

Coda Audio
DC426 Digital Loudspeaker Processor
User Guide version 1.0

C O D A
C O D A A U D I O





Coda Audio DC426 Quick Reference

Editing channels: press channel's GAIN key. First press accesses that channel's gain. To scroll through channel's parameters, use the BACK and NEXT keys. Second press accesses last viewed parameter. Third press will drop back to the default screen.

Accessing menus: press the MENU key. Use the BACK and NEXT keys to select the sub-menu required, and enter the sub-menu using the ENTER key. This applies to all levels of menu. ENTER always confirms selections.

The Menus and their Contents

GLOBAL MEMORY Sub-menu: Recall/Store/Erase input and crossover settings, or combinations of.

INPUT SECTION Sub-menu: Set up input ganging/reset.

CROSSOVER Sub-menu: Set up or adjust crossover design, including routing and output ganging.

INTERFACE Sub-menu: Comms interface setup (RS232).

SYSTEM Sub-menu: Used to view unit's status, and select various global options such as PEQ 'Q' or bandwidth units, delay units, and output metering point (pre/post mute).

SECURITY Sub-menu: Used for locking various operations of the unit, using a 4 digit code.

AES/EBU Sub-menu: Monitor AES input status info. (AES inputs are switched via rear panel.)

Notes

The crossover (output) settings may be stored independently of the input settings, using the **Global Memory** sub-menu.

The output meters show level, in dB from the limiter threshold, and the input meters show level from clipping the A-D converters, pre-gain and all EQ.

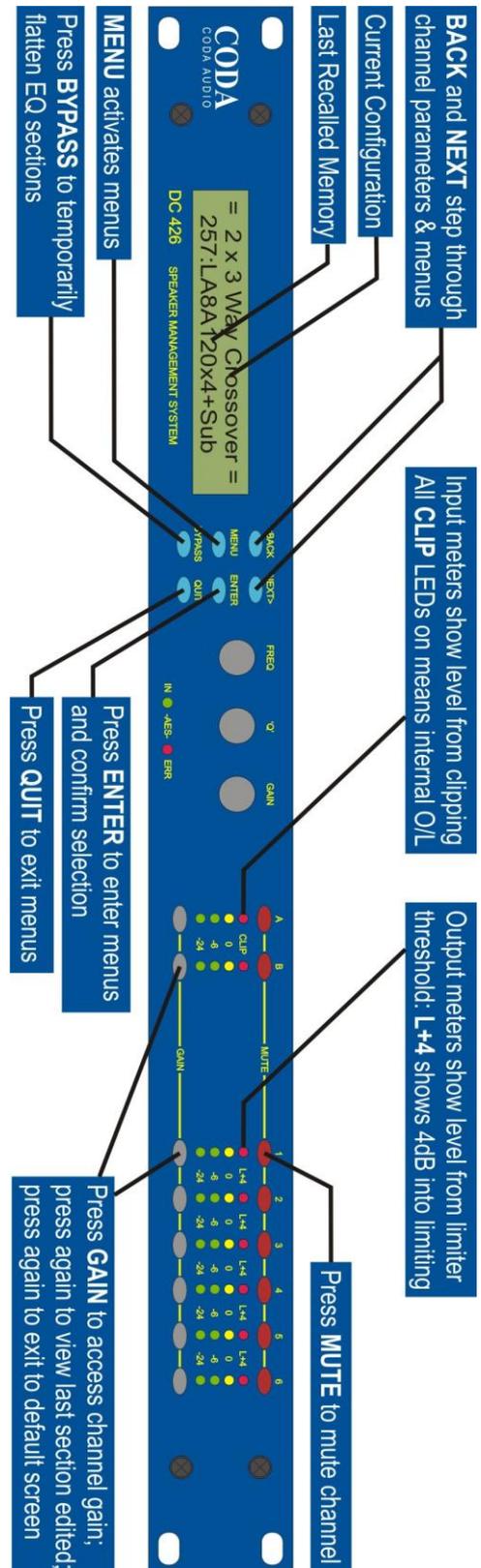
The high and low pass crossover filters are defined independently on each output channel.

To access the limiter attack and release parameters, select "AutoLimiter TimeCst: No" when designing a crossover.

To swap parametric filter units between bandwidth ('BandW') and 'Q', enter **System** Sub-menu, select 'Filter Q/Bandwidth', and select required readout units.

To swap delay time units, enter **System** Sub-menu, select 'Delay Time/Distance', and select required readout units.

Pressing an EDIT key flashes corresponding channels routed to/from that channel.



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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.

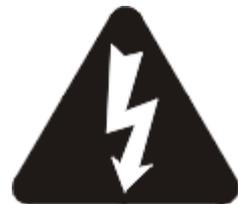
This equipment must be earthed.



**CAUTION
RISK OF ELECTRIC SHOCK
DO NOT OPEN
DO NOT EXPOSE TO RAIN, MOISTURE,
DRIPPING OR SPLASHING**



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



Objects containing liquids, such as vases, must not be placed on this equipment.

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.



Unpacking the unit

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.



Please think of our environment and don't bin any materials, including this manual. When the product has reached the end of its useful life, please dispose of it responsibly through a recycling centre.

Introduction

The DC426 are powerful DSP based audio processors, ideally suited for install applications, where they combine the functions of a multitude of conventional products in a compact 1U unit with extensive remote control capabilities. To achieve this, the unit has two inputs and six outputs which can be configured in a selection of basic crossover modes – 2 x 2 way; 2 x 3 way; and 1 x 6 way. They also offer a “free assign” mode, which allows completely flexible routing of any output from any combination of inputs.

Each input has a gain control, variable delay and a further eight bands of fully parametric equalisation. The parametric filter bands have a large selection of different filter types available.

Each output has a gain control, variable delay, high and low pass crossover filters, nine bands of fully parametric equalisation, polarity switching and, additionally, a fully featured limiter, and a final clip limiter. The crossover filters offer slopes of up to 48 dB/Octave, with a variety of responses available.

Security lock-out is available for all controls.

The DC426 is also equipped with AES/EBU digital inputs and include a sample rate converter, capable of accepting anything from 32 kHz up to 192 kHz.

Features

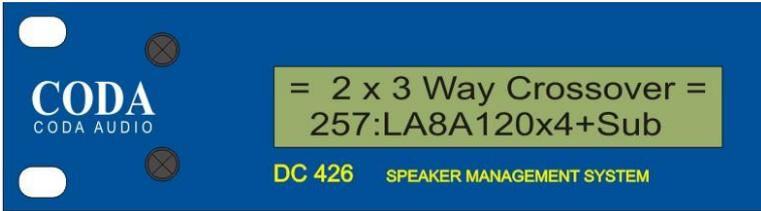
Superb audio quality – carefully optimised double precision signal processing coupled with 24 bit conversion ensure a dynamic range in excess of 117 dB. The high sampling rate of 96 kHz means minimal filtering providing exceptional sonic purity with a bandwidth in excess of 32 kHz.

A flexible input/output multi-mode format caters for any configuration, regardless of scale.

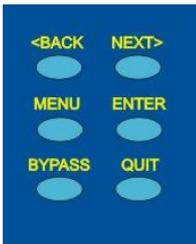
Both routing of inputs to outputs, and ganging (for editing) are completely flexible.

A completely new SHARC™ based DSP platform supplies phenomenal computational power, allowing the unit to provide not only multiple bands of standard parametric equalisation on every input and output, but an additional full spectrum graphic equaliser on each of the two inputs. This additional power also permits both program limiters and no overshoot clip limiters on each output.

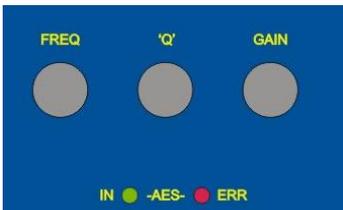
Delay of up to 650 mS may be independently set for each output, with an exceptionally fine minimum increment of 300 nS, which corresponds to a distance change of 0.1 mm! The comprehensive standard specification also includes up to 255 memories. TA's new proprietary Complex Unified Nyquist Truncation algorithm ensures that no matter how much equalisation is applied to a channel, there will always be sufficient headroom available to cater for it.



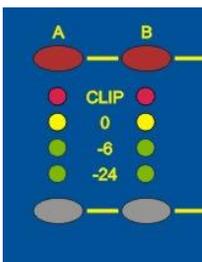
LCD Screen: Shows, by default, the name of the last recalled memory on the bottom line of the screen, and the current routing on the top line.



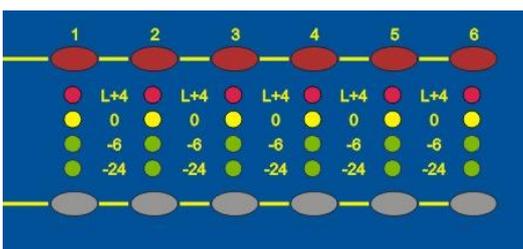
Control Keys: Selection and adjustment of parameters.
 NEXT key moves forward through list of parameters.
 BACK key moves backwards through list of parameters.
 MENU key activates the main menu – a second press selects the last menu edited – a third press selects the last menu item. In this way, three presses on MENU from the default screen will jump back to the last parameter adjusted. Selection of different menus is accomplished using the BACK and NEXT keys, or with the FREQ encoder.
 ENTER key enters the chosen menu, confirms selections, and changes filter types when editing parametric sections.
 BYPASS will flatten the currently selected parametric sections. Note that, for safety reasons, it is not possible to bypass the high and low pass filter sections.
 QUIT exits menus back to the default screen.



Rotary Encoders: Three velocity sensitive encoders adjust the relevant parameters as displayed on the screen.
Status LEDs: The four status LEDs show, from left to right, AES inputs selected (flashing if not locked); ERR flashing if the unit get no digital input.

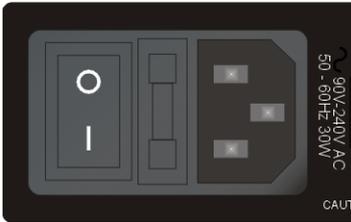


Input Sections: Control and monitor input signal paths.
 Red MUTE buttons illuminate when pressed and mute audio for that channel.
 EDIT buttons illuminate yellow when pressed, and access gain on first press, then last viewed parameter on second press, then exit on third press.
 Input meters show dB from clipping point of the analogue to digital converters. Yellow (0dB) LED illuminates 3 dB from clipping. Red CLIP LED may illuminate independently from the rest of the meter to show digital overflow.



Output Sections: Control and monitor output signal paths.
 Red MUTE buttons illuminate when pressed and mute audio for that channel.
 EDIT buttons illuminate yellow when pressed, and access gain on first press, then last viewed parameter on second press, then exit on third press. Output meters show 0 dB from limiting. The yellow LED illuminates at the onset of limiting. The red LED illuminates at 4 dB into limiting (i.e. 4 dB of gain reduction).

