**User's Manual**

# DXi series

*DX-80I / DX-100I*

## CONTENTS

## Amplificadores profesionales / Professional power amplifiers

Conserve y lea estas instrucciones.
Respete y siga todas las advertencias.
El signo de exclamación en un triángulo equilátero pretende alertar al usuario de instrucciones operativas y de mantenimiento (reparación) en la literatura que acompaña al aparato.
ADVERTENCIA: Los aparatos de CLASE I se deben conectar a una toma de corriente eléctrica con conexión a tierra.

Keep these instructions. Read these instructions.
Heed all warnings. Follow all instructions.
The exclamation point within an equilateral triangle is intended to alert the user of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.
WARNING: Apparatus with CLASS I construction shall be connected to a MAINS socket outlet with a protective earthing connection.

El rayo con punta de flecha dentro de un triángulo equilátero pretende alertar al usuario de la presencia de voltajes peligrosos no aislados dentro de la envolvente del producto, que puede ser de magnitud suficiente para constituir un riesgo de descarga eléctrica para las personas.
ADVERTENCIA: Para evitar lesiones, este aparato debe estar firmemente sujeto al bastidor, de conformidad con las instrucciones de instalación.
El cableado exterior conectado a estos terminales requiere de su instalación por una persona instruida y el uso de cables flexibles preparados.

The lightning flash with arrowhead symbol within an equilateral triangle is intended to alert the user to the presence if uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons.
WARNING: To prevent injury, this apparatus must be securely attached to the rack in accordance with the installation instructions.
The connected outer wiring to these terminals requires of its installation by an instructed person and the use of a flexible the cable already prepared.

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.

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 the 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.

No instale el aparato cerca de fuentes de calor tales como radiadores, calefactores, estufas u otros aparatos que produzcan calor. No bloquee las aberturas de ventilación, e instalar de acuerdo con las instrucciones del fabricante. Los ventiladores de refrigeración succionan aire fresco del frontal y sale caliente por la parte trasera de la unidad a través de las rejillas de ventilación. La parte delantera y trasera del amplificador debe tener una exposición al aire libre (por ejemplo, en un rack las puertas delanteras y traseras abiertas), con cámara de aire de 2 cm a los lados y la parte superior. SI AL AIRE NO SE LE PERMITE ESCAPAR POR ATRÁS, SE SOBRECALENTARÁ LA UNIDAD. Tenga cuidado al montar otro equipo en el mismo rack. Diseñado para funcionar entre 15ºC y 35ºC, con el 75% de humedad relativa máxima.

Do not install near any heat sources such as radiators, heat registers, stoves or other apparatus that produce heat.
Do not block any ventilation openings, install in accordance with the manufacturer's instructions.
The cooling fans suck cool air in through the front and blow hot air out at the rear of the unit through the ventilating grills. The front and rear of the amplifier should have free exposure to the air (i.e. in a rack leave the front and rear doors off), with 2cm air gap at the sides and top. IF AIR IS NOT ALLOWED TO ESCAPE FROM THE REAR, OVER-HEATING WILL OCCUR. Take care when mounting other equipment in the same rack.
Working temperature ranges from 15ºC to 35ºC with a relative humidity of 75%.

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).
Proteja el cable de alimentación de ser pisado o aplastado, especialmente en los enchufes, receptáculos y en el punto en el que salen del aparato. Confíe las reparaciones a personal cualificado. Se requiere servicio cuando el aparato ha sido dañado de alguna manera, como por ejemplo si el cable de alimentación o el enchufe está dañado, se ha derramado líquido o han caído objetos dentro del aparato, el aparato ha sido expuesto a la lluvia o la humedad, no funciona con normalidad o se ha caído. El interruptor de alimentación deberá permanecer fácilmente accesible. Para desconectar completamente este aparato de la red eléctrica, desconecte el cable de alimentación del interruptor de corriente principal. Esta unidad está equipada con un cable de alimentación de 3 hilos. Por razones de seguridad, LA CONEXIÓN A TIERRA NO DEBE DESCONECTARSE EN NINGUNA CIRCUNSTANCIA.

Cuando la unidad esté montada en un rack y permanentemente conectada a red debe ser instalada con un conector de fácil acceso o con un dispositivo de desconexión omnipolar con al menos 3 mm de distancia entre sus contactos. El interruptor de los amplificadores sólo afecta a uno de los polos de la red eléctrica, por tanto, en las unidades con un cable desmontable el dispositivo de desconexión de red (por ejemplo, el enchufe o la conexión a red), debe ser fácilmente accesible para poder estar completamente desconectado de la red. Sin embargo, en unidades con cable fijo deberá usar un dispositivo de desconexión externo (antes descrito). La instalación deberá seguir todas las normas de instalación vigentes.
Utilice sólo accesorios (por ejemplo, soportes o racks) recomendados por el fabricante. Cuando se utiliza un rack o un carro de transporte, tenga cuidado al mover la combinación carro / aparato para evitar lesiones causadas por un vuelco.

Unplug this apparatus during ligtning 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).

Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles and the point where they exit from the apparatus.
Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as if the 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.
The mains circuit breaker shall remain readily accessible.
To completely disconnect this equipment from the AC mains, disconnect the power cord from the mains circuit breaker.
This unit is fitted with a 3-wire power cord. For safety reasons, THE EARTH LEAD SHOULD NOT BE DISCONNECTED IN ANY CIRCUMSTANCE.

Where the amplifier is mounted in a rack and permanently connected to the mains, then the rack should be installed with a readily accessible connector or an ALL POLE circuit breaker with 3mm breaking distances.
The mains switch on the amplifiers only switches one pole of the mains supply, therefore for units with a detachable cord to be fully disconnected from the mains, the mains disconnect device (ie mains plug or mains coupler) should remain readily operable. For units with a fixed mains lead the external all pole circuit breaker with 3mm breaking distances is the disconnect device and therefore the installation of the amplifier shall be carried out in accordance with all the applicable installation rules.
Only use attachments/accessories specified by the manufacturer. Use only with the cart, tripod, bracket or table specified by the manufacturer, or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from a tip over.

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.

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.

## 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 our 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

**DAS Audio Group, S.L.**
C/ Islas Baleares, 24 - 46988 - Pol. Fuente del Jarro - Valencia. España (Spain).

Declara que los amplificadores de la serie *DXi*
Declares that *DXi* amplifier series

cumplen con los objetivos esenciales de las Directivas:
abide by essential objectives relating Directives:

- de Baja Tensión (Low Voltage Directive)                     2014/35/UE

- de Compatibilidad Electromagnética (EMC)                    2014/30/UE

- RoHS                                                         2011/65/UE

- RAEE (WEEE)                                                  2012/19/UE


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

- EN 60065:2014.- Audio, video and similar electronic apparatus. Safety requirements.

- EN 55032:2012.- Electromagnetic compatibility of multimedia equipment. Emission requirements.

- EN 55103-2:2009.- Electromagnetic compatibility. Product family standard for audio, video, audio-visual and entertainment lighting control apparatus for professional use. Part 2:Immunity.

- EN 50581:2012.- Technical documentation for the assessment of electrical and electronic products with respect to the restriction of hazardous substances.

## INTRODUCTION

Thank you for choosing a DXi series amplifier for your application.

Please spend a little time reading through this manual, so that you obtain the best possible performance from the unit and become familiar with its operating requirements.

All DAS Audio Group products are carefully designed and engineered for cutting-edge performance and world-class reliability. If you would like further information about this or any other DAS Audio product, please contact us.

We wish you many years of service from this amplifier and look forward to hearing from you in the near future.

The DXi amplifier series has been designed to combine incredible audio power and performance with ultra-flexible connectivity for both remote control and audio. Exemplary audio processing is assured through the use of a DSP platform, and power amplifier capabilities are taken care of with high efficiency output stages and a generous power supply.

All units named with an "i" include a DSP in the amplifier. *DX-100I* and *DX-80I* include a 4 in 8 out DSP platform.

Accepting analogue, AES3 digital and optional Dante networked audio, the DXi amplifiers can connect to any source and make it available over the network, as well as processing four additional channels with local analogue outputs to connect to DX amplifiers. These auxiliary processed outputs can also be fed onto the network, and used by DX amplifiers that have Dante cards fitted, or by any other Dante enabled device.

Connectivity for remote control is covered by USB, Ethernet and RS485, with relay of either USB or Ethernet control via RS485 to work with legacy products. Configuration of the amplifier's processing is through the industry standard DASnet application, with tuning and monitoring of a network of DXi amplifiers.

With a range of power levels available in the DX Series, the amplifiers can be networked to a single DXi model, creating a powerful, efficient system that's easy to expand and adapt for use in live, install and everything in between.

### INSTALLATION: ELECTRICAL CONSIDERATIONS

The amplifier has been manufactured to comply with your local power supply requirements, but before connecting the unit to the supply, ensure that the voltage (printed on the rear panel) is correct.

The amplifier is fitted with either a 100/120V or 220/240V tapped transformer according to customer requirements.

Make sure power outlets conform to the power requirements listed on the back of the unit. Damage caused by connecting to improper AC voltage is not covered by the warranty.

### Safety Warning

Apparatus with CLASS I construction shall be connected to a MAINS socket outlet with a protective earthing connection.

Where a MAINS plug or appliance coupler is used as the disconnect device, it should remain readily operable.

Where the amplifier is mounted in a rack and permanently connected to the mains, then the rack should be installed with a readily accessible connector or an ALL POLE circuit breaker with 3mm breaking distances.

This unit is fitted with a 3-wire power connector. For safety reasons, THE EARTH LEAD SHOULD NOT BE DISCONNECTED IN ANY CIRCUMSTANCE. If ground loops are encountered consult the section on Line Inputs and Outputs on page 11.

WHERE A FIXED MAINS LEAD IS FITTED, THE WIRING COLOURS ARE:

|  | 230V AREAS: | 120V AREAS: |
|---|---|---|
| EARTH = | GREEN AND YELLOW | GREEN |
| NEUTRAL = | BLUE | WHITE |
| LIVE = | BROWN | BLACK |

TO PREVENT THE LIKELIHOOD OF SHOCK OR FIRE HAZARD, DO NOT EXPOSE THE UNIT TO RAIN OR MOISTURE. DO NOT PLACE OBJECTS CONTAINING LIQUID ON TOP OF THE APPARATUS.

TO AVOID ELECTRICAL SHOCK DO NOT REMOVE COVERS. REFER ALL SERVICING TO QUALIFIED PERSONNEL.

DO NOT USE THE UNIT OF THE ELECTRICAL POWER CORD IS FRAYED OR BROKEN. The power supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords and plugs and the point where they exit from the appliance.

ALWAYS OPERATE THE UNIT WITH THE AC GROUND WIRE CONNECTED TO THE ELECTRICAL SYSTEM GROUND. Precautions should be taken so that the means of grounding of a piece of equipment is not defeated.

DO NOT REMOVE THE LID. Removing the lid will expose you to potentially dangerous voltages. There are no user serviceable parts inside.

ESD strikes to the unit's front panel that are in excess of 4000 volts may cause disturbance to the status LEDs on the unit. This will not affect audio performance and will be corrected on the next power up cycle.

### INSTALLATION: MECHANICAL CONSIDERATIONS

To ensure that this equipment performs to specification, it should be mounted in a suitable rack or enclosure as described below. Like all high power amplifiers, it should be kept away from other equipment which is sensitive to magnetic fields. Also, this amplifier may suffer a substantial reduction in performance if it is subjected to, or mounted close to equipment which radiates high RF fields.

Warning: To prevent injury, this apparatus must be securely attached to the rack in accordance with the installation instructions.

When mounting the amplifier in a rack or enclosure:

Be aware that ...

THE FRONT PANEL IS NOT CAPABLE OF SUPPORTING THE UNIT ON ITS OWN.

Make sure that the rear of the unit is adequately supported. The brackets which are supplied fit standard 19 inch (483mm) rack mounting systems.

ENSURE THERE IS ADEQUATE VENTILATION.

The cooling fans suck cool air in through the front and blow hot air out at the rear of the unit through the ventilating grills. The front and rear of the amplifier should have free exposure to the air (i.e. in a rack leave the front & rear doors off), with 2cm air gap at the sides.

IF AIR IS NOT ALLOWED TO ESCAPE FROM THE REAR, OVER-HEATING WILL OCCUR.

Take care when mounting other equipment in the same rack.

Make sure that the rack unit has a separate earth connection (technical earth).

Please also see the notes regarding maintenance on page 10.

### RF EMISSIONS

The high frequency resonant converters in the *DXi series* amplifiers have been designed to have very low radio frequency (RF) emissions; however even these low level emissions can cause interference with other equipment.

In order for this to be minimised, the amplifier should be mounted in a metal rack enclosure, which should have a separate (technical) Earth. Alternatively a separate earth should be attached to the amplifier at the rear rack mounting bracket.

### DYNAMIC AMPLIFIER PERFORMANCE MEASUREMENTS

The **DXi series** is the very latest example of a 'dynamic amplifier'. This new 'breed' of power amplifiers provide very high peak power levels in a much smaller, and lighter, package than previously possible with conventional amplifiers.

They are designed specifically for today's high power audio installations, which use multiple speakers with electronic crossovers or speaker controllers. These systems can handle very high transient signals that far exceed their RMS power rating. The *DXi series* amplifiers have been designed to match this requirement and can deliver huge levels of power for short durations.

In order to protect themselves and the loudspeakers that they are driving, continuous signals such as sine waves, are automatically detected and reduced (ramped down) to a safe level.

When trying to measure the power output however, continuous signals will give totally incorrect results. A dynamic signal, such as a tone burst, should be used and the levels measured by monitoring the waveform on an oscilloscope. The power envelope can then be accurately measured.

Our power output figures are measured using signals with known Crest Factors and are quoted at the rear of this manual on page 43 and on our website.

Please refer to the technical area of our website for further information.

### INITIAL SET-UP AND SWITCHING ON

Please read all documentation before operating your equipment and retain all documentation for future reference.

Do not spill water or other liquids into or on the unit and do not operate the unit while standing in liquid.

Do not block fan intake or rear ventilation outlets or operate the unit in an environment that could impede the free flow of air around the unit.

If the unit is used in an extremely dusty or smoky environment, it should be cleaned of any collected debris at regular intervals. Please also see the notes regarding maintenance on page 10.

It is important that the power output of your amplifier is matched to the power handling capacity of your loudspeaker. If not, damage to the loudspeaker could occur.

Switching On…

At 'switch-on' the protection circuit will initially activate whilst the circuits stabilise, indicated by the red A/P LED illuminating, in addition to various other LEDs. After a few seconds the red A/P LED will extinguish indicating a satisfactory working condition.Other LEDs may remain illuminated depending upon rear panel switch settings and input connections. If the A/P LED does not extinguish after 5 seconds the unit may be faulty or some external connections may be incorrect or inappropriate. If this occurs you should power down the unit and remove all external connections (except for the mains power supply) and repeat the power up sequence. If the problem persists please contact us.
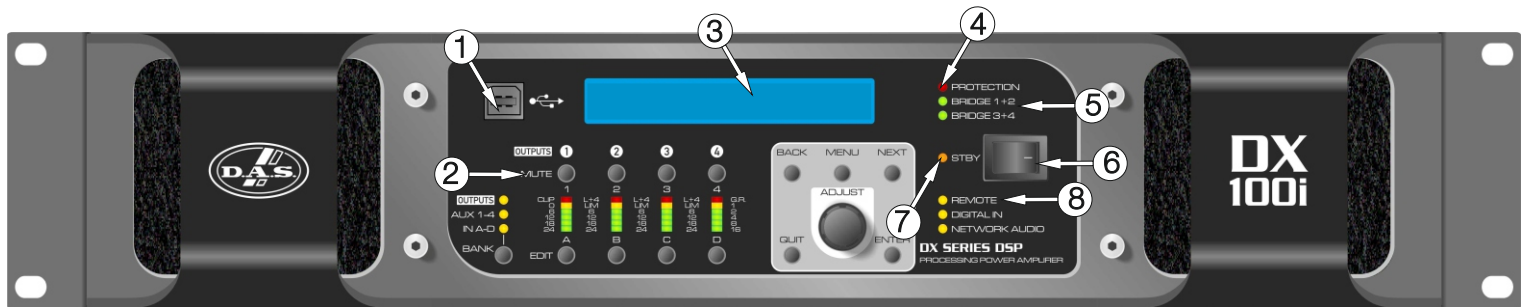
Note: The front panel is common for all models.

