User’s Manual

DSP-4080

Antes de utilizar el equipo, lea la sección “Precauciones de seguridad” de este manual. Conserve este manual para futuras consultas.

Before operating the device, please read the “Safety precautions” section of this manual. Retain this manual for future reference.

DSP-4080

Precauciones de Seguridad

Conserve y lea todas estas instrucciones.
Siga todas las advertencias.
El signo de exclamación dentro de un triángulo indica la existencia de componentes internos cuyo reemplazo puede afectar a la seguridad.

Aparato de Clase I, por tanto debe estar conectado a tierra.

El signo del rayo con la punta de flecha, alerta contra la presencia de voltajes peligrosos no aislados. Para reducir el riesgo de choque eléctrico, no retire la cubierta.

El equipo dispone de un conector estándar IEC60320-14, con portafusible, como conector de alimentación.
Utilice este equipo, sólo, con su apropiado cable de alimentación.

El cableado exterior conectado a estos terminales requiere de su instalación por una persona instruida o el uso de cables flexibles ya preparados.

Este símbolo indica que el presente producto no puede ser tratado como residuo doméstico normal, sino que debe entregarse en el correspondiente punto de recogida de equipos eléctricos y electrónicos.

La posición de encendido está indicada en el interruptor mediante los correspondientes símbolos normalizados (IEC 60417-1:1998 y IEC 60417-2:1998).

Si el aparato es conectado permanentemente, la instalación eléctrica del edificio debe incorporar un interruptor multipolar con separación de contacto de al menos 3mm en cada polo.

No exponga este equipo a la lluvia o humedad. No use este aparato cerca del agua (piscinas y fuentes, por ejemplo). No exponga el equipo a salpicaduras ni coloque sobre él objetos que contengan líquidos, tales como vasos y botellas. Equipo IP-20.

Limpie con un paño seco. No use limpiadores con disolventes.

No instale el aparato cerca de ninguna fuente de calor como radiadores, estufas u otros aparatos que produzcan calor. Debe instalarse siempre sin bloquear la libre circulación de aire.

Desconecte este aparato durante tormentas eléctricas, terremotos o cuando no se vaya a emplear durante largos periodos.

Tenga en cuenta que la tensión nominal de alimentación es el valor indicado en la etiqueta, con un rango ±10% de ese valor (según IEC 60065:2001). Si debe sustituir el fusible preste atención al tipo y rango.

Si el cable o enchufe de alimentación está dañado, debe ser sustituido por un cable o conjunto especial a suministrar por el fabricante o por su servicio postventa.

No existen partes ajustables por el usuario en el interior de este equipo. Cualquier operación de mantenimiento o reparación debe ser realizada por personal cualificado. Es necesario el servicio técnico cuando el aparato se haya dañado de alguna forma, tal como que haya caído líquido o algún objeto en el interior del aparato, haya sido expuesto a lluvia o humedad, no funcione correctamente o haya recibido un golpe.

Safety Precautions

Keep these instructions.
Heed all warnings. Follow all instructions.
The exclamation point inside an equilateral triangle indicates the existence of internal components whose substitution may affect safety.

Class I device. This equipment must be earthed.

The lightning and arrowhead symbol warns about the presence of uninsulated dangerous voltage. To reduce the risk of electric shock, do not remove the cover.

The device have a standard connector IEC60320-14, with fuseholder, for mains.
Only use this equipment with an appropriate mains cord.

The connected outer wiring to these terminals requires of its installation by an instructed person and the use of a flexible cable already prepared.

This symbol on the product indicates that this product should not be treated as household waste. Instead it shall be handed over to the appicable collection point for the recycling of electrical and electronic equipment.

The ON position is indicated in the switch by means of the corresponding standardized symbols (IEC 60417-1:1998 and IEC 60417-2:1998).

If the apparatus is connected permanently, the electrical system of the building must incorporate a multipolar switch with a separation of contact of at least 3mm in each pole.

Do not expose this device to rain or moisture. Do not use this apparatus near water (for example, swimming pools and fountains). Do not place any objects containing liquids, such as bottles or glasses, on top of the unit. Do not splash liquids on the unit. IP-20 equipment.

Clean only with a dry cloth. Do not use any solvent based cleaners.

Do not install near any heat sources such as radiators, heat registers, stoves or other apparatus that produce heat.
The circulation of air must not be blocked.

Unplug this apparatus during lightning storms, earthquakes or when unused for long periods of time.

Take into account that the nominal AC voltage is the value shown in the equipment ±10% (according to IEC 60065:2001). If the fuse needs to be replaced, please pay attention to correct type and ratings.

If the cable or the mains plug are damaged they must be replaced. Contact the manufacturer to provide you with the necessary spare parts.

No user serviceable parts inside. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally or has been dropped.

GARANTÍA

Todos nuestros productos están garantizados por un periodo de 24 meses desde la fecha de compra.

Las garantías sólo serán válidas si son por un defecto de fabricación y en ningún caso por un uso incorrecto del producto.

Las reparaciones en garantía pueden ser realizadas, exclusivamente, por el fabricante o el servicio de asistencia técnica autorizado.

Otros cargos como portes y seguros, son a cargo del comprador en todos los casos.

Para solicitar reparación en garantía es imprescindible que el producto no haya sido previamente manipulado e incluir una fotocopia de la factura de compra.

WARRANTY

All D.A.S. products are warrantied against any manufacturing defect for a period of 2 years from date of purchase.

The warranty excludes damage from incorrect use of the product.

All warranty repairs must be exclusively undertaken by the factory or any of its authorised service centers.

To claim a warranty repair, do not open or intend to repair the product.

Return the damaged unit, at shippers risk and freight prepaid, to the nearest service center with a copy of the purchase invoice.

DECLARACIÓN DE CONFORMIDAD
DECLARATION OF CONFORMITY

D.A.S. Audio, S.A.
C/ Islas Baleares, 24 - 46988 - Pol. Fuente del Jarro - Valencia. España (Spain).

Declara que el ***DSP-4080***:
Declares that ***DSP-4080***:

Cumple con los objetivos esenciales de las Directivas:
Abide by essential objectives relating Directives:

●	Directiva de Baja Tensión (Low Voltage Directive)	2006/95/CE
●	Directiva de Compatibilidad Electromagnética (EMC)	2004/108/CE
●	Directiva RoHS	2002/95/CE
●	Directiva RAEE (WEEE)	2002/96/CE

Y es conforme a las siguientes Normas Armonizadas Europeas:
In accordance with Harmonized European Norms:

●	EN 60065:2002	Audio, video and similar electronic apparatus. Safety requirements.
●	EN 55103-1:1996	Electromagnetic compatibility. Product family standard for audio, video, audio-visual and entertainment lighting control apparatus for professional use. Part 1:Emission.
●	EN 55103-2:1996	Electromagnetic compatibility. Product family standard for audio, video, audio-visual and entertainment lighting control apparatus for professional use. Part 2:Immunity.

Contents

User's Manual

Introduction

The ***DSP-4080*** 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 units have up to four inputs and eight outputs which can be configured in a selection of basic crossover modes – 4 x 2 way; 2 x 3 way + 2 Aux; 2 x 4 way; and 1 x 8 way (as applicable to i/o configurations). 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, including shelving, notch, band-pass, phase and elliptical behaviours.

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 48dB/Octave., with a variety of responses available.

Remote control is catered for in the form of RS232 and RS485 ports, and multiple user memories are provided for the storage and recall of settings. A GPI interface may also be fitted to allow remote memory recalls using simple switch closure apparatus.

Note that only the RS232 and RS485 interfaces offer full remote control of this product – the GPI interface may only be used for memory recall (program change) purposes.

Security lock-out is available for all controls.

The ***DSP-4080*** are also equipped with AES/EBU digital inputs and outputs, and include a sample rate converter, capable of accepting anything from 32kHz up tp 192kHz.

They may be controlled externally by XTA's proprietary AudioCore DSP Windows™ software, along with existing and future 'AudioCore' products.

Features

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

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

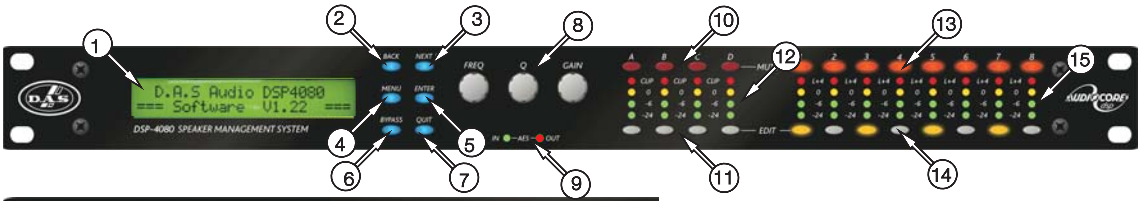
Both routing of inputs to outputs, and ganging (for editing) are completely flexible.

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 four inputs. This additional power also permits both program limiters and no overshoot clip limiters on each output.

Delay of up to 650mS may be independently set for each output, with an exceptionally fine minimum increment of 300nS, which corresponds to a distance change of 0.1mm!

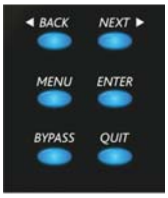
The comprehensive standard specification also includes up to 256 memories, and remote control via RS232 or RS485 ports, with security lockout.

Front panel description



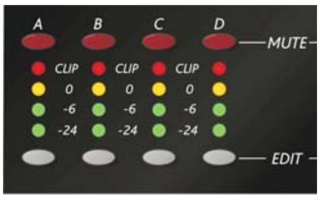
[1]-LCD Screen: Shows, by default, the name of the last recalled memory on the bottom line of the screen, and the current routing on the top line. Also used to show all parameters as they are edited, and all menu selections.

Control Keys: Selection and adjustment of parameters.



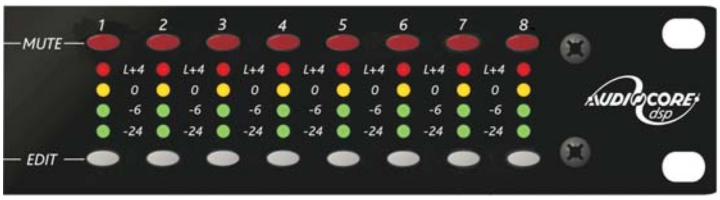
[2]-NEXT key moves forward through list of parameters.
[3]-BACK key moves backwards through list of parameters.
[4]-MENU key activates the main menu – a second press selects the last menu edited – a third press selects the last menu item. In this way, three presses on MENU from the default screen will jump back to the last parameter adjusted. Selection of different menus is accomplished using the **BACK** and **NEXT** keys, or with the **FREQ** encoder.
[5]-ENTER key enters the chosen menu, confirms selections, and changes filter types when editing parametric sections.
[6]-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.
[7]-QUIT exits menus back to the default screen.