Rear Panel Connections



Power Switch: turns the unit's 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.



RS232: 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.



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



AES Input Switch: Recessed switch to select AES digital inputs. Red LED will illuminate in the hole when AES inputs are selected, along with the corresponding front panel indicator.

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

Note the legending on the panel to designate which inputs are used for AES when the digital inputs are enabled.

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



Operating the DC426

Start-up procedure

Switching on the unit will display a brief message detailing the unit type and software version running

== Coda Audio DC426 ==
== Software V 1.26 ==

and all LEDs will briefly illuminate. The unit will then begin its countdown to the wake-up procedure, during which time the audio will fade up to the level last set. Metering will begin to operate when the fade-up starts.

Preliminary Set-up

The procedure below should be followed when first installing a DC426 unit.

- ✓ Design your crossover! To do this, press MENU, and use the BACK or NEXT key to select '**Crossover sub-menu**' and then press ENTER. Use the BACK or NEXT key to select '**Design a crossover**' and then press ENTER. Finally, use the BACK or NEXT key to select the desired routing and follow the set-up wizard to finalise your design.
- ✓ Note that when in a menu, ENTER is always used to confirm selections. The current selection is marked with an asterisk '*'.
- ✓ Use the EDIT keys on each output channel with the BACK and NEXT keys to select the high pass filters, low pass filters, parametrics etc. Note that when designing a new crossover, the high and low pass filters will be set to default values.
- ✓ Use the EDIT keys on each input channel with the BACK and NEXT keys to select the gain, delay and parametrics available on each input.

Note that if no action is taken in menu mode, the unit will return to normal 'default' mode after about twenty (20) seconds. Repeat the above directions to return to menu mode.

Routing Options and Processing Blocks

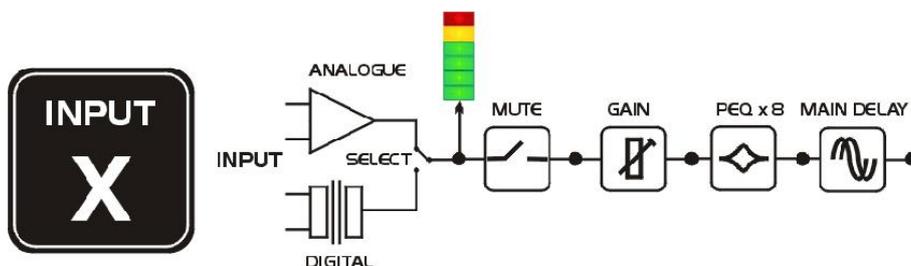
Due to the completely new DSP platform, the routing possibilities within the DC426 have been made completely flexible, with a matrix available allowing any combination of inputs to be routed to any output. The additional DSP power has permitted the inclusion of more processing blocks, even considering the extra inputs and outputs, and the doubling of sample rate.

To reduce set-up time and aid usability, several standard configurations are available as described in a later section.

This section will outline the processing blocks available in relation to the signal path, and explain the various options for routing, including the “Free Assign” mode, which opens up completely flexible channel routing.

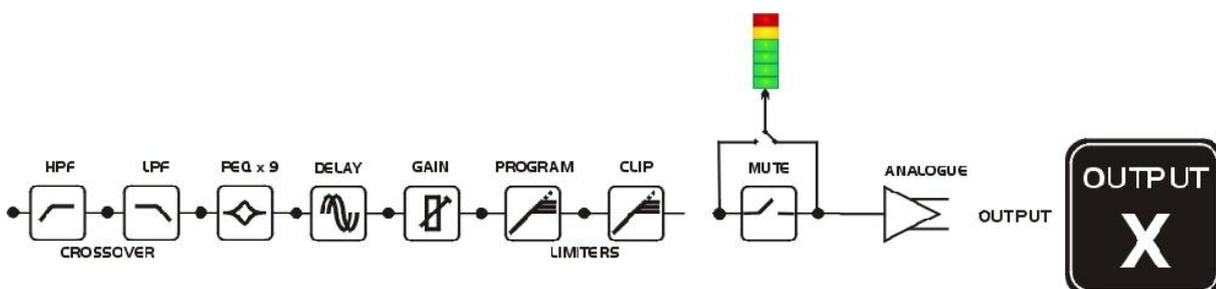
Input Channel Makeup

The diagram below shows the processing available on each of the four input channels, before routing to the matrix.



Output Channel Makeup

The diagram below shows the processing available on each of the eight output channels, after routing from the matrix.



Preset Routing Configurations

In addition to the ability to assign any combination of inputs to any output, a number of preset configurations are provided, for use when designing a crossover from scratch. These have the advantage of suggested settings for the high and low pass filters to useful basic starting points, to filter the different outputs as appropriate for the chosen configuration. These may, of course, be freely modified afterwards should they not suit the requirements exactly.

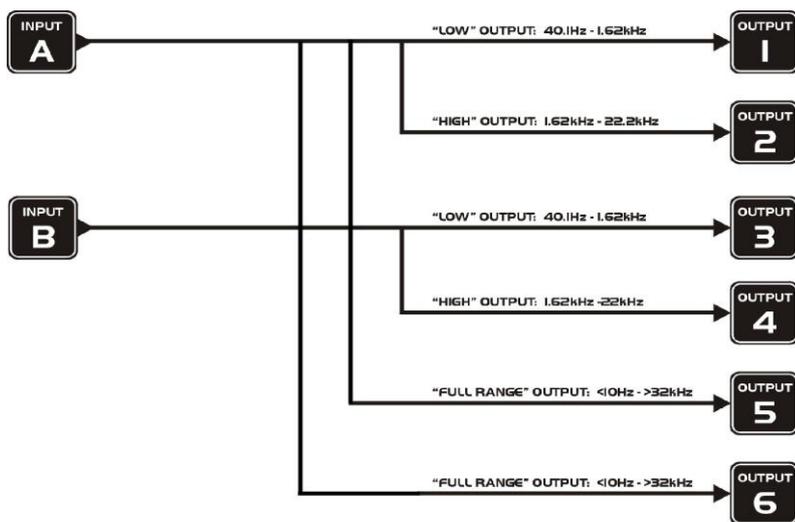
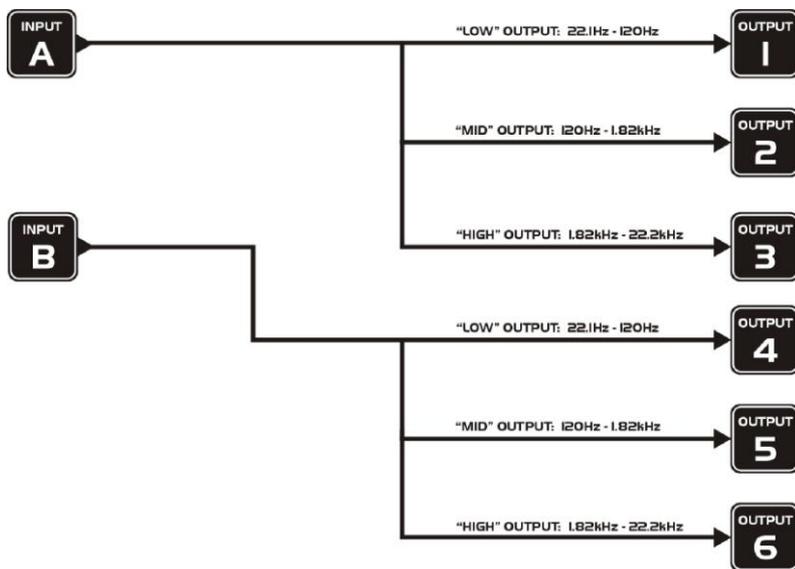


The diagrams on the following pages show the connections made between inputs and outputs, and the suggested values chosen for the high and low pass filters.

2 x 3 way crossover: Inputs A and B feed three outputs each. Default suggested crossover frequencies are shown by each output.

2 x 2 way + Aux. crossover: Inputs A and B feed two outputs each. Default suggested crossover frequencies are shown by each output. Note the 'Aux' outputs are set to full range (Sum of in A+B to Output 5 and 6).

1 x 6 way crossover: Input A fed to all six outputs, with initial settings being all full bandwidth. The crossover points can be adjusted and desired.



Free Assign Routing

If none of the preset configurations are appropriate to the required system setup, it is possible to manually select the routing of the crossover. This is achieved through the **Crossover Menu -> Design A Crossover**.

Pressing ENTER will start the crossover design wizard, with the first option being to choose the routing.

The display will show

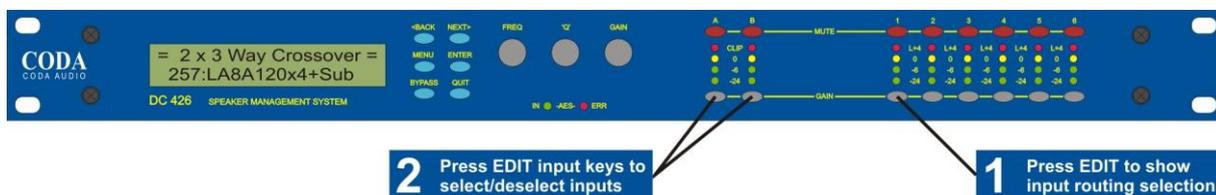
Design A Crossover ->
Routing = 2 X 3 WAY *

or whatever the current configuration is set to. Press BACK until the display shows

Design A Crossover ->
Routing = Free Assign

And then press ENTER. The EDIT key will illuminate for output 1, as will any relevant input EDIT keys, showing which inputs are feeding output 1. The display will also detail the current combination of inputs feeding this output. To change the routing for any output, press its EDIT key, and then choose the required input channel combination by just pressing the input EDIT keys as appropriate. The input combinations can also be stepped through in turn by pressing NEXT, or BACK.

To complete the procedure, press ENTER. The wizard will continue, and if the routing has been changed, all outputs will be muted on exit.



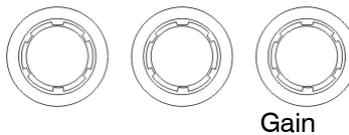


Editing Audio Parameters – Input Channels

Input Gain

The range of the control over the input gain is –40 dB to +6 dB in 0.1 dB steps.

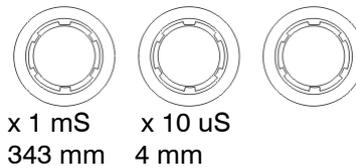
InA Input A Gain
Input Gain = +6.0 dB



Base Delay

The maximum available delay between any input and output is 650.00 mS. For example, if the input delay on channel A is set to 500mS, the maximum available output delay for any output fed from input A will be 150mS. The readout units can be changed between time in milliseconds, distance in feet or distance in metres.

