DP648

Audio Management System Operators Manual

Version 0.4

- 4 Input 8 Output System Processor
- 4 Inputs Patchable from 12 Input Sources
- Analog/ AES/ EBU and optional Dante/ AES67
- Automatic Input failover
- 8 x Outputs : Analog / AES/ EBU and optional Dante/ AES67 card
- Flexible Dynamic Processor - FIR Filtering - Extensive Speaker Preset Library


The DP648 builds on the legacy of the SIDD, C2, D2 processing, the XTA DP448 and DPA amplifiers. Providing ethernet control of this suite of IIR, FIR and dynamic processing essential for sound system management. The Dante/ AES67 option and flexible routing make it a comprehensive system processor, for overall system management or for traditional loudspeaker management.

The DP648 works as a system processor for any audio system, the XTA system scales for more advanced control when partnered with other XTA products. This includes the MX36 console/ source switcher, DPA and DNA amplifiers which come with or without dsp.

***Series 6** Comprehensive Audio System Processing*

CONCERT ARENA THEATRE HoW

www.xta.co.uk

CONTENTS

DECLARATION OF CONFORMITY

We, the manufacturer:

XTA Electronics Limited,
The Design House
Vale Business Park
Worcester Road
Stourport on Severn
Worcestershire
England
DY13 9BZ

acknowledge our responsibility that the following products:

Kind of equipment:	**Audio processor**
Commodity Code:	**8518408099**
Type Designation:	**DP648**
	and all OEM variants of these models

are manufactured:

in accordance with EMC Directive 2004/108/EC,
in compliance with the following norm(s) or document(s):
Technical Regulations: EN55103-1:1996, EN55103-2:1996

and

in accordance with the Low Voltage Directive 2006/95/EC,
in compliance with the following norm(s) or document(s):
Technical Regulations: EN/IEC60065:2002 7th Edition

Signed: ....

Name: Alex Cooper
Position: Research and Development Manager
Date: June 2017

INTRODUCTION

The DP648 has been designed to combine performance with ultra-flexible connectivity for both remote control and audio. Exemplary audio processing is assured through the use of XTA's DSP platform.

Accepting analogue, AES3 digital and optional Dante / AES67 networked audio, the DP648 can connected to assorted sources and make them available over the network audio, aes3 and analog outputs,
Connectivity for remote control is covered Ethernet and GPi. Configuration of the processor's is through the industry standard AudioCore Amped Edition application and globcon XTA MC2 edition,

THANKS

Thank you for choosing a DP648 processor 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 XTA products are carefully designed and engineered for cutting-edge performance and world-class reliability. If you would like further information about this or any other XTA product, please contact us.

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

IMPORTANT SAFETY INSTRUCTIONS

CAUTION: RISK OF ELECTRIC SHOCK.
DO NOT OPEN

The lightning flash with arrowhead symbol within an equilateral triangle is intended to alert the user to the presence 0f uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons.

The exclamation mark 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.

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

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use this apparatus near water.
6. Clean only with a dry cloth.

7. Do not block any ventilation openings, install in accordance with the manufacturer's instructions.

8. Do not install near any heat sources, such as radiators, heat registers, stoves or other apparatus (including processors) that produce heat.

9. Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding-type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.

10. Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles and the point where they exit from the apparatus.

11. Only use attachments/accessories specified by the manufacturer.

12. 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.

13. Unplug this apparatus during lightning storms or when unused for a long period of time.

14. 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. This equipment contains a non-user replaceable lithium battery for memory retention. Should this battery fail and user settings be lost, do not attempt to replace the battery yourself but return the equipment to an authorised service centre.

15. Do not expose this equipment to dripping or splashing and ensure that no objects filled with liquids, such as vases, are placed on the equipment.

16. To completely disconnect this equipment from the AC mains, disconnect the power cord from the mains circuit breaker.

17. This unit is fitted with a 3-wire power cord. For safety reasons, THE EARTH LEAD SHOULD NOT BE DISCONNECTED IN ANY CIRCUMSTANCE.

18. Correct disposal of this product: This symbol indicates that this product must not be disposed of with household waste, according to the WEEE Directive (2012/19/EU) and your national law. This product should be taken to a collection center licensed for the recycling of waste electrical and electronic equipment (EEE). The mishandling of this type of waste could have a possible negative impact on the environment and human health due to potentially hazardous substances that are generally associated with EEE. At the same time, your cooperation in the correct disposal of this product will contribute to the efficient use of natural resources. For more information about where you can take your waste equipment for recycling, please contact your local city office, or your household waste collection service.

INSTRUCTIONS DE SECURITE IMPORTANTES

ATTENTION: RISQUE DE

CHOC ELECTRIQUE.
NE PAS OUVRIR

Le symbole représentant un éclair fléché dans un triangle équilatéral a pour but d’alerter l’utilisateur de la présence d’une “tension dangeruese” non isolée à l’intérieur du boitier, pouvant être d’une force suffisante pour constituer un risqué d’électrocution.

Le point d’exclamation dans un triangle équilatéral a pour but d’alerter l’untilisateur de la présence d’instructions importantes concernant le fonctionnement et la maintenance, dans la documentation qui accompagne l’appariel.

ATTENTION: Appareils de construction de CLASSE I doit être raccordé au réseau électrique via une prise de courant reliée à la terre.

ATTENTION: Pour éviter toute blessure, cet appareil doit être solidement fixé à la torture, conformément aux instructions d'installation.

1. Lisez ces consignes.
2. Conservez ces consignes.
3. Respectez tous les avertissements.
4. Respectez toutes les consignes d’utilisation.
5. N’utilisez jamais l’appareil à proximité d’un liquide.
6. Nettoyez l’appareil avec un chiff on sec.

7. Veillez à ne pas empêcher la bonne ventilation de l’appareil via ses ouïes de ventilation. Respectez les consignes du fabricant concernant l’installation de l’appareil.

8. Ne placez pas l’appareil à proximité d’une source de chaleur telle qu’un chauff age, une cuisinière ou tout appareil dégageant de la chaleur (y compris un ampli de puissance).

9. Ne supprimez jamais la sécurité des prises bipolaires ou des prises terre. Les prises bipolaires possèdent deux contacts de largeur diff érente. Le plus large est le contact de sécurité. Les prises terre possèdent deux contacts plus une mise à la terre servant de sécurité. Si la prise du bloc d’alimentation ou du cordon d’ali-mentation fourni ne correspond pas à celles de votre installation électrique, faites appel à un électricien pour eff ectuer le changement de prise.

10. Installez le cordon d’alimentation de telle façon que personne ne puisse marcher dessus et qu’il soit protégé d’arêtes coupantes. Assurez-vous que le cordon d’alimentation est suffisamment protégé, notamment au niveau de sa prise électrique et de l’endroit où il est relié à l’appareil; cela est également valable pour une éventuelle rallonge électrique.

11. Utilisez exclusivement des accessoires et des appareils supplémentaires recommandés par le fabricant.

12. Utilisez exclusivement des chariots, des diables, des présentoirs, des pieds et des surfaces de travail recommandés par le fabricant ou livrés avec le produit. Déplacez précautionneusement tout chariot ou diable chargé pour éviter d’éventuelles blessures en cas de chute.

13. Débranchez l’appareil de la tension secteur en cas d’orage ou si l’appareil reste inutilisé pendant une longue période de temps.

14. Les travaux d’entretien de l’appareil doivent être eff ectués uniquement par du personnel qualifié. Aucun entretien n’est nécessaire sauf si l’appareil est endommagé de quelque façon que ce soit (dommages sur le cordon d’alimentation ou la prise par exemple), si un liquide ou un objet a pénétré à l’intérieur du châssis, si l’appareil a été exposé à la pluie ou à l’humidité, s’il ne fonctionne pas correctement ou à la suite d’une chute. Pour la mémorisation des paramètres, cet appareil contient une pile au lithium non remplaçable par l’utilisateur. En cas de défaillance de la pile et perte des réglages, n’essayer pas de remplacer la pile par vous-même. Retourner votre appareil vers une station technique habilité.

15. N'exposez pas cet équipement au fait de tomber goutte à goutte ou au fait d'éclabousser et garantissez qu'aucun objet rempli des liquides, comme les vases, n'est placé sur l'équipement.

16. Pour complètement débrancher cet équipement de la conduite principale de courant alternatif, débranchez la corde de pouvoir du disjoncteur de conduite principale.

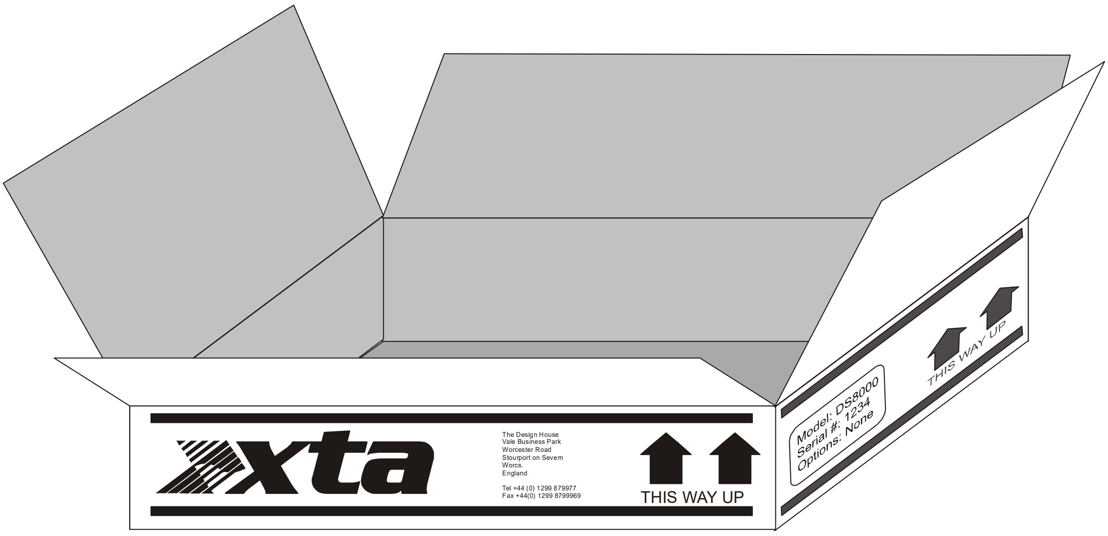
17. Cette unité est correspondue avec une corde de pouvoir de 3 fils. Pour les raisons de sécurité, L'AVANCE DE TERRE NE DEVRAIT ÊTRE DÉBRANCHÉE DANS AUCUNE CIRCONSTANCE.

18. Mise au rebut appropriée de ce produit: Ce symbole indique qu’en accord avec la directive DEEE (2012/19/EU) et les lois en vigueur dans votre pays, ce produit ne doit pas être jeté avec les déchets ménagers. Ce produit doit être déposé dans un point de collecte agréé pour le recyclage des déchets d’équipements électriques et électroniques (EEE). Une mauvaise manipulation de ce type de déchets pourrait avoir un impact négatif sur l’environnement et la santé à cause des substances potentiellement dangereuses généralement associées à ces équipements. n même temps, votre coopération dans la mise au rebut de ce produit contribuera à l’utilisation efficace des ressources naturelles. Pour plus d’informations sur ’endroit où vous pouvez déposer vos déchets d’équipements pour le recyclage, veuillez contacter votre mairie ou votre centre local de collecte des déchets.

Intentionally Blank

Installing Your Processor: Unpacking

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



Additional Symbols and Warnings

只有在高海拔地区使用不超过2000米。

Meaning of the symbol: Evaluation for apparatus only based on altitude not exceeding 2000m, therefore it is the only operating condition applied for the equipment. There may be some potential safety hazard if the equipment is used at altitude above 2000m.

只适合于非热带气候地区使用

Meaning of the symbol: Evaluation for the apparatus only based on temperate climate condition, therefore it is the only operating condition applied for the equipment. There may be some potential safety hazard if the equipment is used in tropical climate region,

Installing Your Processor: Electrical Considerations

The processor 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 processor is fitted with universal supply 100-240V power supply.

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

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

Where the processor 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.

For safety reasons,

THE EARTH LEAD SHOULD NOT BE DISCONNECTED IN ANY CIRCUMSTANCE.

If ground loops are encountered consult the section on connecting your processor on page 14.

The wiring colours are:

230V AREAS: EARTH = GREEN AND YELLOW
NEUTRAL = BLUE
LIVE = BROWN

DO NOT USE THE UNIT IF 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.

Terminals marked with the ⚠ symbol are HAZARDOUS LIVE – external wiring connected to these terminals requires installation by an INSTRUCTED PERSON or the use of ready-made leads or cords.

Installing Your Processor: 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 processors, it should be kept away from other equipment which is sensitive to magnetic fields. Also, this processor 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 processor in a rack or enclosure:

Be aware that...

ENSURE THERE IS ADEQUATE VENTILATION.
The cooling fans (if fitted) suck cool air in through the left and blow hot air out at the side of the unit. The side of the processor 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).

Installing Your Processor: RF Emissions

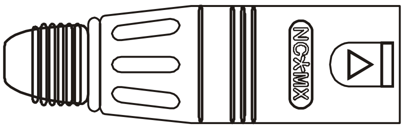
The high frequency resonant converters in the DP648 processor have been designed to have very low radio frequency (RF) emissions; however even these low level emissions can cause interference with other equipment.

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

Connecting To Your Processor: 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:

XLR MALE



PIN1: Screen (see note below)
PIN2: Hot (signal +)
PIN3: Cold (signal -)

The processor 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.

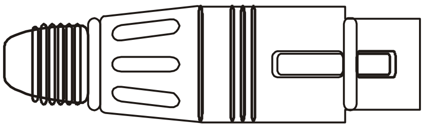
PIN1: Screen - connected to the chassis of the unbalanced equipment - or left disconnected at the unbalanced end.
PIN2: Hot (signal +)
PIN3: Cold (ground 0V)

NOTE: This processor is wired to the latest industry recommendations. PIN1 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) CONNECTIONS INSIDE THE PROCESSOR.

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

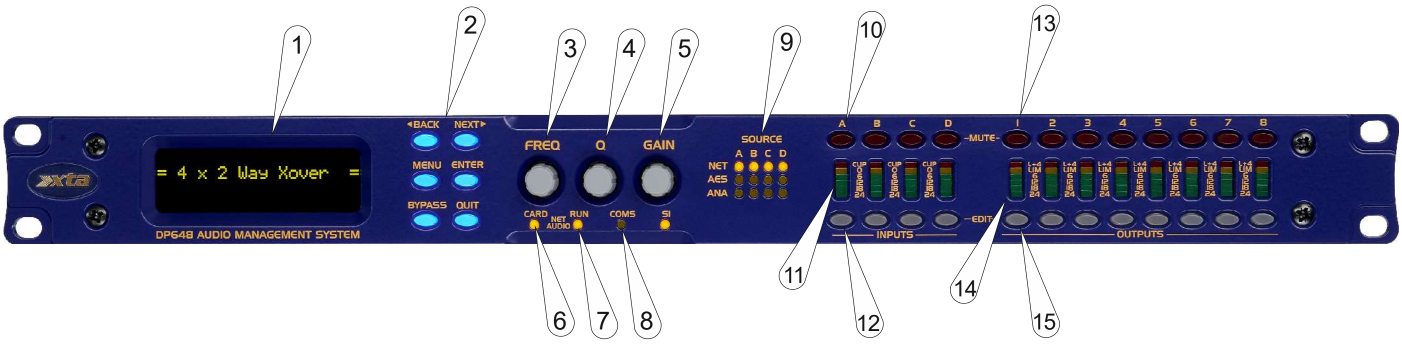
XLR FEMALE



PIN1: Screen (see note above)
PIN2: Hot (signal +)
PIN3: Cold (signal -)

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

Operating Your Processor: Front Panel Controls and Indicators



1: OLED Display :

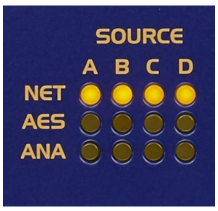
2: Main User Interface: Menu - Back - Next – Enter – Quit these button allow the user to navigate various menus.
NEXT moves forward through a list of parameter or menus
BACK move backward through a list of parameter or menus
The Frequency encoder can also be to move back and forth through the menu system.
ENTER key selects the chosen menu, confirms selections and allows filter types to be changed.
MENU – activates the main menu – a second press selects the last menu edited – a third press selects the last menu item. In this way, three presses on MENU from the default screen will jump back to the last parameter adjusted.
Selection of different menus is accomplished using the BACK and NEXT keys, or with rotating the FREQ encoder.