[8]-Rotary Encoders: Three velocity sensitive encoders adjust the relevant parameters as displayed on the screen.
[9]-Status LEDs: The two status LEDs show, from left to right, AES inputs selected (flashing if not locked), and AES outputs selected.



Input Sections: Control and monitor input signal paths.
Red **[10]-MUTE** buttons illuminate when pressed and mute audio for that channel.
[11]-EDIT buttons illuminate yellow when pressed, and access gain on first press, then last viewed parameter on second press, then exit on third press.

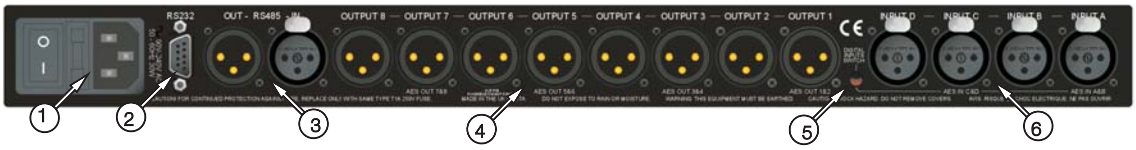
[12]- Input meters show dB from clipping point of the analogue to digital converters. Yellow (**0dB**) LED illuminates 3dB from clipping. Red **CLIP** LED may illuminate independently from the rest of the meter to show digital overflow. All four CLIP LEDs illuminating indicates internal clipping after the ADC.

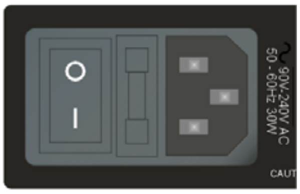


Output Sections: Control and monitor output signal paths.
Red **[13]-MUTE** buttons illuminate when pressed and mute audio for that channel.

[14]-EDIT buttons illuminate yellow when pressed, and access gain on first press, then last viewed parameter on second press, then exit on third press.
[15]-Output meters show dB from limiting. The yellow LED illuminates at the onset of limiting. The red LED illuminates at 4dB, '**L+4**', into limiting (i.e. 4dB of gain reduction).

Rear panel description





[1]:
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.

[2]-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 sockets.
[3]-RS485 IN-OUT: XLR sockets. Used for transmission of remote control data over long distance or multiple unit applications. See page 18 for more information.

[4]-Audio Outputs: 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 outputs are used for AES streams when the digital outputs are enabled. Please see page 23 for more information.

[5]-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.

[6]-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. Please see page 23 for more information.

WARNING:
Class I device. This equipment must be earthed. If the cable or the mains plug are damage they must replaced.
Always replace the fuse with the correct type and rating as shown on the rear panel legend.

Operating the device

Note about operation with AudioCore software:

The following operating information covers setup and control of the *DSP-4080* via the front panel controls only. Please consult the manual supplied with this software for information regarding full computer control.

Start-up procedure

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

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 *DSP-4080*.

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.

Information:
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

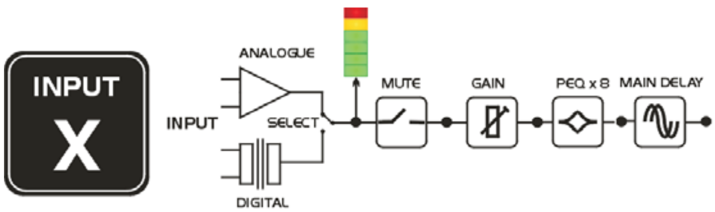
Due to the completely new DSP platform, the routing possibilities within the *DSP-4080* has been made completely flexible, with a matrix available allowing any combination of inputs to be routed to any output. The additional DSP power has permitted the inclusion of more processing blocks, even considering 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.

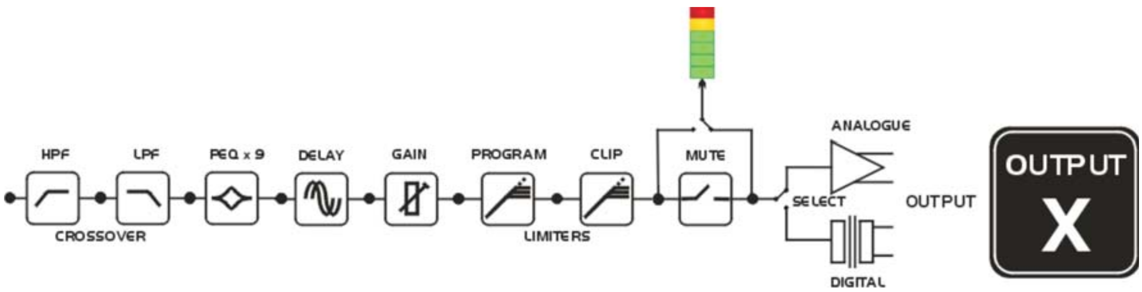
Input Channel Makeup

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.



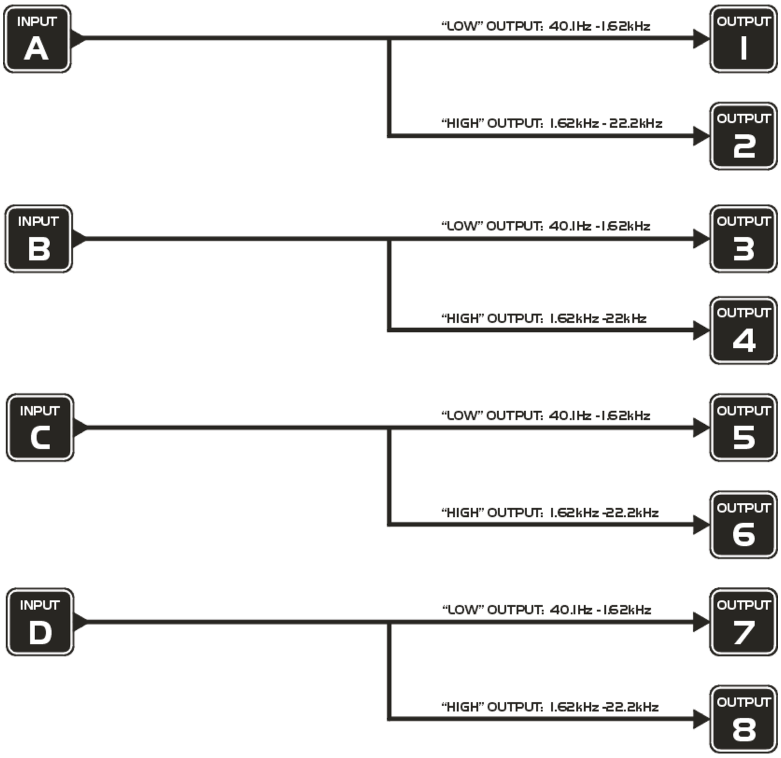
Preset Routing Configurations

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.

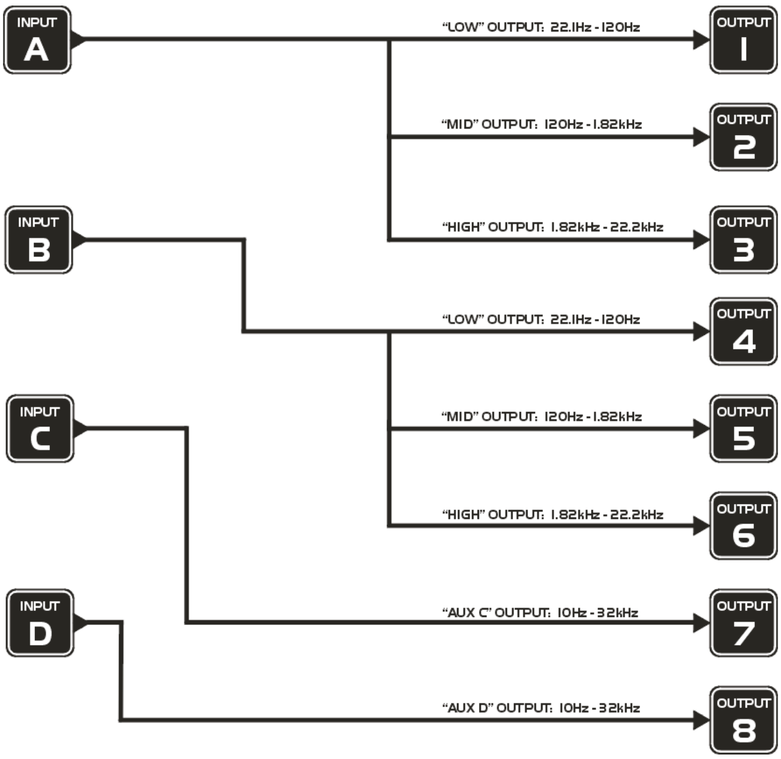
4 x 2 way crossover:

As shown, each input feeds a pair of outputs, odd numbers being the low frequency split, and even numbers being the high part of the spectrum. Default suggested crossover frequencies are shown by each output.



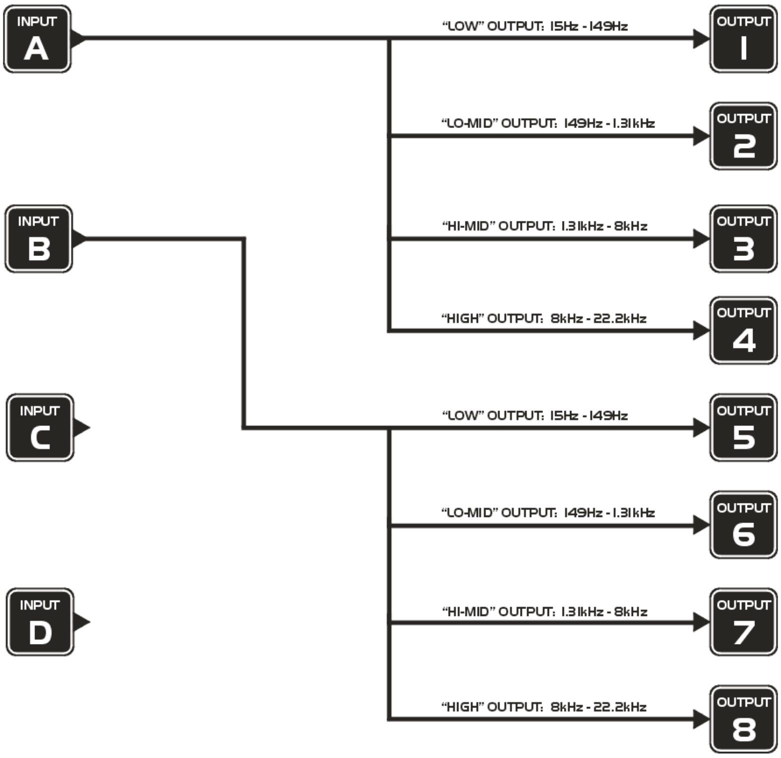
2 x 3 way crossover:

Inputs A and B feed three outputs each, with output 7 being fed from input C, and output 8 from input D. Note the 'Aux' outputs are set to full range. Default suggested crossover frequencies are shown by each output.