1: USB Type "B": Connect to a computer for remote control – a driver will be required. Please see the section on remote control starting on page 36.

2: MUTE keys: This will mute the respective input or output channel according to the current setting of the bank control selection (#14).

3: 2 x 24 LCD: By default this will display preset names and is used to show all parameter information. The LCD contrast and backlight brightness can be adjusted in the System Sub-Menu - see page 32 for details. If the LCD backlight is flashing, it is being remotely identified by software.

4: PROTECTION LED: If a condition exists, either internally or externally, that could cause damage to either the amplifier or the speakers, the protection circuit will disengage the outputs and this LED will illuminate/flash.

Typical conditions that could cause the protection to be triggered include very high frequency or subsonic input signals, DC in the inputs, short-circuited outputs, or internal high temperatures.

The protection circuit can affect all channels or a 'channel pair' depending on the type of fault. This is indicated by the combination of Protection LED illuminating and a corresponding message on the LCD. In this way, it is possible for two channels (a channel pair) to remain functioning even though a fault has caused the other channel pair to mute. A channel pair would be 1+2 or 3+4.

Temperature related faults will reset automatically if the unit has cooled sufficiently. Output short circuits will require manual reset after clearing the fault (switching off at the mains switch and then on again after a few seconds). Short circuits on either channel of a channel pair will only affect that channel pair.

5: BRIDGE pair LEDs: The channel pair LED will illuminate if these channels have been switched into bridged (mono) mode. See page 11 for details of how to connect your speaker to a bridged channel pair, and page 12 for how to enable bridge mode.

6: Power Switch: This double pole switch turns the amplifier fully off and isolates it from the mains supply.

7: STBY LED: The *DXi* amplifiers can be powered down leaving just the DSP active. Apart from the obvious power savings to be made, there may be circumstances when only the DSP is required, such as utilizing it for braking analogue audio channels onto a Dante network. This LED illuminates when the power amplifier sections are turned OFF.
HINT: The amplifier can be brought out of standby by holding ENTER and QUIT at power-up.

8: REMOTE LED: This will flash when the unit is being addressed under remote control.
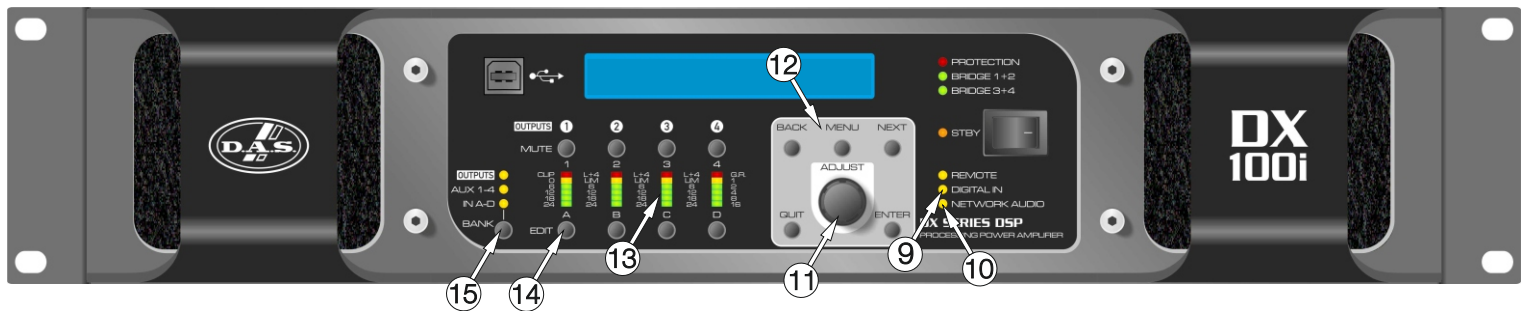
9: DIGITAL IN LED: This illuminates if either AES input is switched in (replacing the respective analogue input channel) and will flash if there is a loss of lock on either input. Note that this does not mean AES is actually routed to the amplifier's DSP, only that a rear panel socket is set to receive an AES stereo signal.

10: NETWORK AUDIO LED: This will illuminate when an audio network connection is correctly made to the amplifier. Note that this does not mean that any network audio channels are actually routed to the amplifier's DSP, only that the connection is available.

11/12: Rotary encoder and navigation keys: MENU is always the way under the bonnet of the amplifier, and then BACK and NEXT to choose the Sub-Menu and ENTER to select the choice. These controls are also used when editing DSP audio parameters, accessed via the EDIT keys.

13: Signal meters: Depending on the current bank selected (#14), these will be showing input levels pre-DSP, output levels for the power amp channels, or output levels of the auxiliary channels.

14: EDIT keys: First press on an EDIT key will show the gain for the selected channel within its respective BANK. Second press will display the last edited parameter - see pages 17 to 18 for more info on full real-time editing of input and output parameters.

15: BANK Select key and LEDs: This key will cycle through three banks: inputs A-D, power amp outputs 1-4 or auxiliary outputs 5-8, for both meter displays, mute functions and during editing. Bank select will default to showing power amp outputs on meters and mutes after 20 seconds of inactivity.
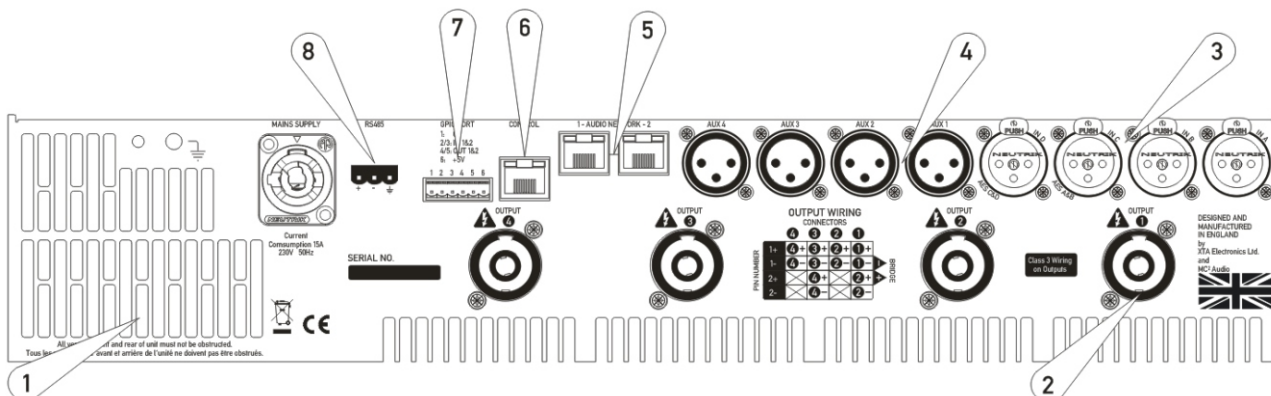
Note: Designed and manufactured in England by XTA Electronic Ltd. and MC² Audio for DAS Audio Group, S.L. The rear panel is common for all models.

1: Fan outlet: The variable speed fans suck air in through the front vents and out through the back of the amplifier. Please see maintenance for recommendations on how to clean this and the front foam sections.

2: Channel A output Speakon socket: Normal output is on pins 1+ hot, 1- cold. Channel B's output is also wired to this socket to enable a single NL4 to provide both channels and to facilitate easier wiring in bridged mode. Channel B is wired pins 2+ hot, 2- cold. Similarly channel C's output Speakon socket carries Channel D's output. Check the table on the rear panel for details.

3: Input XLR sockets: Connect signal inputs to these sockets, wired pin 2 hot, 3 cold, 1 ground. For sensitivity and impedance of these inputs, please see the specifications on page 43. Inputs C & D may also be switched to AES digital inputs, each carrying a stereo AES stream – channels A&B on socket C, channels C & D on socket D. This arrangement allows an analogue stereo source to remain connected to sockets A & B for fallback purposes. To select AES inputs please see the section on page 24.

4: Auxiliary output XLR sockets: These carry the four additional channels of separate DSP processing – they are NOT just link outputs or a copy of the power amplifier channel's processed audio – they are fully independent.

5: Audio network connections: Four additional inputs are available via the Dante network card included. This will also add four network audio outputs, which can be chosen (in banks of four) from a variety of processing points within the amplifier's DSP structure. For more on this feature see the block diagram on page 13 and set-up information from page 27.

6: Ethernet control port: Your amplifier may be remotely controlled by connecting it to a computer via this standard Ethernet port connection.  Please see the section starting on page 35 for how to use this feature. HINT: You can quickly check the amplifier's IP address by pressing ENTER + NEXT.

7: GPIO Port: Your amplifier has a pair of general purpose logic level input and outputs that can be configured to recall memories, put the amplifier in standby, mute and control levels, and also provide feedback about status. Please see the section on page 36 for more details.

8: RS485 Port: Your amplifier may also be controlled via an RS485 connection, and this port may also be used to relay control data from the Ethernet port or the front panel USB port to connect to further devices. Please see the remote control section on page 36 for more information.

### MAINTENANCE

These maintenance instructions are for use by qualified personnel only. Before any routine maintenance, please ensure that your amplifier is disconnected from the mains supply!

The filter behind the air intake apertures on the front of your amplifier should be cleaned or replaced periodically, e.g. 12-24 months. (Filters in amplifiers located in more 'dirty' atmospheres may require more frequent maintenance).

The filter should be 'dry' cleaned, using a vacuum cleaner preferably. Running the unit without a filter is not recommended unless it is within a 'clean room'. Replacement filter material is available.

If the fan vents on the rear of the amplifier develop a build-up of dust/debris on the finger guards, they can be cleaned with a dry paintbrush and a vacuum cleaner.

The casework of the amplifier may be cleaned with a lightly dampened cloth – do not use any solvents as they will damage the paint finish and could remove printing.

If you have any doubts about carrying out maintenance, please refer to a service engineer or contact your local dealer.

### LINE INPUTS AND OUTPUTS

The inputs are made via 3-pin XLR connectors, which are electronically balanced and should be connected via a high grade twin core screened cable, as follows:

**VLR MALE**

Pin 1: Screen (see note below)
Pin 2: Hot  (signal +)
Pin 3: Cold (signal -)

The amplifier is designed to operate with fully balanced equipment and ground loops or loss of performance may be experienced if connected to unbalanced sources. If it is unavoidable however, the following wiring should be used. The cable should still be twin core plus screen.

Pin 1: Screen - connected to the chassis of the unbalanced equipment - or left disconnected at the unbalanced end.
Pin 2: Hot (signal +)
Pin 3: Cold (ground 0V)

NOTE: This amplifier is wired to the latest industry recommendations. Pin 1 is connected directly to the chassis/mains earth. If ground loops (mains hum) are encountered remove the screen connection from the other end of the cable and leave it open circuit. If problems persist, consult your dealer/supplier.
DO NOT TAMPER WITH OR ALTER ANY GROUND (EARTH) CONNECTIONSINSIDE THE AMPLIFIER.

For bridged operation input should be made to channel A (or C) only and the channels set for bridged mode for the appropriate pair of channels.

Outputs are also made via 3-pin XLR connectors wired as follows:

**VLR FEMALE**

Pin 1: Screen (see note above)
Pin 2: Hot  (signal +)
Pin 3: Cold (signal -)

Note that the rear panel outputs are electronically balanced and so are not galvanically (electrically) isolated. Front panel outputs are transformer balanced and so are isolated.

**QPEAKON NL2 & NL4**
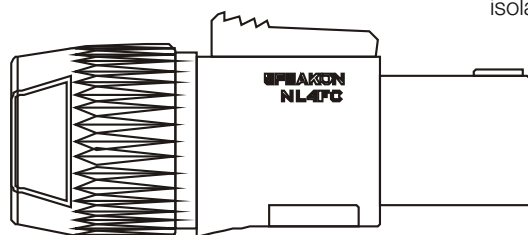
### SPEAKER OUTPUTS

The speaker outputs are via Neutrik Speakon connectors. 2 pole (NL2FC) or 4 pole (NL4FC) connectors can be used.

Pin 1+:  Hot
Pin 1-:  Cold

Additionally, Channel 1 Speakon connector carries Channel 2 output on Pins +2 & -2 to allow easy bi-amping or bridged operation using a single NL4 connector.  Similarly, Channel 3's Speakon connector also carries Channel 4 output.

Output Connector 1

Pin 2+:  Hot Ch. 2
Pin 2-:  Cold Ch. 2

Output Connector 3

Pin 2+:  Hot Ch. 4
Pin 2-:  Cold Ch. 4

For bi-amped operation, connect as above.

There must be no shared connections between channels.

Negative output terminals must not be joined together as they are not both at ground potential. Connecting them together will damage the amplifier and void the warranty!

As the currents involved are very high, and to ensure best performance, the speaker cables should be kept as short as possible and conform to the following minimum requirements:

DX-80I, 14A into 4 Ohm speaker loads
DX-100I, 20A into 4 Ohm speaker loads

When operating the amplifier into loads of less than 4 Ohms, be aware that the current capacity of the speaker cables will need to be increased above the values quoted here.

Do not connect the inputs/outputs to any other voltage source such as a battery, mains source or power supply, regardless of whether the amplifier is turned on or off.

Do not run the output of any amplifier channel back into another channel's input and do not parallel or series-connect an amplifier output with any other amplifier output.

### BRIDGED (MONO) OPERATION

Pairs of channels may be independently bridged – channel pair 1+2, and/or channel pair 3+4.

The method is the same for both channel pairs:

Select the required bridged mode through the front panel MENU system (see page 29 for more information).

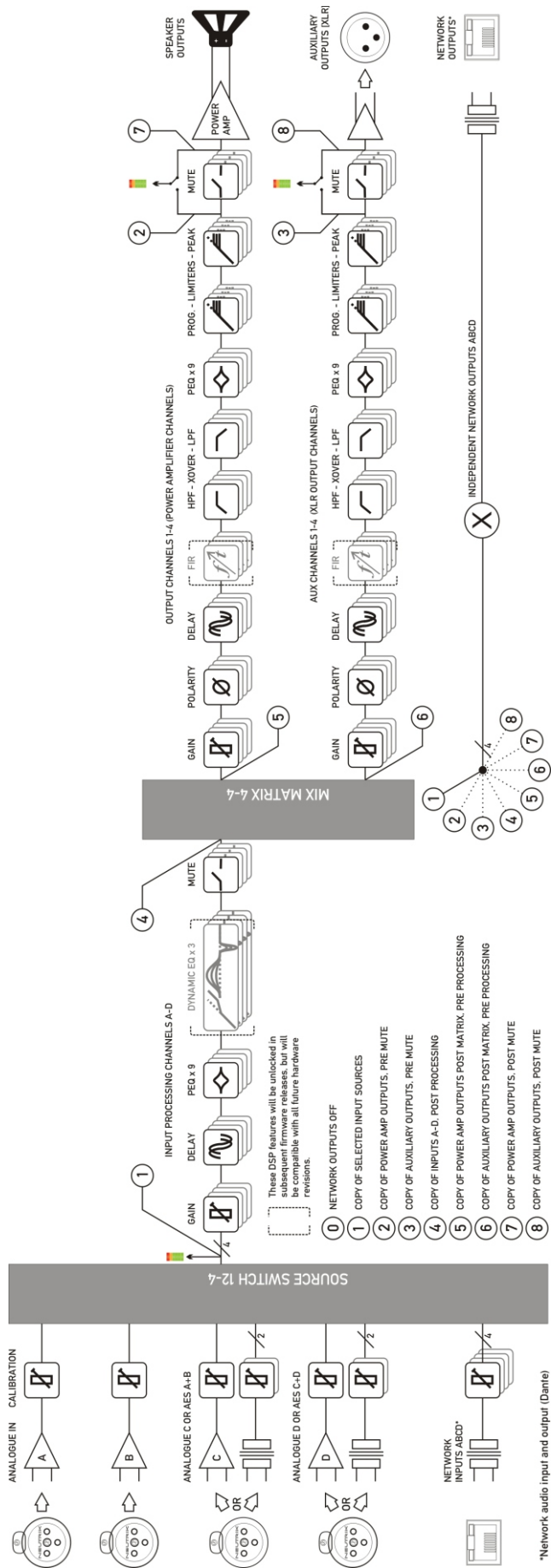Use Channel 1 or 3's Output Speakon connector and connect as follows:

Pin 2+:  Hot
Pin 1-:  Cold

When operating in bridged mode, the minimum impedances are doubled.

