PRO1 Live Audio System

Owner's Manual

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PRO1 Live Audio System — Owner's Manual
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EN Important safety instructions

CAUTION
RISK OF ELECTRIC SHOCK!
DO NOT OPEN!
ATTENTION
RISQUE DE CHOC ELECTRIQUE!
NE PAS OUVRIR!

Terminals marked with this symbol carry electrical current of sufficient magnitude to constitute risk of electric shock. Use only high-quality commercially-available speaker cables with ¼" TS plugs pre-installed. All other installation or modification should be performed only by qualified personnel.

This symbol, wherever it appears, alerts you to the presence of uninsulated dangerous voltage inside the enclosure - voltage that may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions in the accompanying literature. Please read the manual.

Caution
To reduce the risk of electric shock, do not remove the top cover (or the rear section). No user serviceable parts inside. Refer servicing to qualified personnel.

Caution
To reduce the risk of fire or electric shock, do not expose this appliance to rain and moisture. The apparatus shall not be exposed to dripping or splashing liquids and no objects filled with liquids, such as vases, shall be placed on the apparatus.

Caution
These service instructions are for use by qualified service personnel only. To reduce the risk of electric shock do not perform any servicing other than that contained in the operation instructions. Repairs have to be performed by qualified service personnel.

1 Read these instructions.

2 Keep these instructions.

3 Heed all warnings.

4 Follow all instructions.

5 Do not use this apparatus near water.

6 Clean only with dry cloth.

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

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

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

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

11 Use only attachments/accessories specified by the manufacturer.

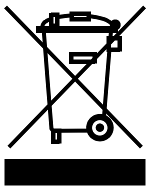
12 Use only with the cart, stand, tripod, bracket, or table specified by the manufacturer, or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.

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

14 Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.

15 The apparatus shall be connected to a MAINS socket outlet with a protective earthing connection.

16 Where the MAINS plug or an appliance coupler is used as the disconnect device, the disconnect device shall remain readily operable.



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4. Warranty claims other than those indicated above are expressly excluded.

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2. Subsequently, the product must be returned in its original shipping carton, together with the return authorization number to the address indicated by MUSIC Group.

3. Shipments without freight prepaid will not be accepted.

§ 4 Warranty Exclusions

1. This limited warranty does not cover consumable parts including, but not limited to, fuses and batteries. Where applicable, MUSIC Group warrants the valves or meters contained in the product to be free from defects in material and workmanship for a period of ninety (90) days from date of purchase.

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4. This limited warranty is invalid if the factory-applied serial number has been altered or removed from the product.

5. Free inspections and maintenance/repair work are expressly excluded from this limited warranty, in particular, if caused by improper handling of the product by the user. This also applies to defects caused by normal wear and tear, in particular, of faders, crossfaders, potentiometers, keys/buttons, guitar strings, illuminants and similar parts.

6. Damage/defects caused by the following conditions are not covered by this limited warranty:

- improper handling, neglect or failure to operate the unit in compliance with the instructions given in Midas user or service manuals;
- connection or operation of the unit in any way that does not comply with the technical or safety regulations applicable in the country where the product is used;
- damage/defects caused by acts of God/ Nature (accident, fire, flood, etc) or any other condition that is beyond the control of MUSIC Group.

7. Any repair or opening of the unit carried out by unauthorised personnel (user included) will void the limited warranty.

8. If an inspection of the product by MUSIC Group shows that the defect in question is not covered by the limited warranty, the inspection costs are payable by the customer.

9. Products which do not meet the terms of this limited warranty will be repaired exclusively at the buyer's expense. MUSIC Group or its authorized service centre will inform the buyer of any such circumstance. If the buyer fails to submit a written repair order within 6 weeks after notification, MUSIC Group will return the unit C.O.D. with a separate invoice for freight and packing. Such costs will also be invoiced separately when the buyer has sent in a written repair order.

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Subject only to the operation of mandatory applicable local laws, MUSIC Group shall have no liability to the buyer under this warranty for any consequential or indirect loss or damage of any kind. In no event shall the liability of MUSIC Group under this limited warranty exceed the invoiced value of the product.

§ 7 Limitation of liability

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§ 8 Other warranty rights and national law

1. This limited warranty does not exclude or limit the buyer's statutory rights as a consumer in any way.

2. The limited warranty regulations mentioned herein are applicable unless they constitute an infringement of applicable mandatory local laws.

3. This warranty does not detract from the seller's obligations in regard to any lack of conformity of the product and any hidden defect.

§ 9 Amendment

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Overview

Chapter 1: Introduction

Welcome to the PRO1 Live Audio System. The PRO1 Live Audio System 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.

The PRO1 Control Centre, which forms an integral part of the PRO1 Live Audio System, was conceived by Midas to offer audio professionals high-performance audio equipment, designed to provide no-compromise sonic quality with a feature set that offers all essential facilities and functions. It represents the very best of British design and engineering combined with contemporary, efficient manufacturing methods, and will give you many years of reliable service.

So, to obtain the best results with a minimum of effort, please read this Owner's Manual and, finally, enjoy your Midas PRO1 Live Audio System!

About this manual

This is the Owner's Manual for the PRO1 Live Audio System. Its purpose is to familiarise the user with the system and show how to install, configure and operate the PRO1 Control Centre.

Note: *The content of this Owner's Manual does not supersede any information for any other equipment supplied with the PRO1 Live Audio System.*

Structure

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

- **Overview:** This gives an overview of the PRO1 Live Audio System and PRO1 Control Centre and contains information about this manual.
- **Getting Started:** This shows you how to set up and power up a PRO1 Live Audio System.
- **Basic Operation Of The PRO1 Control Centre:** This shows you how to use the controls of the PRO1 Control Centre, how to navigate the control surface and GUI, how to route (patch) the channels and buses, and how to carry out basic operations in order to get some audio out of it.
- **Advanced Operations And Features:** This describes the advanced features of the PRO1 and gives detailed operating instructions.
- **Description:** This gives a detailed description of the PRO1 Control Centre hardware and the controls and their functions on both the control surface and GUI. It provides useful reference material.
- **Appendices:** This provides additional reference material and technical information on the PRO1, such as application notes, signal path diagrams, technical specifications, service information, etc.

Conventions

The following lists some of the main conventions used in this manual.

- Hand symbols, such as, (for pushbutton, trackball, glide pad etc.) and (for control knob), are used to show the operation of the physical controls on the control surface. GUI operation is indicated by a pointer, which represents a 'click' or 'drag' operation.
- The graphics shown right are used to differentiate between diagrams of the control surface (left) and GUI (right).

- Unless otherwise stated, illumination of a control (pushbutton, switch, control knob etc.) on the control surface/GUI of the PRO1 indicates an "on", "active" or "enabled" state. Conversely, an extinguished condition indicates the control is "off", "inactive" or "disabled".
- 'Selecting' an option on the GUI enables that option, as indicated by a symbol, such as a tick ✓ or cross X. An empty option is unselected (disabled).
- The following types of pushbutton are used on the control surface:
  - "switch" - a latching pushbutton, that is, one that changes its on/off status.
  - "button" - a non-latching pushbutton.
- Generally, control names are the same whether they are on the control surface or the GUI. However, in cases where they differ, both names will be given, separated by a forward slash "/". The control name shown on the GUI will always be last and enclosed in square brackets "[]".
- Hints and tips, which convey useful information to the user, appear where you see the drawing pin graphic (shown right).

Terminology

To clarify the understanding and use of the digital console and to discriminate between its analogue equivalent, the terminology has been chosen very carefully (see "Glossary" on page 553).

GUI diagrams

This manual contains numerous diagrams that represent the GUI screen displays. Due to the many permutations of control settings, operating status, channel configurations, etc., it is inevitable that these diagrams will look slightly different to those on your control centre.

Anti-aliasing

To make the GUI of the PRO1 as crisp, eye-catching and intelligible as possible it incorporates an anti-aliasing algorithm to ensure the utmost smoothness of straight lines and curves. Unfortunately, the process of reproducing GUI displays for this manual has resulted in an inevitable loss of quality, which in some cases has led to a certain amount of pixelation. Therefore the quality of the actual GUI display on the PRO1 is not truly reflected here.

PRO1 host software

This manual is for an PRO1 Control Centre running host software version 2.00 and later.

To keep the PRO1 Control Centre up to date so that it gives you optimum performance, we recommend that you check the Midas website regularly to see if there are any available host software updates.

Service and support

The PRO1 is a very hi-tech piece of equipment. We provide superb levels of support and service to give users confidence in Midas digital products.

Chapter 2: PRO1 Live Audio System

The PRO1 is a mid-size mixing console designed to work for long periods, not just indoors but under harsh sunlight or near freezing conditions. Built around a lightweight all-aluminium frame, the PRO1 is a standalone console that is easy to configure and operate.

The console forms the core of the PRO1 Live Audio System, whose network carries both proprietary control data and open architecture AES50 digital audio, and uses readily available standard cabling and connectors. The PRO1 uses the reliable Linux operating system.

Additional AES50-compatible sound processing products can be connected to the PRO1 to form extended distribution and mixing console "system".

Features

Please remember, the PRO1 is not just a console, it's a LIVE PERFORMANCE SYSTEM!

- 100 inputs x 102 outputs (maximum capacity) point-to-point routing anywhere within the network
- 24 mic/line inputs with Midas mic preamps
- 48 simultaneous input processing channels
- 24 analogue outputs (including two stereo local monitor outputs)
- Monitor mixing is simple — master, matrix and aux buses can be routed directly from input channels with independent level control, giving 24 monitor mix buses
- Traditional FOH subgroup mixing is simple — any/all aux buses can operate as post-channel fader and pan (aux gain fixed at unity)
- Aux inputs have two modes of operation: effects return and input channel
- Three AES3 outputs
- Two AES3 inputs
- Integral AES50 ports on the Control Centre for I/O expansion and inter-console connectivity
- 27 sample-synchronous, phase-coherent mix buses
- Up to 12 multi-channel FX engines
- Up to 28 Klark Teknik DN370 31-band Graphic EQs
- Surround panning including 5.1, Quad and LCRS
- Full-colour 15" daylight-viewable display screen with DVI out
- Eight VCA (Variable Control Association) groups
- Six POPulation groups
- Automation providing up to 1,000 scenes with snapshot save/recall capability and global edit, presets and show file archiving
- Comprehensive, easy-to-use routing via a GUI Patching screen
- 96kHz 40-bit floating-point processing throughout

- Removable power supply
- Three year factory warranty

Additional I/O Box Options

- DL251: 48 inputs/16 outputs in a fixed configuration I/O
- DL252: 16 inputs/48 outputs in a fixed configuration I/O
- DL351: up to 64 inputs/64 outputs in a configurable I/O
- DL431: 24 inputs in a five way split in a fixed configuration I/O
- DL451: up to 24 inputs/24 outputs in a configurable I/O

Accessories

- Klark Teknik DN9331 Rapide Graphic Controller
- Klark Teknik DN9696 96-track High Resolution Audio Recorder
- Klark Teknik DN9650 Network Bridge (MADI, Dante, Aviom, Ethersound, CobraNet)

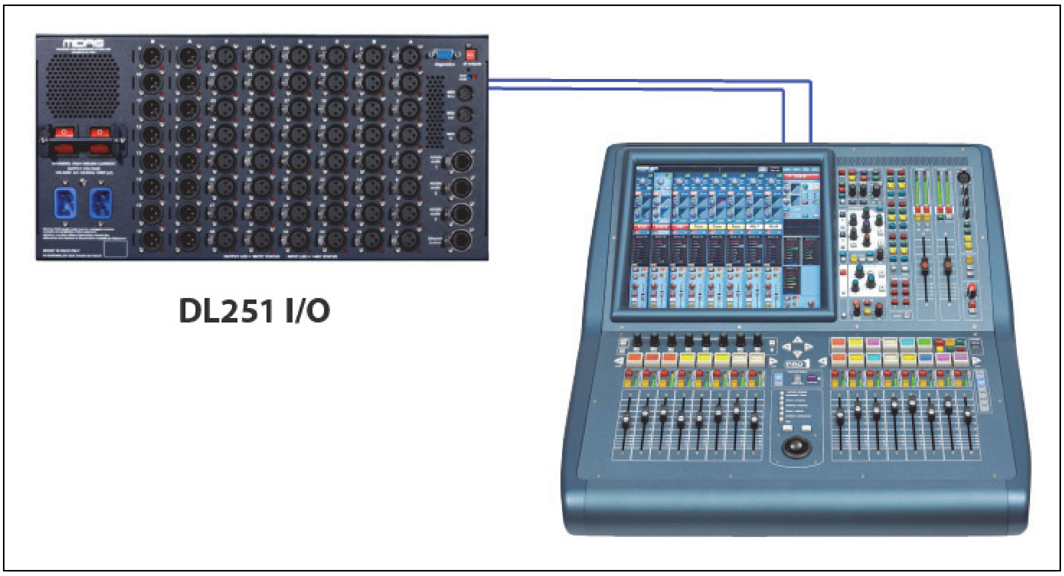
Applications

Although the PRO1 is designed for the traditional touring live sound environment, it is also ideal for medium-sized theatre, small house of worship installations and broadcast. So, being a truly multi-function console in the Midas tradition, the PRO1 is suitable for many applications, such as:

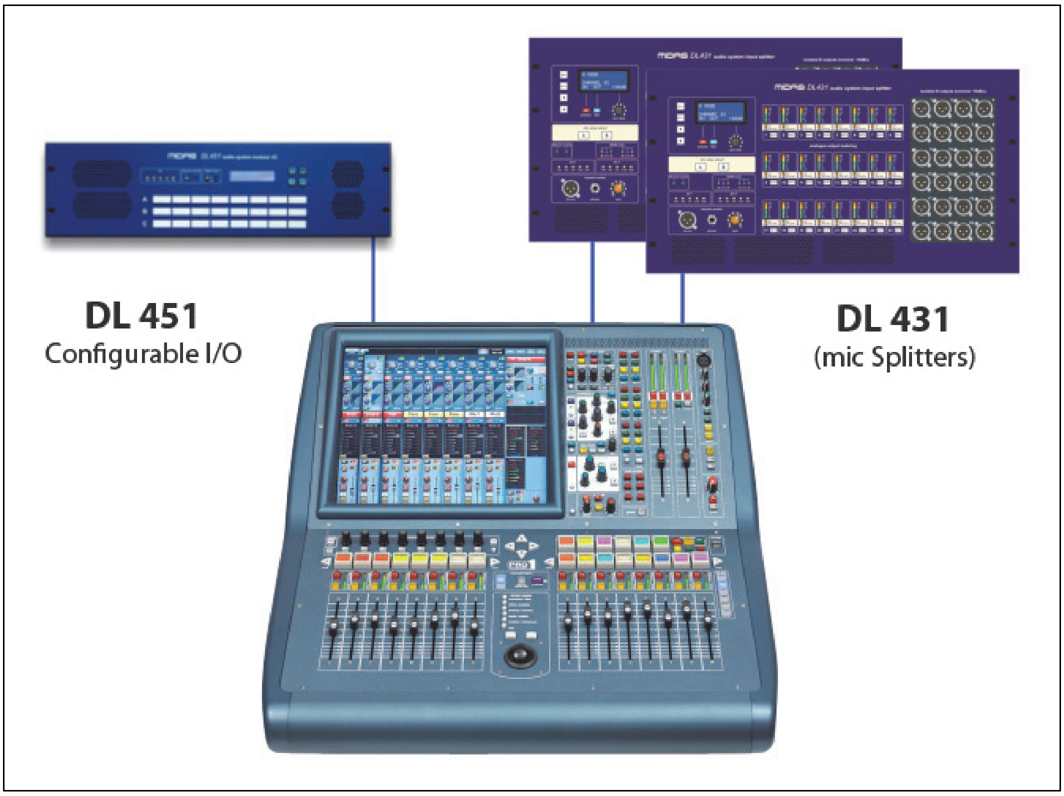
- Live concert sound touring (medium and small scale productions).
- Live concert sound fixed install (medium and small).
- Live sound small theatre MON or FOH duties.
- Live sound house of worship MON or FOH duties.
- Corporate work.
- TV broadcast small outside broadcast (OB) truck.

System configurations

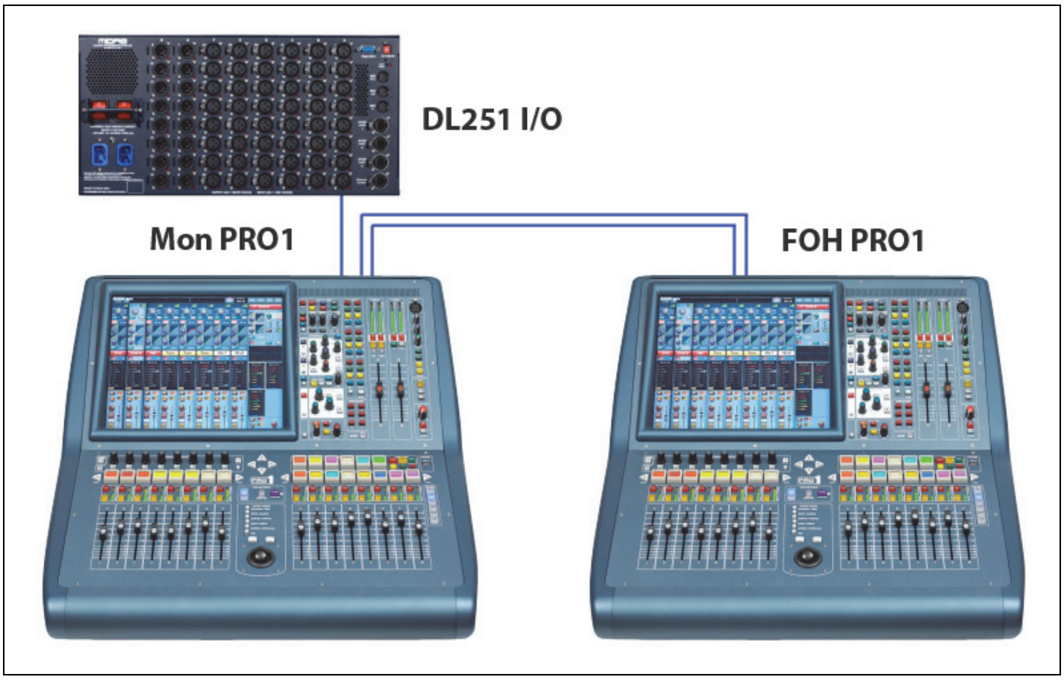
This section shows the basic interconnectivity of a PRO1 Live Audio System and the possible system configurations.



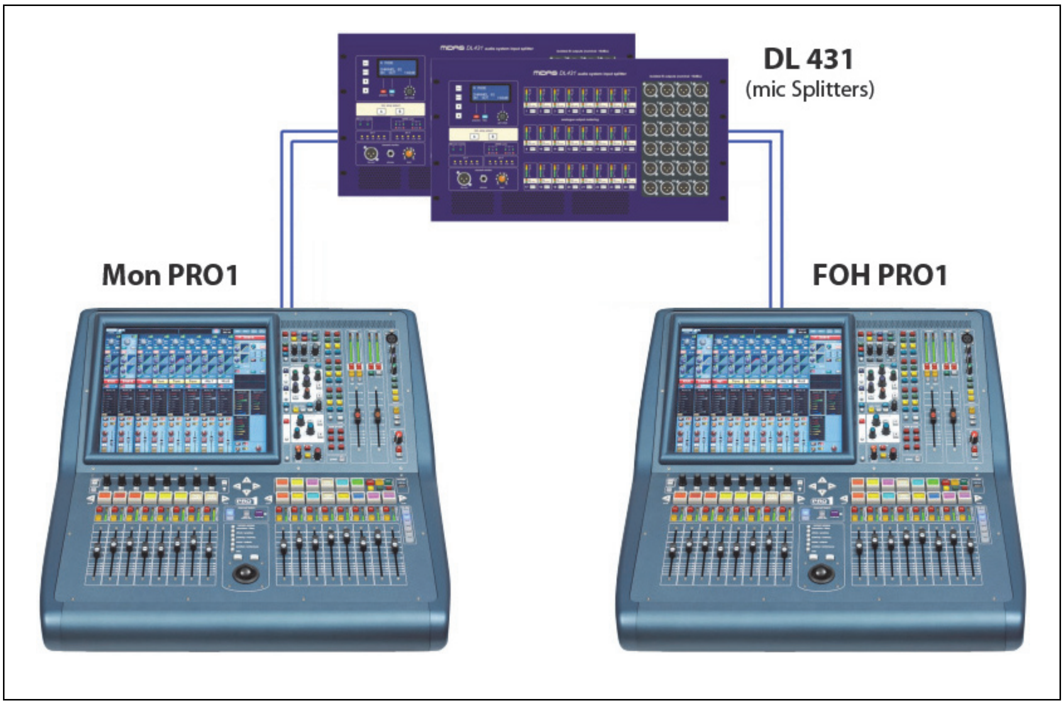
PRO1 with a DL251. This configuration gives 48 mic/line inputs and 16 outputs on stage (DL251) and 24 mic/line inputs and 24 outputs on the console.



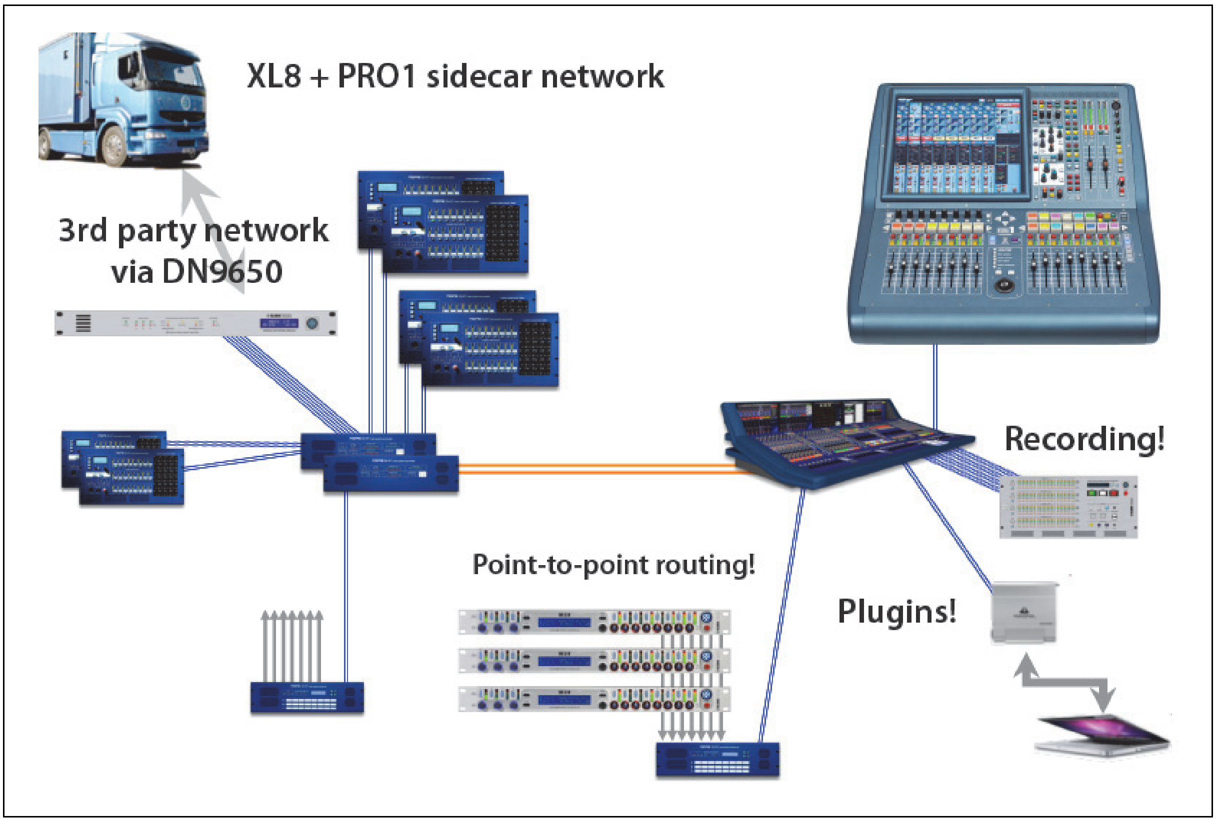
PRO1 system expanded by adding two DL431s and a DL451.



Dual PRO1 FOH and MON system using a DL251 split network



Dual PRO1 FOH and MON system with two XL8 DL431s



PRO1 and XL8 sidecar network with up to 528 inputs and up to 160 simultaneous channels. The PRO1 control centre and engine are connected to the XL8 network as an extender. The channel count depends upon the number of I/Os and their configuration.

Signal flow

The control surface contains the DSP engine, which provides the following time-aligned channels and buses:

- 40 input channels
- 8 aux input channels optionally time-aligned as effects returns or additional input channels
- 16 aux output channels
- 3 master channels
- 8 matrix channels
- 2 solo buses, routable from all locations providing dual monitor formats (in ear/wedge)
- 2 master buses, routable from the 40 inputs and 8 aux inputs, and 16 aux buses
- 8 matrix buses, routable from the 40 inputs and 8 aux inputs, 16 aux buses and 3 masters
- 16 aux buses, routable from the 40 inputs and 8 aux inputs

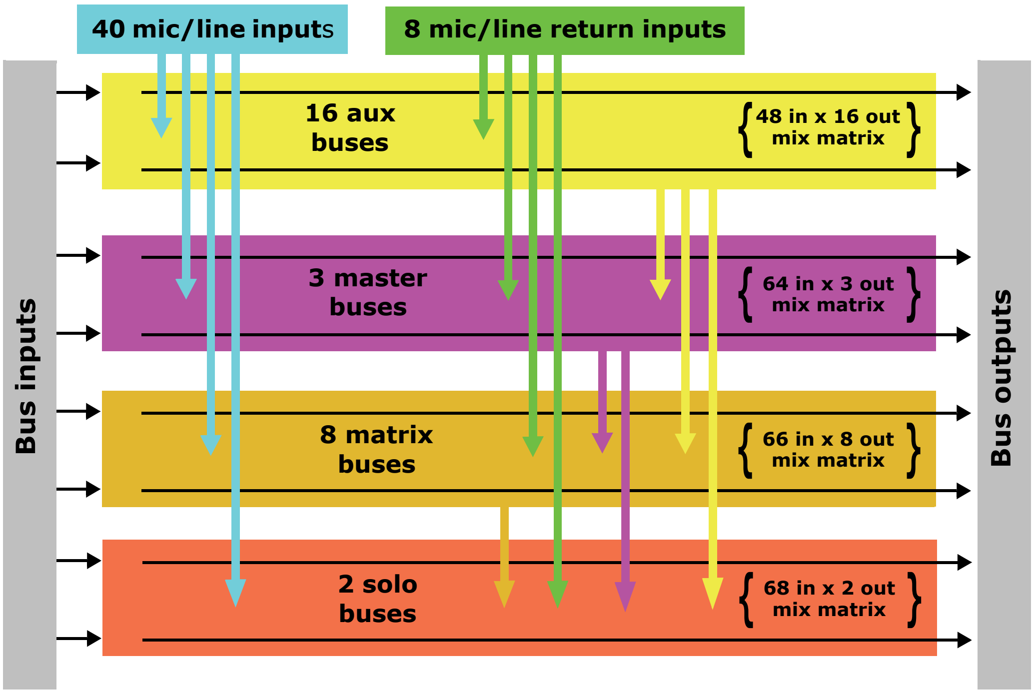
Monitor mixing is easily catered for because the master, matrix and aux buses can all be routed directly from the input channels with independent level controls providing up to 24 monitor mix buses.

Traditional FOH sub group mixing is easily catered for as the aux buses can be individually configured to operate post-channel fader and pan (that is, aux gain fixed at unity).

Auxiliary inputs have two modes, effect return and input channel (default). In input channel mode they are time aligned to the stage and operate as the input channels except that they have no dynamics. In effects return mode they are time aligned to the effects engine and can only route to the matrix and master channels.

Mix matrix

Ultimately, the mix matrix defines the PRO1 Live Audio System's capability. Probably the best way to imagine the mix matrix is to think of an analogue console layout, where inputs run vertically and buses run horizontally. A mix matrix is usually defined as the number of buses and the quantity of simultaneously-mixable inputs there are per bus. The following diagram illustrates the capability within the control centre.



Processing

Although the control centre system allows for considerable insertion of external processing, it also embodies more than enough internal high quality processing to eliminate the need for this.

Input channel processing

Each of the 40 full-function input channels has:

- Analogue and digital gain.
- Phase reverse switch.
- Input delay.
- Swept high pass filter with choice of two filter slopes.
- Swept low pass filter with choice of two filter slopes.
- Frequency-conscious compressor with choice of four compression styles.
- Frequency-conscious noise gate with external side chain.
- Insert point.
- Treble EQ filter with choice of four filter types.
- Parametric hi-mid EQ filter.
- Parametric lo-mid EQ filter.
- Bass EQ filter with choice of four filter types.
- Routing via level controls to 24 mix buses.
- Routing via pan control to left and right master buses.

- Routing to mono master bus.
- Panpot (SIS™).
- Direct output.

Each of the 16 auxiliary inputs has:

- Input gain.
- Source from internal FX or external pool input.
- Fader.
- Panpot (SIS™).
- Routing via level controls to the eight matrix buses.
- Routing via pan control to the left, right and mono master buses.

Mix channel processing

Each of the 16 auxiliary mix buses has:

- Subgroup, auxiliary or mix minus modes.
- Dual mono or stereo pair modes.
- Six-band PEQ.
- Optional 31-band GEQ (replaces PEQ).
- Frequency-conscious compressor with soft clip limiter and choice of five compression styles.
- Insert point.
- Routing via level controls to the 6 matrix buses.
- Routing via pan control to the left, right and mono master buses.
- Direct input.

Each of the eight matrix buses has:

- Six-band PEQ.
- Optional 31-band GEQ (replaces PEQ).
- Five-mode frequency-conscious compressor with soft clip limiter and external side chain.
- Insert point.
- Direct input.

Output channel processing

Each of the eight matrix buses has:

- Six-band PEQ.
- Optional 31-band GEQ (replaces PEQ).
- Five-mode frequency-conscious compressor with soft clip limiter and external side chain.
- Insert point.
- Direct input.

The two master output buses each has:

- Six-band PEQ.
- Optional 31-band GEQ (replaces PEQ).
- Five-mode frequency-conscious compressor with soft clip limiter and external side chain.
- Insert point.
- Direct input.

Effects processing and GEQs

The PRO1 contains six effects processors and eight mono Klark Teknik (KT) GEQs as standard. However, each effect slot can be sacrificed to gain an additional four GEQ effects.

The effects processors (six maximum) can be freely chosen from the following:

- Stereo delay
- Stereo reverb (Klark Teknik DN780)
- Stereo flanger
- Stereo phaser
- Stereo graphic EQ
- Pitch shifter
- Square ONE Dynamics
- Stereo 3-band compressor
- Chamber reverb
- Hall reverb
- Plate reverb
- Vintage room reverb
- Ambience reverb
- Dual stereo chorus
- Dynamic EQ
- Dual stereo delay
- Matrix mixer

Up to 28 mono KT GEQs can be patched into any output. There are many patching options for the effects processors. They can be patched to any source or destination within the system and you can even patch directly to and from an effect externally.

Surround capabilities

Theatres and broadcast have differing requirements for surround, and both are catered for in the PRO1.

Conventional stereo and SIS™ panning is assignable on a channel by channel basis (channel one can be in stereo while channel two can be in SIS™), as follows:

- Stereo left–right routing to master buses.
- SIS™ left–right–centre routing to master buses.

Three additional surround modes operate as follows:

- Quad (left, right, LS and RS).
- Surround (left, centre, right surround).
- 5.1 surround (left, centre, right, subwoofer (LCRS), LS and RS).

Network

The PRO1's digital audio network utilises the physical connectivity of Ethernet (etherCON® connectors and Cat 5e/Cat 6 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 and HyperMAC connections carry digital audio, control data and standard Ethernet traffic bi-directionally down a single cable. Cat 5e cable is used for the 'local' (24-channel) connections. 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 are duplicated for full dual redundancy.

Resilience to failure (redundancy)

The PRO1 Control Centre uses the N+1 principle for the audio, where the AES50 cables include a redundant spare.

The GUI screen can be used to operate the control centre, *even if no control surface hardware is working*.

Latency management

In the interests of reduced latency the primary channel types are restricted to three time zones and the interconnecting buses are restricted to the time in between.

- **First time zone:** Input channels, including aux inputs that are set to input channel mode. (Aux bus time is here.)
- **Second time zone:** Aux channels, including aux inputs that are set to effects return mode. (Master and matrix bus time is here.)
- **Third time zone:** Master and matrix outputs.

Operating system

The operating system of the PRO1 is 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.

GUI

The PRO1 has a daylight-viewable, TFT screen that provides overview and detail status indication. The screen is operated using a trackball and two buttons that, between them, provide the same control as provided by a computer's mouse.

The supplied USB keyboard lets you insert text into pre-defined fields on the GUI screen.

Integration of third party hardware

The PRO1 network includes the capability to interface any third party hardware that uses AES/EBU or AES50 digital audio, or a standard analogue audio interface.

Each PRO1 AES/EBU input and output has a sample rate converter. Synchronisation to external AES3 interfaces can be:

- Global - via inputs on the routers.
- Local to each input.
- Local to each output (synchronisation to adjacent local output).

Multiple local connections can be at different sample rates.

The use of the AES50 protocol for the transmission of digital audio means that any third party digital audio hardware that features this connection can be connected to the Midas network, and will transfer audio to and from the Midas hardware without any additional interfaces or converters (provided it runs in TDM 96kHz mode). This will be particularly useful as the protocol gains acceptance with recording and playback devices, loudspeaker controllers, audio networking systems, digital amplifiers etc.

The PRO1 Control Centre has a DVI connector (called "screen output") on the rear panel, so that the control centre view can be routed to an external monitor.

Chapter 3: About The PRO1 Control Centre

This chapter introduces you to the PRO1 Control Centre and provides a brief hardware description.

Overview of the PRO1 Control Centre

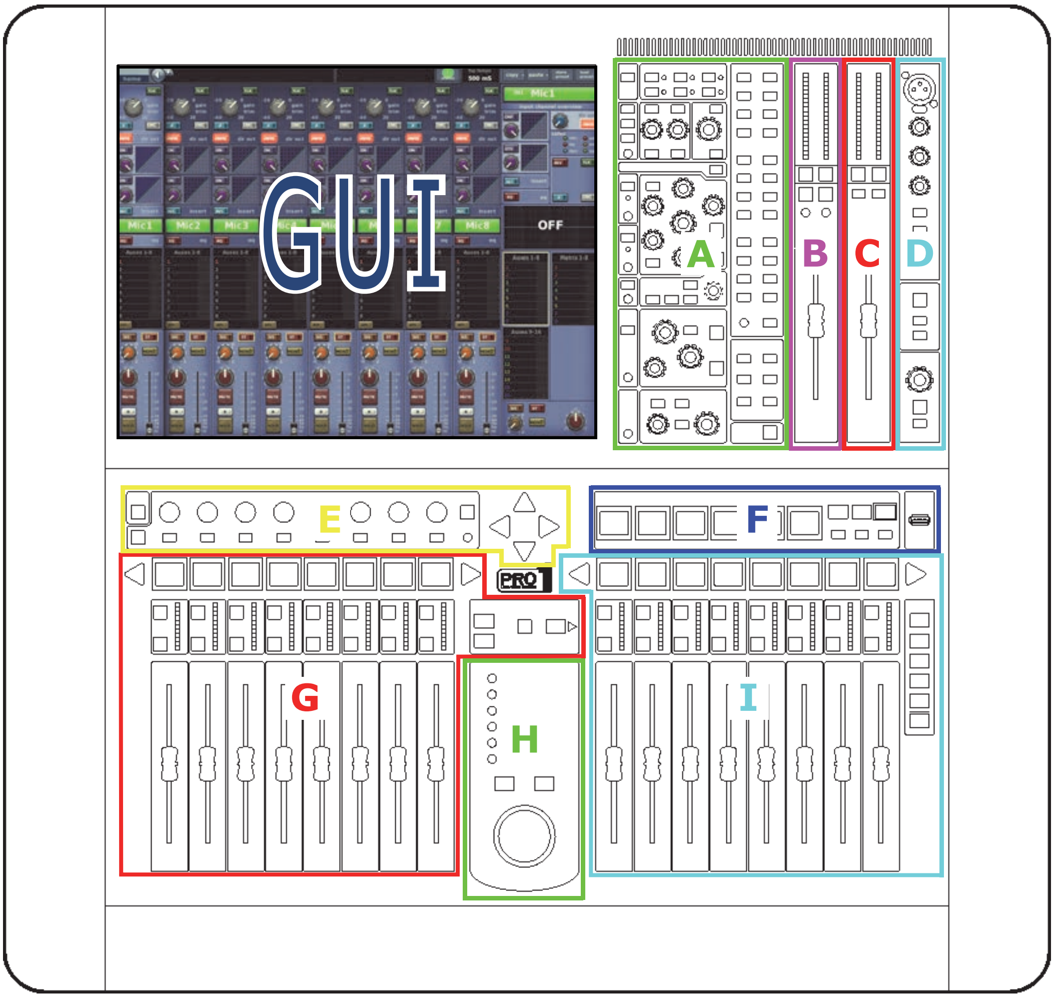
The PRO1 Control Centre comprises a combined control surface and graphical user interface (GUI) that provides an array of easy-to-use controls for the precise manipulation of audio.

The control surface has been designed to emulate the fast access of an analogue control surface by presenting things in a familiar, consistent and logical way. This is enhanced by visual representations on the GUI screen, which also provides extra functionality.

PRO1 control centre

PRO1 control surface

During show time the functions that require fast access are controlled by control knobs (rotary encoders), pushbutton switches, faders etc. More complex functions that do not require this fast access are controlled via the GUI screen using the trackball and left and right navigational keys.



Main areas of control surface

Item	*Description*
A	**Channel processing areas** Contains processing areas, such as the D-zone (dynamic), E-zone (EQ) and mix buses, to the left that provide a more comprehensive control of a channel or mix bus by allowing detailed audio parameter adjustment. The mix bus selection controls are to the right of the channel processing areas, under which are the mute groups and global tap sections.
B	**Masters** This section has a single fader with left and right meters, solo and mute buttons, and left and right master channel selection buttons. It controls the left and right master outputs.
C	**monitors** This section contains a single fader and is used for the stereo left and right monitor a or b channels.
D	**Comms** Talk, solo, headphones, etc.

Item	Description
E	**I zone** Operator-assignable device controls, which also includes a button for direct access to the console overview screen at the upper-left corner and four arrowed scrolling buttons to the right. Utilises the input fast zone's eight channel faders and eight horizontal assignable controls, which operate in conjunction with a vertical channel detail controller and the GUI screen. The usable channel fader control can be increased to 16 by activating the **EXTEND** button which allows the channel fader to extend over the usual mix faders (temporarily overriding them).
F	**POPulation groups, automation and USB connection** Six POPulation groups to the left provide instant navigation to a user-defined set of channels. When a POPulation group is selected, they are mapped to the input bay faders and, when in **EXTEND** mode, also to the mix bay faders. The **automation** and **storage** sections to the right contain scene store/recall controls and USB connection for transporting showfiles, presets and preferences, and carrying out upgrades.
G	**Input fast zone** Contains the operator's 'must have now' controls in eight input fast strips. Provides faders, meters, mutes, solos and selection (LCD) switches for detailed and complex manipulation of multiple channels within the console. Typically, this would be input channels (inputs and aux returns), outputs (aux sends, matrices and masters) or group members POPulation, VCA, etc.). Scrolling is restricted to the channel types currently assigned to the channel faders. This zone also has two left/right navigation buttons for scrolling the channels and a navigation section.
H	**Navigation zone** For GUI screen navigation via a trackball. Includes a **screen access** panel for direct GUI screen access.
I	**Mix fast zone** Eight mix faders (with LCD switches) give the operator primary mix control. Additional input channel faders can be provided by using **EXTEND,** which override the mix faders. They attach themselves numerically to right of the channels already assigned to the channel faders. A navigation section to the right of the mix faders contains channel assign buttons for the mix fader bay.

Connections

The PRO1 has a rear panel and sockets on the control surface for the connection of mains power, AES50 I/O unit(s), USB devices (includes showfile storage and system updates), keyboards, headphones, talk mics, communications, external monitor, AES3 synchronisation, diagnostics (for service personnel only), lamp and word clocks (75R).

For more information, see Chapter 29 "Connections" on page 241.

Rear panel connectors (top) and storage USB connector on the control surface

Keyboard

A QWERTY keyboard (supplied) is used mainly for inputting text, such as when configuring a channel or using automation.

External interfaces

Various devices can be used with the PRO1, such as:

- **External USB mouse:** Instead of using the navigation zone to operate the GUI screen, you can use an external USB mouse. This can be plugged into one of the USB connectors on the rear panel. The USB mouse behaves in the same way as any PC mouse.
- **External USB keyboard:** You can plug a USB keyboard into one of the USB connectors on the rear panel of the PRO1.

- **MIDI:** Standard 5-pin connectors are housed in the rear panel of the PRO1 for use as MIDI in, out and through ports.
- **USB:** Two USB ports are provided on the rear panel of the PRO1. In addition, there is a USB port on the control surface (**storage** section) for removable storage via a USB memory stick.
- **DVI ports for external monitor:** The control centre has a **screen output** DVI connector on the rear panel of the PRO1 for connecting an external monitor. This lets you view remotely what is shown on a GUI screen.

Getting Started

Chapter 4: Setting Up The System

! Before installing, setting up or operating this equipment, make sure that you have read and fully understand all of this section and the "IMPORTANT SAFETY INSTRUCTIONS" at the front of this manual.

This chapter shows you how to set up an PRO1 Live Audio System to its default configuration.

Note: *If you want to set up the PRO1 Live Audio System using a configuration other than the default, please contact Midas Technical Support for details.*

Initial set-up procedure

Initial system set-up basically comprises:

- **Unpacking and checking the equipment** — see "Unpacking the equipment" below.
- **Making up the rack** — see "Making up a rack" below.
- **Installation** — see "Installation" below.
- **Connecting up the equipment** — see "Connecting up" on page 28.
- **Powering the equipment** — see "Powering the PRO1 system" on page 30.
- **Setting up the I/O rack device (initial patching)** — see "Configuring the devices" on page 58.
- **Configuring the DL251 Audio System I/O unit** — see the DL251/DL252 Audio System I/O Operator Manual.

Unpacking the equipment

After carefully unpacking the equipment, save all packing materials, as they will prove useful if you need 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.

Installation

Before installing the equipment:

- Make sure the equipment is correctly connected to the protective earth conductor of the mains voltage supply of the system installation through the mains leads.
- Power to the equipment must be via a fused spur(s).
- Power plugs must be inserted in socket outlets provided with protective earth contacts. The electrical supply at the socket outlets must provide appropriate over-current protection.
- Both the mains supply and the quality of earthing must be adequate for the equipment.

- Before connecting up the equipment, check that the mains power supply voltage rating corresponds with the local mains power supply. The rating of the mains power supply voltage is printed on the equipment.

Handling the equipment

Completely isolate the equipment electrically and disconnect all cables from the equipment before moving it.

When lifting or moving the equipment, always take its size and weight into consideration. Use suitable lifting equipment or transporting gear, or sufficient additional personnel.

Do not insert your fingers or hand in any gaps or openings on the equipment, for example, vents.

Avoid inserting or dropping foreign objects, such as paper, plastic, metal etc., into any gaps or openings on the equipment, for example, vents. If this happens, immediately disconnect the equipment from the AC mains. Then have the equipment inspected by the manufacturer's qualified service personnel.

Do not press or rub on the sensitive surface of the GUI screens.

If the glass of the GUI screen is broken, liquid crystals shouldn't leak through the break due to the surface tension of the thin layer and the type of construction of the LCD panel. However, in the unlikely event that you do make contact with this substance, please wash it out with soap.

Location

Ideally a cool area is preferred, away from power distribution equipment or other potential sources of interference.

Do not install the equipment in places of poor ventilation.

Do not install this equipment in a location subjected to excessive heat, dust or mechanical vibration. Allow for adequate ventilation around the equipment, making sure that its fans and vents are not obstructed. Wherever possible, keep the equipment out of direct sunlight.

Do not place the equipment in an unstable condition where it might accidentally fall over.

Make sure that the mains voltage and fuse rating information of the equipment will be visible after installation.

Electric fields

Caution:
In accordance with Part 15 of the FCC Rules & Regulations, "... changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment."

Should this product be used in an electromagnetic field that is amplitude modulated by an audio frequency signal (20Hz to 20kHz), the signal to noise ratio may be degraded. Degradation of up to 60dB at a frequency corresponding to the modulation signal may be experienced under extreme conditions (3V/m, 90% modulation).

Optional equipment

Unless advised otherwise, optional equipment must only be installed by service personnel and in accordance with the appropriate assembly and usage regulations.

Special accessories

To comply with part 15 of the FCC Rules, any special accessories (that is, items that cannot be readily obtained from multiple retail outlets) supplied with this equipment must be used with this equipment; do not use any alternatives as they may not fulfil the RF requirement.

Making up a rack

If you are making up a rack, there are careful considerations to be addressed beforehand, which are outlined in this section.

Outboard equipment racks

To ensure the correct installation and function of the outboard equipment, such as the DL251 Audio System I/O and DN9696 recorder, racks must meet the following general requirements.

- **Shock mounting (for non-installation environments):** The racks must provide adequate shock protection of the units they house by incorporating appropriately-designed shock protection methods, for example, a foam-suspended rack or a frame suspended on anti-vibration mounts etc.
- **Ventilation:** Midas rack 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 unit **must** be prevented. Midas recommends that racks with fully opening front and rear doors are used.

  Note: *Never combine units in the same rack that have been designed for a ventilation air flow direction other than that for the PRO1 units. To avoid this, we recommend that any non-PRO1 units are housed separately.*
- **Rack mount supports:** Always secure the rear of the units to the rack via their rear rack mount support brackets. These brackets are fitted to every PRO1 unit and are recommended for use in touring applications. The rack mount support fixing hole centres are at a depth of approximately 395 mm from the front panel (this dimension may differ slightly on the DN9696).
- **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 amount of personnel suitable for the task. Also, these handles must be fit for purpose.
- **Clearance at rear of units:** To ensure an adequate clearance at the rear of the units, we recommend that the rack depth, that is, the distance from the front rack strip to the rear of the rack, is a minimum of 700 mm.
- **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 give maximum visibility of the units' LEDs for determining communication status, link status, condition of audio etc.

Connecting up

Connect your system according to your requirements. For examples of system configuration, see “System configurations” on page 7.

To ensure the correct and reliable operation of your equipment, only high quality, balanced, screened, twisted pair audio cable should be used.

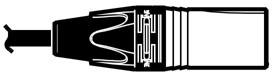
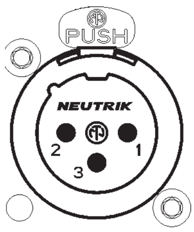
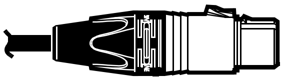
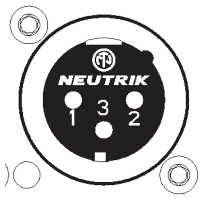
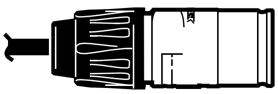
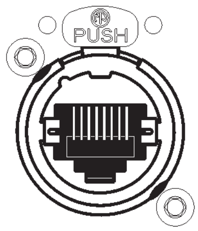
XLR connector shells should be of metal construction so that they provide a screen when connected to the console and, where appropriate, they should have Pin 1 connected to the cable screen.

All Jack connector shells should be connected to the cable screen.

Audio connections

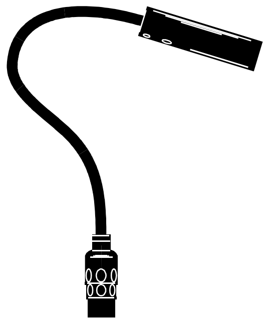
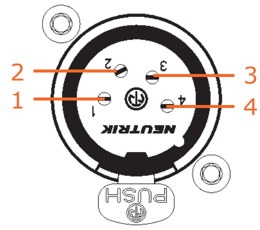
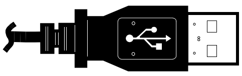
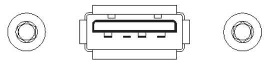
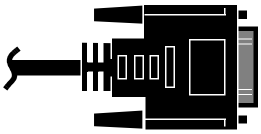
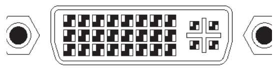
This section gives details of the audio connections of the PRO1 Control Centre.

Table 1: Connector pinouts

Connector on rear panel	***Example of plug***	***Pinouts***	***Example of socket***
Male XLR chassis connector (output)		1 = ground 2 = hot 3 = cold	
Female XLR chassis connector (mic input)		1 = ground 2 = hot 3 = cold	
Ethernet EtherCon® connector		N/A	

Other connections

The section gives details of the other PRO1 Control Centre interconnections.

Description	*Example*	*Pinouts*	*Example of socket*
4-pin, male XLR chassis connector(s) on the rear panel for connecting 12V/5W lamp(s)		1 = N/A 2 = N/A 3 = ground 4 = 12V	
USB type A socket		N/A	
DVI **screen output** socket for connecting		N/A	
foot switch socket for connecting a 1/4” TRS Jack plug		1 (tip) = foot switch 1 2 (ring) = foot switch 2 3 (sleeve) = ground	

The **diagnostics** socket is for service personnel use only.

Powering the PRO1 system

The internal power supplies are of the switch mode type that automatically sense the incoming mains voltage and will work where the nominal voltage is in the range 100VAC to 240VAC.

! **When removing the equipment's electric plug from the outlet, always hold the plug itself and not the cable. Pulling out the plug by the cable can damage it.**

! **Never insert or remove an electric plug with wet hands.**

! **Before switching the PRO1 on/off, make sure that all monitor loudspeaker power amplifiers are turned off or muted.**

The following details the recommended power up and power down procedures for the PRO1 system.

>> To power up the PRO1 system

! **DO NOT switch on the speaker sub-system until after the start-up of the PRO1 system has been completed.**

After all PRO1 system interconnections have been made, start up the system:

1. Make sure that all of the PRO1 system equipment is switched off (the PRO1 Control Centre and speaker sub-system).
2. Switch on the PRO1 Control Centre (see "To switch on the PRO1 Control Centre" on page 31).
3. On the PRO1 Control Centre, move all of the monitor and master channel faders to the minimum position and mute all of the master channels.
4. After the **status** LED (top of the GUI screen on the PRO1 Control Centre) turns green, switch on the speaker sub-system.
5. Switch on the audio source and start playing the audio.
6. On the PRO1 Control Centre, check that the audio inputs are routed to the master channels. Then, unmute the master channels and gradually increase their faders, while listening to the sound levels from the speakers.

If there are no sounds at all from the speakers when the faders are at maximum, move the faders to below the 0dB level and check if the audio is muted somewhere along the input paths and also check that the individual speakers are switched on. If there is still no sound from the speakers, contact Midas Technical Support.

>> To power down the PRO1 system

! **BEFORE switching off any of the PRO1 system components, make sure to mute the audio from the speakers and switch off the speaker sub-system.**

1. Mute the audio from the speakers and switch off the speaker sub-system.
2. Switch off the PRO1 Control Centre (see "To switch off the PRO1 Control Centre" on page 31).

Switching the PRO1 Control Centre on/off

Carry out the following to switch the PRO1 Control Centre on/off in a safe manner.

>> To switch on the PRO1 Control Centre

! Before switching the PRO1 on, make sure that all monitor loudspeaker power amplifiers are turned off or muted.

After connecting up the audio cables, carry out the following:

1. Plug the mains cable into a mains power outlet and then connect the IEC plug to a mains inlet socket on the rear of the PRO1 Control Centre.
2. Make sure that all monitor loudspeaker power amplifiers are turned off or muted and then switch on the mains on/off isolator switch.
3. The control centre will power up; the GUI will display the default screen and all the controls will be set to default. You are now ready to start using the PRO1 Control Centre.

>> To switch off the PRO1 Control Centre

1. Make sure you have saved any shows, scenes or settings you require (see "Saving your show files to a USB memory stick" on page 88).
2. At the GUI, choose **home** ▸ ***Preferences*** ▸ ***Shutdown System.***
3. At the **Shutdown ENTIRE system?** prompt, click **OK.** This initiates the shutdown down sequence.

   During the shutdown sequence the GUI screen will shutdown and all of the LCD select buttons on the control surface will turn red.

   When the shutdown sequence has finished the LCD select buttons on the control surface will turn green, and the appropriate ones will also display text messages, accordingly.

   ! During the shutdown sequence, when the LCD select buttons on the control surface are red, do not switch off the mains power supply, and when they are green, it is OK to switch off the mains power supply.

4. Make sure that the shutdown sequence has finished, and then switch off the mains on/off isolator switch (rear of control centre).
5. Disconnect the mains IEC connector from the socket on rear of PRO1 Control Centre.

Basic Operation Of The PRO1

Chapter 5: Before You Start

This chapter provides useful background information on PRO1 operation.

While this system is a complex, high-tech piece of equipment, we have made it as easy to use and as user-friendly as possible.

Principles of operation

PRO1 Control Centre 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.

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.

Hints and tips

During operation, we recommend that you carry out the following:

- **Check what is hidden** On the PRO1, unlike on an analogue control surface, some of the settings and parameters will be hidden from view (stored in the computer memory of the PRO1). To make sure there are no hidden surprises, such as a reverb send left from a previous mix, we recommend that you view unused parameters at various times during a mix.
- **Check the Console Overview screen** It is a good idea to frequently monitor the **Console Overview** screen (press the **HOME** button underneath the GUI screen), which provides at a glance an overview of the control centre's status and operation. It shows all the meters and the status condition of faders and some switches, such as solos and mutes. However, some things will still remain hidden.

Saving your work

We recommend that you save your work regularly while carrying out the procedures included in this chapter. Not only is this good practice during normal PRO1 operation, but in this instance it may save you from losing some set-ups that could prove useful later on. To do this, create a new show (see "To create a new show" on page 79), and then continue reading through the remainder of this section, following the instructions carefully. Save your work at convenient points (see "To create a new scene using the current settings" on page 81 and "To save a show or create a new one from the current settings" on page 80).

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 to the show file unless you manually store a scene. The show file remains unsaved in RAM after storing a scene.

  Although the state of the control centre is copied every five seconds, it is not stored in a scene. Instead, it is placed in the NVRAM (non-volatile random access memory) of the control centre's memory, which is a type of RAM that doesn't lose its data when the power goes off. If the control centre loses power accidentally, these settings are loaded so that audio parameters are identical, thus avoiding audio level jumps. **When power is lost, the showfile loaded (if any) will not subsequently be restored, and any unsaved changes to it will be lost.**

- **Saving a show** copies the show file onto the internal solid-state disk of the PRO1. This provides you with a 'permanent' copy, provided you shut down the system properly as detailed in the following section.

Shutting down the PRO1 Live Audio System properly

When switching off the PRO1 Control Centre, we recommend that you use the shutdown option of the GUI menu (see "To switch off the PRO1 Control Centre" on page 31).

By using shutdown, the cached copy of the show data, which is maintained by the system, is automatically stored. Shutdown then uses the current showfile, NVRAM data and cache files to restore the PRO1 Control Centre to *exactly* the same state as at power down; even to the point of loading the unsaved show and placing you at the correct scene, with non-stored scene data at the control surface.

If you don't use the **Shutdown** option the audio parameters are still restored, but the show and show status (saved/unsaved) cannot be restored automatically. You must manually reload the show and any unsaved changes will be lost.

Chapter 6: Working With The PRO1 Control Centre

This chapter is intended to familiarise you with the controls (control surface and GUI) of the PRO1 Control Centre.

Although many controls on the PRO1 Control Centre are similar to their equivalent analogue-type counterparts, some have been specifically designed for the PRO1, particularly those for navigation and GUI operation. As you will probably have had experience on analogue consoles, you will already be familiar with most of the PRO1 controls and their operation.

The GUI can be used to replicate most of the control surface functions and even has many of its own. However, the emphasis in this chapter — and throughout the manual — is on using the control surface because, generally, it is quicker and more intuitive. GUI methods will be included where they are anomalous or there is no control surface equivalent.

The navigational controls, such as quick access buttons and scroll buttons, are described in Chapter 7 "Navigation" on page 43, and the ones specifically for automation can be found in “Managing the scenes” on page 81.

For more information on the operation and function of specific controls, see Volume “Description” on page 239.

About the PRO1 controls

The following table shows some of the controls that can be found on the control surface of the PRO1.

Type	*Description*	*Example(s)*
Pushbutton	Generally two-state, that is, on/off or enabled/disabled, and backlit or with an integral LCD for status indication.	
Control knob	In general, the control knobs (rotary controls) are touch-sensitive, their adjustment being shown on the GUI. Some control knobs are backlit to help identify their current role.	
Fader	The high quality motorised faders are, similarly to the control knobs, touch-sensitive.	

Type	Description	Example(s)
LED	Status indication.	
Meter	All of the input, output and monitor peak level meters on the control surface are shown simultaneously on the GUI's **Console Overview** screen. Additional peak level meters are included in the direct input and direct output sections of the input bay and mix bay channel strips, respectively.	

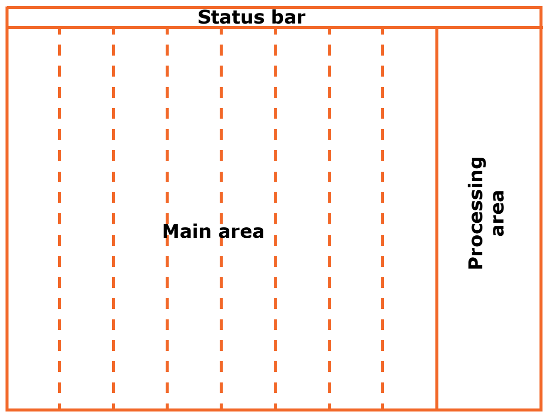
About GUI operation

The GUI is not just an additional feature that enhances control surface operation, it is a fully-featured tool in its own right. Not only does it show what is happening on the control surface, but all of its controls are functional. The GUI contains most of the controls found on the control surface and, in addition, has features that allow configuration of the PRO1 and provide extra functionality.

For information on how to operate the GUI (for example, using click and drag, using the **screen access** buttons), see PRO1 GUI Navigation in the PRO1 Live Audio System Quick Start Guide.

GUI screen layout

The GUI screen has three distinct areas (shown right). The status bar is constantly displayed throughout console operation, while the main and processing areas are context-dependent and may be combined, particularly for GUI menu screens. During channel operation the main area will be divided into eight strips (fast strips) displaying the channels currently assigned to the channel faders at the control surface. The processing area shows a detailed view of the parameters of the currently selected channel section; with no channel selected the processing area will be blank.



To distinguish between channel types, they are colour-coded: inputs = blue; aux inputs = dark green; aux sends = light green; and matrix and masters both = black.

Status bar

The Status bar, which is always displayed at the top of the GUI screen, contains a number of elements as shown in the following diagram.

Item	*Description*
1	**home** button, opens the GUI menu (see “GUI menu flowchart” on page 294).
2	Screen navigation buttons (see “To find a GUI screen that you recently opened” on page 44).
3	Name of current screen.
4	“Not Saved” message appears when the scene/preset library file contains changes that have not been saved.
5	Title of currently selected scene, if any.
6	**status** LED indicates the health and status of the system (see “Diagnostics” on page 329).
7	**Tap Tempo** section, which shows the current tap tempo time (ms).
8	Copy and paste buttons (see “Using copy and paste” on page 84).
9	User library buttons (see “User library (presets)” on page 85).

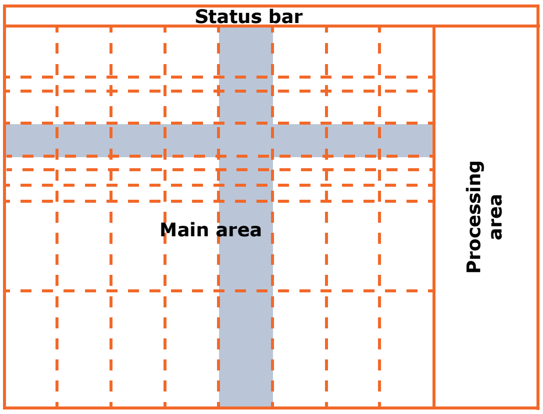
GUI cross-hair

The GUI uses a ‘cross hair’ configuration (shown right) to show which channel is currently selected in the channel fader section and also its current processing area selection.

The processing area can show an output, while the main area displays input channel overviews, and vice versa.

You can navigate the eight channel faders to show output channel overviews.

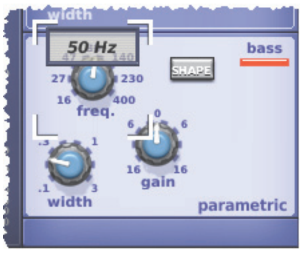
In the example shown right, input channel 5 and its gate detail are selected.

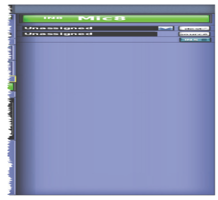


Values displayed on touch (control knob/fader)

You can configure the PRO1 so that, when operating a control knob or fader, the parameter value associated with that control will be displayed on the GUI or LCD select button, respectively. To do this, select the **Display Rotary Values** option (see “Setting the user interface preferences” on page 225).

The following typical examples show the GUI display when adjusting the frequency control knob of the parametric EQ (immediate right) and an assignable control knob to adjust the level of aux 1 (far right).





Fader values (to the nearest dB) are shown at the bottom of their respective LCD select button, regardless of their current channel assignment.

Operating the GUI screen controls

This section shows you how to operate GUI screen elements, such as buttons, control knobs, drop-down lists and sliders.

>> To switch a GUI button on/off

Click the button. If it has a status indicator, this will illuminate/extinguish to show that it is on/off, respectively.

>> To adjust a GUI control knob or fader

Use a drag operation. Move the pointer up/down/left/right for adjustment.

Using drop-down lists

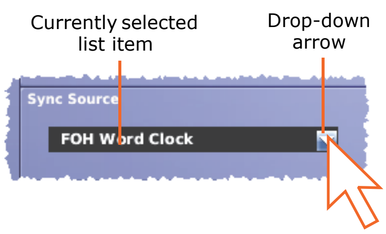
Certain configurable name fields, particularly the signal routing ones, have drop-down lists that offer a number of preset or context-sensitive options to choose from. Long lists — containing more options than can be displayed simultaneously — have sliders that allow you to access all of the options.

>> To select an option from a drop-down list

1 Click the drop-down arrow. The drop-down list will unfold to display some or all of its contents, depending on how many items it contains.

2 Do one of the following:

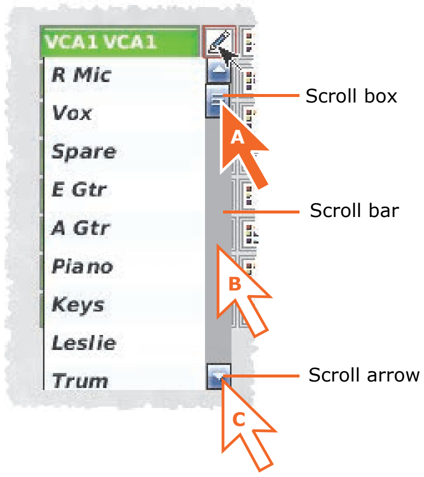
- Click the option you require.
- If necessary, scroll the list (see “To scroll a drop-down list” below) to display the option, and then click it.



>> To scroll a drop-down list

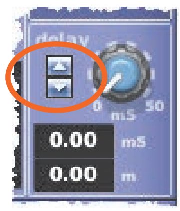
With the drop-down list displayed, do one of the following:

- **A** — Drag the scroll box.
- **B** — Click the scroll bar. The scroll box will 'jump' in the direction of the click to another position in the scroll bar.
- **C** — Click an up/down scroll arrow. The scroll box will 'jump' in the direction of the scroll arrow to another scroll bar position.



Spin buttons

Up/down spin buttons (highlighted right) let you increase/decrease the attribute or value of an item. For example, the amount of time a signal is delayed.



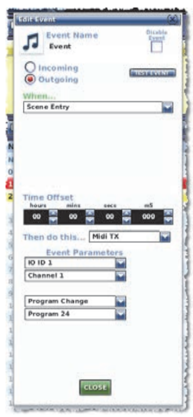
Radio buttons

Radio buttons let you choose a single option from a list of options.

About windows

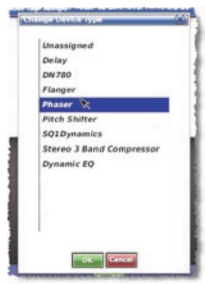
There are three main types of window you will encounter when using the GUI:

- **Parameter** windows contain elements (parameters) that you can select or edit, such as options, lists, tick boxes, text fields etc.
- **Alert** windows contain text that can be a prompt or an error message. Generally, this type of window will contain a user-editable text field and **OK** and **CANCEL** buttons (all controls are inhibited until one of these buttons is clicked or the window is closed).
- **Option** windows have a number of user-selectable options in the form of a list, and may include **OK** and **CANCEL** buttons.



Properties window

Alert window



Option window

>> To close a window

Do one of the following:

- To acknowledge your changes, click **OK.**

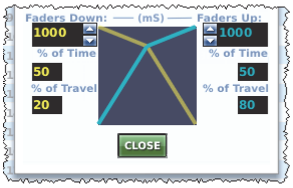
- To cancel your changes, click **CANCEL.**
- Click "(X)" at the upper-right corner of the window.

>> To move a window

Use drag, by clicking on the window's blue bar (top) and dragging the window where you want it.

Graphs

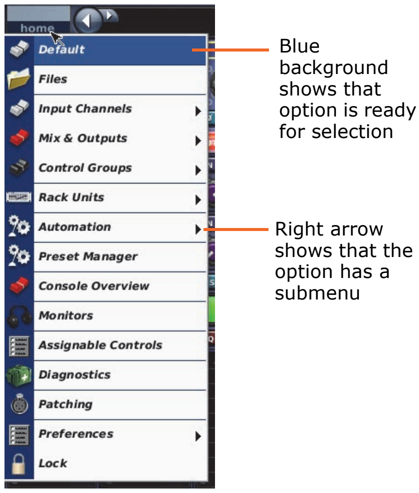
Some graphs, such as the crossfade (immediate right) and surround spatial diagram (far right), can be edited on the GUI using drag.



Using the GUI menu

The GUI main menu (shown right) is opened by clicking the **home** button, which is constantly displayed at the upper-left corner of all GUI screens. To select a submenu option, move the pointer over the arrow to the right of the menu option (the submenu will open automatically to the right of the arrow) and choose the option you want.

Note: *Throughout this manual, menu/submenu option selection sequences are shown in the following format:* ***home* ▸ *Preferences* ▸ *General*** *(for example, for choosing the general preferences screen).*



Text editing

For text editing on the GUI (for example, to configure input and output channel names) you need to attach the supplied external USB keyboard (see "Rear panel connectors" on page 241). You can also use the navigation zone to help you (for example, by highlighting portions of text using drag).

>> To enter/edit text via the keyboard

1. At the GUI, click within the desired text box. This will place an insertion point in it; the pointer will change to an I-beam shape.
2. Using the keyboard, edit/type in the new text. Any existing text can be edited or deleted using the right-click cut, copy and paste options.
3. On the keyboard, press **ENTER** to exit the text box (or click on an empty area of the GUI screen). The pointer will change back to an arrow shape.

Chapter 7: Navigation

This chapter introduces you to PRO1 navigation and shows you how to use the navigational tools of the PRO1 Control Centre to navigate its channels, groups and buses.

For information on other types of PRO1 navigation, refer to the following:

- To navigate the scenes in automation, see “Managing the scenes” on page 81.
- To open the GUI’s **Patching** screen, see “Navigating to the Patching screen” on page 54.
- To navigate the assignable controls, see Chapter 19 "Assignable Controls" on page 169.

An introduction to PRO1 navigation

The PRO1 provides you with unique navigational controls to quickly and easily access the items, such as channels, buses, groups and processing areas, that you will require for mixing.

Navigation is an important feature of the PRO1 Control Centre. 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 amount of channels can be at the control surface at any time, while the others are ‘hidden’. So, navigation is required to access these hidden channels whenever you need them.

The PRO1 has advanced navigational functions that are explained in this chapter.

Channel fader navigation

The channel fader section has eight channels. You can scroll (continuously) from inputs to outputs and vice versa, but only one type can be assigned to the control surface at any time. Channels are handled in banks of eight and are shown on the control surface and GUI in ascending order from left to right.

The following lists the types of navigation available on the PRO1.

- Scrolling single channels
- Scrolling channels banks
- Assigning a mix bus to the channel faders
- Scrolling the mix buses
- Assigning input channels to the channel fader bay or mix bay
- Assigning output channels to the channel fader bay or mix bay
- Navigating the processing area assignments to the channel faders
- Scrolling assignable control assignments
- Swapping the assignable control button assignments
- Assigning VCA groups to the mix bay

- Assigning GEQs to the mix bay
- Recalling the currently selected channel to the channel fader bay

For more information, see the PRO1 Live Audio System Quick Start Guide.

About GUI navigation

While the control surface provides instant, one-button access to many controls, the GUI provides an alternative way of navigating the PRO1 and offers some unique methods of its own. The GUI menu gives you access to all of the screens that you will need and you can even navigate backwards/forwards through the screens that you have recently opened.

Don't forget that you can access some of the GUI screens directly by using the buttons in the ***screen access*** *panel of the navigation zone.*

>> To select a channel/group via the GUI

Click a non-control area of the channel. The local processing area will be assigned to the local channel strip (control surface and GUI).

>> To select a channel/group using the GUI menu

At the GUI, do one of the following:

- To select an input channel, choose **home** ▸ ***Input Channels.*** Then, click the bank containing the input channel you want to open (for example, channels 9-16), and click the desired input channel (for example, IN14).
- To select an output channel, choose **home** ▸ ***Mix & Outputs.*** Then, click the bank containing the output channel you want to open (for example, Aux Sends 1-16), and click the desired output channel (for example, AS7).
- To select a VCA/POPulation group, choose **home** ▸ ***Control Groups*** ▸ ***VCA Groups.*** Then click a non-control area in the desired group (for example, VCA5).

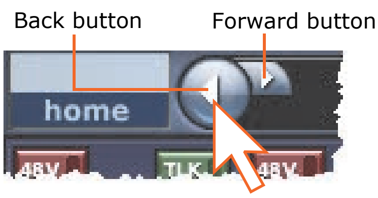
>> To assign a processing area to the output channel strip via the GUI

Click within one of the processing area panels in a GUI fast strip, making sure you avoid a control.

>> To find a GUI screen that you recently opened

Use the back/forward browser buttons to do one of the following:

- To return to the GUI screen you just opened, click the back button (as shown right).
- To open one of the GUI screens you have recently visited, click the back/forward buttons. The back button will take you back through your browser history, while the forward button goes the opposite way.



The back/forward buttons, which are always to the right of the **home** button, are similar to those on standard browsers found on any PC.

Chapter 8: Patching

This chapter describes the patching function of the PRO1.

Introduction

The patching function is fundamental to PRO1 operation as, until the I/Os have been correctly patched, you won't get any audio. Patching is done entirely at the **Patching** screen, which is a GUI menu option. This screen lets you carry out all the routing requirements of the PRO1 by providing 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 any other devices connected within the system. For example, you can adjust the analogue gain, select +48V phantom voltage etc.

Terms used in PRO1 patching

- **Checkpoint:** A patching data store point, created by clicking **CHECKPOINT**.
- **Destination:** The patch connector to which a signal is routed.
- **Device:** A diagram on an I/O tab representing a physical rack unit, such as a line I/O, mic splitter, DN9696, AES50 etc.
- **Drag:** A method of selecting a block of source patch connectors in the **From** section of the Patching screen (see "To select a block of patch connectors in the From section" on page 63).
- **From section:** The left portion of the **Patching** screen, which contains the source patch connectors.
- **Patch connector:** A patching point on any of the tabs. For example, an XLR connector, bus, sidechain compressor, etc.
- **Patching:** The process of routing a channel/signal from a source to a destination(s).
- **Source:** The patch connector from which a signal is routed.
- **Tab:** A 'sheet' in the **From** and **To** sections that contains a specific group of patch connectors.
- **To section:** The right portion of the **Patching** screen, which contains the destination patch connectors.

About the Patching screen

The **Patching** screen has two main areas: a function button panel towards the top of the screen and a patching area below. The function buttons provide the required patching functionality and allow I/O tab devices to be set up. The patching area provides access to all the patch connectors.

The patching area is split equally into two independent sections, called **From** and **To**, which contain the source and destination patch connectors, respectively. The patch connectors are grouped on tabs according to type. Only one tab per section will be visible at any time.

Figure 1: A typical Patching screen display

Item	Description
1	Function button panel, contains the function buttons that enable patching and device configuration (see "Patching screen function buttons" on page 47).
2	Patching area, contains all of the patch connectors on tabs.
3	**From** section, contains a number of tabs that house all of the patch connector sources (see "About the tabs in the From and To sections" on page 48).
4	**To** section, contains a number of tabs that house all of the patch connector destinations (see "About the tabs in the From and To sections" on page 48).
5	Status bar that shows the patching status of the inputs and outputs as a percentage.
6	A pictorial representation of a physical device.
7	Section titles and tab names.

>> To open the Patching screen

Do one of the following:

- At the GUI, choose **home** ▸ ***Patching.***
- Press the **routing / metering** button in the navigation zone.
- At the appropriate GUI screen, click the **src** (source) or **dest** (destination) button. The **Patching** screen will open at the appropriate tab/configuration window.

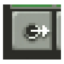
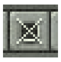
Patching screen function buttons

The function panel buttons of the **Patching** screen are described in the following table. When a button is selected, its background colour changes to a lighter shade.

Legend	*Description*
SINGLE	Lets you patch a single source to a single destination or multiple destinations. See "Single patching (SINGLE)" on page 64.
SEQ.	Lets you select multiple sources and then patch them one by one to their destinations. With this method, each source can only have one destination. See "Sequence patching (SEQ.)" on page 64.
AUTO	Lets you select a block of sources and patch them all automatically in one go, simply by selecting a single destination. Any existing patches within the destination range will be replaced by the new ones. See "Automatic patching (AUTO)" on page 64.
NONE	Clears all currently selected patch connectors from all tabs in the **From** and **To** sections.
LIST	Changes the tooltip type from standard to list to help with sequence patching. This is only available when **SEQ.** is on (see "List tooltip" on page 57).
CHECKPOINT	Sets a patching store point, or snapshot, that contains the patching status at that instant. There is only one checkpoint available, so each time **CHECKPOINT** is clicked the previous checkpoint is overwritten.
RESTORE	Reverts patching status to the last checkpoint or, if no checkpoints have been created, it will revert patching status to the power up condition. All patching done in the intervening period will be lost.
UNDO	Undoes the latest single patch, even if it was part of a multiple patching operation. Repeated clicks will undo the preceding patching operations, going back to the last checkpoint, or power up if no checkpoints have been created.
REDO	Redoes an undo. This can be repeated for each undo in the previous undo operation.
CLEAR SEL.	Clears all current selections and their patches. **Important:** **Unlike the NONE button, which merely removes the current selections (highlighted in yellow), CLEAR SEL. goes a step further by removing the patch as well. This will stop any audio that may have been going through the patched signal.**
CLEAR	Clears all patching (see "To clear all current patching" on page 65). **Important:** **Exercise great caution when using this function. Observe the warning that appears after clicking this button.**
CONFIG	Opens the **AES50 Device Configuration** window, from where you can configure the device (see "The AES50 Device Configuration window" on page 60).

What the Patching screen symbols mean

The following table gives a description of all the symbols that appear on the Patching screen tabs.

Symbol	***Description***
	During patching, this triangle appears under a tab name when its tab sheet contains a selected patch connector.
	Shown at the top of the channel patch connectors, this box aids channel identification by showing the channel's user-configured colour.
	Insert return patch connector.
	Insert send patch connector.
	Bus or channel source patch connector.
	Bus or channel destination patch connector.
	Female XLR chassis patch connector (input).
	Male XLR chassis patch connector (output).
	Jack patch connector.
	Non-functional patch connector, that is, one that cannot be patched.
	Compressor sidechain input patch connector.
	Gate input patch connector.
	DN9696 recorder patch connector.
	Set-up button, which opens the device configuration window (see "Configuring the devices" on page 58 for details).

About the tabs in the From and To sections

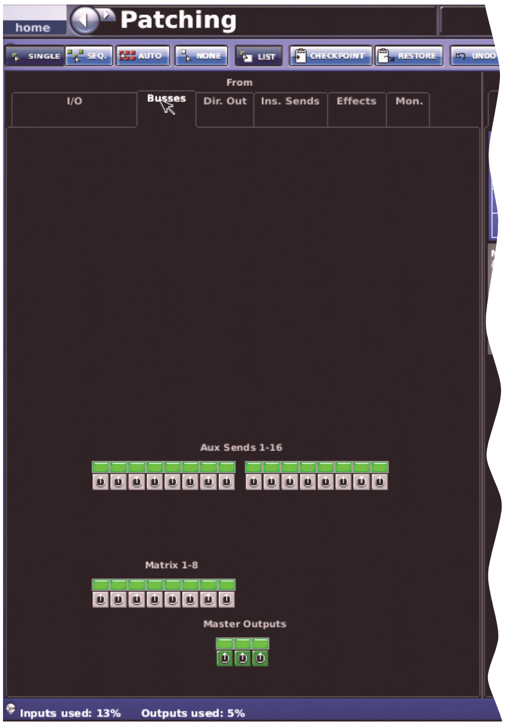
The **From** section (left side of **Patching** screen) contains all of the source patch connectors on tabs, and the **To** section (right side of **Patching** screen) contains all of the destination patch connectors on tabs.

I/O tab — From

The **I/O** tab in the **From** section shows the inputs of the devices fitted within the system (see Figure 1 "A typical Patching screen display" on page 46).

Busses tab — From

The **Busses** tab allows routing from the auxes, matrices and master outputs.



This tab can also be accessed directly via the **dest** button in the **mixes** section of the input channels and in the **direct input** section of the output channels (see “Direct input” on page 291).

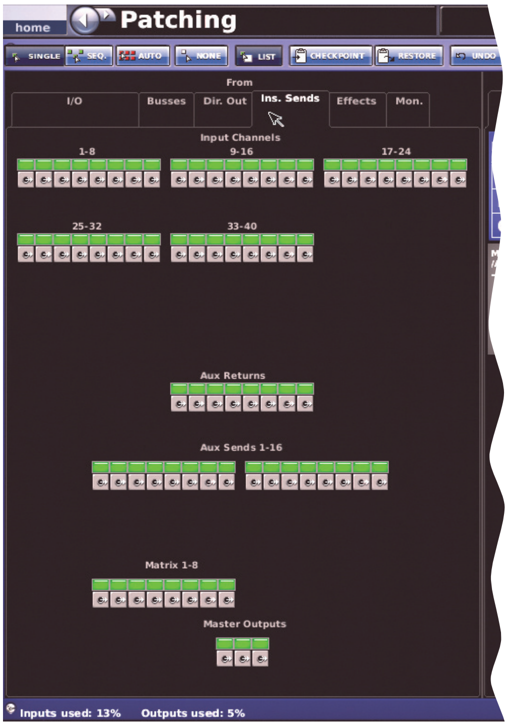
Dir. Out (Direct Out) tab — From

The **Dir. Out** (Direct Out) tab allows you to patch any of the 40 input channels internally. For example, to an effect or to provide a way out of the DL251/DL252 Audio System I/O via a line I/O unit.

This tab can also be accessed directly via the **dest** button in the **direct output** section of the input channels.

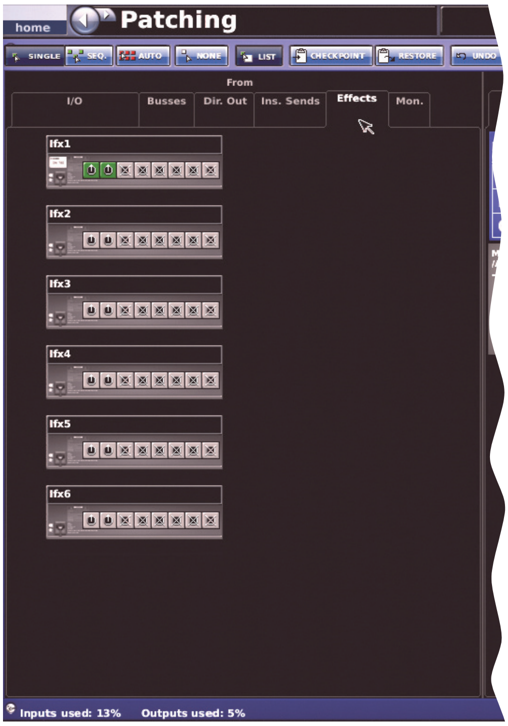
Ins. Sends (Insert Sends) tab — From

The **Ins. Sends** (Insert Sends) tab allows any of the input and output channels to be routed, primarily to an effects device.



Effects tab — From

The **Effects** tab allows patching from any of the effects.



Mon. (Monitor) tab — From

The **Mon.** (Monitor) tab allows routing of the monitor outs and external talk destination.

This tab can also be accessed via the **dest** buttons on the Monitors screen.

Mon. tab output	*Equivalent on the Monitors screen*
Monitor Out A L	**monitor A output L**
Monitor Out A R	**monitor A output R**
Monitor Out B L	**monitor B output L**
Monitor Out B R	**monitor B output R**
Headphone Out L	N/A
Headphone Out R	N/A
External Talk Destination	**external talk output**

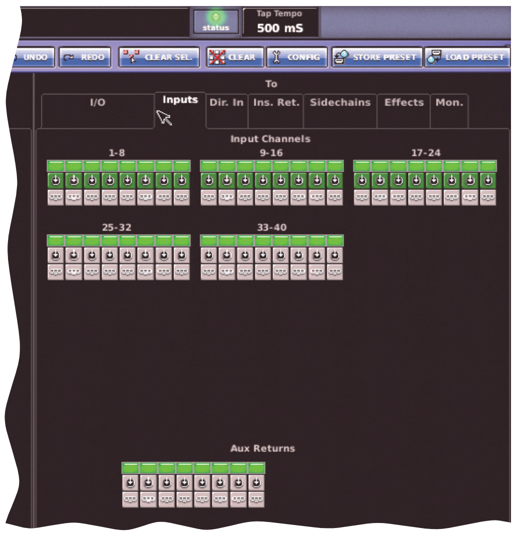
I/O tab — To

The **I/O** tab in the **To** section contains the devices fitted in the stage racks (except the mic splitters).

Inputs tab — To

The **Inputs** tab allows sources to be routed to the input channels and returns.

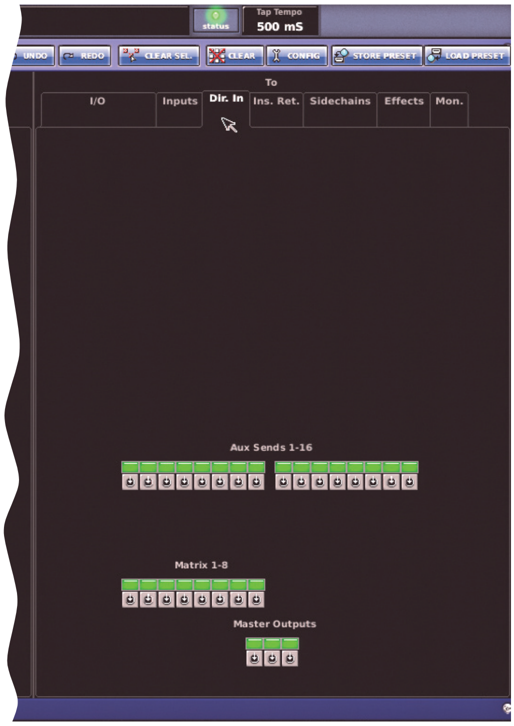
This tab is also accessed from the **src** button in the configuration section of the input channels.