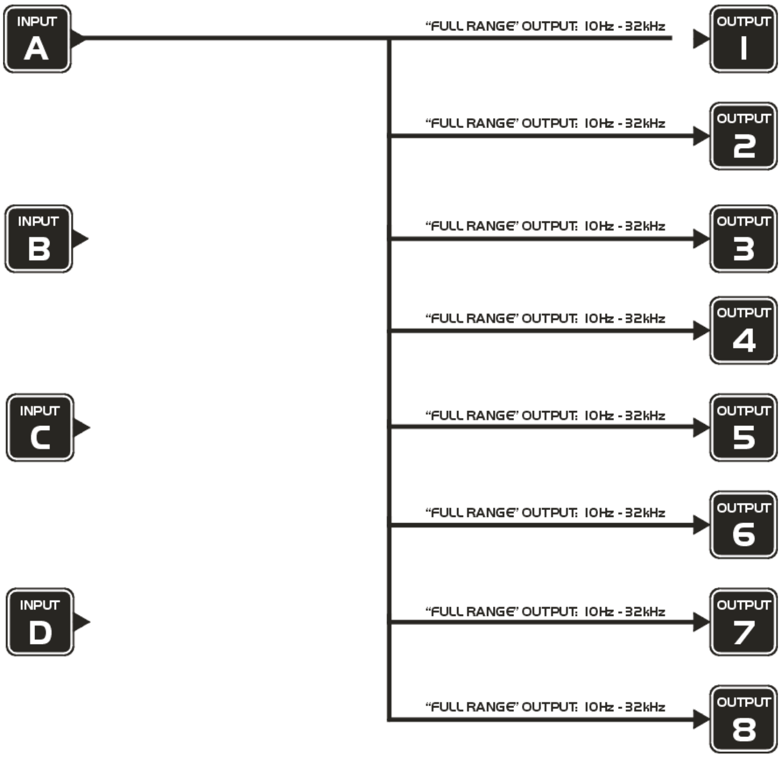
2 x 4 way crossover:

Inputs A and B feed four outputs each, with inputs C & D being unused. Default suggested crossover frequencies are shown by each output.



1 x 8 way crossover:

Inputs A is fed to all eight outputs, with initial settings being all full bandwidth. The crossover points can be adjusted as 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 = 4 x 2 WAY *

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

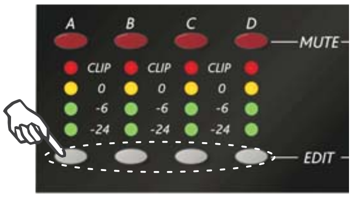
Design A Crossover ->
Routing = Free Assign

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.

Note that:
1st.- Press EDIT to show input routing selection...
2nd.- Press EDIT input keys to select/deselect inputs.

Editing audio parameters - Input channels

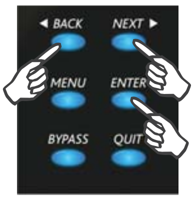


Input gain

The range of the control over the input gain is -40dB to +6dB in 0.1dB steps. Pressing EDIT, the display shows:

IPA Input A Gain
Input Gain = +6.0dB

The GAIN allows to change its value. Pressing ENTER to confirm.



Base delay

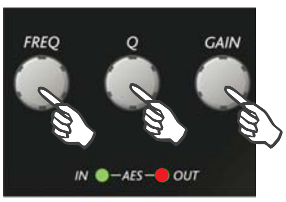
The maximum available delay between any input and output is 650mS. For example, if the input delay on channel A is set to 500mS, the maximum available output delay for any output fed from input A will be 150mS. The readout units can be changed between time in milliseconds, distance in feet or distance in metres.

IPA Input A Delay
Base Delay = 0.00mS

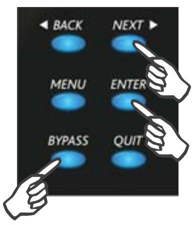
Note: The delay steps are 1mS (343mm) through FREQ, or 10uS (4mm) through Q encoder.

Input parametric EQ

The display shows:

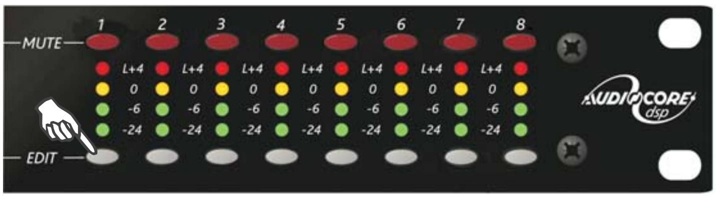


IPA Input A PEQ : 1 <>
1k00Hz Q=3.0 0.0dB



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. For more details about the various types of filter available, please see page 32.

Editing audio parameters - Output channels



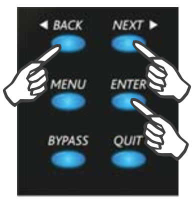
Output gain

The range of the control over the output gain is -40dB to +15dB in 0.1dB steps. Pressing EDIT the display shows:

OP1 Output 1 Gain
Output Gain = +6.0dB

Output polarity

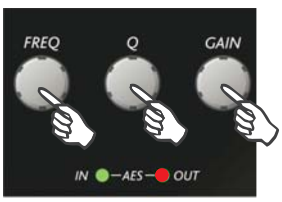
The polarity (or phase) of each output may be switched individually as below.



OP1 Output 1 Polar
Polarity = [+]

Using GAIN, the phase may be changed between '-' (inverted phase) or '+' (non inverted phase).

Output delay



The maximum available delay between any input and output is 650mS. 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.

OP1 Output 1 Delay
Delay = 0.0000mS

Note: The delay steps are 1mS (343mm) through FREQ, 10uS (4mm) through Q, or 0.3uS (0.1mm) through GAIN encoder.

Output high pass filter

The high pass crossover filter on each output has a frequency range of <10Hz up to 32kHz 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 48dB/Octave filters, parametric bands 6 & 7 need to be bypassed, or set to 0dB. If they are not, the message

'Bypass PEQ's 6 & 7 To Access 48dB Slopes'

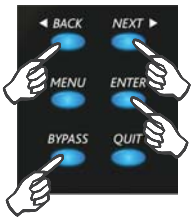
will be displayed.

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

The FREQ allows to change the frequency, and using Q the slope.

Output low pass filter

The low pass crossover filter on each output has a frequency range of 35.1Hz up to >32kHz 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 48dB/Octave filters, parametric bands 8 & 9 need to be bypassed, or set to 0dB. If they are not, the message

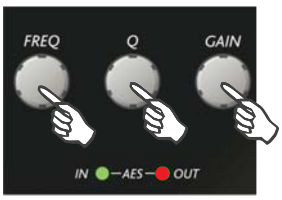
'Bypass PEQ's 8 & 9 To Access 48dB Slopes'

will be displayed.

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

The FREQ allows to change the frequency, and using Q the slope.

Output parametric EQ



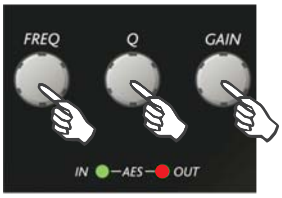
There are nine bands of parametric equalisation available on every output. The behaviour of each individual band can be changed to a variety of different filter shapes, including high and low shelves, notch, and bandpass. Changing the filter type is achieved by pressing BYPASS to bypass the filter and then pressing ENTER during editing any particular band. For more details about the various types of filter available, please see page 32.

OP1 Output 1 PEQ:1<>
1kHz Q = 3 0.0dB

Information:
Note that 2 bands each will be lost when using 48dB slope crossover filters, resulting in a maximum of 5 bands of EQ when both high and low pass are set to 48dB/Octave.

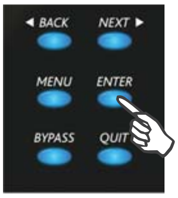
Output limiter

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 2mS and release is "x16" so 32mS. 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=2.0mS Rel=x16 +22dB

Using FREQ, the attack time change. Through Q the release time change, and through GAIN the threshold change. To complete the procedure, press ENTER.



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'**.
More information about the limiters and their use is given in the section on page 27.

OP1 Output 1 ClipLim
Rel.=Medium 2dB Above

Using FREQ, the release time change, and through GAIN the threshold change. To complete the procedure, press ENTER.

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+7 2+4+6+8

…would be a logical ganging arrangement if the crossover was set up as a 4 x 2 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+7+8	**[1x8 way]**
Ganging=1+5 2+6 3+7 4+8	**[4x2 way]**
Ganging=1+3+5+7 2+4+6+8	**[2x4 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

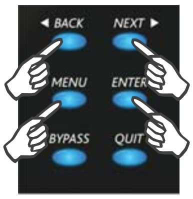
Ganging is cleared by selecting **'Ganging=None'** from the initial choices given above. The **'Input Ganging'** procedure is identical to the crossover ganging, selectable under the **'Input Sub-Menu'**.

Memory structure

The *DSP-4080* 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, two types of memory available: **'Input Only'** and **'Crossover Only'**,also combinations **'Input & Xover'**.