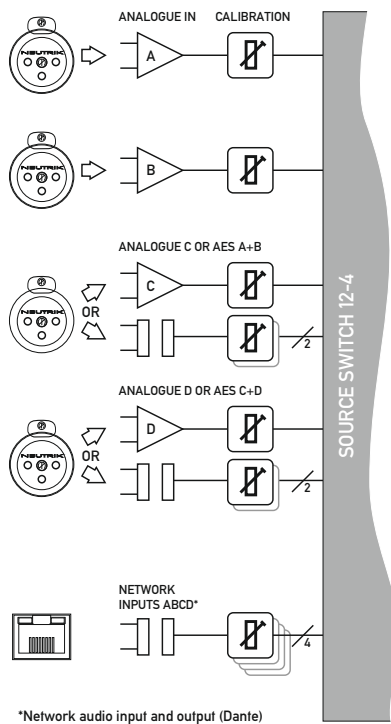**The minimum load in bridged mode is 4 ohms.**

ANALOGUE IN    CALIBRATION

A

B

ANALOGUE C OR AES A+B

C
OR

ANALOGUE D OR AES C+D

D
OR

NETWORK
INPUTS ABCD*

SOURCE SWITCH 12-4

*Network audio input and output (Dante)

## Source Choices and Selection

Your amplifier can source audio from analogue, AES digital, or network locations (DANTE).

Analogue and AES digital audio are standard, and arranged so that AES digital audio can be chosen in pairs of channels to replace either analogue inputs A&B together or analogue inputs C&D together, or all four channels.

Physical input of AES for channels A&B is swapped to the input XLR for channel C and the AES input for channels C&D is on the XLR socket for channel D.

In this way, a pair of analogue inputs can remain connected to channels A&B and a digital stream of the same audio can be connected to input C, with fallback from one to the other possible without repatching.

With the DANTE card, four additional digital sources will be available.
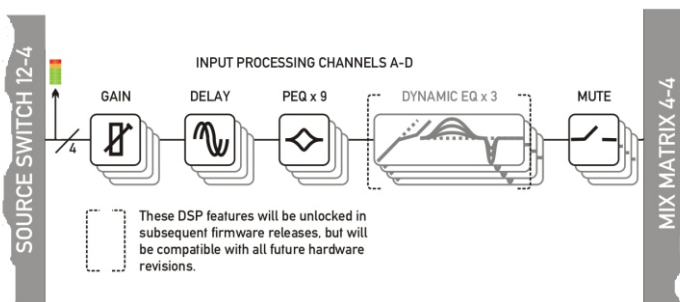
In total, 8 sources can be made available at any one time – four analogue and four digital. This affects the choices that can be made for routing to the inputs of the DSP channels.

The rules are as follows:

Analogue A and B is available at all times;
If AES A&B is selected, Analogue C cannot be used (shares XLR C);
If AES C&D is selected, Analogue D cannot be used (shares XLR D);
If AES A&B is chosen, Dante A&B cannot be used (shares SRC One);
If AES C&D is chosen, Dante C&D cannot be used (shares SRC Two);

Source selection is therefore affected by the selection of AES inputs, which then controls the choice of input source selections on offer to any input processing channel.

SOURCE SWITCH 12-4

INPUT PROCESSING CHANNELS A-D

GAIN    DELAY    PEQ x 9    DYNAMIC EQ x 3    MUTE

These DSP features will be unlocked in subsequent firmware releases, but will be compatible with all future hardware revisions.

MIX MATRIX 4-4

## Input Processing Channels

Your amplifier has four input channels of processing that can be fed from a variety of sources.

These four channels in turn, feed a mix matrix for routing to the power amplifier processing channels and the auxiliary output processing channels.

Each input processing channel consists of the following sections:

Input gain control
Input delay time
Input parametric EQ bands 1 through 9

The dynamic EQ sections will be added in a future firmware update. For more information on adjusting input processing parameters, please see the section beginning on page 17.
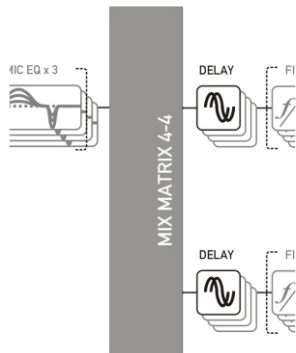
## Mix Matrix Section

The mix matrix allows four independent mixes to be set up for the power amplifier channels and the auxiliary output channels.

These can be either "Boolean" in design (so just A+B+C etc.) or a full mix matrix mode can be used to allow four continuously variable "sends" from the four input processing channels to be combined.

There are also a variety of templates to act as starting points for crossover duties, where useful crossover (high and low pass) frequencies are automatically assigned to groups of outputs, dependant on the format used. For more information on adjusting the mix matrix, please see the output editing section on page 22. Note that the mix matrix will only be available if it has been selected as part of the output/auxiliary configurations – see page 27 to change this.

## Power Amplifier and Auxiliary Output Processing Channels
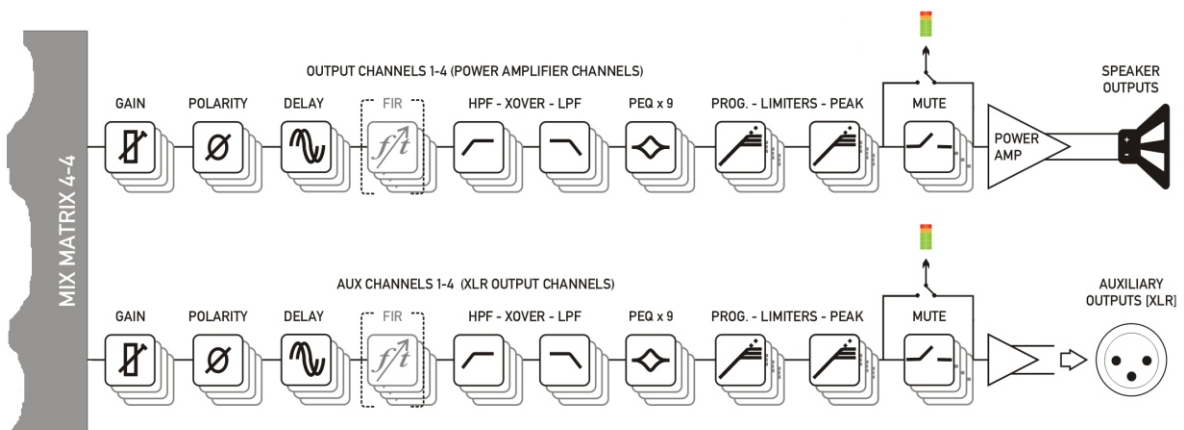


There are eight identical channels of output processing in your amplifier: four dedicated to the power amplifier channels, and a further four that are connected to the auxiliary XLR outputs on the amplifier, providing analogue feeds to other equipment.

Each output processing channel consists of the following sections:

Output delay time
High and low pass crossover filters (up to 48dB/Oct.)
Output parametric EQ bands 1 through 9
Output gain control
Program (RMS) limiter
Peak limiter

FIR filtering capabilities will be added in a future firmware update. For more information on adjusting output processing parameters, please see the section beginning on page 18.

**Independent Network Audio Outputs**
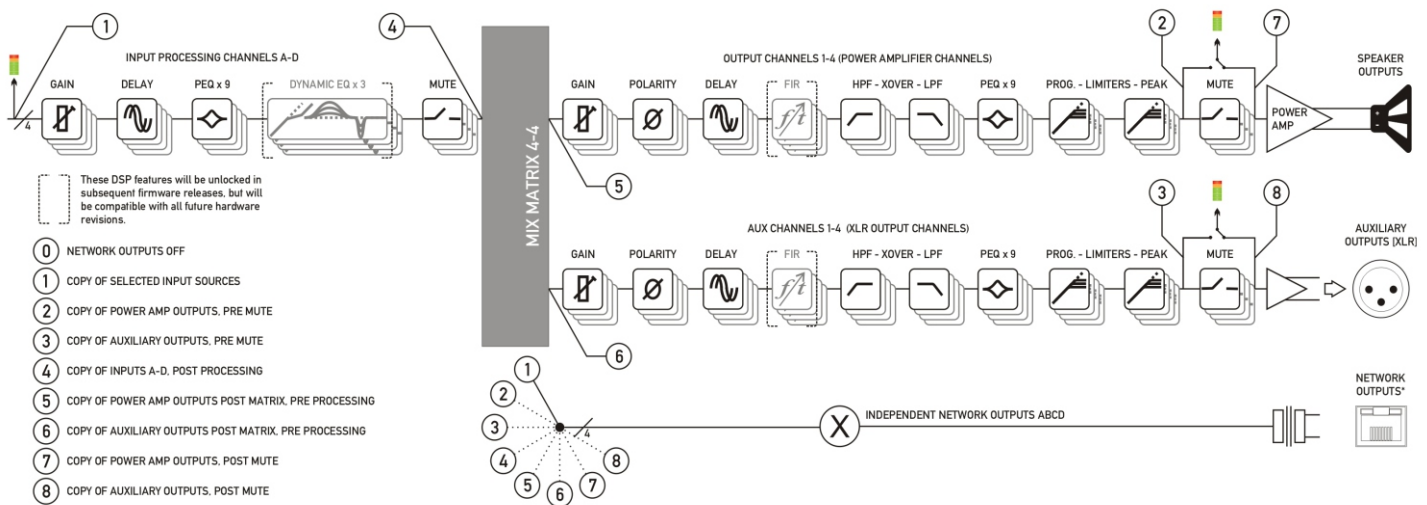


All DXi models have a network audio card fitted (Dante), this means being able to route four channels of audio from the network and, as well, being able to route four channels of audio back onto the network.

This feature is primarily intended for use with the DX series models, which may also be fitted with a network audio interface (optional), allowing them to utilise the additional processing channels in your DXi models via connection with a single Ethernet cable.

Working like this, the network audio outputs would be routed directly from end of the auxiliary output processing, so that they are just a copy of the signals available at the Aux XLR outputs 1-4 (choice #1 in the above diagram).

However, there are other circumstances where it might be more useful to route these outputs from elsewhere within the DSP processing. For example, if the network outputs are being used to also "break-in" analogue or AES channels, for use with other amplifiers, it might be more applicable to route the audio from the outputs of input processing channels (choice #5).

If it's required for the network outputs to break in channels that are not being used by the local processing, this can also be achieved by using the outputs of the mix matrix which feed the auxiliary output processing (choice #3).

Alternately, the mixes being used by the power amp processing can be selected (choice #4) and finally, the pre-mute final output of the auxiliary channels' processing may be selected (choice 2) to prevent muting the local aux outputs from muting the network feeds.

**Note that these routing selections work on a "x4" basis – all four network output feeds are selected from one of the above choices as a group – they cannot be individually assigned to different points.**

For more information on selecting the network audio output routing, please see the section on page 26.

## DIRECTLY EDITING AUDIO PARAMETERS

**Overview**

Editing all audio parameters is available from the front panel of your amplifier using a combination of the select BANK key, reassignable EDIT keys, and the BACK/NEXT/ENTER navigation controls.

First, make sure the correct bank of channels is selected – either inputs A-D, amplifier outputs 1-4 or auxiliary outputs 5-8 by pressing the BANK key.

Next, press the EDIT key that corresponds to the channel to be adjusted. The EDIT key will illuminate, and the gain will always be displayed as the first parameter.

To choose another parameter, use BACK and NEXT to scroll through the available choices.

If there are multiple parameters grouped on a single screen (such as parametric EQ frequency, 'Q' and gain) move between these using the ENTER key.
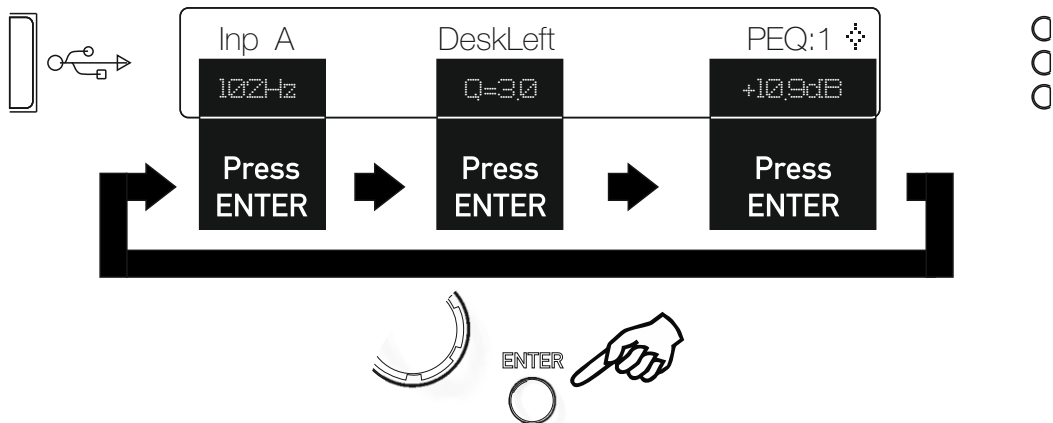


Finally, use the rotary encoder to adjust the value of the parameter.

You can swap bank at any time by pressing the BANK key, and to quickly access the same parameter on another channel (within a bank) double press on the required channel's EDIT key.  If the same parameter doesn't exist in a bank (such as no limiters on the input bank), the gain screen will be shown.

A third press on the same EDIT key will exit editing and return to the current bank's default screen.

Hint: You can quickly check the same setting of any parameter on a bank of channels by double pressing each EDIT key in turn – so to check each output's polarity setting, just press EDIT, press NEXT until Polarity is displayed, then press the next channel's EDIT twice, and so on.


**Input Gain**

Variable between -40dB and +6dB in 0.1dB steps.

```
Inp A              Gain
      Input Gain= 6.0dB
```

**Input Delay**

Max delay time is 650.000mS in 10uS steps. Swap to 1mS steps by pressing ENTER.
Units can be changed to read distance instead of time though the system sub-menu – see page 32 for details.

```
Inp A              Delay
      Delay= 500.00mS
```
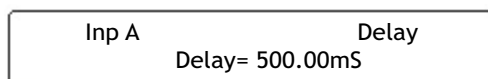
**Input Parametric EQ Bands 1 through 8**

All parametric bands cover a frequency range of 19.7Hz to 32kHz when in standard PEQ mode. Some restrictions apply when other filter types are selected. Move between frequency, 'Q' and filter gain by pressing ENTER. Press and HOLD the ENTER key whilst pressing QUIT to bypass a band and also access different filter types.

**PEQ** Parametric EQ

**LSF** Low Shelf

**HSF** High Shelf

**BPS** Band Pass

**NOT** Notch

**APF** All Pass

**PHF** Phase

**LPF** Low Pass VariQ

**HPF** High Pass VariQ

**LPF** Low Pass Elliptical

**HPF** High Pass Elliptical

```
Inp A              PEQ:1    ✧
100Hz    Q=3.0    10.0dB
```

A filter must be in Bypass before its type can be changed. Once bypassed, the ENTER key will allow access to the type list and this can be adjusted with the encoder, scrolling through the types listed above.

**Output Gain**

Variable between -40dB and +15dB in 0.1dB steps.

```
Out 1       Output 1   Gain
Output Gain= +10.2dB
```

**Output Polarity**

Switch between normal [+] and inverted [-] polarity. Note that channel ganging is temporarily disabled for polarity adjustments.

```
Out 1       Output 1   Polar.
Polarity = [+]
```

**Output Delay**

Max delay time is 650.000mS in 30nS steps. Swap to 1mS steps by pressing ENTER. Units can be changed to read distance instead of time though the system sub-menu – see page 32 for details.

```
Out 1       Output 1   Delay
Delay = 650.0000mS
```

**Output High Pass Filter**

Set the high pass filter frequency – a setting of "<10Hz" bypasses the filter. To change the filter slope and type, press ENTER and then adjust with the encoder. Note that 48dB/Octave filters will only be available if PEQ 6 & 7 are bypassed.

```
Out 1  Output 1           HPF      ⌐
25.8Hz   Linkw - Riley 24dB
```

Parametric bands will remember their settings if bypassed and used in 48dB/Octave crossover filters and these settings will be reinstated if a lower order filter type is subsequently chosen (24dB/Octave or lower).

The message

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

will be shown if the bands aren't already in bypass (or set to 0dB).