__Dir. In (Direct Input) tab — To__

The **Dir. In** (Direct Input) tab allows you to patch, for example, effects to the outputs.

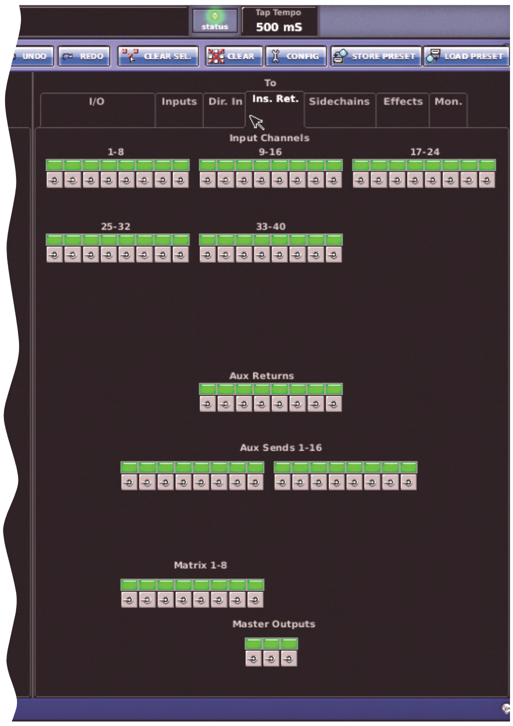
This tab is also accessed from the **src** button in the configuration section of the output channels (see "Direct input" on page 291).



__Ins. Ret. (Insert Return) tab — To__

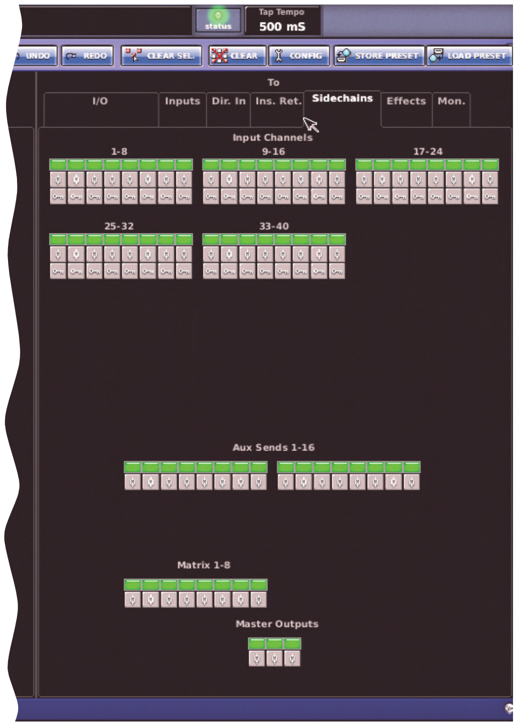
The **Ins. Ret.** (Insert Return) tab allows insert returns to be patched to any of the inputs and outputs.

This screen is also accessed directly via the **src** button in the insert section of both the input and output channels.



Sidechains tab — To

The **Sidechains** tab allows patching to the compressor and gate of the input and output sidechains.



Effects tab — To

See "Effects tab — From" on page 50.

Mon. (Monitor) tab — To

The **Mon.** (Monitor) tab allows routing to the communications and monitors.

This tab can opened using the **src** buttons on the Monitors screen.

Mon. tab input	*Equivalent on the Monitors screen*
Talk Input Source	**talk input**
Talkback Input Source	**talkback input**
Ext Input Source L	**external input L**
Ext Input Source R	**external input R**
PFL Direct Input Source	**pfl direct input**

Mon. tab input	*Equivalent on the Monitors screen*
AFL Direct Input Source L	**afl direct input left**
AFL Direct Input Source R	**afl direct input right**

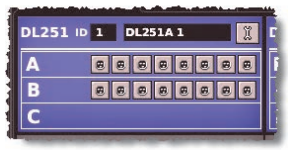
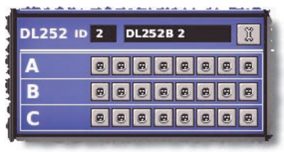
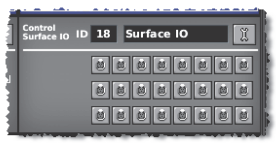
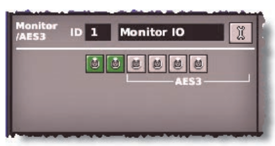
Navigating to the Patching screen

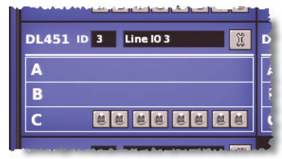
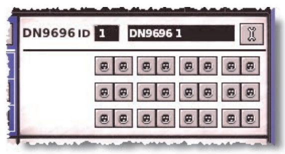
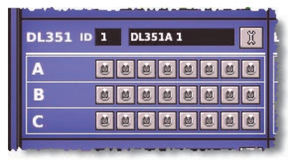
You can access the **Patching** screen from various other screens in the GUI menu, usually by clicking a **source** (source) or **dest.** (destination) button. When you click one of these buttons, not only will the **Patching** screen open, but the appropriate tab in the **From**/**To** section will be open as well.

About the devices on the I/O tabs

The following devices will, if selected (see "Setting up the I/O rack device(s)" on page 60), appear on the I/O tabs of the **From**/**To** sections of the **Patching** screen.

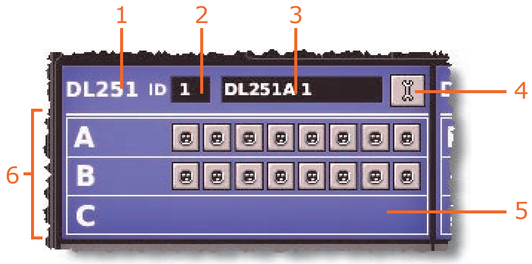
The device options in the following table are available for selection from the **device type** drop-down list in the **AES50 Device Configuration** window (see Figure 2 "The AES50 Device Configuration window" on page 60).

Device type	*Description*	*Diagram*
DL232A	One of three devices that comprise the DL232; the other two being "DL232B" and "DL321C".	
DL251A	One of three devices that comprise the DL251 Audio System I/O; the other two being "DL251B" and "DL251C".	
DL252A	One of three devices that comprise the DL252 Audio System I/O; the other two being "DL252B" and "DL252C".	
N/A	**Control Surface IO** — represents the input and output connectors on the rear panel of the PRO1.	
N/A	**Monitor /AES3** — represents the AES3 I/O connectors on the rear panel of the PRO1.	

Device type	*Description*	*Diagram*
Mic Splitter	DL431 Mic Splitter.	
Line IO	DL451 Modular I/O.	
Generic AES50	Audio-only device that is used to represent the inputs and outputs of any third party AES50 device.	
DN9696	Klark Teknik DN9696 Recorder. Use up to four of these devices (with IDs 1 to 4) to represent the four AES50 ports for up to 96 channels of recording/playback. For more information, go to our website www.klarkteknik.com.	
DL351A	One of four devices that comprise the DL351 Modular I/O. The others are "DL351B", "DL351C" and "Dl351D" (not displayed). For more information, go to our website www.midasconsoles.com.	
None-Unknown	Selects no device.	N/A

Common device features

The device images have certain common features in their layout, as highlighted in the following diagram, which shows one of the I/O devices.



Item	Description
1	Unit type.
2	Unit ID number.
3	Unit name and PRO1-assigned unit number.
4	'Spanner' button, opens the device configuration window (see "Configuring the devices" on page 58).
5	XLR patch connector, which is male or female, depending on section location.
6	Patch connector area. (The line I/O device shows the three card slots, A, B and C.)

Patching tooltips

Patching uses two types of tooltip — standard and list — to convey useful patching information about the patch connectors. A tooltip is a transitory object, in the form of a text box, that only appears while the GUI's pointer is in the proximity of a patch connector.

Standard tooltip

The standard tooltip is the default type that appears during all patching operations. The following diagram shows, typically, the type of information provided by a standard tooltip.

Item	Description
1	Patch connector information panel, contains information on the selected patch connector, such as, name, ID, device name, device ID etc. Depending on the device type, a signal level meter appears if the channel is passing audio.
2	Routing information panel, contains patching information on the selected patch connector. (If this panel is blank, the patch connector is not patched.)
3	The patch connector that the tooltip belongs to.

List tooltip

If you are carrying out a sequence operation, you can use the list tooltip to help in selecting the destinations in the **To** section. This tooltip, which has a distinctive translucent orange background, displays a list of the sources still to be patched. The list is in order of selection, with the first in the queue being at the bottom. You can only use the list tooltip for sequence operations.

Item	Description
1	ID of the patch connector currently under the GUI pointer.
2	List of selected sources still to be patched. Contains channel and device ID information.
3	The source patch connector currently waiting to be patched. Once patched, this will disappear from the list and the one immediately above will become the next available.

>> To select the list tooltip

Press **LIST.** (Pressing **LIST** again will change the tooltip back to the standard type.)

About the patching procedure

Although patching can be thought of as routing/rerouting the console's incoming, internal and outgoing signals, in the context of the **Patching** screen, patching also encompasses the setting up and configuration of the devices. The patching procedure is initially carried out after system installation (see Chapter 4 "Setting Up The System" on page 25 of the operator manual) and comprises:

- **Device configuration:** Configure the devices by adjusting their parameters as necessary (see below).
- **Setting up the I/O rack device(s):** Set up the system devices, such as line I/O and mic splitter, on the I/O tabs in the **From** and **To** sections of the **Patching** screen (see "Setting up the I/O rack device(s)" on page 60).
- **Patching:** Carry out all of the required routing, for example, mics to input channels (see "How to patch" on page 62).

Configuring the devices

You have the option to configure the devices via the **Patching** screen. Parameters, such as gain and +48V phantom voltage, can be set via a device-specific configuration window.

These configuration settings can be independent of channel data, as (until patched) they only control the physical unit. If a device is subsequently patched to one or more channels, the channel(s) control the device, and vice-versa.

The device configuration window also allows control of audio parameters when the device is used as a direct connection to another device, for example, FOH to stage via a digital snake, instead of through the DSP. In this case the settings are also saved in the show file and can be automated, even though the signals are not routed through the control centre DSP.

As the mic splitter control conforms to the **DL431 Mic Splitter Inputs** options (**Use A Inputs** or **Use B Inputs**) in the **Configuration Preferences** section of the **Preferences** screen, it is not possible to control both the A and B settings on the mic splitter from a single console.

Device configuration procedure

The procedure for configuring a device comprises:

- Opening the configuration window of the device.
- Selecting one of the device's cards/channel ranges and configuring the available parameters.
- Repeating for the other cards/channel ranges of the device.
- Repeating for the other devices.
- Closing the device's configuration window.

Note: *The parameters available for a device are dependent on its type.*

About the configuration window

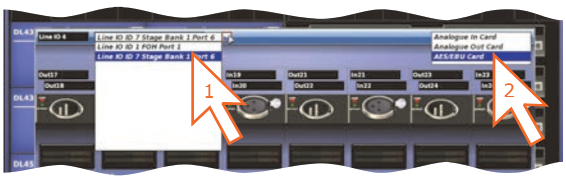
The configuration window, which has a similar format for each device, comprises eight channel panels and drop-down lists for channel range/card selection. A typical Line I/O configuration window is used in the following diagram to show the elements that are common to each device.

Item	*Description*
1	Device ID field, contains the device type and number.
2	Device drop-down list, for device selection. (Diagram shows the drop-down list selected.)
3	Channel range/card selection list.
4	**OK** button, saves the changes and closes the configuration window.
5	Channel panel, contains device-specific controls and graphics.

>> To open the configuration window of a device

Click the device's spanner button  (upper right corner of device).

>> To set up/change the configuration of a device

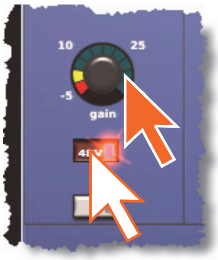


1 In the device configuration window, choose the device from the drop-down list. For example, the FOH line I/O device (ID11) connected to port 2.

2 From the drop-down list at the upper-right corner of the window, choose the card/channel. For example, the "AES/EBU Card".

3 In a channel, configure its parameters. For example, in channel "In1" adjust the gain and switch on the +48V phantom voltage (shown right).

Repeat for the other channels in the card as necessary.



4 Repeat step 2 and step 3 for the other cards as necessary.

5 If necessary repeat step 1 to step 4 for any other devices of the same type.

6 Click **OK.**

Setting up the I/O rack device(s)

You can add, remove and set up the devices, such as line I/Os, mic splitters, DN9696s etc., that are connected to the Stage I/O and FOH I/O racks. This is done via the **AES50 Device Configuration** window (see Figure 2 "The AES50 Device Configuration window" on page 60), which is opened by clicking **CONFIG.** Here, you can set up the device's ID and also the type of cards (modules) fitted in the physical unit. Some of the device fields may be blank, as they are dependent on the device type.

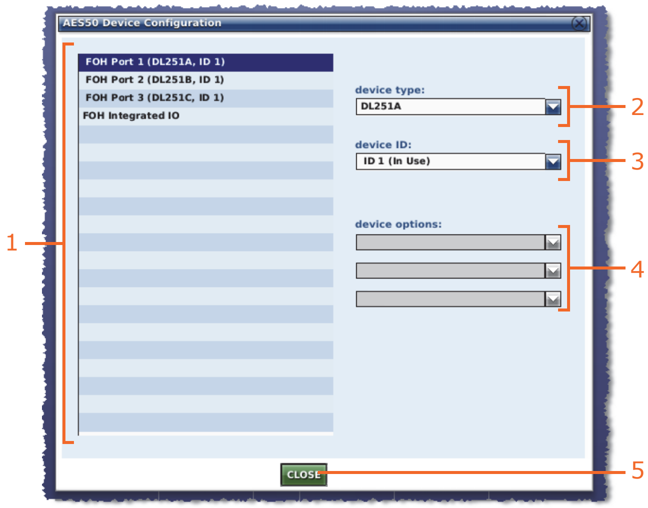


*Figure 2: The **AES50 Device Configuration** window*

Item	Description
1	List of Stage and FOH ports with current device assignments.
2	**device type:** drop-down list, contains a list of the available devices to choose from.
3	**device ID:** drop-down list, contains a full list of IDs for the selected device type. Those already in use will be prefixed with the text "(In use)".
4	**device options:** drop-down list(s) on modular I/O units only (for example, the DL351) from which you can select the card that is actually fitted in the physical unit. The positions of the drop-down lists are relative to the card positions in the physical unit.
5	**OK** button, closes the **AES50 Device Configuration** window.

Device set-up procedure

The device set-up procedure comprises:

- Selecting the port (Stage or FOH) you wish to allocate the device to.
- Selecting the device type.
- Selecting an ID for the device.
- Selecting the options (if any) for the device.

>> To add a device or change its set up

1. Click **CONFIG.**
2. In the **AES50 Device Configuration** window, click the port you want to allocate the device to; these are listed in the far left of the window. For example, choose "FOH Port 3 (unused)". The text in the **device type:** field will change accordingly. Ports that don't have a device allocated to them it will have the text "unused" inside the brackets after their name.
3. In the **device type:** drop-down list, click the type of device. For example, choose "Line IO".
4. In the **device ID:** drop-down list, click the ID you want for the device. For example, choose "ID3".
5. In the **device options:** drop-down list(s), choose the type of each card fitted in the physical unit. For example, choose "Analogue 8 Input".

   Repeat for any other cards.
6. Click **OK.**

>> To remove a device

Select the device from the list in the left of the **AES50 Device Configuration** window. Then, choose "None-Unknown" in the In the **device type:** drop-down list. For more information, see "To add a device or change its set up" above.

How to patch

Patching, basically, involves selecting the source patching connectors in the **From** section of the **Patching** screen and then selecting their destination(s) in the **To** section. You can select patches singly, or in multiples by using the sequence and automatic operations.

About the patch connectors

Each patch connector has three possible states, as indicated by its fill colour. The following table shows what each state signifies (the examples show XLR connectors, although it applies to any type of patch connector).

Symbol	***Description***
	Patch connector is neither selected or patched.
	Patch connector is selected, but can be in either a patched or unpatched condition.
	Patch connector is patched, but not selected.

Working with patch connectors

You can select patch connectors one at a time by clicking on them, or you can select them in blocks by using a drag operation. All of the patch connectors in both the **From** or **To** sections are on tabs so, before you can select a patch connector, its tab must be open.

>> To open a tab in the From or To sections

Click the tab title. For example, **Ins. Sends** (insert sends).

>> To select a single patch connector

Click the patch connector. The effects of clicking a patch connector are shown in the following table.

Table 2: Effects of clicking a patch connector

Click	***To do this in the From section***	***To do this in the To section***
	Select the patch the connector ().	Will do one of the following (provided a source patch connector(s) has been selected in the **From** section): • Select the patch connector () during a single patching operation. • Patch the patch connector () during either a sequence or an automatic patching operation. Otherwise, this has no effect.

Click	To do this in the From section	To do this in the To section
	Deselect the patch connector, which then reverts to its previous state (patched or unpatched).	Remove the patch ( ).
	Select the patch connector ( ) and all the ones it is patched to in the **To** section. (A green triangle under a tab title shows that its tab contains one or more selected patch connectors.)	Remove the patch ( ).

*To quickly check the destinations of a source patch connector, click it. This will select it and all of its destinations. A green triangle will appear under the name of any tab in the **To** section that contains a destination(s).*

>> To select a block of patch connectors in the From section

Use a drag operation to create a bounding box around the desired connections (typically shown right).

This procedure can only be done during sequence and multi-patching operations (initiated by the **SEQ.** and **AUTO** buttons, respectively).

>> To deselect all selected patch connectors

Click **NONE**.

>> To remove a single patch

In the **To** section, click the patch connector from which you want to remove the patch.

>> To remove all the patches of a single source

1 Make sure that no patch connectors are selected. If necessary, click **NONE.**

2 In the **From** section, click the source patch connector from which you want to remove all of the patches. (This will select the source patch connector and also all of its destinations.)

3 Click **CLEAR SEL**.

>> To remove the patches from all selected patch connectors

Click **CLEAR SEL**.

>> To clear a block of patch connectors

1 Click **NONE.**

2 In the **From** section, select the patch connectors you want to unpatch.

3 Click **CLEAR SEL.**

4 Click **NONE.**

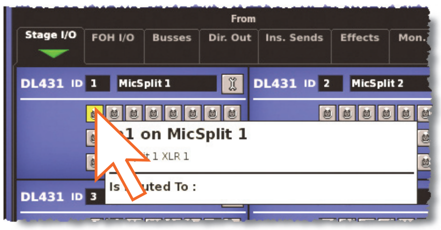
Single patching (SINGLE)

The **SINGLE** function button allows you to patch a single source to a single destination or multiple destinations.

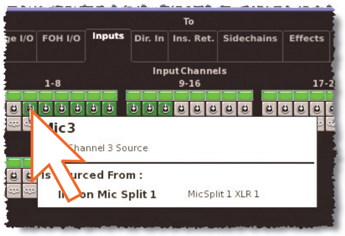
>> To patch a single source to a single destination

The following example shows you how to patch an output from a mic splitter to an input channel.

1. Click **SINGLE.**
2. In the **From** section, click the source patch connector. For example, choose the first patch connector of a mic splitter. Its background will change to yellow and a green triangle will appear under the tab title.



3. Click the destination patch connector. For example, in the **Inputs** tab of the **To** section, choose input channel 3 (Mic3). It will now be patched to the source. If the new patch is carrying a signal, this audio may be heard, depending on the settings of the PRO1 Control Centre.



4. Click the source patch connector (**From** section) again to complete the patch.

   This can also be done by starting another single patch operation or by selecting another destination patch connector in the **To** section.

Note: Note: *You can also carry out single patching operations using the* ***CLEAR SEL.*** *and* ***AUTO*** *functions.*

>> To patch a single source to multiple destinations

1. Patch the source patch connector to its first destination (see "To patch a single source to a single destination" on page 64).
2. In the **To** section, select the other destinations.

Sequence patching (SEQ.)

If you need to do a number of patches, and each has only a single destination, you can use the sequence function by pressing **SEQ.** All of the source patch connectors are selected in the **From** section before patching them, one by one, in the **To** section. This saves you having to go back to the **From** section for the start of each patch. You can change the tooltip to the list type by clicking the **LIST** button (see "Patching screen function buttons" on page 47) to help you.

Automatic patching (AUTO)

You can patch a block of source patch connectors, just by selecting a single destination. This is called "automatic patching". When using automatic patching, note the following:

- Sources are selected in blocks (see "To select a block of patch connectors in the From section" on page 63).
- You can only select one block of sources.

- Destinations are restricted to a single type.
- 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. If there are any sources left over, automatic patching pauses. You can then patch these by selecting another destination.

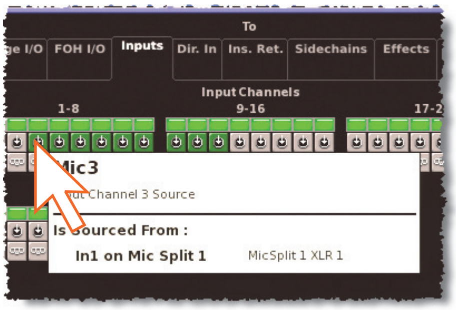
>> To automatically patch a block of source channels

1 Click **AUTO.**

2 In the **From** section, select the source patch connectors (see "To select a block of patch connectors in the From section" on page 63).

3 In the **To** section, choose the destination patch connector that will form the start of the patched range. For example, input channel 3 (Mic3).

4 Click the destination patch connector. The sources will be patch in numerical sequence and in ascending order from here onwards.



Clearing all current patching

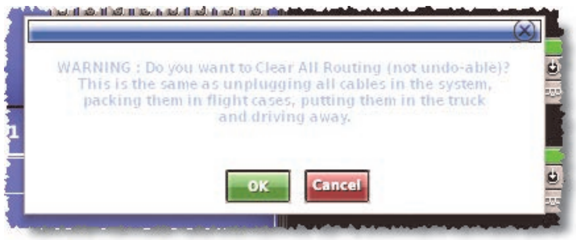
! The CLEAR function button clears all current patching, and must be used with great caution. To alert you of the drastic nature of using this button, a WARNING appears.

>> To clear all current patching

1 Click **CLEAR.** The WARNING window (shown right) will appear.

2 Heed the warning and do one of the following:

- If you want to clear all current patching, click **OK.**
- To cancel the clear operation and close the WARNING, click **CANCEL.**



Chapter 9: Basic Operation

This chapter is intended to familiarise you with the PRO1 Control Centre by showing you how to carry out some basic operations in order to get some audio out of it.

Setting a mic amplifier's input gain

The PRO1 Control Centre has two input gains per channel, one is the remote gain for the analogue mic pre (stage box gain) and the other is the digital trim (console gain). In its default state, the stage box gain is in the channel strip and the console gain is in each input fast strip. However, you can swap these sections over (by using the **gain swap**/**[SWAP]** button) to give you a more global control of the stage box gain.

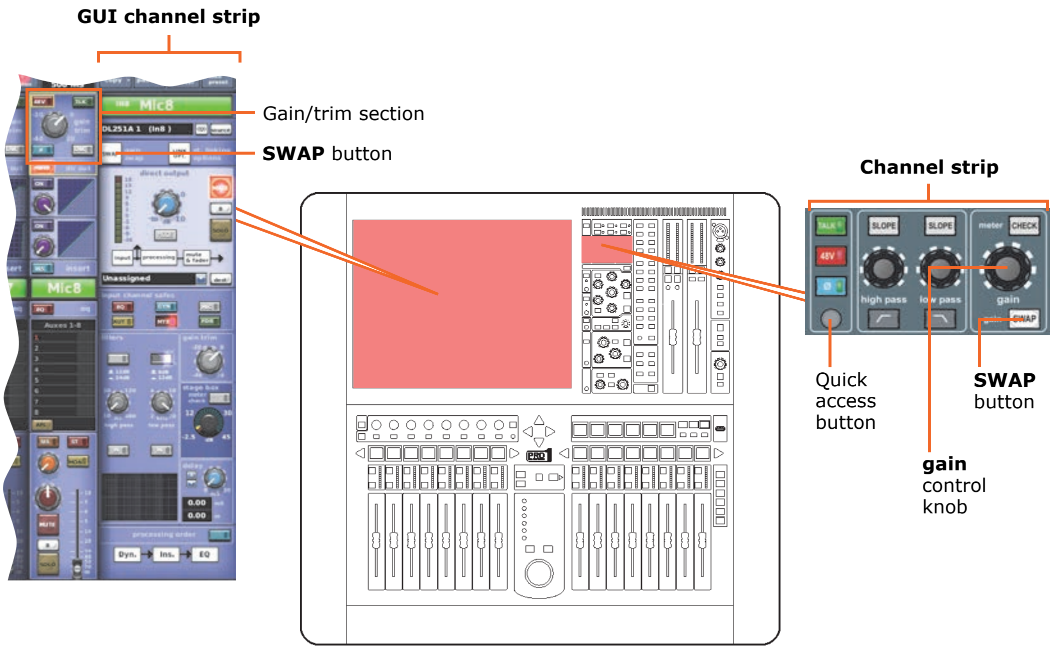


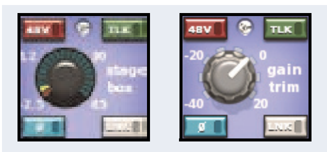
Figure 3: Gain and filters sections of the input strip

Note: *The* ***gain trim*** *control knobs in each input fast strip will adjust whatever has been 'swapped' to their respective strips. The* ***stage box*** *control knob in the input channel strip always controls the alternative 'swap' to the ones shown in the GUI input fast strips.*

>> To set the gain of the stage box/console

1 In the **gain trim** section of an input fast strip, press the quick access button. This selects the input channel and assigns its configuration processing area to the GUI channel strip (shown above), which contains the **SWAP** (gain swap) button.

2 Press the **SWAP** button. This swaps console digital trim to stage box input gain (and vice versa). The diagram right shows the two types of gain that can appear in the input gain/trim section at the top of each GUI input fast strip.



3. Adjust the **gain trim** control knob to set the stage box input gain. Range is shown on the GUI.

   Adjust level to suit the Midas pre-amp characteristic; a suitable level could be one that only just illuminates the yellow LEDs. Drive the mic amps for that 'Midas colouration' — feel free to overdrive if you want.

4. After you have achieved the required gain state, press the **SWAP** button again to swap back to console digital trim.
5. Adjust the **gain trim** control knob to set the console digital trim (gives +20dB to -40dB continuous trim) for preferred gain structure.
6. Set analogue remotes for initial set-up, then adjust digital trim for showtime.

Setting the high and low pass filters

When switched in, the high and low pass filters have two settings each, selectable via their respective **SLOPE** button. These filters can also be set via the GUI.

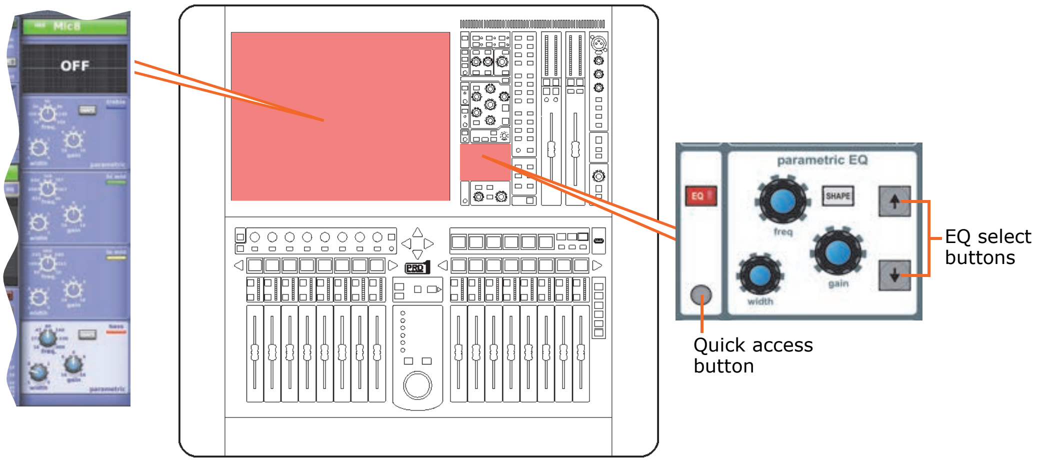
>> To set a filter

1. In the **gain trim** section of an input fast strip (see Figure 3, "Gain and filters sections of the input strip," on page 67), press the quick access button. This selects the input channel and assigns its configuration processing area to the GUI channel strip (shown above), which contains the **filters** section.
2. In the **filters** section of the input channel strip, press the filter select button (**high pass** or **low pass**) to switch the filter in.
3. If necessary press the filter's **SLOPE** button to set its slope (dB); its status is shown on the GUI. For the high pass filter, in = 24dB and out = 12dB, and for the low pass filter, in = 12dB and out = 6dB.
4. Adjust the **high pass**/**low pass** control knob to set filter frequency (Hz). The ranges are 10Hz to 400Hz for the high pass filter and 2kHz to 40kHz for the low pass filter.

Important:
Stage box hi pass — the remote stage box contains a 12dB/Oct 30Hz filter. It is recommended that this is used at all times for optimum A/D performance. However, it may be bypassed if extremely low frequency performance is required, for example, when testing the system.

Input equalisation (E zone)

Use EQ to equalise the input signal via the treble, hi-mid, lo-mid and bass filters, which are in the input channel strip's E zone. Treble and bass each have a parametric filter option with three specific shelving modes. Visual feedback for EQ is via GUI screen only, which also has a graphical representation of the filter.



>> To EQ the input signal

1 In the desired input fast strip, press **EQ** to switch the EQ in. This also selects the channel and assigns the EQ processing area to the GUI channel strip.

2 Do one of the following to select the EQ band:

- In the input fast strip, press the quick access button of the desired band.
- In the E zone, press the **treble**/**bass** up/down arrow buttons until the button/LED of the desired band is illuminated.

On the GUI the currently selected EQ band will have a light coloured background. For example, **treble** in the above diagram.

3 In the E zone, adjust the **freq**, **width** and **gain** control knobs to apply EQ as desired.

4 If you have selected **treble** or **bass,** press **SHAPE** (E zone) to cycle through the different shelving modes so that you can audition them. These are the 'minimum harmonic disruption' types, which are only available for **treble** (bright, classic and soft) and **bass** (deep, classic and warm). For example, the **bright** shelving mode of the **treble** band (shown right) as displayed on the GUI.

Alternatively, you can click the desired **SHAPE** button in the GUI channel strip.

Note: **bright** *and* **deep** *use psychoacoustic phenomena to generate steep slopes that sound natural. These filters are called "minimum harmonic disruption filters".*

Input dynamics processing (D zone)

This section deals with assigning compressor and gate dynamics processors using the controls in the input channel strip's D zone. There are four compressors available — corrective, adaptive, creative and vintage — each with the option of hard knee, medium knee and soft knee (see Appendix A "Application Notes" on page 299). Visual feedback for both compressor and gate is provided by meters in each input fast strip, the dashboard screen and just above/below the graph, which gives a representation of the compressor/gate action.

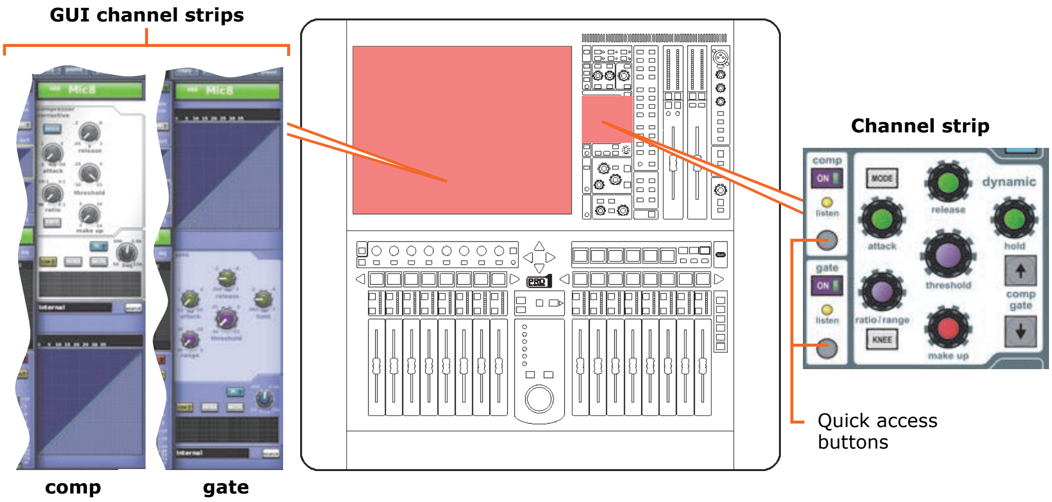


Figure 4: Compressor and gate channel strips

>> To set up a compressor/limiter

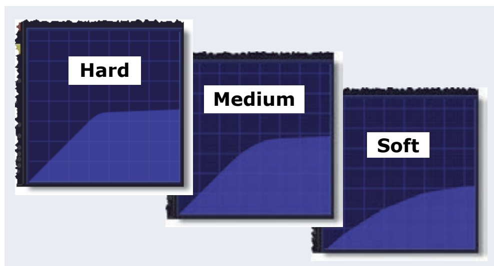
1 Select the desired input channel.

2 In the channel strip, press the quick access button in the **comp** section (see Figure 4 "Compressor and gate channel strips" above) to select the input channel's compressor processing area.

3 Press **ON** in the **comp** section to switch the compressor in.

4 Operate the **attack, ratio/range** (ratio), **release, threshold** and **make up** control knobs (**hold** has no effect) to apply processing. See "Compressor" on page 260. You can set up a limiter by using a high threshold and a steep ratio (greater than 5:1).

5 Press **KNEE** to audition the different algorithms (hard knee, medium knee and soft knee). The effects on the signal are shown right.



6 Press **MODE** to try the different compressor types (**Corrective, Adaptive, Creative** and **Vintage**). For example, Creative (shown right).

>> To set up a gate

1 Select the desired input channel.

2 In the channel strip, press the quick access button in the **gate** section (see Figure 4 "Compressor and gate channel strips" on page 70) to select the input channel's compressor processing area.

3 Press **ON** in the **gate** section to switch the gate in.

4 Operate the **attack, ratio/range** (range), **release, threshold** and **hold** control knobs (**make up** has no effect) to apply processing. See "Gate" on page 264.

Output processing

All outputs — except for the returns — have a six-band PEQ with shelving modes on bands 1, 2 and 6, and have the option of using a GEQ (accessed via a **GEQ** button in GUI channel strip). The returns have a similar EQ to that of the inputs channels.

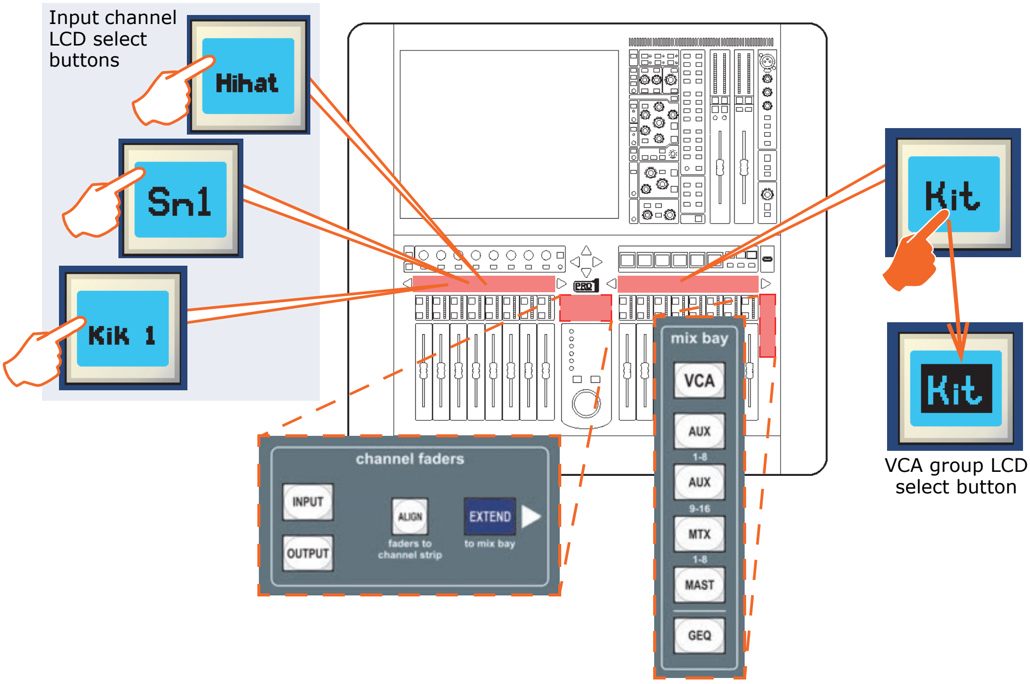
The outputs (except returns) have five compressor modes, which include all of the ones on the input channels, but with the addition of a shimmer mode. The returns have the same compressor modes as those of the input channels.

Using VCA/POPulation groups

VCA and POPulation groups allow simultaneous adjustment of a number of channels. Being instantly recognisable, they provide a quick method of bringing particular channels to the control surface and save you having to remember their name/number.

You can choose channel group associations and also configure the colour and legend of each group's LCD select button, which is used for group member assignment and recall.

VCA groups include fader, solo and mute control, whereas POPulation groups merely bring a group of input channels to a desired area of the control surface for viewing or adjustment.



Assigning VCA group members

>> To assign channels to a VCA/POPulation group

1 If necessary, assign the VCAs to the mix bay by pressing **VCA** in the **mix bay** section.

2 In the mix bay, press and hold down the LCD select button of the desired VCA. For example, a VCA group named "Kit". The button will start flashing when you are in group member selection mode and the inputs will jump to programme mode.

3 While still holding down the LCD select button of the VCA, press the LCD select buttons (just above the channel faders) of the channels that you want as group members. For example, "Kik 1", "Hihat" and "Sn1". If necessary press **INPUT** (**channel faders** section) to assign the input channels to the channel faders and/ or scroll to a new channel bank.

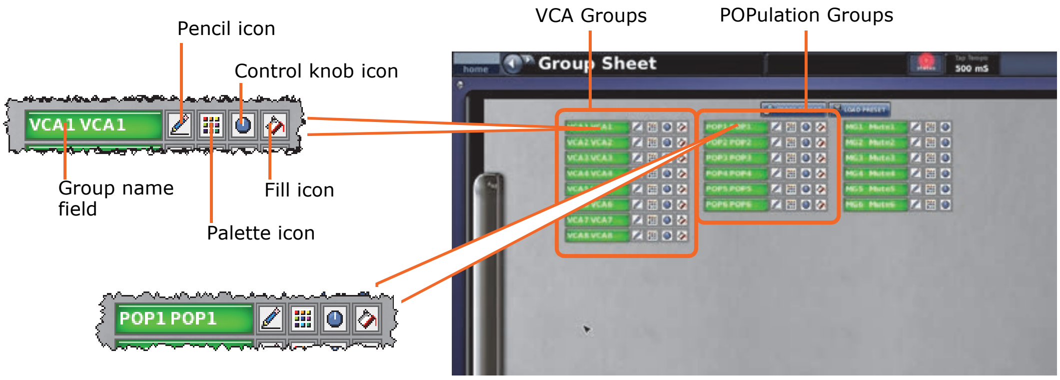
4 Release the group's LCD select button. The group now contains the input channels you have just chosen and the group will be selected.

5 Press the VCA's LCD select button again to exit the group.

*To quickly see which channels are in a particular VCA group, press its **SOLO** button on/off. Monitor this action on the **Console Overview** screen. Only the **SOLO** buttons of channels that are group members will be affected.*

Configuring VCA/POPulation groups

The default name and associated colour of a group, which appear on its LCD select button and on the GUI, can be configured to suit your own preference. You can also globally change the colour of the group members to match the group colour. Configuration is done via the **Group Sheet** screen (shown below).



Note: *Clicking the control knob icon*  *opens the **VCA Groups** screen (a submenu of the **Control Groups** option), which provides group management control.*

>> To access the Group Sheet screen

Do one of the following:

- At the GUI, choose **home** ▸ ***Control Groups*** ▸ ***Group Sheet.***
- In the navigation zone, press the **vcas** screen access button.

>> To set up the name of a VCA/POPulation group

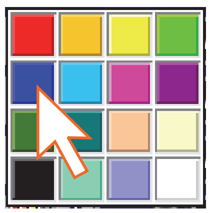
Do one of the following:

- **Choose from a list of pre-configured names** Click the pencil icon  of the group. Then, choose the name from the drop-down list. For example, choose "E Gtr".
- **Type in a new name** Click within the name field of the group and then follow "To enter/edit text via the keyboard" on page 42.

>> To set up the colour of a VCA/POPulation group

1 Click the palette icon  of the group.

2 In the palette (shown right), click the colour you want. For example, choose blue.



>> To change the colour of all of a group's members to match that of its VCA/POPulation group

Click the fill icon of the group. For example, if the colour of the VCA/POPulation is blue all of its group members will now be blue.

Setting up a mix

The PRO1 has 24 configurable mix buses, each of which can be aux mixes, subgroups or mix minus. The aux mixes can also be set up as stereo pairs (restricted to like-coloured buses) or mono. To keep the control surface manageable, access to these mixes is confined to the **mix** section in the channel processing area, which can be scrolled up and down on a 'virtual' surface.

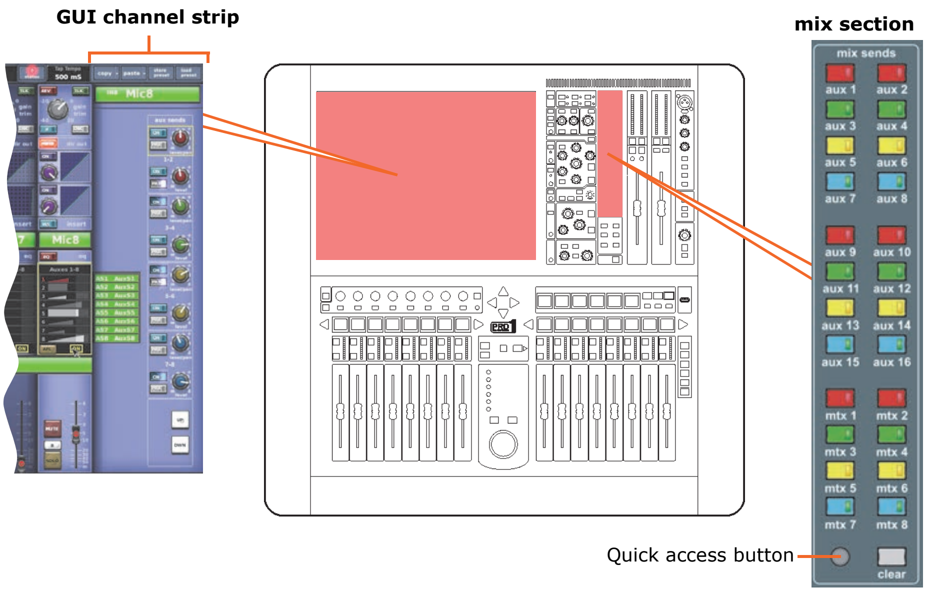


Figure 5: Mix section

On the GUI, the bank of mix buses in each input fast strip (see Figure 6 below) are colour matched, but also show mix bus status information.

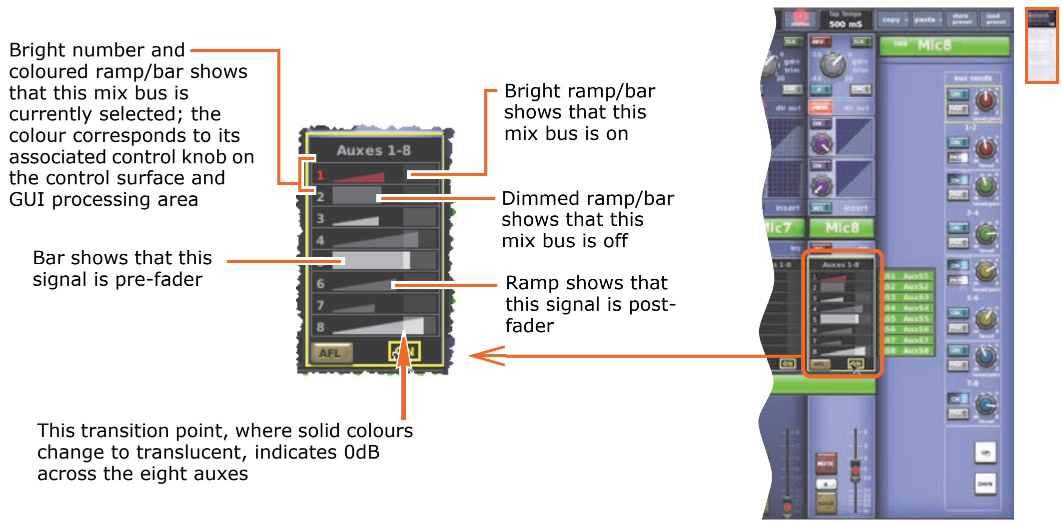


Figure 6: Typical mix section in a GUI input fast strip

>> To select the mix bus mode

1 In the mix bay, select the desired aux channel.

2 In the channel strip, press the quick access button to select the aux's configuration processing area.

3 In the GUI channel strip, click **MODE** repeatedly to cycle through the mix modes (**mix, group** and **mix minus**) to choose the one you want. The currently selected mix mode is shown to the right of the **MODE** button.

group mode is fader only with no pre-fader. In **mix minus** mode all buses are initially routed — you have to switch a bus routing switch on to take it out of the mix. For information about stereo **mix** mode, see "Linking two mixes" on page 76.

>> To set up a mono aux mix

1 Making sure that the mix bus is not linked (to ensure mix is mono), select the bus mode as **mix** (see "To select the mix bus mode" above).

2 In the channel fader bay, select your desired input channel.

3 'Flip' your desired mix send to the channel faders by pressing the appropriate aux button in the **mix sends** section.

4 Use the channel faders and the assignable controls to create your desired mix. For more information, see the PRO1 Quick Start Guide.

Mix bus routing

You can route an aux or matrix (or even master output) to an effect or output. This is a GUI-only operation, which is done via the GUI channel strip or **Patching** screen (see Chapter 8 "Patching" on page 45).

>> To route an aux or matrix to an effect or output

Do one of the following:

- In the GUI channel strip, select the required mix bus destination from the drop-down list.

- In the GUI channel strip, click **dest** (shown right). This will open the **Patching** screen and the appropriate tab.

- In the **Patching** screen, route the aux/matrix. For information on patching, see Chapter 8 "Patching" on page 45.

Linking two mixes

You can link two mixes together, which is a GUI only feature. Pairs can only be created from adjacent mix buses of the same colour, for example, you can link auxes 3 and 4 together, but not 2 and 3.

>> To route an aux or matrix to an effect or output

To link a pair of mix buses, click the **LINK** button in the channel strip of either mix bus. For example, click **LINK** of aux 15 to link it to aux 16 (shown right).

Setting the linked parameter options

The shared parameters default to the user-configurable global default link settings, which are set via the GUI menu (choose **home** ▸ ***Preferences*** ▸ ***General*** and click the **Linking** tab). However, you can override these for the pair via the **Stereo Linking Options** window (shown right), which is opened by clicking the **st. linking options** button (GUI channel strip).

In stereo mix mode the top control knob becomes pan adjust and the bottom one adjusts level. When creating a stereo mix, you can use either the odd or even output to link the two channels, but the mode of the odd channel is used on both.

For details of the parameters that are stereo linked, see Appendix J "Parameters Affected By Stereo Linking" on page 467.

Setting up the effects rack

You can set up the effects rack in the GUI menu's **Effects** screen to contain any of the available effects listed in the **Change Device Type** window (see Chapter 16 "Internal Effects" on page 125) in any of the available rack positions. The diagram right shows the rack populated with all available effects.

The **assignable controls** panel (on control surface and GUI) lets you control the parameters of the selected effect.

>> To add an effect to the effects rack

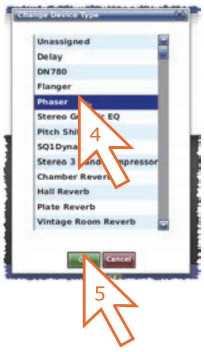
1 At the GUI screen, choose **home** ▸ ***Rack Units*** ▸ ***Effects.*** Alternatively, press the **effects / graphics** screen access button in the navigation zone.

2 Choose the rack position and click within it.

3 In the effect's window, click **Change Device Type.**

4 In the **Change Device Type** window, select the device type. For example, "Phaser".

5 Click **OK.**



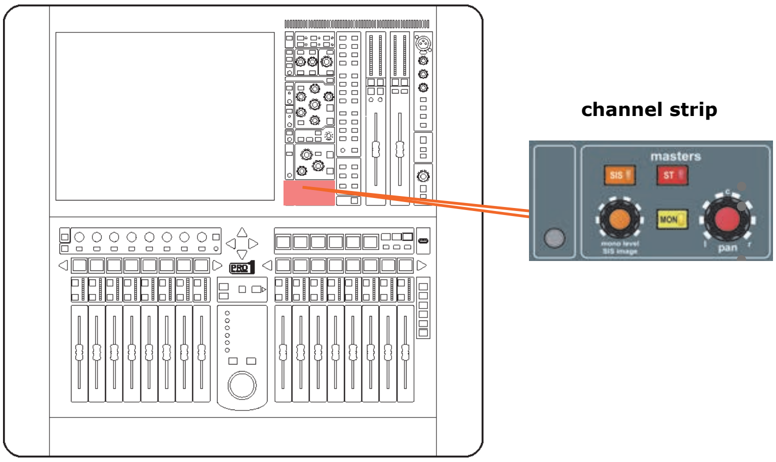
6 Change the parameters of the new effect device as necessary. For example, adjust control knobs, press buttons etc. You can even change the effect's name by editing its name field (upper-left corner of effect). You can do this via the GUI or from the I zone using the output bay GUI (see "PRO1 control surface" on page 18).

7 Click **OK** to exit. The new effect will appear in the effects rack.

You can now patch the new effect, which will be on the **Effects** tabs of both the **From** and **To** sections of the **Patching** screen. For information on how to patch, see Chapter 5 "Patching" on page 23.

Simple routing to master stereo outputs

The following shows you how to obtain audio.



>> To obtain audio

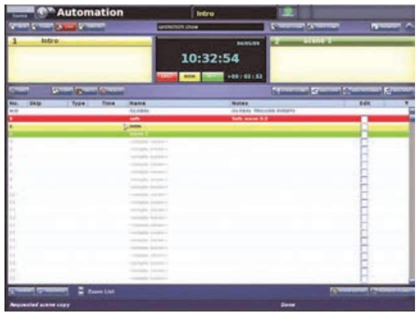
1 Make sure nothing is muted and the master faders are up.

2 In the **masters** section of the channel strip, press **ST** (stereo). You should have audio.

Automation

PRO1 automation is managed from the **Automation** screen of the GUI menu, with support from the **automation** section of the control surface.

The **Automation** screen (a typical example is shown right) has the following functions and features:



- **Show management** — see "Managing the shows" on page 79.
- **Scene management** — see "Managing the scenes" on page 81.
- **Event management** — see "Additional control — managing events" on page 82.
- **Cue list** — shows the scenes and point scenes in performance order. The cue list includes information such as scene number, name, notes etc. You can expand and unexpand the point scenes and 'zoom' the list on/out. It also lets you reorder the scenes.
- **Scope** — the **Recall Scope** and **Store Scope** buttons open the scope screens, from which you can select the automated controls that you want to leave in/out of a scene when it is stored or recalled.
- **Rehearsal** — the **Rehearsal** button lets you carry out a rehearsal, which will 'skip' (leave out) any scenes that you choose.

***Note:** With no show loaded, the **Automation** screen will be blank.*

>> To open the Automation screen

Do one of the following:

- At the GUI, choose **home** ▸ ***Automation*** ▸ ***Automation.***
- In the navigation zone, press the **automation / filing** screen access button.

Managing the shows

The four buttons (**NEW**, **LOAD**, **SAVE** and **SAVE AS**) towards the top of **Automation** screen let you create a new show, load an existing show, update the current show or create a new show using the current settings.

Important:
We recommend that you save your show settings regularly (see "Saving a show versus storing a scene" on page 36). The PRO1 will indicate that there are show settings to be saved by changing the background colour of the SAVE button to red (shown right).

The eye icon in the **Automation** screen (just under the **ADD MIDI** button) opens the **Show** window. This window contains a number of filter options, such as empty scenes, MIDI events etc., that you can choose to exclude from your show.

>> To create a new show

1 Click **NEW.**

2 At the **Enter new show name:** message window, type your chosen name for the new show.

3 Click **OK.** You can now create and manage the scenes for your new show.

>> To save a show or create a new one from the current settings

Do one of the following:

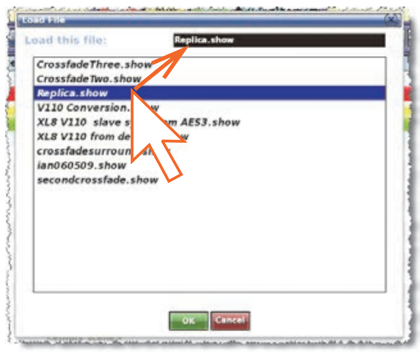
- To update the current show with the latest settings, click **SAVE.**
- To create a new show using the current show settings, click **SAVE AS.** Then, in the **Save File** window (shown right), type in the name of the new show in the **Save this file as:** name field. Click **OK** to save the new show and close the window.
- To overwrite an existing show, click **SAVE AS.** Then, in the **Save File** window (shown right), click the show you want to overwrite to select it, click the **Overwrite existing?** box to select it this option and then click **OK.**

>> To load a show

1 Click **LOAD.**

2 In the **Load File** window, click the show file you want to load (shown right). For example, "Replica.show". The file name will appear in the **Load this file:** name field.

The **Load File** window will contain a list of all the shows currently loaded. If the one you want is not there, import it (see "To load (import) a show file from a USB memory stick" on page 89).



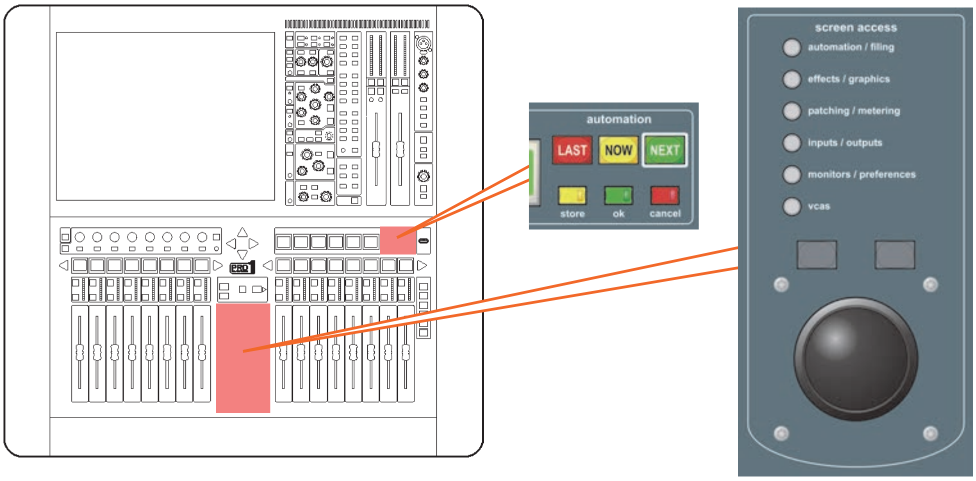
3 Click **OK.** The file will start loading and the window will close.

When the file has finished loading, its name will appear in the show file name field (to the right of the **SAVE AS** button).

Managing the scenes

Up to three scenes in the cue list will be highlighted to indicate your position in the show, as follows:

- **LAST** (red) — the scene immediately before the most recently loaded one.
- **NOW** (yellow) — the most recently loaded scene.
- **NEXT** (green) — the next available 'non-empty' scene.



The **automation** section in the output bay (see "PRO1 control surface" on page 18) supports the **Automation** screen by providing controls for scene navigation, selection and management. The jogwheel is a unique automation controls, whereas the other buttons are replicated on the GUI.

>> To recall a scene

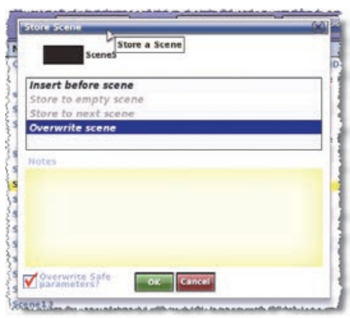
Important:
When recalling a new scene, make sure monitor output levels are low, as the new scene's settings may produce higher audio output levels than the one it is replacing. Also, recalling a scene clears any unsaved adjustments made to the previous scene.

Press **last**, **now** or **next** as desired.

>> To create a new scene using the current settings

1 Do one of the following:
   - In the automation section, press **store.**
   - In **Automation** screen on the GUI, click **STORE SCENE.**

2 In the **Store Scene** window, type in the scene name if necessary.



3 In the **Notes** panel, type in any scene notes as desired.

4 Do one of the following:
   - Click "Insert before scene" to put the new scene in between the one currently highlighted in yellow and the scene immediately before it.
   - Click "Store to empty scene" to put the new scene in the one currently highlighted in yellow, provided it is empty.

- Click "Store to next scene" to put the new scene in the next one, provided it is empty.
- Click "Overwrite scene" to overwrite the scene currently highlighted in yellow.

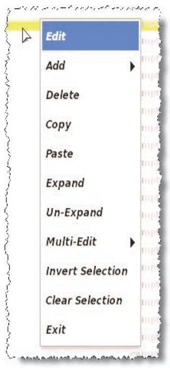
The options are context-sensitive, so some may be greyed-out to show that they are unavailable. An **OK** button will appear at the bottom of the window when a valid store scene option has been selected.

5 Click **OK.**

Additional control — managing events

You can use the MIDI or GPIO functions of the PRO1 to control the parameters of an external device (outgoing), and conversely you can use an external device to control the PRO1 (incoming). Also, by using the PRO1's unique 'internal' event option, you can trigger events from within the showfile itself. All this is done by creating events in scenes/point scenes.

You can have any number and types of events in any scene/point scene; their parameters are set up and edited in an **Edit Event** window. Similarly to scenes/point scenes, you can skip events during rehearsals.



Events (and scenes/point scenes) have a right-click menu (shown right) that lets you to create, edit and copy events.

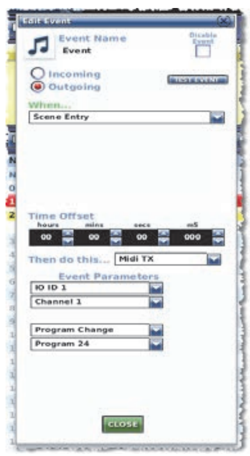
The following shows what some of the event symbols in the **Automation** screen mean: = currently selected event; = MIDI event; = GPIO event; = internal event; = incoming event; and = outgoing event.

>> To create an event

Select the scene in which you want to create the event, and then do one of the following:

- Click the **ADD GPIO**, **ADD INTERNAL** or **ADD MIDI** button as necessary.
- From the right-click menu, choose ***Add* ▸ *Midi Event***, ***Add* ▸ *Internal Event*** or ***Add* ▸ *GPIO Event*** as necessary.

>> To edit an event



1 Open the **Edit Event** window by doing one of the following:

- Right-click the event you want to edit and then choose ***Edit.***
- Select the event you want to edit and then click **EDIT.**

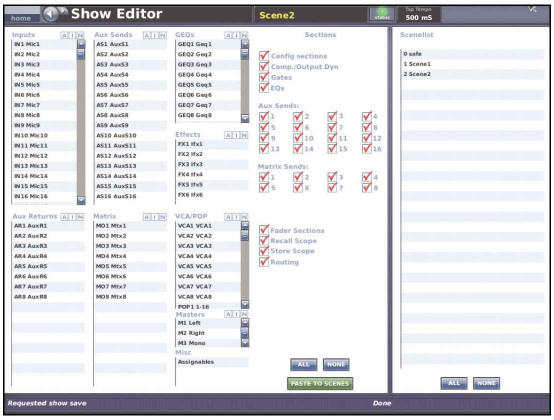
2 In the **Edit Event** window, choose your options as necessary. For example, you can use a program change to trigger the event.

3 At the upper-right corner of the **Edit Event** window, click "X" to close it.

Changes in the **Edit Event** window are *live*, that is, they are immediately reflected in the show file. So you don't have to save the scene, as these are not audio parameters.

Show editor

The show editor is a GUI-only function that allows you to very easily copy and paste settings through scenes. This is done via the **Show Editor** screen.



The **Scenelist** on the right of the screen is a cue list of the current show. Source lists (channels, GEQs, effects and groups) are to the left of the screen, from which you can copy the settings, and in the middle (**Sections**) are the areas you can copy.

>> To open the Show Editor screen

Do one of the following:

- From the GUI menu, choose **home** ▸ ***Automation*** ▸ ***Show Editor.***
- At the **Automation** screen, click **SHOW EDITOR.**

>> To copy and paste sections to a scene(s)

1. In the **Show Editor** screen, choose from the sources (far left) that contain the settings you want to copy from the lists. You can choose any combination.
2. Under the **Sections** heading, choose the sections that you want to copy. Ticked options will be copied. You can use the buttons underneath to help you, as follows:
   - Click **ALL** to select all of the sections.
   - Click **NONE** to deselect all selected sections.
3. In the **Scenelist**, click the scene(s) in which you want to paste the settings. You can use the buttons underneath the list to help you, as follows:
   - Click **ALL** to select all of the scenes in the list.
   - Click **NONE** to deselect all selected scenes.
4. Click **PASTE TO SCENES.**

Configuring the inputs and outputs

Similarly to the VCA/POPulation groups, you can change the name and colour of each of the inputs and outputs. This is done via the GUI at the **Naming Sheet** screen. For configuration details, see "Using VCA/POPulation groups" on page 72.

>> To open the Naming Sheet screen

Do one of the following:

- At the GUI, choose **home** ▸ ***Input Channels*** ▸ ***Naming Sheet.*** Alternatively, choose **home** ▸ ***Mix & Outputs*** ▸ ***Naming Sheet***.
- In the navigation zone, press the **inputs/outputs** screen access button once to open the **Input Sheet** screen or twice to open the **Output Sheet** screen.

Using copy and paste

The **copy** and **paste** buttons (upper-right corner of GUI) let you copy the parameters of a channel's processing area (EQ, compressor, gate etc.) or all of its details areas, and paste them to another channel/ all channels of a similar type. Both buttons have a right-click menu.

>> To copy a processing area to a channel/all channels

1 Navigate the processing area to its local channel strip.

2 Click **copy.**

3 Do one of the following:

- To copy the processing area to another channel, select the channel and then click **paste.** As the copied parameters are still stored, you can paste to as many channels as you want.
- To copy the processing area to all other channels, right-click **paste** to open its menu and then choose ***Paste To All.***

>> To copy all parameters to a channel/all channels

1 Select the channel from which you want to copy all parameters.

2 Right-click **copy** and then choose ***Copy All.***

3 Do one of the following:

- To copy all the parameters to another channel, select the channel and then click **paste.**
- To copy all the parameters to all other channels, right-click **paste** and then choose ***Paste To All.***

Copy and paste rules and restrictions

- You can only copy and paste similar functions. For example, you can't copy the input EQ from one channel to the output EQ of another, as they are different.
- You can only copy and paste across similar channel types. For example, you can't copy from an aux and paste to a matrix.
- Copying and pasting across inputs is restricted to the input bays only.
- Channel names are not copied.
- Compressor and gate side chain listen cannot be copied.

For details of the channel parameters that are copied across, see Appendix I "Parameters Affected By Copy And Paste" on page 441.

User library (presets)

The PRO1 has a user library where you can store settings, such as for the EQ or even the whole channel. For example, you may wish to store the EQ settings of a singer who may be called upon to perform during a future show. You can then easily recall these EQ settings to the appropriate channel when required.

The settings are stored as presets, which are saved in a library. The library files are managed via a **Preset Manager** screen in the GUI menu. This screen has **New**, **Load**, **SAVE** and **Save As** function buttons that let you create new libraries, load existing libraries, save the current library or give it a new name. You can also delete presets from the library.

Before you can save/load a preset, you need to create a new preset library or open an existing one.

>> To create a new preset library

1 From the GUI menu, choose **home** ▸ ***Preset Manager.***

2 In the **Preset Manager** screen, click **New.**

3 In the **Enter new Library name window**, type in your chosen name for the new preset library.

4 Click **OK.**

>> To save a preset to the user library

1 Navigate your chosen settings to the channel strip.

2 At the GUI, click **store preset.**

3 In the **Save Preset** window, do the following:

- In the **Preset Name** field, choose the preset.
- In the **Preset Author** field, type in your name.
- In the **Notes** field, type in any notes to help you identify the contents of the preset.

4 Click **OK.**

If a message window opens containing the text "There is already an existing preset of that name Do you wish to overwrite ?", click **OK** to overwrite the existing preset. Otherwise, create a new one by clicking **Cancel**, choosing another preset name and then clicking **OK.**

>> To load a preset

1 Select the channel in which you want to load the preset.

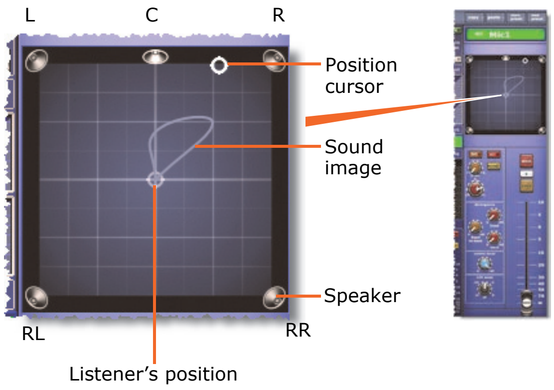
2 At the GUI, click **load preset.**

3 In the **Load Preset** window, choose the preset you want to load.

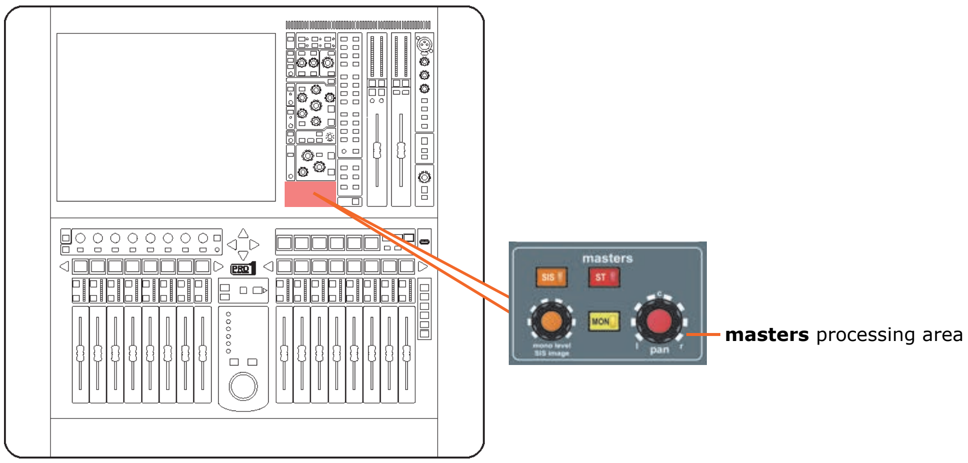
4 Click **OK.**

Surround panning

In addition to stereo and left-centre-right (LCR) panning, PRO1 has three surround panning modes: quad; left, centre, right and surround (LCRS); and 5.1 surround. To help you visualise the surround panning envelope, the masters processing area of the GUI channel strip has a spatial diagram (shown right) that updates when you operate the panning controls.



The 5.1 panning mode uses six matrix channels, while quad mode uses four (left and right on both the front and surround). Although the LCRS mode uses five channels (front left and right, centre and surround left and right), both surround channels are the same. (In an LCRS surround panning arrangement, you can have a single surround speaker positioned directly behind the listener.)



The sound image is controlled via the **masters** processing area. The following below shows the recommended[1] 5.1 surround system configuration.

1. Reference - ITU-R BS.775.1, 1994. *Multichannel stereophonic sound system with and without accompanying picture.* International Telecommunications Union.

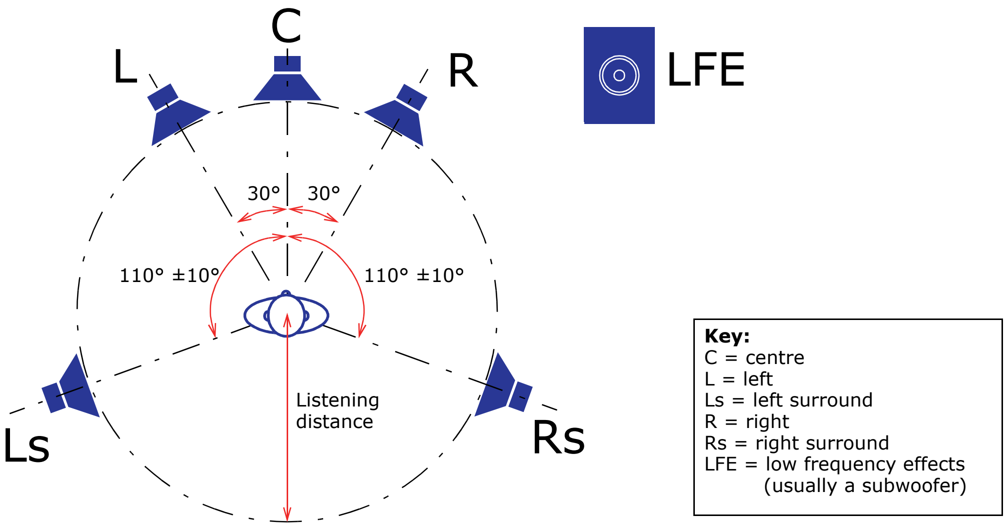
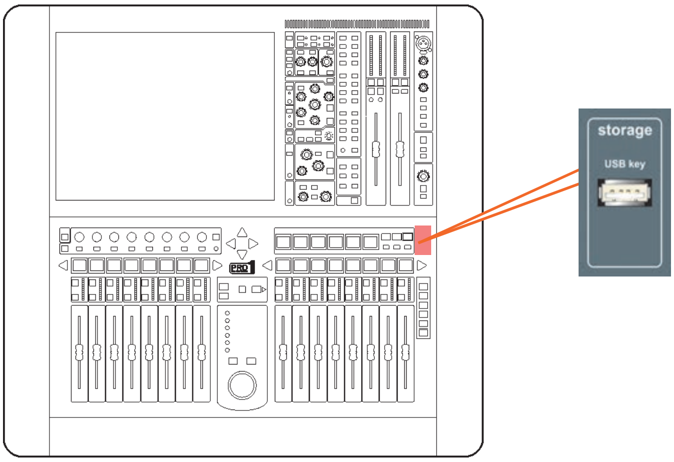


Figure 7: 5.1 surround panning loudspeaker arrangement

In surround mode, the **SIS** button routes the channel to the surround buses in much the same way that the **ST** button routes to the master buses. Surround panning mode is selected via the **Surround Mode** options of the **Preferences** screen (choose **home** ▸ ***Preferences*** ▸ ***General***).

Saving your show files to a USB memory stick

When you are satisfied that your show file is how you want it, we recommend that you save it to a removable storage device (USB memory stick). This provides a valuable back up should the show file stored in the internal memory of the PRO1 be lost, for example, due to inadvertent deletion. You can also load show files onto the PRO1 from the same storage device.



>> To save (export) a show file to a USB memory stick

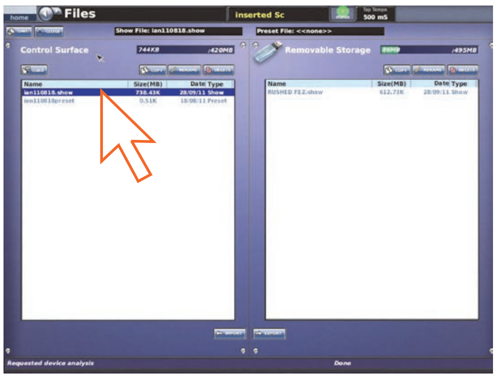
1 If necessary close and save the show file you want to export. You can't export a show file that is open.

2 Insert the USB memory stick into the **USB key** socket (shown above).

3 Do one of the following:

- At the GUI, choose **home** ▸ ***Files.***
- In the navigation zone, press the **automation / filing** screen access button twice.

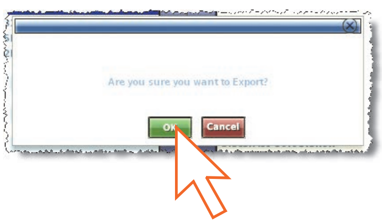
4 You may see an "Analysing..." message in the **Removable Storage** panel, which means that the Midas folder on the USB memory stick is being read. Wait for the message to clear. Then, in the **Control Surface** panel, click the show file you want to copy (shown right).



Both lists show user libraries (presets) as well as show files. If there are neither of these file types on the USB memory stick, the **Removable Storage** panel will be empty.

5 Click **EXPORT.**

6 In the **Are You Sure you Want To Export?** message window, click **OK** (shown right). The file will start copying to the USB memory stick.



7 When your show file appears in the **Removable Storage** panel, it has finished copying to the USB memory stick. Remove the USB memory stick.

>> To load (import) a show file from a USB memory stick

The procedure is similar to the export procedure, as detailed in "To save (export) a show file to a USB memory stick" above, but select the file to be imported to the PRO1 from the **Removable Storage** panel and then click **IMPORT**.

External AES50 synchronisation

If you want to connect AES50 audio between two Midas digital consoles the slave console must be set to external AES50 synchronisation, irrespective of the synchronisation source of the master console.

	Console 2 sync setting			
Console 1 sync setting	***Master***	***Word clock***	***AES3***	***External AES50 from console 1***
Master	Not valid	Not valid	Not valid	Valid connection
Word clock	Not valid	Not valid	Not valid	Valid connection
AES3	Not valid	Not valid	Not valid	Valid connection
External AES50 from console 2	Valid connection	Valid connection	Valid connection	Not valid

A valid connection can be a tie line between the stage routers or the secondary port (Bx/By) of a mic splitter that has its primary port (Ax/Ay) connected to the master console.

Make sure that the DL431 Mic Splitter is set to synchronise from the master console.

Security (locking mode)

To preserve the current state of the PRO1 Control Centre and to prevent unauthorised adjustment of its settings, you can lock it; this is a GUI-only function. When locked, none of the controls on the control surface will function and the PRO1 Control Centre will be totally locked out.

At a GUI screen, choose **home** ▸ ***Lock.***

>> To unlock the PRO1 control centre

At a GUI screen, click the **UNLOCK** button (lower-left corner of screen). The PRO1 Control Centre will revert to the state it was in when it was previously locked.

Advanced Operation And Features

Chapter 10: Stereo Linking

This chapter describes stereo linking, or *channel pairing*, which lets you configure two adjacent channels as stereo. Linking is a GUI-only function, as there are no link buttons on the control surface. The diagram right gives an example of two linked channels (auxes 15 and 16).

Any two adjacent input channels can be linked together, but each odd numbered output channel can only be linked with its adjacent even numbered channel to the right (for example, you can link aux send channel 7 with aux send channel 8, but you cannot link aux send channel 8 with aux send channel 9). So, although each output channel has a link button, the ones on the even numbered channels don't work.

By default, all channels of the PRO1 — except the master left and right channels, which are always stereo linked — are unpaired (mono). When paired, 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 individually. 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.

When linking previously unlinked channels, some normalisation of the prospective left and right control settings — which may be quite different — is required. The PRO1 does this by automatically copying the control settings of the left (odd-numbered) channel to the right channel, with the exception of the pan controls. The pan controls, depending on whether they are in the left or right audio paths, should be manually set to hard left or hard right, respectively.

>> To stereo link two channels

1 If necessary, navigate the desired channels to the channel faders.

2 In the GUI fast strip of the desired channel, click **LNK.** Alternatively, you can click the **LNK** button in the GUI channel strip if the desired channel and its appropriate processing area are selected.

3 In the channels you have just linked, set the **pan** control knob in the left channel fully anti-clockwise and set the one in the right channel fully clockwise. For more information on panning, see Chapter 10 "Stereo Linking" on page 93.

Changing the linking options

You can choose which control options will be linked across the channel pair. There are two ways to do this: globally and per pair. The per pair settings always override the global ones. For details of the linked parameters for each section, see Appendix J "Parameters Affected By Stereo Linking" on page 467.

>> To set the global default stereo linking parameters for a channel type

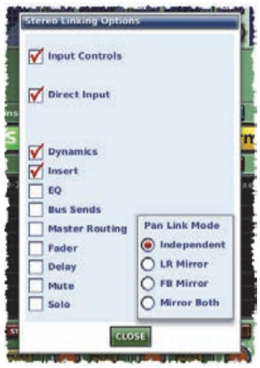
1 At the GUI, choose **home ▸ *Preferences* ▸ *General***, and then click the **Linking** tab to open the **Preferences Link** screen (shown right).

2 In the desired channel section, choose the default stereo linking options you require, and then click the local **Change Existing** button (at bottom of section).

__Note:__ Deselecting a linking option (unticking its box) automatically updates the status of unlinked channels of a similar type accordingly. However, clicking the __Change Existing__ button updates the status of all unlinked __and__ linked channels of a similar type.

Selected control options will be linked across each channel pair of the appropriate type, while unselected controls remain independent on each channel.

>> To set the stereo linking parameters for a channel pair



1 Make sure that one of the paired channels is selected and that its configuration processing area is assigned to the GUI channel strip, and then click the **st. linking options** button in the GUI channel strip.

2 In the **Stereo Linking Options** window (shown right), select the controls that you want to be linked across the channel pair.

3 Click **CLOSE.**

Pan Link options

The **Pan Link Mode** function has four options, which do the following on stereo linked channels:

- **Independent** — Pan controls and front-back panning in surround modes are not linked.
- **LR Mirror** — Pan values are mirrored between the left/right channels and, in surround modes, front-back panning is linked.
- **FB Mirror** — Pan values are not linked and, in surround modes, front-back panning is mirrored and left-right panning is linked.
- **Mirror Both** — Pan values are mirrored between the left/right channels and, in surround modes, are also mirrored front-back.

Chapter 11: Panning

The PRO1 has two main types of panning mode, default and surround. The default mode comprises stereo and LCR panning formats, and only uses the channels for the front loudspeakers, while the surround mode includes channels for the rear surround loudspeakers. For more information, see "Surround panning" on page 86.

The following table shows the panning formats available on the PRO1.

Table 3: Panning formats

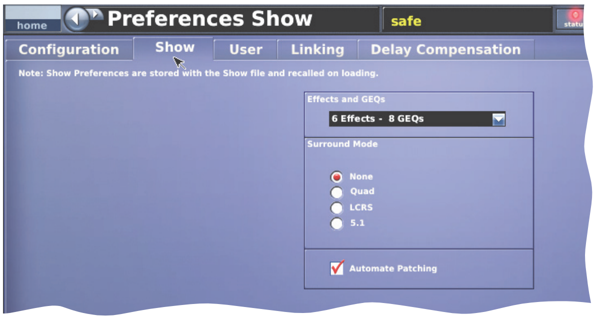
Panning mode	*No. of channels*	*Format*	*Channel types*
Default	2	Stereo	L, R
	3	LCR (SIS™)	L, C, R
Surround	4	Quad	L, R, Lr, Rr
	5	LCRS	L, C, R, Ls, Rs
	6	5.1	L, C, R, Ls, Rs, LFE

Key: L = left; R = right; C = centre; Lr = left rear; Rr = right rear; Ls = left surround; Rs = right surround; LFE = low frequency effects (usually handled by a subwoofer)

>> To select the panning mode

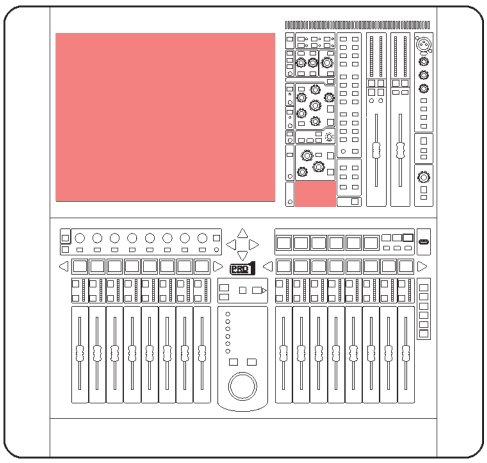
At the GUI, choose **home ▸ *Preferences* ▸ *General.*** Then click the **Show** tab to open the **Preferences Show** screen. Click the desired panning mode in the **Surround Mode** section.

Note: *The* ***None*** *option is the default mode for stereo and LCR (SIS™) (see Table 3 above).*



Stereo panning

The control surface controls for stereo panning are located in the **masters** section of the channel processing area. The **pan** control in each **masters** section may be switched for either conventional or spatial imaging system (SIS™) stereo operation via the **ST** or **SIS** switches, respectively.



SIS™ (LCR) mode

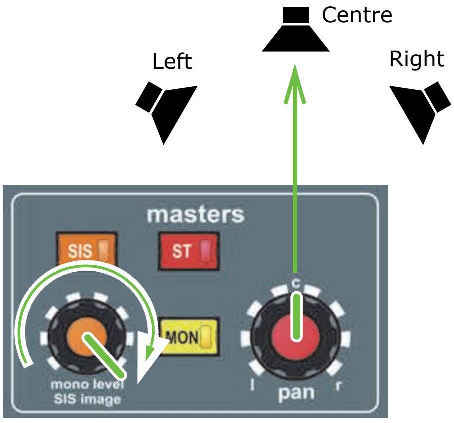
The MIDAS SIS™, which is used for left-centre-right (LCR) loudspeaker systems, configures the channel for LCR mixing. The **SIS** switch activates the spatial imaging system, which uses the **SIS image** control knob to modify **pan** control knob operation so as to place the channel within a three-speaker system.

With **SIS image** control knob set fully clockwise or anti-clockwise, image is full LCR or stereo, respectively. Control knob positions in between generate a composite blend of stereo or LCR panning systems, so that optimum degrees of centre image focus and speaker power can be obtained.

Constant power is maintained at all times so that the *image* can be adjusted during the show without a perceived change in level.

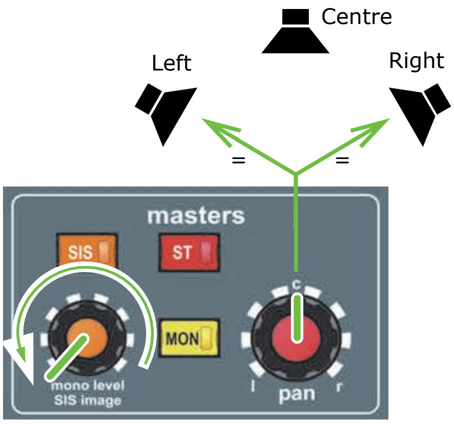
SIS image control knob fully clockwise (LCR)

With the **SIS image** control knob fully clockwise, the **pan** control knob operates in full LCR mode. A centre-panned signal, that is, with the **pan** control knob set to the **c** position, routes to the centre speaker only; there is no signal in the left and right speakers.



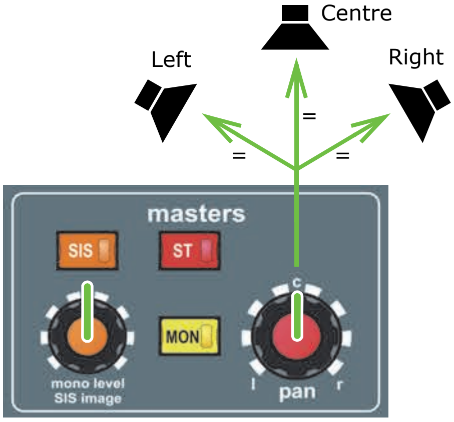
SIS image control knob fully anti-clockwise (stereo)

With the **SIS image** control knob fully anti-clockwise, the **pan** control knob operates as stereo. A centre-panned signal routes to the left and right speakers at equal power.