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



RECALL a Memory
Type= Input & Xover *

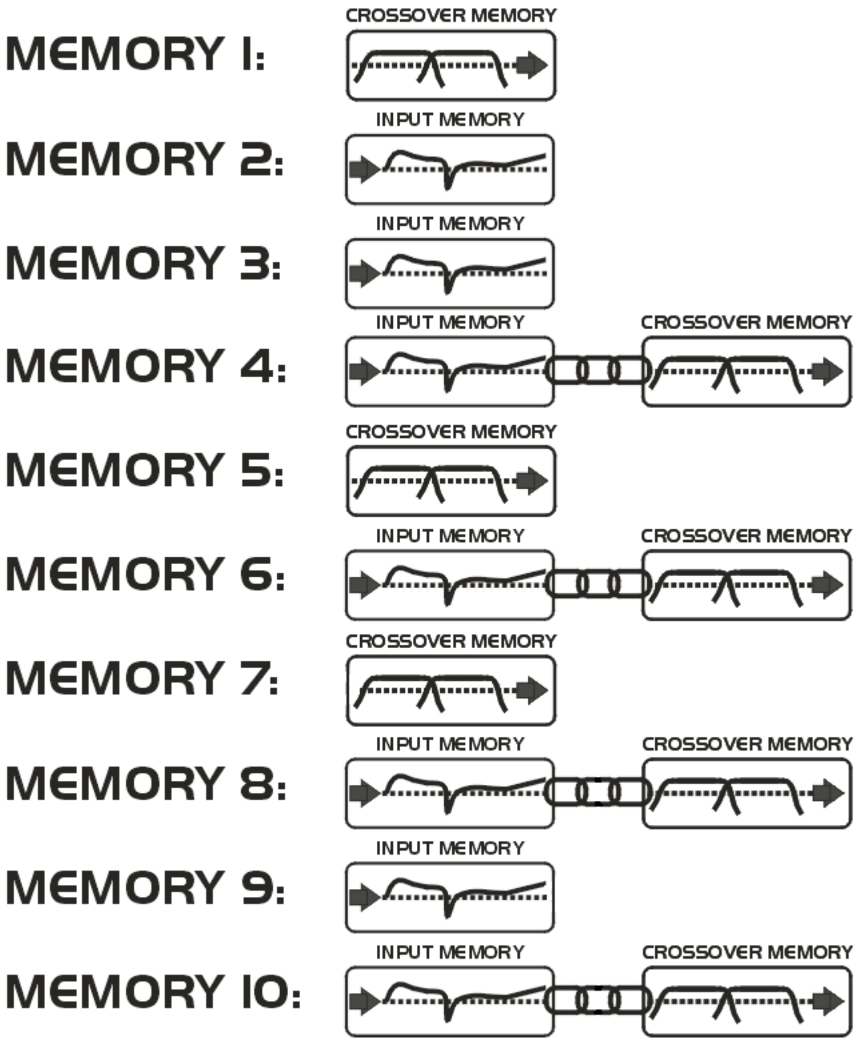
STORE a Memory
Type= Input & Xover *

ERASE a Memory
Type= Input & Xover *

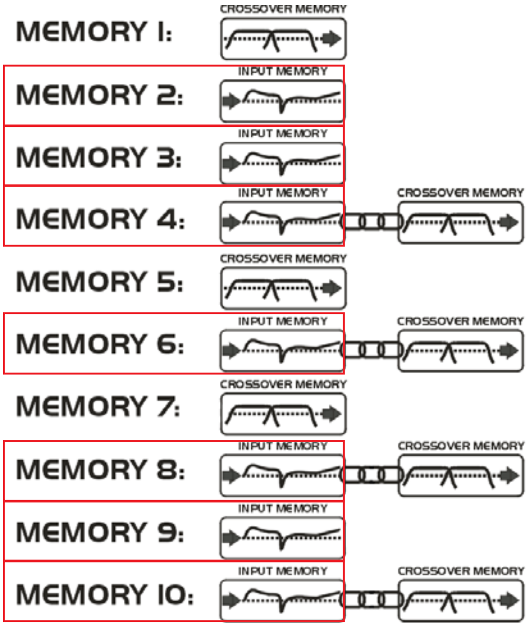
Selecting to **'STORE'** or **'RECALL'** using the **'GLOBAL MEMORY Sub Menu'** option offers the possibility of storing various combinations of the available memory types, and these are selected using the BACK and NEXT keys.

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

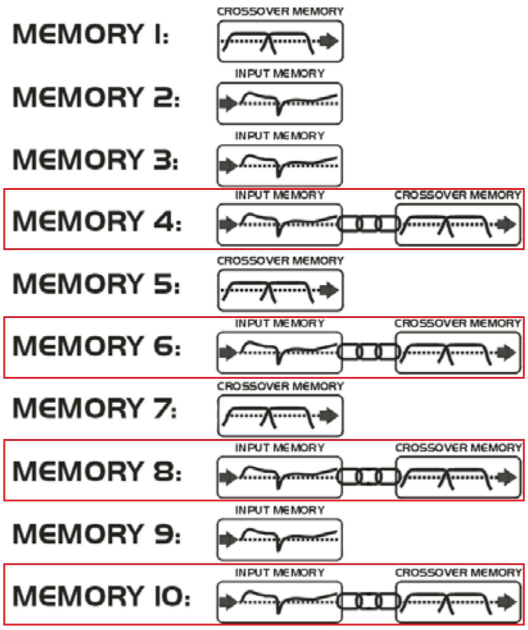
There are 10 memories stored in the unit with various combinations of input and crossover memories.



Through **'STORE'** using **'GLOBAL MEMORY Sub Menu'**, we have stored 10 memories.
If it is required to recall a location that contains **'Input Only'** settings, this will limit the selection as shown below: 2, 3, 4, 6, 8, 9, and 10.



But, recalling 4, 6, 8 or 10 will leave the current **'Xover'** settings untouched.
However, if it is required to recall **'Input & Xover'** settings then 4, 6, 8, and 10 will be the only numbers available, with the option to change all the parameters.



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 *DSP-4080* 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.

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.

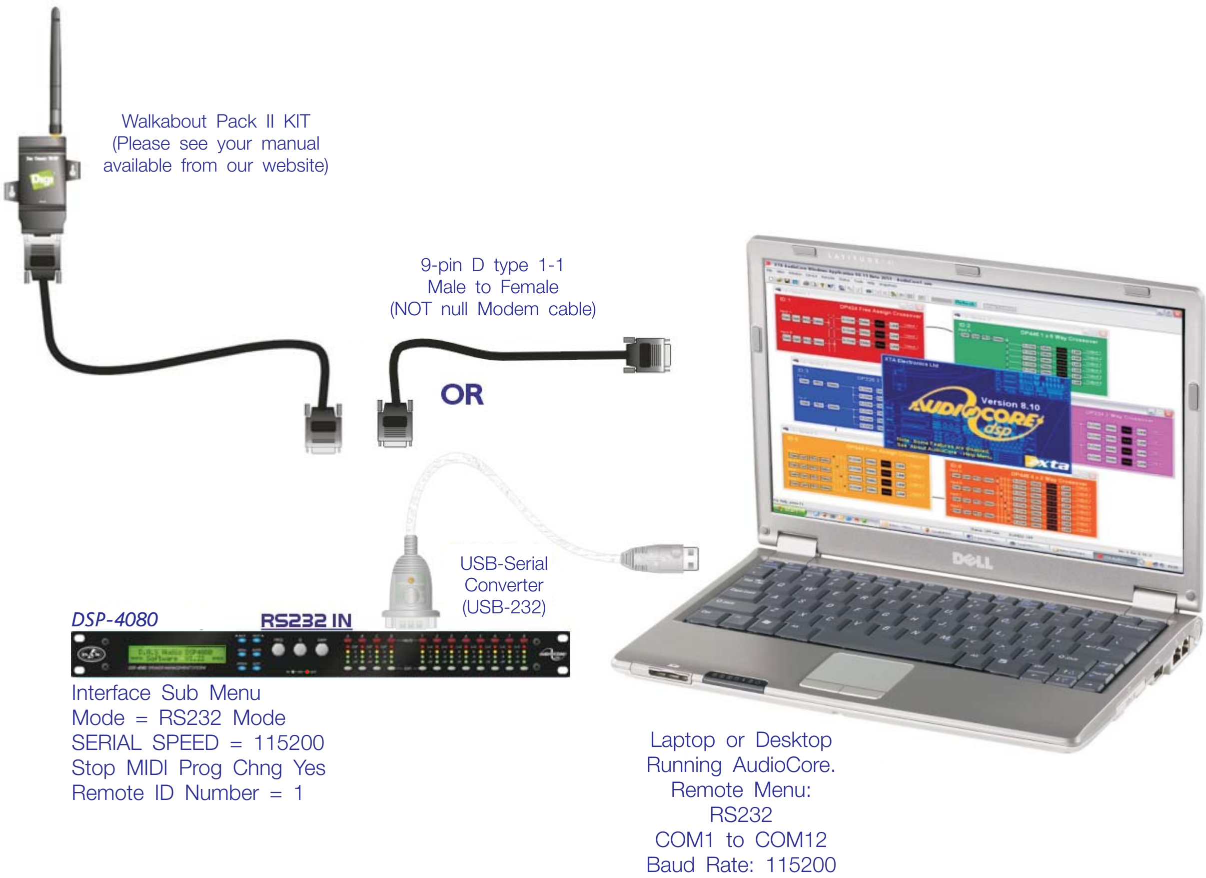
The RS232 connection is suitable for distances of about a maximum of 25 feet (7.62 m) between the PC and the unit. If you experience problems with the connections, consider

- selecting a slower baud rate
- selecting the 'Use Acknowledge Cmd' option in AudioCore (see the Remote Menu > RS232 Configuration window)
- running the unit via the RS485 interface

Note than only one unit at a time may be connected to the computer via this interface. Additional units may be 'daisy-chained' via the RS485 connections from the back of the first one (it acting as a converter for them), but their RS232 ports are not used.

RS232 Connection (Single Unit)

A typical interface set-up might involve running an RS232 link from laptop or a desktop computer to a *DSP-4080* 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).



