# MIDAS HERITAGE-D

# USER MANUAL
# V1.16 Béta

EVI AUDIO FRANCE
Audio professionnel

# Chapter 1: Introduction

## 1.1 About this manual

The manual for the HD96-24 is designed to quickly familiarize the user with the console layout, show how to configure and set the system up and then show how to carry out basic functions needed to start mixing audio.

This document is aimed at professional engineers, such as front of house (FOH) and monitor (MON) engineers, who will be using this equipment in a live sound environment. It is assumed that the reader has previous experience of using professional audio equipment.

This guide has been designed specifically so that mix engineers and system technicians can go straight to the areas applicable to them, that is operation, connecting and setting up the system.

## 1.2 Training

The HD96-24 will continue to develop and improve over many years. It is advised to frequently check the www.midasconsoles.com website for up to date videos, user guides and other helpful information.

To help you find your way around the manual, it has been divided into the following main areas:

- Overview: This gives an overview of the HD96-24 System and contains information about this manual.
- Getting Started: This shows you how to set up and power up a HD96-24 System.
- Operation of the HD96-24 surface: This shows you how to use the controls of the surface, navigate its GUI, route (patch) channels & buses, and carry out various operations.
- Description: This gives a detailed description of the HD96-24 hardware, and the controls and their functions on both the control surface and GUI. It provides useful reference material.
- Appendices: This provides reference material and technical information about the HD96-24, such as application notes, signal path diagrams, technical specifications and service information etc.

## 1.4 System Firmware Version

Our team of software engineers are constantly working to improve and expand the features of the HD96-24. It is crucial to have the latest software version installed on your system in order to achieve the best results from your system. Updates can be found in the Midas mCloud, a new approach to track and store system updates.

# 1.5 Touchscreen

**Warning:** The HD96-24 should not be placed or operated in direct sunlight. If the screen is exposed to direct sunlight it may be come unresponsive and too hot to handle. Please ensure you have suitable cover for your console.

# 1.6 Warranty and Registration

Midas are world renowned for quality and reliability. This product comes with the standard Midas 10-year warranty. Registration of your console is achieved by using the Midas mCloud, a new concept in managing a Digital Mixing System.

# 1.7 Service and support

The HD96-24 is state of the art technology. We provide incredible levels of support and service available via the MIDAS mCloud or by our care team to give owners and users confidence in MIDAS products.

# 1.8 Commonly Used Terms and Definitions

Below are some of the terms used in this manual.

- GUI – Graphical User Interface or Touchscreen.
- Channel – Any input, Output (Aux, Flexi-Aux or Matrix).
- Path – Any Input, Output, VCA or Master.
- POPulation Group – A group of channels used to bring or recall paths to the surface or screen.
- Contributions – Any path that contributes to an output bus.
- Touch – The action of pressing the touch screen to turn on or select a function.
- Select – The same as touch.
- Pinch – Two fingers squeezed together, used to tighten or widen equaliser width (Q).
- Swipe – Moving a page left to right or up and down by pressing, holding and moving in the required direction.
- Press and Hold – Either a way to select all the paths on a current page for multiple editing or a way to engage a parameter function that may be critical if press in error, for example flatting the EQ is a press and hold function.
- Widget – The name for a window or various windows displaying information on the GUI as part of a workflow.
- Workflow – Visualizes the activities needed to mix audio.
- Pot – A physical rotary control used to adjust a level or value.
- Interrogation – Pressing and holding a function to see the parameters associated with that control.

Buttons found on the console surface or in the GUI described in this document will be presented with a box around them like this: BUTTON

# Chapter 2: HD96-24 Overview

## 2.1 Introducing the HD96-24

For decades Midas has been a driving force in the world of pro audio. Building on the incredible success of the XL8 and PRO Series with their exemplary audio performance and road-proven rugged and reliable construction, the Midas PRO Series became the gold standard in concert touring and installed live sound. Offering the same outstanding sample-synchronised and phase-coherent audio performance, interpolated control functions and intuitive navigation, the PRO1, PRO2, PRO3, PRO6, PRO9 and later PRO-X Live Audio Systems have become one of the industry's main choices for live sound mixing.

Now the HD96-24 pushes the boundaries further yet again with a 21" touchscreen for hands on instant access to all controls. Parameter adjustment becomes fast and easy with gesture touch interaction using the precise and accurate multi-touch display which allows up to 10 simultaneous touches. Featuring 144 Simultaneous inputs and 120 (96 x Flexi Aux + 24 x Matrix) time-aligned and phase-coherent busses with no stealing of resource's in channel or bus counts. True and consistent 96 kHz sampling frequency and 64-bit floating-point processing provide exemplary quality audio processing, and the oversampled and interpolated digital signal processing algorithms, combined with the fully interpolated and touch sensitive user controls, result in the smooth continuous response and immediacy of working on an analogue console.

The HD96-24 features the rugged and road-proven KLARK TEKNIK HyperMAC (HMAC) and SuperMAC (AES50-compliant) networking technologies with their ultra-low and deterministic latencies and robust error correction. Its powerful audio networking offers up to 624 inputs and 654 outputs at the 96kHz sample frequency depending on configuration.

The 24 VCA (Variable Control Association) and 24 POP (POPulation) groups, combined with the advanced touch screen navigation system, 28 faders, assignable controls and innovative shortcut area allow simultaneous display and control of all the critical information required to craft an unprecedented mix experience.

# 2.2 Overview

The HD96-24 is a very powerful and flexible audio processing system that provides a complete solution for any audio mixing and signal distribution application in a live sound environment. Common features of the HD96-24 Live Audio System include:

- Graviton Audio Engine incorporated into surface
- 144 flexi inputs
- 120 busses.
- Up to 96 stereo Effects
- 96 Ultima Stereo Dynamic EQs
- 32 Ultima Stereo Multiband Compressors
- 24 VCA (variable control association) groups, and 24 POPulation groups
- Variable phase per input channel
- Input - Aux - Aux - Matrix routing (Delay compensated)
- True 64-bit FPGA processing with GPU coprocessing
- 5 tap off points per channel per send, all delay compensated
- Dedicated 16 into 12 shout mixer with instant configuration recall
- Dual Solo busses featuring duckers and AFL/PFL configurable for input and output
- True Audition™
- True Preview ™
- Channel AI™
- 21" full colour TFT high brightness display screen with capacitive touch sensing
- Advanced and multi-gesture touchscreen user interface
- Award-winning Midas microphone preamplifiers
- HyperMAC and AES50 networking allows up to 576 inputs and 576 outputs sources @ 96 kHz sample rate
- Dual network bridge format converter with up to 128 bidirectional channels and asynchronous sample rate conversion
- Touring grade road case featuring marine grade plywood, aluminium extrusions and composite density protective foam
- Integrated Bluetooth and wireless transceiver module
- Dual ULTRANET Ports providing 32 additional digital outputs
- 28 Midas PRO motorised 100 mm faders
- Fully interpolated touch sensitive controls
- Dual redundant auto-ranging universal switch-mode power supplies
- 10-Year Warranty Program
- Designed and engineered in the UK

# 2.3 Key Features

## New Surface Design

- DreamFlow™ UI
- Full HD multitouch screen - daylight coated
- 40 x 18-bit colour 240x240 channel LCDs
- Total configurability, anything anywhere
- 36 assignable rotaries
- Full channel control with dedicated 4-band EQ and dynamics section
- Manchino workflow heralds new Era in console setup speed
- Completely customisable surface
- 28 faders
- Dante, Madi and USB interfaces available directly in surface with two CM1 slots
- Graviton Audio Engine utilises two HyperMac ports which are configurable for dual redundant or independent operation

Over 39 Effects to choose from

Including Effects from TC Electronic.

- Tc electronic system 6000 FX (VSS4)
- Tc electronic 2290 (coming soon)
- Tc electronic system 3000 FX (VSS3)
- KT DN780 Reverb
- HD2a
- HD 670
- KT 1176
- R-Comp 3
- KT Bus compressor
- Mtec EQP-HD
- Mtec MEQ-HD
- M6 Multiband compressor and dynamic EQ.
- HD Stressor compressor
- HD Wave Designer
- Ultra-Dualistic, Voice Doubler
- XL4 EQ

# 2.4 Applications

### Applications

The HD96-24 is the go-to high-end Midas Digital Console System, akin to the ‘industry standard’ Heritage 3000 and XL4. Although the HD96-24 is designed for the traditional touring live sound environment, it is also ideal for theatre, house of worship installations and broadcast. So, being a truly multi-functional console in the Midas tradition, the HD96-24 is suitable for many applications, such as:

- Live sound touring Front of House (FOH) or Monitors (MONS) duties.
- Live sound theatre FOH or MONS duties.
- Live sound & broadcast for house of worship FOH or MONS duties

# 2.5 System Components

The HD96-24 surface now brings together all DSP and processing into one place. This allows for quicker set-up. With built in Wi-Fi and Bluetooth the HD96-24 can talk to the mCloud for show file back-up, software updates, fault tracking and even the location of the desk anywhere in the world.

All the I/O is handled by the familiar Midas Blue DL series rack which combines the warmth and authenticity of analogue circuitry with the precision and agility of state-of-the-art digital techniques. Extreme care has been taken in the planning and execution of the PCB layout to maintain exceptional grounding and analogue/digital separation with the highest quality of sonic performance.

Compatible DL Series units are:

- DL231
- DL251/252
- DL151/152/153/154/155

The Klark Teknik AS88/DN9680 8 Port AES50 extender and multiplexer can be directly connected to the HD96-24 which Extends AES50 connections up to 500m with optical fibre or 100 m with CAT5 cable. AS88/DN9680 supports both optical fibre and copper snakes using a Gigabit Ethernet digital audio point-to-point link. A dual-fibre Neutrik opticalCON DUO connector is used for optical fibre snake connection, enabling a bidirectional optical link on one multi-mode dual fibre cable

The AS88/DN9680 acts as a bidirectional multiplexer and demultiplexer and combines the eight incoming AES50 streams into the outgoing snake connection, and simultaneously also takes the incoming snake connection and unbundles the eight AES50 streams and routes them to the corresponding AES50 ports with redundant connections available. This allows 192 bidirectional channels of audio to be used

# 2.6 System Busses

The HD96-24 has comprehensive system busses to suit demanding applications, comprising:

- 2 stereo solo busses, routable from all locations and allowing for dual operator
- 3 master busses, routable from the mic/line inputs (up to 144), and 96 aux busses.
- 24 matrix busses, routable from the mic/line inputs (up to 144), 96 aux busses and three master busses.
- 96 aux busses (either standard or flexi-aux, routable from the mic/line inputs (up to 144) or flexi aux bus to aux bus for group or stem style processing.

All of the bus routings provide simultaneous and time aligned mixing of all the sources, which will be switchable for minimum latency requirements.

For monitor mixing, the master, matrix and aux busses can all be routed directly from the input channels, with independent level controls providing up to 123 monitor mix busses. Flexi-Aux busses allow group mixing of channels to be sent to Auxes, Matrices or the Masters, for example, mix and process all your drums via a Flexi-Aux then send to an IEM Aux.

For traditional FOH group mixing, any (or all) of the aux busses can change to operate post-channel fader and pan (that is, aux gain fixed at unity).

# 2.7 Mix Matrix

Fundamentally, the mix matrix defines the capability of the HD96-24. Probably the best way to imagine the mix matrix is to think of an analogue console layout, where inputs run vertically, and busses run horizontally. A mix matrix is usually defined as the number of busses and the quantity of simultaneously mixable inputs there are per bus. The following diagrams illustrate the capability within the HD96-24 system.

# 2.8 Processing

Although the HD96-24 system allows for considerable insertion of external processing, it also embodies ample internal high-quality processing to eliminate the need for this, in the interests of simplicity and reduced overall system size, weight and cost.

## Processing components

- **The processing available is:**
  - Up to 144 x 12 or 24dB/Oct. High pass filters.
  - Up to 144 x 6 or 12dB/Oct. Low pass filters.
  - Compressor/limiters on every input and output channel with side chain filtering and multiple operating “signatures”.
  - Up to 144 gates with side chain filtering.

  - 144 x 4-band parametric input EQ with multiple shelf "modes".
  - 123 x up to 4-band parametric output EQ with HPF/LPF modes.
  - Up to 24 effects processors slots selectable from reverbs, delays, phase shifters, compressors and pitch shifters.
- **Input channel processing**
- Each of the 144 (maximum) full-function flexi input channels has:
  - Analogue gain and digital trim.
  - Polarity reverse switch.
  - Input delay.
  - Swept high pass filter with choice of two filter slopes.
  - Swept low pass filter with choice of two filter slopes.
  - Swept all pass filter with a choice of two options of phase range with low and high optimization.
  - Frequency-conscious compressor with choice of four compression styles.
  - Frequency-conscious noise gate with external side chain.
  - Insert point. (half normalized i.e. the insert send can be used as a channel direct output, even if the insert return isn't switched in).
  - 4 Band Parametric EQ filter with various filter options.
  - Routing via level controls to 120 mix busses.
  - Routing via pan control to left and right master busses.
  - Routing to mono master bus.
  - Direct output. Choice of 3 pick-off points, with independent output level/mute.
  - 3 Slots for FX inserts
- **Mix channel processing**
- Each of the auxiliary mix busses has:
  - Group or Mix mode.
  - Dual mono or stereo pair modes. Mono, Linked or Stereo where a single fader controls 2 linked adjacent paths.
  - Up to 4-band PEQ.
  - Frequency-conscious compressor with choice of four compression styles.
  - Insert point.
  - Routing via level controls to the matrix busses.
  - Routing via pan control to the left, right and mono master busses.
  - Direct input with independent input level/mute.

- **Output channel processing**
- Each of the matrix busses has:
  - Up to 4-band PEQ.
  - Four-mode frequency-conscious compressor with soft clip limiter and external side chain.
  - Insert point.
  - Direct input.
- Each of the three master output busses has:
  - Up to 4-band PEQ.
  - Four-mode frequency-conscious compressor with soft clip limiter and external side chain.
  - Insert point.
  - Direct input.
  - Routing via level controls to the matrix busses.

# 2.9 Audio Physical Connections

The maximum total number of physical analogue XLR connections possible on a HD96-24 is 488 plus talk line in and talkback mic in on the rear of the surface. The maximum number of 488 configurable I/O can be increased up to 654 (488 analogue + up to 128 CM1 + 32 ULTRANET + 4 AES3) depending on the combination of different types of I/O card can be used. All of the configurable I/O are freely routable on a scene-by-scene basis.

# 2.10 Network

The network of the HD96-24 utilises the physical connectivity of Ethernet (EtherCon® connectors and Cat 5e/copper cable) but replaces its data protocol with AES50 protocol (implemented as SuperMac) and the HyperMac high capacity system, which are more suited to high quality, low latency audio distribution. The use of the AES standard allows straightforward interfacing with any third-party hardware that also utilises this connection.

AES50 connections carry digital audio and control data bi-directionally down a single cable. Cat 5e cable is used for the 'local' connections and the dual digital 'snake' (equivalent to a 384-channel analogue multi-core, 192 channels per snake connection) between console and I/O. The combination of audio, control, clock and third-party Ethernet data in a single network means that the hardware interfaces on a single RJ45 connection.

All system connections can be duplicated for full dual redundancy

## 2.11 mCloud Network

The Midas mCloud network is a brand-new concept in file and system management. The HD96-24 has built in Wi-Fi capabilities which allow the surface to share its information over a Wi-Fi connection and any other network connections to the mCloud. Be reassured the connection is completely safe with military grade encryption. Great lengths have been taken to keep information secure.

The mCloud can be used to store your show files, preset files and all other types of data from the console. If you leave your USB stick with your vital settings at home, you can directly log into your mCloud account and load your show file straight to the HD96-24 without breaking into a sweat.

New system updates can be downloaded directly to the surface ready for you to update when you're ready. A list of all previous software versions will be stored on the HD96-24 for peace of mind.

It also allows audio rental companies to keep a track of registrations, software versions, warranties and diagnostic logs. All the admin for running a busy hire company in one place.

Each user of the HD96-24 will be prompted to set up a user profile which also in turn configures your mCloud account.

## 2.12 HD96-24 software

The operating system of the HD96-24 built on Linux, which is an open-source, stable, proven operating system (OS). Linux is used in many mission-critical applications worldwide and has allowed Midas' software engineers to write a ground-up system that contains no 'hidden' or unused code. This has resulted in an efficient, compact application, which is quick in operation, quick booting and comparatively easy to debug.

# 2.13 Graphic User Interface (GUI)

The HD96-24 has a 21” touch screen that provides a quick and intuitive workflow. Modern touch gestures such as pinch and smooth touch screen faders have been included to speed up workflow and let you concentrate on the mix. Not only does the GUI reflect what is happening on the control surface, but it also provides extra functionality via a top and side bar menu. These menus provide access to all the pages that you will require to set up, configure, manage and operate the entire control surface.

Gone are the days of only one touch on a screen at a time. Use both hands to manipulate up to 10 faders at a time if you so wish.

Pinch gesture showing EQ width adjustment.



Independent widget style areas are extensively used to display various different types of information at once, all fully customisable to suit your workflow.

# 2.14 System Card Expansion

The HD96-24 has 2 x CM-1 slots built in for further audio expansion. Adding up to an additional 128 channels of I/O greatly increases networked capabilities. Virtual sound checks and recording have never been easier to set-up and achieve with flexible options. Being able to support new and emerging protocols via its two industry-standard expansion slots give the HD96-24 a greatly extended shelf life.

# CHAPTER 3: System Setup

## 3.1 Initial Set-up Procedure

This chapter shows you how to set up a live audio system to its standard configuration.

### Initial set-up procedure

- Initial system set-up basically comprises:
- Unpacking and checking the equipment.
- Connecting up the equipment.
- Powering the equipment.
- Initial patching.
- Type of snake connection.
- Configuring the rack unit(s).
- User Profile use.

## 3.2 Unpacking the Equipment

After carefully unpacking the equipment, save all packing materials, as they will prove useful should it become necessary to transport the equipment later. Inspect the equipment carefully for any sign of damage incurred during transportation. It has undergone stringent quality control inspection and tests prior to packing and was in perfect condition when it left the factory. However, if the equipment shows any signs of damage, notify the transportation company without delay. Only you, the consignee, may institute a claim against the carrier for damage during transportation.

The screen and LCD displays are protected by a plastic film. Carefully peel this off before use.

**Warning:** This can induce static that can cause temporary distortion of LCD displays. This is rectified after rebooting the console.

## 3.3 Ventilation

The HD96-24 has air intake vents on each side of the console. Air is drawn in through the console side vents and exits via the two fans on the rear of the surface. It is vital none of these airways are blocked as overheating may occur if airflow is restricted.


# 3.4 Racking the I/O

Please take note of the rack requirements as detailed below:

To ensure the correct installation and function of the outboard equipment, any rack has to meet the following general requirements:

### Shock mounting (for non-installation environments)

The rack must provide adequate shock protection of the units it houses by incorporating appropriately-designed shock protection methods. For example, a foam-suspended rack or a frame suspended on anti-vibration mounts.

The Midas I/O units have been designed such that their internal ventilation airflow is drawn in through the front of the unit and expelled though the rear. To facilitate this, rack design must ensure that cool air can flow freely through the rack in the same direction, that is, in through the front of the rack and out through the rear. Situations where the air flows in a circular direction around and through a Midas I/O unit must be prevented. MIDAS recommends that racks with fully opening front and rear doors are used.

### Caution

Never combine units in the same rack that have been designed for a ventilation air flow direction other than that designed for the MIDAS units. To avoid this, we recommend that any non-MIDAS units are housed separately.

## Rack mount supports

Always secure the rear of the Midas I/O units to the rack via their rear rack mount support brackets. These brackets are fitted to every Midas I/O unit and are recommended for use in touring applications.

## Handles on rack case

You must ensure that there are sufficient external handles fitted to the rack casing to enable the rack to be manoeuvred easily and safely, and by the number of personnel suitable for the task. Also, these handles must be fit for purpose.

## Clearance at the rear of units

Ensure an adequate clearance at the rear of the units to provide sufficient free space to enable the cables to achieve their minimum bend radius.

## Securing the cables

We recommend that the cables at the rear of the units be tidied using lacing bars and cable ties. This should provide optimum access to the rear of the units for connecting other cables, switching the units on/off etc., and also to give maximum visibility of the units' LEDs for determining communication status, link status, condition of audio etc.

# 3.5 Connection Instructions

There are currently two ways to connect the system equipment together:

1. HD96-24 surface to a Klark Teknik DN9680 via copper (up to 100 m) or with a multi-mode (MM) fibre optic snake (up to 500 m). Then Klark Teknik DN9680 to I/O box (for example, DL231) via Cat5E (up to 100 m).

2. HD96-24 surface direct to I/O (for example, DL231) via Cat5E (up to 100 m).

# It is imperative only STP Cat5E Rated cables are used!

Length = 100M Point to point as per the Cate5E ethernet protocol - Please take into consideration any in line connections or links reduce the overall cable length.

## AES50 Cat5e STP vs UTP cables.

Music Tribe are standardising the use of Ethercon cables used for AES50 connections and state that customers must use Shielded Twisted Pair (STP) cable only with shielded RJ45 plugs and Ethercon shells.

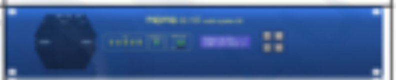
STP cable has the added advantage of a foil or braided shield that guards the cable against electromagnetic interference. A good foil or braided shield and correctly connected shielded plugs and shells also helps protect against Electrostatic discharge (ESD) that can be the cause of dropouts on AES50 connections.

Occasionally shielded Ethercon cables will leave the shield disconnected on one end to help with ground loops, even though it has no benefit for AES50 connections. These connections should have continuity of the shield on both ends including the Ethercon shells. This will ensure the best possible protection against strong ESD impacts, such as handling discharges or even lightning strikes in the neighbourhood.

## All AES/EBU connections must use good quality 110Ω AES/EBU cable to ensure correct operation.

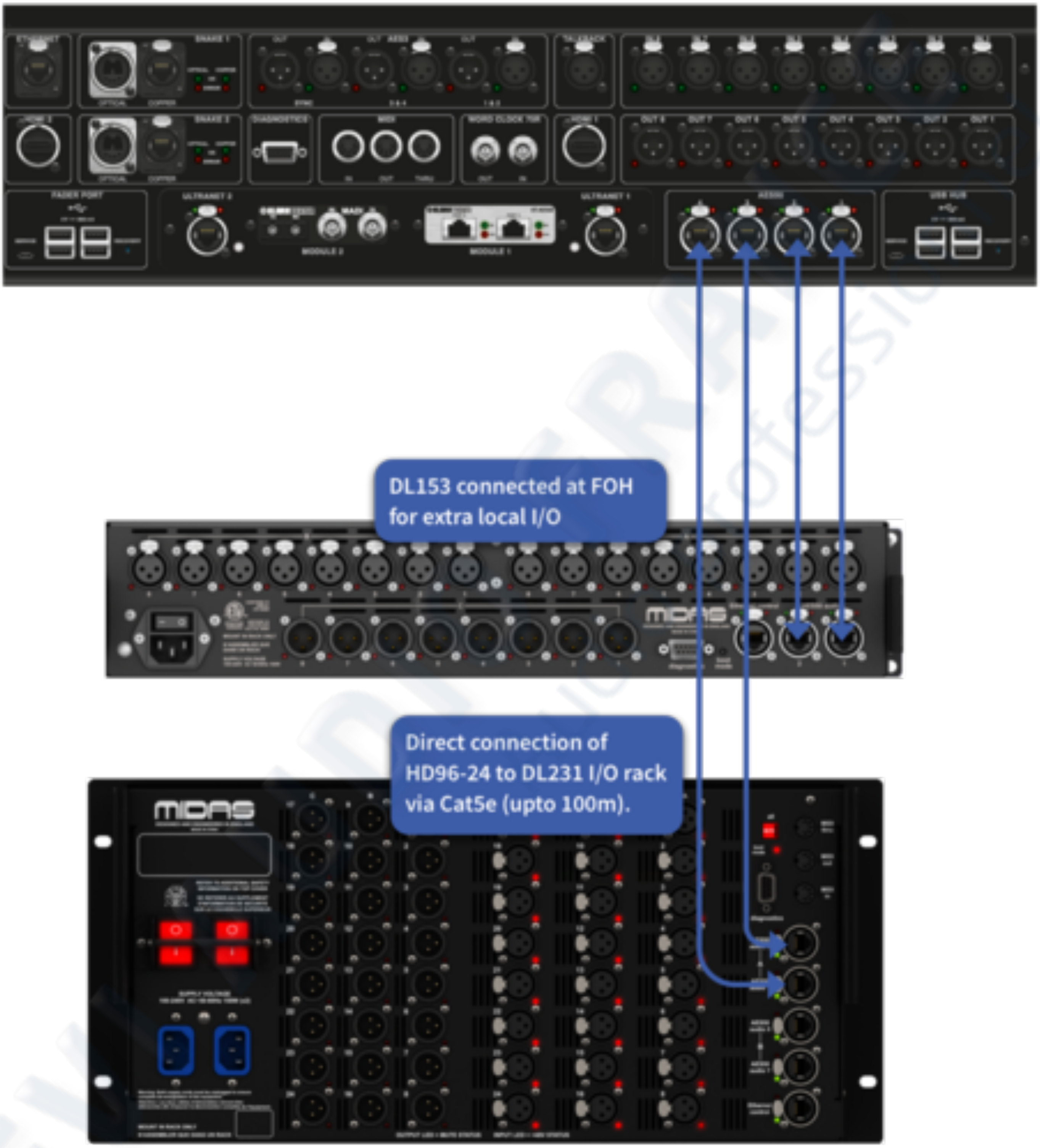
# 3.6 System Components

Below is a list of currently compatible system I/O components with the HD software. When I/O is connected to the HD96-24 system it will be necessary to be update to the latest HD I/O software. The updater is built into the console and guides you through the update process. Once updated, I/O boxes will still be compatible with Pro Series consoles.

| I/O UNIT | |
|---|---|
| **DL231** - 2 award-winning Midas microphone preamplifiers per input with switchable +48 V phantom power 2 dual redundant AES50 network ports with independent phase-locked loop synchronisation 24 electronically balanced output channels can be sourced from microphone preamplifi ers or AES50 ports |  |
| **DL151** - 24 award-winning Midas analogue mic preamps with switchable +48 V phantom power |  |
| **DL152** - 24 active-balanced low impedance line-level outputs |  |
| **DL153** - 16 award-winning MIDAS analogue mic preamps with switchable +48 V phantom power 8 active-balanced low impedance line-level outputs |  |
| **DL154** - 16 active-balanced low impedance line-level outputs<br>8 award-winning Midas analogue mic preamps with switchable +48 V phantom power |  |
| **DL155** - 8 award-winning Midas analogue mic preamps with switchable +48 V phantom power 8 active-balanced low-impedance line level outputs 8 AES3 (AES/EBU) digital inputs and 8 AES3 outputs on XLRF/XLRM |  |
| **DL251** - Audio System I/O This is supplied as a fixed configuration unit with 48 mic/line inputs and 16 outputs. |  |
| **DL252** - Audio System I/O This is supplied as a fixed configuration unit with 16 mic/line inputs and 48 outputs. |  |
| **DL431** - 2 award-winning Midas microphone preamplifiers per input with switchable +48 V phantom power 2 dual redundant AES50 network ports with independent phase-locked loop synchronisation 24 electronically balanced output channels can be sourced from microphone preamplifi ers or AES50 ports |  |

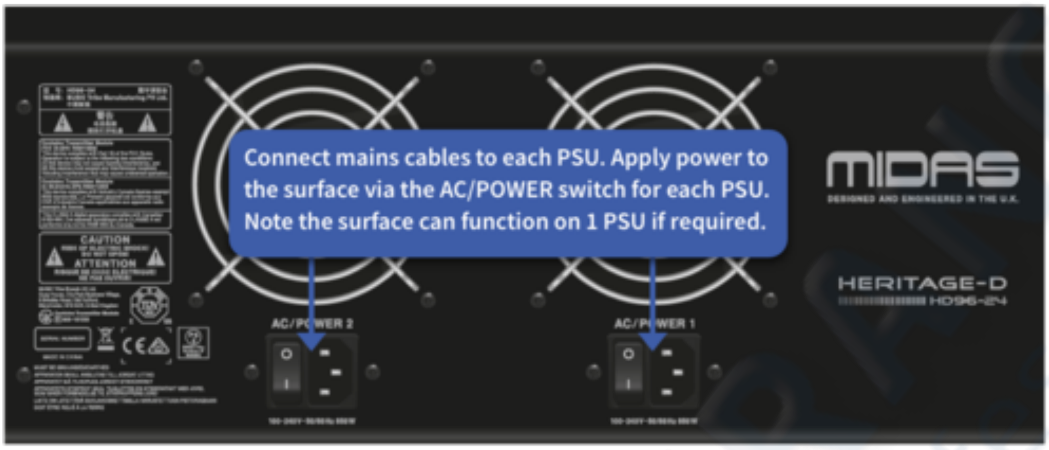
## Connection via HMAC to DN9680, then AES50 connection from DN9680 to DL231

## Direct connection of I/O in AES50 redundant mode



# 3.7 Powering the System

**Important Note:** Make sure your speaker system, in-ears or monitor wedges are muted until the start-up of the system has been completed.




# 3.8 Switching the Control Surface On/Off

After all system interconnections have been made, start up the system by doing the following:

### Switching on the HD96-24

1. Plug the two mains cables into the mains power outlets. Both power supply modules should be supplying power to the HD96-24 surface for correct redundant operation.

2. Plug the connectors of the mains cables into the IEC mains sockets on the rear of the HD96-24.

3. Turn on the power to the HD96-24 surface by switching both D.C. POWER switches on. The surface will boot up. Once the default GUI screen is displayed, it is ready for use.

### To switch off the HD96-24 surface

1. Make sure you have saved any shows, scenes or settings you require.

2. In the GUI, select Menu from the top bar, then press and hold the red SHUTDOWN button at the bottom of the menu until the line traces around the outside of the red button. The shutdown procedure will then initiate.

3. The screen will go blank, the MIDAS logo will briefly be displayed, then the screen will go blank for a second time indicating the shutdown procedure has finished. Once the system has been shut down correctly is it safe to turn off both D.C. power switches (rear of surface).

## 3.9 Setting I/O IDs

Although the programming menu of each type of I/O unit may look slightly different, the procedure for setting up its ID is basically very similar. For full instructions on how to set up the ID of each particular I/O unit, refer to its operator manual.

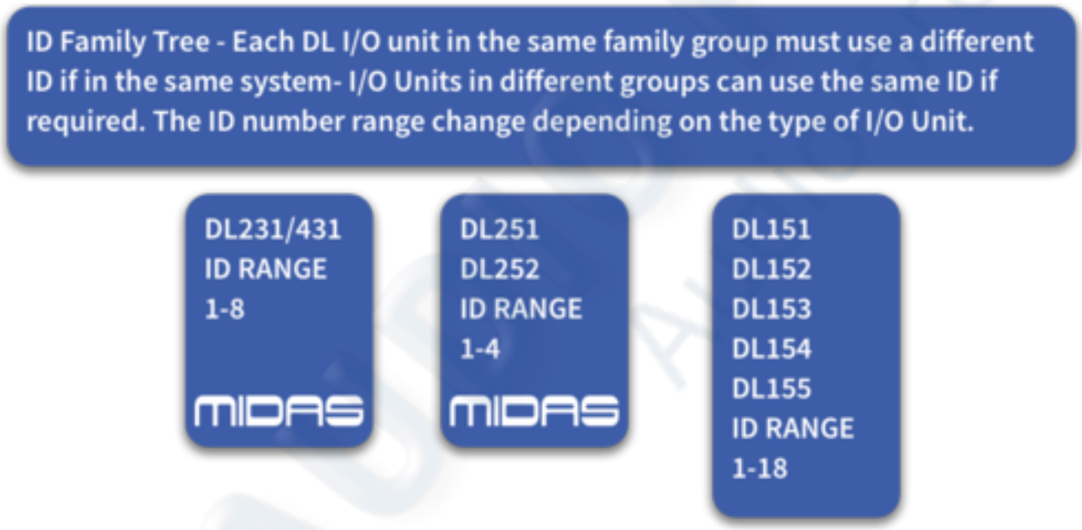
The unit ID number is shown on the LCD screen at the end of the top row of text.

1. Press MENU and hold for approximately two seconds to enter the main menu.

2. Use the UP and DOWN arrow buttons to navigate to the set ID option.

3. Press SELECT to enter the set ID option.

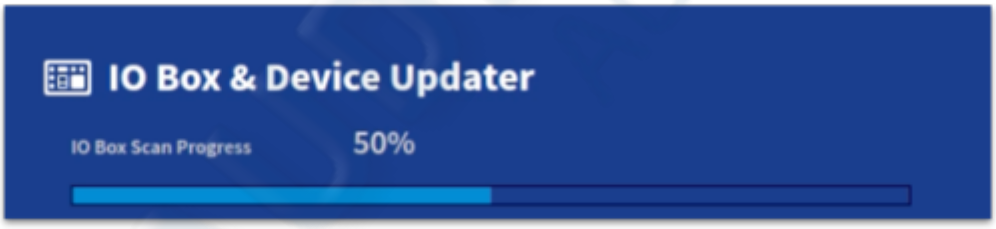
4. Press MENU repeatedly to exit the main menu. (The unit will automatically exit programming mode after 20 seconds of inactivity).



# 3.10 I/O Box Update

**Note:** I/O boxes will need to be updated in order to work with the HD96-24 system. The I/O Box and Device updater can page be found in the Update Manager (side bar menu). With all your I/O connected with the console as master clock, press Sync I/O and follow the instructions. Updated I/O boxes will still work and are fully compatible with Pro Series consoles.


1. Press Sync I/O. Any I/O racks will appear after the I/O scan.




2. FOH I/O connections will be displayed in this section.

3. HMAC I/O connections will be displayed in this section.

4. Each device will display its family type (DL15x, DL231), its current software version and the ID number for the I/O device.

5. Select all currently Sync'd I/O devices ready for update. If you prefer individual racks can be updated.

6. Pressing Update Selected updates all selected I/O to the latest version of HD software.

# 3.11 Internet Connection

The Hd96-24 has a built in Wi-Fi router and Ethernet port to allow a connection to the internet.

### Wired connection:

A DHCP server enables the HD96-24 to request IP addresses and networking parameters automatically from the Internet service provider (ISP), reducing the need for a user to manually assign IP addresses to all network devices. Plugging the Ethernet port into a cat5 socket/port at your venue with an active internet connection will automatically connect you to the internet for updates and mCloud use.

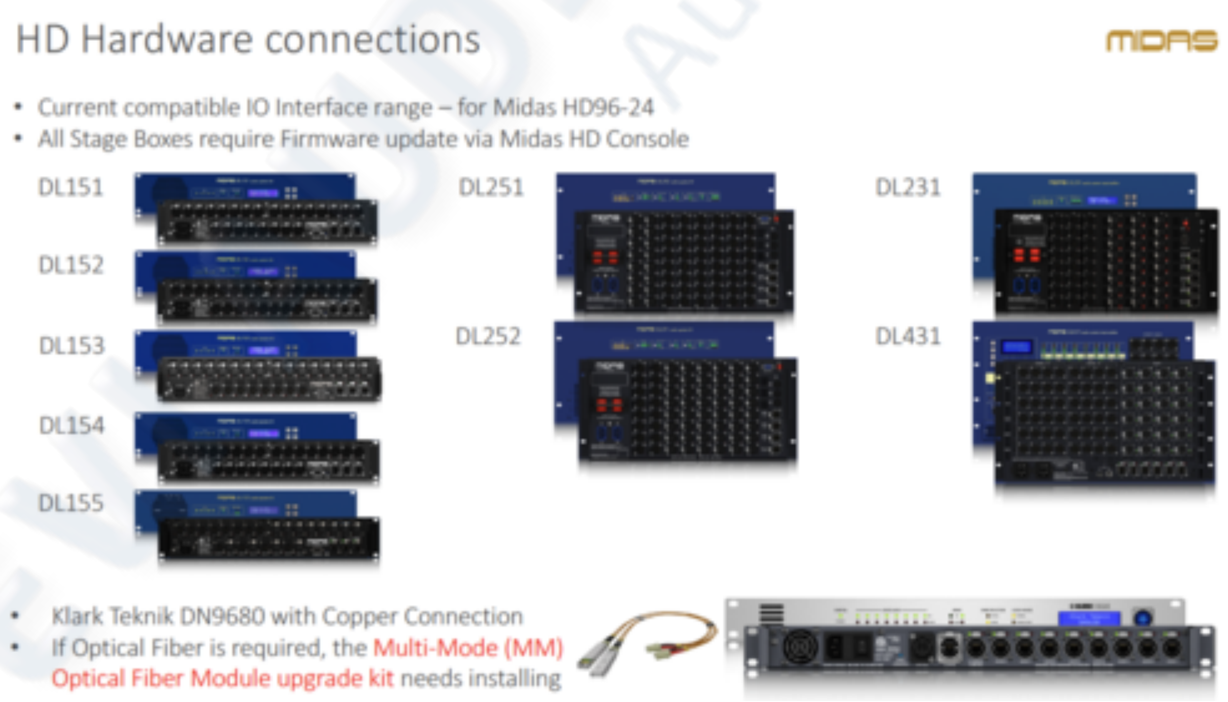
### WiFi connection:

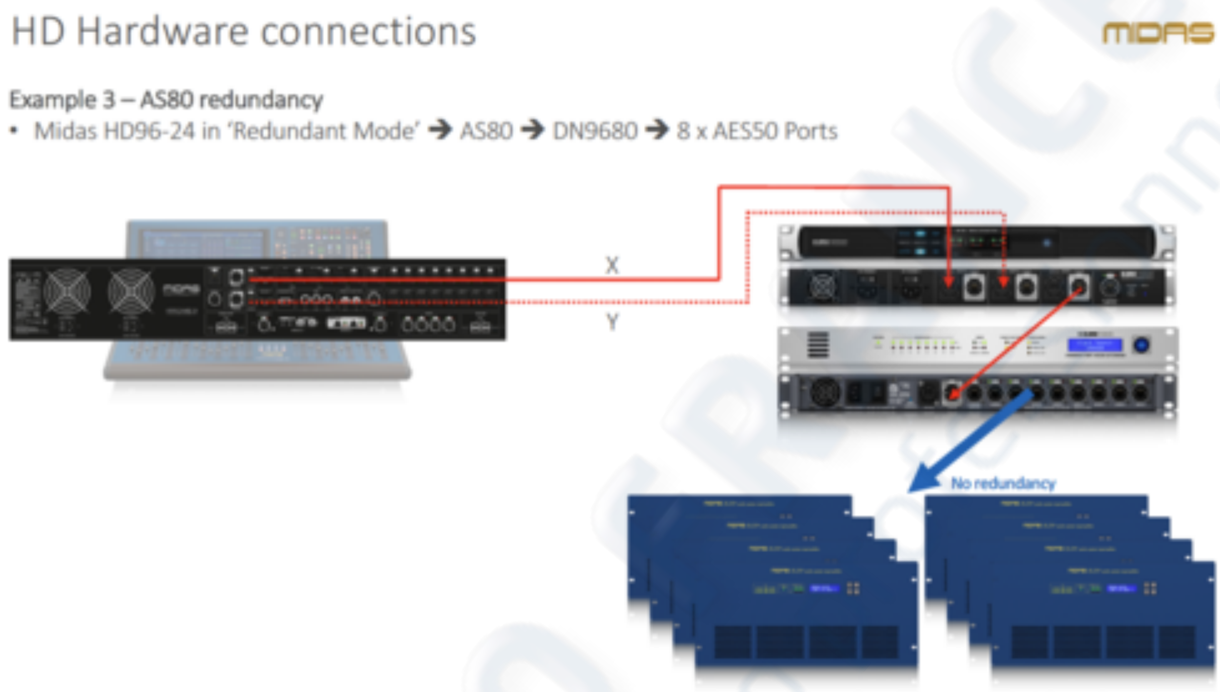
Local WLAN networks will allow you to connect to them with the correct password. Follow the on screen instructions to connect.

**Note:** An active internet connection is required to use mCloud network and for mCloud system updates. Updates and update instructions can also be downloaded from the musictribe.com website.

# 3.12 Hardware Connections

This section details some useful information on hardware connections and how they can be used.

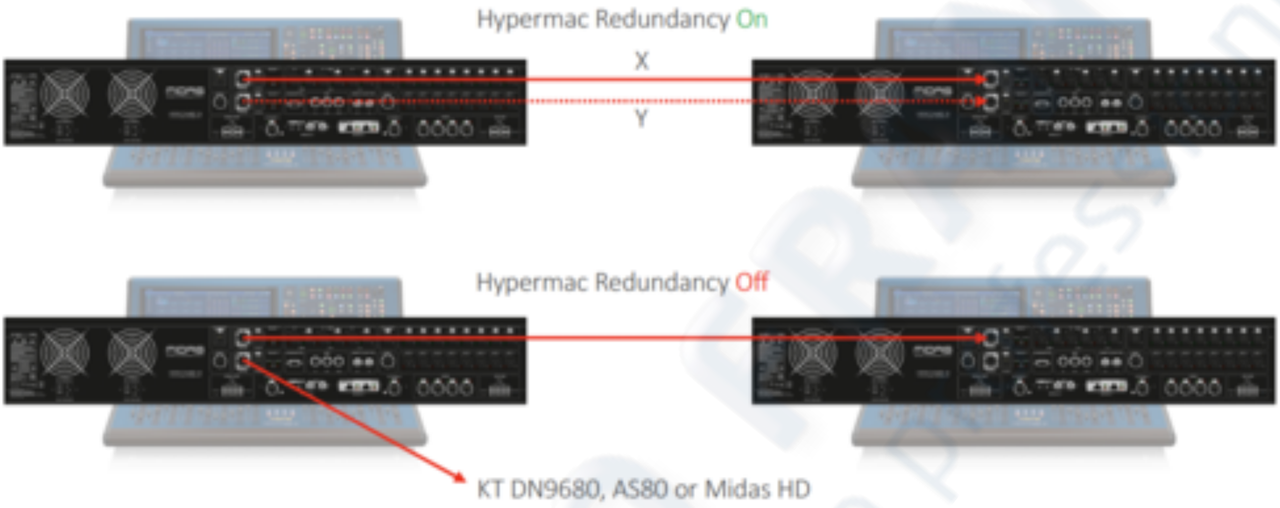



HD Hardware connections
MIDAS
Example 3 – AS80 redundancy
• Midas HD96-24 in 'Redundant Mode' → AS80 → DN9680 → 8 x AES50 Ports
X
Y
No redundancy


HD Hardware connections
MIDAS
Example 2 – AS80 Extension
• HD Snake → AS80 → AS80 → DN9680 → 8 x AES50 Ports
No redundancy
MIDAS
Snake 1
Snake
X
Y

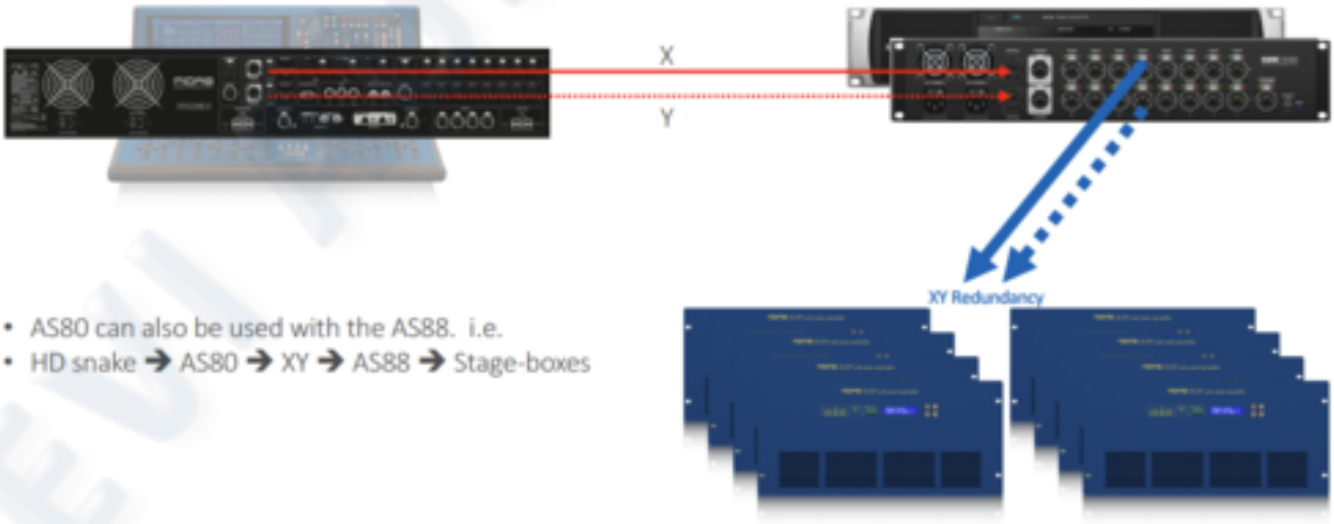
## HD Hardware connections

- Example 5 – Console to Console
- 'Tie line' switched to 'On' in preferences

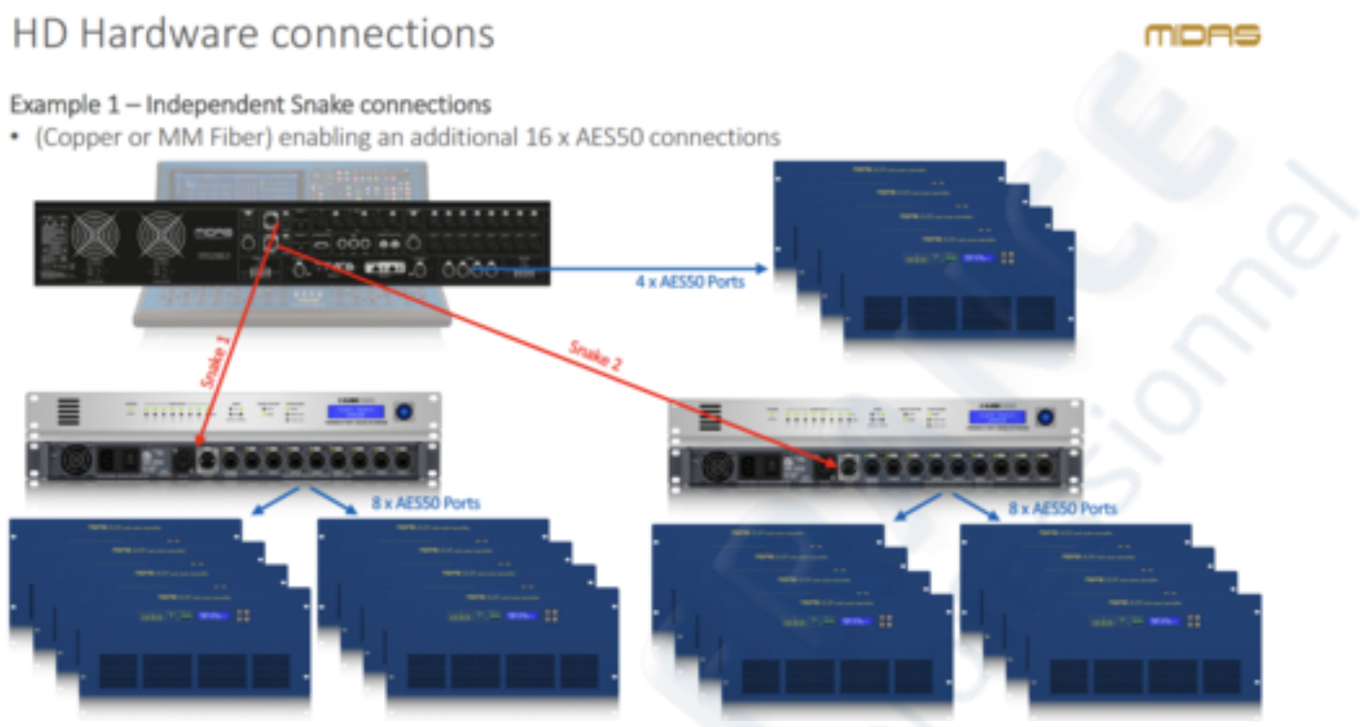


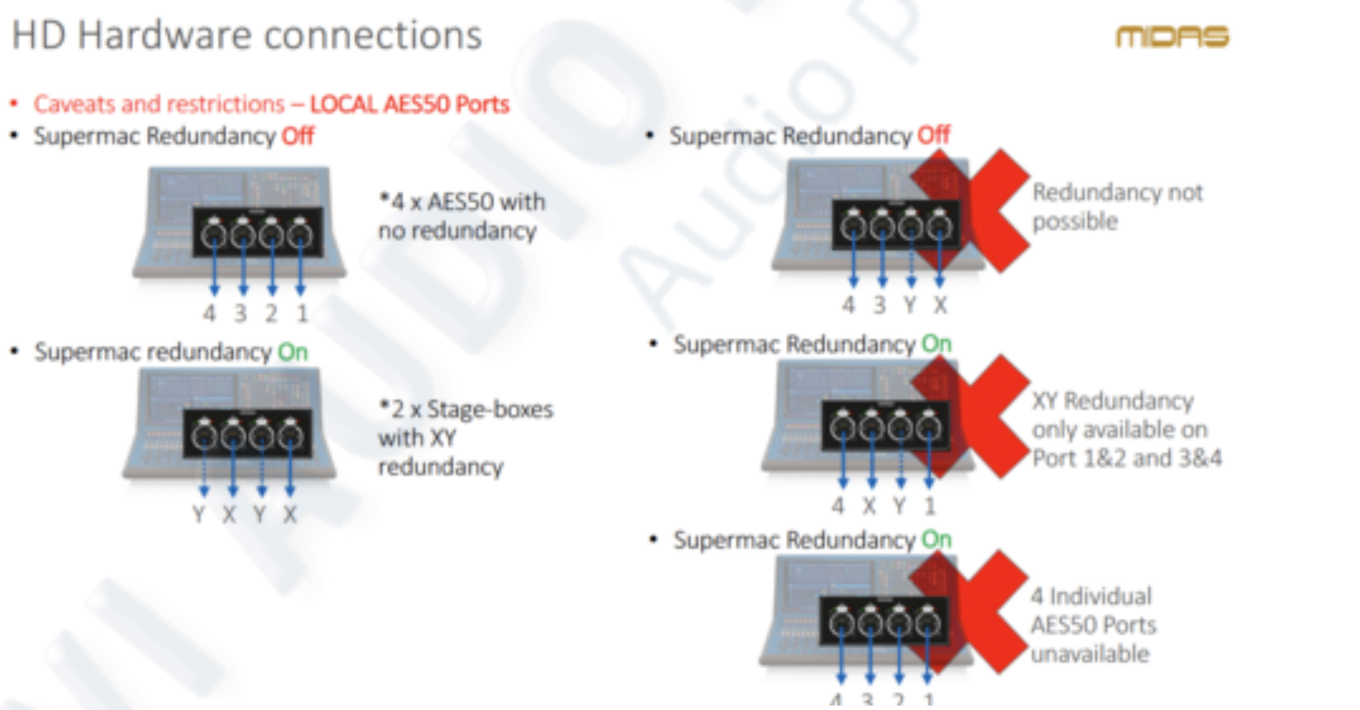

## HD Hardware connections

- Example 4 – Using the AS88 (Release date tbd. Estimated Q4 2021)
- Midas HD96-24 in 'Redundant Mode' → AS88 → 8 x AES50 Ports (X), 8 x AES50 Ports (Y)




- AS80 can also be used with the AS88. i.e.
- HD snake → AS80 → XY → AS88 → Stage-boxes


HD Hardware connections
MIDAS
Example 1 – Independent Snake connections
• (Copper or MM Fiber) enabling an additional 16 x AES50 connections
4 x AES50 Ports
Snake 1
Snake 2
8 x AES50 Ports
8 x AES50 Ports


HD Hardware connections
MIDAS
• Caveats and restrictions – LOCAL AES50 Ports
• Supermac Redundancy Off
*4 x AES50 with no redundancy
4 3 2 1
• Supermac redundancy On
*2 x Stage-boxes with XY redundancy
Y X Y X
• Supermac Redundancy Off
Redundancy not possible
4 3 Y X
• Supermac Redundancy On
XY Redundancy only available on Port 1&2 and 3&4
4 X Y 1
• Supermac Redundancy On
4 Individual AES50 Ports unavailable
4 3 2 1

## HD Hardware connections

MIDAS

**Klark Teknik DN9680**

- Copper Connection
- If Optical Fiber is required, a Multi-Mode (MM) Optical Fiber Module upgrade kit is necessary
- Requires New Firmware (currently HD 0.0.8)
- New – Auto Clock & Switch (Optical & Fiber)
- Not compatible with Pro Series once upgraded

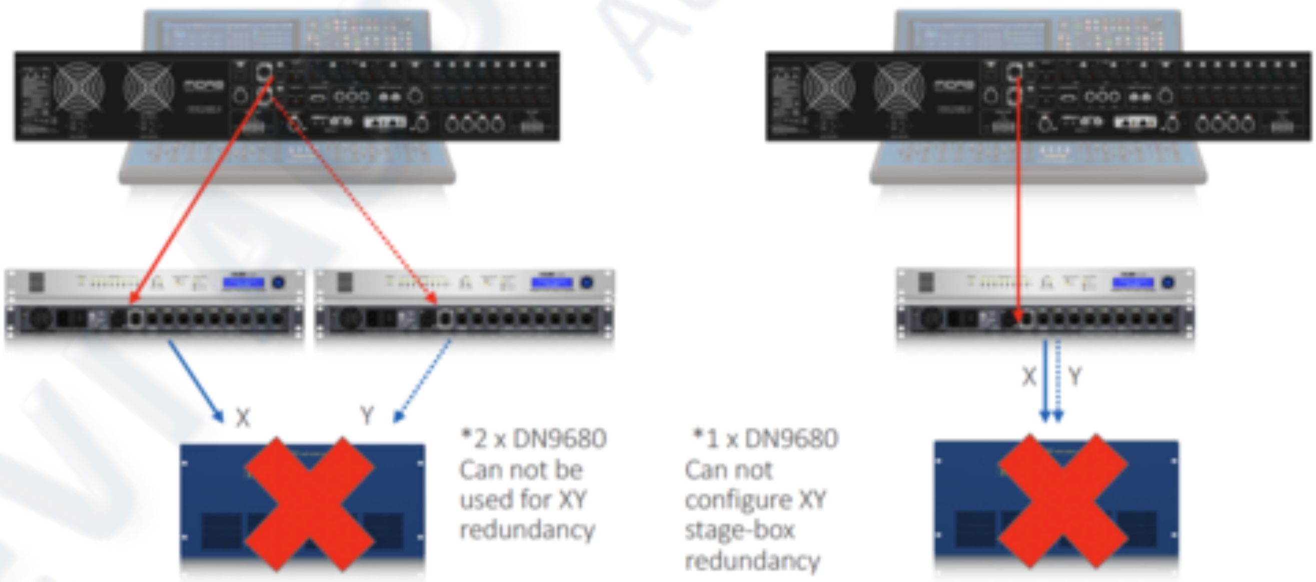
**Klark Teknik AS80** (Est. Release date Q1 2021)

- Snake (Hypermac) Redundancy convertor
- Snake → Snake XY redundant
- Snake XY Redundant → Snake
- Each Port has options for Copper or Fiber
- Redundant PSU
- Not Compatible with Pro Series

## HD Hardware connections

MIDAS

- Caveats and restrictions – DN9680



## HD Hardware connections

- Caveats and restrictions – **Midas HD Connections**
- Dual Hypermac redundancy with DN9680

* Simultaneous Optical and Copper connection for Hypermac redundancy is possible although switching is done manually (on the Midas HD)

- Midas DL251 – AES50 Port C unavailable

# Chapter 4: Before you Start

## 4.1 Principles of Operation

Control surface operation is based on the concept of colours and groups rather than 'layering' or 'paging', which is the case with most digital consoles on the market today. With so many channels available it is far easier to remember them by their user-configured individual/group colour and name rather than their channel number. Tags can also be used to group channels together in order to speed up certain functions, for example, changing the colour of all the drum channels.

The control surface is populated with instantly recognisable controls that are logically distributed in major sections, so that all the controls you need to access most of the time are always on the control surface, while the remainder are only one action away. You can display all I/O meters, both on the control surface and the GUI via the Console View workflow, to give instant monitoring and metering feedback.

## 4.2 Operating Modes

You can change certain aspects of the control surface operation by assigning different tasks to certain areas of the control surface. Later in the Navigation, chapter these functions will be explained and how custom layouts can be created to suit your particular working method.

## 4.5 Hints and Tips

Check the HOME and Console View screen frequently, this can be found in the Home workflow and looking at overview or by pressing the HOME key next to the global shortcuts area. This provides at a glance an overview of the control surface's input/output status.

The multi-edit Manchino page is a great place to set various inputs or outputs to user defined levels or settings e.g. setting all faders to 0dB, setting all contributions into a particular aux to be Pre-fade, or routing a large number of paths to the Stereo bus. Details of how it works can be found in chapter 8 basic operation chapter and in more detail in chapter 33 Manchino advanced.

## 4.6 Setting up a User Profile

The HD96-24 incorporates a system of User Profiles for storing console files and set-up information. Each user of the HD system can have their own profile which keeps all of their show files set ups and presets on the console and via the mCloud network if the HD system has an active internet connection.

After the system has been registered and is turned on for the second time you will be asked to either log in or create and add a new profile.


1. Current selected profile. Press Login and Continue to enter system.
2. Log Out of the system with a short button press.
3. Add a New Profile or Change User profile by pressing here.
4. Securely log out of your mCloud account (password forgotten).
5. Logout of your mCloud account (password is remembered for next log in).
6. Safe Mode (the show database is not available in safe mode, but you can still mix).
7. Activates the lock screen.
8. Shutdown the Console.
9. Allows the setup of the console to be changed.

# 4.7 Saving your work

Shows are automatically stored to your mCloud account and uploaded to the mCloud if the console is connected to the internet for instant show back-up. Auto Save updates your current show and a history of shows can be viewed via your mCloud account on a computer.

We also recommend that you save your work regularly to a USB stick while carrying out the procedures included in this chapter. Not only is this good practice during normal operation, but in this instance, it may save you from losing some set-ups that could prove useful later on. To do this, tap the Automation icon or scene display at the top of the screen, press the Show Manager icon, create a new show and export your show to a USB stick. Now continue reading through the remainder of this section, following the instructions carefully.


1. Press in the scene window to open the scene recall page, press the Show Manager icon to navigate to the show page (or select from the side bar menu).
2. Ensure Auto Save is active (stores every time a scene is stored).
3. A list of shows in your account can be seen here. (Current active show highlighted in light blue).
4. Press New to create a new show.
5. Press Open to load the selected show.

**Note:** Chapter 8 Basic Operation gives more details on opening a show and the various load options.

# 4.8 Saving a show versus storing a scene

It is important to understand the differences between saving a show and storing a scene. Storing a scene saves the current settings of the system to the show file. Scene data is never updated unless you manually store a scene.

In the event of power loss, the console attempts to recover in the same state. i.e. user, and show loaded with identical settings.

Saving a show copies the show file onto the internal solid-state disk of the control surface. This provides you with a ‘permanent’ copy, provided you shut down the system properly as detailed in the following section. You also have the option of saving your show to the Midas mCloud. This gives extra security to your work and allows your show file to be restored to a console even if you have lost your USB stick.

# 4.9 Shutting Down the Control Surface Correctly

When switching off the control surface, we recommend that you use the shutdown option in the GUI menu.

Hold Shutdown for a short time while the line traces around the outside of the button. The surface will then start the shutdown routine, the screen will go blank, the MIDAS logo will briefly be displayed, then the screen will go blank for a second time indicating the shutdown procedure has finished. Only once the system has been shutdown correctly is it safe to turn off the power switch. By using shutdown, the cached copy of the show data, which is maintained by the system, is automatically stored.

If you don’t use shutdown the console will attempt to load the show and scene you were on at time of power down. You do not need to manually load the show.

# 4.10 mCloud System

The mCloud system handles all show file storage at its basic level. Imagine leaving your USB stick at home but not worrying as once you sign into your mCloud account on the surface you can see all your shows in one convenient place. mCloud usernames can be 3 to 18 characters long. Passwords have to be between 5 and 32 characters long.

Below is a list of file and account sync status icons:


Status if connected to the mCloud:

- **Synced** All versions of this show have been pushed to the mCloud; any newer mCloud versions have been synced to the console
- **Conflicted** Edits have been made on both console and cloud: awaiting user to select the correct. Conflicts can also occur using Online Editor (OE) or 2 consoles.
- **Error** The sync service encountered a problem trying to sync this resource (e.g. due to an issue communication with the mCloud)
- **Syncing** Edits are being pushed to and/or pulled from the mCloud.
- **Pending** One or more newer versions have been created on the console and will be synced shortly.

Connected to mCloud means the console can reach the mCloud server, the current user is mCloud-enabled, and a valid password has been entered (or a valid token saved from a previous session)

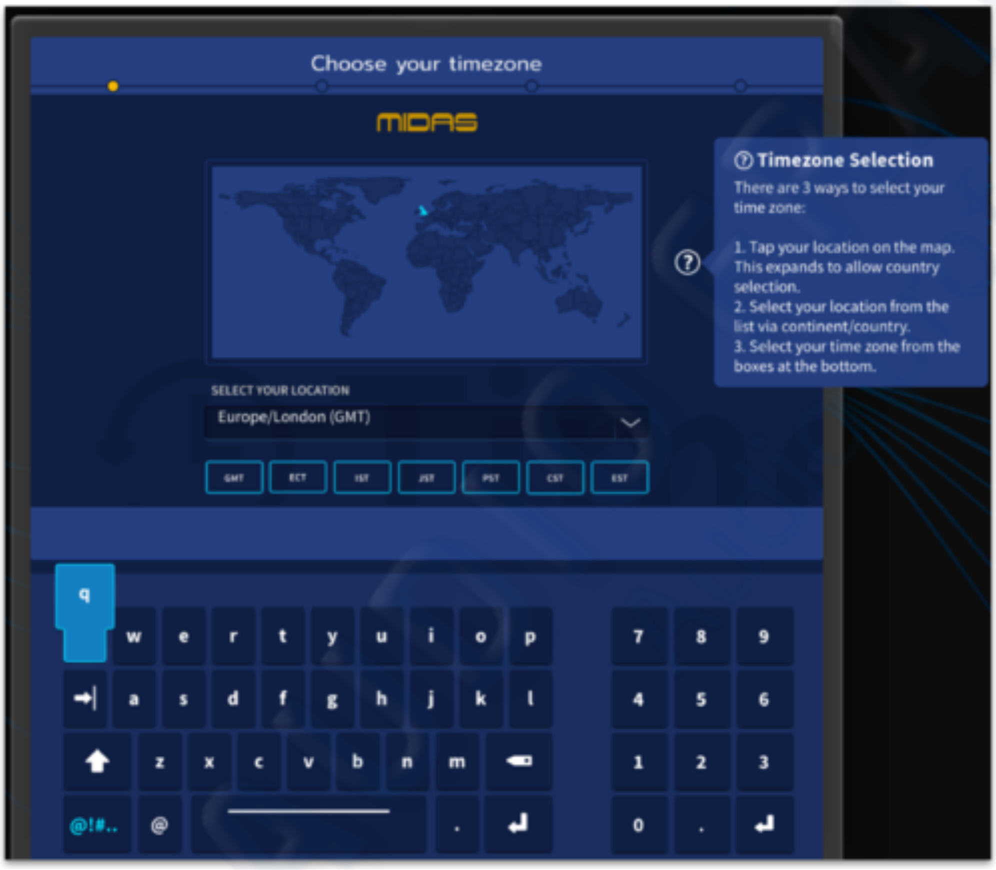
# 4.11 mCloud Support

Please check [www.cloud.midasconsoles.com](http://www.cloud.midasconsoles.com) for help and support with the mCloud system.

# 4.12 Initial Setup

When the HD96-24 is switched on for the first time you will be presented with the time zone selection page. Select time zone, apply and then shutdown the console to change the timezone settings.

In order to enjoy the full benefits of the mCloud system it is advised an internet wired or wireless network connection is available. You will then be guided through the various pages to set up and login into your mCloud account.



# 4.13 Setting up a user profile

The HD96-24 incorporates a system of User Profile for storing information about the console set-up and other User info. Each person using the HD system can have their own profile which keeps all your show files and other useful information on the console and via the mCloud network if the HD system has an active internet connection.

When a system is first turned on you will be asked to create or add a new profile. If you have already set up your mCloud account, you may login via mCloud. Subsequent use of the system will ask you to either login or change user as desired.

Select your user profile if already used with the system by pressing your profile.


Use Login via mCloud to reach the mCloud login page. Enter your name, password and click Sign In.


Choose the Change User option if you wish to switch or also create a new user profile.


Press Create a New User then enter a memorable user profile name. Press Create New User to continue and select your newly created profile to login. You also have the option to create an mCLoud account at the same time. A 4-digit pin can be added at this point for extra security.


**Note:** User Data is not accessible in safe mode. The console may still be used in its current state and will continue to pass audio. You cannot access storage to load or save settings, shows, etc. You can however load a show file from a USB stick if required.

# 4.14 Safe Mode

In an emergency Safe Mode can be used to gain access to the console. If you sign out of your user profile and head to catering once doors have opened it is possible to use safe mode to allow, for example, the system tech to adjust the background music, if required, without having access to your mCLoud account.

To enter Safe Mode, long press the Enter Safe Mode button until the line traces around the button. Next, select the Safe User Icon and press Done to gain access to the system.

**Note:** In Safe Mode, you cannot access or save show data.

# Chapter 5 : About the control surface

## 5.1 Overview of the Control Surface

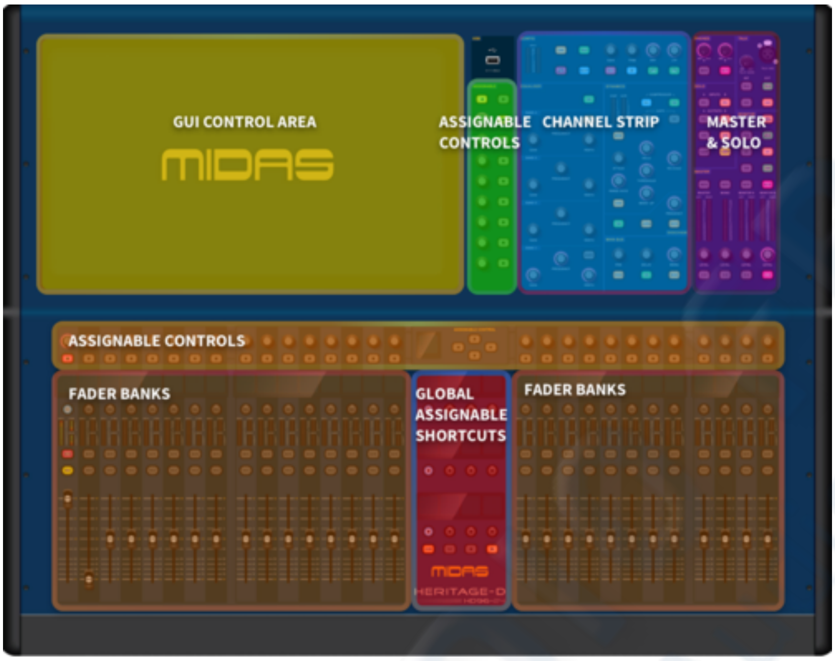
The HD96-24 has been designed to give the operator easy-to-use touchscreen controls along with familiar analogue style controls. This clever combining of working methods ensures any engineer can walk up to this console and instantly feel at home, but when required can delve deeper into the system to achieve spectacular results.

The surface is constructed on a robust MIDAS steel frame chassis similar to those used in established MIDAS analogue products. All associated power supplies, computer motherboards, Wi-Fi router, Bluetooth, memory, graphics cards etc. are housed within the surface, which also contains a digital audio router box that supports local I/O connectors on the rear panel. Substantial forced air-cooling is provided by a bulkhead and large (but slow moving) internal fans. The large capacitive touchscreen displays a large quantity of information and can be customised to match your workflow to make mixing a pleasure. Using modern day gestures from mobile phone and tablet technology, such as pinch and swipe, makes parameter manipulation even faster and more responsive with up to 10 simultaneous touch points.

The HD96-24 system is designed to be easy to see for colour blind people. Great care has been taken to make the system visible to as many types of colour blind people as possible.

# 5.2 Control Surface Layout

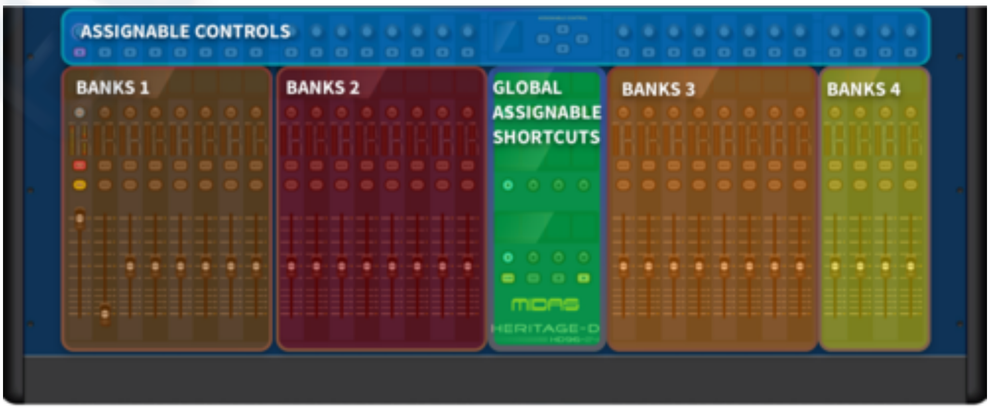
The HD96-24 surface can be spilt in to 7 distinct areas making operation quick and precise with all controls close to hand. The areas are defined in the diagram.




The HD96-24 has 4 fully assignable fader banks split into three banks of eight faders and one bank with four faders, each with individual full colour ultra-bright LCD displays. Any section can be assigned to any function, be it inputs, outputs, POPs, VCA, Matrices or Masters. This concept allows the user to fully customise the surface to suit their mixing preferences.

The global assignable shortcuts area can be used to provide many simple and complex functions with macro style controls at your fingertips. E.g. Pop group selection, triggering macros or automation recall.

The assignable controls above the faders and to the side of the GUI can be fully customised to suit your workflow. Functions can be changed quickly with the cursor arrow controls. For e.g. altering pan position, aux control or gain changes.



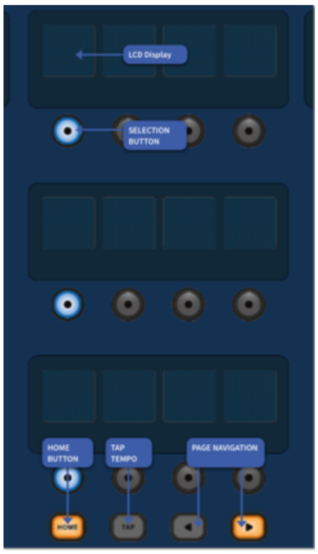
## 5.3 Channel Strip Layout

Each channel strip within a bank provides:


1. **LCD DISPLAY -** A high-resolution display providing metering, channel information and flip status and local parameter values.
2. **SEL -** Selects the channel for a variety of operations, including adjusting parametersfrom the GUI and assigning to the channel detail area.
3. **COMP -** Compressor gain reduction meter.
4. **INPUT -** Input metering.
5. **GATE -** Gate gain attenuation meter.
6. **MUTE -** Press the MUTE button to mute the channel.
7. **SOLO -** Press SOLO to listen to the channel signal
8. **LEVEL -** The fader is touch sensitive providing level control from ∞ to +10dB (or +6dB if contributing to an output bus)

## 5.4 Global Assignable Shortcuts & Home and Tap button function

The Global Assignable Shortcuts area in the centre of the surface allows various functions to be placed within easy reach of the user. Twelve full colour LCD displays with **SEL** selection buttons show a great deal of information and allow for complex operations to be recalled with one button press.