SIS image control knob centred (equal power)

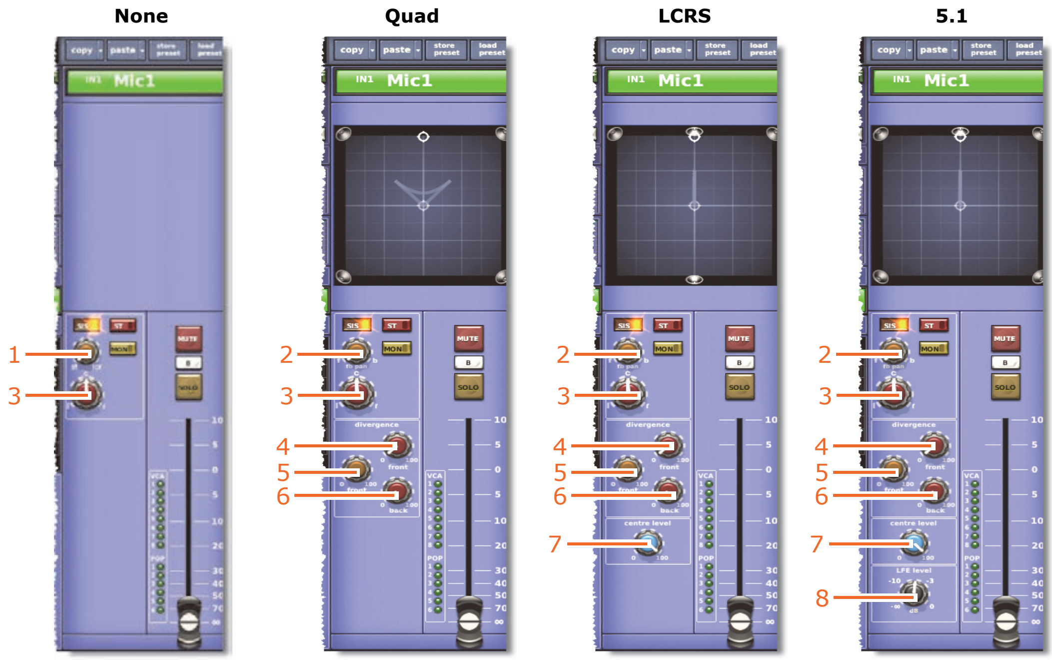
With both the **SIS image** and **pan** control knobs centred, the signal is routed to all three speakers with equal power.



Surround panning

When the PRO1 is configured to operate in one of the surround panning modes, the spatial diagram that appears in the GUI channel strip gives you a visual representation of the sound image in relation to the speakers.

The following diagram shows the appearance of the GUI channel strip for each panning mode, and describes the controls.



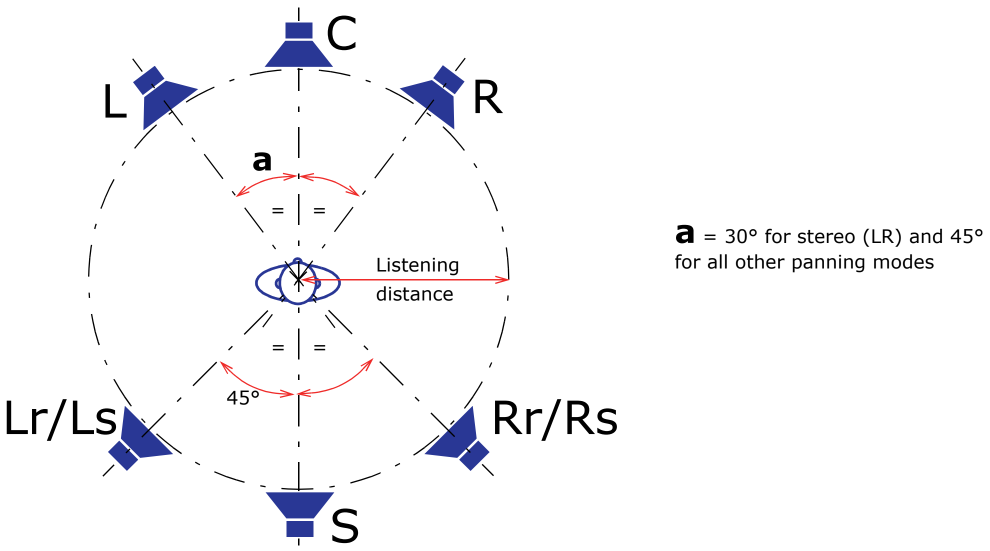
Item	Description	Function
1	SIS™ image control knob	With **SIS** switched on, this adjusts the SIS™ image. With **SIS** switched off, this adjusts the mono level.
2	**fb pan** control knob	Moves the position cursor in the spatial diagram up/down. With **SIS** switched off, this adjusts the mono level.
3	Left-centre-right (LCR) control knob	Moves the position cursor in spatial diagram left/right.
4	**front** control knob	Adjusts the divergence of the front speakers.
5	**front to back** control knob	Adjusts the divergence of the front and rear speakers.
6	**back** control knob	Adjusts the divergence of the rear speakers.
7	**centre level** control knob	Adjusts the divergence of the centre speaker.
8	**LFE level** control knob	Adjusts the signal level of the LFE (usually a subwoofer).

Although the position cursor changes automatically according to the adjustment of the surround panning controls, you can also adjust it on the GUI using drag.

Speaker placement

As the placement of loudspeakers is very important for accurate mix monitoring — especially for multi-channel mixing for surround sound — you should consider speaker positioning, angling and level calibration when setting up your monitor system. If necessary consult the manufacturer of your monitor system for their recommended surround formats.

The following diagram — which is to be read with Table 3 "Panning formats" on page 95 — gives you examples of loudspeaker placements for each panning mode. However, they are only an approximation and should only be used as a guide.



Note: *LCRS has a mono surround channel, which is often fed to two rear 'satellite' speakers.*

Chapter 12: Soloing

Solo enables a signal to be monitored at a level proportional to its level in the mix, in the same stereo position in the mix and with the same signal processing as in the mix. When a solo button is pressed, all signals routed to the monitor output — except the one selected by the solo button — will be cut. Solo allows a sound in one channel to be isolated, which is helpful in fault finding and when equalising a signal.

There are two independent solo systems in the console, solo A and solo B. Both have monitor outputs and solo A has a headphone output, and both can be used to PFL or AFL signals from the same sources throughout the console. This flexible solo bus configuration makes soloing of three-way monitor mixes (in-ears going to solo A and wedge going to solo B) possible.

Solo A

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

- If a solo button is pressed for a short time, the soloing to the selected solo bus remains active when the button is released. If press is sustained (solo button held down) the soloing to the selected solo bus is cancelled when the button is released.
- 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 concurrent, the solo activation that occurred last — while the respective solo add mode (A or B) is inactive — cancels all earlier solos to the same bus before it activates.
- Solos can also be operated from a VCA master when the channel to which they belong is a member of that VCA; this is in addition to the local operation.
- Pressing the solo **CLEAR** button associated with the solo bus they are sending to (A or B) will clear active solos.
- A two-level solo hierarchy exists for each of the solo buses in the console (see “Solo hierarchy” on page 103); the levels are inputs and returns, and aux sends and matrices and masters. 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.
    - With any inputs active, outputs can’t be soloed.
    - with any outputs active, pressing an input solo overrides (cancels) output. When the input solo(s) is cancelled, the output solo(s) are returned.
- Pressing ADD (solo a/b) off cancels all solos and then allows you to solo multiple channels simultaneously.
- Pressing PFL (solo a/b) on changes the point at which the soloed signal is taken from, that is, from post-fader to pre-fader.
- Soloing inputs and outputs:

With solo A off there is no solo in operation.

Table 4: Solo A destination controls

Control	*Options*	*Description*
PFL direct input	-	Direct inject to solo A from linked console, active only while solo A is PFL.
AFL direct input	-	Direct inject to solo A from linked console, active only while solo A is AFL.
Solo add	• On (additive solos) • Off (self-cancelling solos)	Disable self-cancelling solo A solos. (When self-cancelling solos are selected, that is, with solo add mode off, the solo being cancelled will be deactivated before activating a new solo.)
Solo clear	• On (some solo A solos) • Off (no solo A solos)	Single button clearing of currently active solo A solos.
Solo PFL	• PFL (solo pre-fader) • AFL (solo after-fader)	Switch all current and future solo A activations to send the solo A bus pre-fader.
Solo in place (SIP)	• On (SIP active) • Off (SIP inactive)	When active, SIP uses the console's master outputs for the solo A material. It does this by muting all input channels that are not currently soloed to the solo A bus. It is not possible to activate SIP accidentally. The SIP button has a hinged clear plastic cover that has to lifted up before the button can be operated. For SIP purposes, master outputs can be the main master bus or, if configured, a multi-channel output mix. 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 are not 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.

Solo B

The solo B buttons are located next to the individual solo buttons (GUI only). With solo B on, solo goes to the B bus and with solo B off, solo goes to the A bus. Solo B has separate **ADD, CLEAR** and **PFL** controls that, in broadcast mode, can be used to control those functions independently of solo A.

Solo hierarchy

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, namely 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 <--> Side chain listen
  - **Return channels:** Return channel <--> Direct in
  - **Aux buses:** Aux bus <--> Direct in <--> Side chain listen
  - **Matrix outputs:** Matrix bus <--> Direct in <--> Side chain listen
  - **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 active on the console at any 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.

Solo in place (SIP)

Operating a solo button cuts all channels from the main mix except soloed ones; this is called "solo in place" or "SIP". SIP allows you to check the contribution from soloed channels at the actual levels they occur in the mix, that is, taking into account the main fader setting. If solo buttons cut the main output (main mix) they must only be used in rehearsals.

>> To activate SIP

1 Press the solo in place switch. A window will open containing the message "Activate SIP ?".

2 Click **OK.**

Solo modes

There two available solo modes (see "Solo system" on page 110) as follows:

- **Normal** Solo A and B are treated as a single 4-channel solo bus, with the **ADD, CLEAR** and **PFL** controls in the monitor section linked together.
- **Broadcast** Switches all of the channels to solo B, selects the stereo masters for the solo A output and unlinks the **ADD, CLEAR** and **PFL** controls in the monitor section. This allows the main program material to be monitored at the same time as something soloed on the solo B bus.

Chapter 13: Muting

Muting is useful for turning off channels you don't want to listen to. The channel mutes turn off outputs from the channel. See Appendix B "Functional Block Diagrams" for which outputs are affected when channel mute is active. Channel mutes can be activated by:

- Local press.
- Auto-mutes (see "Auto-mute (mute) groups" on page 163).
- VCAs (see "VCA and POPulation groups" on page 159).
- Scene recall (see Chapter 20 "Scenes And Shows (Automation)" on page 177).
- SIP, which uses the control centre's master outputs for solo A material by muting input channels currently soloed to the solo A bus. For SIP muting, channels must be input channels set up to solo the solo A bus.

All of the above mute activation methods, except VCAs, mute the channel outputs and update the channel mute status indicator.

Chapter 14: Monitors And Communications

This chapter describes the monitoring and communications functions of the PRO1.

Monitors (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 master bay (see Figure 8 "Monitor A and B strips" on page 108). Each monitor output has:

- The ability to monitor mono and stereo outputs, and an external input.
- An external talkback input.
- A local monitor output.
- A headphone output.
- Delay compensation.
- Control of the solo buses.

Although the capabilities of both monitors are the same, monitor A is the primary output. They both have a fader control, and there are four balanced XLR outputs on the rear panel).

The monitor output controls *do not* have support from the screens and *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 is the same.

>> To switch control of headphones to fader

Press **C/O**.

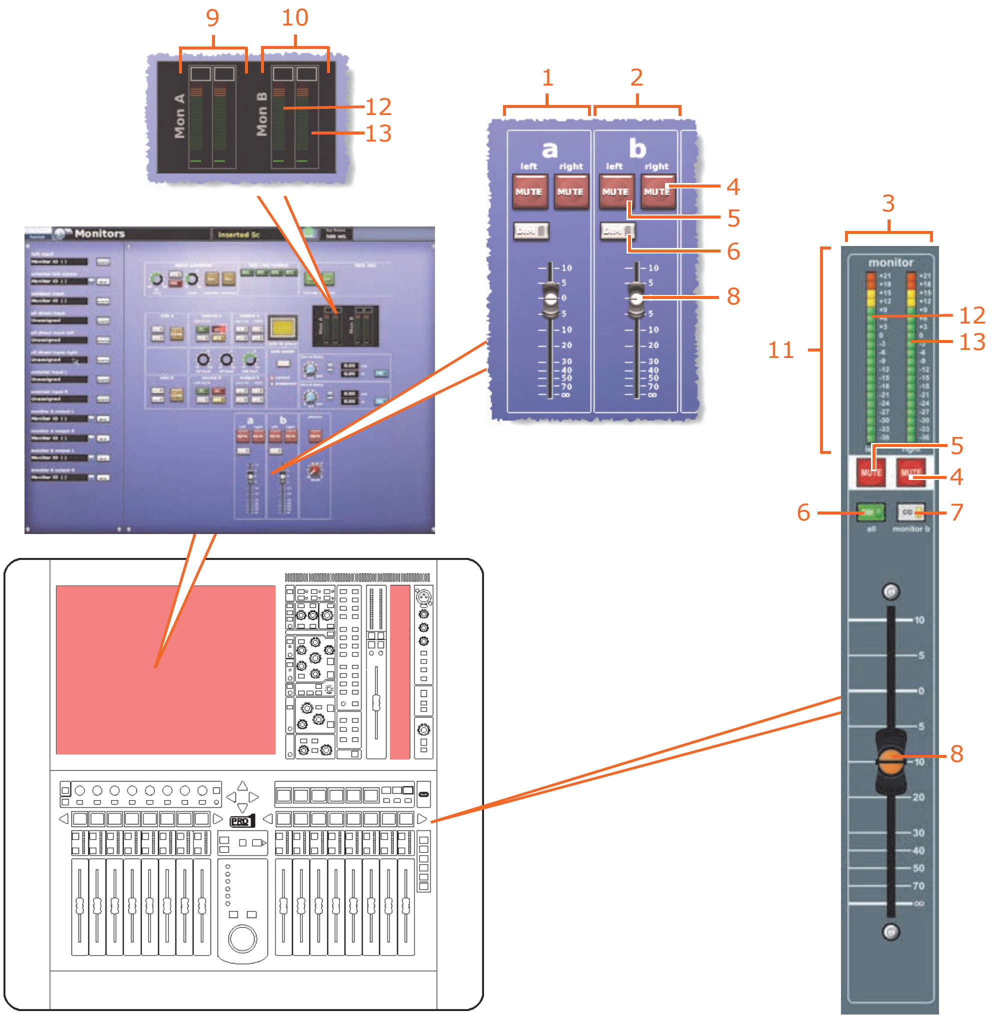


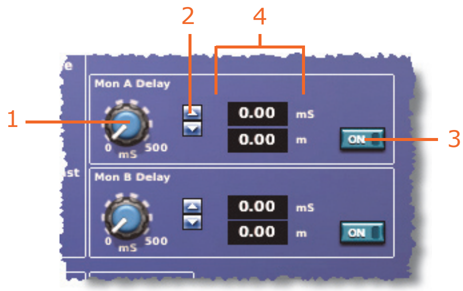
Figure 8: Monitor A and B strips

Item	***Description***
1	Monitor **a** strip on the GUI.
2	Monitor **b** strip on the GUI.
3	**monitor** a/b strip on the control surface. For controlling current monitor assignment (A or B).
4	This **MUTE** button mutes the right monitor signal. On the control surface, it mutes the right monitor signal of the current monitor assignment (A or B).
5	This **MUTE** button mutes the left monitor signal. On the control surface, it mutes the left monitor signal of the current monitor assignment (A or B).
6	This **DIM** button dims the monitor signal output level by 20dB on monitor speakers. On the control surface, it dims the monitor signal of the current monitor assignment (A or B).
7	**C/O** switch that switches control of the monitor's fader and mute to monitor B.he **C/O** (**to fader**) switch (control surface only) swaps control of the solo A to fader.
8	Fader for control of monitor A or B speaker level from -∞ to +10.
9	**left** and **right** meters for monitor **a** (GUI only).

Item	Description
10	**left** and **right** meters for monitor **b** (GUI only).
11	**left** and **right** meters for the monitor (A or B) currently assigned to the **monitor** section on the control surface.
12	Left monitor signal. On the control surface, it is the left monitor signal of the current monitor assignment (A or B).
13	Right monitor signal. On the control surface, it is the right monitor signal of the current monitor assignment (A or B).

Delay (GUI only)

You can delay each monitor output signal (A and B) individually by up to 500 milliseconds (ms). This is done via the two delay sections in the **Monitors** screen. This function does not have support on the control surface.



Monitor A and B delay sections on the GUI

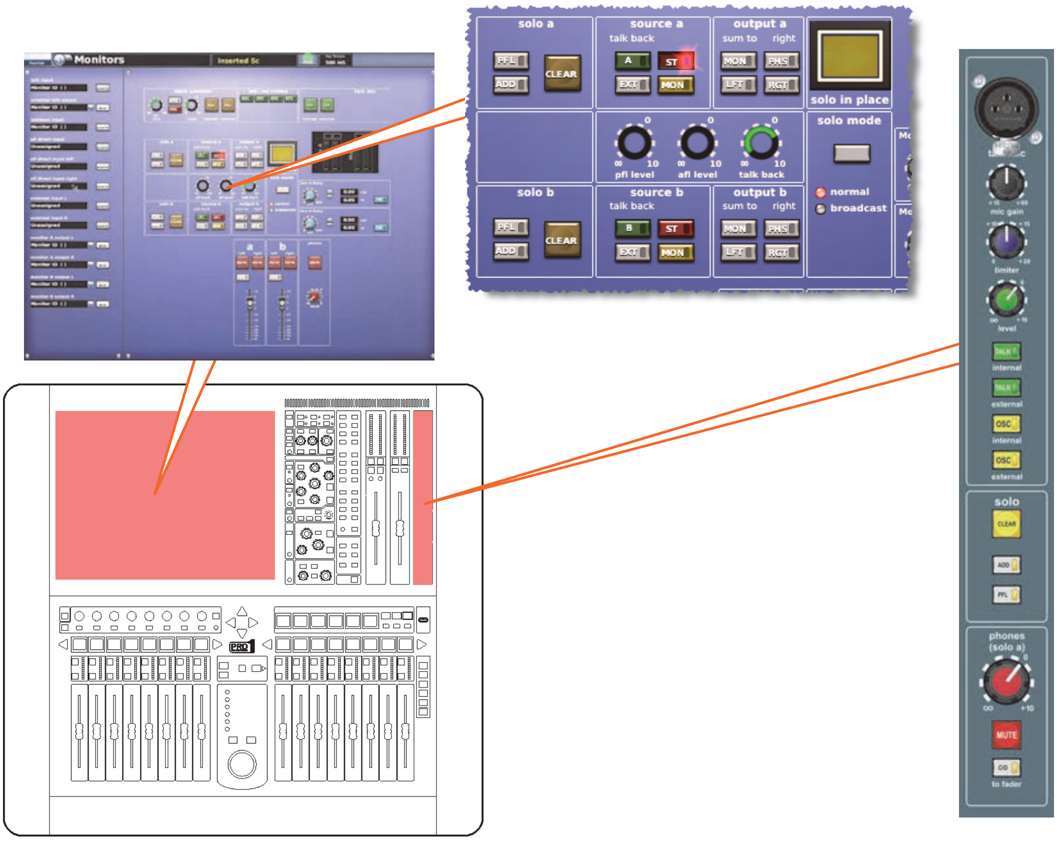
Item	Element	Description
1	Control knob	Adjusts the monitor output signal delay in the range 0ms to 500ms.
2	Up/down spin buttons	Provide finer adjustment of the monitor output signal delay.
3	**ON** switch	Switches the delay on/off.
4	Two delay value fields	Show the current delay value in milliseconds (ms) and metres (m).

Solo system

solo a and **solo b** system sections allow solo signals to be selected independently for each monitor system (A and B). These can be selected as AFL (**PFL** extinguished), PFL (**PFL** illuminated), additive (**ADD** enabled) or interlock cancelling.

The monitor outputs can be configured for normal or broadcast use and both modes change the interleaving logic between differing areas of the monitor output. A mode select button selects either of the two possible options.

Additionally, there is a **solo in place** switch for activating the SIP function.



Solo system controls on the control surface and GUI

solo in place switch

The GUI-only **solo in place** (SIP) switch puts the control centre in SIP mode. In this mode, pressing a **SOLO** button in an input fast strip activates a mute of all other channels, assuming it is set to the appropriate monitor (A or B); talk back remains unaffected.

When SIP is switched on and an input is soloed, all unsoloed inputs are muted, except the auto-mutes. With SIP in operation, pressing a **SOLO** button in a VCA section (for a group) solos all group members, while muting non-group members. When SIP is switched off, any solos are kept active but the mutes are removed (except the ones with auto selected, which are left alone).

As this is an important function that may have detrimental consequences, the button on the control surface is protected by a plastic cover to prevent it being inadvertently switched on/off.

You can protect a channel from this function by switching on its mute safe (see "Safes" on page 254).

>> To activate SIP

1 At the GUI, choose **home ▸ *Monitors.*** Then click **solo in place.**

2 In the "Activate SIP ?" message window, click **OK**.

>> To deactivate SIP

On the GUI, click the **solo in place** button on the *Monitors* screen.

C/O switch

The **C/O** (**to fader**) switch on the control surface swaps control of the headphone output and solo A output between the **phones (solo a)** control knob and the **monitor** fader.

Monitor output (A and B)

The monitor output sections — **output a** and **output b** — have common controls for the monitor A and B sections (GUI only). The monitor's output level is controlled by a non-automated fader, a **DIM** switch and left and right mutes (see Figure 8 "Monitor A and B strips" on page 108).

The buttons in each output section have an integral LED for on/off indication, and have the following functions:

- **MON** switch, sums left and right signals, but with a ±3dB loss. With 0dB on monitor A left and nothing on monitor A right, pressing **MON** gives -3dB on both monitor A left and monitor A right. However, with 0dB on both monitor A left *and* monitor A right, pressing **MON** gives +3dB on them both.
- **PHS** phase reverse switch, reverses the phase of the right monitor signals.
- **LFT** and **RGT** switches, route left and right monitor signals, respectively, to both left and right monitor speaker outputs. These switches can be used in combination, as shown in Table 5 below.

Table 5: Monitor signal routing

LFT button	*RGT button*	*Monitor signal routing*
Off	Off	Left and right monitor signals are routed normally, that is, left monitor signal is routed to the left monitor speaker output, and the right one is routed to the right monitor speaker output.
On	Off	Left monitor signal is routed to both of the monitor speaker outputs.

LFT button	*RGT button*	*Monitor signal routing*
Off	On	Right monitor signal is routed to both of the monitor speaker outputs.
On	On	Left and right monitor signal routing is swapped over, that is, left monitor signal is routed to the right monitor speaker output, and the right one is routed to the left monitor speaker output.

source (a and b) sections

The **source a** and **source b** sections contain monitor input selector switches. 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 external (**EXT**). Additionally, each section has a talkback switch.

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

- Talk **A** and **B** switches, sum the talk back signals to the solo bus. The **talk mic** section (see "Talk mic" on page 116) has a level control knob that is shared between the two monitor paths.
- **ST** switch, routes post-fader stereo master mix to stereo local monitor outputs.
- **EXT** switch, routes stereo external input (two-track return etc.) to stereo local monitor outputs.
- **MON** switch, routes post-fader mono masters mix to stereo local monitor outputs.

solo (a and b) sections

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 buses may accept injected external signals, and two control knob level controls make adjustments.

PFL and AFL levels are adjustable via the **pfl level** and **afl level** control knobs; see "solo system section" on page 113.

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

- **PFL** switch, sends mono pre-fader listen (PFL) solo bus 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.
- **ADD** switch, allows multiple channel access to solo buses. 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 output solos and VCA solos, and will temporarily override them. When input solo is cancelled, output solo or VCA solos will return.
- **CLEAR** switch, illuminates when a solo switch is active in its monitor section and, when pressed, clears any solo switches in that section.

solo mode section

On the GUI, the **solo mode** section has a select button by which you can select one of the following options. The LEDs to the left of the options illuminate to show which option is in operation.

- **normal** — both solo systems (A and B), are active and behave as a single solo system.
- **broadcast** — routes stereo masters to the monitor A output and activates all the solo B controls so that soloed material is routed to the monitor B outputs. This allows the master outputs to be continually broadcast (probably the on-air program), while the other material is soloed.

solo system section

The **solo system** section has three control knobs, as follows:

- **pfl level** control knob — PFL audio bus may accept injected external signals. This control knob adjusts the pre-fader level in the range infinity (∞) to +10dB.
- **afl level** control knob — AFL audio bus may accept injected external signals. This control knob adjusts the after-fader level in the range infinity (∞) to +10dB.
- **talk back** control knob — adjusts the talk back level, in the range infinity (∞) to +10dB.

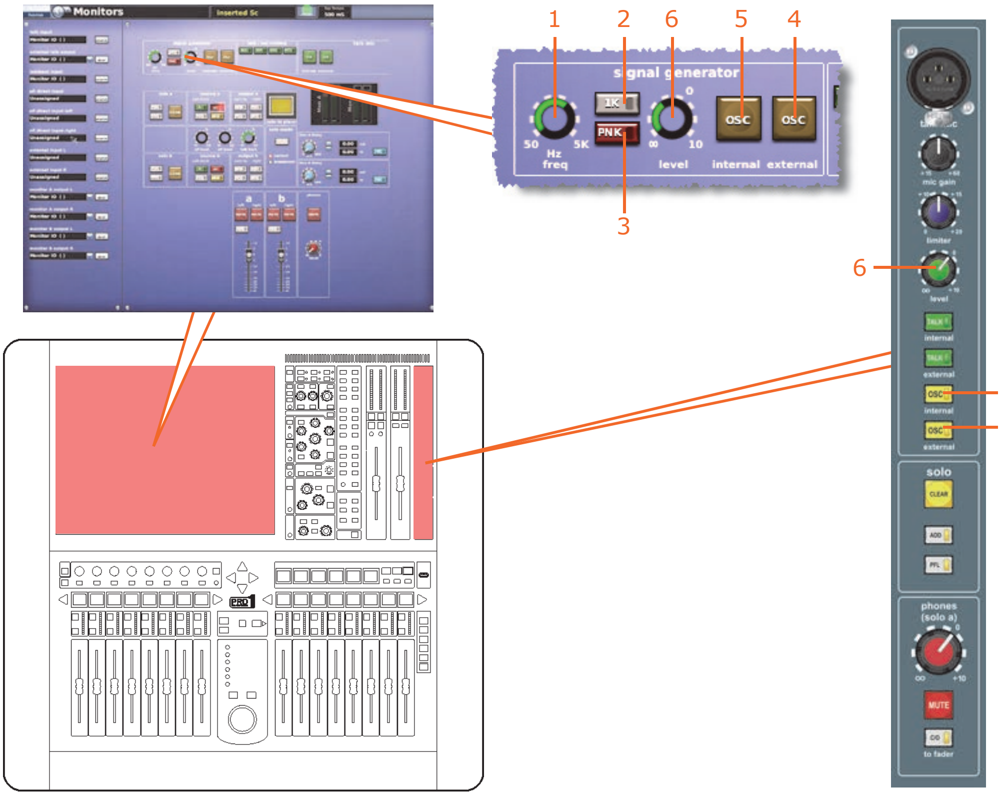
The following four sections in the **Monitors** screen allow you to patch the solo system signals.

- talkback input
- pfl direct input
- afl direct input left
- afl direct input right

For routing details, see Table "Navigating to the Patching screen" on page 54.

Signal generator

The **signal generator** section can output to pink noise (pink noise generator) or sine wave tone (sinusoidal oscillator), and connect to the internal and external talk buses.



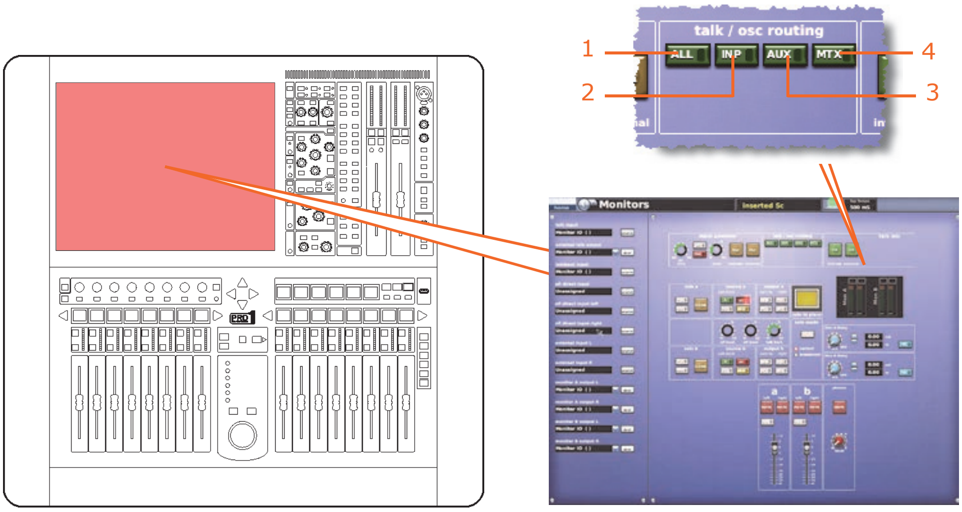
Signal generator controls on the control surface and GUI

Item	Element	Description
1	**freq** control knob	Gives continuous adjustment of the sinusoidal oscillator frequency from 50Hz to 5kHz.
2	**1K** switch	Overrides the swept frequency control (item 1) and provides a fixed 1kHz tone.
3	**PNK** switch	Overrides the sinusoidal oscillator and converts output signal to pink noise.
4	**OSC** (**external**) switch	This OSC external switch connects signal generator output to talk external output XLR.
5	**OSC** (**internal**) switch	This OSC internal switch connects signal generator output to the control centre's internal talk and talk select buses. The internal talk bus can then be mixed onto any of the control centre's buses by pressing the internal talk switches associated with those buses, or mixed onto a group of buses by activating an internal talk group (see "Talk osc/routing" on page 115).
6	**level** control knob	Gives continuous adjustment of signal generator peak output signals from off (∞) to +10dB.

The **OSC** switches (internal and external) are the talk routing switches.

Talk osc/routing

The **talk / osc routing**, or 'internal talk groups' section, sends signal generator and talk mic signals to buses within the control centre. It contains four talk group switches used for selecting the destination of the talk and OSC internal signals. The GUI has an additional four configuration switches for programming.



Talk osc/routing controls on the control surface and GUI

Item	Element	Description
1	**ALL** switch	Routes the talk/OSC internal signal to all outputs.
2	**INP** switch	This input switch routes the talk/OSC internal signal to the input section.
3	**AUX** switch	Routes the talk/OSC internal signal to all auxes.
4	**MTX** switch	This matrix talk switch routes the oscillator or talk signal to the matrix outputs.

Internal talk groups

You can assign talkback or send test signals to any audio bus on the control centre. The preset 'talk' groups allow you to, for example, talk to groups of performers in a monitor mix or make group announcements. Also, by using the internal tone oscillator, you can perform signal path testing and equipment alignment.

There are four 'talk' groups available, which are operated via the GUI. The preset talk groups let you can route to all inputs, all auxes, all matrices or all outputs.

Before you can select a talk group, the **TALK/internal** and/or **OSC/internal** buttons must be switched on. (This also applies to generator routing.) Also, if any internal talk or osc generator routing is active when the **TALK/internal** and/or **OSC/internal** buttons are both switched off, this routing is cancelled.

When a talk group is activated, all talk group member functions are activated.

>> To activate a talk group

1 Make sure either one or both of the 'internal' buttons, that is, the **TALK** (**internal**) button in the **talk mic** section and the **OSC** (**internal**) button in the **signal generator** section, are on.

2 Click the desired talk/oscillator routing button.

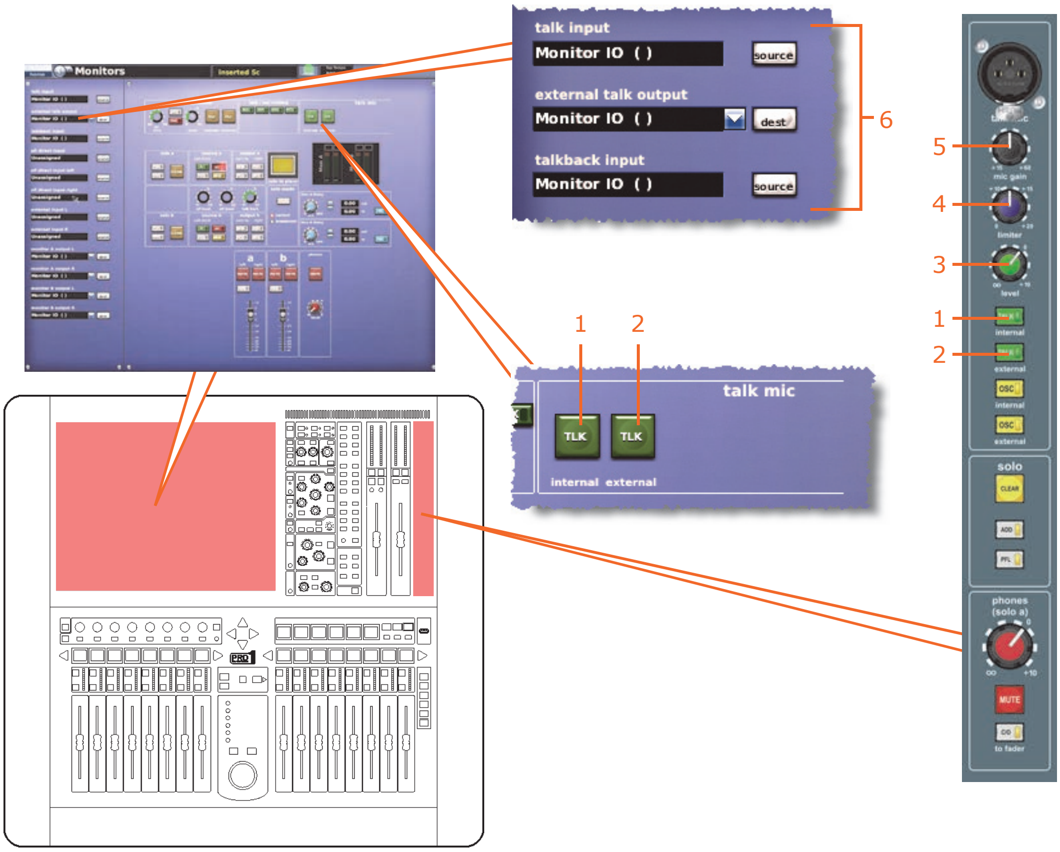
Talk mic

The control surface has an internal talk mic that lets you talk to external locations, and you can also be talked *to* from an external location.

Internal talk mic

This is located in the **talk mic** section, and contains the controls for both the internal talk mic and external talkback functions, which control a talkback microphone connected to the DL251/DL252 Audio System I/O. The talk mic utilises the compressor (limiter), which is in the microphone signal path immediately before the talk level control.

The outputs from the internal mic can be connected to the internal talk bus/talk external XLR output (see Appendix B "Functional Block Diagrams").



Talk mic controls on the control surface and GUI

Item	Element	Description
1	**TALK**/[**TLK**] (**internal**) switch	This talk internal switch connects the talk mic output to the control centre's internal talk system. The internal talk bus can then be mixed onto any of the control centre's buses by pressing the internal talk switches associated with those buses, or mixed onto a group of buses by activating an internal talk group (see "Talk osc/routing" on page 115).
2	**TALK**/[**TLK**] (**external**) switch	This talk external switch connects the talk mic output to the talk external output XLR.

Item	*Element*	*Description*
3	**level** control knob	Gives continuous adjustment to the post-limiter signal from off (∞) to +10dB.
4	**limiter** control knob	Gives continuous adjustment of the peak limiter value from 0dB to +20dB.
5	**mic gain** control knob	Provides continuous mic amplifier gain adjustment of the mic connected to mix bay control surface. Range is +15dB to +60dB and operates in conjunction with the peak limiter.
6	Talk patching sections.	See "About the tabs in the From and To sections" on page 48.

The three control knobs are used in conjunction with mic XLRs (in the screen housing and the rear connector panel) and the two **TALK** routing switches (internal and external).

>> To select the internal talk mic

1. In the **talk mic** section of the master bay, press **TALK** (**internal**) to switch in the talk mic section (see "Talk mic" on page 116).
2. In the **source a** or **source b** section of the master bay, press **TALK.** Your chosen section will determine which system bus (A or B) the talk mic will be sourced from (see "Monitor output (A and B)" on page 111 and "source (a and b) sections" on page 112).

External talkback

The external talkback input is a mic/line input at the stage end of the system that, when enabled in the monitor section (see "Monitor output (A and B)" on page 111), can mix onto the local monitor outputs.

Chapter 15: Graphic Equaliser (GEQ)

This chapter describes the internal GEQs of the PRO Series. Initially, it explains how to use the PRO1 Control Centre to configure and operate the GEQs and then details all of their available control functions.

Overview of the GEQs

The PRO1 Control Centre incorporates a graphic equaliser (GEQ), which is closely based on the Klark Teknik DN370 Graphic Equaliser (see the DN370 operator manual for details). You can configure the PRO1 Control Centre to have set numbers of these GEQs up to a maximum of 28, and these are mutually inclusive of the number of effects you can have. For example, you can have three effects and 20 GEQs, but if you want five effects you can only have 12 GEQs.

Each GEQ is a single-channel, 31-band, third octave graphic equaliser, and GEQ features switched 2nd order high pass and low pass filters and two notch filters with variable frequency ranges.

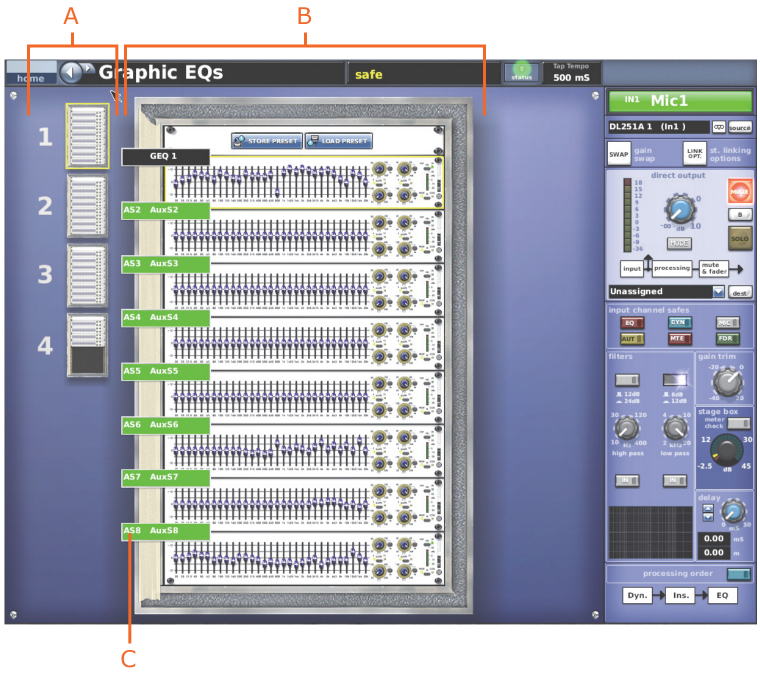
The GEQ is primarily a mono process, but in the case of stereo groups or mix channel outputs, a stereo GEQ is controlled from a single set of controls.

The GEQs are managed via a virtual eight-unit rack on the **Graphic EQs** screen. From here you can open the window of any GEQ, which gives you full control over it.

About the Graphic EQs screen

The main sections of the Graphic EQs screen comprise:

- **GEQ patching source** The border to the left each GEQ unit will display its source, if patched. In the diagram above, GEQ 1 has been patched to "AS3" (aux 3).
- **GEQ rack overview** This section contains an overview of the total number of GEQ racks in use, and also aids GEQ navigation/selection. The number of racks (up to four) is dependent on configuration and the currently selected rack is highlighted in yellow (as is the currently selected GEQ unit).
- **GEQ rack** A 'virtual' rack containing up to eight GEQs. The rack also includes **STORE PRESET** and **LOAD PRESET** user library buttons (see Chapter 24 "User Libraries (Presets)" on page 215).



***Graphic EQs** screen*
***A.** GEQ rack overview (if PRO1 is configured for more than eight GEQs) **B.** GEQ virtual rack **C.** GEQ patching source*

>> To open the Graphic EQs screen

Do one of the following:

- At the GUI, choose **home ▸ *Rack Units* ▸ *Graphic EQs.***
- In the navigation zone, press the **effects/graphics** access button twice.

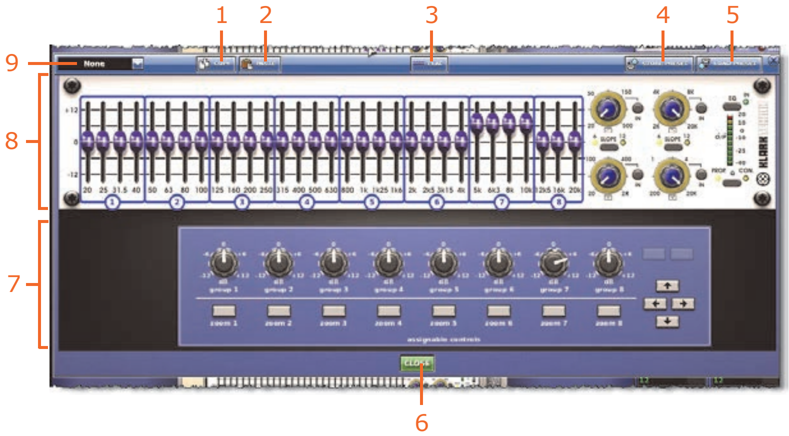
>> To open a GEQ rack

Do one of the following:

- In the **Graphic EQs** screen, click on the desired unit.
- Press **ENTER** in the **assignable controls** section.

About the GEQ window

In the centre of the GUI screen, the GEQ window shows the selected GEQ's front panel. This gives you full control of the GEQ via the GUI controls in the navigation zone by using the trackball and left and right buttons. Below the GEQ is an **assignable controls** panel that lets you select and control the GEQ faders (singly or in groups) and the controls on the right.



GEQ window

Item	Element	Description
1	**COPY** button	Copy and paste function button (see Chapter 18 "Copy And Paste" on page 167).
2	**PASTE** button	Copy and paste function button (see Chapter 18 "Copy And Paste" on page 167).
3	**FLAT** button	Sets all of the GEQ's faders to 0dB.
4	**STORE PRESET** button	See Chapter 24 "User Libraries (Presets)" on page 215.
5	**LOAD PRESET** button	See Chapter 24 "User Libraries (Presets)" on page 215.
6	**CLOSE** button	Closes the GEQ window.
7	**assignable controls** panel	See Chapter 19 "Assignable Controls" on page 169.
8	GEQ panel	Shows the front panel of the GEQ (see "GEQ front panel features" on page 122.
9	Drop-down list	For selecting the source of the GEQ.

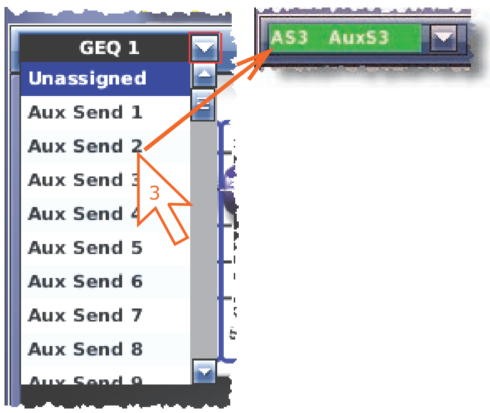
>> To open a GEQ unit window

In the **Graphic EQs** screen, click on a non-control area of the unit you want.

>> To patch a source to a GEQ

1. Open the window of the GEQ.
2. Open the GEQ source drop-down list. (An unpatched GEQ will have "None" displayed in the text field.)
3. In the drop-down list, click the source you want. For example, "Aux Send 3". The new patching assignment will appear in the source name field (as shown right) and in the border on the left of GEQ panel.
4. Click **CLOSE** to accept the change and close the GEQ's window.

The new GEQ patching source assignment will appear in the virtual rack.

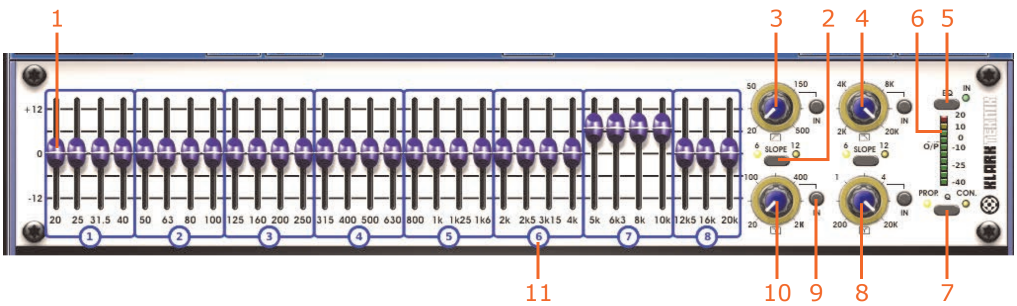


GEQ front panel features

The front panel of the GEQ, as displayed on the GUI, represents the Klark Teknik DN370 Graphic Equaliser. It includes a graphic EQ (faders) section and a filters section.

Thirty one faders provide fine adjustment of each frequency band. 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 an **EQ** switch, 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 a control knob on the GUI screen. To audition the effect of the filters, use either the **EQ** switch (which will also bypass the GEQ) or the individual filter switch.



GEQ front panel

Item	*Control*	*Function*
1	Fader (31-off).	Adjusts signal level.
2	**SLOPE** button	Switches the high or low pass filter between 6dB and 12dB. Adjacent yellow LEDs indicate the active mode.
3	High pass filter control knob	Adjusts the cut off frequency, which is continuously variable from 20Hz to 500Hz.
4	Low pass filter control knob	Adjusts the cut off frequency, which is continuously variable from 2kHz to 20kHz.

Item	Control	Function
5	**EQ** button	Selects the EQ. The adjacent green **IN** LED shows the EQ is on (illuminated) or is being bypassed (extinguished).
6	10-segment meter	Shows the incoming signal level and is pre-EQ (but post-gain control). The LED functions are: two red LEDs illuminate when signal has exceeded +20dBu; two yellow LEDs illuminate when signal level exceeds 0dB (range is between 0dB to +20dB); and the top five green LEDs encompass the signal level range of between 0dB and -40dB, while the bottom one illuminates when the signal has exceeded -40dB.
7	**Q** button	Selects proportional Q (**PROP.**) or constant Q (**CON.**) modes.
8	Notch filter control knob	Adjusts the position of the notch filter within the range 200Hz to 20kHz.
9	**IN** button	Switches the respective high pass/low pass/notch filter in/out.
10	Notch filter control knob	Adjusts the position of the notch filter within the range 20Hz to 2kHz.
11	Fader group ID number	See "Controlling an internal effect/GEQ" on page 173.

Configuring the number of GEQs (and effects)

GEQ (and effects) configuration is a GUI-only operation. We recommend that you configure the number of GEQs and effects before you start using the PRO1 Control Centre.

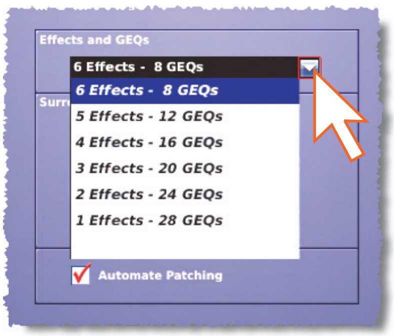
>> To configure the PRO1 Control Centre with the number of GEQs and effects

1 At the GUI, choose **home** ▸ ***Preferences*** ▸ ***General.***

2 Click the **Show** tab to open the **Preferences Show** window.

3 In the **Effects and GEQs** section, open the drop-down list (as shown right).

4 Click on your desired option to select it. For example, click the **3 Effects - 20 GEQs** option to have three internal effects and 28 GEQs.



Copying settings between GEQs

You can copy and paste all the settings of one GEQ to another.

>> To copy the settings of one GEQ to another GEQ

1. In the GEQ rack of the **Graphic EQs** screen, open the window of the GEQ that you want to copy the settings from.
2. In the GEQ window, click **COPY.**
3. Close the GEQ window and then open the window of the GEQ that you want to paste the settings to.
4. In the GEQ window, click **PASTE.**

Chapter 16: Internal Effects

This chapter describes the internal effects of the PRO1. Initially, it explains how to use the PRO1 Control Centre to operate the effects and then details all of their available control functions and their use.

Overview of the internal effects

The **Effects** screen manages up to six user-assignable effects devices, which are a 'bundle' of onboard creative audio effects that provide onboard facilities where outboard effects units would have traditionally been required. The effects are displayed on the screen in a 'virtual' eight-unit rack.

*Typical **Effects** screen (configured for six effects)*

All of the points in the control centre's signal flow where outboard effects can be inserted, such as auxes and returns or insert points on input channels, audio subgroups, mix and master buses, can be patched to effects in the internal effects as well as external world XLRs.

Rack unit number allocation

Each unit position in the rack is allocated a rack number that is recognised by the PRO Series Live Audio System (shown right).

About the effect window

Similarly to the GEQ window on the GUI, the effect window shows a screen-width version of the selected effect, which gives you full control of the effect via the trackball and left and right buttons in the navigation zone. A 'virtual' **assignable controls** panel below lets you select and operate the controls of the effect.

Effect window

Item	*Element*	*Description*
1	**CHANGE DEVICE TYPE** button	Lets you select a different effect.
2	**STORE PRESET** button	User library function button (see Chapter 24 "User Libraries (Presets)" on page 215).
3	**LOAD PRESET** button	User library function button (see Chapter 24 "User Libraries (Presets)" on page 215).
4	**CLOSE** button	Closes the effect window.
5	**assignable controls** panel	See Chapter 19 "Assignable Controls" on page 169.

Item	Element	Description
6	Effect panel	For details of the front panel for each effect, refer to the effect sections later on in this chapter.
7	Effect name	For naming the effect.

>> To open an effect window

In the **Effects** screen, click on a non-control area of the effect you want.

Working with the effects

There are a number of ways of handling the effects, such as setting up, configuration and operation, all of which involve the use of the GUI. However, most of these methods can also be carried out using the I zone; see Chapter 19 "Assignable Controls" on page 169.

>> To open the Effects screen

Do one of the following:

- At the GUI, choose **home** ▸ ***Rack Units*** ▸ ***Effects.***
- In the navigation zone, press the **effects/graphics** access button.

>> To operate an effect control

The method for controlling the effect controls is the same as for any control on a GUI screen. For details, see “About GUI operation” on page 38.

>> To configure an effect

Similarly to the input channels, output channels, groups etc., you can change the name of an effect and the background colour of its text field, as it appears on the GUI (see “Configuring VCA/POPulation groups” on page 73).

>> To change an effect type

For details, see “To add an effect to the effects rack” on page 77.

>> To route an effect

Effects are patched via the Patching screen. For details, see Chapter 8 "Patching" on page 45.

Effect programs

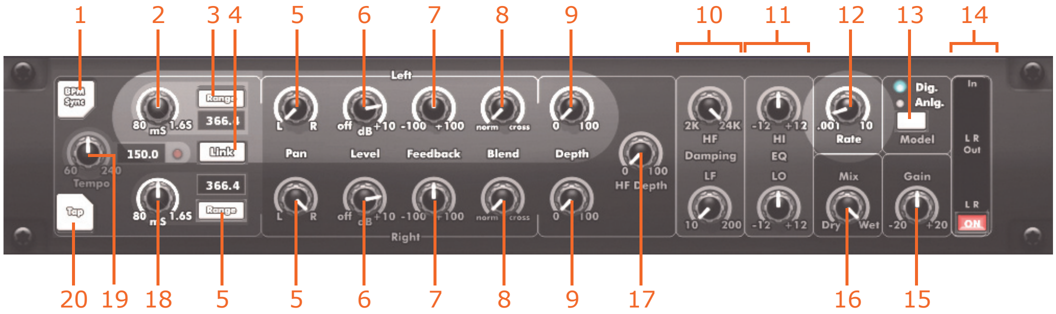
Some types of effect have associated factory presets and user-configurable programs, which you can load within the effect (these are also stored in a show file). You can also save all of the controls from one or more effects in a user preset, which will then contain information about their settings, 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 24 "User Libraries (Presets)" on page 215.

Delay effect

The delay effect provides simple delay line based effects. Delay times can be specified manually or by means of a 'tempo-tap' button. Three-mode delay algorithm:

- **Single** — one delay tap (mono or stereo processing).
- **Dual** — two delay lines (stereo insert only).
- **Ping-pong** — two delay lines with cross feedback.



Item	Control	Function
1	**BPM Sync** button	Activates BPM (TAP TEMPO) mode.
2	Left channel delay time control knob	For entering the desired delay time. Value is shown immediately below, in ms or note duration.
3	**Range** button	Selects one of three delay time ranges (1-25ms, 10-200ms or 80-1600ms). Value is displayed immediately below/above button.
4	**Link** button	Links delay time of left and right channels.
5	**Pan** control knob	Pans channel between L (left) and R (right) outputs.
6	**Level** control knob	Adjusts the output level. Range is from off to +10dB.
7	**Feedback** control knob	Adjusts the amount of negative/positive feedback applied to delay. Controls the number of repeats. Range is from -100% to +100%.
8	**Blend** control knob	Adjusts the feedback blend from **norm** to **cross.**
9	**Depth** control knob	Adjusts depth of delay modulation. Range is from 0 to 100.
10	**Damping** section	Contains a **HF** control knob that adjusts the HF attenuation of delay repeats and an **LF** control knob that adjusts the LF attenuation of delay repeats.
11	**EQ** section	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.
12	**Rate** control knob	Adjusts rate of delay modulation. Range is between 0.001Hz and 10Hz, with 0.7Hz at top dead centre.
13	**Model** select button	Selects digital or analogue delay model. Current selection is shown by the illumination of one of the LEDs above (**Dig.** or **Anlg.**).
14	**ON** switch	Switches the delay effect on/off.

Item	Control	Function
15	**Gain** control knob	Adjusts the amount of gain between -20 and +20, with 0 at top dead centre.
16	**Mix** control knob	Adjusts the mix between dry (0%) wet (100%).
17	**HF Depth** control knob	Adjusts depth of HF damping modulation. Range is 0 to 100, with 50 at top dead centre.
18	Right channel delay time control knob	For entering the desired delay time (in ms).
19	**Tempo** control knob	Adjusts the tempo in tempo mode. Range is from 60 to 240 beats per minute (bpm).
20	**Tap** button	For manually tapping the tempo once the unit is in **BPM Sync** (TAP TEMPO) mode.

Virtual DN780 reverb effect

The virtual DN780 reverb provides emulation of the vintage Klark Teknik DN780 Digital Reverberator/Processor 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.

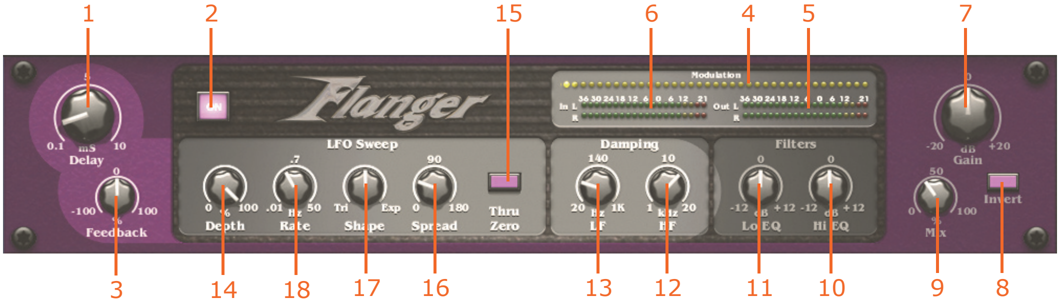
Item	Control	Function
1	**LF** (low frequency) control knob	**REVERBERATION** section control that adjusts the decay time at the low end of the reverb spectrum. Range is from -7 to +7.
2	**HF** (high frequency) control knob	**REVERBERATION** section control that adjusts the decay time at the high end of the reverb spectrum, which sets the absorption characteristic of the simulated space. Range is from -7 to +7.
3	**ROOMSIZE** control knob	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.
4	**DECAY** control knob	**REVERBERATION** section control that sets the overall (mid-band) reverberation decay time. Range is from 0.1 to 18 seconds, depending on room size.
5	**LEVEL** control knob	**REFLECTIONS** section 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.

Item	*Control*	*Function*
6	**PATTERN** control knob	**REFLECTIONS** section control that 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	**PRE DELAY** control knob	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.
8	List of algorithms and algorithm select button	These algorithms emulate the ones on the original DN780. Use the select button to scroll through the list to select the one you want.
9	**MIX** control knob	Controls the **DRY**/**WET** output mix and ranges from 0% to 100%, respectively.
10	**ST** stereo input button	Enhances original algorithm to provide stereo input.
11	Parameter display panel	Shows the current settings for the selected algorithm.
12	**IN** button	Switches in the Virtual DN780 Reverb effect.
13	**LEVEL** control knob	**AUDIO** section control for adjusting the input level. Range is from -∞ to +6dB, with 0dB at top dead centre. This should be set to illuminate the -3dB LED on the input headroom indicator during loud program passages.
14	**IN** button	**AUDIO** section **MUTES** control for removing feed to the reverberation section, enabling the decay qualities of the chosen setting to be confirmed.
15	**REV** button	**AUDIO** section reverb **MUTES** control for providing a rapid means of removing unwanted sounds.
16	Input headroom indicator	**AUDIO** section meters, which comprise two dual-column peak reading LED meters, ranging from 0dB to -27dB in 3dB steps. Each column consists of 10 coloured LEDs. The red LED illuminates at 3dB before the clipping point, which also provides an over-range warning for the arithmetical processor.

The parameter controls give accurate adjustment of all reverberation parameters and allow the engineer to create unique acoustic environments of virtually any type.

Flanger effect

The flanger effect consists of one or, if configured as stereo, two dual-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.

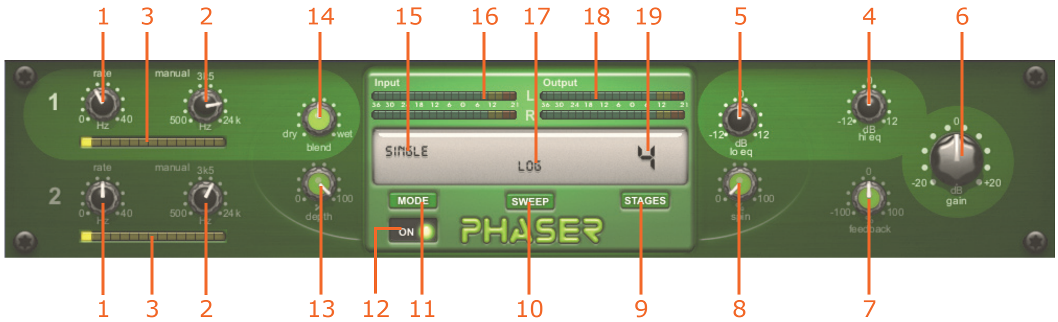


Item	Control	Function
1	**Delay** control knob	Adjust length of modulated delay line in milliseconds. In 'thru-zero' mode, also sets the delay of the dry path. Range is 0.1 to 10, with 5 at top dead centre.
2	**ON** button	Switches the flanger effect on and off. Illuminates when power is on.
3	**Feedback** control knob	Adjusts the amount of negative/positive feedback applied to the flanger. Range is from -100% to +100%, with 0% at top dead centre.
4	Modulation meter	A single row of 37 yellow LEDs are used to show the modulation.
5	**Out** meters	Two rows of 15 green LEDs, one each for L (left) and R (right).
6	**In** meters	Two rows of 15 green LEDs, one each for L (left) and R (right).
7	**Gain** control knob	Adjusts the signal level in dB. Range is from -20dB to +20dB, with 0dB at top dead centre.
8	**Invert** switch	Inverts the wet signal.
9	**Mix** control knob	Adjusts the mix between dry (0%) wet (100%).
10	**Hi EQ** control knob	**Filters** section control for adjusting the amount of HF (high EQ) cut or boost applied to the effect output (in dB). Range is -12dB to +12dB with 0dB at top dead centre.
11	**Lo EQ** control knob	**Filters** section control for adjusting the amount of LF (low EQ) cut or boost applied to the effects output (in dB). Range is -12dB to +12dB with 0dB at top dead centre.
12	**HF** control knob	**Damping** section control for adjusting the high frequency (kHz) tuning of flanger feedback. Range is 1kHz to 20kHz, with 10kHz at top dead centre.
13	**LF** control knob	**Damping** section control for adjusting the low frequency (kHz) tuning of flanger feedback. Range is 20Hz to 1kHz, with 140Hz at top dead centre.

Item	*Control*	*Function*
14	**Depth** control knob	**LFO Sweep** 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%.
15	**Thru Zero** switch	**LFO Sweep** section control for selecting 'thru zero' or normal mode. Illuminates to indicate switch is on.
16	**Spread** control knob	**LFO Sweep** section control for setting the relative phase of left/right modulation. Range is 0 to 180, with 90 at top dead centre.
17	**Shape** control knob	**LFO Sweep** section control for adjusting the shape of the modulation waveform. Range is from Tri (triangle) to Exp (exponential).
18	**Rate** control knob	**LFO Sweep** section control for adjusting the rate of modulation (Hz). Range is between 0.01 and 50, with 0.7 at top dead centre.

Phaser effect

The phaser effect consists of one, or if configured for dual operation, two stereo phasers connected in serial/parallel according to mode setting.

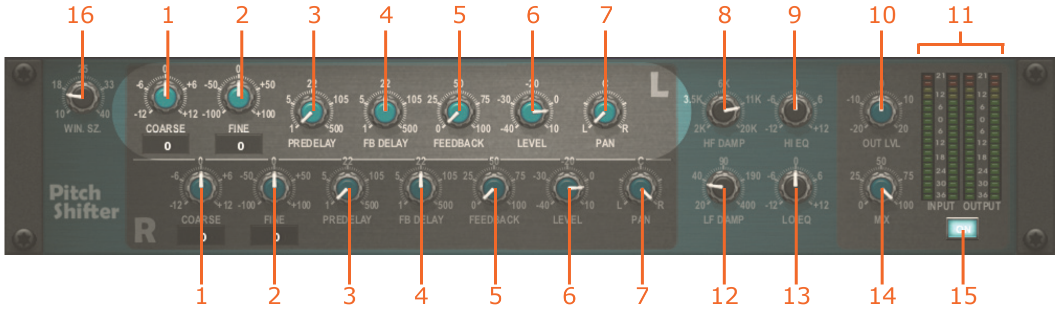


Item	*Control*	*Function*
1	**rate** control knob (channels 1 and 2)	Controls the rate of modulation in Hz. Range is from 0Hz and 40Hz.
2	**manual** control knob (channels 1 and 2)	Sets the sweep offset or performing manual sweep. Range is from 500Hz to 24kHz, with 3k5Hz (3,500Hz) at top dead centre.
3	Modulation meter	A single row of 15 yellow rectangular LED segments are used to show the modulation on each channel.
4	**hi eq** control knob	Adjusts the amount of HF (high EQ) cut or boost applied to the effect output (in dB). Range is from -12dB to +12dB with 0dB at top dead centre.
5	**lo eq** control knob	Adjusts the amount of LF (low EQ) cut or boost applied to the effects output (in dB). Range is from -12dB to +12dB with 0dB at top dead centre.
6	**gain** control knob	Adjusts the signal level (dB). Range is from -20dB to +20dB, with 0dB at top dead centre.

Item	*Control*	*Function*
7	**feedback** control knob	Adjusts the amount of negative/positive feedback applied to the phaser. Range is from -100% to +100%, with 0% at top dead centre.
8	**spin** control knob	Adjusts the amount of relative phase of left/right modulation. Range is from 0% to 100%.
9	**STAGES** button	Selects the number of all pass stages, which sets the number of notches in the frequency response.
10	**SWEEP** button	Sets the modulation waveform shape.
11	**MODE** button	Selects the operating mode: single; dual series; dual parallel; linked series; or linked parallel. When linked, modulation of phasers 1 and 2 are linked.
12	**ON** button	Switching the phaser effect on and off. Illuminates to indicate effect is on.
13	**depth** control knob	Controls the intensity of the effect by setting the depth of modulation. Range is from 0% to 100%.
14	**blend** control knob	Adjusts the mix between dry (0%) wet (100%).
15	Mode	Displays the current mode, selected by the **MODE** switch.
16	**Input** meters	Two rows of 15 green LEDs — one row each for L (left) and R (right) — comprise the input meters.
17	Sweep	Displays the sweep, selected by the **SWEEP** button.
18	**Output** meters	Two rows of 15 green LEDs — one row each for L (left) and R (right) — comprise the output meters.
19	Number of all pass stages	Displays the number of all pass stages, selected by the **STAGES** button.

Pitch shifter effect

The pitch shifter effect is a sound processing device for changing the pitch of an audio signal without changing its duration.



Item	Control	Function
1	**COARSE** control knob	Adjusts the pitch shifting amount in whole tones. Range is from -12 to +12, with 0 at top dead centre. The numerical value is shown underneath.
2	**FINE** control knob	Fine tunes the pitch shifting in 1% increments of a whole tone. Range is from -100 to +100, with 0 at top dead centre. The numerical value is shown underneath.
3	**PREDELAY** control knob	Sets the delay time before the pitch shift. Range is from 1 to 500, with 22 at top dead centre.
4	**FB DELAY** control knob	Sets the delay time on the feedback loop. Range is from 1 to 500, with 22 at top dead centre.
5	**FEEDBACK** control knob	Sets the amount of feedback (output fed back to input) in %. For more details, see "Feedback" on page 135. Range is from 0 to 100, with 50 at top dead centre.
6	**LEVEL** control knob	Sets the output level of the individual channel. Range is from -40 to -10, with -20 at top dead centre.
7	**PAN** control knob	Adjusts the position of the individual channel signal in the unit's stereo output.
8	**HF DAMP** control knob	Adjusts the HF attenuation of delay repeats. Range is from 2k to 20k, with 6k at top dead centre.
9	**HI EQ** control knob	Boosts/attenuates high frequencies. Range is from -12 to +12, with 0 at top dead centre.
10	**OUT LVL** control knob	Sets the overall output level. Range is from -20 to +20, with 0 at top dead centre.
11	**INPUT** and **OUTPUT** meters	Shows the input/output signal levels on dual 20-segment meters (-36dB to +21dB).
12	**LF DAMP** control knob	Adjusts the LF attenuation of delay repeats. Range is from 20 to 400.
13	**LO EQ** control knob	Boosts/attenuates low frequencies. Range is from -12 to +12, with 0 at top dead centre.
14	**MIX** control knob	Controls the balance between dry signal and effect. Range is from 0 to 100, with 50 at top dead centre.

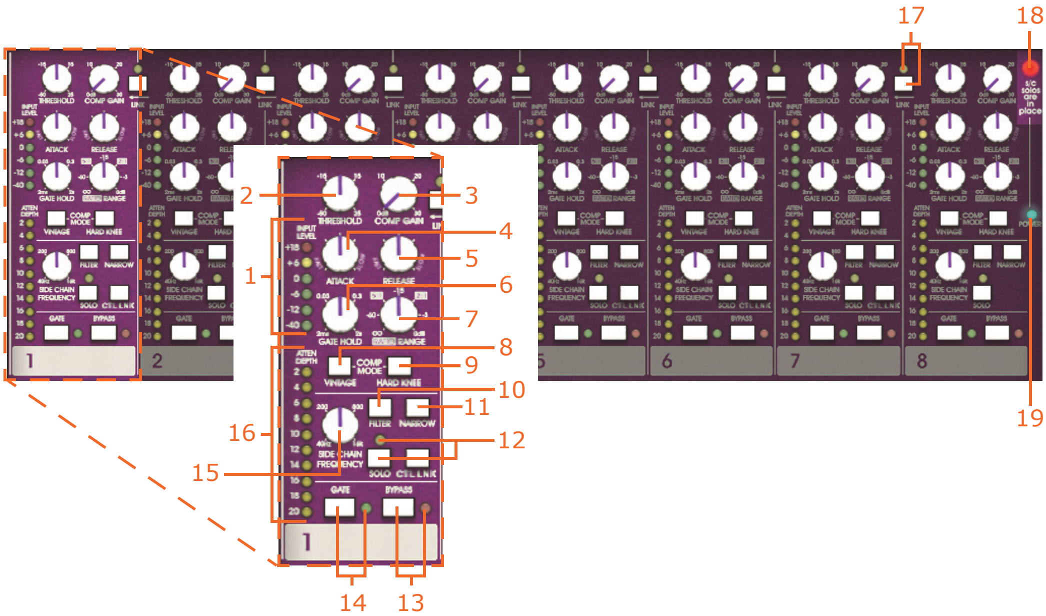
Item	Control	Function
15	**ON** switch	Switches pitch shifter effect on/off.
16	**Win Sz** control knob	Sets the window size used by the pitch shift algorithm. High values result in larger latency, but better quality for low frequency or polyphonic audio source. Low values produce low latency and work best on monophonic audio source.

Feedback

The pitch shifter accepts the input signal and then delays it and plays it back at a different speed, so that its output is delayed and pitch shifted. When this output is fed back into the pitch shifter, further delays and more pitch shifting occur. This can lead to some strange effects, such as feedback.

SQ1 Dynamics effect

The SQ1 dynamics provides an emulation of the Klark Teknik Square ONE Dynamics Processor, which is an eight-channel analogue dynamics processor.



Item	Control	Function
1	**INPUT LEVEL** meter (dBu)	Dedicated peak reading, six-segment audio level LED meter that monitors the input XLR level at all times, no matter how the controls are set.
2	**THRESHOLD** control knob	Adjusts the operating point for the compressor or gate, depending on which is selected.
3	**COMP GAIN** control knob	Adjusts compressor gain — also known as the make up gain — so that the level of the outgoing compressed signal can be matched to the incoming uncompressed signal.
4	**ATTACK** control knob	Adjusts the time it takes the compressor to respond or the gate to open after an over threshold signal, depending on which is selected.

Item	*Control*	*Function*
5	**RELEASE** control knob	Adjusts the time it takes the compressor to recover or the gate to close after the programme material falls back below threshold, depending on which is selected.
6	**GATE HOLD** control knob	Minimises chattering in conjunction with internal hysteresis (see "Intelligent threshold shift (i-TS)" on page 137). Once the signal is detected as below threshold, this control defines a waiting period before the gate starts to close.
7	**RATIO/RANGE** control knob	Adjusts the amount of compression (ratio) applied to signals above threshold or the gain reduction (range) applied to signals below threshold, depending on whether the compressor or gate is selected.
8	**VINTAGE** switch	See item 9 "HARD KNEE switch".
9	**HARD KNEE** switch	Used in combination with the **VINTAGE** switch to provide four compressor operating modes (see "Compressor modes of operation" on page 137)
10	**FILTER** switch	Used in conjunction with the **NARROW** switch and **SIDE CHAIN FREQUENCY** control knob to provide a variable frequency band pass filter that acts on the side chain signals. This switch enables (switch = on) or bypasses (switch = off) the filter.
11	**NARROW** switch	Changes the bandwidth from wide (switch = off) to narrow (switch = on). See Item 10 "FILTER switch".
12	**SOLO** switch with yellow LED	Enabling the **SOLO** switch with SIP disabled, sends post-filter side chain audio to the solo bus output. Please be aware that if you enable the **SOLO** switch with SIP enabled, the side chain signal is routed directly to the output.
13	**BYPASS** switch with red LED	Enables a bypass condition whereby the VCA remains in-circuit, fixed at unity gain. The gain reduction meter continues to operate with **BYPASS** enabled.
14	**GATE** switch with green LED	This switch selects channel operation as compressor mode (switch = disabled) or gate mode (switch = enabled).
15	**SIDE CHAIN FREQUENCY** control knob	Selects the frequency at which the band pass filter acts on the side chain signals (see Item 10 "FILTER switch"). The filter can be used to make the compression frequency selective. Additionally, there is a solo function (see Item 12 "SOLO switch with yellow LED") that places the filtered side chain onto the compressor's solo bus output or, optionally, the main output (SIP mode).
16	**ATTEN DEPTH** meter (dB)	Dedicated attenuation depth (gain reduction) meter that displays the amount of attenuation being applied by the compressor or gate, depending on which mode is selected. The meter comprises 10 LEDs, scaled from -2dB to -20dB in 2dB increments. The compressor gain control will not affect the gain reduction meter reading.

Item	*Control*	*Function*
17	**LINK** switch with yellow LED	Each adjacent pair of channels has an intermediate **LINK** switch that, when enabled, links them together (see "Stereo and multiple channel operation - linking" on page 138).
18	**s/c solos are in place** red LED	**Illuminates to indicate that the master solo in place is enabled. When enabled, all the SOLO switches change function so that no signal will be sent to the master solo output bus, and the side chain signal will be sent directly to the output XLR on the same channel when the SOLO button is pressed, thus replacing the dynamics output signal.**
19	Blue **POWER** LED	Illuminates to indicate that the effect is enabled.

Intelligent threshold shift (i-TS)

i-TS operates in conjunction with **GATE HOLD** to reduce chattering within the gate. Chattering is the undesirable condition that occurs when signals — especially low frequency ones — are very close to the gate threshold. In this situation the gate can become indecisive and repeatedly open and shut on the programme. i-TS ensures that the gate remains open by automatically adjusting the threshold downwards as the signal goes over threshold. When the signal actually falls below threshold (that is, the temporarily adjusted threshold) the i-TS resets, ready for the next gate opening. This improved decision-making ensures that gating is rock solid and attacks start instantly and consistently, even on signals that go slightly over threshold.

i-TS is particularly useful for low frequency material and instruments with oscillating or unpredictable decay envelopes.

Compressor modes of operation

The compressor has been designed to operate as a root-mean-square (RMS) sensing type or peak sensing type compressor. Basically, this means that the unit's circuitry responds to either the effective average value of the signal's waveform or its peak value. The peak sensing type mode of compression has been designed to emulate the qualities and performance of 'vintage' valve-type compressors.

The two main compressor modes each operate in one of two modes, hard knee or soft knee. Thus giving the four following compressor modes, selectable using the **VINTAGE** and **HARD KNEE** switches in combination.

Compressor soft knee and RMS (default setting)

With both the **VINTAGE** and **HARD KNEE** pushbutton switches switched off, the compressor behaves in the default soft knee and RMS mode. This gives the slowest and most subtle feel to the compressor envelopes. The soft knee curvature combines with the adaptive RMS attack and release times to produce gentle envelope curvatures that are ideal for compressing sung vocals, but which can still be aggressive enough to limit transients when needed. The knee curvature also slightly reduces the adaptive nature of the RMS detection, thus providing a little more manual control of the envelope timings than is the case with the compressor set for hard knee and RMS.

Compressor hard knee and RMS

With only **HARD KNEE** switched on, 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 soft knee is still retained, keeping the sound reasonably natural but without any modification of the envelope occurring. Thus, attacks are a little more aggressive but the adaptive nature of the RMS detection is still allowed to operate to its fullest extent. This mode is good for natural sounding limiting of speech.

Compressor hard knee and vintage

With the **VINTAGE** and **HARD KNEE** switches both switched on, the compressor operates with more precise envelope control and, again, with a more defined transition between under and over threshold. This mode uses faster peak sensing (not RMS), like many older compressor designs with exponential attack and release. This produces aggressive compression that gives good fast control and/or limiting of extremely dynamic material. It can also be used to add colour to low frequency signals, making it ideal for controlling instruments such as the bass guitar.

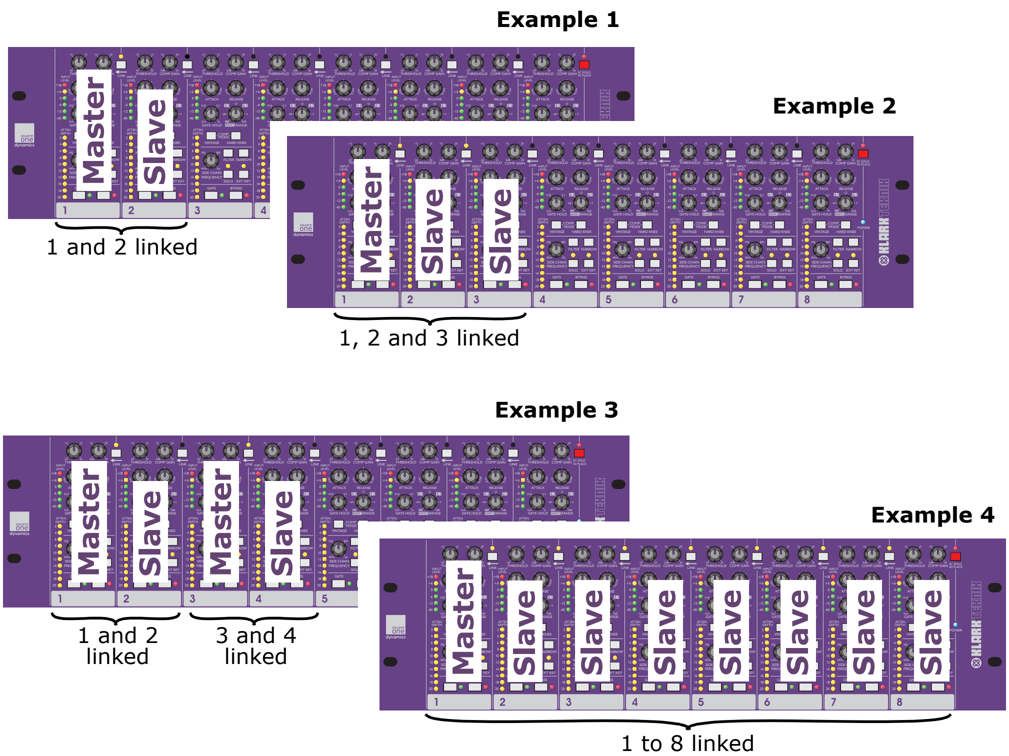
Compressor soft knee and vintage

With only **VINTAGE** switched on, the compressor employs a dual time constant, linear attack profile. The soft knee blurring of threshold occurs once more, as in RMS mode, although the effect is greatly accentuated. 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 (signals go further over threshold) the soft knee effect reduces, gradually returning manual control of the attack and release times to optimise capture of larger transients etc. Thus, like the RMS modes, this compressor mode is very adaptive, making set-up of the envelope controls relatively easy. However, the peak sensing increases harmonic overtones, adding a valve-like brightness and sparkle to the programme, and producing extremely natural and lively sounding compression of acoustic instruments.

Stereo and multiple channel operation - linking

Using the intermediate **LINK** switches, adjacent channels can be linked, in all modes, for stereo or multi-channel operation. Linked channels form a group, the lowest numbered channel in the group being the master and the other group members are slaves. The master channel's settings override those of the slaves with the exception of bypass, solo, ext key and side chain filter, which still act independently. (The slave filters, ext key and solo still function when linked because the slave channel side chain is summed with the master and any other slaves in the group.)

You can have more than one group at any time; Figure 6 "Examples of channel linking" shows some typical group configurations and illustrates that the master is always the lowest numbered channel within the group.



Examples of channel linking

The audio input to the master channel and any slaves in the link are all used to control the action of the dynamic processor. The channel with the highest signal level will have the most effect on the linked group. As all the VCA controls are also linked together, the attenuation or gain applied to the linked channels in the group will always be identical. Each slave's gain reduction meter (**ATTEN DEPTH**) will track its group master channel exactly.

3-Band compressor effect

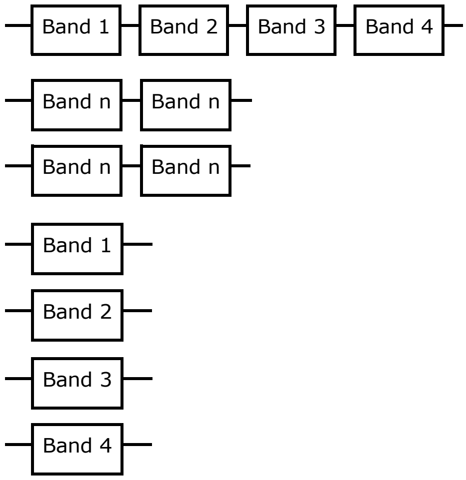
The 3-band compressor effect is for applying different compression profiles to different areas of the audio spectrum.

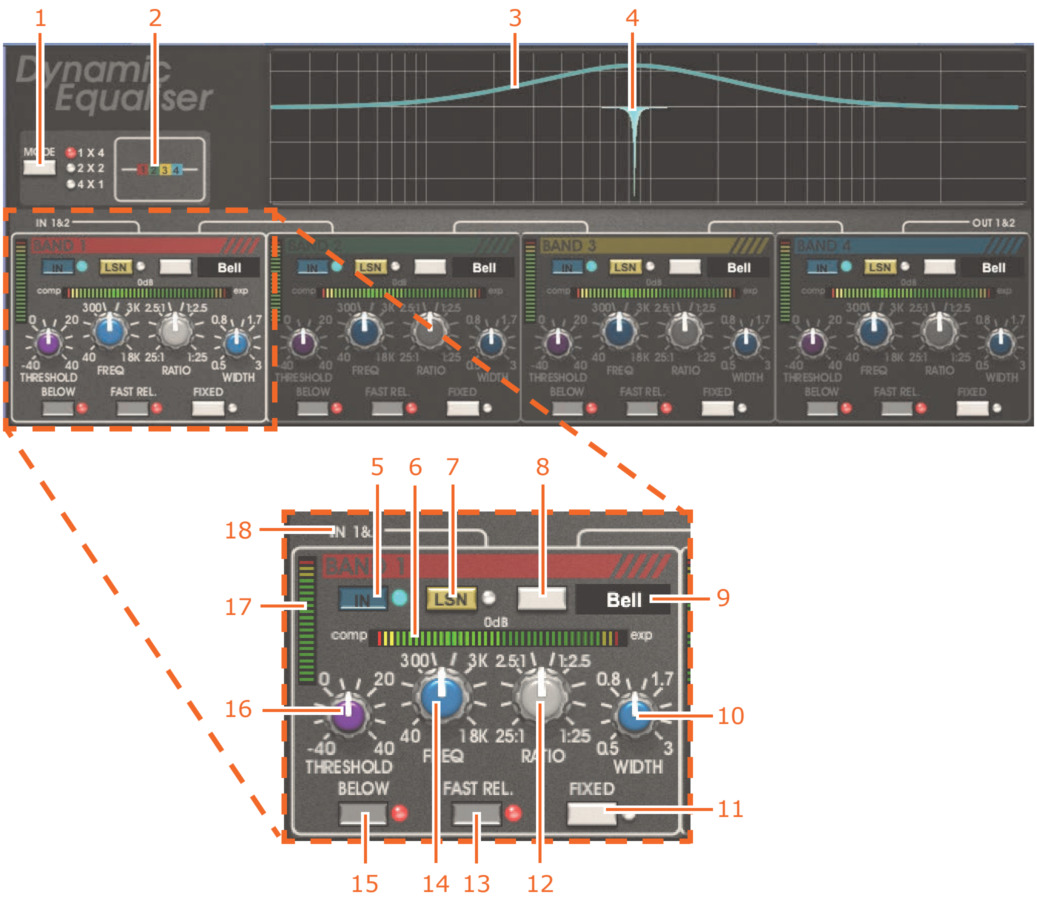
Item	*Control*	*Function*
1	**In** button	Switches the stereo 3-band compressor in/out. It has an adjacent LED (yellow) for in/out indication.
2	**Threshold** control knob	This control is in the **Lo**, **Mid** and **Hi** sections, and is used for setting the threshold at which compression begins. Range is from -50dB to +25dB.
3	**Ratio** control knob	This control is in the **Lo**, **Mid** and **Hi** sections, and is used for setting the compression ratio. Range is from 25:1 to 1:1.
4	**Make-Up** control knob	This control is in the **Lo**, **Mid** and **Hi** sections, and is used for boosting the output level of the compressed signal. Range is from 0dB to +24dB.
5	**Release** control knob	This control is in the **Lo**, **Mid** and **Hi** sections, and is used for setting the release time of the compressor. Range is from 0.05sec to 3sec.
6	**Attack** control knob	This control is in the **Lo**, **Mid** and **Hi** sections, and is used for setting the speed of the onset of compression once the threshold has been exceeded. Range is from 0.2ms to 50ms.
7	**SOLO** button	This control is in the **Lo**, **Mid** and **Hi** sections, and is used for listening to the compressor's sidechain filter. It has an adjacent LED (yellow) for on/off indication.
8	Three-section meter	This has individual meter elements for input (**In**), gain reduction (**GR**) and output (**Out**) positions of the three bands.
9	**Soft Link** button	Links the controls with relative offsets. It has an adjacent LED (yellow) for on/off indication.
10	**Global Knee** control knob	Sets the rate of transition across the threshold. Range is from 4 to 40.
11	**Lo Mid Freq** control knob	Sets the crossover point between the Lo and Mid compressors. Range is from 40Hz to 1kHz.
12	**Hi Mid Freq** control knob	Sets the crossover point between the Mid and Hi compressors. Range is from 640Hz to 16kHz.