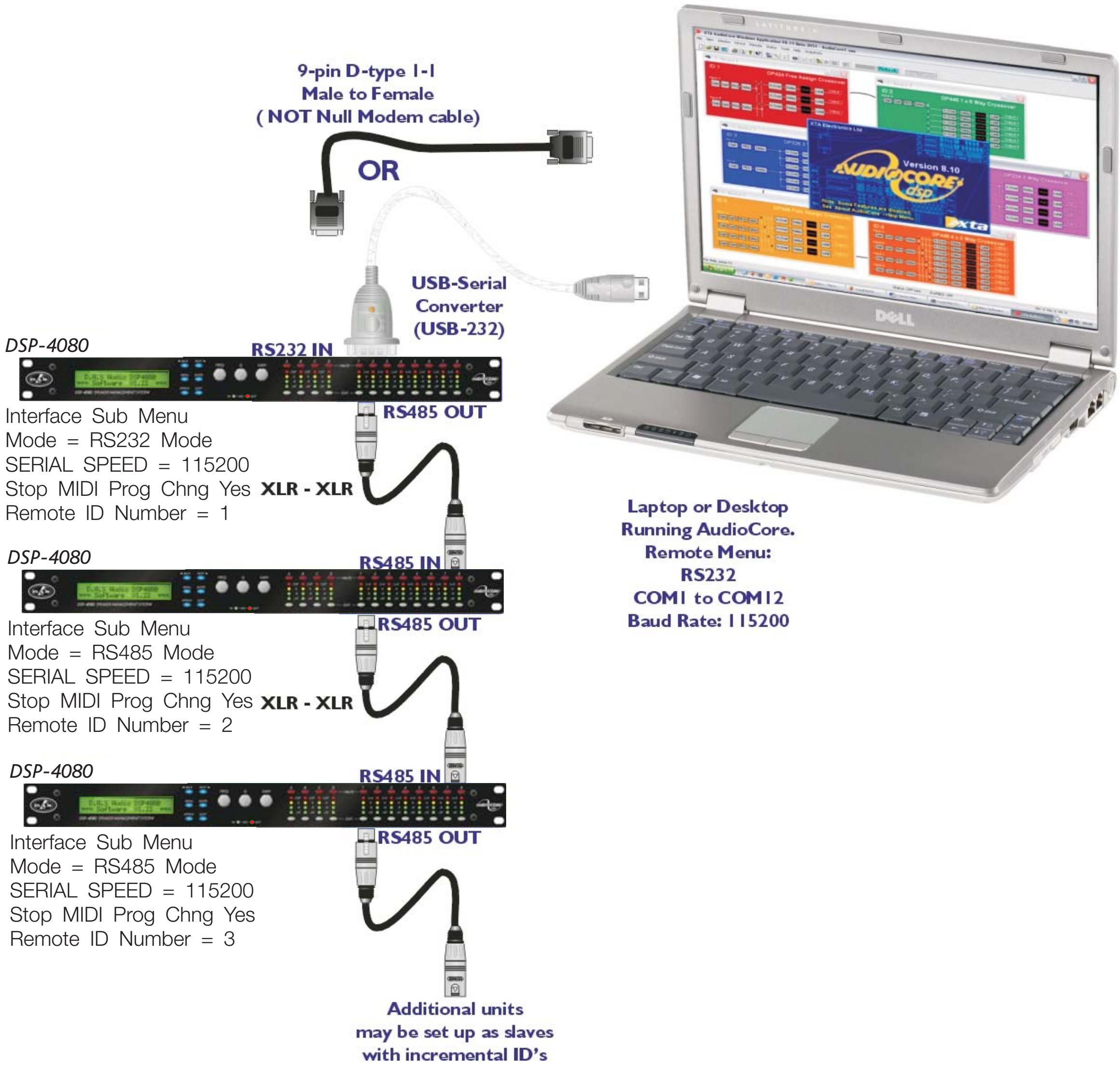
Loading New Software via a PC

The unit's internal software may be updated via the RS232 port ONLY, one unit at a time. We recommend disconnecting all other devices when updating the software.

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.

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 unit. Note the incremental 'ID NUMBER' option in the unit's interface setup.



Shadow ID Numbers

Shadow ID numbers allow extra units to share the same ID and follow the settings of the 'main' ID. This is useful for larger systems (for example anything above a 4-way stereo system) where it is only necessary to set up one side of the system, and allow the other unit to track it identically.

Using the shadow IDs in this way also reduces the apparent system complexity within AudioCore. This is due to the fact that shadow ID's NEVER send back any settings to AudioCore and because of this will NOT appear in the list of connected units.

They can be thought of as listening to and acting upon all information addressed to them, but not replying. Up to 128 shadow units may be connected and assigned the same ID as the 'main' unit, but remember that the maximum total units on any one RS485 network is 128.

RS485 interface

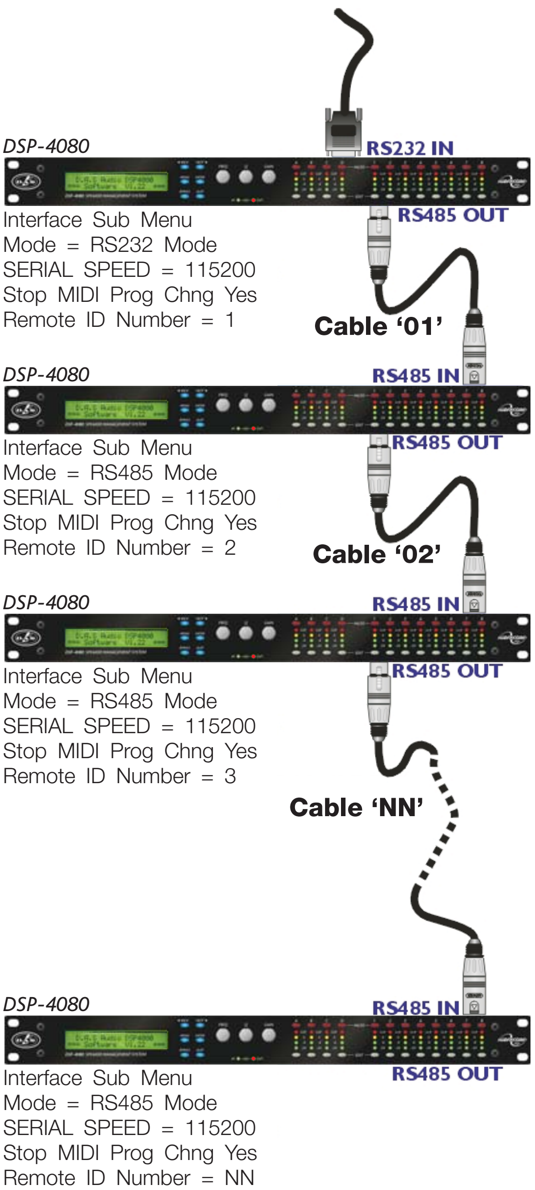
This interface is fitted as standard to *DSP-4080* units and is accessed via the 3-pin XLR sockets on the rear of the unit. Cables to connect units together or to an RS232-485 converter will need to be wired one-to-one. We recommend the use of standard shielded microphone cables, or a balanced feed from a multicore.

RS485 is a fully balanced system, capable of sending data over distances of up to one kilometre. Note, however, that this is the total length of connection. The RS485 output of each unit is purely hardwired from the input and so no electrical regeneration of the signal is provided. What this means is that the distance from the first RS485 output to the last RS485 input must not exceed 1km in total.

As this diagram illustrates – The combined length of cables 01 + 02 + …NN < 1000 metres.

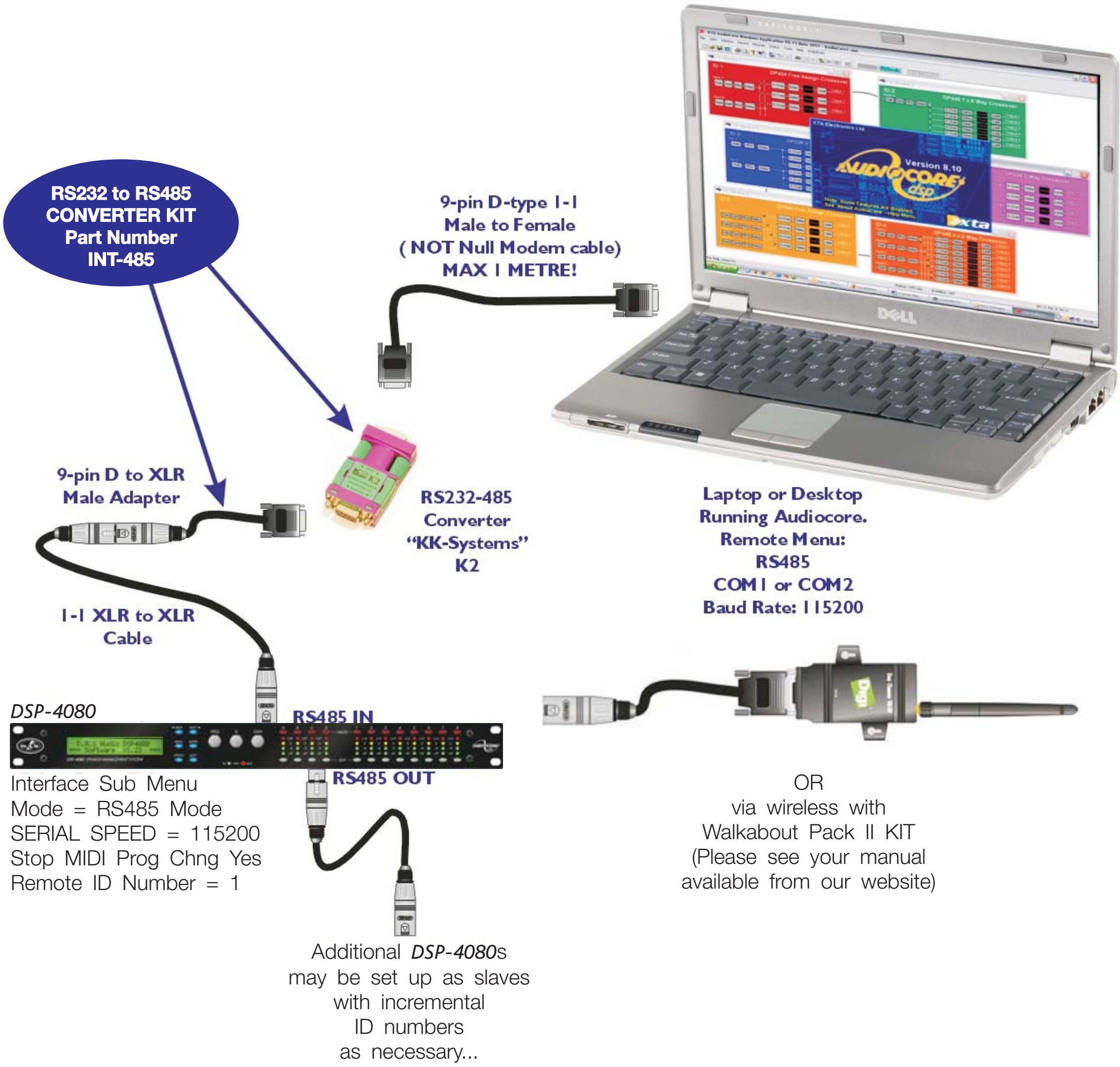
Note that this includes any units set up as shadow IDs.

Units may be connected in a star configuration if required, so a long cable may be used and then a hub formed with units all connected to this, but we do not recommend daisy chaining these configurations together.



RS485 connection

To use RS485 communication directly from a computer, a master unit 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 unit setup.

