operate as either an expander or a gate. For details of how to change the behaviour from expander to gate, see the section entitled "Expander / Gate Option" on page 42.

Press ENTER to confirm the selection of the type. The screen will show

```
Recall[A]Compressor
 234:Memory Name
```

and either turning encoder 3 or using the BACK and NEXT keys will scroll through the list of available memories of this type. Press ENTER once to select the memory. A second press is required to confirm and complete the memory recall.

Pressing QUIT at any time will return the unit to the default screen and exit the entire recall process.

Previewing a Preset or a User Memory

The preview feature allows memories to be sequentially loaded into the audio path without going through the entire recall procedure every time. This allows for quick comparisons to be made between memories, and failsafe confirmation of the settings prior to the recall. Note that the settings loaded for any particular module are not previewed in isolation – the rest of the signal path's contribution will still be audible. So, for example, if a compressor is being previewed, any input EQ, gate, dynamic EQ, or limiter modules already in use will still be heard.

As the sequence of events for previewing presets and user memories is virtually identical, both will be dealt with in this section.

Selecting the memory sub-menu and then either

Preview a Preset or

`Preview a User Memory`

will prompt for the choice of what type of memory is to be previewed. This will be one of the following choices:

```
TYPE = [A]Complete Chan.
TYPE = [B]Complete Chan.
TYPE = [A]Input EQ etc.
TYPE = [B]Input EQ etc.
TYPE = [A]Dynamic EQ
TYPE = [B]Dynamic EQ
TYPE = [A]Expander
TYPE = [B]Gate
TYPE = [A]Compressor
TYPE = [B]Compressor
TYPE = [A]Limiter
TYPE = [B]Limiter
```

The `Input EQ etc.` selection refers to all the processing that may be edited under the input EQ section. This will include the delays, the EQ, the harmonics generator, and the input and output gains.

Note that the memory type for the selection of gate or expander will depend on the current configuration of the gate/expander module for that channel. If a channel is configured as a gate, then only gate memories will be offered. Each channel may be independently set up to operate as either an expander or a gate. For details of how to change the behaviour from expander to gate, see the section entitled "Expander / Gate Option" on page 42.

Press ENTER to confirm the selection of the type. The screen will show

Preview[A]Compressor
234:Memory Name

and the setting appropriate to this module's memory will be loaded into the signal path and become immediately audible.

Remember - pressing QUIT at any time will return the unit to the default screen and exit the entire preview process, restoring the original settings immediately.

Now, either turning encoder 3, or using the BACK and NEXT keys will scroll through the list of available memories of this type. Each one will become audible in turn. Press ENTER to permanently recall the memory. Note that there is no confirmation stage in preview mode.

Let SIDD select the memory type for you!

In any memory mode, storing, recalling or previewing – *SIDD* selects the most likely type of memory you will use based on the module that you are editing. Rather than have to choose the memory type and channel from the list every time, *SIDD* will start you in the appropriate point in the list, leaving you to just press ENTER to confirm the type selection, or change it yourself.

Knowing how *SIDD* does it makes the memory access even simpler.

If you are currently editing a module, for example the compressor for channel A, press MENU, and access a memory function, *SIDD* will select
TYPE = [A]Compressor
in any memory menu that calls for a type selection to be made.

If you are editing a particular module, but want to quickly select a complete channel to be saved, press QUIT to end the editing session before pressing MENU and selecting the menu function. SIDD will then choose
TYPE = [A]Complete Chan.
based on the complete channel type based on the last channel you edited.

If you are not editing at all, pressing MENU and selecting a memory function will always select
TYPE = [A]Complete Chan.
i.e. the start of the list of possible memory types.

Remember - pressing MENU twice from the default screen will jump to the last selected menu function for faster access.

Storing a User Memory

Selecting the memory sub-menu and then either

    Store a User Memory

will prompt for the choice of what type of memory is to be stored. This will be one of the following choices:

    TYPE = [A]Complete Chan.
    TYPE = [B]Complete Chan.
    TYPE = [A]Input EQ etc.
    TYPE = [B]Input EQ etc.
    TYPE = [A]Dynamic EQ
    TYPE = [B]Dynamic EQ
    TYPE = [A]Expander
    TYPE = [B]Gate
    TYPE = [A]Compressor
    TYPE = [B]Compressor
    TYPE = [A]Limiter
    TYPE = [B]Limiter

The Input EQ etc. selection refers to all the processing that may be edited under the input EQ section. This will include the delays, the EQ, the harmonics generator, and the input and output gains.

Press ENTER to confirm the selection of the type. The screen will show

    Store[A]Compressor
    234:Memory Name

allowing selection of the memory location to store into. If the memory already has been used, a name will appear. If the memory is currently unused, then the screen will show

    Store[A]Compressor
    145: {empty}

Press ENTER to confirm the selection of the location to use. At this stage, the name of the memory may be edited, or generated from scratch in the case of an empty location. The screen shows

    Name Memory…
    145:XTA ROCK

with a cursor appearing under the first character of the name. Use encoder 1 to select the character, and the BACK and NEXT keys to move along the name. When satisfied with the name, press ENTER to complete the store procedure. Note that the name does not have to be edited – the last stored name will be kept if ENTER is pressed immediately after choosing the location. If the location was empty, then a completely blank name will be stored.

Erasing a User Memory

Selecting the memory sub-menu and then either

Erase a User Memory

will prompt for the choice of either an individual memory

Erase a User Memory ->
Erase a User Memiory. . .?

or all user memories

Erase a User Memory ->
Erase ALL User Memiories.?

Press ENTER to confirm the selection of the type. If an individual memory has been chosen, the screen will show

Select A Memory to Erase
234: Vocal EQ Set One

allowing selection of the memory location to erase. Use encoder 3 to choose.

Press ENTER to confirm the selection. The screen shows

Erase this Memory. . .?
Press [ENTER] to Confirm

A second press is required to confirm and complete the memory erase.

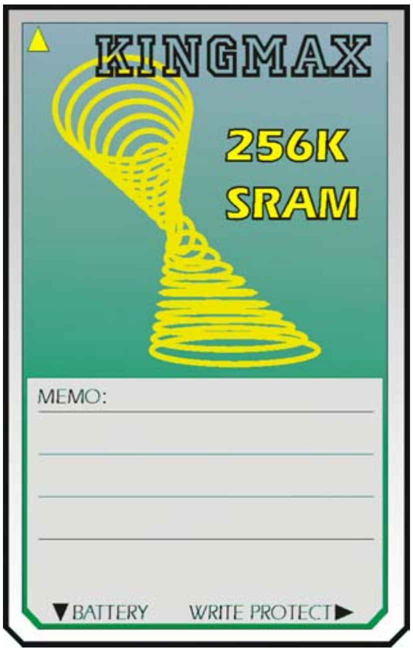
Selection of ALL memories will show

Erase ALL User Memories. . .?
Press [ENTER] to Confirm

Two presses of ENTER will now erase all memories.

Pressing QUIT at any time will return the unit to the default screen and exit the entire recall process.

PCMCIA Cards – What can they be used for?



PC cards, or PCMCIA cards, to give them their full title, may be used to store several types of data for use with *SIDD*. These are:

Memory Set Back-ups
Preset Library Updates
Software Updates
Unit Clone Data

The type of card that SIDD uses is a

PCMCIA Type I Card, 256Kb or greater; SRAM.

XTA can supply you with these cards – the part number is OPT-PC1.

Cards bigger than 256Kb in size may be used, but *SIDD* will only use the first 256K (i.e. the first 25% in the case of a 1Mb card.) In many cases, it is actually cheaper to buy the larger cards, rather than the smaller sizes.

It's worth noting that a card can't store more than one type of data at a time, with the exception of memory sets and clone data. Four complete memory sets and one set of clone data may be stored on a single card. However, software updates/preset files/memory sets may not exist on the same card.

The memory on the card is kept alive by a lithium cell built into the card itself. This has an average life of approximately five years. *SIDD* will warn you if you insert a card with a battery that is low or flat. The battery should be carefully removed and replaced as soon as possible. Most cards only allow for ten minutes or so of unassisted backup, so be sure to have the new battery to hand when removing the old one.

Saving Memories to a PC Card

Saving memories to a PCMCIA card will copy the entire memory set onto the card, erasing whatever was previously stored on the card if it was a different type of data (as described in the previous section). Four (4) memory sets may be saved on the same card – along with a set of clone data if required.

Selecting the memory sub-menu and then

Save Memories > PC Card

will prompt for the insertion of a suitable card into the slot below the encoders, if one has not already been inserted. The card will only go in one way. Make sure the 'Write Protect' switch on the back right hand corner (as seen from the fitted position) is set to 'Off'. *SIDD* will warn you if the card is protected.