The key menus are `Memory',`Configuration', `Interface`, `System`and `Security. Pressing Enter will allow you drill down to the next level of menus. You can quit a menu at any time should you make an error or change your mind.

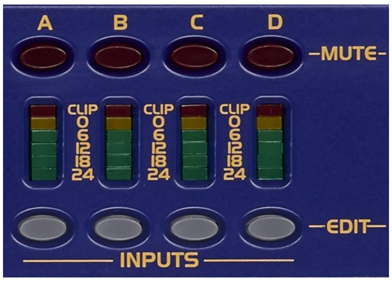
3; FREQ
4: Q
5: GAIN These are three velocity sensitive encoders which adjust relevant parameters on the screen.

6: & 7: CARD – NET AUDIO – RUN : These LED indicate the Network audio status.
The CARD leds indicate the unit is fitted with an Audio Network card, you will also see 2 x additional Network Audio RJ45 connectors on the rear when fitted. Normally only 1 x Control ethernet connector is fitted
If the CARD led does not light and there is no Network audio ethernet connectors on the rear, the unit can be upgraded by contacting your local dealer/ distributor to purchase one.

The `RUN' led indicates there is currently a network audio connection and it is available to patch audio from.
i.e. There is cat5 connected one to one of the rear audio network ethernet connections(e.g. Dante or AES67)
8:COMS : The coms led will illuminate when in communication with remote control software such a AudiocoreAmped edition or globcon.



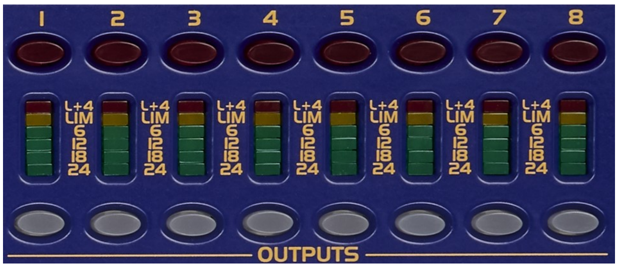
9: Source Matrix
The DP648 Input processing can be patched from any of the available input sources, this led array indicates what source type is patched to Input A,B,C and D.
In the Configuration, Source selection menu - primary and back up sources can be selected.
Should a digital primary input fail or be disconnected from the processor it will automatically switch to the back up (if one is selected). This will be indicated by a flashing primary led at the appropriate crosspoint, the selected backup will then illuminate. This provides quick, at a glance status of source selection.
A NET or AES illuminating indicates AES is connected or Dante connection is valid – it does not indicate the presence of Audio signal, only valid Data.
The Dante / AES67 card is an option, if fitted the CARD – NET AUDIO led will illuminate.



Input Sections: Control and monitor input signal paths.

Red MUTE buttons illuminate when pressed and mute audio for that channel.
EDIT buttons illuminate yellow when pressed, and access gain on first press, then last viewed parameter on second press, then exit on third press.
Input meters show dB from clipping point of the analogue to digital converters.
Yellow (0dB) LED illuminates 3dB from clipping.

Red CLIP LED may illuminate independently from the rest of the meter to show digital overflow.
All four CLIP LEDs illuminating indicates internal clipping after the ADC.



Output Sections: Control and monitor output signal paths.

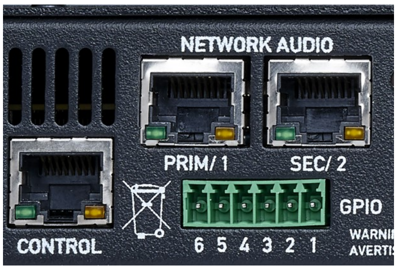
Red MUTE buttons illuminate when pressed and mute audio for that channel.
EDIT buttons illuminate yellow when pressed, and access gain on first press, then last viewed parameter on second press, then exit on third press.
Output meters show dB from limiting.
The yellow LED illuminates at the onset of limiting.

The red LED illuminates at 4dB into limiting (i.e. 4dB of gain reduction).

Rear Panel Sockets and Switches

Power Switch: turns the unit's mains supply off and on.

Mains Fuse: located in a finger-proof holder adjacent to the mains inlet. A spare fuse is also located in this holder. Mains Inlet: connected via a standard IEC socket.



NETWORK AUDIO connections[1]: Should the DP648 be fitted with an Audio network card (Dante / AES67) it will have 2 x network audio ethernet connection. IF fitted this adds four additional inputs available to the input matrix.
This will adds four inputs and eight network audio outputs. For more on this feature see the block diagram on page 17 and set-up information from page 48.

CONTROL – use this port for remote control using Audiocore Amped Edition or globcon. It also be used for firmware updates and preset loading.

7: GPIO Port: Your processor has a pair of general purpose logic level input and outputs that can be configured to recall memories, put the processor in standby, mute and control levels. See appendix starting on page 41 - 43 for more information.

[1] The audio network card is an option and may not be fitted to your pro

Audio Inputs:

3 pin female XLR sockets are provided for each channel. All are fully balanced. Connect signal inputs to these sockets, wired pin 2 hot, pin 3 cold, 1 ground. For sensitivity and impedance of these inputs, please see the specifications on page 72. Inputs C & D may also be switched to AES3 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 36.

Audio Outputs: 3 pin XLR sockets are provided for each channel. All are fully balanced, pin 2 hot, 3 cold, 1 screen. Note the labelling on the panel to designates which outputs are used for AES streams when the individual AES modes are enabled. Please see page 33 for more information.

Operating Your Processor: Initial Set-up and Switching On

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

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

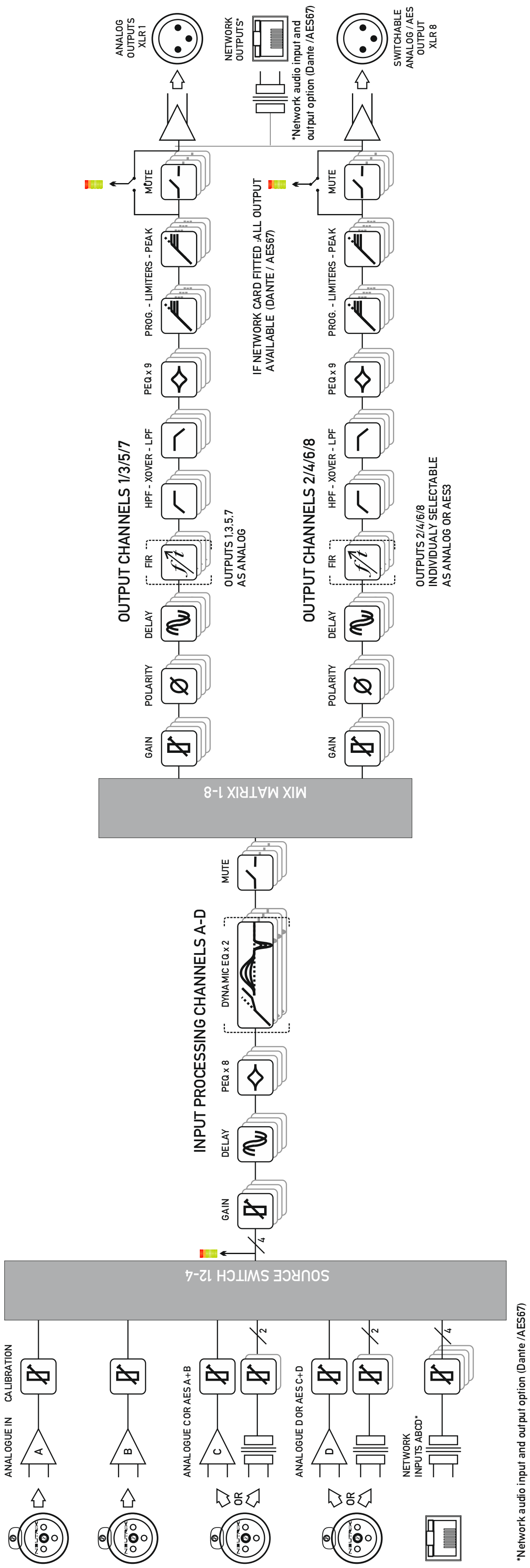
Do not block fan outlet or side ventilation inlets or operate the unit in an environment that could impede the free flow of air around the unit, for example a sealed rack.

If your processor is used in an extremely dusty or smoky environment, it should be cleaned of any collected debris at regular intervals.

Switching On...

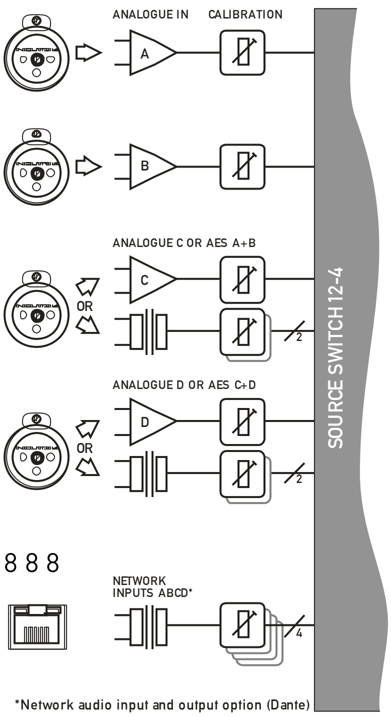
The first time you switch it on, your processor will start-up as a four analogue inputs to eight output system, 4 x 2 way configuration, with no EQ or limiting (apart from self-protective limiters) in place.
The following section explains the DSP and audio features of the processor – please read this carefully as the routing options are very comprehensive!

Inside Your Processor: DP648 Block Diagram



Inside Your Processor: DSP Layout, Configurations and Routing

Source Choices and Selection



*Network audio input and output option (Dante)

Your processor can source audio from analogue, AES digital, or network locations (if a network card is fitted).

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 or analogue inputs C&D.

AES input for channels A&B is on the XLR socket for channel C.
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 automatic failover from one to the other possible without repatching.

Assuming the network audio option is also fitted, four additional digital sources will then be available.

8 sources can be made available at any one time – four analogue and four digital. If an network card is fitted an additional 4 sources are available This affects the choices that can be made for routing to the inputs of the DSP channels.

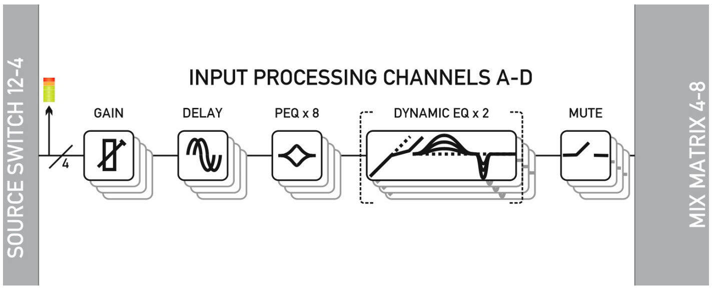
The rules are as follows:

- 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);

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.

Please see page 31 for further info on using the AES and Input source selection menu options.

Inside Your Processor: DSP Layout, Configurations and Routing
Input Processing Channels



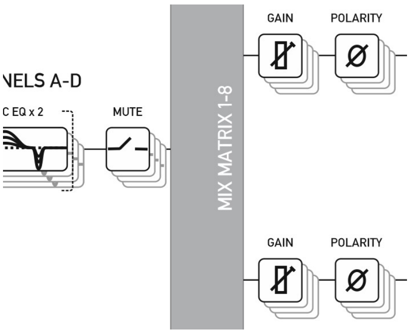
Your processor has four input channels of processing that can be fed from a variety of sources.
These four channels in turn, feed a mix matrix for routing to the eight output processing channels.

Each input processing channel consists of the following sections:

Input gain control
Input delay time
Input parametric EQ bands 1 through 8
2 Bands of Dynamic EQ
Input Mute

The dynamic EQ sections allow boost above, boost below, cut above and cut below modes.
They can be configured as full bandwidth, hi shelf, low shelf and PEQ mode.
This can be configured on the front panel or by remote control.
DEQ can be enabled in the Configuration .....DEQ Active sections menu.
For more information on adjusting input processing parameters, please see the Appendix II beginning on page 60.

Mix Matrix Section

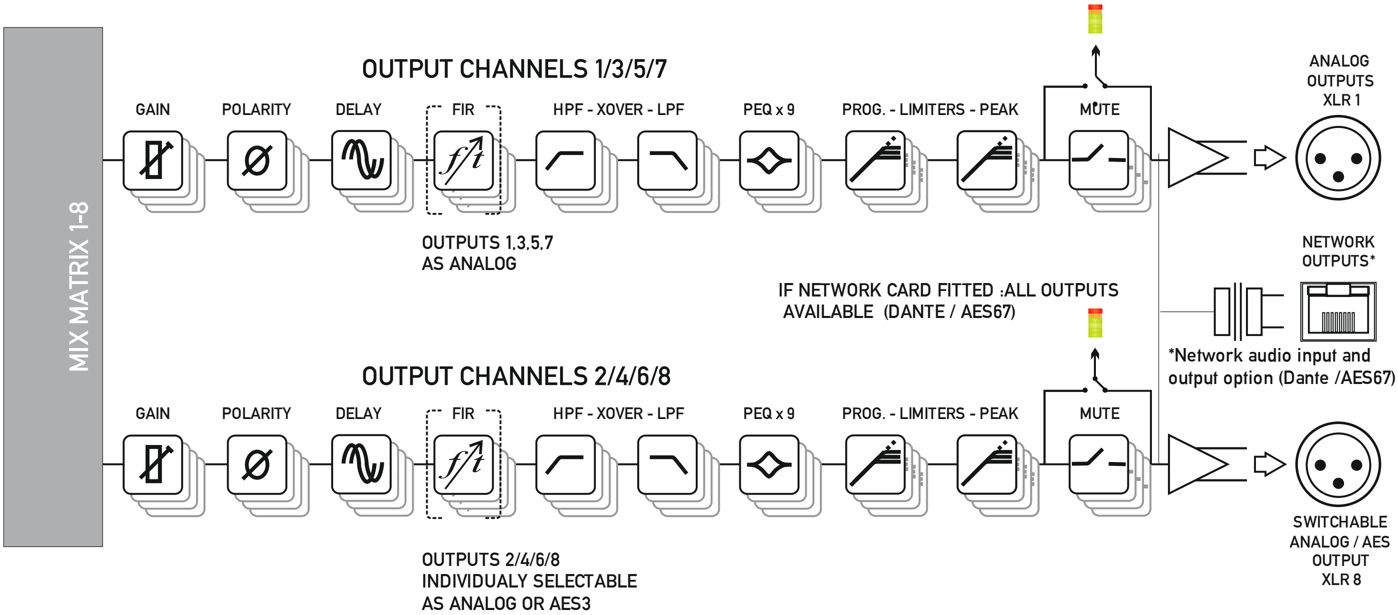


The mix matrix allows four independent mixes to be set up for the 8 output channels.