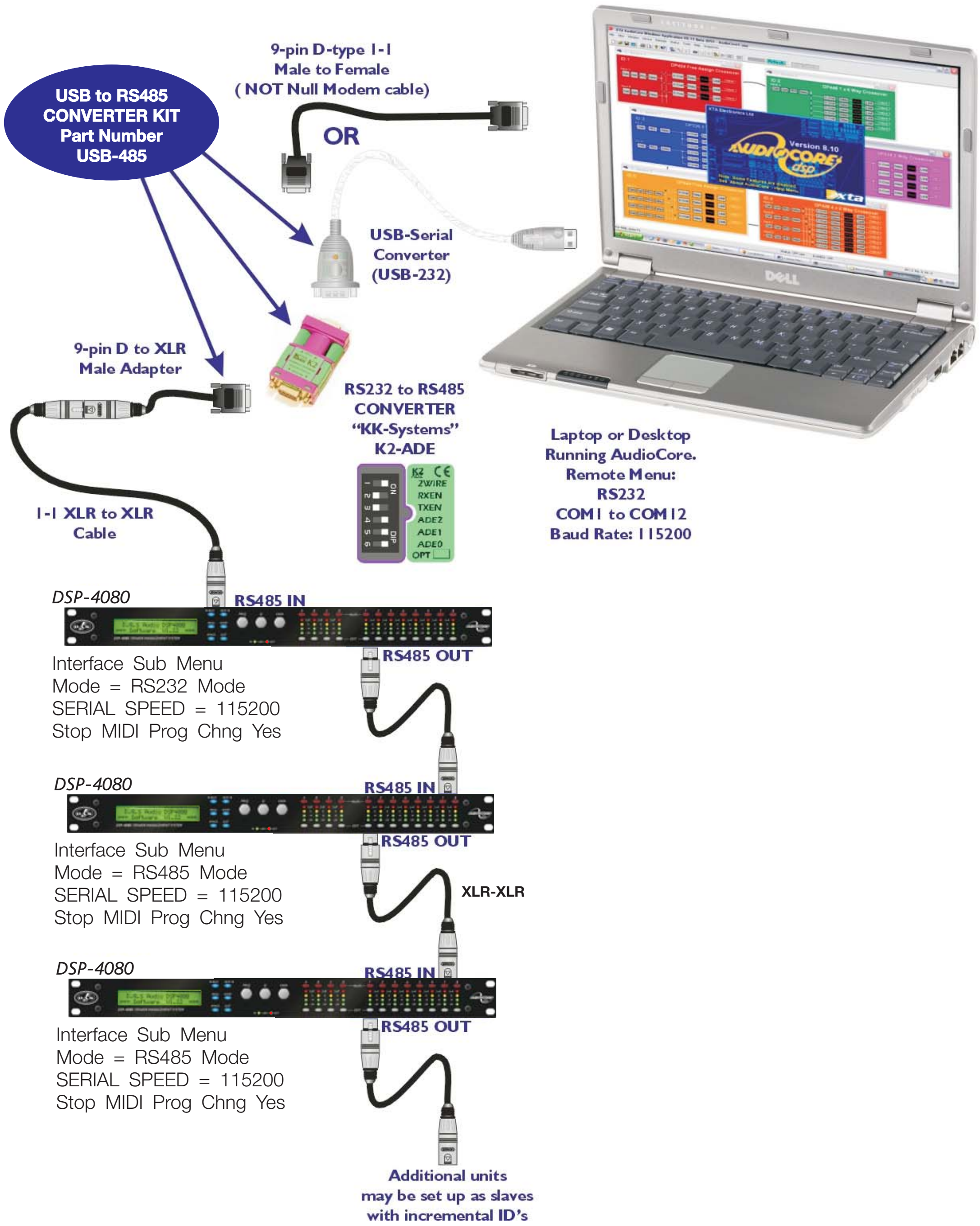


The adapter is available in a kit, which includes an RJ-45 adapter, the XLR to 9-pin adapter, and the converter itself. This complete kit is part number INT-485.

If you need to make up one of the XLR to 9-pin adapters, the pin-out is:

XLR	D-type
1	1
2	3
3	8

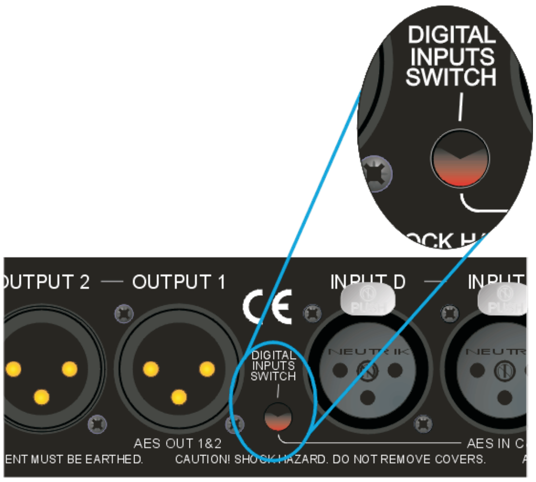
If your laptop or PC does not have a spare serial port (or any serial ports for that matter!), the RS485 converter must be connected through a USB – Serial converter. The RS485 converter that DAS recommend is available in two types – the standard K2, and the more advanced K2-ADE version. **Only K2-ADE version will work with USB-Serial converters**, as these converters do not support the extra handshake lines used with the standard converter.



The adapter is available in a kit, which includes a USB-Serial converter , the XLR to 9-pin adapter, and the K2-ADE converter itself. This complete kit is part number USB-485.

AES inputs and outputs

The *DSP-4080* units have a full AES implementation built in as standard. This allows the unit to both receive digital audio directly, and to transmit digital audio on to other devices. The switching of input and output can be performed independently, and the inclusion of sample rate converters on the inputs allows the unit to accept sample rates from 32kHz up to 192kHz.



AES input

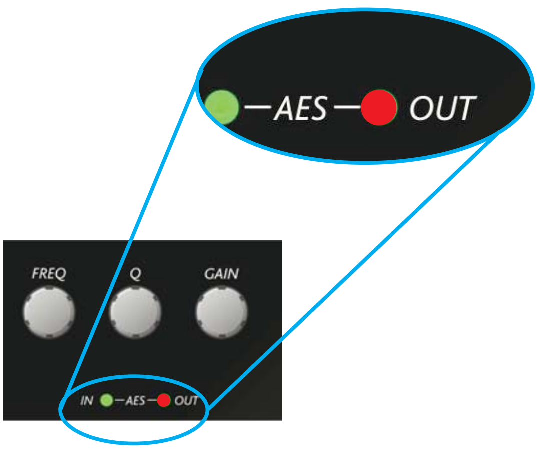
Input selection is via a recessed switch on the rear panel of the unit, between input D and output 1. A red LED inside this aperture illuminates to show that the AES digital inputs have been selected.

A complimentary LED on the front panel 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:

For channels A & B use 'Input A'
and for channels C & D use 'Input C'



AES output

AES outputs are selected through the AES menu:

AES/EBU Sub Menu
Output Selection

Pressing ENTER and then using BACK and NEXT chooses either 'Analogue' or 'Digital'. Press ENTER again to confirm selection.

The AES outputs are marked on the rear panel:
Channels 1 & 2 use 'Output 1'
Channels 3 & 4 use 'Output 3'
Channels 5 & 6 use 'Output 5'
Channels 7 & 8 use 'Output 7'

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 V3: A

This display shows the correct operation of the three AES transmitters V1, V2 and V3. The letter after each is the silicon version (and is of no importance to the user).

Pressing ENTER again will show

AES Device Status
V: 96k0 V: 96k0

This display shows the status of the two AES receivers, input A on the left, and Input C on the right. The sample rate the unit has been able to lock to is shown, or UNLOCKED will be displayed in its place. The unit will lock to sample rates from 32kHz up to and including 192kHz.

The unit's own processing sample rate is 96kHz, and AES output data is always at 96kHz. Internal sample rate converters will translate all incoming rates to 96kHz – one converter for each AES input. This allows the two input streams to be at different rates if necessary.

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 (+ 6dB) 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, C, D, 1, 2, 3, 4, 5, 6, 7** and **8** can be used as a code, as shown below. Press ENTER to accept code and then re-enter it to confirm.

Note:

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 *DSP-4080*, 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 (see page 25 for details) 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 DAS. 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 we provide 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

High performance digital limiters are provided for each output with control over attack time, release time and threshold parameters - see details below. 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. We 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.

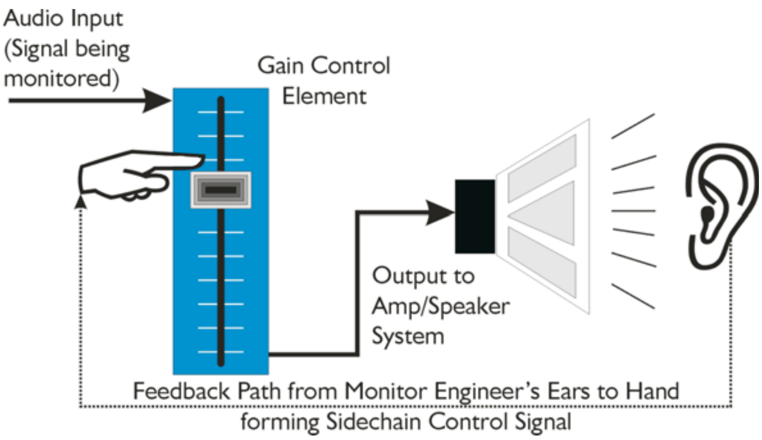
High Pass Filter	Auto Attack Time	Release Time
<10Hz – 31Hz	45mS	x16 (720mS)
31Hz – 63Hz	16mS	x16 (256mS)
63Hz – 125Hz	8mS	x16 (128mS)
125Hz – 250Hz	4mS	x16 (64mS)
250Hz – 500Hz	2mS	x16 (32mS)
500Hz - 1kHz	1mS	x16 (16mS)
1kHz – 2kHz	0.5mS	x16 (8mS)
2kHz – 32kHz	0.3mS	x16 (4mS)

"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.

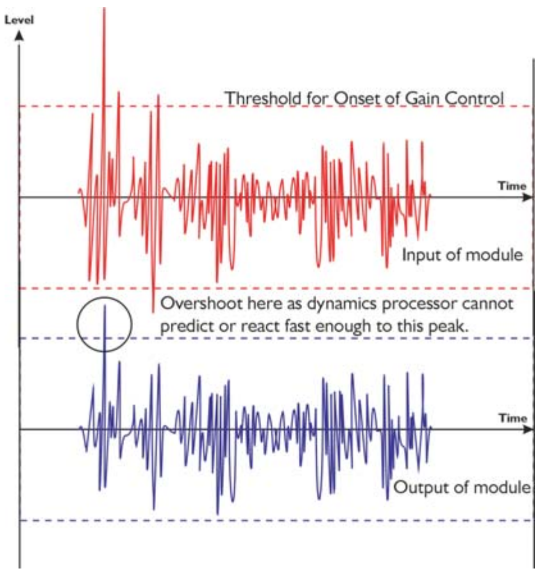
In the case of dynamics processing, being able to delay a signal allows the processor module to delay the main signal in relation to the sidechain (the signal being monitored relative to the threshold), so that it can compensate for peaks prior to the arrival of the main signal.