**Output Low Pass Filter**

Set the low pass filter frequency – a setting of ">32kHz" bypasses the filter. To change the filter slope and type, press ENTER and then adjust with the encoder. Note that 48dB/Octave filters will only be available if PEQ 8 & 9 are bypassed.

```
Out 1  Output 1           LPF      ⌐
9k70Hz   Linkw  - Riley 24dB
```

Parametric bands will remember their settings if bypassed and used in 48dB/Octave crossover filters and these settings will be reinstated if a lower order filter type is subsequently chosen (24dB/Octave or lower).

The message

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

will be shown if the bands aren't already in bypass (or set to 0dB).

**Output Parametric EQ Bands 1 through 9**

All parametric bands cover a frequency range of 19.7Hz to 32kHz when in standard PEQ mode. Some restrictions apply when other filter types are selected. Move between frequency, 'Q' and filter gain by pressing ENTER. Press and HOLD the ENTER key whilst pressing QUIT to bypass a band and also access different filter types.

**PEQ** Parametric EQ

**LSF** Low Shelf

**HSF** High Shelf

**BPS** Band Pass

**NOT** Notch

**APF** All Pass

**PHF** Phase

**LPF** Low Pass VariQ

**HPF** High Pass VariQ

**LPF** Low Pass Elliptical

**HPF** High Pass Elliptical

```
Out 1           PEQ:1
100Hz   Q=3.0   10.0dB
```

Parametric bands missing? If bands  6 and 7, or 8 and 9 are missing when editing, it is because they are being utilised by high order crossover filters – high pass filter orders above 24dB/Octave will disable and hide bands 6 and 7, and low pass filter orders above 24dB/Octave will similarly remove bands 8 and 9.

These bands will be reinstated if the respective crossover filter order is reduced to 24dB/Octave or lower.

A filter must be in Bypass before its type can be changed. Once bypassed, the ENTER key will allow access to the type list and this can be adjusted with the encoder, scrolling through the types listed above.

**Output Limiters: Program Limiter**

Note that the output limiters are calibrated differently for the amplifier output channels and the auxiliary output channels. They cover the same range, but the scaling on the threshold is different as the auxiliary outputs are calibrated to the maximum line output level, whilst the amplifier outputs are calibrated to include the gain of the power amplifier.

```
Out 1  Output 1          Limiter
Ak=2.0mS  Rl=x16   +38.5dB
```

Move between the attack time, the release time and the threshold by pressing the ENTER key.

If the message "Automatic T/C" appears on the limiter edit screen, this means that the limiters time constants have been set to be configured automatically, based on the frequency of this channel's high pass crossover filter.

To turn this feature off, and use manual attack and release times, please see the section on page 27 describing amplifier and auxiliary routing.

Release time is set as a multiplier of the attack time, so is represented as a "time N" readout. The minimum release time is twice the attack time, to minimise audible artefacts of inappropriate limiter time constants. We recommend using the automatic feature unless there is a good reason not to – a badly set up limiter will not only function incorrectly, and not provide the protection you expect, it can also sound pretty terrible!

Setting up limiters has perhaps been seen as a "black art" by some engineers – it is actually a simple process, as long as you have a few basic pieces of information to hand. Please read through the section on limiters and how to set them correctly, starting on page 46 of this manual.

**Output Limiters: Peak Limiter**

The peak limiter immediately follows the program limiter in an output (both power amplifier and auxiliary) signal path. It is designed to control the peaks that pass through the program limiter, due to the attack time set on the program limiter. A slow attack time will allow the program limiter to exceed its threshold for a short period, and this may cause over excursion on LF drivers. This may be controlled by imposing an absolute maximum level, set in dB above the program limiter threshold. This limiter has a zero overshoot characteristic and so only has a release parameter (with no attack time).

```
Out 1   Output 1  PeakLim
Rel.=Medium    +3.5dB Abv
```

Toggle between release time (Slow/Medium/Fast) and the threshold above the Program limiter by pressing the ENTER key. If the release time is replaced with "Rel.=Auto", this means that the limiters time constants have been set to be configured automatically, based on the frequency of this channel's high pass crossover filter. Please see the section on page 27 describing amplifier and auxiliary routing, if you want to revert to manual release time.

## DIRECTLY EDITING AUDIO PARAMETERS (cont'd)

**Output Matrix Gains**

Note that this feature will only be displayed if the configuration for the amplifier or auxiliary outputs has been set to operate in "Full Matrix" mode, as opposed to "Free Assign" or a standard routing configuration (1 x 4 way, 2 x 2 way etc.)

To use  "Full Matrix" mode, please see the section about adjusting the configuration on page 27.

```
Out 1   Matrix Gain  In A
0.0   Off     -40.0  +15.0
```

Initially, the "send" level from Input A will be selected – press ENTER to jump to the next "send" level for this output. Range is from -40dB to +15.0dB, with Mute one step below -40.0dB, whereupon the display will show "Off" as for Input B's send level in the above example.
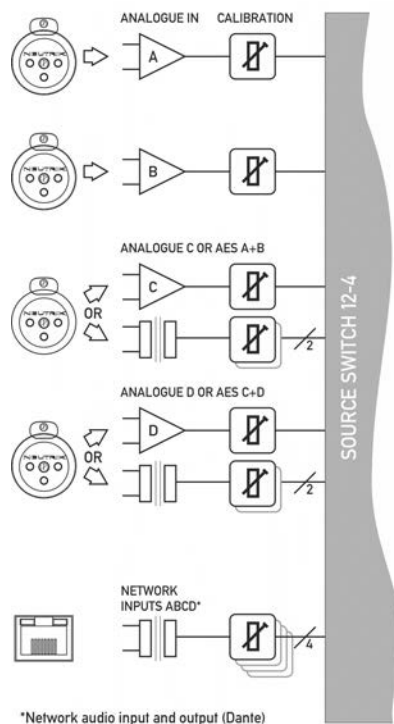
**Overview**

In addition to being able to control all the EQ and limiter functions via the front panel, it is also possible to completely reconfigure the source selections and routing, turn on matrix mixing, and configure ganging options to speed up editing.

The routing and ganging options are independent for the amplifier outputs and auxiliary outputs, but source selection is global across the entire device.

**Selecting Available Sources (Analogue, AES, Network Audio)**



Your amplifier can source audio from analogue, AES digital, or network locations (DANTE).

Analogue and AES digital audio are standard, and arranged so that AES digital audio can be chosen in pairs of channels to replace either analogue inputs A&B together or analogue inputs C&D together, or all four channels.

Physical input of AES for channels A&B is swapped to the input XLR for channel C and the AES input for channels C&D is on the XLR socket for channel D.

In this way, a pair of analogue inputs can remain connected to channels A&B and a digital stream of the same audio can be connected to input C, with fallback from one to the other possible without repatching.

With the DANTE card, four additional digital sources will then be available.

In total, 8 sources can be made available at any one time – four analogue and four digital. This affects the choices that can be made for routing to the inputs of the DSP channels.

The rules are as follows:

Analogue A and B is available at all times;
If AES A&B is selected, Analogue C cannot be used (shares XLR C);
If AES C&D is selected, Analogue D cannot be used (shares XLR D);
If AES A&B is chosen, Dante A&B cannot be used (shares SRC One);
If AES C&D is chosen, Dante C&D cannot be used (shares SRC Two);

Source selection is therefore affected by the selection of AES inputs, which then controls the choice of input source selections on offer to any input processing channel. Switch any required AES sources first, then select the required source combination.

**Switching from Analogue to AES Sources**

To select inputs, from the home screen, press MENU, choose the CONFIGURATION Sub-Menu and press ENTER.

```
CONFIGURATION Sub - Menu
Input AES Selection
```

Use the encoder or BACK and NEXT keys to find the Input AES Selection option and press ENTER.

The screen will now show the current choice for the physical XLR inputs on channels C&D:

```
Input AES Selection
XLR C= Ana: XLR D= Ana *
```

Use the encoder or BACK and NEXT keys to cycle between the four combinations, of analogue (Ana) or Digital (AES). Press ENTER to confirm the choice, and again to confirm if changes have been made.

```
=== Selection Changed === Press
[ENTER] to CONFIRM
```

If a digital source is chosen for either XLR, the AES IN LED on the front panel will begin to flash until a valid lock is seen for the selected inputs.

If the AES input(s) have valid stable signals and there are no other issues, the LED will not be flashing.



**Note that the AES LED being illuminated does NOT mean that an AES source is being used or routed anywhere within the amplifier. It is an indication that valid AES is connected.**

**ADVANCED EDITING FEATURES (cont'd)**

**Choosing Sources to Route to Input Processing Channels**

To choose sources, from the home screen, press MENU, choose the CONFIGURATION Sub-Menu and press ENTER.

> CONFIGURATION Sub -Menu
> Input Source Selection

Use the encoder or BACK and NEXT keys to find the Input Source Selection option and press ENTER.
The screen will now show the current choice for the first input processing channel (A):

> Input Source Selection
> Input A = Analogue A

Use the encoder or BACK and NEXT keys to scroll through the eight choices for this input. These will be:

Four analogue sources:

Analogue A

Analogue B

{Analogue C}

Analogue D

And four digital sources, depending on the AES selection:

AES A

AES B

{Dante C}

{Dante D}

The choices shown above would be shown if AES has been selected on XLR C.  Analogue C is shown in brackets to indicate that this selection will result in no audio, either because the input is currently unavailable (due to input selection choices in the case of Analogue C) or because the input does not exist (due to the network audio card not being fitted in the case of the Dante C and Dante D).

Press ENTER when the required input source is shown, and the next processing channel will be selected, running through input channels A-D. Finally, if anything has been changed, the confirmation screen will show:

> === Selection Changed === Press
> [ENTER] to CONFIRM

Press ENTER to confirm the changes and exit to the home screens.

**Changing Network Output Source**
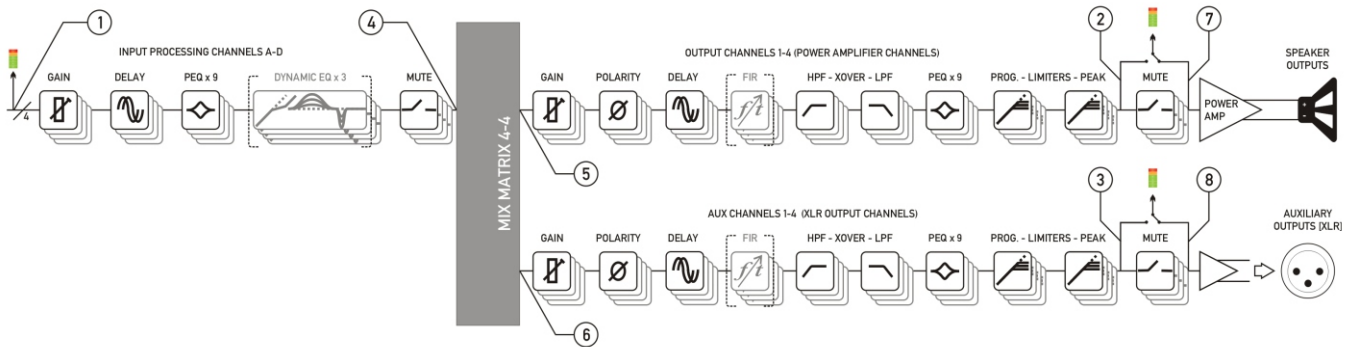


    As your amplifier has a network audio card fitted (Dante) then, as well as being able to route four channels of audio from the network, it is possible to route four channels of audio back onto the network. The possible sources for the network outputs are always selected as a group of four, and there are eight points within your amplifier's DSP than can be chosen.

    To choose sources, from the home screen, press MENU, choose the CONFIGURATION Sub-Menu and press ENTER.

> CONFIGURATION Sub -Menu
> Digital Output Source

    Use the encoder or BACK and NEXT keys to find the Digital Output Source option and press ENTER.

> Digital Output Source
> Source = Aux Outputs   *

    Use the encoder or BACK and NEXT keys to choose between one of the following 5 sources:

    Digital Outputs Muted
    The network outputs source is disconnected and no audio is transmitted. The Dante connection is maintained and visible on the network

    Amp Outputs pre Mute
    Source : audio is sourced just after all amplifier output's processing but pre-mute so the network copies will not be affected by local (or remote) mutes.

    Aux Outputs pre Mute
    Source : audio is sourced just after all auxiliary output's processing but pre-mute so the network copies will not be affected by local (or remote) mutes.

    Amp Outputs+M
    Source : audio is sourced just after all amplifier output's processing post-mute so the network copies will be affected by local (or remote) mutes.

    Aux Outputs+M
    Source : audio is sourced just after all auxiliary output's processing post-mute so the network copies will be affected by local (or remote) mutes.

    Press ENTER to confirm the changes and exit to the home screens.

**Changing Crossover Configuration**

As the amplifier output's configuration setup is identical to the auxiliary output's configuration, this section will just explain the former. The configuration of the amplifier outputs 1-4, and auxiliary outputs 1-4 have separate entries in the Configuration sub-menu, but operate identically.   Once the input sources and AES selections have been set up, the last part of the process is to determine what actual routing configuration is required from the "outputs" of the four input processing channels.

This is the mix matrix. The mix matrix can function either in a "Boolean" operation mode, where input signals still have the capability to be mixed together to feed an output channel's processing, but at fixed standard levels, or as a fully variable mixer.

There is also a selection of useful templates if used in the "Boolean" mode, which will pre-configure not only the routing, but also useful starting points for each outputs' crossover filters.

To change configuration, from the home screen, press MENU, choose the CONFIGURATION Sub-Menu and press ENTER.

```
CONFIGURATION Sub -Menu
Output Routing
```

Use the encoder or BACK and NEXT keys to find the Output Routing option and press ENTER.

```
Output Routing
Routing = 2 x 2 WAY  *
```

Use the encoder or BACK and NEXT keys to choose from the following routing options:

Full Matrix
This mode offers continuously variable send levels from the four input processing channels and the adjustment of the levels is then through individual output editing as explained on page 22.

Free Assign
This mode allows the input sends to be selected as ON or OFF and intelligently adjusts the send levels to maintain 0dB on the output – this is further explained on the next page.

1 x 4 WAY
All four outputs are fed from input processing channel A's output and assigned default crossover points, which may then be adjusted if required.

2 x 2 WAY
Outputs 1 & 2 (or Aux 5 & 6) are fed from input processing channel A's output, and outputs 3 & 4 (or Aux 7 & 8) from channel B. Default crossover points are assigned.

4 x 1 WAY
Input processing channel A feeds output channel 1, B to 2, C to 3 and D to 4.   Crossover points are not assigned and will be set to full range – this can be changed manually if required.

**Using Free Assign Mode**

Free assign mode allows any combination of inputs to be routed to an output, with the send levels being intelligently adjusted to ensure that the output levels sum to 0dB.

So, if two inputs are summed to a particular output, their gains will be dropped by 6dB so that if both input levels are 0dB they will sum to produce a 0dB level at the output (assuming the output gain as not been adjusted by the user).

If three inputs are summed, their gains will be reduced by 9.5dB so if all three inputs are 0dB, so the output will be 0dB.

Lastly, if all four inputs are summed to an output channel, the gains will be reduced by 12dB.

If this method of operation is not desired, the same summing can be achieved without the automatic gains adjustments by switching to Full Matrix mode and manually setting the send levels as required.

Select Free Assign mode when changing the output routing (separately for power amp outputs and auxiliary outputs).

To change configuration, from the home screen, press MENU, choose the CONFIGURATION Sub-Menu and press ENTER.

```
CONFIGURATION Sub -Menu
Output Routing
```

Use the encoder or BACK and NEXT keys to find the Output Routing option and press ENTER.

```
Output Routing
Routing = Free Assign *
```

Use the encoder or BACK and NEXT keys to choose Free Assign and press ENTER.
The first MUTE key will begin to flash and the EDIT keys will illuminate to show which input processing channels are being routed to this output as well as being show on-screen.  In the example below, output 1 is being fed from A+D:
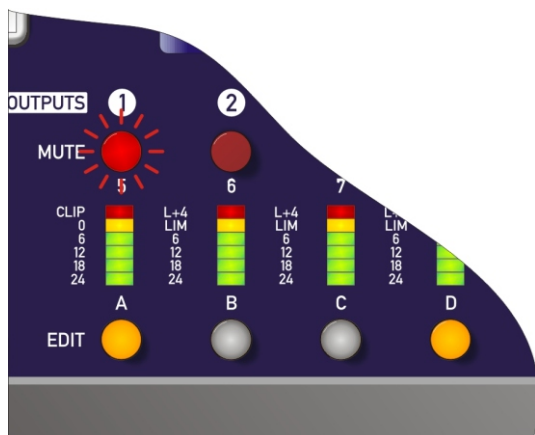


```
Output  Routing
Routing Out1 = A+D *
```

Use the encoder or BACK and NEXT keys to choose the required routing combination.  The EDIT key LEDs will change in sympathy with the input selection.