**HOME** The HOME key is also found in this area. When pressed, the HOME workflow is brought to the GUI.

**TAP** The TAP button is used to set the tempo for effects assigned to the Global TAP tempo function. Commonly 8 taps are required for an accurate tempo.

Arrow Keys

These two keys tab through the various pages of the global assignable shortcuts pages which can be fully customised.

# 5.5 Channel Detail Area

The familiar looking Channel Detail Area is used for hands on control with sections for Config, Equaliser, Dynamics, Phones, Talk, Solo, Monitor, Main Bus and Master. This area makes using the surface easy to use with a familiar analogue feel. The 4 x Change Over C/O buttons for Master, Mono, Monitor A and Monitor B assign the control to the fader below for quick control.

## 5.6 Front and rear panel connections

The surface has connector panels on both the front and rear. The front connector panel to the right of the GUI has an XLR socket and a USB sockets for connecting a talk mic and USB device, respectively. Under the armrests at either side there are two 6.35 mm headphone sockets which link to Mon A and Mon B respectively.


A connector panel on the rear of the control surface has two main sections. On the left are two mains power inlet and ventilation assemblies, with a DC power switch below. The right-section contains connections for the Snake/Fibre Multicore, Ethernet, four AES50 ports, eight analogue audio inputs and outputs, three AES3 inputs and outputs, diagnostics, word clock, Two HDMI™ external monitor outputs, Midi, Talkback, two expansion card slots and USB Hub.

# 5.7 External Interfaces and Peripheral Devices

Various devices can be used with the HD96-24.

**MIDI** Standard 5-pin DIN connectors are housed in the rear panel for use as MIDI IN, OUT and THRU ports. These are also fitted on some I/O units (DL231, DL251 for example) and, therefore, are available at both the FOH and the stage locations.

**USB** 2 x 4way USB 3.0 hubs are provided on the rear of the HD96-24. In addition, a USB port can be found to the right of GUI screen for convenient file transfer.

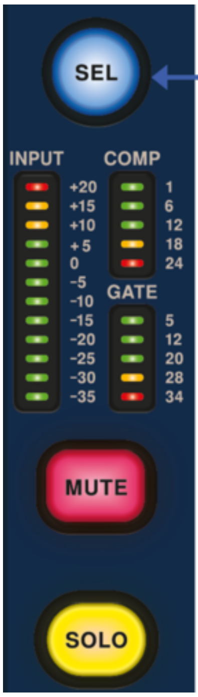
**Note:** The Recovery buttons next to the USB hubs should NEVER be pressed unless advised to by the Midas support team. If, for some reason one of the button has be pushed, audio will stop, and the screen may turn off depending on which button is pressed. The console will then require a power cycle to return to normal function.

**HDMI** the HD96-24 has 2 HDMI™ connections on the rear panel to connect extra displays.

**Ethernet Port** used for connecting the HD96-24 to a network.

**Ultranet** 2 x Ports for connecting Ultranet enabled devices.

# Chapter 6 : Navigation




## 6.1 An Introduction to navigation

The HD96-24 has full multi-touch screen control. This gives the user some of the feeling of working with analogue consoles, whilst at the same time, incorporating modern ways of interacting with everyday products like Smartphones and Tablets. One of the advantages digital consoles have over analogue ones is that their channel count is not limited by the control surface hardware. However, this means that only a certain number of channels can be shown on the control surface at any one time, while the others are available at the touch of the screen (GUI) or assignable shortcuts area.

Pressing the SEL (select) button on any input or output will bring the chosen channel to the surface and GUI controls. Pressing SEL again on a stereo channel will switch the view to the right side of the channel. From here, adjustments to many common parameters can be changed such as Gain, EQ, Dynamics or Pan position. This way of working has a familiar feel that you will understand and be comfortable to use. Mute will stop audio passing and solo will send the audio of the selected path to either the A or B solo bus depending on path settings.


## 6.2 Navigation via the surface detail area

The surface is laid out with a familiar analogue feel for easy operation. Everyday functions are available for hands on operation. All functions are mirrored in the GUI with parameter values highlighted on touch. This allows you to make changes quickly to the selected channel or path.


## Config

1. **Meters** - 12 LEDs Stereo input meters display signals from -35 dB to 20 dB.
2. **Link** - Links the currently selected input channel to the next input. Note outputs always link odd to even, i.e. Aux 1 to 2 etc.
3. **Talk** – Send the talk bus to the select channel.
4. **Gain** – Adjust the gain of the input channel pre-amp.
5. **Trim** – Adjust the trim level for inputs -40 to 20 dB or for Aux, Matrices and Master Busses -12 to 6 dB.
6. **HPF** – High Pass Filter (HPF) control with a range from 10 Hz to 10 kHz (inputs only).
7. **LPF** – Low Pass Filter (LPF) control with a range from 40 Hz to 20 kHz (inputs only).
8. **Dir Mute** – Mutes the direct in/out of the currently selected channel if patched.
9. **Insert** – Switches on the insert point on the selected channel if patched.
10. **48v** – Activates 48v phantom power on the selected input channel.
11. **Ø** - Polarity Switch. Changes the polarity of the selected channel by 180° (often inaccurately called phase reverse as the button only inverts polarity).
12. **HPF** on – Activates the HPF (inputs only).
13. **LPF** on – Activates the LPF (inputs only).


## Equaliser

1. **On** – Turns the Equaliser on for the selected channel.
2. **Gain** – Each band has +- 16.2 dB of range.
3. **Frequency** – Each band as a frequency range of 16 Hz to 25 kHz.
4. **Width** – The width or Q of an EQ band can be changed from 0.3 to 5.3.
5. **Shape** - Changes the shape of Band 4. For inputs, the shape options include, Bell, Bright, Classic and Soft. For outputs, the Shape button has Shelf, LP 6 dB, LP 12 dB and Bell modes.
6. **Shape** - Changes the shape of Band 1. For inputs, the shape options include, Bell, Deep, Classic and Warm. For outputs, the Shape button has Shelf, HP 6 dB, HP 12 dB and Bell modes.

## Dynamics


1. **Compressor GR Meter** – Compressor (Comp) Gain Reduction Meter (Range -1 dB to – 23 dB).
2. **GATE GA Meter** – Gate Gain Attenuator Meter Range (-1 dB to -34 dB).
3. **SEL Comp** – Selects the compressor setting for the selected channel.
4. **ON** – Turns the compressor on for the selected channel.
5. **SEL Gate** – Selects the gate setting for the selected channel.
6. **ON** – Turns the gate on for the selected channel.
7. **Mode** – Selects the mode of the compressor (Corrective, Adaptive, Creative and Vintage) or gate (Gate or Ducker).
8. **Attack** – Controls the attack settings of the gate or comp.
9. **Hold** – Controls the hold value of the gate. Hold is the amount of time the gate is open until the release part of the gate starts.
10. **Release** – Controls the release characteristic of the gate and comp.
11. **Range/Ratio** – Range relates to the gate and controls the amount of signal allowed to pass when the gate is closed. Ratio relates to the comp. With a ratio setting of 3:1 for every 1 dB above the threshold point the signal will be turned down or compressed by 3 dB.
12. **Threshold** – Adjust the point at which either the gate opens, or compression starts to take place.
13. **Knee** – Changes the compression knee setting (Hard, Medium or Soft).
14. **Make-Up** – Adds gain to the compressors output. This allows you to balance the levels of the compressor when on and off by increasing the make-up gain to match the amount of gain reduction taking place. (Range 0 dB to 24 dB).
15. **Sidechain Frequency** – Set the frequency that the sidechain of the gate or comp listens to in order to give tighter control of a certain range of frequencies.
16. **On** – Turns the sidechain of the gate or comp on/off.
17. **Listen** – Sends the selected sidechain frequency to the solo bus for monitoring and to give accurate adjustment.
18. **Width** – Changes the width of the sidechain frequency for the gate and comp (0.1 Oct, 0.3 Oct, 1 Oct and 2 Oct).

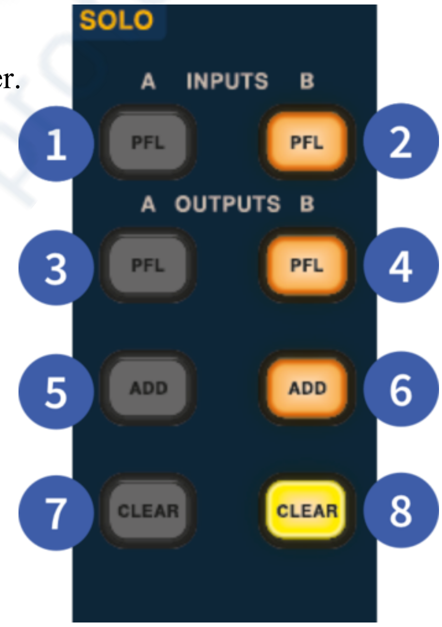
## Phones


1. **Phones A** – Level control.
2. **Phones B** – Level Control.
3. **Phones A** – Mute button.
4. **Phones B** – Mute button.

## Talk

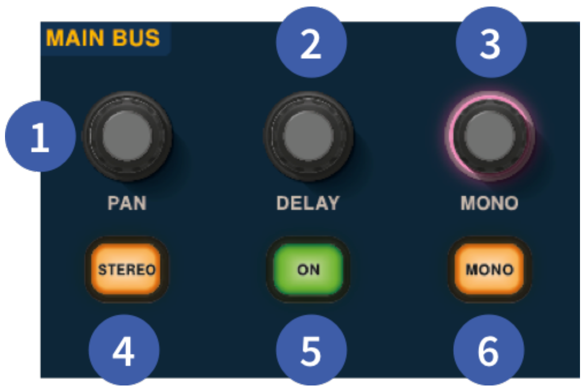

1. **Talk Gain** – Adjust the gain of the surface Talk Mic input.
2. **Talk Mic** – XLR input for local talk mic.
3. **Talk INT** (Internal) – Allows the talk mic to be sent to the internal talk bus. For example, it can be used to send your talk mic into a channel to test signal flow.
4. **Talk EXT** (External) – Allows your local talk mic to be routed to the Ext Talk.
5. **OSC INT** – Activates the oscillator on the selected internal bus.
6. **OSC EXT** – Sends the oscillator to the Ext Talk bus. This can be used to send the oscillator to a channel if required by patching the Ext Talk out in the monitor patching page to a channel.

## Solo



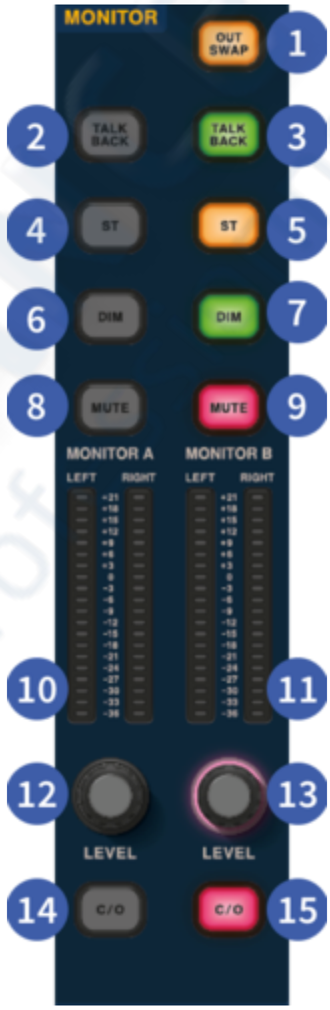

1. A PFL – Any input routed to the monitor A bus is pre fader.
2. B PFL – Any input routed to the monitor B bus is pre fader.
3. A PFL – Any output routed to the monitor A bus is pre fader.
4. B PFL – Any output routed to the monitor B bus is pre fader.
5. Add (A) – Additive mode allows more than one input channel to be listened at once on the A solo bus.
6. Add (B) – Additive mode allows more than one input channel to be listened at once on the B solo bus.
7. Clear (A) – Clears any current solo A selections.
8. Clear (B) – Clears any current solo B selections.

## Main Bus




1. **Pan** – Controls the position of the signal in the stereo field.
2. **Delay** – Adjusts the delay time for the selected channel.
3. **Mono** – Adjust send level to the Mono Bus from the selected channel.
4. **Stereo** – Sends the selected channel to the Stereo Bus.
5. **On** – Turns the selected channel's delay time on.
6. **Mono** – Sends the selected channel to the Mono Bus.

## Monitor



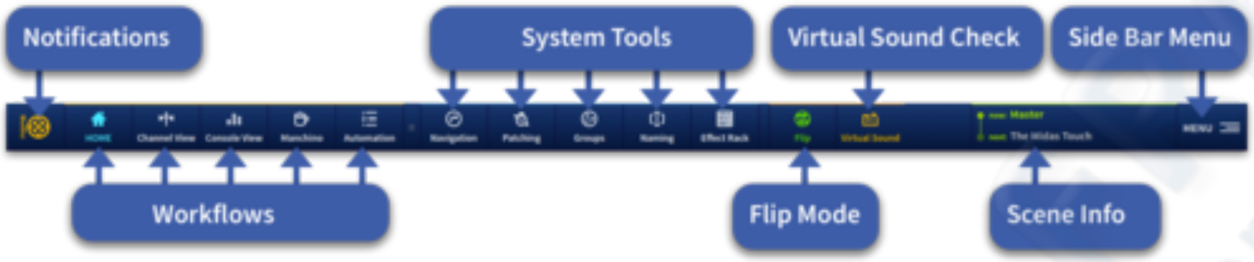

1. **Out Swap** – Completely swaps the A and B Monitor busses over meaning A becomes B and vice versa. For example, this allows you to hear a wedge monitor mix on the in-ear monitor bus if desired without having to re-patch.
2. **Talk Back** – Allows the rear line level talkback input to be directly inputted into the Monitor A bus.
3. **Talk Back** – Allows the rear line level talkback input to be directly inputted into the Monitor B bus.
4. **ST**- Routes the stereo bus to the monitor A bus.
5. **ST**- Routes the stereo bus to the monitor B bus.
6. **DIM A** – Turns the level of the Monitor A bus down by 6 dB.
7. **DIM B** – Turns the level of the Monitor B bus down by 6 dB.
8. **Mute** – Mutes the Monitor A bus.
9. **Mute** – Mutes the Monitor B bus.
10. **Monitor A Metering** – Stereo 20 LED meters, -36 dB to 21 dB.
11. **Monitor B Metering** – Stereo 20 LED meters, -36 dB to 21 dB.
12. **Monitor A Level** – Level control. Maximum level 10 dB.
13. **Monitor B Level** – Level control. Maximum level 10 dB.
14. **C/O** – Sends the monitor A level control to the fader directly below for easy adjustment.
15. **C/O** – Sends the monitor B level control to the fader directly below for easy adjustment.


## Master

1. **Master Mute** – Mutes the Master Stereo Bus.
2. **Mono Mute** – Mutes the Mono Bus.
3. **Master Meters** – Stereo 20 LED meters, -36 dB to 21 dB.
4. **Mono Meter** – 20 LED meters, -36 dB to 21 dB.
5. **Master Stereo Bus Level** – Level control. Maximum level 10 dB.
6. **Mono Bus Level** – Level control. Maximum level 10 dB.
7. **C/O** – Sends the Master Stereo Bus level control to the fader directly below for easy adjustment.
8. **C/O** – Sends the Mono Bus level control to the fader directly below for easy adjustment.

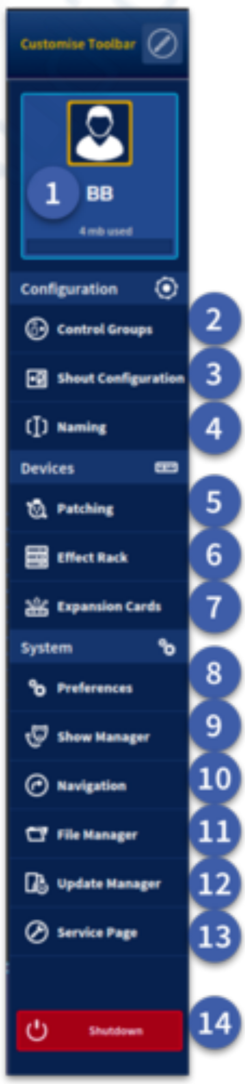
## 6.3 Navigating via the GUI Menu Bar

The top menu bar in the GUI is where various different pages can be selected. These include workflows (such as Automation, FOH, Channel View etc) and tools (such as Navigation, Patching, Groups, Naming and Manchino). There are also buttons for Flip mode, Virtual Sound Check and Current Scene (if an automation file is loaded). The top bar menu can be customised if desired (found in the Console Menu in the Side Bar Menu).




## 6.4 Side Bar Menu

To the right-hand side is the side bar console Menu icon. When selected it opens the side console menu with various options. This is a fixed list. At the top is the Customise Toolbar where the late out of the Menu Bar can be changed explained in Chapter 31.



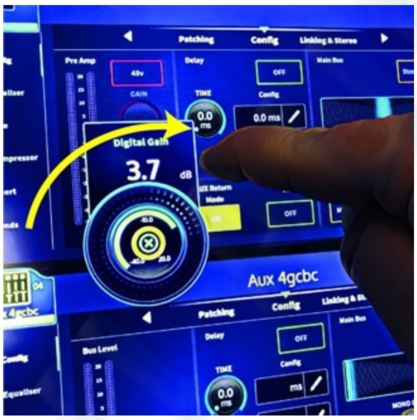

1. **Current User Profile** - Press to go to the User Profile Login Page.
2. **Groups** – Set POP, VCA, Mute and Talk Groups.
3. **Shout Configuration** -  Set up for the 16 x 12 shout mixer.
4. **Naming** – All channels types can be named in this workflow.
5. **Patching** – Set your I/O and other patching.
6. **Effects Rack** – Overview of all the effects currently used in the system.
7. **Expansion Cards** – Settings for the 2 x CM 1 slots and 2 Ultranet ports.
8. **Preferences** – Settings for the console can be found here.
9. **Show Manger** – Load a new show or change the playlist. The Show editor is also in this page.
10. **Navigation** – Surface set up for Fader Banks and Assignables.
11. **File Manager** – Import/export of files plus spreadsheet import of settings.
12. **Update Manager** – All system and I/O updater pages.
13. **Service Page** – For Midas use only.
14. **Shutdown** – Long press to turn the system off.

# 6.5 Widgets and Rotary Controls

A widget is a component or self-contained window within the GUI that enables a user to perform a function. It displays various information about the console's status. This is the heart of function control and makes rapid adjustments of parameters quickly available to the engineer. Within each widget on the left are several different areas or pages to access different parameters. Each area then has further option tabs across the top of the window for various different types of parameter control. In addition, there is the Channel Viewwhich gives the user a more overall view of a full channel and Console View which shows an overall view of a console set-up.

**Note:** if a channel is soloed the icon will turn yellow.


Press and hold any rotary control within the widget GUI area will result in a pop-up rotary control. By moving your finger around this rotary pop-up, you can adjust the parameter with large steps or values. Slide your finger further out from the centre of the rotary pop-up and the response of the displayed control will become finer and values won't change as quickly to allow more accurate and detailed control.



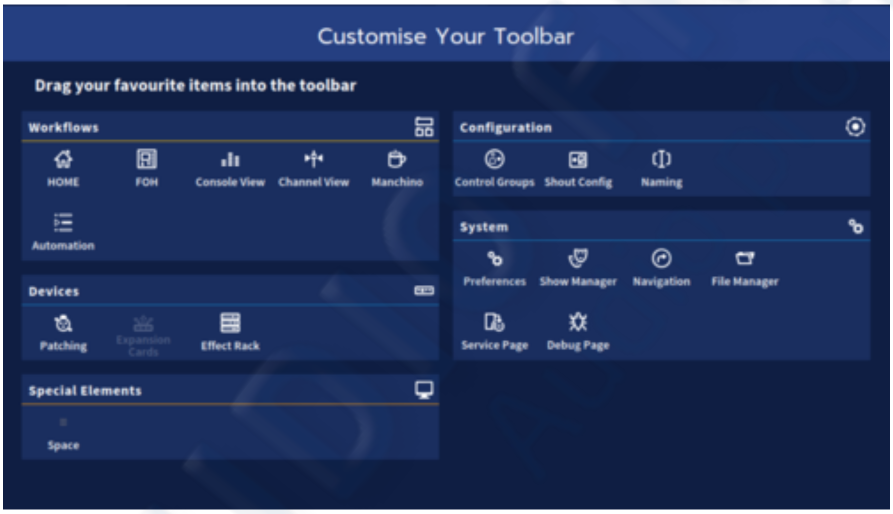


Slide your finger to the padlock in the centre of the rotary pop-up and it will "lock" to the screen. Once locked the rotary pop-up can be freely moved around the screen by dragging it with your finger and will not close until it is removed or un-locked. To remove the locked rotary pop-up simply tap on the X in the centre to remove it. If you change the selected channel the locked-to-screen rotary pop-up changes to the selected channel. An unlimited number of different rotary pop-ups can be locked to screen if required.

**Tip:** This is a great function to quickly set the gains for channels in a line check by selecting each channel in turn and adjusting the “locked-to-screen” rotary or to adjust the Comp threshold of each output as you select them. All rotary functions can be locked to screen.

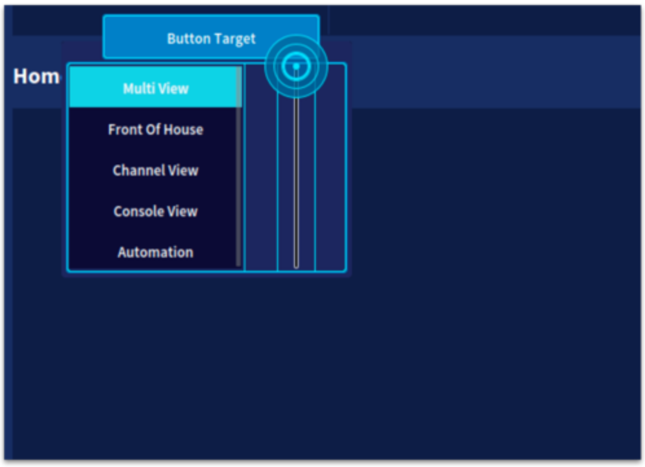
## 6.6 Workflow Modes Overview

Workflows are a group of different widget arrangements. Each tailored towards a different application or type of function. Up to 4 widgets can be accessed within a workflow. Workflows are selected by tapping each icon in the top bar menu. To change the order of workflow press Menu in the top right of the touchscreen, then press Customise Toolbar. This will open the custom toolbar page. You can now freely drag and drop icons to suit your personal preference.



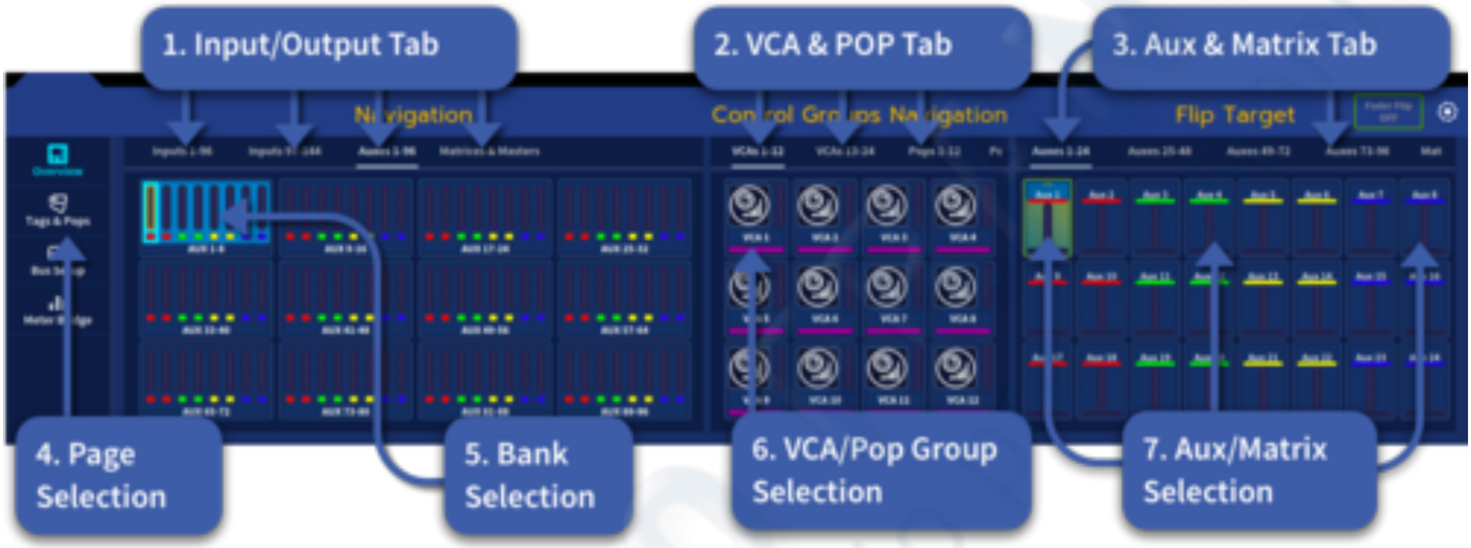

## 6.7 Home View


Home view can be changed to display your preferred workflow view when the HOME button is pressed on the surface or on screen. By default, the HOME view is set to Multi View. This can be changed in the Navigation workflow by selecting any view from the Home Button Target list.



# 6.8 Multi View

Looking at the left side of the GUI is a list of different layouts that affect the top half of the GUI while the bottom 2 widgets can be freely changed to display various different parameters.

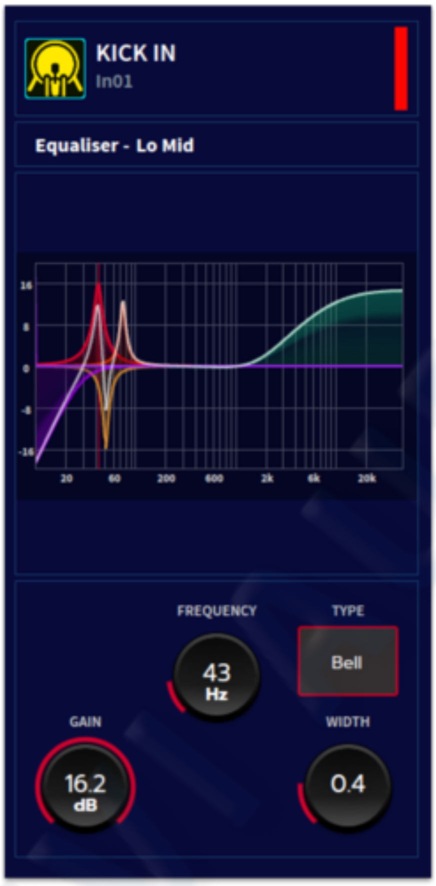



1. Inputs 1-96 on the 1st tab with inputs 97-144 on the 2nd tab. Auxes 1-96 on the 3rd tab and Matrices & Masters finally on the 4th tab.
2. Select VCA or POP group (via the 4 tabs)
3. Select Aux or Matrix sends via the tabs which display in fader/channel overview area
4. Change the selected widget layout.
5. Input Navigation to the left which enables the choice of different banks of 8 channels at a time. Touching any bank will bring up to 16 or 24 channels of information to the surface and GUI (depending on mode selected) - the selected bank of 8, plus the following 8 or 16 paths.
6. VCA and POP Group Navigation selection is another way in which to bring channels to the surface and touchscreen overview area. Simply touch the required VCA or Pop to have its content instantly appear on the surface. Long pressing the POP group or VCA button will quickly open the group assign page.
7. Aux/Matrices selection. - When flip mode is active this becomes a quick way to adjust levels to an aux mix quickly by touch selection, perfect for monitor engineers. If you select when not in flip mode it will bring that channel (and subsequent channels) to the surface.

The two widgets in the lower part of the touch screen can show various functions for the selected input or output. For example, the left side widget can display the Input EQ while the right-hand side shows Aux compression settings. These two widgets are the same for all HOME pages listed below. They will be hidden if the channel overview/GUI fader bank is in use.


## Side Bar Pop Up Display

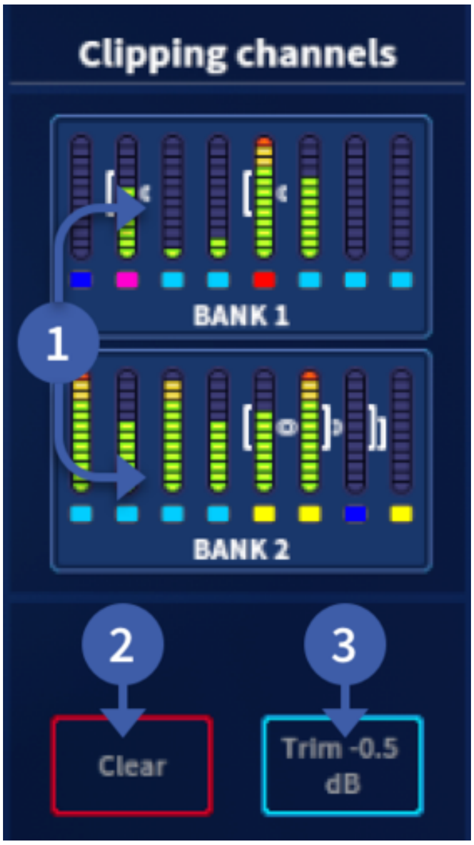
If the GUI channel fader/ Channel overview area is displayed (faders up) in the Home workflow, touching a physical surface control will activate the side bar pop-up display. This will display either the Pre-amp, EQ, Dynamics and Master bus functions and lets you see the information for the physical control you have touched.



## Tags and Pops


1. The Tags and Pops page allows you to call different groups of channels based on user-defined tags which you can create yourself or by pre-defined System Tags. For example, use the system tag 48v to instantly see all channels with 48v applied.
2. The currently selected tags are highlighted to show they are active.
3. Any channels containing the selected tags will be displayed in the banks ready for selection. When a bank is pressed it becomes active via the GUI faders and the surface for quick manipulation.
4. When Match All button is active all contributions must have all the chosen tags attached to them to be displayed. E.g. if 48v and EQ On are both selected only contributions with both 48v and active EQ will be displayed in the bank display directly below. User tags can be used in a variety of ways to organise inputs and outputs to suit your working preferences.

## Clipping Channels




1. The Clipping channels section is located in the centre. This allows any channels that hit the set clipping threshold and number of occurrences (see Clipped Channels Section) of clipping to be brought to the surface and screen quickly in case action needs to be taken to prevent further clipping or distortion. Clipped channels are place in the 2 Area A banks in a standard set up. Touch any bank to make it active on the surface and in the GUI.
2. The Clear button resets the displayed clipped channels.
3. The Trim -0.5 dB button turns the trim down by -0.5 dB on all clipped channels displayed. This is an easy way to reduce the trim of multiple clipping channels at once.

## POP Group Navigation

To the right is POP Groups Navigation. A place to recall pre-defined groups of channels to the surface, layer overview and touch screen faders. The banks can be swiped through to reach all channels and VCAs quickly.


## Bus Setup and Outputs Management


This page allows you to change multiple output settings at once. For example, it can be used to mute outputs, set all output levels to 0 dB or to send the oscillator to any output you wish to test.

1. Tab between Auxes or Matrices.
2. Solo selected Aux or Matrix.
3. Mute selected Aux or Matrix. Interrogation of mute status by press and hold function.
4. Fader level for selected Aux or Matrix.
5. Level change selection. Absolute changes all levels to the same value once the control is turned, Relative (level change is relative to original value with a + or – offset) or Pass-thru (action only takes effect when original level is passed, then level control is taken over by the control).
6. Turns the currently selected Auxes into Groups (Matrices cannot be turned into groups).
7. Turns an Aux into a Flexi Aux (this allows Flexi Aux to Aux routing for group/stem style mixing).

8. Long press to make an aux mono (mono is default).
9. Long press to make an aux linked (odd to even only).
10. Long press to make two auxes into a stereo aux (odd to even only).
11. Toggles Talk on/off for selected aux busses.
12. Flip Mode on/off. This is the same function as the Fader Flip button in the top menu bar.
13. Solo Preferences page.
14. Change page between Configuration Macros & the Oscillator tabs.
15. Turns internal oscillator on.
16. Oscillator level control.
17. Oscillator frequency control.
18. Turns on 1 kHz tone (auto switches Pink Noise off when active).
19. Turns on Pink Noise generator (auto switches 1 kHz tone off when active)
20. Aux or Matrix selection. Press and hold to select all.
21. Send Pan Link makes contributions follow the pan position of the channel they are derived from.
22. Choose the Talk or Shout Mixer (1-12) source.


Select the outputs you wish to change by tapping each one individually or select all outputs in that current tab by pressing and holding on one of the aux or matrices displays. To remove individual outputs when all outputs have been selected, simply tap on that output. To clear all, press and hold on an output.

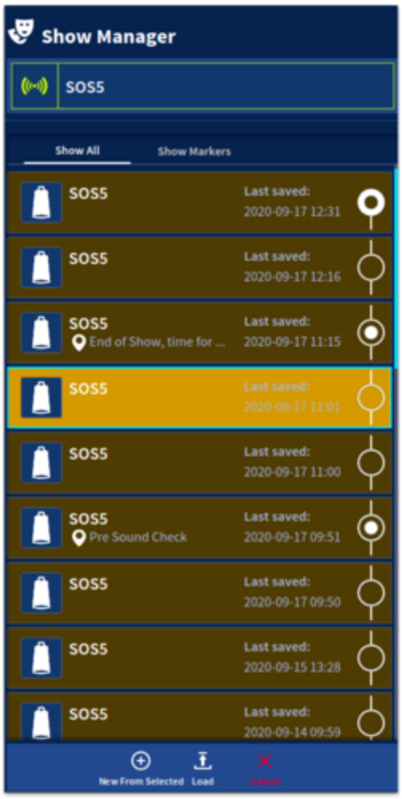
## Meter Bridge

The Meter Bridge page can display metering for any inputs or outputs. The meters also show compressor and gate gain reduction of the selected channel. By pressing on the corresponding input, output, POP or VCA, different combinations of meters can be viewed.

**Note:** In this example, VCA 1 shows a combination of input channels and output channels (the last 4 meters are Stereo Aux outputs).

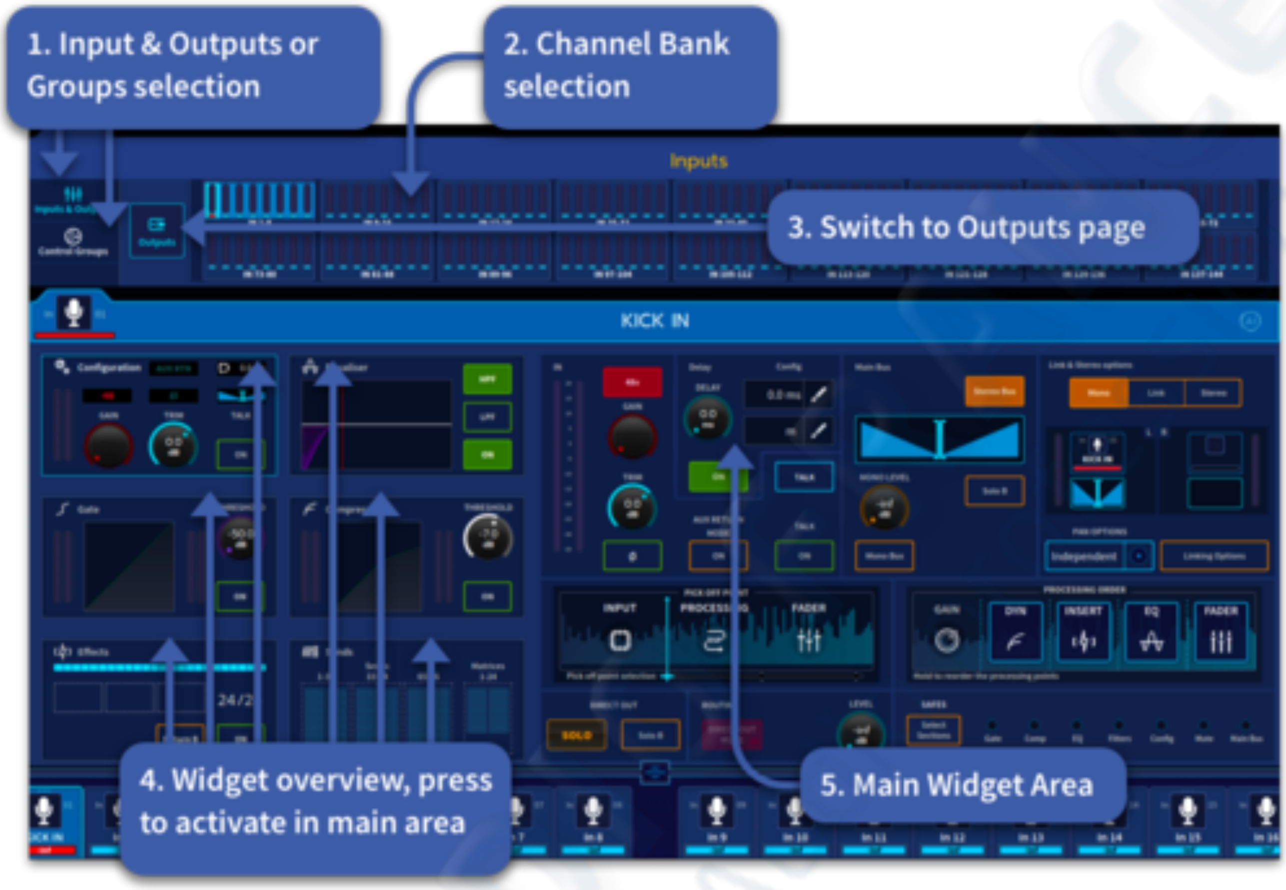

1. Meter Bridge selection.
2. Fixed views of Inputs and Outputs. Press to view.
3. Selected meters area spilled to surface Area A.
4. VCA and POP group number selection.
5. Selection of meters by VCA or POP.
6. Activate Fader Flip mode.

**Note:** When a VCA or POP is selected it also spills to the area it is assigned on the surface.



## 6.9 Channel View

The Channel View workflow gives an overview of many parameters of the selected channel, be it an input or an output. The top half of the page is for channel navigation.

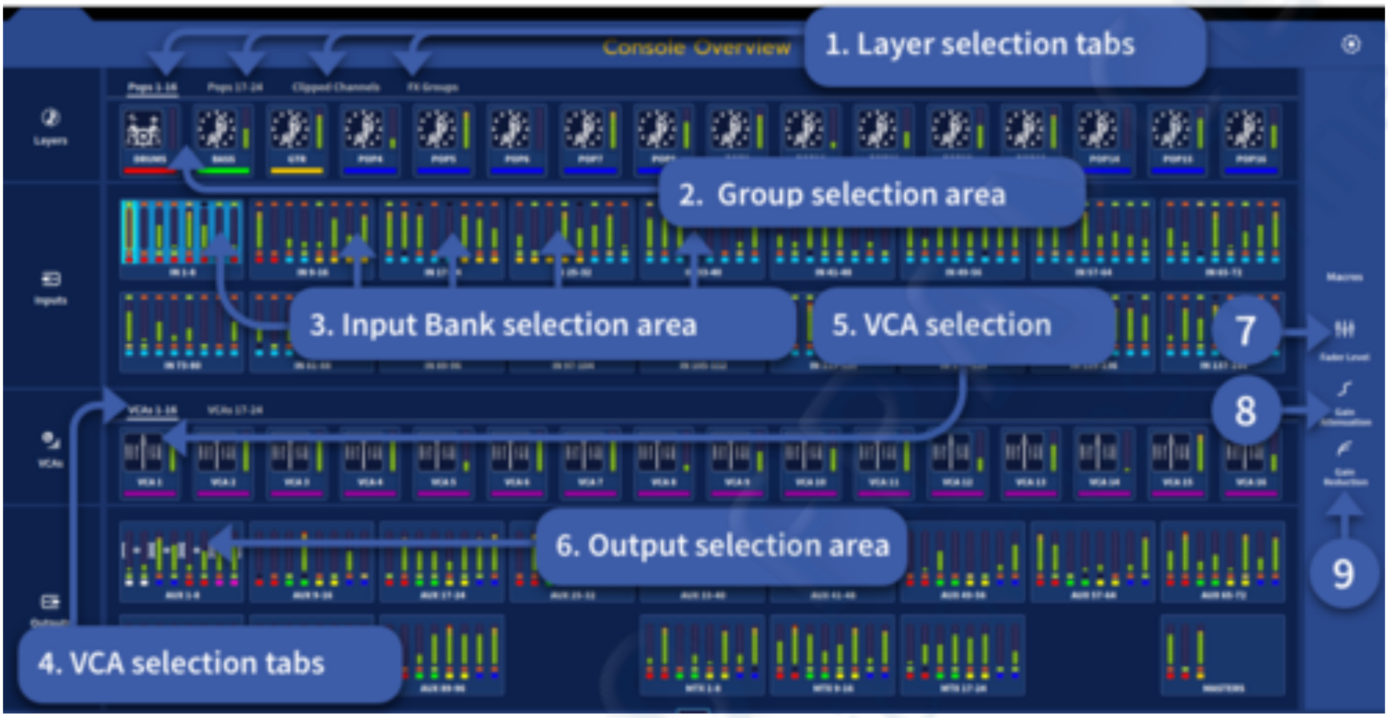



1. Select to navigate via input and outputs or via groups (VCAs & POPs).
2. Channel Bank selection in blocks of 8 for inputs/ outputs or if in the group page for VCA or POP group selection. When any bank or group is touched the associated channels will be displayed to the GUI and laid out on the surface.
3. Switch to Output banks view.
4. Widget overview. The selected widget will be highlighted in blue with its contents displayed in the main widget area to the right.
5. Main widget area where detailed changes can be made to individual channels.

In Channel View, the pull up fader tray is locked in the down position so only the path names and icons are displayed, but allows you to select channels as required to aid navigation.

# 6.10 Console View

The Console View workflow gives you an overview of all the channels in the system in one place. This page is very useful for trouble shooting clipping channels or to see if any inputs or outputs are muted (indicated by the meter outlined in red).




1. Layer selection tabs let you choose from Pops, Mute groups, Talk groups, Clipped Channels or FX Groups.
2. Group selection area lets you choose which group of channels to make active via the GUI and surface. Long pressing a POP group will quickly open the group assign page.
3. Input Bank selection area. All available inputs can be selected here.
4. VCA selection tab. VCAs 1-16 or 17-24 can be selected here.
5. VCA selection. Pressing on any VCA button will bring any associated channels to the GUI and surface. Long pressing a VCA button will quickly open the group assign page.
6. Output bank selection. All available outputs can be selected here.
7. Select to view Fader Level information.
8. Select to view Gain Attenuation of the channel gate.
9. Select to view Gain Reduction of the channel compressor.

In Console View the pull up fader tray is locked in the down position but allows you to select channels as required to aid navigation. If you touch a physical surface control it will activate the side bar pop up to display the chosen control information. This works for Config, EQ, Dynamics and Master bus functions.

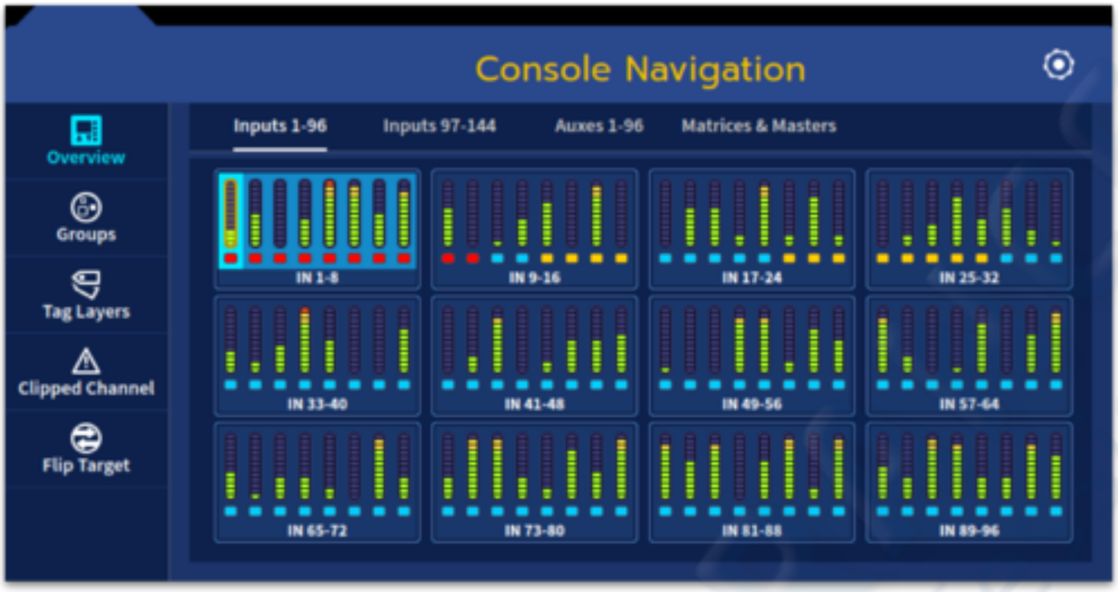
# 6.11 FOH View

The FOH Workflow is a great place to mix a show giving easy access to the fader tray while still allowing full control of the navigation and input channel areas. When the fader tray is lowered current scene and output information can be controlled making this a very flexible view.


1. Console navigation widget.
2. Input and Output information widgets.
3. Current scene automation events view.

**Note:** in the above example no outputs are selected.

The FOH Workflow splits the touchscreen into 4 separate areas.




The top left area has the Console Navigation widget. To the left side you will see icons for Overview, Groups, Tag Layers, Clipped Channels and Flip Target. This allows swift access to any input or output by touching the bank of paths you require.

The Overview page lets you choose from all the available paths and are arranged in tabbed groups for quick access.

The Groups page includes POP and VCA group selection.

The Tag Layers page lets you pick from System Tags or Suggested Tags as described in the Home workflow.

The Clipped Channels page allows all clipped channels to be viewed in one place. The Trim -0.5 dB button reduces all clipped channel trims by -0.5 dB to avoid distortion. The Clear button resets the clipped channels view to show no channels (if channels clip again, they will be displayed).

The Flip Target page is a quick way to select an Aux or Matrix. Used in conjunction with Flip mode it allows aux mixes to be flipped to faders for quick level changes.

To the right, both top and bottom widgets show input or output information. These handle all Configuration, EQ, Gating, Compression, Effect inserts and Send & Matrices/Contributions. These widgets are described in detail in the basic operation chapter of this manual.

Bottom left is the Current Scene page which shows events or notes for the currently loaded scene. See Chapter 19: Automation (Scenes and Shows) for more details.

# 6.12 Automation View

This workflow has the same Console Navigation widget in the top-left side as the FOH Workflow. Directly below is the selected channel detail widget.


The HD96-24 has a powerful scene-based automation system. In order to understand how the system works it can be broken down into the following structure:

1. Show – The whole of the system is stored as a Show, including all global settings, Playlists and Scenes. A show is used to transfer the current set up to a different system.
2. Playlist – A user-defined list comprising of some or all of the available scenes in a specific order.
3. Scene – A snapshot of the console settings taken at one point in time. A scene can be updated, duplicated or deleted if needed.
4. Library Preset – A individual element within a channel, an effect unit's settings or a whole channel strip can be saved to a library and loaded into a similar type of channel or effect if required.
5. Events – Various functions can be triggered with scene recall. Currently only MIDI events are implemented.

For more information on how automation works please see the Automation section of this guide.

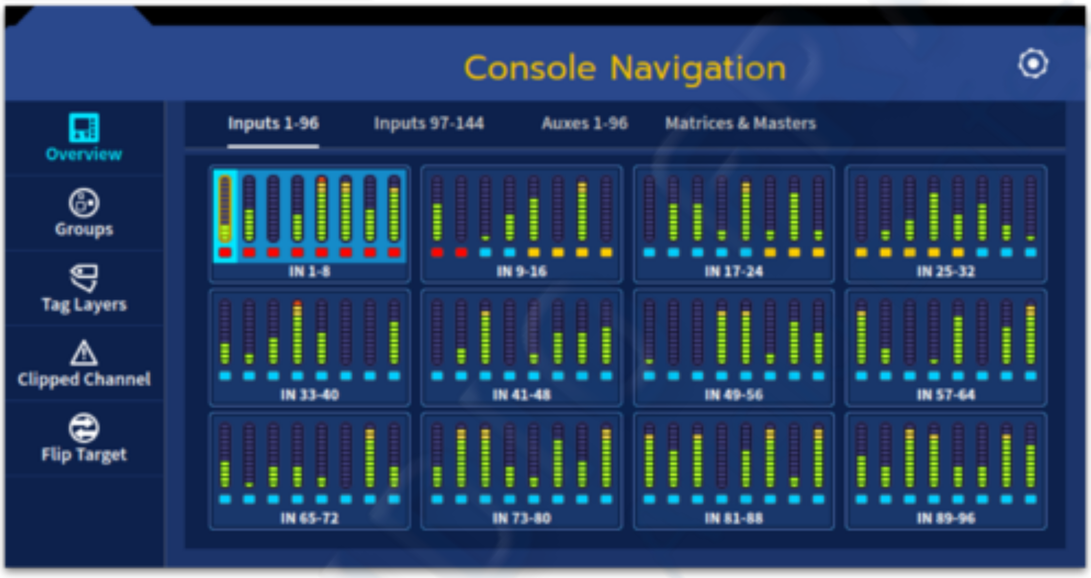
# 6.13 Navigating via Touch Screen Faders

The touch screen has a set of 16 faders which can be used to adjust any levels. Fader values are shown just under the channel name. If a channel is stereo both left and right fader levels are shown as two blocks. To change a level simply touch around the fader cap and move your finger up and down. The GUI faders are extremely responsive, highly accurate and with greater resolution than a physical fader. An upper limit of 10 faders can be moved at any time for maximum hands on control. The touch screen faders are available in the Home, FOH and Automation workflows. The fader tray is locked down in Channel View and Console View.

When a fader is selected it is possible to tighten the accuracy of the fader by dragging your finger outwards to either side fader or faders you have held. This allows fine adjustments of critical levels. While dragging your finger outwards the fader is locked so that accidental level changes can't occur. This rule can be broken if you move your finger more than the predefined tolerance. Both hands can be used at once independently. This means you can move multiple faders with one hand while tweaking a fine fader level with the other hand.

The on-screen faders, layer overview and Mute/VCA assignment page area (which all have their own icon) can be hidden using the pull-down icon revealing another 2 widget areas.


To bring input or output faders to the GUI, either use the various banks in the Console Navigation widget or use VCA or POP (POPulation) groups to allocate pre-defined sets of channels to the surface.


## 6.14 Navigating via the Layer Overview/Channel Strips area

The Layer Overview area is where quick recall of input and output channel strip parameters can be used and recalled. It is only available in the Home, FOH and Automation workflows.

### Channel Strip View

By Pressing in each separate box of a channel strip, different information can be viewed in the widget detail area and channel detail area.

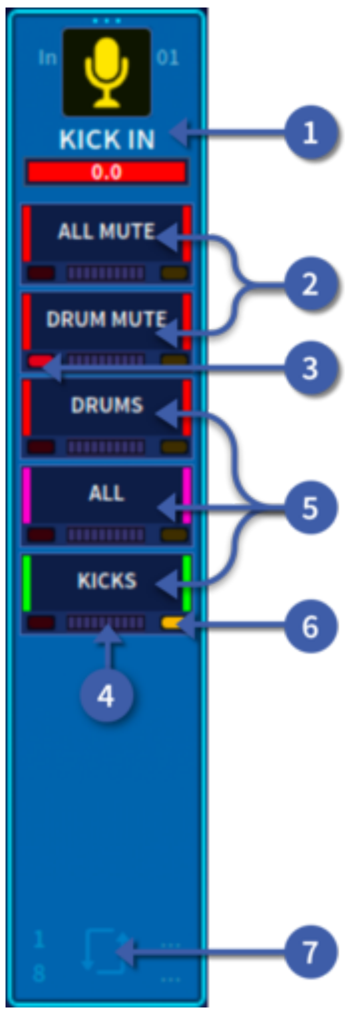

1. Channel Detail Area selection. Icon turns yellow if channel is soloed.
2. Channel Name and Colour Display. Display of fader level and send level if in flip.
3. Mute button.
4. Pan Position.
5. Configuration Widget selection.
6. Gate Widget selection (only for inputs).
7. Compressor Widget selection.
8. Effects Insert Widget selection.
9. Equaliser Widget selection.
10. Channel Solo.

### VCA and Mute area

The 3rd page in the Layer Overview area is the Mute and VCA Contributions.

1. Selected channel information.View Change is only active if the channel is assigned to more than 8 Mute Groups or VCAs. Touch to toggle between 1-8, 9-16 and 17-24.
2. Mute Groups channel is assigned to.
3. Mute status. Red for Mute active.
4. VCA Meter.
5. VCAs the channel is assigned to.
6. VCA solo status. Yellow for current active solo.

View Change is only active if the channel is assigned to more than 8 Mute Groups or VCAs. Touch to toggle between 1-8, 9-16 and 17-24.




When a VCA is added to a POP group, it can be displayed in the Layer Overview Area. Here channels assigned to the VCA are displayed.

1. VCA Name.
2. Channels assigned to the VCA.
3. Channel Mute status. Red for muted.
4. Channel Meter.
5. Channel Solo status. Yellow for current active solo.
6. Touch to toggle between 1-16 and 17-24. If VCA has over 16 member channels.

# 6.15 Navigation via Global Assignable Shortcuts area

The Global Assignable Shortcuts in its basic form allows various groups of channels to be called to the surface, GUI faders and channel overview area. There are many different options of how inputs, outputs and VCAs can be organised and recalled with a few button presses. This is fully customisable in the Navigation page.


The default options based on Pro Series Mode layout (Bank 1&2 Area A, Bank 3 VCA/Outputs and Bank 4 Masters) are:

- **All** – Pressing the All Select button cycles through every input or output of the desk bringing them to the first 2 banks of the control surface and displaying them on the GUI. The following buttons function in the same way but bring different types of paths to the surface.
- **INs** – Cycles through all the inputs on the first 2 banks of the surface.
- **OUTs** – Cycles through all the outputs on the 3rd bank of the surface.
- **VCAs** – Cycles through all the VCAs on the 3rd bank of the surface.
- **TAGs** – Cycles through all the currently selected TAGs on the first 2 banks of the surface.
- **System Tags** – Cycles through all the currently selected System TAGs on the first 2 banks of the surface.
- **All Contribs** – Cycles through all the current input contributions on the first 2 banks of the surface.

**Tip**: A contribution is any channel type that has already been sent to an output bus. i.e. Input to an Aux or Aux to a Matrix.

The Global Assignable Shortcuts area also gives you access to many powerful functions with one button press such as firing scenes, entering flip mode or recalling POP groups. Macro functions such as changing the console layout mode or effect Tap Tempo control can also be deployed here to give the engineer vital tools, quickly to hand. Imagine the power of being able to instantly swap the vocal level, compression setting and EQ of two singers if they switch mics mid show, all with one button!

# 6.16 Navigation via Pinned Fader and Assignable Controls


It is possible to pin a channel or path anywhere on the surface, so that the path remains fixed to a certain position despite swapping layers. For example, keeping your lead vocal close to hand or a crucial aux send.

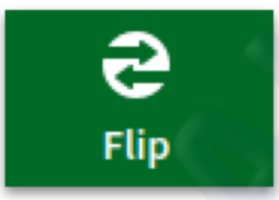
How to pin a path:

First, using the Assignable Controls arrow buttons, navigate to Find/Pin. Then the assignable control buttons under the rotary can be used to select the corresponding channel or path that you wish to pin to the surface. Use the rotary control to select any channel you wish to pin in that position. When the channel pin is activated the chosen path will remain in place regardless of any bank or channel changes from either the surface or GUI and a pin icon will be displayed by the path name on the LCD fader screen.

To remove a pin simply press the associated assignable button again.

For more information on these powerful functions refer to Chapter 18: Assignable Controls and Global Assignable Shortcuts.

# 6.17 Flip Mode




Flip provides a more global approach to aux/matrix mix bus level control. In normal operation you can only use the level control knobs in the assignable strips or the Sends & Matrices widget to adjust the signal level of the paths contributing to the aux/matrix channels. However, by engaging Flip Mode you have the option of controlling them from the surface or GUI faders.

To configure Flip Mode:

In the GUI, select Flip from the top menu bar. The button will illuminate green to show you are in flip mode.

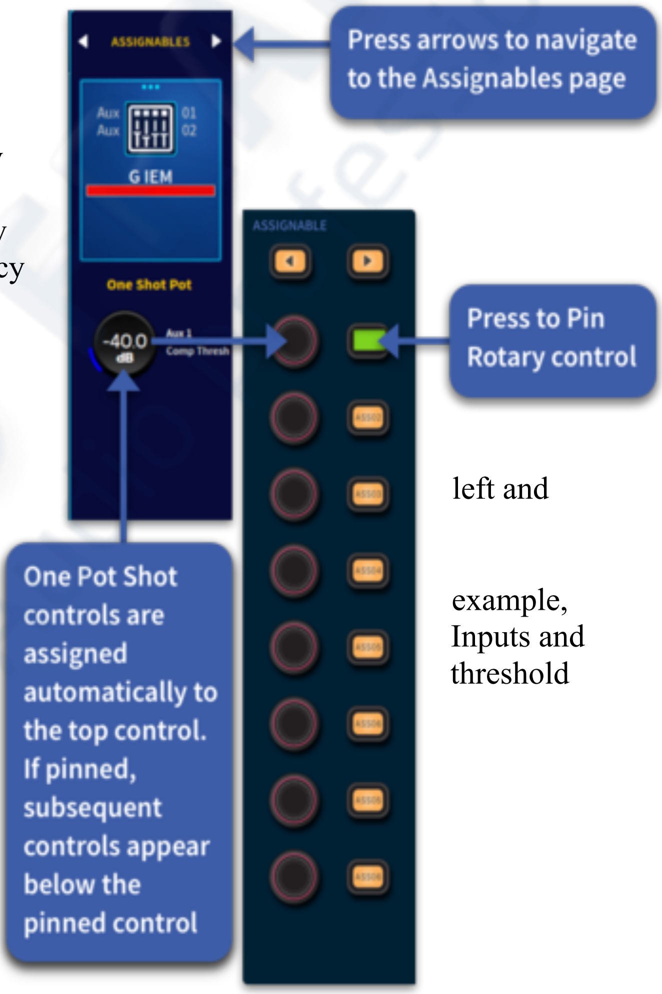
Alternately in the Shortcuts area it is possible to assign one of the 12 shortcuts to toggle Flip Mode. The shortcut button will illuminate green to show you are in flip mode. Another way to flip is in the workflow Home or Monitors which also have a green Fader Flip button.

To flip mixes to fader control select an output on the control surface or via the GUI. The currently selected mix bus will be 'flipped', and the fader LCD screens will display the target output bus.

# 6.18 One Shot Pot

The One Shot Pot allows any rotary function to be automatically assigned to the 8 vertical assignable controls. There is the ability to pin these selected rotaries in place. Up to 8 parameters can be pinned at any one time. An example of this could be pinning the input compressor threshold control or Aux EQ band one gain control. If you select different paths the One Shot Pot function will show the value of the currently selected path(s).

Auto Mode populates rotary controls to the sidebar for the currently selected widget or FX as follows:




- Config tab - Gain, Trim, Delay, Pan, Mono Level, Fader Level
- EQ Tab - Controls for selected frequency band
- Gate - all controls + Sidechain Frequency
- Comp - all controls + Sidechain Frequency
- Sends - level and pan
- Bus Setup - Fader level
- Patching - Gain (when selected)
- All FX - All fader/rotary controls

If there's more than 8 controls in an FX use the left and right arrows to access the extra controls.

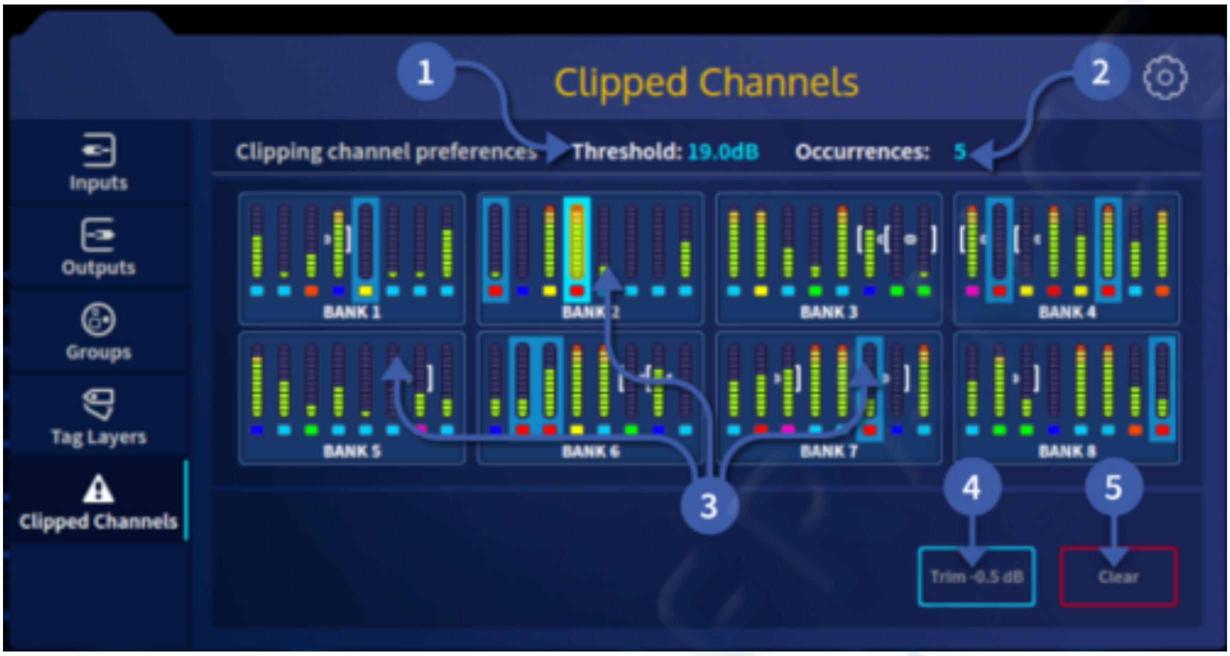
Input and outputs are treated separately, for example, if you wanted compressor threshold for both Inputs and Aux Outputs you would have to pin both threshold controls to the vertical user assignable area.

To use the One Shot Pot:

1. Navigate to the Assignables Page in the side bar menu using the arrows.
2. Select the rotary control you wish to use (must be visible on the GUI) which automatically places the rotary control to the first vertical assignable.
3. The button to the right of the assignable control turns green when pressed which pins the selected function to the assignable control. Pressing the button again clears the Assigned control.

# 6.19 Clipped Channels, fault finding a problem channel

The Clipped Channels page displays all channels whose level has exceeded the pre-defined threshold more times than the set number of occurrences.




1. Threshold can be set by tapping and changing the value with the pop-up keypad. Any channel clipping above the set threshold more than the defined number of occurrences will be displayed here.
2. Occurrences sets the amount of times a channel has to clip in order to appear in the clipped channel banks list. This is sample based so one clipped bass note could be several samples.
3. Bank selection based on the order in which channels have clipped.
4. The Trim -0.5 dB button turns the trim down by -0.5 dB of any clipped channels displayed. This is an easy way to reduce the trim of multiple clipping channels at once.
5. The Clear button resets the displayed clipped channels.

# 6.20 Console Layout Configuration

The HD24-96 can be fully customised to match your workflow preferences. The Navigation page can be accessed from either the top or side menu bar, you have the option to either choose from the available standard layouts (not editable) or creating your own user layout.


1. There are 5 default layout choices, all 5 are read only and cannot be edited (described in detail later in this section).
2. Custom Layouts are displayed here.
3. Delete the selected custom workflow with a long press.
4. Duplicate selected workflow.
5. Edit allows you customise the surface layout/workflow.
6. Create or a add a new workflow.
7. The 4 banks will show what type of information they will display (Area A, Mons, VCAs or Custom etc)
8. There are 3 areas into which layout presets can be assigned. Press the layout icon preset you wish to use, then drag to one of the 3 positions available.
9. Each workflow can be assigned to one surface layout only by pressing the associated button in each of the 3-surface layouts. A blue highlighted button shows which workflow will use which surface layout. For example, in the example above, selecting the Home, Front Of House or Automation workflows will display paths using Pro Series Mode layout; Console View will use a user-configurable custom layout, and Single Channel will use the Theatre Mode.
10. User layouts can also be edited when in use by pressing the Edit button.

**Tip**: Area A assigns inputs or outputs chosen via the GUI or surface to be displayed on both the surface faders and GUI with one touch and give the most intuitive operation.

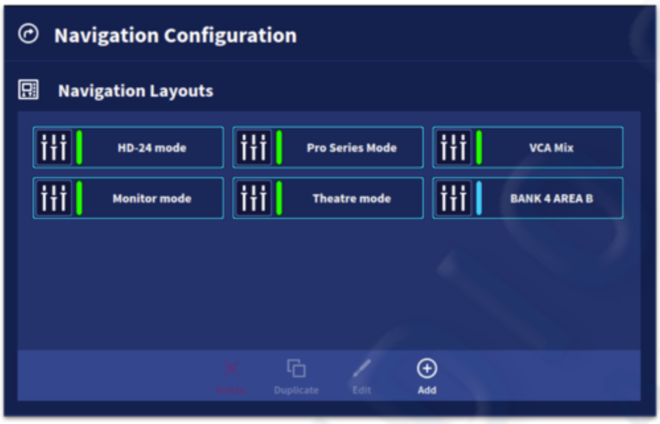
**Note:** Custom surface layouts are automatically stored to the mCloud to be freely used in other shows.

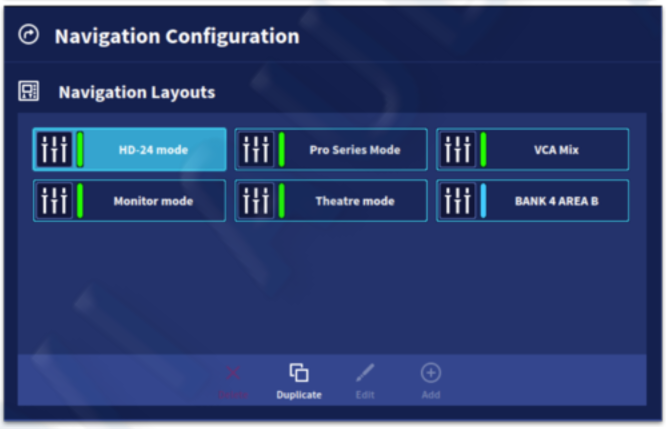
There are 5 standard Workflow options which display the following paths on each of the 4 fader banks:

| Workflow | Bank 1 | Bank 2 | Bank 3 | Bank 4 |
|---|---|---|---|---|
| Pro Series Mode | Area A | Area A | VCA/Outputs | Masters |
| HD24 Mode | Area A | Area A | Area A | Masters |
| Monitor Mode | Area A | Area A | Outputs | Masters |
| Theatre Mode | Area A | VCAs | Outputs | Masters |
| VCA Mix | Area A | Area A | VCAs | VCAs |

# 6.21 Creating a Custom Workflow

To create your own custom workflow press, + Add or select an existing workflow and press to Duplicate the selected workflow.




Selecting a custom workflow and pressing edit will open the Surface/Screen Layout Preview page below.


In the Custom Surface Layout Presets page you can also touch the name area to edit its title. There is the option to Cancel Changes or Store Changes & Save. Surface or Screen Setup lets you choose which part of the console you would like to configure. By default, the screen defaults to Area A. In Bank selection you can freely pick and choose which way you would like the banks to operate. There are 7 Bank Configurable Options plus customised banks options:

1. Area A
2. VCAs
3. VCAs/Outputs
4. Outputs
5. Masters
6. Contributions
7. FX


**Tip**: A contribution is any channel type that has already been sent to an output bus i.e. Input to an Aux or Aux to a Matrix.

There are 3 custom bank layers that can contain any combination of inputs, outputs or VCAs which can then be assigned to any bank from the Custom Bank Slots on from the left-hand side up to can be filled in one Custom Bank. Simply select and long hold the image of the function you wish to assign to a custom bank and drag to the position you wish it to be. Custom banks automatically link. For example, if you add 8 channels to Custom Bank 1 in Bank 1 and 8 channels to Custom Bank 1 in Bank 2. Those 16 channels will be displayed on the surface in banks 1 and 2.

In the Bank/Screen Setup page choosing Screen Selection lets you edit the layout of the 2 banks that can be displayed at any one time on the console screen. The same 7 bank options plus 3 Custom Banks as found in Bank Selection are available for you to use. However, in normal operation it is recommended that the screen GUI is set to Area A so that any selected inputs or outputs can be automatically viewed and edited via the touch screen.

# Chapter 7 : Patching

## 7.1 Introduction


Patching is a GUI-only feature that lets you carry out all system routing requirements. This includes:

- Patching audio from the on-board I/O and external I/O units into the console (Inputs, Direct Inputs, Insert Returns.
- Patching audio from the console via the on-board I/O and external I/O units (Outputs, Bus Outputs, Direct Outputs and Insert Sends).
- Purely internal routing, such as compressor side-chains, external gate triggers and routing to internal effects rack units etc.
- System path setup such as talkback, shout outputs, monitoring routing, solo bus inputs, external monitor inputs etc.

The GUI top bar menu has a Patching icon that takes you to the patching screen. This contains all of the available patching connectors in the system. This screen provides an easy-to-use interface, where you can select your source and destination patching options, facilitated by a panel of function buttons. Additionally, the Patching screen lets you set up the I/O units (devices). For example, you can globally adjust the analogue gain or select +48V phantom voltage for all 24 microphones at once on a DL231 mic box.

## 7.2 Terms used in patching

The following is an explanation of the patching terms

- **Destination:** The patch connector to which a signal is routed.
- **Patching:** The process of routing a channel/signal from a source to a destination(s).
- increase the systems physical connections. There are 4 AES50 ports on the console rear for I/O expansion plus 16 more AES50 available via the 2 HMAC ports.
- **Device:** A block in the I/O tabs that represents a physical rack unit, such as a mic splitter (DL231 etc) or Generic AES50 port.
- **Tab:** A ‘sheet’ in the local, remote or internal page that contains a specific group of patch connectors.
- **Patch connector:** Any patching point where audio can either be sent from or sent to, for example, an XLR connector, Output Bus, sidechain compressor etc.
- **Dragging:** A method of selecting a block of source patch connectors from one side of the patching by swiping your finger across any channels you wish to select.
- **Press and Hold:** Long pressing any connector on a remote or local device will select all connectors within the current page.
- **Local:** External I/O connected directly to the surface.
- **Remote:** External I/O connected via HyperMAC (using an DN9680) to either of the snake connections on the console rear.
- **Internal:** Access to all the internal patching within the system (Inputs, Busses or Direct Outs etc).

# 7.3 About the Patching Screen


Patching screen can be broken down into three main areas types:

1. The function button panel at the top of the GUI has buttons for Unpatch and Locked. To make any patch changes, press the Unlocked button. **Note:** if you navigate away from the patching workflow the page will automatically lock. Locked mode is a good way to see where patches have been made without accidently un-patching a channel.