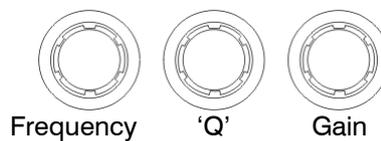
InA Input A Delay
Base Delay = 0.00 mS



Input Parametric EQ

There are eight bands of parameter equalisation available on every input. The behaviour of each individual band can be changed to a variety of different filter shapes, including high and low shelves, notch, and bandpass. Changing the filter type is achieved by pressing **ENTER** during editing any particular band.

InA Input A PEQ:1<>
1k00Hz Q=3.0 0.0 dB

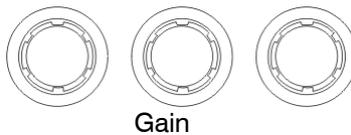


Editing Audio Parameters – Output Channels

Output Gain

The range of the control over the input gain is –40 dB to +15 dB in 0.1 dB steps.

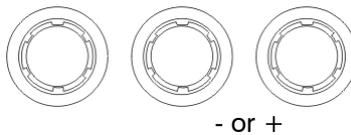
OP1 Output 1 Gain
Output Gain = +6.0 dB



Output Polarity

The polarity (or phase) of each output may be switched individually as below.

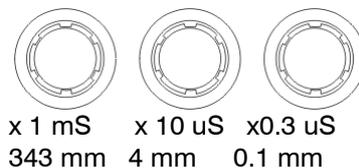
OP1 Output 1 Polar.
Polarity = [+]



Output Delay

The maximum available delay between any input and output is 650.00 mS. For example, if the input delay on channel A is set to 500mS, the maximum available output delay for any output fed from input A will be 150 mS. The readout units can be changed between time in milliseconds, distance in feet or distance in metres.

OP1 Output 1 Delay
Delay = 0.000mS

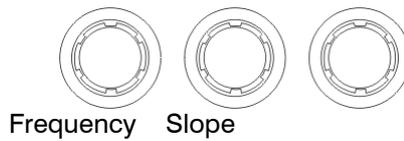




Output High Pass Filter

The high pass crossover filter on each output has a frequency range of <10 Hz up to 32 kHz in 1/36th Octave steps. If you try to set the high pass filter to a higher frequency than the low pass (which would be pointless and result in no output), the message **High/Low Freq. Overlap!** will be displayed. Note that to access the 48 dB/Octave filters, parametric bands 6 & 7 need to be bypassed, or set to 0 dB. If they are not, the message **Bypass PEQ's 6 & 7 To Access 48 dB Slopes** will be displayed.

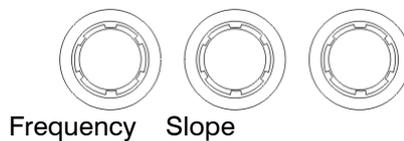
**OP1 Output 1 HPF/~~
<10 Hz Linkw-Riley 48 dB**



Output Low Pass Filter

The low pass crossover filter on each output has a frequency range of 35.1 Hz up to >32 kHz in 1/36th Octave steps. If you try to set the low pass filter to a lower frequency than the high pass (which would be pointless and result in no output), the message **High/Low Freq. Overlap!** will be displayed. Note that to access the 48 dB/Octave filters, parametric bands 8 & 9 need to be bypassed, or set to 0 dB. If they are not, the message **Bypass PEQ's 8 & 9 To Access 48 dB Slopes** will be displayed.

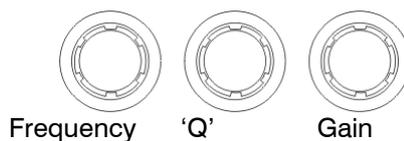
**OP1 Output 1 LPF ~~~\
>32kHz Linkw-Riley 48dB**



Output Parametric EQ

There are nine bands of parametric equalisation available on every output. The behaviour of each individual band can be changed to a variety of different filter shapes, including high and low shelves, notch, and bandpass. Changing the filter type is achieved by pressing **BYPASS** to bypass the filter and then pressing **ENTER** during editing any particular band.

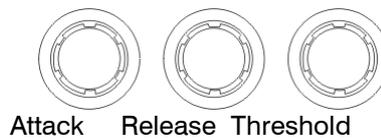
**OP1 Output 1 PEQ:1 <>
1k00Hz Q=3.0 0.0 dB**



Output Limiter

The limiter on each output has adjustable attack and threshold, with a release time that is selectable to be a multiplier of the attack time. For example, as shown below, the attack time is 2 mS and release is “x16” so 32 mS. The attack and release times can be automatically linked to the high pass filter frequency, so that they are set to correct values for the output’s frequency range. If this feature is enabled, the display will show **Automatic T/C** in place of the attack and release times. Selection of automatic time constants is through the **Design a Crossover** wizard, in the **Crossover Sub-Menu**.

OP1 Output 1 Limiter
Atk=2.0 mS Rel=x16 +22 dB

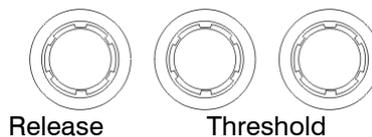


Output “D-Max” (Clip) Limiter

The clip limiter on each output is designed to sit at a threshold just above the standard limiter and has a look ahead attack so that its threshold can never be exceeded. The release time can be automatically linked to the high pass filter frequency, so that it is set to a value appropriate for the output’s frequency range. If this feature is enabled, the display will show **Rel. = Auto** in place of the release time. Selection of automatic time constants is through the

Design a Crossover wizard, in the **Crossover Sub-Menu**.

OP1 Output 1 ClipLim
Rel.=Med. 2 dB Above





Input Ganging and Output Ganging

The method of linking inputs or outputs together during editing is achieved in the same way, so only crossover (output) ganging will be explained here. Having selected **Crossover Ganging** from the menu under the **Crossover Sub-Menu**, the current ganging set-up will be displayed. This will either be a preset selection as would be useful in a standard crossover configuration – for example

<-Crossover Ganging
Ganging=1+3+5 2+4+6

...would be a logical ganging arrangement if the crossover was set up as a 2 x 3 way – linking the control and adjustment of all “Low” outputs together, and that of all “High” outputs together.

However, if the crossover has not been set up with a preset routing configuration, then it may be required to set up the ganging to compliment this configuration. This is achieved using the **Free Assign** mode. This is selected from the preset ganging choices, which are:

Ganging=None	[all outputs independent]
Ganging=Free Assign	[choose ganging]
Ganging=1+2+3+4+5+6	[1 x 6 way]
Ganging=1+2+3 4+5+6	[2 x 3 way]
Ganging=1+2 3+4	[2 x 2 way]

Selecting **Free Assign** and then pressing ENTER will begin the process of ganging outputs together using the following simple rules:

- ✓ All outputs are ganged to the lowest number – so to gang 3 & 5, 5 must be selected and then ganged to 3.
- ✓ Outputs cannot share more than one ganging set – so for example output 3 cannot be ganged to 2 and 4 unless they are ganged together as well. (Effectively 3 and 4 are ganged to 2 in this case)

With these rules in mind, selecting and setting up gangs is quite straightforward.

Press a MUTE key to choose the output to gang – its LED will begin to flash, and an EDIT key will illuminate to show which output it is currently ganged with. To change this selection, just press another EDIT key, remembering that gangs work from the highest to lowest number. So, to gang outputs 1 and 5, press MUTE 5 then EDIT 1 – the display will show

<-Crossover Ganging
Gang Output 5 with 1

Ganging is cleared by selecting **Ganging=None** from the initial choices given above. The **Input Ganging** procedure is identical to the crossover ganging, selectable under the **Input Sub-Menu**.



Menu System Overview

Below is an overview of the menu system – a lot of functions have been assigned menu shortcuts – these are accessible directly from the default screen by pressing MENU followed by the appropriate **MUTE** or **EDIT** button as shown. This table only shows the top level of each menu item – some of these will start wizards or access additional functions, and these will be explained in later sections.

To jump back to the last menu item accessed directly from the default screen, press **MENU** three (3) times – first press is into menus, second is a jump to the sub-menu, third is to the item...

MAIN MENU						
GLOBAL MEMORY	INPUT SECTION	CROSSOVER	INTERFACE	SYSTEM	SECURITY	AES/EBU
	Input Ganging 	Design a Crossover 	External Interface 	System Status 	Unit Locking 	
	Input Reset 	Crossover Ganging 	Wiser 2400 Setup	LCD Contrast		AES Status Info. 
			GPI Interface	LED Brightness		
				Temperature Alarm		
				Wake-up Time		
				Output Meters		
				Filter Q/Bandwidth 		
				Delay Time/Dist'nce 		
				Clip LED Hold Time		
				Set Date & Time		

Note that if no action is taken in menu mode, the unit will return to normal 'default' mode after about twenty (20) seconds. Repeat the above directions to return to menu mode.



Memory Structure

The DC426 has its memories split into sections, allowing independent recall of crossover settings (i.e. all parameters associated with outputs), and input settings.

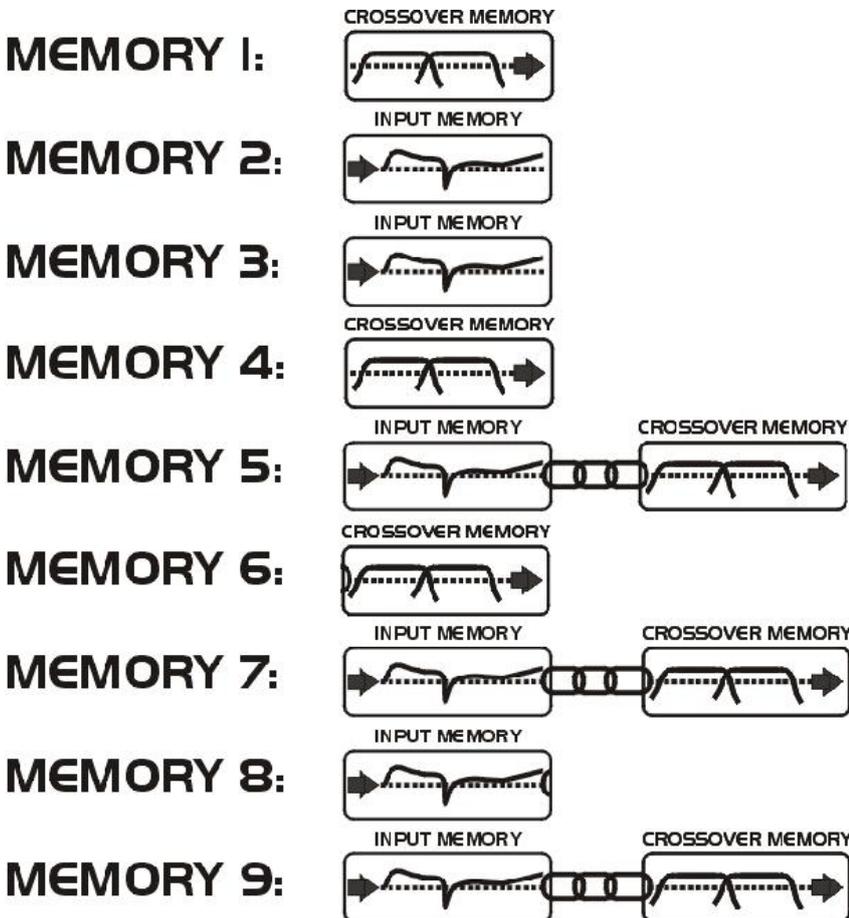
There are, therefore, three types of memory available –**INPUT**, and **Crossover**.