Press ENTER to move on to the next output channel or press the channel's MUTE key to select it. When all channels have been routed, press ENTER – the unit will run through any remaining channels (if output 3 had been manually selected, for example, then ENTER would run through channel 4 before moving on to the final set-up selections).

The next set-up selections are concerned with using automatic time constants for the limiters and selecting bridged configurations.

The screen will first show:

```
          Output Routing
          AutoLimiter TimeCst:Yes*
```

DAS Audio advise setting the limiter time constants automatically. They are based on the frequency of this channel's high pass crossover filter. Unless you have a particular reason to not use the auto setting, set this to YES and press ENTER.

### Bridged Mode for Output Pairs

Output pairs can be selected to run in a bridged configuration to realise higher output powers.
When operating in bridged mode, the minimum impedances are doubled.
The minimum load in bridged mode is 4 ohms.
For details of how to connect to the amplifier when running in bridged mode, please see page 12.

Selecting bridged mode is available through the CONFIGURATION > Output Routing menu and as part of the set-up wizard.  Bridged mode is not available for auxiliary output configuration and should instead be selected on the externally driven "slave " amplifier.

The screen will show:

```
          Output Routing
          Bridging = None    *
```

Use the encoder to select which channel pairs are to be bridged:

1<>2  - bridge outputs 1 & 2 only
3<>4  - bridge outputs 3 & 4 only
1<>2 & 3<>4 – bridge both output pairs

Confirm selection with ENTER – final section of the configuration wizard will then ask about resetting/clearing parameters as explained in the next section. If a bridged mode has been selected there are some changes apparent in the interface to confirm this.

Depending on what bridged mode are active the appropriate LED will illuminate above the power switch:
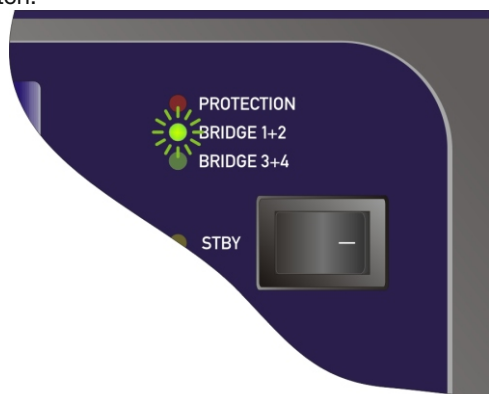


There are some changes operationally when a pair of channels are bridged.

Channel MUTE keys will be automatically ganged;

Meters for the bridge pair will read identically; and lastly

Channel EDIT keys will be automatically ganged, and the display reflects the bridged mode by changing to show the following when a bridged pair are edited:

```
1<B>2  Output 1  Gain
Output Gain = 0.0dB
```

In this example, the bridged pair are outputs 1 & 2 and this will be displayed if EDIT is pressed on either output channel.

Bridged status is stored in an output memory, and ignored if this memory is recalled into the auxiliary channels.

### Resetting Audio Parameters

Resetting parameters is available through the CONFIGURATION > Output Routing menu and CONFIGURATION > Auxiliary Routing menu as part of the set-up wizard.
The final set-up query is about resetting the current parameters. If you have already edited the outputs' DSP settings and are setting up routing last, make sure you select "No" when asked about resetting, or all your filter, delay limiter and crossover parameters will be cleared!

```
Output Routing
Clear Out Settings:Yes*
```

If "Yes" is selected, then, depending on format chosen, all outputs will be set to full range (no crossover), limiters at max threshold (so minimum protection), delays and gains at zero,  polarity normal and all bands of EQ  to PEQ mode, 1kHz, Q of 3.0 and 0dB.  All output/auxiliary channels will also be muted for safety!
Certain formats will preset the crossover frequencies to give a useful starting point:
Selecting a 2 x 2 way configuration will preset the output crossovers as
Outputs 1 + 3:   <10Hz – 1.62kHz, 24dB/Oct. Link-Riley
Outputs 2 + 4:   1.62kHz - >32kHz, 24dB/Oct. Link-Riley
Selecting a 1 x 4way configuration will preset the output crossovers as
Output 1:   <10Hz – 149Hz, 24dB/Oct. Link-Riley
Output 2:    149Hz – 1.31kHz, 24dB/Oct. Link-Riley
Output 3:   1.31kHz – 8.00kHz, 24dB/Oct. Link-Riley
Outputs 4:   8.00kHz – >32kHz, 24dB/Oct. Link-Riley
In both cases, all other DSP functions are reset as in bold above.
Hint: This procedure can be used to quickly clear all EQ – simply run through all the steps of either output or auxiliary routing as appropriate without changing anything and select "Yes" when asked about clearing output settings.

### Ganging Channels for Editing

To make editing of multiple channels simpler (for example, stereo input sources or two-way output editing), it's possible to gang channels together so that any edits applied to one will automatically be applied to the other (or others).
Ganging is separated for inputs, outputs and auxiliary channels, but the method of setting them up is identical so only the output ganging option will be covered here.
To change output ganging, from the home screen, press MENU, choose the CONFIGURATION Sub-Menu and press ENTER.

```
CONFIGURATION Sub -Menu
Output Ganging
```

Use the encoder or BACK and NEXT keys to find the Output Ganging option and press ENTER.

> Output Routing
> Ganging = 1+3 2+4 *

Use the encoder or BACK and NEXT keys to choose the required ganging option and press ENTER. If the ganging mode has changed, confirm this with a final press of ENTER and the ganging is complete.

Note that ganging assumes the settings of the lowest numbered channel in the gang is the initial master channel and so any changes to ganging will immediately copy the settings from the lowest member in the gang to all other ganged channels.

For example – adding channel 3 to a gang of 1+3 will instantly make channel 3's settings identical to channel 1.

Mutes remain unganged at all times (except in bridge mode – see page 29), as does polarity.

Ganged channels are indicated by the EDIT keys illuminating together and the display showing the member numbers of the gang:

> Aux1&3 Extern 1  Gain
> Output Gain = +12.5dB

The label shown ("Extern 1") is always the lowest gang member, no matter which EDIT button has been pressed in the gang.

Labels can only be changed through remote control software or the DeltaDirect iPad app.

Ganging states are stored in output memories and will be recalled when an output memory is recalled to either amplifier outputs or auxiliary outputs.

### Memory Overview

The memory in your amplifier is organised into three different types – input memories, output memories and source memories. Input and output memories are formatted in banks of four channels at a time – be that four input channels or four output channels (amplifier or auxes).

Source memories are a different type and are global to the entire amplifier. It is also possible to store and recall a complete copy of all current settings (so, input, source, amplifier and aux outputs) in a memory location – "Everything".

Input memories contain all the EQ, gains, delays, mutes and names for the four input channels.
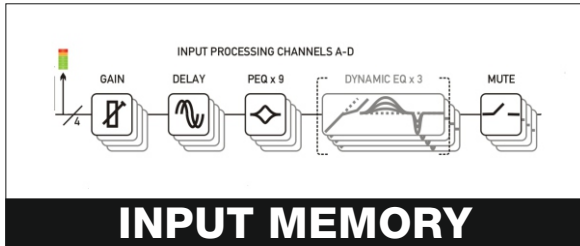


Output memories contain all the EQ, crossover filters, FIR data (if applicable), polarities, gains, delays, mutes, limiters (including auto time constants setting), names, bridge modes, and routing from the mix matrix (including free assign and full matrix gains if applicable). This is for the bank of four amplifier channels, or the bank of four auxiliary output channels.
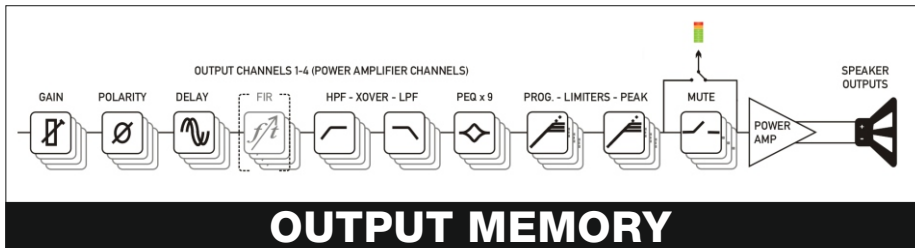


Source memories contain the all data concerned with additional choices of source selection encompassing AES/Analogue switching, Network Audio Output Source and the source switching (choosing which inputs feed the input DSP channels).
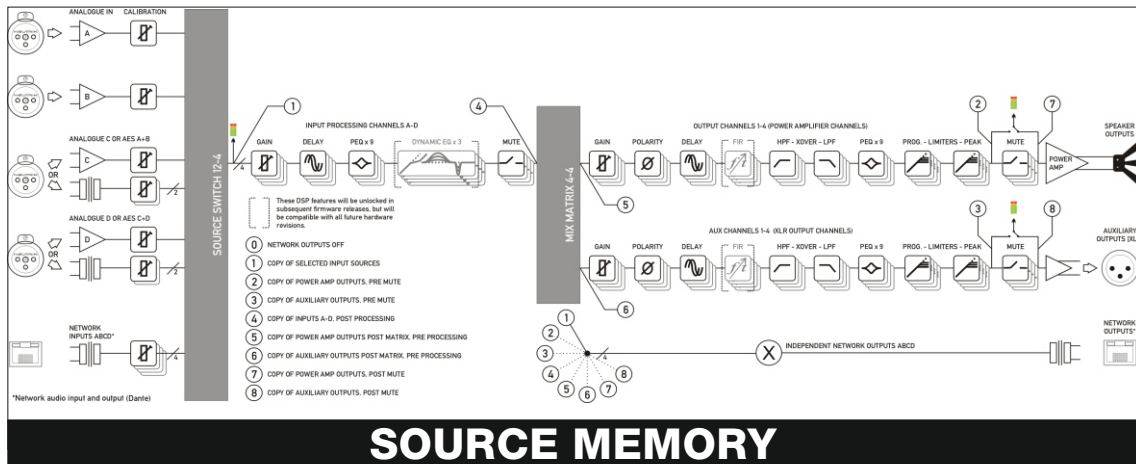


Memories all exist in a numbered list and depending on the type of memory to be recalled, the list will be filtered to only show the applicable memory type.

**Storing a Memory...**

When a memory is to be stored, the type is first selected (Input, Output, Source or Everything) and then in the case of an Output memory, this must be further qualified to determine if the stored data is to originate from the amplifier outputs or the auxiliary outputs.

To store a memory, from the home screen, press MENU, choose the GLOBAL MEMORY Sub-Menu and press ENTER.

```
MEMORY Sub -Menu
Recall an Output Memory
```

Use the encoder or BACK and NEXT keys to choose either Input or Output memory type as appropriate and press ENTER.

```
STORE an Output Memory     Store from
Amp Outputs *
```

In the previous example, the extra step is to choose whether to store the four amplifier outputs' data, or the four auxiliary outputs' data. Choose with encoder and press ENTER.

```
STORE an Output Memory
103:
```

Use the encoder to choose the memory number to use – memories that are already used will show the current name and a question mark on the bottom line and ask for confirmation if chosen:

```
STORE an Output Memory
94:LEFT SIDE 6 SUBS    ?
```

The name may now be edited/entered using a combination of the encoder and the BACK/NEXT keys to move along the name. Press ENTER when complete, or QUIT to exit.

```
Enter Memory Name
94:LEFT SIDE 6 SUBS    ?
```

Memory is then stored, with a confirmation message and the amplifier will return to the default screen on the selected bank.

**Recalling a Memory...**

When a memory is recalled, after choosing the type (Input, Output, Source or Everything), recall of Output memories is further qualified by what bank of outputs the recalled data is to affect – amplifier or auxiliaries.

To recall a memory, from the home screen, press MENU, choose the GLOBAL MEMORY Sub-Menu and press ENTER.

```
┌────────────────────────────────────┐
│        MEMORY Sub -Menu            │
│      Recall an Output Memory       │
└────────────────────────────────────┘
```

Use the encoder or BACK and NEXT keys to choose either Input or Output memory type as appropriate and press ENTER.

```
┌────────────────────────────────────┐
│  RECALL an Output Memory      Recall to │
│ Aux Outputs *                      │
└────────────────────────────────────┘
```

Use the encoder to choose the memory required, remembering that the list   will only contain memories of the chosen type, and so the numbers may not be contiguous (so   there may be gaps!)

Press ENTER and confirm. If the routing configuration of the memory just recalled is different to the current configuration, the outputs will be muted for safety.

```
┌────────────────────────────────────┐
│  RECALL an Output Memory 94:LEFT SIDE │
│ 6 SUBS                             │
└────────────────────────────────────┘
```

Dependant on the bank selected, the display will by default show the name of the last input, amp output or aux output memory recalled, or the same on all three if an "Everything" memory was recalled.

## SYSTEM ADJUSTMENTS

**Overview**

Under the System Sub-Menu you'll find all the non-audio features relating to your amplifier's operation.  As these are self-explanatory, they are summarised in the table below:

| System Sub Menu Option | What it's for... |
|---|---|
| System Status | Check firmware version and system information |
| LCD Contrast | Adjust the viewing angle for the display |
| LED Brightness | Adjust the brightness of meters and switch LEDs |
| Wakeup Time | Add a delay before start of audio fade up or keep muted |
| Output Meter Option | Choose if meters should show pre or post mute |
| Clip Hold Time | Set how long a clip or limit condition is held on meters |
| Filter 'Q'/Bandwidth | Show PEQ filter bandwidth as 'Q' or '1/Q' (bandwidth) |
| Delay Time/Distance | Choose to show delay time in metres, feet or time (mS) |

### Overview

Your amplifier is equipped with multiple options for remote control. It can operate as a single device via a USB connection to the front panel, and this can also be used as a converter to an RS485 network, where other DX series amplifiers or DAS system units may be connected.

It may also be controlled via an Ethernet connection and again will relay data onto an RS485 network if required.

The Ethernet and USB connections can be used in an intelligent switchover mode, where the Ethernet connection is active, but will be overridden by a USB connection, allowing quick connection to a system via the front panel without needing to reconfigure the current remote interface setup.

The RS485 network can be used on its own as third means of communication, if the amplifier is part of a bigger system and an Ethernet connection is not available.

Using an Ethernet connection, multiple amplifiers can be connected directly to a standard Ethernet switch, or WiFi router, allowing for wireless communication.

Finally, the GPIO port on your amplifier can be configured to feed back a variety of alarm signals (such as over temperature threshold, or protection triggered) and offer simple closed contact control of standby mode, mutes or memory recalls. It also accepts variable voltages which allow direct level control of selected channels' gains.

### Remote Control Software Choices

Your amplifier is designed to be fully configured via AudioCore.

Main configuration is performed via AudioCore, either with a wired connection via Ethernet, USB or RS485. Connection to amplifiers is also possible wirelessly using the Ethernet connection if this is via a wireless router.

Tuning of a system can then be performed wirelessly (via a WiFi connection) using just the iPad or tablet.

### Configuration of the Remote Interface

To configure the remote interface, from the home screen, press MENU, choose the INTERFACE Sub-Menu and press ENTER.

```
INTERFACE Sub -Menu
External Interface
```

Use the encoder or BACK and NEXT keys to select External Interface and press ENTER.

```
External Interface
Mode= Ethernet or USB
```

Use the encoder or BACK and NEXT keys to choose the required interface and press ENTER. This will invoke a series of further options depending on the interface chosen – these are explained below:

Mode = OFF: Interface is disabled and no external comms are active. This is useful if you need to fully secure your amplifier as well as locking the front panel (for more information on security and locking see page 42.)