2. The patching areas provides access to all the patch connectors. The two patching areas on both sides can be used in a left to right style to make patches to external devices or internal functions such as sidechain or tape returns. The patching area is split equally into two independent sections, which contain the source and destination patch connectors, respectively. The patch connectors are grouped on pages according to type. Only one page per section will be visible at any time and selected via type or channel number depending on the page and type of device attached.

3. External I/O tab menu. The I/O tabs represent the systems internal patchable items, stage (Remote), 2 x CM-1 slots and FOH/Monitors (Local) racks. Each box shows the internal and external patchable items devices connected in those racks when selected.

**Note:** Each time you access the Patching page, the Locked button will be red, and the page will be display-only. This is a safety precaution to prevent any accidental changes. Before changing any patching, you will need to tap this button, so it turns blue to unlock the patching functionality.

**Tip:** Always patch from the left side to the right.

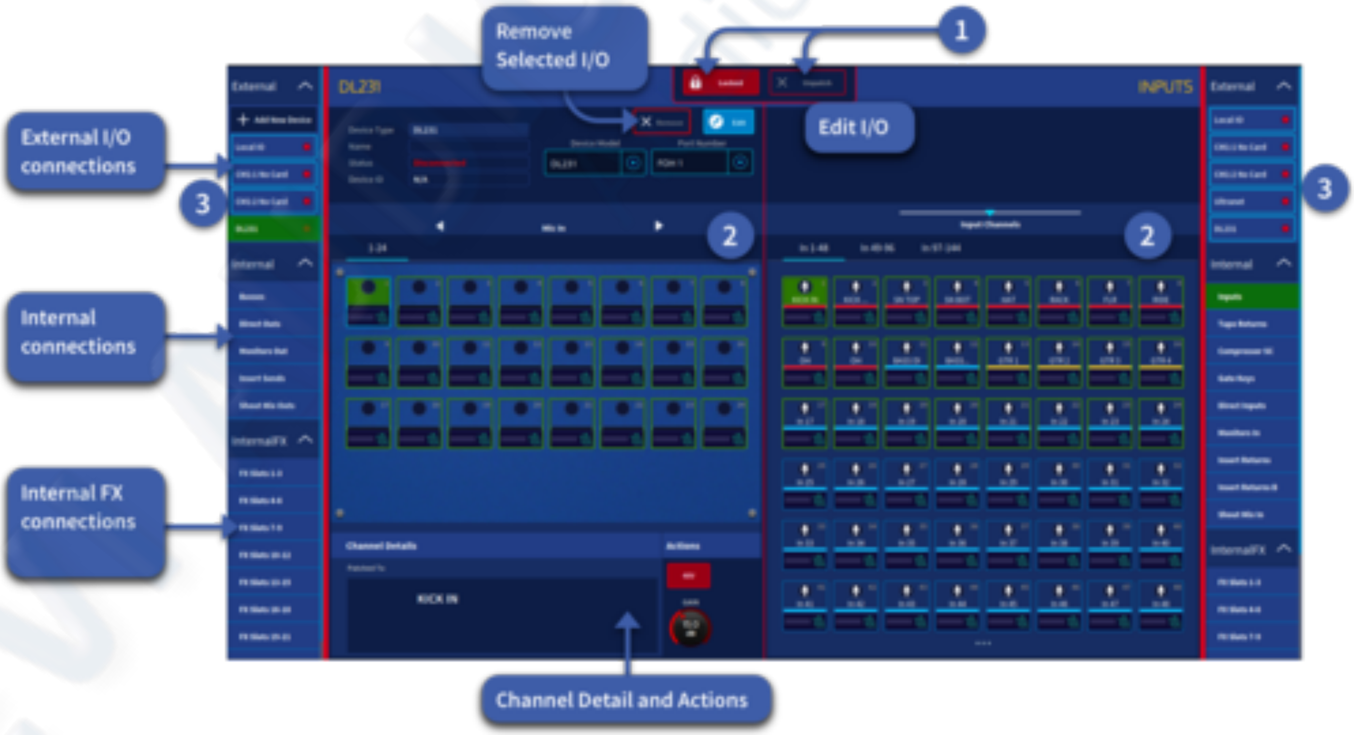
# 7.4 Adding new I/O Devices


Press the + sign to add new I/O boxes. A list of currently supported I/O will appear. Select the type of I/O box required by pressing on the corresponding button. Or use the Auto Detect function which will add the correct I/O device type.

**NOTE:** The DL231/431 has two preamps per channel (A&B) to use the B pre-amp connect to the B (X&Y if using redundancy mode) AES50 connections on the back of the DL231/431. It is not possible to change which preamps (A or B)are used in the HD software.

**NOTE:** The DL 251/252 use two AES50 ports. One I/O device should be added, and two ports assigned in the edit I/O box.

I/O with inputs and outputs will be displayed in different named tabs. Inputs on the left, outputs to the right.



# 7.5 Configuring the I/O devices

The DN9680 will require updating to the latest version from the mCloud in order to work with the HD system. After the update the HD system will install future updates when required. When a DN9680 is connected to Snake port 1 or 2 the 8 AES50 ports can be assigned by pressing the I/O device edit button and selecting the corresponding port. Each Snake connection can be connected to a DN9680 allowing up to 16 AES50 ports to be used.

To edit an I/O device:

1. Select the device from the list.
2. Press Edit.
3. Press in the Name area to call the device a name, in this example IEM RACK.
4. Chance the Device Model if required.
5. Set Port Number, Either FOH 1-4 or Snake 1 or 2 (1 to 8).


Note: If a 251 or 252 rack is connected two ports will be required to be set up. In this example, Snake 2 Port 1&2 have been configured.

# 7.6 Removing I/O Devices

To remove a I/O device:


1. Select the device you wish to remove from the list on the left

2. Press and hold the Remove button until the line traces around the outside.

The I/O device will be removed from the list on the left and all patch information will be lost.

## 7.7 Internal Patch Points

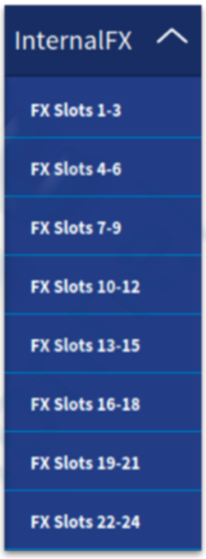
There are many sources that can be patched in the HD24-96 system. Below is a list of patchable items. These pages are available on both side of the patching screen when appropriate. The list is filtered to show the available patching destinations for the tab you have currently selected:

1. **Busses** – Output busses (Aux, Matrix and Masters) pages.
2. **In-Direct Out** – Input channels, direct output pages.
3. **Monitor Out** – Talk back and monitor A/B patching page.
4. **Insert Send** – The send patching point of a channels insert pages.
5. **Shout Mixer Out** – Patch the 12 outputs of the shout mixer.


## 7.8 Internal Patching Destinations

1. **Inputs -** Console inputs
2. **Tape Return –** Input channels, tape returns pages.
3. **Comp SC –** Input channels, compression sidechain input pages.
4. **Gate Key –** Input channels, gate key input pages.
5. **Direct In –** Output busses, direct input pages.
6. **Monitor In –** AFL, PFL, External and Talk back patching page.
7. **Insert Return –** The return patching point of a channels insert pages.
8. **Insert Return B –** The return B patching point of a channels insert pages.
9. **Shout Mixer In –** The 16 inputs to the shout mixer.

# 7.9 Internal FX




The internal Effects can be patched from the patching page to various destinations, the 24 slots are freely patchable. Up to 8 different destination can be patched per slot depending on the type of effect used. Until an effect is placed in a slot it is not patchable. The name of the effect will be displayed to show it is present in a slot.

**Note: It is easier to patch effects in the Effects Rack or directly from the channel**.

# 7.10 CM-1 Port Patching

The two CM-1 ports can have different types of I/O cards fitted. Current expansion cards are:

1. AES50 (48 Bidirectional Channels @48 kHz).
2. Dante (64 Bidirectional Channels @48 kHz).
3. MADI (64 Bidirectional Channels @48 kHz).
4. USB (48 Bidirectional Channels @48 kHz).

The two CM-1 ports will appear in the patching list and will automatically be recognised by the type of card fitted. They can be freely patched as required using the same patching method as described earlier in this section. If 96 kHz is used the above channel counts are halved.

**Tip:** This is a great way to multitrack record a show to an external computer and use the recording for virtual soundcheck.

# 7.11 Patching Tips

## Single patching

To patch a single source to a single destination:

The following example shows you how to patch an output from a mic splitter such as a DL231 to an input channel. Make sure the patch area is unlocked.

1. Select the source patch connector. For example, in the Remote I/O tab of the left side section, click the first patch connector XLR1 of DL231 I/O. Its background will change to blue.

2. Select the destination patch connector. For example, in the Inputs tab of the right-side Internal section, click the patch connector for input channel 1. It will now be patched to the source and will change to solid green to indicate the patch has been made. If the new patch is carrying a signal there will be a meter shown on both patch points (unless the connection is Generic AES50), and this audio may be heard, depending on the settings of the control surface.


**Note:** A quick way to see if any connector is patched to any destination in the system is to look for a green outline around the connector icon.

To patch a single source to multiple destinations

1. Patch the desired source patch connector to one of its destinations, as detailed in "To patch a single source to a single destination".

2. In the inputs section, select the other destinations. They will illuminate green to indicate they are patched.

## Multiple patching

If you would like to make a number of patches, and each has only a single destination, you can use the multiple channel function. All of the source patch connectors are selected in the Device section (Remote or Local) before being patched either by dragging across the connectors, selecting one by one or by pressing and holding to select all inputs (unwanted channels can be deselected after selecting all if desired).

You can only select one block of sources at a time. Destinations are restricted to a single type (for example, inputs). The selected destination forms the start of the automatically patched range of destinations. Sources and destinations are automatically patched in ascending order, the lowest numbered source and the selected destination forming the first patch. Sources will only be patched up to the highest numbered destination of the current destination type.

For example: drag your finger over inputs 1-16 in sequential order on the left side of the screen, and then tap output 9 on the right side of the screen. 1-16 will be patched to 9-24 in sequential order (so 9 to 16, 10 to 17, 11 to 18 etc).


## Unpatch

To undo or disconnect a patch, select the connector you wish to unpatch, then press the Unpatch button. The orange colour will disappear to indicate the patch has been broken. Multiple connectors can be selected and unpatched at once if required.

Note: if a source is selected and patched to multiple channels or destinations clicking on Unpatch will result in all its destinations being unpatched. If you wish to remove the mic preamp from one destination and keep all over patches, click in the Inputs section on the left side of the screen and unpatch them individually.

## 7.12 Channel Detail and Actions Area

When a channel of an I/O box is selected in the patch window, its patch information will be displayed in the bottom section of the GUI. There are several actions that can be altered if the connected I/O box has that functionality.


1. **Patched To** - The destination(s) of the currently selected mic pre are displayed here.
2. **48v -** Applies phantom power to the selected mic pre.
3. **Gain -** The gain of the mic preamp can be controlled from this section.

# Chapter 8: Basic Operation

## 8.1 Scene and show management (Automation)

Automation lets you manage show files and the scenes contained within using user-defined playlists.


To create a show:

1. In the top bar menu press the blank area to the left of the Menu button (name of current active scene is usually displayed here).
2. In the pop-up window press Show Manager.
3. Press the + New symbol to create a new show. Enter a name and press create.
4. Select the show name you wish to open from the list.
5. Press Open to load the selected show, see show Load Options later in this guide.
6. Tap on the name area to edit the text or click on the icon area to change the image. Tags can also be added to the show file here for easy tracking of files. Notes can also be added in this area if desired.
7. A show can be exported to a USB device.
8. To archive any show (available in your mCloud account if required), select the show name and click the Archive button.

**Note:** Opening a show does not load a scene. This needs to be completed after a show has been opened.

**Tip:** If you have many different shows stored you can use the Search Shows bar to find a specific show.

Please refer to Chapter 19 Automation in the user manual for more information on scenes and the automation system.

# 8.2 Clear the Console

- To clear all scene data and current channel settings of the system:
- Create a new show.
- Open the new show.
- Recall the Master Scene. The system will now be cleared of all channel and scene data plus all patching.

# 8.3 Saving and Loading show files to a USB memory stick

To save your work to a USB Stick, first Insert USB stick into the front surface USB port. Navigate to Show Manager from the Automation window in the top bar menu or from the side bar menu.

1. Select the show you wish to export.
2. Select the location on the USB stick and press Export to complete the operation.
3. Press Copy to USB, the exported show will appear in the list.

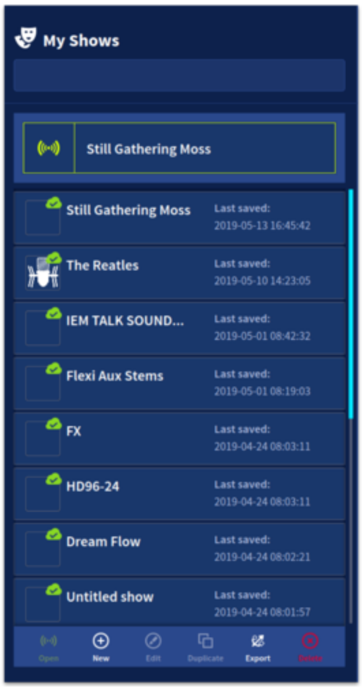

To import a file from a USB stick:

1. Navigate to the File Manger page (found in the side bar menu)
2. Locate the file in the USB stick area
3. Press Copy To Console to move the file into the Console list.

**Note:** This does not open the show. It just copies a version of the show to the console in order to be loaded.


## 8.4 Saving and loading show files to the mCloud

Once you have logged into your mCLoud account on the console all your show files will be available in the Show Manager list. With Auto Save active every time a scene is saved the show will be saved and also pushed to the mCloud. To load a file, select it and press Open. The open show box gives you the option on how the system will load.



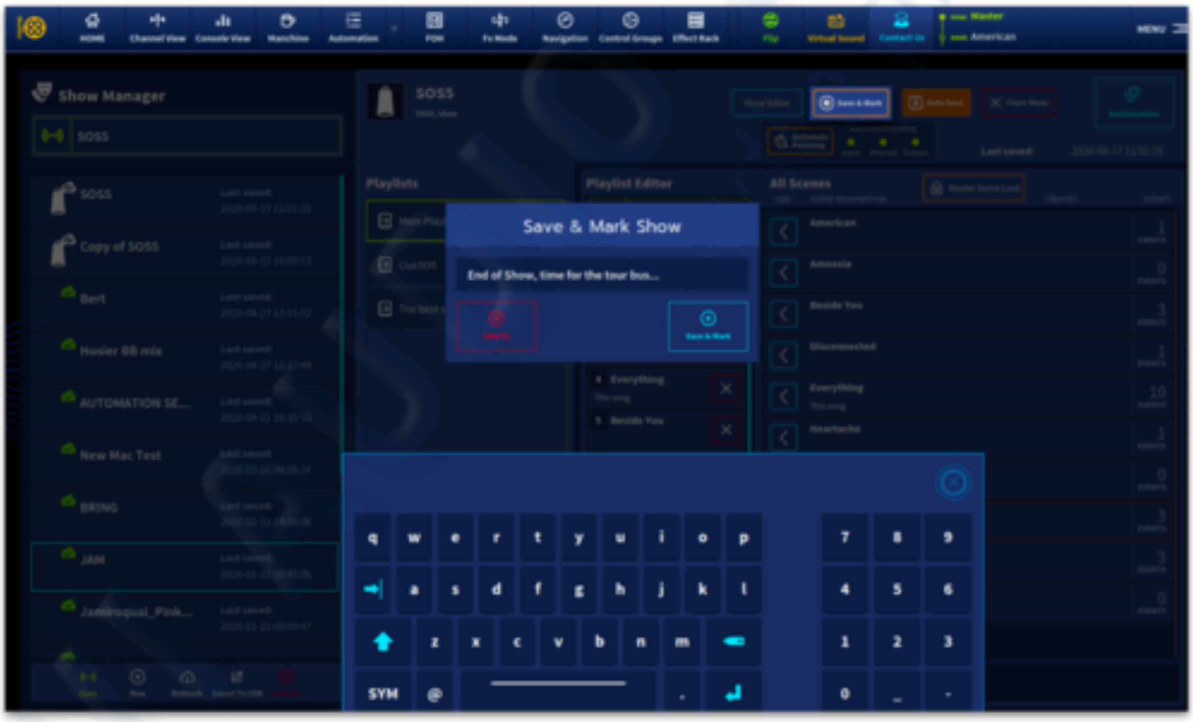
# 8.5 Save & Mark and Rollback

## Save & Mark

If at any point you wish to save your currently loaded show, the Save & Mark button can be used. This function lets you save your work during the day, for example post sound check, then post show. This allows you to use the rollback function if needed to restore an earlier version of your show file:

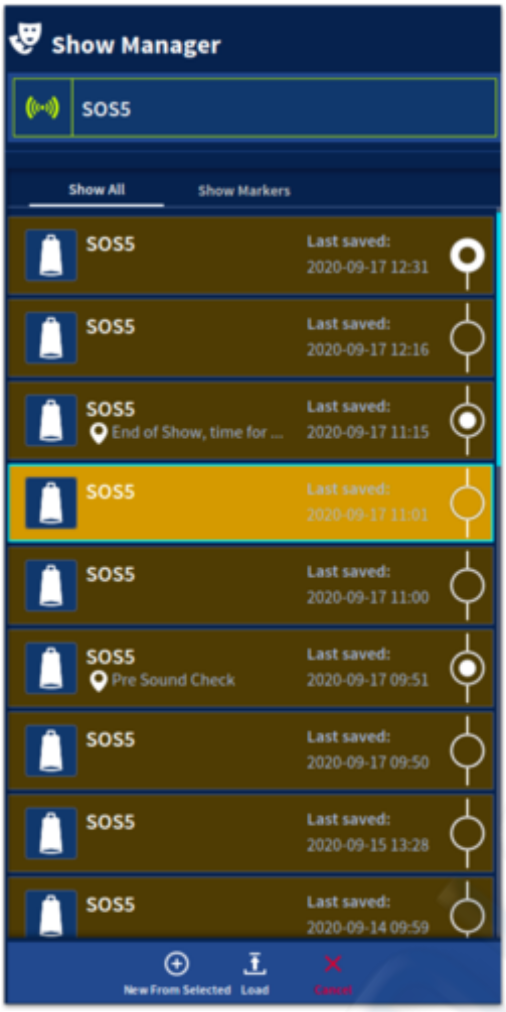
1. Press the Save & Mark button.
2. Enter a name that will help you remember the current scenario.
3. Press Save & Mark to complete the operation (or cancel if you do not wish to proceed).

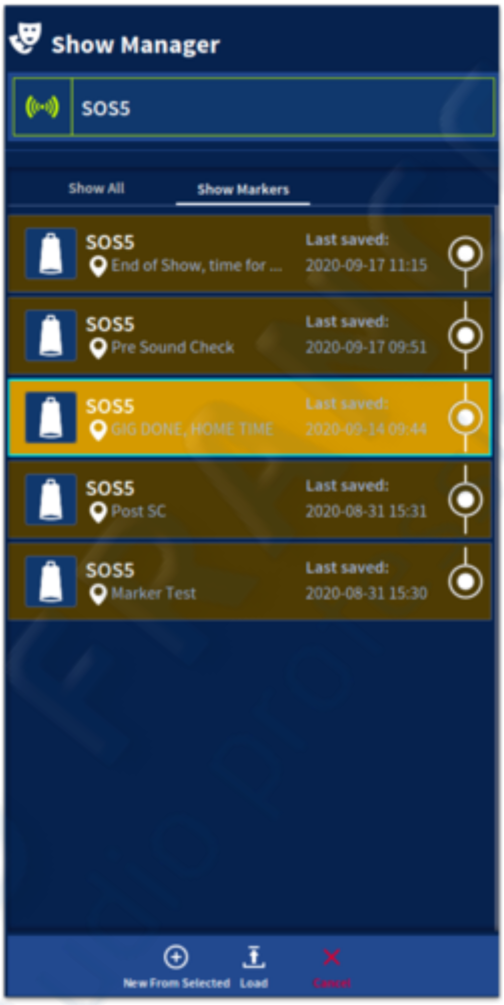
**Note:** Marked shows will appear in the Rollback list (the last 25 saves only) and in your mCloud account (all auto-saved shows and marked shows).




## Rollback

The Rollback feature allows you to look at the last 25 times the selected show was saved. This gives you the option to either load an older version or create a new show from the currently selected version. New shows will be named “Copy of …(name of copied show)”. Shows can be duplicated this way by selecting the last saved version from the top of the list. Marked shows are indicated with a dot in the circle and show the mark information (End of Show etc).




The Show Markers tab displays only marked shows. Older shows can be also loaded, or new shows created from this page.

**Note:** Only the last 25 saved shows will be shown on the console. Older versions will be available in your mCloud account where you can also delete shows if required.

# 8.6 Load Options & Automate Patching Preferences

Once you have logged into your mCLoud account on the console all your show files will be available in the Show Manager list. With Auto Save active every time a scene is saved the show will be saved and also pushed to the mCloud. To load a file, select it and press open. The open show box gives you the option on how the system will load.

**Load Options:**

Safes can either be loaded from the show file or kept in the same state as currently on the console.

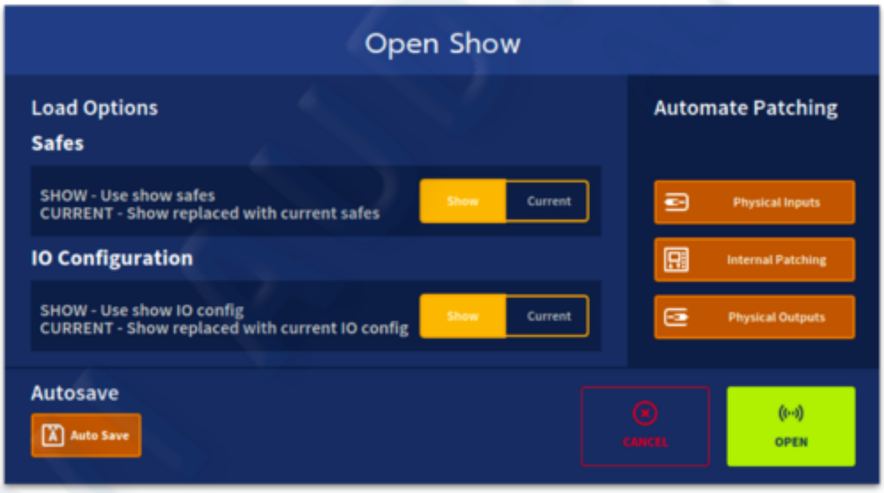
I/O Configuration can be taken from the show file or the currently loaded console setup.

**Automate patching preferences:**

**Physical Inputs** allows any XLR connection patch to an input on the console or I/O box to be stored and recalled in automation.

**Internal Patching** allows all internal patching to be stored and recalled in automation. For example, inserts, buses to effects and internal aux returns.

Physical Outputs stores and recalls any XLR connection patch information from an output on the console or I/O box.




**Note:** As a general rule, load Safes and I/O config default to load from the Show file. Automate patching on and Auto-save are also active.

# 8.7 Naming Page

The HD-96-24 has several different ways to name inputs and outputs. The first is in the Naming page which can be found in the top menu bar on screen and accessed by touching the icon. Once in the Naming page all channel, auxes, matrices and masters can be seen. Tap on the Inputs or Auxes Matricices Mastsers page then you can select which path you wish to name. Type the name in using the pop up on-screen QWERTY keyboard or via a connected USB keyboard if desired. Once you have finished naming the item, pressing Return will automatically take you to the next path to be named. You can also select a colour at this point from the colour bar above, which will be displayed on screen and on the fader LCD screens.


1. Inputs page selection.
2. Output page selection.
3. Channel name selection (keyboard will pop up on selection).
4. Naming area.
5. Channel colour selection area.

Pressing Return when a channel is named will take you to the next channel sequentially. **Tip:** The Groups workflow is a great way of colouring multiple channels at once to increase workflow set-up speed (details later in this chapter). The Manchino workflow also enables name inputting and easy manipulation of the same control across multiple channels. Groups of either inputs or outputs can be edited all at once and is a very powerful way to make global changes in one operation.

Another way to name a channel is to navigate to any of the workflows found in the top bar menu. There are several ways to navigate to the different channel, auxes, matrices or masters by touching on the associated navigation blocks.

**Note:** To name a VCA or POP group use the Groups workflow.


In the Home workflow you will find input navigation in the top left widget and output selection can be found in the top right of the widget. To name inputs:

1. Touch the block named IN 1-8. This brings channels 1-16 on to the GUI fader/overview area.
2. Press the Image or icon on channel 1 in the GUI or press the SEL button of the corresponding path on the surface.
3. On the far-right side of the touch screen is the channel detail area, pressing the name in this area will open up the Channel List Naming Page. Channels can also be named by touching the name in the widget area.
4. Touching the icon will open the image change page.
5. Touching in the tags are will take you to the channel tags page.

There are three different pages to use, selected by the three icons in the middle of this display called:

Title

Channel Tags

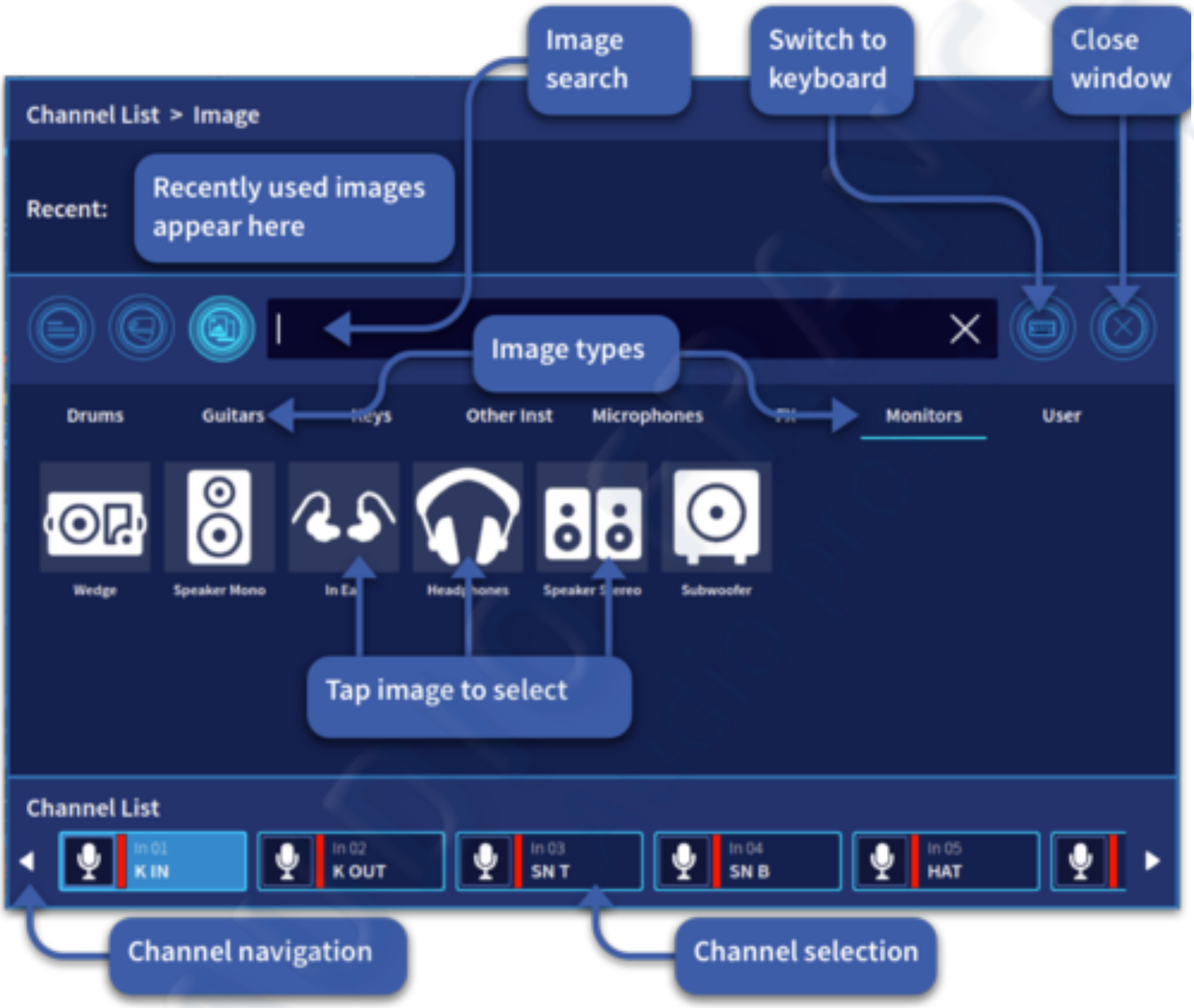
Image

## Title

Here you can name and colour the selected path using the Qwerty keyboard and colour bar above the keyboard. Use the channel navigation to scroll through the channel list.

## Image

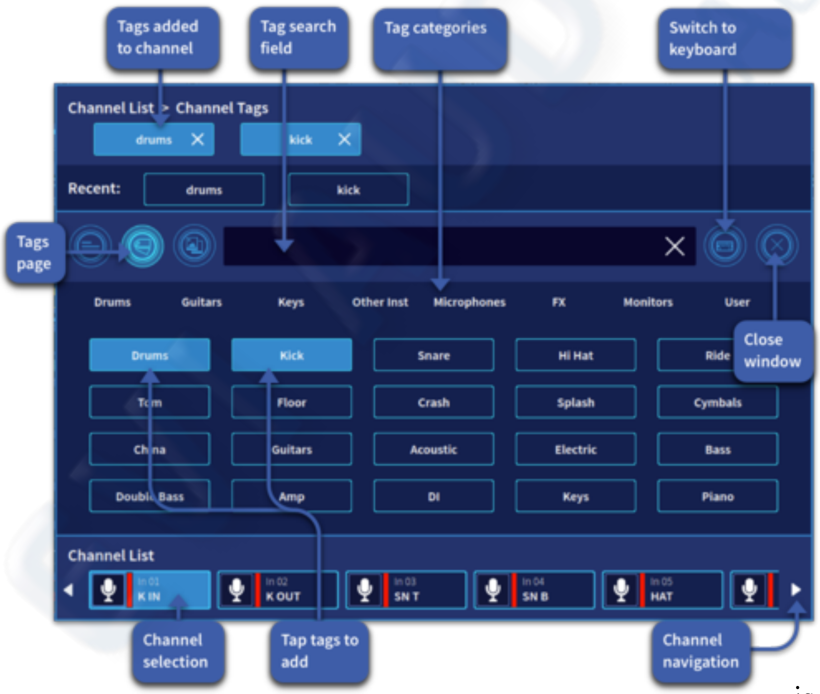
Here you can select a suitable image to be displayed with the selected path. There are several categories of images and icons to choose from. This gives a quick visual indication of the type of path when looking at the touchscreen.




**Note:** All the workflows work in a similar way for naming purposes.

# 8.8 Channel Tags

The new concept of tagging channels greatly increases ease of navigation by grouping channels in different ways. There are limitless ways to use this function. Any path can have multiple tags assigned to them. For example, all monitor wedge outputs could be tagged together for swift recall. For a large orchestra all the different instrument sections can be tagged and within each section all different type of mics can be tagged for quick changes to be made. System tags can be used to quickly see which channels have 48v applied, whether a channel has an active gate or if an effect insert has been enabled on a channel. Countless custom tags can be created to use as you desire making this a very powerful and flexible navigation system. In the Tags Layers page found in the Home, FOH, and Automation modes, there is a choice of either System, Suggestedor Favourite. Suggested tags are built from tags you have previously used or assigned. Touching on a tag opens up a block showing all the channels associated with that tag. Touching another tag will add any additional channels into the block section. Note that if a channel is in both tag groups it will only appear once.System tags show common functions of the console and is a great way to fault find. Tags are automatically assigned when the associated functions are used on an input or output, for example turning a compressor on will add a Comp On tag to that path and remove it again when the compressor is switched off. When the tag name is pressed the following useful information is displayed in the eight-section navigation blocks for easy recall to the surface. The list below is designed to aid in finding problem inputs or outputs quickly. To add a tag to a channel. Select the Tag page by touching the image of the channel you wish to tag in the channel detail area. Select the tag or tags you wish to add. They will appear at the top of the page and can also be removed by clicking on the X within each tag.




- Routed to Mono – When a channel is routed to the Mono Bus.
- Patched – When an input or output is patched to a source or destination.
- EQ On – When an input or outputs EQ is active.
- Gate On – When an input channel Gate is active.
- Comp On – When an input or output Compressor is active.
- 48V – When 48v Phantom power is enabled on an input.
- Phase Ø – When an input or output has polarity (phase) reversed applied.
- Insert On – When an input or output has its insert return switched in.
- Routed to Stereo – When a channel is routed to the stereo bus.

Favourite tags are displayed here automatically and can be cleared by pressing the corresponding X


1. Tags can be searched and filtered by typing in the text entry field.
2. Press + to create a new tag.
3. Match All means a path must contain all the chosen tags.
4. Clear all the currently selected tags.
5. Tags will appear here and can be used to recall the tagged channels to the surface quickly.
6. Channels with matching tags are displayed here in banks of 8.

# 8.9 Channel Configure (Config)

First select the channel or output you wish to edit. In the channel or output widget the top option on the left is Configuration. This area has five pages which can be changed by touching the next or previous name in the list or by using the arrows either side.

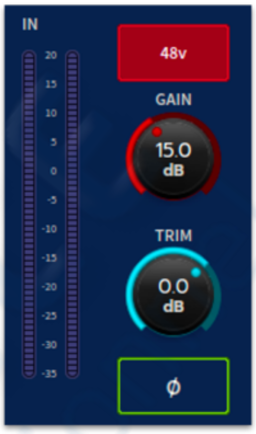
## For Inputs

### Configuration

The Configuration page allows you to set many of the basic functions needed to start mixing audio.


1. 48v.
2. Mic Preamp Gain (available when patched).
3. Digital Trim.
4. 180° Polarity change.
5. Delay On/Off. (max 50.0 ms).
6. Delay Time (displayed in milliseconds and meters).
7. Aux Return mode.
8. Talk On/Off.
9. Stereo Bus (routes the channel to the Stereo Master Bus).
10. Pan (position within the Stereo Master Bus).
11. Mono Level.
12. Mono Bus (routes the channel to the Mono Bus).
13. Solo B (sets whether the channel will be soloed on the Monitor A or Monitor B bus).
14. Talk assignment (press to change to one of the 12 shout mixer paths).

# 8.10 Setting a Pre-Amp's input gain




The surface has two types of input level adjustment per channel, one is the remote Gain for the analogue microphone pre-amp (stage box gain) and the other is the digital Trim (console gain). In the CONFIG section there are dedicated controls for both Gain and Trim.


Adjustments of any selected channel Gain can be made here in 2.5dB steps for most I/O boxes (-2.5dB to +45dB for Gain and -40dB to +20dB for Trim). Both Gain and Trim values can be seen in the channel view widget which can be automatically displayed when the gain or trim pot is touched.

**Tip:** Locking the Gain rotary to screen is a great way to quickly set the gains for channels in a line check by selecting each channel in turn and adjusting the locked-to-screen rotary as required.

# 8.11 Linking & Stereo

The Linking page is where channels can be set to be Mono, linked as two Mono Channels or combined on one fader to function as a fully stereo channel, plus which channel parameters will be linked or act independently from each other.


1. Mono. The channel is Mono.
2. Link. Two adjacent mono channels linked together.
3. Stereo. Two adjacent channels linked as above but controlled by one a single fader. Use the SEL button to switch between the left and right view of the channel or press the channel name on widget.
4. Pan position and meter information.
5. A choice of Independent or Mirror pan link mode.

6. Selectable options which will link or unlink various channel controls. See chapter 9 on Linking for more information.

# 8.12 Patching

This page gives an overview of where the channel is patched to and from. It's a great place to check all connections are correct for the selected channel. Listed below are the various options available to patch the selected channel to and from various internal sources and destinations or to external devices.


1. 48V on/off.
2. Patch Source information. Press to navigate to the patching page (this is the same for all similar icons in the console).
3. Analogue mic preamp Gain (-2.5 dB to 45 dB).
4. Tape Return 48V (can be used as an alternative input source if required).
5. Tape Return patching information.
6. Tape Gain (-2.5 dB to 45 dB).
7. Compressor Side Chain patching information.
8. Gate Key patching information.
9. Insert Send On/Off
10. Insert Send patching information.
11. Insert Return A patching information.
12. Insert Return B patching information.

# 8.13 Direct Output


1. Choose the pick off point. Either Post Input, Post Processing or Post Fader.
2. Pressing Solo sends the direct output signal to Monitor A
3. If Solo B is active, Solo will send the signal to the Monitor B bus.
4. The Direct Out Mute button can only be unmuted if the direct output is patched.
5. Patch point information.
6. Direct Output level control (-inf to 10 dB).
7. Direct Output meter.

# 8 .14 Options


1. Processing order can be changed by long pressing and dragging the icons. DYN, Insert and EQ can be arranged in any order. It is possible to drag the Fader before the Insert only for a post fader insert point (Fader can only be dragged before Insert).
2. Below are the channel automation safes. The options are All, Gate, Comp, Equaliser, Filters, Config, Mute or Main Bus (Fader level, Pan position, Stereo Bus switch, Mono Bus switch and Mono Bus Level control).

**Note:** When an automation safe is on, the parameters associated with that safe type will not be recalled when a scene is recalled.

# 8.15 Configuration Pages for Outputs


1. Digital Trim. Range is from -12 dB to +6 dB.
2. Phase (180° Polarity change.)
3. Delay On/Off.
4. Delay Time (displayed in milliseconds and meters, maximum 500.0ms).
5. Group mode (turn an aux into a subgroup).
6. Talk On/Off.
7. Stereo Bus (sends the channel to the Stereo Master Bus).
8. Pan (position within the Stereo Master bus).
9. Level control to the Mono Bus.
10. Mono Bus (sends the channel to the Mono Bus).
11. Sends to Solo B for monitoring purposes.
12. Flexi Aux on (Flexi Aux allows group/stem style Aux to Aux mixing).
13. Talk Bus/Shout mixer output assign. (See Shout Config for details).
14. Sends Pan Follow makes contributions follow the pan position of the channel they are derived from.

# 8.16 Linking & Stereo (Outputs)


1. Mono. The channel is Mono.
2. Link. Two mono channels linked together; the faders can be linked by using the Fader option.
3. Stereo. Two linked channels controlled by a single fader.
4. Current pan position and level meter information.
5. A choice of Independent or Mirror pan link mode.
6. Options to link or unlink parameters when a pair of inputs are linked.

**Note:** Remember Outputs have to link odd to even. I.E. 1-2 or 9-10 etc.

# 8.17 Patching (Outputs)


1. Aux Compressor Side Chain patch information.
2. Send and Return compensation on/off. In general, this will need to be on in order to preserve audio time alignment.
3. Aux Insert Send patch information.
4. Insert On/Off.
5. Insert Return A patch information.
6. Insert Return B patch information.
7. Bus Output mute.
8. Bus Output patch information.
9. Post-fader Bus Output level (fader level).

# 8.18 Direct Input


1. The DIRECT INPUT MUTE can only be unmuted if the direct input is patched.

2. Direct input level (-inf to 10 dB).

3. Choose the Inject point. Either Post Input (pre-processing) or Post Processing.

4. Pressing Solo will send the direct input signal to Monitor Bus A unless the Solo B button is active, then the signal is sent to Monitor Bus B for audition.

# 8.19 Options


1. Processing order can be changed by long pressing and dragging the icons in an order you like. It is possible to drag the Fader before the Insert for a post fader insert point (Fader can only follow Insert).
2. Below are the channel automation safes. The options are All, Comp, Equaliser, Config, Mute or Main Bus. See chapter 19 on Safes for more information.

# 8.20 Oscillator (Output Widget)

1. Oscillator Level control.
2. Frequency control (50 Hz to 5 kHz).
3. 1 kHz fixed tone.
4. Pink Noise on/off.
5. Internal send on/off.
6. Oscillator to Talk Internal on/off
7. Talk on/off.


# 8.21 Setting the HPF and LPF

The high and low pass filters can be switched on/off independently of each other. Each filter has two slope settings. The filters are replicated on the GUI, which also shows the value of the frequency and filter slope cutoff.

First select the channel you wish to adjust by either pressing the appropriate SEL button on the surface or via the GUI.


1. In the Equaliser section the High Pass Filter (HPF) and Low Pass Filter (LPF) can be toggled on and off with the ON button depending on selection.
2. Frequencies are defined with the corresponding HPF/LPF Freq controls. The ranges are 10Hz to 10 kHz for the high pass filter and 40Hz to 20kHz for the low pass filter. When either filter is switched on it is highlighted purple in the widget.
3. There are two filter slopes to choose from for each filter: 12dB/Oct or 24dB/Oct for the HPF and 6dB/Oct or 12dB/Oct for the LPF.
4. In the Equaliser display widget, the HPF and LPF settings will be displayed on screen. They are displayed in purple when active and flatten when turned off. The filters can be altered by touching the corresponding purple dot to change the frequency of the filter.

**Note:** The frequency for both filters can also be set by using the pop-up rotary by touching the FREQ control display in the Equaliser widget (in between the Filter on and Filter slope). It can also be locked to the screen. Please refer to the Navigation section on how the pop-up rotaries function.

# 8.22 Input Equalisation (EQ)

To equalise the input signal, use the 4-band parametric EQ, which is situated in the Equaliser section of the channel overview or Equaliser widget. Band 1 and 4 each have a parametric filter option and three specific shelving modes. Visual feedback for EQ is either via Equaliser widget, and channel detail area. In addition, it is possible to hear changes to EQ in the solo bus before the change is committed to the input or output with the innovative EQ audition mode.

To EQ the input signal:


1. Select the bank you wish to pick a channel to EQ.
2. Select the channel you wish to adjust via the GUI or by pressing the surface SEL button.
3. Navigate to the Equaliser widget either by touching any control in the Equaliser section on the surface or touching Equaliser in the input widget screen. There are 3 buttons to either: Turn the HPF, LPF or EQ on.
4. In the Equaliser section, press the green EQ button to activate the EQ (either on the surface or on screen). The EQ button's LED will illuminate green when on and show a green outline when off.
5. In the Equaliser section, adjust the Frequency, Width and Gain control knobs to change EQ parameters as required or, in the Equaliser widget, touch the EQ band you wish to adjust. To adjust the width of the selected EQ band, use the pinch gesture to increase or decrease the width of the EQ band. A pinch gesture is a continuous gesture that tracks the distance between the first two fingers that touch the screen when moving left and right.
6. To Flatten the EQ curve press and hold the red Flatten button in the GUI until the line traces around the outside of the button. Once this two second press has elapsed the EQ will be reset. All the EQ bands frequency and widths will remain in the same positions, but the gain will be reset to 0db for each band. This flatten function does not affect the HPF and LPF settings.
7. Engage True Audition mode.

**Note:** EQ Gain, Freq or Width in the Equaliser widget can be set by using the pop-up rotary by touching the corresponding Gain, Freq or Width display in the Equaliser widget. They can also be locked to the screen if required. Please refer to the Navigation section on how the pop-up rotaries function.

**Note:** The minimum harmonic disruption filters are bright and deep, which are available for treble and bass, respectively. These filters use psychoacoustic phenomena to generate steep slopes that sound natural.

Audition the different filters, including the 'minimum harmonic disruption' types, by scrolling through them using the Shape button in the Equaliser section or in the Equaliser widget.

Band 1 HP 6dB, HP12dB, Bell or Shelf filters (Deep, Classic and Warm).
Band 2 Bell with Gain, Frequency and Width controls.
Band 3 Bell with Gain, Frequency and Width controls.
Band 4 LP 6dB, LP12dB, Bell or Shelf filters (Bright, Classic and Soft).

## 8.23 True Audition

In the top right of the EQ widget is the innovative EQ Audition button. In this mode the EQ display turns yellow and any changes to EQ will only be heard in the solo bus. For example, this allows you to fine-tune the EQ without affecting the sound in an artist's in-ear mix. Once you have made your changes either press the green APPLY button to commit instantly or press the red X to cancel the audition EQ and return you to your live mix.


Warning: If you press the green commit button any changes will happen immediately and can affect everyone listening to that channel. Big EQ changes will be very noticeable.

The Flatten EQ and EQ ON buttons can also be heard just in the solo bus when in preview mode. This allows you to hear the difference the whole EQ is making on a signal before committing to it.

# 8.24 Output Processing

The outputs have a four-band parametric equaliser with shelving modes on bands 1, 2 and 4 and operate in a similar way to the input EQ. The options are:

Band 1 HP 6dB, HP12dB, Bell or Shelf filters.
Band 2 Bell or HP24dB (When HP24dB is in use Band 1 is disabled)
Band 3 Bell with Gain, Frequency and Width controls.
Band 4 LP 6dB, LP12dB, Bell or Shelf filters.

# 8.25 Input Dynamics Processing (Dynamics Section)

This section shows you how to use the compressor and gate dynamics processors via the surface Dynamics section and various GUI screens. The rotary pop-up functions can also be used to adjust certain settings of the gate and compressor which can also be locked to the screen if desired.

## Setting Up A Compressor

**Surface Controls**

**FOH Workflow View**


## Channel View

To set up a compressor/limiter:

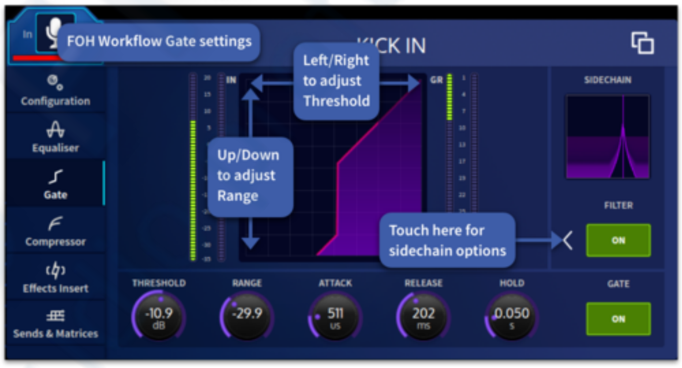

1. Select the channel you wish to adjust. Then navigate to the Compressor widget either by touching any control in the Dynamics section on the surface or touching Compressor in the input widget screen. Alternatively, in the Channel View workflow touch the small compressor section to the left to see the Compressor and sidechain section enlarged to the right.
2. On the surface Dynamics section press the SEL button associated with the compressor, this assigns the compressor functions to the physical controls, then press the ON button. The Button will illuminate green when the compressor is on. Alternatively, in the Compressor widget touch the Compressor ON button in the right-hand bottom corner. This will illuminate green when on and only show a green outline when off. You will also see the compressor curve turn green when the compressor is active. Sidechain filter ON and Filter Edit buttons are located in the bottom left of the Compressor widget or to the side if in Channel View workflow. The sidechain curve will turn purple when turned on and active.
3. In the Dynamics section, adjust the Attack, Ratio, Release, Threshold and Make Up controls to apply processing. You can also set up a limiter by using a high threshold and a steep ratio (greater than 10:1). The Hold control has no function when using the compressor and is only used with the gate. Threshold can be adjusted in the GUI by swiping up and down in the GUI display.
4. Press KNEE in either the Dynamics section or in the Compressor widget to audition the different slope algorithms (hard knee, medium knee and soft knee).
5. Press MODE in either the Dynamics section or in the Compressor widget to audition the different compressor types (Corrective, Adaptive, Creative and Vintage). More information on the different compression modes can be found in chapter 29.
6. Press Sidechain FILTER in the Compressor widget to access the sidechain function. From this page you can set the side chain, listen to the source, toggle the sidechain on or off, set the frequency and Width. Press the blue down arrow Comp button to return to the main compressor settings.

**Note:** The outputs have the same four compressor modes as the input channels. Gates are not available on outputs. If you do need a gate on an output, use the SQ1 in gate mode.

## Setting Up A Gate

### Channel View


### FOH Workflow


1. Select the channel you wish to adjust. Navigate to the Gate widget (Channel View or FOH Workflow) either by touching any control in the Dynamics section or touching Gate in the input widget screen.
2. In the Dynamics section press the SEL button associated with the Gate, this assigns the gate functions to the physical controls. Press the ON button and it will illuminate green. Alternatively, in the Gate widget touch the gate ON button in the right-hand bottom corner which will illuminate in green when on and show a green outline when off. You will also see the Gate curve turn purple when the gate is active. Sidechain filter ON and Filter Edit buttons are located in the bottom left of the Gate widget.
3. In the Dynamics section, adjust the Attack, Hold, Release, Range/Ratio (Ratio) and Threshold controls to apply processing. The Make-Up control has no function when using the Gate as it is only used with the compressor. You can adjust the threshold of the gate in the GUI by swiping left or right in the gate graphical display. You can also adjust the Range in a similar way by swiping up and down in the same GUI window.
4. In the Channel View workflow sidechain functions are presented to the right side of the GUI. In the FOH workflow press the Sidechain Filter arrow to access the sidechain functions of the gate. In both areas you can set the side chain, listen to the source, toggle the sidechain on or off, and set the frequency and width of the sidechain. Press the purple down arrow button to return to the main Gate settings.

**Gate Surface Controls**

The familiar surface controls mirror the GUI and give a more traditional approach to working with the HD96-24 system. Press Sel to activate the gate controls on the surface for the selected channel. The controls function in the same way as the GUI.

# 8.26 Using VCA & POP groups

The HD96-24 has 24 POP groups and 24 VCAs. VCA/POP groups allow simultaneous control over a large number of channels. This provides a quick method of bringing particular channels to the control surface and saves you having to remember their name/number. You can choose channel group associations and also configure the colour, name and image of each group's LCD select button to make them instantly recognisable. Select Groups in the main GUI top bar menu. This will display the groups information and setup page as shown below. When no group is selected, handy instructions are visible for your reference.

In the Global Assignable Shortcuts area, the associated POP or VCA group (when setup) is selected by the SEL button under the LCD screen. Global Assignable Shortcuts are setup in the Navigation page (see Chapter 6 Navigation or Chapter 18 for Global Assignable Shortcuts set up details).

Holding the SEL button takes you to the Groups page to freely assign channels. Any group can have any channels (input/output) assigned to them, although in normal practice is more likely that they will only have one or the other.

The difference between VCA groups and POP groups are that VCA groups include fader, solo and mute control. POP groups are limited to bringing channels to the control surface and have no other function. POP groups let you create a group of related instruments that you require on the control surface for quick access. For example, all channels associated with a person's monitor mix can be assigned to a POP group and brought to the surface with one button press or all the drum microphones can be grouped together.

**How to configure a VCA/POP group:**

Select Groups in the main GUI top bar menu. This will display the groups information and setup page as shown below. When no group is selected, handy instructions are visible for your reference.



1. Select POPs from the list on the left-hand side.
2. Select the first POP Group you would like to assign inputs, outputs or VCAs to.
3. On the right-hand side you can touch the large POP image or name to access the naming, tagging and Image page. Naming, tagging and Image has been explained in an earlier section. The name can also be edited by pressing the name next to the Add Channels to Group area.
4. Transfer Colour At this point you have the option to transfer the colour to channels which will be added to the POP Group. This can be done at any point by long pressing the button.
5. With the POP group highlighted tap the Inputs, Outputs or VCAs you wish to be assigned to that POP Group. Note the order in which you tap the inputs, outputs or VCAs is the order they will appear on the surface. To change this order press and hold on the desired input, output or VCA until the icon wiggle animation starts. It now can be freely moved to which ever position you like or removed by dragging it out of the drop zone. Each POP group can contain all different types of inputs, outputs or VCAs. If more than 16 paths are added to a POP group, each press of the POP group button will toggle through the different banks of channels associated within that POP group (once assigned to the global shortcuts area). There is no limit on how many channels can be added to a Pop or VCA.
6. Remove Gaps If there are blank channels in the selected POP group, they will be removed with subsequent channels moved down to fill the gaps.
7. Order Ascending This function re-orders the POP group or VCA to be in numerical order (lowest channel first).
8. Order Descending This function re-orders the POP group in reverse numerical order (highest channel first)
9. The Empty Group button allows any POP or VCA groups to be cleared quickly and operates when holding the button for a short period.
10. Decide where the POP group unfolds. Choose either Area A or Area B.

There is also an option to add group tags. This means for example if you create a POP group for the drums you can then add a “drums” tag to all the drum kit. This is a great way to organise large shows to simplify operation.

If you wish to move or reorder the channels simply hold on to the channel icon you wish to move until it wiggles, then drag it to the new position. Channels will automatically move up and down to allow the selected channel to be dropped in its new position.

The same principles apply to the VCAs. The only difference being only Inputs and Outputs can be assigned to VCAs.

## 8.27 Alternative way to assign VCA/POP groups

Pressing and holding down the SEL button of the desired group (VCA or POP) in the global assignable shortcuts area will take you to the Groups page and automatically select that group to edit.

**Note:** If more than 16 channels are assigned to a POP or VCA group pressing the groups SEL button will page through all assigned channels on the surface. Again, these can be fully reordered as desired.

Pressing the VCA SEL button will also show channels assigned to it and will page through the channels if more than 16 channels are assigned.

## 8.28 Setting up an Aux/Group mix

The control surface has 96 configurable aux busses each of which can be used as a Subgroup or as a Flexi-Aux to send to other auxes. All of these busses can also be set up as mono, stereo or paired/linked together. 24 matrix outputs can also be accessed directly from input channels via level controls. Similarly, to the inputs and groups, identification of mixes is by colour coding.

In the input channel there is a Sends page which allows the selected channel to be turned On, set to Pre-Fader, sent to any Aux or Matrix using the Absolute, Relative or Pass-thru method. It can also be used to adjust the pan in stereo Aux or Matrices.

The overview displays in the GUI channel strip show the status of these busses, which are colour coordinated to match those in the sends section of the control surface.

# 8.29 Routing to master stereo outputs

The following shows you how to route audio. Before proceeding with this operation, make sure nothing is muted and master faders are up.

To send audio to the L/R Master Output do one of the following:

Select the channel and press the ST (stereo) button in the Main Bus section on the surface.

In the GUI select the channel you wish to adjust and navigate to the Config page in the widget. Press the STEREO BUS button.

# 8.30 Manchino Multi Edit Page (Basic introduction)

The Manchino page allows you to select and edit any number of inputs or outputs at once in one very easy place. For example, all channels can easily be selected and sent to the stereo bus. A selection of channels could be made Pre-Fader all at once in many aux sends with a few touches of the screen.

On the left side of the GUI there are two tabs which are for either inputs or outputs. Functions for inputs and outputs vary depending on the parameters associated with them. For example, 48v can't be added to an AUX master.

To select the inputs or outputs you wish to edit simple touch their associated button on screen, by pressing and holding on a channel or bus button all channels or busses can be selected at once. Individual channels can be removed by tapping on them.

If you select a few different channels and wish to clear your selection, press and hold on one of the selected channels to clear all. If you select a couple of channels or busses then wish to add all the rest of the channels or busses, press and hold on a new unselected channel or bus. This will then select all other items in that page.

With no channels selected certain buttons can be pressed to see the current status of the currently viewed channels. E.G. press and hold the phase button to see which channels have been phase inverted or press and hold the Stereo Bus button to see which channels are routed to the main master buss. This is a great way to fault find by seeing the overall status of functions on many channels at a glance.

See Chapter 33 for a more in-depth view of the power of the Manchino page.

# Security (locking mode)

If you wish to lock the console, log out of your user profile by clicking on your Username in the top bar or side bar menu. On your return enter your password to operate the system again.

Alternatively, as a quick way to lock the surface while you are away from the console, navigate to the user profile page (press on your Username icon) and press Lock Screen. This stops fader positions and controls being moved (apart from headphones level and Talk Gain which are hardware controls).

Press Unlock Console to regain access to the system.

# Chapter 9: Stéréo Linking

## 9.1 Input Linking & Stereo Page

By default, all of the channels of the HD96-24 system are mono (unpaired). However, adjacent channels can be linked together to form a stereo pair, which is known as “stereo linking” (or “channel pairing”).

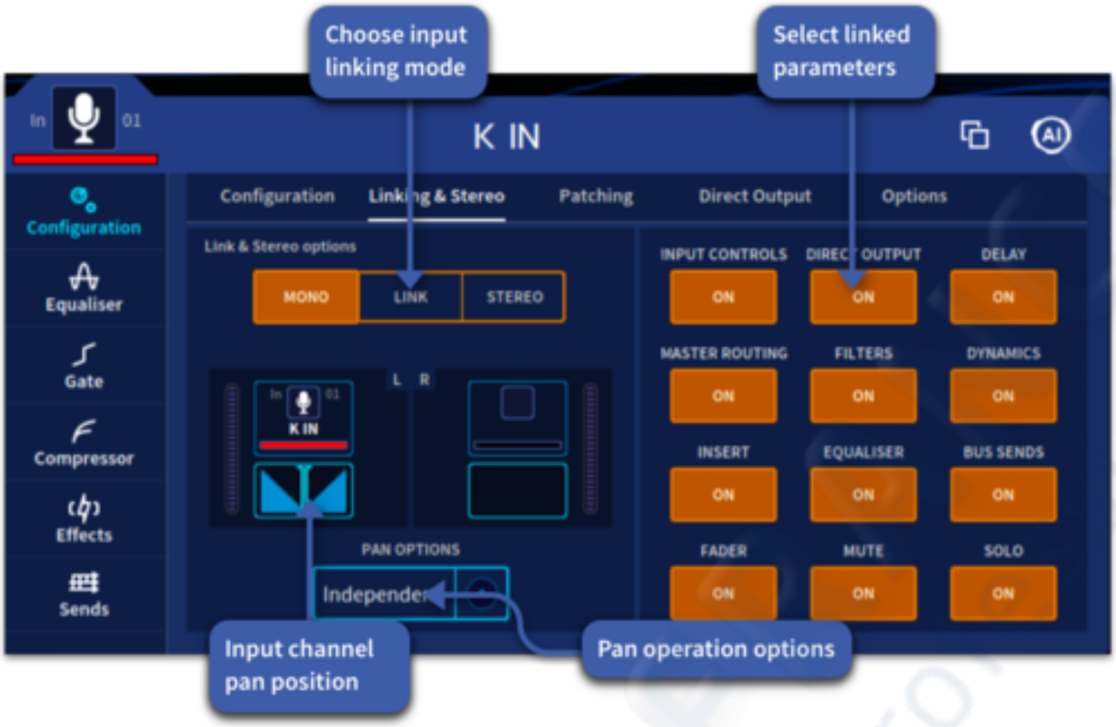
You can choose which controls/parameters are linked across the channel pairs. However, these can be overridden from the Linking & Stereo widget on per pair basis. There are two options for linking channels which are Link or Stereo.

All link states plus the parameter settings are saved and recalled by Automation, and so can be set on a scene-by-scene basis.

When paired by the Link option, by holding the link button for a few seconds, the controls for each signal path act simultaneously on both the left and right signal paths. Individual trims, for example, adjusting the mic amp gains to balance stereo mix inputs, can be applied to the left and right audio paths independently. The channels are not truly mono at this time, and any settings necessary to preserve the audio prior to trimming, such as dynamics side chain linking, are maintained. Note that any two adjacent input channels can be linked together i.e. 1 to 2 or 2 to 3 and so on. Only odd to even outputs can be linked. I.e. Aux 1 to Aux 2, Aux 3 to Aux 4 etc. You cannot link Aux 2 to Aux 3 for example.

When Stereo is selected both chosen channels are placed onto one fader creating a true stereo fader and the pans are automatically set to hard left and hard respectively, though these positions can be subsequently changed if required. There are options to change how the pans for each channel behave. Either Independent, so that any position can be set on each side or to Mirror where the position of one channel's pan will symmetrically mirror the other. For example, setting one pan to the 10 o'clock position will set the other pan to 2 o'clock. Pre-amp settings remain independent to allow changes to be made.

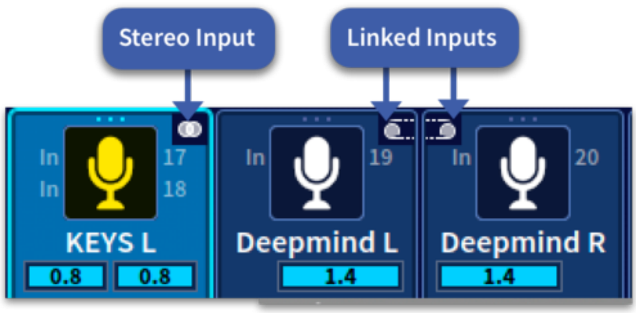
When linking previously unlinked channels, some normalisation of the prospective left and right control settings, which may be quite different, is required. The HD96-24 does this by automatically copying the control settings of the left channel (with the exception of the pan controls) to the right channel. The pan controls should be manually set.




To Link or make Stereo channels:

- Select the left side of the input or output you wish to link in either the GUI or on the surface.
- Navigate to the Link & Stereo options area in the Configuration page found in the Home, Channel View or FOH workflows.
- In the Linking & Stereo page hold for a short time the Link or Stereo button until the line traces around the button and the button changes.

The channels are now linked or made into a fully stereo channel. The GUI icon will show that the two channels are tied together if link mode is used. If stereo is used the right side will disappear from the surface and a symbol denoting a stereo channel will appear on the icon. When any path is Stereo linked, only the name for the left side of the pair is displayed.

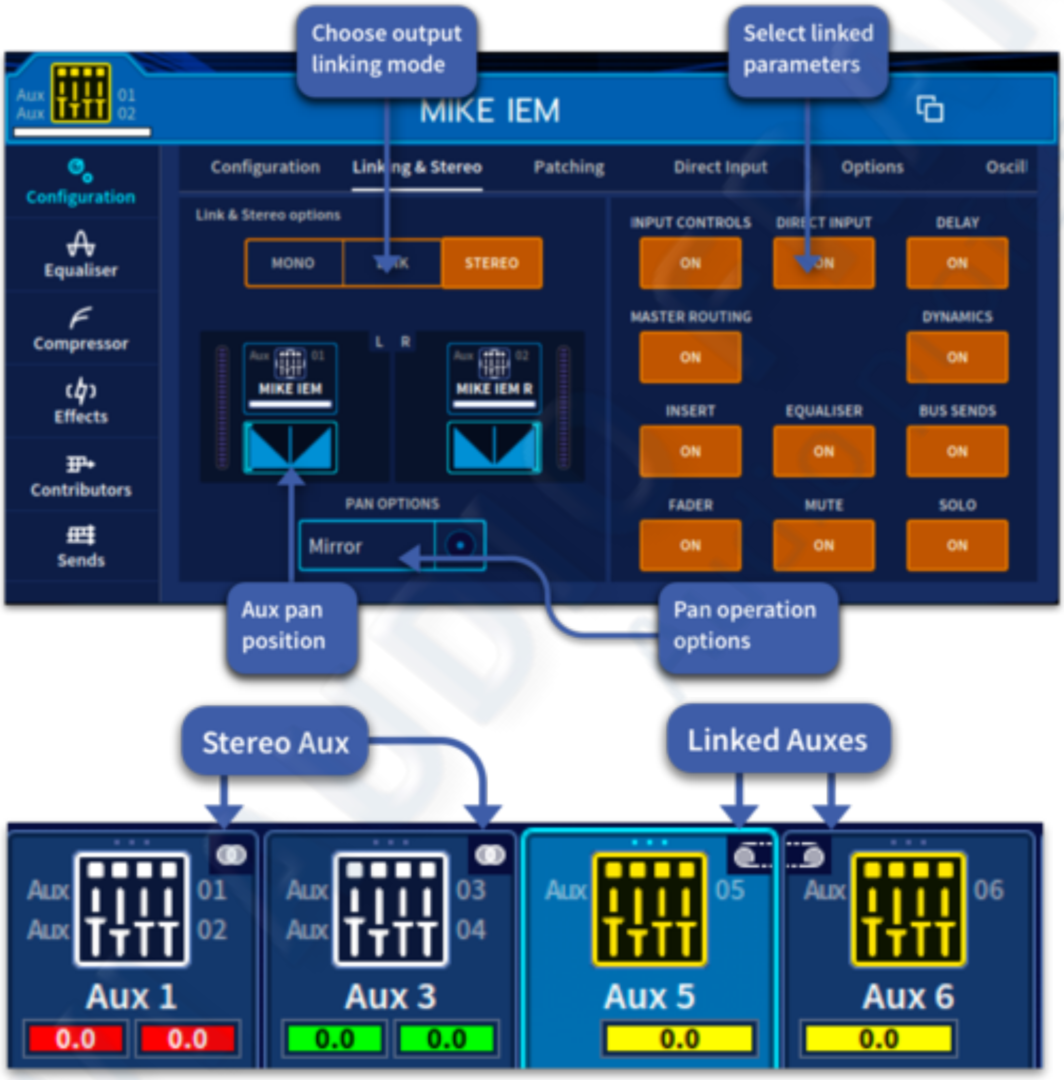



On the console surface, the LINK button will be unlit if a path is mono or lit when a path is set to either Link or Stereo. While pressing the surface button will create a linked channel, creating a Stereo channel can only be done on screen.

# 9.2 Aux Linking & Stereo page

Available link options are displayed for Aux paths on the right-hand side of the widget. Choose link mode as described in input channel linking.

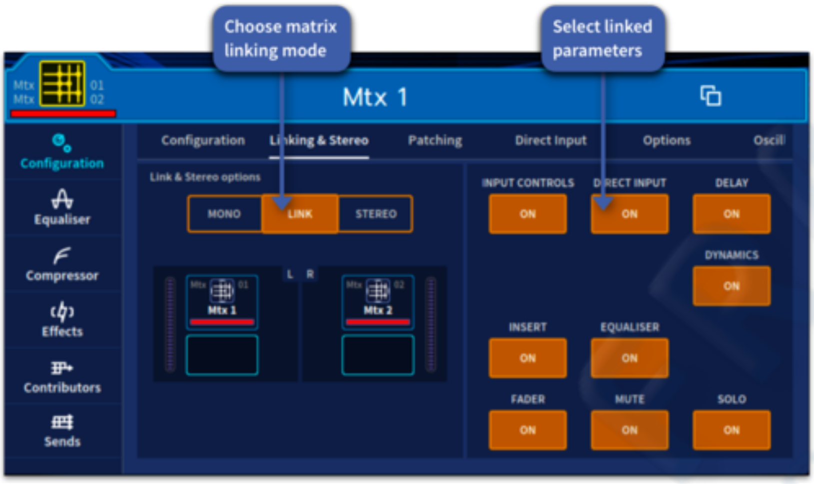
**Note:** Any two adjacent input channels can be linked together i.e. 1 to 2 or 2 to 3 and so on. Only odd to even outputs can be linked, such as Aux 1 to Aux 2, Aux 3 to Aux 4 etc. You cannot link Aux 2 to Aux 3 for example.



## 9.3 Matrix Linking & Stereo page

Available link options are displayed for Matrices on the right-hand side of the widget. Choose link mode as described in Input channel linking.

**Note:** Matrices cannot be panned as they are last in the signal path.

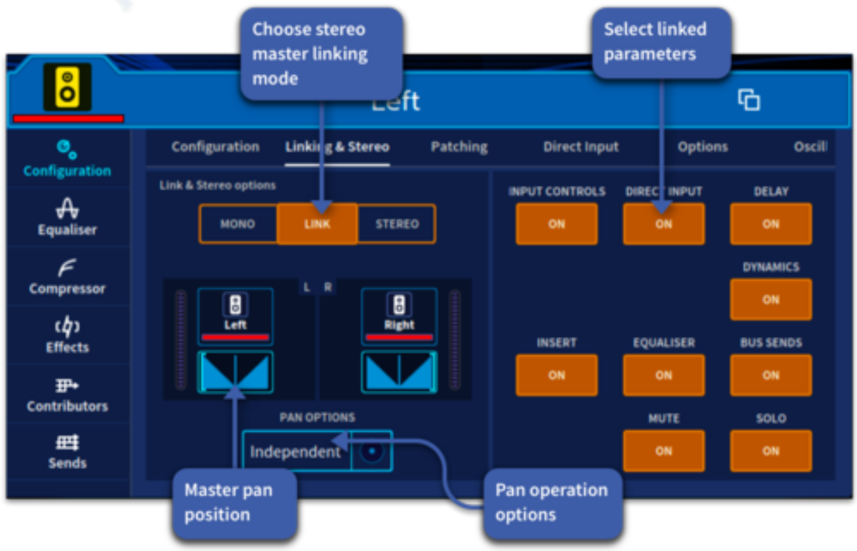



## 9.4 Linking the Master Channels

You can link the left and right master channels in a two-way link (left and right) or as a single stereo fader, both of which use the linking parameters set for the left master channel.

To link the left and right master channels

1. In the left master channel (control surface), press its SEL button (just above the fader) to select it.
2. Navigate to the Master Channel Configuration page in either the Home, Channel View or FOH widgets.
3. Set the link type for the left master channel by long holding Link or Stereo until the line has traced around the outside and the button turns orange.



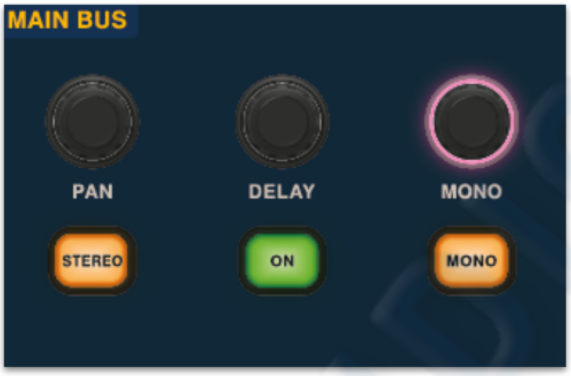
# Chapter10: Panning

## 10.1 Stereo panning

The surface controls for stereo panning are located either in the Main Bus sections, on the Assignable Controls area above each fader screen when selected or in the Configuration widget section of each input (available in all workflow areas). The surface LCD displays also show pan information.

Pan is scaled from -100 to +100 with 0 at the central point.

The Pan control has a snap to centre position when you are close to the 0 point in all views. The pan indicator in all GUI views also changes to orange when in the central position.




# Chapter11: Soloing

## 11.1 Soloing

With the Solo button you can isolate a single channel, which is helpful in fault finding or when equalising a signal. Pressing a Solo button cuts all signals routed to the monitor output, except those channels where a Solo has been engaged. You can then monitor a signal at a level proportional to its level in the mix, in the same stereo position in the mix and with the same reverberation as in the mix. It can also be used to listen to a group of channels assigned to a VCA, such as a drum kit which can then be heard in full via a VCA. . There are 2 monitor busses on HD96 – Solo A which interrupts Monitor A and Solo B which interrupts Monitor B. Both busses operate in the same manner, so they have a similar set of controls for each. When a Solo is cleared, that monitor bus reverts to its previous monitoring selection.


- PFL Inputs A - Pre-Fade Listen active for an input on solo bus A. If unlit, the solo will be AFL.
- PFL Inputs B - Pre-Fade Listen active for an input on solo bus B. If unlit, the solo will be AFL.
- PFL Outputs A - Pre-Fade Listen active for an output on solo bus A. If unlit, the solo will be AFL.
- PFL Outputs B - Pre-Fade Listen active for an output on solo bus B If unlit, the solo will be AFL.
- ADD Solo Bus A - Enables multiple solos to be heard at once. If unlit, only one solo at a time can be heard.
- ADD Solo Bus B - Enables multiple solos to be heard at once. If unlit, only one solo at a time can be heard.
- CLEAR A - Cancels all solos on A and returns Monitor A to its previous monitoring selection.
- CLEAR B - Cancels all solos on B and returns Monitor B to its previous monitoring selection.