These, and all combinations of memory types, appear in the **GLOBAL MEMORY Sub Menu**, and its operation warrants a little more explanation.

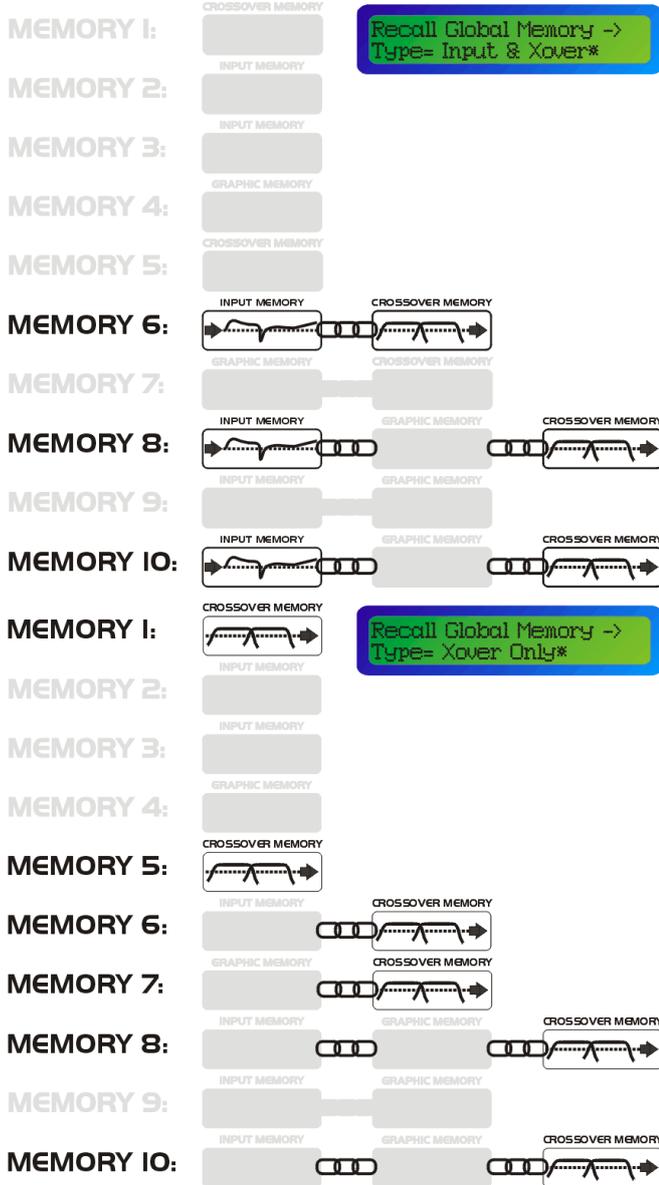
Selecting to **Store** or **Recall** using the **Global Memory** option offers the possibility of storing various combinations of the available memory types, and these are selected using the **BACK** and **NEXT** keys.

To explain how this all works, please consider the following example.

There are 10 memories stored in the unit with various combinations of input and crossover memories.



As can be seen, different memory locations contain different combinations of the three memory types available. If it is required to recall a location that contains Input and Crossover settings, this will limit the selection as shown overleaf...



There will be three memories to choose from in this case, as location 6 is an Input & Crossover memory, whilst 8 and 10 will appear as they contain Input and Crossover information as well.

Selection of Crossover memories only will additionally include locations 1 and 6 in the list of memories available for recall, as shown below.

In this way it is possible to recall part of a memory, as long as it contains the memory type required.

Note that storage and erasure of memories does not follow quite the same rules, being simpler in its operation.

Selecting Input and Crossover during a Store will skip any memories that have other combinations in them.

Selecting Erase for any combination will show only locations that have EXACTLY that combination – it is not possible to erase just one part of a combination memory.

The DC426 has 256 memory locations, but these are dynamic in nature – obviously a memory containing Input and Crossover settings takes up more space than one containing just Input settings. Storage of 56 memories containing Input and Crossover settings is possible.



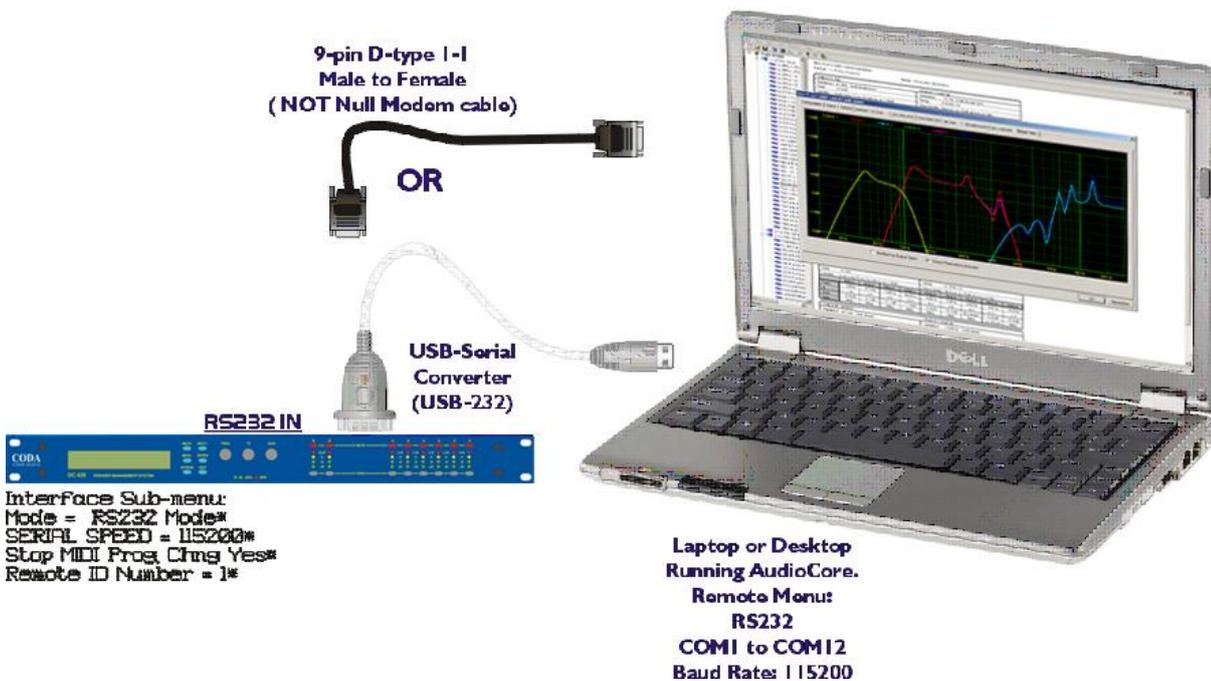
Remote Control Interface Operation

RS232 Interface

This interface is fitted as standard to all units and is accessed via the 9-pin D-type connector on the rear of the unit. Note that to connect to a computer's COM (serial) port correctly, a one-to-one cable must be used, and NOT a 'null modem' cable. A 'null modem' cable has the 'transmit' and 'receive' wires swapped over and will not work.

RS232 Connection (Single Unit)

A typical interface set-up might involve running an RS232 link from laptop or a desktop computer to a **DC426** unit 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).



Loading New Software via a PC

The unit's internal software may be updated via the RS232 port **ONLY**, one unit at a time. The unit's interface must be set as RS232 Master on ID1 for the PC loader program to recognise it and allow the update to be sent.

AES Inputs

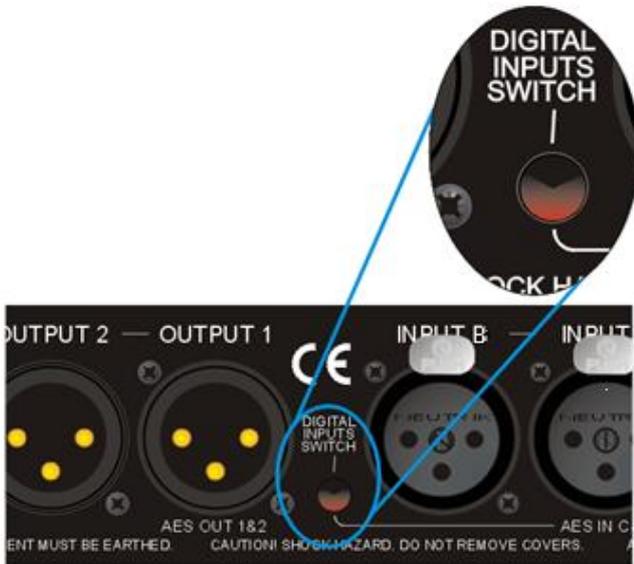
The DC426 has a full AES implementation built in as standard. This allows the unit to receive digital audio directly. The switching of input can be performed independently, and the inclusion of sample rate converters on the inputs allows the unit to accept sample rates from 32 kHz up to 192 kHz.

AES Input

Input selection is via a recessed switch on the rear panel of the unit, between input D and output 1. A red LED inside this aperture illuminates to show that the AES digital inputs have been selected.

A complimentary LED on the front panel below the PCMCIA card also illuminates. The switch controls the rear panel LED directly, whilst the front panel one is via the processor, allowing it to relay a little more information.

If it is flashing, this means that AES inputs have been selected but have not locked. Once a stable AES signal is being received, it will be permanently illuminated.



The AES inputs are marked on the rear panel.



AES Diagnostics and Status Information

Also under the AES/EBU Sub Menu is the AES Status Information option, which can be used to check the incoming sample rate(s) and confirm that the data is being received correctly.

Pressing **ENTER** will first show
AES Device Status
V1: E V2: E

This display shows the correct operation of the AES transmitters V1 and V2. The letter after each is the silicon version (and is of no importance to the user).

Security and Locking

After selecting the **Security Sub Menu** and pressing **ENTER**, select one of the lock types, choosing the most appropriate one for your application. As ever, **ENTER** will confirm your selection.

User Specific

Upon pressing **ENTER** to select this type of lock, each parameter group is presented in turn. Choose the type of lock (as above) using the **FREQ** encoder, and press **ENTER** to confirm each parameter. After the last parameter, the unit requests a password. The description of this operation is given at the end of this section.