Mode = Ethernet Only: The Ethernet control port on the rear panel is active. Selecting this choice will then ask for the following further information to be confirmed:

RS485 Relay:   Choose ON to relay all remote control data onto the RS485 network for connection to further devices (other DAS Audio units)

RS485 Speed: Only shown if RS485 Relay = ON: set RS485 baud rate (recommended to use 57600)

Remote ID Number:   Set a device ID (still required even on Ethernet connections). This must be unique to the connected unit – setting the same ID on multiple units may cause comms problems for all devices.
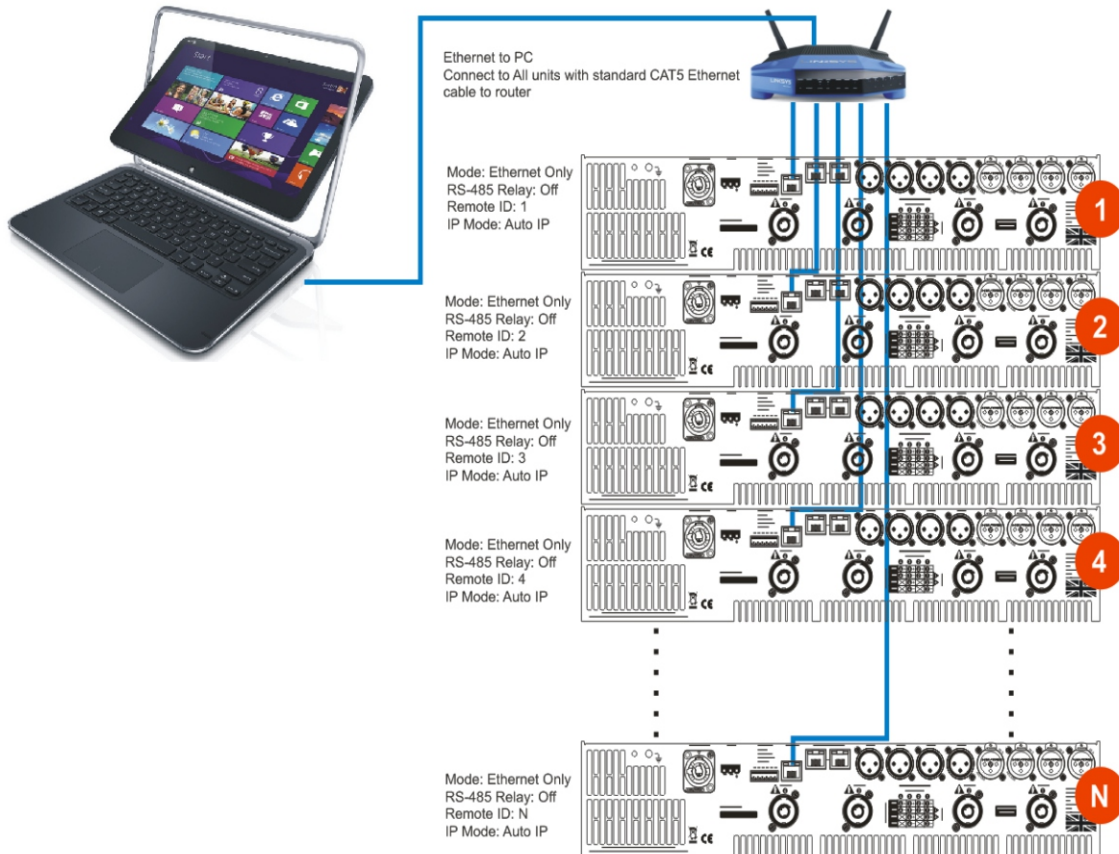
IP Mode: Choose either Static or Auto IP.   Auto IP will generate an IP address based on unique hardware features such as MAC address.   Choose Static if you need to select the IP address to lie within a specific range, for instance when working within a larger infrastructure.

GtWay:   The gateway address is used for external access to the Internet and should be left at the default setting.

SubNt: The subnet mask is used to subdivide IP addresses into groups that allow further sub-groups addressing to be defined, so further extending the address range.

Leave at default 255.255.255.0 unless specifically required.

IP: The IP address specifies the amplifier's unique identifier on the Ethernet network.   It is used in conjunction with the amplifier's Remote ID number to identify individual devices on the network. Make sure this is not set to the same value as any other devices or comms problems will occur.

Mode = USB Only: The USB socket on the front panel is active.  Selecting this choice will then ask for the following further information to be confirmed:

USB Speed: As the USB connection appears as a virtual COM port in remote software, it still has an associated serial speed.  Leave at the default of 57600.  This is not the actual USB serial data rate – it's the virtual COM port data rate!

RS485 Relay: Choose ON to relay all remote control data onto the RS485 network for connection to further devices (other Delta DSP amplifiers or XTA processors)

RS485 Speed: Only shown if RS485 Relay = ON: set RS485 baud rate (recommended to use 57600)

Remote ID Number: Set a device ID (still required even on Ethernet connections).  This must be unique to the connected unit – setting the same ID on multiple units may cause comms problems for all devices.
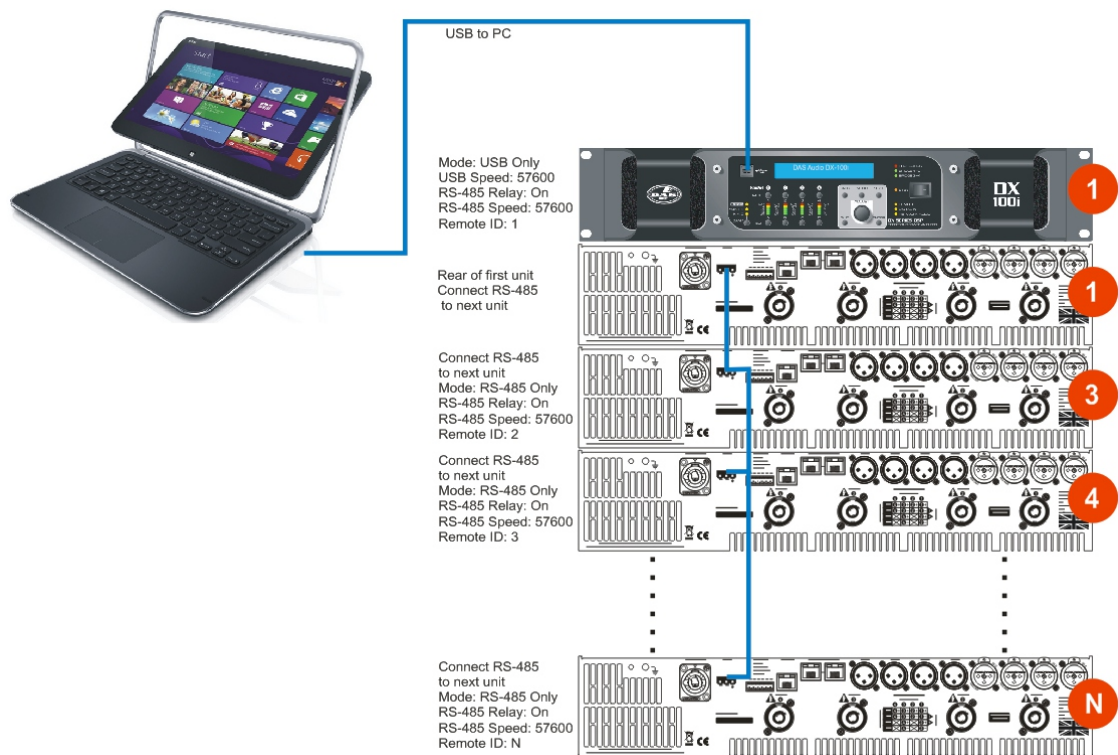
Mode = Ethernet Or USB: The Ethernet control port on the rear panel is active, so long as the amplifier is not connected via USB, whereupon it automatically takes priority.   Selecting this choice will then ask for the following further information to be confirmed:

RS485 Relay: Choose ON to relay all remote control data onto the RS485 network for connection to further devices (other Delta DSP amplifiers or XTA processors)

USB Speed: As the USB connection appears as a virtual comm port in remote software, it still has an associated serial speed.   Leave at the default of 57600. This is not the actual USB serial data rate – it's the virtual COM port data rate!

RS485 Speed: Only shown if RS485 Relay = ON:   set RS485 baud rate (recommended to use 57600)

Remote ID Number:   Set a device ID (still required even on Ethernet connections).   This must be unique to the connected unit – setting the same ID on multiple units may cause comms problems for all devices.

IP Mode: Choose either Static or Auto IP.   Auto IP will generate an IP address based on unique hardware features such as MAC address. Choose Static if you need to select the IP address to lie within a specific range, for instance when working within a larger infrastructure.

GtWay: The gateway address is used for external access to the Internet and should be left at the default setting.

SubNt: The subnet mask is used to subdivide IP addresses into groups that allow further sub-groups addressing to be defined, so further extending the address range.

Leave at default 255.255.255.0 unless specifically required.

IP: The IP address specifies the amplifier's unique identifier on the Ethernet network. It is used in conjunction with the amplifier's Remote ID number to identify individual devices on the network.   Make sure this is not set to the same value as any other devices or comms problems will occur.

**Remember that this mode is purely an automatic priority switching version of an Ethernet or USB connection – it is not possible to interconnect amplifiers via the Ethernet cables either directly or via a router – use the RS485 bus for device interconnection.**

**All three connection methods will relay data either explicitly (RS485 using the hardware bus connection) or electronically (USB or Ethernet through the processor) if required.**

Mode = RS485 Only: The RS485 port on the rear panel is active.   Selecting this choice will then ask for the following further information to be confirmed:

RS485 Speed: Set RS485 baud rate (recommended to use 57600)

Remote ID Number: Set a device ID (still required even on Ethernet connections). This must be unique to the connected unit – setting the same ID on multiple units may cause comms problems for all devices.
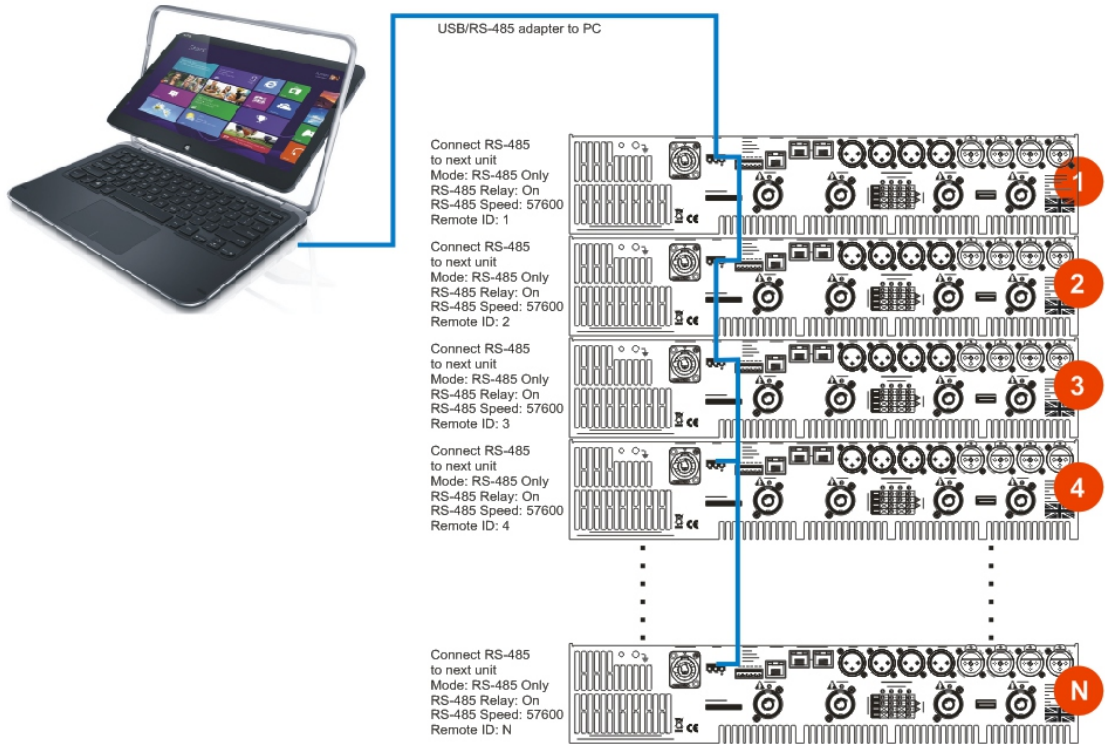
**Wireless Connection with Wifi Router**

Free yourself of cables and go wireless! Plug the Ethernet connection from your amplifier(s) into a wireless router and you are good to go!

Main remote configuration of a single amplifier, or a network of amplifiers is performed through AudioCore, with system tuning using either AudioCore.

## RESETTING BACK TO DEFAULTS AND CLEARING SECURITY

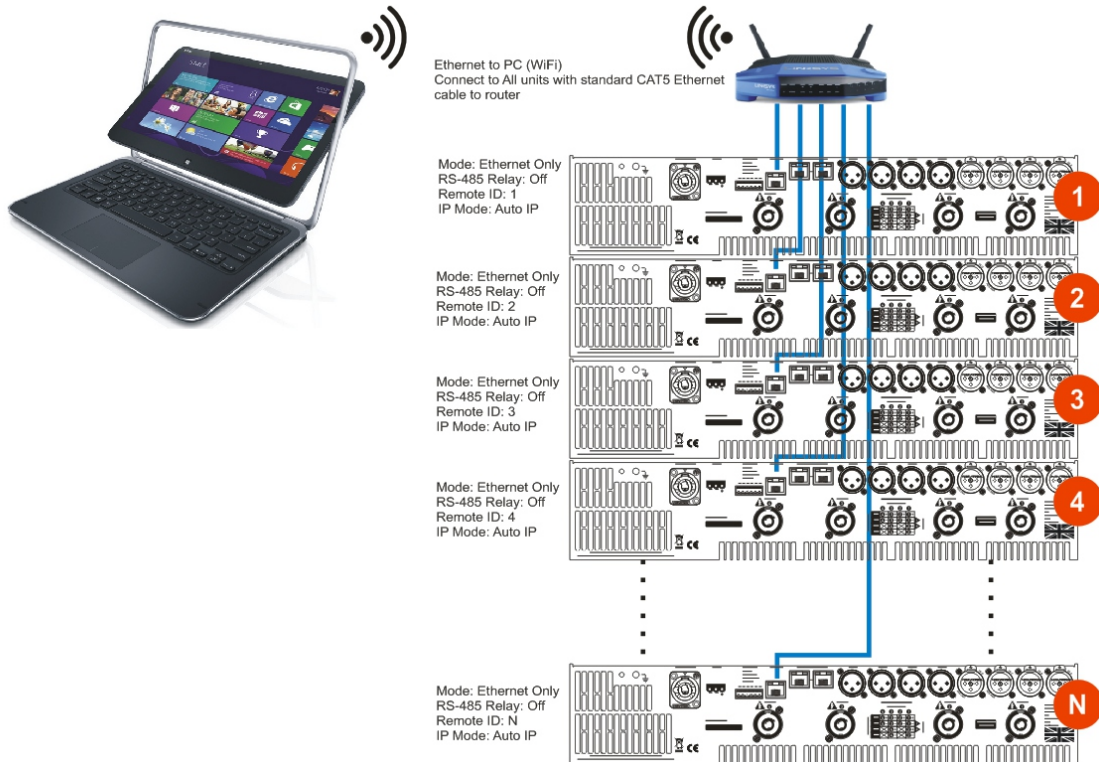If you need to quickly clear all current settings and memories, there are a variety of setup control codes that can do this for you.

### Default Settings

Your amplifier will start-up as a "four analogue inputs to four output system", with no EQ or limiting (apart from self-protective limiters) in place. The aux outputs will "mirror" the analogue inputs (and so initially act just like "link" outputs, again with no EQ or limiting in place).

### Entering Start-up Control Codes

Turn the amplifier on with the BACK and ENTER keys held. The display will show:

```
Enter Setup Code:
      [   ]
```

Use the EDIT and MUTE keys to represent the characters A, B, C D for EDITs and 1, 2, 3, 4 for MUTEs.

Type in one of the following codes to initiate a setup function:

D4B13D – clear all user memories and reset to defaults, keeping security and factory alignment settings

A2341B – clear current settings to default (leaves memories, alignments and security)

AA2121 – clear security via the XTA/MC2 supplied crack code

### Clearing  Security – Forgotten Unlock Code

Note that if you see the message "Parameters Locked!" during editing, or parameters can be viewed but not adjusted, then this is due to preset data locking or hiding and is specific to the currently running preset – this is not device security, and we cannot override it – it is specified by the speaker manufacturer.

If you have forgotten the pass-code and your amplifier is locked, you will need to contact us to unlock it after performing the following procedure.   There is no master unlock code for all amplifiers – the code is generated based on an algorithm using amplifier internal settings for greater security.