The screen will then show

Save Memories > PC Card
1: Memory Set Name

Use encoder 3 to choose the location on the card for the memory set to be saved – this may be over another previously saved set, or a new one. *SIDD* will warn you if you are about to overwrite a previously saved memory set. Press ENTER to confirm the choice of location. The screen will show

Name Memory Set...
1: Memory Set Name

with a cursor appearing under the first character of the name. Use encoder 1 to select the character, and the BACK and NEXT keys to move along the name. When satisfied with the name, press ENTER to complete the save procedure. Note that the name does not have to be edited – the last saved name will be kept if ENTER is pressed immediately after choosing the location. If the location was empty, then a completely blank name will be stored.

NOTE: It is crucial that the 'Write Protect' switch on the card is set back to the 'On' position before the card is removed from the unit. Data corruption and loss of all the information on the card may result if this is not done!

If all four memory locations on the card have been used, *SIDD* will warn you and prompt for you to choose one of the locations to overwrite. If you don't want to overwrite any of them, press QUIT, insert a blank card, and start again.

Loading Memories from a PC Card

Note: Before loading memories from a PC Card, note that the entire memory contents already present in SIDD will be overwritten. This is the case even if the memory set contains fewer memory locations, or only locations currently unused.

Selecting the memory sub-menu and then

    Load Memories < PC Card

will prompt for the insertion of a suitable card into the slot below the encoders, if one has not already been inserted. The card will only go in one way. If the wrong type of card is inserted (such as a clone card or program update card), *SIDD* will warn you with a message.

Assuming the card is of the correct type, the screen will show

Load Memories < PC Card
1: Name of Memory Set

Use encoder 3 to choose the location on the card of the memory set to be loaded. Press ENTER to confirm selection. The screen will show the name of the chosen set, and require a further press of ENTER to confirm the loading of the chosen set.

User Memory Resources

As described in the previous section "Memory Configurations and Presets – How Are They Organised?" on page 32, there are 256 available memory locations in *SIDD*. The dynamic allocation of memory to these locations depends on what is stored in each. Obviously a complete channels worth on data will take up more space than just a limiter modules settings. Sixty (60) complete channel memories can be stored.

With this in mind, selecting the memory sub-menu and then

    User Memory Resources

will display

User Memmory resources
Free Memories = 225 (89%)

If the memories are mainly complete channels worth of data, then the percentage of free memory will be much lower than if only modules at a time have been stored. Half the locations being free will not mean 50% of the memory is free. If you are getting short on memory, either copy the entire memory onto a PC card(see page 39), or erase some memories you no longer need (see page 37).

Configuration Sub-menu

The functions available in the Configuration Sub-menu are as follows.

Expander / Gate Option	Set whether the mode of the expander/gate module on each channel.
Stereo Ganging	Set parameter ganging on each module corresponding one on other channel
Function Linking	Set sidechain linking on each module with corresponding module on other channel
Channel Routing	Enable one channel to be used as an external sidechain input to the other and… Select routing of listen signal, aux delay output and main gain position in signal path
Channel Names	Name each channel from a list or create name from scratch
Function Modes	Select the level of editing complexity available for each module

Global or Individual Settings

Many of the options in this menu initially ask the question

Set Globally?

This is a shortcut that allows all the relevant modules to be configured identically in one operation. This proves useful in cases where a function needs to be cleared on all modules simultaneously. If it is required to set each module separately, select

Set Individually?

Using either encoder 3 or the BACK/NEXT keys. Press ENTER to continue.

Expander / Gate Option

Both channels may have their expander/gate module configured independently to operate as either a noise gate or an expander. The parameters available for each module during editing are dependant on this setting, with gates and expanders being treated as completely separate dynamics processors as far as memories are concerned.

Selecting the configuration sub-menu and then

`Expander / Gate Option`

will show the current mode of the two channels of the unit. This is self-explanatory, and will be one of the four following. Note that the asterisk next to one selection indicates the current format.

```
Mode: A = Gated, B = Gated
Mode: A = Expand, B = Gated
Mode: A = Gated, B = Expand
Mode: A = Expand, B = Expand *
```

Use encoder 3 to choose a different configuration, and press ENTER. The screen will display

```
Format Modified!!
Press [ENTER] to Confirm
```

Pressing ENTER again will, after a short delay, change the format and return to the default screen.

What is the Difference Between Stereo Ganging and Function 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 'Stereo 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. Once any module has been ganged, either EDIT key may be used for editing. Both EDIT LEDs will illuminate. Note that the STEREO LED will only illuminate if **all** the modules are ganged. The LISTEN function always remains independent even if entire channels are linked.

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 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.

Stereo ganging is also indicated on each modules summary information screen – the channel ID changes from 'A' or 'B' to 'S' for stereo.

SIDD's function linking option allows the sidechains of each module to be linked to maintain stereo image integrity. Each module can be linked independently, so that, for example, limiters may be linked but compressors stay independent. Function linkage is indicated 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.

Stereo Ganging

Selecting the configuration sub-menu and then
    Expander / Gate Option

will ask the question mentioned earlier about global or individual setting of the modules. Use encoder 3 or the BACK/NEXT keys to choose global or individual. Press ENTER to proceed.

If the gangs are to be set globally, the screen will show

Stereo Ganging
Set Stereo Ganging: ON*

Choose ON or OFF and press ENTER to confirm choice. The unit will request a further press of ENTER before setting all the stereo gangs, and returning to the default screen.

If the gangs are to be set individually, the screen will show

Stereo Ganging
Input Ganged = NO*

Selecting YES or NO and pressing ENTER will proceed through the Dynamic EQ, Gate/Expander, Compressor and Limiter, finally returning to the default screen.

Function Linking

Selecting the configuration sub-menu and then
Function Linking

will go through each module in turn – select 'Linked' or 'Not Linked' as required. Note that the Dynamic EQ module is slightly different. It offers three choices -

Function Linking
DynamicEQ1: No Linking

Function Linking
DynamicEQ1: Linked High

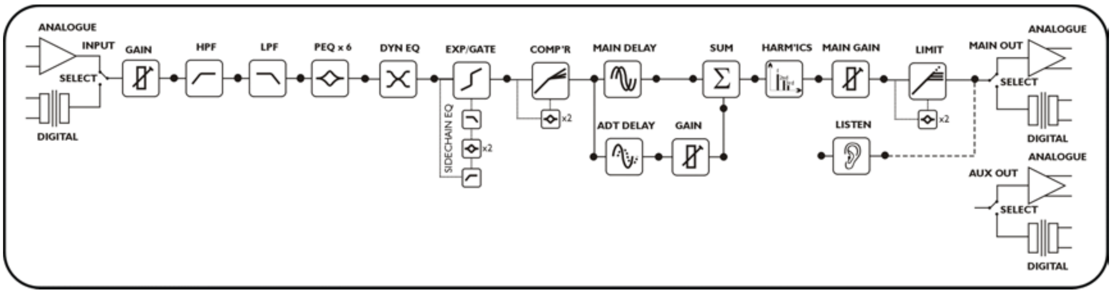
Function Linking
DynamicEQ1: Linked Low

These refer to the way the linkage will operate. If 'Linked High' is chosen, then the channel with the highest sidechain signal over the threshold will determine the amount of dynamic EQ compression or expansion applied. If 'Linked Low' is chosen, the converse will be true – the channel with the lowest sidechain signal over the threshold will determine the amount of dynamic EQ compression or expansion applied.

For all the other modules, the choice is simply 'Linked' or 'No Linking'. The sidechain with the highest signal over the threshold will determine the gain reduction applied to both.

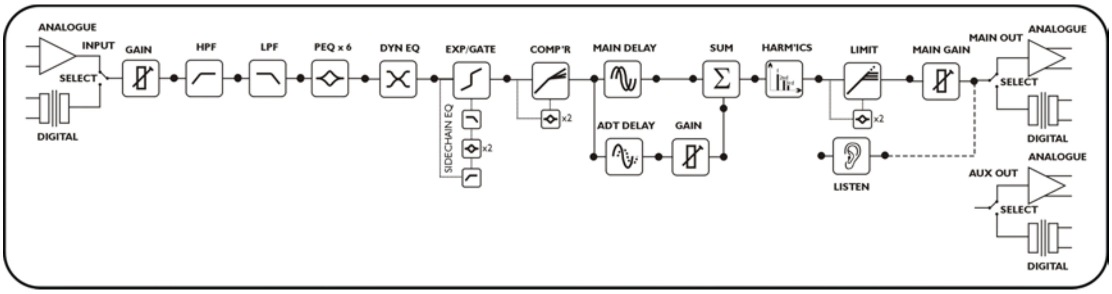
SIDD's Configurations

To facilitate use in as many situations as possible, SIDD may be configured to operate in several different ways. The amount of processing and overall order of the main modules does not change, but the routing of the ADT module, and the 'Listen' signal may be altered.