This option allows the user to specify, for each type of parameter, whether it is to be completely accessible ('No Lock'), viewable but not adjustable ('Control'), or effectively unavailable ('Display'). The ability to operate mutes, store or recall memories, or even access the menus may also be locked.

Xover Only

All input parameters are available, but only the gain trim (± 6 dB) is available on the outputs, effectively locking all the crossover settings. All mutes remain active.

Xover + Trim

All input parameters available, but no output parameters – the crossover sections are completely locked. All mutes remain active.

Xover + Trim + Mute

As for 'Xover + Trim' but additionally, output mutes are locked. Input mutes remain active.

Changes Only

All parameters may be viewed, but none may be adjusted. This applies to both inputs and outputs. All mutes remain active.

Changes + Views

No parameters are accessible – in effect the **EDIT** keys do nothing. All mutes remain active.

Changes + Mutes

All parameters may be viewed, but none may be adjusted. This applies to both inputs and outputs. All mutes are also locked.

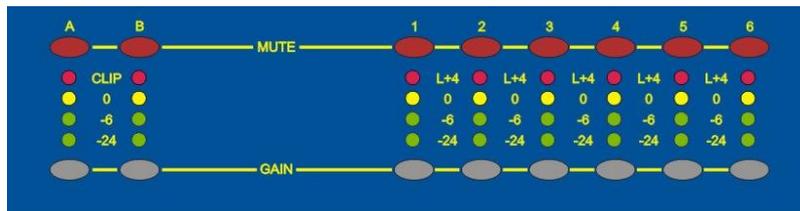
EVERYTHING

No parameters are accessible – in effect the **EDIT** and **MUTE** keys do nothing.

Entering the Password to Complete the Locking Operation

After selection of the lock type from the list above, a four-digit security code will be asked for. This can be entered by using the **FREQ** control to select a character, and the **BACK** and **NEXT** keys to move to the next character.

Alternatively, the **EDIT** keys can be used to enter a code by pressing any combination of the eight buttons. Each **EDIT** key represents its channel labelling, so any combination of **A, B, 1, 2, 3, 4, 5, 6**, can be used as a code, as shown below. Press **ENTER** to accept code and then re-enter it to confirm.



To prevent external computer control being used to adjust locked settings, be sure to set the external interface to OFF before locking out the unit.



Unlocking the Unit

To unlock the unit press **ENTER** and then type the code in. This can be entered by using the **FREQ** control to select a character, and the **BACK** and **NEXT** keys to move to the next character. Alternatively, the **EDIT** keys can be used to enter a code by pressing any combination of the eight buttons. Each **EDIT** key represents its channel labelling, as described in the locking section.

Forgotten the Password?

Don't panic! Your unit can still be unlocked. In an attempt to improve the security system on the DC426, and prevent a standard master password from becoming common knowledge, the units now have a random password key generator.

The procedure for unlocking a unit using the password override is explained below:

Switch the unit on with the **MENU** key held in momentarily. After a few seconds, the unit will ask for a security code. Use the **EDIT** keys in the same manner as for entering lock codes and enter 2121.

The display will show:

**Enter XTA Supplied Code:
Break Code = 12345 [NNNN]**

The Break Code (in the example 12345) should be noted and supplied to Coda Audio. We have software to generate the corresponding Pass Code which should be typed in, followed by **ENTER**. This will unlock the unit and wipe the previous password.

Note the following about this procedure:

Once the Break Code has been noted, do **NOT** press **MENU** again during the operation of the unit (except to get back to this point on power up), or a different code will be generated. The unit may be used as normal, but every press of **MENU** will change the Break Code, so the Pass Code Coda Audio supply will not work!

The unit may be switched on and off as necessary – just be sure **NOT** to press **MENU**, or the entire Break Code procedure will have to be repeated.

Advanced Audio Features

Program Limiter and “D-Max” Limiter

The **DC426** has two levels of dynamic protection on its outputs – a traditional program limiter, and a newly introduced “D-Max” limiter.

Program Limiter

High performance digital limiters are provided for each output with control over attack time, release time and threshold parameters. This level of control allows the user to balance the required subjective quality of the limiter against the driver protection requirements. It does also mean that an incorrectly set limiter may sound awful! In particular, as with all limiters, using too fast an attack or release time will result in excessive low frequency distortion.

In the **Design a Crossover** sub-menu there is an option for automatic limiter time constants. Use this option if you are unsure how to set the time constants manually. Coda Audio recommend the use of the automatic setting.

In this mode the time constants will be automatically set from the High-Pass filter frequency according to the table below.

The time constants are set by the high pass filter frequency for that channel.

High Pass Filter	Auto Attack Time	Release Time
<10Hz – 31Hz	45mS	x16 (720mS)
31Hz – 63Hz	16mS	x16 (256mS)
63Hz – 125Hz	8mS	x16 (128mS)
125Hz – 250Hz	4mS	x16 (64mS)
250Hz – 500Hz	2mS	x16 (32mS)
500Hz - 1kHz	1mS	x16 (16mS)
1kHz – 2kHz	0.5mS	x16 (8mS)
2kHz – 32kHz	0.3mS	x16 (4mS)

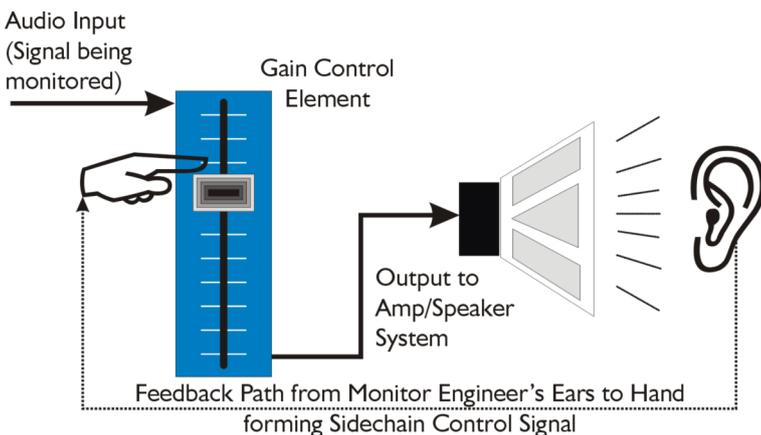


“D-Max” Clip Limiter

The main limitation with traditional dynamics control is the inability of the processing to react truly instantaneously to the signal. 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.

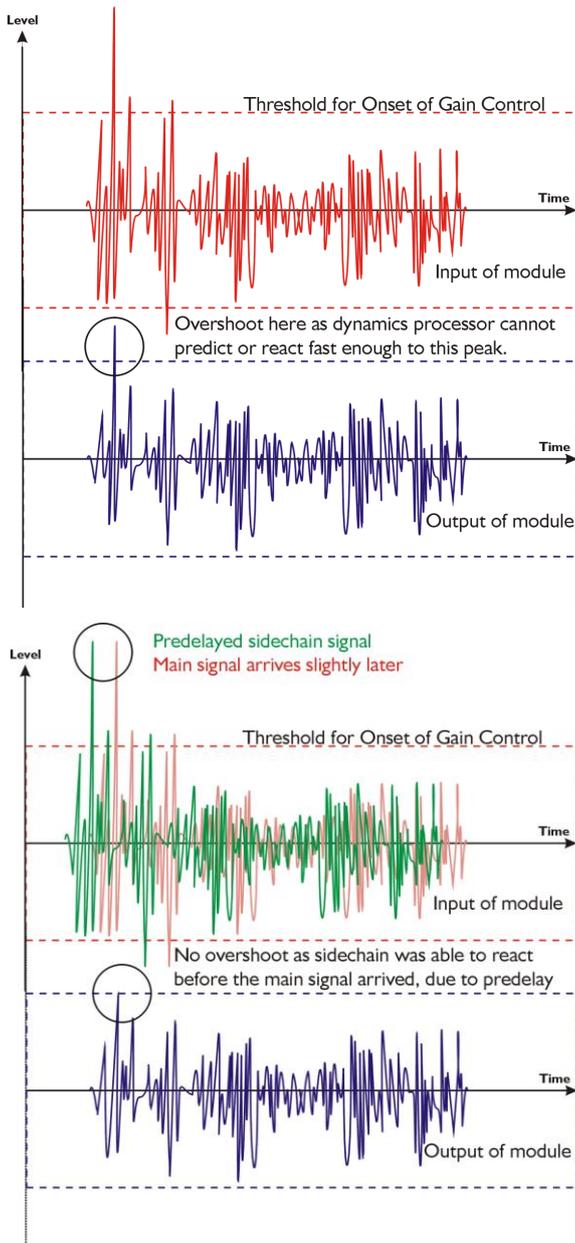
Consider the situation of a monitor engineer listening to a band perform. Having no access to dynamics processors, he has had to resort to manually ‘riding the faders’ in an attempt to keep control of the levels. Should the level of one of the channels on his desk reach an unacceptably high level, he 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.



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

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

The DC426's "D-Max" limiter pre-delays the sidechain signal, resulting in a "zero overshoot" limiter, which is able to catch all peaks and provide a reliable absolute maximum setting for the output of any channel.

The pre-delayed sidechain is shown in green, with the main signal in red. As the main signal arrives slightly after the sidechain, the output from the unit does not suffer from the overshoot problem.

Remember that this delay is only in the order of tens of uS, 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.

The "D-Max" limiter which appears in output lists just following the traditional limiter, has only two parameters to adjust:

OP1 Output 1 ClipLim
Rel.= Medium 10dB Above

The release time (either **Fast**, **Medium**, or **Slow**) and the threshold. Note that the threshold is set to be a minimum of 2dB above the threshold of the program limiter – setting the threshold to **10dB Above**, as in the example, means that no more than 10 dB of overshoot above the threshold of the program limiter will ever be allowed.

The release time may also be set to follow the High Pass filter of the output – this is achieved through the **Design a Crossover** sub-menu, and will result in the display changing to show

OP1 Output 1 ClipLim
Rel.= Auto 10dB Above



Setting Accurate Limiter Thresholds

The limiter built into the **DC426** is intended to be used for loudspeaker driver protection, as opposed to amplifier protection. All modern professional power amplifiers designed for live sound use have their own limiters, which are tailored to protecting the amplifier from clipping.

The following section describes how to set up the units' limiters to provide exceptional protection against driver overheating, and cone over-exursion.

Most speaker systems are given a power rating in Watts RMS. This is the maximum continuous power that the system will handle and often appears very conservative. In reality, as music program is far from continuous in nature, the peak power of the system is much higher – up to ten times the continuous figure. Any limiter, which is to protect the driver from damage, must be able to fulfil the following tasks.

- Have an attack time which is calculated to allow transients through but keep the RMS level below the speaker manufacturers specification;
- Have a release time which is sufficiently long to avoid the limiter itself modulating the program;
- Be intelligent enough to adjust the envelope of the limiter according to the frequency content of the program material.