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 matrix gains on page 34. Note that the mix matrix will only be available if it has been selected as part of the `output routing' in the Configuration menu – see page 34 to change this.

Inside Your Processor: DSP Layout, Configurations and Routing
Output Processing Channels



There are eight identical channels of output processing in your processor, these provided analogue XLR outputs to other equipment. XLR Outputs 2, 4, 6 and 8 can be switched to AES3 outputs carrying channel pairs.
If fitted with an audio network card the 8 outputs are presented on the audio network for patching to other units e.g. via Dante/AES67

Each output processing channel consists of the following sections:

- Output delay time
- FIR processing
- 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

The DP648 offers extensive FIR filtering capabilities. FIR filtering is not user adjustable and is typically part of OEM presets only, a preset designer is available on request. For more information on adjusting output processing parameters, please see the section beginning on page 24.

Operating Your Processor: Menu structure overview

Pressing the Menu key, then using BACK , NEXT and the GAIN encoder allows you to navigate the processor menu structure. The Channel EDIT key (orange / clear button), allows access to individual output parameters. Below is an overview of the menu structure, after which follows a section on individual parameter adjustments.

Memory Menu	What it's for...
Recall Output Type	Recall an user output memory or Preset
Recall Input Type	Recall a user Input Memory
Recall Source Type	Recall a user Source Memory
Recall Everything	Recall a user Everything Memory
Store Output Type	Store / Overwrite / Rename a User Output Memory
Store Input Type	Store /Overwrite / Rename a User Input Memory
Store Source Type	Store /Overwrite / Rename a User Source Memory
Store Everything	Store /Overwrite / Rename an Everything Memory
Erase a Memory	Erase a user memory

Configuration Menu	What it's for...
AES Input Selection	Switch Input XLR C and or D from Analog to AES
Source Selection	Select Analog, AES, Dante routing to Input ABCD (inc Failover options)
Input Ganging	Enable ganging of input processing parameters
DEQ Active Sections	Enable 0,1 or 2 bands of Input Dynamic EQ
Output Routing	Adjust ABCD routing to outputs 1 to 8.
Output Ganging	Enable Ganging of Output processing parameters
AES Output Select	Individually Switch Output XLR 2,4,6,8 to AES outputs

Interface Menu	What it's for...
IP Control Options	Configure control ethernet Interface i.e. ID, static or dhcp mode.
Remote	Enable or Disable remote control via Audiocore Amped Edition / globcon XTA MC2 Edition
GPIO Options	Enable and Configure GPIO ports
Dante Options	Enable Dante cards options. EG AES67 mode
OSC Options	Enable & Configure OSC ports, OSC control panel IP address

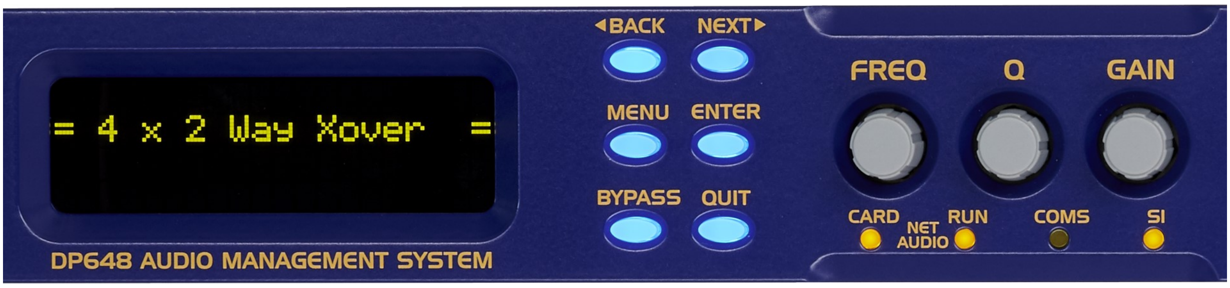
System Menu	What it's for...
System Status	Check firmware version and system information
LEDBrightness	Adjust the viewing angle for the LED
Display Brightness	Adjust the brightness display
Wakeup Time	Add a delay before start of audio fade up or keep muted
Output Meter Option	Choose if meters should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

Security Menu	What it's for...
Unit Locking	Enable / Configure parameter or complete unit locking. Configure password to unlock.

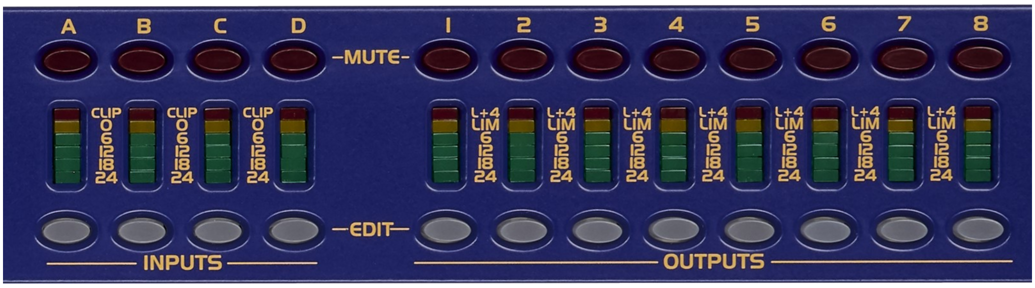
Operating Your Processor: Directly Editing Audio Parameters

Overview

Editing all audio parameters is available from the front panel of your processor using a combination of the channel EDIT keys, and the BACK/NEXT/ENTER navigation controls.



First, select the channels you wish to adjust –
 either inputs A-D or outputs 1-8 by pressing the corresponding channel EDIT.



Initially the GAIN will be displayed for the selected channel – this can be adjusted using the GAIN encoder.

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

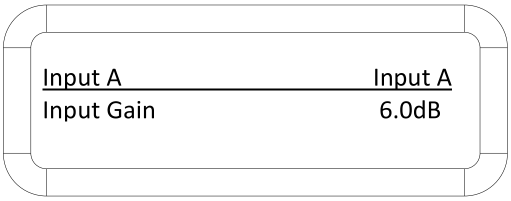
If there are multiple parameters grouped on a single screen (such as parametric EQ, frequency, 'Q' and gain) use the corresponding FREQ, Q or GAIN encoder to adjust it.

You can quickly access other channel at any time by pressing another channel EDIT, and to quickly access the same parameter on another channel, 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 default screen.

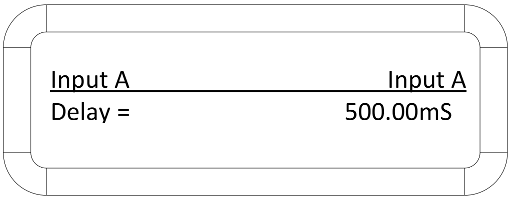
Hint: You can quickly check the same setting of any parameter on a bank of channels by double pressing each EDIT key in turn – so to check each output's polarity setting, just press EDIT, press NEXT until "Polarity" is displayed, then press the next channel's EDIT twice, and so on.

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



Input Delay

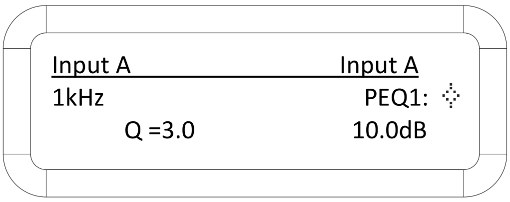
Max delay time is 650.000mS in 10uS steps. Swap to 1mS steps by pressing ENTER.
Units can be changed to read distance instead of time though the system sub-menu..



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.

- **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

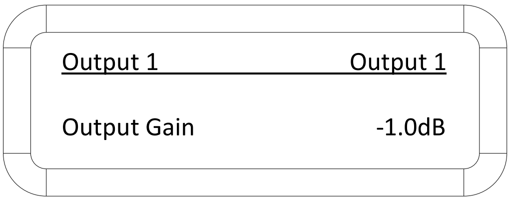


Input Parametric EQ Bands 1 through 8 (continued_

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.

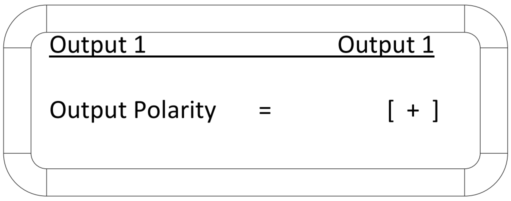
Output Gain

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



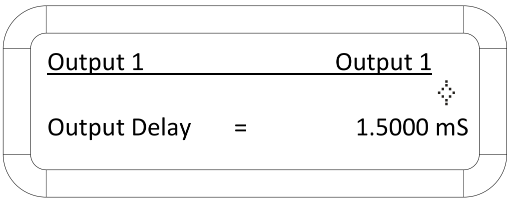
Output Polarity

Switch between normal [+] and inverted [-] polarity.



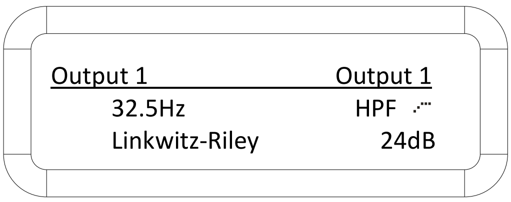
Output Delay

Max delay time is 650.000mS. Use FREQ for 1mS steps, Q for .0208mS steps and GAIN for 300nS steps
Units can be changed to read distance instead of time though the system sub-menu .



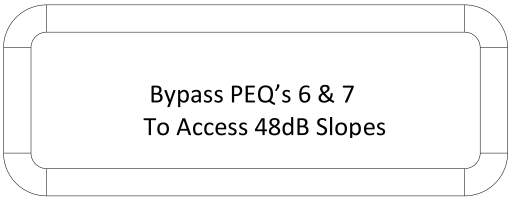
Output High Pass Filter

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.



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

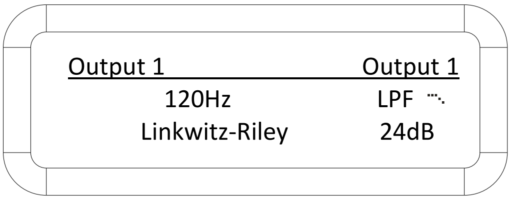
The message



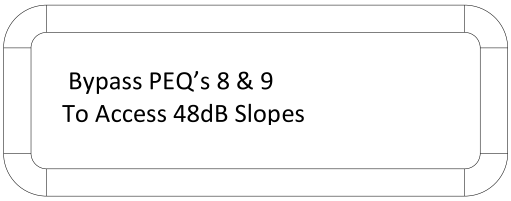
will be shown if the bands aren't already in bypass (or set to 0dB).

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.



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

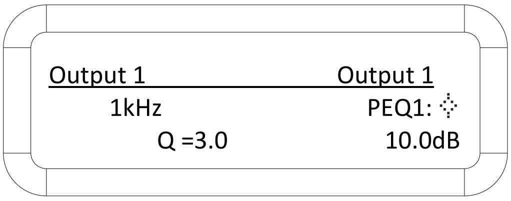


The message will be shown if the bands aren't already in bypass (or set to 0dB).

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.

- **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

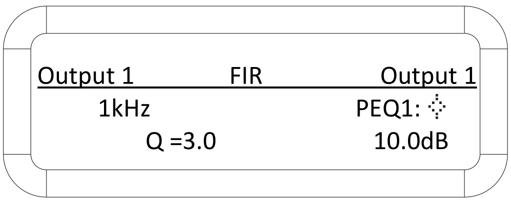


Parametric bands missing? If bands 6 and 7, or 8 and 9 are missing when editing, it is because they are being utilised by high order crossover filters – high pass filter orders above 24dB/Octave will disable and hide bands 6 and 7, and low pass filter orders above 24dB/Octave will similarly remove bands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

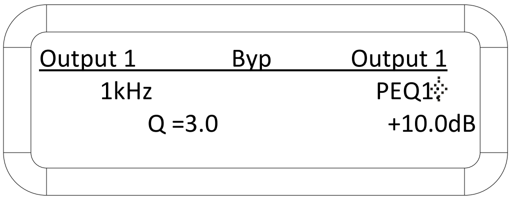
These bands will be reinstated if the respective crossover filter order is reduced to 24dB/Octave or lower.

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.

Some speaker manufacturers might use FIR filters, If an output includes an FIR filter `FIR' will be displayed as below:-

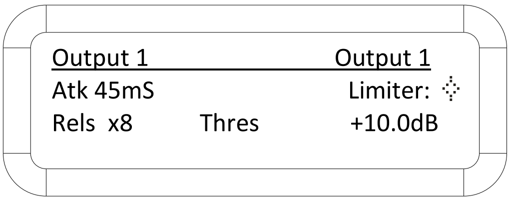


Output filters can be bypassed by pressing the BYPASS key.
When a filter is bypassed, `BYP' will be displayed, as below :-



Output Limiters: Program Limiter

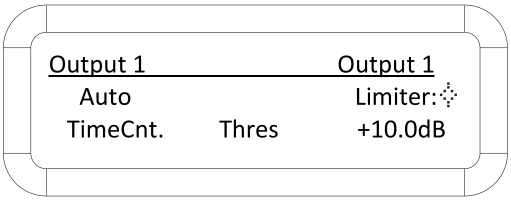
The limiter on each output has adjustable attack and threshold, with a release time that is selectable to be a multiplier of the attack time. For example, as shown below, the attack time is 45mS and release is "x8" so 32mS. The attack and release times can be automatically linked to the high pass filter frequency, so that they are set to correct values for the output's frequency range.



To adjust Attack time use the FREQ encoder
To adjust Release time use the Q encoder
To adjust the Threshold use the GAIN encoders

If the message "Automatic Cnt" 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 filter.

If this feature is enabled, the display will show **Auto TimeCn** in place of the attack and release times. Selection of automatic time constants is through the **Design a Crossover** wizard, in the **Crossover Sub-Menu**.



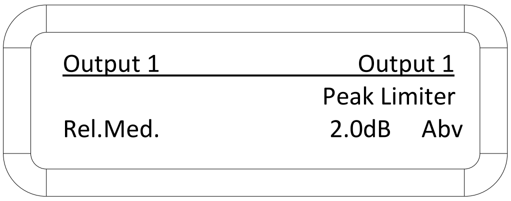
To turn this feature off, and use manual attack and release times, please see the section on page 42 describing Output routing.

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 54 of this manual.

Output Limiters: Peak Limiter

The peak limiter immediately follows the program limiter in the output 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 <u>above the program limiter threshold.</u> This limiter has a zero overshoot characteristic and so only has a release parameter (with no attack time).

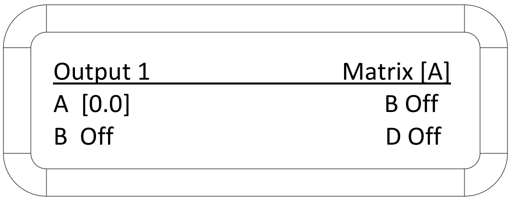


Use the Q encoder set release time (Slow/Medium/Fast)
Use the GAIN encoder to adjust how many dB the peak limiter is above the program limiter
If the release time is replaced with “Rel.=Auto”, this means that the limiters time constants have been set to be configured automatically, based on the frequency of this channel’s high pass crossover filter. Please see the section on page 54 describing processor and auxiliary routing, if you want to revert to manual release time.

Output Matrix Gains

Note that this feature will only be displayed if the configuration for the processor outputs has been set to operate in “Full Matrix” mode , as opposed to “Free Assign” or a standard routing configuration (1 x 8 way, 4 x 2 way etc.)

To use “Full Matrix” mode, please see the section about adjusting the configuration on page 34.