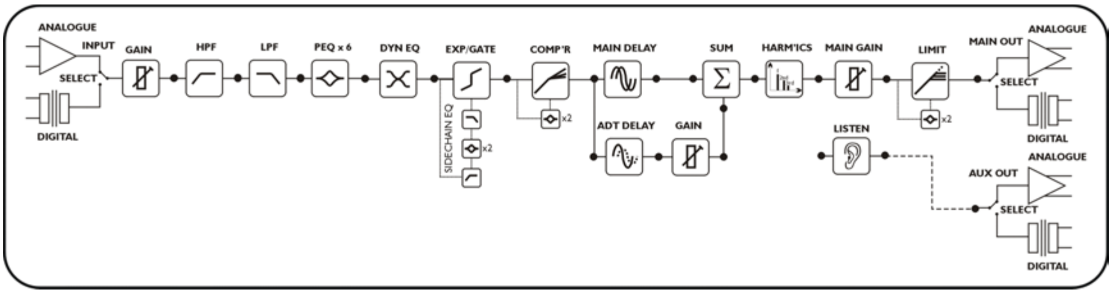
Configuration 1: A.D.T. Format; Gain Pre-Limiter; Listen on Mains

In this configuration, the separate delay line with gain control is set up to operate as an auto double tracking module, which is routed back into the main signal path. This allows for 'thickening' effects to be applied to the main signal after the majority of the dynamics processing. The gain control is pre-limiter so that the limiter can be set to operate as a 'brickwall' device, not allowing any signals above its threshold to pass through. Any gain applied with the output gain control effectively moves the signal nearer the limiter threshold. The 'listen' signal will appear in place of the main signal when 'LISTEN' is pressed during editing of any module.



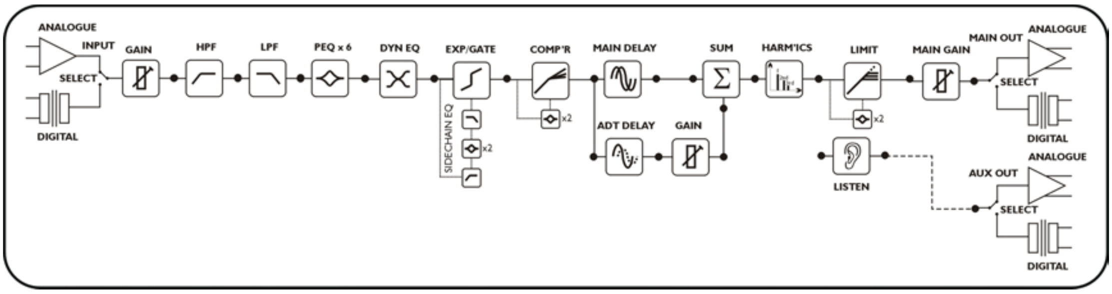
Configuration 11: A.D.T. Format; Gain Post-Limiter; Listen on Mains

In this configuration, the main gain control is just after the limiter, so that adjusting the gain will increase the level at the outputs, but not affect the limiter threshold. The ADT module is the same, and the listen signal still gets switched into the main signal path when 'LISTEN' is pressed.



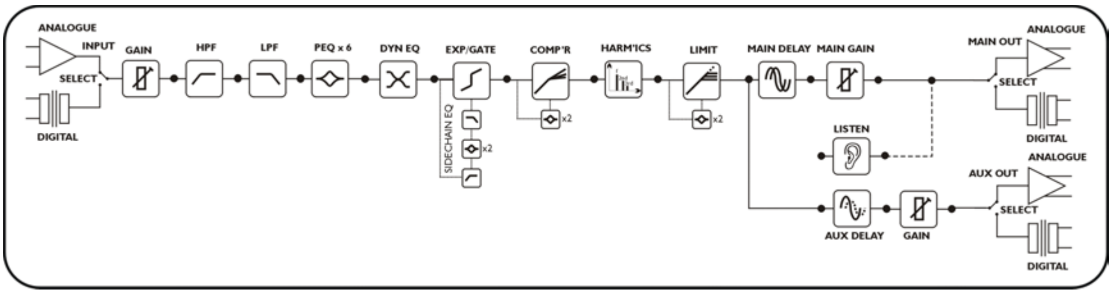
Configuration III: A.D.T. Format; Gain Pre-Limiter; Listen on Auxes

In this configuration, the main gain control is just before the limiter, with the ADT module as before. However, the listen signal now gets switched through to the auxiliary outputs when 'LISTEN' is pressed, so the main signal path is not interrupted. This allows independent monitoring of any sidechain signal during editing.



Configuration IV: A.D.T. Format; Gain Post-Limiter; Listen on Auxes

In this configuration, the main gain control is after before the limiter, with the ADT module as before. The listen signal is again routed to the auxiliary outputs when 'LISTEN' is pressed.



Configuration V: Split Format; Gain Post-Limiter; Listen on Mains

In this configuration, the main delay is moved to the post-limiter position, and the ADT delay, is routed to the auxiliaries. Pressing 'LISTEN' will interrupt the main signal path and switch in the sidechain signal, leaving the auxiliaries untouched.

Configuration Menu cntd. : Channel Routing

The channel routing deals with not only the configuration setup described in the last section, but also the sidechain source for the entire channel.

Selecting the configuration sub-menu and then
Channel Routing

will start by asking for the sidechain source for each channel:

Input Routing
Channel A Source: A+B*

Choosing 'B' for channel A's source effectively makes channel B the external sidechain input for channel A. This might prove useful if, for instance, the gate was being used to trigger a bass guitar main signal, keyed from the bass drum microphone. It is also possible to mix the two channels sidechains together – the 'A+B' option effectively performs the same function as setting all the Function Linking options on.
Press ENTER to confirm selection and proceed.

After choosing the input sources for the sidechains, the output mode must be set. This will either be 'ADT' or 'SPLIT'. The screen will show

Output Mode
Output Mode = ADT*

or

Output Mode = SPLIT

Choosing SPLIT and pressing ENTER will conclude the routing options and *SIDD* will ask for one final press of ENTER before updating the configuration. The outputs will mute for a couple of seconds whilst this happens.

Choosing ADT will then proceed to the choice of limiter position.

Output Mode
Limiter Posn = Post Gain

or

Limiter Posn = Pre Gain*

Finally, the choice of routing the listen signal via the auxiliaries or the main outputs is offered.

Output Mode
Listen Outputs = Main

or

Listen Outputs = Auxes*

One final press of ENTER will update the complete configuration. Again, the outputs will mute for a couple of seconds whilst this happens.

Remember - pressing QUIT at any time will return the unit to the default screen and exit the entire configuration process: no changes take effect until all the settings have been selected.

Channel Names

The channel name appears on the default screen after the unit has finished waking up, as well as being displayed as part of each module's summary information screen.

Selecting the configuration sub-menu and then
Channel Names

will ask which channel to name. Choose the channel and press ENTER. *SIDD* contains a list of predefined names covering a large number of applications. These are designed to save the time and effort of typing one in by hand. If required, a name can be created from scratch. Once used, it will subsequently appear as part of the list of predefined names, allowing it to be reused again at a later stage.

Having selected the channel, the screen will show

Channel Names
Select Predefined List*

Encoder 3 can be used to change this to

Channel Names
Select User Edit*

Dealing with the predefined list first, selecting it and pressing ENTER will then display

Channel Names
Select 51:BACKLINE L :

With encoder 3 being used to scroll through the list. Press ENTER to confirm selection of the new name.

If the user edit option was chosen, a cursor appears under the first character of the name (whatever that currently is set to).

Channel Names
Edit Name: _ACKLINE R :

The BACK and NEXT keys move along the name and encoder 3 chooses the character. Press ENTER when finished editing, and to accept the new name.

Function Modes

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 current function mode for any module is shown during editing, on the module summary information screen.

Selecting the configuration sub-menu and then
Function Modes

will ask the question mentioned earlier about global or individual setting of the modules. Choose global or individual and press ENTER to proceed.

If the gangs are to be set globally, the screen will show

Function Modes
All Functions = Expert*

Choose level of editing complexity and press ENTER to confirm choice.

If the gangs are to be set individually, the screen will show

Function Modes
[A]Input EQ = Expert*

Choose level of editing complexity and press ENTER to proceed through the Dynamic EQ, Gate/Expander, Compressor and Limiter.

Note that parameters set up and then hidden by entering a simpler editing mode remain in effect – they are just inaccessible.

Security Sub-menu

Please read the entire section about locking/unlocking the unit before entering the security menu, as an inadvertent lockout may require a call to XTA, and we're not here 24 hours a day!

The security system built into *SIDD* is designed to prevent any unwanted tampering with the front panel, via the use of a four character code.

Locking the Unit

Selecting the security sub-menu and then

Security Sub-Menu

will offer the single option to lock the unit:

Unit Security Menu:
Lock Unit

Pressing ENTER will then ask for the entry of a lock code. The module editing keys come into play at this point, representing the set of possible letters/numbers that may be used. The diagram below shows how the keys map to their respective characters.

It is simply a case of typing in the required code using the keys, and then confirming the code by typing it in again. If the confirmation if wrong, the unit will not lock. Assuming the code has been confirmed, all editing will have been disabled. It is still possible to view all the parameters of each module by pressing the module edit keys, and scrolling BACK and NEXT through the parameters.

Any attempt to adjust a parameter, bypass a function or enable the sidechain listen will display the message
!! Unit is Locked !!