The program limiters are capable of performing all these tasks. The only parameter that the user must set manually is the threshold, and it is crucial that this is done correctly. Consider the table below.

dB	Ratio	Vrms	Pwr 8 Ω	Pwr 4 Ω	Pwr 2 Ω
45	177.83	137.74	2371.71	4743.42	9486.83
44	158.49	122.77	1883.91	3767.83	7535.66
43	141.25	109.41	1496.45	2992.89	5985.79
42	125.89	97.52	1188.67	2377.34	4754.68
41	112.20	86.91	944.19	1888.39	3776.78
40	100.00	77.46	750.00	1500.00	3000.00
39	89.13	69.04	595.75	1191.49	2382.98
38	79.43	61.53	473.22	946.44	1892.87
37	70.79	54.84	375.89	751.78	1503.56
36	63.10	48.87	298.58	597.16	1194.32
35	56.23	43.56	237.17	474.34	948.68
34	50.12	38.82	188.39	376.78	753.57
33	44.67	34.60	149.64	299.29	598.58
32	39.81	30.84	118.87	237.73	475.47
31	35.48	27.48	94.42	188.84	377.68
30	31.62	24.49	75.00	150.00	300.00



Using this table it is a straightforward procedure to work out the required setting of the limiter thresholds for the system.

- ✓ First, check the RMS power rating of the speaker system, and its impedance.
- ✓ Look up this value in the table above, using the closest value below the rated power of the speaker system. Note the corresponding 'dB' value.
- ✓ Check the gain of your amplifier, which needs to be in 'dB'.
- ✓ Subtract FROM this gain figure that obtained from the table to find the required absolute setting for the limiter thresholds.

Note that, for safety, always set the limiter threshold 1 or 2 dB below the maximum allowable worked out using the above method.

ALWAYS REFER TO YOUR SPEAKER MANUFACTURER FOR LIMITER SETTINGS.

Crossover Filter Slopes

It should also be noted that the turnover frequency displayed on the screen is the -3 dB point for all types except Linkwitz-Riley where the -6 dB point is shown. If the -6 dB point is to be used for the Bessel or Butterworth filter, take the required crossover frequency, multiply this by the appropriate factor from the following table and then select the closest available frequency on the display.

Filter Type	High pass factors	Low pass factors
Bessel 12dB/Oct.	1.45	0.69
Butterworth 12dB/Oct.	1.31	0.76
Bessel 18dB/Oct.	1.37	0.73
Butterworth 18dB/Oct.	1.19	0.84
Bessel 24dB/Oct.	1.35	0.74
Butterworth 24dB/Oct.	1.15	0.87
Bessel 48dB/Oct.	1.39	0.72
Butterworth 48dB/Oct.	1.08	0.93

Please note that unlike conventional analogue crossovers, crossover points and slopes are set with absolute accuracy since component tolerance problems do not occur.



Time Alignment

A further advantage of the **DC426** over conventional products is the provision of an independently adjustable delay section for each output. This allows the true arrival time from multiple drivers to be precisely aligned rather than relying on the compromise 'phase adjust' approach. Delay time is adjustable in 0.3 μ S steps (0.1 mm).

To convert from units of time (i.e. milliseconds) to units of distance use the following formula:

1 millisecond = 343 mm (1.126 ft) @ 20 ° C (68 ° F)

To calculate time delay for a known distance, use:

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

where °C is the temperature in °C.

To simplify this equation at 20 ° C.

Delay time in milliseconds =
(Distance in meters x 2.192) or (Distance in feet x 0.955)

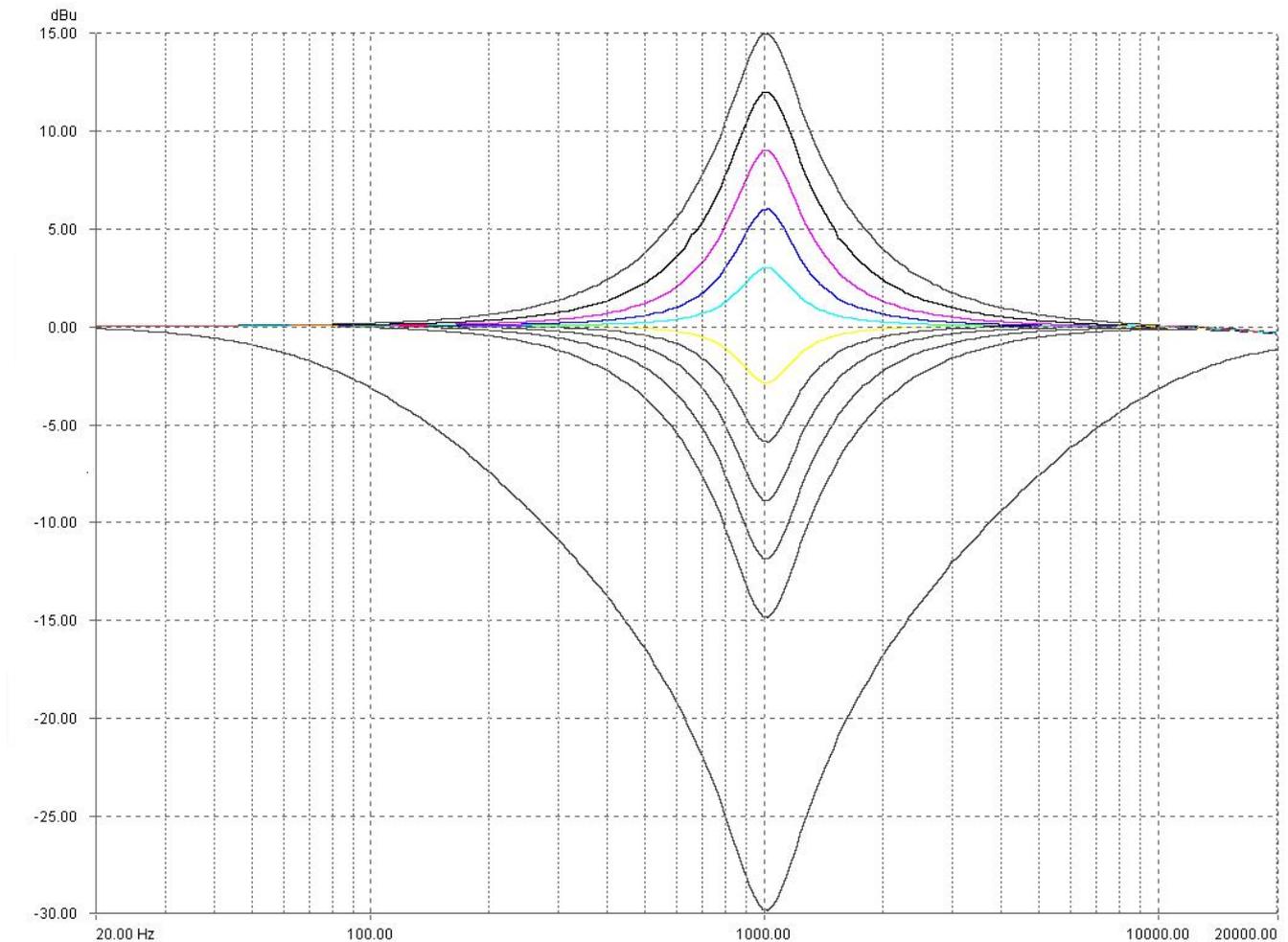
Note: Centigrade = (Fahrenheit - 32) x 0.5555.

Parametric Filter Types and their Uses

A wide selection of filter types has been made available under the PEQ section when editing input or output filters. Scrolling through the various filter types is achieved by repeated presses of the ENTER key. Note that this will only change filter types if the filter is BYPASSED or the GAIN set to 0 dB. Bypassing the filter, then changing types using the ENTER key will automatically set the gain back to 0 dB.

Each filter type will be explained in turn in the following section.

Standard Parametric EQ

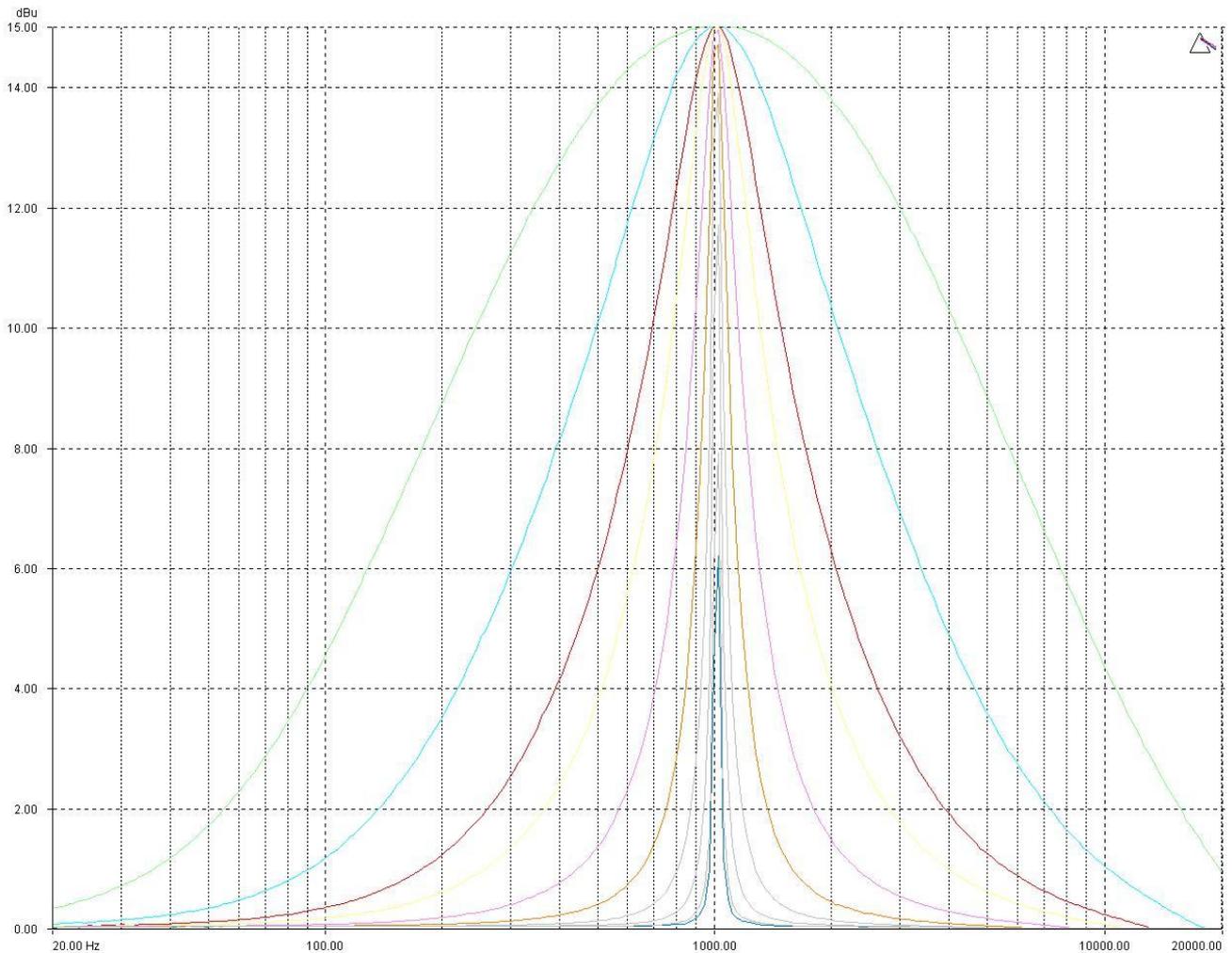


InA Input A PEQ:1<>
1k00Hz Q=3.0 0.0dB

The standard parametric band has adjustable frequency, 'Q' (or Bandwidth) and Gain controls. These affect a range of frequencies symmetrically about the centre frequency as shown in the graph.