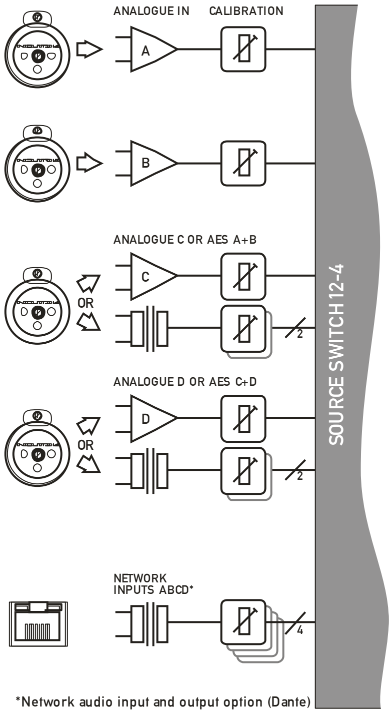
Initially, the “send” level from Input A will be selected – press ENTER to jump to the next “send” level.
Range is from -40dB to +15.0dB, with Mute one step below -40.0dB, whereupon the display will show “Off” as for Input B’s send level in the above example.

Operating Your Processor: Advanced Editing Features

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.
Menus exist to independently configure input and output ganging, output routing and source selection.

Selecting Available Sources (Analogue, AES, Network Audio)



Your processor can source audio from analogue, AES digital, or network locations (if a network card is fitted e.g. Dante / AES67).

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.

Assuming the network audio option is also fitted, four additional digital sources will then be available.

Either 8 or 12, (if a audio network card is fitted), sources can be made available at any one time. 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);

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 MENU
AES Input Selection

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

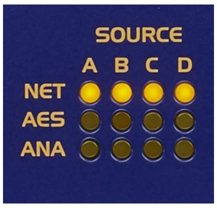
The screen will now show the current choice for the physical XLR inputs on channels C&D:

CONFIGURATION MENU .
XLR C : XLR D
AES : ANALOG

Use the encoder or BACK and NEXT keys to cycle between the four combinations, of ANALOG or AES.
Press ENTER to confirm the choice, and again to confirm changes or Quit to cancel.

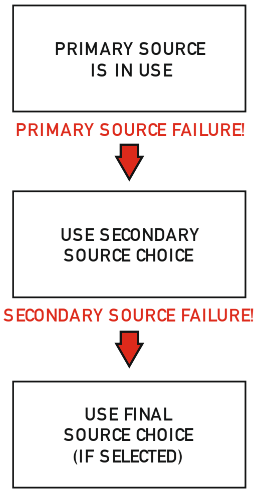
=== Selection Changed ===
Press [ENTER] to CONFIRM

If an INPUT A,B,C or D has AES selected as a primary source and there is a loss of lock or connection to that signal – the associated led in the crosspoint matrix below will flash.



What is Failover?

Your processor supports comprehensive source failover to secondary or even tertiary sources should the primary source fail. Using this system assumes you have more than one source format available (so, for example, an AES stream, plus an analogue backup, or a Dante stream plus analogue backup.)

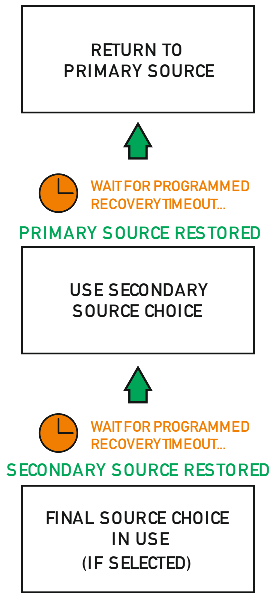


In fact, your processor is capable of setting up a three tier system, so even in the case of a secondary source failure, it can default back to an analogue source. Analogue is always the lowest priority failover source.

A source failure is defined differently for AES inputs and for Dante inputs. For AES sources it is indicated by a loss of signal, so either a failure of the upstream device or disconnection. For Dante sources it is indicated by loss of signal, disconnection and additionally by a loss of subscription.

As analogue sources cannot be verified for signal integrity, if analogue is chosen as the primary source, the failover system will not be offered.

The failover system can also be programmed to restore the source, should secondary or primary sources return, with a hold time to prevent erratic behaviour in the case of intermittent faults.
The Source Matrix LEDs on the front panel indicate indicate the failure of a primary or Secondary source.

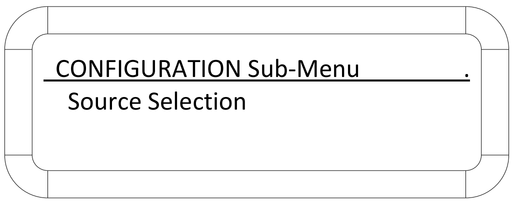


Failover can also be set to NEVER, this will require user intervention to return to the primary source. This can be done by remote control software or by resetting the current status in the Source selection.
Essentialy Configuration Menu...Source Menu.......
Accepting all current status by pressing enter at each step.

Failover selections are stored as part of a "Routing" memory" and are therefore included in storage of an "Everything" memory.
For more information on how to store and recall settings, please see page 38.

Source Selection and Failover

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



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):

```
Source    Selection    A          .
Primary
                   = Analogue A  *
```

Use the encoder or BACK and NEXT keys to scroll through the eight source choices for this input.

These are four analogue sources:	Analogue A	Analogue B	[Analogue C]	[Analogue D]
Four possible AES input selections:	AES A	AES B	[AES C]	[AES D]
If fitted with an Audio Network Card	Network A	Network B	Network C	Network D

The choices shown above would be displayed if AES has been selected on XLR C. Analogue C is shown in square brackets to indicate that this selection will result in no audio because the input is currently unavailable.

If a source is shown in square brackets e.g. [NETWORK A] it is because it is not available for selection, reasons could such no network audio card fitted or an AES XLR is currently in analog mode.

Press ENTER when the required input source is shown. If the source chosen is analogue then no failover operation can be used on that channel, and the next processing channel will be selected and so on, running through input channels A-D.

Finally, if anything has been changed, the confirmation screen will show:

```
Source Selection                .
  Selection Changed
    Press [ENTER] to CONFIRM
```

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

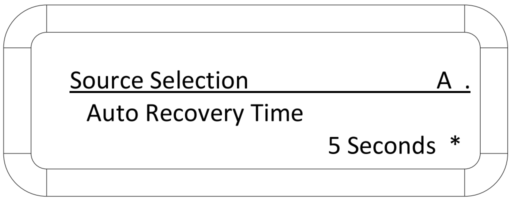
If a digital selection has been made for a primary source on any channel e.g. as AES or NETWORK, then pressing ENTER after this selection you will be asked if Failover is required.

```
Source Selection              A  .
  Failover Secondary
                    ANALOG A
```

Use the BACK/ NEXT or GAIN encoder, to select a secondary source to automatically switch to, should your primary fail. If you do note require failover , select NO. Then press ENTER

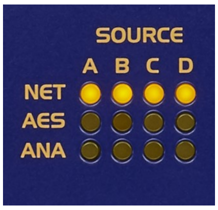
Use the BACK/ NEXT or GAIN encoder to scroll through the 8 or 12 choices for this input. Selecting an analogue source as the secondary will finish the selection process for this input and move onto the primary choice for the next input. If a digital source is chosen for the secondary, then a final step will determine if the tertiary (final) failover source, which can only be analogue.

When the Source Selection for Input D is complete, the user will be prompted to select the recovery time.
This is the time after failover for a secondary to revert to the primary input when it has returned, the options are :- 1,3,5,10,15,20,30seconds or never.
This will only be shown if a channel has failover enabled.



Use the BACK/ NEXT or GAIN encoder to select the time for the system to hold before automatically reselecting the (primary) higher order source. It is possible to prevent the higher order source taking over again by selecting "Never" from the list. Press ENTER to confirm the changes and exit to the home screens.

If Failover is set to NEVER, this will require user intervention to return to the primary source.
This can be done by remote control software or by resetting the current status in the Source selection Menu. Essentially running through the source selection menu again and accepting the current situation.



Once this primary / secondary selection has been completed, the source routing matrix on the front panel will be updated accordingly, any primary / secondary failures will be indicated by a flashing led.

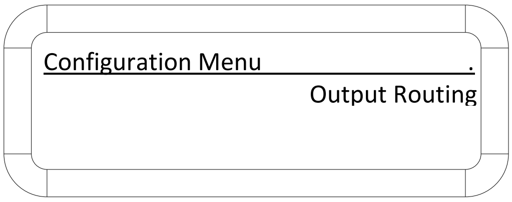
Changing Crossover Configuration

Once the input sources and AES selections have been set up, the next part of the process is to determine the routing configuration, this determines the outputs the four input processing channels are routed to.

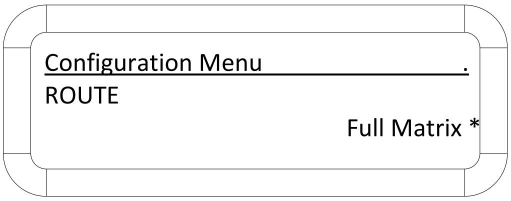
This is the mix matrix. The mix matrix can function either in a "Boolean" operation mode, where input signals still have the capability to be mixed together to feed an output channel's processing, but at fixed standard levels, or as a fully variable mixer.

There is also a selection of useful templates if used in the "Boolean" mode, which will preconfigure not only the routing, but also useful starting points for each outputs' crossover filters and ganging if required

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



Use the BACK/ NEXT or GAIN encoder to find the Output Routing option you require and press ENTER.



Use the BACK/ NEXT or GAIN encoder to choose from the following routing options:

Full Matrix
This mode offers continuously variable send levels from the four input processing channels.
The adjustment of the levels is then through individual output editing accessible via pressing the output channel select. As explained on page 34.

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 8 WAY
All eight outputs are fed from input processing channel A's output
Should you select output filters to be cleared – output filters will be set for full range operation.

2 x 4 WAY
Input A feed outputs 1 – 4 & Input B feeds output 5 - 8.
Default 4 way crossover points are assigned should you select filters to be cleared when prompted.

4 x 2 WAY
Outputs 1 & 2 are fed from Input channel A Outputs 3 & 4 are fed from Input channel B.
Outputs 5 & 6 are fed from Input channel C Outputs 7 & 8 are fed from Input channel D.
Default 2 way crossover points are assigned should you select filters to be cleared when prompted.

2 x 3 WAY + Aux
Input processing channel A feeds output channel 1,2 & 3
Input processing channel B feeds output channels 4,5 &6

Input C feeds Output 7, Input D feed output 8.
Default 3 way crossover points are assigned should you select filters to be cleared when prompted.
Output 7 & 8 will be cleared as full range should you select output filter to be cleared.

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.

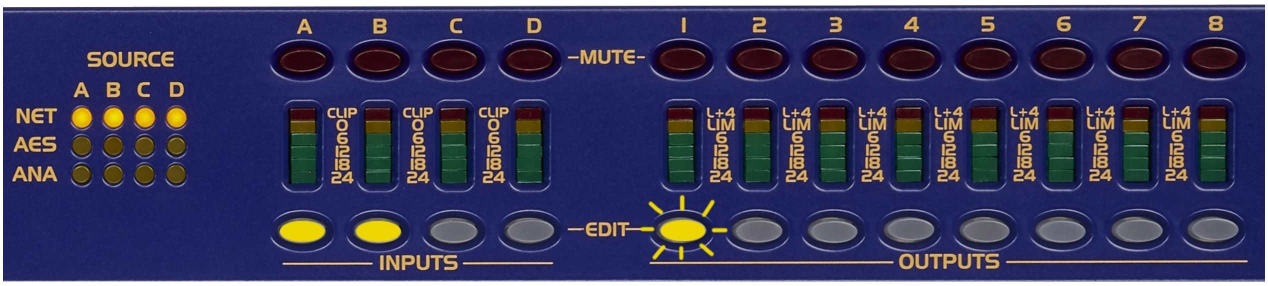
To change configuration, from the home screen, press MENU, choose the CONFIGURATION Sub-Menu and press ENTER.

Configuration Menu .
Output Routing

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

Configuration Menu .
ROUTE
Free Assign*

Use the BACK/ NEXT or GAIN encoder to choose Free Assign and press ENTER.
The Output One Edit will now flash and an Input 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+B:



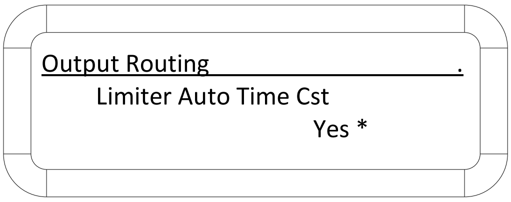
Output Routing .
Routing Output 1 =
Input A + B

Use the BACK/ NEXT or GAIN encoder to choose the required routing combination. The input EDIT key LEDs will change to show the current input selection and the OLED display.
Press ENTER to move on to the next output channel or press the channel's EDIT 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).

This work flow allow you quickly change all output routing or just change 1 output, subsequently just accepting the current status by pressing enter.

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

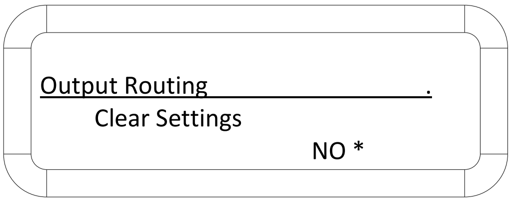
The screen will first show:



We 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.

Resetting Audio Parameters

Resetting parameters is available through the CONFIGURATION > Output 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 potentially all your filter, delay limiter and crossover parameters will be cleared!



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 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 + 5 + 7: <10Hz – 1.62kHz, 24dB/Oct. Link-Riley
Outputs 2 + 4 + 6 + 8: 1.62kHz - >32kHz, 24dB/Oct. Link-Riley

Selecting a 2 x 4way configuration will preset the output crossovers as
Output 1 + 5: <10Hz – 149Hz, 24dB/Oct. Link-Riley
Output 2 + 6: 149Hz – 1.31kHz, 24dB/Oct. Link-Riley
Output 3 + 7: 1.31kHz – 8.00kHz, 24dB/Oct. Link-Riley
Outputs 4 + 8: 8.00kHz – >32kHz, 24dB/Oct. Link-Riley

In both cases, all other DSP functions are reset as in **bold** above.

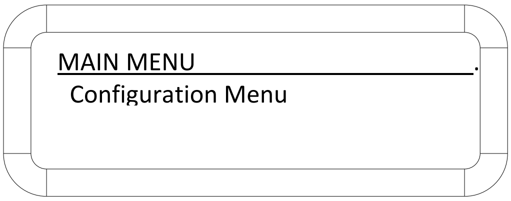
Hint: This procedure can be used to quickly clear all EQ – simply run through all the steps, routing as appropriate or without changing anything and select "Yes" when asked about clearing output settings.

Ganging Channels for Editing

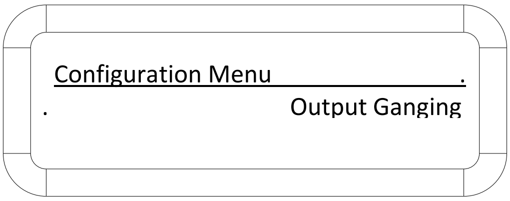
To make editing of multiple channels simpler (for example, stereo input sources or two-way output editing), it's possible to gang channels together so that any edits applied to one will automatically be applied to the other (or others).

Ganging rules can be applied to Input and or Output, the method of setting them up is identical so only the output ganging option will be covered here.

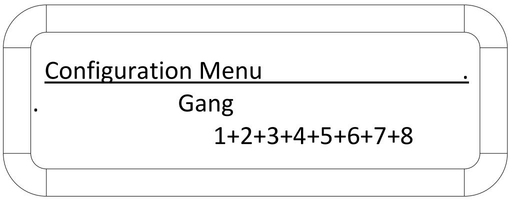
To change output ganging, from the home screen, press MENU, choose the CONFIGURATION Sub-Menu and press ENTER.