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. Flexible routing options allow for the following configuration modes:

- One chain of stereo 4-band processing.
- Two chains of stereo 2-band processing.
- Four chains of stereo single-band processing.





Item	*Element*	*Function*
1	**MODE** button	Selects the routing configuration. Audio routing paths are illustrated by item 18.
2	Band selection indicator	Shows the selected routing configuration mode.
3	Limit curve	Shows the minimum cut/maximum boost EQ response for the selected band.
4	Dynamic curve	Shows the real time dynamic EQ response curve. This curve will vary between flat (no EQ) and the outer curve (full EQ) depending on the signal level, ratio and threshold settings.
5	**IN** button and LED indicator	Switches the individual band on/off.
6	**comp**/**exp** meter	Shows the current cut/boost of the selected EQ band.
7	**LSN** button and LED indicator	Sidechain listen button that routes the bandpass filtered sidechain signal to the unit output.
8	EQ type	Selects the type of equaliser (shown in item 9) from **Bell**, **Low Shelf**, **High Shelf** and **Full Band.**
9	Name field	Shows the currently selected EQ type.
10	**WIDTH** control knob	Sets the bandwidth of the EQ band and sidechain filter for the selected band.
11	**FIXED** button and LED indicator	Forces selected band to use a fixed EQ gain, so that when it is enabled it behaves as a static EQ. This is useful for previewing the effect of the EQ curve.
12	**RATIO** control knob	Sets the ratio of the compression or expansion applied to the selected band. The centre position produces a ratio of 1:1, which will have no effect. As the control is turned anti-clockwise, more compression is applied, up to a ratio of 25:1. If the control is turned clockwise from the centre position, expansion is gradually applied up to the ratio of 1:25.
13	**FAST REL.** button and LED indicator	Enables fast envelope release setting.
14	**FREQ** control knob	Sets the centre frequency of the selected EQ band.
15	**BELOW** button and LED indicator	Sets whether the compressor/expander operates above (off) or below (on) the threshold (see Figure 9 "Transfer characteristics" on page 143).
16	**THRESHOLD** control knob	Sets the threshold level.
17	Meter	Shows the level of the sidechain signal relative to the threshold setting, that is, it shows the signal level within the frequency region selected by the frequency (**FREQ**) and width (**WIDTH**) controls.
18	Band I/O	Shows the audio routing path between inputs, outputs and the four EQ bands.

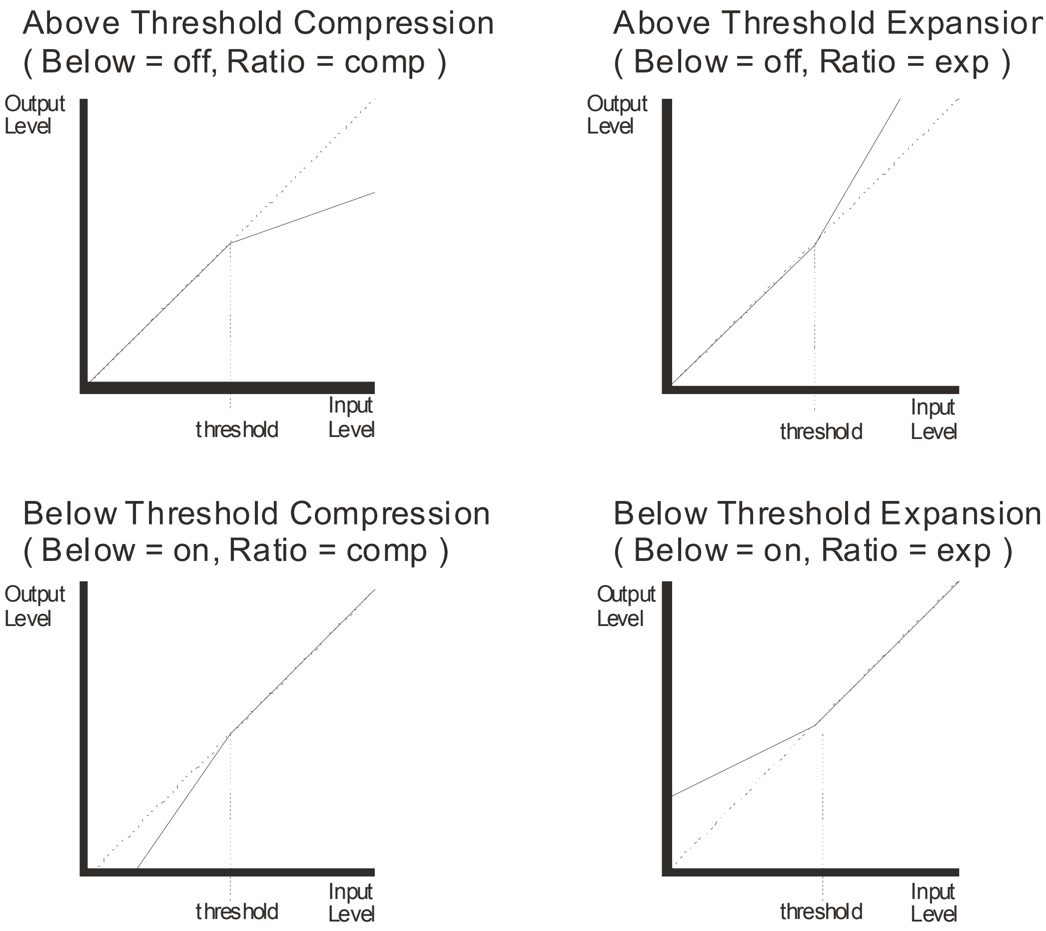


Figure 9: Transfer characteristics

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 diffuse early reflections with the additional flexibility of separate reverb tail level and decay control.

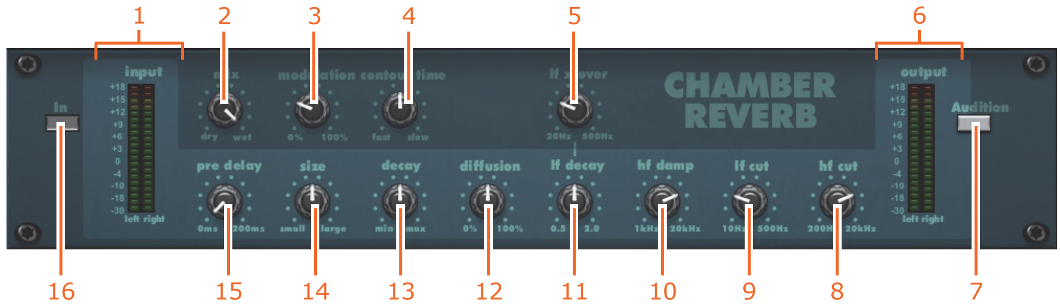
Reflective surface materials and air absorption properties can be simulated by adjusting the high and low frequency cut amount and high frequency damping.

Item	*Element*	*Function*
1	**input** meter	Two adjacent 11-LED columns — one each for left and right — comprise the input meters.
2	**Mix** control knob	Adjusts the dry/wet signal ratio.
3	**Modulation** control knob	Specifies the combined rate and depth of modulation applied the reverb tail.
4	**In** switch	Switches the Plate Reverb effect in/out.
5	**Audition** button	This momentary-action button triggers a short internally generated sound to aid reverb evaluation (as a check).
6	**output** meter	Two adjacent 11-LED columns — one each for left and right — comprise the output meters.
7	**HF Cut** control knob	High frequency cut control knob applies a 6dB/Oct low pass filter to the input signal, in the range 200Hz to 20kHz.
8	**LF Cut** control knob	Low frequency cut control knob applies a 6dB/Oct high pass filter to the input signal, in the range 10Hz to 500Hz.
9	**HF Damp** control knob	High frequency damping, progressively reduces the high frequency content over time, in the range 1kHz to 20kHz.
10	**Tall Gain** control knob	Increases the level of the reverb tail, between off and 0dB.
11	**Diffusion** control knob	Increases the density of both the early reflections and reverb tail, between 0 and 100%.
12	**Decay** control knob	Adjusts the decay time relative to the room size, from minimum to maximum.
13	**Size** control knob	Specifies the room size (also affects decay), from small to large.
14	**Predelay** control knob	Specifies the time before the reverb begins, between 0ms and 200ms.

Chamber reverb

The chamber reverb emulates the sound of echo chambers found in early recording studios. This is characterised by a rapid build up of reflection density within a small to medium sized space coupled with a relatively colourless 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.



Item	*Element*	*Function*
1	**input** meter	Two adjacent 11-LED columns — one each for left and right — comprise the input meters.
2	**mix** control knob	Adjusts the dry/wet signal ratio.
3	**modulation** control knob	Specifies the combined rate and depth of modulation applied the reverb tail.
4	**contour time** control knob	Controls the time over which the reflection density increases during the initial portion of the reverb tail.
5	**lf x-over** control knob	Specifies the cross-over frequency for the low frequency decay (**lf decay**), in the range 20Hz – 500Hz.
6	**output** meter	Two adjacent 11-LED columns — one each for left and right — comprise the output meters.
7	**Audition** button	This momentary-action button triggers a short internally generated sound to aid reverb evaluation (as a check).
8	**hf cut** control knob	High frequency cut control knob applies a 6dB/Oct low pass filter to the input signal, in the range 200Hz to 20kHz.
9	**lf cut** control knob	Low frequency cut control knob applies a 6dB/Oct high pass filter to the input signal, in the range 10Hz to 500Hz.
10	**hf damp** control knob	High frequency damping, progressively reduces the high frequency content over time, in the range 1kHz to 20kHz.
11	**lf decay** control knob	Specifies the ratio of decay for low frequency content, in the range 0.5 – 2.0.

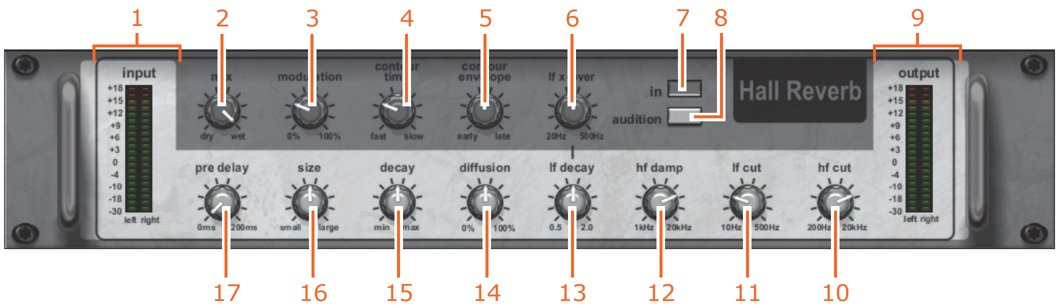
Item	*Element*	*Function*
12	**diffusion** control knob	Increases the density of both the early reflections and reverb tail, between 0 and 100%.
13	**decay** control knob	Adjusts the decay time relative to the room size, from minimum to maximum.
14	**size** control knob	Specifies the room size (also affects decay), from small to large.
15	**pre delay** control knob	Specifies the time before the reverb begins, between 0ms and 200ms.
16	**In** switch	Switches the Plate Reverb effect in/out.

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.

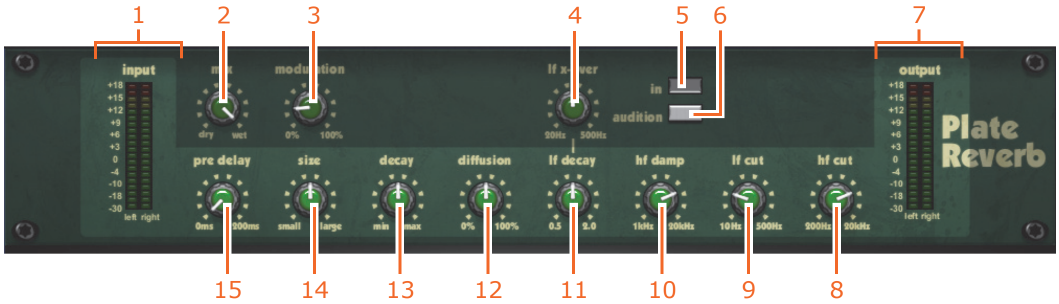


Item	*Element*	*Function*
1	**input** meter	Two adjacent 11-LED columns — one each for left and right — comprise the input meters.
2	**mix** control knob	Adjusts the dry/wet signal ratio.
3	**modulation** control knob	Specifies the combined rate and depth of modulation applied the reverb tail.
4	**contour time** control knob	Controls the time over which the reflection density increases during the initial portion of the reverb tail, between fast and slow.
5	**contour envelope** control knob	Controls the time over which the reflection density increases during the initial portion of the reverb tail, between fast and slow.
6	**lf x-over** control knob	Specifies the cross-over frequency for the low frequency decay (**lf decay**), in the range 20Hz – 500Hz.
7	**in** switch	Switches the Plate Reverb effect in/out.
8	**audition** button	This momentary-action button triggers a short internally generated sound to aid reverb evaluation (as a check).
9	**output** meter	Two adjacent 11-LED columns — one each for left and right — comprise the output meters.
10	**hf cut** control knob	High frequency cut control knob applies a 6dB/Oct low pass filter to the input signal, in the range 200Hz to 20kHz.
11	**lf cut** control knob	Low frequency cut control knob applies a 6dB/Oct high pass filter to the input signal, in the range 10Hz to 500Hz.
12	**hf damp** control knob	High frequency damping, progressively reduces the high frequency content over time, in the range 1kHz to 20kHz.
13	**lf decay** control knob	Specifies the ratio of decay for low frequency content, in the range 0.5 – 2.0.
14	**diffusion** control knob	Increases the density of both the early reflections and reverb tail, between 0 and 100%.
15	**decay** control knob	Adjusts the decay time relative to the room size, from minimum to maximum.
16	**size** control knob	Specifies the room size (also affects decay), from small to large.
17	**pre delay** control knob	Specifies the time before the reverb begins, between 0ms and 200ms.

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 transudcer 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 build up 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.



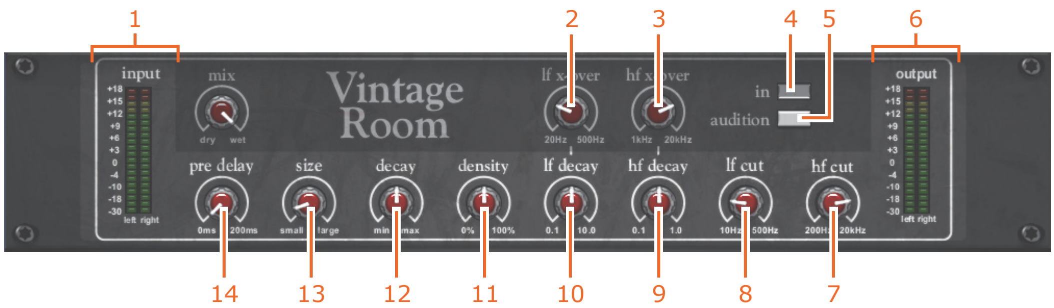
Item	Element	Function
1	**input** meter	Two adjacent 11-LED columns — one each for left and right — comprise the input meters.
2	**mix** control knob	Adjusts the dry/wet signal ratio.
3	**modulation** control knob	Specifies the combined rate and depth of modulation applied the reverb tail.
4	**lf x-over** control knob	Adjusts the cross-over frequency for the low frequency decay, in the range 20Hz to 500Hz.
5	**in** switch	Switches the Plate Reverb effect in/out.
6	**audition** button	This momentary-action button triggers a short internally generated sound to aid reverb evaluation (as a check).
7	**output** meter	Two adjacent 11-LED columns — one each for left and right — comprise the output meters.
8	**hf cut** control knob	High frequency cut control knob applies a 6dB/Oct low pass filter to the input signal, in the range 200Hz to 20kHz.
9	**lf cut** control knob	Low frequency cut control knob applies a 6dB/Oct high pass filter to the input signal, in the range 10Hz to 500Hz.
10	**hf damp** control knob	High frequency damping, progressively reduces the high frequency content over time, in the range 1kHz to 20kHz.
11	**lf decay** control knob	Adjusts the ratio of decay for low frequency content, between 0.5 and 2.0.

Item	Element	Function
12	**diffusion** control knob	Increases the density of both the early reflections and reverb tail, between 0 and 100%.
13	**decay** control knob	Adjusts the decay time relative to the room size, from minimum to maximum.
14	**size** control knob	Specifies the room size (also affects decay), from small to large.
15	**pre delay** control knob	Specifies the time before the reverb begins, between 0ms and 200ms.

Vintage room

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.

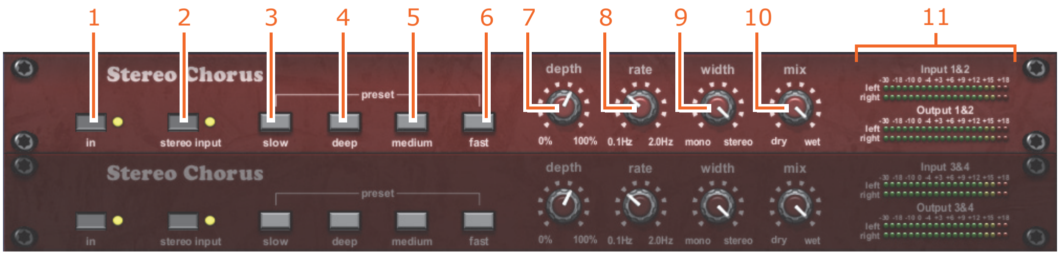


Item	Element	Function
1	**input** meter	Two adjacent 11-LED columns — one each for left and right — comprise the input meters.
2	**lf x-over** control knob	Adjusts the cross-over frequency for the low frequency decay.
3	**hf x-over** control knob	Adjusts the cross-over frequency for the high frequency decay.
4	**in** switch	Switches the Vintage Room reverb effect in/out.
5	**audition** button	This momentary-action button triggers a short internally generated sound to aid reverb evaluation (as a check).
6	**output** meter	Two adjacent 11-LED columns — one each for left and right — comprise the output meters.
7	**hf cut** control knob	High frequency cut control knob applies a 6dB/Oct low pass filter to the input signal, in the range 200Hz to 20kHz.

Item	*Element*	*Function*
8	**lf cut** control knob	Low frequency cut control knob applies a 6dB/Oct high pass filter to the input signal, in the range 10Hz to 500Hz.
9	**hf decay** control knob	Adjusts the ratio of decay for high frequency content, between 0.1 and 1.0.
10	**lf decay** control knob	Adjusts the ratio of decay for low frequency content, between 0.1 and 10.0.
11	**density** control knob	Increases the reflection density of the reverb tail, between 0 and 100%.
12	**decay** control knob	Adjusts the decay time relative to the room size, from minimum to maximum.
13	**size** control knob	Specifies the room size (also affects decay), from small to large.
14	**pre delay** control knob	Specifies the time before the reverb begins, between 0ms and 200ms.

Stereo chorus

Emulation of dual stereo chorus but with having two units in one rack space.



Item	*Element*	*Function*
1	**in** switch	Switches the stereo chorus effect in/out.
2	**stereo input** switch	Changes the operation to be stereo in and stereo out.
3	Preset **slow** switch	Sets a slow rate with minimal depth.
4	Preset **deep** switch	Sets a slow rate with maximum depth.
5	Preset **medium** switch	Sets a medium rate with minimal depth.
6	Preset **fast** switch	Sets a fast rate with minimal depth.
7	**depth** control knob	Adjusts the amount of modulation applied to the pitch in the range 0 to 100%.
8	**rate** control knob	Adjusts the speed of modulation applied to the pitch in the range 0.1Hz to 2Hz.
9	**width** control knob	Adjusts the stereo spread of the output signal from mono to stereo.
10	**mix** control knob	Adjusts the dry/wet signal ratio.
11	Meters	Input and output meters.

Dual stereo delay

The dual stereo delay effect is a simpler, more concise, version of the current delay device with the advantage of having two units in one effect device rack space. The dual stereo delay is a dual stereo in and dual stereo out device with metering for each discrete input and output.

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 rotaries will change from seconds (milliseconds) to musical note durations as they do with the current effects units. However, the seven-segment LED 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 should be greyed out.

Item	*Element(s)*	*Function*
1	On/off switch	Switches the delay effect in or out.
2	**delay time** controls	The control knob adjusts the delay time in the range 0 seconds to 2 seconds in 5 milliseconds increments. In global tap mode, adjusts the note intervals. Switching the **X2** switch changes the delay time scale to 0 seconds to 4 seconds in 10 millisecond intervals.
3	**tap** button	Use this button to tap in the delay time or tempo manually.
4	**up** and **down** buttons	Increases/decreases the delay time by 1 millisecond 0.1 bpm
5	Display	Shows current delay time and selected unit.
6	**feedback** control knob	Adjusts the delay feedback loop gain in the range 0 to 100%.

Item	*Element(s)*	*Function*
7	**lf cut** controls	This low frequency control knob applies a 6dB/Oct high pass filter to the delay and optionally feedback signal (by switching on the **post fb** switch below). The default is post-delay and pre-feedback. Range is from 10Hz to 500Hz.
8	**hf cut** controls	This high frequency control knob applies a 6dB/Oct low pass filter to the delay and optionally feedback signal (by switching on the **post fb** switch below). The default is post-delay and pre-feedback. Range is from 200Hz to 20kHz.
9	**mix** control knob	Adjusts the dry/wet signal ratio.
10	Meters	Input and output meters.

Matrix mixer

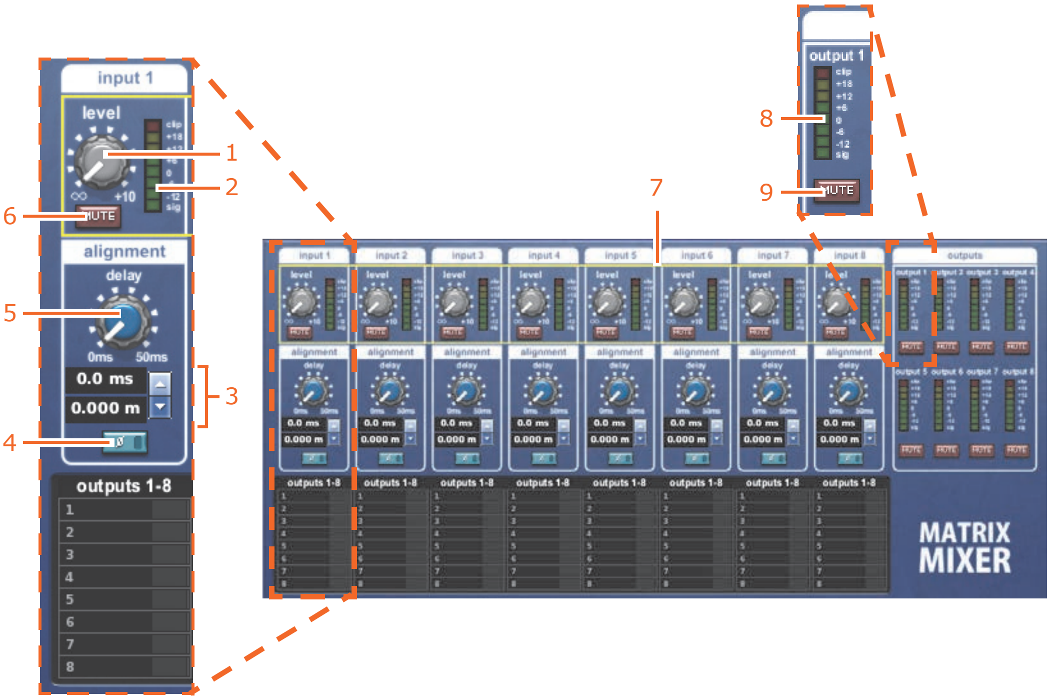
The matrix mixer is an eight mono I/O device with discrete metering for each input and output. The display of the matrix mixer comprises controls that duplicate the equivalent ones on the control surface and can be used as an alternative method of operation. You can link the output EQ settings across channels and also link odd and even outputs as a stereo pair, which is a GUI-only function.

Unlike the other internal effects, the matrix mixer has two screens (input and output), which require specific navigational methods (see “Navigating the input and output screens” on page 156). Both screens provide an overview of the other to save you having to navigate between them in order to obtain incidental information.

Note: *The global tap option does not apply to the matrix mixer.*

Input screen

The input screen shows the signal level, delay and output send contributions for the inputs and, to the right, an overview of the outputs with facility for muting.



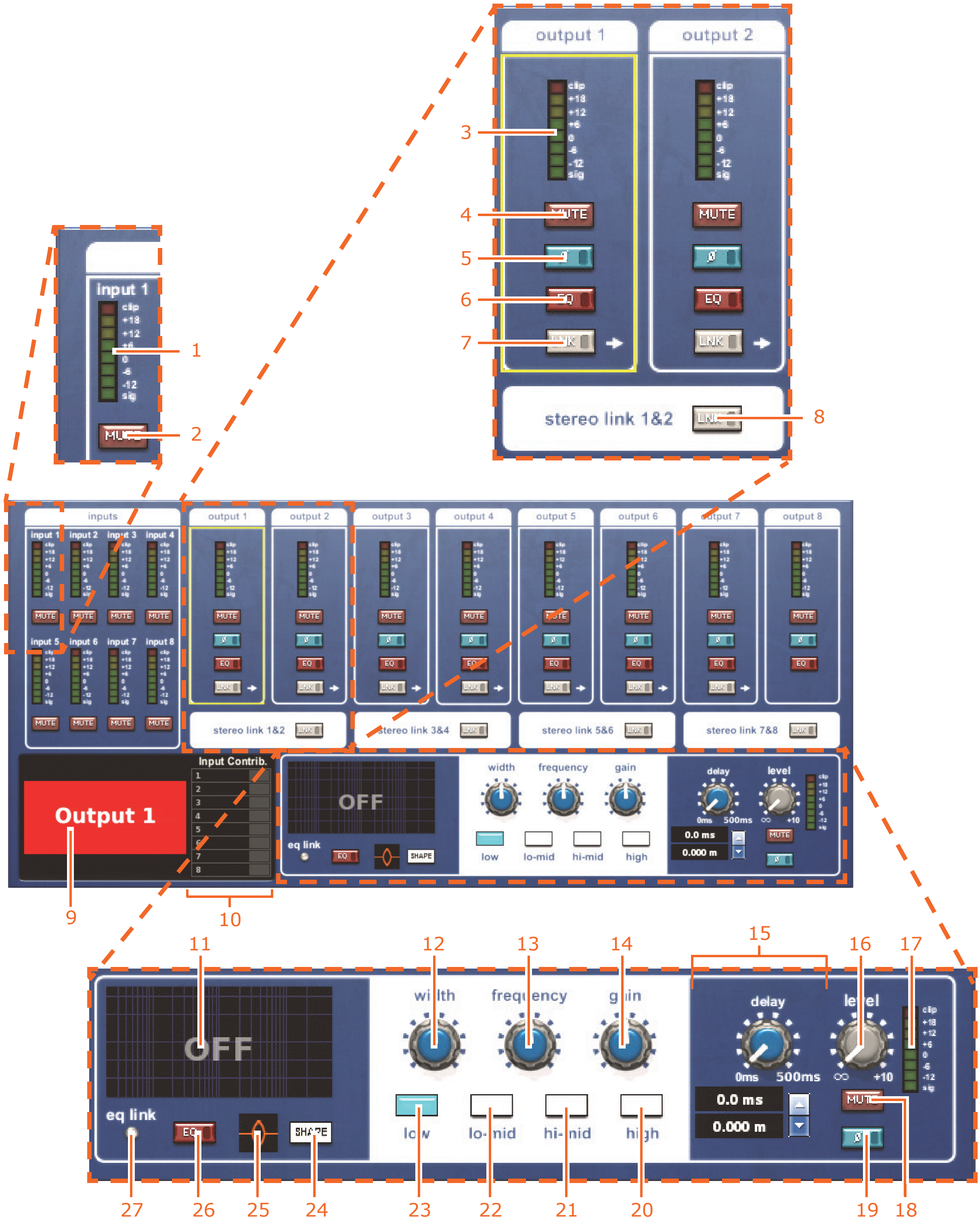
Item	Element	Function
1	**level** control knob	Continuous adjustment of the input level from ∞ (infinity) to +10dB.
2	Meter	11-segment meter for showing the input channel level.
3	Up/down arrows	Increases/decreases the input delay time (shown in milliseconds and metres) in 0.01 millisecond increments. You can also type in the value in the fields.
4	Phase switch ø	Adjusts the input signal phase by 180°.
5	**delay** control knob	Adjusts the input delay time within the range 0ms to 50ms in 0.01 ms increments.
6	**MUTE** switch	Mutes the input channel.
7	Yellow box	Shows which controls are currently assigned to the **assignable controls** panel.
8	Meter	11-segment meter for showing the output channel level.
9	**MUTE** switch	Mutes the output channel.

>> To switch an output mix send on/off

You can have any combination of the eight outputs contributing to each input. To switch an output on/off, navigate to the desired output using the left/right navigation (see "Navigating the input and output screens" on page 156) and click the associated button in the **assignable controls** panel.

Output screen

The output screen shows the output channel controls with a processing area underneath for the selected channel. An inputs section to the left provides an input overview, with facility for muting.

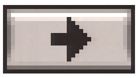
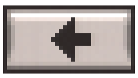


Item	*Element*	*Function*
1	Meter	8-segment meter for showing the input channel level.
2	**MUTE** switch	Mutes in input channel.
3	Meter	8-segment meter for showing the output channel level.
4	**MUTE** switch	Mutes the output channel.
5	Phase switch ø	Adjusts the output signal phase by 180°.
6	**EQ** switch	Switches the output channel EQ in/out.
7	**LNK** switch	Links the EQ of the local output channel to the adjacent output channel to the right.
8	**LNK** switch	See "Stereo linking" on page 157.
9	Output channel identifier	Shows the name of the currently selected output and indicates, by its background colour, which pair it belongs to.
10	Input contributions panel	Shows the level contributions of the eight input channels for the selected output channel. However, when a pair of output channels are stereo linked, its function changes (see "Stereo linking" on page 157.).
11	Graph	Displays the EQ envelope of the selected output channel.
12	**width** control knob	Adjusts the filter width of the EQ band for the selected output channel, in the range 0.1 – 3.0 Oct.
13	**frequency** control knob	Adjusts the filter frequency of the selected EQ band for the selected output channel, using the same frequency ranges as the standard input channel EQ.
14	**gain** control knob	Adjusts the filter gain of the selected EQ band for the selected output channel, in the range -16dB to +16dB.
15	**delay** controls	Provides adjustment of the output delay via control knob or up/down arrows for the selected output channel, in the range 0ms – 500ms. Delay time accuracy is to 0.01 millisecond. Delay is also shown as a distance (metres).
16	**level** control knob	Continuous signal level adjustment of the selected output channel, from ∞ (infinity) to +10dB.
17	Meter	8-segment meter for showing the level of the selected output channel.
18	**MUTE** switch	Mutes the selected output channel.
19	Phase switch ø	Adjusts the output signal phase by 180° on the selected output channel.
20	**high** button	Switches the EQ controls to work with the High EQ band on the selected output channel.
21	**hi mid** button	Switches the EQ controls to work with the Hi-Mid EQ band on the selected output channel.
22	**lo mid** button	Switches the EQ controls to work with the Lo-Mid EQ band on the selected output channel.
23	**low** button	Switches the EQ controls to work with the Low EQ band on the selected output channel.
24	**SHAPE** button	Selects the shelving mode for the high and low EQ bands on the selected output channel. For information on the shelving modes, see "EQ (E zone)" on page 266.

Item	*Element*	*Function*
25	Icon	Identifies the currently selected EQ shelving mode (see “EQ (E zone)” on page 266). Default parametric icon is .
26	**EQ** switch	Switches the EQ in/out for the selected channel.
27	**eq link** LED	Illuminates to indicate that the EQ is on for the selected channel.

Navigating the input and output screens

The up and down navigation arrow buttons on the Matrix Mixer operate in the same way as on any internal effect (see “Rack and unit control navigation” on page 174). However, as this effect is unique in that it has two screens, the left and right buttons function in a slightly different manner, as described in this section.

Navigation button	*Function*
	Scrolls consecutively down through the input screen sections (input, alignment and output mix sends) and then left to right through the outputs of the output screen, crossing over screens in between. See Figure 10 on page 157.
	Scrolls in the opposite direction to the right arrow button.

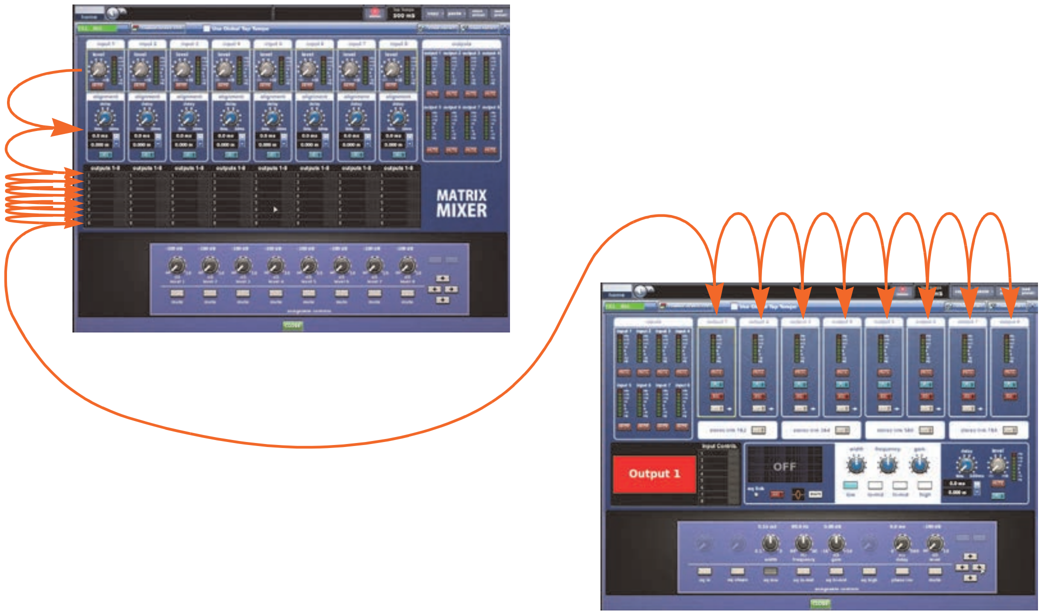
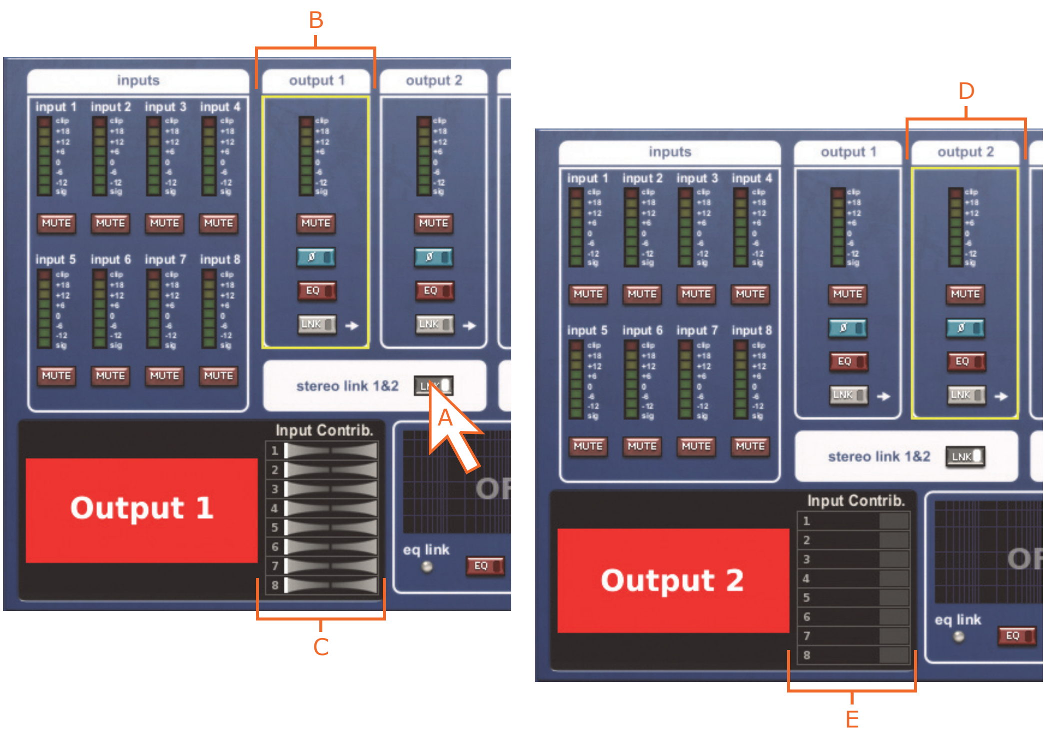


Figure 10: Scrolling order of right arrow navigation button

Stereo linking

Clicking a stereo linking button (**LNK**) stereo links the local pair of output channels, so that the input mix send controls operate as pan on the left (odd numbered) output channel and level on the right (even numbered) output channel, which is similar to the normal mix sends on the console.



Stereo linking a pair of output channels (for example, outputs 1 and 2)
*Click the local stereo link button (**A**) to link the two outputs, so that output 1 (**B**) now becomes panning (**C**) and output 2 (**D**) remains as signal level (**E**).*

Chapter 17: Control Groups

PRO1 control groups comprise VCA, POPulation, MCA, auto-mute and talk groups. This chapter explains the functions of each of these groups and shows you the areas on the control surface and GUI that are used for their operation and management.

VCA and POPulation groups

A VCA/POPulation group can contain any combination of channels. When you select a group via the control surface its channel members are unfolded to the channel faders from right to left. Recalling another VCA/POPulation group deselects the one currently selected.

The GUI provides full support for both groups on the **VCA Groups** screen. This replicates the VCA hardware controls and provides extra functionality. This screen also provides group member management.

For information on group selection and navigation, see Chapter 6 "Working With The PRO1 Control Centre" on page 37 and Chapter 7 "Navigation" on page 43.

For details of VCA/POPulation group configuration and operation, see "Using VCA/POPulation groups" on page 72.

>> To select a VCA/POPulation group

Press the LCD select button of the desired group. If you are selecting a VCA group, VCAs must be assigned to the mix faders; if necessary, press the **VCA** button in the **mix bay** section first. If a group is already selected, this will be replaced by the new selection.

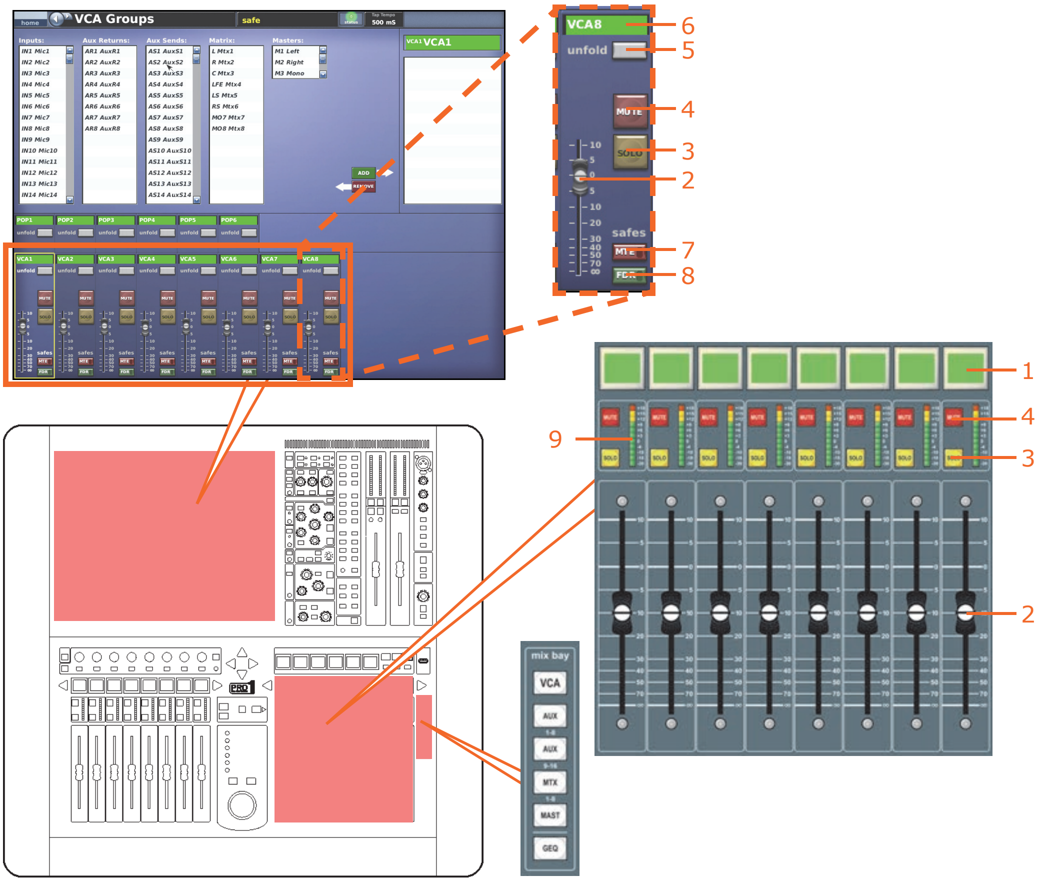
Pressing the LCD button a currently selected group will deselect it.

>> To open the VCA Groups screen

At the GUI, choose **home** ▸ ***Control Groups*** ▸ ***VCA Groups.***

VCA groups

There are eight VCA groups, which are accessible via the mix fader LCD select buttons when configured for VCA operation.



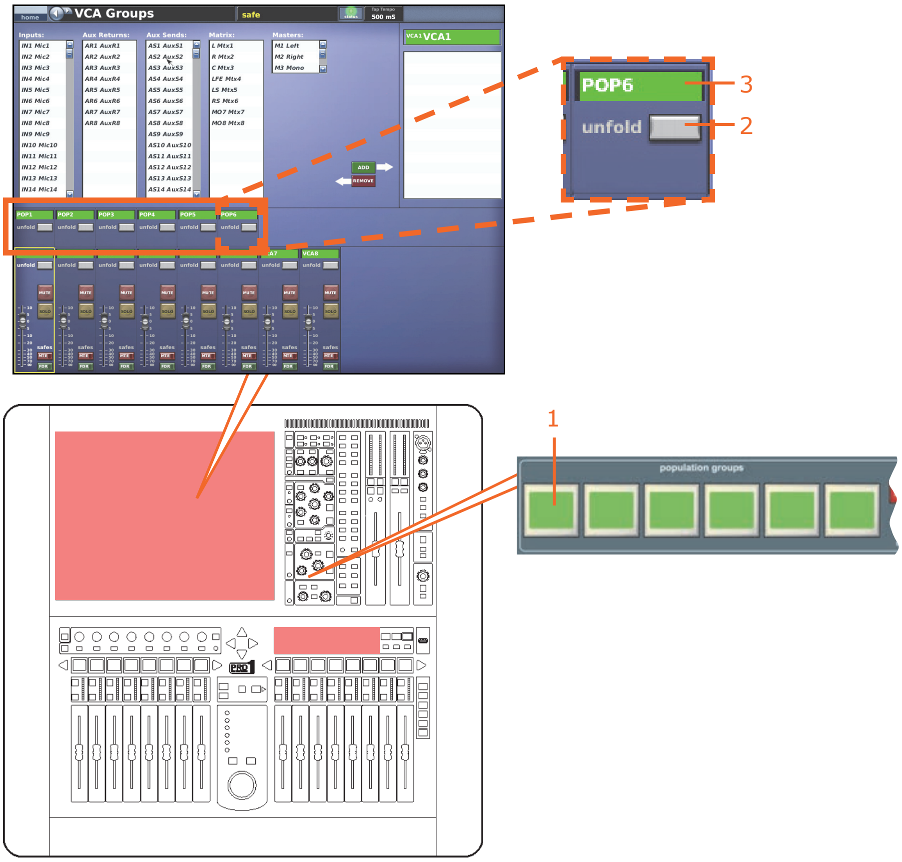
VCA groups on the control surface and GUI

Item	Description
1	LCD select button, selects the VCA group and is also used for group member assignment.
2	VCA group control, adds its level control on top of the local channel fader controls of the group members.
3	**SOLO** button activates signal routing from all assigned channels to the monitor A section of the control centre. It is used to monitor the VCA master faders by creating a mix on the solo buses, which consists of all input channels and audio mix groups that are assigned to control from corresponding VCA masters.

Item	*Description*
4	**MUTE** switch. This is, technically, not a mute but a fader minus infinity (-∞) switch that overrides the VCA group master (without moving its physical position). The VCA group mutes can be stored and recalled as part of the scene automation. When on, it mutes all post-fader signals from channels that have been assigned to the VCA group master (regardless of local press, scene mute and SIP mute, which affect only the mute status indicator while channel is muted by VCA). However, it does not update the mute status indicator on the channel (only channel outputs). On removal of the VCA mute, the channel outputs are updated to the current state of the channel mute status indicator. The VCA control group mute has been removed from scene recall and auto-mute action. When mute safe is active all channel mute activation methods, other than by local press, are ignored. De-activating the mute safe condition re-evaluates and applies the current status of auto-mute and SIP mute.
5	**unfold** button, assigns the VCA group to the control surface, unfolding the group members to the channel faders.
6	VCA group ID, fixed and user-configured name of group.
7	**MTE** button for selecting mute safe. Integral LED illuminates when the button is switched on.
8	**FDR** button for selecting fader safe. Integral LED illuminates when the button is switched on.
9	11-segment LED meter.

POPulation groups

There are six POPulation groups. Although similar to VGA groups in their methods of group management and selection, they have limited functionality.



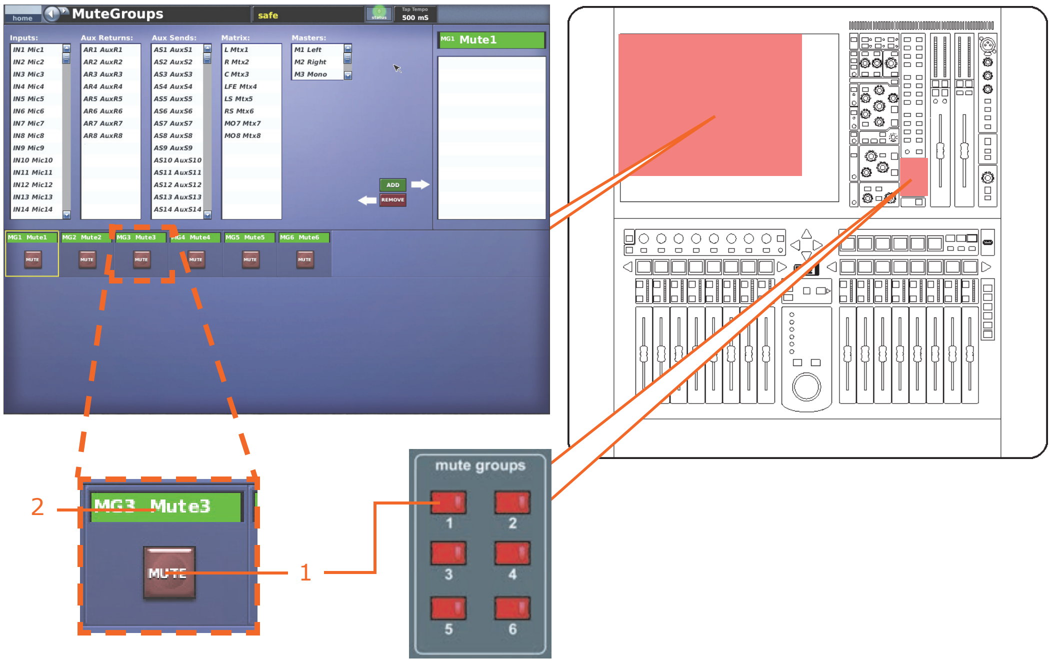
POPulation groups on the control surface and GUI

Item	Description
1	LCD select button, selects the VCA group and is also used for group member assignment.
2	**unfold** button, assigns the POPulation group to the control surface, unfolding the group members to the channel faders.
3	POPulation group ID — fixed and user-configured name of group.

Auto-mute (mute) groups

You can simultaneously mute any channels you want by assigning them to an auto-mute group. You can have up to six auto-mute groups, each one being enabled by its own **MUTE** button in the **population and mute groups** section.

Auto-mute groups are managed via the **Mute Groups** screen of the GUI menu, from where you can assign channels to any of the groups. You can configure the name and background colour of a mute group at the Groups Sheet screen (see "Configuring the groups" on page 166).



Item	Description
1	**MUTE** switch, mutes/unmutes all of the assigned channels. Also, the same channel can be assigned to more than one auto-mute group — the channel should be auto-muted while any of the mute groups to which it is assigned are muted.
2	Auto-mute group name (default and user-configured).

An auto-mute on can happen because of:

- Activating an assigned auto-mute.
- Assigning an already active auto-mute.
- Recalling a scene that assigns an already active auto-mute.

An auto-mute off can happen because of:

- Deactivating all of the assigned auto-mutes.
- Unassigning all of the active auto-mutes.
- Recalling a scene that de-assigns all of the active auto-mutes.

>> To open the Mute Groups screen

At the GUI, choose **home** ▸ ***Control Groups*** ▸ ***Mute Groups.***

>> To program mute group at the control surface

1 In the **auto mute groups** section (master bay), press and hold the **SELECT** button of your chosen auto-mute group.

2 Do one of the following:

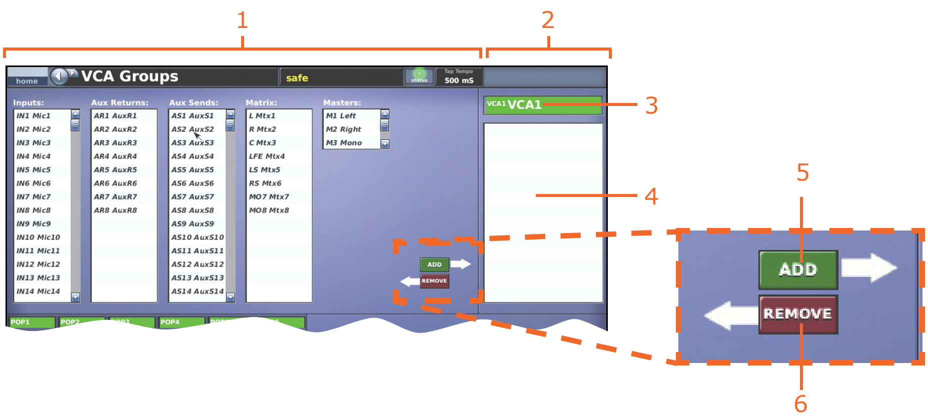
- To add inputs to the group, press the LCD select button of each input channel you want in the group. If necessary navigate the input channels you want to the control surface.
- To add outputs to the group, press the quick access button (bottom of each output fast strip) of each output you want in the group. If necessary navigate the output channels you want to the control surface.

About the control group screens

The control group screens each have two main areas. The management section at the top allows you to choose the group members, and is common to all control group types. While the bottom section contains the controls and sections specific to each type, which are described later on in this chapter.

Management section

The group management section of each control group screen has two main panels that, with the aid of the **ADD** and **REMOVE** buttons, let you choose group members.



Typical management section of control group screen

Item	Control	Function
1	Group member selection lists	These group type-dependent lists let you choose which inputs, returns, auxes, matrices and masters you want as members in the selected group. If necessary, use the slider on the right of a panel to access all non-members. You can add more than one member at once by using the Ctrl and Shift keys on the keyboard. Shift will select a range of members and Ctrl will act as an individual toggle.
2	Group member panel	Shows the current group members and lets you
3	Name field	Name of selected group.
4	Group member panel	List of the current members of the selected group.
5	**ADD** button	Moves the non-members (selected in the group member selection lists) to the group member panel, adding them to the group.
6	**REMOVE** button	Moves the members (selected in the group member panel) back to their respective group member selection lists, removing them from the group.

>> To program a group at the GUI screen

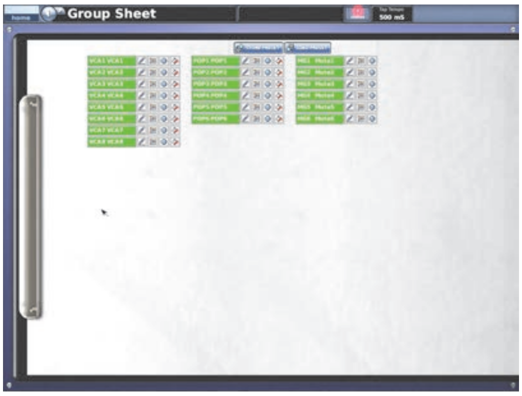
1. Open the screen of the desired control group. For example, if you want to program a VCA group, open the **VCA Groups** screen.
2. Click the desired group. For example, **VCA1.**
3. Click the channels that you want to add to the group.
4. Click **ADD.** The channels will be moved to the group member panel.

If you want to remove any members from the group, click the channels that you want to remove from the group (group member panel). Then, click **REMOVE.** The channels are moved back to their respective lists.

Configuring the groups

home ▸ *Control Groups* ▸ *Group Sheet*

Similarly to the **Input Sheet** screen, the **Group Sheet** screen lets you change the name and background colour of each group as they appear on the GUI screen and LCD select switch. Additionally, you can change the colour of all the current members of the group to match the group colour by clicking the fill button .



>> To open the Group Sheet screen

At the GUI, choose **home ▸ *Control Groups* ▸ *Group Sheet.***

Chapter 18: Copy And Paste

The PRO1 has a number of copy and paste features that make it easy to copy useful settings to other areas. You can copy and paste the following:

- Processing areas across channels — see “Using copy and paste” on page 84.
- Parameters across GEQs — see “Copying settings between GEQs” on page 124.
- Parameters through scenes — see “Show editor” on page 83.
- Scenes — see “Copying and deleting scenes” on page 185 and “To copy and paste sections to a scene(s)” on page 83.
- Events — see “To save a show or create a new one from the current settings” on page 80.
- Presets — see “To create a new preset library from the current one” on page 217.

For details of the copy and paste, see the Appendices.

Copying through channels/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, although the areas (parameters) that are copied across are similar.

Chapter 19: Assignable Controls

This chapter describes the **assignable controls** section, which has full GUI support and lets you do the following:

- Operate specific processing area controls of the channels currently assigned to the channel faders (see "Operating the channel fader assignments" on page 170).
- Control any rotary on the control surface (see "Controlling a rotary control" on page 171).
- Operate the controls of an internal rack unit (see "Controlling an internal effect/GEQ" on page 173).

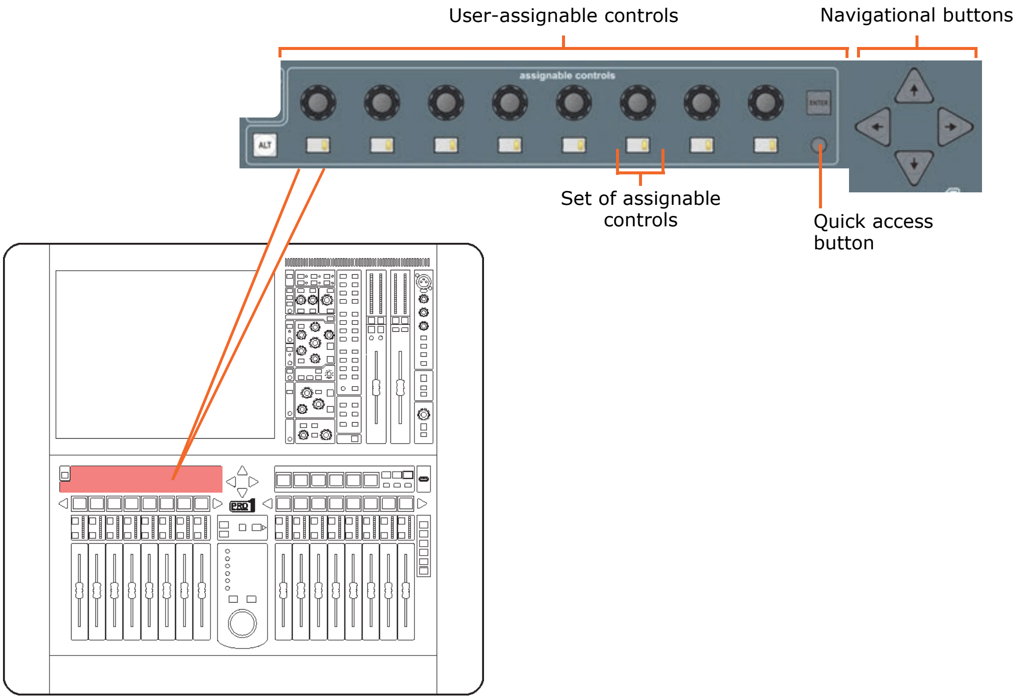


*Figure 11: The **assignable controls** section on the control surface*

Operating the channel fader assignments

During normal operation the assignments of the **assignable controls** section follow the cross-hair (see "GUI cross-hair" on page 39) processing area selection. This lets you control a predetermined button and control knob within the selected processing area. You can change the button assignment within a processing area (using the **ALT** button), but the control knob assignment remains fixed.

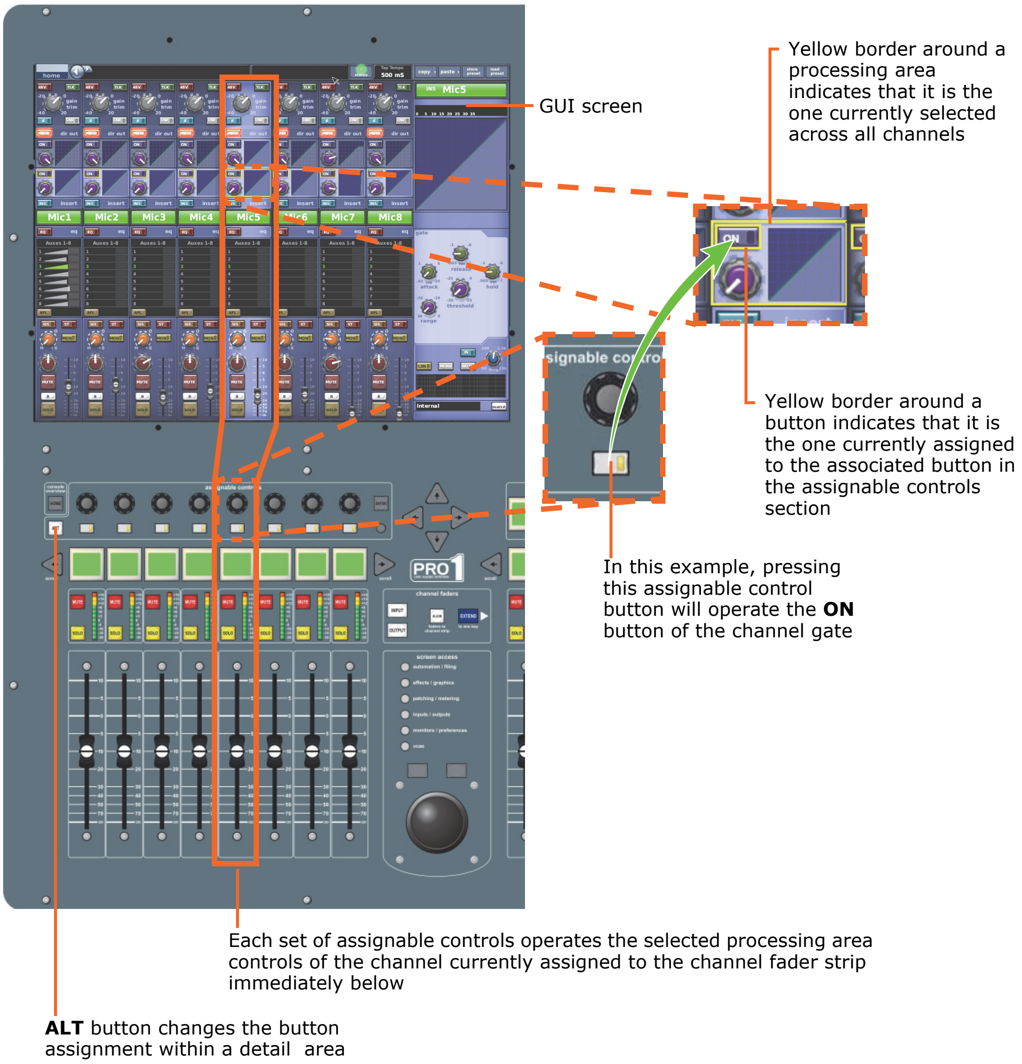
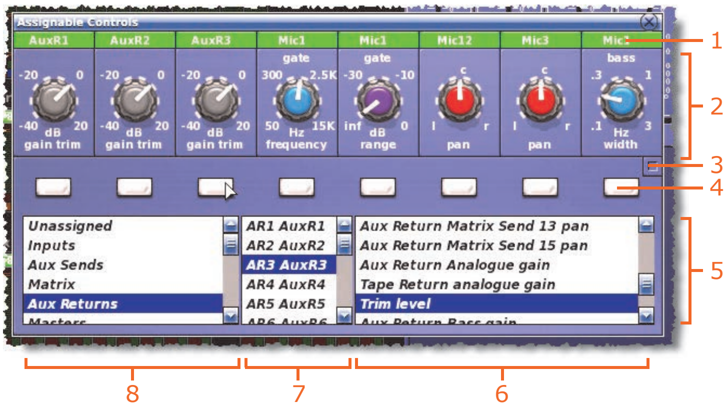


Diagram showing the processing area control assignments of the assignment controls

Controlling a rotary control

An important function of the PRO1 is one that lets you operate any of the control knob functions on the control surface, such as **gain trim**, compressor/gate **threshold** and **level**, and even any of the internal effects (but not the GEQs), from the **assignable controls** section. This gives you quick access to the controls that are currently the most useful to you. The **Assignable Controls** window lets you assign your desired controls and also shows you the current assignments and status of each one.

In addition, you can assign the 'tap' function of the Delay effect to any pushbutton in the **assignable controls** section. This function — which is also available via the **tap L**/**tap R** parameters when operating the Delay effect from here — lets you input the delay time manually by tapping the pushbutton instead of choosing a value (see "To manually set the tap time of an effect" on page 172).

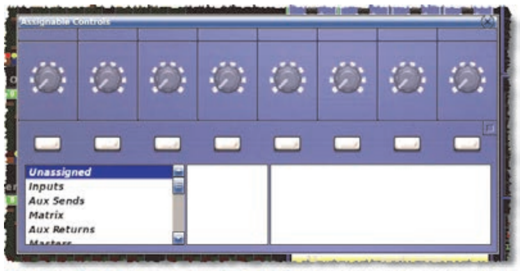


Assignable Controls window

Item	Element	Function
1	Name field	Channel name, with background colour to match the default/user-defined channel colour.
2	Control knobs	User-assigned controls.
3	Button	Collapses the control selection lists (item 5).
4	Button	Assign/unassign button.
5	Control selection lists	These offer you all the available options from which to choose your rotary control assignment.
6	Option list	Available control options for the selected channel/effect.
7	Option list	Available channels/effects for the selected channel type/effect.
8	Option list	Available channel types/effect. The options are: ***Unassigned; Inputs; Aux Sends; Matrix; Aux Returns; Masters;*** and ***Effects.***

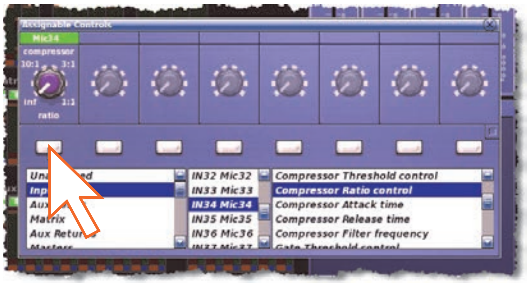
>> To open the Assignable Controls window

Press the quick access button in the **assignable controls** section (see “The assignable controls section on the control surface” on page 169). In the example shown right no controls have been assigned.



>> To assign a control to a set of assignable controls

Open the **Assignable Controls** window and select the desired control using the three panels at the bottom of the window (see “About the assignable controls panel on the GUI” on page 173). For example, choose the compressor ratio control of input channel 34. Then click one of the overlying assignment buttons (as shown right).



>> To unassign a control

1. In the **Assignable Controls** window on the GUI, click ***Unassigned*** in the far left panel (bottom of window).
2. Click the desired assignment button.

>> To manually set the tap time of an effect

1. Assign the desired effect’s delay time parameter to the **assignable controls** section. (Choose ***Effects*** in the left panel and then the desired channel and delay time parameter from the other panels.)
2. Tap the assign/unassign button of the assigned control (just as you would the **Tap** button of the effect) to achieve the desired tap time.

   The PRO1 measures the interval between taps. It uses the most recent taps to calculate the average tap time, which is constantly updated according to each subsequent tap. This value is displayed on the effect’s front panel in the appropriate **Range** field and is also indicated by the control knob immediately left.

Controlling an internal effect/GEQ

As the internal effects and GEQs of the PRO1 are primarily GUI-only features, control surface support is provided by the **assignable controls** section, which lets you operate their parameters using physical controls.

With an internal rack unit selected on the GUI, a specific set of its parameter controls will be automatically assigned to the **assignable controls** section. To encompass all of the rack unit's parameter controls they are bundled into predetermined sets — known as 'pages' for effects and 'groups' for GEQs — which are navigated using the arrow buttons (control surface or GUI).

About the assignable controls panel on the GUI

The **assignable controls** panel is displayed at the bottom of the effect/GEQ window. In addition to replicating the **assignable controls** section on the control surface, it displays additional information, such as button and control knob assignments, current 'page' number, etc.

Assignable controls panel

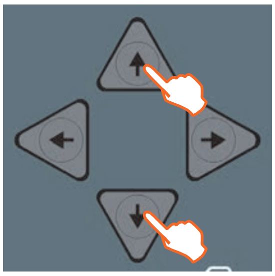
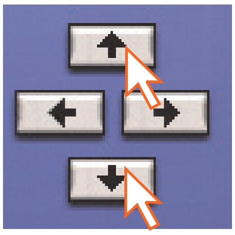
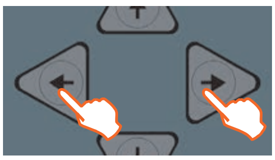
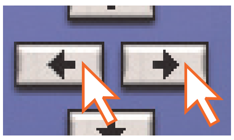
The following diagram explains the elements of the **assignable controls** panel. It uses the one for the effects as an example, but this also applies to the GEQs.

Item	Description
1	For the effects only, this shows which 'page' of parameters is currently assigned and is in the format [page number]/[total number of pages]. For example, the diagram above is displaying page 1 of a total of 2 pages.
2	Single set of controls (button and control knob).
3	Control button.

Item	Description
4	**Effects only:** Description of the effect's button currently assigned to the button. **GEQ only:** Will show either the text "zoom n" (where n is number from 1 to 8) or "overview" to indicate which display you are in, that is, overview or zoom, respectively.
5	Navigation buttons, which replicate the arrow buttons to the right of the assignable controls section on the control surface (see "Rack and unit control navigation" on page 174).
6	Parameter description of the overlying control knob's assignment.
7	Control knob. Includes gradations and dimensions applicable to the assigned parameter.
8	When a control is unassigned, the assignable controls section displays it as a 'ghost' image. (This also applies to the buttons.)

Rack and unit control navigation

To navigate the rack and unit controls, use the four navigational buttons (see "The assignable controls section on the control surface" on page 169) to the right of the **assignable controls** panel on the control surface, which are replicated on the GUI.

Operating this pair of navigational buttons (control surface or GUI) . . .	*Does this . . .*
 or 	At the **Effects** or **Graphic EQs** screen, scrolls up/down through the units in the rack(s), even with the unit window open. Scrolling finishes at the first and last units.
 or 	At the **Effects** screen, scrolls left/right to predetermined control parameters of the selected effect, even with the unit window open (see Figure 12 "Changing the effect parameter assignments" on page 175). However, the function of these buttons on the **Graphic EQs** screen is more extensive (see "Controlling a GEQ" on page 175).

*Don't forget that you can operate the selected effect or GEQ in their respective rack unit view (for example, the **Effects** screen shown right) via the control surface or GUI.*

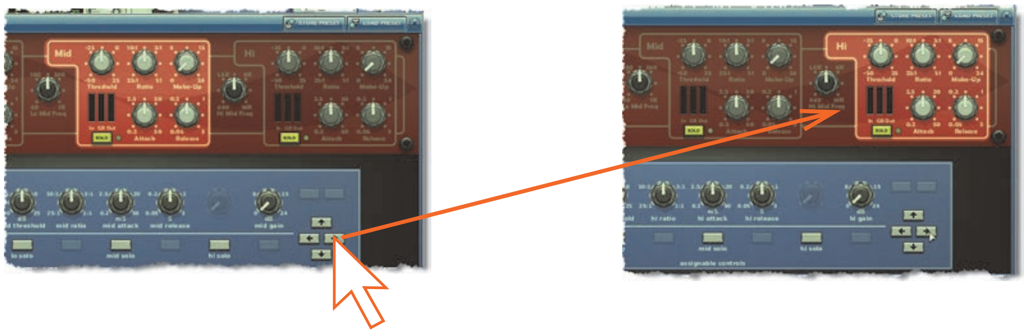


Figure 12: Changing the effect parameter assignments

Controlling a GEQ

You can use the **assignable controls** section to adjust the faders and controls of an internal GEQ, such as the high/low pass filters, notch filters, slope, etc. To accommodate all of the faders there are, effectively, two levels of display, known as “overview” and “zoom”. The overview display appears when you open the window of the GEQ, and lets you adjust a group of GEQ faders simultaneously by the same amount.

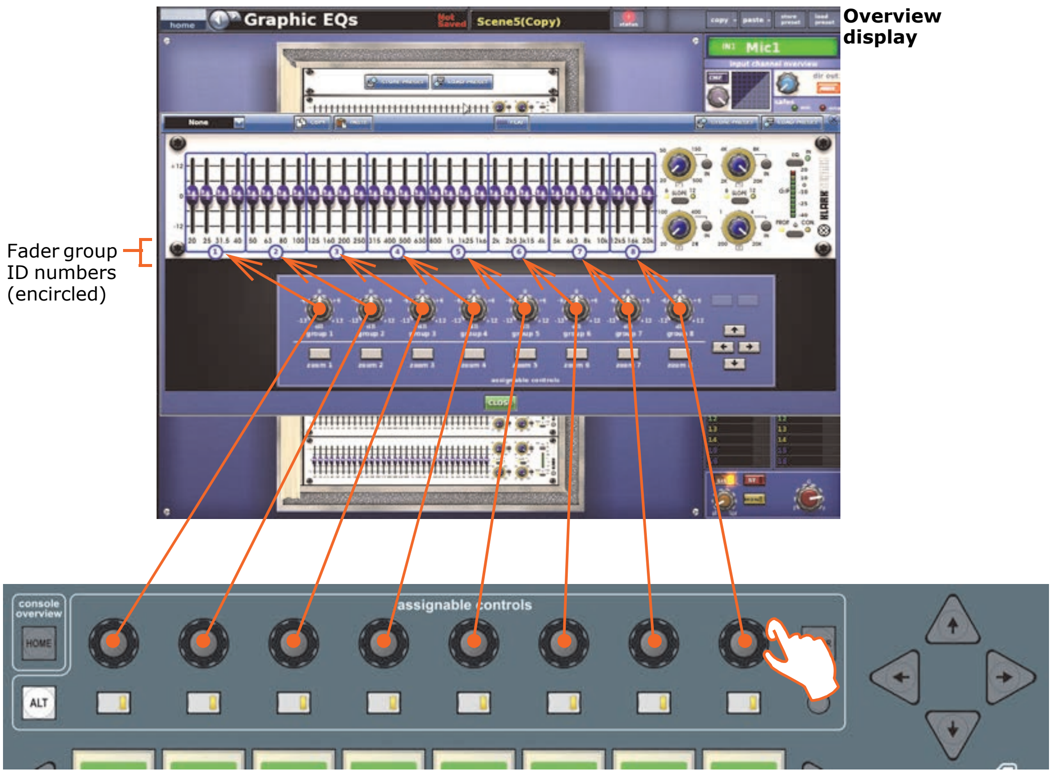


*Figure 13: Fader group control knob assignments in the overview display of the **Graphic EQs** screen*

In the zoom display, each of the assignable control buttons 'zooms in' on a specific group of faders. Each fader is assigned its own control knob in the assignable controls section, so that it can be operated individually. The right navigation arrow button opens the GEQs control panel (far right).

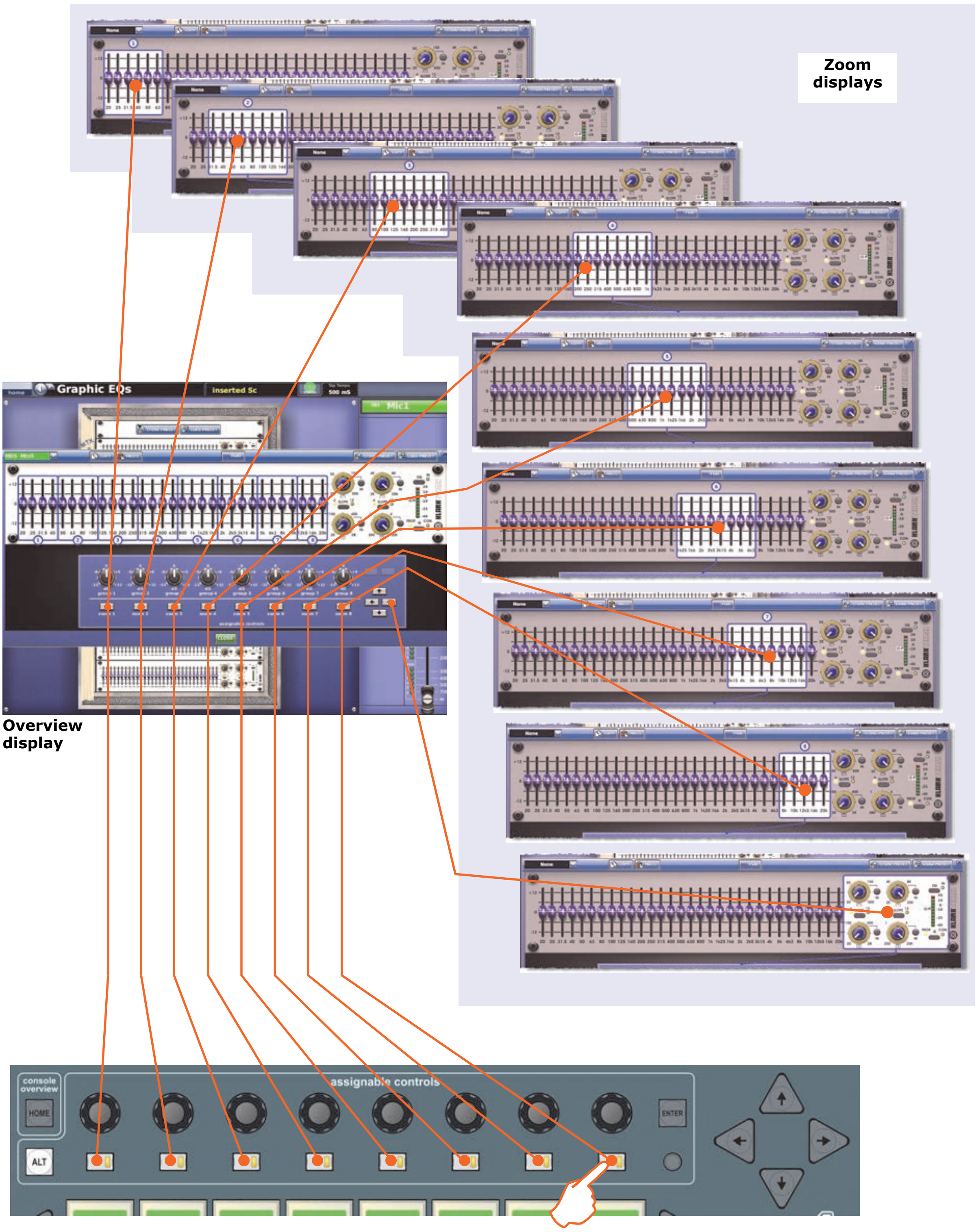


Figure 14: I zone LCD button assignments in the overview display

Chapter 20: Scenes And Shows (Automation)

This chapter shows you how to use scenes and shows, which are part of PRO1 automation.

About automation

Automation is predominantly a GUI-only function that allows complex editing of scenes and the creation of show files via the GUI menu. The control surface provides limited control via the **automation** section, which facilitates fast store/recall operation during show time and rehearsals.

The automation system of the PRO1 can store and recall up to 1000 scenes, each one being a snapshot of the control centre's settings at the instant the scene was created. By recalling scenes, users can — with certain exceptions — restore the control centre 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 for theatre productions, where each act requires reconfiguration of audio I/O.

All of the scenes for a show are contained within a show file. Show files are stored in the PRO1, so that they can be loaded when required, and they can also be transferred to/from external USB storage devices.

Shows can contain events. An event may trigger, for example, the sending of a MIDI message, a GPIO output voltage, or action an internal function. They can also be used for crossfades between scenes.

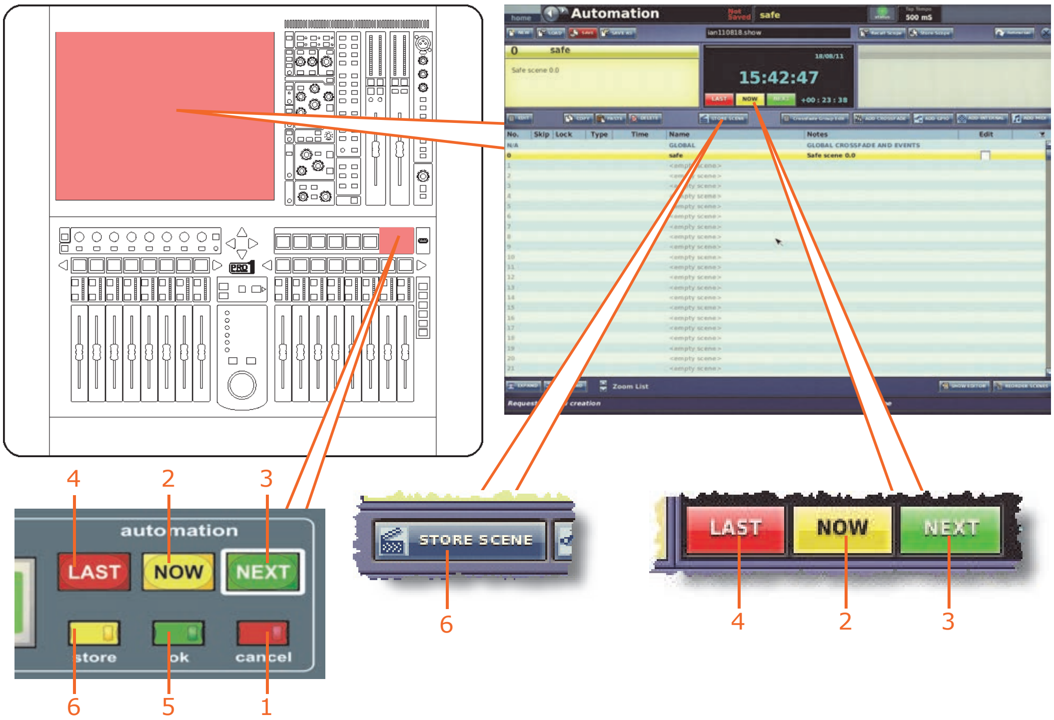
For theatre applications, channel settings can be recalled (across all scenes) from the user library (see Chapter 24 "User Libraries (Presets)" on page 215) so that one generic show can cope with different performers on a night-by-night basis.

You can copy certain parameters through scenes by using the **Show Editor** screen. For more information on the **Show Editor** screen and for details on how to use it for copying and pasting throughout scenes, see "Show editor" on page 83. For details of the parameters that can be copied through the scenes, see Appendix I "Parameters Affected By Copy And Paste" on page 441.

Throughout this chapter, wherever scenes are mentioned this also applies to point scenes, unless otherwise stated.

Automation controls

Although automation is supported on the master bay control surface by the **automation** section, most of the functionality is accessed via the GUI. The **Automation** screen allows editing of scenes, show files and events, and provides access to the scope editing screens. Additionally, the GUI has a **Files** screen for show file management and transfer.

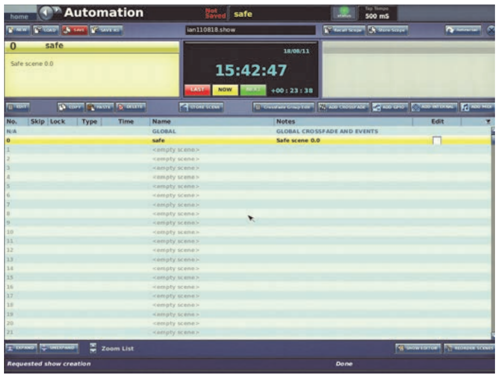


Automation controls on the control surface and GUI

Item	Control	Function
1	Red **cancel** button	Cancels a store operation and closes the **Store** window (illuminates to prompt when this button is active).
2	Yellow **NOW** button	Recalls the currently highlighted scene in the cue list, clearing any unsaved adjustments.
3	Green **NEXT** LCD button	Recalls the next scene/point scene (highlighted in green) in the cue list.
4	Red **LAST** button	Recalls the previous scene (highlighted in red) in the cue list.
5	Green **ok** button	Confirms an action (illuminates to prompt when this is necessary).
6	**store**/[**STORE SCENE**] button	Opens the **Store Scene** window (see "To create a new scene using the current settings" on page 81) and lets you store the current settings to the currently selected scene.

Automation screen

The **Automation** screen (shown right) contains information on the following:



- Scenes — see “Scenes” on page 180.
- Shows — see “Show files” on page 188.
- Scope — see Chapter 21 "Scope (Automation)" on page 193.
- Events — see Chapter 23 "Crossfades (Automation)" on page 207.

For details of how to open the Automation screen, see “To open the Automation screen” on page 79.