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.

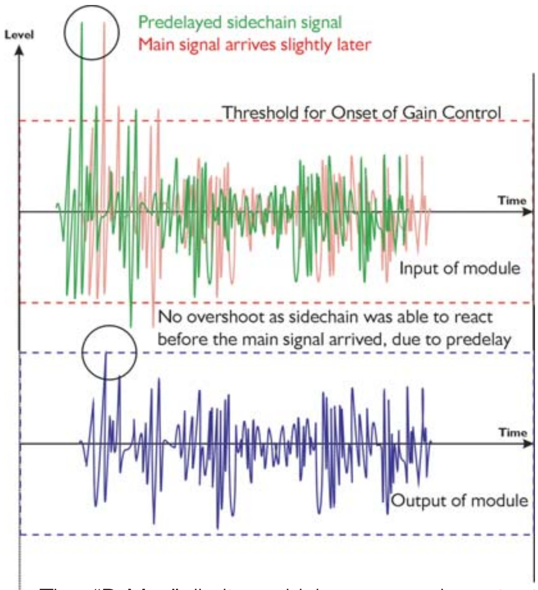
In this case, the delay between the signal actually going over the threshold, the engineer registering the situation, and then turning the signal down will be in the order of several hundred milliseconds at best. This will only be true if he is not distracted – in reality, it may be several seconds before 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 *DSP-4080*'s “D-Max” limiter predelays 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 predelayed sidechain is shown in green, with the main signal in red. As the main signal arrives slightly after the sidechain, the output from the 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 10dB 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 limiters built into the *DSP-4080* are 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-excursion.
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 manufacturer's 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.

Alert:
ALWAYS REFER TO D.A.S. FOR LIMITER SETTINGS.

Crossover Filter Slopes

It should also be noted that the turnover frequency displayed on the screen is the -3dB point for all types except Linkwitz-Riley where the -6dB point is shown. If the -6dB 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.

Please see page 12 for details of how to adjust the high and low pass crossover filter settings.

Time Alignment

A further advantage of the *DSP-4080* 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 precisely aligned rather than relying on the compromise 'phase adjust' approach. Delay time is adjustable in 0.3uS steps (0.1mm).
Please see page 12 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) @ 20Cº (68Fº)

To calculate time delay for a known distance, use:

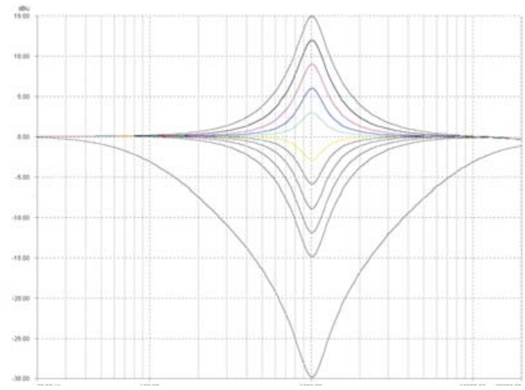
$$\text{Time delay} = \frac{\text{Distance (in metres)}}{20.06 \times \sqrt{273 + {}^{\circ}C}}$$

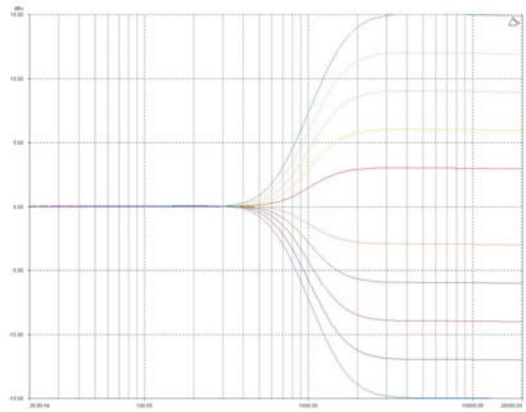
where ºC is the temperature in ºC.

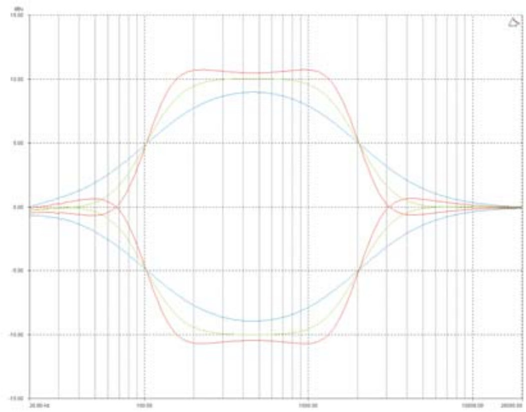
To simplify this equation at 20Cº.

Delay (in mS) = 0.955 x Distancia (in feet)
(Distance in metres x 2.192) or (Distance in feet x 0.955)

Note: [ºC] = 0.5555 x ([ºF]-32)







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 0dB. Bypassing the filter, then changing types using the ENTER key will automatically set the gain back to 0dB.

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 freqency 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 freqency 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.

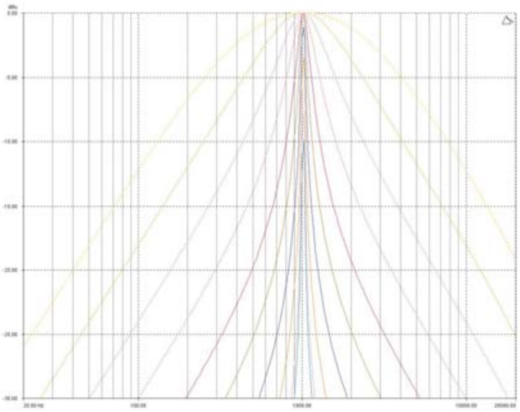
Remember that 'Q' is 1/Bandwidth, so the higher the 'Q', the lower the Bandwidth, and the smaller the range of frequencies affected.

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

Assymetrical responses may be achieved by adjusting the 'Q' of each filter independently.



<- *Bandpass filter*

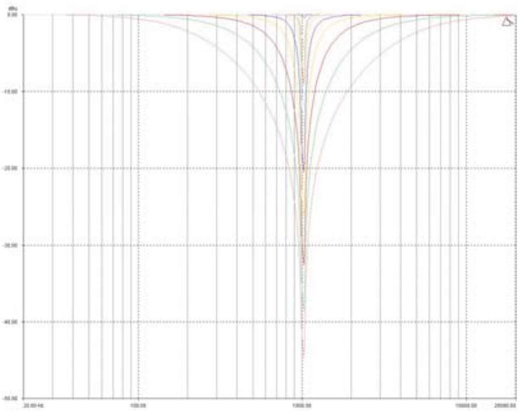
InA Input A BPS:1/
1k00Hz Q = 3.0 Bandpass

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

The bandpass filter has adjustable frequency and 'Q' (or Bandwidth) controls. These affect a range of frequencies symmetrically about the centre frequency as shown in the graph, gradually cutting the level, but providing no gain.

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

Note that the response is fundamentally NOT a flat-topped response (so it is not constructed from a high pass and low pass). See previous page for details of how to construct a flat-topped filter response.



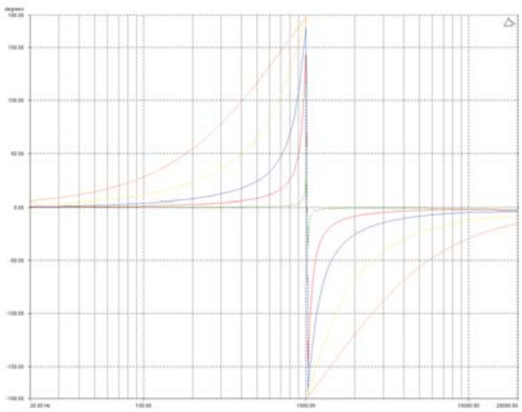
<- *Notch filter*

InA Input A NOT:1\/
1k00Hz Q = 0.75 Notch

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

The notch filter has adjustable frequency and 'Q' (or Bandwidth) controls. These affect a range of frequencies symmetrically about the centre freqency as shown in the graph.

Remember that 'Q' is 1/Bandwidth, so the higher the 'Q', the lower the Bandwidth, and the smaller the range of frequencies affected. The notch filter depth varies with bandwidth – the wider the filter, the lower the depth will be.



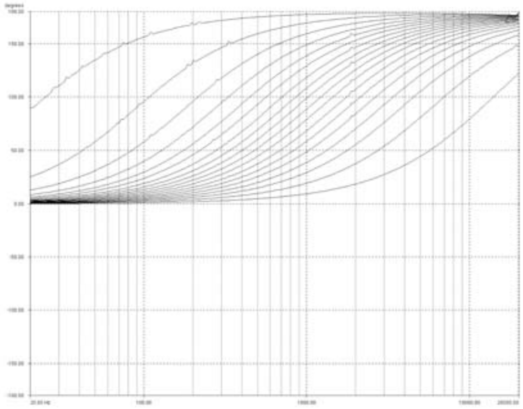
<- *All pass filter*

InA Input A APF:10
1k00Hz Q = 3.0 Allpass

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

The allpass filter has adjustable frequency and 'Q' (or Bandwidth) controls. These affect the frequency at which the phase effectively flips 180º, and the 'speed' at which this transition occurs.

The graph shows an allpass filter centred at 1kHz, with various 'Q' settings – the higher the 'Q' the faster the transition.



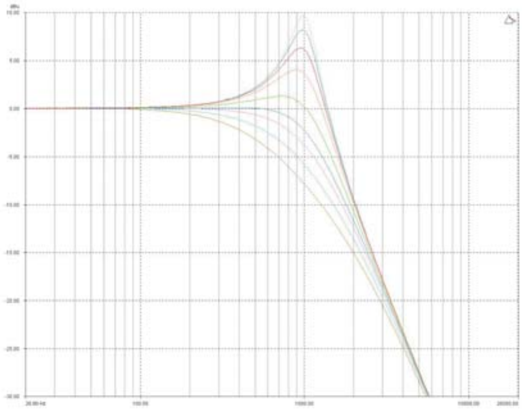
<- *Phase filter*

InA   Input A   PHF:1*0*
1k00Hz   150º   Phase

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

The phase filter has adjustable frequency, and phase shift controls. This introduces a phase shift that gradually changes from 180º above the centre frequency to the specificed value at the centre frequency, and tending towards 0º below the centre frequency.