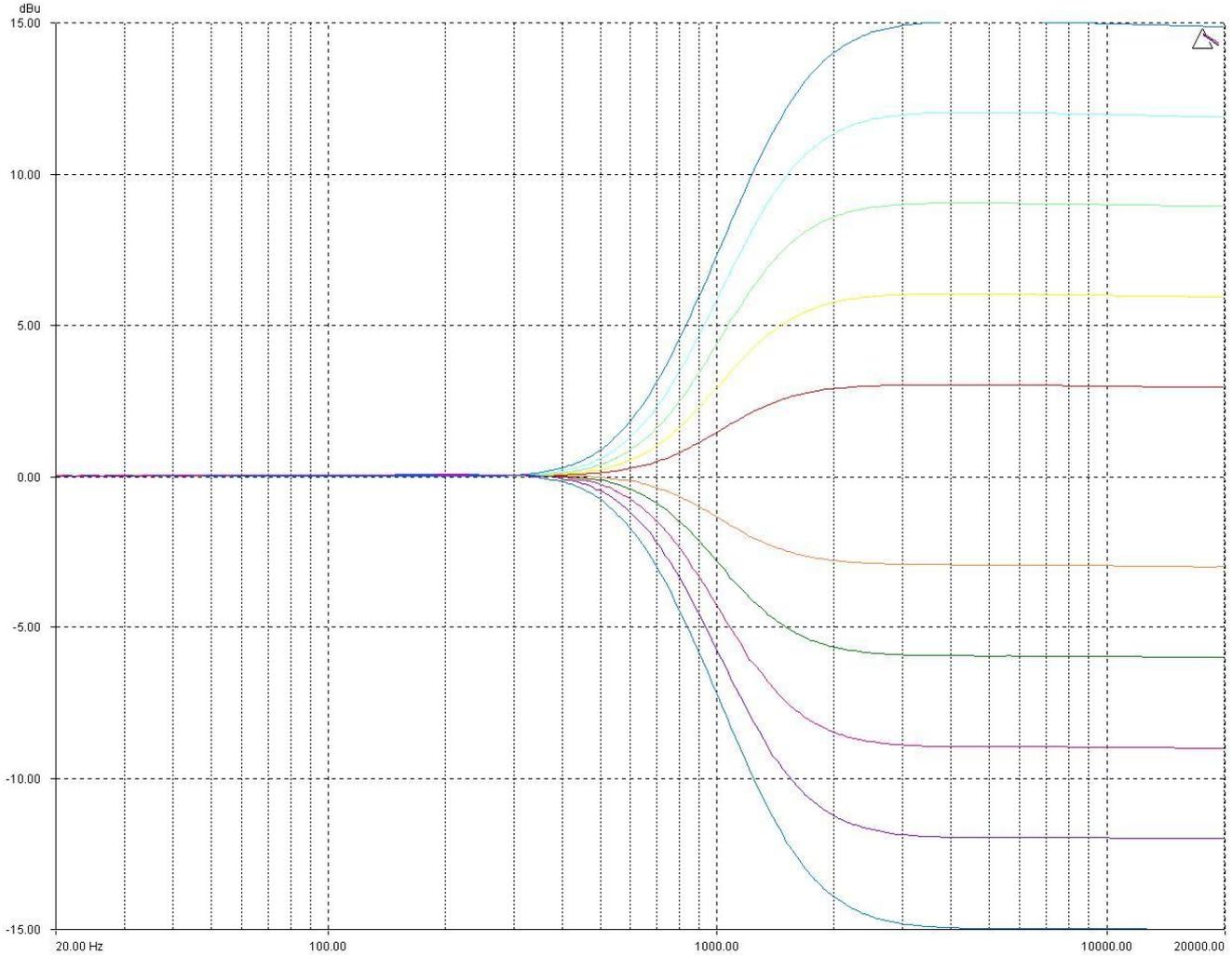
Various levels of cut and boost are shown to the left, along with various 'Q' settings (gain boosts only are shown below). Remember that 'Q' is 1/Bandwidth, so the higher the 'Q', the lower the Bandwidth, and the smaller the range of frequencies affected.





Shelving EQ (High Shelf shown)

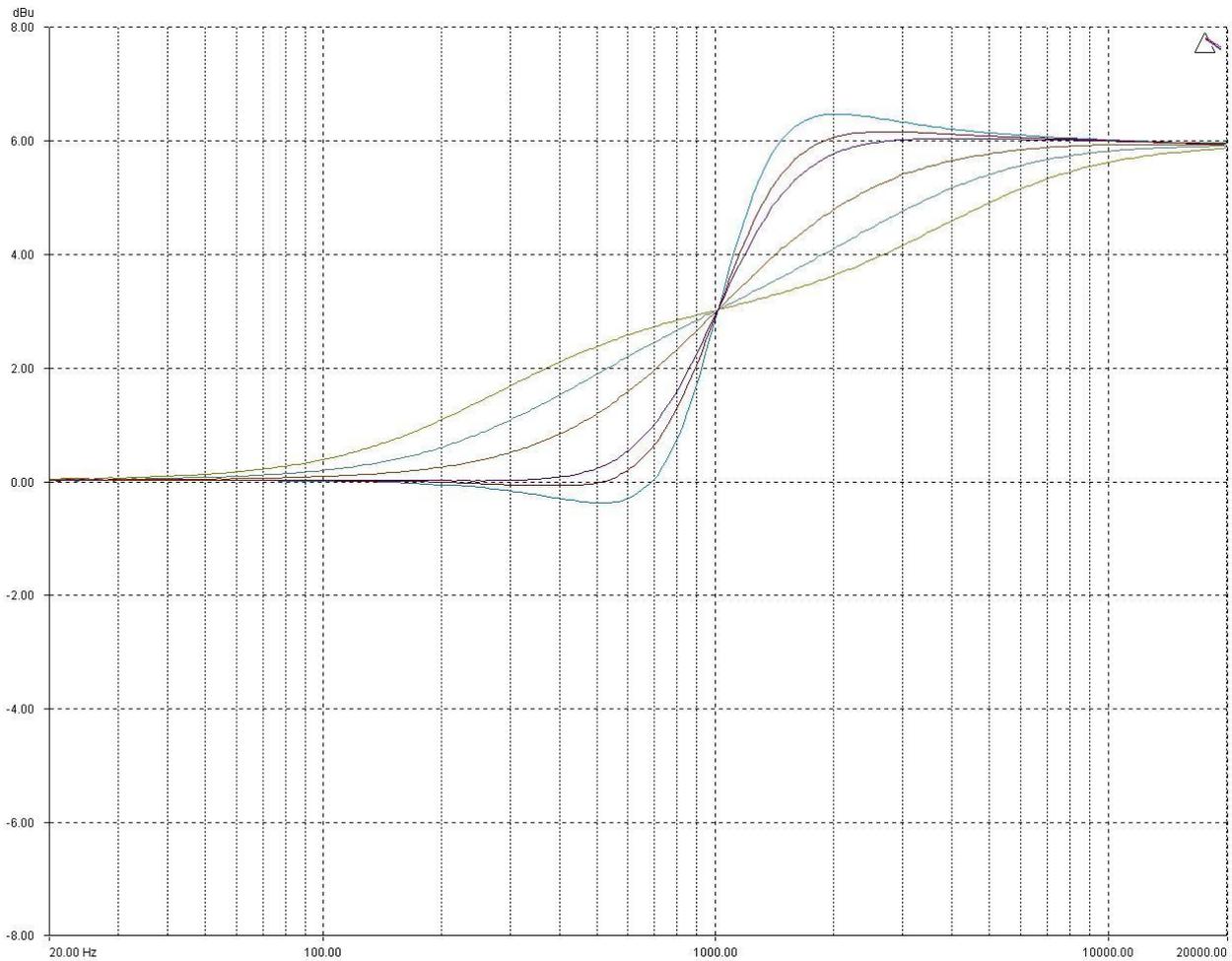
InA Input A HSF:1-<:
1k00Hz Q=3.0 0.0dB



Remember – to change filter types, press **BYPASS** to bypass the filter, and then use **ENTER** to select the filter type.

The shelving EQ has adjustable frequency, 'Q' (or Bandwidth) and Gain controls. These affect a range of frequencies from the turnover frequency as shown in the graph. For a high shelf, frequencies above the turnover frequency will be affected. For a low shelf, frequencies below the turnover frequency will be affected.

Various levels of cut and boost are shown to the left, along with various 'Q' settings (gain boosts only are shown below).