Use the BACK/ NEXT or GAIN encoder to find the Output Ganging and press ENTER.



Use the BACK/ NEXT or GAIN encoder 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 as does polarity.

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

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.

Ganging states are stored in output memories, Input memories and everything memories and will be recalled accordingly.

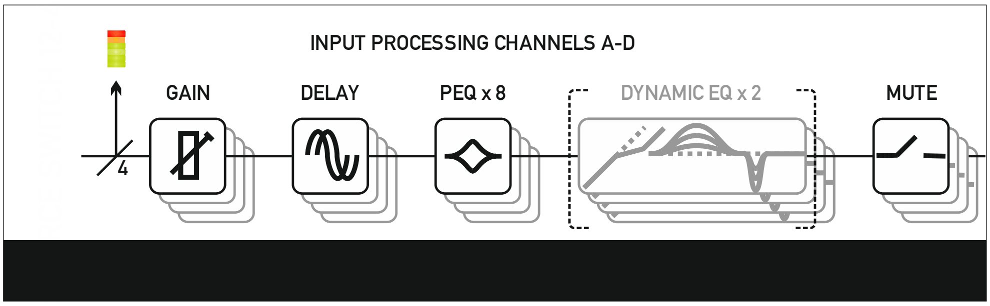
Operating Your Processor: Storing and Recalling Settings

Memory Overview

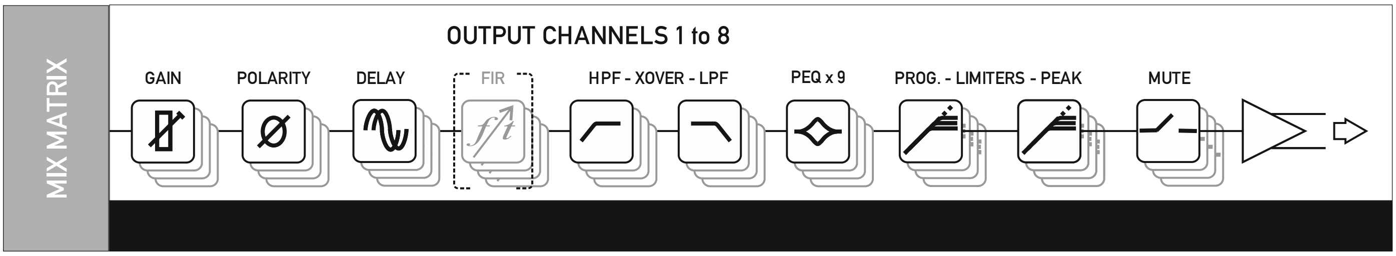
The user memory space in your processor can store four different types – input memories, output memories, source memories and everything memories.

It is possible to store and recall a complete copy of all current settings (source, input, output) in a memory location – these are "Everything" memories.

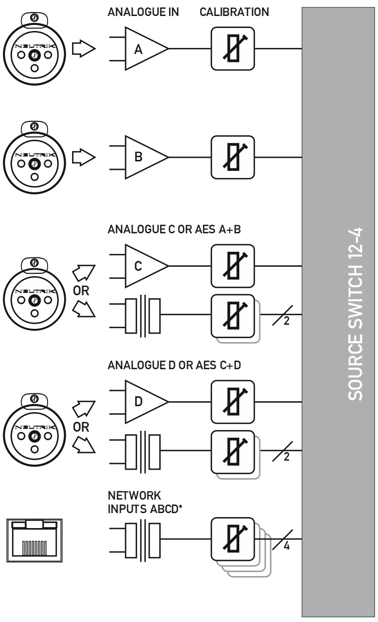
Input memories contain all the EQ, DEQ, gains, delays, mutes, input ganging and names for the four input channels.



Output memories contain all the EQ, crossover filters, FIR data (if applicable), polarities, gains, delays, mutes, limiters (including auto time constants setting), names, output ganging and routing from the mix matrix (including free assign and full matrix gains if applicable). It does not include the state of AES out XLR 2/4/6/8.



There is an additional type of output memory stored in the high memory locations #257 and above.
These are a specific type of memory which contain speaker configuration containing 1,2,3 or 4 output channel – no routing and no ganging. When recalled the user will be asked for the base channel to load them.
These high memory preset are permanently encoded, please see Appendix V how to create the *.bin file and load into a unit. They can be created from the Audiocore Amped Edition preset library using the `Preset Selector software..



Source memories contain the source selection configuration for all four input channels A, B, C and D. This includes primary, secondary and tertiary failover you have configured.

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.

Location 0 – 100 are user memories and can be either input, source or everything memories.

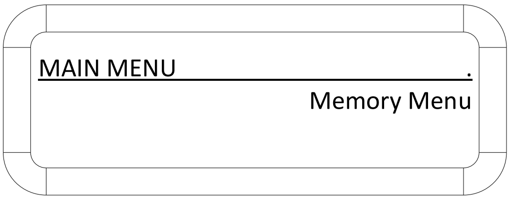
Location #257 and above are presets or manufacturers preset.

Up to 100 user memories and 100 preset memories are possible depending on memory type and FIR storage requirements.

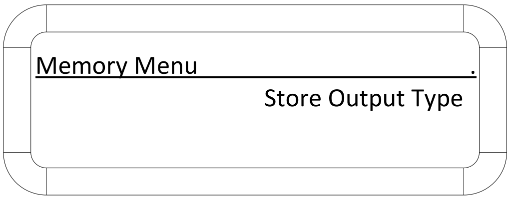
Storing a Memory…

When a memory is to be stored, the type is first selected (Input, Output, Source or Everything)

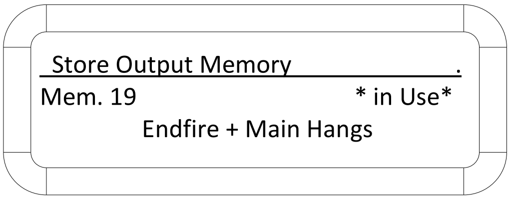
To store a memory, from the home screen, press MENU, choose the MEMORY Sub-Menu and press ENTER.



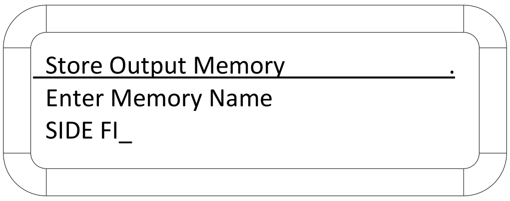
Use BACK/ NEXT or GAIN encoder to choose either Store an Output, Input, Source or Everything memory type as appropriate and press ENTER. Note all the Store, Recall and Erase option are accessible in the this manner.



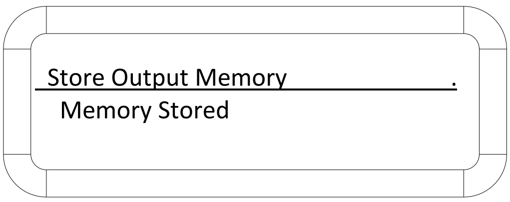
Use BACK/ NEXT or GAIN encoder to choose the memory number – memories that are already used will show the current name and and `*in Use' and ask for confirmation if chosen:



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.

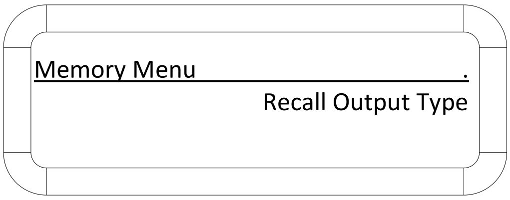


Memory is then stored, with a confirmation message and the processor will return to the default screen.

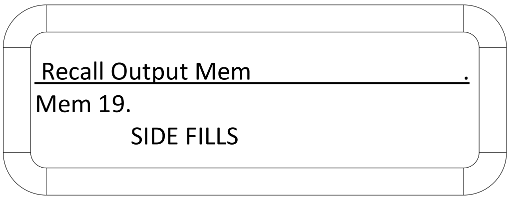


Recalling a Memory...

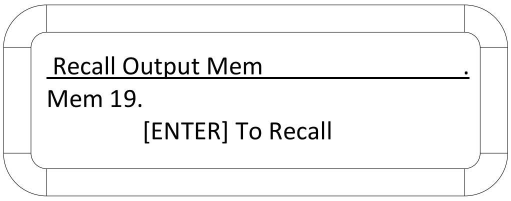
To recall a memory, from the home screen, press MENU, choose the MEMORY Sub-Menu and press ENTER.



Use BACK/ NEXT or GAIN encoder to choose either select Recall Output , Input, Source or Everything type as appropriate and press ENTER. This allow you to filter by memory type.



Use BACK/ NEXT or GAIN 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.

Overview

Your processor is equipped with a CONTROL ethernet connector for software remote control, bulk preset loading and firmware loading.

Software remote Control is done using AudioCore Amped Edition V10 onwards and approximately Q4 2025 globcon XTA Mc2 Edition.

Firmware is loaded using an the XTA IP loader.

Bulk preset loading from a *.bin files is done the XTA preset loader.

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

The GPIO port on your processor offers simple closed contact control / logic control of mutes or memory recalls. It also accepts variable voltages which allow direct level control of selected channels' gain, additionally it allows memory selection.

The optional Dante interface gives your processor network audio capabilities to input and output streams, and can be configured for AES67 compatibility. It is possible to fix the IP address of the Dante card IP address of your processor's control port and Dante interface .

The processor CONTROL and AUDIO NETWORK connection is via different ethernet connectors, *when using Static IP addresses they must not be fixed the same.*

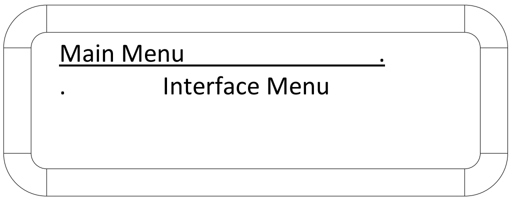
The selection of which Dante audio streams are "subscribed" is performed using Audinate's "Dante Controller" free application. For further information on its operation, please see their own extensive documentation.

Remote Control Software Choices

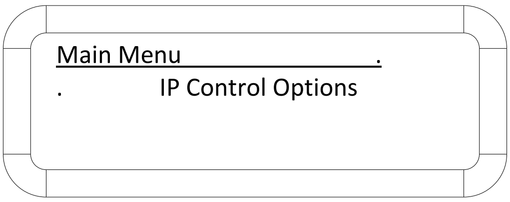
Your processor is designed to be fully configured via AudioCore Amped Edition (V10 onwards) and control by globcon XTA MC2 edition is scheduled.

Configuration of the Remote CONTROL ethernet Interface

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



Use BACK/ NEXT or GAIN encoder to select IP Control Options and press ENTER.



Selection of IP control option will invokes a series of options to configure for the ethernet CONTROL port on the rear these are explained below:

Remote ID Number : The user must assign individual ID number to units. This must be unique to the each unit – setting the same ID on multiple units will cause comms network issues.

IP Mode: Choose either DHCP/AUTO or Static
Auto IP will generate an IP address based on unique hardware features such as MAC address, if a DHCP server is available this will provide an IP address. Auto IP / link local IP address are in the 169.254.*.* range.

Choose Static if you need to select the IP address to lie within a specific range, for instance when working within a larger infrastructure. Should you select Static you will be prompted to select the following :-

Static IP Adress : The IP address specifies the processor's unique identifier on the Ethernet network. It is used in conjunction with the processor'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.
All unit must have unique IP addresses and unique XTA ID numbers.

Subnet Mask : 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.

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

IP Speed : This determine the speed the processor will communicate with other network interface eg on switch or a computer. Option are 100Mb or 10Mb. We recommend the 100Mb for normal use.

The processor can be configured using a direct ethernet cable from you control computer.
Ensure all control software has public and private firewall permissions, sometime computers might view link local ip addresses as maliscious.

In the Interface Menu is a `REMOTE CONTROL' entry, this turns ***ON / OFF*** *via AudioCore Amped Edition / globcon.*

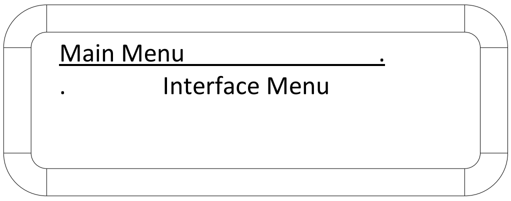
Mode = OFF: Interface is disabled and no external comms are active. This is useful if you need to fully secure your processor as well as locking the front panel (for more information on security and locking see page 68.)

This is useful when GPi and OSC control might be need by a user but AudioCore / globcon control is forbidden.

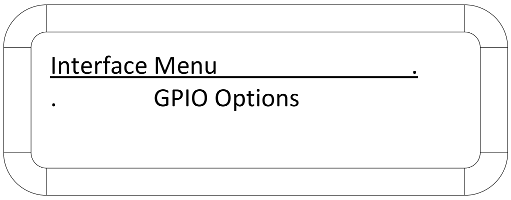
For optimum performance connect as below :-

The General Purpose Input and Output pins on your processor can be programmed in a variety of ways to adjust the operation of the processor.

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

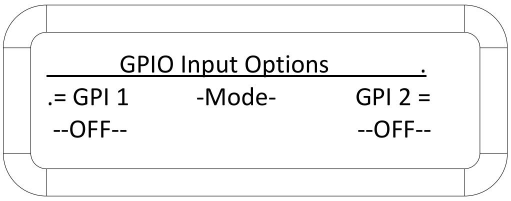


Use the encoder or BACK and NEXT keys to select GPI Interface and press ENTER.



This will invoke a series of further choices to determine the operation of the inputs, as outlined below:

Operating Mode



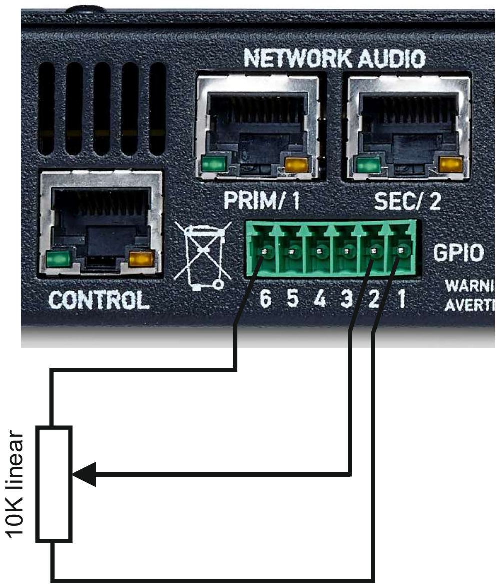
Use the encoder to select from the list of combinations of operating modes for the two GPI inputs. The available combinations are given at the end of this appendix. Press ENTER to confirm selection and move to the next option.

At the end of this section is a list of GPi mode combinations

VCA Mode

Input gains can be adjusted in channel pairs, with level control over the range of +6dB to -18dB in 0.5dB steps, via an analogue voltage applied to the GPI input. The channel choices are A+B, C+D or A+B+C+D.
Recalling a memory or adjusting gain either remotely by another method or via the front panel will override the VCA setting until the control is next adjusted (last takes priority).

A typical connection to a GP input port for VCA control would be to connect a linear potentiometer as shown below:



VR1 – the potentiometer should be a linear taper, with have a value of 10k or greater to avoid cabling losses affecting the value.