This graph shows the phase shift relative to the input (ignoring processing delays), in 10 steps – the filter will actually provide higher resolution than this, operating in 2 steps. The filter is centred at 1kHz in this example.



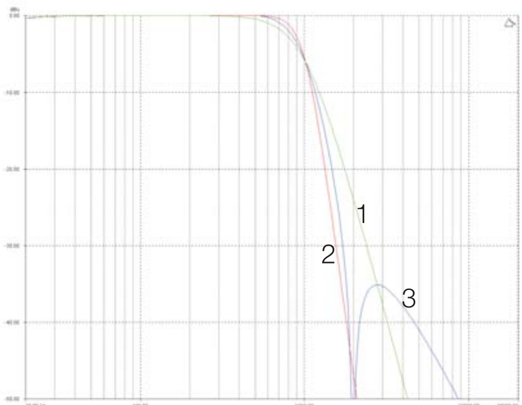
<- *Low/High pass variable 'Q' filter* (low pass shown)

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.



<- *Elliptical filters*

InA   Input A   HPF:1/~~
1k00Hz   HP Ellip

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

The elliptical filter has adjustable frequency only. This sets the 3dB point of the filter which features a steep roll-off (of approx 36dB/Octave down to one Octave above (or below in the case of a HPF) the turnover frequency.

However, the gain does increase again above this frequency to settle at 12dB down from 0dBr. This filter can be used in conjunction with the standard crossover filters to produce sharper roll-offs than would be otherwise achievable.

The use of this filter is explained in the following example.

The graph shows three different low pass filter shapes. Number 1 is a standard 24dB/Octave Linkwitz-Riley curve. Number 2 is a 48dB/Octave Linkwitz-Riley.

As can be seen, number 2 is significantly steeper in roll-off than number 1, as would be expected.

However, number 3 is a 12dB/Octave Butterworth filter with an Elliptical Low Pass filter following it. This produces a combined roll-off very close to that of the 48dB/Octave (with the side effect of the response rising again after the initial drop from the pass band).

Specifications

Inputs: 4 electronically balanced
Impedance: > 10k ohms.
CMRR : >65dB 50Hz - 10kHz.
MUTE : ON/OFF
Outputs: 8 electronically balanced
Source Imp: < 60 ohms
Min. Load: 600 ohms
Max. Level: +20dBm into 600 ohms
MUTE : ON/OFF
Frequency Resp.: ±½dB 20Hz-20kHz
-3dB @ 32kHz
Dyn. Range: >116dB 20Hz-20k unweighted
Distortion: < .02%@1kHz,+18dBm
Maximum Delay: 650 mS
Min Step Size: 0.3uS
Input Gain: +6dB to -40dB in 0.1dB steps
Output Gain: +15dB to -40dB in 0.1dB steps

Parametric Equalisation
8 per Input / 9 Sections per Output
Filter Gain: +15dB to -30dB in 0.1dB steps
Freq. Range: 19.7Hz - 32kHz, 1/36 octave steps
Filter Q / BW: 0.4 to 128 / 2.5 to 0.008
(Sections switched to shelving response)
Low frequency: 19.2Hz - 1kHz
High frequency: 1kHz - 32kHz
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:
1^{st} Order 6dB/Oct.
Bessel/Butterworth/Linkwitz-Riley 12-24-48dB/Oct.
Bessel/Butterworth 18dB/Oct.

Limiters
Program Limiter:
Threshold: +22dBu to -10dBu
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: 4 x 4 point, -24dB to digital clip
Output meter: 8 x 4 point, -24dB to +4dB into limit

Connectors
Inputs: 3 pin female XLR
Outputs: 3 pin male XLR
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: < 30 watts.
Weight: 3.3kg. Net (4.7kg. Shipping)
7.26lb. Net (10.34lb. Shipping)
Size: 44 x 482 x 300 (mm)
1.75"(1U) x 19" x 11.8"

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

Latency: 1.5mS (analogue in – analogue out @ 96kHz)

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, and GEQ 'Q' setting.

CROSSOVER Sub-menu: Set up or adjust crossover design, including routing and auto limiter setting. Also set up output ganging.

INTERFACE Sub-menu: Comms interface setup (RS232 and RS485), G.P.I. interface configuration, and wireless interface.

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: Switch outputs from analogue to digital and 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.

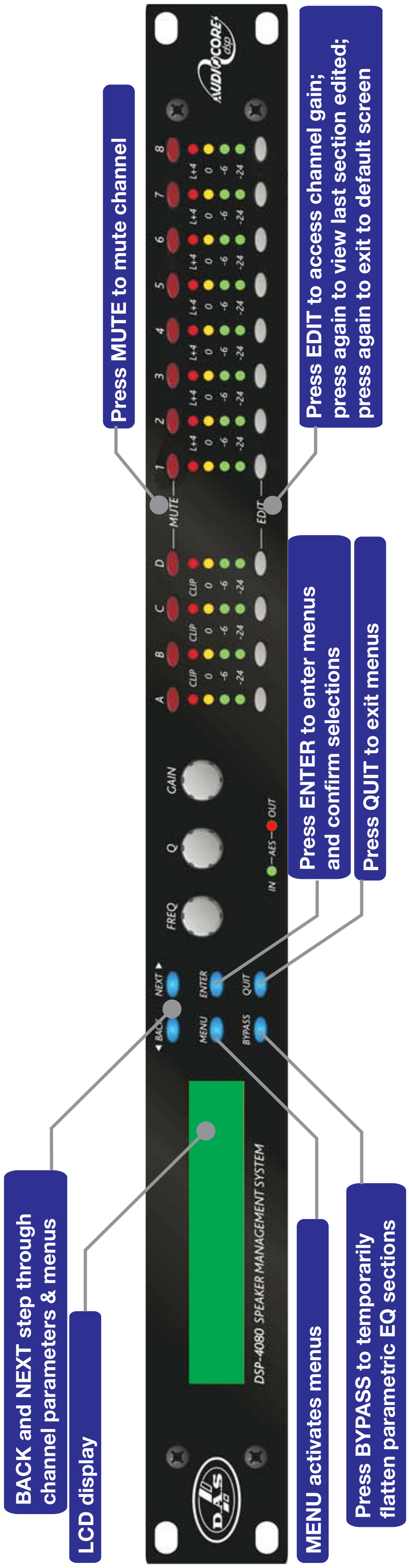
The high and low pass crossover filters are defined independently on each output channel.

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.



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.

MAIN MENU						
GLOBAL MEMORY	INPUT SECTION	CROSSOVER	INTERFACE	SYSTEM	SECURITY	AES/EBU
Recall a Memory MENU D	Input Ganging MENU 2	Design a Crossover MENU 1	External Interface MENU 2	System Status MENU 1	Unit Locking MENU 5	Output Selection MENU 6
Store a Memory MENU D	Input Reset MENU 4	Crossover Ganging MENU 3	Wiser 2400 Setup	LCD Contrast		AES Status Info. MENU 6
Erase a Memory			GPI Interface	LED Brightness		
				Temperature Alarm		
				Wake-up Time		
				Output Meters		
				Filter Q/Bandwidth MENU 4		
				Delay Time/Dist'nce MENU 5		
				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.

Menu system shortcuts

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. The entire list of features accessible in this way is given below.

MENU	A	-----
MENU	B	Store Input Memory
MENU	C	Store Crossover Memory
MENU	D	Store Global Memory

MENU	A	-----
MENU	B	Recall Input Memory
MENU	C	Recall Crossover Memory
MENU	D	Recall Global Memory

MENU	1	System Status
MENU	2	External Interface Set-up
MENU	3	-----
MENU	4	Filter Q/Bandwidth Display Readout
MENU	5	Delay Units Time/Distance Readout
MENU	6	AES Input Status
MENU	7	-----
MENU	8	-----

MENU	1	Design a Crossover
MENU	2	Input Ganging
MENU	3	Crossover (Output) Ganging
MENU	4	Input Reset
MENU	5	Unit Locking
MENU	6	AES Output Mode
MENU	7	-----
MENU	8	-----

Menus in detail

GLOBAL MEM.	
Recall a Memory MENU D	Recall Input and Crossover Memories or combinations of.
Store a Memory MENU D	Store Input and Crossover Memories or combinations of.
Erase a Memory	Erase Input and Crossover Memories or combinations of.

INPUT SECTION	
Input Ganging MENU 2	Gang (link) inputs together so their parameters track.
Input Reset	Start wizard to reset sections of input parameters.

CROSSOVER	
Design a Crossover MENU 1	Set up a new crossover from scratch. This selection starts a wizard to guide through the process. Also select this to alter the set-up of the current crossover.
Crossover Ganging MENU 3	Gang (link) outputs together so their parameters track.

INTERFACE	
External Interface MENU 2	Starts a wizard to configure the baud rate, ID and port selection of the remote interface.
Wiser 2400 Setup	Configures wireless interface (if connected).
GPI Interface	Configure the GPI inputs used for closed contact memory recall (hardware option).

SYSTEM	
System Status MENU 1	Displays a series of information screens including software version, temperature, hardware and firmware versions, date and time. Press **NEXT** to jump through info.
LCD Contrast	Adjust the viewing angle of the screen.
LED Brightness	Adjust the brightness of all the meters and button LEDs.
Temperature Alarm	Set the threshold for the unit to flash a warning temperature message on the screen.
Wake-up Time	Adjust the time before the audio fades in on start-up – can also be set to keep mutes on when powered up.
Output Meters Opt'n	Select the monitoring point for the meters – either pre or post mute (so meters can be set to work even when outputs muted)
Filter Q/Bandwidth MENU 4	Select the readout units for the 'Q' setting of parametric filters – 'Q' is 1/Bandwidth (in octaves) – small 'Q' values mean wide response variations.
Delay Time/Dist'nce MENU 5	Select the readout units for all delay values – either time, or distance in feet or metres.
Clip LED Hold Time	Select the time that the input CLIP LEDs stay illuminated for after an overload has passed.
Set Date & Time	Adjust the real time clock settings.

SECURITY	
Unit Locking MENU 5	Protect the unit against unauthorised access with a password - please see page 25 for more details.

AES/EBU	
Output Selection MENU 6	Switch the outputs of the unit to digital. (Inputs are selected via rear panel switch)
AES Status Info. MENU 6	Information screen showing details of the incoming AES streams (sample rate/lock).

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UM_DSP4080_01_EN

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