Unlocking the Unit

Pressing the MENU key when the unit is locked will display the message

Enter Code TO...
Unlock Unit [ ]

Use the module editing keys to re-enter the code previously used to lock the unit. The unit will unlock upon the entry of the last code character. There is no limit to the number of times the password may be entered incorrectly.

Forgotten the Code?

If you forget the code used to lock your unit, it is still possible to unlock it by entering the master code. By default, this is 'AABB' on all units as they leave the factory. However, realising that this code may quickly become common knowledge, it is possible to change this master code to a user defined one.

The procedure is as follows:

Turn on the unit with the ENTER key held in.

After running through the start-up procedure, the unit will display

```
Enter Master Key :
   [       ]
```

Enter the code 'AABB' using the module edit keys.

The unit will then display

```
Enter New Master Key :
     [       ]
```

Enter a new four digit code to replace the current master code.

You will be asked to re-enter the code for confirmation – after doing so the message

```
New Master Key is SET
```

will be displayed, and the unit will then start as normal. From now on, the new master code may be used to override the lockout on the unit.

If the security override code has been changed from the default 'AABB', you will have to contact XTA after invoking the following procedure:

Turn on the unit with the MENU key held in.

After running through the start-up procedure, the unit will display

```
Enter Set-Up Code:
   [       ]
```

Enter the code '8713' using the module edit keys.

The unit will then display

```
  Enter XTA Supplied Code:
Break Code = XXXXX [       ]
```

and note the 'XXXXX' encrypted code.

Ring XTA and quote the code – we will give you the complimentary break code.

This code will need to be entered at the above stage to reset the master reset to 'AABB'.

The unit may then be unlocked using 'AABB' as required.

System Sub-menu

The functions available in the Configuration Sub-menu are as follows.

System Status Info.	Display temperature log and service data
LCD Contrast	Adjust viewing angle of the display
LED Brightness	Adjust brightness of metering
Program Update	Install new software via PCMCIA card
Preset Update	Install new presets via PCMCIA card
Wake-up Time	Adjust time after power-up before audio enabled
Filter Q or BW	Change parametric filter width units
Delay in Time/Distance	Change delay units readout
Meter Operational Mode	Disable meters when module bypassed
Unit Cloning	Store/Recall all memory & options via PCMCIA card

System Status Information

Selecting the system sub-menu and then
System Status Info.

Will automatically begin to scroll through a variety of data relating to the operating state of the unit. This is listed in the table below.

Display Shows…	This means…
===System OK===	No following status information is invalid…
==Audiocore Series==	Unit is part of Audiocore series of processors
DP324 *SIDD*	Model number and name
===Software V170==	Current software version
XTA Presets V000	Current preset library version
Numeric Code: 1395	Software version's revision sub-code
Firmware No: 268	Units audio/host processor hardware revision
Stack Probe: 74	Host processor program health
Current Temper. = 23°C	Units internal temperature now
Maximum Temper. = 26°C	Units internal temperature since powered up today
Max Ever Temper. = 45°C	Maximum temperature ever attained

LCD Contrast

Selecting the system sub-menu and then
LCD Contrast

will allow adjustment of the viewing angke of the display, from its default setting of 0 through to 100. Use encoder 3 to adjust.

LED Brightness

Selecting the system sub-menu and then

    LED Brightness

will illuminate all the LEDs and allow adjustment of the brightness, from 0, through its default setting of 8, to 15. Use encoder 3 to adjust.

Program Update

The operating system software may be updated from a PCMCIA card.
SIDD will warn you if
you try to load an older version than is currently running
you try to load from a card that contains the wrong type of data
you try to load from a blank or corrupted card

Selecting the system sub-menu and then

    Program Update

will prompt for the insertion of a PCMCIA card containing the new version of software. After inserting a card, press ENTER. The display will show

```
Loading New Software
   Erasing Flash
```

followed by

```
Loading New Software
 Block nnnn OF 2046
```

The number 'nnnn' will increment until the load is finished. *SIDD* will reboot when finished. During the process, the meters will freeze and there will be no audio. The entire process, from pressing ENTER until the re-boot, should take no longer than one minute.

During the program update do NOT switch off or disturb the power connection to the unit! The software will become irretrievably corrupted if this happens, the unit inoperable, and your memories may be erased!

Contact XTA if this happens for details on how to reboot your unit and avoid losing your memory settings.

If the program card is inserted before MENU is pressed, the menu system is bypassed and jumps to the update function directly.

Preset Update

A new library of presets may be loaded from a PCMCIA card in exactly the same manner as a software update is loaded.

Selecting the system sub-menu and then
    Preset Update

will prompt for the insertion of a PCMCIA card containing the new preset file. After inserting a card, press ENTER. The display will show

Preset Update
Loading Presets

The unit will reboot after the load process has completed.

Wake-up Time

This is the countdown timer that appears after the unit is powered up, prior to the audio fading up. The time is adjustable in ten second increments from zero, to sixty seconds .

Selecting the system sub-menu and then
    Wake-up Time

will display

Wake-up Time
Wake-up Time: 20 Secs *

Use encoder 3 to select the required time and press ENTER to confirm selection.

Filter Q or Bandwidth (BW)

The readout of units for the parametric filter width may be swapped between 'Q' (a unitless value) and bandwidth (in octaves). 'Q' is simply the reciprocal (i.e. 'one over') of the bandwidth.

Selecting the system sub-menu and then
    Filter Q or BW

will display

Filter Q or BW
Filter Mode: BandWdth *

Use encoder 3 to select the required time and press ENTER to confirm selection.

Delay in Time/Distance

The readout units of the main delay and ADT/split delay may be changed from milliseconds to either feet or metres.

Selecting the system sub-menu and then
    Delay in Time/Distance

will display

Delay in Time/Distance
Display in : Time (mS)*

Use encoder 3 to select 'Time (mS)', 'Metres' or 'Feet'. Press ENTER to confirm selection.

Meter Operational Mode

To minimise the possibility of confusion during high pressure live situations, *SIDD*'s Meters can be set to only operate when the relevant module is in circuit. This cuts down on the number of LEDs illuminated and makes it easier to see exactly how much processing is being applied at any given time.

Selecting the system sub-menu and then
    Meter Operational Mode

will display

Meter Operational Mode
Meters OFF in Bypass *

Use encoder 3 to select 'Meters OFF in Bypass' or 'Meters Always Active'. Press ENTER to confirm selection.

Note that the 'OPEN' LEDs associated with the expander/gate will always illuminate when 'BYPASS' is pressed, as the gate/expander is effectively opened. Input and output metering cannot be disabled under any circumstances.

Unit Cloning

To configure one unit identically to another, the quickest way is with the cloning function. This copies all memories, the current audio settings and all menu options, with the exception of the remote interface settings and security settings. Once written to a PCMCIA card, this card may be used to replicate the settings on any number of units making them audibly and operationally identical to the original.

Storing Clone Data:

Selecting the system sub-menu and then

Unit Cloning

will prompt for the insertion of a PCMCIA card. After inserting a card, press ENTER. The display will show

Unit Cloning
Make a Clone Card*

Note that if the card already contains clone data, the display will show

Unit Cloning
Clone Unit From Card*

Use encoder 3 to change to "Make a Clone Card" and press ENTER. Make sure the 'Write Protect' switch on the back right hand corner (as seen from the fitted position) is set to 'Off'. *SIDD* will warn you if the card is protected.

Pressing ENTER will copy the clone data to the card. The unit will then exit to the default screen.

NOTE: It is crucial that the 'Write Protect' switch on the card is set back to the 'On' position before the card is removed from the unit. Data corruption and loss of all the information on the card may result if this is not done!

Recalling Clone Data:

Remember that all the settings in the unit will be overwritten when the clone data is copied!

Selecting the system sub-menu and then

Unit Cloning

will prompt for the insertion of a PCMCIA card. After inserting a card, press ENTER. The display will show

Unit Cloning
Clone Unit From Card*

Press ENTER to accept, and then again to confirm. The settings from the card will be copied and the unit will reboot.

Ext. Interface Sub-Menu

Interface Operation

SIDD has as standard three external interface systems MIDI, RS232 and RS485. This allows complete control via computer (cable or radio) and MIDI 'Program Change' command. Setting up the interface depends on whether the unit is in a single unit system or a multiple unit system.

Single Unit System

Select

Ext. Interface Menu

and press ENTER.

A series of options will now be offered – select as shown below.

Interface Setup
External Mode:Master*

Press ENTER. Now, choose the interface to use (RS232, RS485 or MIDI)

Interface Setup
Master Source: RS232*

Press ENTER. If RS232 or RS485 was chosen, set the baud rate to match your computers COM port setting

Interface Setup
RS232+RS485 Baud:38400*

Press ENTER. Now, choose the ID number that the unit is to use (analogous to a MIDI channel, only there are 32 ID numbers available). It is good practice to choose ID #1 for a single unit system.

Interface Setup
Remote ID Number = 1 *

Multiple Unit System
For multiple unit systems one unit is set as a master, this is the unit connected to the computer. This unit should be set up as for a single unit system. The remaining units should be set as slaves and connected via RS485 XLR connectors in a chain to the master unit.