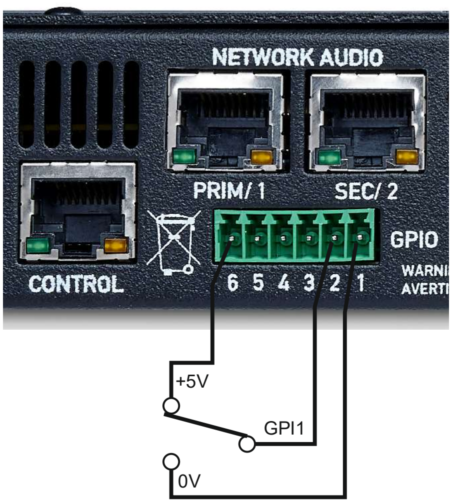
As the inputs are active over a 3V3 range and the output on port pin 6 is +5V, the top 33% of the potentiometer's range will not be active.

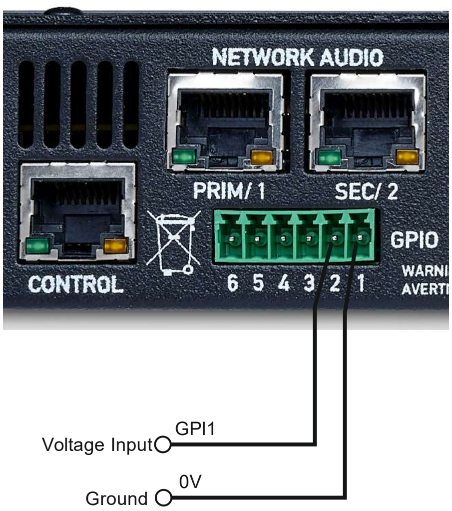
Both GPI inputs can be used in VCA mode to control inputs A+B on input 1, and C+D on input 2.

The GPIO pins are numbered right to left as follows :
1 = 0V
2= GPI1
3=GPI2
4= GPO1
5= GPO2
6= +5V

Memory Recall Mode (2 Memories)
The GPI interface can be configured to switch between Everything memory 1 and Everything memory 2, or switch between everything memories 1 – 8.
This is done by applying 0V or a positive voltage to a GPI pin.

Changing Memory Selection by another method (remote message or front panel) will override the GPI setting until the state of the GPI input is next changed (last takes priority).
The switching voltage can be provided by the DP648, the GPi header provides a 0V and +5V connection for this purpose.
Alternatively an external logic voltage could be provided.
If GPi1 = 0V = Memory 1 If GPi1 = +V = Memory2
The GPi has a +V pullup, so simply connecting Gpi1 to 0V will switch to memory 2.
A typical connection to a GP input port for Mem 1/ 2 selection or muting functions are shown below:





It is also possible to supply an external voltage trigger for the GP Input, as long as the ground for the trigger source is also common. The logic level is 3v3, but the input is tolerant of up to 15v. Do not apply higher voltages to this input or damage to the processor may occur!

Memory Recall Mode (8 Memories)

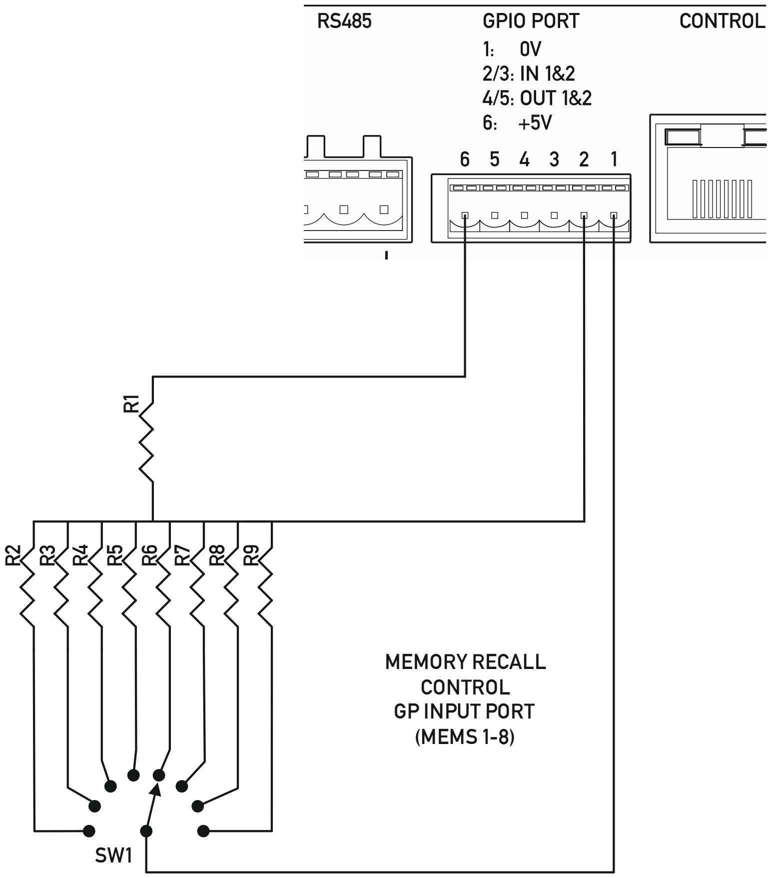
The processor can recall memory settings based on either a logic level (memory 1 or 2) like in the previous diagram or a using voltage level (memories 1 to 8 are available).

Select the required recall mode and press ENTER – either “Mem 1-8” or “Mem 1/2”.

Recalling a memory by another method (remote message or front panel) will override the GPI setting until the state of the GPI input is next changed (last takes priority).

When toggling between two memories in mode “Mem 1/2”, the simple switch logic method can used as described on the previous page.

For the “Mem 1-8” mode, it is recommended that a multi-position switch is used, as the input relies on a voltage threshold to act as the trigger for a memory recall.



Such an arrangement is shown below:

Typical resistor values to allow recall of the first eight memories would be as follows:

R1 = 1K
R2 = 0R (short circuit)
R3 = 47R
R4 = 150R
R5 = 250R
R6 = 410R
R7 = 540R
R8 = 850R
R9 = 1k20R

Mutes Mode

The processor can be muted by applying 0V logic level to the selected GPI pin. This can either be a "System Mute" (as with AudioCore/globcon) so when enabled all outputs are muted and previous mute state is restored when disabled. Changing mutes by another method (remote message or front panel) will override the GPI setting until the state of the GPI input is next changed (last takes priority).
A typical connection to a GP input port for mutes control is the same as recalling Memory 1 / Memory 2.
See the Memory section for example connections.

If mute mode is selected :
If GPi = 0V = Mute
If GPi = +V = On

The GPi input have pull up to +V, ground / connection to OV is required to enable mute.
The GPi input is 15V tolerant .

Mode Combinations

The following combinations of operation are available for the GPI port pins:

GPI1 Mode VCA:A+B	GPI2 Mode Sys. Mute
GPI1 Mode VCA:C+D	GPI2 Mode Sys. Mute
GPI1 Mode VCA:A+B+C+D	GPI2 Mode Sys. Mute
GPI1 Mode VCA:A+B	GPI2 Mode VCA:C+D
GPI1 Mode VCA:A+B	GPI2 Mode Mem 1/2
GPI1 Mode VCA:C+D	GPI2 Mode Mem 1/2
GPI1 Mode VCA:A+B+C+D	GPI2 Mode Mem 1/2
GPI1 Mode VCA:A+B	GPI2 Mode Mem 1 – 8
GPI1 Mode Mem 1/2	GPI2 Mode Sys Mute
GPI1 Mode Mem 1 - 8	GPI2 Mode Sys Mue
GPI1 Mode Mute A+B	GPI2 Mode Mute C+D
GPI1 Mode VCA:C+D	GPI2 Mode Mem 1/2
GPI1 Mode VCA:A+B+C+D	GPI2 Mode Mem 1/2
GPI1 Mode VCA:A+B	GPI2 Mode Mem 1-8
GPI1 Mode VCA:C+D	GPI2 Mode Mem 1-8
GPI1 Mode VCA:A+B+C+D	GPI2 Mode Mem 1-8

Dante / AES67 Interface Configuration

Your Dante interface will work straight out of the box, but should you wish to adjust settings, from the home screen, press MENU, choose the INTERFACE Sub-Menu and press ENTER.

Main Menu .
. Interface Menu

Use the BACK/ NEXT or GAIN encoder to find the Dante Interface option and press ENTER.

Main Menu .
. Dante Options

The Dante interface allow enabling capability AES67[2]

Dante Interface .
. AES67 Mode =
Disabled

Changing this will require a reboot of the card which will happen automatically at the end of the wizard. Press ENTER to choose Network Mode:

Dante Interface .
. Network Mode =
DHCP

Unless you have a valid reason for changing this to "Static" and entering a fixed IP address, SubNet Mask and Gateway IP address, this should be left on DHCP. Changing this will also require a reboot of the card at the end of the wizard.

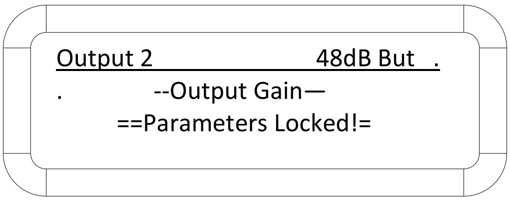
[2] Audinate explain that "The purpose of the AES67 standard is to tie several existing protocols together to create an interoperability specification. The goal of AES67 is that every audio device can eventually connect together with a standard IT network, and share audio. Neither AES67 nor AVB are competitive equivalents to Dante. AES67 and AVB are both a collection of standards, which are not actual implementations. Dante is a commercially supported solution, and more than just a standard. It is important to emphasize that AES67 will be incorporated as an option within Dante, rather than an alternative to it."

Operating Your Processor – Security and Preset Locking

Overview

The security system in your processor offers a wide range of locking options to prevent unwanted access to settings. This system is in addition to the "Parameter Hiding" that may be present within individual speaker presets, as programmed specifically by a speaker manufacturer.

If, during editing, a parameter is displayed like this



or the gain control is only allowing a "Trim" range of ±6dB, this may mean that the individual preset currently running on the outputs has preset locking. Preset locking is specific down to a filter/parameter level, so it's entirely possible for some parametric EQ bands on just one output to be locked, whilst others are still available.

Similarly, gain may be just "trimmable" on some outputs but fully locked or fully unlocked on others.

Preset locking can differ in individual presets and XTA cannot defeat this system, as it is designed to be secure and maintain carefully designed output settings for a speaker manufacturer's particular system. It is only available on output memory settings and does not encompass input or source data.

Note that recalling and saving a copy of a preset in user memory will maintain all locking data, as will copying and pasting output data using AudioCore Amped Edition V10 or globcon XTA MC[3] edition

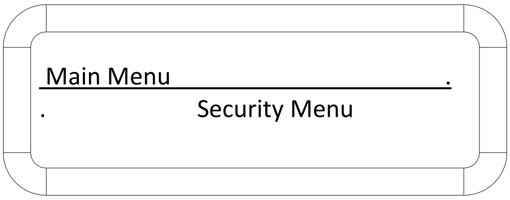
The security system, in contrast, operates across all controls at the same time, and is designed as an anti-tamper system for the front panel.

If the security system to lock the front panel has not been enabled then the locking is only within the current preset. This can be checked by attempting to unlock the unit as explained on page 52 – if you are immediately asked for a pass code then the unit is locked.

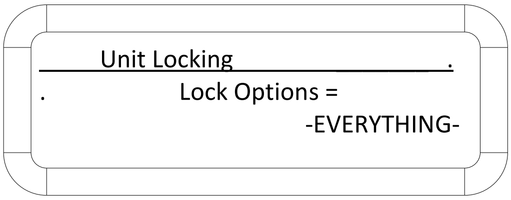
[3] Audinate explain that "The purpose of the AES67 standard is to tie several existing protocols together to create an interoperability specification. The goal of AES67 is that every audio device can eventually connect together with a standard IT network, and share audio. Neither AES67 nor AVB are competitive equivalents to Dante. AES67 and AVB are both a collection of standards, which are not actual implementations. Dante is a commercially supported solution, and more than just a standard. It is important to emphasize that AES67 will be incorporated as an option within Dante, rather than an alternative to it."

Locking the Front Panel

To configure security, from the home screen, press MENU, choose the SECURITY Sub-Menu and press ENTER.



Use the BACK/ NEXT or GAIN encoder to select the required lock type:

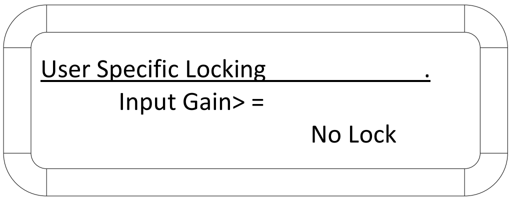


There is a list of useful locking scenarios including the ability to specify individual locking elements. Note that this only works to a unit level, i.e. control from the front panel – not an individual channel level as is the case with presets. The types of locking available from the front panel are as follows :-

User :			
User Specific	see notes after this list.		
Xover only :	Output setting locked & not viewable	Output mutes not changeable	Inputs free to use
Xover + Trim	Output setting locked & not viewable	Output Mutes Work	Inputs free to use
Xover + Trim + Mutes	All output setting locked & mutes	Output Mutes not changeable	Inputs free to use
Changes	Input & Outputs locked & viewable	Input & Output Mutes work	
Changes + View	Input & Outputs locked, no viewing	Input & Output Mutes work	
Changes + Mutes	Input & Outputs locked & viewable	Input & Output Mutes not changeable.	
Everything	Nothing changeable or viewable		

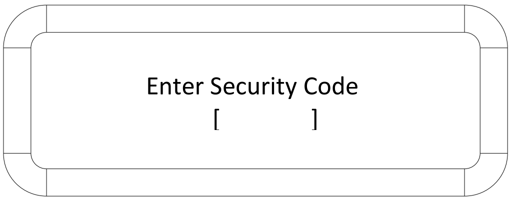
Parameter can still be adjusted by remote control, GPi or OSC – these can also be disabled if required, see the following menus to enable / disable:-
[Interface]…. [Remote] : To enable / disable AudioCore / globcon control
[Interface]…. [GPIO Option] : To enable / disable GPi control
[Interface]…. [OSC Options] : To enable / disable OSC control

Choosing "User Specific" allows sections of the interface to be tailored including EQ sections (inputs and outputs), crossovers, mutes, phase, delays, limiters, memory recall and storage. In the case of directly editable parameters, where appropriate the choice will include "Control" or "Display" as well as "No Lock":



"Control" means the parameter value remains viewable, but cannot be adjusted, and "Display Locked" means the value is hidden and replaced by the "Control Locked" message.

Having chosen the locking scenario, and pressed ENTER, the unit will then request a password to complete the procedure:



Use the EDIT keys to enter a 6 character code, Inputs ABCD and Outputs 1 – 8.
As you enter the final character, the unit will ask for a confirmation and the code must be entered again to lock the unit.

If you wish to disable remote control as well as front panel locking, make sure you set the remote interface to "Off" before locking the unit, using the "Everything" setting.

Unlocking Procedure

Navigate back to the Security Sub-Menu and re-enter your passcode to unlock the unit. There is no limit on the number of attempts to unlock the unit.

Operating Your Processor: Resetting Back to Defaults & Clearing Security

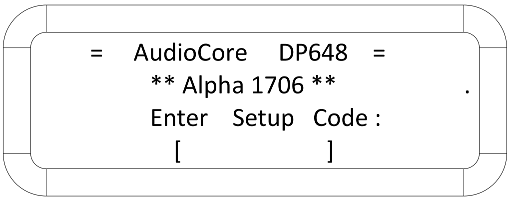
If you need to quickly clear all current settings and memories, there are a variety of setup control codes that can do this for you.