Using the right-click menu

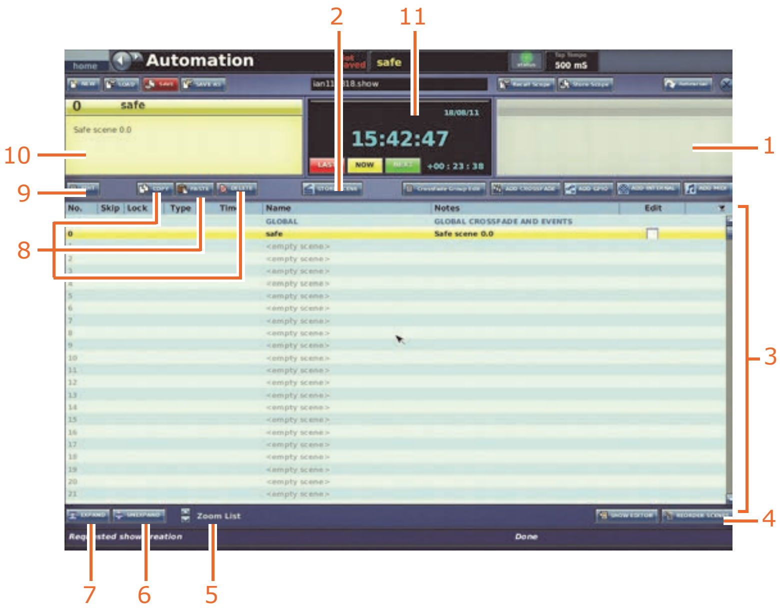
You can access some of the functions of the function buttons and also additional functions by right-clicking the desired scene. This opens a menu that has the following options:

- ***Edit:*** Opens the **Edit Scene Properties** window.
- ***Add:*** Opens a submenu with the following options:
  - ***Overwrite Scene:*** Overstores the scene with any changes made. For example, if you are working on scene 2, and you have made changes to it, right-click on scene 3 and then select **Add** > **Overwrite Scene,** and scene 3 will be overstored with the changes made to scene 2.
  - ***Insert Scene:*** Inserts the scene you have just copied immediately before this one.
  - ***Midi Event:*** Creates a MIDI event in the scene.
  - ***Internal Event:*** Creates an internal event in the scene.
  - ***GPIO Event:*** Creates a GPIO event in the scene.
  - ***Crossfade Event:*** Creates a crossfade event in the scene.
- ***Delete:*** Deletes the selected scene (see “Copying and deleting scenes” on page 185).
- ***Copy:*** Copies the selected scene (see “Copying and deleting scenes” on page 185).
- ***Paste:*** Pastes the scene you have just copied.
- ***Expand:*** Expands the scene/point scene (see “To expand a scene/point scene” on page 183.
- ***Un-Expand:*** Closes the point scenes of the scene/point scene (see “To expand a scene/point scene” on page 183).
- ***Multi-Edit:*** Opens a submenu with the following options:
  - ***Set List:*** Opens the **Set List** window.
  - ***Show Editor:*** Opens the **Show Editor** window.
- ***Invert Selection:*** Any scenes that have been ‘checked’ (that is, their check box in the **Edit** column contains an “X”) become unchecked, and vice versa.

- ***Clear Selection:*** Unchecks any scenes that have been checked.
- ***Exit:*** Closes the right-click menu.

Scenes

The scene management areas of the **Automation** screen are is intended for fast operation during show time and rehearsals. They let you edit, copy, delete, store and recall scenes, and can be broadly subdivided into the following areas.



*Scene-related elements of the **Automation** screen*

Item	***Element***	***Description***
1	Scene panel	Contains scene number, title and notes pertaining to the 'next' scene.
2	**STORE SCENE** button	See "Automation controls" on page 178.
3	Scene cue list	See "Scene cue list" on page 182.
4	**REORDER SCENES** button	See "Changing the order of the scenes" on page 186.
5	**Zoom List** spin buttons	See "Using zoom" on page 187.
6	**UNEXPAND** button	See "To close the point scenes of a scene/point scene" on page 183.
7	**EXPAND** button	See "To expand a scene/point scene" on page 183.
8	**DELETE, PASTE** and **COPY** buttons	See "Copying and deleting scenes" on page 185.
9	**EDIT** button	See "Editing scene properties" on page 184.
10	Scene panel	Contains scene number, title and notes pertaining to the 'now' scene.
11	Show information panel	See "Date and time" on page 182 and "Automation controls" on page 178.

For details of how to recall a scene and create a new scene from the currently selected one, see "Managing the scenes" on page 81.

Scene contents

A scene contains all of the control centre settings that existed at the point of creation, except:

- Anything that is explicitly taken out of store or recall using the automation scope controls.
- All solo, monitor and comms section controls.
- All surface selection or navigational control settings.

Additionally, each scene can contain:

- Scene information, including name and notes.
- Events — MIDI, GPIO, internal and crossfades.

Point scenes

For every scene there are 10 point scenes available, and each point scene has another 10 point scenes. Point scenes are the same as scenes. They allow each scene to be divided into smaller sections.

Numbering and navigation

As scenes need to be recalled in sequence, each scene requires a sequential number. So, although there is a maximum of 1000 scenes, the range of scene numbers is much greater to allow for gaps to be left for adding scenes without having to renumber the subsequent scenes — a major requirement in scripted shows. To facilitate this, each scene has an associated four-digit, two-decimal place scene number, giving a possible 99 point scenes per main scene. The scene number locates the scene in the sequence of stored scenes and is the basis of scene navigation.

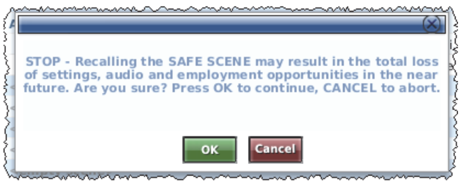
Safe scene (scene 0)

All scene numbers are available for storing scenes in except for scene 0, which is called the "safe" scene. This scene is the control centre's initial snapshot scene, which cannot be overwritten by the user. It is the only snapshot present when there is no show loaded and, when recalled (see important note below), it sets the control centre — regardless of scope settings (unless hardware safes are in place) — to a safe state in which it is not passing any audio.

The settings include:

- All mutes off.
- Gains are set to 0dB.
- All levels are at minus infinity (-∞) dB.
- All faders are at minus infinity (-∞) dB — except VCA faders, which remain at 0dB.

Important:
A warning (shown right) will appear when you try to recall the safe scene. Please heed the warning and click OK to continue or Cancel to abort.



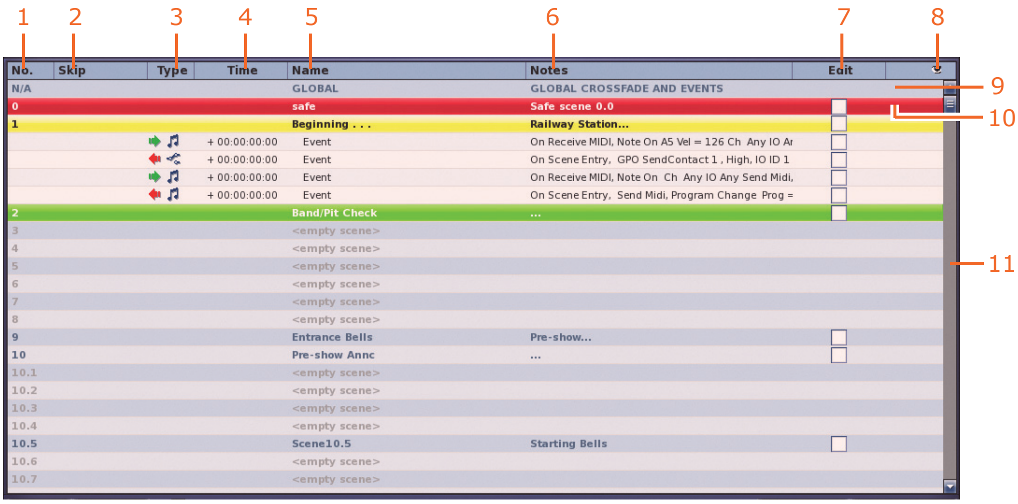
Date and time

The current date and time, and the duration of the current scene are displayed towards the top of the **Automation** screen.

Scene cue list

The scene cue list provides you with an overview of the show. It tells you at a glance where you are in the performance and provides scene information, such as scene number and title, some of which are editable. Other features allow you to alter settings, 'skip' scenes and edit scene properties.

The scene cue list provides you with an overview of the show. It tells you at a glance where you are in the performance and provides scene information, such as scene number and title. Other features let you alter settings, 'skip' scenes, edit scene properties and choose what to leave out of the cue list.



Elements of the scene cue list

Item	Element	Description
1	**No.** column	Number column shows the scene number and point scene number.
2	**Skip** column	Skip column. When you see a skip arrow in this column it means that this scene will be missed out during a rehearsal. For example, during rehearsal you may need skip scene 3 by going straight from scene 2 to scene 4 (auto status). Also, indicates scene selection when it contains an event (yellow circle).
3	**Type** column	Shows the type of events and whether they are incoming or outgoing.
4	**Time** column	Displays the time before an event is triggered. A blue countdown time bar shows the time remaining.
5	**Name** column	Title of scene/point scene or name of event.
6	**Notes** column	Scene notes.

Item	*Element*	*Description*
7	**Edit** column	Contains a tick box per scene/event, which is used for selection purposes when reordering scenes, see “Changing the order of the scenes” on page 186.
8	Eye symbol	Opens the **Show** window (see “Configuring the scene cue list view” on page 184).
9	**GLOBAL** scene	See “Additional control — managing events” on page 82.
10	**safe** scene	See “Safe scene (scene 0)” on page 181.
11	Scroll bar	Lets you scroll to the other scenes.

>> To select a scene/point scene

The ‘now’ scene is the currently selected scene (highlighted in yellow).

Do one of the following:

- At the GUI, click the scene/point scene in the **Automation** screen.
- At the GUI, click the **LAST**/**NOW**/**NEXT** button in the **Automation** screen as necessary.
- In the **automation** section (control surface), press the **last**/**now**/**next** buttons as necessary.

When recalling a scene, you can avoid replacing the current settings by using scope masks, see Chapter 21 "Scope (Automation)" on page 193.

>> To expand a scene/point scene

Select the scene/point scene and do one of the following:

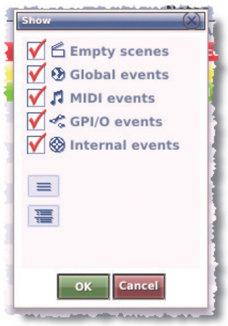
- Click **EXPAND.**
- Right-click the scene to open the right-click menu. Then, choose **Expand.**

>> To close the point scenes of a scene/point scene

Select the scene/point scene and do one of the following:

- Click **UNEXPAND.**
- Right-click the scene to open the right-click menu. Then, choose **Un-Expand.**

Configuring the scene cue list view



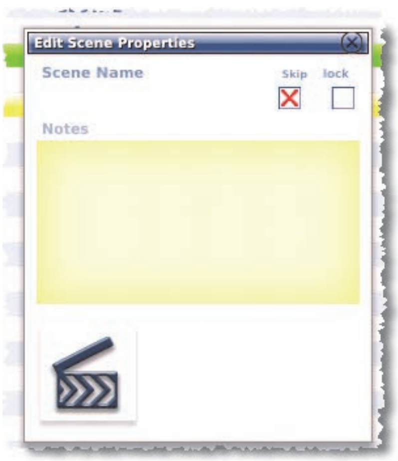
You can exclude any of the following elements from the scene cue list (see "Scene cue list" on page 182), which is done via the **Show** window (shown right):

- **Empty scenes**
- **Global events**
- **MIDI events**
- **GPI/O events**
- **Internal events**

>> To configure the scene cue list view

1. Click the eye symbol (right of the cue list title bar) to open the **Show** window.
2. In the **Show** window, select (tick) the desired option(s).
3. Click **OK.**

Editing scene properties



You can change the name of a scene, add/edit its notes (also editable in the **Store** window when storing a scene), choose to skip the scene during a rehearsal and choose to lock pending events and scene crossfades. This is all done via the **Edit Scene Properties** window (shown right).

The flexibility of the automation system means that complex schemes of events can lead to situations where the console is operating in potentially strange ways. For example, it is possible that an event could be set to an unintentionally long delay time or a jump event creates an endless loop. The lock option, when selected, provides a 'break' in this loop.

The **Scene Name** and **Notes** sections are edited just as you would any other text field.

>> To open the Edit Scene Properties window

Do one of the following:

- Select the scene and then click **EDIT.**
- Right-click on the scene and then choose **Edit** from the right-click menu.

>> To skip a scene during rehearsal

Open the **Edit Scene Properties** window of the desired scene, and then click the **Skip** box to place a red cross inside it. After you close the window, a skip arrow symbol will appear in the scene's **Skip** column.

>> To lock a scene's pending events or crossfades

Open the **Edit Scene Properties** window of the desired scene, and then click the **lock** box to place a red cross inside it. After you close the window, a lock symbol will appear in the scene's **Lock** column.

Adding a new scene

You can add a new scene anywhere in the cue list. The new scene can be inserted in the cue list or you can overwrite an existing scene, which replaces it with the new one.

>> To insert a new scene

1 Right-click the scene before which you want to insert the new one.

2 From the right-click menu, choose **Insert Scene.**

>> To overwrite an existing scene with a new one

1 Right-click the scene you want to overwrite.

2 From the right-click menu, choose **Overwrite Scene.**

Note: *If you are creating the first scene, use* ***STORE SCENE*** *(see "To create a new scene using the current settings" on page 81).*

Copying and deleting scenes

You can copy and delete single scenes/point scenes from the cue list.

>> To copy a scene

1 Do one of the following:

- Select the scene you want and click **COPY.**
- Right-click the scene and then choose **Copy** from the right-click menu.

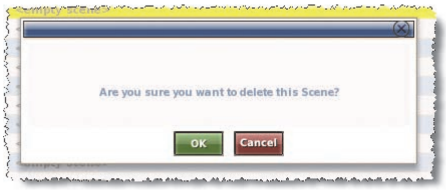
2 In the cue list, select the scene/point scene before which you want to paste the copied scene. Then, click **PASTE.**

>> To delete a scene

1 Do one of the following:

- Select the scene and then click **DELETE.**
- Right-click on the scene and then choose **Delete** from the right-click menu.

2 In the message window (shown right), click **OK**.



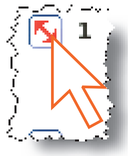
Changing the order of the scenes

You can change the order of the scenes in the cue list. This is done using **REORDER SCENES** button. You can reorder as many scenes as you want by selecting them in the order you want them to appear in the reordered list.

>> To reorder the scenes

1 In the Automation screen, click **REORDER SCENES.** The grey double arrowhead symbol will appear in each box in the **Edit** column.

2 In the cue list, click the box of the first scene you to reorder (as shown right). A "1" will appear in the **Edit** column, signifying that it will be the first of the reordered scenes, and the scenes will be reordered from this point on in the cue list.

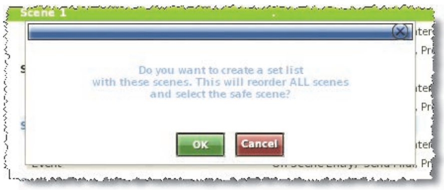


3 Repeat for the next scene you want to reorder. This will be labelled "2" in the **Edit** column.

4 Repeat for the remaining scenes/point scenes.

5 Click **REORDER SCENES.** The message window (shown right) will open.

6 Click **OK.**



Overriding store scope

You can choose to ignore the parameters selected at the **Store Scope** screen (Chapter 21 "Scope (Automation)" on page 193), so that these 'safed' parameters *will* be stored in the scene. This is selectable as a global option (for all scenes) in the **Preferences** screen and also on a per scene basis in the **Store Scene** window.

Note: *This feature does not affect scene recall.*

>> To override the safe parameters (selected in store scope) for a single scene

Open the **Store Scene** window (see "To create a new scene using the current settings" on page 81) and select the **Overwrite Safe parameters?** option.

>> To override the safed parameters (selected in store scope) for every scene

At a GUI screen, choose **home ▸ Preferences ▸ General** and then click the **User** tab to open the **Preferences User** screen. Then, select the **Overwrite "Safe" parameters** option in the **On Scene Store** section.

Using patching in automation

The Automate Patching option switches on per-scene automatic routing, and must be used with caution. To alert you to the drastic consequences of using this option, a WARNING window appears.

You can change the patching of certain sources and destinations on a per-scene basis. For example, you have an input channel's compressor side chain patched from one source in one scene and from a different source in another scene.

For details of the parameters that can be patched per scene, see Appendix G "Parameters Affected By Automate Patching" on page 393.

>> To use patching in automation

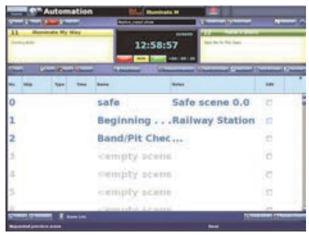
At a GUI screen, choose **home ▸ Preferences ▸ General** to open the **Preferences** screen, and select the **Automate Patching** option in the **Configuration Preferences** section.

Using zoom

You can enlarge the cue list to zoom in on certain scenes or make the scenes in the cue list smaller so that you can view more scenes simultaneously. This is done using the **Zoom List** spin buttons.

>> To enlarge the scene view (zoom in)

In the **Automation** screen, click the up (top) **Zoom List** spin button. The diagram right shows a typical **Automation** screen at maximum zoom.



>> To reduce the scene view (zoom out)

In the **Automation** screen, click the down (bottom) **Zoom List** spin button. The diagram right shows a typical **Automation** screen at minimum zoom.

Show files

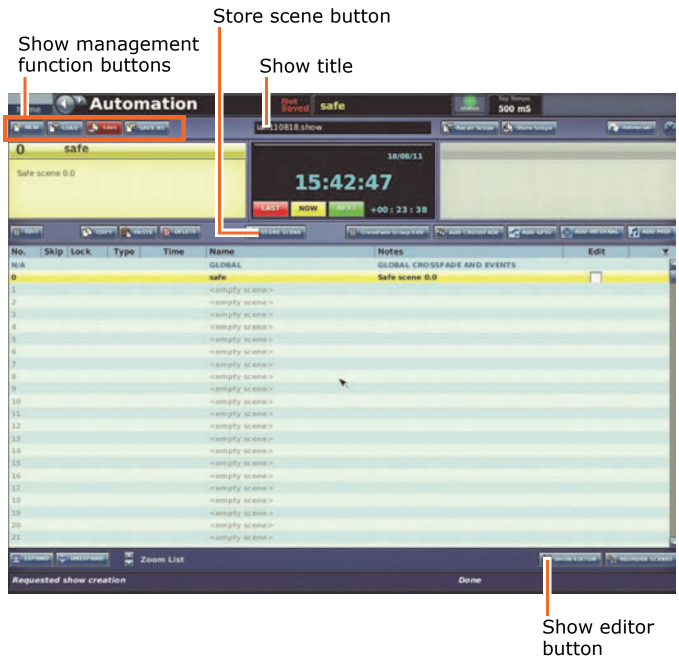
Show files are only handled via the GUI, using the **Automation** and **Files** screens of the GUI menu.

Managing show files

The **Automation** screen lets you create new shows, load existing ones and update the current show file with the latest settings.

For details of how to use the show function buttons to create a new show, save a show, create a new show from the current settings and load a show, see “Managing show files” on page 188.

The following table gives a description of each function button on the **Automation** screen that is used for the show files.



Legend	*Description*
NEW	For creating a new show (see “To create a new show” on page 79).
LOAD	For loading a stored show by restoring all stored snapshots and associated automation data from the selected show file (see “To load a show” on page 80).
SAVE	For backing up all stored snapshots and associated automation data to the selected/current show file (see “To save a show or create a new one from the current settings” on page 80). This button changes to red to show that there are show settings to be saved. We recommend that you save your show at regular intervals.
SAVE AS	Create a new show using the settings of the current one (see “To save a show or create a new one from the current settings” on page 80).
STORE SCENE	For details, see “To create a new scene using the current settings” on page 81.
SHOW EDITOR	For details, see “Show editor” on page 83.

Managing show files on the Files screen

Show files can be transferred between the PRO1 and an external USB device, such as a USB memory stick. This allows you to backup and archive your show files, so none will be lost, and also transfer them to other PRO1s. You can even create templates for new shows, so that you don't have to start from scratch, or modify existing show files. All this is done via the Files screen (see "About the Files screen" on page 219).

For details of how to back up/export your files and also how to import them from an external source, see "Saving your show files to a USB memory stick" on page 88.

PRO1s are compatible with other Midas digital consoles, such as the PRO Series and XL8.

Rehearsals

Rehearsal mode allows you to skip scenes/point scenes to match the arbitrary nature of the performance sequence during rehearsals.

>> To select a scene to 'skip' and to 'unskip' a scene

See "Editing scene properties" on page 184.

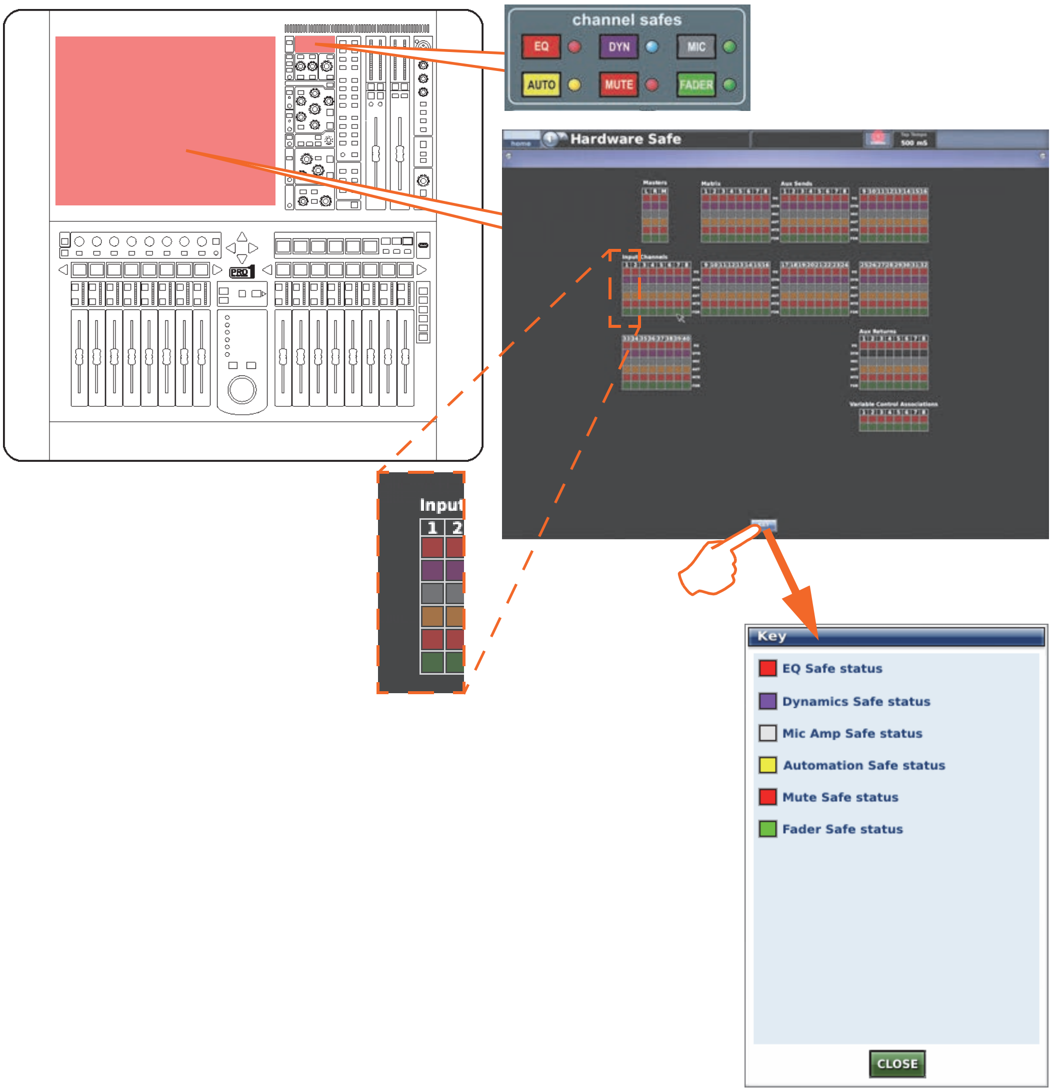
>> To carry out a rehearsal

1. Click **REHEARSAL.**
2. Carry out the rehearsal as necessary. (Note how the scenes selected as 'skipped' are missed out during the show's rehearsal, as you use the last, now and next buttons.)
3. To end the rehearsal, click **REHEARSAL.**

Safes

Important:
Safes are intended for emergency use only and are not to be confused with scope (see Chapter 21 "Scope (Automation)" on page 193).

PRO1 safes prevent certain controls from being recalled with a scene. Safe activation and status are provided on both the control surface (**channel safes** section) and the GUI (**Hardware Safe** screen).



*The **channel safes** screen on the control surface and the GUI **Hardware Safe** screen. Click **KEY** to open the **Key** window.*

There are six types of channel safe: EQ, dynamic, mic, auto, mute and fader. These are available on all channels except aux returns, which don't have dynamic safes. The VCAs have mute and fader safes. Solo (for monitor areas A and B) is always out of scene on any channel.

For information on the channel and group parameters for each safe, see Appendix H "Parameters Protected By Safes" on page 397.

The **Hardware Safe** screen shows all of the available safes for the channels and VCAs. The **KEY** button opens a **Key** window that shows you what the safe buttons represent.

>> To open the Hardware Safe screen

At the GUI, choose **home** ▸ ***Automation*** ▸ ***Hardware Safe.***

>> To switch a safe on/off

Do one of the following:

- At the GUI's **Hardware Safe** screen, click the desired safe to switch it on/off. This can be done for any safe.
- Use the appropriate button in the **channel safes** section with the appropriate channel assigned to the control surface.
- At the GUI's **VCA Groups** screen, click the desired safe button.

Chapter 21: Scope (Automation)

This chapter shows you how to use the scope feature of PRO1 automation to include/exclude specific parameters on scene store/recall.

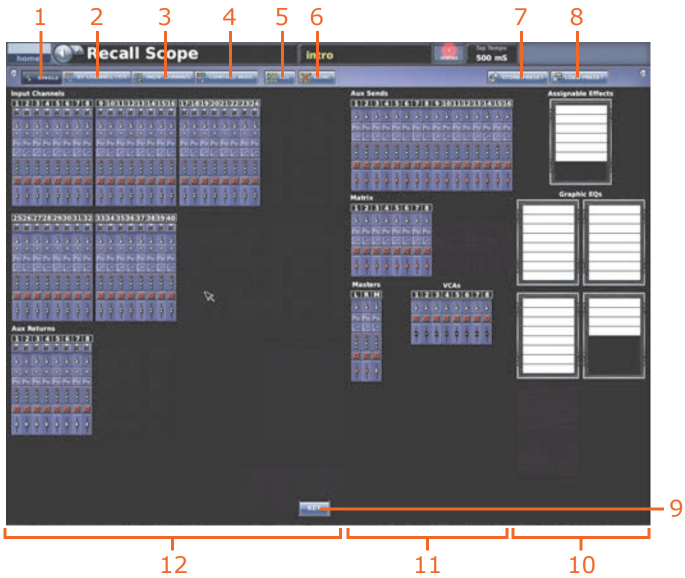
Although scope has two functions, recall and store, the emphasis in this chapter is on recall scope, which will be the most commonly used of them both. Store scope will only be required in certain circumstances, and even then it must only be used with caution (see “Using store scope” on page 199).

About scope

Scope lets you define the extent of the automated controls for all channels, buses, groups, assignable effects and GEQs. To do this it has a **Recall Scope** screen from which you can select the controls that are excluded from the scene when it is recalled and you can also view the current scope status.

About the Recall Scope screen

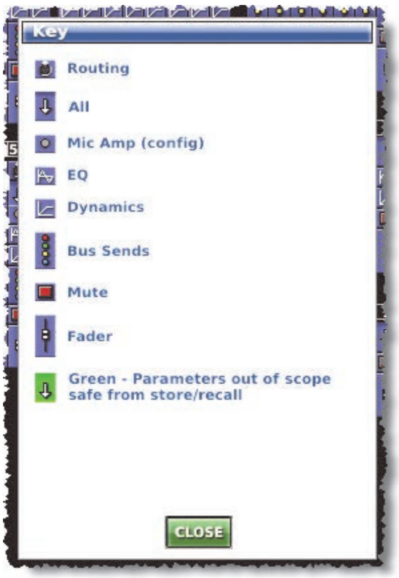
The **Recall Scope** screen has a number of type-specific areas, such as **Input Channels, Aux Sends, VCAs** and **Graphic EQs,** which contain user-selectable parameter sections that you can make ‘out of scene’ on scene recall.



*Elements of the PRO1 **Recall Scope** screen*

Item	*Element*	*Description*
1	**SINGLE** button	Scope function button for selecting single parameter sections on the scope screen.
2	**BY CHANNEL TYPE** button	Scope function button for selecting the same parameter section in all channels of a single type on the scope screen.

Item	*Element*	*Description*
3	**INDV. CHANNEL** button	Scope function button for selecting all of the parameter sections of a single channel on the scope screen.
4	**CONSOLE WIDE** button	Scope function button for selecting all of the parameter sections of every channel on the scope screen.
5	**ALL** button	Scope function button that selects all of the parameter sections on the scope screen, that is, for every channel, assignable effect and internal GEQ.
6	**NONE** button	Scope function button that deselects all selected parameters sections on the scope screen.
7	**STORE PRESET** button	See Chapter 24 "User Libraries (Presets)" on page 215.
8	**LOAD PRESET** button	See Chapter 24 "User Libraries (Presets)" on page 215.
9	**KEY** button	Opens the Key window (below), which shows you what each parameter section symbol represents.
10	Rack units	Area for the assignable effects and GEQs.
11	Output channels	Area for the aux sends, matrices, VCAs and masters.
12	Input channels and groups	Area for the inputs and aux returns.



*The **Key** window, which is opened by clicking the **KEY** button*

The **Recall Scope** screen has a section for each of the following.

Description	***Input Channels***	***Aux Returns (returns)***	***Aux Sends (auxes)***	***Matrix (matrices)***	***Masters***	***Variable Control Associations (VCA and POP. groups)***	***Assignable Effects (internal effects) and Graphic EQs***	***Symbol***
Routing	Yes	Yes	No	No	No	No	N/A	
All	Yes	Yes	Yes	Yes	Yes	Yes	N/A	
Mic Amp (config)	Yes	Yes	No	No	No	No	N/A	
EQ	Yes	Yes	Yes	Yes	Yes	No	N/A	
Dynamics	Yes	No	Yes	Yes	Yes	No	N/A	
Bus Sends	Yes	Yes	Yes	No	Yes	No	N/A	
Mute	Yes	Yes	Yes	Yes	Yes	Yes	N/A	
Fader	Yes	Yes	Yes	Yes	Yes	Yes	N/A	
Assignable Effects/Graphic EQ	N/A	N/A	N/A	N/A	N/A	N/A	Yes	

>> To open the Recall Scope screen

At either GUI screen, choose **home** ▸ ***Automation*** ▸ ***Recall Scope.***

Selecting scope parameter sections

The scope function buttons provide a number of ways of selecting/deselecting scope parameters, such as singly, by channel type, all on the control centre etc.

When a parameter is selected, its background changes from blue to green.

Selection is cumulative, so you can combine any of the selection/deselection functions until all the desired parameter sections have been selected. When you have finished the selection process, just go to the next screen you require; you don't have to save your selection, as it remains stored in the current condition until you alter it again.

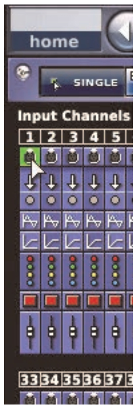
The "Assignable Effects" panel lets you choose which effects are 'out of scene' on scene recall.

For details of the parameters affected by scope, see Appendix F "Parameters Affected By Scope" on page 349.

>> To select a single parameter section

1 Click **SINGLE.**

2 Click the desired parameter section (shown right).



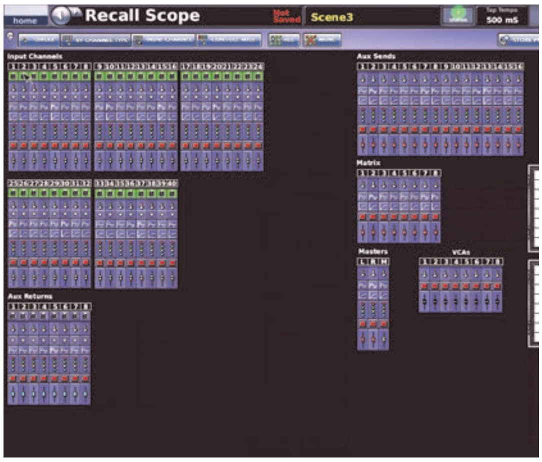
In some cases more than one parameter section may be selected. This may occur if:

- The parameter section belongs to a channel that is stereo linked. The equivalent parameter section of the other paired channel will also be selected.
- Other channels are patched to the same source as the channel in which you are making your selection. This only applies to the **All** and **Mic Amp** parameter sections.

>> To select a parameter section in all channels of a single type

1 Click **BY CHANNEL TYPE.**

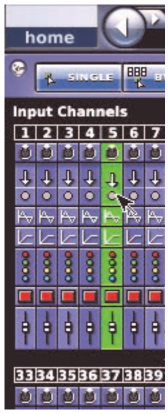
2 Click the desired parameter section in any channel of the desired type. For example, click the routing parameter section in any of the input channels. The routing parameter section on all input channels will be selected (as shown right).



>> To select all of the parameters of a single channel

1 Click **INDV. CHANNEL.**

2 Click any parameter section in the desired channel. For example, click the **All** parameter of input channel 5; all the parameter sections in channel 5 are selected (as shown right).

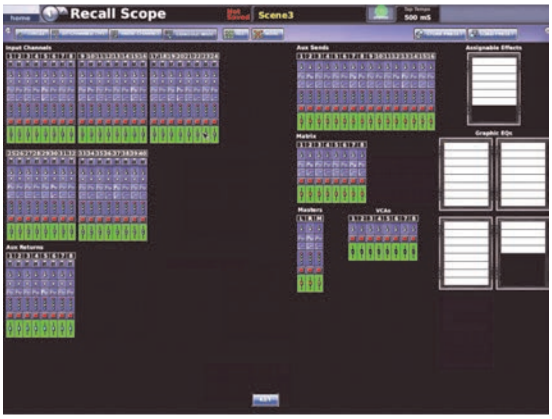


If the channel is stereo linked, all of the parameter sections in its paired channel will also be selected.

>> To select a single parameter section console wide

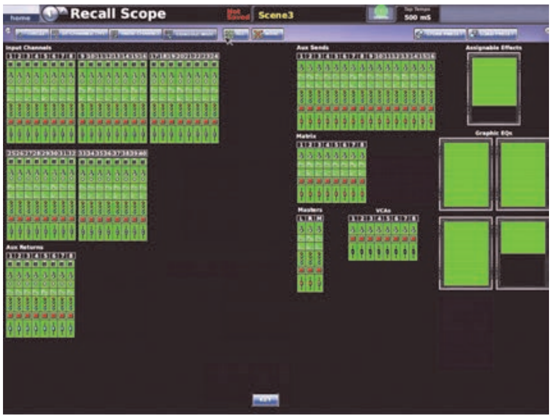
1 Click **CONSOLE WIDE.**

2 In any channel, click the desired parameter section. For example, clicking the fader parameter of input channel 1 selects the fader parameter of every channel.



>> To select every parameter section on the console

Click **ALL.** Every parameter section on the **Recall Scope** screen is selected (as shown right).

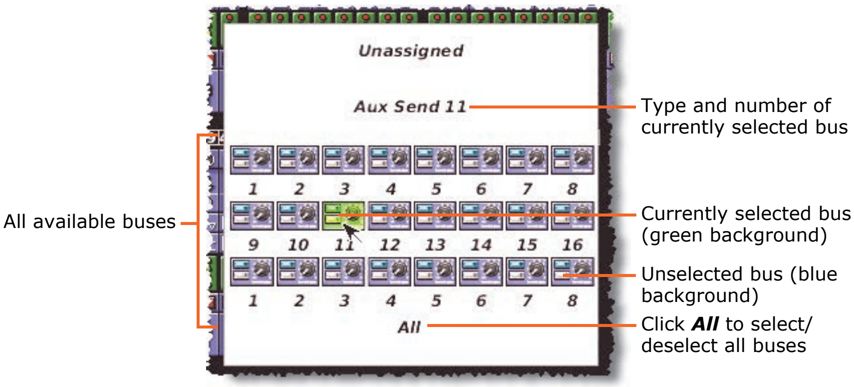


>> To deselect a parameter section(s)

Follow the procedures for selecting parameters, but only click ones that are already selected.

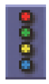
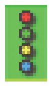
Selecting bus parameters

As a channel's bus parameter section represents all of its available buses collectively, you can only select either all or none of its buses using the scope function buttons. However, by clicking the bus parameter section you can open its 'bus select' window, which lets you select any buses you want out of scene. In this window each aux/matrix bus is represented by an single icon, and only available buses are shown.



Typical 'bus select' window

The background colour of the bus parameter for each channel shows the selection status of its respective buses. There are three states, as shown in the following table.

Bus selection status	***Symbol***
None of the buses are selected	
One or more, but not all, of the buses are selected	
All of the buses are selected	

>> To select/deselect buses

In the 'bus select' window, do one of the following:

- To select/deselect a single bus, click on its icon.
- To select/deselect all buses, click ***All.***

Saving scope parameters in a scene

Scope parameters have to be saved in a scene.

>> To save your selected parameters in a scene

1. Save the parameters you want into a scene (see “To create a new scene using the current settings” on page 81).
2. Select the desired recall scope parameters (see “Selecting scope parameter sections” on page 195).
3. Overwrite the scene by clicking the “Overwrite scene” option (see “To create a new scene using the current settings” on page 81).

Using store scope

! 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 store and aways store everything.

Please use store scope with great care. All of the methods of the recall scope operation, as detailed in this chapter, apply equally to store scope.

Chapter 22: Events (Automation)

This chapter shows you how to use the events of the PRO1's automation.

About events

There are four types of event — MIDI, GPIO, internal and crossfade — that you can have in a scene, and you can have any combination of each. You can choose whether the event is triggered on the PRO1 or on an external device.

For more information on events and also how to create, edit and copy/paste an event, see "Additional control — managing events" on page 82.

MIDI

MIDI performs two functions on the PRO1. It allows the PRO1 to trigger external MIDI-equipped equipment on each scene change and it also allows external MIDI equipment to trigger a PRO1 scene change.

MIDI output from the PRO1 can include a globally-enabled outgoing message that contains the recalled scene number and is sent out for all recalled scenes. Also, up to eight messages with user-selectable content are stored per scene and sent out whenever the scene is recalled.

MIDI input can be globally set up to cause scenes to be recalled when specific incoming MIDI messages are encountered.

GPIO

The general purpose input and output (GPIO) on the PRO1 is used to control/respond to various devices. You can use GPIO inputs to control PRO1 parameters from an external device, for example, you can use an external switch to switch the PRO1's talkback on/off or you can use an external switch or joystick to control the PRO1's parameters. You can also use the PRO1's keys and faders to send control signals to an external device.

Internal

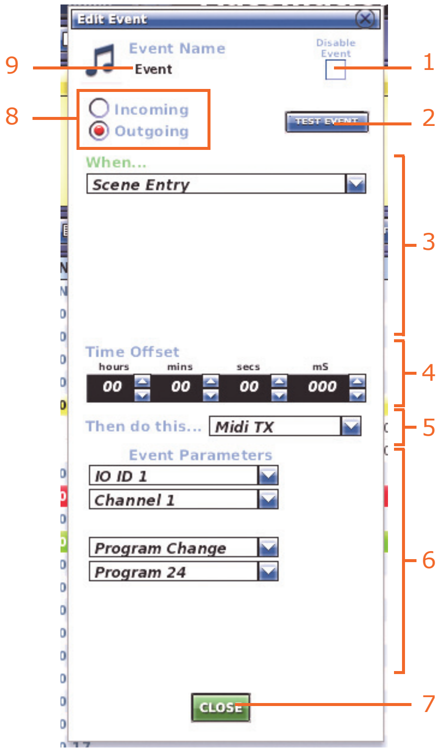
You can create an event on the PRO1 without using an external device; this type of event is called an "internal" event. This means that an event is triggered and carried out entirely on the PRO1.

Crossfades

For information on crossfades, see Chapter 23 "Crossfades (Automation)" on page 207.

About the Edit Event window

You can edit the parameters of an event on the GUI in the **Edit Event** window.



Item	*Description*
1	**Disable Event** tick box, for choosing whether the event is missed out (skipped) during a rehearsal.
2	**TEST EVENT** button, for executing the selected outgoing MIDI or GPIO event using the current event parameters.
3	**When...** section for selecting the parameters that trigger the event. Displays either one or three drop-down lists, depending on the whether the event is selected as **Outgoing** or **Incoming,** respectively.
4	**Time Offset** section, for setting the period of time that the event happens after it has been triggered. Zero = no offset.
5	**Then do this...** section, for choosing the type of event that you want.
6	**Event Parameters** section (see the "Programming events" on page 203).
7	**CLOSE** button, for closing the **Edit Event** window.
8	Incoming/outgoing selection section, for selecting whether the event is triggered on the PRO1 or on an external device.
9	Text field, for displaying the user-configured event name.

Figure 15: Edit Event window

Programming events

Each type of event is programmed in a similar way, regardless of whether it is an incoming/outgoing MIDI or GPIO event, or an internal event. However, the options in the **Edit Event** window may vary depending on the chosen event.

>> To program an event

1. Open the **Edit Event** window (see "To edit an event" on page 82).
2. In the **Event Name** section, type in the event name. If you want to skip this event during a rehearsal, select the **Disable Event** option.
3. To select whether the event occurs on the PRO1 or an external device, click **Incoming** or **Outgoing**, respectively. (This is not applicable to internal events.)
4. Select the desired parameters in the **When...**, **Then do this...** and **Event Parameters** sections. To help you, refer to Table 6 "Outgoing event options" (below), Table 7 "Incoming event options" on page 204 and Table 8 "Description of all event option parameters" on page 204.
5. If you want to incorporate a time delay between the event being triggered and the event itself, select a time in the **Time Offset** section (click the up/down spin buttons).
6. Click **CLOSE.**

Table 6: Outgoing event options

When ...	***Then do this...***	***Event Parameters (from top list downwards)***				
Scene Entry,	***Midi TX***	***IO ID1*** to ***IO ID 18***	***Channel 1*** to ***Channel 16***	***Note Off,***	***A0*** to ***C7***	***Velocity 0*** to ***127***
Scene Exit,				***Note On,***	***A0*** to ***C7***	***Velocity 0*** to ***127***
Scene Recall,				***Aftertouch,***	***A0*** to ***C7***	***Pressure 0*** to ***127***
Scene Entry And Recall				***Pressure,***	N/A	***Pressure 0*** to ***127***
Scene Entry And Exit				***Control Change,***	***All Notes Off, Reset All***	N/A
Scene Entry, Exit And Recall				***Program Change***	***Program 0*** to ***128***	N/A
	GPIO TX	***IO ID1*** to ***IO ID 18***	***Static Low, Static High***	***Closure 1*** to ***Closure 8***	N/A	N/A
	Last	N/A	N/A	N/A	N/A	N/A
	Next	N/A	N/A	N/A	N/A	N/A
	Now	N/A	N/A	N/A	N/A	N/A
	Jump	List of scene titles	N/A	N/A	N/A	N/A
	Notes Event	**Event Parameters** section changes to a **Notes** window, where you can enter event notes				
	X-Fade Event	**Event Parameters** section changes to crossfades parameter display				

Table 7: Incoming event options

When … (from top list downwards)			***Then do this…***	***Event Parameters (from top list downwards)***
Any IO Box, ***IO ID1*** to ***IO ID 18***	***Any MIDI Channel,*** ***Channel 1*** to ***Channel 16***	***Note Off,*** ***Note On,*** ***Aftertouch,*** ***Pressure,*** ***Control Change,*** ***Program Change,*** ***Pitch Wheel***	***Midi TX*** ***GPIO TX*** ***Last*** ***Next*** ***Now*** ***Jump*** ***Notes Event*** ***X-Fade Event***	For details, see Table 6 "Outgoing event options" on page 203

Table 8: Description of all event option parameters

Parameter	***Description***
Aftertouch	How hard a key is pressed after it has been touched, that is, it changes the pressure after the note has been hit. Typically, aftertouch is useful for adding tremolo or vibrato effects to a sound, just as a violin can add volume or pitch changes to a sustained note using finger vibrato or addition bowing intensity. Parameters for **Aftertouch** are notes A0 to C7, with each having a possible pressure of between 0 and 127. You can also choose between **Enable MIDI Byte 1** and **Enable MIDI Byte 2.**
Any IO Box	The trigger can be on any IO port of any IO box.
Any MIDI Channel	Any of the MIDI channels.
Channel *n*	One of the 16 MIDI channels.
Closure *n*	Provides a contact closure that can be programmed to open or close in response to a MIDI event.
Control Change	Select the control changes that can be applied to a note in progress. For example, by altering the volume (not velocity) or adding sustain to a note (holding it for longer). Parameters are **All Notes Off** and **Reset All.** You can also choose between **Enable MIDI Byte 1** and **Enable MIDI Byte 2.**
FOH MIDI PORT	The trigger is via the MIDI port of the FOH DL251/DL252 Audio System I/O.
GPIO TX	Selects a GPIO event.
IO ID*n*	The trigger is via a specific IO port.
Jump	Opens a specified scene on the PRO1.
Last	Opens the last (previous) scene on the PRO1. This scene is the same one that would be opened if you pressed the **last** button.
MIDI TX	Selects a MIDI event.

Parameter	Description
Next	Opens the next scene on the PRO1. This scene is the same one that would be opened if you pressed the **next** button.
Now	Opens the 'now' scene on the PRO1. This scene is the same one that would be opened if you pressed the **now** button.
Note Off	Informs the instrument to stop playing a note at a specified pitch and velocity. Parameters for **Note Off** are notes A0 to C7, with each having a possible velocity of between 0 and 127. You can also choose between **Enable MIDI Byte 1** and **Enable MIDI Byte 2.**
Note On	Informs the instrument to start playing a note at a specified pitch and velocity. Parameters for **Note On** are notes A0 to C7, with each having a possible velocity of between 0 and 127. You can also choose between **Enable MIDI Byte 1** and **Enable MIDI Byte 2.**
Notes Event	Using this option, you can displays notes that may be useful at a certain point in the scene.
Pitch Wheel	Use the pitch wheel to trigger the event. The pitch wheel is a wheel type device, normally found to the left of a synthesizer keyboard, that manipulates the pitch of a played note(s).
Pressure	Pressure applied to the key that is being pressed. This affects, for example, the vibrato of the note being played. Parameters are between 0 and 127. You can also choose between **Enable MIDI Byte 1** and **Enable MIDI Byte 2.**
Program *n*	One of the 128 programs.
Program Change	Changes the device to a particular patch/voice/preset etc. Parameter are 0 to 127. You can also choose between **Enable MIDI Byte 1** and **Enable MIDI Byte 2.**
Scene Entry	Triggers the event when a scene is opened.
Scene Exit	Triggers the event when a scene is closed.
Scene Recall	Triggers the event when the 'now' scene is reloaded, but not jogged to.
Scene Entry And Exit	Triggers the event when a scene is opened or closed.
Scene Entry And Recall	Triggers the event when a scene is opened or the 'now' scene is reloaded (but not jogged to).
Scene Entry, Exit And Recall	Triggers the event when a scene is opened, closed or the 'now' scene is reloaded (but not jogged to).
Static High	External device is closed/switched off.
Static Low	External device is opened/switched on.
X-Fade Event	Lets you configure the crossfade event.

Chapter 23: Crossfades (Automation)

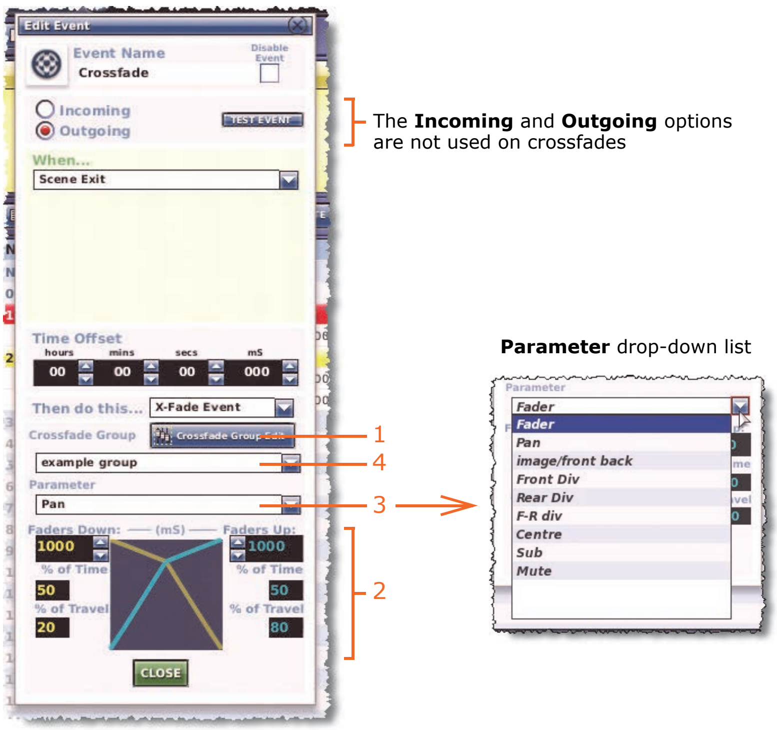
This chapter shows you how to use crossfades.

Crossfades are events that are triggered using the standard event mechanism, and are managed via the **Automation** screen. A crossfade event is managed in a similar way to any other event, such as GPIO and MIDI, and is detailed later on in this section.

A crossfade event is a trigger to change the value of a control — most often but not always a fader — between two levels, that of the current control position and that of the position in another scene, over time. If the level in the next scene is higher than the current level, the crossfade uses the 'fade up' time; if it is lower, it uses the 'fade down' time.

About the crossfade Edit Event window

You can edit the parameters of a crossfade event in the **Edit Event** window.



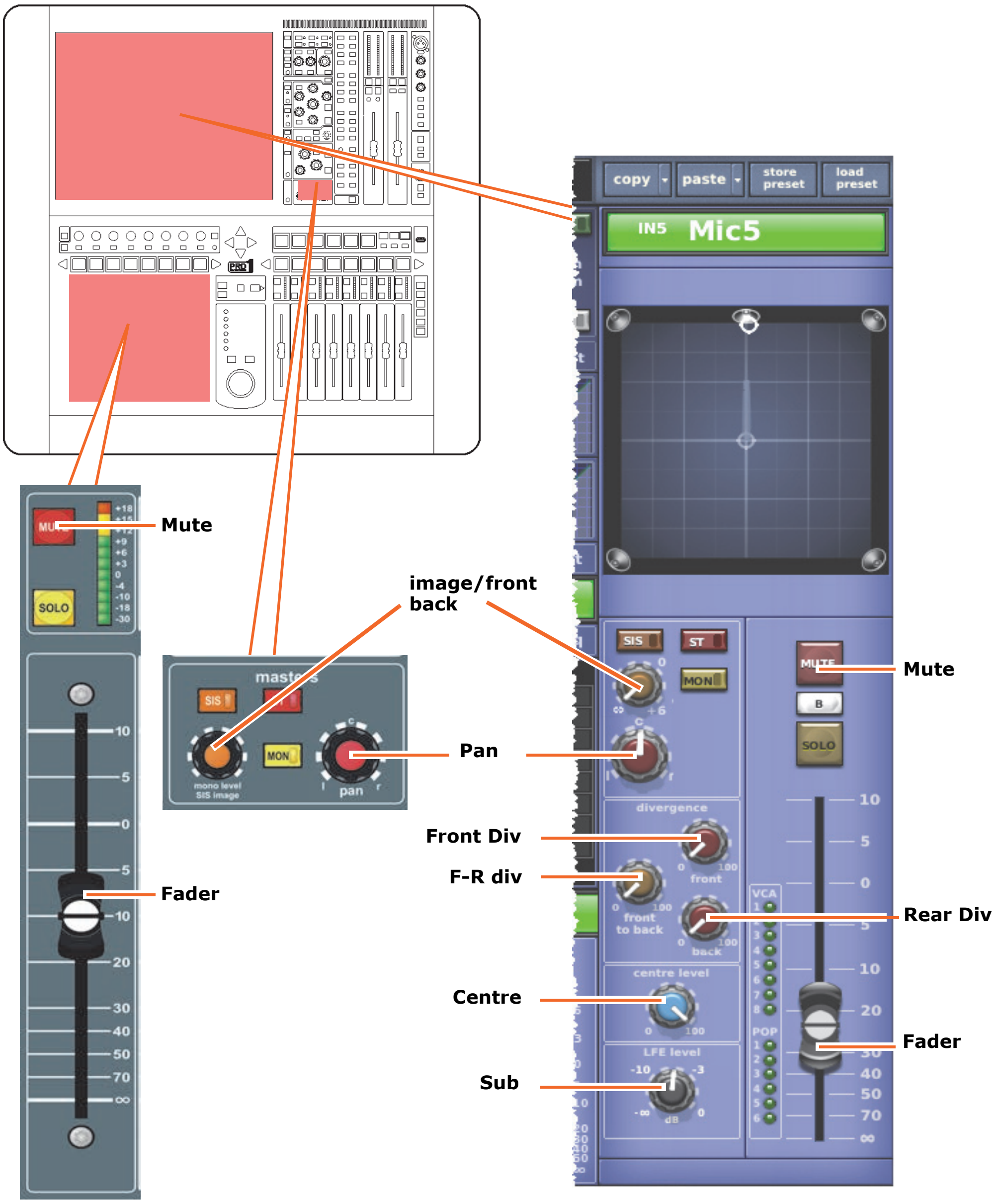
*Crossfade **Edit Event** window (for details of the Edit Event window elements that are common to each event type, see Figure 15 "Edit Event window" on page 202)*

Item	***Element***	***Description***
1	**Crossfade Group Edit** button	Opens the **Crossfade Groups** screen (see "Crossfade groups" on page 211).
2	Crossfade set up section	See "Crossfade set up section in the Edit Event window" on page 209.

Item	*Element*	*Description*
3	**Parameter** drop-down list	For choosing the level control that you want the crossfade to operate on (see "About the crossfade parameters" on page 208).
4	**Crossfade Group** drop-down list	Contains all of the available user-configured crossfade groups. Also includes the default **example group** that contains all possible crossfade sources.

About the crossfade parameters

The following diagram shows the PRO1 configured for 5.1 surround mode, which utilises each available parameter option. The presence of the **divergence**, **centre level** and **LFE level** sections are dependent on the currently selected surround mode.



Using a crossfade mute

The **Mute** option of the **Parameters** list lets you initiate a mute at the end of a crossfade down operation. For example, if you set a crossfade of two seconds, the mute will turn on after this time has expired (provided it was off). If the crossfade is a 'crossfade up', the mute will turn off during the crossfade up time (provided it was on).

Crossfade set up section in the Edit Event window

The crossfade set up section (bottom of the **Edit Event** window) is where you set up how the crossfade operates. Here you can set up the duration of the crossfade and the rate at which it occurs. You can configure two crossfade rates per crossfade or keep its rate constant throughout.

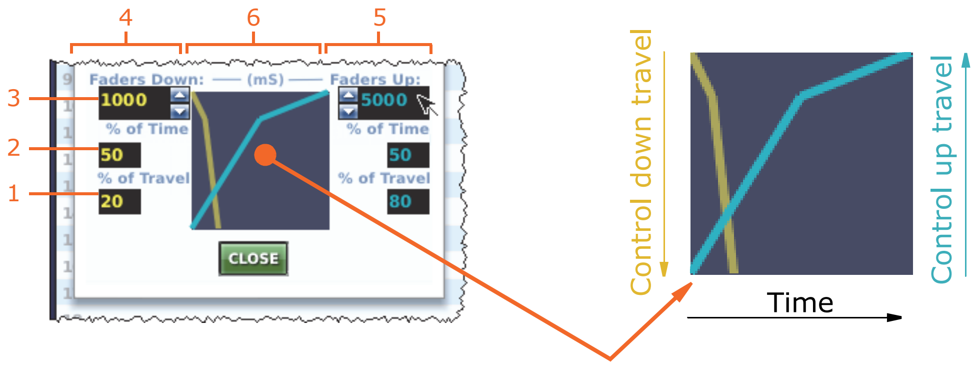


Figure 16: Crossfade set up section

Item	***Element***	***Function***
1	**% of Travel** number field	Sets the initial travel distance of the control as a percentage of the total distance of travel.
2	**% of Time** number field	Sets the time for the initial travel of the control as a percentage of the overall time.
3	**Faders Down** field	Sets the time taken for the total travel (milliseconds) of the control for a 'fader down' (or whichever control is used) crossfade event. For the **Faders Up** field, this sets the time taken for the total travel (milliseconds) of the control for a 'fader up' crossfade event.
4	Faders down section	For setting the crossfade down parameters.
5	Faders up section	For setting the crossfade up parameters.
6	Graph	Up/down crossfade graph.

>> To quickly adjust the time and travel of the faders up/down

You can quickly adjust the **% of Time** and **% of Travel** parameters by dragging the graph. Click anywhere on the line of the graph in the **Edit Event** window and drag to where the parameters are as desired. Clicking while pressing the left button adjusts the down travel, and doing the same with the right button adjusts the up travel.

>> To create a crossfade event

1 Open the **Edit Event** window.

2 If you want to disable this event, select the **Disable Event** option.

3 In the **When...** section, select the parameters that will trigger the event.

4 If you want to incorporate a time offset delay between the event being triggered and the start of the event, set a time in the **Time Offset** section.

5 In the **Then do this...** drop-down list, select the **X-Fade Event** option.

6 Do one of the following:

- Select a crossfade group from the **Crossfade Group** drop-down list.
- Create a new crossfade group. Click the **Crossfade Group Edit** button to open the **Crossfade Groups** screen and then follow the instructions in "Crossfade groups" on page 211.

7 In the **Parameters** section, select the level control on which you want the crossfade event to occur (see "Programming events" on page 203). For example, a fader.

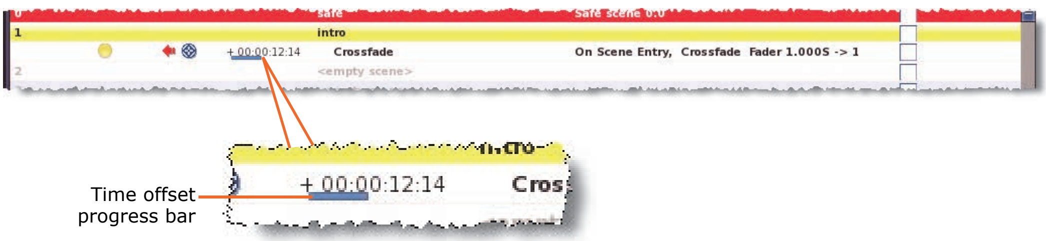
8 In the crossfade set up section (see "Crossfade set up section in the Edit Event window" on page 209), set up the crossfade parameters, such as time, % of travel etc.

9 Click **CLOSE.**

To set up a crossfade to have a constant rate across its full travel, set both the **% of Time** and **% of Travel** fields to 50%.

How a crossfade operates

When the crossfade event is triggered, the time offset (if configured) will start.



In the cue list, a blue status bar in a crossfade event will show the progress of a time offset

After the time offset has finished, the crossfade will start; this will be either a down or up crossfade, depending on the current control level. During the crossfade, the control level will alter at the configured rate, shown on both the control surface and GUI. Progress is shown in real time on the **Automation** screen.

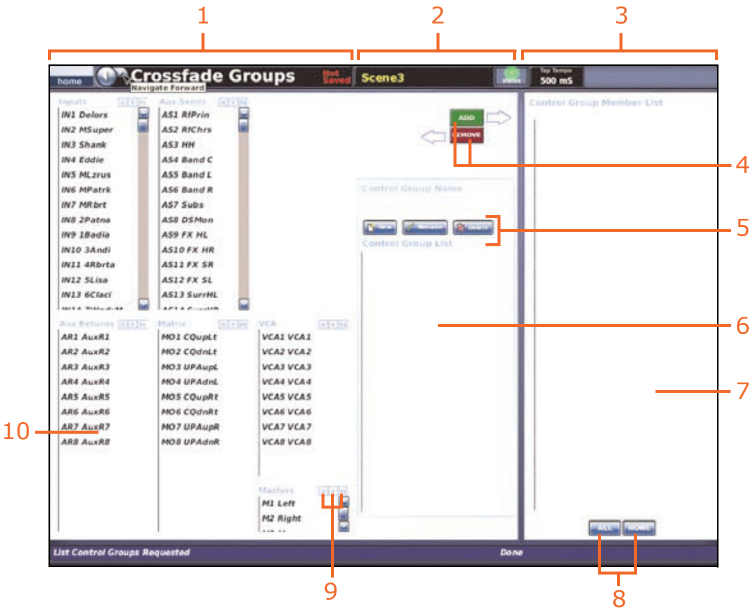
Typical crossfade graph in the ***Automation*** *screen. The blue vertical bar will travel from left to right according to the time elapsed and at the configured crossfade rate.*

__Note:__ The graph display shows the current longest crossfade in progress. So, if a delayed crossfade starts during the current one, and it is longer than the current one, the graph will change to show the new crossfade.

You can manually override the crossfade using the controls in the automation section of the output bay (see "Manually controlling a crossfade" on page 214).

Crossfade groups

Crossfade groups let you choose which channels/buses/groups will contain the crossfade. These groups are managed at the GUI's **Crossfade Groups** screen.



***Crossfade Groups** screen*

Item	Description
1	Crossfade control group member source panel. Contains panels of channels, buses and groups from where you select the sources for your crossfade group.
2	Crossfade group management section, where you can create new crossfade groups or delete existing ones, add/remove members to/from the currently selected group, and select the desired group from the **Control Group List.**
3	Crossfade group member panel.
4	**ADD** and **REMOVE** buttons. These buttons add or remove the currently selected members to or from the **Control Group Member List**, respectively.
5	**New, Rename** and **Delete** buttons. These buttons let you create, rename or delete crossfade control groups, respectively.
6	**Control Group List,** which shows the existing crossfade control groups.
7	**Control Group Member List,** which shows the current members of the selected crossfade control group.
8	**ALL** and **NONE** buttons. These buttons select all or none of the members in the **Control Group Member List**, respectively.

Item	Description
9	**A, I** and **N** buttons. Each section in the crossfade source panel has a set of these buttons, which select all members in the list, invert the current list selection or deselect all members in the list, respectively.
10	A panel containing a list of channels, buses or groups available for crossfade group membership.

>> To open the Crossfade Groups screen

At the GUI menu's **Automation** screen, click **Crossfade Group Edit.**

>> To create a new crossfade group

1 At the **Crossfade Group** screen, click **New.**

2 In the **Enter new control group name:** prompt window, type in your chosen name for the new crossfade group.

3 Click **OK.** The new group will appear in the **Control Group List.**

>> To edit a crossfade group

1 In the **Crossfade Group List** panel, select the crossfade group you want to edit.

2 Do one of the following:

- To add members to the group, select the members from the lists of inputs, auxes, returns, matrices, masters and groups at the left of the screen and then click **ADD.** The selected items will be moved to the **Control Group Member List.**
- To remove members from the group, select the members in the **Control Group Member List** that you want to remove and then click **REMOVE.** The selected items will be moved back to their respective panels in the left of the **Crossfade Group** screen.

If you want, you can edit the "example group" crossfade control group.

>> To rename a crossfade group

1 In the **Crossfade Group List** panel, select the crossfade group you want to rename.

2 In the **Rename control group as:** prompt window, type in the new name for the crossfade group.

3 Click **OK.** The new name will appear in the **Control Group List.**

If you want, you can rename the "example group" crossfade control group.

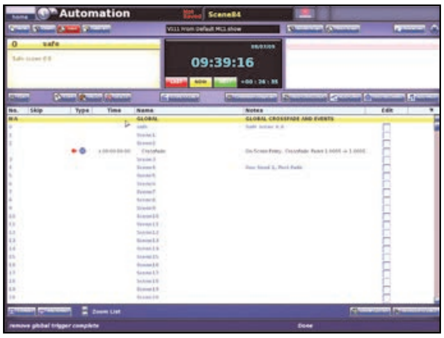
>> To delete a crossfade group

In the **Crossfade Group List** panel, select the crossfade group you want to delete and then click **Delete.**

You cannot delete the "example group" crossfade control group.

Global events

The **GLOBAL** scene, at the top of the cue list in the **Automation** screen, lets you have the same crossfade(s) in every scene. However, any scene-based crossfade(s) will override the global one(s) if both are present.



>> To set up a global crossfade

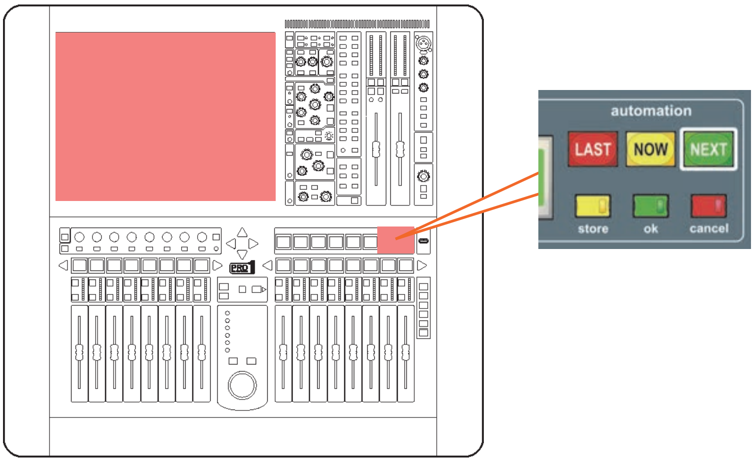
Select the **GLOBAL CROSSFADE AND EVENTS** scene and do one of the following:

- Click **ADD CROSSFADE.**
- From the right-click menu, choose ***Add ▸ Crossfade Event.*** The crossfade will appear in the **GLOBAL CROSSFADE AND EVENTS** scene.

***GLOBAL** scene (highlighted in yellow) in the **Automation** screen*

Manually controlling a crossfade

The controls in the **automation** section let you manually override the crossfade.



*The **cancel, now** and **ok** buttons in the automation section*

Control	***Function during crossfade***
cancel button	Pauses the crossfade. Pressing the **cancel** button again, while the crossfade is paused, cancels the crossfade. ***Note:*** *The level of the control on which the crossfade is operating will remain at the point at which it was paused. If you restart the crossfade the control will travel over the full crossfade period, that is, if you stop (rather than pause) a five-second crossfade at two seconds and restart it, it will take the control five seconds to move to the final position, and not three seconds.*
now button	Pressing this button while the crossfade is paused (by pressing the **cancel** button), continues the crossfade.
ok button	Jumps to the end of the crossfade, effectively cancelling the remaining time to the end of the crossfade.

Chapter 24: User Libraries (Presets)

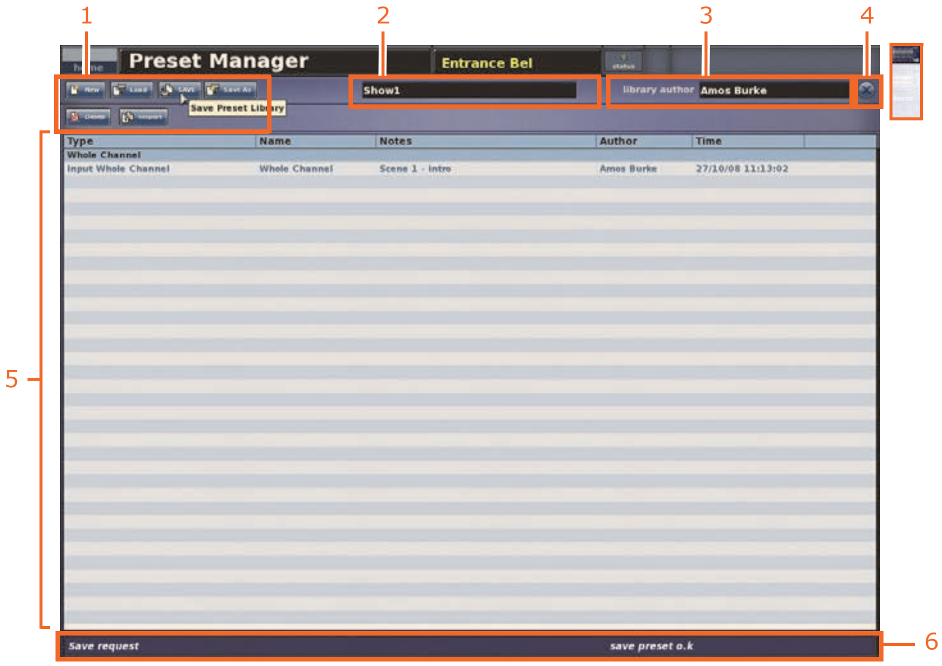
User libraries is a GUI only feature that provides a method of handling presets. The GUI menu has an option that opens the **Preset Manager** screen.

For more information on presets, including details of how to save and load a preset, see "User library (presets)" on page 85.

About the Preset Manager screen

Using the **Preset Manager** screen, you can create new user libraries, load existing ones or save the current one. You can create a new user library from scratch or save the current one under a new name. Similarly to show files, you can import libraries from external storage devices (USB memory sticks).

The **Preset Manager** screen also allows you to delete presets from the currently loaded library.



Item	Description
1	**Preset Manager** screen function buttons for creating/loading/saving/importing user libraries. For a description of each button, see Table 9 "Description of Preset Manager screen function buttons" on page 216.
2	User library name.
3	**library author** field, displays the name of user library author/creator.
4	Close button, closes the **Preset Manager** screen.
5	List of presets in the user library. For a description of the column titles, see Table 10 "Description of preset list titles" on page 216.
6	Operation status information.

Table 9: Description of Preset Manager screen function buttons

Button	***Use to***
New	Create a new preset library.
Load	Load a preset library.
SAVE	Save any unsaved changes that have been made to the currently loaded preset library.
Save As	Create a new preset library from the current one.
Delete	Delete the currently selected preset.
Import	Import all presets from a preset library into the one currently loaded.

Table 10: Description of preset list titles

Title	***Description***
Type	Preset type.
Name	User-entered preset name.
Notes	User-entered preset notes.
Author	User-entered name of preset author/creator.
Time	Time that preset was created.

>> To open/close the Preset Manager screen

To open the Preset Manager screen, at the GUI choose **home ▸ *Preset Manager.*** To close it, Click **X** at the upper-right corner of screen.

Managing user libraries

The background of the **SAVE** button in the **Preset Manager** screen changes to red when changes have been made to the current user library, but they haven't yet been saved. We recommend that you save these changes regularly.

>> To create a new preset library

1 In the **Preset Manager** screen, click **New.**

2 In the "Enter new Library name:" message window (shown right), do the following:

- Type in the name of the new preset library in the text field containing the text "Libraryname".
- Type in your name in the text field containing the text "Author".

You can also replace an existing preset library. To do this, type in its exact name in the "Libraryname" text field and then click the **Overwrite existing?** box to tick it.

3 Click **OK.** A new **Preset Manager** screen will open.

*Don't forget that you can use the right-click **Cut**, **Copy** and **Paste** options when editing the text fields.*

>> To load a preset library

1 In the **Preset Manager** screen, click **Load.**

2 In the **Load File** window, click the preset library you want to load. Its name will appear in the "Load this file:" text field.

3 Click **OK.**

>> To save changes to the currently selected preset library

At the **Preset Manager** screen, click **SAVE.**

If the **SAVE** button is red (shown right), there are unsaved changes; this button changes back to blue after the library has been saved (updated).

>> To create a new preset library from the current one

1 In the **Preset Manager** screen, click **Save As.**

2 In the **Save File** window, type your chosen name for the new preset library in the "Save this file as:" text field.

You have the option to overwrite one of the existing preset libraries. Do this by clicking it in the **Save File** window and then ticking the **Overwrite existing?** option.

3 Click **OK.** The new preset library will be selected.

>> To import a preset library into the one currently selected

1. In the **Preset Manager** screen, click **Import.**
2. In the **Load File** window, click the preset library whose contents you want to import.
3. Click **OK.**

   If the currently selected preset library has unsaved changes, the window message "The Preset Library has not been saved Do you wish to continue?" will appear. To continue importing, click **OK.**

If you want, you can save the changes by clicking **SAVE.**

Deleting presets from a user library

In addition to storing and loading presets (see "User library (presets)" on page 85), you can delete presets from a preset library.

>> To delete a preset from a user library

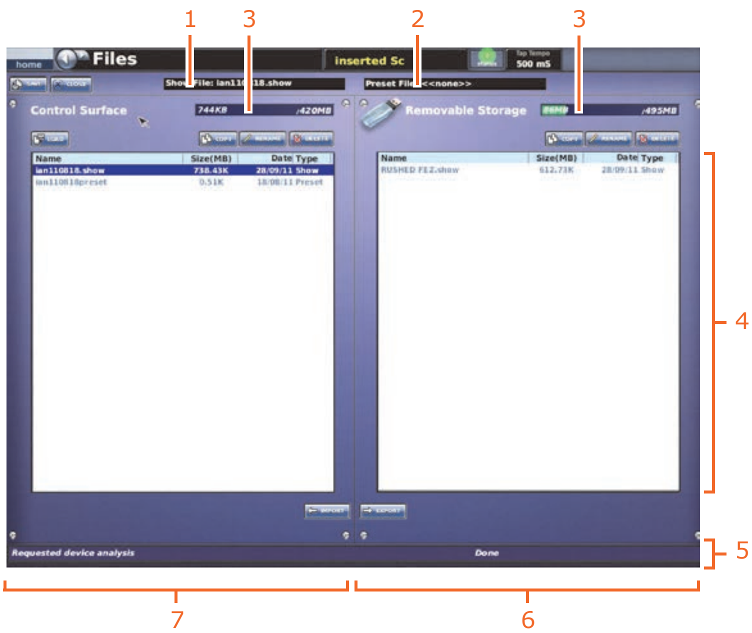
1. In the **Preset Manager** screen, click the preset you want to delete.
2. Click **Delete.**
3. In the "Are you sure you want to delete this preset?" message window, click **OK.**

Chapter 25: File Management

This chapter shows you how to import/export your show and preset files.

About the Files screen

The **Files** screen manages files on the control centre (**Control Surface** panel) and any removable storage device (**Removable Storage** panel) that is currently plugged into one of the USB ports (see “Control surface” on page 246). Each panel lists the files contained on its own storage media. The files can be imported/exported across the panels, and can also be copied, renamed or deleted within their own panel.



Item	Description
1	Title of show file currently loaded, if any.
2	Title of user library preset file currently loaded, if any.
3	Memory usage panel. The number on the right shows the total amount of storage space (in MB), or memory, there is on its respective storage media. The green bar on the left shows how much memory has been used.
4	File lists, display the files on their respective storage media. Each list shows the name, size, creation date and type (for example, show and preset) of each file.
5	Information panels give feedback on file management status.

Item	*Description*
6	**Removable Storage** panel, shows the files currently stored on the storage media (typically, a USB memory key) that is plugged into the active USB socket on the control centre (see "Control surface" on page 246). This panel only appears when the storage media is connected to the control centre, otherwise it is blank. Immediately after plugging in the storage media, you will probably see the message "Analysing...", while its contents are being read.
7	**Control Surface** panel, shows the files stored on the control centre's SSD.

The following table describes all of the buttons on the **Files** screen.

Button	*Function*
SAVE	Saves the currently loaded show in its current state. Turns red when there are changes to be saved.
LOAD	Loads the selected show file.
COPY	Copies selected file to its local section, appending "_copy1" to its name. Subsequent copies of the same will file will increment the 'copy' number by one, that is, "_copy2", "_copy3" etc.
RENAME	Lets you to rename the selected file in its local section.
DELETE	Deletes the selected file in its local section.
EXPORT	Copies file selected in **Control Surface** section to the **Removable Storage** section, effectively copying it onto the removable device. This provides a useful backup facility.
IMPORT	Copies the file selected in the **Removable Storage** section to the **Control Surface** section (PRO1).

Chapter 26: Using Other Devices With The PRO1

This chapter explains how to use other external devices with the PRO1.

Using multiple digital consoles

! CHANGING THE SYNCHRONISATION CAN RESULT IN LOUD NOISES FROM THE SYSTEM. ALWAYS MUTE THE PA AT THE AMPLIFIER/SPEAKER BEFORE CHANGING THE SYNCHRONISATION SOURCE OR MASTER/SLAVE STATUS.

You can use an PRO1 together with one or more digital consoles, which can be other Midas digital consoles or indeed any digital console. For example, you can use two PRO1s together in a dual FOH and MON system. To do this the digital consoles must be synchronised. The synchronisation method can be via AES50, AES3 or wordclock. You can even use DL431 Mic Splitters via their AES50 connections.

Synchronising the consoles

Before you start, choose which Midas digital console you want as master.

For information on external AES50 synchronisation between two Midas digital consoles, see “External AES50 synchronisation” on page 89.

>> To configure system synchronisation at the consoles

1 **Mute the PA at the amplifier/speaker.**

2 On the master Midas digital console, configure it by choosing **home ▸ *Preferences* ▸ *General*** at the GUI and then clicking the “Master” option (shown right) on the **Configuration** tab.

3 Configure a slave Midas digital console by choosing **home ▸ *Preferences* ▸ *General*** at the GUI and then selecting the following options:

- Under the **Master/Slave Status** heading of the Configuration tab, click the “Slave” option.
- Open the **Sync Source** drop-down list and select the desired synchronisation source.

Repeat for any other Midas digital consoles. Configure any non-Midas digital consoles as appropriate.

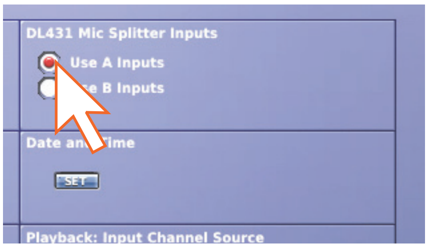
Sharing the A and B inputs of the mic splitter

If you are using two Midas digital consoles, you must configure them to use the A or B inputs of the mic splitters. Although it doesn't matter which inputs each console uses, they can't use the same ones. Also, both consoles must be synchronised.

>> To configure two PRO1s for use with the mic splitters

In a dual FOH and MON system it may be easier and more convenient to always set the FOH control centre to master, and to use the mic splitter A inputs for FOH and the B inputs for MON. Also, although the sync method doesn't matter in this case, it is easier to sync the two consoles using the DL431 Mic Splitters, as described below.

1. **Mute the PA at the amplifier/speaker.**
2. Configure the **AES50 Sync** option of the DL431 Mic Splitter's main menu to **Cable Sync A.**
3. On the FOH control centre, configure the port that the DL431 Mic Splitter is connected to (see "Device set-up procedure" on page 61).
4. Configure the FOH control centre by choosing **home ▸ *Preferences* ▸ *General*** at the GUI and then selecting the following options on the **Configuration** tab:
   - Choose the "Use A Inputs" option (shown right).
   - Under the **Master/Slave Status** heading, choose the "Master" option.



5. On the MON control centre, configure the port for the DL431 Mic Splitter (see "Device set-up procedure" on page 61).
6. Configure the MON control centre by choosing **home ▸ *Preferences* ▸ *General*** at the GUI, and then selecting the following options on the **Configuration** tab:
   - Under the **DL431 Mic Splitter Inputs** heading, choose the "Use B Inputs" option.
   - Under the **Master/Slave Status** heading, choose the "Slave" option.
   - In the **Sync Source** drop-down list, choose the port that you configured for the DL431 Mic Splitter in the previous step.

Using an external USB mouse

You can operate the GUI screen using an external USB mouse instead of its trackball/glide pad. Just plug the mouse into one of the USB sockets on the rear panel of the PRO1. Alternatively, you can plug it into the socket in the **storage** section (to the right of the **automation** section).

Using a USB keyboard

You can use a USB keyboard (other than the one supplied). Just plug it onto one of the USB sockets on the rear panel of the PRO1. Alternatively, you can plug it into the socket in the **storage** section (to the right of the **automation** section).

Using an external monitor

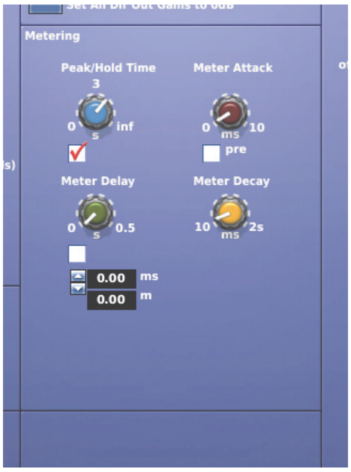
You can plug a monitor in the DVI socket on the rear panel of the PRO1 for viewing what is shown on the GUI screen.

Chapter 27: Changing The Preferences

This chapter shows you how to change the user settings of the PRO1 to suit your own preferences and the current working environment.

For information on configuring the number of GEQs and effects, see "Configuring the number of GEQs (and effects)" on page 123.

Setting the meter preferences



The **Metering** section of the **User** tab provides global parameter adjustment of all of the meters on the control centre.

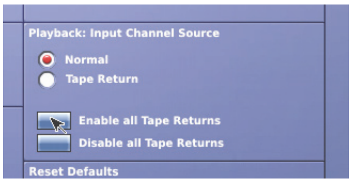
- **Peak/Hold Time** control knob, sets the time (seconds) the peak LED stays on for, in the range 0 (no peak hold) to infinity (peak always on). This only affects the meters on the GUI. Click the **enable** tick box below to switch on this function.
- **Meter Attack** control knob, adjusts the time it takes the meters to rise, in the range 0 (no delay) to 10 milliseconds. Enable the **pre** option to switch the input channel, aux return and output channel meters to be before the EQ/dynamics/insert. Although the input channels and aux returns are always pre-fader, this option also changes the output channels to pre-fader.
- **Meter Delay** control knob, adjusts the meter delay time, in the range 0 to 0.5 seconds. For example, if the control centre is FOH, this function allows you to synchronise the meters with what you are hearing. This is because the sound from the performers on the stage will take a certain amount of time to reach you, whereas the meters pick up that sound at source. Click the tick box below to select this function. You can adjust the meter delay by using the spin buttons underneath, which display the delay time in milliseconds (ms) and distance in metres (m).
- **Meter Decay** control knob, adjusts the time it takes the meters to fall, in the range 10 to 25 milliseconds.

Configuring a virtual soundcheck

The **Virtual Soundcheck Record** option of the **User** tab lets you set the virtual soundcheck options.

- **All Dir Out Pre-processing** — switches all the direct outputs to pre-processing.
- **Unmute all Dir. Out** — unmutes all direct outputs.
- **Mute all Dir Out** — mutes all direct outputs.
- **Set All Dir Out Gains to 0dB** — sets the gain of all direct outputs to 0dB.

Configuring playback



The **Playback: Input Channel Source** option of the **Configuration** tab lets you set the record and playback options of a virtual soundcheck.

- Lets you select the input channel source as **Normal** or **Tape Return** via the radio buttons.
- **Enable all Tape Returns** — enables all tape returns.
- **Disable all Tape Returns** — disables all tape returns.

Restoring the PRO1 defaults

! The Restore Default Preferences button will reset all console preferences, and must be used with great caution. To alert you of the drastic consequences of operating this button, a WARNING window appears.

! The Restore Default Globals button resets all console defaults, including patching and I/O set-up, and must be used with great caution. To alert you to the drastic consequences of operating this button, a WARNING window appears.

You can restore console defaults by using the options in the **Restore Defaults** section of the **Configuration** tab. However, these options must be used with great care.

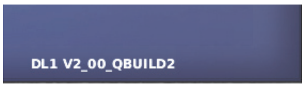
>> To reset console preferences to default

Open the **Preferences** screen and click **Restore Default Preferences.** Acknowledge the warning window as appropriate.

>> To reset all console settings to default

Open the **Preferences** screen and click **Restore Default Globals.** Acknowledge the warning window as appropriate.

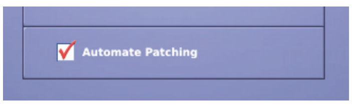
Checking the build information



The lower-right corner of the **Configuration** tab is predominantly a service-only feature, and shows the current build and host software versions of the PRO1 Control Centre (typically shown right).