Turn the amplifier on with the BACK and ENTER keys held. The display will show:

```
Enter Setup Code:
      [   ]
```

Use the EDIT and MUTE keys to represent the characters A, B, C D for EDITs and 1, 2, 3, 4 for MUTEs.

Type in AA2121.

The display will now show:

```
Enter XTA Supplied Code:
Break No.=64223  [    ]
```

Make a note of the "Break Number" shown (64223 in this example) and contact us with this – we can then supply the code to unlock your amplifier.

Do not enter any menus or make any further adjustments to the amplifier or this code may "roll over" and you will need to go through this procedure again.   Either switch the amplifier off or press QUIT to exit without entering a code.

Resetting security leaves does not delete any settings – it just clears the locking passcode.

## SPECIFICATIONS

### Main Specifications

| Parameter (Units) | DX-80I | DX-100I |
|---|---|---|
| Output Power per channel [Crest Factor = 4.8] (Watts) | | |
| 8 Ohms | 1000 | 1400 |
| 4 Ohms | 2000 | 2800 |
| 2.7 Ohms | 2200 | 3700 |
| 2 Ohms | 2000 | 3500 |
| Output Power per channel bridged [Crest Factor = 4.8] (Watts) | | |
| 8 Ohms | 4000 | 5400 |
| 4 Ohms | 4000 | 7000 |
| THD+N, 4 Ohms (%) | | |
| @1kHz, 1dB below max output power < | 0.18 | 0.08 |
| @20Hz - 20kHz, 1dB below max output power < | 0.2 | 0.1 |
| Gain Options (dB) | 32 | 32 |
| Sensitivity Options for max power (dBu) | 8.3 | 10.7 |
| Sensitivity Options for max power (Volts) | 2.0 | 2.66 |
| Frequency Response, +0/0.5dB (Hz) | 20 – 20000 | 20 – 20000 |
| Power Consumption, Nominal @ 240V, 4 Ohms (A) | 5.0 | 7.5 |
| Power Consumption, Nominal @ 120V, 4 Ohms (A) | 10.4 | 15.5 |
| Dimensions H x W x D (mm) | | |
| Amplifier | 88 x 482 x 428 | 88 x 482 x 428 |
| Boxed | 230 x 580 x 560 | 230 x 580 x 560 |
| Boxed Shipping – all except UK | 250 x 610 x 600 | 250 x 610 x 600 |
| Weight (kgs) | | |
| Amplifier | 10.3 | 10.9 |
| Boxed – shipping | 11.8 | 12.4 |

### Additional Specifications

| Parameter (Units) | DX-80I | DX-100I |
|---|---|---|
| Input Impedance – Active Balanced (Ohms) | 20k | 20k |
| Input CMRR (dB) | > 60 | > 60 |
| Damping Factor, 1kHz, 8 ohms | > 400 | > 400 |
| Signal Limiters Present | Yes | Yes |
| Protection Present – Short Circuit / DC Output / Temperature | Yes | Yes |
| Mains In-rush Control Present | Yes | Yes |
| Output Power per channel, 8 Ohms (Watts) | | |
| Continuous music [Crest Factor of 2.8 or 9dB] | 975 | 1360 |
| Continuous music [Crest Factor of 4.8 or 14dB] | 1000 | 1400 |
| Continuous music [Crest Factor of 7.8 or 18dB] | 1025 | 1440 |
| Output Power per channel, 4 Ohms (Watts) | | |
| Continuous music [Crest Factor of 2.8 or 9dB] | 1950 | 2620 |
| Continuous music [Crest Factor of 4.8 or 14dB] | 2000 | 2700 |
| Continuous music [Crest Factor of 7.8 or 18dB] | 2050 | 2780 |
| Output Power per channel, 2.7 Ohms (Watts) | | |
| Continuous music with Crest Factor of 2.8 [9dB] | 1960 | 3600 |
| Continuous music with Crest Factor of 4.8 [14dB] | 2010 | 3700 |
| Continuous music with Crest Factor of 7.8 [18dB] | 2060 | 3880 |

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

## SPECIFICATIONS (cont'd)

Power Consumption and Thermal Emissions – DX-80I

| Mains (V) | Load (R) | Current Draw (A) | | | | Thermal Emissions (W) | | | |
|---|---|---|---|---|---|---|---|---|---|
| | | No Sig'l | Light | Average | Heavy | No Sig'l | Light | Average | Heavy |
| 240 | 8 | 1.5 | 2.2 | 3.4 | 6.3 | 360 | 378 | 410 | 486 |
| 240 | 4 | 1.5 | 2.8 | 5.0 | 10.3 | 360 | 394 | 453 | 593 |
| 240 | 2 | 1.5 | 3.1 | 5.8 | 12.4 | 360 | 402 | 474 | 647 |
| 120 | 8 | 3.3 | 4.7 | 7.1 | 12.9 | 400 | 418 | 450 | 526 |
| 120 | 4 | 3.3 | 5.9 | 10.4 | 21.0 | 400 | 434 | 493 | 633 |
| 120 | 2 | 3.3 | 6.5 | 12.0 | 25.1 | 400 | 442 | 514 | 687 |

Power Consumption and Thermal Emissions – DX-100I

| Mains (V) | Load (R) | Current Draw (A) | | | | Thermal Emissions (W) | | | |
|---|---|---|---|---|---|---|---|---|---|
| | | No Sig'l | Light | Average | Heavy | No Sig'l | Light | Average | Heavy |
| 240 | 8 | 2.1 | 3.2 | 5.1 | 9.6 | 504 | 533 | 582 | 701 |
| 240 | 4 | 2.1 | 4.1 | 7.5 | 15.7 | 504 | 557 | 647 | 863 |
| 240 | 2 | 2.1 | 4.6 | 8.9 | 19.1 | 504 | 570 | 682 | 953 |
| 120 | 8 | 4.7 | 6.9 | 10.6 | 19.6 | 560 | 589 | 638 | 757 |
| 120 | 4 | 4.7 | 8.7 | 15.5 | 31.9 | 560 | 613 | 703 | 919 |
| 120 | 2 | 4.7 | 9.7 | 18.2 | 38.7 | 560 | 626 | 738 | 1009 |

No Sig'l = Quiescent, Light = Crest Factor of 7.8(18dB),
Average = Crest Factor of 4.8(14dB), Heavy = Crest Factor of 2.8(9dB)
For details of measurement methods please refer to the Technical Support area of our website.

## SPECIFICATIONS (cont'd)

Auxiliary Outputs and DSP Performance

| Parameter (Units) | DX-80I/DX-100I |
|---|---|
| Source Impedance – Active Balanced (Ohms) | < 60 |
| Minimum Load (Ohms) | 600 |
| Maximum Output Level (dBu) | +18 |
| ADC, DAC and DSP sample rate (kHz) | 96 |
| AES accepted sample rates (kHz) | 32-192 |
| Dynamic Range (20Hz-20kHz Unwtd, dB) | >114 |
| Distortion (@1kHz, +10dBm, %) | <0.001 |
| | |
| Signal Processing | |
| Delay up to 650mS (inputs), 650ms(outputs) - independant | |
| Filters Parametrics – 8 Per input / 9 per output<br>    Each parametric can be switched to Bandpass, Allpass, Notch, VariQ,<br>    Shelf and Elliptical response<br>    Phase filtering – 2 degree steps on each input and output | |
| FIR Filtering – available on all outputs – total maximum taps available: 4000* | |
| Crossover Filters (per output)<br>Bessel / Butterworth 6/12/18/24/48dB per octave and Linkwitz-Riley 12/24/48dB per octave | |
| Dynamic EQ**<br>3 bands (per input)<br>Parametric behaviour – 19.7Hz – 32kHz<br>'Q' 0.4 – 128,<br>Max gain automatic gain adjustment 18dB<br>Max ratio (cut above mode) 4:1<br>Max ratio (all other modes) 2:1<br>Attack 70uS – 2.0 Seconds<br>Release 11mS – 3.4 Seconds | |
| Mix matrix mode (auxes and amplifier outputs)<br>Input sends to each output continuously variable from -40.0dB to _15.0dB in 0.1dB steps plus mute | |
| Limiter<br>Threshold +18dBu to -12dBu (auxiliary outputs); +42dBu to +20dBu (amplifier outputs)<br>Attack time 0.1 to 91 milliseconds<br>Release time 2, 4, 8, 16 or 32 times the attack time | |
| Clip/D-max Limiter<br>Look-ahead attack time,Fast, Medium or Slow release times | |
| | |
| *FIR Filtering enabled in later firmware release – 4000 taps with DEQ globally disabled | |
| ** DEQ enabled in later firmware release | |

Your amplifier has two levels of dynamic protection on its outputs – a traditional program limiter, and a peak 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 for the type of signal in the pass-band will result in excessive low frequency distortion. There is provision, within the remote software application, to set 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 corresponding channel's High-Pass filter frequency according to the table below.

| 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) |

### Peak 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.
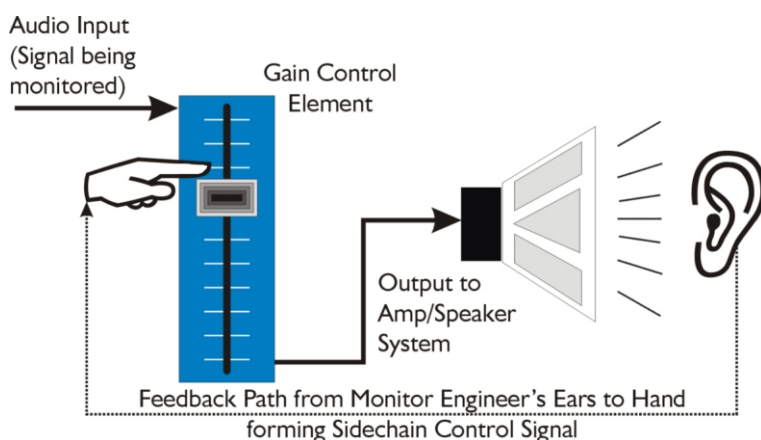


There is a hidden sidechain in operation even in this case. The main signal path is fed through the monitor desk and the gain controlled by adjusting the fader. The sidechain is formed by the feedback path between the engineer's ears checking the level and his brain instructing his hand to turn the fader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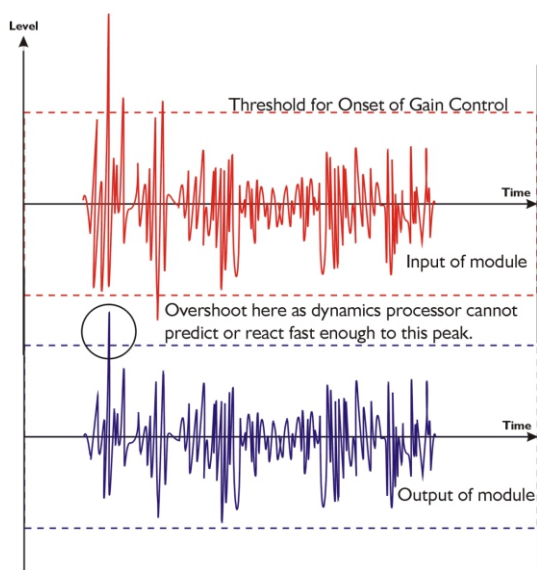


For an analogue dynamics processor, the situation is much better. Controlling the gain electronically, and not relying on a human sidechain feedback mechanism, it can react much more quickly.

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.
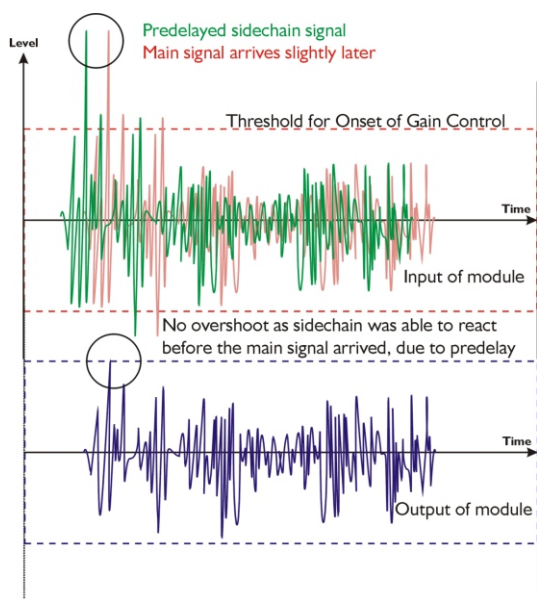
The peak limiter predelays the sidechain signal, resulting in a "zero overshoot" limiter, which is able to catch all peaks and provide a reliable absolute maximum setting for the output of any channel.

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 microseconds, and is a pre-delay – 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 peak limiter follows the RMS limiter, has only two parameters to adjust – the release time 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" means that no more than 10dB of overshoot above the threshold of the program limiter will ever be allowed.

The release time can also be automatically set if the RMS limiter has automatic time constants enabled and so are set by the high pass filter frequency for that channel.

**Setting Accurate Limiter Thresholds – Program Limiter**

The limiters built into your amplifier are intended to be used for loudspeaker driver protection, as opposed to amplifier protection. The amplifier has additional limiters which can adapt automatically to both temperature and incoming mains conditions to stay operational and playing music for as long as possible. 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 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 RMS 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.

**Amplifier Outputs' Program Limiter Lookup Table**

Note that the setting of the auxiliary outputs' limiters need to be referenced to the external amplifier's gain for correct operation. A separate lookup table and explanation is given overleaf.

Consider the table below.

| dB Limit | Power 8Ω | Power 4Ω | Power 2.7Ω[1] | Power 2Ω |
|----------|----------|----------|---------------|----------|
| 48 | 4721 | 9442 | | |
| 47 | 3753 | 7506 | | |
| 46 | 2982 | 5964 | Bridge Mode Operation | |
| 45 | 2370 | 4741 | Not Possible | |
| 44 | 1884 | 3768 | | |
| 43 | 1496 | 2993 | | |
| 42 | 1189 | 2377 | 3396 | 4755 |
| 41 | 944 | 1888 | 2698 | 3777 |
| 40 | 750 | 1500 | 2143 | 3000 |
| 39 | 596 | 1192 | 1702 | 2383 |
| 38 | 473 | 946 | 1352 | 1893 |
| 37 | 376 | 752 | 1074 | 1504 |
| 36 | 299 | 598 | 853 | 1194 |
| 35 | 237 | 474 | 678 | 949 |
| 34 | 188 | 377 | 538 | 754 |
| 33 | 150 | 299 | 428 | 599 |
| 32 | 119 | 238 | 340 | 475 |
| 31 | 94 | 189 | 270 | 378 |
| 30 | 75 | 150 | 214 | 300 |
| 29 | 60 | 120 | 178 | 240 |
| 28 | 47 | 94 | 139 | 188 |
| 27 | 38 | 76 | 113 | 152 |
| 26 | 30 | 60 | 89 | 120 |
| 25 | 24 | 48 | 71 | 96 |
| 24 | 19 | 38 | 56 | 76 |
| 23 | 15 | 30 | 44 | 60 |
| 22 | 12 | 24 | 36 | 48 |
| 21 | 10 | 20 | 30 | 40 |
| 20 | 7.50 | 15 | 22 | 30 |

[1] 2.7Ω is the ideal subwoofer load of 3 x 8Ω drivers in parallel for most efficient power delivery

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

The section in grey will only be relevant when pairs of output channels are bridged. Under these conditions, the available limiter threshold range will increase by 6dB to +48dB.

The minimum impedance for bridged channels is 4Ω, so this section of the table is left intentionally blank. Driving the amplifier into loads lower than those recommended will result in the channels muting and may result in damage to the affected channels.

**Auxiliary Outputs' Program Limiter Lookup Table**

Note that the setting of the amplifier outputs' limiters does not need referenced to the amplifier's gain as this is predetermined for the internal power amplifier sections. A separate lookup table and explanation is given on the previous page.