The HD24 Control Surface has two independent solo systems, Solo A and Solo B. Solo A interrupts the Monitor A bus when a solo is activated; the Solo B interrupts the Monitor B bus when a solo is activated. Any path can solo on either A or B. Both have monitor and headphone outputs, and both can be used to PFL or AFL signals from the same sources throughout the control centre. This flexible solo bus configuration makes soloing of monitor mixes incredibly easy— in-ears going to solo A and wedge going to solo B — thus the ability to choose whether an input or output goes to solo A or solo B or on a channel by channel or Aux by Aux basis.

**Note:** The headphone level controls are independent of the Solo A and B bus level. This means if you use the C/O button for a Monitor Bus and use the fader to control a local monitor speaker or a PFL IEM system, the headphones will still be audible (if the level is turned up) even if this fader is all the way down.

# 11.2 Using Solo A/B

In its default state, all paths have their solo assignment set to Solo A. To change this, select the Configuration tab on the channel widget (in Home, Channel View or Automation workflows) and press the SOLO B button.


With Solo A/B on, solo goes to the selected solo bus with the following conditions:

If the solo button is pressed and released, the soloing to the selected solo bus remains active when the button is released.

If the solo button is held down, it will only be active for the length of the button press.

Pre-fader audio is sent to the selected solo bus if the associated PFL control for that bus is active.

Post-fader audio is sent to the selected solo bus if the associated PFL control is inactive.

Unless multiple solo activations to the same solo bus are active, the last solo selection, while the respective solo add mode (A or B) is inactive will cancel all earlier solos to the same bus before it activates.

Soloing a VCA will also solo all the input channels contained within it. This is in addition to the local operation. Channels will still be sent to which ever solo bus they are assigned too.

Pressing the solo Clear button associated with the Solo Bus (A or B) clear all active solos for that bus.

A solo hierarchy exists for each of the solo busses in the system. Activating a solo with a higher precedence in the hierarchy deactivates all solos with less precedence and inhibits them from being operated. As soon as the higher precedence solos are cleared, the stages of the inhibited solos are restored, and they resume normal operation.

Some modifications to this hierarchy are possible. For example, mix busses can be used as sub-mixes (hierarchy is as described) or outputs (having same precedence as master outputs).

Turning ADD (solo A/B) off cancels all solos.

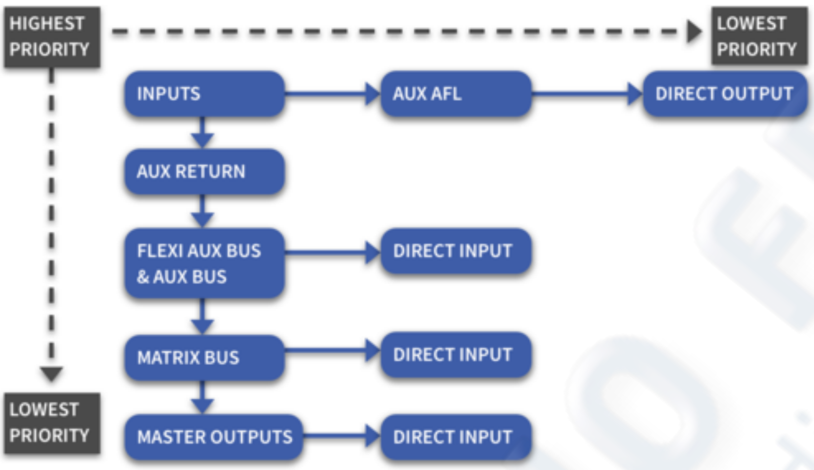
When Soloing inputs and outputs (with solo ADD switched on) the following rules apply:

- With any inputs active, you can't solo outputs.
- With any outputs active, pressing an input solo temporarily suspends the output solo. Then, if you cancel the input solo(s), the previous output selection solo(s) is reinstated.

# 11.3 Solo Hierarchy

Solo signals are divided into five layers: Input Solo, Aux Return Solo, Aux & Flexi Aux Bus Solo, Matrix Bus Solo and Master Buss Solo. The illustration below shows the priority of the Solo function. After you have switched layers from the lower to upper levels by cancelling a solo for the upper layer, the previous Solo status of the layer immediately below will be restored. For example, if you switch layers in the order of:

Input Solo →Aux Return Solo → Flexi Aux & Aux Bus Solo → Matrix Bus Solo → Master Bus Solo, you can then successively cancel the Solo button to successively restore the Solo status of the previously selected layer.




The solo system add-mode hierarchy works as follows:

The highest level of solos will be the inputs and returns. When active, these will override and inhibit the remaining solo sources (flexi auxes, auxes, matrices and masters).

Within the constraints of the two-level solo hierarchy, only one source can be active on any channel at any instant:

- Input channels: Input channel <--> Aux AFL <--> Direct out
- Aux Return channels: Return channel
- Aux busses: Aux bus <--> Direct in <-->
- Matrix outputs: Matrix bus <--> Direct in <-->
- Master outputs: Master bus <--> Direct in <--> Side chain listen

An additional constraint is placed on the side chain listen. This is due to the nature of the DSP, where only one side chain listen can be auditioned at any one time, regardless of whatever else is active in the same solo hierarchy level.

If an input channel solo is active via a VCA master solo, soloing the input temporarily overrides the VCA master solo. However, soloing a direct input or AFL solo on the same channel or a side chain solo on any channel, cancels both the input solo on that channel

## 11.4 VCA Solo

If input and outputs are assigned to the same VCA it is possible to solo it and hear all the contributions together. If a path is soloed within a VCA, when that path is un-soloed the VCA solo will return. If Solo ADD mode is engaged only inputs will be sent to the solo bus. If two channels are in a VCA together and the first is set to Monitor A bus while the second is set to Monitor B bus. The two channels will still be soloed in the Monitor busses they are assigned to and not merged to the same Monitor bus.

## 11.5 Solo In Place (SIP)

In the Midas analogue console days, the yellow solo in place button (SIP) would be found with a protective cover over it. It was also known as the 'P45 button' (P45 is a UK form given after being fired) because if SIP was accidently used in a show, everything apart from the soloed instrument would disappear along with your mixing career!


Saying that, SIP can be a very useful tool in sound check to solo an instrument or voice while cutting all channels from the main mix (except soloed ones) by pressing a solo button. SIP lets you check the contribution from soloed channels at the actual levels they occur in the mix, that is, considering the main fader setting.

To prevent accidental SIP activation, the SIP button has a long button press to save it accidentally being turned on.

For SIP purposes, master outputs can be the main master bus.

To be eligible for SIP muting, channels must be input channels and set up to solo to the solo A bus; channels with any other combination cannot be subjected to SIP muting. Channels eligible for SIP muting that are currently or subsequently muted by a means other than SIP (that is, local button press, auto-mute or scene recall) remain muted, regardless of the SIP status. On removal of the overriding mute, the mute is restored according to the current SIP status.

# 11.6 Solo Management

The solo signals can be selected for each monitor system (A and B) to be AFL, PFL, additive or interlock cancelling. PFL and AFL audio busses may accept injected external signals, and two-level controls to make adjustments.

PFL and AFL levels are adjustable via the PFL level and AFL level controls.

The function of the buttons in each solo section is as follows:


1. Solo Dim – Turns the level of the solo bus down in the monitor bus in order to hear talk signals.
2. Input PFL - Sends mono pre-fader listen (PFL) solo bus input signals to headphones and local monitor outputs. With PFL switch disabled (LED extinguished), stereo after fader listen (AFL) solo bus signals are sent to headphones and local monitor outputs. If an input is linked or made stereo the pan position in the stereo image is followed in the monitor bus.
3. Output PFL - Sends mono pre-fader listen (PFL) solo bus output signals to headphones and local monitor outputs. With PFL switch disabled (LED extinguished), stereo after fader listen (AFL) solo bus signals are sent to headphones and local monitor outputs. If an output is linked or made stereo the pan position in the stereo image is followed in the monitor bus.
4. Solo Add - Allows multiple channel access to solo busses. When solo Add mode is off, pressing a solo switch cancels any currently active solos. Multiple solos (for example, stereo left and right signals) can be monitored in this mode provided solo switches are pressed at approximately the same time. When solo Add mode is on, auto-cancelling is defeated, which allows multiple channel or output soloing. In this mode, input solos have priority over outputs and VCA solos and will temporarily override them. When the input solo is cancelled, output solo or VCA solos are reinstated.
5. Solo Clear – This switch illuminates when a solo switch is active in its monitor section and, when pressed, clears any solo switches in that section.

# 11.7 Sources

The Source A and Source B sections contain monitor input selector switches in the Monitor Configuration page. On both the A and B systems, these define the source for the monitor section from the possible 'primary' choice of stereo master (ST), mono master (MONO) or Solo-Source External (EXT). Additionally, each section has a talkback button (talkback is a line level input on the back of the surface).

The function of the buttons in each source section is as follows:


1. Talkback [A and B] switches, sums the line level talkback input from the rear of the surface to the solo bus.
2. Stereo - Routes post-fader stereo master mix to stereo local monitor outputs.
3. Solo-Source External switch, routes stereo external input (two-track return etc.) to stereo local monitor outputs.
4. Solo Mono switch, routes post-fader mono masters mix to stereo local monitor outputs.

# 11.8 Summing

Each monitor bus has the following functions:


1. Mono – Sums left and right to mono. Great for checking stereo signals for polarity and phase issues.
2. Polarity - Inverts the right side of the monitor bus by 180°.
3. Left to Right – Sends the left monitor bus signal to both left and right sides of the selected monitor bus.
4. Right to Left - Sends the right monitor bus signal to both left and right sides of the selected monitor bus.

**Note:** pressing both Left and Right to Both swaps the left and right signals in the solo bus.

## 11.9 Solo System


The solo system section controls are found in the Monitor Control Settings page:

1. Solo In Place – Activates solo in place. Use with caution and remember to turn off.
2. PFL level control — PFL audio bus may accept injected external signals. This control adjusts the pre-fader level in the range infinity (∞) to 10 dB.
3. AFL level control — AFL audio bus may accept injected external signals. This control knob adjusts the after-fader level in the range infinity (∞) to 10 dB.
4. Input Select Follows Solo – When an input is soloed it is also selected.
5. Output Select Follows Solo – When an output is soloed it is also selected.
6. Input Solo Follows Select – When an input is selected it is automatically sent to the solo bus.
7. Output Solo Follows Select - When an output is selected it is automatically sent to the solo bus.
8. Monitor Output Swap - When active the Monitor A and B are reversed (including all patching). For example, if using an in ear monitor and listening monitor speaker this button allows you to quickly send monitoring signals going to the speaker to be audible in your in ear instead.

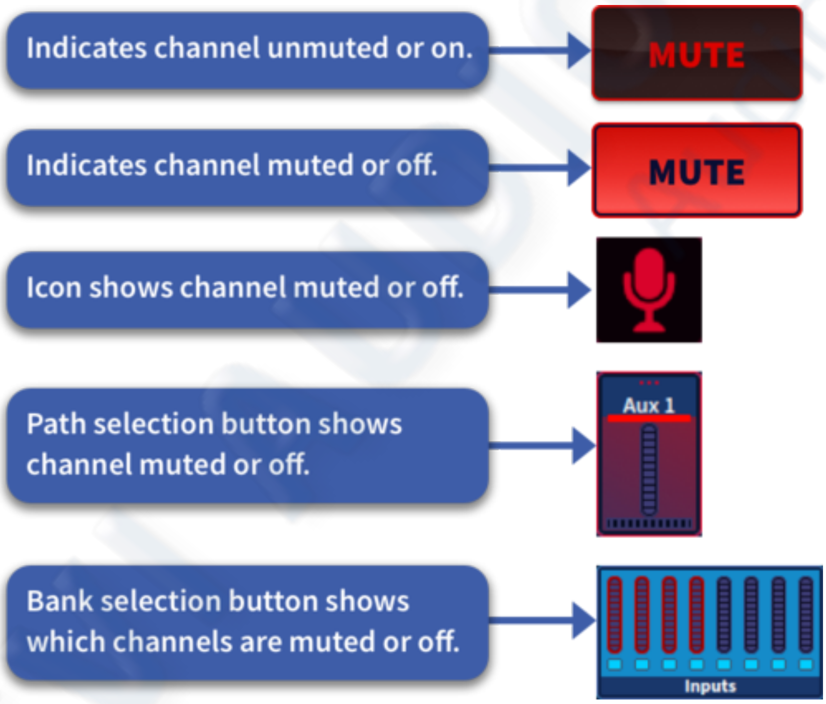
# Chapter12: Muting

## 12.1 Muting

You can cut (mute) the output signal of a channel. This is generally used for unused mics, guitar changes etc. Channel mutes can be activated by any of the following, which (except the VCAs) mute the channel outputs and update the channel mute status indicator:

- Local MUTE button press.
- Mute groups chapter 16
- VCAs — see “VCA and POP groups” in Chapter 8 Basic Operation.
- Scene recall (automation) — see Chapter 19 "Scenes and Shows (Automation)" in chapter 19.
- SIP — see “Solo in place (SIP)” in chapter 11.

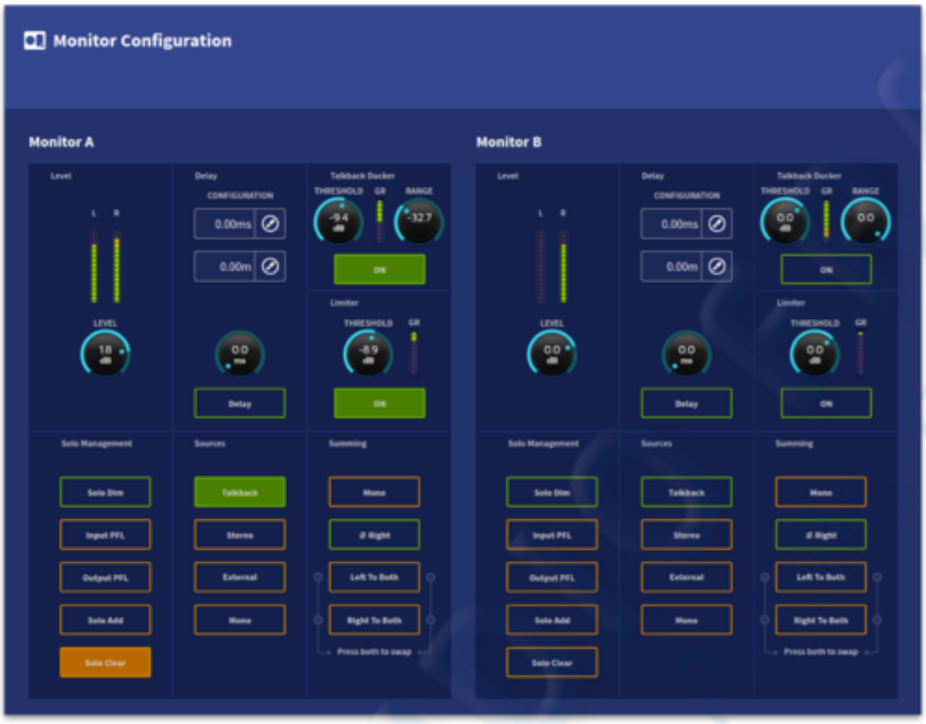
When a channel is muted its icon, gain, bank metering and selection icons will also turn red to indicate the channel is muted as listed below.



# Chapter13: Monitors and Shout Configuration

## 13.1 Monitor Configuration (A and B)

To match the two-bus solo system there are two monitor outputs, A and B, which control their respective output levels. These are controlled from the monitor section on the surface or within the Monitor Configuration page in the GUI.




Each monitor output has:

- The facility to monitor mono and stereo outputs.
- A stereo external input.
- An external talkback input (line level on the rear of the surface or patchable).
- A headphone output.
- Delay Configuration.
- Control of the solo busses.
- Polarity, this flips the right side 180°.
- Left to both A/B.
- Right to both A/B.
- Talkback Ducker
- A Limiter to protect hearing.

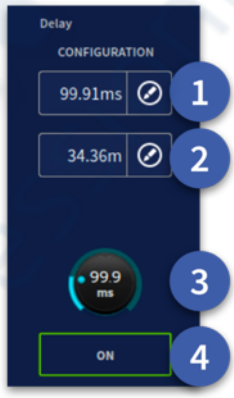
For more information on this section please refer to Chapter 26 Preferences. Although the capabilities of both monitors are the same, Monitor A is the primary output. They both have a fader control if required using the C/O buttons.

The monitor output controls will affect the levels shown in the Monitor Configuration page but are not affected by automation.

The Monitor A and Monitor B meters monitor the peak signal levels of stereo left and right for both monitor paths. The metering capabilities of both monitors are the same.

## 13.2 Delay

You can delay each monitor output signal (A and B) individually by up to 500 milliseconds (171.92 m). This is done via the two delay sections in the Monitor Control Settings page. Pressing the delay time allows a pop-up control to appear. This may be locked to the screen. This function does not have a dedicated control on the surface but can be assigned using the one pot shot method. The delay is displayed in both ms and meters for accurate delay time input.




1. Delay time in ms (delay time can be manually added by tapping the paint icon).
2. Delay time in meters (delay time can be manually added by tapping the paint icon).
3. Delay time control (hold for pop up control).
4. ON button for delay time.

**Tip:** You could delay the monitor output to the distance from stage so that the audio is in time at the mixing position. If a stage is 35 meters from the mixing position delay the monitor output by 101 ms. The rough rule is 34 cm distance = 1 ms. Delay time can be entered in meters.

## 13.3 Talkback Ducker and Limiter

For both Monitor A & B busses there is an independent Talkback Ducker and limiter. The Ducker drops the level of the currently monitored source when audio is present on the talkback. This is highly effective to always ensure you hear any communications via the talk or shout system. The following limiter protects the monitor bus from large peaks in audio. For example, if you have a drummer with a loud click track running in an aux mix you can set the limiter to catch this if you solo that aux by accident.

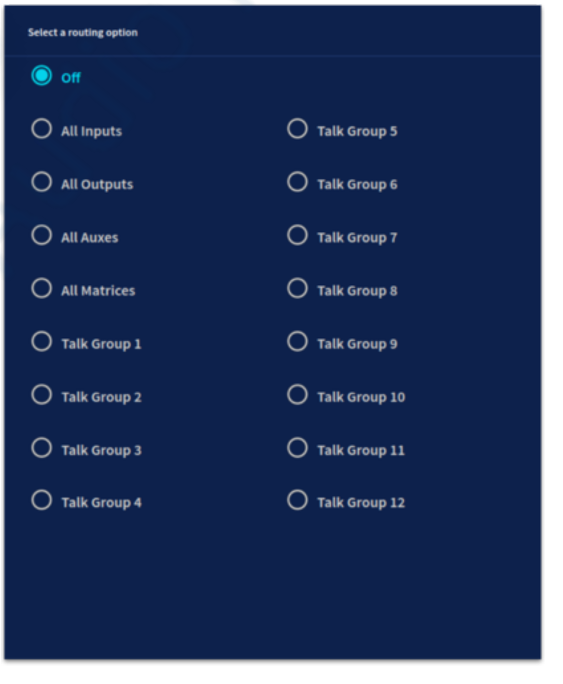

1. Threshold – Sets the level at which ducking happens.
2. Range – Sets the maximum amount of gain reduction.
3. GR meter – Displays the amount of ducker gain reduction.
4. On – Ducker on/off.
5. Threshold – Sets the level at which the limiter starts working.
6. GR meter – Shows the amount of limiter gain reduction.
7. On – Limiter On/Off.

## 13.4 Talk and Oscillator Configuration Page

The basic talk system allows you to send the internal talk mic to any of the routing options. For example, talk to all the auxes or to one of the Talk Groups (see system routings options).


1. Talk Internal – This button sends the talk mic to the selected routing option.
2. Talk Ext – This sends the Talk Bus to the Talk External patch point found in the patching page.
3. Level – Controls the level of the talk bus up to a maximum of 10 dB.
4. Limiter – Adjusts the limiter on the Talk bus.
5. Mic Gain – Control the level of the talk mic gain.
6. Talkback Level – Control the level of the line level Talkback found on the rear of the console.

## 13.5 Talk Osc Routing




System Routing options for the talk bus are:

- All Inputs (can be used to check external inserts or record sends)
- All Outputs (can be used to check all sends and outputs quickly)
- All Auxes (Talk to all auxes quickly)
- All Matrices (Check signal flow to the matrices)
- Talk Group 1-12 (Talk groups can be used to communicate with set groups of auxes, for example 1 group for the artist on stage, another to talk to all the technical staff).

## 13.6 Oscillator

The signal generator section can output pink noise (pink noise generator) or sine wave tone (sinusoidal oscillator).

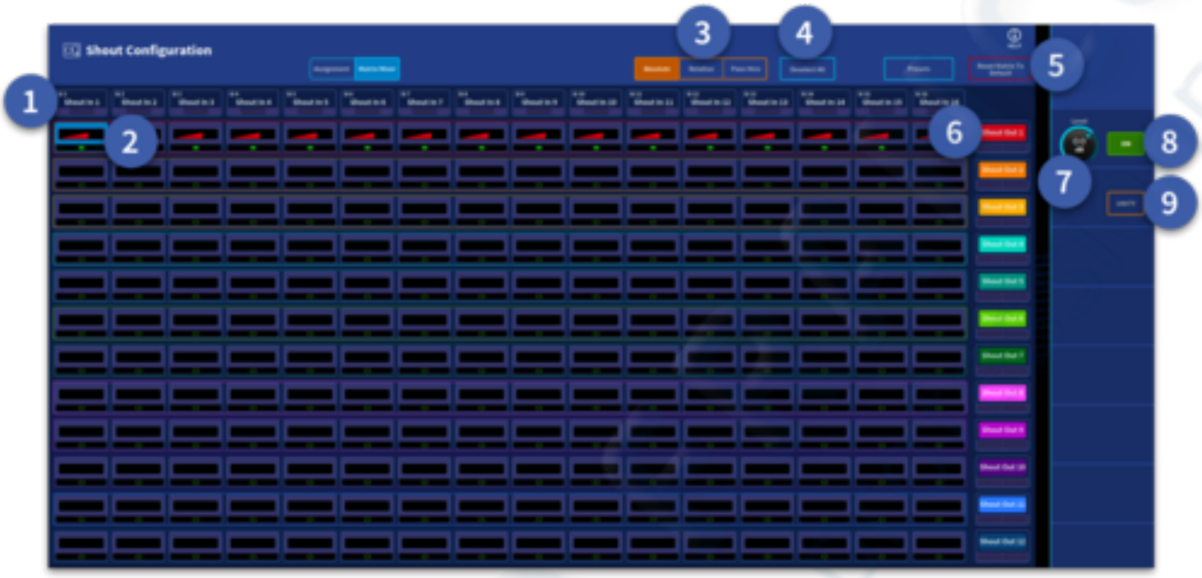

1. Oscillator Internal - Connects the signal generator output to the internal Talk and talk select busses. The internal talk bus can then be mixed onto any of the HD96-24's busses by pressing the talk button in the Aux widget.
2. Oscillator External - connect to the external talk bus to send to a patched XLR output.
3. Level control- Gives continuous adjustment of signal generator peak output signals from off (4) to +10 dB.
4. Frequency (Freq) - Gives continuous adjustment of the sinusoidal oscillator frequency from 50 Hz to 5 kHz.
5. 1 kHz switch - Overrides the swept frequency control (item 4) and provides a fixed 1 kHz tone.
6. Pink switch – Overrides the sine wave oscillator and converts the output signal to pink noise.

## 13.7 Shout Mixer Configuration

The Shout Mixer Configuration page, found in the side bar menu, allows complex communication groups to be created and stored. 16 shout inputs (independent of the input and output channel count) can be used to create complicated talk systems with 12 dedicated outputs to allow different mixes as required.

# 13.8 Matrix Mixer

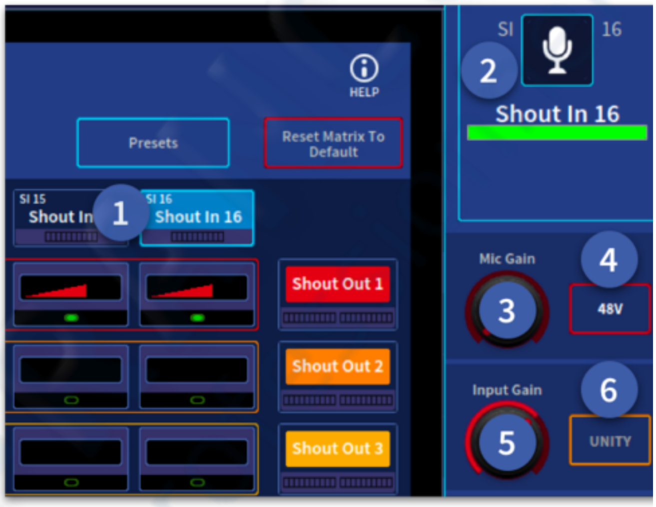
The Matrix Mixer allow routing of the 16 shout inputs signals to any of the 12 Shout out paths. The Shout outputs and inputs are available in the patching page. The 16 shout ins and 12 shout outs can be named, colored, given icons and tagged in the same manner as other channels by clicking in the naming field in the side bar display.




1. Shout in 1-16 – Shout Ins can have any source patched to them including directly from any microphone pre-amp currently configured in the system (saving input channels) or from any output bus. The Shout ins can then be sent to a shout output. Long pressing a Shout In button selects it in all shout out busses (vertical selection).
2. Shout level and On Display – Shows the level (Sloping shape) and On/Off status (green dot) to the assigned shout out bus.
3. Absolute/Relative/Pass-thru - Absolute changes the value to the current selected value (lowest value dB becomes the starting point). Relative keeps the value differences intact, once the maximum value is reached levels will become absolute. Pass-thru only affects values when the value reaches the same value as the lowest selected value while the control is increased.
4. Deselect All – Clears all current selections.
5. Reset Matrix to Default – Sets the matrix back to zero. Use with caution!
6. Shout Out 1- 12 – Shout outs can be patched in the patching page. Long pressing a Shout Out button selects all shout in paths for quick level control (horizontal selection).
7. Level – Adjusts the levels within the Matrix Mixer in by Absolute, Relative or Pass-thru mode. Maximum level into a shout out is 6 dB per shout in.
8. On – Turns selected shout sends on in the selected shout out path.
9. Unity – Sets the selected shout in sends to 0 dB with a long press.

# 13.9 Shout Inputs

A shout input can be used to input a pre-amp or any output. The controls functions are listed below.



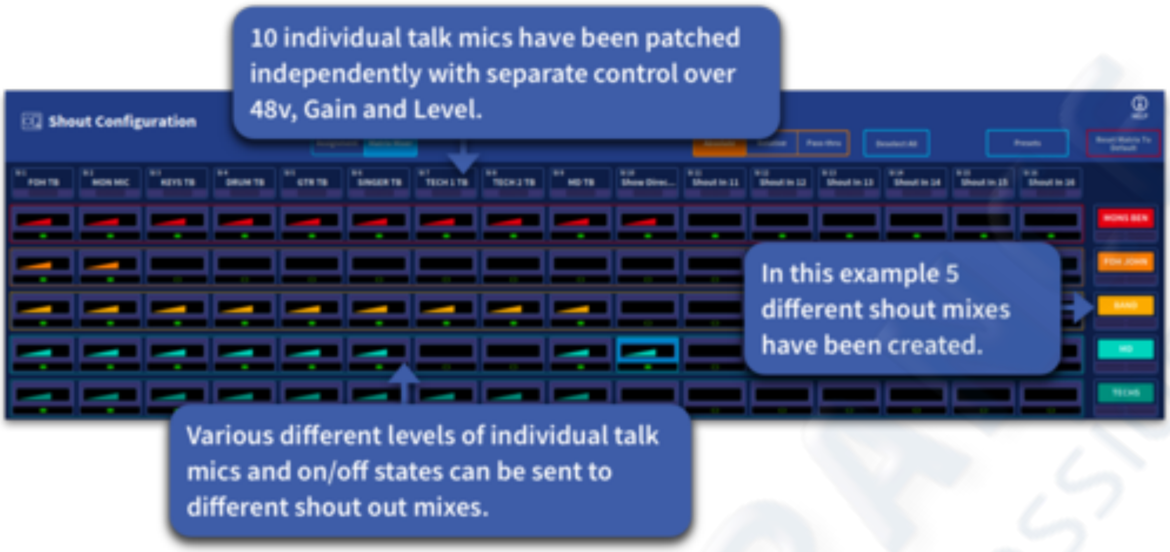

1. Shout In – When a shout in is selected, parameters can be accessed in the side bar. Metering of the input is displayed under the name.
2. Shout In Name – Here a Shout Input can be named, given an icon, change the colour and add tags in the same way as a regular channel.
3. Mic Gain – When patched to a microphone pre-amp the Mic Gain becomes available.
4. 48v – Add 48v Phantom power to the shout input when patched to a microphone pre-amp.
5. Input Gain - Trim the level of the post gain pre-amp signal from infinity (∞) to 6 dB.
6. Unity – Sets trim to 0 dB with a long press.

# 13.10 Shout Outputs

A shout output is used to send groups of mixed inputs (can be a direct input of a talk mic or an aux output) to various different destinations as required.


1. Shout Out – When a shout out is selected parameters can be accessed in the side bar. Metering of the output is displayed under the name.
2. Shout Out Name – Here a Shout Output can be named, given an icon, change the colour and add tags in the same way as a regular channel.
3. Level – Change the output level of the Shout Out from infinity (∞) to 6 dB (Default 0 dB).
4. Threshold – Adjust the limiter threshold from infinity (∞) to 25 dB.
5. Patch to TB – This sends the selected Shout Out to the Talkback bus. This is useful for monitor engineers who wish to listen to a shout mix in the solo bus. By default, this button turns Talkback on in Monitor Bus A. This can be changed to Monitor Bus B in the Monitor Configuration page if required.
6. Limiter – Activate the limiter on the shout out path.

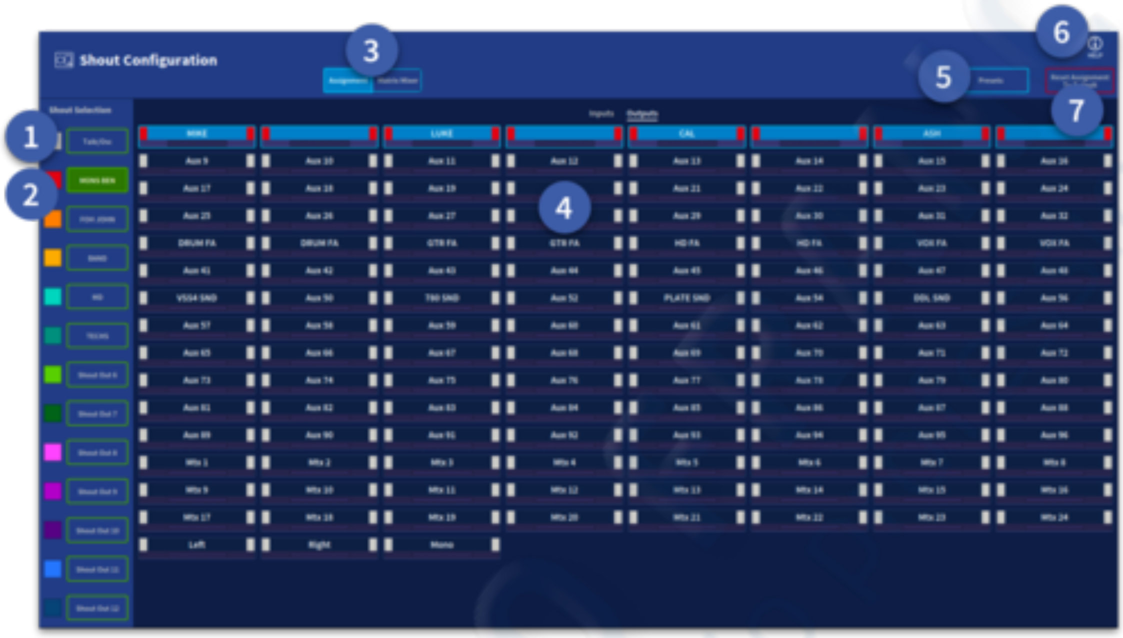
**Tip:** When mixing in-ear monitors (IEMs), talk systems become essential. The following example allows multiple mics to be separately mixed and sent to everyone easily.




- The Talk Mic on the front of the surface could be patched to a Shout Mixer input using the Ext Talk in. This allows you to use the EXT button on the surface to turn your talk mic on and off and gives you instant gain control.
- Patch all your band members, engineers, MDs and technicians talk mics directly into the shout inputs 1-16 (10 microphones have been patched in the example below).
- Set mic gain and add 48v as required.
- Give separate shout out paths to people who require different mixes (See example below with 5 different groups of people).
- Turn on and mix different levels to each shout mixer path recipients' preference.
- Assign the shout out paths to all the different auxes feeding IEMs (Assignment Page). In the example below band shout out would be assigned to all the band IEMs, Techs shout out to Tech IEMs etc.
- To hear the shout out 1 in monitor bus A, press the Patch to TB button.
- If you are using Monitor A bus for IEMs and Monitor B bus for a listening wedge or speaker for example, turn Talkback on for the Monitor A bus only (default if Patch TB button is used). This allows the Talkback Ducker to be used so that when people talk it dips the currently soloed mix allowing any instructions from artists or crew to be heard clearly.

# 13.11 Assignment Page

In the diagram below the 4 stereo in ear monitor outputs are assigned to Shout Out 1 which allows the monitor engineer to hear the output of that shout group in the monitor bus (when Shout out 1 is patched to PFL in).



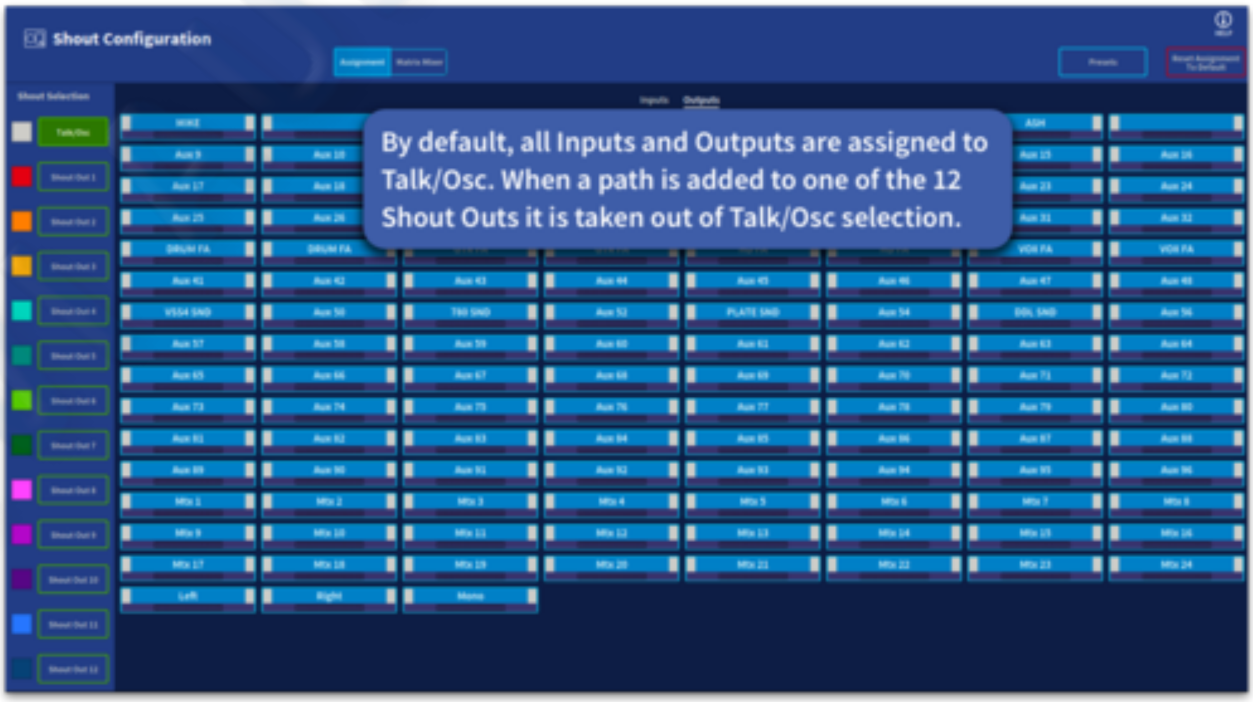

1. Talk/Osc – By default all inputs and outputs are assigned to the Talk/Osc bus.
2. Shout Out 1-12 – Any combination of inputs and outputs can be assigned to one of the 12 shout out paths.
3. Assignment/Matrix Mixer – Changes the view between Assignment Page and the Matrix Mixer.
4. Inputs/Outputs – All inputs or outputs can be seen at once via the Inputs/Output selection button.
5. Presets – Opens the preset manager window (see following section).
6. Help – Opens the helpful information window shown below.
7. Reset Assignment to Defaults – This resets the Assignment page to all inputs and outputs assigned to the Talk/Osc if required.

# 13.12 Presets

Presets allow different configurations of the shout mixer to be instantly loaded. For example, having a preset for sound check and one for the show. Presets can be assigned to global assignable keys for instant changes of shout mixer set-up.


1. Presets – Press to open the presets manager window.
2. Delete – Deletes the currently selected preset.
3. Save – Saves the current Shout Configuration as a new preset (a keyboard pops up to allow naming. Press return on the keyboard after typing a name to store the preset).
4. Rename – Rename the currently selected preset.
5. Load – Load the currently selected preset.
6. Stored Preset – Presets are stored in this list. When a preset is selected it is highlighted with a blue edge.

The Reset Assignments to Defaults button resets the shout mixer as shown below:



# Chapter 14: Graphic Equaliser( GEQ)

## 14.1 Overview of the GEQs

The HD96-24 features a stereo GEQ which is modelled on the legendary KLARK TEKNIK DN370 Graphic EQ which can be added to any path via the Effects Rack. Two stereo GEQs per effects slot can be used. The GEQ window shows a screen-width version of the selected GEQs front panel. This gives you full control of the GEQ via the GUI.

Each GEQ is a dual channel, 31-band, third octave graphic equaliser and features switched 2nd order hi-pass and low-pass filters with two notch filters with variable frequency ranges. The GEQ is primarily a mono process, but in the case of stereo paths, a stereo GEQ is controlled from a single set of controls with the ability to unlink the left and right sides if required.

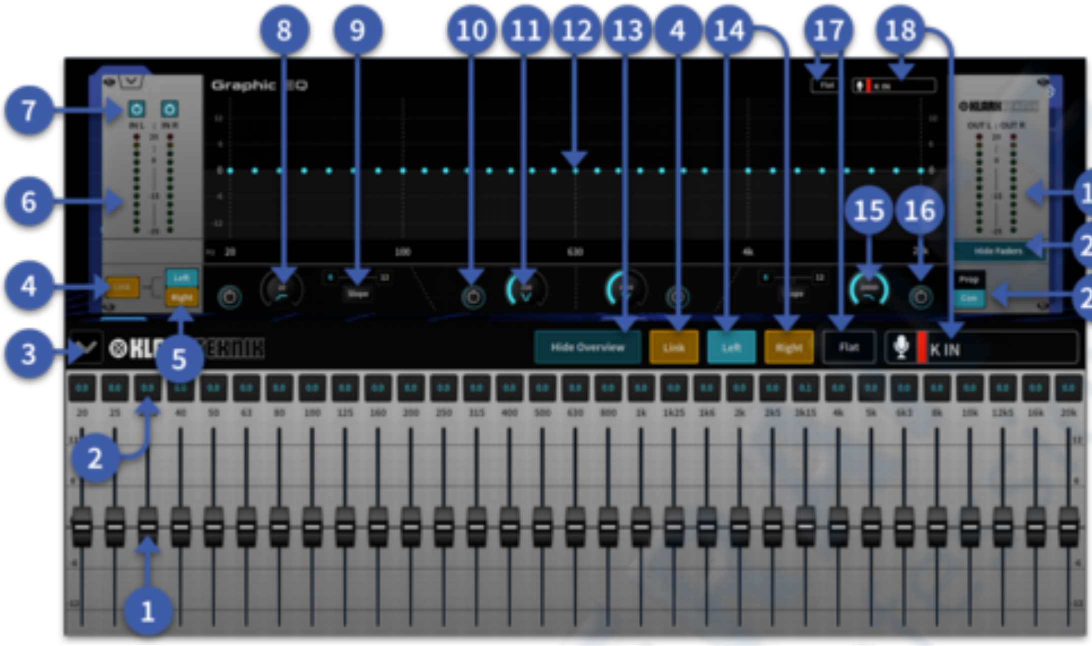
## 14.2 Features

The GEQ consists of a graphic EQ (faders) section and a filters section. Thirty-one faders provide fine adjustment of each frequency band with a range of -12 dB to +12 dB. The 31 frequency bands are spaced 1/3 octave apart on the standard ISO 266 frequency centres. All the functions of the GEQ can be bypassed via the GUI power switches, such that the output will be the same as the input.

The GEQ has one high pass filter, one low pass filter and two variable frequency notch filters. Each filter is adjusted via the controls in the GUI screen.

Each fader can be selected by touching the fader cap and works in the same way as GUI faders, i.e. you can drag up and down. Once a fader is selected, dragging your finger outwards allows finer control of level the further you drag your finger out.

## 14.3 Controls




1. Fader (31) – Adjusts the signal by +-12 dB. Touch fader cap to select.
2. Band Gain Display – Displays the amount of level change.
3. Hide Faders – Close the fader window.
4. Link – The option to link left and right GEQ controls.
5. Left/Right selection – In mono mode select which side is displayed.
6. 12-segment input meter - Shows the incoming signal level.
7. GEQ Power Buttons – Buttons are linked together in link mode or can be used indvidually if GEQ is in dual mono mode.
8. High Pass Filter - Adjusts the cut off frequency, which is continuously variable from 20 Hz to 500 Hz.
9. Slope Button - Switches the high or low pass filter between 6dB and 12dB.
10. Power Button - Switches the respective notch filter in/out.
11. Notch Filter Adjusts the position of each notch filter within the range 20 Hz to 2 kHz or 200 Hz to 20 kHz.
12. GEQ Display – Shows each bands level in a graphical format when the faders are hidden.
13. Hide Overview – Removes the upper overview of the GEQ leaving the faders on view.
14. Left/Right – In dual mono mode (un-linked) lets you select the left or right side of the GEQ.
15. Low Pass Filter - Adjusts the cut off frequency, which is continuously variable from 2 kHz to 20 kHz.
16. Power Button - Switches the respective high pass/low pass in/out.
17. Flat – Resets all 31 bands of the GEQ to 0 dB.
18. Channel – Displays the path the GEQ is inserted on.
19. 12-Segment LED Output meter. Shows the outgoing signal level and is post-EQ.
20. Hide Faders – Removes GUI GEQ faders leaving the overview above.
21. Prop/Con Selects proportional Q (Prop) or constant Q (Con) modes.

## 14.4 Proportional-Q Vs Constant-Q Response

KLARK TEKNIK's enhanced Proportional-Q (Prop) equalisation offers key advantages over standard Constant-Q (Con) graphic equalisers. The Con response boosts or cuts an increasingly wide band of frequencies, resulting in more of the frequency spectrum being lost when using the Con equaliser to eliminate problem frequencies.

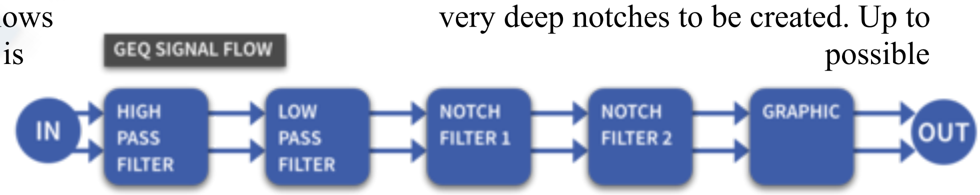
In contrast, with a Prop response, at low amounts of cut or boost the width of the filter is relatively broad allowing for gentle contouring of the frequency spectrum, but becomes progressively narrower as the amount of boost or cut is increased, giving a more 'focused' response ensuring that problem frequencies can be attenuated quickly and effectively. At the same time, the enhanced Prop equalisation response used on the GEQ minimises interaction between adjacent frequency bands, allowing subtle tonal correction without frequency response ripple, so that more of the musical content is preserved.

**Note:** At high gain settings, a proportional-Q equaliser "automatically" increases Q for more dramatic problem solving such as suppression of feedback or unwanted resonances.

## 14.5 GEQ Filters

The 12 dB/oct high-pass filter features a 20 Hz to 500 Hz range and is invaluable for the smooth rejection of unwanted low and subsonic frequencies, particularly relevant for use with modern compact wedge monitors. The 12 dB/oct low-pass filter features a 2 kHz to 20 kHz range and can be used to improve intelligibility by tailoring the upper frequency response to match that of the monitor speaker.

The two 17 dB notch filters feature overlapping frequency ranges of 20 Hz to 2 kHz and 200 Hz to 20 kHz. The notch filters allow the surgical removal of resonances and feedback with minimal effect on the rest of the program material, and fast control of 'between fader' frequencies. The ability to overlap the notch filters, both with each other and with the graphic equaliser bands, allows very deep notches to be created. Up to 46 dB of attenuation is possible when using the notch filters in conjunction with the graphic equaliser bands.




## 14.6 GEQ Patching

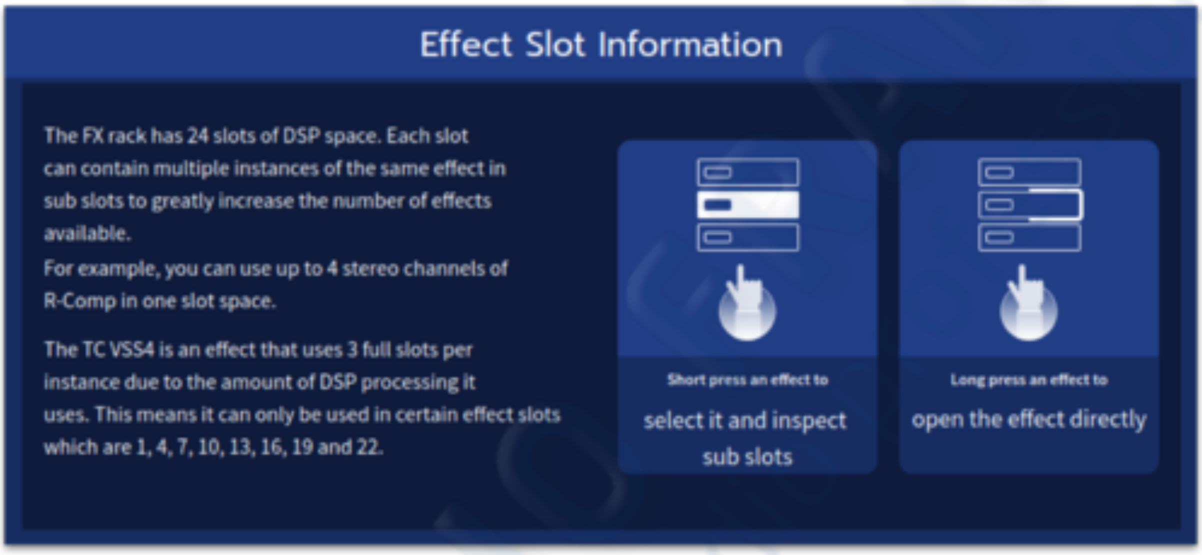
There are serveral ways to patch a GEQ:

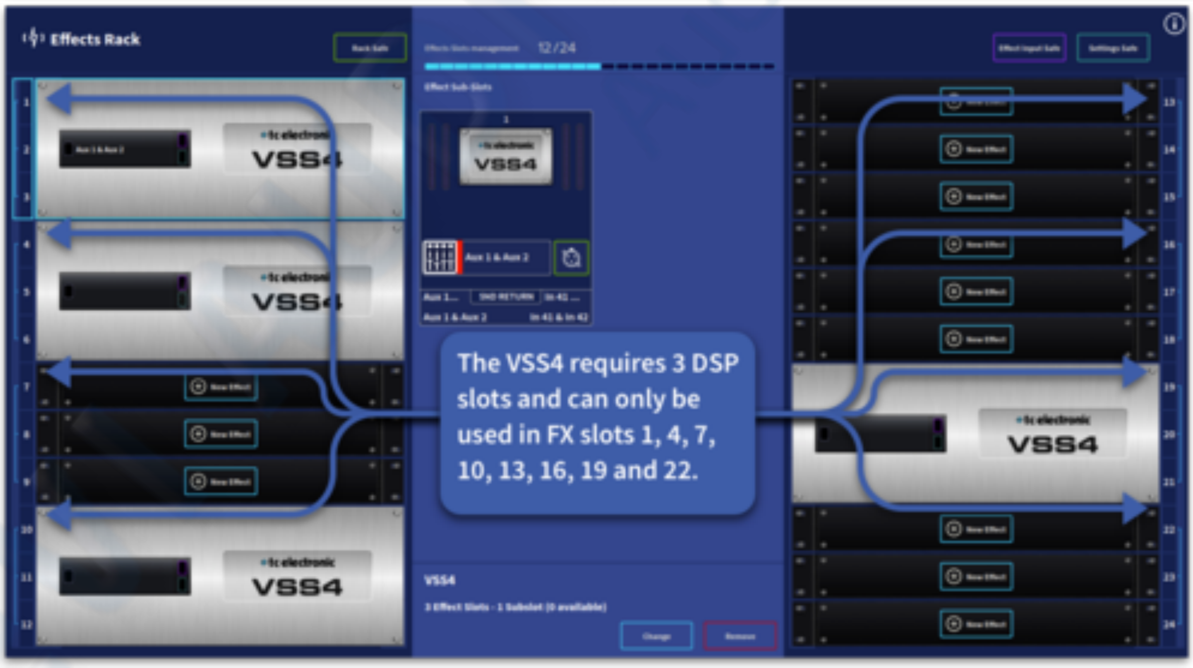
- Insert a GEQ on a path by navigation to either the Home, FOH or Channel View workflows. Select the path you wish the GEQ to be inserted on and add the GEQ in the effects insert page.
- Add a GEQ in the FX Rack page which guides you through the patching options. The GEQ will now also be available In the Patching Workflow to patch as desired.

# Chapter 15: Internal Effects

## 15.1 Effects overview

Effects can be assigned in two main ways. Either by adding in a channel or in the Effects rack. They can be used as an insert on any path or patched via an Aux send and returned to a channel of your choice. For example, this can be used to send several paths to the same reverb. Up to 24 DSP slots are available and effects are assigned dependant on their DSP value. For example, many effects can have many instances in an Effect slot (sub-slots) or take up 3 entire slots (VSS4) if they require the processing power.



The Effect Rack can be found in the side bar menu and gives an overview of all the effect within the system. It can be added to the top bar menu by dragging it from the Customise Toolbar menu.

### About the effect window

The effect window displays the selected effect, which gives you full control of the effect via the GUI touch controls with pop up rotary control. There is also the ability to use the side Assignable Controls which let you select and operate the controls of the effect as desired.

## 15.2 Working with channel inserted effects

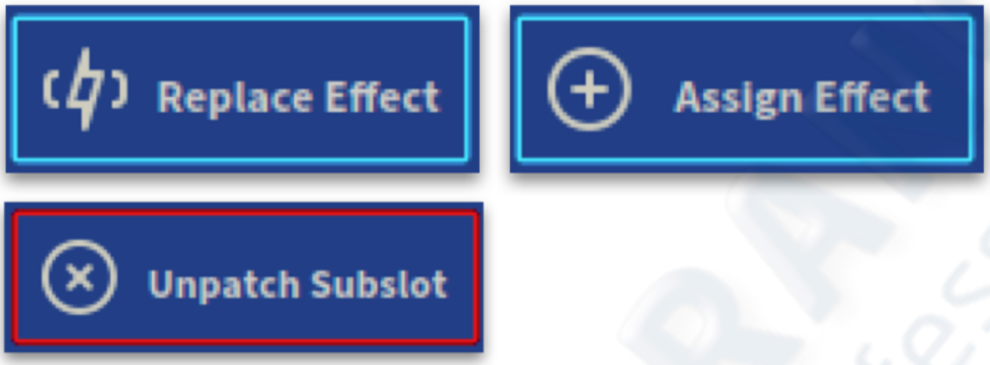
Each channel has 3 slots for effects to be inserted into (more than 3 inserts can be chained together using the patching page if required).

To add an effect as an insert navigate to the Effects Insert page within the Home, Channel View, Automation or FOH widgets.


1. Press the Add button in the slot you wish the effect to be inserted on. The effect browser window will open (described in the next section).
2. The slot will then display the chosen effect (after closing the Effects Browser), press the image to edit effect parameters.
3. Check the individual effect is turned on to use the inserted effect.
4. Check the channel INSERT is also turned on in order to use all the inserted effect.
5. Press the Edit button to open the effect browser to edit settings.
6. The available Effects Slots within the system are displayed at the bottom of the window. The remaining slots turn red when only 4 or less slots are available.
7. Effect rack slot position number and sub slot location.

## 15.3 Effects Browser

The Effects Browser allows you to see an overall view of the effects and how the DSP has been allocated. It allows you to insert an effect on the current channel, replace an effect or unpatch from sub slots.



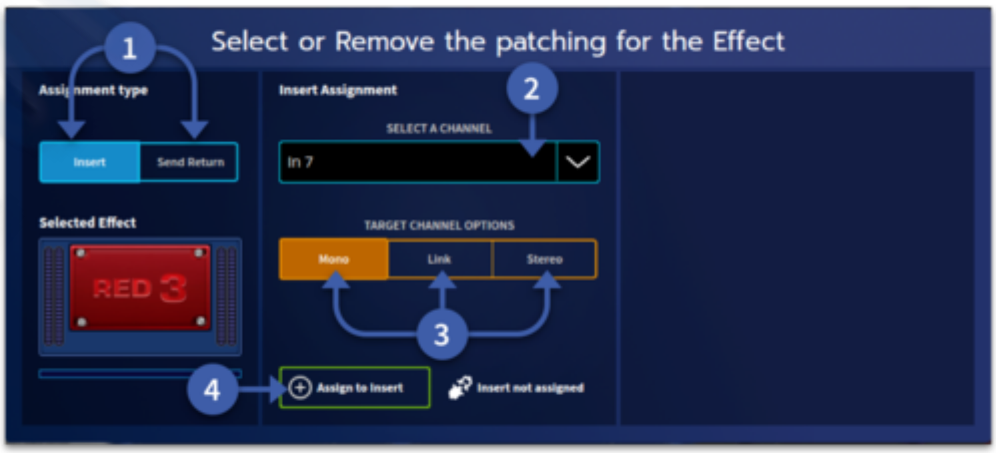

The blue boxes display how many effects can be used per DSP slot. In the example below 4 x R-Comps can be seen in slot 1 and an instance of VSS4 is filling slot 4,5 and 6.

# 15.4 Effect Rack

The Effect Rack has 24 slots of DSP space. Each slot can contain multiple instances of the same effect in sub slots to greatly increase the number of effects available. For example, you can use up to 4 stereo channels of R-Comp in one slot space. Short press an effect to select it and to see if sub-slots are available. Long press an effect to open the effect directly. The effect in the first sub-slot will open.The Effects rack management bar shows how many slots you have used and can be found in the Effect rack and also the effects page in the Channel View, FOH and Automation workflows. The slot management display turns from blue to red as DSP runs out.


To add an effect:

1. Press the + New Effect button.
2. Select the effect from the icons list using the tabs to navigate effect types.
3. Press Add Effect to confirm effect selection, the effect patching will appear.




To patch an effect:

1. Select the Assignment type, Either as an Insert or Send & Return effect.
2. Select the channel you wish to insert or use as the send.
3. Set the target as Mono, Link (two mono channels) or a Stereo channel.
4. Press and hold to confirm the patch (once patch is confirmed it can be removed).
5. If using as a Send & Return effect, choose the channel to use as an Aux Return.
6. Set the target return as Mono, Link (two mono channels) or a Stereo channel.
7. Press and hold to confirm the return patch. To remove a patch press and hold Remove.

# 15.5 Effect Sub-slots

Each slot in the effects rack has the ability to host up to 4 stereo effects if there is sufficient DSP available depending on effect type chosen. For example, 4 stereo R-Comps can be used in 1 effects slot space.

To indicate the status and of each sub-slot the following icons are used:


1. Effect is on. The white dot shows currently selected on the path in view.
2. Effect is assigned but not active.
3. Effect is on.
4. No effect assigned.
5. Effect slot position number.

# 15.6 Effects Rack Safes

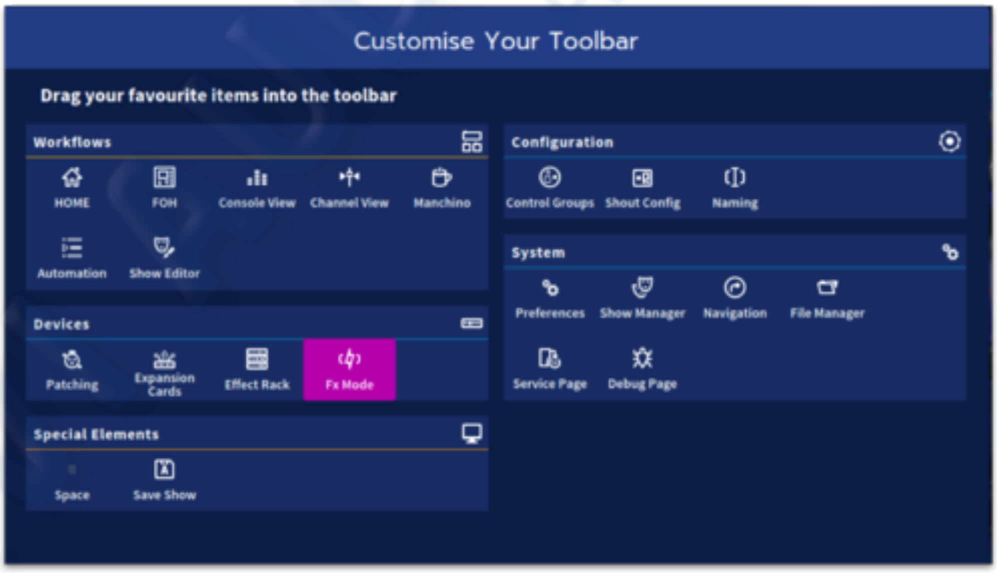
The Effects rack can be made "Safe" from certain parameter recall when using automation. Below is a description of how each function works:


1. Rack Safe – This "safes" the whole Effects rack from automation and stops any changes to effect types. Long press to engage. Press and hold to activate.
2. Effect Input Safe – This "safes" all input patching to effects. Effect returns need to be placed in safe mode independently via the channel config page. Press and hold, then select the effect(s) you wish to add to Patching Safe or select an effect then press Patching Safe.
3. Settings Safe – This "safes" any parameter changes to an effect if the same effect is used throughout all scenes. Note that if an effect is changed in a slot with settings safe engaged, i.e. a delay to a reverb, the settings will change. Press and hold, then select the effect(s) you wish to add to Settings Safe or select an effect then press Settings Safe.
4. Indicates a slot is in Effects Input Safe mode.
5. Indicates a slot is in Settings Safe mode.

## 15.7 FX Mode

FX Mode allows any effect to be viewed when a channel is selected. This works for inserted effects or channels with an effect return patched. This aids in fast navigation to FX when you are in the mix and can be quickly toggled on and off.
FX mode can be activated in two ways:
In the Effect Rack, press the purple FX Mode button.


FX Mode can also be added to the top bar menu by adding the icon from the custom toolbar menu.




When FX Mode is active, 4 boxes can be seen (Return, 1, 2 and 3). If you have selected a channel with FX inserted, you can choose the insert slot which selects the FX that will be opened when you return to that channel.


With FX Mode is active, if a channel is in Aux Return Mode, when you select the channel again the FX patch to that channel will open. This is great for your reverb and delay returns. **Note:** If the FX return channel also has an insert, you can select if you wish the return or the insert to be opened on channel selection.

# 15.8 Effect Programs

Some types of effect have associated factory presets and user-configurable programs, which you can load within the effect (these are also stored in a show file). You can also save all of the controls from one or more effects in a user preset, which will then contain information about their settings, including the loaded factory preset or user configurable program.

For details of each effect type, refer to its section in this chapter. For information on presets, see Chapter 23 “User Libraries (Presets)”.

# 8.1 Reverbs

## 8.1.1 DN780 Reverb

The Virtual DN780 Reverb provides emulation of the vintage KLARK TEKNIK DN780 Digital Effect unit. The DN780 is not just a reverberation device, it also gives the user a unique and flexible means of producing realistic acoustic simulations for environments of all types and sizes. The provision of effects programs further extends this versatility, making it a very powerful acoustic processing package. The parameter controls give accurate adjustment of all reverberation parameters and allow the engineer to create unique acoustic environments of virtually any type.


1. **IN level meter** - Input meters, which comprise a dual-column peak reading LED meters, ranging from 0dB to -30 dB in 3 dB steps.
2. **Level control** - Input level. Range is from -4 to +6 dB, with 0 dB at top dead centre. This should be set to illuminate the -3 dB LED on the input headroom indicator during loud program passages.
3. **Pre Delay control** - Controls the amount of delay (in milliseconds) between the initial signal and the onset of reverberation. On certain program types, pre-delay is inserted between early reflections and reverb to improve authenticity. Its range is algorithm dependent. Low level, phase-dependent clicks are produced when pre-delay is altered during the program.
4. **Room size control** - Adjusts the average dimension of the simulated space. Range is from 8 to 90 metres. A momentary mute is implemented when this control is adjusted.

5. **Decay control** - Control that sets the overall (mid-band) reverberation decay time. Range is from 0.1 to 18 seconds, depending on room size.
6. **Pattern control** - Controls the 'density' of early reflections. Selects the number and spacing of Early Reflections/ADT/Multi-tap delays. Range is from 1 to 9.
7. **Level control** - Control that acts as a 'depth' control by altering the apparent distance between the sound source and the listener. Alternatively, adjusts the input level for Sound-On-Sound/Infinite Room. Range is from 0 to 9.
8. **Low Freq (low frequency) control** - Adjusts the decay time at the low end of the reverb spectrum. Range is from -7 to +7.
9. **High Freq (high frequency) control** - Adjusts the decay time at the high end of the reverb spectrum, which sets the absorption characteristic of the simulated space. Range is from -7 to +7.
10. **MIX control** - Controls the Dry/Wet output mix and ranges from 0% to 100%, respectively.
11. **OUT Level meter** - The output meters, which comprise a dual-column peak reading LED meters, ranging from 0dB to -30 dB in 3 dB steps
12. **Preset Buttons** - These algorithms emulate the ones on the original DN780. Use the buttons to select the type of effect you desire.
13. **Input button** - Removes feed to the reverberation section, enabling the decay qualities of the chosen setting to be confirmed.
14. **Reverb button** - Reverb Mute, a rapid means of muting the reverb algorithms.
15. **Audition** - A rim shot sample is triggered to test reverb characteristics.
16. **Stereo input button** - Enhances original algorithm to provide stereo input.
17. **Bypass** - turn the reverb off.

### Pre-delay

0 to 990 milliseconds (ms) of pre-delay is available allowing a very wide range of control. Delays of less than 30ms closely integrate the direct and reverberant sounds; often a desirable feature on percussive sounds. Delays of 50ms or more cause the direct and reverberant sounds to separate and convey a feeling of depth and distance to the simulated environment.

### Pattern

The pattern control alters the 'density' of the early reflections. It is adjustable from 0 to 9 as shown on the display, with 0 giving a low density or 'grainy' character to the early reflections and 9 producing a high-density effect.

Level

The level control functions convincingly as a 'depth' control, altering the apparent distance between the sound source and the listener. It is adjustable from 0 to 9 as shown on the display, with 0 being relatively distant and 9 bringing the sound source closer.

### Decay

The reverberation decay time is adjustable from 0.1 to 18 seconds, depending on room size, changing the reverberant field from a virtually dead sound to a totally surreal effect. Short decay times, under one second, are essential for authentic small room simulation and also extremely useful for ambience applications where classic reverberation is not wanted. Reverb times of 1 to 4 seconds cover the majority of normal applications where classic reverberation is required. Longer decay times are available for special effect applications.

### LF key

LF is adjustable to ±7, depending on room size and decay time, as shown in the GUI. An increase in LF decay time is generally desirable on simulations of large halls, since low frequency sounds suffer less than higher frequencies from absorption in air. Very small spaces usually need the 'thin' sound created by reducing LF decay.

### HF key

HF is adjustable to ±7. The HF decay control sets the absorption characteristic of the simulated space. In reality, large environments feature considerably reduced high frequency decay times due to air absorption. A smaller room will feature greater HF decay time if the walls are tiled and the room is empty than if the room contains soft furnishings and curtains. The wide range of control provided will allow a suitable setting to be chosen to enhance realism in most applications.

### Room size

Room size is adjustable from 8 to 90 linear metres, representing a wide range of room volumes. Since the acoustic character of a given environment depends not only on the reverberation time and construction of the room, but also to a great extent on its volume. The room size control is, in fact, essential if authentic simulation of a range of different sized environments is required. Small room sizes give a confined, 'box-like' sound. Medium room sizes suggest a room or small hall, whereas large room sizes suggest a large hall or cathedral. Again, there is no substitute for experimentation.

### Special effects programs

This section gives details of the effect's programs available on the DN780 Reverb effect.

### Direct Signal

Effects such as ADT and Echo rely on a suitable level of direct (dry) signal being added on the mixing console. Since this is largely a question of taste, no precise instructions are included here. It is recommended that, as a general principle, direct signal is initially set at a normal operation level without any effect present. The effect is then increased in level as required.

### Delay

- PRE-DELAY control adjusts the delay time within the range 0 to 2.0 seconds.
- REV button mutes the effect.

Preset parameters: On selecting this effect program, delay is set to 200 ms.

Stereo mix: The signal's at left and right outputs are both delayed by the same amount as set using the PRE-DELAY control, that is, they are essentially monophonic.

## ADT

- PRE-DELAY control adjusts the delay time before the second voice is heard. Delay is adjustable within the range 0 to 127ms.
- PATTERN control selects the number and spacing of the second voices. Selection is from Pattern 1 (two voices) to Pattern 5 (eight voices).
- REV button mutes the effect.

Preset parameters: On selecting this effect program, delay is set to 40ms and pattern is Pattern 5 (a wide multi-voiced effect).

Stereo mix: Left and right output signal's use different delay taps to achieve a stereo effect. Using only one output halves the number of 'voices', that is, Pattern 1 (one voice) to Pattern 5 (four voices).

### Application Notes:

1. Try delays from 25 to 50ms. Short delays reduce the effect, long delays produce echo.
2. Direct signal must be added at a suitable level on the mixing console. Try 50/50 direct/effect mix on Pattern 1, much less direct on Pattern 5.
3. For conventional ADT, try 'Delay' of 40ms, Pattern 1, and use one output only, panned, say, fully right. Pan direct signal fully left and use a 50/50 direct/effect mix.

## Multi-Tap Echo

- **PRE-DELAY** control adjusts the time delay interval between the direct signal and the first repeat. Delay is adjustable from 0 to 990ms.
- **PATTERN** control selects the number and spacing of the repeats. Pattern 1 (two repeats) to Pattern 9 (eight repeats).
- **DECAY** control sets the feedback (regeneration) level for repeat echoes.
- **HF** control allows the high frequency filtering to be applied to the regenerated signal.
- **REV** button mutes the effect.

**Preset parameters:** On selecting this effect program: delay is set to 196ms; pattern is Pattern 4; decay is 73; and HF is 0. These settings give an effect similar to a typical multi-head tape echo, but with full stereo image.

**Stereo mix:** Different delay taps are used for left and right outputs to achieve a stereo effect. Using only one output halves the number of taps, that is, Pattern 1 (one tap), Pattern 9 (four taps).

**Application Notes:**

1. Set delay time as required, generally fairly short for multi-echoes (higher pattern numbers), and longer for repeat echo. 'Fine tune' delay setting to set exact musical timing for single tap repeat echoes.
2. Direct signal must be added at a suitable level on the mixing console.
3. For single tap repeat echo, start with Pattern 1, with 'Delay', 'HF' and 'Decay' all set at maximum. Reduce parameters as required. Use one output only.

## Sound-On-Sound

- **PRE-DELAY** control sets the 'loop length' and hence the timing of the effect between 0 and 2.0 seconds.
- **LEVEL** control provides 10 level increments of signal input to the 'digital loop'. Return level to '0' after use to avoid noise build-up.
- **DECAY** control sets the 'erasure' of the loop from '0' (100% erasure) to '99' (zero erasure).
- **REV** button clears the memory of unwanted effect.

**Preset parameters:** On selecting this effect program: pre-delay is set to 2.0s; level is 0; and decay is 99. These settings represent maximum loop length with zero erasure. Please note that no sound will be heard until 'level' is increased.

**Stereo mix:** Outputs left, and right are essentially identical. However, to avoid the possibility of slight phase cancellations, it is recommended that only one output is used on this program.

**Application Notes:**

1. Since the 'level' inside the signal processor increases as fresh input is added, input level must be lower than that recommended for normal use; try -15dB on the meters.
2. Correct pre-delay ('loop length') should be set before creating the effect as attempts to alter this later will usually destroy part of the recorded sound.
3. Remember to return level to '0' immediately after use to avoid noise build-up.

## Infinite Room

**LEVEL** control provides 10 level increments of signal input to the 'infinite room'. Return level to '0' after use to avoid noise build-up.

**REV** button clears memory of unwanted effect.

**Preset parameters:** On selecting this effect program, level is 0. Please note that no sound will be heard until 'level' is increased.

**Stereo mix:** Infinite room is a spacious, full stereo effect.

**Application Notes:**

1. Since the 'level' inside the signal processor increases as fresh input is added, input level must be less than that recommended for normal use; try -15dB on the meters.
2. Remember to return level to '0' immediately after used to avoid noise build-up.

## Alive, Non-Linear and Reverse

- **PRE-DELAY** control sets the delay between the direct signal and the onset of the effect. Maximum pre-delay is 990ms.
- **PATTERN** control changes the density of the reflections, with 0 giving a low
- density or 'grainy' character and 9 producing a high-density effect.

- **DECAY** control sets the length of the effect, from '1' (short) to '12' (long). The display simply shows these increment numbers and is not calibrated in seconds.
- **LF** control adjusts the low frequency content of the effect.
- **HF** control adjusts the high frequency content of the effect.
- **REV** button clears memory of unwanted effect.

**Preset parameters:**

1. **Alive -** on selecting this effect program: pre-delay is 0s; pattern is 9; decay is 8.3; LF is +0; and HF is +2.
2. **Non-Linear -** on selecting this effect program: pre-delay is 0s; pattern is 4; decay is 5.4 LF is 0; and HF is 0.
3. **Reverse -** on selecting this effect program: pre-delay is 0s; pattern is 4; decay is 12; LF is 0; and HF is +2.

**Stereo mix:** All these effects are in full stereo and are completely mono-compatible.

**Application Notes:**

1. These three effects will find instant application in any recording studio engaged in contemporary music production, as they allow pronounced acoustical enhancement without the 'muddying' effect of longer, conventional decay envelopes. This makes possible a bright and 'punchy' mix. These effects work well on most instruments but try "Non-Linear" for explosive snare sounds and "Reverse" on vocals.
2. The Alive program produces a more natural, live ambience, which is less coloured than the other two effects and has wide-ranging applications.

**Note:** The DN780 Alive preset was used to great effect on the snare drum sound for David Bowie & Mick Jagger version of "Dancing in The Street".