Using patching in automation

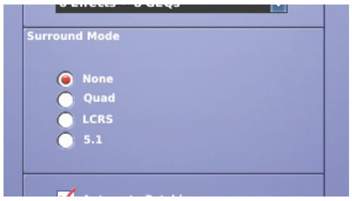
! The Automate Patching option switches on per-scene automatic routing, and must be used with caution. To alert you to the drastic consequences of using this option, a WARNING window appears.



The **Automate Patching** option of the **Show** tab, when enabled, lets you store patching information in scenes. For information, see "Using patching in automation" on page 187.

Selecting the surround mode

The **Surround Mode** section of the **Show** tab lets you select the type of surround mode that the PRO1 uses. For information, see "Selecting the surround mode" on page 225 and "Surround panning" on page 98.



Setting the time and date

The **Date and Time** section of the **Configuration** tab lets you change the PRO1's time and date.

>> To set the time and date of the PRO1

1 At a GUI screen, choose **home** ▸ ***Preferences*** ▸ ***General.***

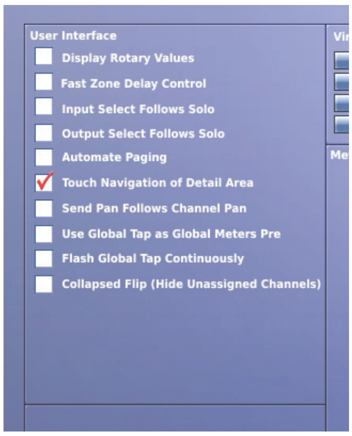
2 In the **Date and Time** section of **Configuration Preferences**, click **SET.**

3 In the time and date window, enter the time (hours and minutes) and date. Make sure you enter the time correctly, according to the currently configured **Format,** that is, 12-hour (a.m. or p.m.) or 24-hour.

4 Click **OK.**

Setting the user interface preferences

The **User Interface** section of the **User** tab lets you set some of the PRO1's operating parameters to suit your own preferences.

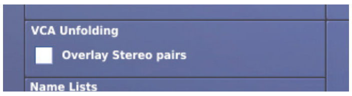


- **Display Rotary Values** — the current value of a parameter, adjusted via a control knob or fader, can be displayed as a numerical value on the GUI or LCD select button, respectively. See "Values displayed on touch (control knob/fader)" on page 40.
- **Fast Zone Delay Control** — choosing this option means that you place the delay control for the inputs onto the surface controls. To do this you have to cycle the gain **SWAP** button (see "Using gain swap" on page 257) through digital gain, analogue gain and then delay. Without selecting this option, the inputs delay control is a GUI only feature.
- **Input Select Follows Solo** — when you solo an input channel, such as an aux return, this channel is automatically selected.
- **Output Select Follows Solo** — when you solo an output channel, such as a matrix, this channel is automatically selected.
- **Automate Paging** — choose this option to store channel paging in automation. So that, on scene recall, the control surface (channels assigned to it) will revert to the state it was when the scene was last saved. When unselected, scene recall does not affect channel paging.
- **Touch Navigation of Detail Area** — choose this option to navigate the local processing (detail) area on the control surface (right of the GUI screen) to the GUI processing area when one of its touch-sensitive controls is operated.

- **Send Pan Follows Channel Pan** — if a channel is contributing to a stereo mix bus (for example, stereo aux channel), choosing this option will cause the pan for the contribution to mirror that of the channel pan.
- **Use Global Tap as Global Meters Pre** — choose this option to change the function of the **TAP** button in the **global** section (right of the **advanced mix bay navigation** section) to operate as a global meters pre button (see Meter Attack in "Setting the meter preferences" on page 223).
- **Flash Global Tap Continuously** — choose this option so that the **TAP** button in the **global** section (right of the **advanced mix bay navigation** section) will flash to reflect the current global tap tempo; this value is always displayed (in milliseconds) in the **tap tempo** section at the top of the GUI screen.
- **Collapsed Flip (Hide Unassigned Channels)** — choose this option so that only channels assigned to a mix bus will be visible when flipped to that bus. Fader bay scrolling changes to banks of four. This affects solo, such that any AFL solo is derived post- the aux send contribution level (and its pan if applicable).

VCA unfolding

The **VCA Unfolding** section of the **User** tab lets you select **Overlay Stereo Pairs** to unfold only left channels of channel pairs when a VCA group is selected. Use the navigation buttons to display any desired right channel.



Changing the default input/output names

The **Name Lists** section of the **User** tab lets you change the names that appear in the lists on the **Input Sheet** and **Output Sheet.** These lists provide you with a number of default names from which you can choose when naming your inputs and outputs in the GUI menu.

>> To change the set names in the Input/Output Sheets

1. At the GUI, choose **home** ▸ ***Preferences*** ▸ ***General.***
2. In the **Names Lists** section of **User Interface Preferences**, click **SET NAMES.**
3. In the **Set Name Lists** window (shown right), click within the field containing the name you want to change. Enter the new name (see "Text editing" on page 42).

   Repeat for any other names you want to change.
4. Click **CLOSE.**

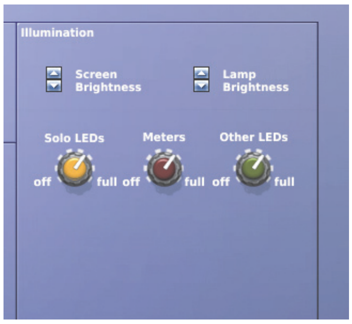
On-scene store

The **On Scene Store** section of the **User** tab lets you choose whether or not parameters protected by a channel safe will be written to a scene when the scene is stored (see "Overriding store scope" on page 186).

Changing the signal processing preferences

The **DSP Timeout** section of the **Configuration** tab lets you set the amount of time (between 10 and 600 seconds) the DSPs will continue to run after an update is received from the control surface, before the audio is muted.
The **inf.** tick box is for selecting infinity, which will allow audio to continue indefinitely if power to the control surface is lost.

Adjusting PRO1 illumination



The **Illumination** section of the **User** tab lets you adjust the brightness and contrast of the GUI screen, the brightness of the LEDs (including meters) on the control surface and the brightness of the lamp.

To increase/decrease the brightness of the GUI screen or lamp, click the desired up/down spin buttons.

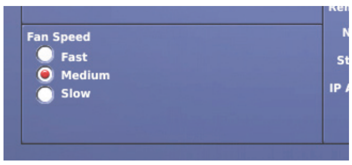
To increase/decrease the brightness of the solo LEDs, meter LEDs or the other LEDs on the control surface, use drag to adjust the appropriate control knob (from **off** to **full**).

Selecting the function of the foot switch(es)

On the **Configuration** tab you can select the foot switch function as either of the following:

- **Next Scene** Choose this option to go to the next scene when the foot switch is operated.
- **Tap Tempo** Choose this option to use the foot switch to set the tempo.

Selecting the fan speed



In the **Fan Speed** section of the **Configuration** tab you can select the speed of the internal cooling fan of the PRO1 Control Centre according to the operating conditions. If you are operating the PRO1 in a warm or hot environment we recommend that you select the **Fast** option. If the noise of the fan operation is causing a problem, select the **Slow** option.

Remote control server

In the **Remote Control Server** section of the **Configuration** tab, you can enable remote control of the console by the Midas software for Apple iPad. To do this carry out the following:

1 Connect your console to a secure wireless network.

2 Do one of the following:

- Enter an appropriate IP address for your network using the up/down spin buttons of the **IP Addr.** section*
- Tick the **DHCP** box so that your router automatically assigns an IP address to your network**

3 Click **Start.** The server will take several seconds to start up. Once up and running, the status will confirm the IP address.

You can stop the server at any time by clicking **Stop.** This will disable external control of the console.

Important:
We recommend that you do not connect the console to an unsecured or public network, as other users could potentially take control of the console.

* Avoid using addresses in the range 192.168.20.x, as they are reserved for console internal communication. Netmask is predefined as 255.255.255.0.

** In DHCP mode, if the address you are assigned is in the range 169.254.x.x this indicates the console could not find a DHCP server (and the server will not function). Check connections and consult the manufacturer's instructions for your router.

Configuring the channels, groups and internal units

You can change the default name and colour of the input and output channels, groups, internal rack units and GEQs of the PRO1 that appear on the control surface (LCD select buttons) and GUI. This is done via the 'Sheet' screen of each item, which is accessible via the GUI menu.

The procedure for configuring the VCA/POPulation groups is shown in "Configuring VCA/POPulation groups" on page 73, and this is principally the same for each of the above items.

Chapter 28: Delay Compensation (Latency)

A time delay is induced in a channel's signal by placing, for example, an insert or GEQ in its path. This delay affects system latency and can also produce undesirable audio effects. To overcome this the PRO1 incorporates a system of user-configurable delay compensation parameters. These are presented to the user in the form of button-selectable options on the GUI and can be switched on or off to suit the current application.

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.

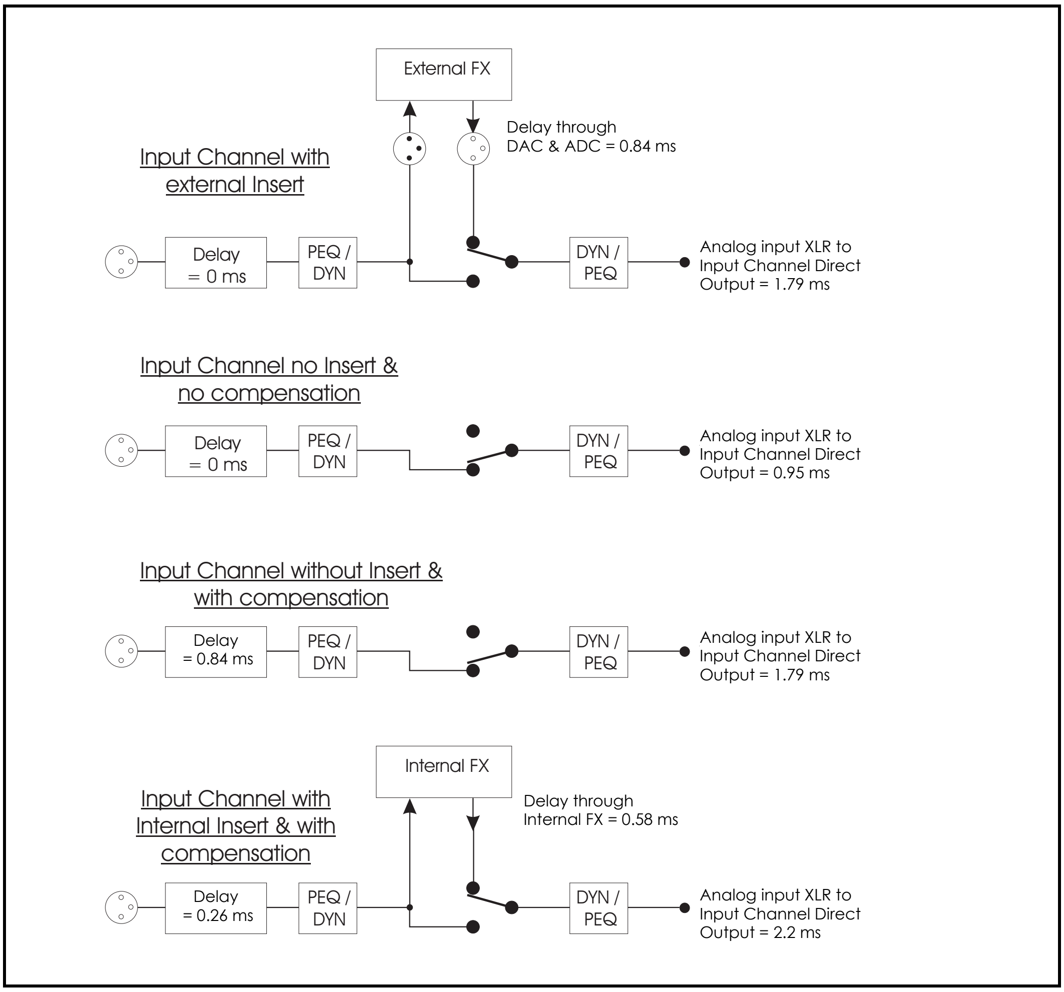


Figure 17: Input channel insert compensation

To avoid the comb filtering effect, the PRO1 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 console, 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 console to be configured for the lowest latency for a given application.

GEQ compensation

Output bus channels have the ability to have a GEQ inserted into them, which incurs an additional delay in their signal path. With GEQ compensation active, a delay is inserted into the output buses, which is removed when a GEQ becomes active. This ensures that all bus outputs of the same type are aligned regardless of whether they use a GEQ or not.

GUI Delay Compensation options

PRO1 delay compensation (latency) is configured in the **Delay Compensation** section (shown right) of the **Delay Compensation** screen.

For a description of the delay compensation options and details of when best to use them, see Table 11 (below). In this table the ***Description*** column explains what happens when the delay compensation option is selected (switched on) and the ***Latency (ms)*** column shows the value that the overall system latency is increased by.

>> To access the delay compensation options

At the GUI, choose **home ▸ *Preferences* ▸ *General*** and click the **Delay Compensation** tab to open the **Preferences Delay** screen.

Table 11: Delay compensation options

Section	***Option***	***Description***	***Recommendations***	***Latency (ms)***
Input Channels	Insert	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.	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.	0.84
Aux Sends	Monitor Mode (Align with Masters)	See “Monitor Mode (Align with Masters)” on page 232.	N/A	N/A

Section	***Option***	***Description***	***Recommendations***	***Latency (ms)***
	Insert	Compensates for inserts placed in aux buses. 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.	If there are no inserts on any aux channels, switch this option off to reduce overall system latency. If any aux channel has an insert, switch this option on. If the **Monitor Mode (Align with Masters)** option is selected, switch this option off.	0.84
	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.	If no effects are used between the aux and return channels, switch this option off to reduce overall system latency. If any effects are used between any auxes and returns, switch this option on. If the **Monitor Mode (Align with Masters)** option is selected, switch this option off.	0.84
	Graphic EQ	This setting controls the delay compensation aligning aux bus outputs for channels that use GEQ with those that do not.	If no aux buses use GEQ, switch this option off to reduce overall system latency. If any aux bus has a GEQ inserted, switch this option on to ensure all aux bus outputs are time-aligned.	0.5
Master and Matrix	Insert	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 off, the outputs of any master or matrix channels using inserts will be delayed relative to the equivalent channels not using them.	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.	0.84
	Graphic EQ	This option controls the delay compensation that aligns master and matrix bus outputs for channels using GEQ with those that do not.	If no master or matrix buses are using GEQ, switch this option off to reduce overall system latency. If any master or matrix buses have a GEQ inserted, switch this option on to time-align all master and matrix bus outputs.	0.5

Monitor Mode (Align with Masters)

The default console bus structure is organised such that inputs can be routed to masters, and also simultaneously routed to masters and matrix channels via aux buses (see Figure 19 "Routing via aux buses") or via aux and return buses (see Figure 18 "Routing via aux and return buses"), while maintaining the same overall input to output latency in both paths.

This may not be the desired structure if auxes and matrix outputs are both being used for monitor mixes, where it is desirable for auxes, masters and matrix outputs to be time-aligned with the minimum possible latency.

With this option switched off, the delay element that is used to delay the paths from input channels to master/matrix channels is removed. With all insert and GEQ delay compensation switched off, the latency between a system input XLR and a system output XLR is being fed by an aux, master or matrix channel is 1.79ms (see Figure 20 "Latency for input to aux, master and matrix outputs").

If using this option, it is advisable to use the same GEQ and insert compensation settings for aux and master/matrix channels to maintain their identical output latency.

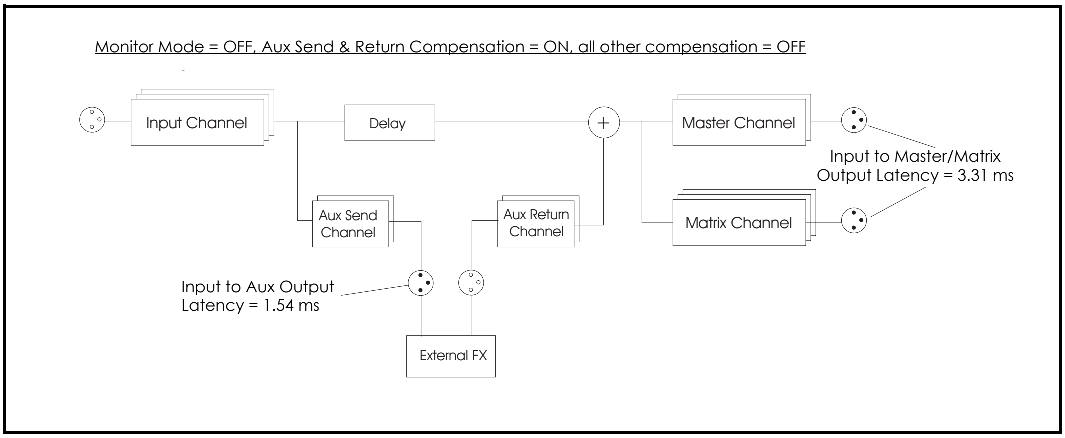


Figure 18: Routing via aux and return buses

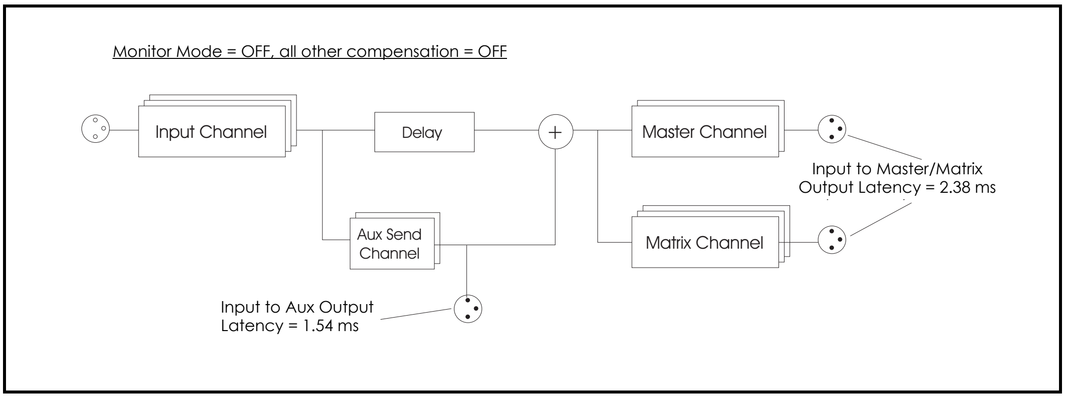


Figure 19: Routing via aux buses

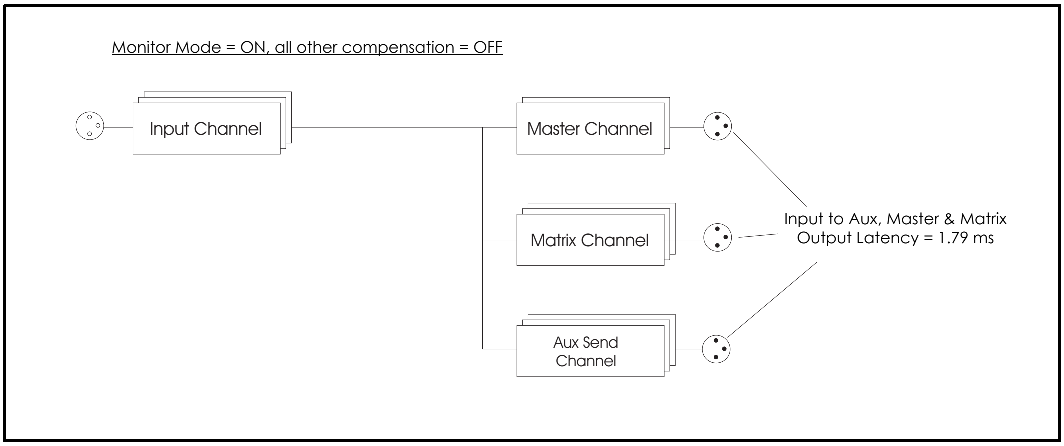


Figure 20: Latency for input to aux, master and matrix outputs

Zones

The PRO1 system can be divided into conceptual ‘zones’, as follows:

- **System Input Zone:** DL251 Audio System I/O or surface analogue/AES3 inputs, which are normally routed to Input channels. These inputs are *primary* system inputs and the console output latency is measured relative to these inputs.
- **Mix Zone:** Aux outputs, return inputs and master/matrix direct inputs, which can be freely patched to and from internal or external effects, while maintaining output signal alignment.
- **Output Zone:** System outputs, that is, master and matrix outputs when **Monitor Mode (Align with Masters)** compensation is switched off, or aux, master and matrix outputs when **Monitor Mode (Align with Masters)** compensation is switched on.

Aux direct inputs are fixed to the System Input Zone so that DL251/DL431/DL451 inputs routed to an aux direct input will automatically line up with inputs routed to auxes via input channels.

Return inputs and master and matrix direct inputs can be configured to operate in either the System Input Zone (for example, as additional console inputs) or the Mix Zone (for example, as effect returns) and are configurable on a per channel basis.

Examples of patches using the Mix Zone that are all fully compensated when the **Send-FX-Return** option of the **Aux Sends** section of the delay compensation is switched on are:

- Aux -> Internal/External Effect -> Return
- Aux -> Internal/External Effect -> Master Direct Input
- Aux -> Internal/External Effect -> Matrix Direct Input
- Aux with Insert -> Internal/External Effect -> Return

Input channel direct outputs are simply a copy of the input channel output or mic input signal, depending on the direct output mode for a particular channel. It is not possible to delay these signals to line-up with the main system outputs or aux outputs, so patching from a direct output to an effect and back in to a return, master direct input etc., cannot be fully compensated for.

The input to direct output latency depends on the direct output mode and the **Inserts** option of the **Input Channel insert** delay compensation status according to the following table.

	Input channel insert compensation (ms)	
Direct Output mode	***Off***	***On***
Pre-processing	0.59	0.59
Post-processing	0.948	1.78

Masters to matrix tap-off-point

The signal path that feeds master bus signals onto matrix channels is fully compensated for so that signals fed direct 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.

The user is given the choice of tap-off point, so they may 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, as shown in the following diagram.

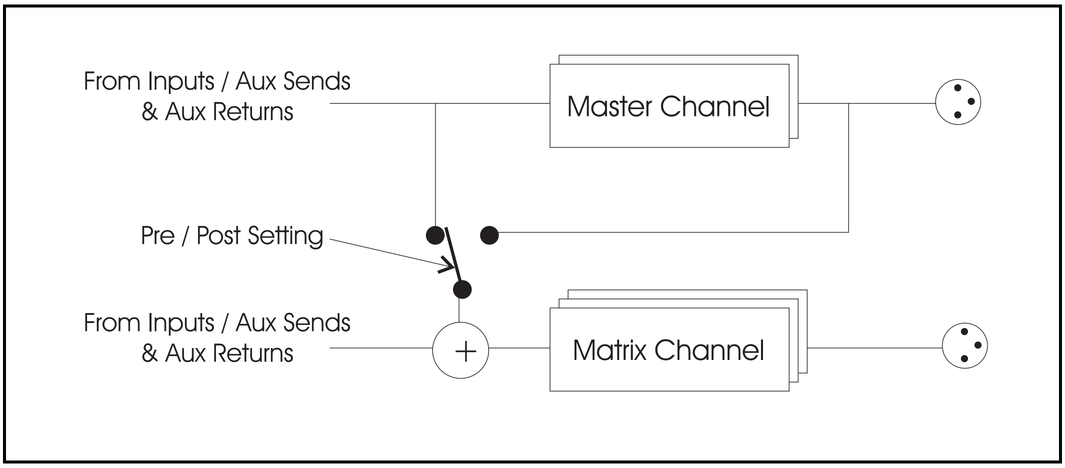


Figure 21: Master and matrix tap-off points

Sending pre-processed master bus signals to matrix channels reduces the overall system latency. If post-processed tap-off points are used, the system must compensate for both the latency of the matrix and master channels, and if insert and GEQ compensation are both required on master and matrix channels, this can push the maximum system latency up to 8.49 ms. With no other insert or GEQ compensation switched on, using post-processed tap-off points produces a system latency of 2.72 ms, as opposed to pre-processing tap-off points, which would produce a latency of 2.38 ms. When **Monitor Mode (Align with Masters)** is switched on, these figures are 1.79 ms and 2.14 ms, respectively.

Typical configurations

Here are some actual examples of typical configurations to illustrate the effects of delay compensation. Please note the following:

- All XLRs are located at stage end, unless FOH is stated.
- INS can mean an internal effect or an external effect with analogue, or AES3 I/O at either FOH or stage position.
- Abbreviations are: IP = input channel; AS = aux (send) channel, AR = (aux) return channel; AR (Input) = (aux) return channel set to input mode; MAST = master channel; MTX = matrix channel; DI (mix) = direct input set to Mix Zone (DI can be either pre- or post-); and DI (input) = direct input set to System Input Zone (DI can be either pre- or post-).

FOH Mix Setup

The following table shows the delay compensation settings for this mix.

Option	*On/off status*
Master to Matrix Post-processing	On
Input Channels	
Insert	On
Aux Sends	
Monitor Mode (Align with Masters)	Off
Insert	On
Send-FX-Return	On
Graphic EQ	On
Master and Matrix	
Insert	On
Graphic EQ	On

The following example signal paths all measure the same latency of 815 samples @ 96kHz = 8.49 ms:

- FOH XLR – IP – MAST – XLR
- XLR – IP – MAST – XLR
- XLR – IP – MTX – XLR
- XLR – IP – AS - MAST – XLR
- XLR – IP – AS – INS – MAST – XLR
- XLR – IP (With INS) – AS (With INS) – INS – AR – MAST (With INS) - XLR
- XLR – IP (With INS) – AS (With INS + GEQ) – INS – AR – MAST (With INS + GEQ) - XLR
- XLR – IP – AS – INS – MAST DI (Mix)
- XLR – IP – AS – INS – MTX DI (Mix)
- XLR – MAST DI (Input)
- XLR – MTX DI (Input)
- XLR – AS DI – MAST – XLR
- XLR – AR (Input) – MAST – XLR
- XLR – IP – MAST – MTX
- XLR – IP – AS (With GEQ) – MAST (With GEQ)

- XLR – IP – AS (With GEQ) – MTX (With GEQ)

FOH Mix Low Latency

The following table shows the delay compensation settings for this mix.

Option	*On/off status*
Master to Matrix Post-processing	Off
Input Channels	
Insert	Off
Aux Sends	
Monitor Mode (Align with Masters)	Off
Insert	Off
Send-FX-Return	On
Graphic EQ	Off
Master and Matrix	
Insert	Off
Graphic EQ	On

The following example signal paths all measure the same latency of 366 samples @ 96kHz = 3.81 ms:

- FOH XLR – IP – MAST – XLR
- XLR – IP – MAST – XLR
- XLR – IP – MTX – XLR
- XLR – IP – AS - MAST – XLR
- XLR – IP – AS – INS – MAST – XLR
- XLR – IP – AS – INS – MAST (With GEQ) – XLR
- XLR – IP – AS – INS – MAST DI (Mix)
- XLR – IP – AS – INS – MTX DI (Mix)
- XLR – MAST DI (Input)
- XLR – MTX DI (Input)
- XLR – AS DI – MAST – XLR
- XLR – AR (Input) – MAST – XLR
- XLR – IP – MAST – MTX
- XLR – IP – AS – MAST (With GEQ)
- XLR – IP – AS – MTX (With GEQ)

Monitor Mix

The following table shows the delay compensation settings for this mix.

Option	*On/off status*
Master to Matrix Post-processing	Off
Input Channels	
Insert	Off
Aux Sends	
Monitor Mode (Align with Masters)	On
Insert	On
Send-FX-Return	Off
Graphic EQ	On
Master and Matrix	
Insert	On
Graphic EQ	On

The following example signal paths all measure the same latency of 300 samples @ 96kHz = 3.125 ms:

- FOH XLR – IP – MAST – XLR
- XLR – IP – MAST – XLR
- XLR – IP – MTX – XLR
- XLR – IP – MAST- MTX – XLR
- XLR – IP – AS – XLR
- XLR – IP – AS (With GEQ) – XLR
- XLR – AS DI – AS – XLR
- XLR – AS DI – AS (With GEQ) – XLR
- XLR – AR (Input) – MAST – XLR
- XLR – AR (Input) – MAST (With GEQ) – XLR
- XLR – AR (Input) – MTX (With GEQ) – XLR
- XLR – AR (Input) – MAST - MTX - XLR
- XLR – AR (Input) – MAST - MTX (With GEQ) - XLR

Monitor Mix (Low Latency)

The following table shows the delay compensation settings for this mix.

Option	***On/off status***
Master to Matrix Post-processing	Off
Input Channels	
Insert	Off
Aux Sends	
Monitor Mode (Align with Masters)	On
Insert	Off
Send-FX-Return	Off
Graphic EQ	Off
Master and Matrix	
Insert	Off
Graphic EQ	Off

Input Ch Insert (OFF)

Aux Insert (OFF)

Aux Send & Return (OFF)

Aux GEQ (OFF)

Masters / Matrix Insert (OFF)

Masters / Matrix GEQ (OFF)

Master to Matrix tap-off = pre-processing.

Aux alignment (OFF)

The following example signal paths all measure the same latency of 172 samples @ 96kHz = 1.79 ms:

- FOH XLR – IP – MAST – XLR
- XLR – IP – MAST – XLR
- XLR – IP – MTX – XLR
- XLR – IP – MAST- MTX – XLR
- XLR – IP – AS – XLR
- XLR – AS DI – AS – XLR
- XLR – AR (Input) – MAST – XLR
- XLR – AR (Input) – MAST - MTX - XLR

Description

Chapter 29: Connections

This chapter describes the external connections of the PRO1.

For information on powering the PRO1 up/down, PRO1 system interconnections and rear panel connections, see the PRO1 Quick Start Guide.

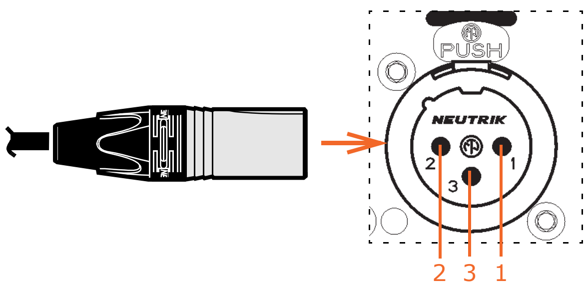
Rear panel connectors

PRO1 rear panel

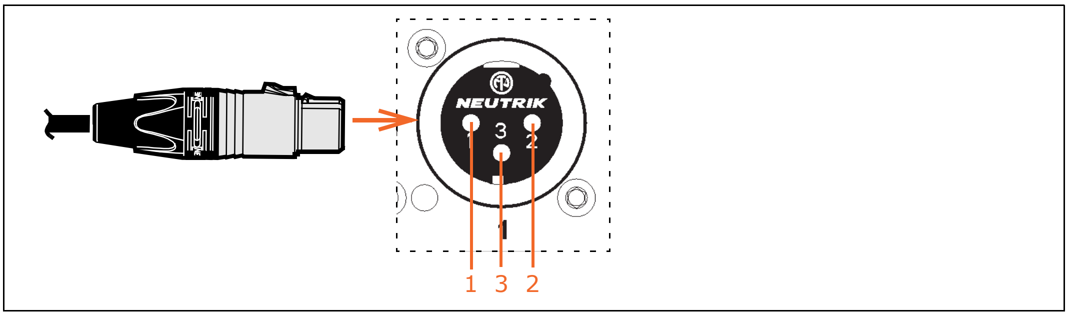
To ensure the correct and reliable operation of the equipment, only high quality balanced, screened, twisted pair audio cable should be used. The AES3 adapter cable is to be 110 Ohms.

To fully comply with national legislation including (but not limited to) transposition of EC EMC Directive 2004/108/EC by EU member states and FCC Part 15 for the United States of America, only the supplied cable may be used with the AES3 outputs.

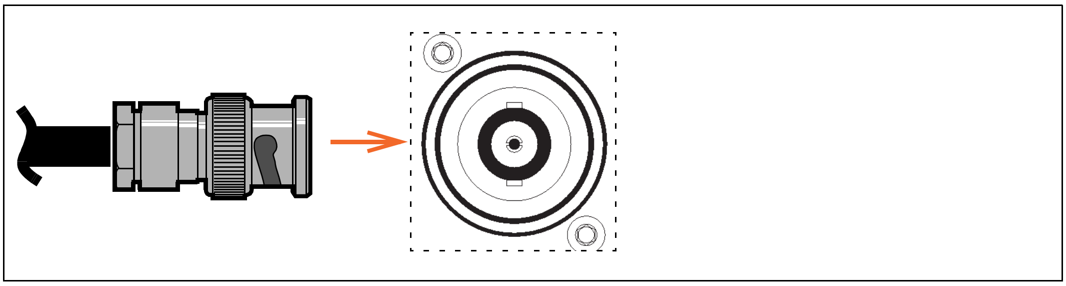
XLR connector shells should be of metal construction so that they provide a screen when connected and, where appropriate, they should have Pin 1 connected to the cable screen.



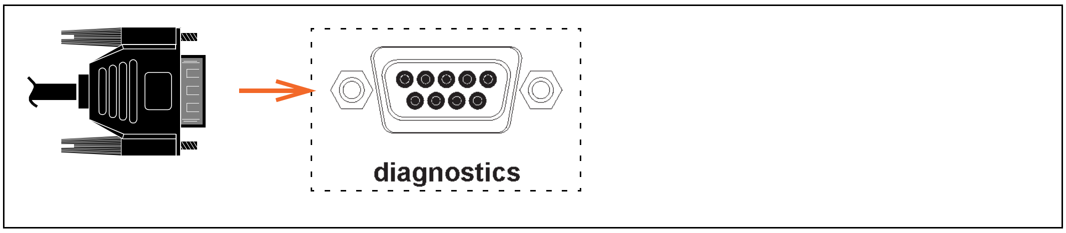
*Mic/line input audio connector. Male XLR plug and female XLR chassis connector with the following pinouts: **1.** Ground **2.** Hot **3.** Cold*



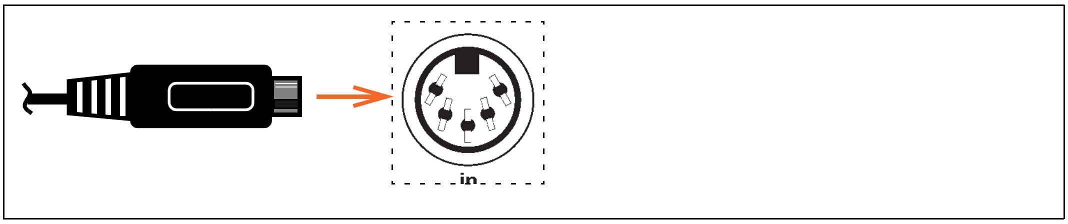
Line output audio connector. Female XLR plug and male XLR chassis connector. ***1.*** *Ground.* ***2.*** *Hot.* ***3.*** *Cold.*



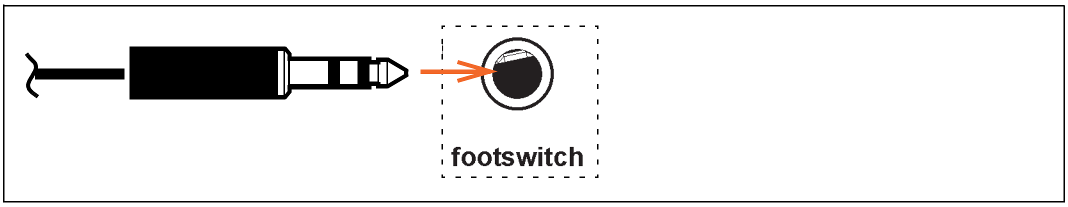
BNC connector and 75 Ohms coaxial cable for word clock and video sync.



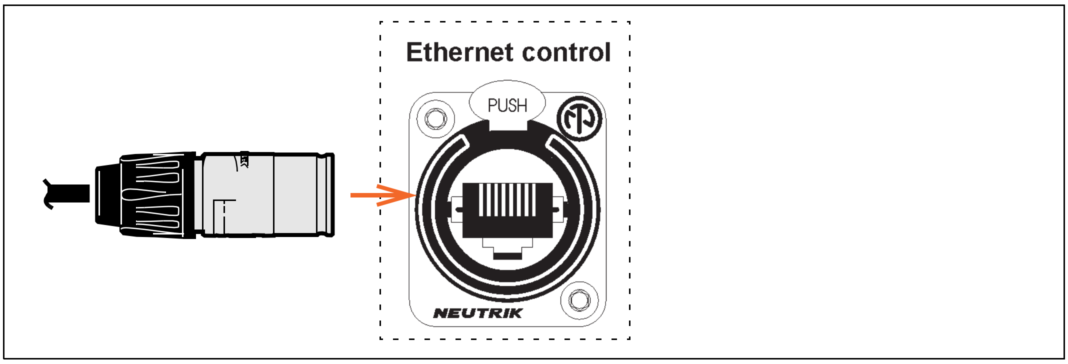
9-way, D-type connector for diagnostics purposes. (For use by service personnel only.)



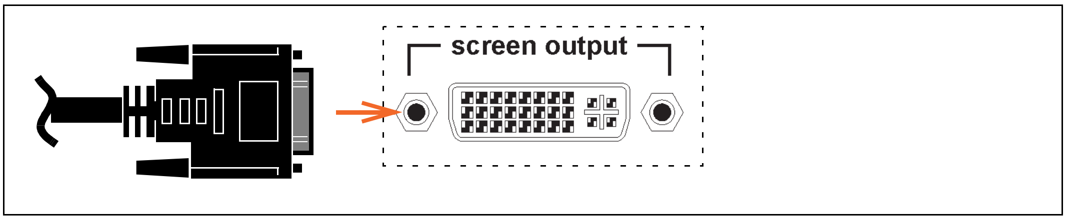
MIDI connector (in, out or thru). 5-pin plug and socket.



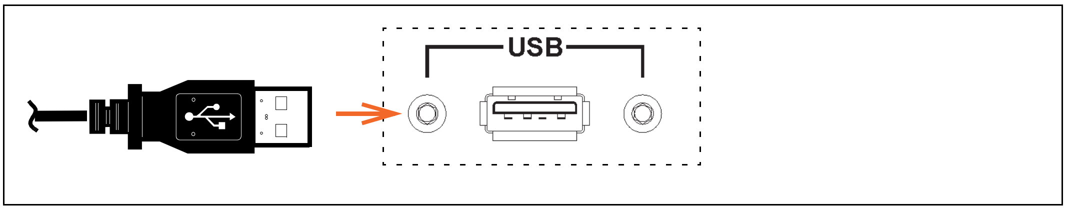
Footswitch 1/4” TRS connector



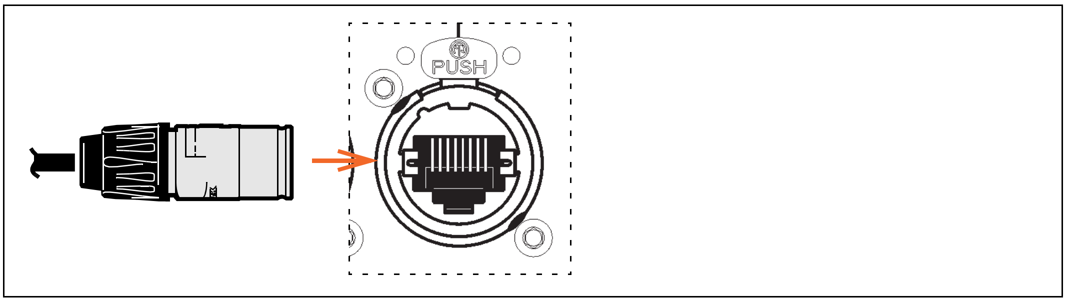
A 100Mb/s Ethernet control port for connection of a Neutrik etherCON Ethernet connector.



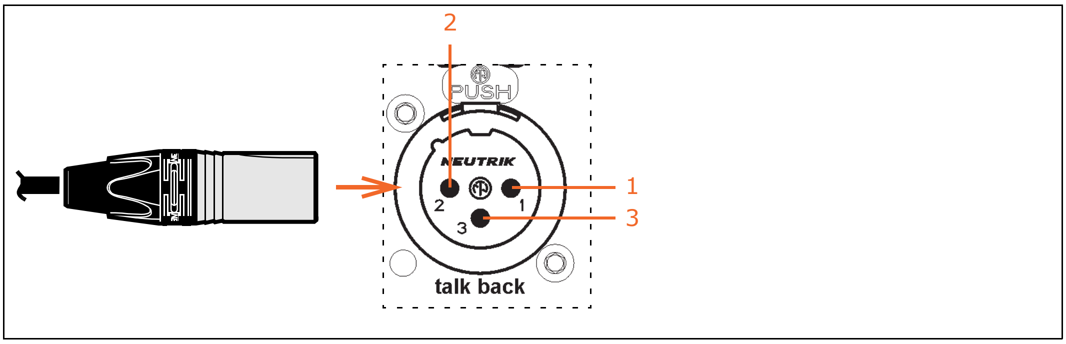
DVI connector.



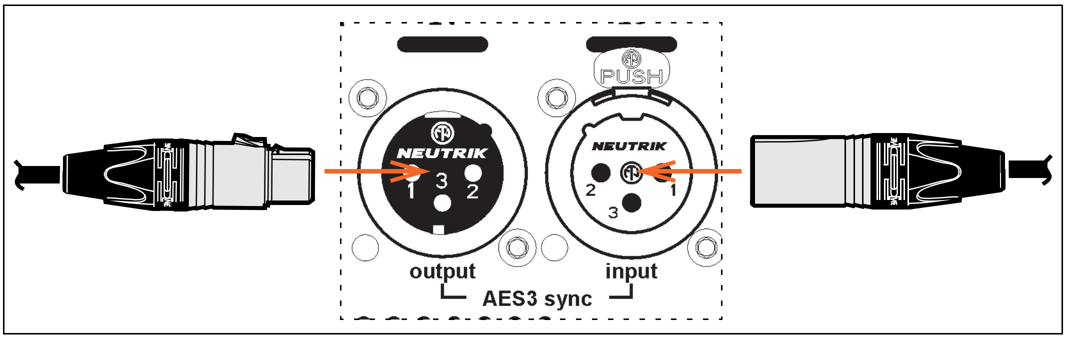
USB type A connector.



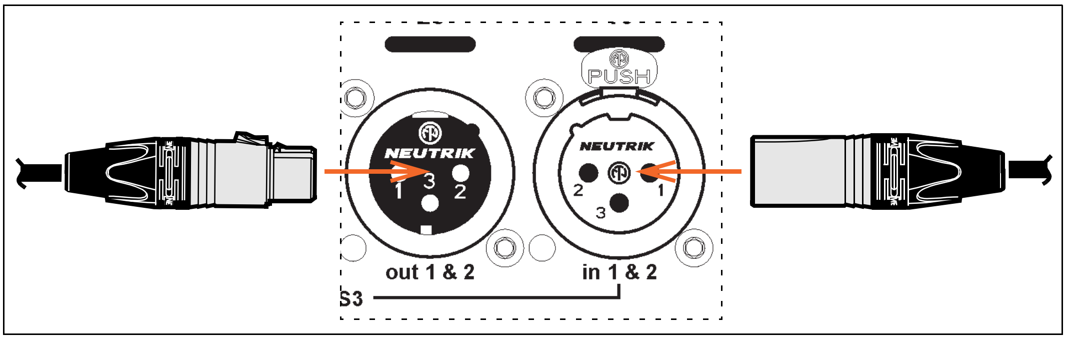
Bi-directional digital audio EtherCon® ports.



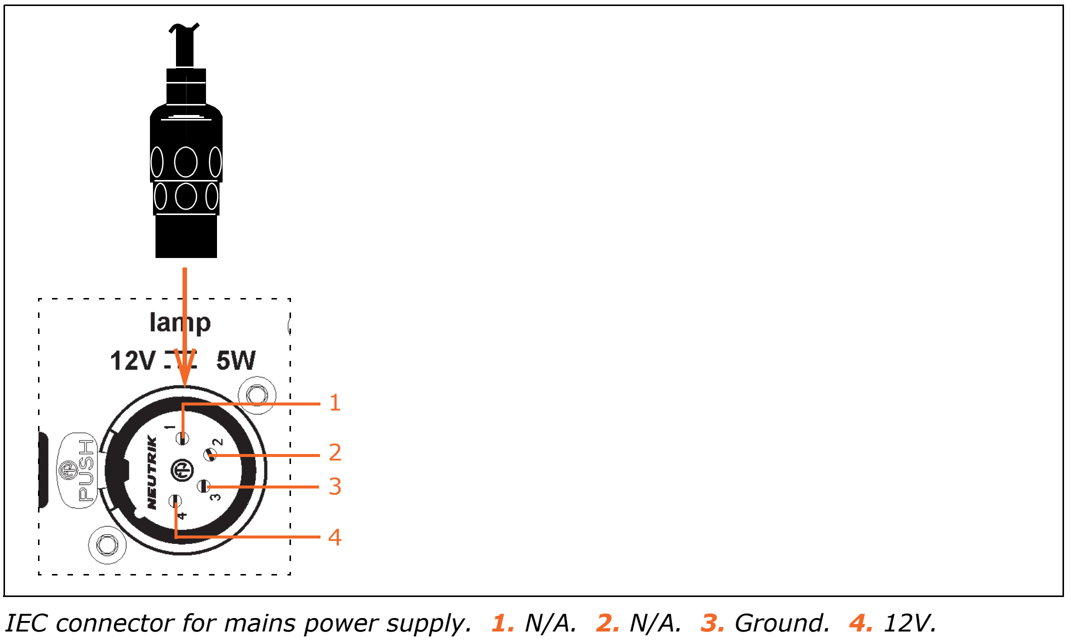
The talk back input connector accepts a male XLR plug. ***1.*** *Ground.* ***2.*** *Hot.* ***3.*** *Cold.*



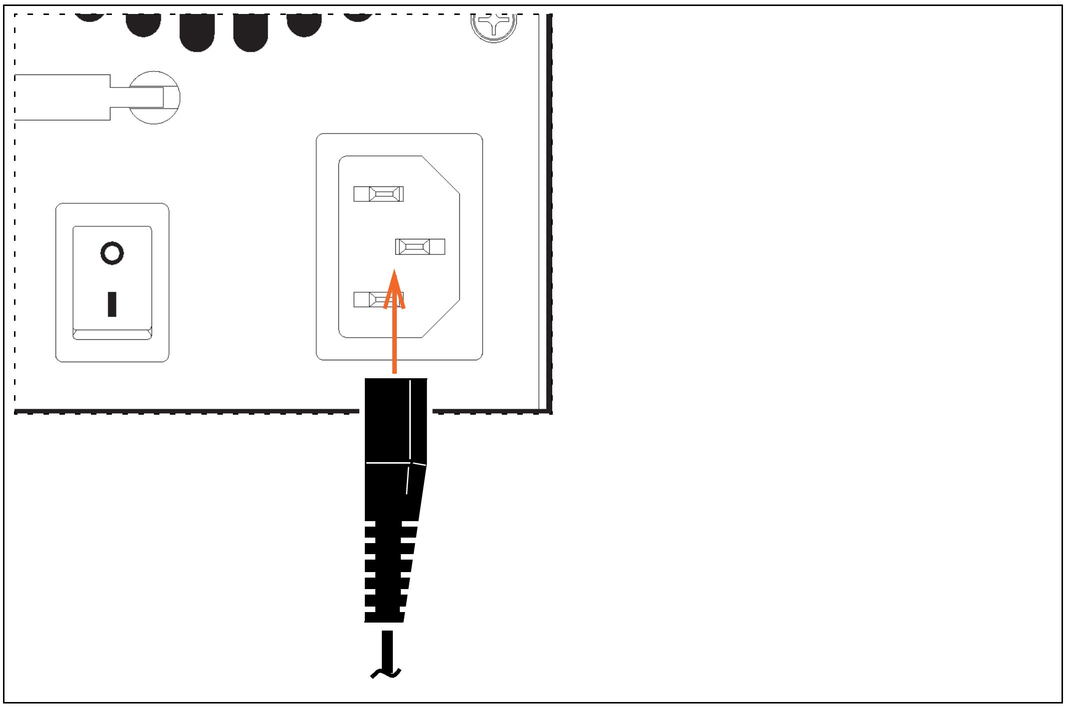
The AES3 sync input and output connectors are for synchronisation with external devices that can transmit/receive a 96kHz AES3 signal. They accept a female XLR plug and male XLR plug, respectively.



The two pairs of AES3 input and output connectors are for synchronisation with external devices that can transmit/receive a 96kHz AES3 signal. They accept a female XLR plug and male XLR plug, respectively.



*IEC connector for mains power supply. **1.** N/A. **2.** N/A. **3.** Ground. **4.** 12V.*

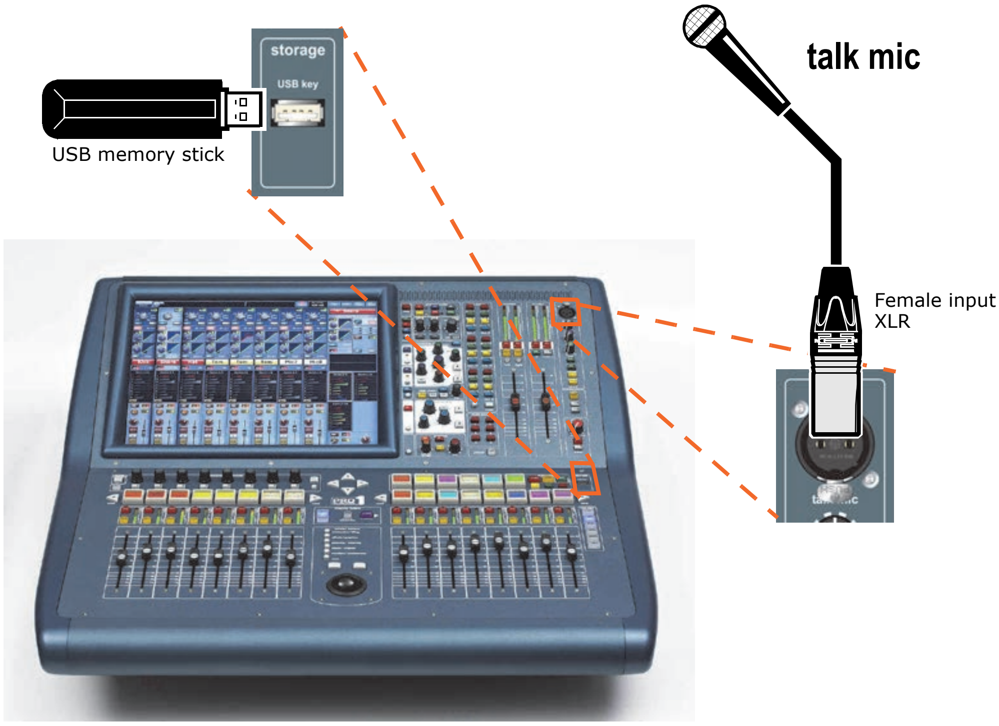


IEC connector for mains power supply.

Control surface

The PRO1 has a rear connector panel that caters for the connection of mains power leads, 19" rack unit(s), USB memory keys, keyboards, headphones, talk mics, communications, external monitor, AES3 synchronisation, diagnostics (for service personnel only), lamp and word clocks (75R).

The control surface houses a USB connector for uploading/downloading show files and updating the system software, and a connector for a talk mic.



Talk mic connector and storage USB connector on the control surface

There are two similar connector panels at the front of the PRO1 Control Centre, which are situated at either end under the armrest as shown in the following diagram.

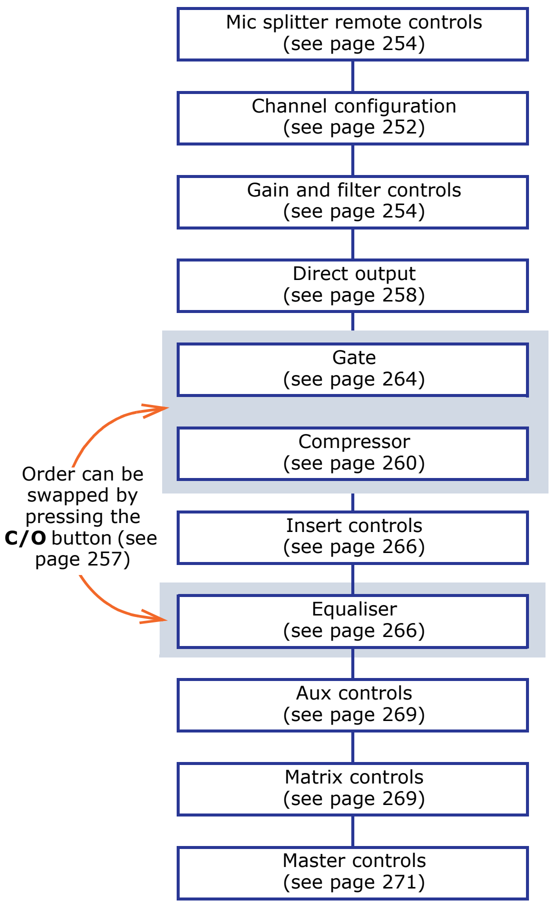
Chapter 30: Input Channels

By default, all of the input channels are mono, although any two adjacent channels (odd left and even right) can be linked to form a stereo pair. The order of processing in the signal path of both channel types is basically the same.

The order of the descriptive sections in this chapter loosely follows the signal path taken by the input channels. However, this varies according to signal processing order and the operation of certain controls.

Input channel routing

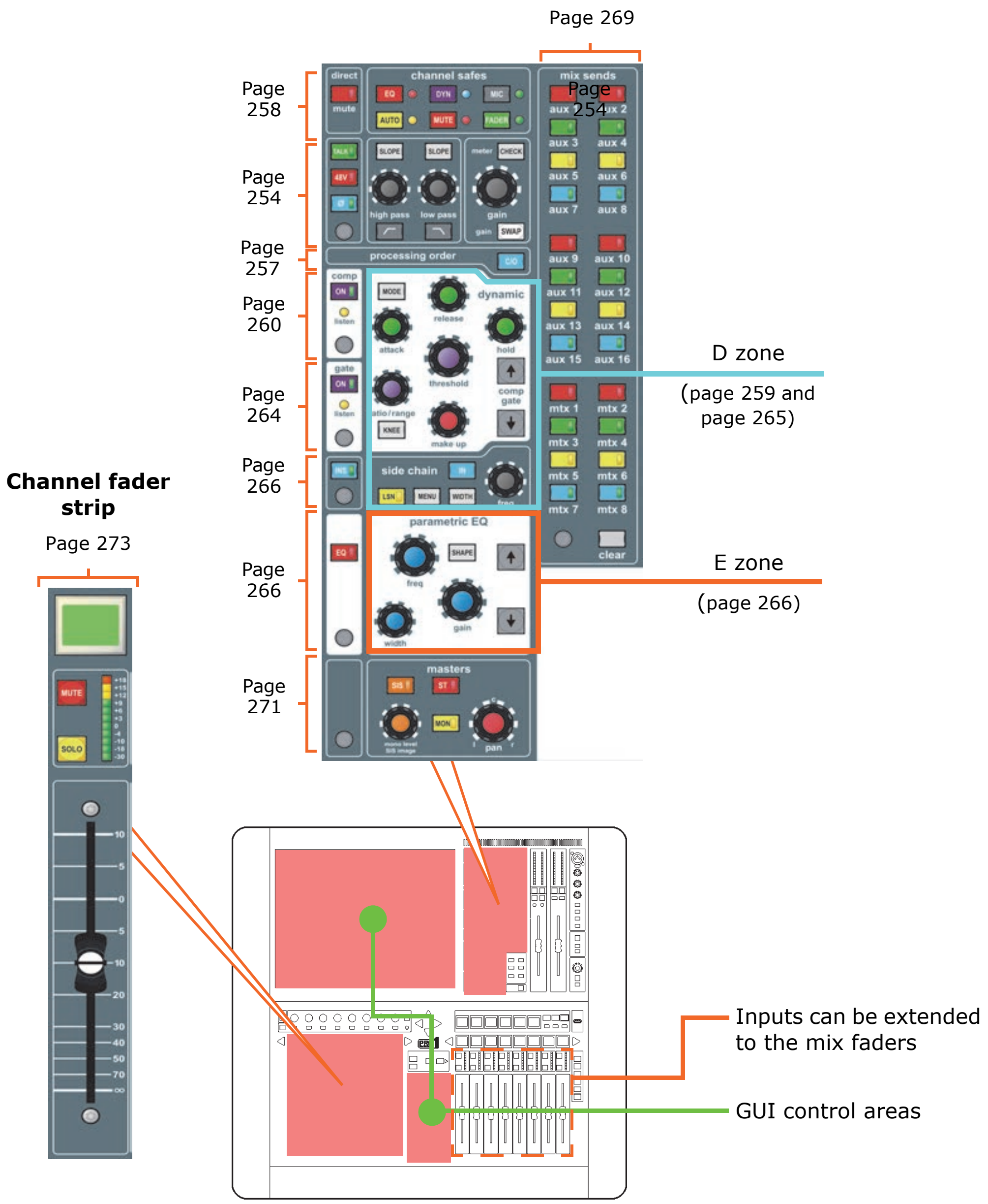
The diagram right shows the default signal path, on which the structure of this chapter is based. This chapter will explain each of these groups of controls, showing the pertinent controls on both the control surface and GUI.



Input channel areas

The input channels are assigned to the channel bay faders. However, by using the **EXTEND** button in the **channel faders** section (see the PRO1 Control Centre Quick Start Guide for details) the mix bay faders can be used as well. Detail adjustment of the input channels is augmented by the channel processing areas. The GUI provides extensive input channel support and also provides extra functionality.

Channel processing areas



Areas on the control surface concerned with input channels

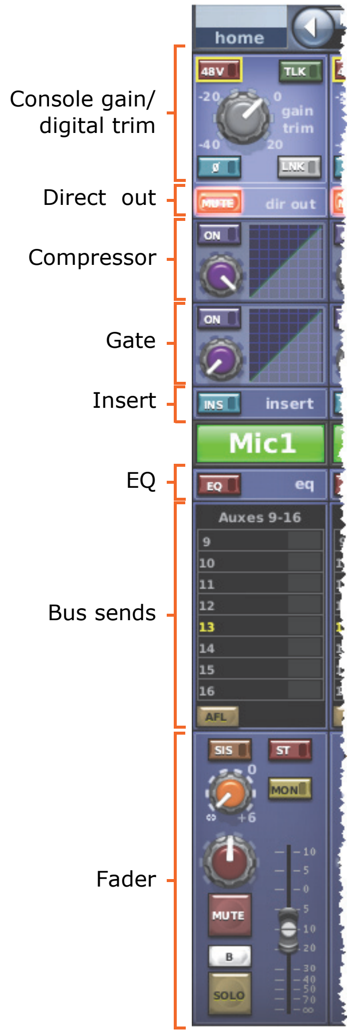
Inputs on the GUI

The GUI replicates the channel faders by displaying the eight channels currently assigned to them. When an input channel is selected the GUI's channel strip displays the channel's **input channel overview.** From this display, you can access processing areas by clicking within specific sections (avoiding any controls).

For details of how to operate the GUI, see Chapter 6 "Working With The PRO1 Control Centre".

GUI input fast strips

The input fast strips on the GUI (a typical example is shown right) give an overview of their equivalent versions on the control surface.



The **gain trim** section changes its appearance to suit the type of control that has been 'swapped' to it (see "Using gain swap" on page 257).

Some processing areas are configuration dependent, such as the bus sends (depends on surround configuration) and console gain/digital trim.

The appearance of the console gain/digital trim section (also for channel processing area, see "GUI channel strips" on page 249) depends on whether or not a signal source is patched to the input channel and the nature of that source, and the current status of the Gain Swap parameter.

Due to the free-routing nature of the console's architecture, it is possible that any given input to a signal chain can be sourced from a variety of devices that have various remote-controlled mic amps. However, the GUI will offer the correct controls according to the source that has been routed.

GUI channel strips

When an input channel is selected, its overview appears in the channel strip. This is called the "input channel overview" (see Figure 22, "Processing areas available from the input channel overview display. A. Depends on what device is connected (for example, DL251 Audio System I/O or DL451 Modular I/O). B. The display in this area depends on surround configuration.," on page 250) and provides limited controls and status information. Clicking a non-control area within a specific section will open that section's processing area, which contains the full set of controls. The following processing areas are available, which are shown in Figure 22 "Processing areas available from the input channel overview display. A. Depends on what device is connected (for example, DL251 Audio System I/O or DL451 Modular I/O). B. The display in this area depends on surround configuration." on page 250:

- Configuration (direct out, safes and gain trim - channel ID, channel source, filters, linking, swap, delay and processing order)
- Compressor
- Gate
- EQ
- Inserts

- Mix buses
- Masters (faders, solo, panning etc.)

For details of how to navigate the GUI channel strip, see “About GUI navigation” on page 44.

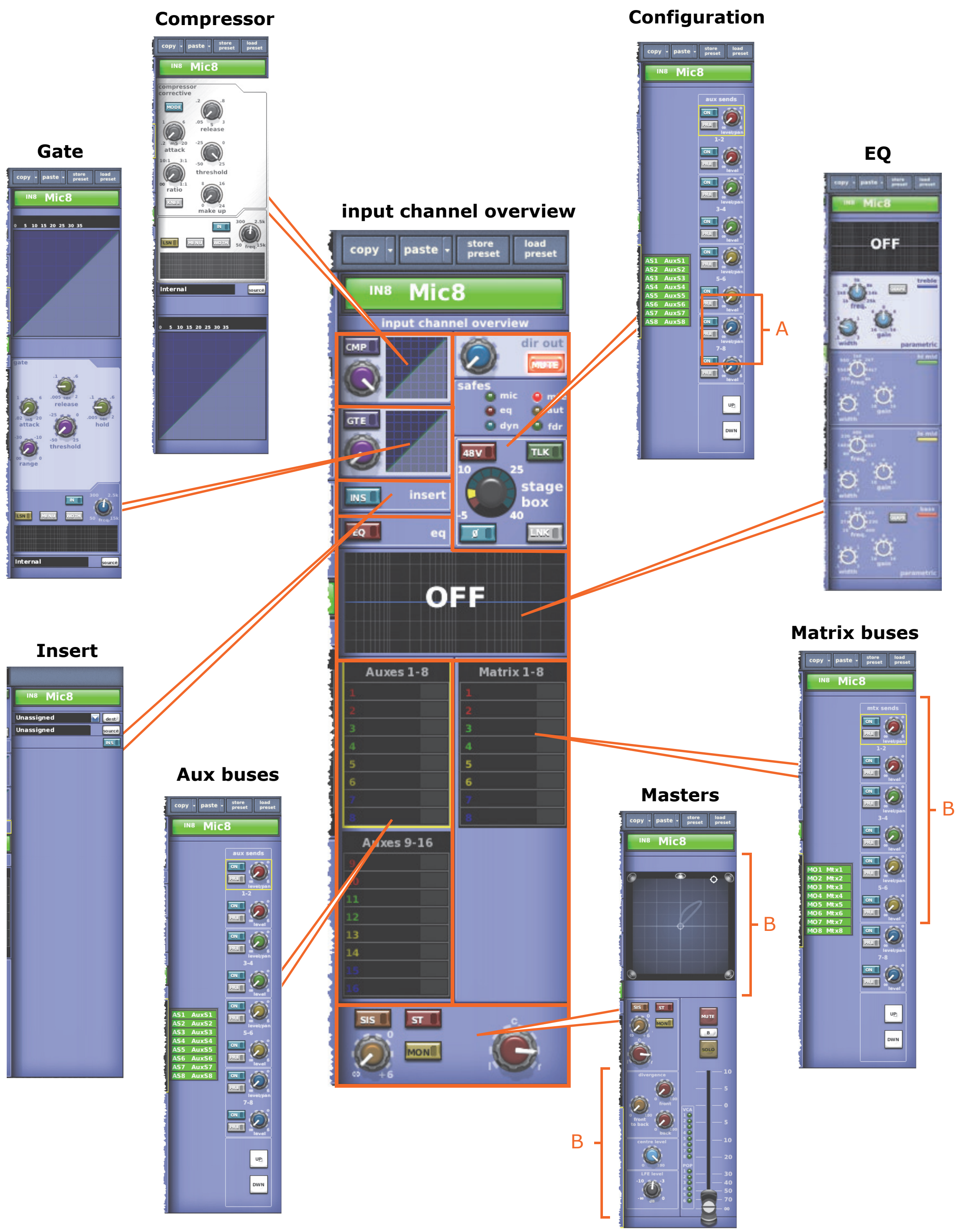
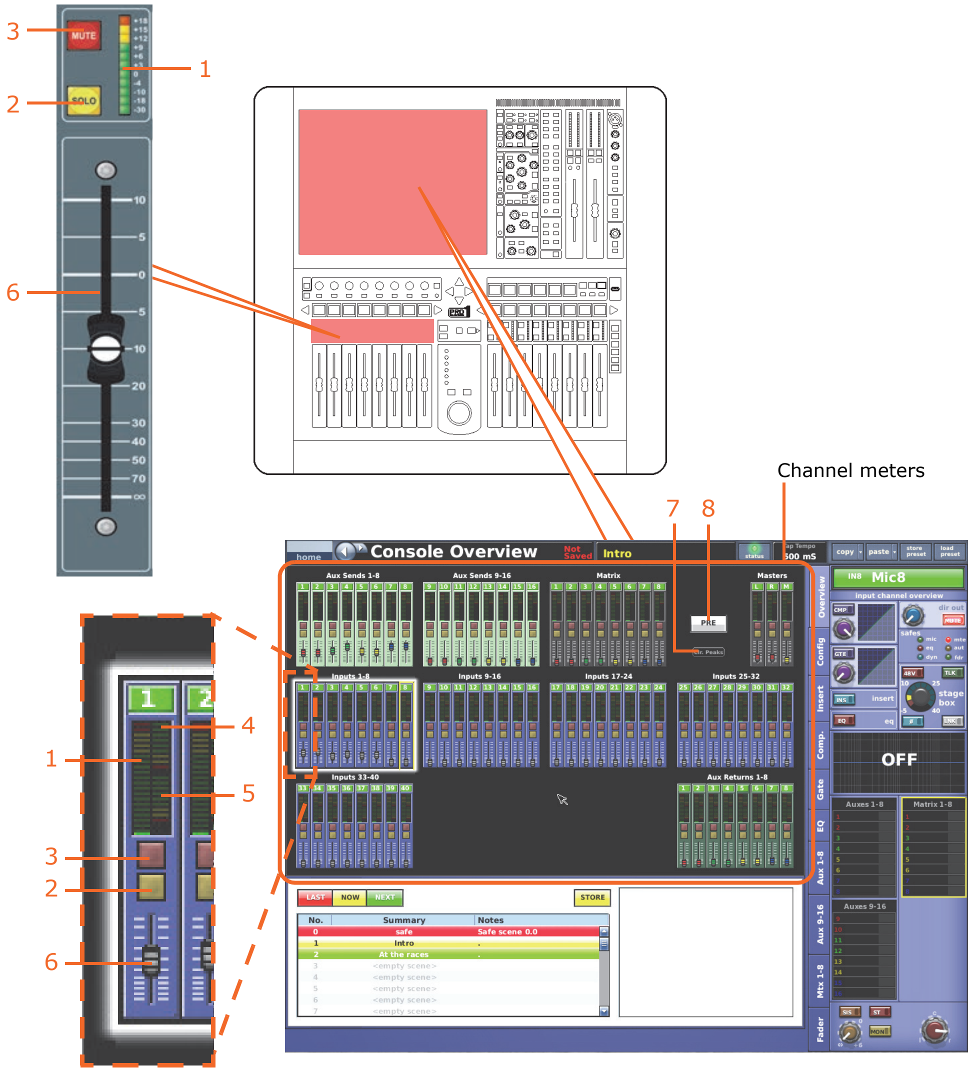


Figure 22: Processing areas available from the ***input channel overview*** *display.* ***A.*** *Depends on what device is connected (for example, DL251 Audio System I/O or DL451 Modular I/O).* ***B.*** *The display in this area depends on surround configuration.*

Input metering

The **Console Overview** screen shows all of the meters all of the time. Meters can be switched globally to monitor the raw A/D input point, and are also individually switchable using the meter **CHECK** button in the **gain** section (see "Mic amp input gain (preliminary input processing)" on page 254).



Item	Description
1	11-segment meter for showing the input channel level.
2	**SOLO** switch for switching the input channel's solo on/off.
3	**MUTE** switch for muting the input channel.
4	LED meter for showing the gain reduction when using a compressor.
5	LED meter for showing the gain reduction when using a gate.
6	Input channel fader.
7	**Clr Peaks** button, which momentarily clears the meter peaks.
8	**PRE** switch, which is a global meter switch that switches all inputs to monitor the raw A/D input point.

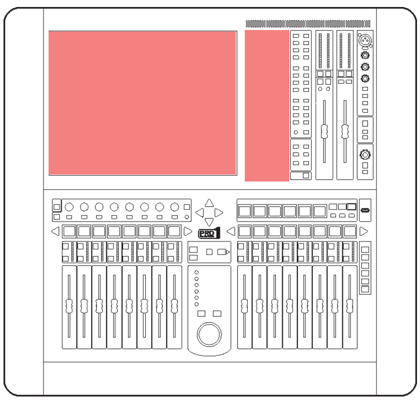
In flip mode the appearance of the **Console Overview** screen changes to indicate mix send selection. It does this by changing the fader section background colour of the relevant channels to match the associated colour of the selected mix send, which is also shown in text towards the upper-left corner of the screen (highlighted right). In the example display shown right, aux send 5 (yellow) is currently selected.

Channel configuration controls

There are a number of input channel controls that are loosely termed 'channel configuration' controls. These comprise:

- **Input channel ID (GUI only):** name and identification. Both the name and colour of the name field are user-configurable. For details, see "Input channel ID (GUI only)" on page 253.
- **Input channel source (GUI only):** shows where the input is routed (patched) from, that is, the physical location the input channel is getting its audio from, and provides direct access to the **Patching** screen. For details, see "Input channel source select (GUI only)" on page 253.
- **Gain swap:** swaps what the rotary gain is controlling from remote (stage box) gain to digital trim (console gain), and vice versa. For details, see "Mic amp input gain (preliminary input processing)" on page 254.
- **Stereo linking:** links adjacent channel for stereo operation. For details, see "Stereo linking (GUI only)" on page 253.
- **Input channel direct output:** routes signal path from a selected point to an I/O. For details, see "Direct output" on page 258.
- **Input channel safes:** switches that protect specific controls from being changed by the automation system. For details, see "Safes" on page 254.
- **Gain and filter:** mic amp gain and filter controls.
- **Inserts:** allows configuration of the send and return points when an insert is used.
- **Input channel delay (GUI only):** user-defined delay to be added to the input signal processing. For details, see "Input channel delay (GUI only)" on page 253.
- **Processing order:** selects whether the EQ or the dynamics comes first in an input channel's signal path.

Their control is divided between control surface and the GUI, although some are GUI only. All of them are in the configuration processing area, with the exception of the inserts, which have their own processing area (see Figure 22 "Processing areas available from the input channel overview display. A. Depends on what device is connected (for example, DL251 Audio System I/O or DL451 Modular I/O). B. The display in this area depends on surround configuration." on page 250).

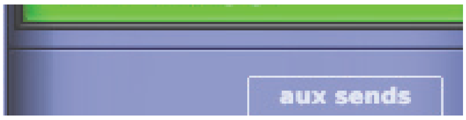


Input channel ID (GUI only)

You can change the channel name via the GUI, which can be done directly in the input channel overview or in any of the processing areas (see “Text editing” on page 42).

To change the background colour of the input channel name field (default is green), open the **Naming Sheet** screen of the GUI menu.

Input channel source select (GUI only)



The channel’s source is shown in the text field; if none has been selected, it will contain the text “Unassigned” (as shown right). You can select the source for this channel by clicking **source**, which opens the **Patching** screen (see Chapter 8 "Patching" on page 45). Also, by clicking the recorder button  you can set the input source to tape returns to obtain an alternative input, for example, from a hard disk recorder for a virtual soundcheck.

Input channel delay (GUI only)

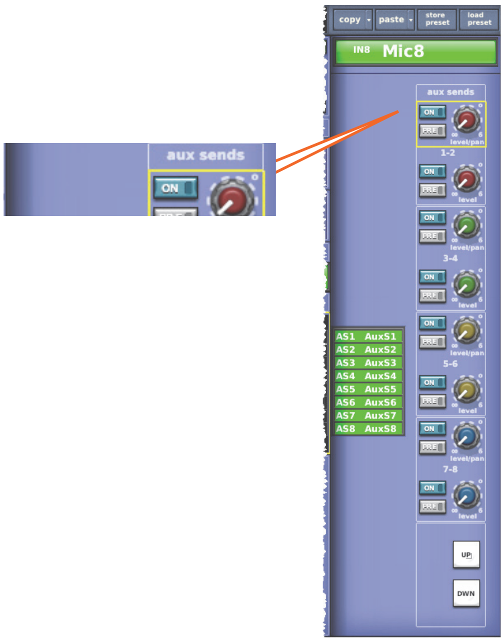
The input channel delay can only be changed via the **delay** section of the configuration processing area (GUI channel strip). This section has a control knob for adjusting the delay in the range 0ms to 50ms; this value is displayed in both milliseconds (ms) and metres. You can fine tune the delay value using the spin buttons to the left of the control knob.

The **delay** section allows you to 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 drum kit mic set up, you may have a mic close to a snare drum and a couple of overhead mics. In this case, setting an input channel delay on the snare drum — to bring it more in line with the overheads — will probably produce a better sound.

Stereo linking (GUI only)

The linking/gain swap section of the configuration detail has a **LINK OPT.** button that opens a **Stereo Linking Options** window from where you can choose which parameters you want to link between the pair.

For more information, see Chapter 10 "Stereo Linking" on page 93.

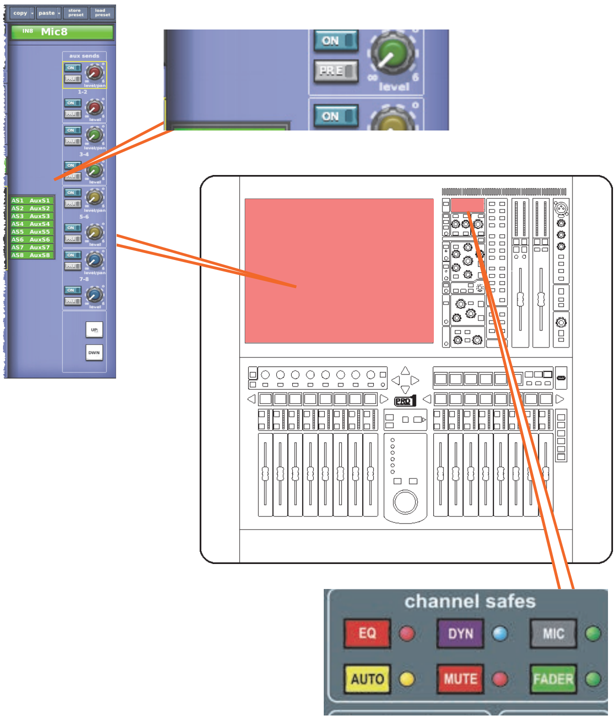


Safes

Each input channel has six different safes that protect specific controls/areas from the automation system.

You can switch the safes on/off by using the buttons in the **channel safes** section of the channel strips or via those in the **input channel safes** section on the GUI, which also illuminate when they are on.

For more information on what areas are protected by each safe, see Appendix H "Parameters Protected By Safes" on page 397.

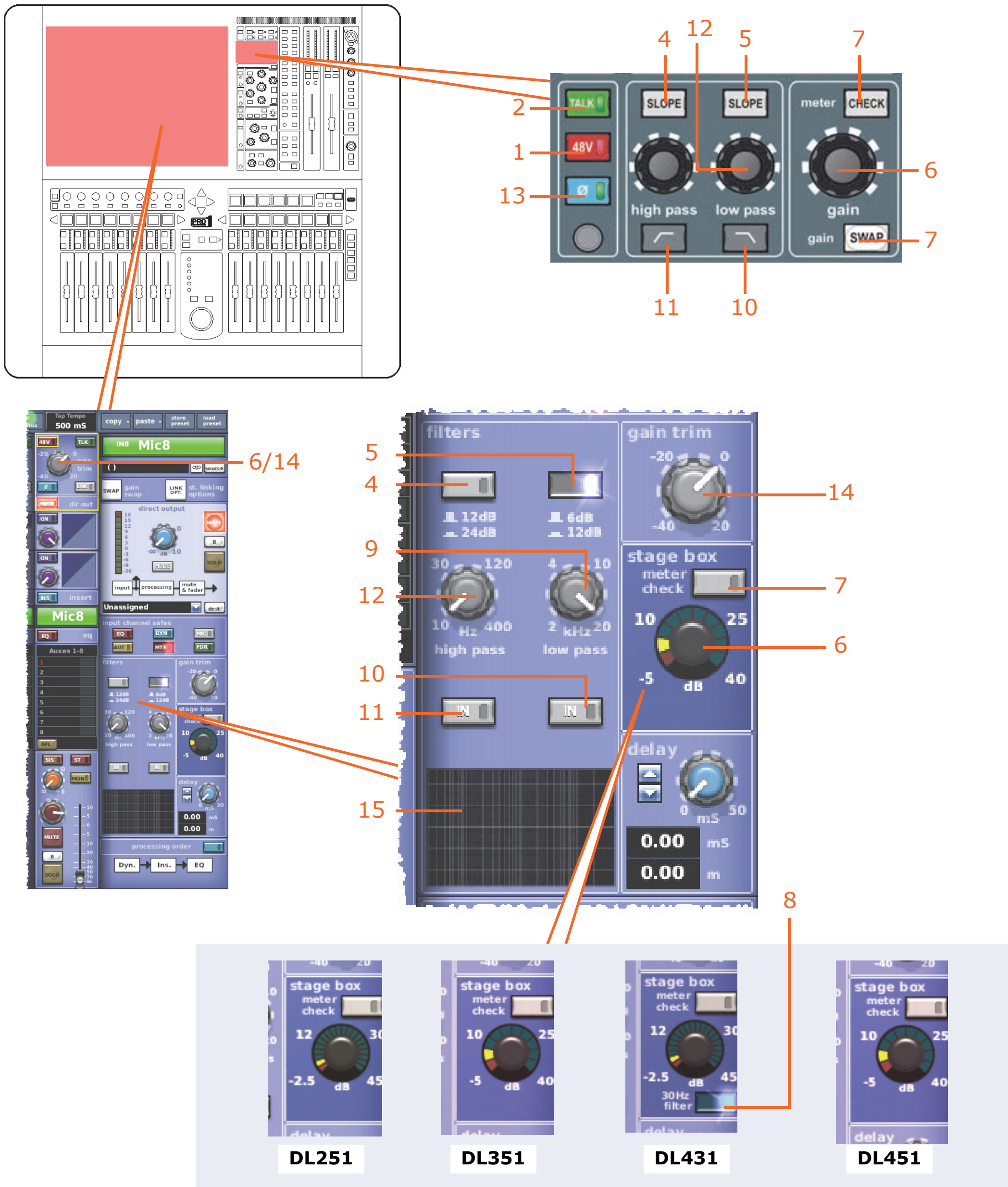


Mic amp input gain (preliminary input processing)

There are two types of mic input channel controls: digital and remote. Most of the 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 devices connected to the PRO1. For example, the analogue input module (DL441) has a 48V phantom voltage button and a gain control. The controls are adjusted via the device's configuration window (see "Configuring the devices" on page 58).

By default, console digital trim is adjusted by the gain trim control knob in each input fast strip and the remote gain control is adjusted by the stage box control knob in the input channel strip. However, by pressing the gain swap button these functions are swapped over, so that the gain trim control knob now controls the remote gain, and the stage box control knob controls the digital trim. Pressing gain swap again reverts them to default. As the legends of these two control knobs on the control surface are permanently fixed, their current 'swap' status can only be determined by the illumination of the **SWAP** button and what is shown for these controls on the GUI.



Mic amp input gain on the control surface and GUI

Item	*Control*	*Function*
1	**48V** switch (stage box only)	Connects 48 volts of phantom power to the XLR mic input channel connector. Suitable for a condenser microphone or DI box.
2	**TALK** switch	Connects talk mic and/or tone and noise generators to the input channel.
3	Gain swap button	See "Using gain swap" on page 257.
4	**SLOPE** switch (digital trim only)	Selects the value of the **high pass** filter. Where, switch on (illuminated) = 24dB slope and switch off (extinguished) = 12dB slope.

Item	*Control*	*Function*
5	**SLOPE** switch (digital trim only)	Selects the **low pass** filter. Where, switch on (illuminated) = 12dB slope and switch off = 6dB slope.
6	**stage box** control knob	Adjusts the input gain of the remote amplifier in 5dB steps, ranging from -5dB to +40dB. Note that the **stage box** control knob on the control surface will only adjust the gain currently selected to the GUI input channel strip, that is, stage box or digital trim.
7	**CHECK** switch (stage box only)	Monitors the mic amp input after the 30Hz filter, but before any further processing. (The 30Hz subsonic filter switch accesses the high pass filter on DL431 Mic Splitter when connected to a PRO1. In this case, the gain steps would be 2.5dB from -2.5dB to +45dB.)
8	**30Hz** subsonic filter switch	Acts on remote amplifier (mic splitter) to remove very low frequencies in the audio signal — usually caused by noise on stage. This avoids wasting valuable headroom trying to digitise it.
9	**low pass** control knob (digital trim only)	Adjusts frequency of low pass filter in the range 2kHz to 20kHz.
10	Low pass filter switch /[**IN**] (digital trim only)	Activates low pass filter in the input channel signal path before the insert points and EQ.
11	High pass filter switch /[**IN**] (digital trim only)	Activates high pass filter in the input channel signal path before the insert points and EQ.
12	**high pass** control knob (digital trim only)	Adjusts frequency of high pass filter in the range 10Hz to 400Hz.
13	Phase switch ø	Applies a 180° inversion of the input signal polarity within the input amplifier, such that channel signal will have opposite polarity to the input signal. This is used to correct input signal phase problems when trying to sum signals that are 180° out of phase. For example, where two mics are facing each other when using a mic on both the top and bottom of a snare drum. Ordinarily, the two mics would be out of phase - causing cancellation when the control centre sums the two signals into the output. Reversing the phase of one signal causes the mics to have the same phase, thus avoiding cancellation.
14	Gain trim (digital trim) control knob	Applies continuous trim adjustment (small digital steps) of the input signal level in the range –40dB to +20dB. Gives a further 60dB of fine adjustment (DSP) on top of the remote amplifier gain setting. This control knob on the input fast strips can be controlling stage box gain, digital trim or delay, depending on the current state of the swap.
15	Graph	Shows the effects of currently applied filter.

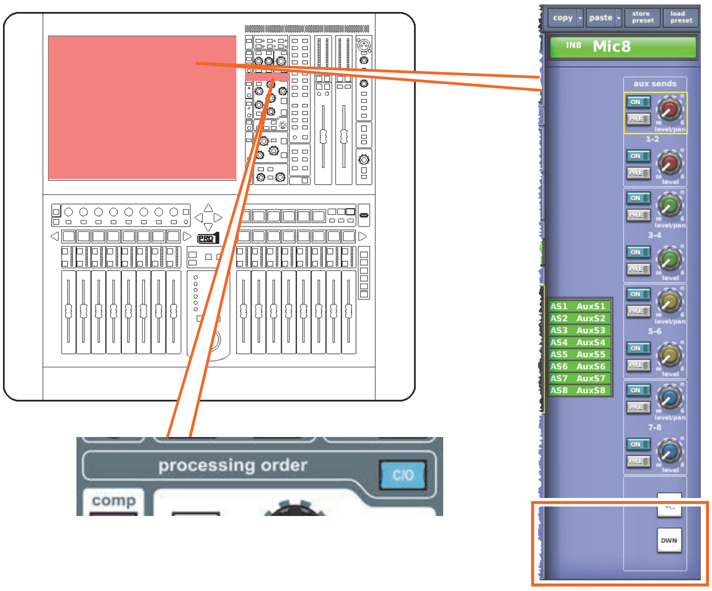
Using gain swap

Operating the gain **SWAP** button, swaps the function of the gain control (top of input fast strips) between that of the digital trim and stage box gain. In addition, if the **Fast Zone Delay Control** option (see "Setting the user interface preferences" on page 225) is enabled, input delay is also included in the swap.