Consider the table below.

| dB | Ratio | Vrms | Pwr 8Ω | Pwr 4Ω | Pwr 2.7Ω | Pwr 2Ω |
|----|-------|------|--------|--------|----------|--------|
| 45 | 177.83 | 137.74 | 2371.71 | 4743.42 | 7027.28 | 9486.83 |
| 44 | 158.49 | 122.77 | 1883.91 | 3767.83 | 5881.95 | 7535.66 |
| 43 | 141.25 | 109.41 | 1496.45 | 2992.89 | 4433.92 | 5985.79 |
| 42 | 125.89 | 97.52 | 1188.67 | 2377.34 | 3512.98 | 4754.68 |
| 41 | 112.20 | 86.91 | 944.19 | 1888.39 | 2796.60 | 3776.78 |
| 40 | 100.00 | 77.46 | 750.00 | 1500.00 | 2222.22 | 3000.00 |
| 39 | 89.13 | 69.04 | 595.75 | 1191.49 | 1765.18 | 2382.98 |
| 38 | 79.43 | 61.53 | 473.22 | 946.44 | 1042.13 | 1892.87 |
| 37 | 70.79 | 54.84 | 375.89 | 751.78 | 1113.75 | 1503.56 |
| 36 | 63.10 | 48.87 | 298.58 | 597.16 | 884.68 | 1194.32 |
| 35 | 56.23 | 43.56 | 237.17 | 474.34 | 702.43 | 948.68 |
| 34 | 50.12 | 38.82 | 188.39 | 376.78 | 556.19 | 753.57 |
| 33 | 44.67 | 34.60 | 149.64 | 299.29 | 443.38 | 598.58 |
| 32 | 39.81 | 30.84 | 118.87 | 237.73 | 352.21 | 475.47 |
| 31 | 35.48 | 27.48 | 94.42 | 188.84 | 279.76 | 377.68 |
| 30 | 31.62 | 24.49 | 75.00 | 150.00 | 222.22 | 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 this gain figure FROM 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.

As an example, for a subwoofer rated at 2000W and 4Ω, working with an amplifier which has 32dB of gain, the limiter threshold would be calculated as follows:

"First, check the RMS power rating of the speaker system, and its impedance." 2000W, 4Ω
"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." 41dB
"Check the gain of your amplifier, which needs to be in 'dB'." 32dB
"Subtract this gain figure FROM that obtained from the table to find the required absolute setting for the limiter thresholds." 41 – 32 = +9dB
"Note that, for safety, always set the limiter threshold 1 or 2 dB below the maximum allowable worked out using the above method." with safety, +8dB

**Setting Accurate Limiter Thresholds – Peak Limiter**

Assuming the RMS limiter has been set correctly and, just as importantly, attack and release times have been chosen as appropriate to the driver to be protected, the peak limiter is typically set to limit overshoot to 3dB above the RMS limiter threshold.

This would allow peaks of twice the RMS power level to reach the outputs. If the driver has a peak power capability of more than double the rated RSM power, then this value can be increased.

To calculate the setting for the peak limiter it's:
$$10 \times (Log_{10}(Peak\ Power\ /\ RMS\ Power))$$

So for example, a 15" driver has a quoted RMS power handling of 800W, and a peak power handling of 1600W, the calculation is
$$(1600/800) = 2$$
$$Then\ Log_{10}(2) = 0.3010$$
$$Then\ 10 \times 0.3010 = 3.010\ or\ 3dB$$

Speaker manufacturers may quote AES power in place of RMS power and "Program" instead of "Peak". These terms, whilst not strictly interchangeable, are similar as a "pair" of measurements. AES tends to be a slightly more conservative rating given the definition of how it is measured.

If AES power is quoted, then it normally is paired with the "Program" rating and so the calculation of the threshold for the peak limiter is still valid.

**Setting Appropriate Attack and Release Times**

As stated earlier in this appendix, having control over the attack and release times of the program limiters 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. When setting limiter attack and release times during the crossover configuration there is an option for automatic limiter time constants. Use this option if you are unsure how to set the time constants manually. See page 27 for details on how to turn this option on.

We recommend the use of the automatic setting.

In this mode the time constants will be automatically set from the high pass crossover filter frequency according to the table below:

| High Pass Filter | Auto Attack Time | Auto 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) |

Only the release time may be adjusted for the peak limiters, as attack time is always set to "zero-overshoot" and so cannot be changed. The release time may be set to "slow", "medium" or "fast" – we recommend using the automatic setting which is selected for both limiters at the same time as part of the crossover configuration, detailed on page 27.

## ANNEX II:  UPGRADING FIRMWARE AND LOADING PRESETS

Loading new firmware or presets uses the same "Flash Loader" utility that is bundled with the firmware or presets "bin" (binary) file. Firmware (or Program) files have the prefix "LTAM", and preset files have the prefix "LTAR".

**Amplifier Communications Set-up**

Updates can be performed through an Ethernet, US or RS485 connection. Choose the correct port on the amplifier and configure it accordingly though the Interface Sub-menu. See page 35 for more info on choosing the interface.

**Download the Files**

Go to DX-80I / DX-100I related downloads to find the latest firmware or presets and software:

http://www.dasaudio.com/en/p/dx-80i-3/

Products > Amplifiers > DX Series Amplifiers > DX-80i

# DX-80I

**FEATURES**

**TECHNICAL DATA**

**DOWNLOADS**

**Configuration Guide using DX Amplifiers**
Configuration guide using DX series amplifiers with SF, HQ, Artec, UX and WR series products.

**Photo DX Family**
High Resolution, 1.8 MB

**Photo DX-80i**
High Resolution, 0.7 MB

**DX Audiocore Software**
Audiocore V9.23 / DX Flashloader V5.52

**DX Firmware files**
DX Firmware V1.20 files

**DX Firmware update**
DX Firmware V1.20 update

**DX-80i Spec Sheet**
English, 2 MB

Other products
of this series:
Series

Unzip the files – you will have the loader application, and the firmware or preset file as appropriate.

**Connect To The Amplifier**

Plug in the appropriate cable according to your chosen connection method. Remember – if you are connecting via an USB or RS485, you will need to make sure your amplifier is set to ID1 for the loader to find                                                                                                                it .

Start the loader application:
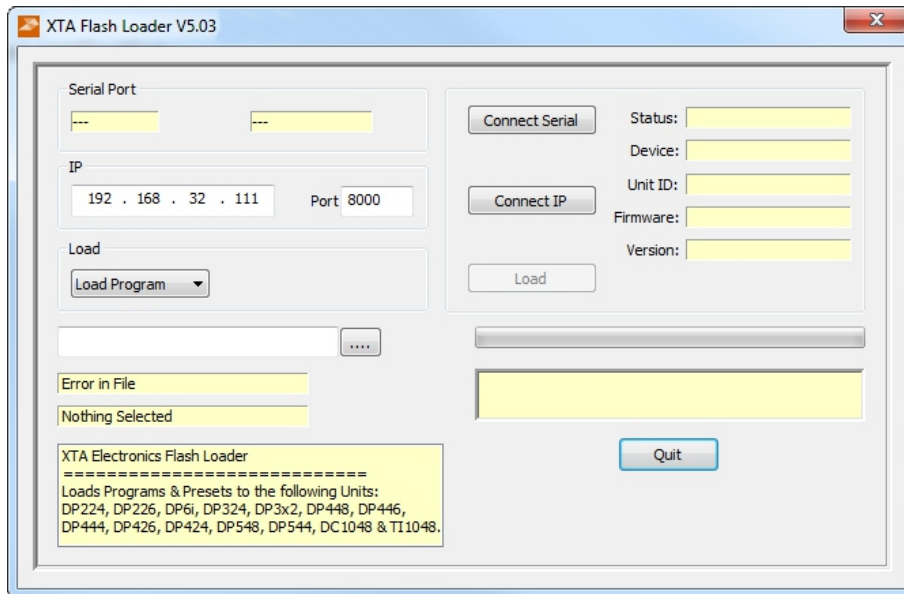


Press the "Connect Serial" button if using USB or RS485, or "Connect IP" if using an Ethernet connection.  We recommend you choose either a USB or wired Ethernet connection.

After a few seconds, the unit should be located and its status and type should be displayed, along with the current firmware and some other information.

**Choose What To Load**

From the drop-down list under "Load" choose either "presets" or "program" (firmware), and then locate the correct file by pressing the "…" button.
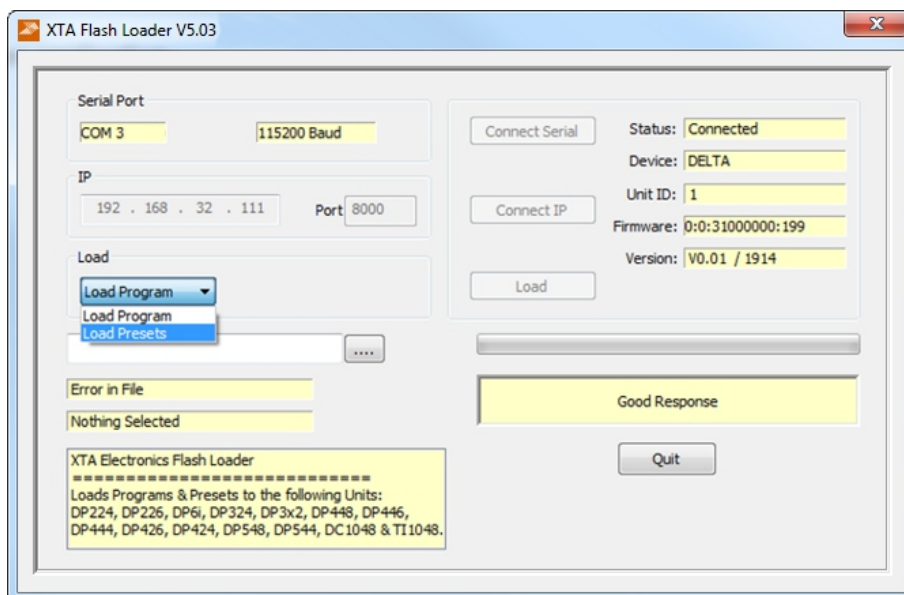


The file type will have been filtered for you – LTAM*.bin for firmware (program) files or LTAR*.bin for preset files.

**Start The Update!**

**Just before you press the "Load" button – please close any other applications running on the PC – interruptions in the process could render your amplifier inoperable. Do not disconnect the cable, turn off the power or attempt to close the loader app once loading has begun!**

Now that you have been warned (!) – press the "Load" button.  Confirm the load, and the amplifier will switch to Standby and begin to load the new firmware. A message detailing progress will accompany the bar on the loader app:

Remote Load  374B
Block 0 of 4096

Upon completion, the app will close and your amplifier will restart – that's it!

**What If It All Goes Wrong...**

Don't panic!  Your amplifier can be recovered, but please be aware that…

THE AMPLIFIER OPERATES AT VOLTAGES IN EXCESS OF THE INCOMING MAINS VOLTAGE, AND THESE VOLTAGES REMAIN PRESENT FOR AN EXTENDED PERIOD AFTER MAINS POWER IS REMOVED!

READ THESE INSTRUCTIONS CAREFULLY AND HEED ALL WARNINGS!

Disconnect your amplifier completely from the incoming mains supply.

Ensure that the HT present LEDs visible here through the front left vent with the foam removed…
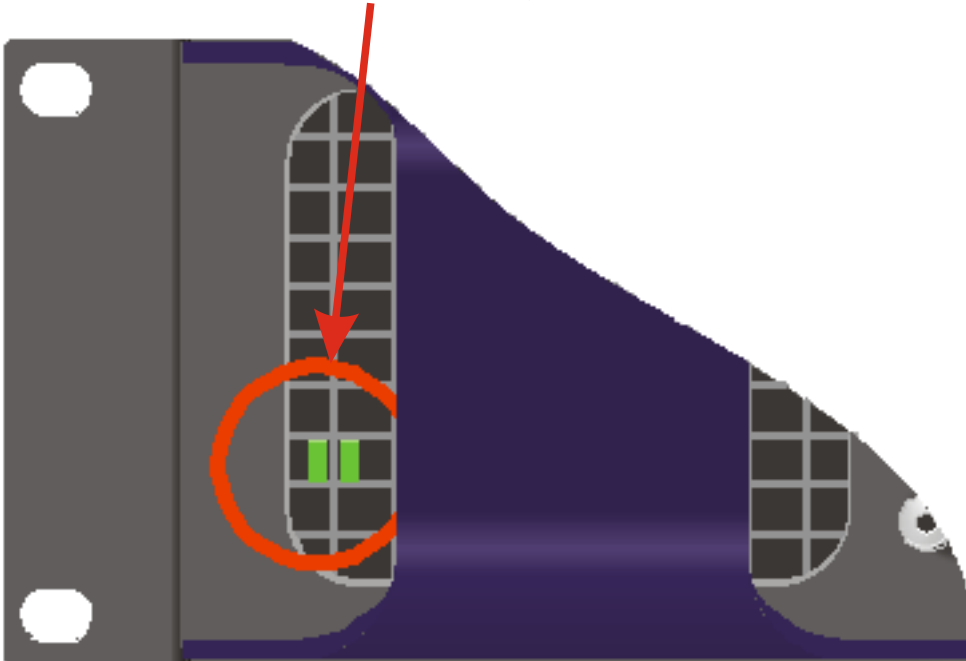


…are fully extinguished before continuing. This will take at least five minutes so please be patient for your own safety.

It is now safe to remove the top cover of your amplifier – there are four screws (Torx) along each side and eight on the top of the cover. Be careful not to drop any screws inside!

Locate the "Rescue Recovery" link CN11 – between ICs U24 and U28 on the top circuit board - and remove it. This will make your amplifier run in recovery mode – the original firmware version loaded at the factory will now be active.
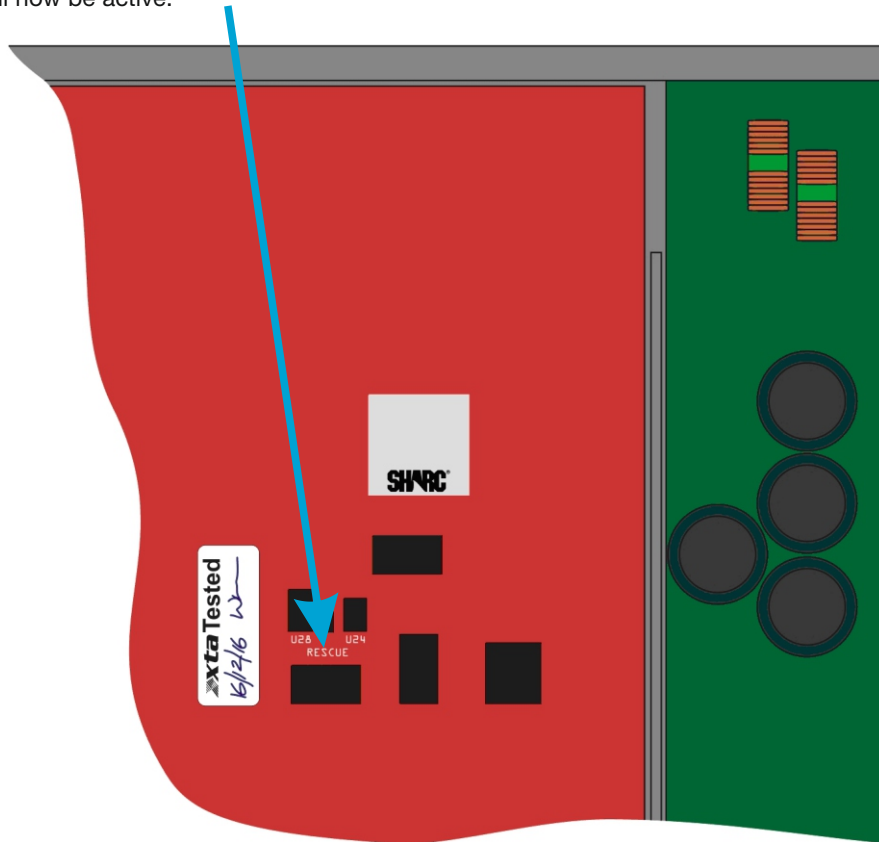


Replace the top cover temporarily, reapply mains power and switch on. Once booted, check the interface is set as required and perform a program update again using the Flash Loader application.

The display will show an additional "R" next to the Remote Load message to show rescue mode is active:

```
Remote Load  R374B
Block 0 of 4096
```
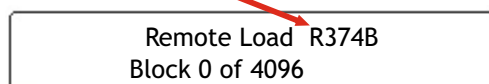
Upon completion, the app will close.   Disconnect the power and wait for at least five minutes until the HT Present LEDs have fully extinguished again.

Remove top cover and replace the "Rescue Recovery" link once more.

Replace top cover and screw down, and replace the filter foam in the left hand front vent.

Your amplifier should now be fully operational again – no settings will have been lost in the process.

**www.dasaudio.com**

UM_DXI_01_EN

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