## 8.1.2 TC Electronic VSS4 Reverb

VSS4 is a genuine Stereo Reverb that radically departs from blurry sustain concepts of the past. Based on source-related reflections from multiple angles, the precision of VSS4 is comparable to real-world mono or stereo sources positioned in an authentic space.

VSS4 is your first choice for spot and close microphone pick-ups as well as synthetic sources. It is not only capable of adding spaciousness, but also distinctive character, localization and depth to the source. If the reflection patterns are too immense for a particular situation, you can tailor it to precisely complement the material processed.


1. **Main** button - Main settings page of the VSS4 reverb.
2. **Set-up** - Set-up page for pan and levels.
3. **Colour** - Colour page to adjust crossover and filters.
4. **Gloss** - Gloss page to adjust diffusion and modulation.
5. **Presets** - Presets page recall, select desired preset and press load.
6. **Master Decay** (Master Reverb) - Range: 0.1 to 20 sec Adjusts the Master Reverb Decay time. Decay time can be further adjusted using the multiplier for the following bands: Lo, LoMid, HiMid and Hi.
7. **Pre Delay** (Master Reverb) - Range: 0 to 300ms, Pre-Delay on the complete wet signal including Early Reflections.
8. **Hi Cut** (Master Reverb) - Determines the Hi Cut frequency for the Diffuse Field part of the Reverb.
9. **Reverb Delay** (Master Reverb) - Determines the delay before the reverb effect begins.
10. **Reverb Size** (Master Reverb) - Adjusts the overall perceived size of the Diffuse Field part of the algorithm. Generally, you would attempt to define the perceived room size by altering the Decay time. However, with the Size parameter you can achieve a perceived alteration of the room size thus keeping the Decay time. The default value is meant as the size that the specific Location Type is intended to have.
11. **Reverb Width** (Master Reverb) - Adjusts the stereo width of the reverb.
12. **Lo Colour** (Master early) - Lo Colour adjustment of the Early Reflections.
13. **Hi Colour** (Master early) - Hi Colour adjustment of the Early Reflections.

14. **Early Start** (Master early) - Adjusting the Early Start time is an efficient way of getting rid of the first reflections that normally colour the source the most. By adjusting the Start time, the first reflections are discarded but the timing of the later reflections remain unchanged. Therefore, this adjustment is typically more acoustically precise and useful than a normal Pre-delay control.
15. **Early Stop** (Master early) - The Early Stop parameter can reduce the later reflections in the Early Reflections pattern. When using large location types this can be a useful parameter to smooth the overall perception of the room.
16. **Input Meters** - Two rows of 28 LEDs, which show input metering.
17. **Output Meters** - Two rows of 28 LEDs, which show output metering.
18. **Preset Display** - Displays the currently selected preset.
19. **Reverb Level** - During perceptual experiments, we have found that reducing particular orders of reflections can be useful for optimizing the room response to a given microphone placement. The 0% setting will enable the full response while 100% is full reduction.
20. **Reverb Mute -** Reverb can also be muted.
21. **Early Level** - Adjusts the level of the Early Reflections from the two sources.
22. **Early Level Mute** - Can also be muted.
23. **Dry Level** - Attenuates the Dry signal level on the Output side thus leaving the Reverb and Early Level unaffected. Off equals a “kill-dry” setting.
24. **ER Decrease -** Reflection patterns are rendered to a high order. 1st order reflections have hit one surface before arriving at the listener, 2nd order reflections two surfaces etc. High order reflections are often more diffuse than low order ones. In VSS4 this effect is emulated by assigning individual diffusion characteristics to each reflection.


25. **In Level** (setup) - Controls input level ∞ dB to 0 dB.
26. **Out Level** (setup) - Controls output level ∞ dB to 0 dB.
27. **Location** (setup) - Choose the type of reverb location.
28. **Source 1** (setup) - Pan position of source 1. Adjust to change the width of the reverb in the stereo field display in the pan position window.
29. **Source 2** (setup) - Pan position of source 2. Adjust to change the width of the reverb in the stereo field display in the pan position window.


30. **Lo Cut** (Colour) - Range: 20Hz to 200Hz Determines the Lo Cut frequency.
31. **Lo Damp** (Colour) - Range: 0 to -18dB Attenuation of the frequencies below the selected frequency via the Lo Cut parameter.
32. **Lo Xover** (Colour) - Sets the Cross-over frequency between the Lo and LoMid Decay. 20 Hz - 500 Hz.
33. **Mid Xover** (Colour) - Sets the Cross-over frequency between the LoMid and HiMid Decay. 200 Hz - 2 kHz.
34. **Hi Xover** (Colour) - Sets the Cross-over frequency between the HiMid and Hi Decay. 500 Hz - 20 kHz.
35. **Hi Cut** (Colour) - Range: 20 to 20kHz. Attenuates the high-end frequencies.
36. **Hi Soften** (Colour) - Hi Soften is a special filter used to "soften" the high frequencies of Reverb diffuse field. This is not a simple Hi Cut filter but a complex set of filters working together to remove the frequencies that give a "brittle" or “harsh” sounding Reverb. Hi Soften is scaled/linked to the Hi Cut and Hi Decay parameters.
37. **Lo Decay** (Colour) - Range: 0.01 to 2.5. Decay multiplier in relation to the Master Decay, for the frequencies below the Lo Xover setting.
38. **Lo Mid Decay** (Colour) - Decay multiplier in relation to the Master Decay, for the frequencies above the Lo Xover and below the Mid Xover settings.
39. **Hi Mid Decay** (Colour) - Decay multiplier in relation to the Master Decay, for the frequencies above the Mid Xover and below the Hi Xover settings.
40. **Hi Decay** (Colour) - Decay multiplier in relation to the Master Decay, for the frequencies above the Hi Xover setting.


41. **Reverb Type** (Gloss) - Choose between Coloured or Normal.
42. **Diffusion** (Gloss) - Range: +/-25. Adjusts the amount of diffusion applied.
43. diffuse fields.
44. **Mod Rate** (Gloss) - Range: +/-50. Adjusts the Rate of the selected modulation.
45. **Mod Depth** (Gloss) - Range: 0 - 200%. Adjusts the Depth of the selected modulation.


46. **Preset Selection** (Presets) - Up to 10 presets within each of the 5 banks.
47. **Load** (Presets) - Load selected preset.

## 8.1.3 TC Electronic VSS3 Reverb

The TC VSS3 reverb is a simplified version of the VSS4 reverb. It still has the great TC sound but is less DSP intensive and only takes 1 slot space instead of the 3 that the VSS4 takes. The VSS3 enables you to add the softest and cleanest ambience to your work that you have ever heard. The wide range of presets will give you the exact reverb you are looking for.

**Constructing a Reverb Preset**

The relationship of Early Reflections and the Reverb tail is very important in this algorithm. Adjusting the balance between the Early Lev and the Rev Lev parameters is one of the easier ways to make a HUGE difference in the sound of your reverb! When you start building your preset you should try this:

- First turn the Rev Lev all the way down. In a send/return configuration, push up the return level. You should now hear Early Reflections and no Reverb Tail.
- Then begin changing the Early parameters until you select a room shape that compliments the program material.
- Re-adjust the wet/dry balance until it is pleasing, then bring up the Rev Lev until the tail of the reverb becomes audible. Add just enough tail to make it work together.
- Adjust the Decay time accordingly.
- On some presets you may choose to have very little Early Reflections or none at all. Certain “ambience” style presets might have little or no “tail”. This is up to you.

The VSS3 was designed to have the smoothest Reverb tail ever developed, but it is the Early Reflections that define the “personality” of the room, so try and experiment with this relationship! By using these parameters correctly, you can create a BIG sound without having a mix swimming in reverb wash.

Early Reflections define the actual feel of the room, where the Reverb tail is the less defined “bowl” of reflections that follows. The major part of the Early Reflection patterns of the VSS3 are simulations of existing rooms and are based on a large number of reflections (40-100), which have been processed through an advanced algorithm.

There are a number of different types and sizes covering a lot of different acoustic spaces that you need for music and postproduction. As the patterns are simulations of real rooms, the delay times of the first reflections are sonic and spatially “connected” to the direct signal. Using Pre Delay together with Early Reflections should therefore be considered very carefully, as the acoustic space created by the pattern tends to “collapse” if too much Pre Delay is added. If you want the well-known “slap back” reverb effect, you should use Rev Delay on the Reverb tail instead and reduce the level of the Early Reflections.


1. **Main** - Selects the Main page for parameter editing.
2. **Presets** - Selects the Presets page for parameter editing.
3. **Reverb type** - Displays the current reverb type.
4. **Preset Display** - Displays the current preset.
5. **Early Control** - Adjust early reflection levels. From ∞ To 0 dB.
6. **Reverb Level** - Adjusts the level of the reverb. From ∞ To 0 dB.
7. **Early Delay** - Adjusts the early delay parameter. From 0 ms to 100 ms.
8. **Reverb Delay** - Adjusts the amount of time before the reverb is heard. From 0 ms to 200 ms.
9. **High Decay** - Applies filtering to the HF portion of the reverb. From 0.10 x to 2.5 x.
10. **Decay** - Adjust the reverb tail length. From 0.10 s to 20 20 s.
11. **Input Metering** - 28 LEDs, which show stereo input metering.
12. **Output Metering** - 28 LEDs, which show stereo output metering.
13. **Power Button** - Bypass the effect with the power button.
14. **Dry Level** - Allows the original dry signal to be mixed in with the wet signal. From ∞ To 0 dB.


15. **Reverb type select** - Select the type of reverb, choose from: Rooms, Halls, Large, Plates, Vocal, Drums and Effects.
16. **Reverb Preset** - Select the reverb preset from the list dependant on which reverb type is selected.
17. **Load** - Load the selected preset.

## 8.1.4 TC Electronic M350

The M350 is an extremely user-friendly reverb processor which is equally suited for live and studio applications. It gives you 11 TC-quality reverb effects. 2 x M350 reverbs can be used in 1 DSP slot.


1. **Effect Select** - 11 types of reverb are available. Ambience, Church, Gated, Hall, Lo-Fi, Mod, Plate, Room, Spring, Tile and Default. Tap to select.
2. **Pre-Delay** - From 0.100 s (min) up to 20 s dependent on which effect type is chosen.
3. **Decay Time** - From 0.100 s (min) up to 20 s dependent on which effect type is chosen.
4. **Colour Filter** - Controls the relative colour of the selected reverb from dark to crisp and bright.
5. **Mix Ratio** - Adjusts the wet/dry balance. Turn fully right for a completely wet signal.
6. **Input meter** - 14 LEDs, which show stereo input metering.
7. **Output meter** - 14 LEDs, which show stereo output metering.
8. **Power** - Bypass the effect.
9. **Close** - Minimises the effect (the same for all FX).

| Type | Decay min – max (in seconds) |
|---|---|
| Ambience | 0.100 – 2.000 |
| Church | 0.100 – 20.000 |
| Gated | 0.100 – 20.000 |
| Hall | 0.100 – 12.000 |
| Lo-Fi | 0.100 – 7.000 |
| Mod | 0.100 – 20.000 |
| Plate | 0.100 – 20.000 |
| Room | 0.100 – 5.000 |
| Spring | 0.100 – 6.000 |
| Tile | 0.100 – 2.000 |
| Default | 0.100 – 6.000 |

## 8.1.5 Ambience Reverb

The ambience reverb adds warmth and depth to source material without adding the obvious artefacts commonly associated with artificial reverbs. It simulates smaller rooms using diffused early reflections with the additional flexibility of separate reverb tail level and decay control. Reflective surface materials and air absorption properties can be simulated by adjusting the high and low frequency cut amount and high frequency damping.


1. **In level meter** - Input meters, which comprise a dual-column peak reading LED meters. Each column consists of 14 coloured LEDs. The red LED illuminates at 3 dB before the clipping point.
2. **Pre Delay** - Controls the amount of delay (in milliseconds) between the initial signal and the onset of reverberation.
3. **Size control** - Adjusts the average dimension of the simulated space. Range is from 0% to 100% of the largest space possible. A momentary mute is implemented when this control is adjusted.
4. **Decay control** - Control that sets the overall (mid-band) reverberation decay time. Range is from 0.5 to 10 seconds, depending on room size.
5. **Audition** - A rim shot sample is triggered to test the reverb's characteristic.
6. **LF Cut** - Control that adjusts the Low Frequency cut from 10 Hz to 500 Hz.
7. **HF Cut** - Control that adjusts the High Frequency cut from 0.2 kHz to 20 kHz.
8. **HF Damp** - Dampens the HF to produce a darker sounding reverb. Settings from 1 kHz to 20 kHz.
9. **Diffusion control** - Reducing diffusion creates more discrete echoes, for vocals and other sustained sounds, reduced diffusion can give smooth reverberation that doesn't overpower the source and maintains clarity.
10. **Modulation control** - To create more variation in reverb's sound, modulation adds subtle changes to the reverb characteristics.
11. **On** - Turns on the reverb effect.
12. **Tail Gain** - Used to reduce the level of the reverb's tail from 0 dB to -inf dB
13. **Mix control** - Controls the dry/wet output mix and ranges from 0% to 100%, respectively.
14. **Out level meter** - Output meters, which comprise a dual-column peak reading LED meters, ranging from 18 dB to -30 dB in 3 dB steps. The red LED illuminates at 3 dB before the clipping point, which also provides an over-range warning for the arithmetical processor.

## 8.1.6 Vintage Reverb

The vintage room reverb effect provides an incredibly natural sounding reverb in the style of the earliest digital reverberators that became popular during the 1980s. Its strength is in recreating natural acoustic ambiences with a very warm and dense characteristic without sounding particularly artificial.

Reflective surface materials and air absorption properties can be simulated by adjusting the high and low frequency cut amount. Low frequency decay and cross-over parameters allow relative control over the low band reverb tail length. This can be used to either simulate real room responses, which often have a longer decay time at low frequencies or alternatively can be useful to reduce low frequency energy in a live environment where it may already be present due to the natural reverberation of the venue. High frequency decay and cross-over parameters provide additional control over the high band reverb tail length.


1. **LF Cut** - Control that adjusts the Low Frequency cut from 10 Hz to 500 Hz.
2. **Input level meter** - Input meters, which comprise a dual-column peak reading LED meters. Each column consists of 14 coloured LEDs. The red LED illuminates at 3 dB before the clipping point.
3. **Output level meter** - Output meters, which comprise a dual-column peak reading LED meters, ranging from 0dB to -30 dB in 3 dB steps.
4. **Power -** Bypass the reverb effect.
5. **HF Cut** - Control that adjusts the High Frequency cut from 0.2 kHz to 20 kHz.
6. **LF Decay** - Adjusts the length of the low frequency section of the reverb in comparison to the rest of the reverb tail.
7. **LF X-over** - Changes the X-Over point of the reverb from 20 Hz to 500 Hz.
8. **Audition** - A rim shot sample is triggered to test the reverb's characteristic.
9. **HF X-over** - Changes the X-Over point of the reverb from 1 kHz to 20 kHz.

10. **HF Decay** - Adjusts the length of the high frequency section of the reverb in comparison to the rest of the reverb tail.
11. **Pre Delay** - Controls the amount of delay (in milliseconds) between the initial signal and the onset of reverberation.
12. **Size control** - Adjusts the average dimension of the simulated space. Range is from 0% to 100% of the largest space possible. A momentary mute is implemented when this control is adjusted.
13. **Decay control** - Control that sets the overall (mid-band) reverberation decay time. Range is from 0.5 to 10 seconds, depending on room size.
14. **Density control** - Lower densities give more space between the reverb's first reflections and subsequent reflections. Higher densities place these closer together. Generally, as with diffusion, higher densities work better for percussive content, and lower densities for vocals and sustained sounds.
15. **Mix control** - Controls the dry/wet output mix and ranges from 0% to 100%, respectively.

## 8.1.7 Chamber Reverb

The chamber reverb emulates the sound of echo chambers found in early recording studios. This is characterized by a rapid buildup of reflection density within a small to medium sized space coupled with a relatively colorless and smooth decay.


Reflective surface materials and air absorption properties can be simulated by adjusting the high and low frequency cut amount and high frequency damping. Low frequency decay and cross-over parameters allow relative control over the low band reverb tail length. This can be used to either simulate real room responses, which often have a longer decay time at low frequencies, or alternatively can be useful to reduce low frequency energy in a live environment where it may already be present due to the natural reverberation of the venue.

1. **INPUT Level meter** - LED Metering shows input level.
2. **OUTPUT Level meter** - LED Metering shows output level.
3. **LF CUT** - Control that adjusts the Low Frequency cut from 10 Hz to 500 Hz.
4. **LF DECAY** - Adjusts the length of the low frequency section of the reverb in compression to the rest of the reverb tail.
5. **LF X-OVER** - Changes the X-Over point of the reverb from 20 Hz to 500 Hz.
6. **HF DAMP** - Dampens the HF to produce a darker sounding reverb. Settings from 1 kHz to 20 kHz.
7. **HF CUT** - Control that adjusts the High Frequency cut from 0.2 kHz to 20 kHz.
8. **PRE DELAY** - Controls the amount of delay (in milliseconds) between the initial signal and the onset of reverberation.
9. **DECAY** - Control that sets the overall (mid-band) reverberation decay time. Range is from 0.5 to 10 seconds, depending on room size.
10. **SIZE** - Adjusts the average dimension of the simulated space. Range is from 0% to 100% of the largest space possible. A momentary mute is implemented when this control is adjusted.
11. **AUDITION** - A sample is triggered to test reverb characteristic.
12. **MIX** - Controls the dry/wet output mix and ranges from 0% to 100%, respectively.
13. **DIFFUSION** - Reducing diffusion creates more discrete echoes, for vocals and other sustained sounds, reduced diffusion can give smooth reverberation that doesn’t overpower the source and maintains clarity.
14. **CONTOUR TIME** - Increases the centre part of the reverbs tail in comparison to the rest of the reverb.
15. **MODULATION** - To create more variation in reverb sounds, modulation adds subtle changes to the reverb characteristics.
16. **POWER** - Turns on the Reverb effect.

## 8.1.8 Hall Reverb

The hall reverb simulates the response of a real concert hall adding a sense of space to the source material with less initial density than a chamber reverb. The slower build-up of reflections and generally longer decay times associated with this type of algorithm allows for increased clarity of the source, while offering a richer, more lush overall sound that is less dense in character.

This effect features contour controls to adjust the envelope shape during the initial portion of the reverb tail and also the time over which the reflection density increases.

Reflective surface materials and air absorption properties can be simulated by adjusting the high and low frequency cut amount and high frequency damping. Low frequency decay and cross-over parameters allow relative control over the low band reverb tail length. This can be used to either simulate real room responses, which often have a longer decay time at low frequencies or alternatively can be useful to reduce low frequency energy in a live environment where it may already be present due to the natural reverberation of the venue.


1. **IN Level meter -** LED Metering shows input level.
2. **PRE DELAY -** Controls the amount of delay (in milliseconds) between the initial signal and the onset of reverberation.
3. **LF CUT -** Control that adjusts the Low Frequency cut from 10 Hz to 500 Hz.
4. **LF X-OVER -** Changes the X-Over point of the reverb from 20 Hz to 500 Hz.
5. **LF DECAY -** Adjusts the length of the low frequency section of the reverb in compression to the rest of the reverb tail.
6. **HF CUT -** Control that adjusts the High Frequency cut from 0.2 kHz to 20 kHz.
7. **HF DAMP -** Dampens the HF to produce a darker sounding reverb. Settings from 1 kHz to 20 kHz.
8. **OUT Level meter -** LED Metering shows output level.
9. **CONTOUR TIME -** Increases the centre part of the reverbs tail in comparison to the rest of the reverb.
10. **CONTOUR ENV -** Controls the shape of the envelope for the centre of the reverb.
11. **SIZE -** Adjusts the average dimension of the simulated space. Range is from 0% to 100% of the largest space possible. A momentary mute is implemented when this control is adjusted.

12. **DECAY -** Control that sets the overall (mid-band) reverberation decay time. Range is from 0.5 to 10 seconds, depending on room size.
13. **DIFFUSION -** Reducing diffusion creates more discrete echoes, for vocals and other sustained sounds, reduced diffusion can give smooth reverberation that doesn't overpower the source and maintains clarity.
14. **MODULATION -** To create more variation in reverb sounds, modulation adds subtle changes to the reverb characteristics.
15. **MIX -** Controls the dry/wet output mix and ranges from 0% to 100%, respectively.
16. **AUDITION -** A sample is triggered to test reverb characteristic.
17. **ON -** Turns ON the Reverb effect.

## 8.1.9 Plate Reverb

The plate reverb effect simulates the actual plate reverb devices that were used in studios in the 1960s and 1970s. They were literally a plate of metal that was suspended under tension with a transducer to transmit audio to the plate while two or more contact microphones were attached to the plate to pick up the results. The plate reverb has a very rapid buildup of reflections and, as a result, is very dense initially with a fairly smooth decay characteristic. For this reason, it is typically the first reverb choice for percussion instruments.

Reflective surface materials and air absorption properties can be simulated by adjusting the high and low frequency cut amount and high frequency damping. Low frequency decay and cross-over parameters allow relative control over the low band reverb tail length. This can be used to either simulate real room responses, which often have a longer decay time at low frequencies, or alternatively can be useful to reduce low frequency energy in a live environment where it may already be present due to the natural reverberation of the venue.


1. **IN level meter** - Input meters, which comprise a dual-column peak reading LED meters. Each column consists of 14 coloured LEDs. The red LED illuminates at 3 dB before the clipping point.
2. **Pre Delay** - Controls the amount of delay (in milliseconds) between the initial signal and the onset of reverberation.
3. **Size control** - Adjusts the average dimension of the simulated space. Range is from 0% to 100% of the largest space possible. A momentary mute is implemented when this control is adjusted.

4. **Decay control** - Control that sets the overall (mid-band) reverberation decay time. Range is from 0.5 to 10 seconds, depending on room size.
5. **Diffusion control** - Reducing diffusion creates more discrete echoes, for vocals and other sustained sounds, reduced diffusion can give smooth reverberation that doesn't overpower the source and maintains clarity.
6. **Modulation control** - To create more variation in reverb's sound, modulation adds subtle changes to the reverb characteristics.
7. **LF Cut** - Control that adjusts the Low Frequency cut from 10 Hz to 500 Hz.
8. **HF Cut** - Control that adjusts the High Frequency cut from 0.2 kHz to 20 kHz.
9. **HF Damp** - Dampens the HF to produce a darker sounding reverb. Settings from 1 kHz to 20 kHz.
10. **LF X-over** - Changes the X-Over point of the reverb from 20 Hz to 500 Hz.
11. **LF Decay** - Adjusts the length of the low frequency section of the reverb in comparison to the rest of the reverb tail.
12. **Power On** - Turns on the reverb effect.
13. **Mix control** - Controls the dry/wet output mix and ranges from 0% to 100%, respectively.
14. **Audition** - A rim shot sample is triggered to test the reverb's characteristic.
15. **Out level mete**r - Output meters, which comprise a dual-column peak reading LED meters, ranging from 18 dB to -30 dB in 3 dB steps. Each column consists of 14 coloured LEDs. The red LED illuminates at 3 dB before the clipping point, which also provides an over-range warning for the arithmetical processor.

# 8.2 Delays

## 8.2.1 Midas Delay

The Midas delay effect provides a simple delay line effect. Delay times can be specified manually. To use the surface tap button, switch on Use Global and BPM Sync. Around 5 taps provides the most accurate tempo timing setting.

There are two models of delay, Analogue and Digital. The two different effect styles become more apparent when modulation is applied to the delay tail.


1. Close - For closing the effect.
2. Tempo - Adjusts the tempo in tempo mode. Range is from 60 to 360 beats per minute (bpm).
3. Use Global - Follows the global tap tempo when active and allows the surface tap button to be used.
4. Range - Selects one of three delay time ranges (1-25 ms, 10-200 ms or 82-1640 ms). Value is displayed immediately below/above button.
5. BPM Sync - Allows BPM (Tap Tempo) mode.
6. Link - Links delay time of left and right channels.
7. Time Left - For entering the desired delay time in ms or a note value.
8. Time Right - For entering the desired delay time in ms or a note value.
9. Pan - Pans channel between L (left) and R (right) outputs.
10. Level - Adjusts the output level. Range is from off to +10 dB.
11. Feedback - Adjusts the amount of negative/positive feedback applied to delay. Controls the number of repeats. Range is from -100% to +100%.
12. Blend - Adjusts the feedback blend from norm to cross.
13. Mod Depth L & R - Adjusts depth of delay modulation. Range is from 0 to 100.
14. HF Depth - Adjusts depth of delay modulation. Range is 0 to 100.
15. Mod Rate - Adjusts rate of delay modulation. Range is between 0.001 Hz and 10 Hz.
16. Mix - Adjusts the mix between dry (0%) wet (100%).
17. Filtering - Contains a HF damping control that adjusts the HF attenuation of delay repeats and an LF damping control that adjusts the LF attenuation of delay repeats.
18. EQ - Contains a HI control knob that adjusts the amount of HF (high EQ) cut or boost applied to the output. LO control knob adjusts the amount of LF (low EQ) cut or boost applied to the output. Range of both is -12 to +12, with 0 at top dead centre.
19. I/O - Switch for de-activating or bypassing delay.

20. Gain - Adjusts the amount of gain between -20 and +20, with 0 at top dead centre.
21. Model - Selects digital or analogue delay resolution. Current selection is shown by the illumination of one of the LEDs (Dig. or Analogue).
22. Input Meters - Two rows of 28 LEDs, which show input metering.
23. Output Meters - Two rows of 28 LEDs, which show output metering.

## 8.2.2 Stereo Delay

The Stereo Delay is a simple effect with easily recognisable controls. Tap control function and easily adjusted filters give the Stereo Delay a clean but controlled sound. It uses less DSP then the Midas Delay which allows two delays to be place in 1 DSP slot.

### BPM display mode:

- Tempo is accurate to 0.1 bpm.
- With global tap enabled the display shows global tempo regardless of delay time setting.
- With global tap disabled the display shows the equivalent tempo assuming a delay of one beat. For example, if the delay time is 500ms the tempo is calculated as 60/0.5 = 120 bpm.
- Up/down buttons adjust local or global tap tempo by 0.1 bpm.

### Millisecond display mode:

- With global tap enabled the display shows current delay (in milliseconds) based on global tempo and selected musical interval. For example, if a 1/8 dot interval is selected on the delay control and the global tempo is 120 bpm the delay value shown will be 0.75 x 60/120 bpm = 375 ms.
- With global tap disabled the display shows the actual delay time set on the unit.
- Up/down buttons adjust delay units by 1 millisecond increments.

If the global tap option is enabled the delay time display will change from seconds (milliseconds) to musical note durations as they do with the current effect's units. However, the display will continue to follow the display mode selected. Also, if the global tap option is enabled the tap button on the unit will not affect the global tempo and is greyed out.


1. **Tap** - Tap tempo button. Requires a minimum of 3 taps to attain an accurate tempo.
2. **Global** - Follows the global tap tempo when active and allows the surface tap button to be used.
3. **Mode** - Switches between BPM and ms in the display.
4. **Delay Time** - Shows the current delay time in either BPM or ms depending on settings.
5. **BPM/ms** - Shows current tempo display mode.
6. **Down Arrow** - Moves the tempo down by either 1 ms or 0.1 BPM depending on mode.
7. **Up Arrow** - Moves the tempo up by either 1 ms or 0.1 BPM depending on mode.
8. **X2** - Doubles the current ms or divides by 2 if in BPM mode.
9. **Tempo** - Adjust the tempo with a pop up rotary control (can be locked to screen). 1ms - 2 s or 1 BPM - 2000 BPM. X2 Double the maximum possible time.
10. **Low Freq Cut** - Swipe the graph to change the low frequency cutoff point. 10 Hz - 500 Hz.
11. **High Freq Cut** - Swipe the graph to change the high frequency cutoff point. 200 Hz - 20 kHz.
12. **Low Freq Cut** - Press number for rotary control of low frequency cutoff point.
13. **High Freq Cut** - Press number for rotary control of high frequency cutoff point.
14. **Low Freq Cut Post FB** - Places the low cut filter after the feedback tail.
15. **High Freq Cut Post FB** - Places the high cut filter after the feedback tail.
16. **Bypass** - Bypass the delay, signal still runs through the effect.
17. **Mix** - Adjusts the wet/dry balance. Turn fully right for a completely wet signal.
18. **Feedback** - Adjusts the amount of negative/positive feedback applied to delay. Controls the number of repeats. Range is from 0% to 100%.
19. **Input Meter** - 29 LEDs which show stereo input metering.
20. **Output Meter** - 29 LEDs which show stereo output metering.

## 8.2.3 TC Electronic 2290

The TC 2290 is a classic studio and live sound delay unit which is widely viewed as one of the best delay effects ever made.

### Insert vs Aux Effect

The TC2290 can be inserted directly into an effect slot on a single channel, which passes the entire signal through the effect. In this case, note that the direct input signal becomes mono before it is panned, either statically or modulated. This occurs when the PAN/DYN section DIRECT button is engaged.

However, the TC2290 can also be added to an aux bus and one or more channels can send a portion of their signal to this bus to be processed by the effect. The output of the effect is then mixed back in with the rest of the tracks. This differs from an insert effect in that the TC2290 isn't affecting the track's entire signal, so the direct signal cannot be modulated using the MOD buttons in the PAN/DYN section. In this setup, the Mix parameter should always be set to 100%.

### Mono/Stereo Operation

The TC2290 can be used both as a mono instance on mono channels and a stereo instance on stereo channels. In the case of a mono out instance, the output signal is made by outputting the left channel only. In this case, panning should not be used.

### Meters

The meter section gives feedback about the incoming and outgoing audio signals. The input level displays the audio as it enters the plug-in and is not affected by adjustments to the input level control or any other parameter. The output meter is affected by the results of the effect as well as the output level control parameter.

### Glob. Tap

The TC 2290 can follow the global tap tempo set from the console. When engaged the Delay Time control adjusts the divide parameter from: 1/16T, 1/16, 1/16 dotted, 1/8T, 1/8, 1/8 dotted, ¼T, ¼, ¼ dotted, ½T, ½, ½ dotted, 1T, 1, and 1 dotted. This is a great way to create “Dub” effects when using T or dotted timings.

### Modulation

This section controls parameters of the modulation effects. Note that the modulation is actually engaged with the MOD buttons located in the PAN, DYN and DELAY sections. The types of modulation effects include:

- Delay time modulations – chorus, flanger, pitch, auto-doubling.
- Pan position modulations – auto-panning of the direct signal, delay signal, or both.
- Dynamic modulation – tremolo, delay compressor/expander, ducking and gating.

Each of these parameter sets consists of the following values:

Waveform – determines the modulation waveform, between sine wave (SINE), random wave (RAND), input signal envelope controlled (ENV), or input level triggered (TRIG). The modulation target determines the function of ENV and TRIG.

Speed – The SPEED parameter is shown in Hz (cycles per second). Depending on the modulation target, when the ENV or TRIG Waveform is selected, the parameter controls the speed from no effect to maximum effect. A setting of “1” means a ramp time of 1 second, whereas a setting of “5” means a ramp time of 1/5 of a second.

Depth – The DEPTH value is displayed in percentage of maximum modulation.

The yellow OSC/THRESHOLD LED shows modulation speed when using periodic modulations (SINE, RANDOM) and indicates when the input level passes the threshold for ENV or TRIG effects.

The **Mod. Set** window shows all the current modulation values in one place all at the same time.


1. **Input Metering -** Metering shows input level (not affected by Input gain).
2. **Delay On -** Turns the delay effect on.
3. **Input Gain -** Controls the level within the effect. -100 dB to 0 dB.
4. **Preset Selector -** Displays current preset and allows preset selector to be expanded.
5. **Feedback Level -** Selects feedback level which can be adjusted from 0% to 99% using the Threshold control.
6. **Feedback High -** Select the HPF option and adjusts the high pass filter from 2 kHz, 4, kHz, 8 kHz, 33 kHz with the Threshold control.
7. **Feedback Low -** Select the LPF option and adjusts the low pass filter from off (0.0), 100 Hz, 200 Hz, 400 Hz with the Threshold control.
8. **Feedback Dyn.Mod -** Adjust the Feedback Dynamic modulation level from 1-9.
9. **Data Control & Threshold LED -** Used to change Level, High, Low and Dynamic Mod features. Green threshold indicates when an input reaches the trigger level.
10. **Feedback -** Displays settings for the feedback section.

11. **INV. -** Pressing the INV button inverts the feedback signal, which may not be noticeable with echo effects, but is more pronounced when applied to modulation such as flanger.
12. **Feedback Set**. - Displays settings for the feedback section.
13. **Modulation Delay -** Select Delay modulation (chorus, flanger, pitch, auto-doubling).
14. **Pan Modulation -** Select Pan modulation (auto-panning of the direct signal, delay signal, or both).
15. **Dyn. Modulation -** Select Dynamic Modulation (delay compressor/expander, ducking and gating).
16. **Speed -** The Speed parameter is shown in Hz (cycles per second). Depending on the modulation target, when the ENV or TRIG Waveform is selected, the parameter controls the speed from no effect to maximum effect. 0.1 Hz to 10 Hz range.
17. **Amount -** Displays values associated with the Speed control.
18. **Depth -** Displays the threshold value of the selected modulation path. 1-9 range.
19. **Amount -** Displays the % of depth 0%-100%.
20. **Threshold -** Controls the threshold of the selected modulation path.
21. **Depth -** Displays the threshold value of the selected modulation path. 1-9 range.
22. **Amount -** The yellow OSC/THRESHOLD LED in the upper left corner of this section shows modulation speed when using periodic modulations (SINE, RANDOM) and indicates when the input level passes the threshold for ENV or TRIG effects.
23. **INV. -** When engaged, the INV DLY MOD button inverts the flanger sweep start and the envelope pitch shift direction. This only applies to Delay Mod Waveforms ENV and TRIG. (Does not affect Pan or Dyn mod)
24. **Sine -** Selects sine wave modulation.
25. **Rand -** Random modulation values.
26. **Env -** Input signal envelope controlled. The modulation target determines the function of Env.
27. **Trig -** Input level triggered. The modulation target determines the function Trig.
28. **Pan Modulation -** Pan Mod on. The parameters are adjusted in the modulation section.
29. **Pan Delay -** Pan Delay on. The parameters for each effect are adjusted in the modulation section.
30. **Pan Direct -** The DELAY/DIRECT button determines if the PAN effect is applied to the delay signal only, the direct signal only, both or neither. This also applies to the static pan set in the plug-in, so if neither DELAY nor DIRECT is lit, there is no panning at all.
31. **Dyn Modulation -** Dyn Mod on. The parameters are adjusted in the modulation section.
32. **Dyn Reverse -** The REVERSE button causes the selected Dynamic effect to function in an opposite way. With the Waveform set to SINE or RAND, a tremolo effect is achieved which produces a modulated increase/decrease in volume. When the REVERSE button is activated, this creates a modulation that enhances delay volume when direct volume is suppressed and vice versa.
33. **Time -** The main function of this section is to control the delay time. The yellow LED above the display will flash in rhythm with the current tempo, and the exact time in ms will be displayed.
34. **Delay Mod -** Enables modulation of the delay signal.
35. **Glob. Tap -** Follows global tap tempo from the main surface tap button. Tempo divides can be set with the Delay time control.
36. **Delay Time -** Adjust the delay time from 0 ms to 9999 ms.
37. **Mod. Set Display -** Displays all the settings for the various modulation parameters.
38. **Output Gain -** Controls the output level of the effect. -100 dB to 0 dB.

39. **Output Gain -** Controls the mix (or ratio) of wet (processed) and dry (unprocessed) signals. 0.0% to 100%
40. **Pan -** Adjust the Pan position. L100 to 0 to R100.
41. **Mute In -** Muting the input will allow the echo tail to fade naturally after the effect is bypassed.
42. **Mute Out -** Mutes the effects output.
43. **Inv. -** When the Inv delay button is activated, the output of the delay signal is phase inverted.
44. **Deep Mod -** This disables the automatic modulation depth mapping known as "Golden Ratio". This makes it possible to do much deeper modulation with wild pitch shifts, but it is somewhat uncontrollable.
45. **Output Meter -** Metering shows input level.


1. **Preset Bank Selection Area -** Select a bank containing presets.
2. **Preset Selection Area -** Displays the current bank of presets, different tabs for various different settings.
3. **Load Preset -** Once a preset is selected press load. Preset name is displayed on the main effect display.

# 8.3 Dynamics

## 8.3.1 HD-2A Levelling Amplifier

The pure tube signal path is recreated in HD-2A Levelling Amplifier. The ultra-smooth optical attenuator is closely modelled on our LA style compressor. It provides breathing, natural and effortlessly musical compression. Inspired by the original LA-2A.

The Comp/Limit switch determines whether the effect acts as a compressor or a limiter. The Limit setting uses a much higher compression ratio. Gain affects the input level of the signal and controls the threshold at which compression takes place. Power On toggles the effect in or out of the signal path. Increasing Peak Reduction means that signal peaks are compressed more. Mix allows parallel compression to be used. The Gain Reduction control changes how Gain Reduction metering information is displayed.

**TIP**: The HD-2A can be used to smash (heavily compress) a drum group but allow the dry transients of the uncompressed signal to be mixed back in giving a huge drum sound. Set the Mix dial to around 50% and experiment.


1. **Limit/Compress** - Sets the compression ratio of the leveller. When set to Compress, the compression ratio is approximately 3:1 and when set to Limit, the ratio is around ∞:1.
2. **Gain** - The Gain knob increases the output level to compensate for the reduced signal level that results from compression.
3. **VU Meter** - A VU meter that displays either the amount of gain reduction, or output level, depending upon the setting of the Meter Function switch.
4. **Peak Reduction** - This control sets the amount of signal compression by adjusting the trigger threshold. Increasing the value lowers the threshold, and therefore increases the amount of compression.
5. **Mix** - Dry/wet mix. Allows Parallel compression, a form of upward compression, is achieved by mixing an unprocessed 'dry' signal with a heavily compressed version of the same signal.
6. **Metering Selection** - This rotary control sets the mode of the VU Meter. When set to Gain Reduction, the VU Meter indicates the Gain Reduction level in dB. When set to +10 or +4, the VU Meter indicates the output level in dB.

7. **Power** - Bypass control for the compressor.

## 8.3.2 KT 1176 Limiting Amplifier

When Urei released the 1176LN Limiting Amplifier in the late 60s, it broke new ground. The Field Effect Transistors employed had just been invented and the 1176 was one of the first audio processors to benefit from this new technology. Our digital reincarnation is based on the early Rev. E model and authentically captures the smooth character of the original class-A output stage and its FET's legendary Fast Attack. Inspired by Urei 1176LN. With an added MIX new sound creation possibilities are available using parallel compression techniques.


1. **Input** - Input - ∞ to 0 dB determines the level of the signal entering the 1176 KT, as well as the threshold. Higher settings will therefore result in increased amounts of limiting or compression.
2. **Mix** - Dry/wet mix. Allows Parallel compression, a form of upward compression, is achieved by mixing an unprocessed 'dry' signal with a heavily compressed version of the same signal.
3. **Output** - Output - ∞ to 0 dB determines the final output level of signal leaving the 1176 KT. Once the desired amount of limiting or compression is achieved with the use of the Input control, the Output control can be used to make up any gain lost due to gain reduction.
4. **Attack** - Sets the amount of time it takes the 1176 KT to respond to an incoming signal and begin gain reduction. The 1176 KT attack time is adjustable from 20 microseconds to 800 microseconds. The attack time is fastest when the Attack knob is in its fully clockwise position and is slowest when it is in its fully counter clockwise position.
5. **Release** - Sets the amount of time it takes the 1176LN to return to its initial (pre-gain reduction) level. The 1176LN release time is adjustable from 50 milliseconds to 1100 milliseconds (1.1 seconds). The release time is fastest when the Release knob is in its fully clockwise position and is slowest when it is in its fully counter clockwise position.
6. **Ratio** - The 1176LN Ratio buttons allow four different modes of operation: All – The famous all buttons pressed mode (extreme limiting).
   4 - Selects a 4:1 ratio (moderate compression). 8 - Selects an 8:1 ratio (hard

compression). 12 - Selects a 12:1 ratio (smooth limiting). 20 - Selects a 20:1 ratio (hard limiting).
7. **Level** - Displays the chosen metering option.
8. **GR** - The amount of gain reduction.
9. **(+) 8** - When +8 is selected, a meter reading of 0 corresponds to a level of +8 dB.
10. **(+) 4** - When +4 is selected, a meter reading of 0 corresponds to a level of +4 dB.
11. **Power** - Bypass control for the compressor.

## 8.3.3 KT Bus Compressor

Based on the distinguished master buss center compressor of the 4000 G console, the KT Buss Compressor captures the unique sound of the original's circuit input and twin VCA gain-reduction amplifier design. Cherished by top engineers for its ability to "glue together" tracks, the KT Bus compressor is ideal for gluing IEM mixes or a master bus mix together.

**Tip**: Try the KT Bus Comp on an IEM mix with the attack set to 1 (to allow transients through), release set to Auto, Ratio at 2.1, HPF at 125 Hz and 100% Mix. Adjust the threshold until around 4 dB of gain reduction takes place. This will glue an IEM mix together making it sound more controlled while still allowing low end energy to pass into the mix preventing the pumping effect.


1. **Threshold** - The threshold determines when the compressor starts to compress the signal.
2. **Make Up Gain** - Adds gain to compensate for the gain reduction.
3. **Attack** - By slowing the attack rate, the compressor gradually comes to full compression, instead of compressing immediately. Transient response is less affected, so maintaining the presence of each note.
4. **Release** - By slowing the release rate, the compressor recovers more slowly from compression, so it does not turn off completely when the signal returns below the threshold. The Auto release characteristic for G Bus Compressor has an exceptional quality that is optimized for full mixes.
5. **Ratio** - The ratio control determines how much compression is applied to the signal. Ratios available are 2:1, 4:1, and 10:1.
6. **HPF** - HPF frequency setting for sidechain filter. Anti-clockwise turns of the sidechains HPF filter. The sidechain HPF allows removal of low-frequency content from the control sidechain, reducing excessive gain reduction on bass-heavy audio signal's without reducing bass content of the audio signal itself.
7. **Gain Reduction Meter -** Shows the amount of gain reduction to the signal.
8. **Mix** - Dry/wet mix. Allows Parallel compression, a form of upward compression, is achieved by mixing an unprocessed 'dry' signal with a heavily compressed version of the same signal.
9. **IN** - Bypass control for the compressor.

## 8.3.4 R-Comp

The R-Comp is a unique Compressor and Limiter which offers both mono operation and true stereo operation from a single set of controls. It's a relatively clean VCA compressor which generally gives a small bite to things around 4-8 kHz. It can grab quite fast if that's needed. Great on the mix bus to add a little thickening and polish.


1. **Input** - input -18 to +18 with 0 dB at top dead centre.
2. **Ratio** - The ratio control determines how much compression is applied to the signal.
3. **Attack** - By slowing the attack rate, the compressor gradually comes to full compression, instead of compressing immediately. Transient response is less affected, so maintaining the presence of each note.
4. **Release** - By slowing the release rate, the compressor recovers more slowly from compression, so it does not turn off completely when the signal returns below the threshold.
5. **Auto Release** - Auto release when engaged averages the release time of the compressor automatically. The release control is bypassed.
6. **Threshold** - The threshold determines when the compressor starts to compress the signal.
7. **Input/Output meter** - VU displays, which shows input/output metering. Switch to change level view.
8. **Gain Reduction meter** - Shows the amount of gain reduction to the signal.
9. **Make Up Gain** - Adds gain to compensate for the gain reduction.
10. **Dry/Wet** - Allows Parallel compression, a form of upward compression, is achieved by mixing an unprocessed 'dry' signal with a heavily compressed version of the same signal.
11. **Power** - Bypass control for the compressor.

## 8.3.5 HD 670 Compressor

The Fairchild 670 tube compressor not only achieves record bids in high-end vintage gear auctions, it also delivers some of the finest colorations in compressor history. A six-step switch determines the timing, and the two large Input and Threshold controls adjust the levels. Our model is true to the original signal path, and conveniently provides models for dual, stereo-linked or M/S operation. Inspired by Fairchild 670.

Input gain determines how much of the input signal passes through the effect. On toggles the compressor in or out of the signal path. Threshold determines how loud the signal has to be before compression is applied. Time switches the compression's release time. Bias changes the simulated tube bias algorithm and adjusts the knee of the compression from soft to hard via 4 settings. Out gain affects the gain of the output signal. Balance set to its default setting controls the amount of additive signal deflection know as 'thud' which happens due to an attack. At this setting it is minimized. Setting this control counter clockwise from this position results in a thud of one polarity on transients and going clockwise produces a thud of opposite polarity.

The terms lateral (side-to-side) and vertical (up-and-down) refer to the mechanical modulations in a vinyl record groove that are transduced into electrical audio signals by the phonograph stylus and cartridge.

The HD 670 can perform dynamics processing on the lateral (LAT) and vertical (VERT) components of stereo signal's independently. In other words, the monophonic (middle) and/or stereo (side) components of a stereo source signal can be compressed or limited separately from the other component.

can also be used for creative effects outside of the phonograph environment.

Lat/Vert processing is accomplished by first routing the stereo source signal through a sum/difference (mid/side) matrix which separates the stereo source into lateral (middle, or centre without stereo) and vertical (side, or stereo without mono) signal components. The lateral/vertical components are then compressed or limited independently. Finally, the mid/side components are recombined into a normal stereo signal via a second sum/ difference matrix.

In lateral/vertical mode the HD 670 operates as two monophonic compressors with independent control of the middle and side components of the two input signals. The input signals are processed by the sum/difference (mid/side) matrix before and after the compressors, but there is no interaction between the two compressors.

**Tip:** For the most useful results set the Time Constant to 1, this is the best setting for general live sound compression.


1. Metering - Metering shows either Input, Gain Reduction or Output.
2. Input Gain - Adjust the input gain to the compressor for more distortion characteristics if desired.
3. Time Constant - P1 Attack=200 µs Release=300 ms P2 Attack=200 µs Release=800 ms P3 Attack=400ms Release=2 sec, P4 Attack=800 ms Release=5 sec P5 Attack=200 Release is Program dependant P6  A400 R Program Dependant.
4. Balance - Adjusts the release characteristic of the compression by creating either an asymmetrical positive or negative DC bias.
5. D.C. Bias - This control changes the knee of the compressor, hard knee when the control is fully CCW, through to soft knee when fully CW.
6. Thresh - Adjust the level at which compression takes place.
7. Output - Adjusts the output of the compressor to allow level matching.
8. Mix - Adjusts the Wet/Dry blend of compression for parallel compression effects.
9. On/Off - Turns the effect on or off.
10. LAT/VERT - In this mode, the 670 splits the two channels via a matrixing network, separately limits them, then recombines them through another matrix to reconstitute the stereo signal. This method of processing allowed the 670 to maximize the use of the stereo space, making mixes louder with less distortion.

## 8.3.6 Smart Dynamics Processor

The Smart Dynamic Processor is based on adaptive signal processing techniques which are able to simplify the user workflow, without compromising sound quality and versatility. In particular, the Smart Dynamics Processor is an intelligent dynamic range processor based on the Midas Channel Compressor and is therefore characterized by the same unique sound signature. It can operate in two different processing modes: smart transient mode or smart loudness mode.

The Smart Transient mode is designed to work on the envelope of transient sounds, like drums or slap bass, and modify its different components in order to achieve the desired sonic result.

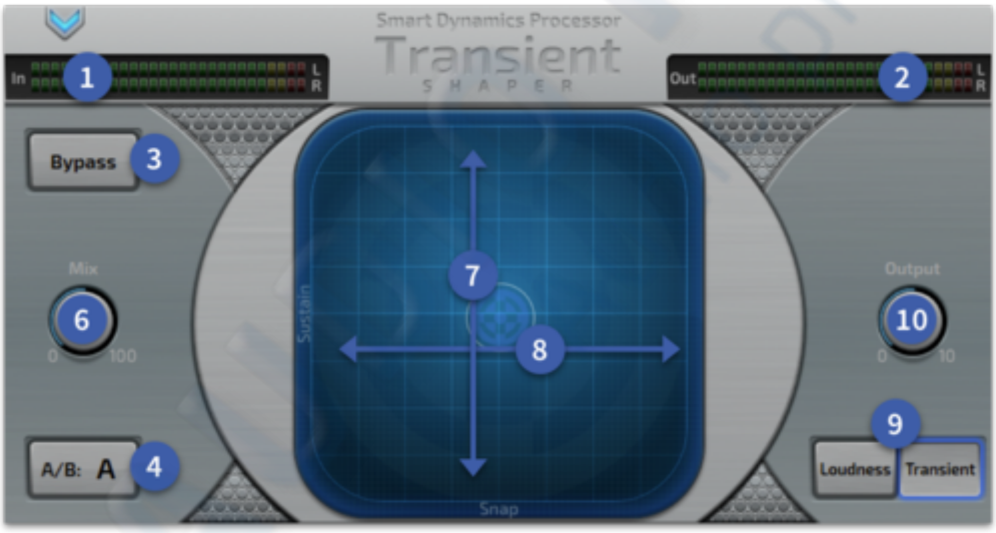
When operating in this mode, 4 different Semantic Controls associated with the desired sonic result are available:

- the Snap control, insert space which can be used to amplify or to reduce the attack component of a transient signal;
- the Sustain control, which can be used to amplify or to reduce the sustain component;
- the Balance control allows the blending of the dry signal with processed signal in order to get either a more natural or enhanced sound;
- the Output level controls can be used to correct the final output level and avoid clipping.

The Smart Loudness mode is instead designed to increase naturally the loudness of any type of sound, without introducing distortion. This is achieved through a combination of intelligent upward compression and adaptive limiting. When operating in this mode, the overall perceived loudness of the track can be increased by operating the Loudness control (5). The output level can then be adjusted to avoid clipping.


1. **Input meters** - Two rows of 28 LEDs, which show input metering.
2. **Out meters** - Two rows of 28 LEDs, which show output metering.
3. **Bypass** - Switches the effect on and off.
4. **A/B** - A/B is comparing two versions of the effect to see which sounds best.

5. **Transient Fader** - The Smart Transient mode is designed to work on the envelope of transient sounds, like drums or slap bass, and modify its different components in order to achieve the desired sonic result.
6. **Mix** - The Mix control allows the blending of the dry signal with processed signal in order to get either a more natural or enhanced sound. Range -0% (dry) to 100% (wet).
7. **Sustain** - Sustain control, which can be used to amplify or to reduce the sustain component.
8. **Snap** - Snap control, insert space which can be used to amplify or to reduce the attack component of a transient signal.
9. **Maximiser/Transient** - Toggles between the Maximiser and Transient effect.
10. **Output** - Adjusts the level of the output of the effect.



## 8.3.7 Dual De-Esser

De-essing is an audio effect designed to reduce the amount of excessive sibilance in an audio signal, usually when dealing with the human voice. The Dual De-Esser accurately and seamlessly removes sibilance from audio tracks.

The Lo-Band (around 6k for male or 7k female voices) and Hi-Band (around 11k) controls select which portion of the audio spectrum is affected by the De-Esser on each channel. By increasing the amount of reduction, the level to which the de-essing affects the input signal can be controlled.


1. **Input Metering** - Metering shows input level.
2. **Low GR Meter** - Shows the amount of Low Band S reduction. 0 to -14 dB.
3. **Low GR threshold** - Threshold controls the point at which the Low Band begins responding to signal level.
4. **Low GR threshold** - Displays the amount of gain reduction to be applied for the Low Band.
5. **Power** - Bypass the effect.
6. **Low Band On** - Low Band active when illuminated.
7. **Low Band Range** - Low Band active when illuminated.
8. **High Band On** - High Band active when illuminated.
9. **High GR threshold** - Threshold controls the point at which the High band begins responding to signal level.
10. **High GR Level** - Displays the amount of gain reduction to be applied for the High band.
11. **High GR Meter** - Shows the amount of Low Band S reduction. 0 to -14 dB.
12. **Output Metering** - Metering shows output level.

## 8.3.8 DSR De-esser

The de-esser is designed to reduce sibilance in human voices, such as excessive presence of S sounds.

Due to the special sibilant detection algorithm, the de-esser is completely input level independent. This means that the reduction of sibilants will not change if the input level is changed. For example, changing the microphone amplifier gain does not affect the detector.

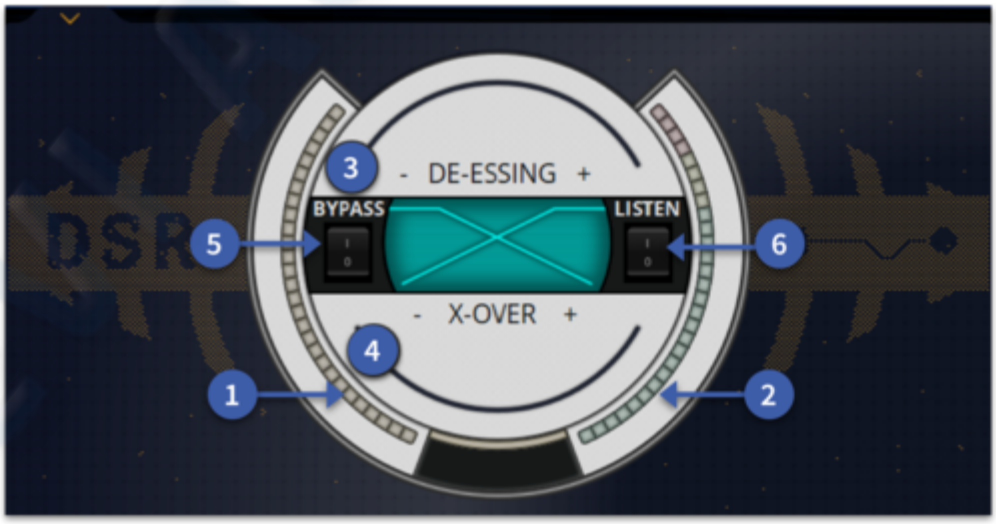
Although the de-esser is primarily designed to be applied on human voices, it can also be used creatively on other instruments.

**X-over Control** - Part of the detection algorithm is a matched lowpass/highpass crossover filter. The crossover frequency is adjusted by the crossover knob. When a normal, non-sibilant, sound is present in the input the energy will be mostly focused in the lower section on the frequency spectrum. On the other hand, when a sibilant sound is present most of the energy will be present in the higher section of the frequency spectrum. Therefore, the crossover knob is useful for tuning the detector so that the de-essing mechanism is only triggered by high-frequency energy content, as it would occur with an S sound.

The optimal crossover frequency value is indicated by the high pass filter being lit blue only when a sibilant sound is present.

**De-Essing Control -** When a sibilant is successfully detected the amount of reduction can be adjusted by using the de-essing knob. At minimum position, the reduction is 0 and equivalent to bypass, i.e. the audio is not affected. As the user turns the knob clockwise, the number of sibilants will be reduced.

**Listen** - When the Listen button is pressed the signal from the high pass filter is routed directly to the output.



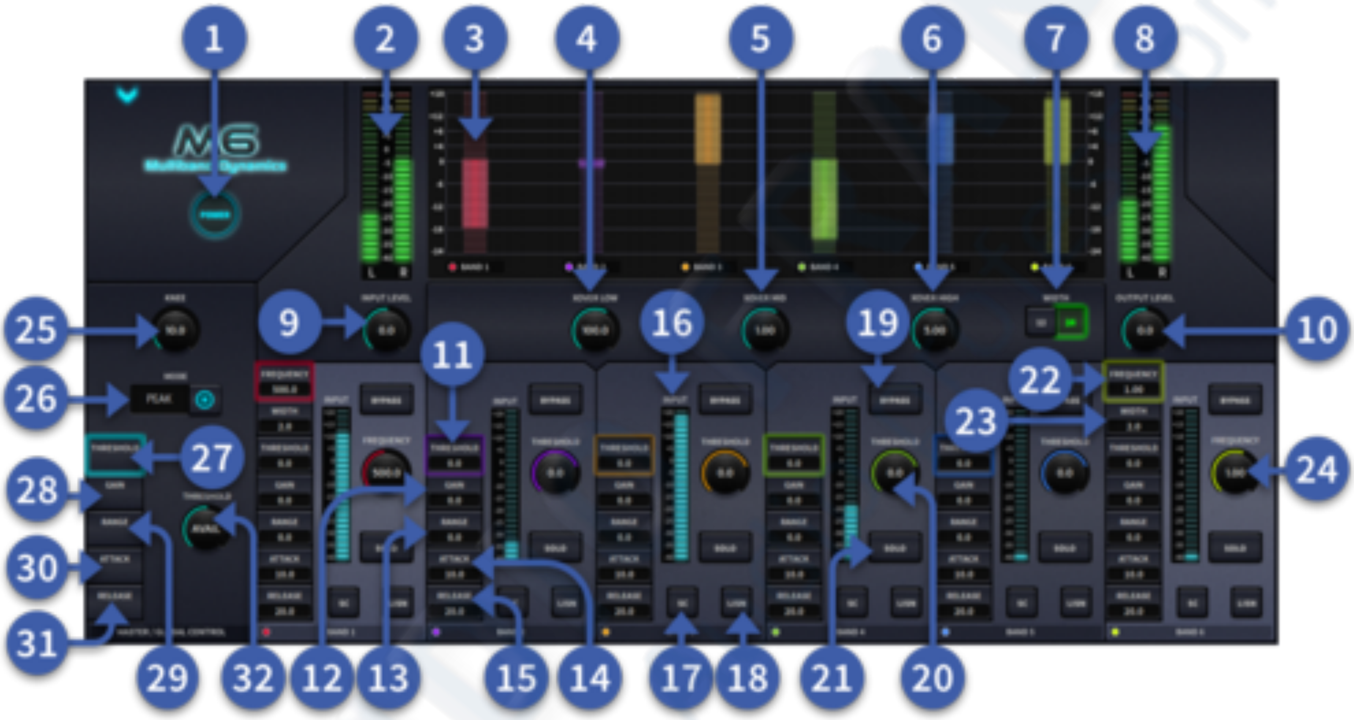

1. Gain Reduction meters - 28 LEDs, which show gain reduction metering.
2. Out meters - 28 LEDs, which show output metering.
3. DE-ESSING - When a sibilant sound is successfully detected the amount of reduction can be adjusted by using the de-essing slider. At minimum position, the reduction is 0

and equivalent to bypass. Moving the slider clockwise will reduce the amount of sibilance.

4. X-Over - Part of the detection algorithm is a matched lowpass/highpass crossover filter. The crossover frequency is adjusted by the crossover slider. When a normal, non-sibilant, sound is present in the input the energy will be mostly focused in the lower section on the frequency spectrum. When a sibilant sound is present most of the energy will be present in the higher section of the frequency spectrum. Therefore, the crossover knob is useful for tuning the detector so that the de-essing mechanism is only triggered by high-frequency energy content, as it would occur with an *s* sound.
5. Bypass - When pressed the input signal is unaffected.
6. Listen - When the Listen button is pressed the signal from the high pass filter is routed directly to the output of the de-esser, enabling fine-tuning of the crossover. Note that this happens in place, i.e. the main output is filtered, therefore the user should be very careful when using it.

## 8.3.9 M6 Multiband Dynamics

M6 is a six-band multiband compression and expansion tool. The M6 combines multiband compression, equalization, de-essing, expansion, and limiting capabilities into one easy to use interface. The M6 is a powerful effect when the need to perform separate EQ and dynamic processes to different bands on the same input source. The M6 has four bands in the centre controlled by crossover regions (2-4), plus two additional frequency selectable bands that are not part of the multiband crossover (1&6). The two floating bands have dedicated frequency and width controls, unlike the four middle bands. This can be useful in situations when you need additional control over the dynamics of a frequency range that is not being handled sufficiently by the four main bands such as de-essing a vocal.



1. Power - Turns the effect on or off.
2. Input meter - 25 LEDs, which show input metering.
3. Band 1 - 6 Gain Exp/Comp Meter - 43 LEDs, which show gain expansion and compression.
4. X-OVER Low Frequency - Sets the Low crossover frequency from 25 Hz to 250 Hz.
5. X-OVER Mid Frequency - Sets the Mid crossover frequency from 250 Hz to 2.5 kHz.
6. X-OVER High Frequency - Sets the High crossover frequency from 1 kHz to 16 kHz.
7. Slope Width - Sets the crossover slope at either 12 dB or 24 dB.
8. Output meter - 25 LEDs, which show output metering.
9. Input Level Trim - Trim the input level -18 dB to + 18 dB.
10. Output Level Trim - Trim the output level -18 dB to + 18 dB.
11. Threshold - Threshold controls the point at which a band begins responding to signal level. Range: 0 to -60 dB.
12. Gain - Gain controls the output gain of each compressor band. Ideal for sculpting the shape of a sound. Range: +18 to -18 dB.
13. Range - Range controls the maximum gain change. Negative range values result in compression. Positive range values result in expansion. Range:+18 to -24 dB.

14. Attack - Attack controls the speed at which dynamic processing begins. Range: 0.50 to 500 ms.
15. Release - Release sets the recovery speed of the gain attenuation when the input drops below the threshold. Range: 5 to 500 ms.
16. Band 1-6 Input meter - 25 LEDs, which show individual band input metering.
17. Side Chain Function - Bands can be individually side chained to a different path.
18. Band Listen - Listen to the signal in isolation of the processing band or SC if on.
19. Band Bypass - Bypass defeats a band’s dynamics processing and Gain control.
20. Current Band 2-5 - Shows the currently selected control for band 2 - 5.
21. Band Solo - Solo is used to listen to an individual band, post-process.
22. Frequency - Band 1 & 6 Frequency selection 20 Hz to 20 kHz.
23. Width - Sets the width of band 1 & 6 from 0.3 (Narrow) to 60 (wide).
24. Current Band 1 & 6 - Shows the currently selected control for band 1 & 6.
25. Global Knee - Determines the compression knee characteristics From soft (4) to hard (40).
26. Mode - Choose compression mode, either Peak, RMS or Vintage.
27. Global Threshold - Controls all band Thresholds simultaneously, keeping their relative values intact.
28. Global Gain - Changes all band Gain values simultaneously, keeping their relative values intact.
29. Global Range - Changes all band Range values simultaneously, keeping their relative values intact.
30. Global Attack - Changes all band Attack values simultaneously, keeping their relative values intact.
31. Global Release - Changes all band Release values simultaneously, keeping their relative values intact.
32. Global Control - Shows the currently selected global control.

## 8.3.10 PIE (Prime Input Expander)

The Prime Input Expander or PIE for short can greatly increase perceived level of vocals in a busy live mix. It can be used to remove unwanted noise from a stage. Think of it like a gate tailored for vocals. The side chain can be used to further tighten control over the vocal range by using carefully selected settings. For example, try setting the side cain to 2K in BP mode with a width of 1. Then set the depth to -12 and set the threshold so that when the vocalist sings the PIE opens.


1. Slow/Fast - select the speed at which the PIE reacts to input.
2. Input Meter - shows input level.
3. Threshold - sets the level at which the PIE opens and lets audio pass. - 36 dB to 12.
4. Depth - sets the level of attenuation from 0 dB (no change to -20 dB).
5. RMS/PEAK - select RMS or Peak detection. RMS gives smoother results, peak works well for dynamic signals.
6. Output Meter - shows output level.
7. Power - can be used to bypass the effect.
8. S/C External - when engaged this uses any externally patched input signal to trigger the effect.
9. S/C LISTEN - this button is used to listen to the sidechain, either external or with the internal filter settings.
10. Width - sets the Q of the internal side change frequency. Turn CCW for tight control, CW for a wider Q value.
11. Freq - sets the frequency of the side chain filter from 20 Hz to 20 kHz. This is dependent on the type chosen.
12. Type - choose from Low Pass (LP), High Pass (HP), Band Pass (BP) or Flat. When Flat is selected the frequency and width have no effect.

## 8.3.11 Stressor Compressor

The Stressor is a characteristic emulation inspired by a famous modern compressor used on many recordings and in the live sound world. It can be used to tame aggressive vocals or add pizzazz to flat sounding drums. The Nuke ratio turns the Stressor into a brutish brick wall limiter that will not surrender.

Sidechain HPF and Sidechain EQ further expand the creative use of the stressor by tailoring the frequency response.

**Tip:** Set the Attack to 3, Release to 4 with a ratio of 3:1. Adjust the Input to give around 7 to 10 dBs of gain reduction on loud peaks. This is a great start for rock style vocals. A ratio of 10:1 gives a classic 1176 Opto style compression.


1. Input - Adjust the input into the effect to control the amount of compression. Range 1-10.
2. Attack - Changes the attack time, fast settings to reduce transients, longer settings to add attack to the source material . Range 1-10.
3. Release - The release is variable from 50 mS to 3 seconds and varies depending on ratio setting. Range 1-10.
4. Output - Adjusts the output of the Stressor to control. Use output to make up any gain loss and for comparison tests. Range 1-10.
5. Mix - The Mix control allows the blending of the dry signal with processed signal in order to get allow transients to be mixed back into the signal for parallel compression. Range -0% (dry) to 100% (wet).
6. Ratio - Press to select the 8 different ratio settings.
7. SC HPF - The Side-Chain High Pass Filter cuts low information in the stressors sidechain by choosing the exact point on the frequency spectrum where you want compression to be triggered without losing any low-end.
8. Sidechain Gain - Adjust the gain of the side chain by +- 30 dB.
9. Sidechain Frquency - Set the sidechain frequency from 40 Hz to 10 kHz.
10. Sidechain BW - Set the bandwidth of the sidechain EQ from 0.1 (tight) to 5 (wide).
11. Gain Reduction - 16 LEDs show gain reduction.
12. Active - Turn the Stressor on. When in active audio will pass un-processed for comparison purposes.
13. Input - 16 x 2 LEDs, which show stereo input metering.
14. Output - 16 x 2 LEDs, which show stereo output metering.

## 8.3.12 Dual Band & Brickwall limiter.

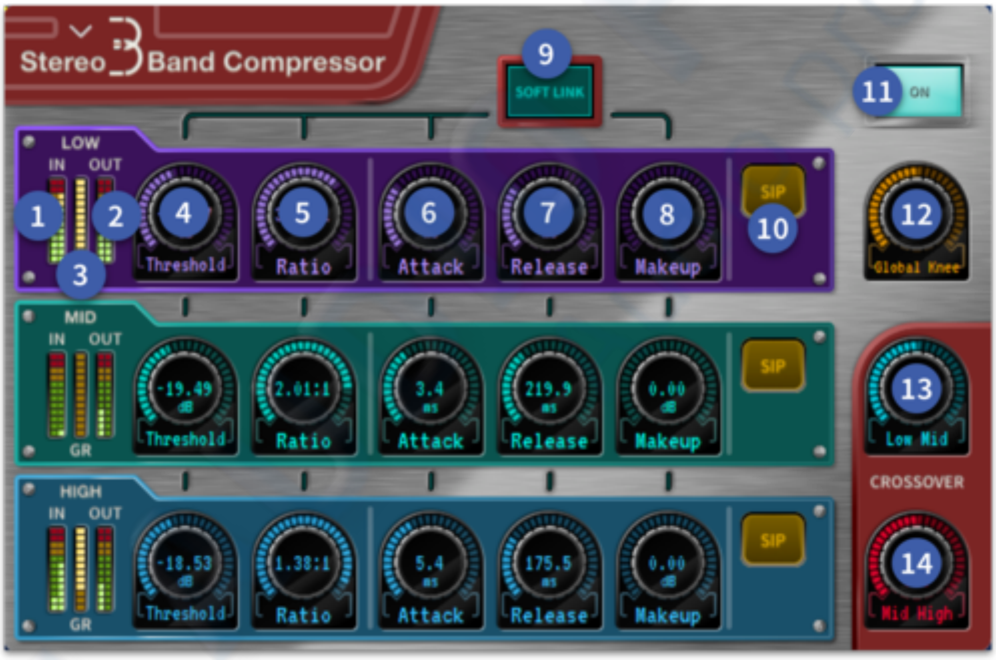
The Dual Band and Brickwall limiters are designed to ensure the output level never exceeds a set limit. They can be used to protect loudspeakers or the output to an in ear monitor system. The innovative dual band limiters allow the user to limit the low and high bands independently of each other.

Tip: Use the Dual Band limiter on a Stereo DJ feed or signal to protect the speaker system from independent High and Low peaks.


1. Input and Low Band GR Meter - 14 LEDs which show stereo input metering. 14 LEDs which show gain reduction (GR) of the Low Band.
2. Output and High Band GR Meter - 14 LEDs which show stereo output metering. 14 LEDs which show gain reduction (GR) of the High Band.
3. Input Gain - Adjusts the input level to the limiter. -20 dB to 40 dB.
4. Threshold - Threshold controls the point at which the limiter begins responding to signal level.
5. Attack - Attack controls the speed at which dynamic processing begins.
6. Release - Release sets the recovery speed of the gain attenuation when the input drops below the threshold.
7. Crossover - Sets the frequency at which the two bands are split.
8. Crossover Enable - Engages the crossover for frequency split Dual band limiting. 2x full band limiters in series when The Crossover is disabled.
9. Mode - Switch between 2 Band or Brick wall mode.
10. Power - Turn the limiter off.
11. Soft - The soft knee setting applies limiting gradually until the ratio is reached. This makes the transition from uncompressed to compressed audio smoother and less abrupt.
12. Input Gain - Adjusts the input level to the limiter. -20 dB to 40 dB.
13. Output Gain - Output gain sets the maximum level.
14. Attack - Attack controls the speed at which dynamic processing begins.
15. Release - Release sets the recovery speed of the gain attenuation when the input drops below the threshold.


## 8.3.13 Stereo 3 Band Compressor

The 3-band Compressor effect is a minimum phase shift (analogue style) implementation that guarantees coherent band summing, even at the most extreme crossover point settings. Each band provides full control of its compressor's action, with partially adaptive time constants ensuring the most natural results from even the most variable sources.




1. Band Input Meter - 12 Stereo LEDs showing each band's input metering.
2. Band Output Meter - 12 Stereo LEDs showing each band's output metering.
3. Gain Reduction Meter - 12 LEDs showing band gain reduction.
4. Threshold - The threshold determines when the compressor starts to compress the signal.
5. Ratio - The ratio control determines how much compression is applied to the signal. Ratio range from 1:1 to 25:1.
6. Attack - Changes the attack time, fast settings to reduce transients, longer settings to add attack to the source material . Range 0.2 ms to 50 ms.

7. Release - The release is variable from 50 ms to 3 seconds and varies depending on ratio setting.
8. Makeup - Adjusts the output level. Use output to make up any gain loss and for comparison tests. Range 0 dB to 24 dB.
9. Soft Link - Links all bands together for global parameter control.
10. Solo In Place - Press and hold to send the selected band to the solo bus for audition.
11. Power - Bypasses the effect.
12. Global Knee - Compression knee adjustment from 4 dB (Soft) to 40 dB (Hard).
13. Low Mid - Adjust the low/mid crossover point frequency. 40 Hz to 1 kHz.
14. Mid High -Adjust the mid/high crossover point frequency. 640 Hz to 16 kHz.

### 8.3.14 Klark Teknik SQ1 Compressor/Gate

A veritable toolbox of dynamics processing, SQ1D provides either compression or gating with side chain filtering and monitoring. SQ1D features multiple compression modes with combinations of RMS and Peak sensing and Hard and Soft Knee characteristics, ideal for both corrective and creative dynamics processing. In conjunction with the fully-featured noise gate and side chain options.

SQ1 features a "Vintage" switch allowing the selection of the default RMS sensing compression mode, or when engaged, an emulation of many older compressor designs with exponential attack and release envelopes. When used in conjunction with the "Hard Knee" switch, SQ1D provides several different forms of compression to suit a wide range of programme material. By default, SQ1D offers a Soft Knee response, however when the "Hard Knee" switch is activated, a Hard Knee response is provided instead.