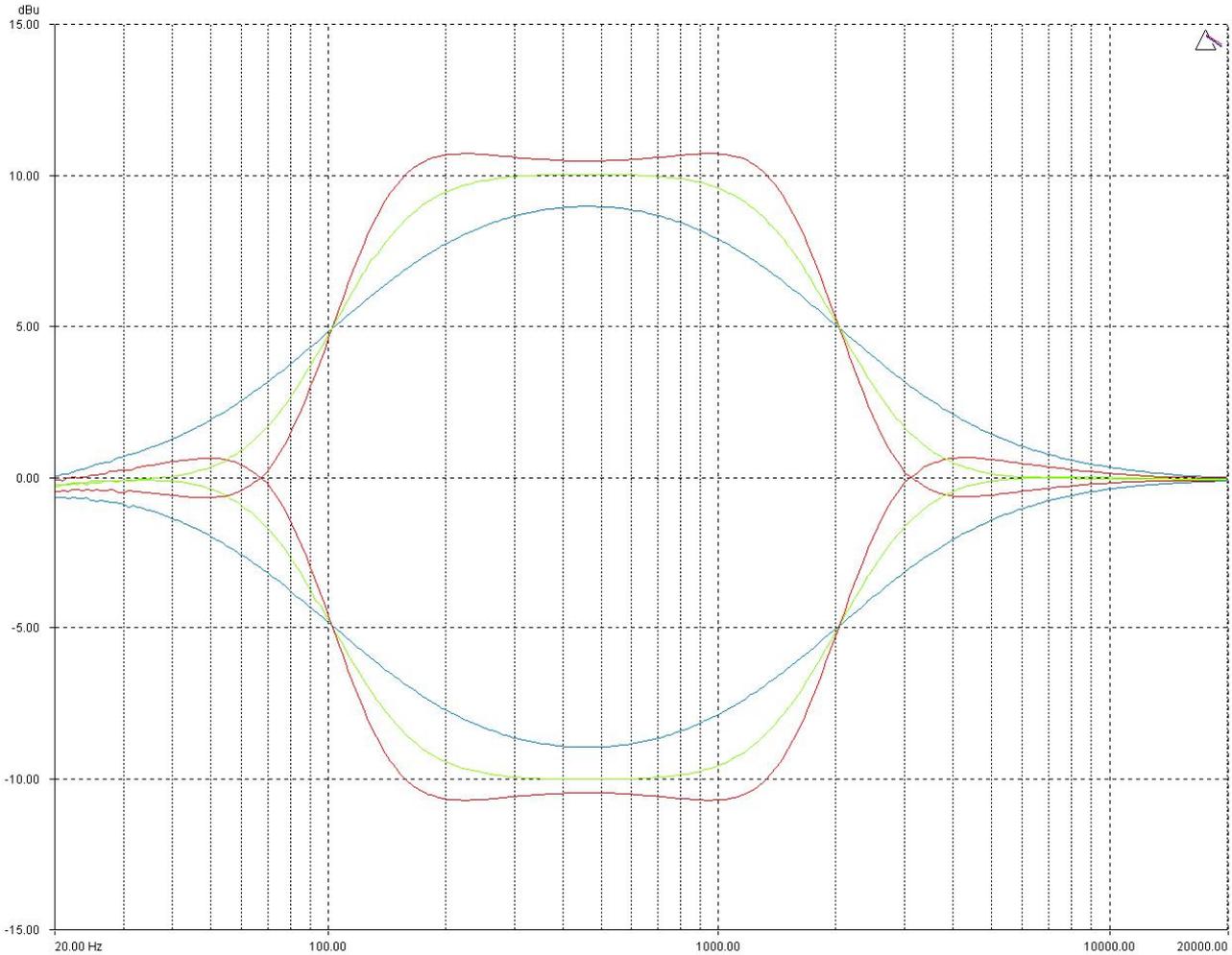


Remember that 'Q' is $1/\text{Bandwidth}$, so the higher the 'Q', the lower the Bandwidth, and the smaller the range of frequencies affected.

Note that 'Q' settings above 0.75 will result in slight overshoot in the filter response (as seen at the highest setting to the right). This is normal behaviour and does not indicate instability.



Creating a flat-topped EQ Response



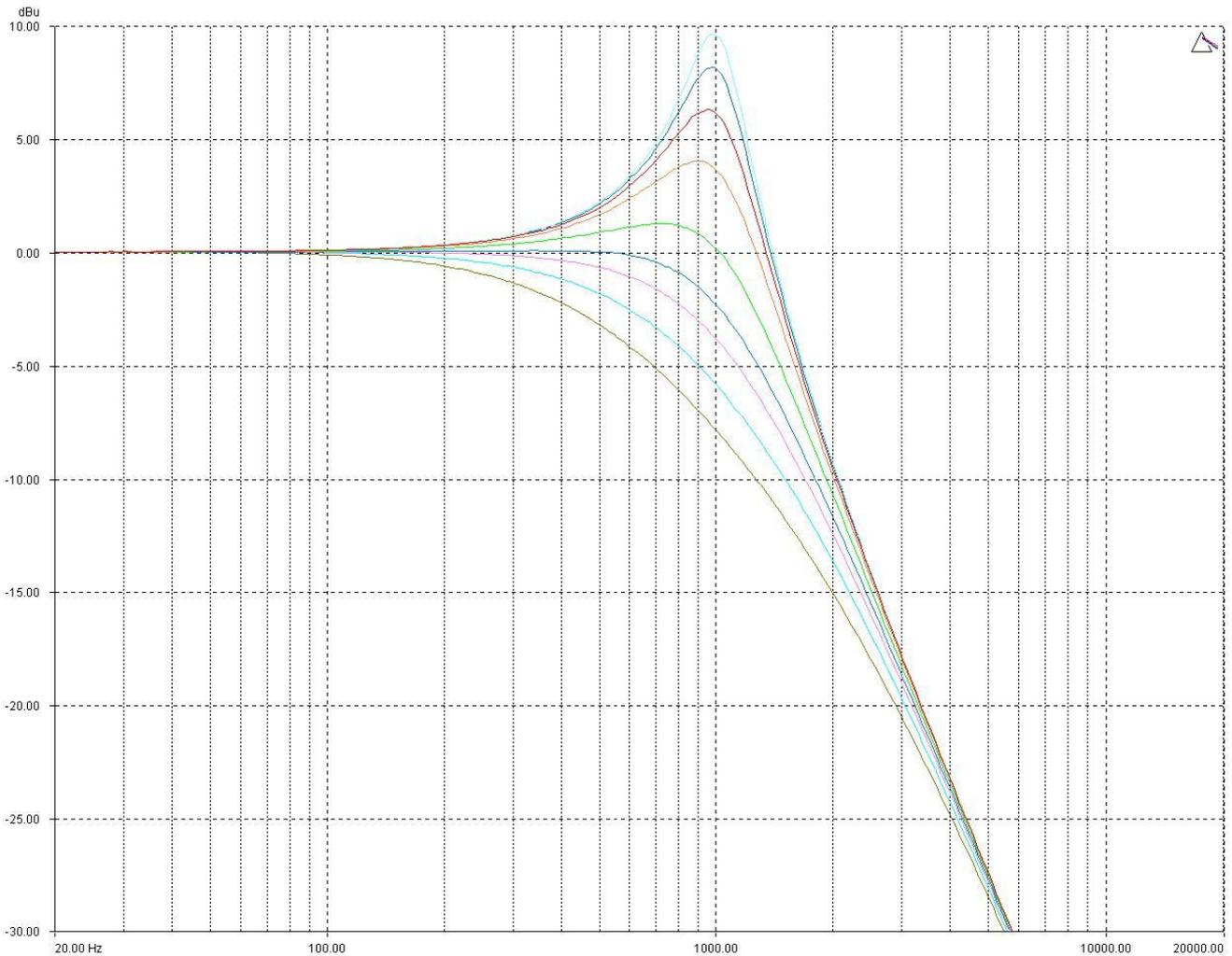
To create a flat-topped EQ filter response such as that shown to the left, use two EQ bands, **BOTH** configured as low shelves. For an overall **BOOST**, set the Lower frequency filter to **BOOST** the desired amount, and the Upper frequency filter to **CUT** by the same amount.

This example shows one filter at 100 Hz and the other at 2 kHz, with the 100 Hz filter at -10 dB, and the 2 kHz filter at +10 dB. Varying the 'Q' affects the slope of the response – values above 0.75 will cause overshoot as shown.



Low/High Pass variable 'Q' Filter (Low Pass shown)

InA Input A LPF:1~~\
1k00Hz Q=3.0 LPF VarQ



Remember – to change filter types, press **BYPASS** to bypass the filter, and then use **ENTER** to select the filter type.

The low and high pass variable 'Q' filters have adjustable frequency and 'Q' (or Bandwidth) controls. The 'Q' control adjust the damping of the filter, so that low 'Q' settings show less overshoot at the turnover frequency, but also slower roll-off.

Remember that 'Q' is 1/Bandwidth, so the higher the 'Q', the lower the Bandwidth, and the smaller the range of frequencies affected. The filter is primarily 12dB/Octave, but in achieving this sort of roll-off with a high 'Q' value will result in quite a large overshoot in level at the turnover frequency. This type of filter is often also called a resonant filter.

Specifications

Inputs: 2 electronically balanced ◆
Impedance: > 10 kOhms.
CMRR: >65 dB 50 Hz – 10 kHz.

Outputs: 6 electronically balanced ◆
Source Imp: < 60 Ohms
Min. Load: 600 Ohm
Max. Level: +20 dBm into 600 Ohm

Frequency Resp.: $\pm 1/2$ dB 20 Hz-20 kHz
 -3 dB @ 32 kHz

Dyn. Range: > 116 dB 20 Hz-20 k unwtld.
Distortion: < .02 % @ 1 kHz, +18 dBm

Maximum Delay: 650 mS
Min Step Size: 0.3 μ S

Input Gain: +6dB to -40 dB
 in 0.1dB steps

Output Gain: +15 dB to -40 dB
 in 0.1dB steps and mute

Parametric Equalisation

8 per Input / 9 Sections per Output

Filter Gain: +15 dB to -30 dB
 in 0.1 dB steps

Freq. Range: 19.7 Hz – 32 kHz,
 1/36 octave steps.

Filter Q/BW: 0.4 to 128/2.5 to 0.008
 (Sections switched to shelving response)

Low frequency: 19.2 Hz – 1 kHz

High frequency: 1 kHz – 32 kHz

Shelf gains: ± 15 dB in 0.1 dB steps.

High and Lowpass Filters

Filters: 1 of each per output.

Freq. Range HPF: 10 Hz – 16 kHz,
 1/36 octave steps.

Freq. Range LPF: 35 Hz – 22 kHz,
 1/36 octave steps.

Responses:

1st Order 6 dB/Oct.

Bessel/Butterworth/Linkwitz-Riley

12-24-48 dB/Oct.

Bessel/Butterworth 18 dB/Oct.

Limiters

Program Limiter:

Threshold: +22 dBu to -10 dBu

Attack time: 0.3 to 90 milliseconds

Release time: 2/4/8/16/32 x Attack time

“D-Max” Limiter:

Attack Time: -60uS

Release Time: Slow/Medium/Fast

Display:

2 x 24 Character LCD

Input meter:

2 x 4 point,
 -24 dB to digital clip.

Output meter:

6 x 4 point,
 -24 dB to +4 dB into limit.

Connectors

Inputs: 3 pin female XLR

Outputs: 3 pin male XLR.

External: 9 pin DEE connector (RS232)

Power:

3 pin IEC

Power:

60 to 250 V ± 15 %
 @ 50/60 Hz.

Consumption:

< 30 watts.

Weight:

3.3 kg. Net (4.7kg. Shipping)

Size:

1.75"(1U) x 19" x 11.8"
 (44 x 482 x 300mm)
 excluding connectors

◆ Transformer options available

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

Latency:

1.5 m (analogue in – analogue out
 @ 96 kHz)

CODA

C O D A A U D I O



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