Default Settings

Your processor will start-up as a "four analogue inputs to eight output system", with no EQ or limiting (@ maximums) 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 processor on with the MENU key held. The display will show:



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:

3D44AB – ***clears all user memories and reset to defaults***, keeping security and factory alignment settings

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

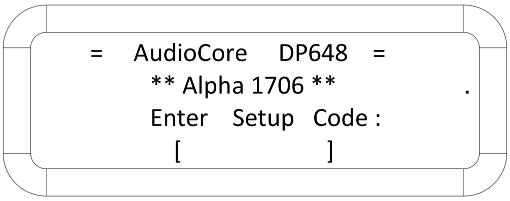
AA2121 – clear security via the XTA supplied crack code (see page 52 for more information)

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 passcode and your processor is locked, you will need to contact us to unlock it after performing the following procedure. There is no master unlock code for all processors – the code is generated based on an algorithm using processor internal settings for greater security.

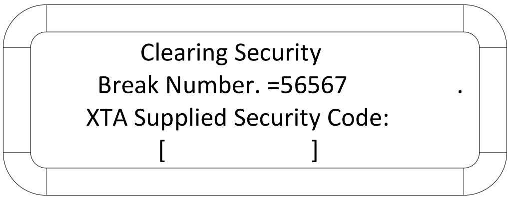
Turn the processor on with the BACK and ENTER keys held. The display will show:



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:



Make a note of the "Break Number" shown (56567 in this example) and contact XTA directly with this – we can then supply the code to unlock your processor.

As a rule we verify the user owns the processor or is authorized to unlock the unit.

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

Resetting security ONLY clears the locking passcode. All current setting and user memories are unchanged.

Looking After Your Processor: Maintenance

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

If the fan vents on the side of the processor 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 processor 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.

Looking After Your Processor: Warranty

Your processor is guaranteed for a period of five (5) years from the date of manufacture.
Please note that this does not apply to OEM versions of the processor – please consult your manufacturer for their warranty terms. We hope that it gives you many more years of reliable service than this, but should anything go wrong, please contact us to advise you about repairs or any spares you might require.

Please do not attempt to repair the processor yourself as doing so will invalidate the warranty.

Our contact details are:

XTA Electronics Limited,
The Design House
Vale Business Park
Worcester Road
Stourport on Severn
Worcestershire
England
DY13 9BZ

Tel: +44(0)1299 879977
Fax: +44(0)1299 879969

email: sales@xta.co.uk for general enquiries

Our website is a great place to get started if you have any questions regarding the general use of your processor or need copies of this manual in digital form, or datasheets and photographs.

www.xta.co.uk

Appendix I: Limiters and How to Set Them Correctly

Your processor 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.

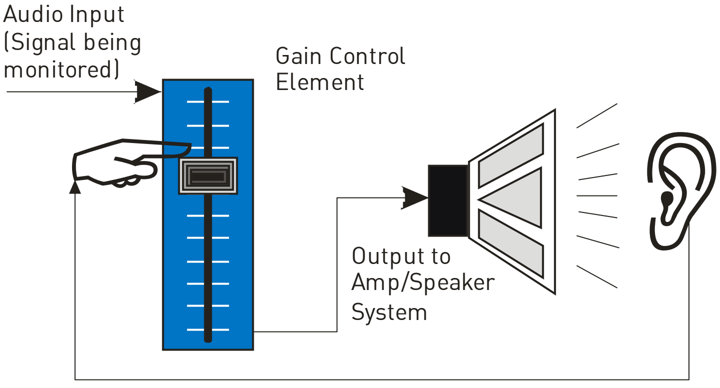
High Pass Filter	Auto Attack Time	Release Time
<10Hz – 31Hz	45mS	x16 (720mS)
31Hz – 63Hz	16mS	x16 (256mS)
63Hz – 125Hz	8mS	x16 (128mS)
125Hz – 250Hz	4mS	x16 (64mS)
250Hz – 500Hz	2mS	x16 (32mS)
500Hz - 1kHz	1mS	x16 (16mS)
1kHz – 2kHz	0.5mS	x16 (8mS)
2kHz – >32kHz	0.3mS	x16 (4mS)

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.

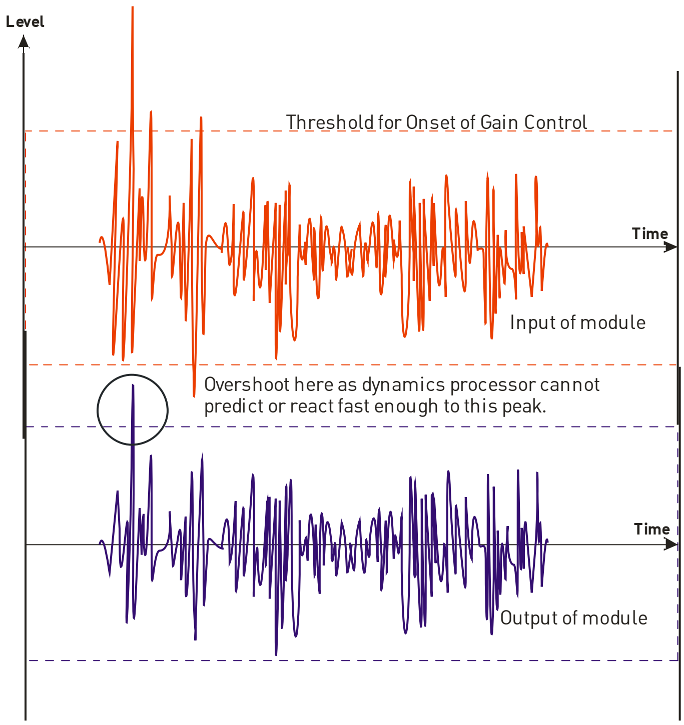
Consider the situation of a monitor engineer listening to a band perform. Having no access to dynamics processors, he has had to resort to manually 'riding the faders' in an attempt to keep control of the levels. Should the level of one of the channels on his desk reach an unacceptably high level, he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



Feedback Path from Monitor Engineer's Ears to Hand forming Sidechain Control Signal

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.

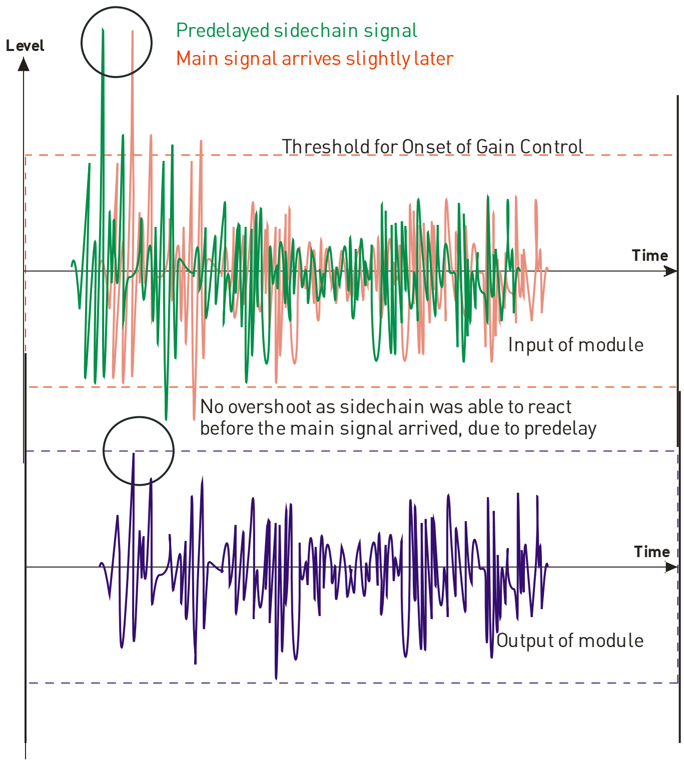


For an analogue dynamics processor, the situation is much better. Controlling the gain electronically, and not relying on a human sidechain feedback mechanism, it can react much more quickly.

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



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

Remember that this delay is only in the order of tens of 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

Introduction

The limiters built into your processor are intended to be used for loudspeaker driver protection, as opposed to processor protection. The processor 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.

Processor Outputs' Program Limiter Lookup Table

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.

Program Limiter Lookup Table

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

Consider the table below.

dB	Ratio	Vrms	Pwr 32Ω	Pwr 16Ω	Pwr 12Ω	Pwr 8Ω	Pwr 4Ω	Pwr 2.7Ω	Pwr 2Ω
45	177.83	137.74	593	1186	1581	2372	4743	7027	9487
44	158.49	122.77	471	942	1256	1884	3768	5882	7536
43	141.25	109.41	374	748	997	1496	2993	4434	5986
42	125.89	97.52	298	595	793	1189	2377	3513	4755
41	112.20	86.91	236	472	629	944	1888	2797	3777
40	100.00	77.46	188	375	500	750.00	1500	2222	3000
39	89.13	69.04	149	298	397	596	1191	1765	2383
38	79.43	61.53	118	236	315	473	946	1042	1893
37	70.79	54.84	94	188	250	375	752	1114	1504
36	63.10	48.87	75	149	199	299	597	885	1194
35	56.23	43.56	59	119	158	237	474	702	949
34	50.12	38.82	47	94	125	188	377	556	754
33	44.67	34.60	38	75	100	150	299	443	599
32	39.81	30.84	30	60	79	119	238	352	475
31	35.48	27.48	24	47	63	94	189	280	378
30	31.62	24.49	19	38	50	75	150	222	300

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 processor, 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 4R, working with an processor 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, 4R**
"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 processor, 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:

$$10 \times (\mathrm{Log}^{10}(\mathrm{Peak\_Power} / \mathrm{RMS\ Power}))$$

So for example, a 15" driver has a quoted RMS power handling of 800W, and a peak power handling of 1600W, the calculation is

$$(1600/800) = 2$$
$$\text{Then } \mathrm{Log}^{10}(2) = 0.3010$$
$$\text{Then } 10 \times 0.3010 = 3.010 \text{ or } 3\text{dB}$$

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 <u>all</u> 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 42 for details on how to turn this option on.

<u>We recommend the use of the automatic setting.</u>

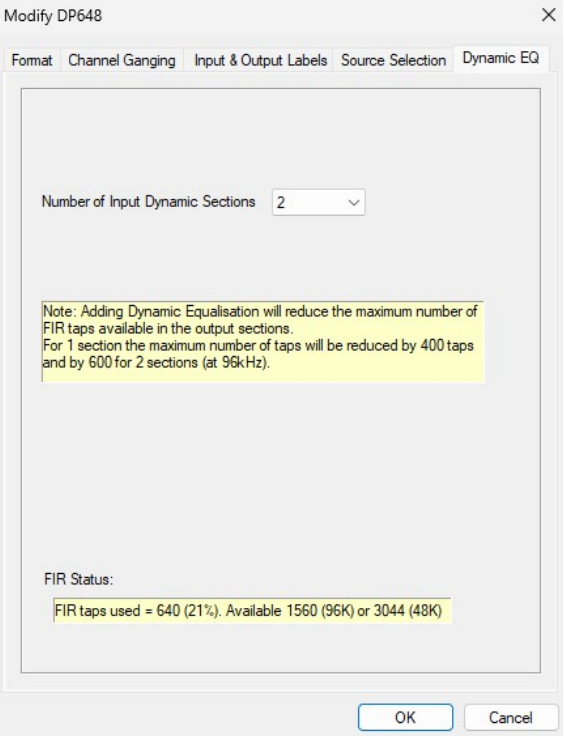
In this mode the time constants will be automatically set from the high pass crossover filter frequency according to the table below:

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 42.

Appendix II:

Dynamic EQ?



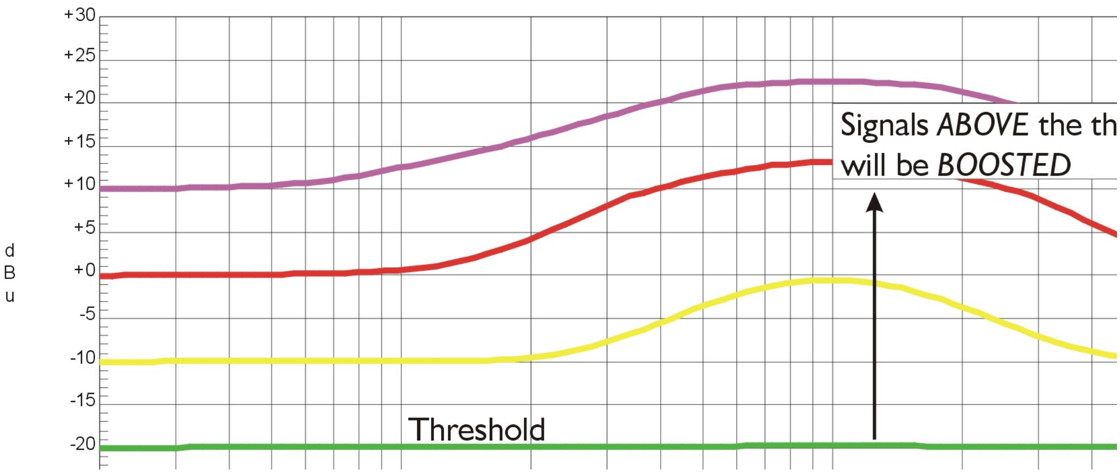
Dynamic EQ uses a DSP resource which is dynamically made available.
This DSP resource is shared with the processors FIR capability.
The more FIR taps used in the output processing the less DEQ bands are available.
Some manufacturer loudspeaker setting might use custom FIR filters for response optimsation, these can be loaded from the Audiocore Amped Edition library or bulk loaded using the Preset loader.

The Audiocore output tabs advise the user FIR taps which are loaded, the Audicore right-click modify, DEQ tab advises the total FIR use and allows no, one or two bands to be enabled .
This modify - DEQ tab is pictured on the left, this where the user can select the number of DEQ bands.
This can also be done on the front panel in the Configuration - DEQ Active Sections menu.

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

Quadrant/Mode I: "Boost Above"

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



Uses of "Boost Above" mode.

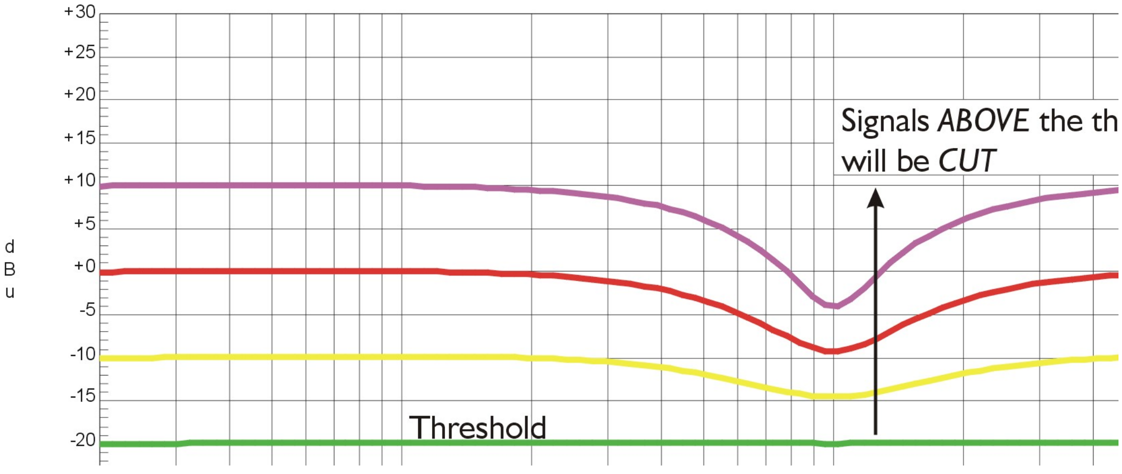
This mode is more useful that it might first appear – the ability to add EQ only at higher signal levels allows some very effective emphasis of certain parts of the spectrum to be added, without the side effect of a permanent audible peak.

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

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

Quadrant/Mode II: "Cut Above"

This is one of the more traditional modes of operation. Having selected the frequency band to work with, the dynamic eq will listen to this band and act upon it by cutting(compressing) any frequencies present in it that go above the predetermined threshold. Consider the example below where the threshold is set to –20dB, and the selected frequency band is centred around 1kHz, with a 'Q' of 1.0.