With both mode switches inactive the compressor behaves in the default RMS and Soft Knee modes. This gives the slowest (and most subtle) feel to the compressor envelopes. The soft knee curve combines with the adaptive RMS attack and release times to produce gentle envelope responses that are ideal for compressing vocals, but which can still be aggressive enough to limit transients when needed. The Soft Knee curve also reduces the adaptive nature of the RMS detection slightly, providing a little more manual control of the envelope timings.

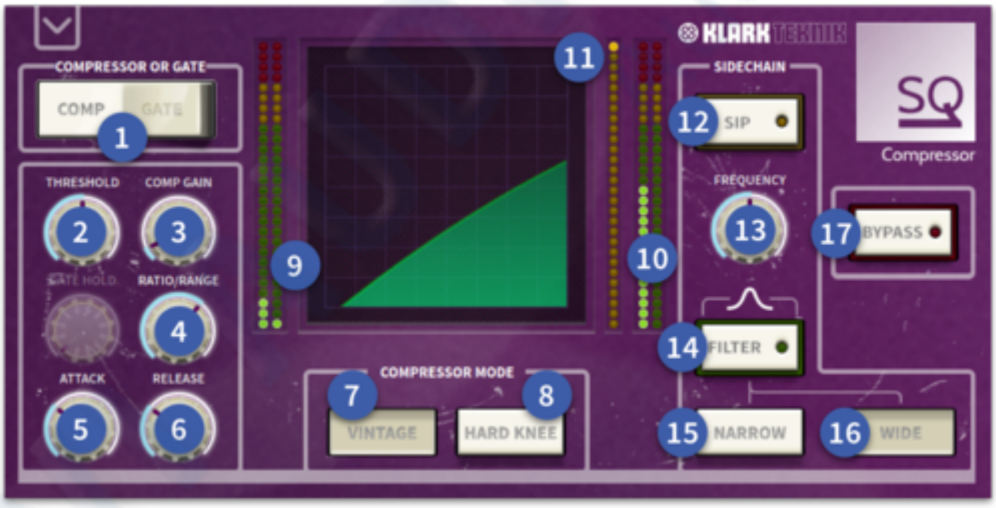
When the Hard Knee response is used with RMS mode, the compressor operates in a more clinical way with a more defined transition between under threshold and over threshold - this is better suited to limiting style compression. A small amount of the soft knee curve is still retained keeping the sound reasonably natural but with no modification of the compressor envelope. This means that attacks are more aggressive, but it also allows the adaptive nature of the RMS detection to operate to its fullest extent. This mode is good for natural sounding limiting of speech.

With the "Vintage" switch active and the default Soft Knee response selected, the compressor employs a dual time constant, linear attack profile. This produces extremely subtle attack and release curves during the onset of compression that are largely independent of the envelope control settings. As the compressor is driven harder (i.e. signal's further over threshold) the soft knee effect reduces, gradually returning manual control of the attack and release times to optimise capture of larger transients. Like the RMS modes, this compressor mode is very adaptive making set up of the envelope controls relatively easy. The peak sensing, however, increases harmonic overtones, which adds a "Vintage" brightness and sparkle to the audio, producing extremely transparent and lively sounding compression of acoustic instruments.

When both the "Vintage" and "Hard Knee" switches are selected, the compressor operates with more precise envelope control and a defined transition between under- and over-threshold. This produces aggressive compression that gives fast control of extremely dynamic material. It can also be used to add colour to low frequency signal's making it ideal for controlling instruments such as bass guitar. With the highest ratio setting, this mode allows the compressor to effectively be used as a limiter.

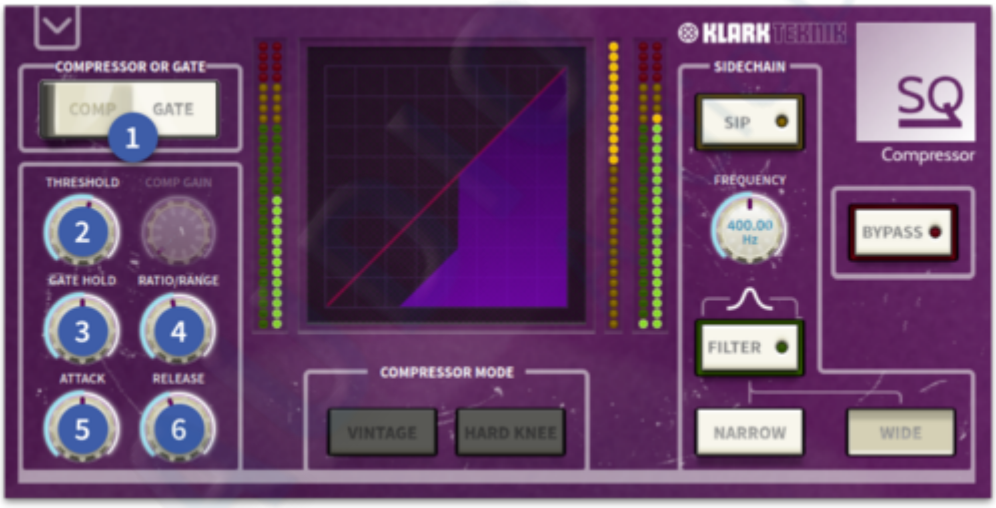
The Gate has the iTs feature which operates in conjunction with the Gate Hold control to reduce false triggering of noise gates. When signal's (especially low frequency ones) are very close to the gate threshold, a gate can repeatedly open and close on the audio. iTS ensures that the gate remains open by automatically adjusting the gate threshold downwards the moment the signal goes over the threshold setting. When the signal eventually falls below the (new) temporarily adjusted threshold, the iTS function resets, ready for the next gate opening. This hysteresis action results in more decisive operation of the noise gates, and attack envelopes also start instantly and consistently, even on signal's that are only slightly over threshold.

The side chain is equipped with a selectable bandpass filter with wide and narrow responses. This powerful feature greatly extends the capabilities of the multiple compression and gating modes. Frequency-selective compression is a very effective and creative alternative to equalisation, with the wide bandpass filter response being ideal to enhance the essential character of an instrument, allowing it to stand out in a mix without the use of harsh- or unnatural-sounding boosted EQ responses. The SQ1 can also be used as a frequency-selective De-esser by selecting the narrow bandpass filter response to reduce sibilance or other undesirable artefacts from vocals, or to remove specific resonances from instruments or programme material. When the SQ1 is operating as a noise gate, the side chain filter can be used to select the key frequency in audio that can be used to trigger the operation of the gate, especially useful with drums kits and percussion instruments.




1. **Comp or Gate** - The SQ1 can operate as either a Compressor (Comp) or Gate.
2. **Threshold** - The threshold determines when the compressor starts to compress the signal.
3. **Comp Gain** - Adds gain to compensate for the gain reduction.
4. **Ratio** - The ratio control determines how much compression is applied to the signal.
5. **Attack** - Controls the speed that the compressor responds to signal's that exceed the level set by threshold.

6. **Release** - Sets the time taken for the signal to return to normal after the input level has fallen below threshold.
7. **Vintage** - When engaged the Vintage switch delivers an emulation of many older compressor designs with exponential attack and release envelopes.
8. **Hard Knee** - When the Hard Knee response is used with RMS mode, the compressor operates in a more clinical way with a more defined transition between under threshold and over threshold.
9. **Input Meter** - 28 LEDs display stereo input level.
10. **Output Meter** - 28 LEDs display stereo output level.
11. **Gain Reduction Meter** - 28 LEDs display Gain Reduction level.
12. **SIP (Solo In Place)** - A Solo button is included which will route the side chain signal's directly to the solo bus for monitoring of the side chains frequency response.
13. **Side Chain Frequency** - Sets the Frequency of the side chain with wide a response. 40 Hz to 16 kHz.
14. **Filter** - The side chain filter can be used to select the key frequency component in the audio that can be used to trigger the operation of the gate, especially useful with drums kits and percussion instruments.
15. **Narrow** - Switches the side chain filter to a narrow band.
16. **Wide** - Switches the side chain filter to a wide shape.
17. **Bypass** - Switches SQ1 in/out.




1. **Comp or Gate** - The SQ1 can operate as either a Compressor (Comp) or Gate.
2. **Threshold** - Determines when the gate starts to open.
3. **Gate Hold** - Determines the amount of time the gate is held open after the signal falls below the Threshold.
4. **Range** - The ratio control determines how much compression is applied to the signal.
5. **Attack** - Controls the speed that the gate responds to signal's that exceed the level set by threshold.
6. **Release** - Sets the time taken for the signal to return to normal after the input level has fallen below threshold.

## 8.3.15 Wave Designer

The Wave Designer is a creative tool for modifying signal transients and dynamics. Use it to make a bass drum sound huge in the mix or level out volume inconsistencies of slap bass tracks. Adjusting the Attack control can add punch or suppress exceedingly dynamic signals. Increasing the Sustain control acts in a similar way as a compressor, allowing the peaks to carry longer before decay. The effect can also be used to reduce the sustain for a more staccato sound. The Gain control compensates for level changes caused by the effect.

**Tip:** Try fast attack and sustain settings (turn CCW) on heavily distorted guitars to bring out more definition in individual notes being played.

The Dry/Wet control allows the original signal to be blended back into the signal path to give more artistic control, all perfectly in time. This allows some of the original transients to pass through opening up further imaginative sounds.


1. Attack - Adjust the attack portion of the signal. Turn CW for shorter attack and CCW to allow more transient through.
2. Sustain - Adjust the sustain portion of the signal. Turn CW to shorten the sustain of the signal. Turn CCW to increase the sustained portion of the signal.
3. Gain - Adjust the gain or level of the signal when in manual mode. +-24 dB.
4. Mix - Adjusts the wet/dry balance. Turn fully CW for a completely wet signal.
5. Power - Bypasses the effect.
6. Input Meter - Stereo input metering.
7. Output Meter - Stereo output metering.

# 8.5 EQs

## 8.5.1 MTEC EQP-HD EQ Program Equalizer

Recording engineers claim the Pultec EQP-1a to be the “secret sauce” of sound enhancement. We scrutinized this classic to create an exact physical model that reproduces the multi-faceted sound in painstaking detail. Even the transformers and tube output stage have been faithfully modelled. Inspired by Pultec EQP-1a. It can be divided into 3 discrete bands in a triangle type configuration as highlighted below.

**Band 1** Low Freq determines the curve on which the left-hand Low Boost and Low Atten controls are effective. These controls can be used to either boost or attenuate the signal as required.

**Band 2** Hi Frequency selects the frequency to be changed. Boost increases level at chosen frequency. Bandwidth adjusts the width of the high-frequency boost curves from sharp (narrow) to broad (wide).

**Band 3** Atten and Atten Sel cut the high frequency content at 5 k, 10 k or 20 k.


1. **LF Boost -** This control determines the amount of low shelf gain to be applied to the frequency set by the CPS switch.
2. **LF Atten -** This control determines the amount of low shelf cut to be applied to the frequency set by the CPS switch.
3. **CPS Low Frequency -** This switch determines the frequency of the low shelf portion of the equalizer. Four frequencies are available: 20, 30, 60, and 100 CPS. CPS stands for Cycles Per Second, which is now more commonly referred to as Hertz and abbreviated as Hz.
4. **Bandwidth -** This control sets the proportion of frequencies surrounding the centre frequency (determined by the KCS switch) to be affected by the high boost (this is a bandwidth control).
5. **HF Boost -** This control sets the amount of gain for the high frequency portion of the equalizer.

6. **HF Atten -** This control determines the amount of high shelf cut to be applied to the frequency set by the Atten Sel switch.
7. **KCS High Frequency -** This switch determines the frequency of the high boost portion of the equalizer. Seven frequencies are available: 3, 4, 5, 8, 10, 12, and 16 KCS. KCS stands for Kilo Cycles per Second, which is now more commonly referred to as Kilohertz and abbreviated as kHz.
8. **Atten Sel -** This switch (ATTEN SEL) determines the frequency of the high frequency attenuator. Three frequencies are available: 5, 10, and 20 KCS.
9. **On/Off -** Switch the effect out of circuit.
10. **Bypass -** Bypass the EQ but leaves the sonic character of the unit active.

## 8.5.2 MTEC MEQ-HD Mid-Range Equalizer

Inspired by Pultec MEQ5 Mid-Range Equalizer with its richly colourful tube-amplified sound. With two bands of midrange boost and one band of midrange dip, the MEQ-5 is designed to enhance and control the “power region” where sound energy is often concentrated.

Pultec wrote the book on passive equalization. By digitally “rebuilding” every aspect of the original Pultec classic, we captured the very essence in our parametric equaliser. Our digital re-incarnation is based on the original model and authentically emulates the smooth character of the rather unique components.

Low frequency, mid frequency and hi frequency all dictate at which frequencies the lo boost, mid cut and hi boost push controls work.


1. **Low Mid Gain -** The Low Mid Peak band adds up to 10dB of boost.
2. **Low Mid Frequency -** 5 Frequency choices: 200 Hz, 300 Hz, 500 Hz, 700 Hz and 1000 Hz.
3. **Mid Gain -** The Mid Dip band offers up to 10dB of attenuation.
4. **Mid frequency -** 11 Frequency choices: 200 Hz, 300 Hz, 500 Hz, 700 Hz, 1 kHz, 1.5 kHz, 2 kHz, 3 kHz, 4 kHz, 5 kHz and 7 kHz.
5. **Hi Mid Gain -** The Hi Mid band offers 5 cutoff points and 10dB of Peak (boost).
6. **Hi Mid frequency -** 5 Frequency choices: 1.5 kHz, 2 kHz, 3 kHz, 4 kHz and 5 kHz.
7. **Bypass -** On/Off control but leaves the sonic character of the unit active.
8. **On/Off -** Bypass the EQ completely.

## 8.5.3 Dynamic EQ

The dynamic EQ is a 4-band parametric dynamic equaliser, which is able to provide frequency selective compression or expansion. The dynamic EQ features proportional-q filters that, when boosting or cutting by small amounts, reduce the bandwidth of the filter compared to the setting at maximum cut/boost. Filter coefficients are calculated at the audio rate to provide a lightning fast attack time, which is essential for transparent operation. Each band features a full-band EQ type that switches out the EQ filter so that the band operates as a non-frequency selective, or 'full-band' compressor/expander. The Fixed button turns any dynamic EQ band into a non-dynamic EQ if required. The amount of gain is fixed as set by the controls.


1. **Close Effect -** Closes the currently open effect.
2. **Input Meter -** 20 LEDs, which show input metering.
3. **Comp Gain Reduction Meter -** 20 LEDs, which show the selected band gain reduction metering.
4. **EXP Expansion Meter -** 20 LEDs, which show the selected band expansion metering.
5. **Threshold -** Turn to set the signal threshold level above (or below) which the effect is to take place. Turning anti-clockwise reduces the signal level needed to exceed the threshold, and therefore increases sensitivity. Signal’s exceeding threshold initiate the selected processing. Turning clockwise increases the signal level needed and reduces sensitivity. When set fully clockwise, the threshold is set to +20dBu.
6. **Freq -** Turn to set the centre frequency of the dynamic activity. Note that the range of each band is 40 Hz to 18 kHz.
7. **Ratio -** Sets the Ratio for compression or Expansion.
8. **Width -** Controls bandwidth or ‘Q’. Turn clockwise to widen or defocus the affected band of frequencies. Turn anti-clockwise to narrow or sharpen the affected band.
9. **Bell -** Switches the selected band to a bell shape.
10. **LF Shelving -** Switches selected band to a Low Frequency shelving response.
11. **HF Shelving -** Switches selected band to a High Frequency shelving response.

12. **Flat -** Switches selected band to a flat (wide-band) response.
13. **Below -** When the switch is depressed to select Below the threshold's sensitivity setting remains the same, but the processing is 'turned on its head' as compression or expansion is now only applied to signal's which are below the threshold. Below threshold compression now works like a frequency sensitive gate in that the further below the threshold the signal is, the more it is attenuated. There being progressively less attenuation as signal level increases to approach the threshold setting. Below threshold expansion acts to boost the selected frequency range up to (but not above) the threshold.
14. **Fast Rel -** When depressed for Fast, the compress/expand release time is quicker.
15. **Fixed -** Makes each band into a fixed EQ and removes the dynamic function.
16. **Listen -** Long press this button to latch it on and send the side chain signal of that band to the solo bus. This will include any filtering and allows you to set up the signal that will subsequently be removed or expanded by the dynamics processing.
17. **IN -** Activate each band of dynamic EQ individually.
18. **ON -** Turns the whole dynamic EQ on/off.

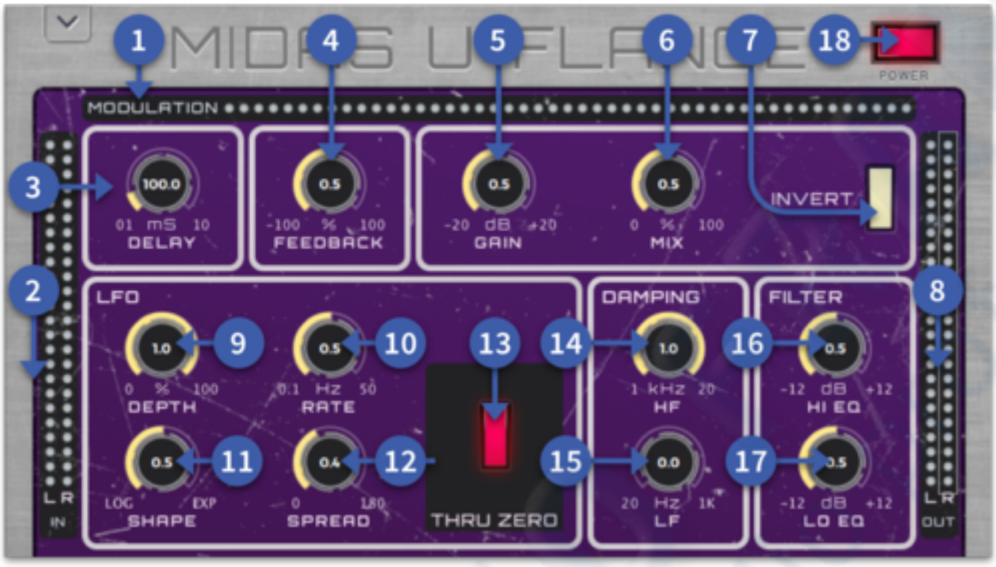
## 8.5.4 XL4 EQ

The XL4 EQ is **identical** in operation and sound to the Midas channel EQ. Details on its use can be found in the Basic Operation chapter. It can be used to add extra bands of Midas EQ to an input or output if required.

# 8.6 Modulation

## 8.6.1 Midas U Flange

The flanger effect consists of one or, if configured as stereo, two-tap delay lines. One tap is fixed, and the other tap position is modulated to provide 'thru-zero' flanging or single tap modulation when 'thru-zero' is off. The perfect effect for that 80's guitar solo or to give flat keyboard sounds some movement.




1. **Modulation Meter** - A single row of 48 orange LEDs is used to show the modulation position.
2. **In Meters -** Two rows of 24 LEDs, which show input metering.
3. **Delay -** Adjust length of modulated delay line in milliseconds. In 'thru-zero' mode, also sets the delay of the dry path. Range is 0.1 to 10, with 5 at top dead centre.
4. **Feedback -** Adjusts the amount of negative/positive feedback applied to the delay. Controls the number of repeats. Range is from -100% to +100%, with 0% at top dead centre.
5. **Gain -** Adjusts the signal level in dB. Range is from -20dB to +20dB, with 0dB at top dead centre.
6. **Mix -** Adjusts the mix between dry (0%) wet (100%).
7. **Invert -** Inverts the wet signal's polarity.
8. **Out -** Two rows of 24 LEDs, which show output metering.
9. **Depth -** LFO section control for adjusting the intensity of the effect by setting the depth of modulation as a percentage. Interactive with Delay, as for Chorus. Range is 0% to 100%.
10. **Rate** - LFO section control for adjusting the rate of modulation (Hz). Range is between 0.01 and 50, with 0.5 at top dead centre.
11. **Shape** - LFO section control for adjusting the shape of the modulation waveform. Range is from Tri to Exp.
12. **Spread** - LFO section control for setting the relative phase of left/right modulation. Range is 0 to 180.

13. **Thru Zero** - LFO section control for selecting ‘thru zero’ or normal mode. Illuminates to indicate switch is on.
14. **HF** - Damping section control for adjusting the high frequency (kHz) tuning of flanger feedback. Range is 1 kHz to 20 kHz.
15. **LF** - Damping section control for adjusting the low frequency (kHz) tuning of flanger feedback. Range is 20 Hz to 1 kHz.
16. **Hi EQ** - Filters section control for adjusting the amount of HF (high EQ) cut or boost applied to the effect output (in dB). Range is -12 dB to +12 dB with 0 dB at top dead centre.
17. **Lo EQ** - Filters section control for adjusting the amount of LF (low EQ) cut or boost applied to the effects output (in dB). Range is -12 dB to +12 dB with 0dB at top dead centre.
18. **Power** - Switches the flanger effect on and off. Illuminates in red when power is on.

## 8.6.2 Midas U Phase

The phaser effect consists of one, or if configured for dual operation, two stereo phasers connected in serial/parallel according to mode setting. This effect is fabulous at adding beatlesque creative movement to vocals.


1. Modulation Meter - A Pair of 25 orange LED segments are used to show the modulation on each channel.
2. Input Meters - Two rows of 24 LEDs, which show input metering.
3. Rate - Controls the rate of modulation in Hz. Range is from 0 Hz and 40 Hz.
4. Manual - Sets the sweep offset for performing manual sweep. Range is from 500 Hz to 24 kHz.
5. HI EQ - Adjusts the amount of HF (high EQ) cut or boost applied to the effect output (in dB). Range is from -12 dB to +12 dB with 0 dB at top centre.
6. LO EQ - Adjusts the amount of LF (low EQ) cut or boost applied to the effects output (in dB). Range is from -12 dB to +12 dB with 0 dB at top centre.

7. Mode - Selects the operating mode: single; dual series; dual parallel; linked series; or linked parallel. When linked, modulation of phasers 1 and 2 are linked.
8. Sweep - Sets the modulation waveform shape.
9. Stages - Selects the number of all pass stages, which sets the number of notches in the frequency response.
10. Depth - Controls the intensity of the effect by setting the depth of phasing filters. Range is from 0% to 100%.
11. Spin - Adjusts the amount of relative phase of left/right modulation. Range is from 0% to 100%.
12. Feedback - Adjusts the amount of negative/positive feedback applied to the delay. Controls the number of repeats. Range is from -100% to +100%, with 0% at top dead centre.
13. Blend - Adjusts the mix between dry (0%) wet (100%).
14. Gain - Adjusts the signal level (dB). Range is -±20dB, with 0dB at top centre.
15. Out - Two rows of 24 LEDs, which show output metering.
16. Power - Switch the Phaser effect on and off. Illuminates red to indicate effect is on.

## 8.6.3 Pitch Shifter

The stereo pitch shifting effect is often used in two different ways. One is to set the Mix knob lower and only use the Cent knob to make a small off set in pitch between the wet and dry sounds. This results in a "voice doubling" effect that thickens the overall sound in a subtler way. The extreme use of the effect is to turn the Mix knob fully clockwise, so the entire signal is affected. This way, the signal can be shifted into other keys up to an octave above or below the original. When used on a voice, this results in a chipmunk sound or a low Darth Vader effect. When the Semi and Cent controls are set at 12 o'clock, the pitch is not altered. Making adjustments by semitone will have a very pronounced effect, whereas changes to the Cent knob will be very minor. The Delay control creates a time difference between the wet and dry sound. The HI CUT controls allow the effected signal to be band-limited. The Dual Pitch effect allows the left and right channels to be adjusted independently and allows panning of the two channels. The Mode button switches between Modern or Vintage options. Vintage mode adds a feedback control which allows shimmer style pitchshifting as the signal is sent back into the effect. As you turn the control CCW more signal is sent back through.


1. **Semi -** Adjusts the pitch shifting amount in whole tones. Range is from -12 to +12, with 0 at centre.
2. **Cent -** Cent tunes the pitch shifting in 1% increments of a whole tone. Range is from -100 to +100, with 0 at top dead centre.
3. **Delay -** Sets the delay time before the pitch shift. Range is from 1 ms to 2 s.
4. **Level -** Sets the level in %. Range is from 0 to 100%.
5. **Pan -** Adjusts the position of the individual channel signal in the unit's stereo output.
6. **HF Cut -** Adjusts the HF attenuation of delay repeats. Range is from 2k to 20k.
7. **Mode** - Toggle between Modern or Vintage mode.
8. **Mix** - Controls the balance between dry signal and effect. Range is from 0% to 100%, with 50 at centre.
9. **Input and Output Meters** - Shows the input/output signal levels on dual 14-segment meters.
10. **Bypass -** Switches pitch shifter effect on/off.
11. **Feedback-** Repeats the pitch shifted effect to add a shimmer type sound. Range is from 0 to 100%.

## 8.6.4 Chorus

The chorus effect makes one sound source (such as a voice) sound like many such sources singing (or playing) in unison. Since performance in unison is never precise, chorus effects simulate this by making independently adjusted duplicates of the input signal and mixing them back in with the original sound.


1. **Input meters -** Two rows of 14 LEDs, which show input metering.
2. **Stereo Input -** Activate for use on stereo sources.
3. **On/Off -** Effect bypass button.
4. **Slow -** A Slow modulation effect preset.
5. **Deep -** A Deep shift effect preset.
6. **Medium -** Standard setting for creating a rich chorus.
7. **Fast -** Fast effect preset for quick accentuation of sounds.
8. **Depth -** Affects the depth of the chorus from 0% to 100%.
9. **Rate -** Controls the rate of modulation in Hz. Range is from 0.1 Hz to 2.0 Hz.
10. **Width -** Fully CCW mono's the effect, fully CW makes the stereo effect more pronounced.
11. **Mix -** Adjusts the wet/dry balance. Turn fully right for a completely wet signal.
12. **Out -** Two rows of 14 LEDs, which show output metering.

## 8.6.5 Rotor Motor

Rotary Motor emulates the sound of a Leslie rotating speaker. The Rotary Motor provides more flexibility than the original and can be used with a variety of instruments, and even vocals, to create a whirling, psychedelic effect. The Low Speed and High-Speed controls adjust the rotational speed of the Slow and Fast Speed selection and can be toggled with the Slow/Fast switch. The Acceleration controls adjusts how quickly the speed increases and decreases from the Slow mode to the Fast mode. The rotation effect can also be disengaged with the Stop switch, which will stop the movement of the speakers. Distance adjusts the distance between the Rotary speakers and the virtual microphone. Balance adjusts the level between the low and high virtual microphones.


1. **Low Speed -** Adjusts the speed of Slow mode. Range 0.1 Hz - 4 Hz.
2. **High Speed -** Adjusts the speed of Fast mode. Range 2 Hz - 10Hz.
3. **Acceleration -** The Acceleration control adjusts how quickly the speed increases and decreases from the Slow mode to the Fast mode. 0.0% to 100%.
4. **Distance -** Distance adjusts the distance between the Rotary speakers and the virtual microphones. 0.0% to 100%.
5. **Balance -** Adjusts the balance between the virtual horn and virtual drum controlling the signal tone. -100 (drum) to 100.0 (horn).
6. **Mix -** Controls the mix (or ratio) of wet (processed) and dry (unprocessed) signals. 0.0% to 100%.
7. **Motor -** Allows The rotation effect of the motor to be disengaged (Stop).
8. **Speed -** Selects either the slow or fast speeds for rotation.
9. **Input Metering -** Metering shows either input level.
10. **Output Metering -** Metering shows either output level.

## 8.6.6 Ultra-Dualistic Voice Doubler

The Ultra-Dualistic Voice Doubler can be used to make a vocal sound double tracked (also known as ADT Auto Double Track) and was widely used from the 80's onwards to make vocals sound thicker and larger than life. This effect is based on a legendary processor that gives more realism to the vocal processing by modulating each side of the delayed signal. It can also be used to add warmth and width to keyboard or string parts.


1. **Input Meter -** Stereo input metering.
2. **Time Left -** Adjusts the delay time of the left voice from 1-25 ms.
3. **Time Right -** Adjusts the delay time of the right voice from 1-25 ms.
4. **Level Left -** Adjusts the left effect level from ∞ to 10 dB.
5. **Level Right -** Adjusts the right effect level from ∞ to 10 dB.
6. **Pan Left -** Adjusts the pan position of the left channel.
7. **Pan Right -** Adjusts the pan position of the right channel.
8. **Mod Depth L -** Changes the modulation depth of the left channel. 0-30%.
9. **Mod Depth R -** Changes the modulation depth of the right channel. 0-30%.
10. **Mod Rate -** Adjust the overall modulation depth from 0 to 5 Hz.
11. **Mix -** Adjusts the wet/dry balance. Turn fully right for a completely wet signal.
12. **Gain -** ± 20 dB gain control of the output.
13. **Power -** Bypasses the effect.

# 8.7 Distorsion / Exciter

## 8.7.1 Enlightenment Bass

The Enlightenment Bass adds rich harmonic content based on the lowest fundamental in order to increase the perception of bass in the mix.

Adjust Intensity to around -6dB to start. A small amount of Intensity goes a very long way. Excessively high settings will increase the perceived bass level but may result in bass distortion!


1. **Input Metering -** Metering shows input level.
2. **Power On -** Bypass the effect.
3. **Intensity -** Controls the amount of harmonic content created.
4. **Crossover Frequency Display -** Shows the crossover frequency. 32 Hz to 245 Hz.
5. **Crossover Frequency -** Sets the frequency at which the effect starts to work (below the crossover point).
6. **Sub Level -** Adjusts the amount of sub bass in the signal path.
7. **Output Gain -** Adjust the overall level of the signal.
8. **Solo -** Solo the sub signal to fine tune the sound.
9. **Output Metering -** Metering shows output level.

## 8.7.2 Glow

From the depths of Mordor the Glow effect brings intense heat and grit to your sounds. Used on a drum group, Glow can add anything from a dash of saturation and glue to an enriched harmonic explosion. You can use it to add a boost to the low mid frequency content and can add a tiny bit of crunch to poke this area through in the mix. Turn up the input trim and balance with the output trim to add more harmonics and compression. Heat can be used to add a tube like glow while intense adds pleasing saturation to your signal.

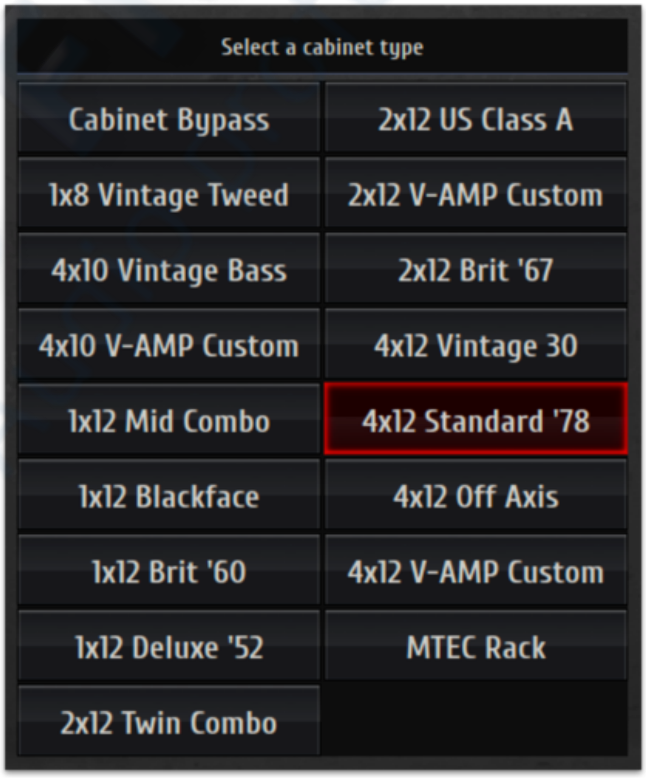

1. **Input Meter** - Magma meters, which show input metering.
2. **Glow** - Controls the amount of glow magma glue being applied. Range 1 - 5.
3. **Type** - There are two mystical harmonic types. Heat and Intense.
4. **Drive** - Adjusts the harmonic drive from 1 to 5.
5. **Input Trim** - Adjust the trim level into the effect. Range +-12 dB.
6. **Mix** - Adjust the wet/dry mix of the band from 0% to 100%.
7. **Output Trim** - Adjust the trim level output of the effect. Range +-12 dB.
8. **On/Off** - On/Off control for the distortion.
9. **Output** - Magma meters, which show output metering.

.

### 8.7.3 Rack Amp

The Rack Amp is designed to reproduce the sound of plugging into a guitar amplifier. Great attention to detail has been taken in modeling the 16 cabinet types to emulate many classic guitar and bass amps. Feeding any Guitar or Bass DI signal into the effect gives great creative scope to control tone. The effect also works on stereo sources like keyboards and can be used to add some extra grit to the sound.


1. **Input Meter -** 14 Stereo LEDs showing input metering.
2. **Cabinet Type -** Choose from the 16 different cabinet types.
3. **Preamp -** Controls the amount of input gain prior to distortion.
4. **Buzz -** Changes the low-end breakup.
5. **Punch -** Adjusts the midrange distortion.
6. **Crunch -** Modifies the high-frequency content and distortion for smooth or cutting notes.
7. **Drive -** The Drive control simulates the amount of power amp distortion from a tube amp.
8. **Level -** The overall output is controlled by the Level knob.
9. **Low EQ -** Low EQ is a shelf that cuts or boosts below 200 Hz kHz by ± 12 dB.
10. **High EQ -** High EQ is a shelf that cuts or boosts above around 1 kHz by ± 12 dB.
11. **Output Meter -** 14 Stereo LEDs showing output metering.
12. **Power -** Bypasses the effect.

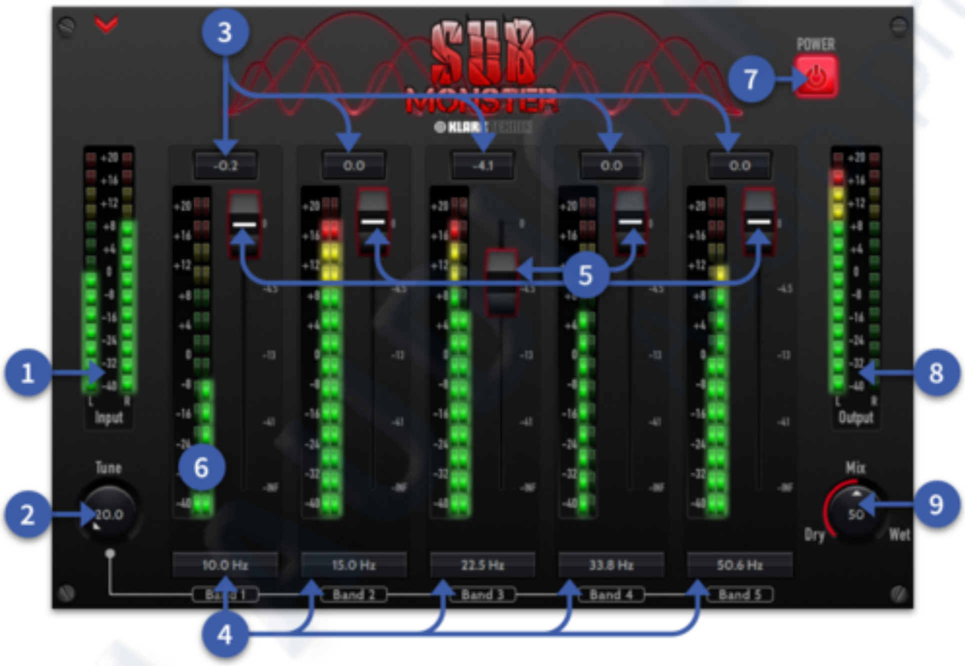


## 8.7.4 Sub Monster

The Submonster is a sub-harmonic synthesizer effect which boosts the low frequencies of an audio signal by generating an additional signal an octave below a given frequency range. The aim of this effect is to add low end to a signal which may have weaker low frequency content or to generally increase the bass frequencies.

The Submonster works particularly well on drum tracks, especially bass drums, and when applied to an overall mix. When placed on individual instruments it can create a sub-octave doubling effect, which works well when applied to monophonic sounds. The effect has five tunable bands of sub-octave synthesis, each with its own gain control. The frequencies on which the bands are centered are controlled using the tune parameter, which controls all five bands center frequencies to reduce overlap.

The tune parameter ranges from 20 Hz to 60 Hz which corresponds to the frequency of the lowest generated sub-harmonic band, with each subsequent band having a minimum and maximum frequency of 1.5x the previous band's minimum and maximum frequency. This gives a total range of sub-harmonic synthesis between 10 Hz and 150 Hz.




1. **Input Meter -** 14 LEDs, which show input metering.
2. **Tune -** Adjust the target frequency from 20 Hz to 60 Hz. Harmonics are created by generating an additional signal an octave below the selected frequency range.
3. **Band 1 - 5 Gain Display -** Shows the amount of added sub-harmonic gain per band.
4. **Band 1 - 5 Frequency -** Displayed frequency of created sub-harmonic: Band 1 = 10 Hz - 30 Hz, Band 2 = 15 Hz to 45 Hz, Band 3 = 22.5 Hz to 67.5 Hz, Band 4 = 33.8 Hz to 101.2 Hz, Band 5 = 50.6 Hz to 151.9 Hz.
5. **Band 1 - 5 Fader level -** Set level of the chosen band from -40 dB to 0.00 dB.
6. **Band 1 - 5 Meters -** 14 LEDs, which show band level metering.
7. **On/Off -** Turns the effect on or off.
8. **Output Meter -** 14 LEDs, which show input metering.
9. **Mix -** Adjusts the Wet/Dry blend of the effect when inserted on a channel

## 8.7.5 Tape Saturation

As its name suggests, the Tape Saturation effect emulates analogue tape saturation; when the number of magnetised particles required to fully record and reproduce an audio signal exceeds the amount available. This can be heard as analogue 'warmth', and similar effects can be achieved in the digital domain by emulating various analogue tape artefacts.

An example of this is tape's tendency to compress the high frequencies in 'transients', or peaks in the audio signal, and while this is a side-effect of using analogue tape, it can also be desirable when trying to achieve a vintage tape sound. This effect can be altered by use of the Transient Smoothing control. Another way to achieve that nostalgic tape sound is by the creative use of biasing; most professional tape machines are set up to compensate for this, in other words, to be slightly overbiased.

The Overbias control is used to emulate this by reducing the amount of tape distortion at the expense of the high frequencies and transients. The limit of the high frequency response is also controlled by the overall frequency response of the tape process, and this in turn is affected mainly by utilising different tape speeds. Slower tape speeds 3¼ or 7½ ips) have less high frequency definition and a boost at low and mid frequencies, whereas higher speeds (15 / 30 ips) have better high frequency representation and less extreme lower ends resulting in more accurate audio reproduction.

Another important element to tape emulation is the output transformer which supplies a low end 'bump' in the frequency response and increases harmonic distortion of frequencies between approximately 50 – 100 Hz. The amount of distortion can be controlled by effective use of the Transformation control.


1. **Input meter -** 14 LEDs, which show input metering.
2. **Input Gain -** Sets the level into the tape machine.
3. **Input Drive -** Increases the effect of tape saturation applied.
4. **Overbias -** Over bias is used to emulate nostalgic tape sound. The Over bias control is used to emulate this by reducing the amount of tape distortion at the expense of the high frequencies and transients.
5. **Transformation -** The amount of transformer drive applied. 1-10 Range.

6. **Tap Speed -** Changes the tape speed. 3¼, 7½, 15 or 30 ips.
7. **Transient Smooth -** Changes the speed in which Transients are handled to compress the high frequencies in a subtle way. Fast, Med or Slow options.
8. **Transformer -** Changes the speed in which Transients are handled to compress the high frequencies in a subtle way. Fast, Med or Slow options.
9. **Effect -** Toggles the effect on/off.
10. **Output Drive -** Sets the output level from -12 to +12 with 0 at top dead centre.
11. **Output Meter -** 14 LEDs, which show output metering.

## 8.7.6 UNCL.HD MULTIBAND DISTORTION

This effect goes to 11, well actually to 12. Which is 2 louder than 10. It's great for adding subtle warmth to full on distortion to any input. Guitar amp simulation can also be added to further increase the sonic palate available to the engineer. There are three bands, adjustable by a 24 dB per octave crossover filter. Each band has an automatic compressor function controlled by the 'Squash' parameter, to add more punch to the sound before it goes through the distortion.

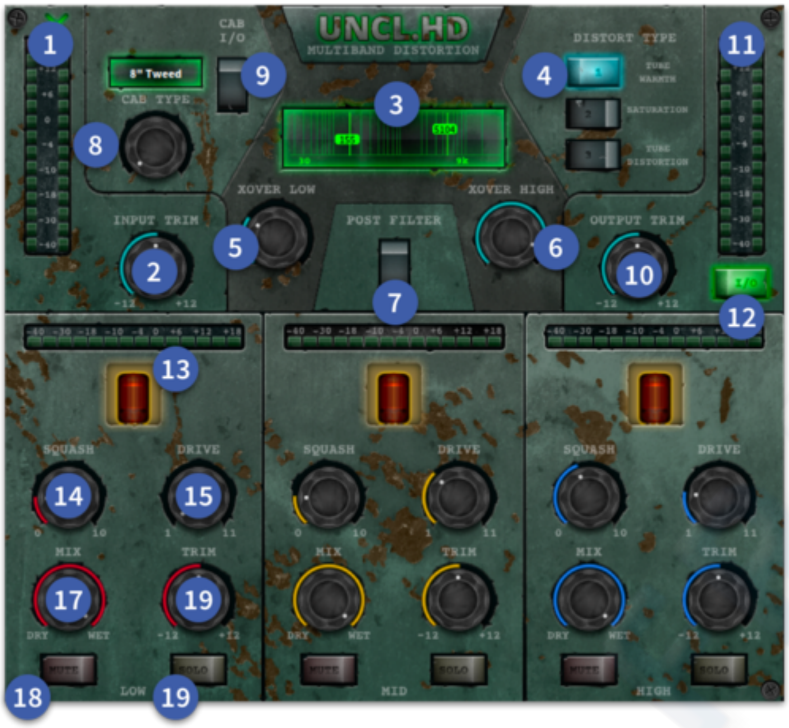
The 'Drive' parameter controls the amount of distortion introduced to the sound. This parameter, together with the right distortion type, can instead bring some soft saturation to the sound.

In the bottom section there is a Mix control and a Trim parameter for each band, to further balance and manipulate the effect. Moreover, the mute/solo buttons remove or isolate a band for more precise sound design.

There are three distortion types. going from soft saturation to more aggressive distortion, and, apart from monitoring the level of the sound on each band, can also be used to monitor the distortion applied to it with the three bands provided.

The effect also features a Post Filter section, in case additional control is required over the extra harmonics created by the distortion, and a Cabinet Unit applied to the output, which can add the characteristic timbre of 11 different cabinet types.

Finally, the level of the sound can be controlled by the input and output gain parameters.




1. Input meter - 14 LEDs, which show input metering.
2. Input Trim - Adjust the trim level into the effect. Range +-12 dB.
3. Frequency Range - Displays the frequency range of each band.
4. Type - There are three distortion types. Tube Warmth, Saturation and Tube Distortion.
5. X-Over Low - Sets the low band range, adjustable with a 24 dB per octave crossover filter. 30 Hz to 9 kHz.
6. X-Over High - Sets the high band range, adjustable with a 24 dB per octave crossover filter. 30 Hz to 9 kHz.
7. Post Filt - The Post Filter section can be used for extra control over the extra harmonics created by the distortion.
8. Cab On - Activates the guitar amp cab simulation.
9. Cab Type - A guitar cabinet can be applied to the output, which can add the characteristic timbre of 11 different cabinet types which are: 1 = 8" Tweed. 2 = 4x10" Bass. 3 = 4x10" Custom. 4 = 12" Mid Combo. 5 = 12" Black. 6 = 12" Brit'60. 7 = 4x12" Brit'96. 8 = 4x12" Std'78". 9 = 4x12" Off Axis. 10 = 4x12" Custom. 11 = Rack Amp.
10. Output Trim - Adjust the trim level output of the effect. Range +-12 dB.
11. Output - 14 LEDs, which show input metering.
12. On/Off - On/Off control for the distortion.
13. Meter (High/Mid/Low) - Displays the amount of compression occurring via LED and tube warmth.
14. Squash (High/Mid/Low) - Controls the amount of compression being applied.
15. Drive (Low/Mid/High) - Adjusts the drive for each band from 1 to 11 (1 louder than 10).
16. Mix (Low/Mid/High) - Adjust the wet/dry mix of the band from 0% to 100%.
17. Trim (High/Mid/Low) - Adjust the level of the band. Range +-12 dB.
18. Mute (Low/Mid/High) - Mutes the selected band.
19. Solo (Low/Mid/High) - Solos the selected band for audition.

## 8.7.7 M-Harmonics

This effect adds rich harmonics to signals. From vintage tones to full on tube style fuzz. Choose a blend of 2nd or 3rd harmonics to add depth to drums or other sources. The Drive control increases the amount of the effect. The colour allows tonal changes to be made. Turning the colour control CCW increases bass information while turning CW adds more treble detail.


1. **Input Meter -** LED Metering shows input level.
2. **Input Meter VU -** VU Metering shows input level.
3. **On/Off button -** Acts as bypass when needed.
4. **Output meter -** Metering shows output level.
5. **Input Gain -** ∞ to 0 dB. Input gain control.
6. **Harmonics -** Turn CCW to add 2nd harmonic content. Turn CW to add 3rd harmonic to the signal.
7. **Drive -** Adding drive adds more of the selected harmonics to the signal. From 10% to 100%.
8. **Colour -** Turn CCW to increase low end content, turn CW to add high end content
9. **Mix control -** Adjust the wet/dry mix of the band from 0% to 100%.
10. **Output Gain -** ∞ to 0 dB. Output gain control.

# Chapter 16: Talk and Mute Groups

## 16.1 Talk Groups

The HD96-24 has 12 Talk Groups which can be used to directly communicate with pre-determined groups of paths using the dedicated talkback input or another pre-defined microphone input. Both inputs and outputs can utilise the Talk Group system. For example, when using in ear monitors have a talk group for the backline techs and another for the band for independent communication.

**Tip:** Talk In can be used for your local talk microphone. The Talkback input can be used to send a microphone from another person or console, for example, a Front of House (FOH) engineer's microphone can be feed into the Talkback input of the monitor engineer's console which allows both the Monitor engineer and FOH engineer to talk to the stage.

For more complex talkback set ups the shout mixer may help (see Chapter 13 Shout Configuration).


To set up Talk Groups:

1. Select the Groups icon from the top or side bar menu.
2. Press the Talk icon in the left-side to open the Talk Groups Selection page.
3. Select the Talk Group you wish to edit. It now can be named, coloured and given a different icon if required by pressing on the large icon on the right-hand side of the page. There is an option to Transfer Colour to channels if desired and also the ability to follow tags (any channels added with the same tag will automatically be added to the talk group).

4. Touch the channels you wish to add to the Talk Group. To the right are tabs for input and output page selection. A maximum number of 32 inputs and outputs can be displayed at once in a Talk Group, swipe right to view channels 33 onwards.
5. To remove a channel from a Talk Group simply hold onto the desired icon until it wiggles and drag it away from the Drag & Drop area, or tap the icon used to initially assign it to the group.
6. Transfer the colour of the talk group to the channels within it.
7. Remove Gaps If there are blank channels in the selected talk group subsequent channels will be moved down to fill the gaps.
8. Order Ascending re-orders talk group to be in numerical order (lowest channel first).
9. Order Descending re-orders the talk group in reverse numerical order (highest channel first).
10. Pressing and hold the Empty Group button for a short time until the line completely traces around the outside of the button to clear a talk group.
11. Select Inputs or Outputs.
12. Tabbed selection of input or outputs.
13. The name, tags and icon of the Talk Group can be edited in two places.
14. Turn the selected Talk Group ON if needed.

To use Talk Groups:


1. In the Global Assignable Shortcuts area allocate the Talk Groups you wish to use to a shortcut destination.
2. Once a talk group is assigned press the SEL button to talk to the selected paths.

# 16.2 Mute Groups

The HD96-24 has 12 dedicated Mute Groups that can be used alongside VCA mutes to turn on and off selected paths or channels. For example, muting multiple instruments on stage or turn effects off at the end of songs.

To set up Mute Groups:


1. Select the Groups icon from the top or side bar menu.
2. Press the Mute icon in the left-side to open the Mute Groups Selection page.
3. Select the Mute Group you wish to edit. It now can be named, coloured and given a different icon if required by pressing on the large icon on the right-hand side of the page. There is an option to Transfer Colour to channels if desired and also the ability to follow tags as described earlier.
4. Now touch the channels you wish to add to the Mute Group. To the right are tabs for input and output page selection. A maximum number of 32 inputs and outputs can be displayed at once in a Mute Group, swipe right to view channels 33 onwards. To remove a channel from a Mute Group, hold the desired icon until it wiggles and drag it away from the Drag & Drop area, or tap the same icon used to initially add the path.
5. Transfer Colour of the Mute Group to all channels contained within it.
6. Remove Gaps If there are blank channels in the selected talk group, they will be removed with subsequent channels moved down to fill the gaps.
7. Order AscendingThis function re-orders talk group to be in numerical order (lowest channel first).
8. Order Descending This function re-orders the talk group in reverse numerical order (highest channel first).
9. Empty the Mute Group of all content by pressing and holding the Empty Group button for a short time until the line completely traces around the outside of the button.
10. Select Inputs or Outputs to be added to a Mute Group.
11. Tabs selection of channels.

12. There are two areas in which Mute Groups can be named, coloured or given an icon.
13. Tap to edit Name, Colour or Icon.
14. Turn the selected Mute Group ON if needed.


To use Mute Groups:

1. In the Global Assignable Shortcuts area allocate the Mute Groups you wish to use to a shortcut destination.
2. Once a Mute Group is assigned press the SEL button to mute the nominated group. The SEL button and the LCD display will change to show the state of mute/unmute. You will also see all channels in the Mute group turn red to indicate they are muted or off in the GUI and on the surface if currently displayed.

# Chapter 17: Copy and Paste

## 17.1 Using Copy and Paste

The copy and paste button (upper-right corner of the widget) lets you copy the parameters of one single channel's processing areas such as the Configuration, EQ, Compressor, Gate or Sends and paste them to one of the channels of a similar type. This is a GUI only feature.

## 17.2 Copy and Paste rules and restrictions

Only the same channel type can be copied and pasted. For example, you cannot copy from an aux and paste to a matrix or you can't copy the compressor or EQ from one input channel to an output compressor or EQ.


To copy an individual processing area to another channel such as EQ, Gate or Comp:

1. Navigate to the Channel View or FOH widget and select either the Config, Equaliser, Gate, Compressor or sends section.

2. Press the copy and paste icon in the upper right side. The icon will turn yellow to indicate the copy function is active. The selected channel will also turn yellow. The copy and paste drop down box will appear under the home workflow.
3. To copy the individual processing area to another channel, select which areas you wish to copy, then select the channel you wish to paste to in the channel overview area. This will shade in a lighter yellow. You can paste to as many channels as you want using the bank navigation to select any channel in the system. Press Paste Selected to confirm the action.
4. Press the Copy and Paste icon again to exit copy and paste mode. The source channel will then return to its default blue colour.
5. The Copy All button allows you to copy a complete channel to other channels of the same type.
6. Cancel closes the copy and paste window. It does not undo the copy and paste function.

## 17.3 Channels Versus Scenes

The fundamental difference between copying through channels and copying through scenes is that the former is location-based, while the latter can be thought of as being time-based. However, the areas (and parameters) that are copied across are similar.

## 17.4 Copy and Paste Parameters

**CONFIG** – Preamp Gain, Trim (not 48v), Polarity, Delay time and on/off state, talk bus setting, aux return state, direct in out, processing order, Insert and link options.

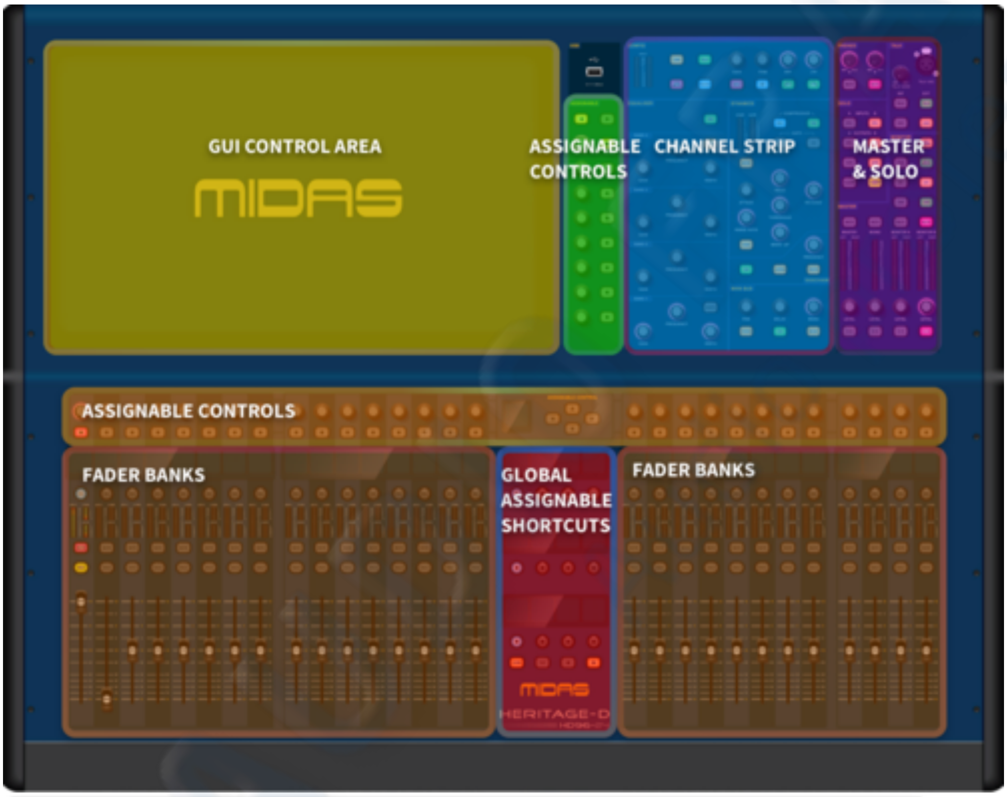
**EQ** – Equaliser parameters.

**FILTERS** – Phase, LPF and HPF parameters. **GATE** – Gate and side chain parameters. **COMP** - Compressor and side chain parameters. **SENDS** – Send levels, pan, on/off state, pre/post fade, tap off points. **MAN BUS** – Pan, Level to Master and Mono bus and on/off state. **MUTES** – Mute status (not including Mute Groups).

# Chapter 18: Assignable Controls & Shortcuts

## 18.1 About the Assignable Rotary

The Assignable Controls found above the fader banks have a variety of functions which can be navigated via the 4 Assignable arrows found above the Global Assignable Shortcuts area and can be fully customized in the Navigation Page.




## 18.2 Controlling the Assignable rotary controls

The Assignable Controls can be scrolled through by using the up and down arrows. The left and right arrows open up various different settings associated to the selected item. As standard there are 8 pages of functions with extra parameters accessed via the left and right arrow buttons. Up to 20 pages maximum can be assigned with various different functions as needed.

Orange = to stereo bus

| Rotary Control | Rotary Function | Button Type | Button Function |
|---|---|---|---|
| Pan | Adjust Pan position | Stereo | |
| Preamp/Tape Gain | Adjust the Gain | 48V | Red = 48V active |
| HPF/LPF Frequency | Adjust the HPF/LPF Freq | HPF/LPF On | Toggle the HPF/LPF On/Off |
| Comp Thresh | Adjust the Threshold | Comp On | Toggle the Comp On/Off |
| Comp Ratio | Adjust the Ratio | Comp On | Toggle the Comp On/Off |
| Comp Gain | Adjust the Gain | Comp On | Toggle the Comp On/Off |
| Gate Thresh | Adjust the Threshold | Gate On | Toggle the Gate On/Off |
| Gate Range | Adjust the Range | Gate On | Toggle the Gate On/Off |
| EQ Band 1 | Adjust Band 1 Gain | EQ On | Toggle the EQ On/Off |
| EQ Band 2 | Adjust Band 2 Gain | EQ On | Toggle the EQ On/Off |
| EQ Band 3 | Adjust Band 3 Gain | EQ On | Toggle the EQ On/Off |
| EQ Band 4 | Adjust Band 4 Gain | EQ On | Toggle the EQ On/Off |
| Trim | Adjust the Trim level | Send On | Select Output On/Off |
| Trim | Adjust the Trim level | Send Pre | Toggle Pre/Post fade |
| Trim | Adjust the Trim level | Send AFL | AFL the channel |
| Find | Scroll to change 'Pinned' channel | Pin | Pin Channel to fader |
| Find | Scroll to change 'Pinned' channel | Reset Pin | Reset 'Pinned' channel. |

## 18.3 Pinned Channels



Any channel type can be pinned to any position on the surface.

Use the Assignable Control up and down arrows to navigate to the Find/Pin page.

Once a channel has been pinned, by pressing the button above the LCD, that channel will not change until the pin has been removed. Once a channel is pinned the rotary encoder can be used to select the channel to be pinned. This can be any input, output or VCA.

Remember last pin when turned on (found in preferences, general configuration) allows you to remove a pin channel temporally and add the same pinned channel back in the same position when set again. With this function off, all pins will be reset when removed.

## 18.4 Flip Target


Flip target allows any channels send contribution to an aux or matrix to be pinned to the surface.

Flip Target is ideal, for example, if you would like to keep the level of a vocal microphone send fader level to a monitor speaker aux send to always have direct access to control feedback.

Pin to selected output in preferences needs to be on if you wish to keep a channel pinned in flip when flip mode is turned off.


To remove a pinned channel in flip, turn the rotary control fully counterclockwise until the pin is removed. To use Flip Target.

1. Pin to selected output on in preferences.
2. Pin the input channel you wish to use in the place you wish the fader to be.
3. Press the right arrow in the Assignable Control section to display Flip Target/Pin.
4. Turn the rotary encoder of the pinned channel to the Aux or matrix you require.

In this example Input 37 send contribution to Aux 37 is pinned to the surface.

# 18.5 Global Assignable Shortcuts

Global Assignable Shortcuts are a great way to execute common functions or pre-programmed macros within the console with one button press. Below is a list of functions currently available. This list will increase as the HD system improves over time so be sure to keep your software up to date for the most up to date new features.

To set up the Assignable Rotary & Button functions use the Navigation Page.

1. The current Global Assignable preset (GA) is displayed here.
2. Select the GA preset you wish to edit.
3. Delete a GA preset
4. Duplicate a GA preset
5. Edit takes you to the GA preset set-up page.
6. Set Active makes the selected GA preset operational across all layouts.



1. **Cyclic** – When you reach the end of the list it jumps back to the start for continuous operation.
2. **List** – Scroll up and down within the fixed list to access functions.
3. **Add Page** – To add a new page. A up to maximum of 20 pages can be added, selected by the page tabs.
4. **Remove Page** – Remove the currently selected page.
5. **Rotary Function** – Select the function of the Rotary on the currently selected page.
6. **Button Function** – Select the function of the Button on the currently selected page.

Below is a list of the parameters available in the Navigation Page which can be added.

**Available Rotary Parameters:**

- Pan
- Preamp/Tape Gain
- HPF/LPF Frequency
- Compressor Settings
- Gate Settings
- EQ Gain
- Trim

- Delay
- Find
- Pinned Channel
- DFA (Doesn't Function for Audio)

None

**Available Button Parameters:**

- Stereo
- 48V
- HPF/LPF On
- Compressor On
- Gate On
- EQ On
- Send On/Pre/ AFL
- Send Enables
- Delay On
- Pin Channel
- Phase Invert On
- Solo B
- Talk on
- Safes
- DFA
- None

Note: The DFA controls contain many advance audio tools, use carefully to demonstrate to your artist the Midas sound. Use the left/right arrows to navigate. Use with caution!

A description of the Global Assignable Shortcuts page:

1. Assignables – either Automation, Groups or Other.
2. Tabbed pages available within the Assignables Type.
3. Shortcuts available to add to the surface. Can be selected with a long press then dragged to the page you have selected.
4. Cyclic or List. This determines if the pages will cycle back to the first page after the last page is reached or whether the pages appear in a list only. I.E. once you reach the last page you can only go backwards.
5. The assignable shortcuts page currently selected to edit.
6. Delete the currently selected page.
7. Add a new page to the end of the list (Maximum 20).
8. Currently allocated assignable shortcuts (12 per page).
9. Store Changes & Save.
10. Cancel Changes.

# 18.6 Assignables Types

## Assignables Types: Automation

**Macros**

**Store Scene** – Overwrite current settings to the current scene.

**Store to New** – Store current settings to a new scene.

**Current Scene** – Displays the current scene name.

**Previous Scene** – Recalls the previous scene in the current playlist.

**Next Scene** – Recalls the next scene in the current playlist.

**Events**

Any Events created in scenes will be displayed here.

**Note:** Events can be named in the Automation workflow to make navigation of scene events easier in this page.

## Assignables Types: Groups

Page 1 POP Groups.

- **All** – Scrolls through all channels on the surface and GUI using all banks defined as Area A.

- **INs** – Scrolls through all inputs on the surface and GUI using all banks defined as Area A.
- **OUTs** - Scrolls through all outputs on the surface and GUI using all banks defined as Area A.
- **VCAs** - Scrolls through all VCAs on the surface and GUI using all banks defined as Area A.
- **TAGS** – Spills to the surface all tagged channels based on selection from the Tags & Pops widget via the Suggested tab in the Home workflow.
- **System Tags** – Shows the selected system tagged channels as selected from the Tags & Pops widget.
- **Favourite Tags** – Shows selected Favourite Tagged channels on the surface.
- **FX** – Shows send/return channels to FX and any channel with an FX inserted.
- **All Contribs** – will show all contributions of a selected aux/bus
- **In Contribs** – will show input contributions of a select aux/bus
- **Out Contribs** – will show contributions of a select aux/bus
- **Clipped Channels** – Brings all currently clipped channels to the surface.
- **POP 1-24** - The next 24 Assignables are for each POP group individually. They can be assigned to any area in the assignable shortcuts area.

Page 2 Mute Groups.

- **Mute groups 1 to 12** – Engage the selected mute group to turn off predefined sets of channels.

Page 3 Talk Groups.

- **Talks to Outputs** - When active the talk back bus is sent to all Outputs.
- **Talks to Inputs** - When active the talk back bus is sent to all Inputs.
- **Talks to Auxes** - When active the talk back bus is sent to all Auxes.
- **Talks to Matrices** - When active the talk back bus is sent to all Matrices.
- **Talks to Group 1 to 12** - When active the talk group turn on the talk bus to predefined sets of channels.

Page 4 Outputs.

- **Aux Output (1-96)** – A dedicated output selection shortcut for every aux in the system.
- **Matrix Output (1-24)** – A dedicated output selection shortcut for every Matrix in the system.
- **Left, Right and Mono** – A dedicated output selection shortcut for every Master bus in the system.

## Assignables Types: Others

Page 1 Presets

- **Previous Preset** – Recalls the Previous Preset in the current preset list.
- **Next Preset** - Recalls the Next preset in the current preset list.
- **Pro Series Mode** – Sets the surface to Pro series mode – Bank 1&2 Area A. Bank 3 VCA or Outputs. Bank 4 Masters.

- **HD24 Mode** – Sets the surface to Pro Series Mode – Bank 1,2 and 3 Area A. Bank 4 Masters.
- **Monitor Mode** – Sets the surface to Monitor Mode – Bank 1&2 Area A. Bank 3 Outputs. Bank 4 Masters.
- **Theatre Mode** – Sets the surface to Theatre Mode – Bank 1 Area A. Bank 2 VCAs. Bank 3 Outputs. Bank 4 Masters.
- **VCA Mix Mode** – Sets the surface to VCA Mix Mode – Bank 1&2 Area A. Bank 3&4 VCAs.
- **Flip Mode** – Activate flip mode for the currently selected channel.
- **Pin Layer** – Locks the currently selected layer to the surface until the layer is unpinned.
- **Custom Desk Layouts** - Puts the surface into a previously set up custom layout.

Page 2 Others

- **Tap Tempo** – Tap tempo control

# Chapter19: Automation

## 19.1 About Automation

The HD96-24 has a new Automation system that is predominantly a GUI-only feature that allows complex editing of scenes and the creation of show files via the GUI. Some controls for storing and recall of scenes can be added to the Global Assignable Shortcuts area.

The automation system of the HD96-24 can store and recall a vast number of scenes, each one being a snapshot of the HD96-24 settings at the instant the scene was created. By recalling scenes, users can, with certain exceptions, restore the HD System to the state that existed at that time the scenes were stored. This makes it ideal for multi-act tours by providing quick and accurate access of settings for the band with a minimum of sound check time, as well as complicated touring bands, where each song requires different effect settings with various channel changes such as mutes or level control of monitor sends.

# 19.2 Master Scene

### Master Scene

The Master scene is a new concept in how to update and control automation data.


By default, the Master Scene is locked. Each new scene stored will recall **all** the information stored in it. Recalling the Master Scene when locked will bring back all default values similar to the 'safe' scene in Pro Series.

With the Master Scene unlocked and used as the starting point of your show, any changes to the Master Scene will be seen as the new default setting when saving new scenes. However, it should be noted that any stored changes in consequent scenes will be propagated to future scenes rather than the default setting.

The Master Scene can be used as a default for all recalled scenes that do not have different parameter/setting information stored in that scene. For example, if the EQ of channel 1 is changed in scene 3 and wish to copy that EQ change to every scene, you can by saving to the Master Scene. If scene 2 has a separate EQ stored to the EQ you have just stored to the Master scene, the stored EQ of that scene will be recalled, not the EQ from the Master Scene.

**Note:** When unlocked The Master Scene recalls anything saved to it.

Master Scene events are created just as you would for any other scene. Events of a similar type in other scenes will override the Master ones once stored.

It is possible to delete all changes in a channel (so it follows the master scene again) or individual change, such as EQ or Aux level change in a scene if you don't require the automation change anymore by using the Show Editor (Chapter 20).

## 19.3 Auto-Save

When on this saves the current show every time a scene is saved or if a playlist is opened, created or edited. This powerful feature makes life easier for the user knowing the show is always saved.


All of the scenes for a show are contained within a show file. Show files are stored in the HD surface, so that they can be loaded when required. They can also be transferred to/from external USB storage devices and via the Midas mCloud.

Events provide an additional scene control by which you can use the MIDI, Target Tags and Notes functions to trigger events on internal and external devices from within the show file.

## 19.4 Create or manage a show

To create or manage a show:


1. In the top bar menu press the blank area to the left of the MENU button (name of current active scene is usually displayed here).
2. In the pop-up window press Show Manager. This opens the Show Manager page (you can also select Show Manager in the side bar menu).
3. Press the + New symbol to create a new show. It will be given the default name of Untitled Show, which can be edited later.
4. Export the show to USB if required.
5. Press Open to open the selected show.
6. Tap on the name area to edit the text. Tags can also be added to the show file here for easy tracking of files. Notes can also be added in this area if desired.
7. To archive any show, press and hold the Archive button.

**Note:** Opening a show does not load the master scene or any other scene into current active memory. This needs to be completed after a show has been opened.

**Tip:** Remember if your console is online and you are signed into your mCloud account all your show files are synced and backed up to the mCloud for extra safety.

## 19.5 Automation controls

Although automation is supported on the surface in the shortcuts area, it requires large amounts of screen support. The GUI provides this in the form of an Automation workflow that gives full scene and show file creation via the Show Manager. Additionally, the events and notes pages can be created, edited and deleted from this workflow.

# 19.6 Scene Creation and Recall

Once a show has been created the next step is to create scenes, this can be achieved in two ways. By creating in the Scene window as described below or by creating in the Show Manager page as described in the next Playlists section. Many of these controls can be programmed in the Global Assignable Shortcuts area for quick use during a show.


1. In the top bar menu press the Scene button, this opens the scene recall widget.
2. Press Store Options to reveal the various scene management options.
3. Pressing Master Scene stores all changes in the current scene apply to the master scene (even if you are in the Master Scene).
4. Press New Scene to make a complete copy of the currently loaded scene. The new or duplicated scene will appear at the end of the current scene playlist.
5. Store to Current Scene updates and stores the scene you are currently in.
6. The Go button is used to recall any scene you select by navigating up and down the playlist by swiping in the play list area.
7. Previous recalls the scene directly before the current active scene.
8. Next recalls the next scene (after the current scene) in the playlist.
9. Show Manager opens the show manager and playlist manager page.
10. Cancel and return to Store Options if required.

# 19.7 Playlists

A playlist is a way of organising scenes into various different orders, for example if a band changes their set list you can create a new playlist for that setlist. This give you the ability to go back to a previous playlist if required. You can create as many playlists as you like.

To create a play list:


1. After creating and loading a show press New in the Playlists area.
2. The currently selected playlist is highlighted in green.
3. Press Open to load the currently selected Playlist
4. Press Rename in order to edit the name.
5. Press Delete to remove the playlist.
6. This area displays a list of scenes in the current open playlist. Playlist order can be changed by holding onto the scene you wish to move until it wiggles, then move it up and down in the list.
7. This area displays all scenes available within the currently loaded show.
8. This area shows the number of events stored in that scene.
9. This arrow moves the scene into the currently open playlist at the end.
10. This X removes the scene from the playlist (it does not delete the scene)
11. Remove All clears all scenes from the current playlist.
12. To delete a scene, select the scene you wish to delete in the All Scenes window. Then press and hole the Delete button to erase the scene permanently.

## 19.8 Safes

**Important:** Safes are intended for emergency use only and are not to be confused with scope (see Chapter 20 Show Editor (Scope). Safes are incorporated into the HD96-24 system to prevent certain controls from being recalled with a scene. Safe activation and status are provided via the GUI only.

There are 7 types of channel safe: Gate, Comp, EQ, Filters, Config, Mute and Main Bus. Although some types of safes are channel-specific, any channel can be made safe from off-channel mute, fader and automation control. Also, solo (for monitor areas A and B) is always out of scene on any channel.

## 19.9 Scene Equaliser Preview

The Equaliser scene preview function allows you to see changes in different scenes without having to recall a scene. This powerful feature lets you look ahead at possible changes, edit and store any changes to the scene in preview.


1. Select the scene Preview function.
2. Select the scene from the list, the graph will display the EQ of the selected scene which can be changed but will not be audible.
3. Press cancel if you do not wish to apply the changes.
4. Press Store to save any changes to the EQ settings from the scene in preview.

When the scene True Audition button is also engaged with the preview button, a list of scenes will appear next to the equaliser window called Preview Audition. The EQ window will turn yellow. Selecting different scenes will show you any changes to the EQ in the selected scene and be audible in the solo bus only. This means you can listen to and edit changes for future scenes without affecting the main mix or monitor mixes.