Always check the GUI for 'swap' status.

Processing order

This section has a **C/O** switch that changes order of processing from EQ/INS/DYN (default) to DYN/INS/EQ and vice versa, which is shown in the GUI channel strip.

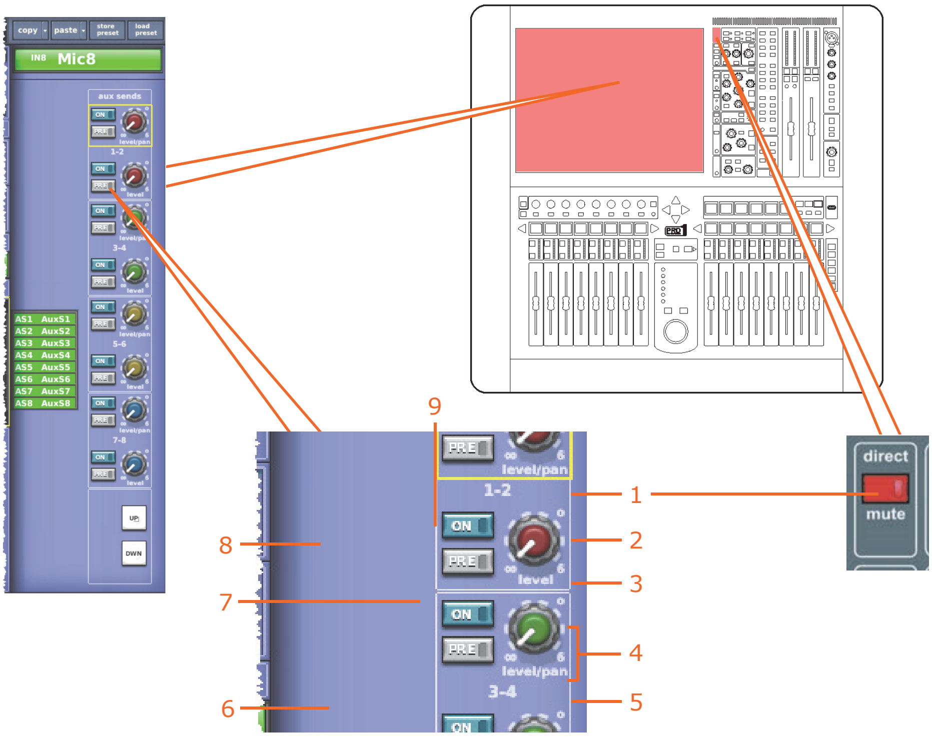


Direct output

The direct output section lets you to take a signal directly out of a defined point in the input channel's signal path and route it to either an internal assignable effect or a physical output (a physical connection at one of the line I/O boxes). This function is optional and assigned on a channel-by-channel basis.

This section is deliberately distanced from the main channel panel controls because it is a limited resource and unused on many channels.

Selection of signal path position (item 4) and destination (item 5) can only be carried out via the GUI.



Item	Description	Function
1	**MUTE** switch	Mutes any assigned direct output by removing signal from the output. However, it will not operate (will remain illuminated) if nothing is assigned. It is included in the scene recall system but is not affected by the channel mute safe or the auto-mute masters (unless the source tap-off point is after the main channel mute).
2	**B** switch	Changes the operation of the **SOLO** switch so that it routes signals to the monitor B section of the control centre.
3	**SOLO** switch	Activates signal routing to the Monitor A (or if the B switch is illuminated, Monitor B) section of the control centre.
4	Tap-off point diagram	Shows where the direct output is sourced from in the signal path, as selected by the mode button (see item 7).
5	**dest** button	Opens the **Patching** screen so that you can select the destination of the direct output.

Item	Description	Function
6	Direct output drop-down list	Displays the destination(s) of the direct output. For example, to an O/B vehicle, while simultaneously going into a DN9696.
7	**MODE** button	Changes the source tap-off point for the signal. There are three options: post-fader and mute; pre-mute and post-processing; or pre-mute and pre-processing. This function is not used if the direct output is not unassigned to channel.
8	11-LED meter,	Monitors the direct output level in the range +18dB to -36dB.
9	Control knob	Adjusts direct output level. Range is infinity (∞) to 10dB.

Dynamics (D zone)

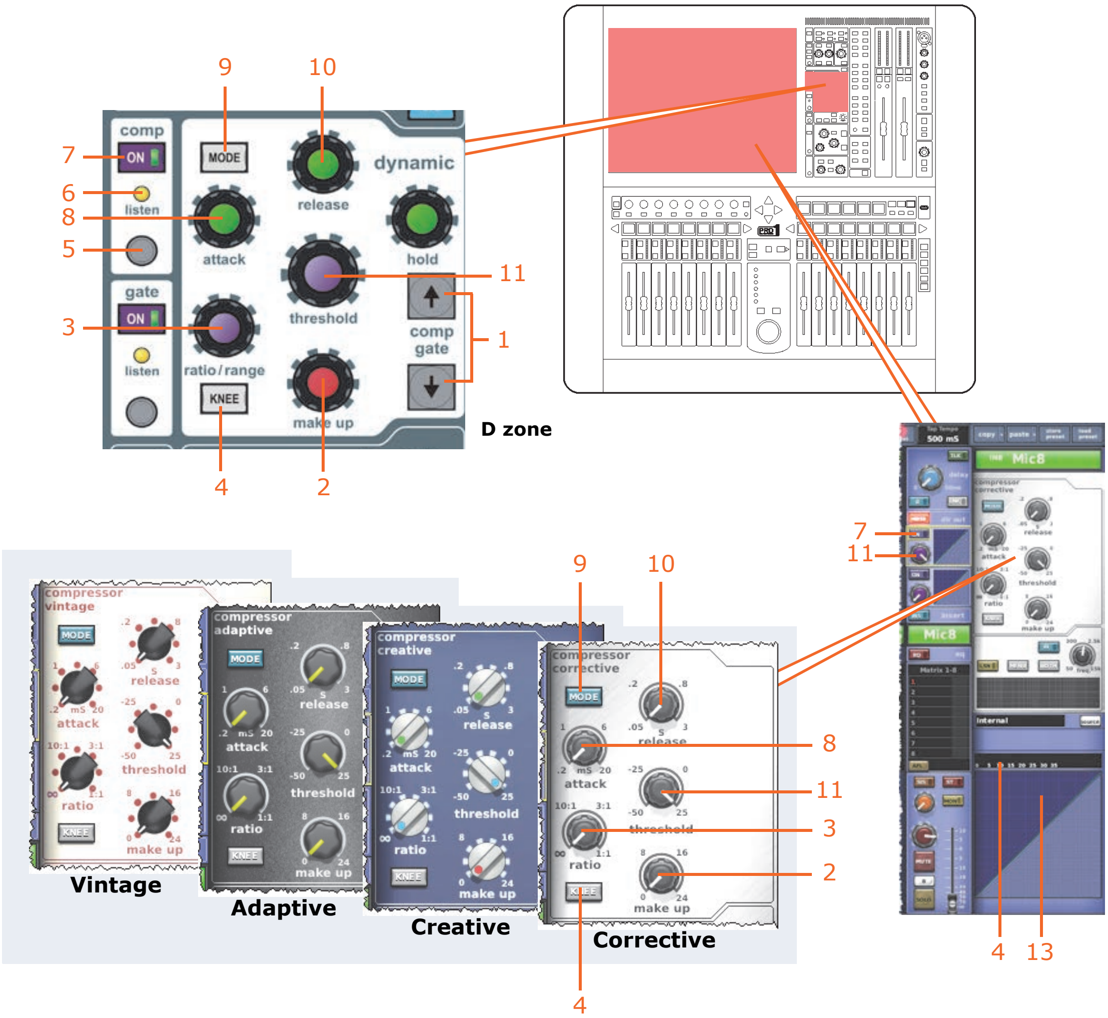
The **dynamic** section — or D zone — controls two dynamic devices present in the input channel signal path, that is, the compressor and gate. While most D zone controls are shared between the two dynamic devices, some are device-specific. The GUI treats both devices independently, the processing area of the one currently displayed in the channel strip being the one currently selected to the D zone. Swapping between the two dynamic devices can be done by clicking in the compressor/gate areas of an input fast strip, by pressing the input or gate quick access buttons, or the up/down select buttons in the D zone itself when the input channel strip is in use.

Activating the dynamic device’s **ON** button activates the device, but also affects the audio.

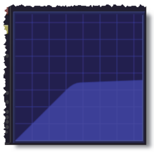
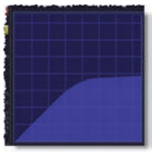
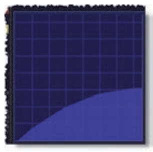
By default the source for the compressor's sidechain and gate's key (sidechain) is the channel itself, but by pressing or clicking on the menu button in the dynamics processing area you are taken to the patching page where you can choose another source for these inputs. For side chain details, see “Side chain” on page 265.

Compressor

The input channel compressor has four styles — corrective, adaptive, creative and vintage — which are selectable via the **MODE** button. Each has a distinctive sound and a different appearance in the GUI channel strip. While the dynamic section is addressing the compressor, all of its controls are enabled except the **hold** control knob.



Item	*Description*
1	**comp**/**gate** up and down select buttons, for swapping **dynamic** section control from compressor to gate, and vice versa.
2	Compressor **make up** gain control knob, compensates for the reduced *loudness* of a compressed signal. Range is from 0dB to 24dB.
3	Compressor **ratio** control knob, adjusts amount of compression applied to signals above threshold. Range is from infinity (∞) to 1:1 (maximum), which sets the compressor to limiter mode.

Item	Description
4	Compressor **KNEE** switch, controls how compressor starts to apply gain as the signal goes through the threshold (see "About the compressor graph" on page 262). There are three knee types as follows, accompanied by their typical affects on the compressor graph: • **Hard knee** Compressor immediately applies gain reduction at selected ratio once attack time has elapsed. This knee has hardly any curve at all.  • **Medium knee** Intermediate knee type.  • **Soft knee** Compressor, starting from slightly before threshold, gradually makes the transition to applying gain reduction at selected ratio. This knee is noticeably more rounded.  For application notes, see "Knee" on page 300.
5	Quick access button, which directly selects the compressor or gate processing areas on the input channel strip.
6	To aid set up, the compressor has a side chain listen that sends the side chain onto a solo bus. This side chain **listen** LED indicator illuminates to warn you that soloed material is from the side chain, and not the main channel. For information on the side chain, see "Side chain" on page 265.
7	**ON** switch, enables the compressor in the signal path. When switched off, compressor is bypassed. (Both the **comp** and **gate** switches can be on at the same time.)
8	Compressor **attack** control knob, adjusts time for compressor to respond after an over-threshold signal. Range is from 0.2ms to 20ms (milliseconds).
9	**MODE** switch, selects compressor mode. There are four compressor types available: **corrective**, **adaptive**, **creative** and **vintage**. See "PRO1 compressor modes (dynamic)" on page 299 for details.
10	Compressor **release** control knob, adjusts time for compressor to recover after programme material falls back below threshold. Range is from 0.05s to 3.00s (seconds).
11	**threshold** control knob, sets the signal level above which gain reduction starts to be applied. Range is from -50dB to +25dB.
12	Compressor 'gain reduction' meter.
13	Compressor graph (see "About the compressor graph" on page 262).

About the compressor graph

The GUI shows a graph of the compressor envelope (input level against the output level) to help you visualise the effect that the compressor is having on the signal (typically as shown right).

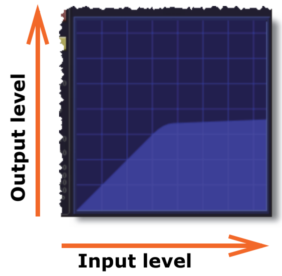


Figure 23 "Compressor graphs showing the effect of ratio" shows what happens to the compressor graph with and without ratio. Initially, both graphs have the same slope, which is pre-threshold and unaffected by compression; this has a gradient of 1:1 — 'what you put into the compressor, you get out'. Without ratio the slope is constant and remains unaffected by compression. However, with ratio the gradient changes at threshold, which is the point where compression starts to be applied. After this the gradient of the post-threshold signal is at the selected ratio value.

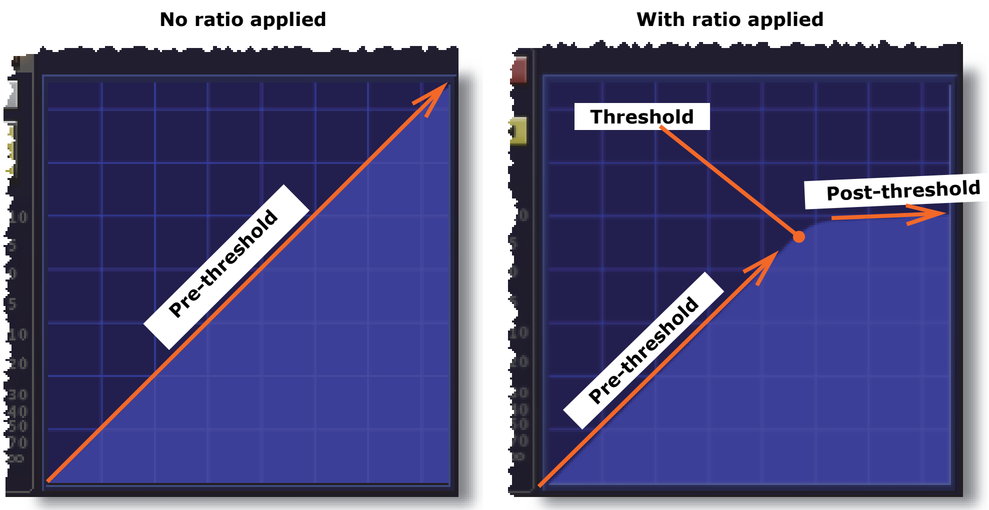
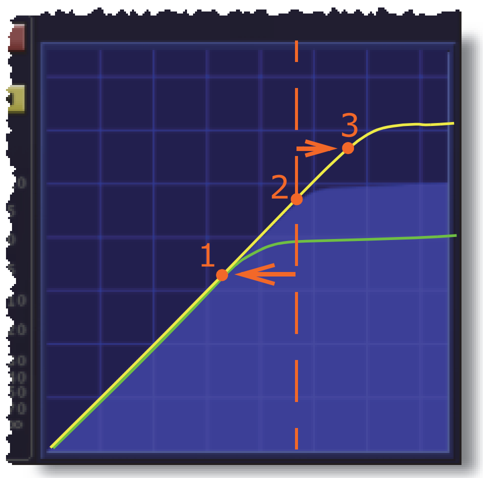


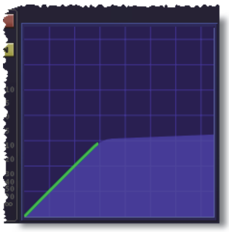
Figure 23: Compressor graphs showing the effect of ratio

The diagram right shows the effects of threshold adjustment. Position **2** is the threshold of the actual compressor graph shown on the GUI. If you reduce threshold (for example, to position **1**), compression starts earlier and less signal is passed 1:1; the signal path would follow the green line. Conversely, increasing threshold (for example, to position **3**) delays compression and more signal is passed 1:1; the signal path would follow the yellow line.



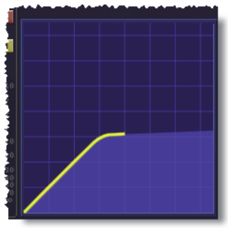
With a signal running through the compressor, a coloured line on the graph follows the contour of the shaded graph area. The line's colour changes according to signal level to show the level of compression, that is, uncompressed (green), within knee area (yellow) or at full compression (red). An example of each is shown in the following diagram.

Uncompressed



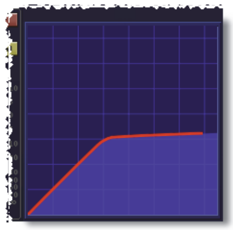
If signal doesn't reach threshold (point where gradient changes), the line is **green**. As the threshold is not exceeded, the signal is uncompressed.

Within knee area



If signal goes into knee area to point where gradient changes (more obvious with medium and soft knees), compression starts to be applied and line colour changes to **yellow**.

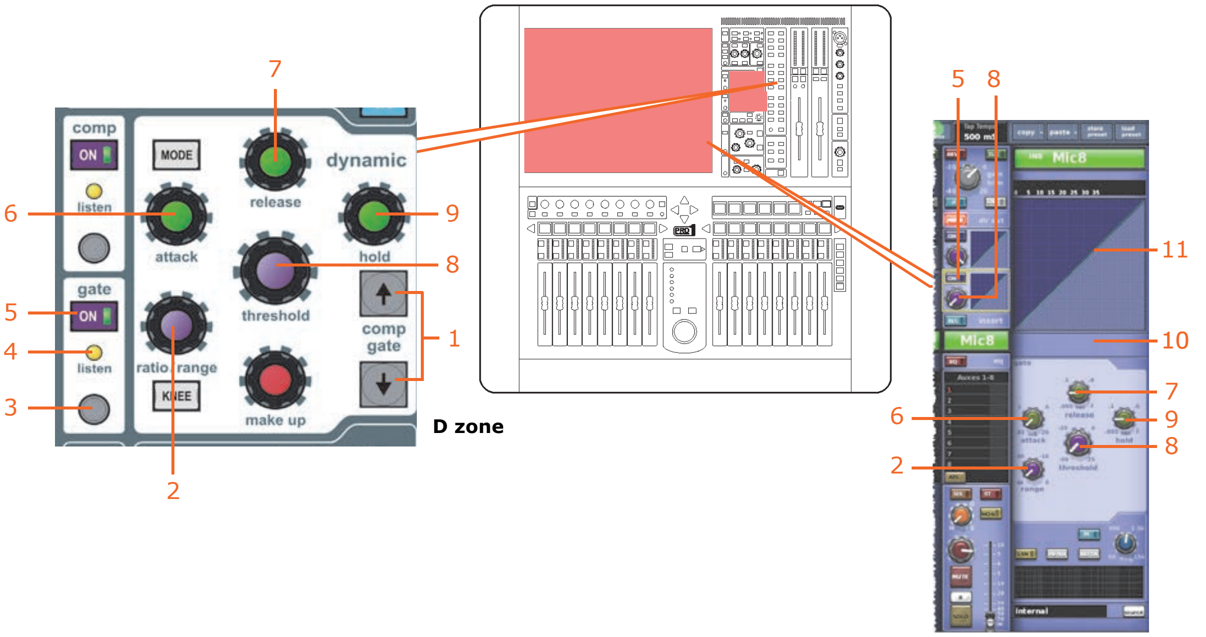
Fully compressed



If signal reaches the point where gradient changes (over-threshold), full compression at selected ratio is applied and line colour changes to **red**.

Gate

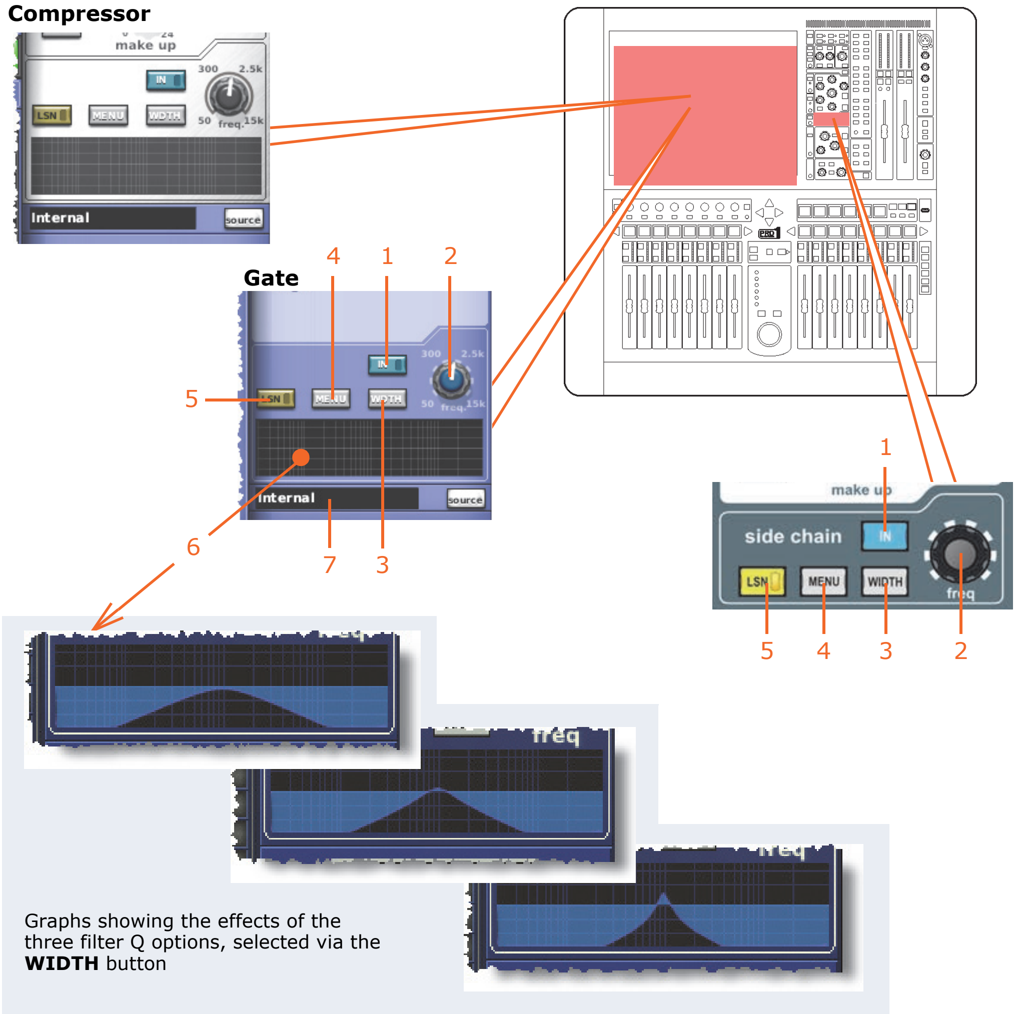
Unlike the compressor, gate mode has only one style. While the **dynamic** section is addressing the gate, all of its controls are enabled except the **make up** control knob and the **MODE** and **KNEE** buttons.



Item	Description
1	Up/down select buttons, for swapping **dynamic** section control from compressor to gate, and vice versa.
2	Gate **range** control knob, adjusts 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	Quick access button, which directly selects the compressor or gate processing areas on the input channel strip.
4	To aid set up, the gate has a side chain listen that sends the side chain onto a solo bus. This side chain **listen** LED indicator illuminates to warn you that soloed material is from the side chain, and not the main channel. For information on the side chain, see "Side chain" on page 265.
5	**ON** switch, enables gate in the signal path. When switched off, gate is bypassed. (Both the **comp** and **gate** switches can be on at the same time.)
6	**attack** control knob, adjusts time taken for gate to open after an over-threshold signal. Range is from 0.02ms to 20ms (milliseconds).
7	**release** control knob, adjusts time taken for gate to close after programme material falls back below threshold. Range is from -0.005s to 2.000s (seconds).
8	**threshold** control knob, sets signal level at which gate opens. Range is from -50dB to +25dB.
9	**hold** control knob, minimises chattering in conjunction with internal hysteresis. Once the signal is detected as below threshold, this defines a waiting period before the gate starts to close. Range is from -0.005s to 2.000s (seconds).
10	Gate meter.
11	Gate graph display. Similar to the compressor graph (see "About the compressor graph" on page 262), this shows the effects of adjusting the gate control knobs.

Side chain

You can manipulate the side chain filter from the **side chain** section (channel strip and GUI). The side chain filter is a swept band pass type, which acts on the dynamics side chains of the compressor and gate, and covers the full audio spectrum.



Graphs showing the effects of the three filter Q options, selected via the **WIDTH** button

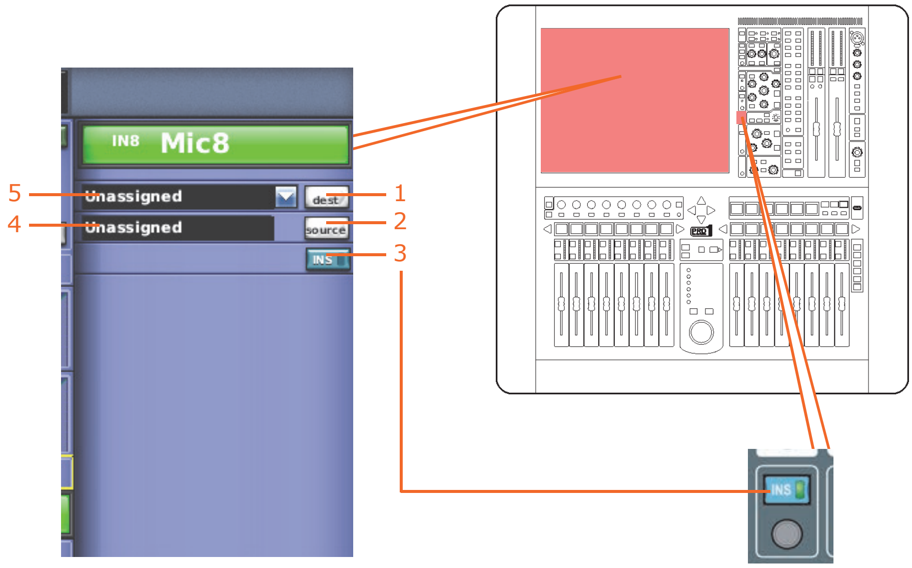
Item	*Description*
1	**IN** switch, switches the side chain filter into the sidechain signal path.
2	**freq** control knob, adjusts the side chain filter frequency in the range 50Hz to 15kHz. (Visually, this moves the envelope on the graph left or right.)
3	**WIDTH** button, changes the filter Q. There are three options, and the effects of each are shown in the side chain graph (see above). This is only enabled when the side chain filter is switched on.
4	**MENU** button, opens the **Select Side-Chain Source** window from which you can select the side chain source for the selected input channel (see “Side chain” on page 265). Pressing this button with the **Select Side-Chain Source** window open, closes the window.
5	**LISTEN**/[**LSTN**] switch, places the side chain pushbutton onto the channel filter bus, allowing the audio signal to be monitored via headphones. This effectively replaces the channel solo audio path with a post-filter (pre-dynamic) signal.

Item	Description
6	Graph, shows the effects of the side chain filter on the signal.
7	Side chain source field, shows you where the side chain of the compressor/ gate is sourced from. If you see the text "internal" here, it means that the source is from the channel itself.

For details of how to select a side chain pick-off point, see "Side chain" on page 265.

Insert

Input channel insert section provides a send and return out of the signal path, primarily so that an effects device 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** switch can be found on both the GUI and also in each input fast strip. This section is optional and assigned on a channel-by-channel basis.

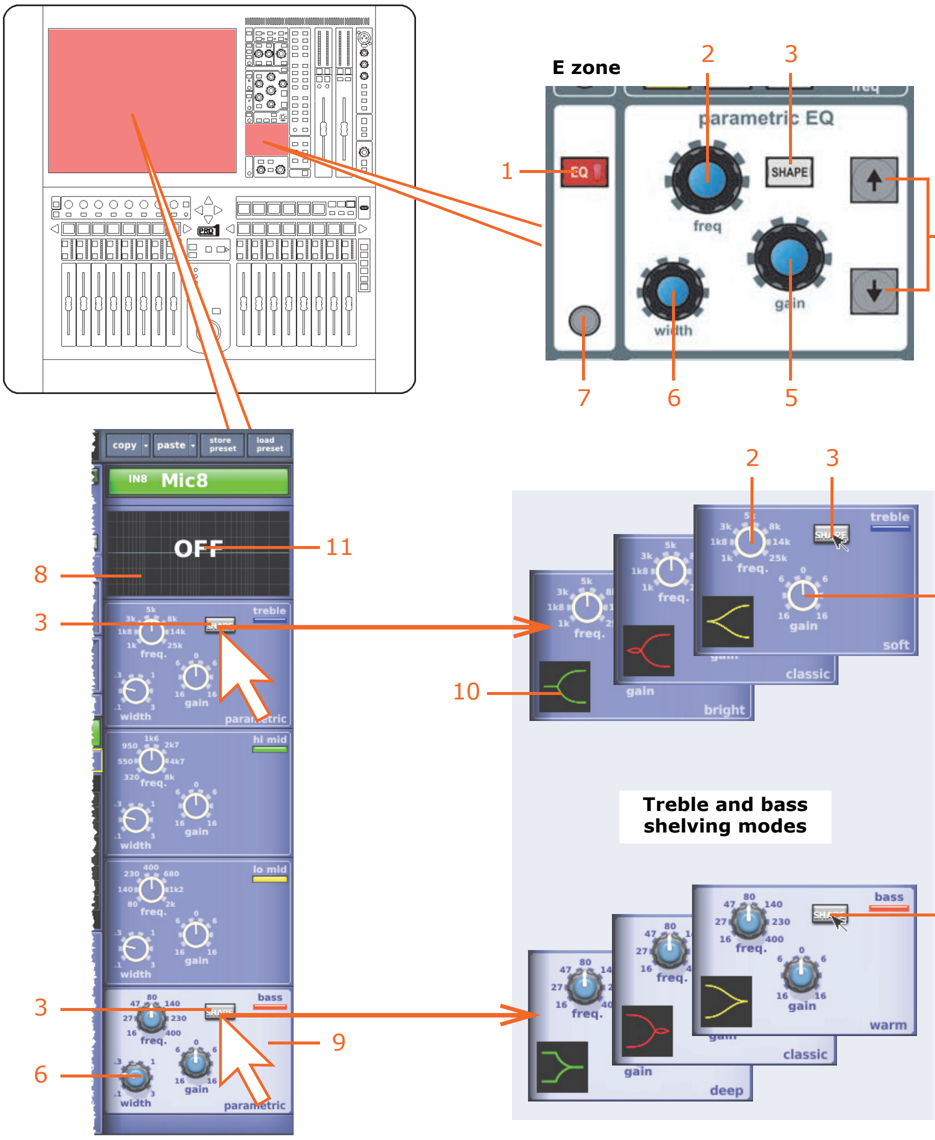


Item	Description
1	**dest** button opens the **Patching** screen from where you can select the destination of the insert send.
2	**source** button opens the **Patching** screen from where you can select the source of the insert return.
3	**INS** switch, connects (inserts) returned programme material to the channel signal path, provided both the insert send and insert return points have been assigned.
4	Insert return field shows you the source of the insert return.
5	Insert send field has a drop-down list, which shows the destination(s) of the insert return.

EQ (E zone)

The input channel equaliser (EQ) is a four-band swept parametric EQ (PEQ) that allows tonal control of the input signal via the parametric EQ section, or E zone, in the input channel strip. The four bands are treble, hi-mid, lo-mid and bass, with an additional three shelving modes available for treble and bass. Any combination of the four bands can be used to control the signal, although only one band can be adjusted in the E zone at any time.

The E zone contains all of the PEQ controls, along with a shelving mode selection button and a set of band selection buttons.



Item	Description
1	**EQ** on/off switch.
2	**freq** control knob adjusts the band's centre frequency. The frequencies that each band covers increases as you move from the bass band to the treble band. The graph in the EQ processing area on the GUI channel strip gives you a visual indication, once some gain has been applied, of where the band is located.
3	**SHAPE** button, changes shelving mode on treble and bass bands. For recommended usage, see Table 12 "Recommended band mode usage" on page 268. For a description of each mode, see "PRO1 input channel EQ modes" on page 302.
4	Up/down band selection buttons, cycle through the bands, changing what the E zone controls are controlling.

Item	*Description*
5	**gain** control knob, adjusts the gain of each band in the range -16dB to +16dB. And on the graph in the EQ processing area of the GUI channel strip, this causes the envelope to move up/down.
6	**width** control knob, adjusts the signal bandwidth in the range 0.1 Oct to 3.0 Oct. On the graph in the EQ processing area (GUI channel strip), causes the base of the envelope to widen. (Not available for treble and bass shelving modes.)
7	Quick access button, selects the EQ processing area (E zone) of the input channel strip.
8	Graph of EQ envelope.
9	Highlighted section indicates the band the E zone controls are currently controlling.
10	Icon representing the shape of the signal's envelope.
11	"OFF" is displayed when the EQ is switched off.

In the GUI channel strip, the EQ processing area displays all four bands simultaneously and has a graph that shows a colour-coded EQ envelope for each selected band. Here, you can view the settings of the four bands simultaneously. The GUI also shows the ranges available for each control knob and indicates the active band, which is distinguished by its cream-coloured background.

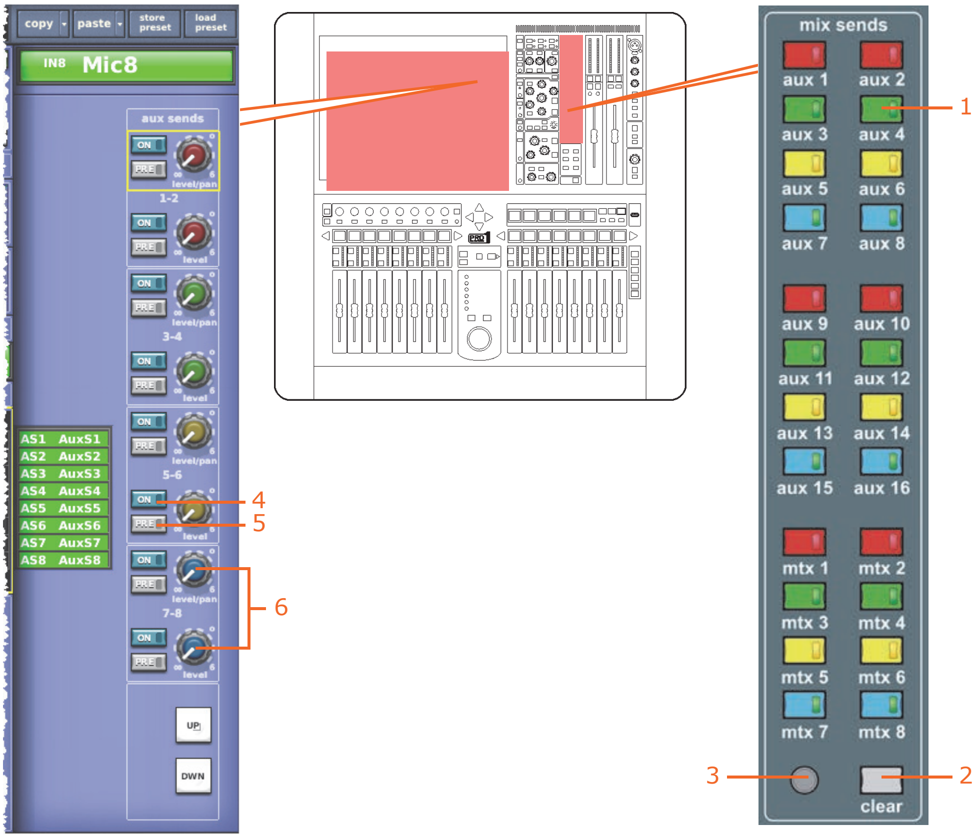
The following table illustrates the recommended uses of the treble and bass shelving modes.

Table 12: Recommended band mode usage

Band	*Mode*	*Best*
Treble	Bright	On single source material
Treble	Classic	Good for single source and pre-mixed material
Treble	Soft	For gentle shaping of pre-mixed material
Bass	Deep	On single source material
Bass	Classic	All round EQ
Bass	Warm	For gentle shaping of pre-mixed material

Mixes

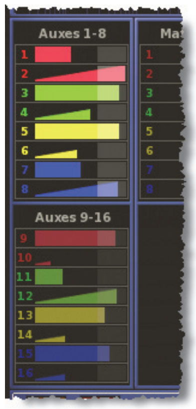
Each input channel can send an individually controllable contribution to each of the 16 aux buses (**aux sends**) and eight matrix buses (**mtx sends**). The contributions to the buses are controlled by mix controls that give continuous adjustment (in the range +6dB to off). The controls in the mix fader bay include **level**/**pan** and **level** control knobs, with the pan control coming into operation when the corresponding bus is stereo linked to its neighbour.



Item	Description
1	Aux/matrix buttons to activate flip on the channel faders.
2	**clear** button, cancels flip.
3	Quick access button for assigning the mix sends to the GUI channel processing area.
4	**ON** switches, switch bus assignment on/off.
5	**PRE** buttons; when on, signal is pre-fader.
6	After a input channel has been selected to the channel strip, **level/pan** and **level** control knobs offer control of relative contribution levels onto the active buses. **level/pan** operates odd numbered controls, while **level** operates the even ones.

The **mix** section in the channel processing area controls a bank of eight buses per selected input channel and this layout is replicated in the GUI channel strip.

However, the GUI's **input channel overview** gives a simultaneous display of the status of all 24 buses. It displays the levels sent to the buses and shows which are on/off and whether they are pre- or post-fader.



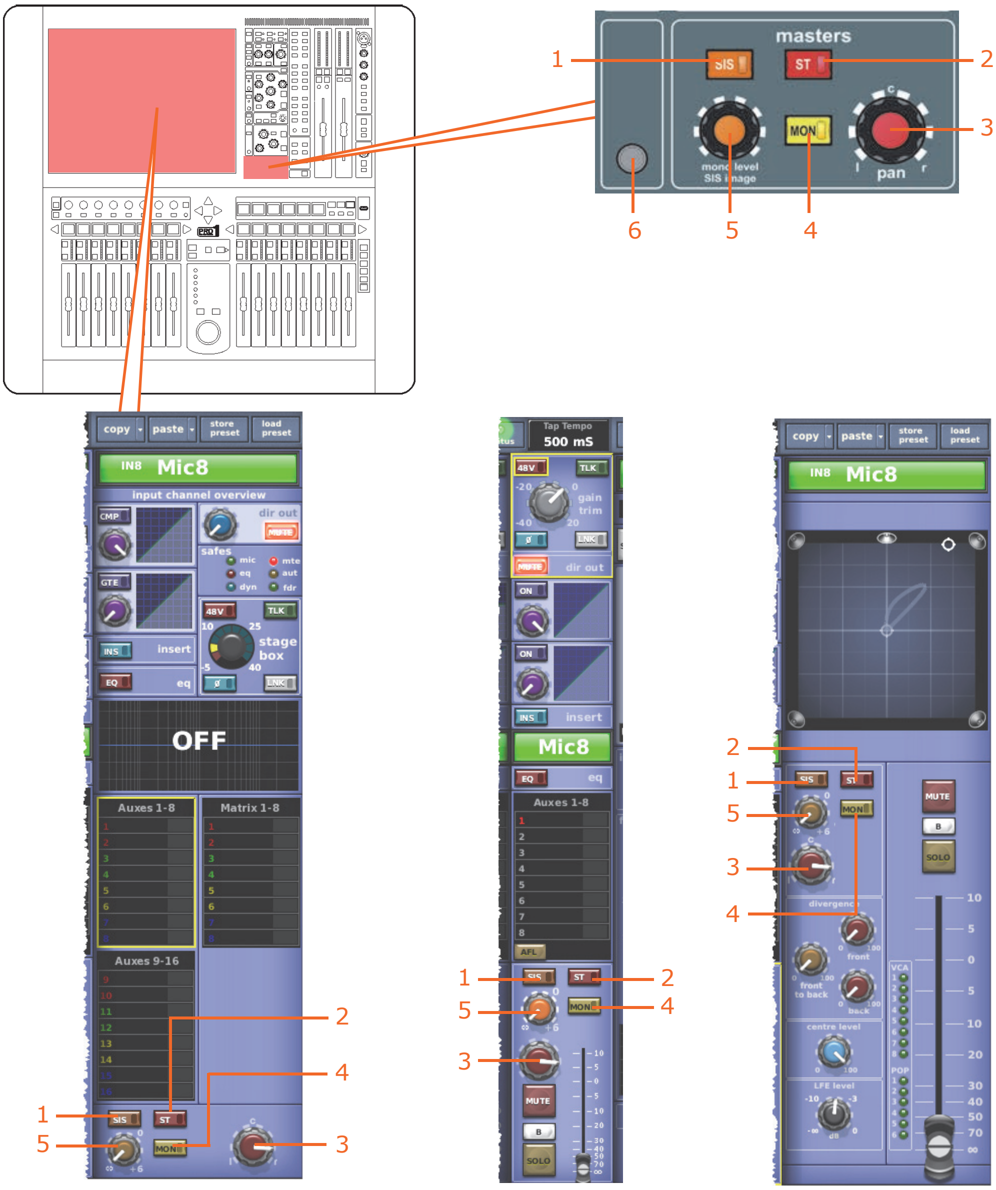
Don't forget, you can edit the levels on the GUI using drag.

Mix busses can operate in one of three modes and each mix bus can optionally be stereo linked to the bus next to it. So, depending on the set up, some of the mix bus controls change their function and some might not be available. In each of the modes the following controls are available:

Bus type	***Control***
Mono mix	**level** control knob and **ON** and **PRE** buttons. Also, if **PRE** is not enabled, the level sent to the mix bus is affected by the main fader control.
Stereo mix	**level** control knob, **level/pan** control knob (level control knob for the odd numbered bus) and **ON** and **PRE** buttons. Also, if **PRE** is not enabled, the level sent to each side of the stereo bus is affected by the main fader and pan controls.
Mono group	**ON** button only. The level sent to the group bus is controlled by the main fader control.
Stereo group	**ON** button only. The level sent to each side of the stereo bus is controlled by the main fader and pan controls.
Mono mix minus	**ON** button only, which is labelled **minus** on the GUI fast inputs and channel strip overview. The level sent to the bus is controlled by the main fader control. This switch works the opposite way round to normal, and when it is on the audio is not sent to the bus.
Stereo mix minus	**ON** button only, which is labelled **minus** on the GUI fast inputs and channel strip overview. The level sent to the side of the stereo bus is controlled by the main fader and pan controls. This switch works the opposite way round to normal, and when it is on the audio is not sent to the bus.

Master controls

In general, there are three routing switches to the master buses and also pan control. Pan provides master panning as three-way or two-way (depending on SIS™ setting) and also provides two-way panning for any stereo mix groups stereo and subgroups etc. (When used in fader flip mode, sends to mix buses are controlled from the channel master pan and fader.



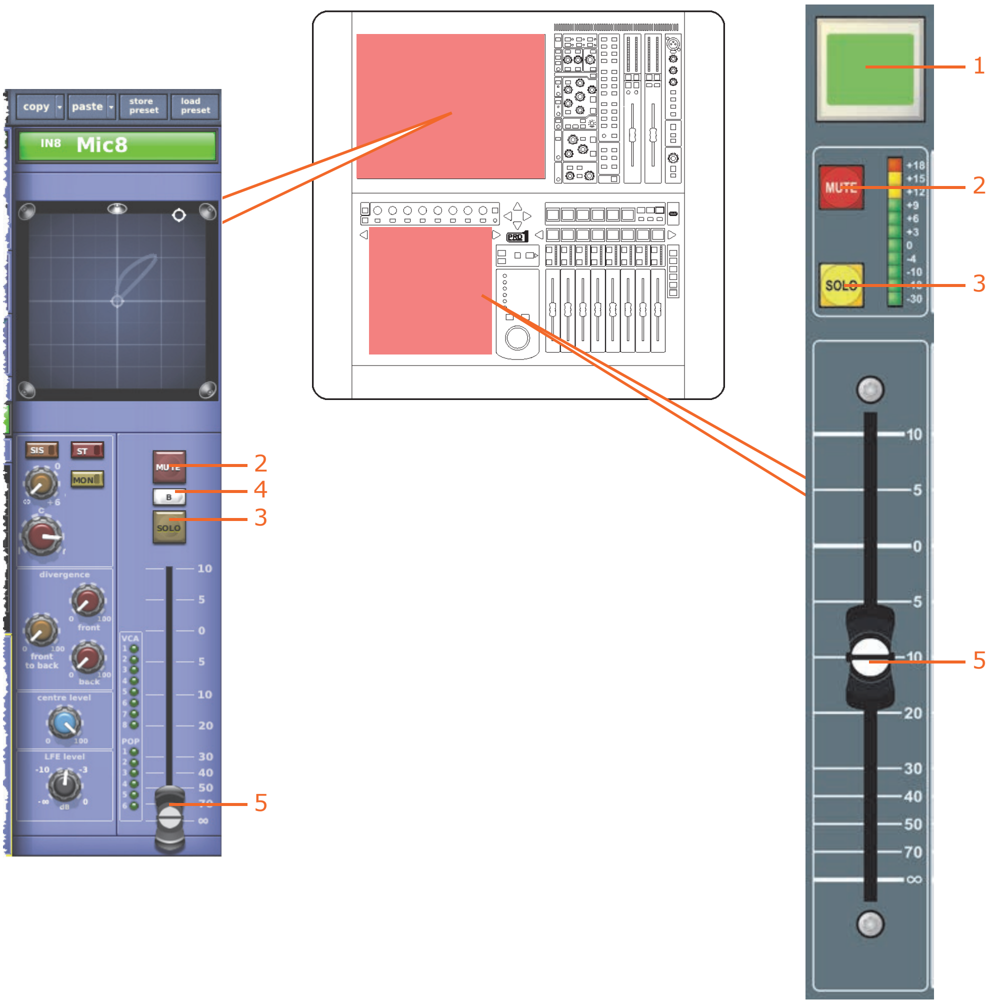
Item	Description
1	**SIS** (spatial imaging system) switch, enables SIS™ mode. This mode operates with the **pan** and **mono level SIS image** control knobs, and acts as an LCR master bus enable, overriding stereo and mono master bus assignments. However, their status remains in memory so that when **SIS** is disengaged, the mono and stereo settings can return. Pressing **SIS** alters the gradations of the **mono level SIS image** control knob on the GUI.
2	**ST** (stereo) switch, connects post-fader channel signal to master stereo bus via pan control.

Item	*Description*
3	**pan** control knob, adjusts the relative levels sent to a left-right bus pair or the master left-centre-right (LCR) buses. In SIS™ mode, it can also control the 'image' to give a constant power crossfade from LCR to stereo.
4	**MON** (mono) switch, connects post-fader channel signal to mono master bus.
5	**mono level/SIS image** dual-function control knob. In mono mode, it acts as a mono level control knob to adjust the mono signal level. In SIS™ mode, it becomes a SIS image control knob that modifies **pan** control knob operation to place the channel within a three-speaker system (see "Stereo panning" on page 95).
6	Quick access button, selects the master processing area on the GUI channels strip.

For more details, refer to "Stereo panning" on page 86 and "Spatial imaging system (SIS™)" on page 313.

Solo, mute and fader

Each input fast strip contains the **MUTE** and **SOLO** buttons motorised fader. The fader is replicated in each input fast strip on the GUI and also in the masters processing area (GUI channel strip). The fader controls the channel signal level and provides instant feedback of level settings.



Item	*Description*
1	LCD select button.
2	**MUTE** button, mutes all post-processing signals exiting the channel. In addition to scene recall, it can also be remotely muted from the auto-mute masters.
3	**SOLO** button, activates signal routing to the monitor A or B section of the control centre (depending on the status of the solo B button) by sending the input channel signal to PFL mono and AFL stereo buses. The solo system is auto-cancelling, so that each new solo cancels the last one. Input solos override any active VCA and bus solos. This button is a latching type. Give a short press to latch it and a long press to self-cancel when it is released.
4	Solo **B** button (GUI only), changes the operation of the **SOLO** switch so that it routes signals to the monitor B section of the control centre.
5	Motorised fader.

The LCD select buttons in the input fast strips are used for input channel navigation and group selection. They also provide useful feedback for the user.

For more information on navigation, see "About the PRO1 controls" on page 37.

Aux returns

The aux returns are input channels and their functionality is as broadly described for the mic inputs in the earlier sections of this chapter.

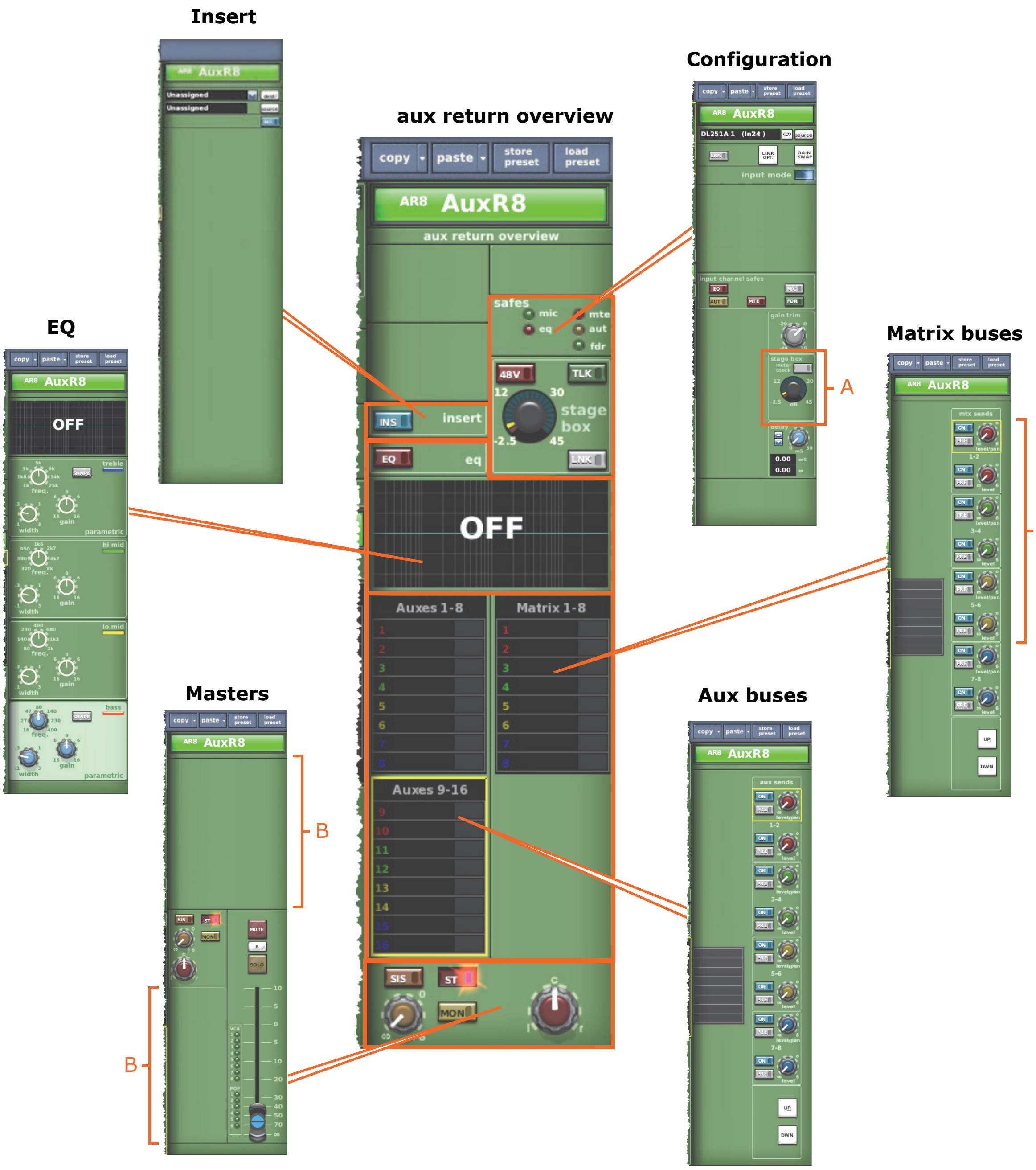


*Figure 24: Processing areas available from the **aux return overview** display. **A.** Depends on what device is connected (for example, DL251 Audio System I/O or DL451 Modular I/O). **B.** The display in this area depends on surround configuration.*

Chapter 31: Output Channels

This chapter shows you the areas on the control surface that are used to manage the outputs and also describes their function. There are three type of output: auxes, matrices and masters.

The structure of this chapter is loosely based on the signal path of the output channels and also the processing areas, which are opened via the output channel overview displays in the GUI channel strip.

Output channel routing

The following table shows the approximate signal path of each output type.

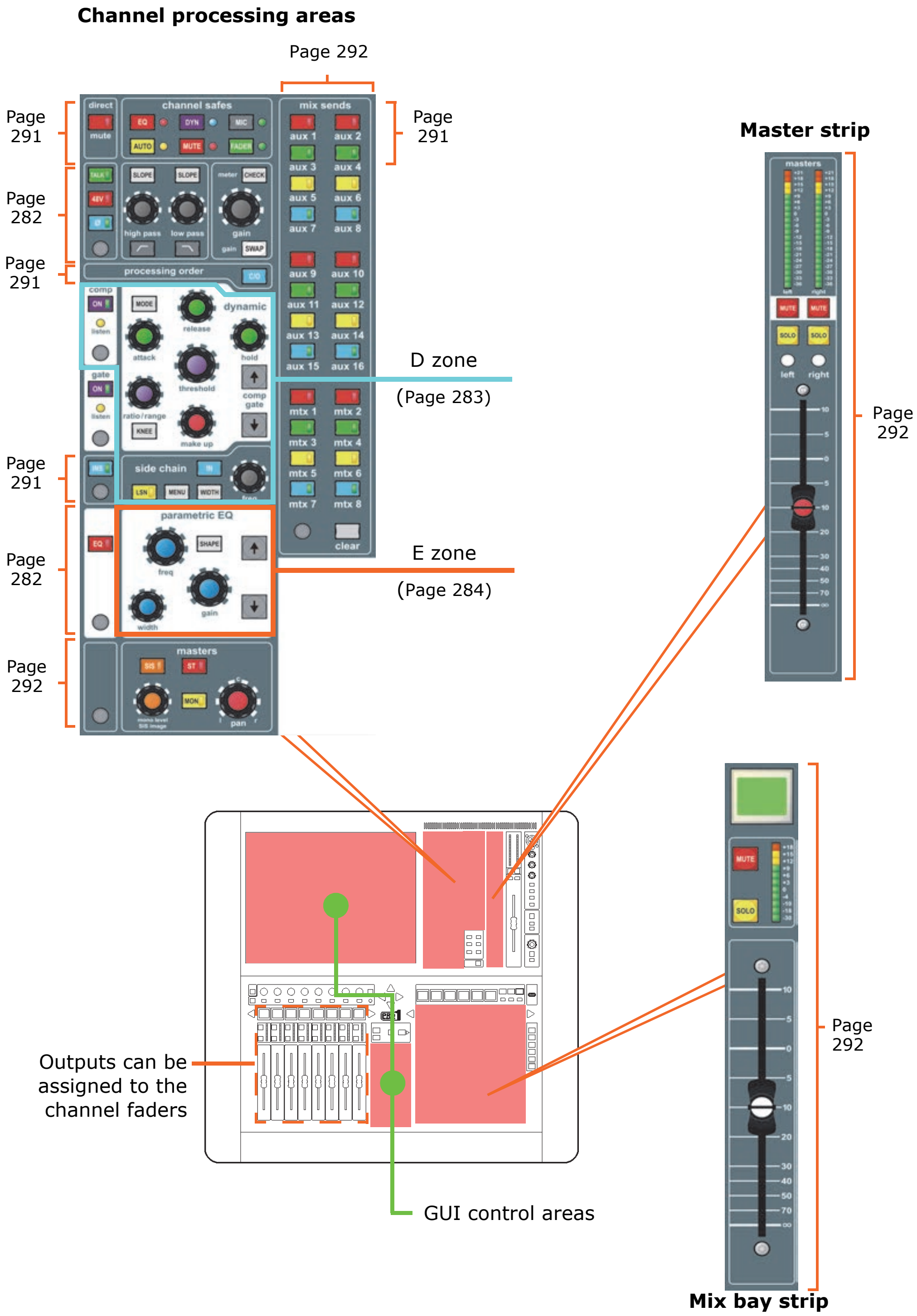
Channel controls	*Aux*	*Matrix*	*Master*
Configuration	See page 289	See page 289	See page 289
Direct input	See page 291	See page 291	See page 291
***#Dynamics (dual compressor)**	See page 283	See page 283	See page 283
Insert	See page 291	See page 291	See page 291
***EQ**	See page 284	See page 284	See page 284
Mixes	Matrix sends only (see page 292)	N/A	N/A
Master controls, solo select and fader	See page 292	See page 292	See page 292

* Order can be swapped (see "Processing order" on page 291).

Includes side chain section.

Output channel areas on the control surface

In the default mode of operation the output channels are assigned to the mix faders. However, by using the **OUTPUT** and **EXTEND** buttons (see the PRO1 Control Centre Quick Start Guide) they can also be extended to the channel faders as well. Detail adjustment is augmented by the channel processing areas. The hardware controls are replicated on GUI, which also provides extensive support and additional functionality.



Outputs on the GUI

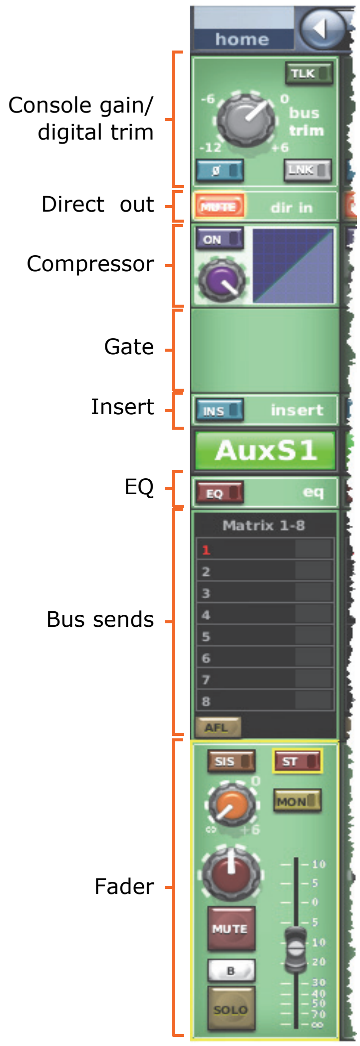
The outputs are represented on the GUI in a similar way to the inputs, although each has a different coloured background. Each output channel has a GUI fast strip and these are displayed in banks of eight (except for the three masters). When a channel is selected, its overview is displayed in the GUI channel strip; selecting a non-control area in an overview section will open its associated processing area. This will be detailed more fully later on in this chapter.

All of the outputs are shown in on the **Console Overview** screen of the GUI and each has its own meter. For more information, see "Input metering" on page 251.

GUI output fast strips

The output fast strips on the GUI (a typical example is shown right) give an overview of their equivalent versions on the control surface. These show the gain, bus controls, pan control knob and fader.

Some processing areas are configuration dependent, such as the bus sends (depends on surround configuration) and console gain/digital trim.



GUI channel strips

When a channel is selected, its 'overview' appears in the channel strip. For diagrams showing the 'overview' of each output and all their processing areas, see Figure 25, Figure 26 and Figure 27.

As with the input channels, the output overview provides a limited set of controls and status information (see "Inputs on the GUI" on page 249).

Processing areas

The following processing areas are available from 'overview' displays in the GUI channel strip. To see the available processing areas for each output, see Figure 25, Figure 26 and Figure 27.

Channel controls	*Aux*	*Matrix*	*Master*
Insert only	N/A	N/A	N/A
Configuration only	N/A	N/A	N/A
Insert and configuration	Yes	Yes	Yes
Compressor	Yes	Yes	Yes
EQ	Yes	Yes	Yes
Buses	Yes (matrix only)	N/A	Yes (matrix only)
Solo, mute, safes and fader only	N/A	Yes	N/A
Masters and solo, mute, safes and fader	Yes	N/A	Yes

For details of how to navigate the GUI channel strip, see "About GUI navigation" on page 44.

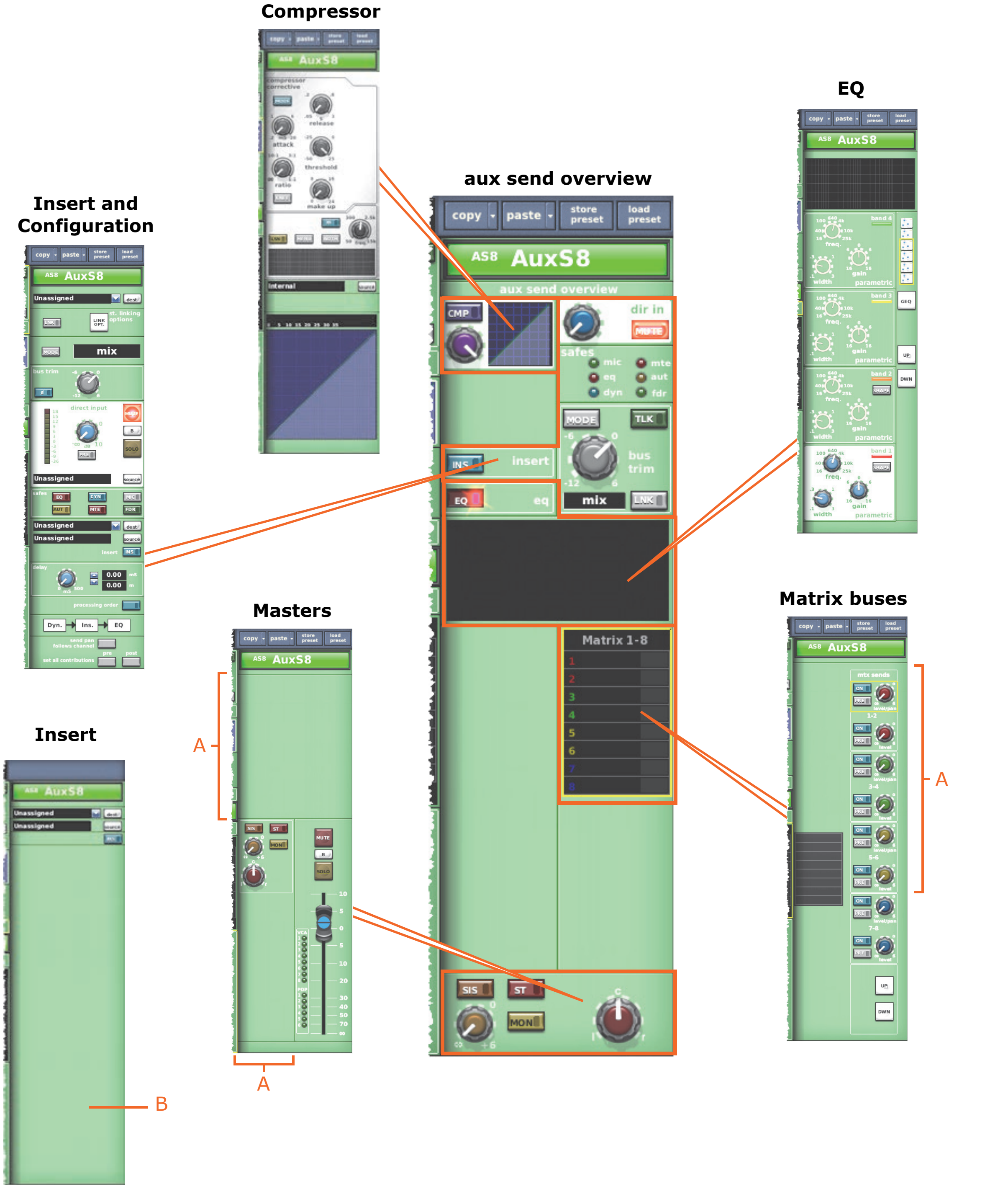


Figure 25: Processing areas available from the ***aux send overview*** *display.*
A. *The display in this area depends on surround configuration.*
B. *The insert processing area is selected via the quick access button in the insert processing area or by clicking in the insert section of a channel strip on the GUI.*

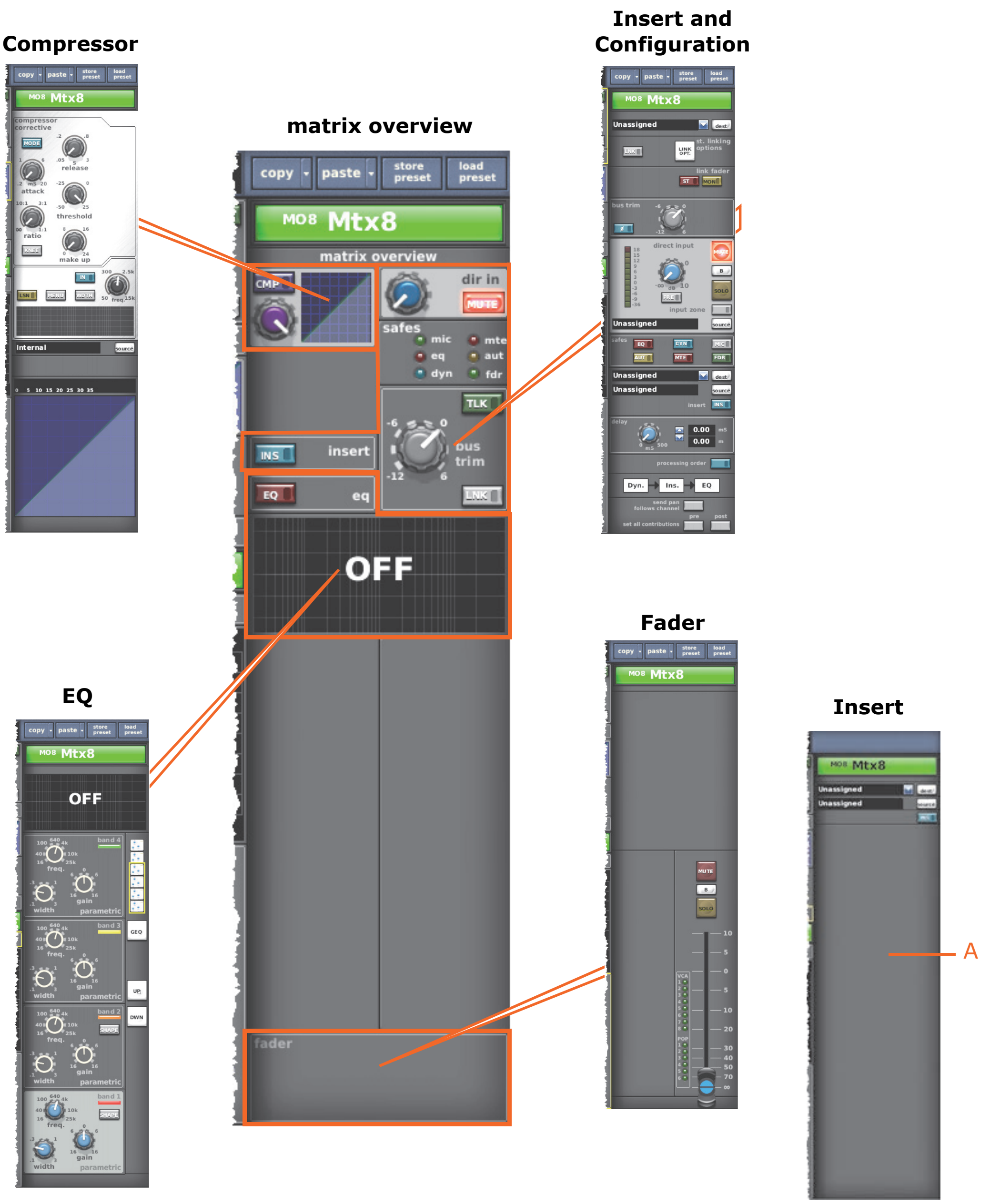


Figure 26: Processing areas available from the ***matrix overview*** *display*
A. *The insert processing area is selected via the quick access button in the insert processing area or by clicking in the insert section of a channel strip on the GUI.*

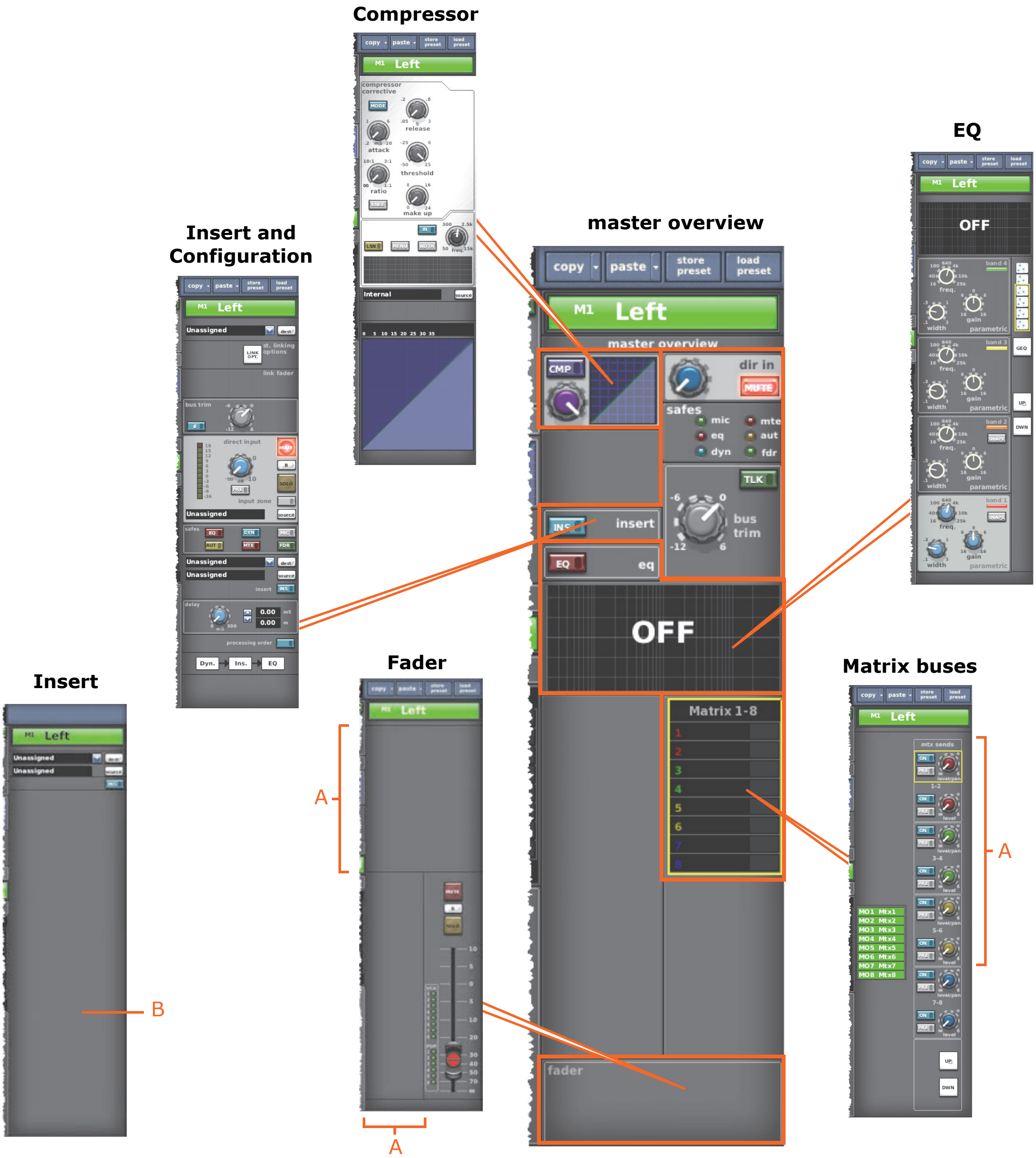


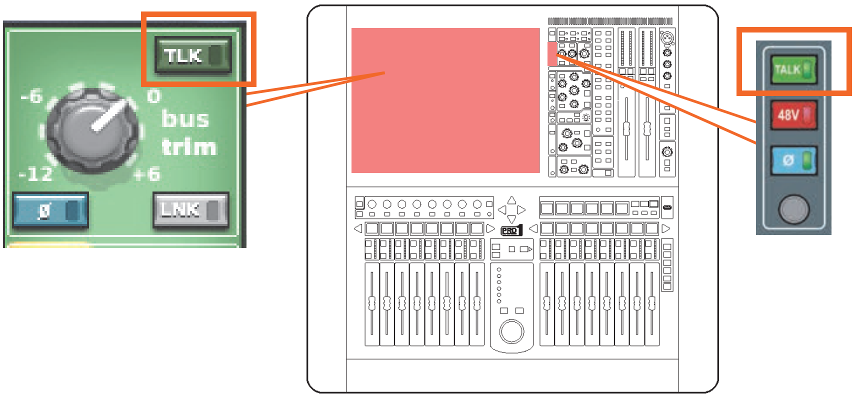
*Figure 27: Processing areas available from the **master overview** display.*
A. *The display in this area depends on surround configuration.*
B. *The insert processing area is selected via the quick access button in the insert processing area or by clicking in the insert section of a channel strip on the GUI.*

Output metering

All of the outputs are shown in on the **Console Overview** screen of the GUI, each with its own meter. For more information, see "Input metering" on page 251.

Talk

There is a talk switch in the channel processing area on the control surface and also on the output 'overview' displays of the GUI channel strip.

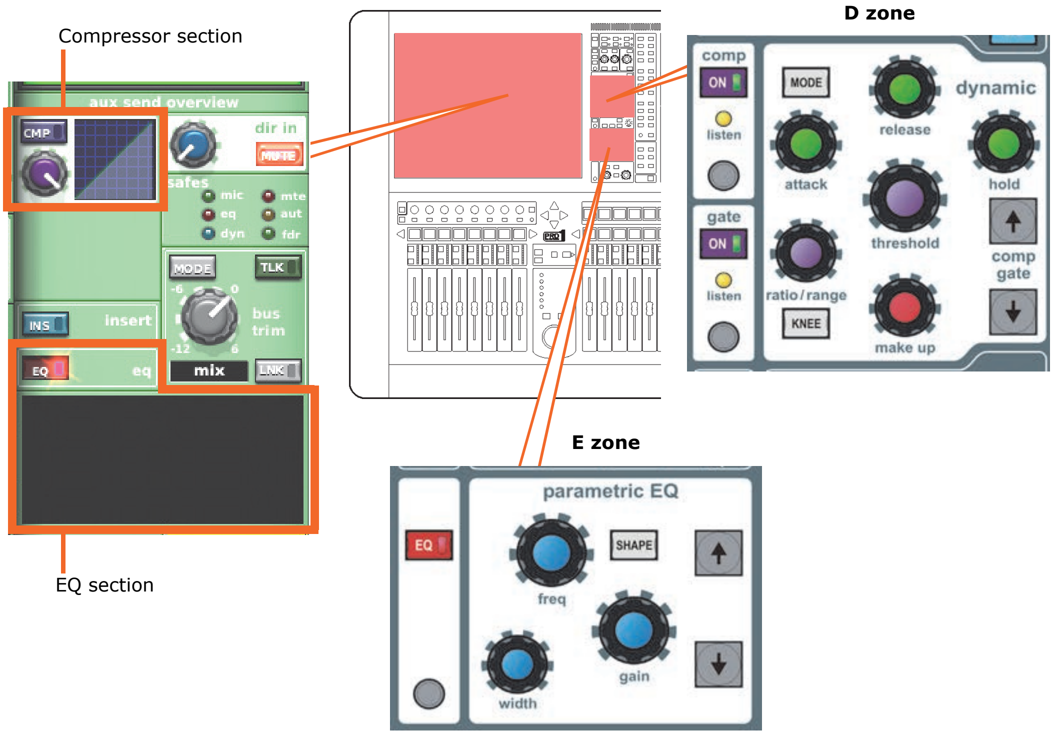


If the **TALK**/[**TLK**] **(internal)** switch in the **talk mic** section is active, the talk buttons will illuminate to prompt the operator to select a bus that the talk signals should be routed to. These are also used to set up a talk group after pressing one of the **talk/osc routing** panel buttons.

Dynamics and EQ

The control surface has a combined dynamics and EQ section that contains **DYN** and **EQ** on/off buttons, and a **listen** LED (yellow) 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 strip overview display the aux, matrix and master outputs each have a compressor section and an EQ section (both are highlighted in the diagram below). Clicking within either of these sections will open their respective processing areas, which are described in the following subsections.



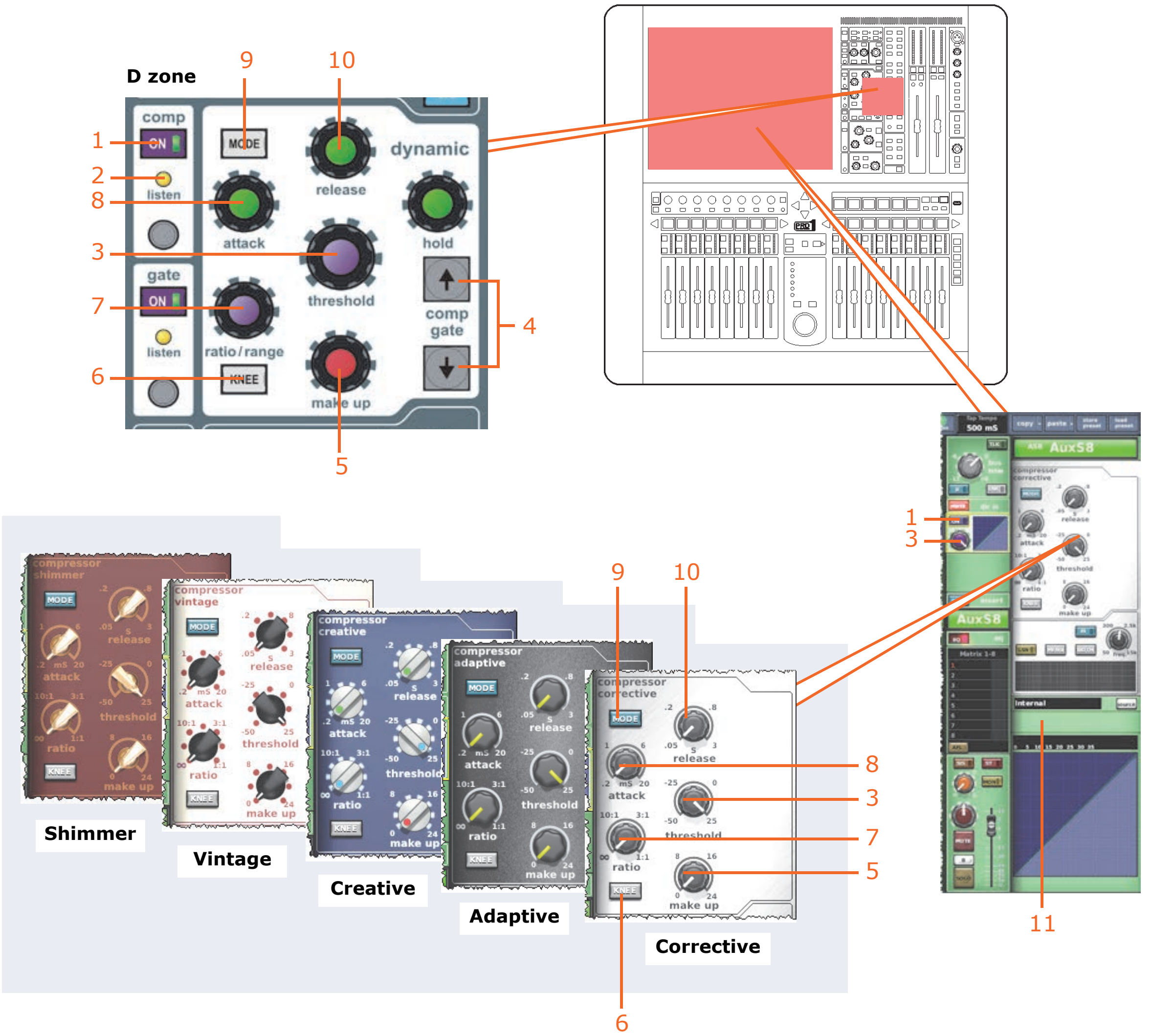
Compressor (D zone)

For the outputs, the **dynamic** section (D zone) only has a compressor in the output channel signal path. As the D zone is also used for the gate on the input channels, some controls may be redundant.

The output channel compressor has five styles — corrective, adaptive, creative, vintage and shimmer — which are selectable via the **MODE** button. Each has a distinctive appearance in the GUI channel strip. 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 "About the compressor graph" on page 262.

The side chain is similar to the one used for the input channels, see "Side chain" on page 265.



Item	Description
1	**ON** switch, enables the compressor in the signal path. When switched off, compressor is bypassed.
2	To aid set up, the compressor has a side chain listen that sends the side chain onto a solo bus. This side chain **listen** LED indicator illuminates to warn you that soloed material is from the side chain, and not the main channel.
3	**threshold** control knob, sets the signal level above which gain reduction starts to be applied. Range is from -50dBu to +25dBu.

Item	Description
4	**comp**/**gate** up and down select buttons, for swapping **dynamic** section control from compressor to gate, and vice versa.
5	Compressor **make up** gain control knob, compensates for the reduced *loudness* of a compressed signal. Range is from 0dB to 24dB.
6	Compressor **KNEE** switch, controls how compressor starts to apply attenuation as the signal goes through the threshold (see "About the compressor graph" on page 262). For more information, see "Knee" on page 300.
7	Compressor **ratio/range**/[**ratio**] control knob, adjusts amount of compression applied to signals above threshold. Range is from infinity (∞) to 1:1. When set to infinity it sets the compressor to limiter mode.
8	Compressor **attack** control knob, adjusts time for compressor to respond after an over-threshold signal. Range is from 0.2ms to 20ms (milliseconds).
9	**MODE** switch, selects compressor mode from the five compressor types available (see "PRO1 compressor modes (dynamic)" on page 299).
10	Compressor **release** control knob, adjusts time for compressor to recover after programme material falls back below threshold. Range is from 0.05s to 3.00s (seconds).
11	Compressor 'gain reduction' meter (not displayed).

EQ (E zone)

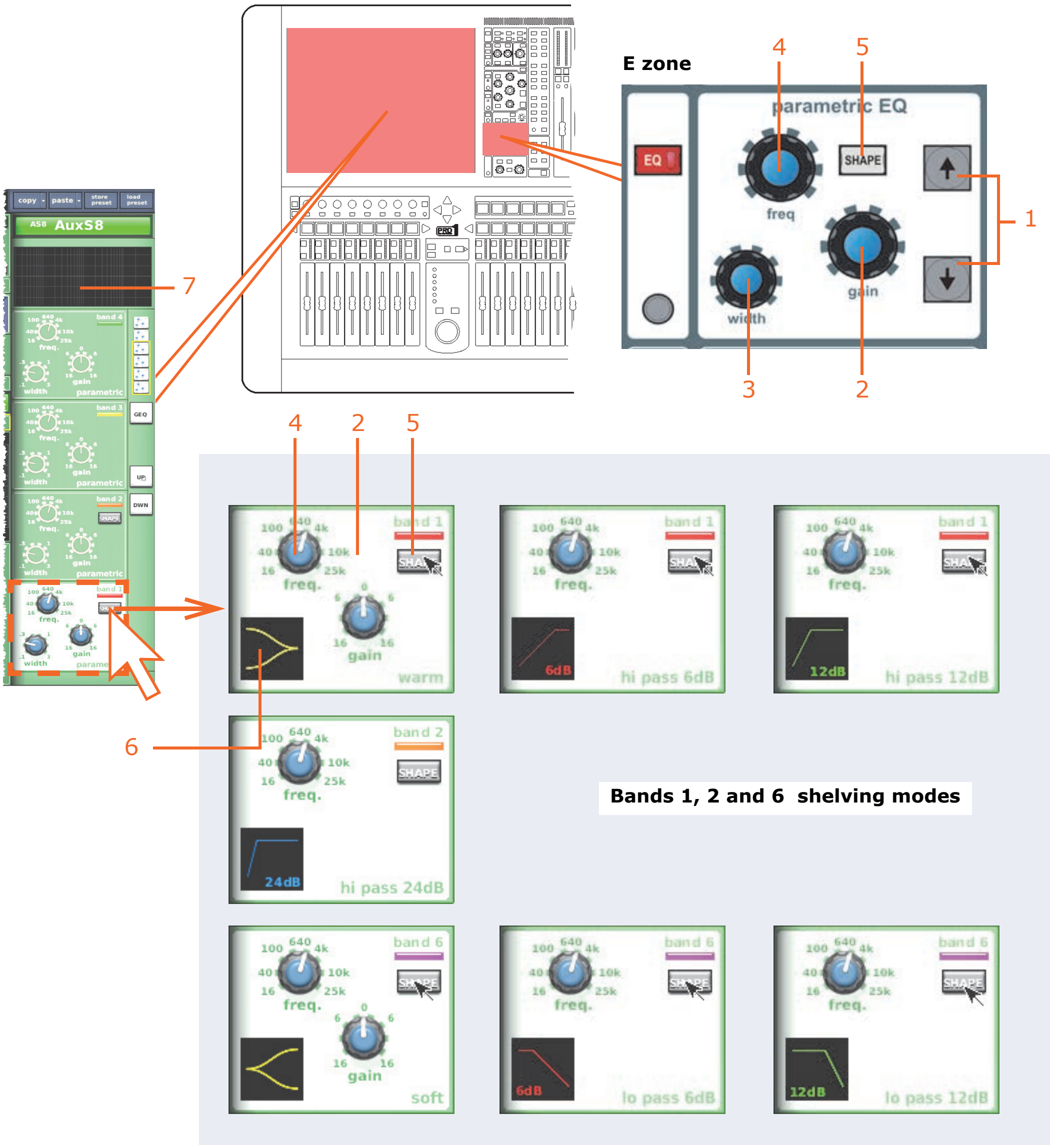
For tonal control of the aux, matrix and master output signals, the output channel EQ has the option of a six-band sweep parametric EQ (PEQ) or an assignable graphic EQ (GEQ).