Set up the slave units as follows.

Select
Ext. Interface Menu

and press ENTER.

A series of options will now be offered – select as shown below.

Interface Setup
External Mode:Slave*

Press ENTER. If RS232 or RS485 was chosen, set the baud rate to match you're the baud rate set on the master unit.

Interface Setup
RS232+RS485 Baud:38400*

Press ENTER. Now, choose the ID number that the unit is to use (analogous to a MIDI channel, only there are 32 ID numbers available). Normally, the master unit is assigned to ID #1. It makes sense to assign the next unit physically connected to the master as ID #2, and so on. This makes it easier to troubleshoot, should any units not respond.

Interface Setup
Remote ID Number = 2 *

For information about how to connect the units together in various situations, see the section entitled "Typical Interface Setups" on page 63.

AES/EBU Sub-Menu

SIDD may be factory fitted with an AES/EBU digital interface that provides digital audio data connections for inputs, the main outputs and the auxiliary outputs.

The AES/EBU digital signal is a stereo data stream, and so only one input XLR and two output XLR's are used for the digital interface.

Digital Connections

Connection of AES/EBU signals is via the existing rear panel XLR connectors.
The connections are as follows:

Input A:	Switchable between digital (stereo) input and analogue input A
Input B:	Analogue input B at all times
Main Out 1:	Switchable between digital output (for main outputs 1 & 2) and main analogue output 1
Main Out 2:	Analogue main output 2 only
Aux Out 1:	Switchable between digital output (for aux outputs 1 & 2) and auxiliary analogue output 1
Aux Out 2:	Analogue auxiliary output 2 only

Routing Options

Selecting the AES/EBU sub-menu and then
`Routing Options`

allows selection of the input source – either analogue or digital, and similarly the output source, again either analogue or digital.

AES Diagnostics

Selecting the AES/EBU sub-menu and then
`AES Diagnostics`

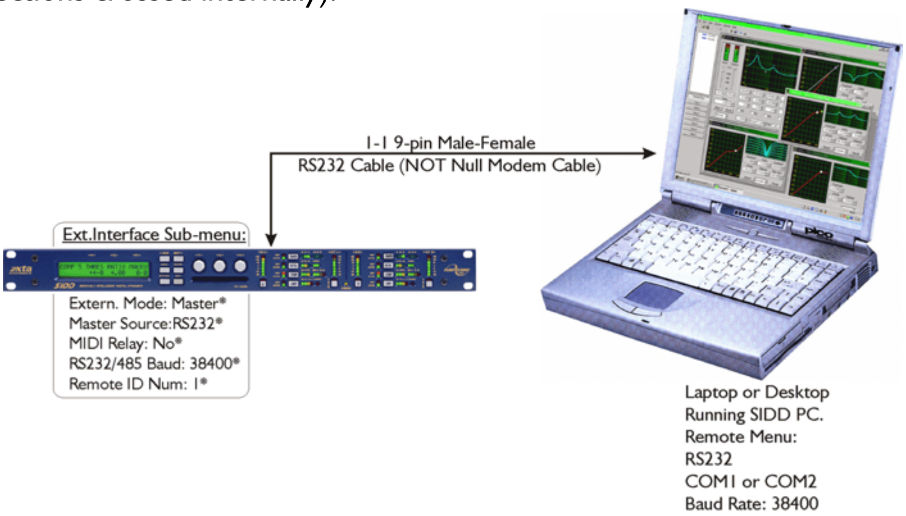
reports validity flag, clock speed and errors present in the incoming AES digital signal.

The AES options are only accessible when the interface has been fitted.

Typical Interface Set-ups

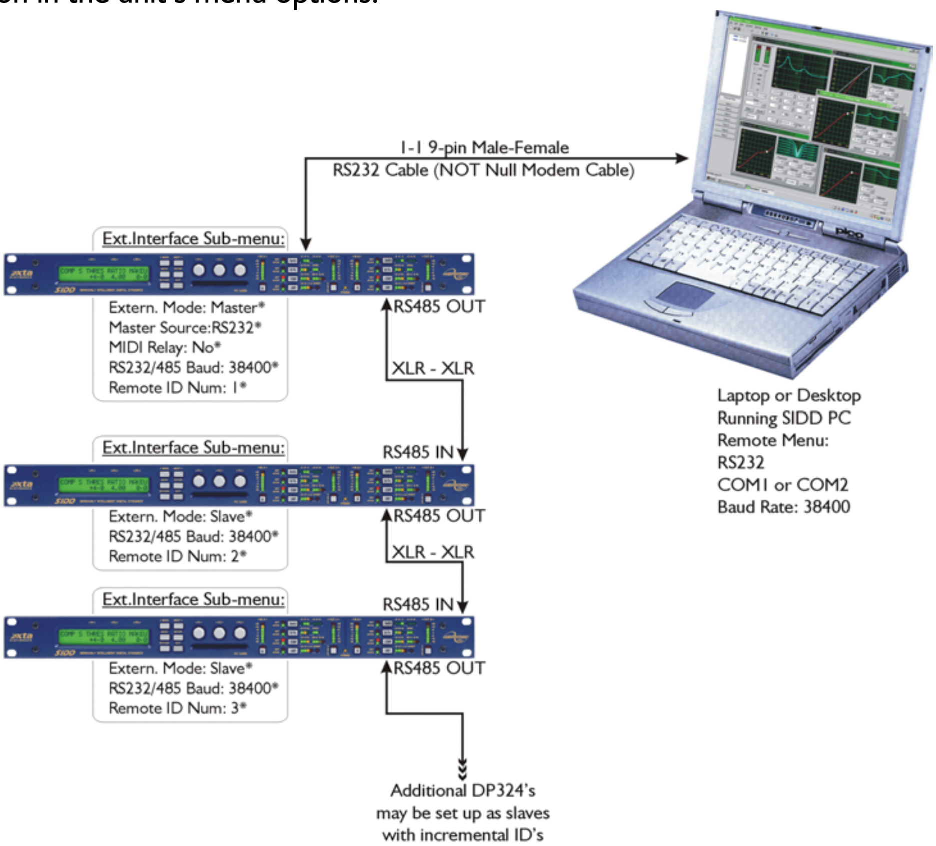
RS232 Connection (Single Unit)

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 1-1 connection type, NOT a null modem cable (which has connections crossed internally).



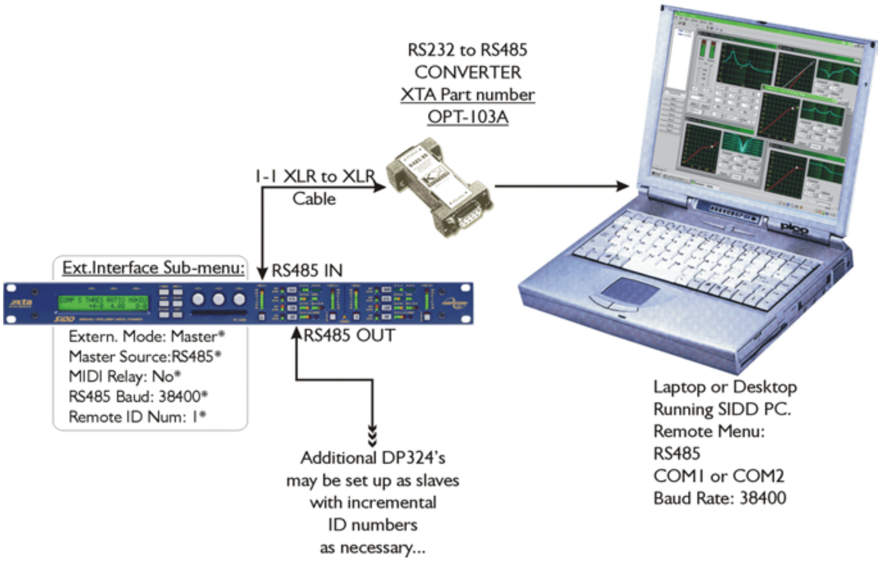
RS232 Connection (Multiple Units)

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.



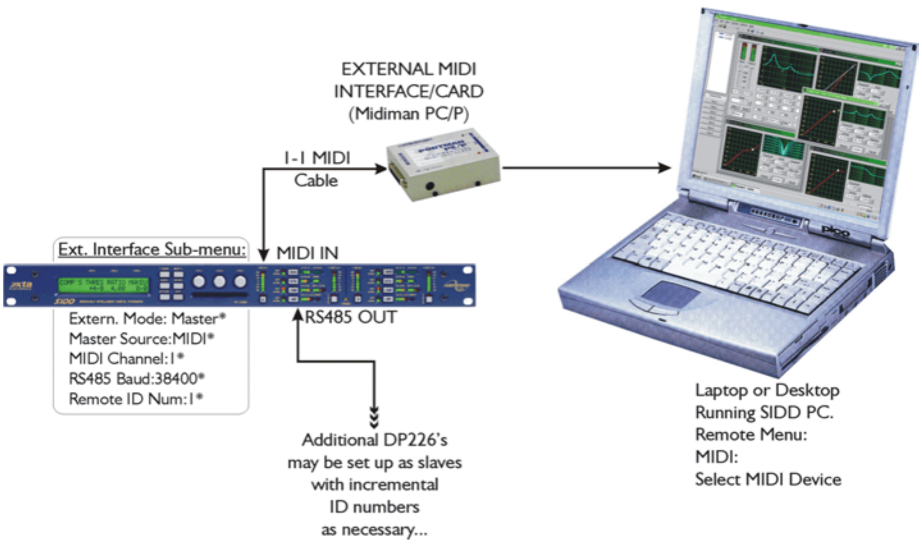
RS485 Connection

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.