Signals below the threshold will pass unaltered, but as increasing signal is applied, those frequencies centred around 1kHz will be cut or compressed. The ratio in the above example is set at 2:1 so, as with any compressor, the amount of gain reduction applied depends on how much the signal exceeds the threshold. The red line represents a signal at 0dB, which is 20dB above the threshold. At 1kHz, therefore, the signal has been compressed to –10dB or 2:1.

Uses of "Cut Above" mode.

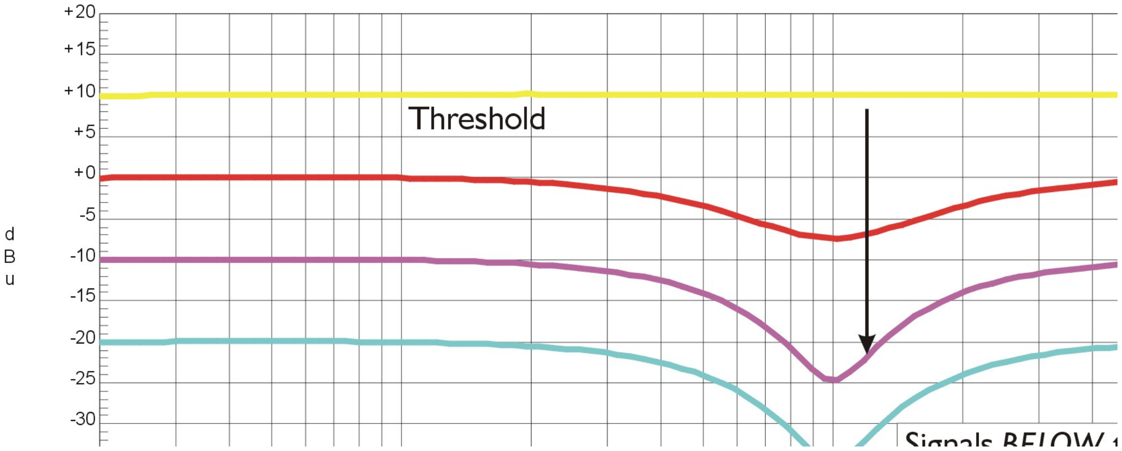
Traditional use of 'frequency conscious' compression is to control or 'tame' a certain band of frequencies within the program material. Insertion of EQ into the sidechain will make the compressor respond to the required band, but it will cause broadband compression of the signal, so any peaks will cause the entire signal to be compressed. This produces the familiar problem of dulling the material if it is bass-heavy, or causing unnecessary dips and changes in ambience when attempting to remove sibilance.

The difference with dynamic EQ is that only the band selected is compressed. This means that it becomes possible to compress the low frequency content of material without affecting the high frequencies at all. The result is increased volume and perceived level without sacrificing clarity. Any instance where the desired result is to control a band of frequencies, such as de-essing, or de-popping, without affecting the surrounding frequency ranges is an ideal use for this mode.

Try de-essing with the filter centred at 8-9kHz, and a relatively narrow 'Q' of
3.2, and a maximum gain of 12dB, attack 1mS, release 100mS.

Quadrant/Mode III: "Cut Below"

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



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

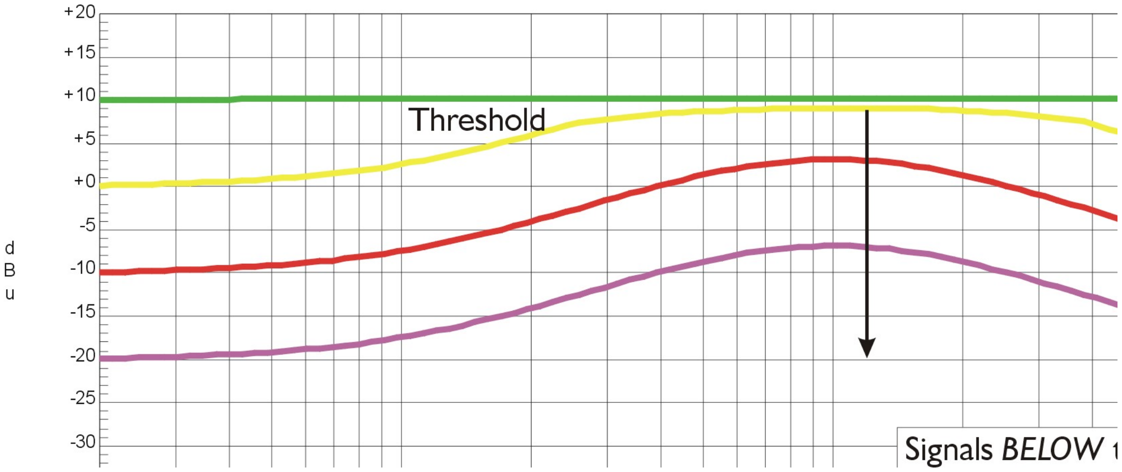
Uses of "Cut Below" mode.

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

Try the filter set to a wide band at 8kHz, and a maximum gain of 12dB, attack 25mS, release 100mS. The threshold setting is more crucial in this mode than usual, with the trade-off being effective removal of noise against possible intrusive dulling of the program material.

Quadrant/Mode IV: "Boost Below"

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



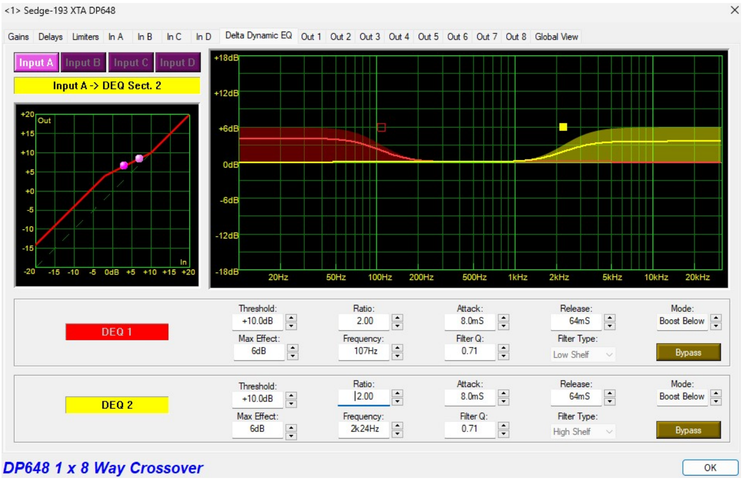
Signals above the threshold pass unaltered but, as the signal drops below the threshold, frequencies around the 1kHz region will be progressively boosted (or expanded). How much boost is applied will depend on the ratio set and how far below the threshold the signal actually is.

Uses of "Boost Below" mode.

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

DEQ Filter Types and Use of Full Bandwidth Mode

In addition to a standard parametric "bell" response, each DEQ band can be set to either a high shelving or low shelving response, and also a "full bandwidth" mode. The shelving responses operate in the similar manner to the "bell" response, insofar as there are parametric adjustments for frequency, 'Q' and maximum allowable gain.



The AudioCore screenshot on the left shows a DEQ screen with band 1 set to operate as a low shelving filter in "Boost Below" mode – this will progressively introduce the low shelving response to the incoming audio as it drops below the set threshold. Using this and another band set to a high self will allow a classic loudness curve to be realised, but with the added advantage of being related to the audio signal, not just determined by the gain control of the channel.

Additionally, using a band set to high shelf and "Cut Below" can be used as a frequency selective gate, only gating high frequencies in the absence of signal, to reduce noise – this is the basis of single ended noise reduction systems.

The screenshot also shows band 1 set to "Full Bandwidth" mode, which effectively turns this band into a gain cell like a standard compressor or expander. If the mode is set to "Cut Above" then this is exactly what the band becomes – a compressor.

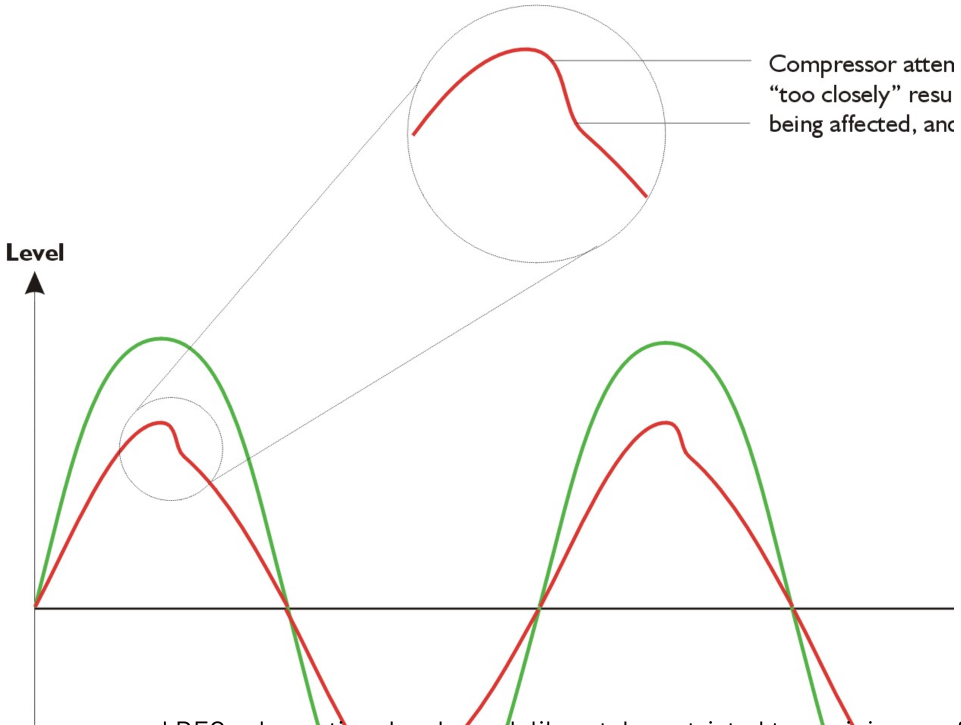
Setting the mode to "Cut Below" turns this band into a downward expander – as signals drop below the threshold they will be progressively reduced in level as determined by the "Ratio" up to a maximum gain reduction as set by the "Max Effect" parameter.

Both "Boost" quadrants are still available in "Full Bandwidth" mode, but we recommend you exercise caution when experimenting with these, as settings may cause large sudden gain changes!

In the screenshot above band 2, in yellow, is Cut Above mode, PEQ mode with wide Q. This an effective mode where stringent SPL monitoring might be in place e.g. live events / festivals, Band 2 could be configured to act quickly to control shouts of make `make some noise!' by singers or Mc's or feedback. The compressor in red operating much slower for overall levelling (up to attack time = 2.9seconds).

Setting the Attack and Release times

As with all dynamic processing (DEQ, compression, limiting), using too fast attack and release times on low frequency program (such as a bass guitar) will cause the compressor to respond to individual cycles of the signal, rather than the overall envelope. This will result in obvious distortion, which might be described as sounding like clicking superimposed on the original signal.



The compressor and DEQ release time has been deliberately restricted to a minimum of 45mS to prevent excessive distortion on low frequency signals, even with fast attack times and high ratios. None the less, it is still possible to introduce some distortion if care is not taken with the settings. The best way to ensure that the signal is not being excessively distorted is to make good use of the 'BYPASS' button, constantly comparing the original signal with the effected version.

Appendix III: OSC - remote control by OSC

OSC stands for **Open Sound Control**. It is a communication protocol used for networking control of computers, sound synthesizers, digital consoles and the DP648. It's an alternative to the much older MIDI standard, designed to be more flexible and powerful, especially for modern, networked application.

The DP648 can be directly controlled by OSC commands generated by control panel designed and deployed on PC/ Android / Apple devices using Hexlers Touch OSC app. These commands are received by the DP648 Control ethernet port. These app can be found at the websiste below :=

https://hexler.net/touchosc

The DP648 OSC menu allow the user to configure :-

OSC RX Port : The processor port it will receive message FROM a control panel, normally match the panel TX port.
OSC TX Port : The processor port it will send replies to the control panel, normally matches the panels RX port.
RX IP Address : The IP address of the control panel sending OSC message to the processor port.

The OSC protocol gives system designer the ability to provide a custom user control to non technical user and venue staff. EG simple gain /mute or memory recalls.

The user has the flexibility to enable / disable processor Interfaces as required. eg once a system is configured Audiocore / globcon control can be disabled thus preventing unauthorized control but GPi and OSC control remain enabled for venue staff / evacuation system integration .

Please separate OSC protocol instruction, programming guide and example panel available on the XTA website.

Appendix IV: 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”.

Processor Communications Set-up

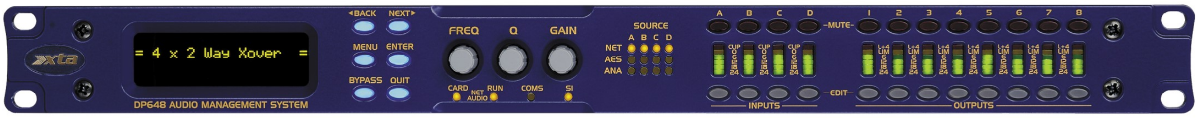
Updates can be performed through an Ethernet connection.

Download the Files

Go to the Support > Latest Software section of the XTA website to find the latest firmware or presets – these will be bundled with the “Flash Loader” application that is used to connect and update your processor.

DP648

Audio System Processor



Specification

Inputs
4 x XLR Electronically balanced Analog Inputs
XLR C & D remote / front panel switchable to AES/EBU
Impedance >10k Ohms CMRR >65dB 50Hz - 10kHz

Outputs
All electronically balanced
AES/EBU (XLR 2,4,6,8) remote / front panel switchable
Source Imp < 600 Ohms
Minimum Load 600 Ohm
Maximum Level +20dBm into 600 Ohm load
Sampling Rate 96kHz internal, up to 192kHz can be accepted and converted
Frequency Response ±0.5dB 10Hz - 32kHz
Dynamic Range >117dB 20Hz -20kHz.
Unwtd Distortion < 0.001% @ 1kHz, +10dBm
Maximum Delay 650 mS. (increment 0.325 µs steps)
Output gain Adjustable +15dB to -40dB in 0.1 dB steps and mute

Equalisation
Parametric Filters - 8 Per input / 9 per output
9 different filter types including PEQ, Bandpass, All pass, Notch, Vari Q, Shelf and Phase filtering - 2 degree steps on each input and output
2 x DEQ pre input, Configurable as cut above , cut below, boost above and boost below as hi shelf , low shelf, parametric EQ and full bandwidth modes. This enables compressors, de-essers, multiband compression etc to be configured.*
*DEQ / FIR processing pool is dynamically assigned

Crossover Filters
Bessel / Butterworth 6/12/18/24/48dB per octave and Linkwitz-Riley 12/24/48dB per octave
FIR processing up to 5600 taps @ 48K and 2900 taps @ 96K available spread across all outputs.*

Limiters
Threshold +22dBu to -10dBu
Attack time 0.3 to 90 milliseconds
Release time 4, 8, 16 or 32 times the attack time
Clip/D-max Limiter - Look-ahead attack time, Fast, Medium or Slow release times

Connectors
Inputs 3 pin female XLR
Outputs 3 pin male XLR
1 X Control port : RJ45 for Audiocore Amped Edition, globcon and firmware updates
Option Dante Card : 2 x RJ45 – configurable as primary / redundant or 2 switch ports.
Power 3 pin IEC Power 60VAC - 240VAC
Consumption < 40 watts

Weight 3.5kg. Net (5kg. Shipping)

Size 1.75”(1U) x 19” x 12” (44 x 482 x 305mm) excluding connectors

Due to continuing product improvement, all specifications subject to change. E&OE

www.xta.co.uk