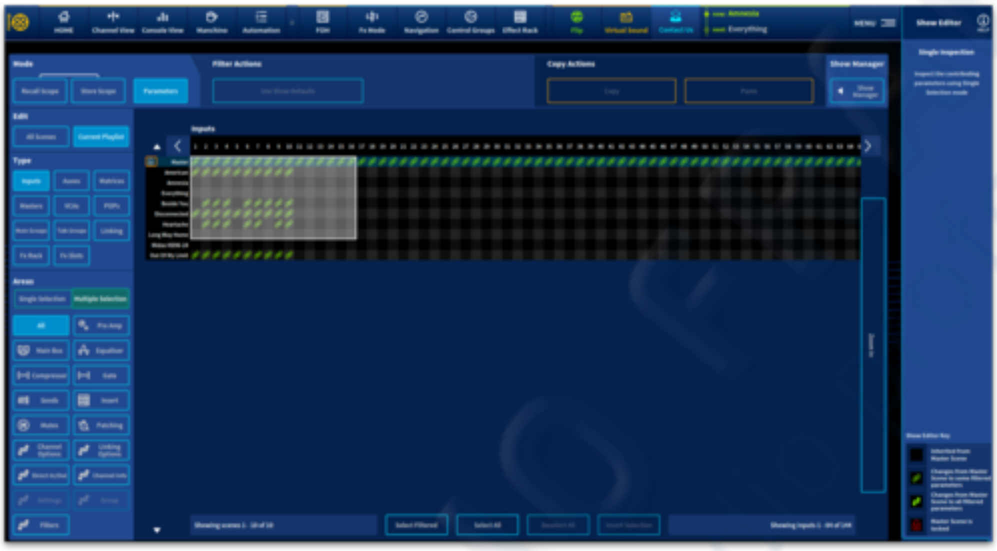

1. Press to engage True Audition mode.
2. Press to also engage scene Preview mode.
3. Select the scene you wish to audition and edit (only in the solo bus), the graph will display the EQ for the selected scene.
4. Press cancel to clear any changes.
5. Press Store to commit changes to the scene in preview.

**Note:** Pressing Store does not affect the EQ in the currently active scene.

# Chapter 20: Show Editor (Recall and Store Scope)

## 20.1 About scope

The show editor also allows you to see which parameters differ from those saved in the Master Scene with green arrows. If desired, parameters changes can be removed. This allows the master scene to regain control of the parameters in scene recall.



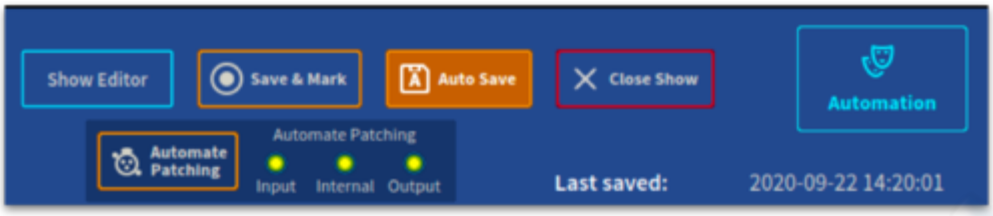
**About scope**

Scope has two functions, recall and store, the emphasis in this chapter is on recall scope, which will be the most commonly used. Store scope will only be required in certain circumstances, and even then, it must only be used with caution (see "Store Scope Mode").

**Note:** The show editor has a help section with detailed instructions built into the console.

# 20.2 About the Show Editor

The Show Editor is found in the Show Manager page. Press to open the overview page.

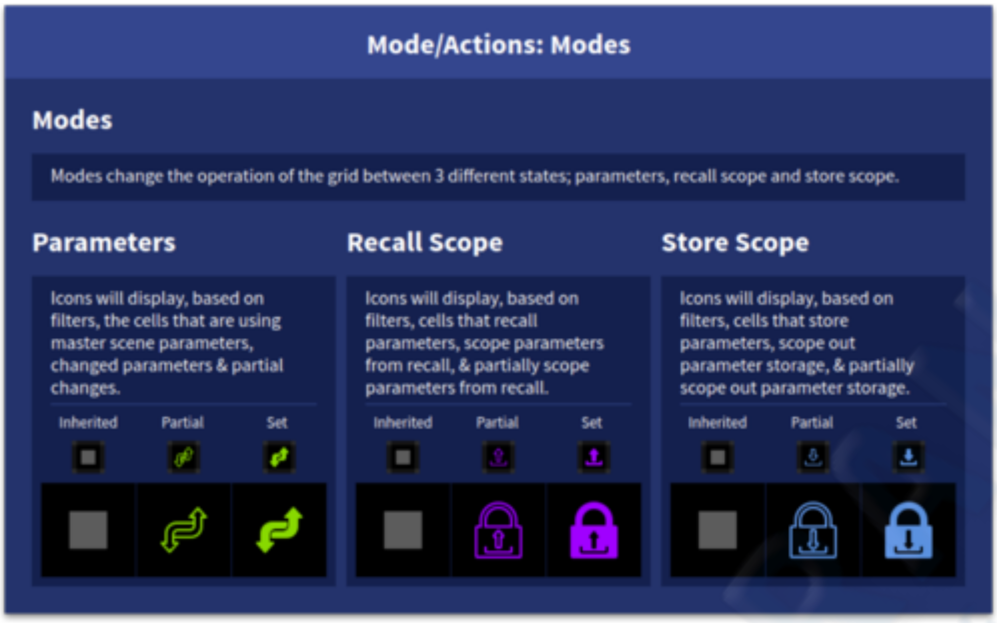



Scope lets you define the extent of the automated controls for all Inputs, Auxes, Matrices, Masters, VCAs, POPs and Misc (Physical Devices, Linking and FX Rack). The Show editor page allows you to select the controls that are excluded from the automation when a scene is recalled, exclude controls from being stored in a scene or return parameters back to use show defaults and follow the master scene as required.

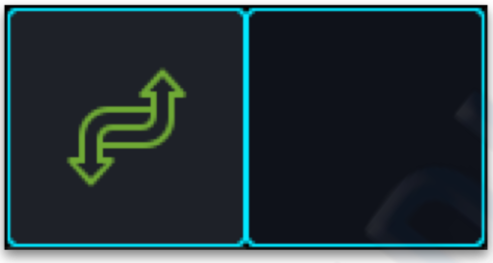
The Show Editor page has a number of type-specific areas which include Inputs, Auxes, Matrices, Masters, VCAs, POPs, Mute Groups, Talk Groups, Linking, FX Rack and FX Slots which contain user-selectable parameter sections that you can make 'out of scene' on scene recall.

A grid system is utilised in order to show which channels are either in Recall Scoped, Store Scoped or have different parameters to those stored in the master scene. These are indicated by different colour icons:

- Recall Scope – Purple Padlock (Solid indicates **all** currently selected area parameters are active for that cell, outlined padlock indicates **some** of the selected area parameters are active in Recall scope).
- Store Scope Blue Padlock (Solid indicates **all** currently selected area parameters are active for that cell, outlined padlock indicates **some** of the selected area parameters are active in Store scope).
- Parameters – Green Arrows (Solid indicates **all** currently selected area parameters are active for that cell, outlined arrows indicates **some** of the selected area parameters are active).




Press the Channel/Scene cell you wish to select. When a cell is selected it is highlighted with thin blue line around the edges. Pressing the channel number or scene number selects all cells related to the chosen channel (vertical) or scene (horizontal).



# 20.3 Parameter Mode Overview

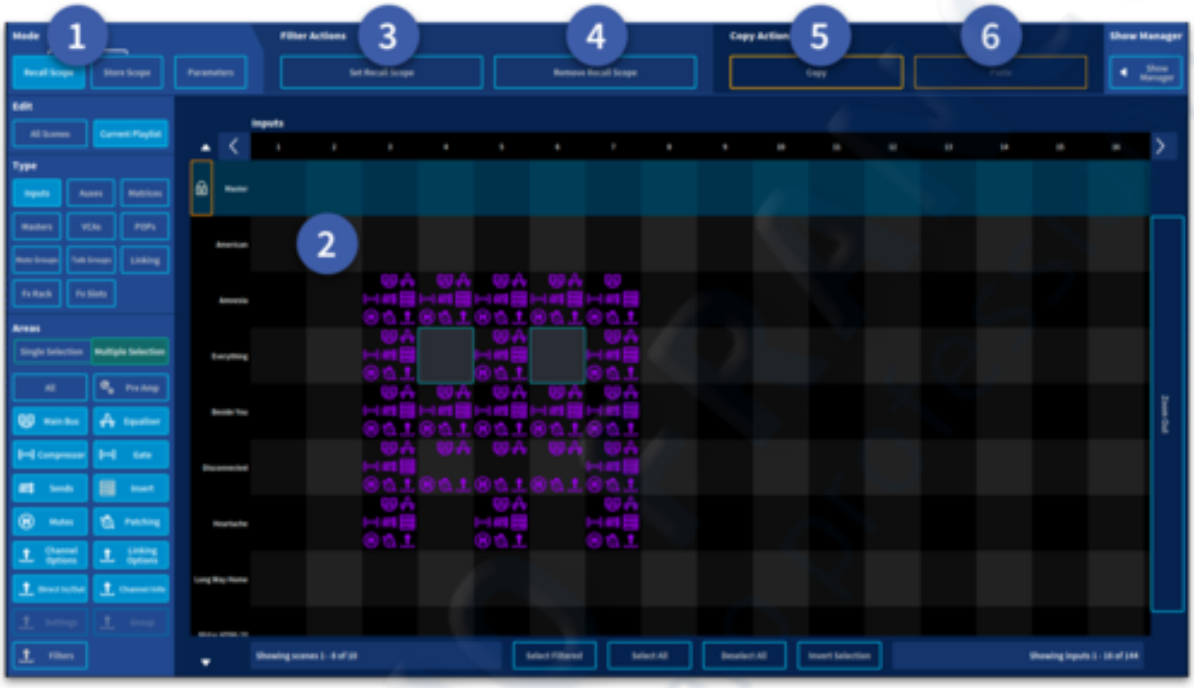
This view shows parameters and indicates if there is a difference to the Master Scene.

1. **All Scenes** – Any changes made in All Scenes mode will affect all playlists within that show.
2. **Current Playlist** – Change made will only affect currently loaded playlist.
3. **Type** – All the parameters here can be individually added to the recall or store scope. Selecting a type will change which parameters can be scoped in the "Areas" section. Types available are: Inputs, Auxes, Matrices, Masters, VCAs, POPs and Misc.
4. **Areas** – Depending on the Type selected, various parameters can be selected to either see if a parameter has been stored and is different from the master scene (green arrows or area icon). Areas available are All, Pre-Amp, Main Bus, Equaliser, Compressor, Gate, Sends, Insert On, Mutes, Patching, Channel Options, Linking Options, Direct In/Out, Channel Info, Physical Devices, Group, Linking, Settings, FX Rack and Filters. Single Selection mode allows one parameter to be looked at in isolation. Multiple Selection mode allows combinations of different types to be used to make multiple adjustments at once.
5. **Recall Scope** – In this mode channels/scenes in Recall Scope will be shown by a purple padlock.
6. **Store Scope** – In this mode channels/scenes in Store Scope will be displayed by a blue padlock.
7. **Parameters** – In this mode channels/scenes that differ from the master scene will be displayed by green arrows.
8. **Use Show Defaults (Filter Actions Parameters)** – When in Parameters mode using this function will reset any selected cells and use settings from the Master Scene once again.
9. **Select Filtered** – Any cells in use (with corresponding icon) will be highlighted.
10. **Select All** – Selects all cells.
11. **Deselect All** – Clears all current selections.

12. **Invert Selection** – Deselects all currently selected cells and selects all unselected cells.
13. **Channel Number** – Displays all the channels available. Use the arrows to navigate to any number off screen. In Zoom pressing the channel number selects all scenes for the selected channel.
14. **Scenes List** – Displays all current scenes (depending on Show/Scene selection) in order. In Zoom pressing a scene name selects all channels horizontally across the editor page.
15. **The Grid** – Shows all affected cells depending on mode selection.
16. **Zoom Area** – The highlighted area shows where the zoom will focus on.
17. **Zoom In/Out** – Turns the cell zoom on and off as required.
18. **Show Manager** – Returns back to the Show Manager.
19. **Help** – Press for information about the Show Editor.
20. **Single Inspection** – When in Single Selection mode this column displays the parameters in the selected area.
21. **Show Editor Key** - An explanatory list of symbols used in the grid.
22. **Master Scene Lock Icon** – This icon indicates if the Master Scene Lock is engaged.

# 20.4 Recall Scope Mode

Recall scope can stop the loading of information from a scene by setting individual items or cells within a channel to not recall. The blue padlock indicates that a channel or scene has part or all of channel in the recall scope.




All the Edit, Type and Areas function in the same as the Parameters mode.

1. **Recall Scope** – Press to enter Recall Scope Mode.
2. **The Grid** – Channels with cells in Recall Scope are highlighted with a purple padlock. Icons are used to show which Areas are in Recall Safe.
3. **Set Recall Scope** – This activates Recall Scope on any selected cells.
4. **Remove Recall Scope** – This clears any selected active Recall Scopes.
5. **Copy** – Copies any cell selections to the clip board.
6. **Paste** – Pastes any copied selections to the chosen destination.

**Note:** Copy Actions works with all 3 modes, Recall Scope, Store Scope and Parameters.

# 20.5 Store Scope Mode

**Although store scope is sometimes useful in very specific situations, it must always be used with care. This is because it is possible that control settings will not be stored at all and will consequently be lost. Therefore, it is much safer to use recall scope and always store everything.**

Please use store scope with great care and observe the caution above. All of the methods of the recall scope operation, as detailed in this chapter, apply equally to store scope. It is advised that the master scene is kept out of the Store Scope so that settings are stored.

Sometimes it's worth using this feature to specifically stop something from being overwritten. This could actually be the master scene in some cases if you hand over a show to an operator and don't want changes to be made. For a theatre show, store scope all outputs to protect all the system EQ and time aligning. Leaving the user to change the inputs accordingly day to day.


1. **Store Scope** – Press to enter Store Scope Mode.
2. **The Grid** – Channels with cells in Store Scope are highlighted with a blue padlock. Icons indicate which areas will **NOT** be stored.
3. **Set Store Scope** – This activates Store Scope on any selected cells.
4. **Remove Store Scope** – This clears any selected active Store Scopes.

# Chapter 21: Events (Automation)

## 21.1 About Events

There are 3 types of event: Notes, Events and Global Events. Multiple events can be added to a scene. You can have any combination of each kind of event. You can choose whether MIDI events are triggered via the HD96-24 surface or via connected I/O.

- **Notes:** Each scene has a Notes area. This is the place to write information about that scene.
- **Events:** There are 4 types of scene-based events that happen on a specific scene at a specific time if required. MIDI, TAG, NOTE or a Scene Control Event.
- **Global:** There are 4 types of global-based events that happen every time any scene is fired or can be delayed if required. MIDI, TAG, NOTE or a Scene Control Event.

**ALL:** Displays all events types, global and scene based.


The 4 event types, whether they are global or scene-based, control various different parameters as listed below:

1. **MIDI Event** - All different types of MIDI information can be sent or received.
2. **Tag Event** - Tagged channels can be controlled by Scene Recall or MIDI events.
3. **Note Event** - Notes can be displayed on scene recall or after a set time (Event Delay).
4. **Scene Control Event** - When a scene is recalled an action can be triggered, such as skipping to a later scene after certain amount of time.

**Tip:** There is no limit to the number of events that can be added to a scene.

## 21.2 About the Edit Event Window


1. Green box indicates current scene.
2. The padlock indicates the Master Scene is locked.
3. Blue box indicates scene to be edited.
4. Add new automation event in selected scene.
5. Trigger the currently selected scene.
6. Copy the selected scene.
7. Edit the selected event. Edit also allows you to edit event names.
8. Delete the selected event.
9. Turn the event on and off as required.
10. Time stamp for selected event.
11. When an event is triggered the circle symbol will fill to show the event is active.
12. Event name (can be changed via the edit button).
13. Type of event, either MIDI, Tag or Note.
14. Expand select event to display the contents.
15. Trigger the currently selected scene action. Use this to test functionality.

# 21.3 MIDI

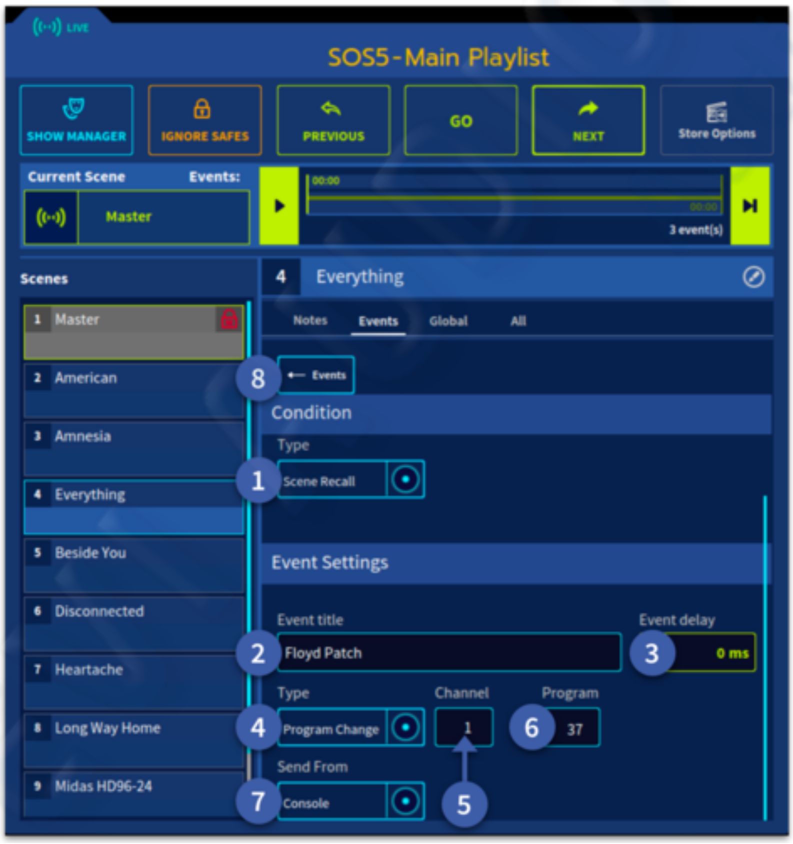
MIDI performs two functions on the HD96-24. It allows the HD96-24 to trigger external MIDI-equipped equipment on each scene change and it also allows external MIDI equipment to trigger a HD96-24 scene change.

MIDI output from the HD96-24 can include a globally enabled outgoing message that contains the recalled scene number and is sent out for all recalled scenes.

Below is a list of possible MIDI action types in addition to Scene Recall:

- Note On
- Note Off
- Poly Pressure
- Control Change
- Program Change
- Channel Pressure
- Pitch Bend

MIDI input can be globally set up to cause scenes to be recalled when specific incoming MIDI messages are encountered.




1. Event title edit:
2. Event delay allows the triggering of an event to be delayed in a predetermined time in ms.
3. MIDI type selection.
4. MIDI channel selection (1-16)
5. Different MIDI options will be displayed here depending on the MIDI type selected.
6. Slider to adjust data depending on MIDI type selected.
7. Choose which MIDI out port is used depending on connected I/O.
8. Back to the main events page.

# Chapter 22: TBA

## New Document

A line of text in a paragraph.

## New Document

## New Document

## New Document

## New Document

# Chapter 23: TBA

## New Document

A line of text in a paragraph.

## New Document

## New Document

## New Document

# Chapter 24: File Management

## 24.1 Show Manager

1. Show displays all current shows in the console and on your mCloud account.


2. Presets is a list of various types of presets found within the console and synced to your mCloud account.
3. The icon indicates the type of file. In this example a show file is displayed.
4. Selected file is highlighted and displays the last modified time and date.
5. The cloud icon tells you the mCloud sync status.
6. Inspect lets you see more detail about the selected file.
7. Selected file will be copied to USB when pressed.
8. Rename the selected file.
9. Duplicate the selected file.
10. Delete the selected file.
11. USB device name.
12. Eject USB device.
13. Back up one layer within the USB device.
14. Selected file show for USB device.
15. Copy selected file to console.
16. Create a new folder.
17. Inspect a USB file to see more detail about the file.
18. Delete the selected file from the USB device.

## 24.2 Preset Manager


1. Select Presets to view the preset list within the console and your mCloud account.
2. Different preset types are displayed here and can be selected.
3. A list of presets of the selected type will be displayed with mCloud sync status of each file.

# 24.3 Spreadsheet Import

The File Manager page also allows the import of data from a spreadsheet. The mCloud website [www.midasconsoles.com](http://www.midasconsoles.com) has a downloadable spreadsheet template which allows the user to name, colour, set VCAs, Set POPs, Select Aux types and many other features. This spreadsheet can then be loaded directly into the console as a starting point for building your new show.


**Warning** – Start by loading a new blank show. When loading a spreadsheet, the whole system is cleared, any current settings will be lost, and the new settings loaded directly onto the surface. At this point the new settings are not saved and will need to be saved to a master scene, then also saved as a show.

1. Select USB and the folder with the spreadsheet you wish to use.
2. Select the spreadsheet from the list.
3. Press Copy To Console to import all the settings of the spreadsheet, remember to save to the Master Scene and then as a Show in order to recall settings at a later date.

# Chapter 25: Multiple Consoles and Other Devices

## 25.1 Sharing A and B mic Preamp Inputs

The DL231 has two preamps per channel, A and B. To use either the A or B side connect directly to the preamp (A or B) you wish to use.

**Note:** Only A or B has control of 48v which is set in the I/O box.

DL231 outputs can be patched from any connected console but the DL231 decides which console or direct split of the inputs is in use. The DL231s outputs can be set to either:

1. AES50 A
2. AES50 B
3. Mic Amp A
4. Mic Amp B

## 25.2 Ultranet

The HD96-24 system has two ultranet ports which allow up to 16 outputs per connection. To connect to an Ultranet enabled device such as a Behringer P16-M connect the output from ultranet port 1 to the input of the P16-M.


To patch a P16-M:

1. Select the Busses or Direct Out tab from the left side of the patching workflow
2. Select Ultranet from the right-side patching tab.
3. After unlocking the patching page select the Output(s) or Direct Out(s) you wish to patch
4. Select the Ultranet port you wish to use (Port 1 or 2).
5. Now select the Ultranet outputs you wish to use (1-16).


To use other Ultranet enabled devices navigate to Expansion Cards page in the side bar menu. Here you will find the ultranet tab which the selection of various Ultranet enabled products.

1. Select either Port 1 or 2.
2. Navigate the 4 pages, each has the settings for 4 Ultranet outputs.
3. This button navigates you directly to the patching page if required.
4. Displays the current patch path.
5. Selects the Ultranet device type from the drop down list.
6. Select the profile you require.
7. Select an EQ preset if required.

## 25.3 Using External USB Equipment

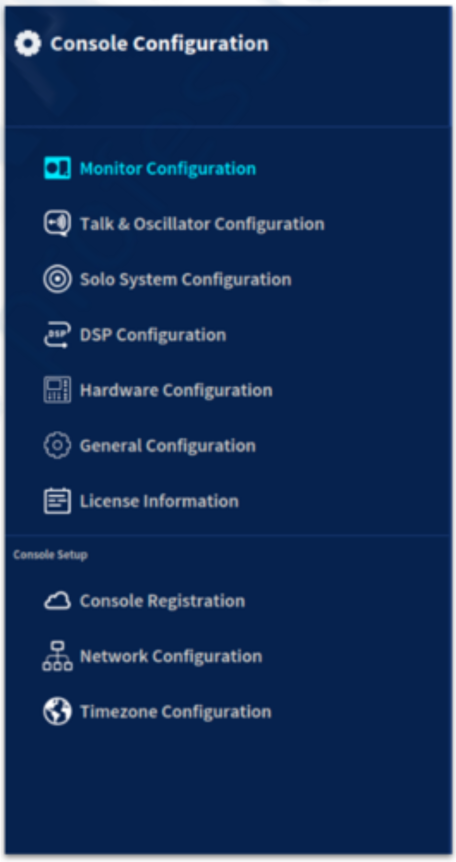
A standard USB keyboard can be used in either the Fader Port USB hub (Not the USB Hub).

**Warning:** Do not use a USB mouse. Although a mouse can be plugged in and used in a USB port there will be no mouse pointer and it is possible to click buttons by accident.

## 25.4 Using External Displays

Currently HDMI 1 is not in use but will be used in the future to display scene and metering information. HDMI 2 mirrors the GUI and can displays all the same information as the main touch screen. In the future functionality of these ports will be greatly increased. For example, HDMI 1 will be used to display metering and scene information in an impending update.

# Chapter 26: Préférences (Console Configuration)




## 26.1 Monitor Configuration

In the Monitor Configuration page various functions can be set globally. The controls are identical for the Monitor A and B bus. See Chapter 13 for more information on the Monitor Configuration and its use.


1. Monitor Level Meter.
2. Adjust the level of the Monitor bus.
3. Display of the Monitor bus delay in millisecond.
4. Display of the Monitor bus delay in meters.
5. Monitor bus delay control, up to 500ms delay is available.
6. Monitor bus delay on/off button
7. Dim the Solo level by -20 db.
8. Turns Input Pre Fade Listen (PFL) on.
9. Turns Output Pre Fade Listen (PFL) on.
10. Enter Solo Add mode on the chosen monitor bus. This allows multiple channels to be soloed at the same time.
11. Clears all currently selected solos.
12. Talkback to monitor bus.
13. Listen to the Stereo Bus when no other path is soloed.
14. Listen to an External Source (found in the Monitor In patching page).
15. Listen to the Mono Bus.
16. Sum the Monitor bus to Mono.
17. Invert the polarity of the right side of the Monitor bus.
18. Send the Left signal to both sides of the Monitor Bus. Press 18 and 19 together to swap.
19. Send the Right signal to both sides of the Monitor Bus. Press 18 and 19 together to swap.
20. Talkback Ducker threshold adjust the level at which the monitor bus is ducked to allow the Talkback Bus to be audible.
21. Talkback Ducker Gain Reduction (GR) meter.
22. Range adjusts the amount of attention in dBs applied to the solo bus when the talkback bus is active.
23. Activates the Talkback Ducker.
24. Limiter threshold adjusts the point at which the limiter is active on the selected solo bus. Can be used to protect hearing when changing between quiet and loud in ear mixes.
25. Limiter Gain reduction meter.
26. Limiter on/off.

**Tip** – Delay time is worked out at 1m needing a delay of 2.92 ms or just under 1 ms per foot (E.G. In metric measurements FOH @ 50m from stage = 146.2 ms or FOH 150 ft from stage = 133.9 ms)

## 26.2 Talk & Oscillator Configuration

1. Engage the Talk bus to internal.
2. Engage the Talk bus to external.
3. Talk bus Limiter to protect against loud signals.
4. Talk bus level control.
5. Talk Mic Gain control.
6. Talkback level control.
7. Send the Oscillator to the internal bus.
8. Send the Oscillator to the external bus.
9. Level control for the Oscillator.
10. Frequency control of the Oscillator (50 Hz to 5 kHz).
11. Oscillator fixed to 1 kHz.
12. Turn the Oscillator into a Pink Noise generator.
13. Various routing options to send the Talk and Oscillator to.


## 26.3 Solo Bus Configuration

The Solo System page allows the use of Solo in Place. This is also where you can set the PFL and AFL Levels and how you interact with the SEL and SOLO buttons.


1. Monitor Output Swap switches the Monitor A and B bus if required.
2. Long hold to Activate Solo In Place mode.
3. PFL Level control.
4. AFL Level control.
5. Input Select Follows Solo. When you press Solo on an input channel it also selects it.
6. Output Select Follows Solo. When you press Solo on an Output bus it also selects it.
7. Input Solo Follows Select. When you press SEL on an input channel it also Solos it.
8. Output Solo Follows Select. When you press SEL on an Output bus it also Solos it.

# 26.4 DSP Configuration


1. Clocking selection for Primary source.
2. Clocking selection for Secondary source.
3. Snake 1 Connection choice. Either Optical (Fibre up to 500 m) or Copper (Cat 5 up to 100m).
4. Snake 2 Connection choice. Either Optical (Fibre up to 500 m) or Copper (Cat 5 up to 100m).
5. Remote Snake 1 Connection choice. Either Optical (Fibre up to 500 m) or Copper (Cat 5 up to 100m).This selects the type of connection used for the DN9680.
6. Remote Snake 2 Connection choice. Either Optical (Fibre up to 500 m) or Copper (Cat 5 up to 100m). This selects the type of connection used for the DN9680.
7. Choose whether the console is the clock Master or Slave to external clock source.
8. Choose whether the clock is Manually or Automatically selected.
9. Choose whether the console uses the Primary or Secondary clock source if set to slave.
10. Console Clock Status. Displays the state of the console with regard to clock source. Also visible in the top bar menu, press to bring you directly to this page.
11. Set metering to Pre Fader.
12. Enables metering delay (up to 500ms).
13. Set meter Attack time.
14. Set meter Decay time. (0 to 2 s).
15. Metering Delay control (up to 500ms).
16. Delay Compensation Options (See chapter 27 for more detail on Delay Compensation).
17. HyperMAC redundancy sets both HyperMAC ports to mirror each other for true redundancy. A KT AS80 will be required to use this function.
18. SuperMAC redundancy sets all 4 local AES50 ports to mirror each other for true redundancy, 1&2 and 3&4.This only works for DL15x, DL231 and DL25x I/O devices.
19. Analyser enable turns the spectrum analyser on or off in the channel EQ displays.
20. FFT size adjusts the details of resolution of the frequency content.
21. Window Type. Choose from Hanning, Hamming, Blackman or Rectangular.
22. Overlap size. Adjust the % of refresh rate.

## Clocking

In order for the HD96-24 to function correctly, Clocking must be synchronised and stable. The HD96-24 can utilise it's own internal clock or synchronise from an external source.

Setting the Clock source to Master utilises the internal clock. (Primary & Secondary settings do not apply).

Setting the Clock source to Slave will utilise either the Primary or Secondary selected sources. Options for these sources are available in the lists. You can choose clock source from Word Clock, AES3, Local AES50 or Snake connections.

Depending on the Manual/Automatic Setting, Manual will force the selection to the chosen Primary or Secondary setting. If the clock is dropped with the chosen source, it will remain searching for the clock at this source unless manually changed.

When switched to automatic, the console will attempt to sync to the primary clock. If no clock is found after a short while (approximately 4-10 seconds) it will switch to the secondary option. If this happens to fail as well it will revert to primary and process will start again and continue until a stable clock is found.

**Tip**: One of the options in the Secondary list is 'internal'. This setting is useful as a backup to a primary source selection When set to Automatic, should the primary source fail, the clock will revert to the secondary - Internal (which is equivalent to the Master Clock).

As default the console is set to slave, if it receives no clock it defaults to the internal clock. When the console is set to be slave it can slave from FOH AES3 ports, HyperMAC ports, SuperMAC ports, Word Clock, AES3 with a secondary Slave option available should the 1st fail. Clocking can come via 3 sources:

1. **Console**
2. **I/O Stage Box**
3. **Other device (via AES3 or Word Clock Input).**

### Metering Preferences

The Metering Preferences section of the Preferences screen provides global parameter adjustment of all of the

meters on the control surface.

Meter Delay control adjusts the meter delay time, in the range 0 to 2 seconds. For example, if the control surface is at the FOH position, this function allows you to synchronise the meters with the audio being heard. This is because the sound from the artist on the stage will take a certain amount of time to reach you, whereas the meters pick up that sound at source. Click the Delay Enable box to select this function. The delay time is displayed in milliseconds (ms).

Meter Decay control adjusts the time it takes the meters to fall, in the range 10 to 25 milliseconds.

Pre Fader option can be used to switch the output channel meters to pre-fader.

## Redundancy

HyperMAC redundancy sets both HyperMAC ports to mirror each other for true redundancy. A KT AS80 will be required to use this function.

SuperMAC redundancy sets all 4 local AES50 ports to mirror each other for true redundancy, 1&2 and 3&4. This only works for DL15x, DL231 and DL25x I/O devices.

### Spectrum Analyser

FFT size - Controls the resolution of the analyser, higher value gives more precise frequency detection but slower response rate Overlap - Refresh rate of analysis. Higher values give better response rate.

Windows - Different windows types have different properties. Each window represents a trade-off between frequency resolution, amplitude roll off and noise floor. Hamming and Hanning windows are suitable for most applications as they have good frequency resolution and reduce noise leakage, Hanning provides better elimination of noise. Blackman-Harris is similar to Hamming and Hanning but has a less steep noise reduction if greater frequency content is required. Rectangular provides no noise reduction which is suitable for broadband frequency content.

# 26.5 Hardware Configuration

Brightness page In the Hardware configuration pages, you will find controls to adjust many aspects of the system's appearance. In the Brightness page there are controls to adjust brightness of individual elements or globally between 0 and 100% (0.5% – 100% for the Screen).

- **Global** – Adjusts the whole system's luminosity at once.
- **Screen** – Adjust the brightness of the GUI Screen.
- **Backlight** – Adjust the backlight level of the GUI Screen.
- **Surface** – Adjusts the Surface controls light levels as a whole.
- **Knobs** - Adjusts the brightness of the rotary controls.
- **Buttons** – Adjusts the brightness of the buttons.

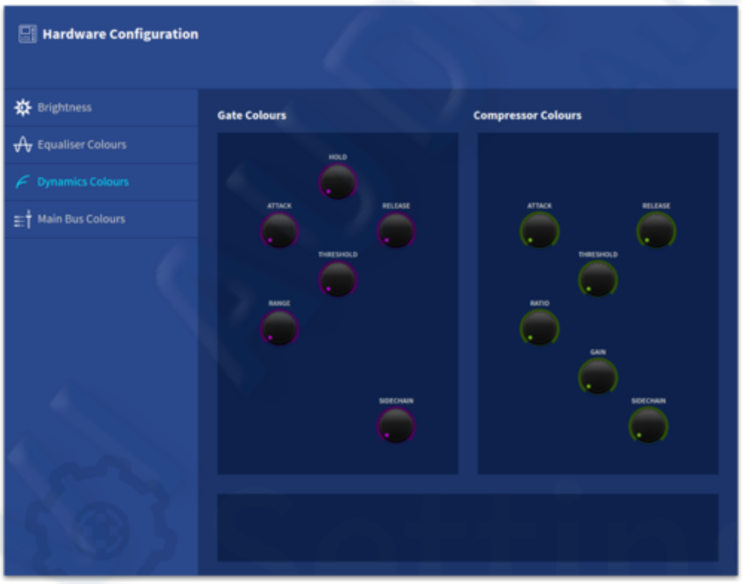
- **LCDs** – Adjust the brightness of the LCD screens.
- **Meters**- Adjust the brightness of the meters.
- **Logo** – Adjusts the luminosity of the Midas logo on the rear of surface from 0-100%.
- **Lamp** – Adjusts the luminosity of the lamps.


## Equaliser Colours

The colours for each EQ band and filters on the surface controls and also via the GUI Equaliser display can be customised by selecting the control and picking a colour from the pop-up menu. This choice is global across the console.


## Dynamics Colours

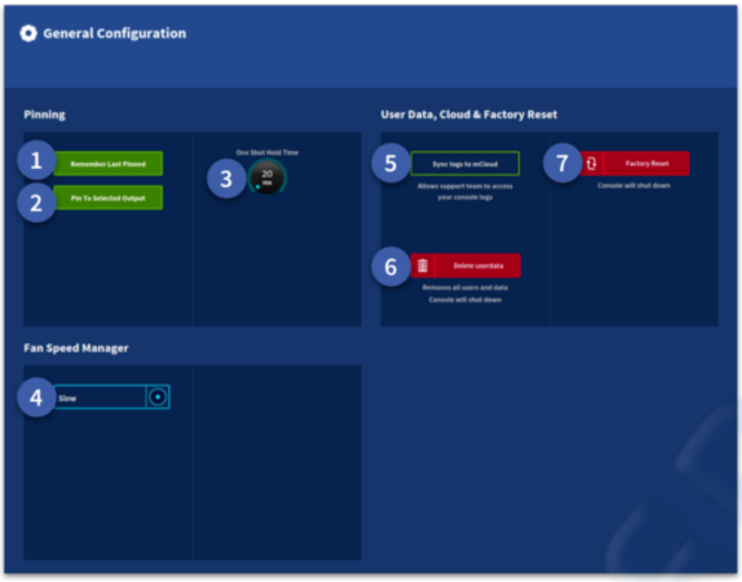
The colours for the Gate and Compressor section can be customised by selecting the control and picking a colour from the pop-up menu. This choice is global across the console controls and GUI.




## Main Bus Colours

The colours for the Pan, Delay and Mono bus can be customised by selecting the control and picking a colour from the pop-up menu. This choice is global across the console controls and GUI.

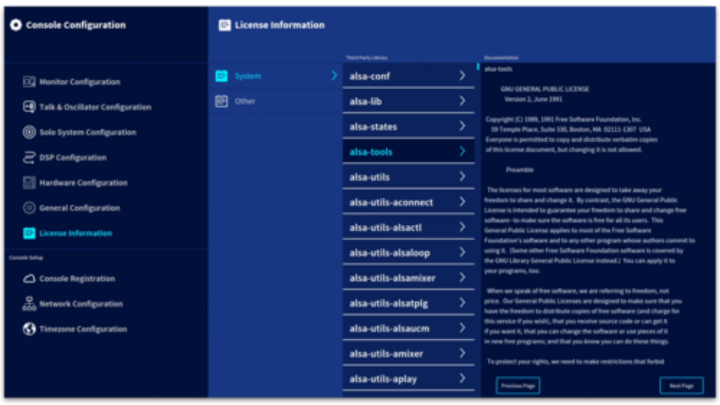
# 26.6 General Configuration




1. In the Pinning section there are options to decide whether the console will remember the last select Pin when reactivating the pinning system.
2. Pin To Selected Output Pins the channel in Flip Mode. This function allows the selected input channel to be pinned to the surface in the current flipped Aux send state. For example, use this mode to pin your main vocal level in a monitor wedge to a fader so it is always available. If this option is off, any previous flip pinned channel will revert to an un-flipped state.
3. One Shot Hold Time relates to the amount of time the One Shot Pot remains available to use after which the function will disappear.
4. Fan Speed Manager selects the speed of the internal fans. If the console is overheating it will automatically increase the fan speed if required. It is not possible to turn the fans completely off.
5. Sync logs to mCloud allows the support team to check if you have a problem by looking at the console logs.
6. Delete User Data clears all the current user data in the console. This includes all show files and shout mixer presets associated with the user. This action only deletes the User and its data from the console and does not delete your mCLoud user or mCloud show files.
7. Factory Reset also removes user data. It resets the Wi-Fi regulatory domain and the onboarding process.

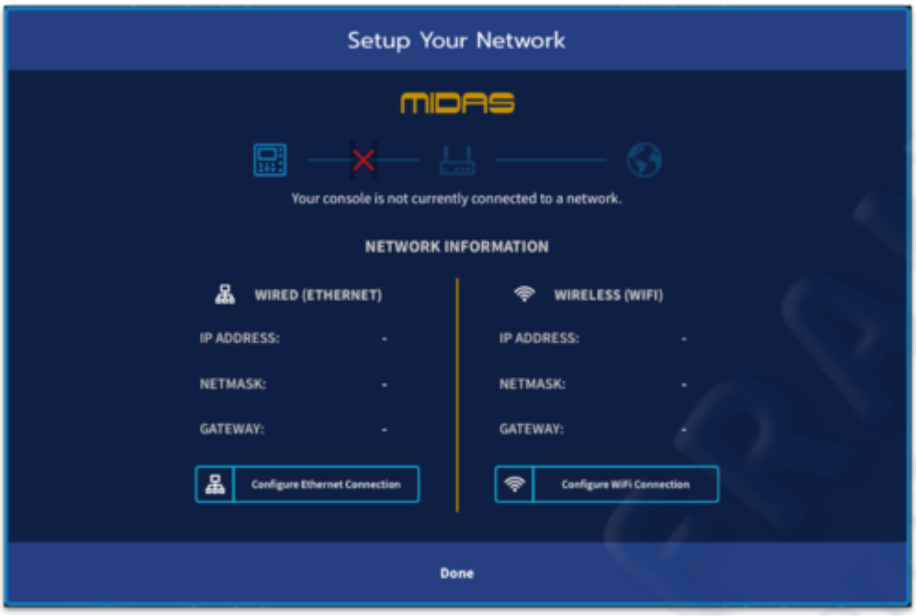
# 26.7License Information

In this section information on various 3rd party apps can be found.



# 26.8 Network Configuration

The Network configuration page presents you with the option to connect over Wi-Fi or Ethernet.

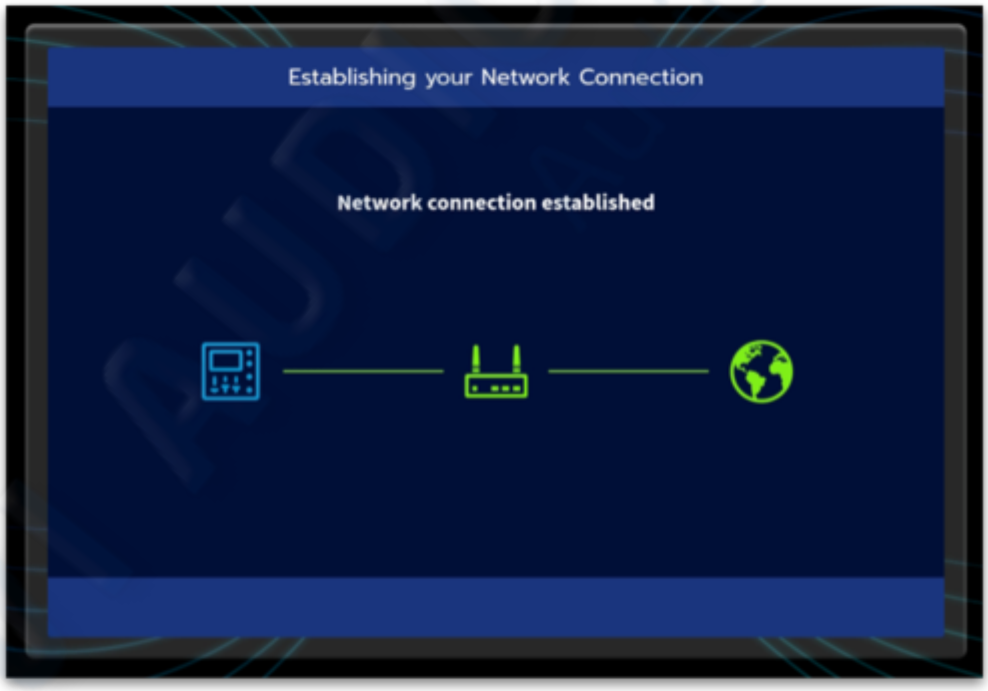




Select a network and enter the network password. Press the Connect To Wi-Fi to establish a connection.

**Note:** You can set up a personal hotspot on a mobile/cell phone to connect to the mCloud if required.


The system will tell you once a network connection has been established.




Press Configure Wi-Fi if you wish to forget the current connection.


An Ethernet connection can be made if you have a wired connection available . This can be set up manually or automatically with DHCP. Press Apply and Connect to make the connection.

# 26.9 Time Zone Configuration

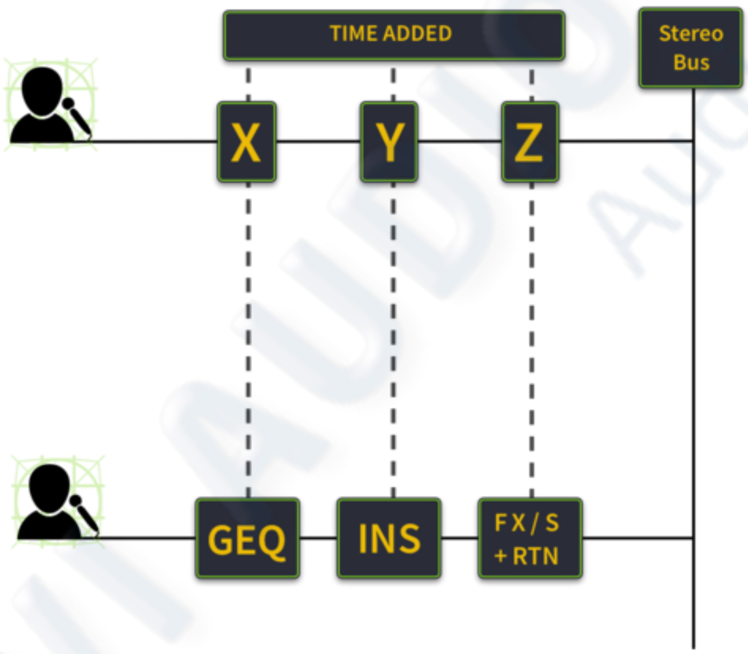
To change the time of the console simply select your current time zone. This means all your files and presets will be time stamped with the correct time and date for future reference if required.

# Chapter 27: Delay Compensation (Latency)

## 27.1 About Delay Compensation

A time delay is induced in a channel's signal by placing, for example, an insert (internal or external) or GEQ in its path. This delay affects system latency and can also produce undesirable audio effects such as comb filtering. To overcome this the HD96-24 system incorporates a system of user-configurable delay compensation parameters. These are presented to the user in the form of button-selectable options in the GUI and can be switched on or off to suit the current application with a range of presets to help you choose.

In the diagram below, two microphones are both being sent to the stereo bus. The first channel has no processing turned on. The second has a GEQ, Insert and a Send and Return FX active which all add different amounts of time. This means the second channel arrives slightly later than the first which could cause phasing or comb filtering issues. In order to compensate for each processing block. The same time difference are added to the first channel (**X**, **Y** and **Z**). Now the two channels arrive at the stereo bus at the same time. Delay compensation insures, depending on settings that all your Inputs and outputs are perfectly aligned creating that glorious Midas sound.




To access the delay compensation options:

In the GUI, choose Menu > Preferences > and then click the DSP Configuration tab.


1. **Input Channel** - Time-aligns the output of all input channels, regardless of whether or not they have an active insert. When this option is switched off, any input channels with inserts will be delayed relative to those input channels that do not have inserts. A choice of 1,2 or 3 inserts can be compensated for. Tip - If no inserts are used in the input channel layer, switch this option off to reduce the overall system latency. If there is an insert on any input channel, switch this option on.
2. **Flexi Aux Channel** - Compensates for inserts placed in flexi aux busses. To do this it modifies the delay that sits between the input channel outputs and aux, master and matrix channel inputs, so that signals fed from flexi auxes to masters will line up with signals fed from inputs through auxes to masters. A choice of 1,2 or 3 inserts can be compensated for.
3. **Aux Channel** - Compensates for inserts placed in aux busses. To do this it modifies the delay that sits between the input channel outputs and master/matrix channel inputs, so that signals fed from inputs to masters will line up with signals fed from inputs through auxes to masters. A choice of 1,2 or 3 inserts can be compensated for. Tip If no effects are used between the aux and return channels, switch this option off so that overall system latency will be reduced. If any effects are used between any auxes and returns, switch this option on.
4. **Master/Matrix** - Time-aligns the output of all the master and matrix channels, regardless of whether or not they have an active insert. With this option switched on, the outputs of any master or matrix channels using inserts will be delayed relative to the equivalent channels not using them. A choice of 1,2 or 3 inserts can be compensated for. Tip If no inserts are used in the master/matrix channel layer, switch this option off to reduce overall system latency. If Inserts are used in any master/matrix channels, switch this option on.
5. **Number of inserts to be compensated for** - A choice of 1,2 or 3 inserts can be compensated for.
6. **Align All Outputs** - Time-aligns the output of all the flexi aux, aux, master and matrix channels, regardless of whether or not they have an active insert. With this option switched on, the outputs of any flexi aux, aux, master or matrix channels using inserts will be delayed relative to the equivalent channels not using them.
7. **Aux Channel > Send/FX Return** - This option compensates the inputs to master and matrix paths for the signal path between an aux through an effect, and back through a return to the master and matrix channels. Tip - If no effects are used between the aux and aux return channels, switch this option off so that overall system latency will be reduced. If any effects are used between any auxes and returns, switch this option on.
8. **Flexi / Aux Bus Compensation** - This allows flexi aux to be time compensated to auxes for stem style mixing.
9. **Master > Matrix post-processing** - You have a choice of tap-off point, so you can choose to send either a pre-master or post-master channel processed signal to the matrix channels. This is a global setting and affects all master -> matrix contributions.

## 27.2 Insert Compensation

If a channel insert is active, it takes a finite amount of time for the signal to be sent through an internal or external effect and returned to the channel. Therefore, with no insert compensation, channels with inserts assigned are delayed more than channels that don't have an insert assigned to them. If two correlated signals with different delays are mixed together, this can produce comb filtering.

To avoid the comb filtering effect, the HD system insert compensation works by delaying all channels except the ones that have inserts assigned. In practice, the actual delay used for compensation depends on the type of insert (internal/external) and its location (stage/FOH). Each channel type or layer within the control centre, such as, input, aux, master or matrix, has its own parameter controlling the delay compensation for that layer. This provides the user with the maximum flexibility and allows the control centre to be configured for the lowest latency for a given application.

## 27.3 GUI Delay Compensation Presets

The system delay compensation (latency) is configured in the Delay Compensation section found in Preferences in the GUI side bar menu within the DSP page.

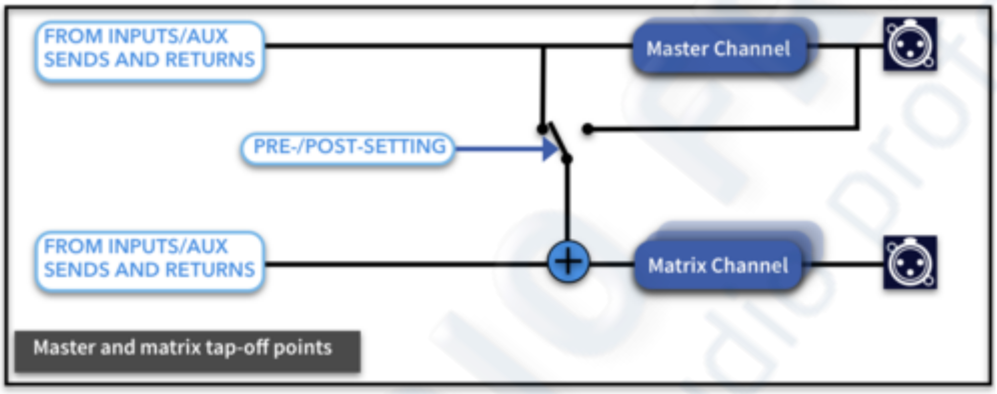
For a description of the delay compensation options and details of when best to use them, see the table above. In this table the Description column explains what happens when the delay compensation option is selected (switched on).

1. **Minimum** – Turns all delay compensation off (not recommended as comb filtering may occur).
2. **Monitor Low Latency** – Can be used when mixing in ear monitors.
3. **Monitor** – Recommended for mixing on monitor speakers.
4. **Mixed FOH/Mon** – Use when mixing both FOH and Monitors from one console.
5. **FOH Low Latency** – Can be used if required (less compensation used).
6. **FOH** – Recommended for the best quality Midas sound (if in doubt use this setting).
7. **Load** – Load the selected Preset.

## 27.4 Master to Matrix Post-Processing

This is the last option in the delay compensation page. You have a choice of tap-off point, so you can choose to send either a pre-master or post-master channel processed signal to the matrix channels. This is a global setting and affects all master -> matrix contributions.

The signal path that feeds master bus signals onto matrix channels is fully compensated for, so that signals fed directly to matrix channels or indirectly to matrix channels via master channels will always line up at the outputs, as will signals sent only to masters or only to matrix channels.



# Chapter 28 : Panel Connections

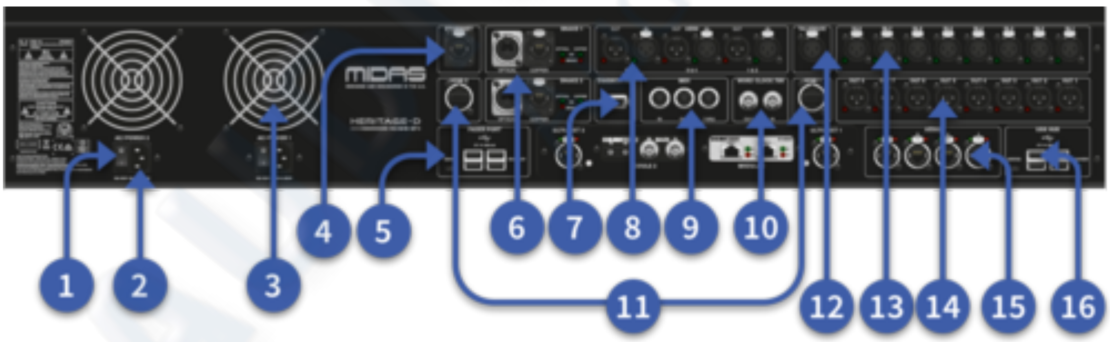
## 28.1 Front Panel Connections


A USB drive can be used on the front panel to save and load shows, user libraries or other settings. Any standard USB key can be used with no limit on size of key.

The Talk Mic input can be used to add a talk back mic, a gain control next to the input can be used to adjust the mic pre amp to a suitable level.

6.35mm Headphone sockets can be found under each side of the surface. The left side signal is taken from the monitor A bus and the right side headphone socket from the monitor B bus.

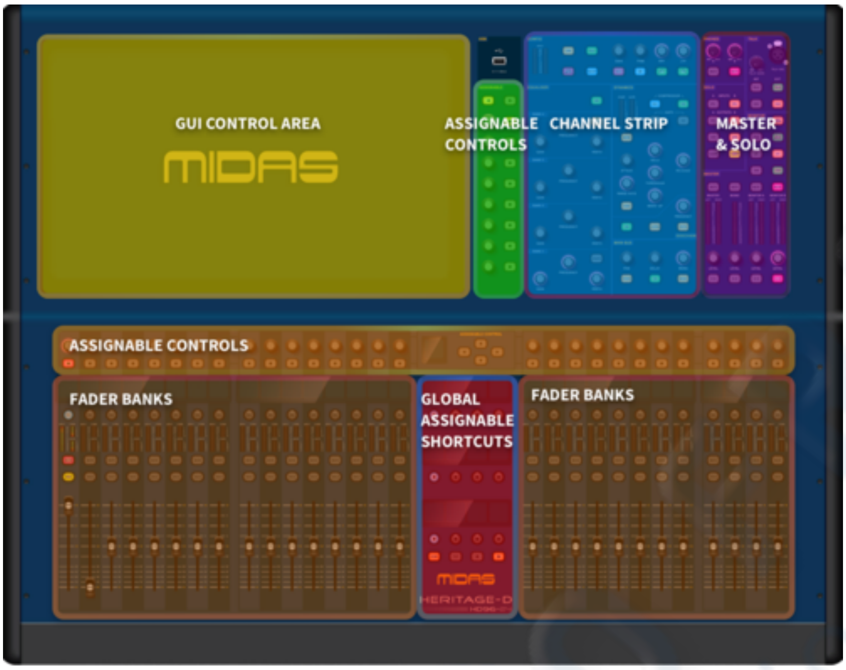
## 28.2 Rear Panel Connections

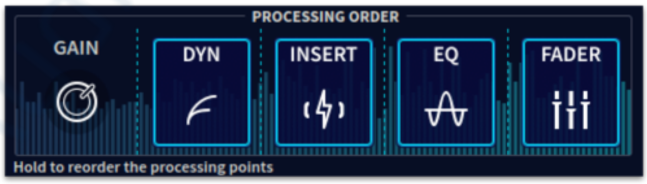




1. Power on switch
2. 100v-240v IEC socket
3. Internal fan
4. Ethernet port
5. Fader port USB hub
6. 2 x Hyper-mac snake connections (192 channels per connection via optical fibre or Cat5e copper)
7. Diagnostics port (for Midas use)
8. AES3 1-4 Digital I/O and AES Sync I/O.
9. MIDI In, Out Thru
10. Word clock I/O BNC 75 Ω
11. HDMI outputs for connecting external monitors
12. Talkback input
13. 8 x Midas microphone input preamplifiers
14. 8 x Output XLRs
15. 4 x AES50 ports (24 bi-directional channels each)
16. USB Hub for connection of keyboards or other USB devices

# Chapter 29: Inputs

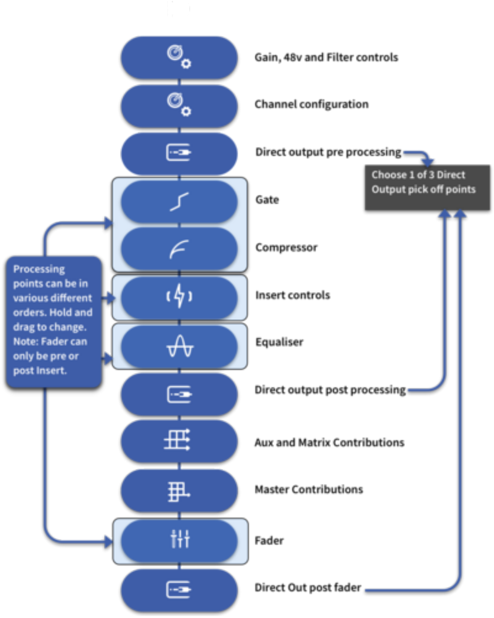
## 29.1 Input channel routing



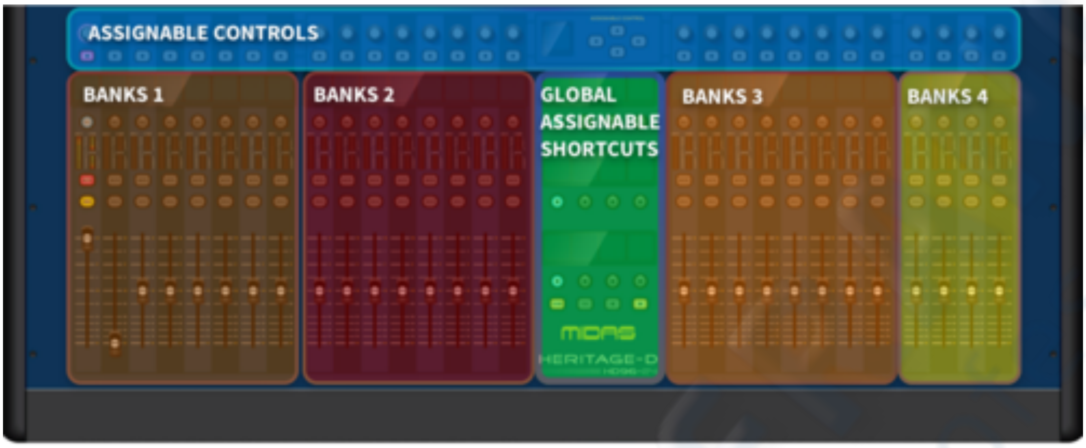

All inputs channels are mono by default, although any 2 adjacent channels can be either linked to form a stereo pair or a full stereo channel. This is achieved in the Configuration widget and described in the Stereo Linking chapter. The order of processing in the signal path of all channel types is basically the same.

The diagram below shows the default signal path. This chapter will explain each of these groups of controls, showing the significant functions on both the control surface and GUI.



## 29.2 Input Channel Areas of the Control Surface

The surface has 28 possible locations for inputs to be freely assigned to. Channels can be located in any of the 3 Banks of 8 and/or 1 Bank of 4 as discussed in the Navigation section earlier in this manual. When an input channel is selected, its channel settings are brought to the surface controls and to the GUI input widget.




## 29.3 Input Widget Area


When an input channel is selected via the touch screen or surface SEL button, its overview appears in the Input Channel widget or Channel View widget (dependant on workflow selected, FOH view shown above) and provides all the information needed for adjusting the different sections relating to input function.

The following processing sections are available:


1. Configuration (Config, Linking & Stereo, Patching, Direct Output and Options)
2. Equaliser
3. Gate
4. Compressor
5. Effects
6. Sends

For details of how to navigate the GUI channel strip, see “Navigation via the GUI”

Configuration

Linking & Stereo

Patching

Direct Output

Options

Pages in the Configuration tab

Equaliser

Gate

Compressor

Effects Insert

Sends Overview

## 29.4 Channel View Workflow

The Channel View widget utilises a large portion of the GUI in order to display as much information as possible on the selected channel.


1. Touching into each area opens up those settings into the right-hand side for detailed editing. This workflow is great for sound check given an overview of all the essential tools needed to craft a great mix.
2. The different processing areas function in the same way as the surface controls and separate input widget pages.
3.

**Note:** To view the right side of a stereo channel press the SEL button a second time.

## 29.5 Side Bar Area


The HD 96-24 has various side bar views which can be easily changed and display a wide variety of information in different ways. The following sections detail the various different displays.

**Tip -** Auto One Shot Pot mode automatically maps rotary controls from the currently selected widget area to the 8 assignable pots to the left of the screen.

## 29.6 Channel View


This view displays currently selected channel information and aids navigation by pressing to access various different pages depending on which workflow is active.

1. **Side Bar View** - navigation arrows select view.
2. **Channel Info** – Displays Channel Name, Icon, Colour and Tags. Press in the area type you wish to edit which then opens the channel list edit page.
3. **Configuration** - Opens the configuration widget.
4. **Gate** – Press to open the current channel's gate settings.
5. **Compressor** - Press to open the current channel's compressor settings.
6. **Equaliser** - Press to open the current channels EQ settings.
7. **Effects** - Press to open the current channels effect page settings. The green boxes indicated an effect is active the bottom green box shows the insert is active.


## 29.7 Sends Overview

This view displays the currently selected channel's Aux and Matrices send level and Pre/Post information.

The Channel Detail Area gives a simultaneous display of the status of all busses. It displays the if the bus is stereo or mono, levels sent to the busses, shows which are on/off, if the Aux is a Flexi Aux (FA) and if they are pre or post-fader.

- Mix bus sends are brightly colored when they are on.
- Mix bus sends are dimmed and grey when they are off.

1. Five blocks aid navigation. They are split into Aux 1-24, 25-48, 49-72, 73-96 and Matrices 1-24.
2. Each of the 5 blocks, once selected, displays 3 sections of 8 showing Level, Pan, Mono/Stereo state, Flexi Aux (FA) status and Pre/Post send state (Solid for Pre fade sends, Ramp for Post fade sends).

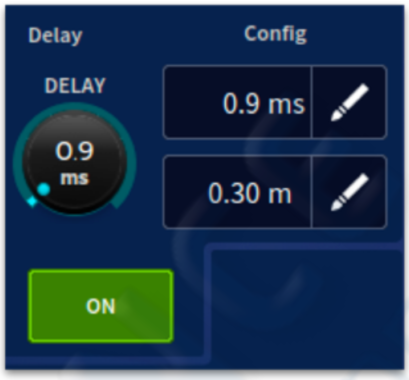

# 29.8 Send Detail Area

This view shows the send levels for the currently selected channel for 8 mono or 4 stereo output’s at a time.

1. Navigate left and right using the arrows in blocks of 8.
2. Each Aux Displays in either Mono or Stereo. There is a level and pan control (if stereo). A slope indicated a post fade send. A block indicates a pre fade send. There are 3 meters which display channel Send level, Aux Output Gain Reduction and Output level.

**Tip:** This is a great place to mix/control In Ear Monitors (IEMs) from. When increasing the volume of the selected channel in an aux mix you can also see how this affects the outputs gain reduction.

## 29.9 Input Channel Delay




The input channel delay can be changed via the delay section of the configuration processing area (GUI channel strip) or from the delay control in the Master Section on the control surface. The control for adjusting the delay in the range 0ms to 50ms; this value is displayed in both milliseconds (ms) and meters (m). You can fine tune the delay value in the GUI by dragging your finger further out from the pop-up control. The delay on/off button lets you bypass the delay to hear the difference in real time.

**Note:** Pressing the delay time will display a numeric keypad to manually input delay time.

The delay section lets you incorporate a time delay on an input channel, which is used mainly for mic placements and time aligning to reduce comb filtering. For example, on a bass drum you may have an internal microphone and a mic in the bass drum front head hole. In this case, setting an input channel delay on the internal drum, to bring it more in line with the hole mic, may produce a better sound.

## 29.10 Aux Return Mode


When the Aux Return Mode button is active on an input channel it prohibits that channel becoming a tape return when virtual sound check is engaged. This means aux returns channels can be used for effects with virtual sound check playback. Aux Returns are also delay compensated to allow for effect timing variations. This function can also be used if you need to let a member of a band play along to the virtual sound check by placing their own channel's in Aux Return Mode. After they have finished rehearsing return the channels to the normal state.

# 29.11 Effects Inserts

The Effects Insert page allows you to add 3 effects units if DSP is available. Additional effects can be chained together in the patching page if required.

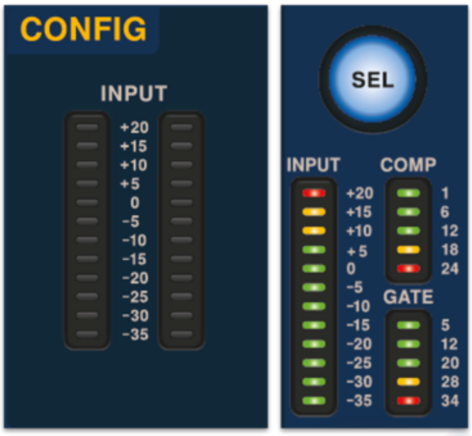
Pressing the Add button opens up the Select or remove page. Tap on the effect you wish to use and press the Add button to load that effect into the chosen slot. If enough DSP is available, the effect will load. The red bar will wiggle if there is not enough DSP left. The amount of DSP left is shown by the Slots Available indicator.



Touching each filled effect slot or the Edit button will open the effect ready for parameter adjustment. For details of effect types and control see the effects chapter. To remove an effect, touch Edit on the slot you wish to clear. The press and hold Remove. A line will trace around the Remove button and after this has completed the effect will be removed. This is to stop accidental removal of effects.

## 29.12 Input Metering

In the Configuration widget mic gain is displayed on a 53 segment LED style display which gives an accurate representation of the level of signal within the system . This level information is also displayed on the surface in various different areas including the Channel Banks fader area and in the Config area to the right of the touch screen. The meters will show either stereo or mono signals depending on channel's link configuration.


1. The surface uses a 12-segment LED meter in both the banks and surface config area. These default to monitor pre-fader signal level. Meter range is +20 dB to -35 dB, in 5 dB increments on small meter displays (shown below); 1 dB increments on the large meter displays (above).

## 29.13 Input Inserts

Input channel insert section provides a send out of - and return to - signal path, primarily so that external dynamic or effects processors can be added to the signal's processing. The send destination and return source may only be set from the GUI screen, although the INS button can be found on both the GUI and also in the Config area. This section is optional and assigned on a channel-by-channel basis.

A return B path is also available. This takes the place of the Return A path when activated in the channel widgets Effects tab in all workflows.

**Tip:** An example of how to use this could be to run two separate external inserts on a channel with the 2nd external unit as a spare. If the first external unit fails, turn on the Return B path.

## 29.14 Direct Output


The direct output section provides a way of leaving the surface via an I/O box or a multitrack recording system. It lets you take a signal directly out of a defined point in the input channel's signal path and route it a physical output (a physical connection via one of the I/O boxes).

The pick off point by default is set pre-processing ready for virtual sound-check use. It can be set post processing or post fader if required.

**Note:** The Direct Out Mute can only be un-muted if the direct out is patched.

## 29.15 Options - Processing Order


The processing order section in the options page of the configuration widget can change the position of the EQ, Insert or Dynamics in any order you like in the input channel's signal path by holding and dragging to the desired order.

It is also possible if the Insert icon is before the Fader icon to make a post fader insert by dragging the Insert after the fader.

**Note:** Gate always precedes compression in the DYN block, no matter how the processing order is set. Fader can only move before **Insert** to create a post fader insert point.

## 29.16 Safes

Each input channel has 8 different Safes that protect specific controls/areas or the whole channel against changes from the automation system.

You can switch the safes on/off by using the Safe buttons in the Options tab. The same buttons are also displayed on the Channel View widget.


The 8 options are:

- All
- Gate
- Comp
- Eq
- Filters
- Config (All config parameters apart from elements in the Main Bus)
- Mute (Channel Mute and Direct Out Mute)
- Main Bus (includes Pan, Pan Mode, Send to Stereo, Channel Fader Level, Send to Mono Bus and Mono Bus Level)

## 29.17 Mic Amp Input Gain

There are two types of mic input channel controls: digital (trim) and remote analogue preamp (gain). Most channel controls are digital, which directly affect the parameters stored within the DSP. However, a few controls can also be thought of as remote controls, which control the physical components of the mic splitters and even components that are in the signal path before it enters the digital domain.

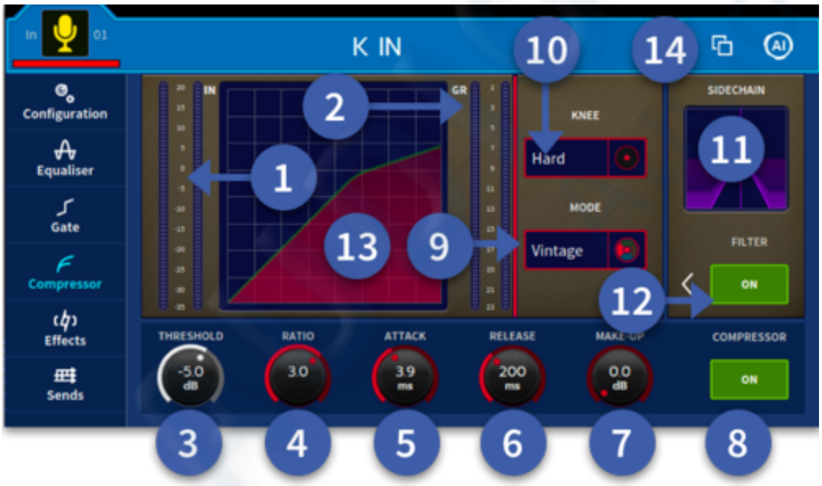
The remote controls are dependent on the types of I/O box. For example, the analogue input of a 251 has a 48V phantom voltage button and a gain control. The controls are adjusted via the device's configuration window (see Configuring the Devices in Chapter 7).

# 29.18 Dynamics and Mode Descriptions

The dynamic section controls two dynamic processors present in the input channel signal path, that is, the compressor and gate. The GUI treats both devices independently, the processing area currently displayed in the channel widget or single channel view being the one currently displayed in dynamics area on the surface, highlighted by different coloured controls, the default colours are Purple for Gate settings and Green for Compressor settings. Swapping between the two dynamic devices can be done via the SEL key in the dynamics section or via the GUI (see Chapter 6 Navigation). Note: swapping the gate/comp selection on the surface will swap that displayed processor in the widgets; swapping on the widget will NOT change the surface selection. Operating the dynamic device's ON button activates the device, but also affects the audio. You can select the source of both the compressor and gate, and also use the sidechain for both.

The sidechain filter is a swept band pass type, which acts on the dynamic's sidechains of the compressor and gate/ducker which covers the full audio spectrum. There are 4 filter Q widths available, 0.1 Oct (tightest), 0.3 Oct, 1 Oct and 2 Oct (widest). This allows you to narrow the frequency response of the sidechain for more control. The sidechain graph shows the effects of the sidechain filter on the signal, selectable via the Width button.

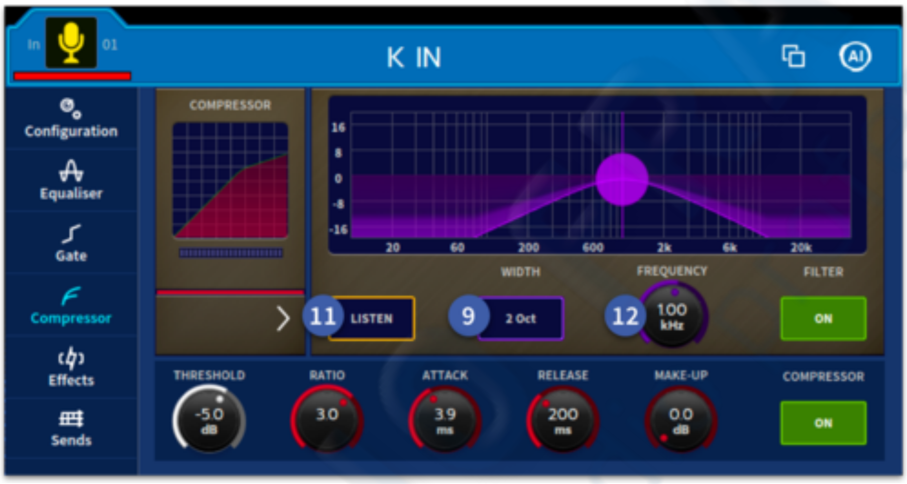
Sidechain Listen button, places the sidechain on to the channel filter bus, allowing the audio signal to be monitored via the solo bus.