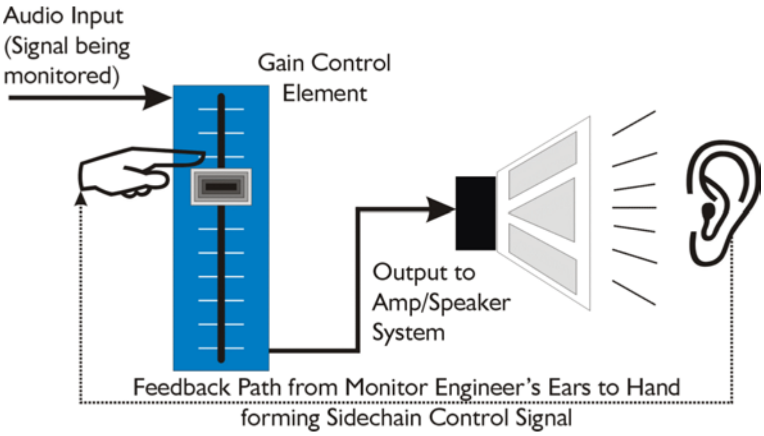
Note that two way communication is NOT possible via the MIDI interface – only download to the unit is possible, and no settings are returned to the connected computer.

Look Ahead Delay – Pre-emptive Action

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

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

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

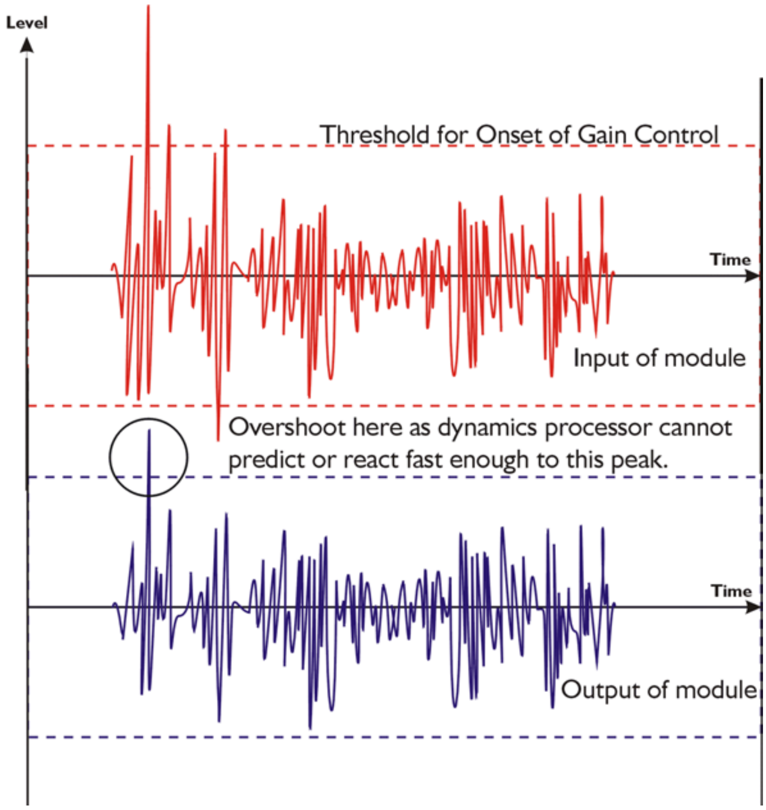


There is a hidden sidechain in operation even in this case. The main signal path is fed through the monitor desk and the gain controlled by adjusting the fader. The sidechain is formed by the feedback path between the engineer's ears checking the level and his brain instructing his hand to turn the fader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

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

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

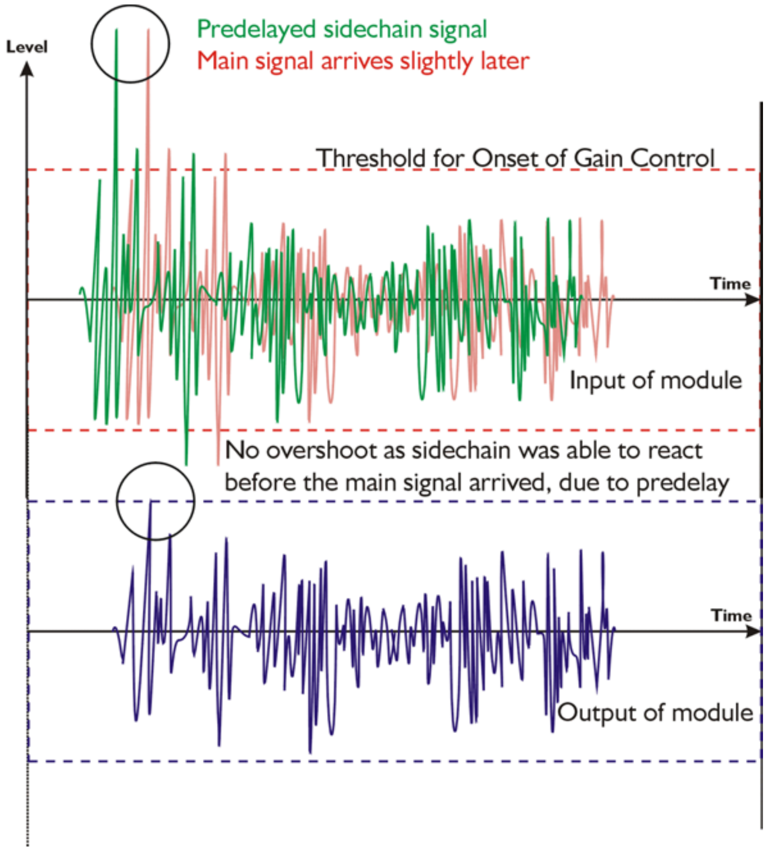
[1] XTA would like to point out that whilst SIDD might sound male, not all engineers are necessarily male. Some might well be female, or at least have long hair.



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

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.

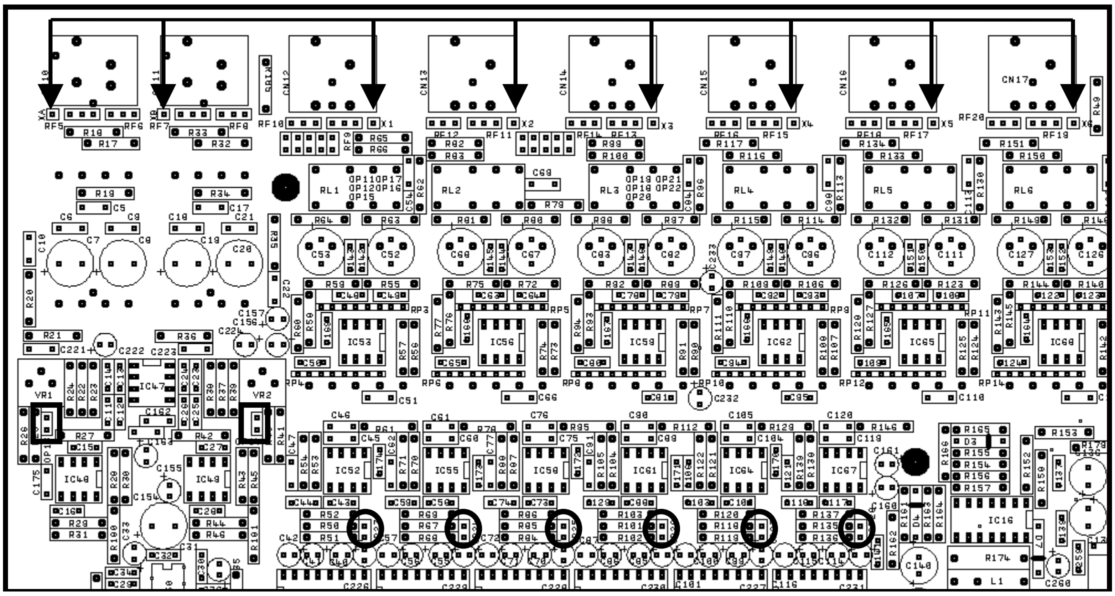
Times when it is useful include:

- 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

Operating Notes

Operating Level
With any audio signal processing equipment it is necessary to ensure adequate signal level is used through the device, to avoid sacrificing noise performance. It is suggested that the operating level chosen should give adequate level to just light the +12dB LED on the headroom meter with maximum program level being used. Since the meter is deliberately set to show clipping 3dB early, this still provides 9dB of headroom before clipping occurs. With equalisation in use it may be necessary to further reduce the input level, as gain within the unit may cause digital clipping, indicated by the top red LED's lighting independently of the rest of the meter.
It should be noted that the figure quoted for the maximum input level options is the clipping point for that option (not a safe operating level). Always ensure that this clipping point is no lower than that for the following equipment in the signal chain, and allow extra margin if equalisation sections are boosted.

Grounding
The Screen (shield) pins on all audio connectors are normally connected directly to the ground pin of the IEC mains inlet. The chassis is also directly connected to this pin. Never operate this unit without the mains safety ground connected. Signal ground (0V) is in turn connected to the chassis ground.
To avoid ground loops, cable shields should be connected to ground at one end only. The normal convention is that the shield is only connected at the output XLR.
Provision is also made for separately isolating each input and output shield pin permanently within the DP324 by breaking the appropriate PCB track, where marked with a box and an arrow next to each XLR connector using a small drill bit or cutter. See the following diagram for details.



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

Time Alignment
A feature of SIDD is the provision of an independently adjustable delay section for each output. Delay time is adjustable in 21μS steps (7mm).

Please see page 13 for details of how to adjust the delay times.

To convert from units of time (i.e. milliseconds) to units of distance use the following formula:
1 millisecond = 343mm (1.126ft) @ 20°C (68°F)

To calculate time delay for a known distance, use:

$$\text{Time delay} = \frac{\text{Distance in meters}}{20.06 \times \sqrt{273 + {}^\circ C}}$$

where °C is the temperature in °C.

To simplify this equation at 20°C.
Delay time in millisecs = (Distance in meters x 2.192) or (Distance in feet x 0.955)

Note: Centigrade = (Fahrenheit –32) x 0.5555.

Specifications

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

Outputs: 4 electronically balanced◆
Source Imp: < 60ohms
Min. Load: 600ohm
Max. Level: +20dBm into 600 ohm

Frequency Resp.:+½dB 20Hz-20kHz
Dyn Range:>110dB 20Hz-20k unwtd
Distortion:< .02%@1kHz,+18dBm
Maximum Delay: 300 mS
Min Step Size: 21µS
Input Gain: ±15dB in 0.1dB steps
Output Gain: -60dB - +15dB in 0.1dB steps, and 'Off'

Parametric Equalisation
6 per Channel
Filter Gain: +15dB to -30dB in 0.1dB steps.
Freq. Range: 20Hz - 20kHz, 1/36 octave steps. (368 positions)
Filter Q / BW: 0.4 to 128 / 2.5 to 0.008
(Sections switched to shelving response)
Low frequency: 20Hz - 1kHz
High frequency: 1kHz - 20kHz
Shelf gains: ±15dB in 0.1dB steps.

High and Lowpass Filters
Filters: 1 of each per output.
Freq. Range HPF: 10Hz - 16kHz
1/36 octave steps.
Freq. Range LPF: 35Hz - 22kHz
1/36 octave steps.
Responses:
Bessel/Butterworth 12db/Octave

Dynamics Processing
Dynamic EQ module x 2
Gate/Expander x 2
Compressor x 2
Limiter x 2

Additional Processing
Harmonics Generator
A.D.T.Module

Display:2x24 character backlit LCD

Input meter: 2 x 7 point, -18dB to +18dB.
Output meter: 2 x 7 point, -18dB to +18dB.

Gain reduction and Threshold metering for each module.

Connectors
Inputs: 3 pin female XLR
Outputs: 3 pin male XLR.
MIDI In: 5 pin DIN
External: 9 pin DEE connector (RS232)
RS485:3 pin male XLR (out) 3 pin male XLR (in)
Power: 3 pin IEC

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

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

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

Warranty

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

Options and Accessories

Part Number	Part Description
ITX-100	DP324 Transformer balanced inputs (factory fitted only)
OTX-100	DP324 Transformer balanced outputs (factory fitted only)
AES-226	AES/EBU Digital inputs/outputs (factory fitted only)
OPT-PC1	'Type I' 256k SRAM PCMCIA card
OPT-103A	RS232 to RS485 converter (from PC to DP324)
OPT-103B	9 pin 'D' type to XLR male adaptor cable
OPT-103C	9 pin 'D' type to 9 pin 'D' type cable. (1m)

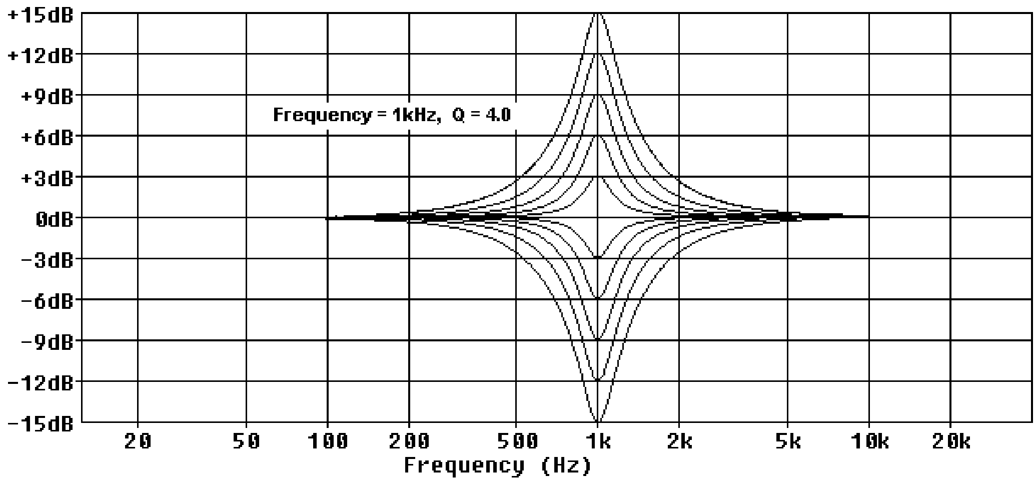
Appendices

Appendix 1: Limiter threshold in dB to Vrms lookup table.

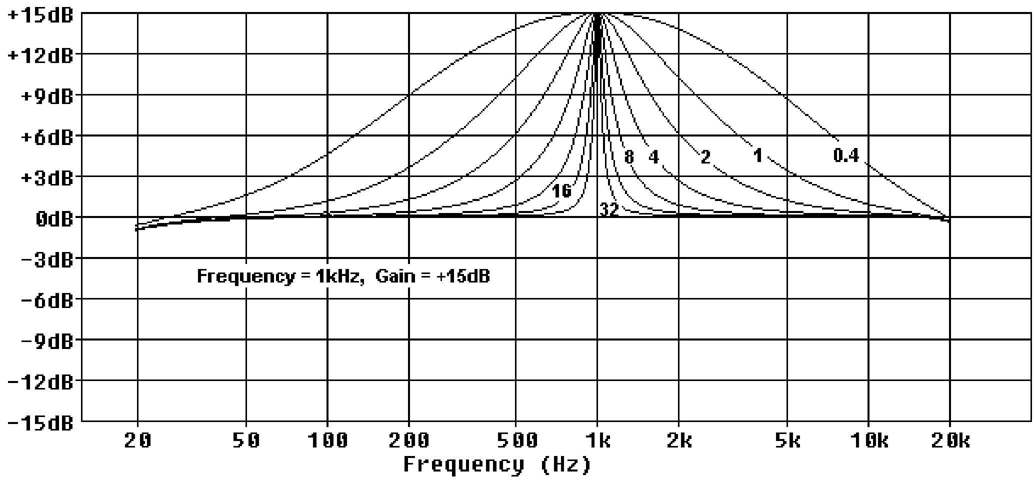
dB	Vrms		dB	Vrms
+22	9.75		+5	1.38
+21	8.69		+4	1.23
+20	7.75		+3	1.09
+19	6.90		+2	0.98
+18	6.15		+1	0.87
+17	5.48		0	0.77
+16	4.89		-1	0.69
+15	4.36		-2	0.62
+14	3.88		-3	0.55
+13	3.46		-4	0.49
+12	3.08		-5	0.44
+11	2.75		-6	0.39
+10	2.45		-7	0.35
+9	2.18		-8	0.31
+8	1.95		-9	0.27
+7	1.73		-10	0.24
+6	1.55			

Calculation:
Vrms = 0.7746 x 10 ^ (dBu + 20)