1. Input Meter - Input metering to the compressor.
2. Gain Reduction (GR) - Gain reduction meter.
3. Threshold - Sets the signal level above which gain reduction starts to be applied. Range is from -50 dB to +25 dB.

4. Ratio - Adjusts amount of compression applied to signals above threshold. Range is from infinity 1:1 to 25:1. When set to 10:1 or higher, the compressor acts as a limiter.
5. Attack - Adjusts time for compressor to respond after an over-threshold signal. Range is from 0.2 ms to 100 ms.
6. Release - Adjusts time for compressor to recover after programme material falls back below threshold. Range is from 0.05 s to 3.00 s.
7. Make-Up - Compensates for the reduced loudness of a compressed signal. Range is 0dB to +24dB.
8. On Switch - Enables the compressor in the signal path.
9. Mode Selection - Selects compressor mode. There are four compressor types available: Corrective, Adaptive, Creative and Vintage.
10. Knee Type - Switches between the 3 types of Knee settings and controls how the compressor starts to apply gain as the signal goes through the threshold. See Knee (below) for a description of the types.

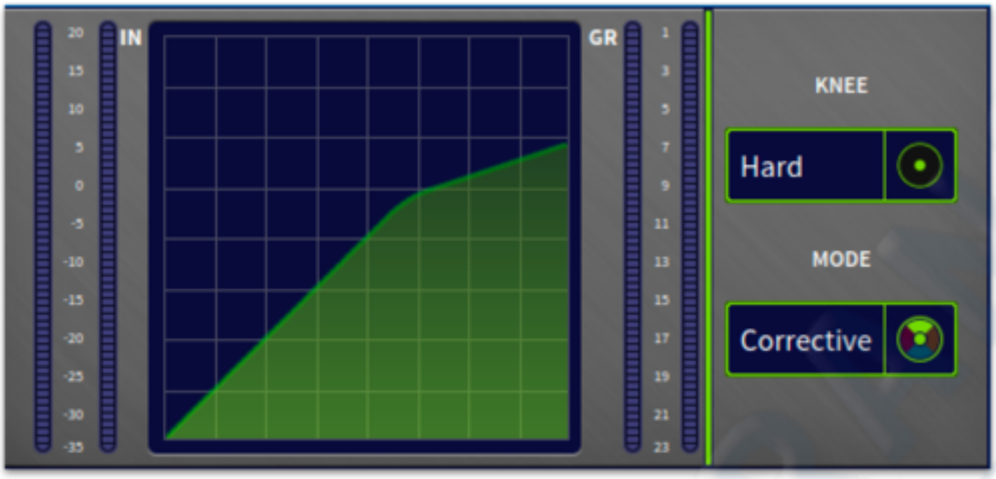



11. Sidechain Control - To aid set up, the compressor has a sidechain listen that sends the sidechain onto a solo bus. The listen LED illuminates to warn you that soloed material is from the sidechain, and not the main channel.
12. Filter Control - On Switch - Adjusts the sidechain filter frequency in the range 50Hz to 15kHz. (Visually, the arrow moves the sidechain parameters view in and out of view)
13. Comp Graph - See "Compressor graph".
14. Copy & Paste - Copy and Paste button. See Copy and Paste chapter for details.
15. Compressor SEL – assigns the compressor to the surface control (can be gate or compressor).

**Tip:** Try compressing a Bass Guitar using a Bass Drum to trigger the sidechain of the compressor. This will let the Bass Drum cut through a mix a little more.

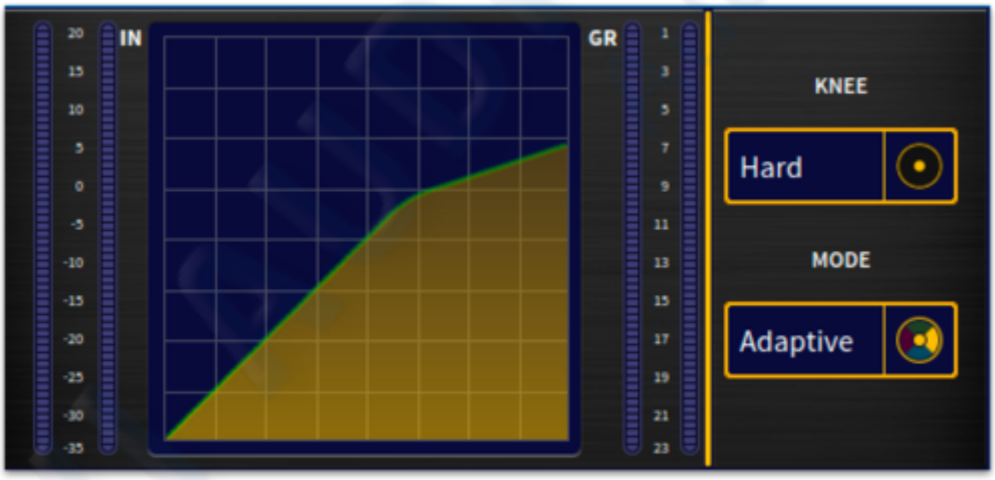
# Compressor

## Corrective mode (exponential peak - fast)



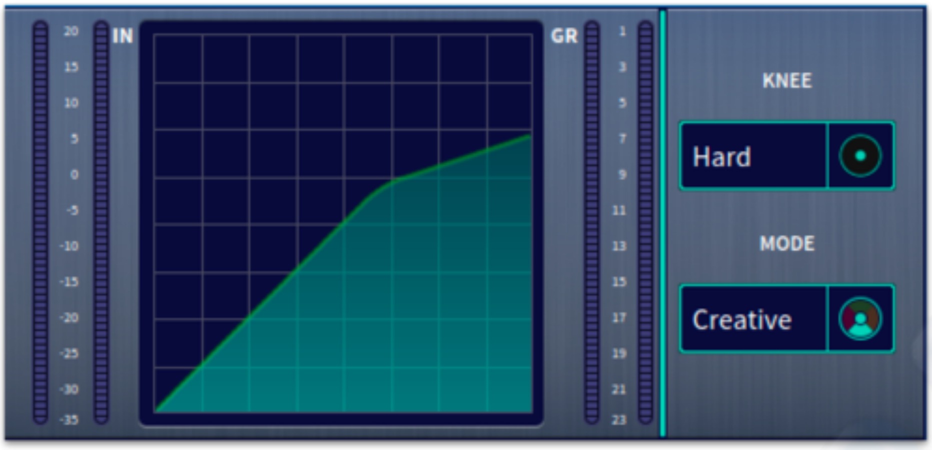

This is a peak sensing compressor (like many older designs) with exponential attack and release. It produces aggressive compression that gives good fast control/limiting of dynamic material. It can be used to add color to low frequency signals, thus making it ideal for controlling extremely dynamic instruments like bass guitar. The compressor tends to sound best with fast attack time settings that capture transients and with release adjusted to taste to either emphasise or reduce distortion and pumping effects.

## Adaptive mode (exponential RMS - accurate)




This is a root-mean-square (RMS) sensing compressor with exponential attack and release. The RMS averaging process interacts with the attack and release to produce a very adaptive envelope character. This allows faster attacks on large (over-threshold) signal changes and produces slower attacks on small signal changes, regardless of attack time setting. It is also sonically accurate and works well for both compression and limiting of vocals and many other sources.

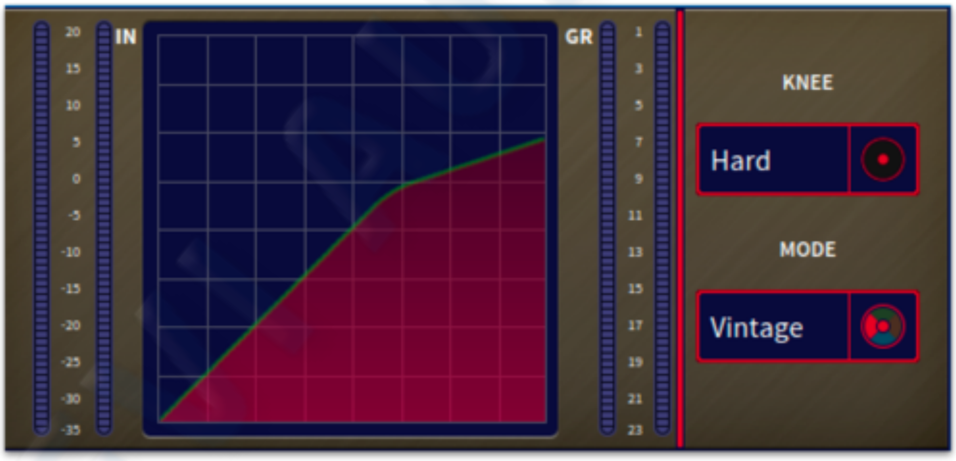
## Creative mode (linear peak - slow)




This is a peak sensing compressor with linear (dB rate) attack and second order release. The compressor is very transparent, providing some dynamic control but without unduly affecting the intentional dynamic content of the source material. The linear attack provides a constant rate of attack, such that large changes in source signal level take longer to become compressed than smaller changes. Adding soft knee noticeably delays these attacks, which can be particularly useful on drums where compression can be applied to emphasize transients giving more punch while retaining a good deal of artistic dynamic range from the drummer.

The compressor normally sounds best with slower attack time settings, when it can be used on difficult instruments, such as the acoustic guitar, with relatively fast release to keep equal perceived loudness within a mix without producing excessive flutter or distortion.

## Vintage mode (adaptive peak - bright)



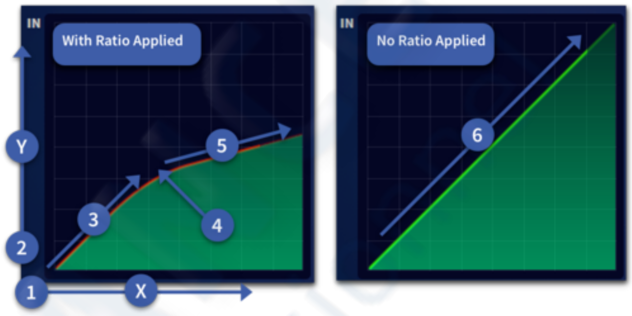

This is a peak sensing compressor with a partially adaptive nature. It produces extremely subtle attack and release curves during the onset of compression that are largely independent of the envelope control settings. However, as it is driven harder, that is, signals are further over-threshold, the attack and release times become more aggressive and gradually return to manual control so that the operator can optimise the capture (or otherwise) of larger transients etc. The peak sensing algorithm intentionally increases harmonic overtones during compression, which adds a tube or valve-like brightness and sparkle to the signal, producing extremely natural and lively sounding compression of acoustic instruments.

## Compressor Graph

This section uses examples to illustrate the effect on the compressor graph of adjusting the compressor's parameters.

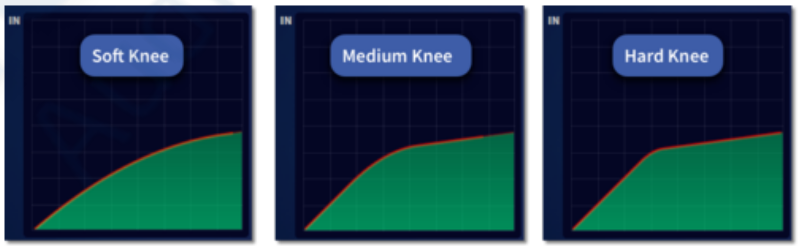
### Ratio

The following diagram shows a signal on the compressor graph with ratio applied; it shows the point of threshold and how ratio affects the gradient of the signal following this. The graph on the right shows an uncompressed signal, that is, with no ratio applied.




1. Input level. The 'x-axis' of the graph.
2. Output level. The 'y-axis' of the graph.
3. This portion of the graph is pre-threshold and is unaffected by compression, that is, with a gradient of 1:1.
4. This portion of the graph is post-threshold and shows the effects of compression. The gradient is the same as the compression ratio.
5. Threshold, the point where the gradient changes and where compression starts to be applied.
6. Graph with no ratio applied, that is, 1:1 gradient. No effect on audio.

### Knee




The soft knee curves behave in a traditional way to blend the compression ratio around the threshold setting but more importantly they also have a significant effect on the attack envelope shapes. The soft knee typically slows down attack speed on signals in the knee area, which is desirable for natural sounding compression because it compliments the reduced ratio effect of the soft knee. This produces very gentle compression in the knee region.

The Knee switch has three settings: Hard (4 dB); Medium (12 dB); and Soft (40 dB). In hard setting the compressor still retains some soft knee characteristics. This is because the implementation of an extremely hard knee produces undesirable sounding distortion on low frequency programme material.

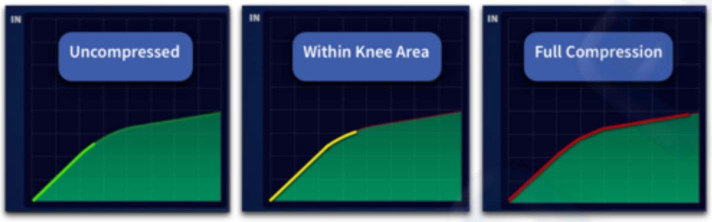
- Hard knee Compressor immediately applies gain reduction at selected ratio once attack time has elapsed.

- Medium knee Intermediate knee type.
- Soft knee Compressor, starting from slightly before threshold, gradually makes the transition to applying gain reduction at selected ratio.

## Compressive Display Types

With a signal running through the compressor, a coloured line on the graph follows the contour of the shaded graph area. The line's colour changes according to which of the three signal levels it is at.

- Uncompressed If signal doesn't reach threshold (point where gradient changes), the line is green. As the threshold is not exceeded, the signal is uncompressed.
- Within knee area If a signal goes into the knee area to point where the gradient changes (more obvious with medium and soft knees), compression starts to be applied and the line colour changes to yellow.
- Full compression If the signal reaches the point where the gradient changes (over threshold), full compression at selected ratio is applied and the line colour changes to red.




Typical graphs showing the line representing the signal level as it follows the contour of the compressor envelope and changing colour as it passes through the knee.

# Gate

While the dynamic section is addressing the gate, all of its controls are enabled except the Make Up control and KNEE buttons.

1. Threshold - Sets signal level at which gate opens. Range is from -50 dB to +25 dB.
2. Range - Control adjusts the amount of gain reduction applied to the signal below threshold. Controls the maximum gain reduction that is possible. Range is from minus infinity (-∞) to zero.
3. Attack - Adjusts time taken for gate to open after an over-threshold signal. Range is from 0.02 ms to 20 ms
4. Release - Adjusts time taken for gate to close after programme material falls back below threshold. Range is from 2 ms to 2.000 s.
5. Hold - Minimises chattering in conjunction with internal hysteresis. Once the signal drops below the threshold, this defines a waiting period before the gate starts to close. Range is from -0.005s to 2.000s
6. On - Enables the gate in the signal path. When switched off, gate is bypassed.
7. Input level - Shows input into the gate.
8. Gain Reduction - Shows attenuation of signal in dB.
9. Sidechain Display - Shows the SidechainSfrequency graphically.
10. Filter On/Off - Enables the Sidechain filter into the signal path. Use arrow to expand Sidechain view
11. Mode - Switch between Gate or Ducker

## Ducker

In radio this can typically be achieved by lowering (ducking) the volume of a secondary audio track when the primary track starts and lifting the volume again when the primary track is finished. A typical use of this effect in a daily radio production routine is for creating a voice-over.

Use of the sidechain allows the ducker on the channel you wish to be turned down to be triggered by the channel you wish to have priority.

## Sidechain

The sidechain filter is a swept band pass type, which acts on the compressor and gate/ducker which covers the full audio spectrum. There are four filter Q widths available, 0.1 Oct (tightest), 0.3 Oct , 1 Oct and 2 Oct (widest). This allows you to narrow the frequency response of the sidechain for more detailed control. The sidechain graph shows the effects of the sidechain filter on the signal, selectable via the Width button.

Sidechain Listen button, allows the audio signal to be monitored via the solo bus for fine tuning the frequency selection.

**Tip:** For snare drum gating try setting the width at 0.3 Oct and the frequency around 2k. This can help cut out spill from other close drums and give a cleaner drum sound. Snare bottom mics could also be triggered or keyed from the snare top mic to allow the gate to open faster and allow super quick transients through.

# 29.19 EQ

The input channel equaliser is a four-band parametric EQ that allows tonal control of the input signal via the surface parametric EQ section or via the GUI. An additional three shelving modes are available for bands 1 and 4. Any combination of the four bands can be used to manipulate the signal's frequency response.

The Input and Channel View widgets contain the EQ on/off switch and all parameters to adjust the EQ section.

The Equaliser section on the surface contains all of the EQ controls, along with a shelving mode selection button and another set of band selection buttons.

The Input and Single Channel View widgets display all four bands simultaneously and has a display that shows a coloured EQ envelope for each selected band. Here, you can view the settings of the four bands simultaneously plus the two filters show in purple.


1. On - Enables the EQ in the signal path.
2. Gain control - Adjusts selected band's signal gain in the range -16dB to +16dB. Changes will be displayed on the graph.
3. Frequency - Adjust frequency within the 16 Hz to 25 kHz range.
4. Width - Adjusts the signal bandwidth in the range 0.3 Oct to 5.3 Oct. In the EQ display width will be displayed and tighten/widen as changed. (Not available for treble and bass shelving modes.)
5. Band 1 – Selects band 1.
6. Band 2 – Selects band 2.
7. Band 3 – Selects band 3.
8. Band 4 – Selects band 4.
9. HPF Point - Visual display of High Pass Filter which can be moved by touch.
10. LPF Point - Visual display of Low Pass Filter which can be moved by touch.
11. Flatten EQ - Press and hold to flatten EQ Gain values. All Frequency and Q values are unaffected by flattening.
12. Phase - Phase (see Phase and All Pass Filter section).


1. EQ Audition Mode - EQ Preview (see EQ Audition section).
2. EQ Preview - Preview EQ in different scenes and apply to current scene.
3. Copy & Paste - EQ copy & paste (see Copy & Paste Chapter 17).
4. AI settings - Opens the EQ AI options.
5. Band 1 & 4 shape - Changes the band 4 filter shape between Bell, Bright, Classic or Soft. Changes the band 1 filter shape between Bell, Deep, Classic or Warm.


1. Filter On/Off - High/low pass filter on/off.
2. LPF/HPF Freq select - HPF 10 Hz to 400 Hz - LPF 2 kHz to 20 kHz.
3. Filter slope - Either 12 dB/Oct or 24 dB/Oct for the Hi Pass Filter; 6dB/Oct or 12dB/Oct for the Lo Pass Filter.

The HD96-24 input EQ comprises four bands: treble; hi mid; lo mid; and bass. The default operation for all four sections is full parametric sweep (peak), with the following controls:

- Gain: continuous adjustment of boost and cut from + 16dB to - 16dB with a 0dB centre.
- Width: continuous adjustment f bandwidth from 0.3 to 5.3 octaves (this only operates in bell mode for Bass and Treble).
- The Treble EQ band can be switched from bell to any of three other shelving modes:
- Soft, Classic, and Bright.
- The Bass EQ band can be switched from bell to any of three other shelving modes:
- Warm, Classic, and Deep.

# 29.20 Filters

The difference between the shelf filters is subtle and, if you do not have time to experiment, it is probably best to use classic because this is the best all round filter. However, when you do have time to experiment you may find the other types each have their uses. The minimum harmonic types, and in particular the bass, can sound

very natural, even with very aggressive EQ, but the psycho-acoustic principles that they operate on do not always work as well on multiple source or pre-mixed material.

### Soft treble

The soft treble response provides a very gentle gradient between EQ'd and non-EQ'd frequency areas. This produces the absolute minimum of phase shift but does not provide much differentiation, thus frequencies outside the area of interest are often unintentionally EQ'd. This is best used to provide gentle shaping of pre-mixed material.

### Classic treble

The classic treble response provides a much steeper gradient between EQ'd and non-EQ'd frequency areas, as made famous by previous Midas consoles like the XL4. This provides better differentiation and minimal phase shift, but there is some undershoot error, that is, when boosting the treble, the midrange is slightly cut and vice versa. This is the best all round EQ and especially effective when microphones are

covering multiple sources.

### Bright treble

The bright treble response provides a slightly steeper gradient than the classic and it is uniquely shaped to provide minimum harmonic disruption to the EQ'd material. As for the classic EQ, this provides better differentiation and minimal phase shift, but now the mid-range is not changed as much. This filter is best used on single source material and especially good for acoustic performances.

### Warm bass

The warm bass response provides a very gentle gradient between EQ'd and non-EQ'd frequency areas. This produces the absolute minimum of phase shift, but does not provide much differentiation, thus frequencies outside the area of interest are often unintentionally EQ'd. This is best used to provide gentle shaping of premixed material.

### Classic bass

The classic bass response provides a much steeper gradient between EQ'd and non-EQ'd frequency areas and is modelled on the XL4. This provides better differentiation and minimal phase shift, but there is some undershoot error, that is, when boosting the bass, the mids are slightly cut and vice versa. This is often desirable on bass EQ and it is the best all round, general purpose EQ curvature.

## Deep bass

The deep bass response provides a slightly steeper gradient than the classic and it is uniquely shaped to provide minimum harmonic disruption to the equalised source. As for the classic EQ, this provides better differentiation and minimal phase shift, but there is no undershoot error. Powerful boost/cut can be used that still sounds very natural and do not alter the mid-range. This is best used on single source material.

## High Pass filter (HPF)

The HPF attenuates (not boosts) all frequencies below a certain level (cut-off frequency) while allowing all those above it to pass through. The harshness or smoothness with which the sound is removed beyond this point is determined by the dB/octave, with 6dB being the most common. The HPF is generally used to take rumble or hum out of any sound source but may also produce a sound effect by manipulation of the controls.

## Low Pass filter (LPF)

The LPF attenuates (not boosts) all frequencies above a certain level (cut-off frequency) while allowing all those below it to pass through. The harshness or smoothness with which the sound is removed beyond this point is determined by the dB/octave slope selection, with 6dB being the most common. The LPF is generally used to reduce noise in quiet passages with excessively high frequencies but may also produce a sound effect, like a filtered drum roll, by manipulation of the controls.

# 29.21 Phase (All Pass Filter) Controls

The Variable Phase feature in each channel's equalizer section allows the user to alter the phase of a signal by a variable amount. The effect works by using two all-pass filters in series and controlling the center frequency of the filters to change the phase shift. The all-pass structure allows for a flat magnitude-frequency response; however, the filters delay different frequencies by different amounts resulting in a frequency-dependent phase shift.

Additional features have been added to this fundamental design by allowing control over the frequency range of the center frequencies via the High and Low settings. High can be used on instruments and vocals with a frequency range in the upper audio spectrum. While Low is best used on instruments with a greater low frequency content such as bass guitar or kick drum. The Phase Frequency Range allows a greater range of frequencies to be covered by the control. You can switch between a 0 to 90° or 0 to 180° phase shift range by using the 180/Deg or 90/Deg Phase Adjust Range buttons.


**Tip:** This phase control can be used to control phase alignments to inputs for multi mic setups, on a drum kit for example or multiple microphones on the same guitar cab.

**Note:** The Phase filter replaces the LPF filter. The Treble band can be used as a high filter by changing the shape to Bright, Classic or Soft modes.

1. The Phase function (All Pass Filter) is the last tab in the EQ widget display.
2. Phase On button. When Phase is activated it turns the LP Filter (Low Pass Filter) off.
3. Phase degree display, depending on the setting of the phase adjust range, either 0° to 90° or 0° to 180° scale. The line shows the degree of phase change.
4. The degree of phase change. Can be adjusted via this control or dragging control 3 up and down.
5. Phase adjust range by either 180°or 90°.
6. Phase adjust Frequency range. Can be set to High (for vocals, keyboards or guitars etc) or Low (for bass guitar or bass drums).

# 29.22 Mixes

Each input channel can send a variable contribution to each of the 96 flexi-aux busses (aux sends) and to each of the 24 matrix busses. The bus contributions are controlled by either the assignable controls or flip to faders that give continuous adjustment (in the range +6dB to off) of subgroup levels sent to matrix mixes. The controls in the assignable sections can include pan and level control for each bus, whose function depends on the current bus mode in operation.


1. Desired busses are flipped using the Flip button in the GUI or from the global assignable shortcuts. When input faders are flipped the LCD displays show Flipped To, which indicates Flip mode is active and the target aux or matrix. Select the Auxes or Matrices via the 4 tabs.
2. Select the mix or mixes you wish to adjust.
3. Aux Send On.
4. Aux Send Pre/Post.
5. Tap off point selection.
6. Choose how changes are made either Relative, Absolute or Pass Thru.
7. Send level changes are made here.
8. Long press to set sends to unity, 0 dB.
9. Send Pan position.

# 29.23 Input channel Pick Off points

Input channels can have the Auxes' pick off point changed on a per channel basis and can be changed in many different ways via the Manchino page, see chapter 33 for information. To change individual Pick Off points of input channels navigate to the Sends page in either the Home, Channel View, Automation or FOH workflow. Then select the bank the Auxes or Matrices you wish to alter.


**Note:** the tap off points cannot be changed for Flexi-Auxes or Matrices.

Select the Pick Off point to change to one of the following options for Aux and Matrix Sends:

1. Tap off menu
2. Post Filter.
3. Post EQ.
4. Post Insert.
5. Post Dynamics.
6. End of Chain. (After all channel processing).

## 29.24 Master Sections and Pan Control

The masters have extensive support on both the control surface and the GUI. In general, there are three routing switches to the master busses and pan control. Pan provides master panning and also provides two-way panning for any stereo mix groups, subgroups etc. When used for monitor applications, aux, mix and main busses are controlled from the channel master pan and fader. AFL solos also operate as a default from the main solo switch.


1. To Stereo Bus - This stereo switch connects the post-fader channel signal to the master stereo bus via pan control.
2. Pan control - Adjusts the relative levels sent to a left-right bus pair or the master left-centre-right (LCR) busses. In SIS™ mode, it can also control the 'image' to give a constant power crossfade from LCR to stereo.
3. To Mono Bus - This mono switch connects post-fader channel signal to the mono master bus.
4. Mono level - This function controls the mono signal level of the selected channel.
5. Solo B – Assigns the current channel to the Solo B Bus. This means when the channel is soloed it will appear on the Solo B bus and not the Solo A bus.

## 29.25 LCD Displays

The LCD Displays can show a variety of information including, name, colour, icon, pan, VCA assignment and if the console is in flip mode, which the auxiliary, if the channel is selected or matrix it is flipped to.


The LCD displays also have link status and a level meter to the side in either Mono or Stereo depending on the channel configuration.

# Chapter 30: Outputs

## 30.1 Outputs

There are five types of output: flexi aux, auxes, group, masters and matrices. All output channel strips can be seen in the GUI by selecting the outputs tab in the navigation page. Different widgets can be recalled by touching in each section of the output channel strip overview area.

It’s important to know the difference between the different output types:

| PATHS | | | | | |
|---|---|---|---|---|---|
| FROM | TO AUX | TO FLEXI AUX | TO GROUP | TO MASTER | TO MATRIX |
| INPUT | YES | YES | YES | YES | YES |
| FLEXI AUX | YES | NO | NO | YES | YES |
| AUX | NO | NO | NO | YES | YES |
| GROUP | NO | NO | NO | YES | YES |
| MASTER | NO | NO | NO | NO | YES |
| MATRIX | NO | NO | NO | NO | NO |

## 30.2 Output on Channel Banks

Any output, be it Flexi Aux, Aux, Group or Matrix can be assigned to any of the 4 banks on the control surface. In general bank 3 is used to show aux or matrix outputs with bank 1&2 generally assigned to show inputs but this is fully customisable. This flexibility allows quick adjustments of contributions to auxes or matrices using flip mode.


**Tip:** This is a great way to manage multiple in-ear mixes with up to 8 stereo Aux mixes displayed on a bank at any time. The Output global shortcut can be used to tab through all the aux sends or a POP group can be set up to recall Aux sends to the bank you require.

## 30.3 GUI Channel Strips


Similarly, to the inputs, when an output bank is selected (Flexi Aux, Aux, Matrix or Master), the selected outputs ‘overview’ appears in the GUI output overview area in the bottom half of the GUI, as illustrated in the figures on the following pages. This overview display provides controls and status information and gives direct access to the processing widgets.

The following processing areas are available from ‘overview’ displays in the GUI channel strip:

- Configuration
- Compressor
- Insert
- EQ

For details of how to navigate the GUI channel strips, see “Navigation via the GUI/Widgets” in Chapter 6.

## 30.4 Master Section on the Control Surface

The MASTER area allows control of the main Stereo and Mono Bus outputs.


1.**Mute** - Main stereo master bus mute.
2.**Mute** - Mono Bus mute.
3.**Meter** - Master Stereo Bus LED meter.
4.**Meter** - Mon Bus LED meter.
5.**Level control** - Continuously variable control of Stereo Master Bus.
6.**Level control** - Continuously variable control of Mono Bus.
7.**C/O** - The changeover button switches the Master Stereo Bus rotary to the surface fader directly below. This overrides any input or output active on the surface in that position.
8.**C/O** - The changeover button switches the Mono Bus rotary to the surface fader directly below. This overrides any input or output active on the surface in that position.

## 30.5 Output Metering

Signal level monitoring of the stereo outputs is available for the stereo masters on the surface and in the GUI. Metering for all other outputs is a GUI only feature. These meters can be found in various places like the channel overview area, workflows and widgets. These meters give an accurate display of the levels within the system.

**Tip:** The Meter Bridge in the Home workflow is a great place to see large meters of all channels or use the Console Overview workflow to see everything at once.

# 30.6 Talk & Bus Trim (GUI only)

There is a Talk switch in the Pre-amp section, each output widget (in all workflows) and can also be found in the assignable controls section above the surface faders if set-up.


If the Talk switch is active, the Talk button can be used to send the talkback mic to that channel for testing signal flow purposes.

1. Press the Talk or Shout button to select if the bus uses the Talk Bus or one of the 12 Shout Mix busses.
2. The tab expands to show all options and allows selection of 1 Shout Busses or the Talk bus.

The shout mixer is covered in detail in Chapter 13 Monitors and Shout Configuration.

## Bus Trim (GUI only)

The bus trim section in Configuration (all output channels) has a control for fine adjustment of the gain. Range is from -12 dB to +6 dB also with the option to change the phase.

# 30.7 Dynamics and EQ

The surface has a combined dynamics and EQ section that contains Dynamics and EQ On buttons, and a listen LED (Orange) that illuminates when listen is active in the output processing area to show when a channel has its dynamic side chain soloed.

In the GUI channel overview display the aux, matrix and master outputs each have a compressor section and an EQ section. Clicking within either of these sections will open their respective processing areas, which are described in the following subsections.

## Compressor (Dynamics)

For the outputs, the dynamic section only has a compressor in the output channel signal path. As the dynamics section on the surface is also used for the gate on the input channels, some controls may be redundant.

The output channel compressor has four styles: corrective, adaptive, creative and vintage. These are selectable via the Mode button. While the dynamic section is addressing the compressor, all of its controls are enabled except the hold control knob.

For details of the compressor graph, see “Compressor graph” in Chapter 29. The side chain is similar to the one used for the input channels, see “Side chain”.

## EQ (Equaliser)

For tonal control of the aux, matrix and master output signals, the output channel EQ has the option of a 4 band swept parametric EQ (PEQ).

The parametric EQ section of the channel allows tonal control of the input signal. The EQ contains all of the PEQ controls, along with a shelving mode selection button and another set of band selection buttons.

All of the outputs, except returns, have 4-band PEQs. Two of the 4 bands have three shelving modes each, while another has just one. Any combination of the six bands can be used to control the signal.

For more information, see “EQ” in Chapter 29.

## 30.8 Mute, Safes, Level and Solo

Each output (control surface and GUI) has controls for muting, soloing, and output signal level control. This is supported on the GUI in the appropriate widget area. In addition, the Configuration page in Automation workflow, Home workflow, FOH workflow and Channel View page have access to the channel safes when needed to stop certain parameters from being recalled on a scene change.


## 30.9 Output Channel Names

You can change the channel name in the GUI and surface LCD channel display. This can be done in the Channel View page or in any of the processing widgets by long pressing the channel name. Here you can change the background colour of the output channel name field and/or the channel name. You can also open the manchino page in the GUI menu or click the large channel icon in the side bar display to name channels.

Chapter 8 Basic Operations explains how to name all channels.

## 30.10 Output Channel Source/Destination

The channel's destination is shown in the configuration widget, patching page. If no destination has been selected, it will display no patch information. You can select the destination for this channel by clicking the patch image, which opens the patching workflow (see Chapter 7 Patching).

## 30.11 Stereo Linking

The Linking & Stereo section of the Configuration page in widget has a Mono, Link or Stereo button for linking the selected output channel to the adjacent (higher numbered) output channel. You can select the linked parameters and pan law settings.

On the surface in the CONFIG area you can link a selected mono channel by pressing the Link button. This links the selected mono channel to the next, if the next channel is also mono.

For more information, see Chapter 9 Stereo Linking.

# 30.12 Output Configuration Patching

The patching page contains controls for the direct Input, time compensation and output mute and level.


1. Aux Compressor Side Chain patch information.
2. Send and Return compensation on/off. In general, this will need to be on in order to preserve audio time alignment.
3. Aux Insert Send patch information.
4. Insert On/Off.
5. Insert Return A patch information.
6. Insert Return B patch information.
7. Bus Output mute.
8. Bus Output patch information.
9. Post-fader Bus Output level (fader level).

# 30.13 Direct Input

The direct input section provides an internal connection via the patching page to allow an external source to be patched into an output bus. There is a Level control (-¥ to +10 dB) plus a Mute button (mute duplicated on the surface).

The direct input is often used to bus another monitor desk's aux outputs into the HD96-24 system via the patching page.


1. Direct Input Mute – Mutes the Direct Input if required.
2. Direct Input Level Control – Adjusts the direct input level up to 10 dB.
3. Direct Input Patch information.
4. Direct Input Point – Choose where the signal is injected in the signal path, pre or post channel processing.
5. Metering for selected output.

## 30.14 Safes

Each output channel has six types of output channel safes that each protects a specific control/area from the automation system.

You can operate the safe switches via the GUI in the Output Configuration widget page, via the Manchino Page or via the configuration tab in Channel View, which also provide on/off status information. In the example below in Channel View, the Comp and Config are placed in recall safe.


1. Select Options – Opens the Safes selection tab.
2. ALL- This turns on recall safe for the whole channel.
3. Comp - The compressor settings are placed in recall safe.
4. EQ – All the EQ settings, Phase and Filters are put in recall safe.
5. Config – All config settings are placed in recall safe.
6. Mute – The channels mute is put in recall safe.
7. Main Bus – The channel's Main Bus settings (Pan, level, Send on and Mute to the Stereo Master and Mono Bus) are placed in recall safe.
8. Recall Safe indicators – This area show which sections of the channel are in recall safe.

## 30.15 Inserts

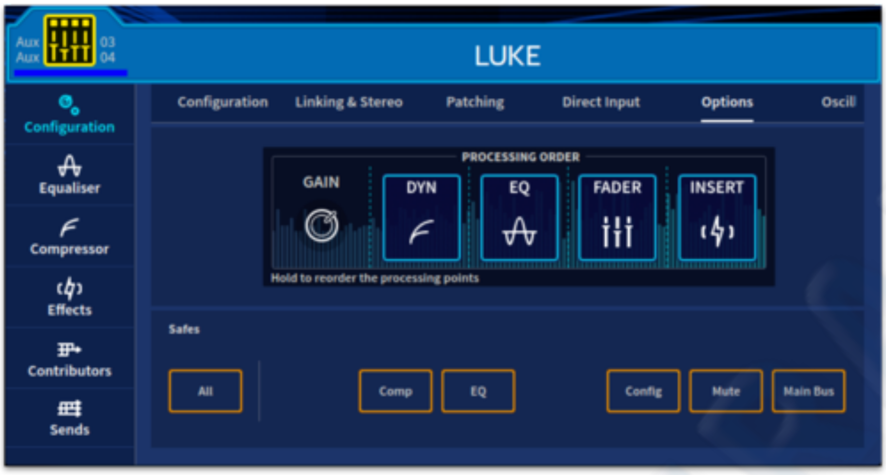
You can configure the send and return points of the aux, matrix and master outputs in the Insert section of the Configuration widget page or via the configuration tab in Channel View. A Return B path is also available which can be used to have a back-up external insert patched if needed. This can be found in the Effects tab.

## 30.16 Output Channel Delay

Similar to the input channels, all of the output channels have a delay that can be incorporated into the signal path. However, this can be a much larger delay, being in the range 0ms to 500ms (milliseconds).

# 30.17 Processing Order

Similarly, to the input channels, you can change the processing order on all of the output channels. Hold and drag to reorder the processing blocks. The Insert point can only be moved post fader as shown below.



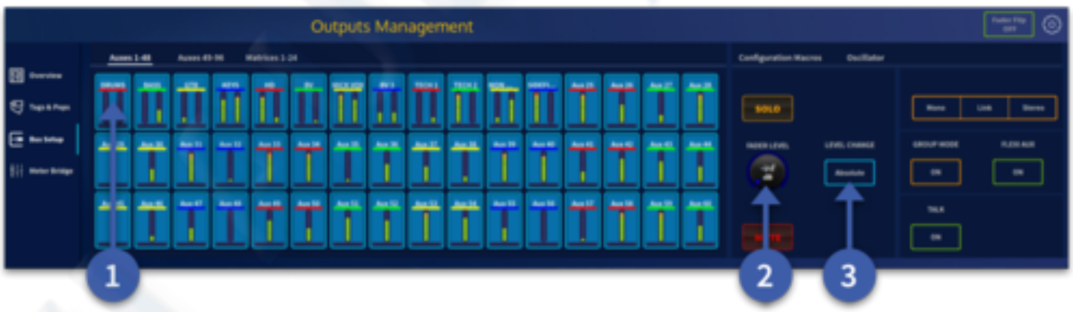

# 30.18 Outputs Macro Interrogation Control

The Output management page found in the Home workflow can be used in two ways:

## Path Select Control

Select the path you wish to change the select the parameter (Solo, Mute, Fader Level, Group mode and Talk). Fader level can be changed in 1 of 3 modes: Absolute, Relative or Pass-thru.

To select a path and change fader level:




1. Press the path or paths you wish to select or press and hold to select all paths on current page (as shown below). Pressing individual selected paths again will de-select them.
2. Adjust fader level.
3. Change mode if required (Absolute, Relative or Pass-thru).

**Tip:** This is a great place to quickly set all your aux paths to 0 dB, send pink noise to multiple paths or to mute all auxes/matrices quickly if needed.

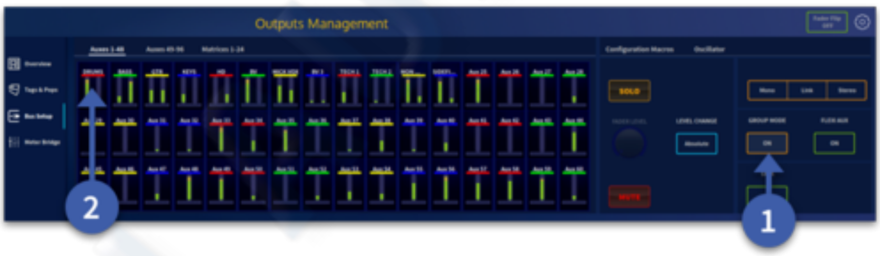
## Outputs Macro Interrogation Control

1. Press and hold the macro you wish to Interrogate (SOLO, MUTE, GROUP MODE or TALK ON). The path colours will change to show which macro is selected. SOLO = Yellow, MUTE = Red, GROUP MODE = Black, FLEXI AUX = Blue and TALK = Green.
2. Select the path or paths to instantly activate the selected macro function. Long holding on a selected path will select all paths on current page.

### Mute Macro


### Solo Macro


### Group Mode Macro

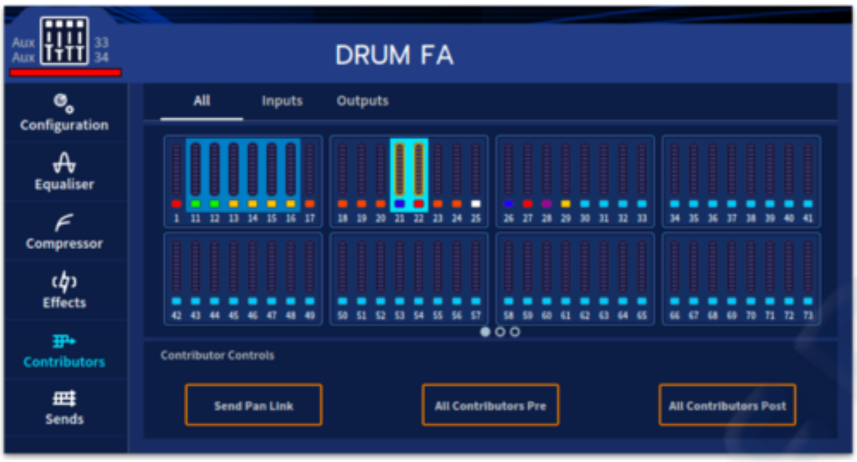



### Talk On Macro

# 30.19 Contributors

The Contributors page found in the Home and FOH workflows displays all channels currently sending or active in that output. This allows these channels to be called to the surface quickly in banks of 8. In the example below channels 1-40 have been turned on ready to send to the aux mix.

Contributor Controls allow changes to be made quickly to all contributions to the selected Aux or Matrix. The options are:




1. SEND PAN LINK – All Contributions to the selected Aux or Matrix follow the main channel pan.
2. ALL CONTRIBUTORS PRE – turn all current contributors to pre fade.
3. ALL CONTRIBUTORS POST – turn all current contributors to post fade.

# 30.20 Flexi-Aux

A Flexi-Aux allows you to, for example, group multiple keyboards into a stereo aux and process with EQ or Compression over the whole group. This Flexi-Aux can then be sent to any output, all perfectly in time. This means, for example if a band member requires all the keyboards levels up in their IEM mix, it can be achieved with one fader movement.

In the image below a compressed drum flexi aux is being sent to the first 4 in ear mixes.

**Note:** The green LED on an aux indicates the send is on. The orange LED indicates the Aux is pre fader.

# Chapter 31 :GUI Menu ( Side Menu)

## 31.1 GUI Side Menu

The GUI is a very powerful multi-functional tool that forms the core of the HD96-24 system. It gives you total control and monitoring of the operating environment, enhances control surface operation (you can operate the HD96-24 fully by GUI-only) and allows the use of internal and external devices. To facilitate this the GUI incorporates a simple-to-use GUI menu.

The GUI menu presents you with a list of options from which to choose, depending on your requirement. The following lists some of the functions that the GUI menu provides:


1. **Customise Toolbar** - The top menu bar can be re-ordered as required.
2. **User Profile** - Click on this icon to change or log out of the current user profile. This windows also displays how much memory has been used by the current user. You can also make changes to the mCloud settings here.
3. **Groups** - This page allows all POP, VCA, Mute and Talk Groups to be set up. (See Chapter 8 for POP & VCA and Chapter 17 for Talk and Mute groups.)
4. **Shout Configuration** - The shout mixer settings can be found in this page.
5. **Naming** - Name all channels. (See chapter 8)
6. **Patching** - This page takes you to the patching page where all internal and external connections can be made. (see chapter 7).
7. **Effects Rack** - Overview of current internal FX and DSP use. (See chapter 15)
8. **Expansion Cards** - This page shows the current settings of the CM1 card slots and Ultranet settings.
9. **Preferences** - User and operating preferences. Adjust GUI screen brightness, select delay compensation, select clocking preferences etc. (See chapter 26)
10. **Show Manager** - Show settings, Show editor and Playlist control.
11. **Navigation** - Select the layout of channels, busses and groups on the surface and how each workflow is set out.
12. **File Manager** - File management (internal, mCloud and external).
13. **Update Manager** - Update the console and connected I/O boxes.
14. **Service Page** - For Midas use only.
15. **Shutdown** - Press and hold to turn of the system safely. Please wait until you see the “Safe to turn off” message.

# Chapter 32: Configuring Virtual Soundcheck

## 32.1 Configuring a Virtual Soundcheck

Virtual sound check has become part of the modern day engineer's toolkit in order to test the sound in a room before the band arrive.

The HD96-24 virtual soundcheck system is designed to make this process quick and easy.

Virtual Sound Check sends audio from the patched I/O Port to the Input Channels in place of the live inputs, temporarily overriding the normal preamp input patch.

The large yellow Virtual Sound button in the top bar switches the system into virtual soundcheck mode, instantly changing all inputs to tape returns after a long press. Channels in Aux Return Mode are not affected by the Virtual Sound Check which allows effects to be used and tweaked as the they would be for the normal show. Aux Return mode can also be used to isolate a channel from virtual sound check and can be used, for example to let a band member play along to the multitrack recording.

To use virtual sound check, connect your computer or sound card to the CM1 slots via Madi, USB, Dante or AES50 depending on the card fitted in the CM1 slots. A KT9630 or KT9650 can be used to patch into the system via AES50. Madi and Dante allow 64 tracks of audio at 48 kHz per CM1 slot, two cards can be used for 128 channels of playback and record if required. USB allows 48 channels and 48 channels per AES50 port at 48 kHz. If 96 kHz sample rate is used the channel count is halved.

In this example 48 channels are patched from the USB card @ 48 kHz in CM1 slot 2 to tape returns 1-48.

**Tip:** 48 kHz is perfectly acceptable for virtual sound check purposes and allows greater channel counts. To record patch directly from the Pre-amp to the CM-1 outputs you wish to use. This keeps the direct outputs for other uses.

In the patching page select your input device in the left hand window and patch the inputs to the tape returns on the right side. This allows your normal channel inputs to stay patched to the I/O currently in use.

To engage virtual sound check press and hold the Virtual Sound button until it is solid yellow.

Make sure your DAW computer software is correctly patched and set-up will allow you to play back your recorded audio. If required, the Tape Gain can be adjusted in the Configuration Patching Widget of the Home workflow.

**Tip:** To stop channels switching to Virtual Sound Check tape returns turn Aux Return Mode on each channel you wish not to switch. (found in most workflows). For example, any effects returns or talk mics.

# Chapter 33:

# Manchino Multi Edit Page ( Advanced)

## 33.1 About the Manchino Page

The definition of Manchino (Manchester Cappuccino):

*1. Four shots of espresso with a dash of foam used to keep engineers focused in the creation of the Midas HD96-24 system.*
*2. The name of the multi edit page that revolutionises how engineers interact with a mixing console.*

The Manchino page is the heart of the HD96-24 system and can be used to do countless tasks and make complex changes possible but with only a few short touches.

The Manchino Page has 8 separate tabbed pages (Linecheck, Equaliser, Compressor, Gate, Patching, Naming, Sends and Safes/Scope) which control various functions throughout the console. Tabs for Inputs, Outputs and Matrices/Masters plus VCA and POP group navigation allow multiple channel selections at one.

In this chapter each page will be described in depth. All pages use the same channel selection system for navigation which is described in the first Linecheck page. There are some differences between Input, Auxes and Matrices & Masters controls such as Auxes, Matrices & Masters don't have HPF & LPF or a gate. Many of the functions, such as 48v or EQ On have a press and hold second function which lets you see which paths are currently in the active or On state. Long press functions are listed below:

| LINECHECK | EQUALISER | COMPRESSOR | GATE | SAFES |
|---|---|---|---|---|
| 48v | EQ ON | COMP ON | GATE ON | ALL |
| Ø | | FILTER ON | FILTER ON | GATE |
| HP ON | | | | COMP |
| LP ON | | | | EQUALISER |
| STEREO | | | | FILTERS |
| | | | | CONFIG |
| | | | | MUTE |
| | | | | MAIN BUS |


1. Long press a function button to see the status on the paths to the left side.
2. In this example functions highlighted in red are showing that 48v is on.

Manchino operates in two ways:


A SOLO channel inspector type mode where a single path’s controls can be looked and changed in one place whist being heard in the solo bus.

Multiple Select mode allows various groups of channels to be selected and controlled in several different ways.

# 33.2 Linecheck Page

1. Three tabs to select Inputs, Auxes or Matrices & Masters. Only one page can have selections at any one time. For example, you can't edit Inputs and Auxes at the same time.
2. Paths can be selected individually or selecting all by pressing and holding. After selecting all, channels can be pressed again to deselect or press and hold to deselect all.
3. Deselect All clears all current path selections.
4. Solo mode allows only one channel to be selected and edited at once. Solo also sends the selected channel to the solo bus for audition.
5. Multiple Select allows any combination of channels to be selected within the current page.
6. Path selection by VCA. For example, Select the guitar VCA and all the guitar channels will be selected.
7. Path selection by POP group. For example, Select the drums POP group and all drums channels will be selected.
8. Select which page of the Manchino you wish to use from Linecheck, Equaliser, Compressor, Gate, Patching, Naming and Sends.
9. Turn various safes on for the currently selected channels.
10. Display of the currently selected channels in banks of 8.
11. Arrows to scroll through the various banks of channels if more than 8 channels are selected.
12. Gain level for the selected channels.
13. 48v button for the selected channels. Solid red if all channels have 48v applied. Translucent red if more than one channel is selected with different 48v states.
14. Trim control for the selected channels
15. Ø Polarity control, Solid green if all channels have Ø applied. Translucent green if more than one channel is selected with different Ø states.

16. High Pass Frequency (HP) for the selected channels. 10 Hz to 10 kHz.
17. HP ON. Solid green if all channels have HP on. Translucent green if more than one channel is selected with different HP states.
18. Low Pass Frequency (LP) for the selected channels. 20 kHz to 40 Hz.
19. LP ON. Solid green if all channels have LP on. Translucent green if more than one channel is selected with different LP states.
20. Pan control changes the pan position in the stereo bus for all selected paths.
21. Stereo button indicates solid orange if all selected channels are assigned to the stereo bus. Translucent orange if more than one channel is selected with different stereo send states.
22. Fader level adjusts the position of the fader to the main stereo bus.
23. Unity sets selected channels to 0 dB.

**Tip:** A great place to set all faders to 0dB ready for monitor mixing or turn all your High Pass Filters on.

**Note:** Press and hold on a VCA or POP to take you to the set-up and naming if required.

# 33.3 Equaliser

## Equaliser Solo View

In the Equaliser page, If one path is selected the EQ for that path is displayed and can be edited in a similar way to the main Equaliser found in the various workflows and widgets.

**Note:** With no channels selected press and hold EQ ON to see which channels the EQ is turned on. No other buttons in this page display overview information.


1. Select a single path to edit.
2. In single channel edit the Equaliser window resembles the normal Equaliser widget found in other workflows with bands selectable by the tabs across the top of the EQ. Individual bands can be edited to right hand side with the vertical assignable controls.
3. Gain control for the selected band.
4. EQ on/off control.
5. Frequency control for the selected band.
6. High Pass Filter (HP) select.
7. Band width control.
8. Bass band select.
9. Lo Mid band select.
10. Mid Hi band select.
11. Treble band select.
12. Low Pass Filter (LP) select.
13. Eq shape control for Bass (Bell, Deep, Classic and Warm) and treble bands (Bell, Bright, Classic and Soft). All EQ shapes are described in the Input EQ section.

## Equaliser Multiple Select View

If two or more paths are selected the Equaliser Page allows multiple paths to be edited as one. For example, select a monitor speaker aux POP group and take a little 1 kHz out of every mix.


1. Select 2 or more paths to edit at once.
2. Each path's EQ will be displayed in the window. Only 16 paths will be displayed if more than 16 paths are selected. The other paths are still selected even though not in view and changes will be made to all selected paths.
3. Select the band you wish to change.
4. Absolute changes all values to the same value as set by the control.
5. Relative keeps any differences between levels intact. For example, add +3 dB of gain to the bass band will add +3 dB to any cut or boost on the selected paths. If a kick drum was at +2 dB in the bass band it would become +5dB, if a snare had a cut in the bass band at -2 dB it would become +1 dB.
6. Pass thru allows EQ changes to be made once the control "Passes through" the current set level, then follows the control as it changes.

The vertical assignable controls are available to the right of the screen. They are controlled in the same manner as Solo View.

# 33.4 Compressor Page

### Compressor Solo View

In the Compressor page, If one path is selected the compressor for that path is displayed and can be edited in a similar way to the main Compressor found in the various workflows and widgets with some parameters controlled by the vertical assignable controls.

1. Compression graph (can be used to adjust threshold by swiping up and down).
2. Sidechain frequency display (swipe left to right to change frequency).
3. Input level meters.
4. Gain reduction meters.
5. Compression knee type selection (Soft, Medium or Hard).
6. Compression mode (Corrective, Adaptive, Creative or Vintage modes).
7. Threshold (-50 dB to +25 dB). Adjust the point at which compression occurs.
8. Compression On button.
9. Ratio control (1:0 to 25:1).
10. Attack control (200 us to 100 ms).
11. Release control (50 ms to 3 s).
12. Make-up gain (0 dB to 24 dB).
13. Sidechain source selection
14. Sidechain listen, sends the signal from the channel’s sidechain to the solo bus for audition.
15. Sidechain width control (0.1 Oct, 0.3 Oct, 1 Oct or 2 Oct).
16. Sidechain frequency control (50 Hz to 15 kHz).
17. Sidechain On button.

## Compressor Multiple Select View

If two or more paths are selected the Compressor Page it allows multiple paths to be edited as one. For example, select the playback system POP group and change the compression mode to all of the members of the group all at once. This allows you to hear the differences between the various modes in one swift action across all tracks.

**Tip:** If the attack portion of the drum kit sound is being lost it is possible to slow the attack and/or adjust the threshold of all input channel compressors in order to let more transients through as a whole.

1. Individual compression graphs for all selected channels including gain reduction meters.
2. Switch between Compression and Sidechain view.
3. Select Knee type for all selected channels (Soft, Medium or Hard).
4. Pick compression mode (Corrective, Adaptive, Creative or Vintage modes).
5. Adjustments to level can be changed by an Absolute, Relative or Pass-thru value as described earlier in this user guide.

# 33.5 Gate Page

In the Gate page, If one path is selected the gate for that path is displayed and can be edited in a similar way to the main gate found in the various workflows and widgets with some parameters controlled by the vertical assignable controls.


1. Gate graph (can be used to adjust threshold by swiping up and down).
2. Sidechain frequency display (swipe left to right to change frequency).
3. Input level meters.
4. Gain attenuation meters.
5. Gate type selection (Gate or Ducker).
6. Sidechain source selection
7. Sidechain listen sends the signal from the channel's sidechain to the solo bus for audition.
8. Sidechain width control (0.1 Oct, 0.3 Oct, 1 Oct or 2 Oct).
9. Sidechain frequency control (50 Hz to 15 kHz).
10. Threshold (-50 dB to +25 dB). Adjust the point at which compression occurs.
11. Gate On button.
12. Ratio control (-100.0 to -0:0).
13. Attack control (20 us to 20 ms).
14. Release control (2 ms to 2 s).
15. Hold (0.002 s to 2 s).
16. Sidechain filter On button.

**Gate Multiple Select View**

If two or more path's are selected in the Gate Page it allows multiple gates to be edited as one. For example, select the drum POP group and turn off all the gates at once for mic tapping/line checking, once drum channels are tested, turn all the gates on again.


1. Individual gate graphs for all selected channels including gain attenuation meters.
2. Switch between Gate and Sidechain view.
3. Adjustments to level can be changed by an Absolute, Relative or Pass-thru value as described earlier in this user guide.

## 33.6 Naming Page

The Manchino page can be used to name all paths within the system. A number of clever naming functions speed up show creation.

1. Select a single path to name individually or select multiple paths to use the Auto-Number and Auto Append L/R functions.
2. Text enter window.
3. QWERTY Keyboard.
4. Auto-Number allows multiple channels to be named with an ascending numerical value. For example, select 8 input channels that will be used for a hard disk playback system. Name the channel HD and input 1 in the Order From box. After confirming the action, the channels will be named HD1, HD2, HD3 etc. If you input a space after typing HD there will be a space after the number (HD 1, HD 2 etc).
5. Order From window.
6. Select the colour for the selected paths.
7. Auto-Append L/R allows 2 channels to be automatically named with an L and R. For example, select 2 channels, name them Piano, then select the Auto-Append L/R button and enter. The channels will now be named Piano L and Piano R.

# 33.7 Manchino Sends Page

The sends page revolutionises how engineers interact with a console. Multiple send levels can be changed to many destinations, all relative to the original values. For example, when 3 members of the band all ask the monitor engineer for “more drums” in their in-ear monitors (IEMs) at once simply select the Drum POP group, select their IEM Auxes, then add level, which relatively adds the same amount of level to each drum, to each person in one smooth action.


1. Select either Multiple Select or Solo mode for Aux and Matrix paths. Solo allows you to listen to the selected output path in the solo bus. Multiple Select allows a number of output paths to be selected at once (press and hold to select all in a similar way to selecting all inputs).
2. Select via the tabs Aux 1-64, Aux 65-96 or Matrices 1-24. Please note that the press and hold select all function only selects the auxes viewable in that page. For example, if you select all Auxes 1-64, Auxes 65-96 in the second tab won’t be selected.
3. Indicates the selected input channels send to Aux is turned on or active.
4. Indicates, when illuminated orange that the selected channels are sending Pre Fader.
5. Aux overview and selection area. Press and hold to select all current viewable Auxes or Matrices.
6. Adjustments to level can be changed by an Absolute, Relative or Pass-thru value as described earlier in this user guide.
7. Unity sets level to 0 dB for swift set up operation.
8. If multiple channels are selected and 1 aux or matrix, selected channels can be unfolded to the vertical assignable controls to the left of the GUI for individual channel level and pan changes. See the next section for more details on unfolding.
9. Send level control, function changes according to the settings.
10. C/O changes the send level control to the send pan control.
11. Turns the channels send on to the selected Auxes or Matrices.
12. Changes selected channels send between PRE or Post Fader.
13. Sets the channel to the default pick off point.

14. Sets the pick off point to Post Filter.
15. Sets the pick off point to Post Equaliser.
16. Sets the pick off point to Post Insert.
17. Sets the pick off point to Post Dynamics.
18. Turns an Aux into a Flexi Aux (Allowing stem style Aux to Aux mixing).
19. Group mode turns an Aux into a fixed send group (think old subgroups on an analogue mixing desk).

## Sends Unfold Channels

When you unfold an Aux, it means you can see the separate input channels sending to that Aux in banks of 8 on the vertical assignable controls. For example, below the drum channels are unfolded from Aux 1&2.

1. Press to unfold select channels from the selected Aux.
2. Channels active in the chosen Aux will be displayed here in banks of 8.
3. Use the arrows to navigate through the banks of channels.
4. Each of the 8 displayed unfolded channels can have their send level to the selected Aux changed.
5. Pressing the C/O button turns the controls from send level to pan control within the chosen Aux.

# 33.8 Safes

In this page it is possible to globally apply Safes to the console. Safes are set relative to the type of channel and stop scene information from being recalled.


1. All – Safe from automation all channel parameters.
2. Gate – Safe from automation for input channels gate settings.
3. Comp – Safe from automation for compressor settings.
4. Filters – Safe from automation for Filter settings.
5. Equaliser – Safe from automation for EQ settings.
6. Config – Safe from automation config settings.
7. Mute – Safe from automation mute settings.
8. Main Bus – Safe from automation of pan and fader level to the Masters.

# Chapter 34: Update Manager

## 34.1 Console Updater

The HD96-24 is at the start of its journey and will grow and develop over many years. Essential updates will bring new features, increase productivity and workflows.

There are three ways to update the console, either using the internet via the built in Wi-Fi (Remote), via mCloud profile or with a USB file download from the Midas website. You will also find the release notes and full update procedure documentation.

To update via mCloud (an active internet connection is required):

1. In the GUI, navigate to Update Manager found in the Left hand side menu.
2. Select the Console Updater from the options available.
3. Select the Remote, then press the check for updates button (Any new updates will be displayed).
4. Select the update from the list and press the Install button.
5. Press confirm to execute the update or cancel if you do not wish to proceed.
6. Once executed the warning 'Installing Release' will appear and the progress bar will indicate progress of the update.
7. When the updater is finished a warning will appear which says, 'Restart is required to complete the update'. Press Restart to finalise the update process.

To update via a USB stick:

1. Download the software from the Midas website and make sure the update file is in the top layer. The folder must be called: midas (all lower case).
2. Turn on the HD96-24 system and wait until the system has finished loading.
3. Insert USB stick into the front USB port on the surface.
4. In the GUI, navigate to File Manager found in the Left hand side menu.
5. Select the Console Updater from the options available.
6. Select Local then USB from the options and find the update file in the Midas folder.
7. Select the update and press Add, this moves the update file to the console.
8. Select the update from the list and press the Install button.
9. Press confirm to execute the update or cancel if you do not wish to proceed.
10. Once executed the warning 'Installing Release' will appear and the progress bar will indicate progress of the update.
11. When the updater is finished a warning will appear which says, 'Restart is required to complete the update'. Press Restart to finalise the update process.


1. Local updates from a USB stick.
2. Remote updates from the Midas mCloud.
3. Add updates from a USB stick to the console.
4. Current available updates are listed here
5. Release notes for the currently selected update are shown here.
6. Install the selected update with a long button press.
7. Current software version.

## 34.2 Surface Updater

1. Reboot surface- In the unlikely event of the Dream flow GUI freezing it is possible to restart the system. Audio will continue to pass during this process. A long press of the Reboot Surface button is required to restart the surface.
2. Calibrate Faders - After a long button press the surface will enter fader calibration mode. The faders will then self-calibrate. This process takes a few minutes. The calibration is finished once all faders have stopped moving.


3. Update All - This function updates all the surface boards if required. This update is only required if a board has been replaced within the console.

## 34.3 I/O Box & Device Updater

Note: I/O boxes will need to be updated in order to work with the HD96-24 system. The I/O Box and Device updater page can be found in the Update Manager tab (side bar menu). With all your I/O connected with the console as master clock, press Sync I/O and follow the instructions. Updated I/O boxes will still work and are fully compatible with Pro Series consoles.


1. Press Sync I/O. Any I/O racks will appear after the I/O scan.
2. FOH I/O connections will be displayed in this section.
3. HMAC I/O connections will be displayed in this section.
4. Each device will display its family type (DL15x, DL231), its current software version and the ID number for the I/O device. In this example the I/O use Pro Series software and would require updating.
5. Select all currently Sync'd I/O devices ready for update. If you prefer individual racks can be updated.
6. Update Selected updates all selected I/O to the latest version of HD software.

# Chapter 35: Channel AI

## 35.1 What is Channel AI?

The thought of Artificial Intelligence (AI) in a mixing console could be seen as thought provoking, futuristic or even scary. Channel AI is designed to help not hinder, to suggest, not take over and to give a guide to help improve your sound and workflow. The innovative ability to analyse a channel and let the system tell you what the instrument type may be has many applications. This can save vital set-up time.

### What is Channel AI?

Channel AI allows an input channel to be analysed or profiled as we call it. This is achieved by capturing a short excerpt from the incoming audio signal, i.e. a 4 second long sample. Onset segmentation is performed on the captured audio. Instrument classification is performed on onset per onset basis. The instrument classifier returns three outputs: channel name, spill detection and confidence (quality of the captured audio sample). Currently the instrument classifier can detect the following instruments: Kick; Snare; Hat/Cymbals; Toms; Bass; Guitar; Piano; Vocals. This database will increase over time to include more instruments and their many different variations. Suggestions of the instrument type, spill and confidence are based on the Midas database of profiled instrumentsand on models of sound perception that our research team developed.

Based on further analysis of the captured audio, settings are generated for the following signal processing units: Trim, Equaliser, Gate and Compressor. These presets are adaptive rather than and are likely to vary depending on the characteristics of the captured audio. For instance, if you profile a channel during a quiet passage of a song, the trim parameter will be set to higher value than if you profile a loud passage of the same song.

## 35.2 How to Turn Channel AI On

Press the Channel AIicon to active the function. Once on, channel AI can be used on any input channel. Press the icon again to turn Channel AI off.

# 35.3 Profile a Channel


When profiling a channel, the signal is captured post preamp and is not affected by any channel processing performed on that channel. For the best results the input level of the channel to be profile should be between -30 dB and 20 dB making sure to avoid clipping. It is important to note that AI should be used with the default processing order **DYN > INSERT> EQ** in order to achieve the best results.

1. **Profile**

Press the Profile button when audio is playing. Each time a channel is profiled a new profile is created. Each profile and its associated presets are determined by the audio playing at the moment when the channel is being profiled. This means different presets can be created for the same channel by profiling the channel multiple times. When selecting a profile, details associated to the chosen profile are displayed in the respective fields.

2. **Profile selection**

You can create up to 4 profiles for each channel by pressing the profile button. Profiles are numbered in chronological order from 1 (oldest) to 4 (latest). Upon channel selection, the latest profile created is automatically selected.

By selecting a profile cell, it is possible to access all the settings related to it, i.e.:

- Name of the detected instrument
- Spill indicator
- Confidence indicator
- Adaptive suggestions for the selected channel

3. **Delete**

Each profile can be deleted by pressing the delete button located beside the profile button. When a profile is deleted, all the other profiles get numbered according to the same criteria used when they were generated.

4. **Auto Name**

After a channel has been profiled the system will tell you what it believes the instrument or vocal is. This is achieved by in depth analysis of the audio signal and based on the Midas database. If the system doesn't recognise the instrument it will be listed as unknown.

## 5. **Auto Set-up**

The Auto Set-up function aims to speed up your workflow. It allows with a single touch of a button to name the channel and apply all default preset settings to the channel. This includes setting channel name, trim value, Filters, EQ, Compression and Gate settings. Triggering Auto-Setup does not result in any audible changes since you will need to activate the respective DSP units (i.e. EQ, Gate and Compressor) which are turned off by default. This allows you to decide if the suggested EQ, Gate and Compression settings work in context before applying them to the channel. All this means, is that you are always fully in control of the system.

## 35.4 Adaptive Configuration Suggestions

When selecting a profile from the list, the presets associated to this profile will be displayed in the Configuration Suggestions field. However, this will not result in any changes to your settings, unless you decide to select a preset from the Configuration Suggestions list and explicitly apply to the channel by touching on the Apply button. The Configuration Suggestions field is context dependent. In order to display the presets associated to a particular digital signal processing unit (e.g. EQ, Compressor, Gate) you will need to select the respective DSP unit for that channel. For instance, if you select the Equaliser all EQ presets for the selected channel and profile will displayed.

## 35.5 Individual Component Processing

Individual components can be applied by navigating to that section and applying the presets in the list.

### Trim suggestions.


After profiling the channel navigate to the config widget then:

1. Select the Trim type required from the configuration suggestions' list.
2. Press apply to set the trim level.

All instruments will have the following suggestions:

- Trim (0dB): should statistically normalize the input to target 0 dBu
- Trim (10dB): should statistically normalize the input to target 10 dBu

**Note:** If you change the preamp gain of an input you will also need to re-profile the channel in order to get the maximum benefits from the system.

## EQ Suggestions


1. Select the Profile you wish to use.
2. If the instrument has been detected correctly, all the suggestions names and settings with gains and BW for the various bands will appear in the list depending on instrument detected.
3. Press Apply to recall the selected preset.

*Presets currently include:*

- **Kick:** Flat, Sub, Deep, Tight, Rock, Click, Corrective.
- **Snare:** Deep, Tight, Presence, Snap, Corrective.
- **Tom:** Deep, Tight, Snap, Corrective.
- **Hi Hat/Cymbals:** Snap, Thin, Bright, Muffled, Corrective.
- **Overheads:** Thin, Warm, Bright, Corrective.
- **Guitar:** Warm, Smooth, Tight, Aggressive, Bright, Corrective.
- **Bass:** Deep, Tight, Smooth, Bright, Corrective.
- **Piano/Keyboard:** Deep, Warmth, Bright, Corrective.
- **Vocal:** Warm, Presence, Bright, Male, Female, Corrective.

The Flat preset moves the 4 x EQ frequency bands to the suggested settings without boosting or cutting. The gain is set to zero and the width is set to 0.7 for all bands. This makes a good starting point to EQ a signal. The other presets may also add gain and width changes to the EQ bands it suggests. The corrective presets only attenuates problematic frequencies but it doesn't boost any frequencies.

## Compressor Suggestions


1. Select the Profile you wish to use.
2. If the instrument has been detected correctly, the suggested name and presets will appear in the list. This is dependant on instrument detected.
3. Press Apply to recall the selected preset.

Hat/cymbals, tom, bass, guitar, piano, vocals and drum mic will have the following suggestions:

- Aggressive (average gain reduction between 5 and 9 dB)
- Corrective (average gain reduction between 3 and 5 dB)
- Limit (average gain reduction between 0 and 1 dB, no make-up gain)

- Timings (attack and release only, no changes to threshold, ratio and makeup)

Kick, snare, and tom will have the following suggestions:

- Aggressive (average gain reduction around 8 dB) Hard Knee
- Corrective (average gain reduction around 6 dB) Med Knee
- Gentle (average gain reduction around 4 dB) Soft Knee
- Limit (average gain reduction between 0 and 1 dB, no make-up gain) Hard Knee
- Timings (attack and release only, no changes to threshold, ratio and makeup)

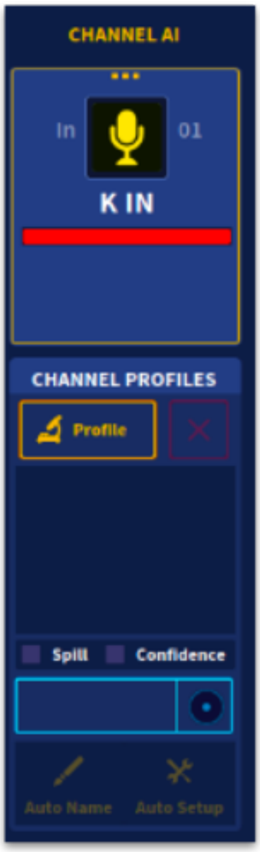
All the settings are fully adaptive.

## Gate Suggestions


1. Select the Profile you wish to use.
2. If the instrument has been detected correctly, all the suggestions names and settings will appear in the list depending on instrument detected.
3. Press Apply to recall the selected preset.

Kick, Snare and Tom drum mics will have a suggestion called "Natural”; whose settings are fully adaptive.

# 35.6 Spill




Spill (also known as bleed and leakage) is the occurrence in live sound mixing whereby sound is picked up by a microphone from a source other than that which is intended. Spill is usually seen as a problem, and various steps are taken to avoid it or reduce it. In the HD96-24 system the spill LED indicates the quality of the captured profile, i.e. indicates the amount of spill from other instruments.

Spill indicator (off = no spill, green = normal spill, yellow = medium spill, red = heavy spill)

# 35.6 Confidence

Confidence is a preference for what is known as the indicator of how sure the system is about the predictions made. A low confidence profile might, therefore, predict the wrong instruments and hence the associated suggestions will be wrong.

Confidence indicator (green = high confidence, yellow = medium confidence, red = low confidence)