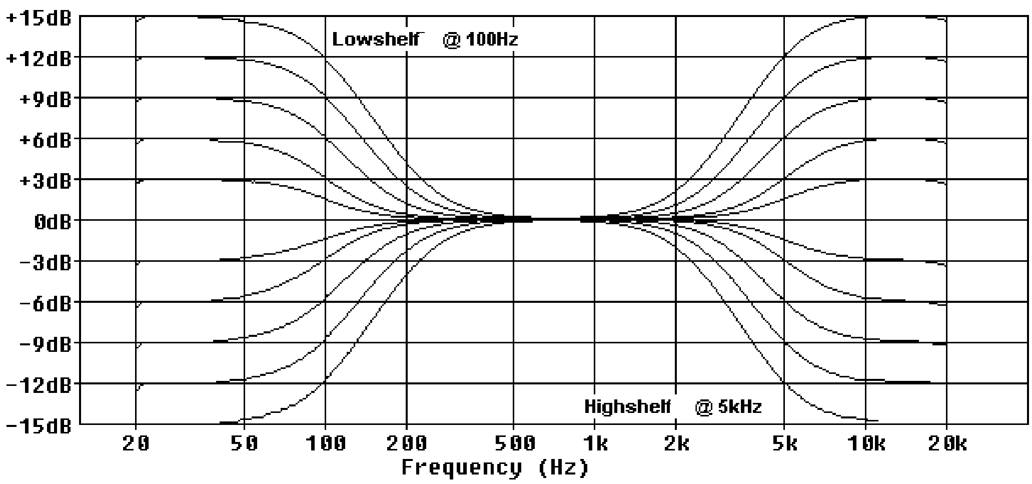
Appendix 2: Equalisation Curves



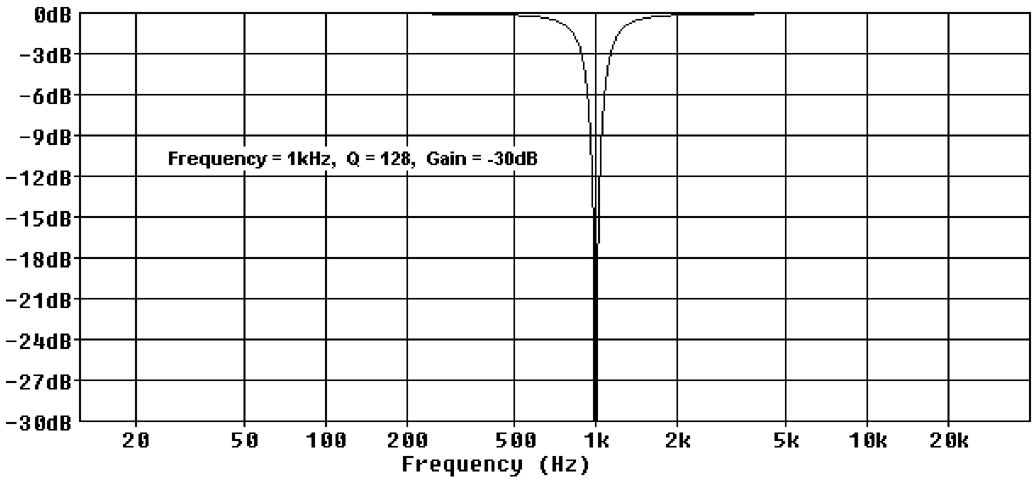
Parametric Filter Gain Curves



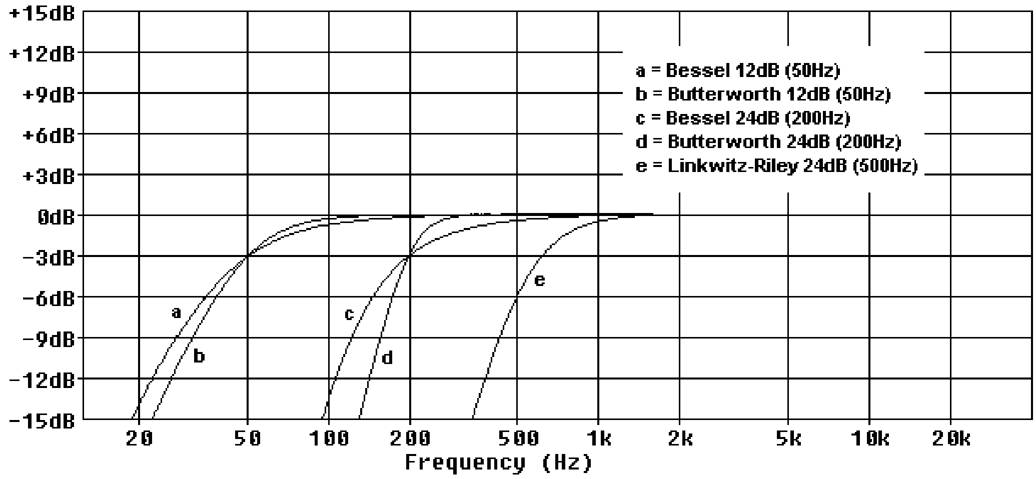
Parametric Filter 'Q' Curves



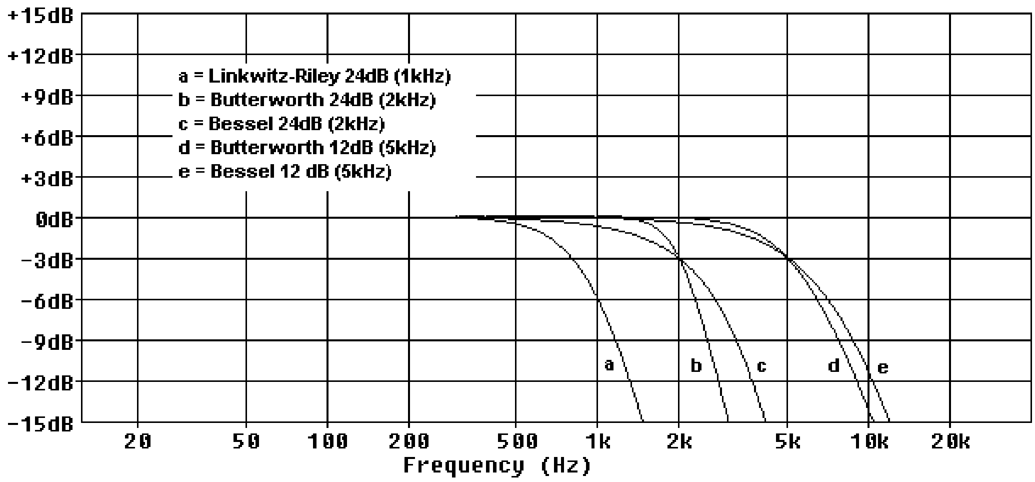
Parametric Filter High and Low Shelving Responses



Parametric Filter High ‘Q’ – Notch Filter Response



High Pass Filter Response Curves



Low Pass Filter Response Curves

Please note that *SIDD* does not currently support 24dB/Octave filters.

Appendix 3: Frequently Asked Questions

How do I update *SIDD*'s software from the net?

The XTA website http:www.xta.co.uk has the latest release of *SIDD* software posted on it along with the latest release of the computer control software. Download the DP324 software, as explained on the software page to a known folder on your 'C' drive. Locate the downloaded files on your computer and view the 'Read me.txt' file which will take you through, step by step, how to upgrade the DP226.
You will need a 9-pin D to 9-pin D, male to female serial lead wired pin 1 to pin1. XTA can supply this cable for you – part number OPT-103C.

What is the best way to connect *SIDD* to a computer?

The best way is via the 'External' RS232 port. This will require a 9-pin D to 9-pin D lead, as would be used for updating the unit with new software. If it is required to connect a chain of units to the computer for remote control of an entire system, use the RS485 XLR sockets to chain the slave units together and to connect them to the master. This configuration, and several others are explained in the section "Typical Interface Setups" on page 63.

What kind of PCMCIA card do I need for use with *SIDD*?

For all applications, whether cloning units, storing sets of memories, storing presets, or updating software, the unit accepts 'Type I' 256K SRAM cards. Note that any card will only accept one type of data – it is not possible to mix input memory sets, clone data or any other combination of the data types above on the same card. XTA can supply this card for you – part number OPT-PC1.

How do I copy settings from one *SIDD* to another?

This may be accomplished using a function built into the unit known as 'Unit Cloning'. See the section entitled "Unit Cloning" on page 58.

Can ordinary microphone cables be used for RS485 connections?

Yes. RS485 is a very robust electrical communication specification, operating point to point over distances up to 1km. As the system only requires a balanced pair of interconnects with a screen, normal male to female XLR leads are fine. In fact, as long as individual pairs are screened, a feed in a multicore may be used.

Can *SIDD* be controlled via Audiocore?

No. *SIDD* has been designed in conjunction with its own proprietary software package, which is designed to permit full two way communications at all times. As Audiocore does not operate in this way (the PC is seen as the master in Audiocore systems), it cannot be adapted to control *SIDD*. However, multiple *SIDD*'s may be connected to the same network (via RS485 In-Out connections). Audiocore will ignore the units, and *SIDD*'s software will ignore any other DP-series units.
Note that the maximum number of connected units must still not exceed 32 in total.

What are the maximum cable lengths for RS232 and MIDI connections?

The specification for RS232 connections details that the cable run should be no greater than 25 metres. However, newer drivers normally allow runs of approaching 100 metres, as long as good quality cable is used throughout. Dropping the baud rate will permit greater distances to be achieved in noisy environments or with less well-shielded cables. The specification for MIDI communications is 15 metres. This is an absolute maximum – XTA recommend taking 10 metres as a better maximum, unless a MIDI 'thru' can be inserted to regenerate the signal.