PEQ

The parametric EQ section (E zone) of the channel strip (mix and master bays) allows tonal control of the input signal. The E zone contains all of the PEQ controls, along with a shelving mode selection button and another set of band selection buttons.

All of the outputs have six-band PEQs. Two of the six bands have three shelving modes each, while another has just one. Any combination of the six bands can be used to control the signal, although only one band can be adjusted in the E zone at any time. However, the EQ processing area (GUI channel strip) displays four bands at any time, and also has navigational controls.

Note: *While 24dB in band 2 is selected, band 1 is nullified.*



Item	***Description***
1	Up/down band navigation buttons (see "Navigating the PEQ output bands" on page 287). Also used in conjunction with the blue adjacent LEDs to show which band is currently selected. Illuminated up arrow means that band 5 is selected, and illuminated down arrow means that band 2 is selected.
2	**gain** control knob, adjusts signal gain in the range -16dB to +16dB. On the graph in the EQ processing area (GUI channel strip), causes the envelope to move up/down, inverting as it passes the origin.
4	**width** control knob, adjusts the signal bandwidth in the range 0.1 Oct to 3.0 Oct. On the graph in the EQ processing area (GUI channel strip), causes the base of the envelope to widen. (Not available for shelving modes.)
5	**freq** control knob, adjusts signal frequency. The range is band-dependent. On the graph in the EQ processing area (GUI channel strip), causes the envelope to move left/right.

Item	Description
6	**SHAPE** button, changes the shelving mode on treble and bass bands. For a description of each mode, see "PRO1 output channel EQ modes" on page 303.
7	Shelving symbol.
8	Graph of EQ envelope (see "EQ graph" on page 287). When "OFF" is displayed, EQ is switched off.

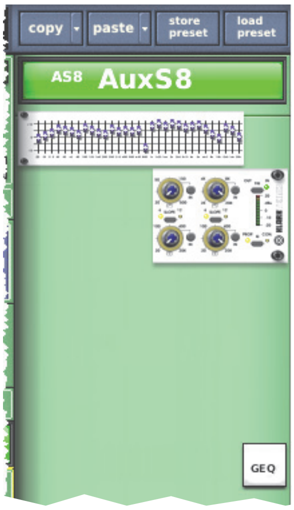
GEQ

You can access the output GEQ (shown right) from the EQ processing area of the GUI channel strip.

The GEQ is similar to the ones found in the **Graphic EQs** screen and can also be operated using the assignable controls. For information, see Chapter 15 "Graphic Equaliser (GEQ)" on page 119.

>> To open the GEQ window

1. Select the output. Its 'overview' display will appear in the channel strip.
2. In the GUI channel strip, open the EQ processing area (see "To assign a processing area to the output channel strip via the GUI" on page 44).
3. Open the GEQ display (shown right) by clicking **GEQ** in the processing area.
4. Open the GEQ screen (shown above right) by clicking on a non-control area of the GEQ image in the processing area.

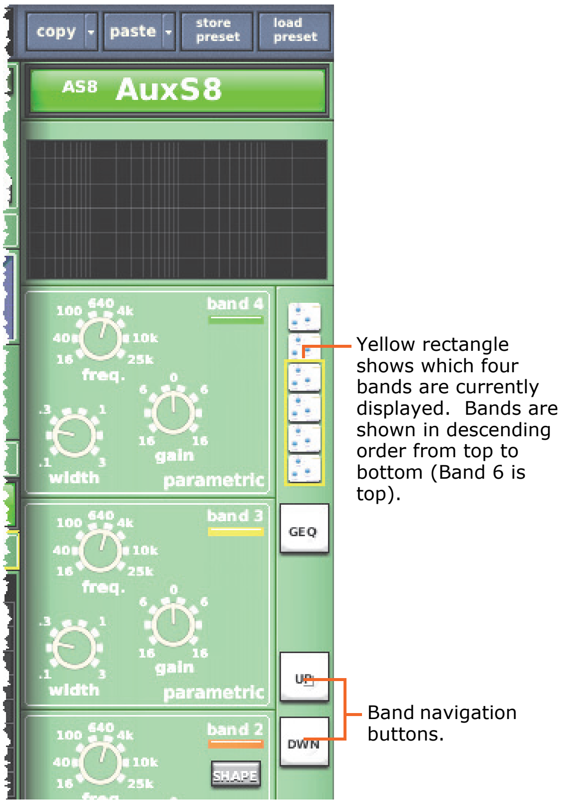


>> To close the GEQ window

1. In the GEQ screen (shown above), do one of the following:
   - Click **OK.**
   - Click X at the upper-right corner of the EQ window.
2. In the EQ processing area, click **GEQ** to open the outputs overview display.

Navigating the PEQ output bands

You can change band selection by clicking the **UP**/**DOWN** buttons in the EQ processing area. This will change selection by one band at a time.

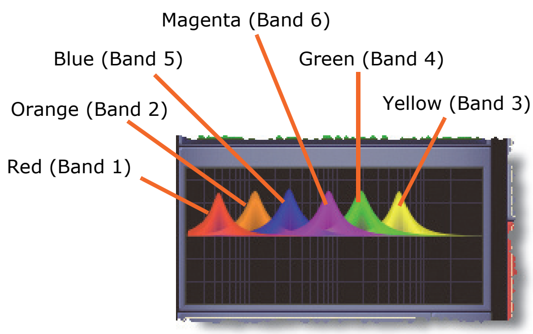


EQ graph

The controls in the output EQ sections, that is, the EQ **gain**, **freq** and **width** control knobs, have a similar functionality to the ones in the input EQ sections. For details, see "EQ graph" on page 287.

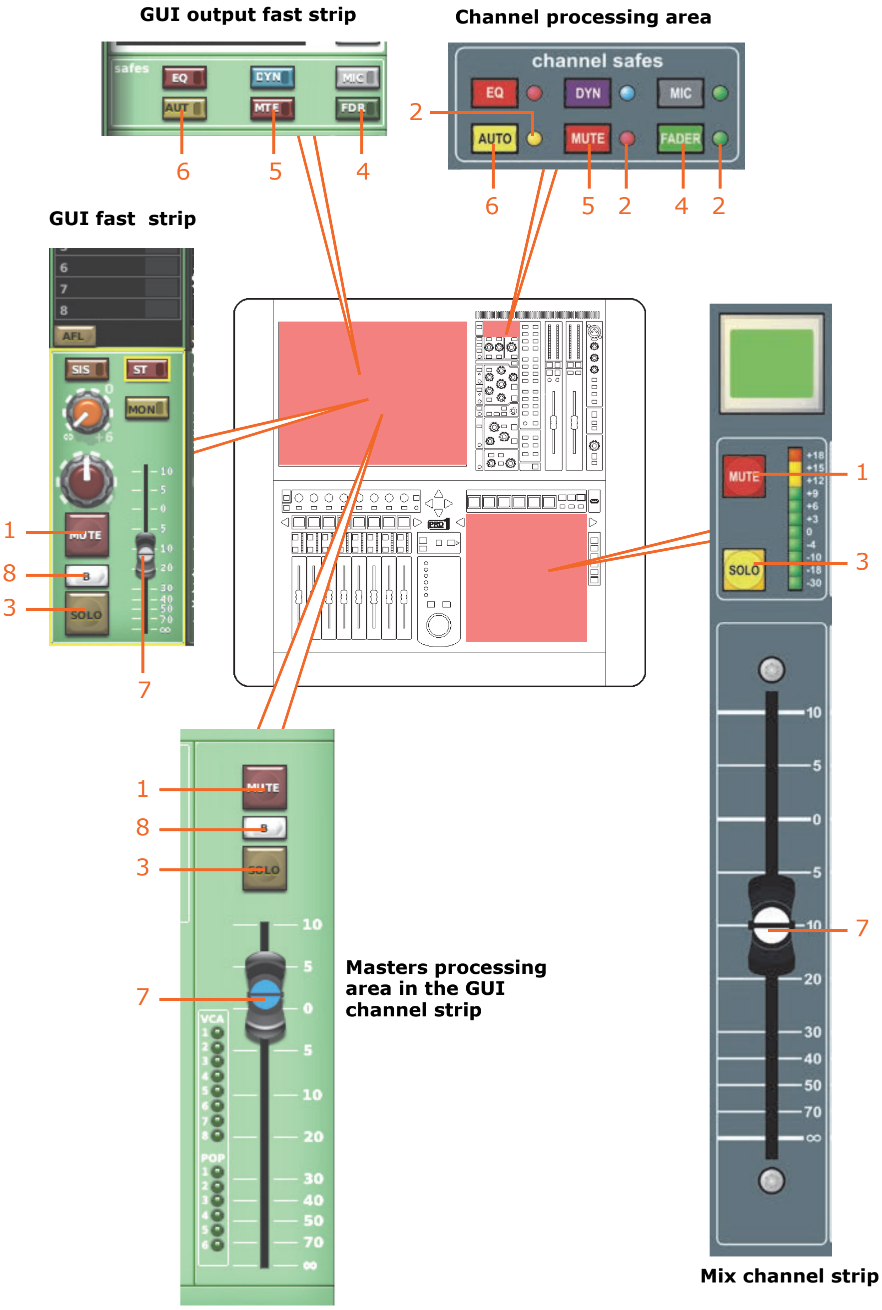
What the graph colours represent

Any combination of EQ envelopes for the four bands can be displayed, and each one is represented by a different colour (as shown right).



Mute, safes, level and solo

Each output fast strip (control surface and GUI) has controls for muting, soloing, safes and output signal level control. This is supported on the GUI in the appropriate processing area. They also each have a **SELECT** button for navigation. In addition, the channel strips (control surface and GUI) have the full complement of safes.



Item	Description
1	**MUTE** switch, mutes all post-processing signals leaving the channel. (In addition to scene recall, muting can be remote from the auto-mute masters.)
2	Safe LEDs, illuminate when their associated safe is on.
3	**SOLO** switch, activates signal routing to the monitor A section of the control centre.

Item	Description
4	**FADER**/[**FDR**] switch, switches fader safe on so that the fader is removed from scene recall.
5	**MUTE**/[**MTE**] switch, switches mute safe on so that mute is removed from the scene recall and auto-mute action.
6	**AUTO**/[**AUT**] switch, switches auto safe on so that the channel is removed from scene recall (this does not affect the action of the auto-mutes and VCA control groups) and control is removed from VCA control group faders.
7	Fader for adjusting output signal level. Has the same function as the **level** control knob (see item 3).
8	Solo **B** switch (GUI only), changes the operation of the **SOLO** switch so that it routes signals to the Monitor B section of the control centre.

Output channel configuration controls

There are a number of output channel controls that are loosely termed 'channel configuration' controls. The following table shows the configuration controls available on each output and references the pertinent section within this chapter.

Table 13: Output channel configuration controls

Channel controls	*Aux*	*Matrix*	*Master*	*Refer to*
Output channel ID	Yes	Yes	Yes	Page 290
Output channel source/destination	Destination	Destination	Destination	Page 290
Stereo linking	Yes	Yes	Yes	Page 290
Mix	Yes	N/A	N/A	Page 290
Link fader	N/A	Yes	N/A	Page 290
Bus trim	Yes	Yes	Yes	Page 290
Direct input	Yes	Yes	Yes	Page 291
Safes (EQ, dynamics, mic, auto, mute and fader)	All six	All six	All six	Page 291
Insert	Yes	Yes	Yes	Page 291
Delay	Yes	Yes	Yes	Page 291
Processing order	Yes	Yes	Yes	Page 291

For routing information, see Chapter 8 "Patching" on page 45.

Output channel ID (GUI only)

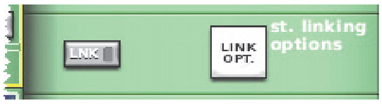
You can change the channel name that appears in the GUI channel strip (see “Text editing” on page 42). This can be done in the **output channel overview** or in any of the processing areas.

You can also change the background colour of the output channel name field, which is done in the **home** ▸ ***Mix & Outputs*** ▸ ***Naming Sheet*** screen of the GUI menu (see “Configuring VCA/POPulation groups” on page 73).

Output channel source/destination (GUI only)

The channel’s destination is shown in the text field of the configuration processing area. If no destination has been selected, it will contain the text “Unassigned” (as shown right). You can select the destination for this channel by clicking **dest**, which opens the **Patching** screen (see Chapter 8 "Patching" on page 45).

Stereo linking (GUI only)



The linking section of the configuration processing area has a **LINK** switch for linking the selected output channel to the adjacent (higher numbered) output channel. The **LINK OPT.** button opens a **Stereo Linking Options** window from where you can select which parameters you want to link. For more information, see Chapter 10 "Stereo Linking" on page 93.

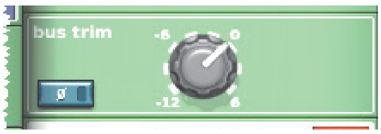
Mix mode (GUI only)

The **mix** section (aux only) has a **MODE** button for scrolling through the three mix modes, that is, mix, mix minus and group.

Fader linking (GUI only)

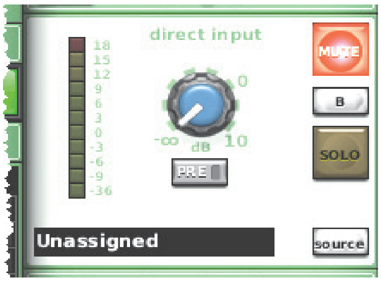
The **link fader** section (matrix only), has an **ST** and a **MON** button for linking the matrix channel fader to the stereo or mono master faders, respectively. Control of the stereo master faders reverts to the highest fader.

Bus trim (GUI only)



The **bus trim** section has a control knob for fine adjustment of the gain, in the range -12dB to +6dB and a phase reverse button.

Direct input



The **direct input** section provides an internal connection to effects etc. or an external input into the output from an effect or line I/O unit. It allows you to take a signal directly out of a defined point in the input channel's signal path and route it to either an internal assignable effect or to one of the physical outputs (a physical connection at one of the line I/O boxes). This function is optional and assigned on a channel-by-channel basis.

This section is deliberately distanced from main channel panel controls because it is a limited resource and unused on many channels.

Selection of signal path source can only be carried out via the GUI.

This section has similar functionality to the **direct output** section on each input channel, see "Direct input" on page 291.

Safes

Each output channel has six types of output channel safes that each protects a specific control/area from the automation system.

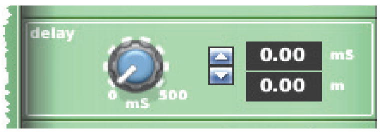
You can only operate the safe switches via the channel strips (control surface and GUI), which also provide on/off status information. The status of some of the safes is displayed via LEDs in the output fast strips and master strips on the control surface.

For more information on what areas are protected by each safe, see Appendix H "Parameters Protected By Safes" on page 397.

Insert (GUI only)

You can configure the send and return points of the aux, matrix and master outputs in the **insert** section of the configuration processing area. (The insert on/off button **INS** will only work if the insert destination and source points have been patched.)

Output channel delay (GUI only)



Similarly 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). For details, see "Input channel delay (GUI only)" on page 253.

Processing order

Similarly to the input channels, you can change the processing order on all of the output channels. For details, see "Processing order" on page 257.

Mixes

Each of the aux and master output channels can send a variable contribution to the mixes on each of the eight matrix buses. The buses are controlled in pairs via mix controls that give continuous adjustment (in the range +6dB to off) of sub group levels sent to matrix mixes. The controls in the **mix** section (mix and master bays) include **level**/**pan** and **level** control knobs for each bus pair, whose function (auxes only) depends on the current bus mode in operation.

The mixes on the outputs are similar in functionality to inputs, except there is no after fader listen. For details, see “Mixes” on page 269.

Masters

Towards the bottom of each input fast strip are the **masters** section and pan control, LCD select button, mute and solo, and the input fader.

For each output, the **masters** section (mix and master bays) functions in the same way as for the inputs. For more information, see “Master controls” on page 271.

Chapter 32: GUI Menu

The GUI is a very powerful multi-functional tool that forms the core of the PRO1 Control Centre. It gives you total control and monitoring of the operating environment, enhances control surface operation (you can even operate the PRO1 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:

- **Configuration** Configure the routing, associations and the names and colours of channels, groups, graphic EQs and internal effects, set up for multi-console operation, set up the connected devices etc.
- **Navigation** Select the channels, buses and groups you want, go quickly to a GUI screen display, go to recently opened screens, move through the scenes in a show and go to the patching screens you want.
- **Management** Manage show files (internal and external), automation and the monitoring system.
- **User and operating preferences** Adjust GUI screen brightness and contrast, select delay compensation, select fader flip etc.
- **Information** View current software information.
- **Security** Lock the screens to prevent unauthorised access.
- **Shutdown sequence** Shut down the control centre properly.
- **Upgrading the software** Install the very latest version (or any previous version) of PRO1 software.

For details of how to access the GUI menu, see "Using the GUI menu" on page 42.

GUI menu flowchart

The GUI menu and all of its available submenus are shown in Figure 28 "GUI menu flowchart" on page 294.

Icons to the left of the options help to identify the option type and aid navigation. A black triangle to the right of a menu option shows that it has a submenu.

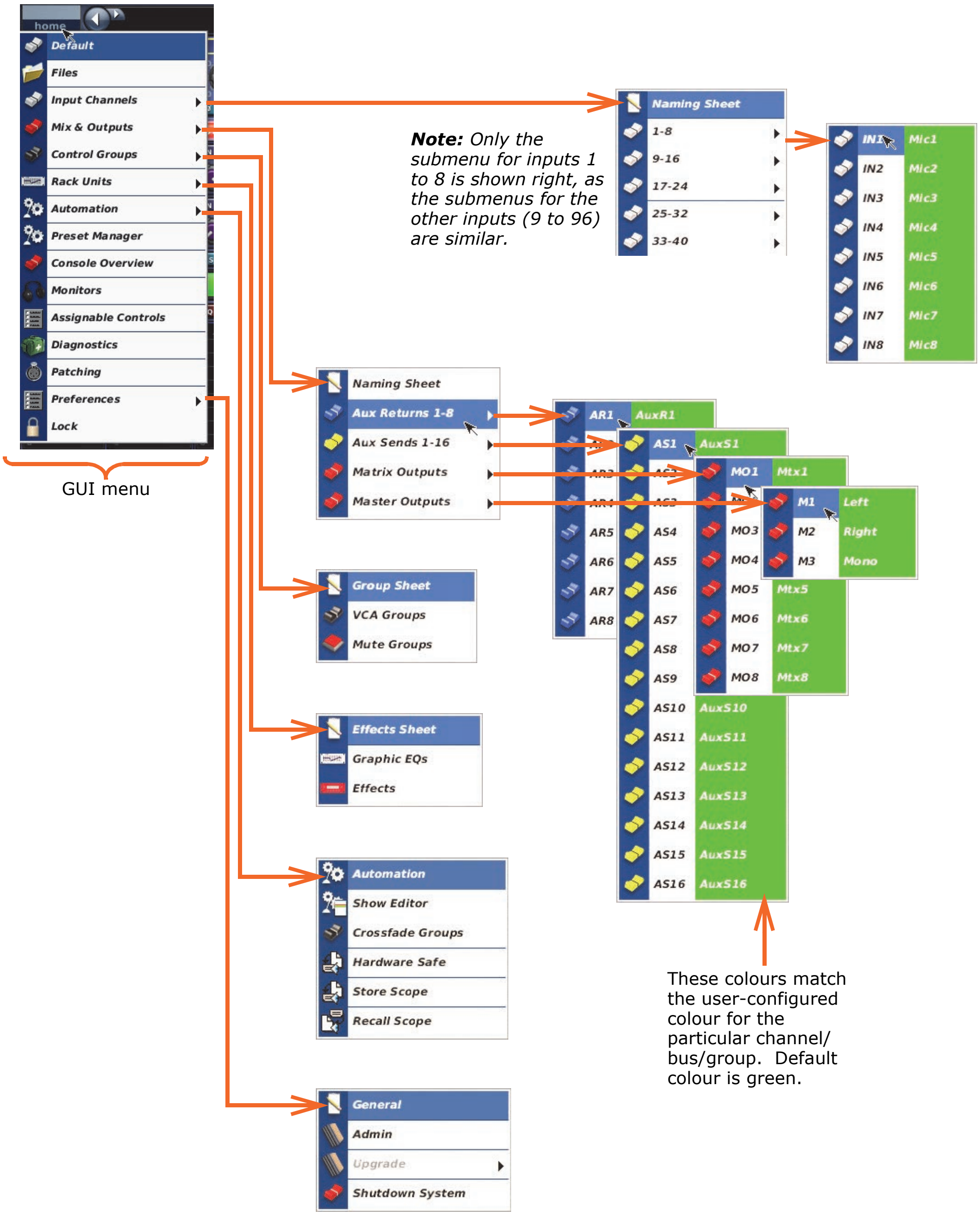


Figure 28: GUI menu flowchart

GUI menu options

When you open the GUI menu, you are presented with a main list of options that open specific screens or submenus, as shown in the following table. You can access some of the screens directly from the **screen access** buttons in the navigation zone (see "PRO1 control surface" on page 18).

Option	***Description/function***
Default	Opens the default display.
Files	Opens the **Files** screen (see "Managing show files on the Files screen" on page 189).
Input Channels	Input channel option, which opens a submenu with the following options: • ***Naming Sheet*** — lets you configure the 56 input channels (see "Configuring the channels, groups and internal units" on page 228). • ***1-8*** through to ***49-56*** — click one to open the associated bank of inputs, or open the submenu. Each submenu contains the eight single inputs belonging to its bank; click on one to select its channel.
Mix & Outputs	Output channel option, which opens a submenu with the following options: • ***Naming Sheet*** — lets you configure the output channels (see "Configuring the channels, groups and internal units" on page 228). • Output channel options — click one to open the associated bank of outputs (returns, auxes, matrices or masters), or open the submenu.
Control Groups	Control groups (VCA/POPulation, auto-mute and talk) option, which opens a submenu with the following options: • ***Group Sheet*** — lets you configure each group (see "Configuring the channels, groups and internal units" on page 228). • ***VCA Groups*** — click to open the VCA Groups screen, or open the submenu, which contains an option each for the 12 VCA groups and eight POPulation groups; click on one to select its group. See "VCA and POPulation groups" on page 159. • ***Mute Groups*** — click to open the **Mute Groups** screen (see "Auto-mute (mute) groups" on page 163).
Rack Units	Control groups (VCA/POPulation, auto-mute and talk) option, which opens a submenu with the following options: • ***Effects Sheet*** — lets you configure each of the 16 GEQ and 16 internal effect 'virtual' rack units (see "Configuring the channels, groups and internal units" on page 228). • ***Graphic EQs*** — opens the **Graphic EQs** screen (see "About the Graphic EQs screen" on page 119). • ***Effects*** — opens the **Effects** screen (see "Overview of the internal effects" on page 125).

Option	Description/function
Automation	Automation option, which opens a submenu with the following options: • ***Automation*** — opens the **Automation** screen (see Chapter 20 "Scenes And Shows (Automation)" on page 177). • ***Show Editor*** — opens the **Show Editor** screen (see "Show editor" on page 83). • ***Crossfade Groups*** — opens the **Crossfade Groups** screen (see "Crossfade groups" on page 211). • ***Hardware Safe*** — opens the **Hardware Safe** screen (see "Safes" on page 190). • ***Store Scope*** — opens the **Store Scope** screen (see "Using store scope" on page 199). • ***Recall Scope*** — opens the **Recall Scope** screen (see "About the Recall Scope screen" on page 193).
Preset Manager	Opens the **Preset Manager** screen (see Chapter 24 "User Libraries (Presets)" on page 215).
Console Overview	Opens the **Console Overview** screen (see "Input metering" on page 251).
Monitors	Opens the **Monitors** screen (see Chapter 14 "Monitors And Communications" on page 107).
Assignable Controls	Opens the **Assignable Controls** window (see Chapter 19 "Assignable Controls" on page 169).
Diagnostics	Opens the **Diagnostics** screen (see Appendix D "Troubleshooting" on page 329).
Patching	Opens the **Patching** screen (see Chapter 8 "Patching" on page 45).
Preferences	Preferences option, which opens a submenu with the following options: • ***General*** — opens the **Preferences** screen (see Chapter 27 "Changing The Preferences" on page 223). • ***Admin*** — opens the administrator's window. This is a supervisor-only function, which is accessed by typing in a password. • ***Upgrade*** — opens a list of TAR files from which to choose when updating the PRO1's software (see "Updating your system" on page 344). • ***Shutdown System*** — opens the shutdown message window, which initiates an expedient shutdown of the PRO1 (see "To switch off the PRO1 Control Centre" on page 31).
Lock	Locks the GUI (see "Security (locking mode)" on page 89).

Appendices

Appendix A: Application Notes

This chapter provides more in-depth information on certain areas and function of the PRO1.

Spatial imaging system (SIS™)

Although conventional consoles can be used for three-channel mixing, the methods for doing so are complicated and unorthodox. This forces the engineer to work in unaccustomed ways, limiting creative flexibility, and making use by visiting operators impractical. The spatial imaging system (SIS™) has been developed to overcome this.

Backing vocalists can be panned slightly towards the centre cluster to improve intelligibility, while keeping the featured vocal 'front and centre'.

Musical instruments can be placed in a conventional mix then easily switched to the centre for solos.

In theatrical productions, SIS™ allows you to pan an actor's voice across three channels following their on stage movements. In stereo-only productions, the centre output can be used to provide a mono-to-mix-base feed activating a single 'left + right to centre' switch.

The ability of SIS™ to feed centre-panned signals equally to both left and right outputs, as well as the centre, is particularly useful for distributing the load of high energy, centre-panned sounds across all FOH loudspeaker arrays.

PRO1 compressor modes (dynamic)

This section aims to provide an understanding of the compressor modes contained within the PRO1 Control Centre.

Description

The PRO1 compressors have five primary operating modes (only four on inputs). These change the signature (or shape) of the attack and release envelope curves, interactions and timings. Before dealing with this in detail, some of the generic terms are defined and explained:

Threshold

The threshold adjusts the operating point of the compressor. Signals that go over this point, or *over-threshold*, will be affected by compressor action. Signals that stay below threshold will not trigger any compression although they may still be affected by compression releases from previous over threshold signals.

Ratio

The compression ratio control provides control of the amount of compression that is applied to over-threshold signals. This is expressed as a ratio of signal level changes from input to output. For example, when the compressor is set to 2:1, every 2dB input level change will only generate a 1dB output level change (assuming the signal levels are over threshold).

Attack

The attack control adjusts the time taken for the compressor to respond to an over-threshold signal. The shape of the attack can be selected from one of the five mode combinations mentioned above, making the compressor easily adaptable for a wide number of creative and corrective applications.

Release

The release control adjusts the time the compressor takes to recover after the programme material falls back below threshold. Both attack and release also respond to changes in programme level that remain over-threshold. For example, a signal that reduces in level but remains above threshold will still trigger a release, but in this case it will only be a partial release - because the compressor will still be required to generate gain reduction, but now, as appropriate for the new lower signal level.

Knee

Most compression sounds more natural in soft knee mode. Soft knee compression blurs the distinction between over-threshold and under-threshold signals, such that signals that are a long way below threshold remain unaffected by compression, and signals that near the threshold get compressed, but at greatly reduced ratios. When signals are just over-threshold the compressor ratios are still somewhat reduced; it is only when signals go well over threshold that the full ratio compression is applied. When using a harder knee setting the compressors operate in a more clinical way with a more defined transition between under-threshold and over-threshold; this is better suited to limiting style compression.

Gain

The gain control provides adjustment of the *make up* gain so that the level of the outgoing compressed signal can be matched to the incoming uncompressed signal.

Side chain filter

A band pass filter is provided that acts on the side chain signals. This can be used to make the compression frequency selective. The controls for this are frequency, adjustable from 50Hz to 15KHz, and bandwidth selectable as wide, medium or narrow. Additionally, there is a listen function that places the filtered side chain onto the solo bus and a side chain filter in to activate or eliminate the filter action.

Compressor envelope modes

The five envelope *modes*, or *signatures*, are the key to the sonic character of the PRO1 compressors, and they allow adjustment far beyond the normal capabilities of simple attack and release settings. They largely fall into two application types:

1. Compressors that are good at capturing and controlling dynamic transients: corrective mode and vintage mode.
2. Compressors that emphasise dynamic transients and provide creative control of levels within a mix: adaptive, creative and shimmer modes.

The Vintage and Adaptive compressors tend to morph a little between these two categories depending on threshold control settings. This makes them easy to use intuitively with minimal fine-tuning of the envelope control settings.

Further refinement and enhancement of the envelope modes is provided by combination settings of the three-position **KNEE** switch. It is best to understand the operation of these two functions in more depth before looking at the detail of the compressor signature switching.

Knee

The soft knee curves behave in a traditional way to blend the compression ratio around the threshold setting (as described above), 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 (4dB); medium (12dB): and soft (40dB). 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.

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 and/or limiting of dynamic material. It can be used to add colour to low frequency signals making it ideal for controlling extremely dynamic instruments like the 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 minimise 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. The attack control is still active, allowing some user intervention although the adaptive nature makes envelope control setting fairly non-critical. The compressor is therefore very fast and simple to set up on most programme material. It is also sonically accurate and works well for both compression and limiting of vocals and many other sources. The most natural sounding compression is normally achieved with *soft knee* settings.

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 programme 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 emphasise transients giving more punch while retaining a good deal of artistic dynamic 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 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 *valve-like* brightness and sparkle to the programme, producing extremely natural and lively sounding compression of acoustic instruments.

Shimmer mode (overshoot peak - slow) - output only

This is a peak sensing compressor with an exponential release and unusual second order attack character that tends to *overshoot*.

If used sparingly, the compressor sounds very soft and natural and can provide additional control of material that already has a fairly low dynamic content. It can sound very transparent on vocals where it retains a good degree of life in the performance.

If used at higher ratios with slow attack and fast release times, the compressor can produce a very soft bouncy sound character.

PRO1 input channel EQ modes

This section aims to provide an understanding of the input channel EQ modes contained within the PRO1 Control Centre.

Basic specification

The PRO1 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.
- **Width:** continuous adjustment f bandwidth from 0.1 to 3.0 octaves (this only operates in parametric mode for Bass and Treble).
- **Treble:** continuous adjustment of the frequency range that the treble equaliser acts on from 1kHz to 25kHz.
- **Hi mid and lo mid:** hi mid frequency control gives continuous adjustment of the frequency range that the hi mid equaliser acts on from 320Hz to 8kHz. Lo mid frequency control gives continuous adjustment of the frequency range that the lo mid equaliser acts on from 80Hz to 2kHz.
- **Bass:** continuous adjustment of the frequency range that the bass equaliser acts on from 16Hz to 400Hz.

The treble EQ band can be switched from parametric to any of three other shelving modes: Soft; Classic; and Bright.

The bass EQ band can be switched from parametric to any of three other shelving modes: Warm; Classic; and Deep.

Description

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 so 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 mids are 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 there is no undershoot error corrupting the mids. This 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 pre mixed 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 EQ'd material. 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 does not corrupt the mids. This is best used on single source material.

PRO1 output channel EQ modes

This section aims to provide an understanding of the output channel EQ modes contained within the PRO1 Control Centre.

Basic specification

The PRO1 output EQ comprises six bands strategically positioned at certain frequencies ranging from the low end (bass) to the high (treble) of the frequency band. The default operation for all six sections is full parametric sweep (peak), with the following controls:

- **Gain:** continuous adjustment of boost and cut from + 16dB to - 16dB.
- **Width:** continuous adjustment of bandwidth from 0.1 to 3.0 octaves.
- **Frequency:** continuous adjustment of the frequency range that the band EQ acts on from 16Hz to 25kHz.

Band 1 can be switched from parametric to any of three shelving modes: warm; high pass filter 6dB; and high pass filter 12dB.

Band 2 can be switched from parametric to high pass filter 24dB.

Note: *When the 24dB/octave high pass filter is selected in band 2, band 1 becomes inaccessible.*

Band 6 can be switched from parametric to any of three shelving modes: soft; lo pass filter 6dB; and lo pass filter 12dB.

Description

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.

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 pre mixed 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.

The PRO1’s high pass filters have gain roll off before the corner frequency, which is variable.

Lo pass filter (LPF)

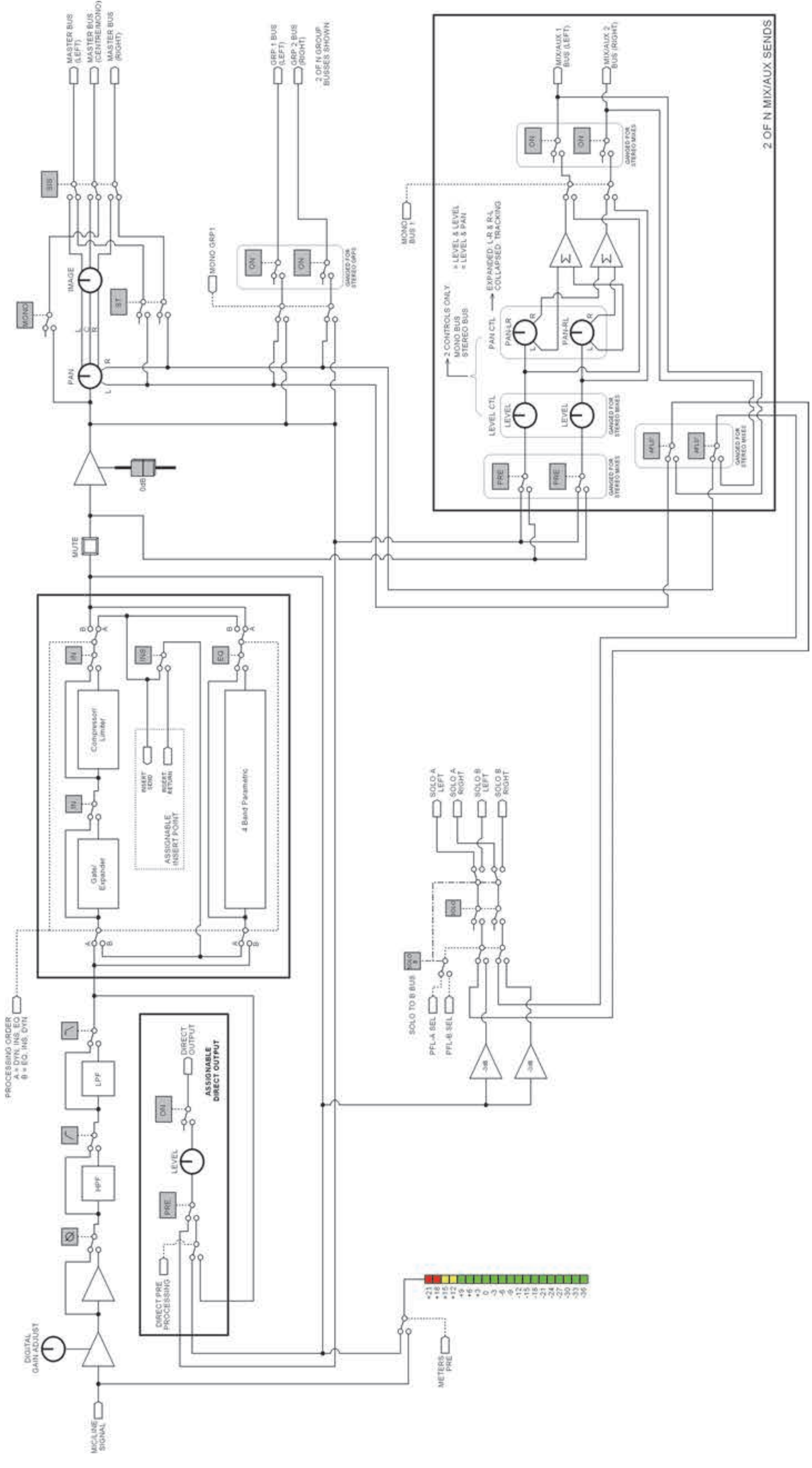
The LPF attenuates (not boosts) all frequencies above a certain level (cut-off frequency) while allowing all those below it to pass through. The harshness or smoothness with which the sound is removed beyond this point is determined by the dB/octave, with 6dB being the most common. The LPF is generally used to reduce noise in quiet passages or to take the *fizz* off any source with excessively high frequencies, but may also produce a sound effect by manipulation of the controls.

The PRO1’s low pass filters have gain roll off after the corner frequency, which is variable.

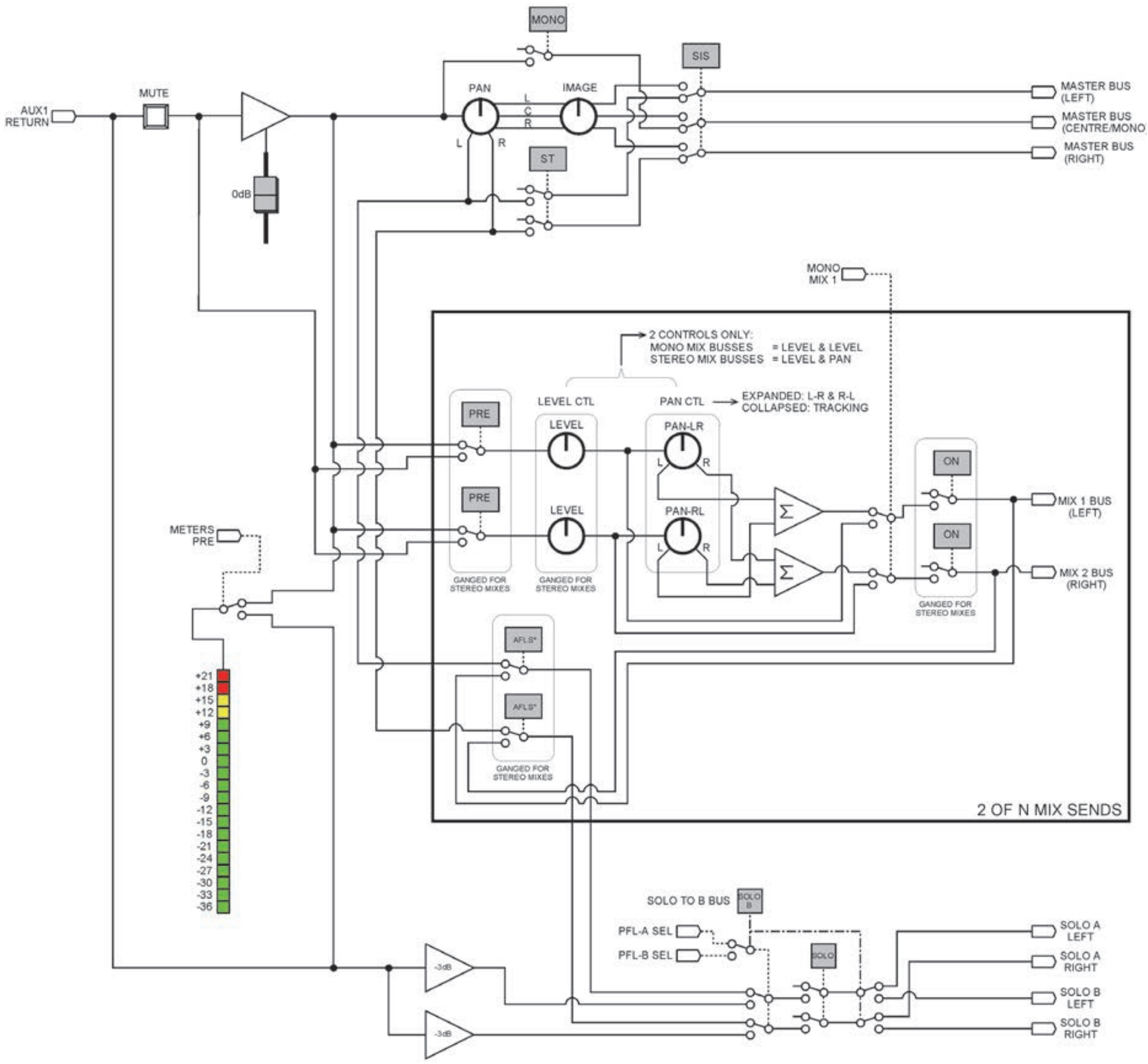
Appendix B: Functional Block Diagrams

This chapter contains the PRO1 signal path diagrams.

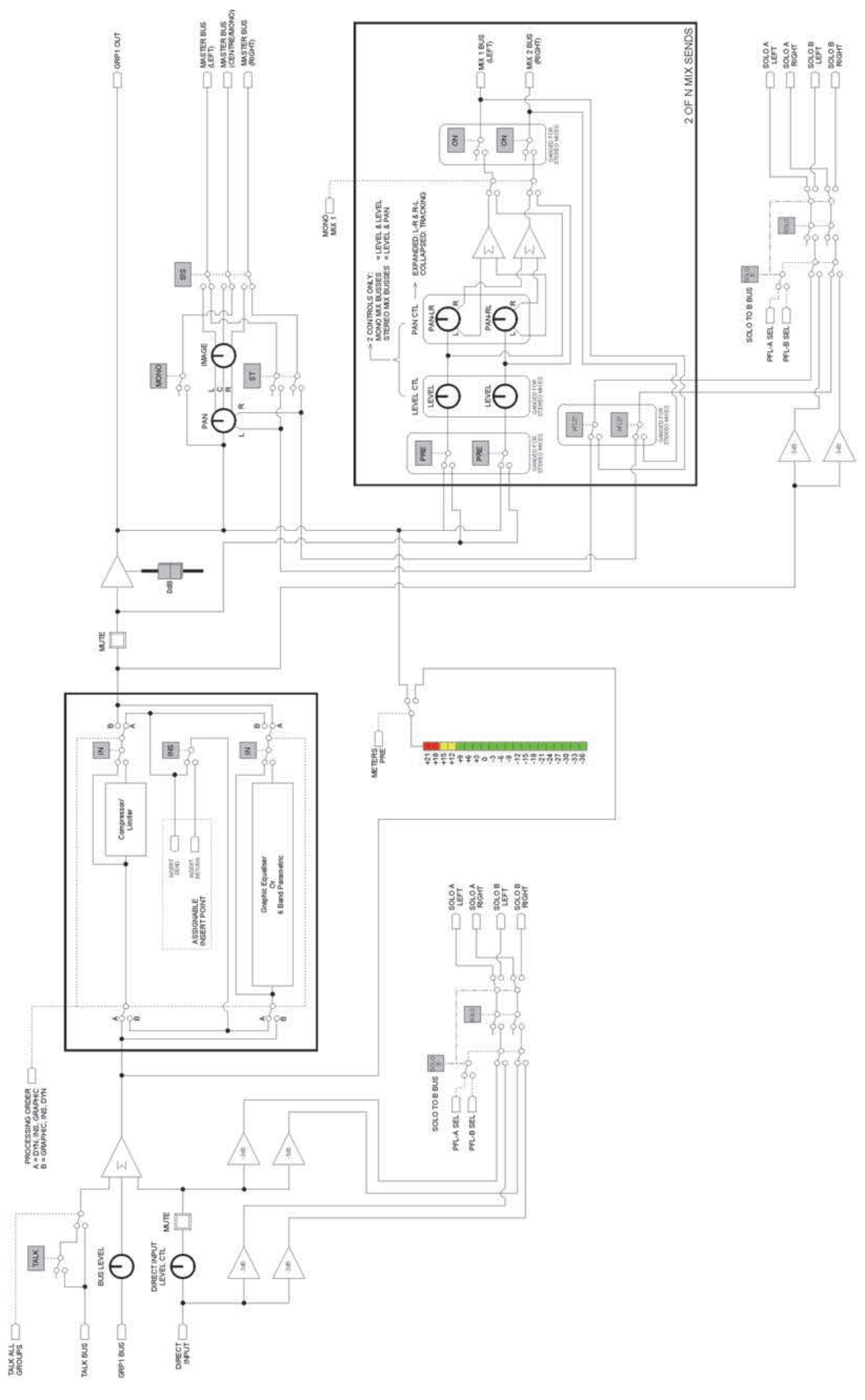
Mono input channel signal flow



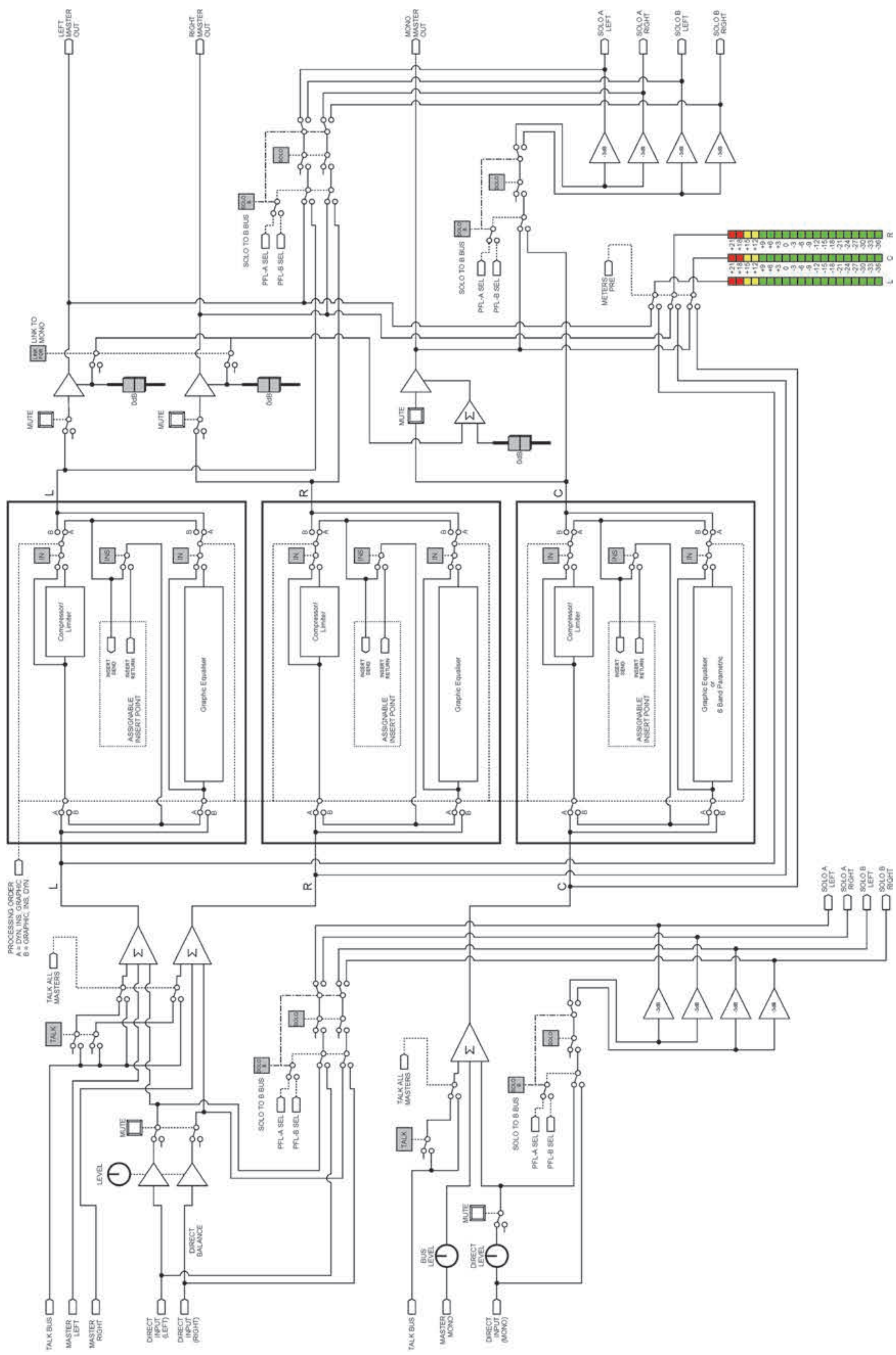
Returns signal flow



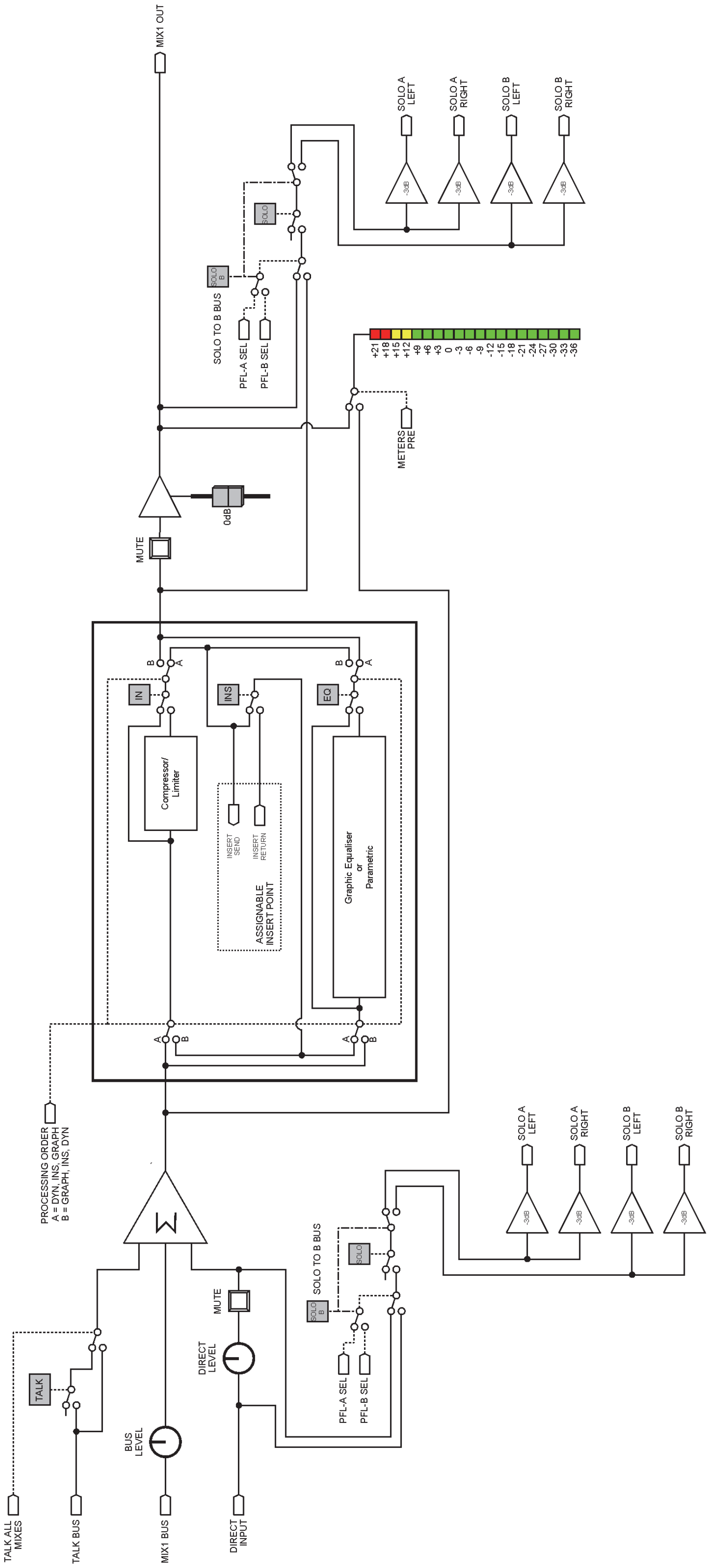
Aux/group signal flow



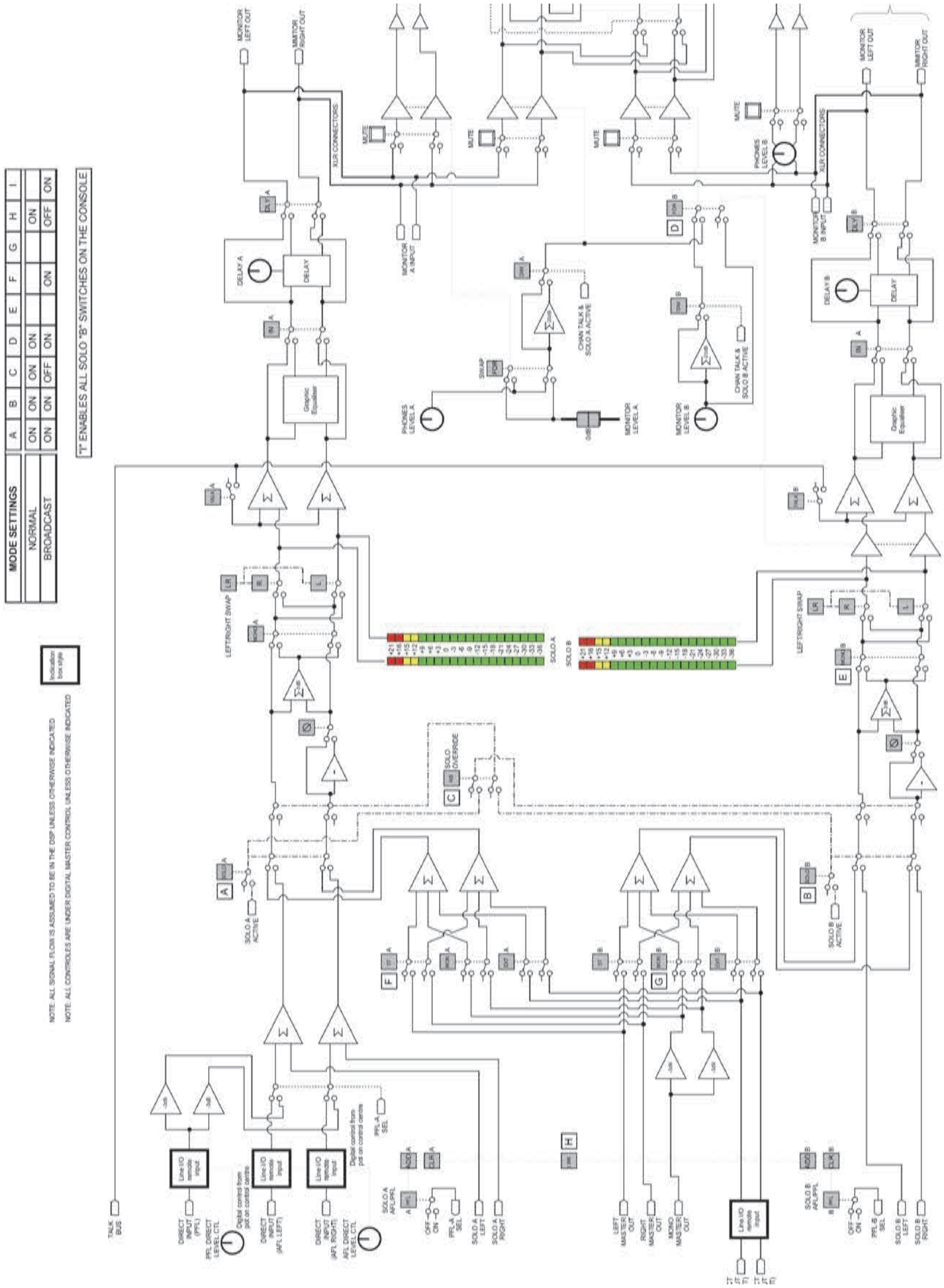
Master signal flow



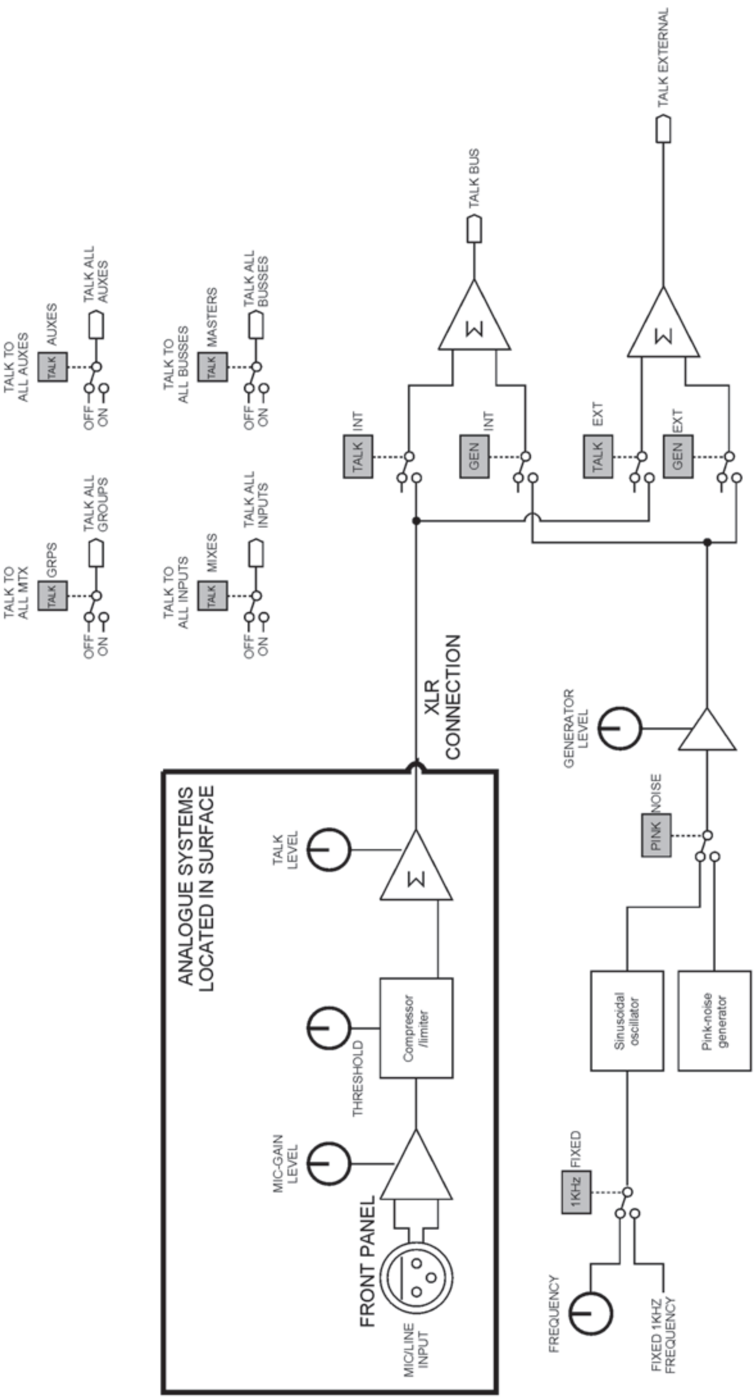
Mono mix signal



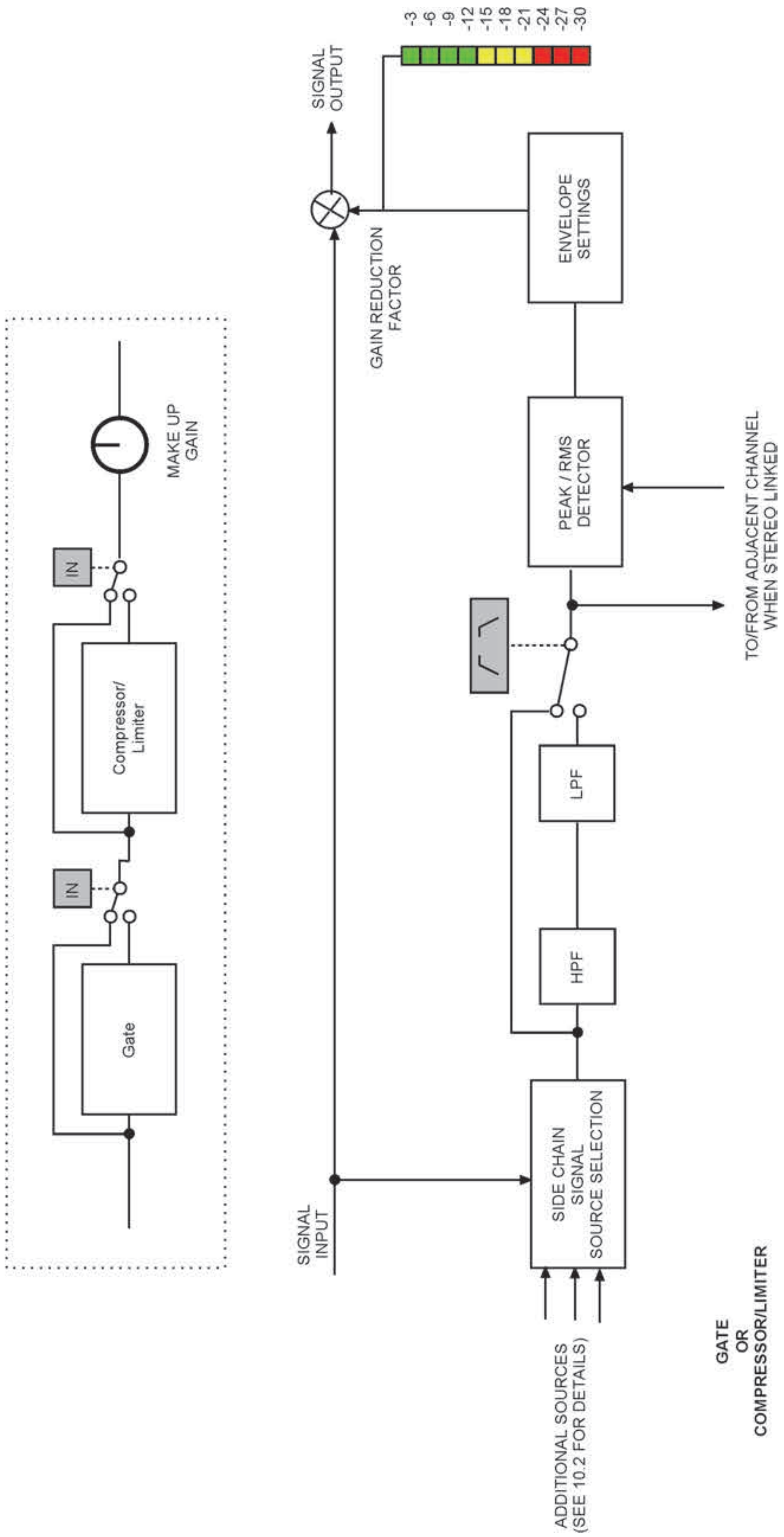
Monitor signal flow



Comms signal flow



Dynamics signal flow



Appendix C: Technical Specification

This appendix provides the full technical specification for the PRO1 Live Audio System, which includes the DL251 Audio System I/O.

PRO1 general statistics

XLR connections	1 x 5U rack I/O box houses: • 48 x (XLR) mic/line inputs • 16 x (XLR) line outputs Control surface houses: • 24 x (XLR) mic/line inputs • 16 x (XLR) line outputs • 4 x (XLR) monitor outputs • 3 x (XLR) master outputs • 2 x (XLR) AES3 stereo outputs • 2 x (XLR) AES3 stereo inputs • 1 x (XLR) AES3 sync output • 1 x (XLR) AES3 sync input • 1 x (XLR) talk out • 1 x (XLR) talkback • 1 x (XLR) talk mic
System expansion	Multiple 3U rack configurable I/O boxes (optional DL451 rack boxes) can be connected that house 24 x I/O slots in 8 x 8 wide blocks of: • 8 x (XLR) mic/line inputs or • 8 x (XLR) line outputs or • 8 x (Jack) line in and 8 x line out • 4 x (stereo) AES3 in and 4 x (stereo) AES3 out Multiple 6U rack splitters (optional DL431) house: • 24 x splitter mic/line inputs • 2 x 24 splitter outputs • 1 x 24 transformer isolated splitter outputs
Input audio processing	40 x dual slope high and low pass filters 40 x 4-band parametric EQs with 3 shelf modes 40 x 4-mode creative input compressors 40 x input gates
Mix/output audio processing	24 x output 6-band parametric EQs with shelf and multiple high and low pass modes 24 x 5-mode creative output dynamics 8 x assignable Klark Teknik output GEQs

Assignable audio processing	4 x assignable stereo effects (each of these can be reconfigured to generate 4 further GEQs, making a total of 24 available on the console)
Mixing control assistance	6 x auto-mutes 6 x surface POPulation groups 8 x VCA faders 8 x VCA groups 1000-scene snapshot automation
Resilience	N+1 cable redundancy

PRO1 general specifications

Sampling frequency	96kHz
Latency delay	<2ms input to master (no compensation)
Dynamic range	106dB, 22Hz to 22kHz (no pre-emphasis)
Maximum voltage gain	80dB inputs to subgroups and masters 86dB inputs to aux and matrix
Crosstalk at 1kHz	-100dB physically adjacent input channels
Crosstalk at 10kHz	-90dB physically adjacent input channels
Fader/pan cut off at 1kHz	-100dB
Fader/pan cut off at 10kHz	-100dB
Display screen	1 x 15” daylight-viewable colour screen
Dimensions	PRO1 control centre: 586 mm wide x 570 mm deep x 260 mm high (23.1” x 22.4” x 10.2”) DL251 I/O box: 5U x 200 mm deep
Net weight	PRO1 control centre: 21.5 kg DL251 I/O box: 10 kg
Shipping weight	Touring (in flight case): 97.50 kg Install (in carton): 28.70 kg
Power requirements	100V to 240V a.c. ±10%, 50 to 60Hz
Operating temperature range	+5°C to +45°C
Storage temperature range	-20°C to +60°C

PRO1 audio performance specifications

Frequency response

Input	*Output*	*Gain*	*20Hz*	*20kHz*
DL251 I/O box	DL251 I/O box	0dB	0dB to -1.0dB	0dB to -1.0dB
DL251 I/O box	DL251 I/O box	40dB	0dB to -1.0dB	0dB to -1.0dB
Surface I/O	Surface I/O	0dB	0dB to -1.0dB	0dB to -1.0dB
Surface I/O	Surface I/O	40dB	0dB to -1.0dB	0dB to -1.0dB
DL451 I/O box	DL451 I/O box	0dB	0dB to -1.0dB	0dB to -1.0dB
DL451 I/O box	DL451 I/O box	40dB	0dB to -1.0dB	0dB to -1.0dB
DL431 splitter	DL351 I/O box	0dB	0dB to -1.0dB	0dB to -1.0dB
DL431 splitter	DL351 I/O box	40dB	0dB to -1.0dB	0dB to -1.0dB
DL431 splitter	DL431 A out	0dB	0dB to -0.5dB	0dB to -0.5dB
DL431 splitter	DL431 A out	40dB	0dB to -0.5dB	0dB to -0.5dB
DL431 splitter	DL431 B out	0dB	0dB to -0.5dB	0dB to -0.5dB
DL431 splitter	DL431 B out	40dB	0dB to -0.5dB	0dB to -0.5dB
DL431 splitter	DL431 C out	-6dB	0dB to -1.0dB	0dB to -1.0dB

Gain error at 1kHz

Input	*Output*	*Gain*	*Maximum*	*Minimum*
DL251 I/O box	DL251 I/O box	0dB	+1.0dB	-1.0dB
DL251 I/O box	DL251 I/O box	40dB	+1.0dB	-1.0dB
Surface I/O	Surface I/O	0dB	+1.0dB	-1.0dB
Surface I/O	Surface I/O	40dB	+1.0dB	-1.0dB
DL451 I/O box	DL451 I/O box	0dB	+1.0dB	-1.0dB
DL451 I/O box	DL451 I/O box	40dB	+1.0dB	-1.0dB
DL431 splitter	DL451 I/O box	0dB	+1.0dB	-1.0dB
DL431 splitter	DL451 I/O box	40dB	+1.0dB	-1.0dB
DL431 splitter	DL431 A out	0dB	+0.5dB	-0.5dB
DL431 splitter	DL431 A out	40dB	+0.5dB	-0.5dB
DL431 splitter	DL431 B out	0dB	+0.5dB	-0.5dB
DL431 splitter	DL431 B out	40dB	+0.5dB	-0.5dB
DL431 splitter	DL431 C out	-6dB	+1.0dB	-1.0dB

Input CMRR

Input	*Output*	*Gain*	*100Hz*	*1kHz*
DL251 I/O box	DL251 I/O box	0dB	60dB	60dB
DL251 I/O box	DL251 I/O box	40dB	90dB	90dB
Surface I/O	Surface I/O	0dB	60dB	60dB
Surface I/O	Surface I/O	40dB	90dB	90dB
DL451 I/O box	DL451 I/O box	0dB	80dB	80dB
DL451 I/O box	DL451 I/O box	40dB	90dB	90dB
DL431 splitter	DL451 I/O box	0dB	80dB	80dB
DL431 splitter	DL451 I/O box	40dB	90dB	90dB
DL431 splitter	DL431 A out	0dB	80dB	80dB
DL431 splitter	DL431 A out	40dB	90dB	90dB
DL431 splitter	DL431 B out	0dB	80dB	80dB
DL431 splitter	DL431 B out	40dB	90dB	90dB
DL431 splitter	DL431 C out	-6dB	110dB	90dB

Distortion at 0dBu

Input	*Output*	*Gain*	*1kHz*	*10kHz*
DL251 I/O box	DL251 I/O box	0dB	0.01%	0.01%
DL251 I/O box	DL251 I/O box	40dB	0.03%	0.03%
Surface I/O	Surface I/O	0dB	0.01%	0.01%
Surface I/O	Surface I/O	40dB	0.03%	0.03%
DL451 I/O box	DL451 I/O box	0dB	0.01%	0.01%
DL451 I/O box	DL451 I/O box	40dB	0.03%	0.03%
DL431 splitter	DL431 A out	0dB	0.01%	0.01%
DL431 splitter	DL431 A out	40dB	0.03%	0.03%
DL431 splitter	DL431 B out	0dB	0.01%	0.01%
DL431 splitter	DL431 B out	40dB	0.03%	0.03%
DL431 splitter	DL431 C out	-6dB	0.01%	0.01%
DL431 splitter	DL451 I/O box	0dB	0.01%	0.01%
DL431 splitter	DL451 I/O box	40dB	0.03%	0.03%

Distortion at +20dBu

Input	*Output*	*Gain*	*1kHz*	*10kHz*
DL251 I/O box	DL251 I/O box	0dB	0.03%	0.03%
DL251 I/O box	DL251 I/O box	40dB	0.03%	0.03%
Surface I/O	Surface I/O	0dB	0.03%	0.03%
Surface I/O	Surface I/O	40dB	0.03%	0.03%
DL451 I/O box	DL451 I/O box	0dB	0.03%	0.03%
DL451 I/O box	DL451 I/O box	40dB	0.03%	0.03%
DL431 splitter	DL431 A out	0dB	0.03%	0.03%
DL431 splitter	DL431 A out	40dB	0.03%	0.03%
DL431 splitter	DL431 B out	0dB	0.03%	0.03%
DL431 splitter	DL431 B out	40dB	0.03%	0.03%
DL431 splitter	DL431 C out	-6dB	0.03%	0.03%
DL431 splitter	DL451 I/O box	0dB	0.03%	0.03%
DL431 splitter	DL451 I/O box	40dB	0.03%	0.03%

Mixing noise (all bus types) 22Hz to 22kHz unweighted

Number Of Inputs	*Gain*	*Fader Position*	*Pan*	*Output Noise*
12	0dB	-infinity	Central	-91dBu
12	0dB	0dB	Central	-78dBu
24	0dB	-infinity	Central	-91dBu
24	0dB	0dB	Central	-75dBu
48	0dB	-infinity	Central	-91dBu
48	0dB	0dB	Central	-72dBu

Signal path noise 22Hz to 22kHz unweighted

Inputs 150R terminated.

Input	*Output*	*Gain*	*Output Noise*	*EIN*
DL251 I/O box	DL251 I/O box	0dB	-85dBu	-85dBu
DL251 I/O box	DL251 I/O box	45dB	-81dBu	-126dBu
Surface I/O	Surface I/O	0dB	-85dBu	-85dBu
Surface I/O	Surface I/O	45dB	-82dBu	-127dBu
DL451 I/O box	DL451 I/O box	0dB	-89dBu	-89dBu
DL451 I/O box	DL451 I/O box	40dB	-87dBu	-127dBu
DL431 splitter	DL431 A out	0dB	-98dBu	-98dBu
DL431 splitter	DL431 A out	40dB	-88dBu	-128dBu
DL431 splitter	DL431 B out	0dB	-98dBu	-98dBu
DL431 splitter	DL431 B out	40dB	-88dBu	-128dBu
DL431 splitter	DL431 C out	-6dB	-123dBu	-117dBu
DL431 splitter	DL451 I/O box	0dB	-89dBu	-89dBu
DL431 splitter	DL451 I/O box	40dB	-87dBu	-127dBu

Dynamic range 22Hz to 22kHz unweighted

Input	*Output*	*Gain*	*Maximum Output*	*Dynamic Range*
DL251 I/O box	DL251 I/O box	0dB	+21dBu	106dB
DL251 I/O box	DL251 I/O box	45dB	+21dBu	103dB
Surface I/O	Surface I/O	0dB	+21dBu	106dB
Surface I/O	Surface I/O	45dB	+21dBu	103dB
DL451 I/O box	DL451 I/O box	0dB	+21dBu	110dB
DL451 I/O box	DL451 I/O box	40dB	+21dBu	108dB
DL431 splitter	DL431 A out	0dB	+26dBu	124dB
DL431 splitter	DL431 A out	40dB	+26dBu	114dB
DL431 splitter	DL431 B out	0dB	+26dBu	124dB
DL431 splitter	DL431 B out	40dB	+26dBu	114dB
DL431 splitter	DL431 C out	-6dB	+21dBu	144dB
DL431 splitter	DL451 I/O box	0dB	+21dBu	110dB
DL431 splitter	DL451 I/O box	40dB	+21dBu	108dB

PRO1 system inputs and outputs

DL251 I/O box - analogue inputs

Connector	3-pin XLR balanced
A/D converter	24-bit, 96kHz and 128 times oversampling

DL251 I/O box - analogue outputs

Connector	3-pin XLR balanced
D/A converter	24-bit, 96kHz and 128 times oversampling

DL251 I/O box - MIDI

MIDI	In, Out and Thru on 5-pin DIN

DL251 I/O box - digital system inputs and outputs

System connector	2 x AES50 (24 channels of bi-directional digital audio) on EtherCon® XLR
N+1 connector	1 x AES50 (24 channels of bi-directional digital audio) on EtherCon® XLR providing redundant back-up

PRO1 control surface - DSP/router system inputs and outputs

System connector	2 x AES50 (24 channels of bi-directional digital audio) on EtherCon® XLR
N+1 connector	1 x AES50 (24 channels of bi-directional digital audio) on EtherCon® XLR providing redundant back-up
System expansion connector	3 x AES50 (24 channels of bi-directional digital audio) on EtherCon® XLR
Word clock IN connector	BNC
Word clock OUT connector	BNC
AES3 sync IN connector	3-pin XLR
AES3 sync OUT connector	3-pin XLR

PRO1 control surface - analogue audio system inputs

Connector	3-pin XLR balanced
A/D converter	24-bit, 96kHz and 128 times oversampling
Talkback connector	3-pin XLR balanced line
Talk connector	3-pin XLR balanced mic with 48V phantom voltage

PRO1 control surface - analogue audio system outputs

Connector	3-pin XLR balanced
D/A converter	24-bit, 96kHz and 128 times oversampling
Monitor connector	3-pin XLR balanced line
Talk connector	3-pin XLR balanced line
Headphone connector	1/4” Jack (stereo)

PRO1 control surface - digital audio system inputs and outputs

Input connector	AES3 (two channels of digital audio) on 3-pin XLR
Sample rates	Accepts any frequency 32kHz - 96kHz
Bypass	Sample rate converter can be bypassed
Output connector	AES3 (two channels of digital audio) on 3-pin XLR
Sample rate	48k, 96k, or auto-tracking to inputs
Bypass	Sample rate converter can be bypassed
Word length	16, 20 or 24-bit

PRO1 control surface - control data system inputs and outputs

System connector	EtherCon® XLR

PRO1 control centre - miscellaneous inputs and outputs

Monitor output connector	DVI
USB host connection	USB 2.0 full speed (12.0Mbs) 5V, 1A maximum load
Lamp connector	4-pin XLR
MIDI	In, Out and Thru on 5-pin DIN

Inputs and output characteristics

Analogue input characteristics

Input Type	*Load Z*	*Gain*	*Maximum Level*	*Connector*
DL251 I/O box	10k	-22.5dB to +65dB	+24dBu	XLR
Surface I/O	10k	-22.5dB to +65dB	+24dBu	XLR
DL451 I/O box	10k	-25dB to +60dB	+26dBu	XLR
DL431 splitter	5k	-22.5dB to +65dB	+24dBu	XLR
Talk mic	600R	+15dB to +60dB	+6dB	XLR
Monitor	10k	0dB	+21dBu	XLR

Analogue output characteristics

Output Type	*Source Z*	*Gain*	*Maximum Level*	*Connector*
DL251 I/O box	50R	0dB	+21dBu	XLR
Surface I/O	50R	0dB	+21dBu	XLR
DL451 I/O box	50R	0dB	+21dBu	XLR
DL431 splitter (main)	150R	0dB	+24dBu	XLR
DL431 splitter (isolated)	75R	-6dB	+18dBu	XLR
Talk	50R	0dB	+24dBu	XLR
Monitor	50R	0dB	+24dBu	XLR
Headphones	10R	+10dB	750mW	1/4” Jack

Digital I/O characteristics

Type	*Channels*	*Data Length*	*I/O*	*Description Notes*	*Connector*
AES3	2	24-bit	Input	Conforms to AES3-2003	XLR
AES3	2	24-bit	Output	Conforms to AES3-2003	XLR
AES50	24	24-bit	Bi-directional	Conforms to AES50-2006	EtherCon® XLR

Miscellaneous digital characteristics

Type	*I/O*	*Description Notes*	*Connector*
Word clock	IN	Accepts TTL level, 96kHz square wave; impedance 75 ohms	BNC
Word clock	OUT	Provides TTL level, 96kHz square wave	BNC
AES sync	IN	Accepts a 96kHz digital audio signal conforming to AES3-2003	XLR
AES sync	OUT	Provides a 96kHz grade II reference signal conforming to AES3-2003	XLR

Main processing functions

Main input channel functions

Input channel hi pass	10Hz to 400Hz swept in digital domain Slope selectable 12dB/Oct or 24dB/Oct
Input channel lo pass	2kHz to 20kHz swept in digital domain Slope selectable 6dB/Oct or 12dB/Oct
Input channel treble	Parametric Operation Frequency 1kHz to 25kHz swept Gain +16dB to −16dB BW 0.1 Oct to 3 Oct Shelf Operation Frequency 1kHz to 25kHz swept Gain +16dB to −16dB Soft, Classic or Bright (minimum harmonic disruption) curves
Input channel hi mid	Parametric Operation Frequency 320Hz to 8kHz swept Gain +16dB to −16dB BW 0.1 Oct to 3 Oct
Input channel lo mid	Parametric Operation Frequency 80Hz to 2kHz swept Gain +16dB to −16dB BW 0.1 Oct to 3 Oct
Input channel bass	Parametric Operation Frequency 16Hz to 400Hz swept Gain +16dB to −16dB BW 0.1 Oct to 3 Oct Shelf Operation Frequency 16Hz to 400Hz swept Gain +16dB to −16dB Warm, Classic or Deep (minimum harmonic disruption) curves

Input channel compressor	Peak, Linear, RMS, Vintage modes Threshold -50dBu to +25dBu Attack 200µs to 20ms Release 50ms to 3 sec Ratio 25:1 to 1:1 Knee 4dB, 12dB or 40dB Gain 0dB to +24dB Side chain source selectable + filter Frequency 50Hz to 15kHz swept Bandwidth 1/3, 1 or 2 Oct
Input channel gate	Peak mode Threshold -50dBu to +25dBu Attack 20µs to 20ms Hold 5ms to 2 sec Release 2ms to 2 sec Range 100dB to 0dB Side chain source selectable + filter Frequency 50Hz to 15kHz swept Bandwidth 1/3, 1 or 2 Oct

Auxiliary input channel functions

Aux channel treble	Treble Parametric Operation Frequency 1kHz to 25kHz swept Gain +16dB to −16dB BW 0.1 Oct to 3 Oct Shelf Operation Frequency 1kHz to 25kHz swept Gain +16dB to −16dB Soft, Classic or Bright (minimum harmonic disruption) curves
Aux channel hi mid	Parametric Operation Frequency 320Hz to 8kHz swept Gain +16dB to −16dB BW 0.1 Oct to 3 Oct
Aux channel lo mid	Parametric Operation Frequency 80Hz to 2kHz swept Gain +16dB to −16dB BW 0.1 Oct to 3 Oct
Aux channel bass	Parametric Operation Frequency 16Hz to 400Hz swept Gain +16dB to −16dB BW 0.1 Oct to 3 Oct Shelf Operation Frequency 16Hz to 400Hz swept Gain +16dB to −16dB Warm, Classic or Deep (minimum harmonic disruption) curves

Output channel functions

Output channel band 6	Parametric Operation Frequency 16Hz to 25kHz swept Gain +16dB to −16dB BW 0.1 Oct to 3 Oct Lo Pass Operation Frequency 16Hz to 25kHz swept Slope 6dB/Oct or 12dB/Oct Shelf Operation Frequency 16Hz to 25kHz swept Gain +16dB to −16dB Mode soft curve
Output channel bands 3, 4 and 5	Parametric Operation Frequency 16Hz to 25kHz swept Gain +16dB to −16dB BW 0.1 Oct to 3 Oct
Output channel band 2	Parametric Operation Frequency 16Hz to 25kHz swept Gain +16dB to −16dB BW 0.1 Oct to 3 Oct Hi Pass Operation Frequency 16Hz to 25kHz swept Slope 24dB/Oct
Output channel band 1	Parametric Operation Frequency 16Hz to 25kHz swept Gain +16dB to −16dB BW 0.1 Oct to 3 Oct Hi Pass Operation Frequency 16Hz to 25kHz swept Slope 6dB/Oct or 12dB/Oct Shelf Operation Frequency 16Hz to 25kHz swept Gain +16dB to −16dB Mode soft curve
Output channel GEQ	8 available in place of PEQ (above) 31 Bands, 1/3 Oct, Proportional Q Lo Pass Frequency 2kHz to 20kHz swept Slope 6dB/Oct or 12dB/Oct Hi Pass Frequency 20Hz to 500Hz swept Slope 6dB/Oct or 12dB/Oct
Output channel dynamic	Pk, Linear, RMS, Vintage, Shimmer modes Threshold -50dBu to +25dBu Attack 200µs to 20ms Release 50ms to 3 sec Ratio 25:1 to 1:1 Knee 4dB, 12dB or 40dB Gain 0dB to +24dB Side chain source selectable + filter Frequency 50Hz to 15kHz swept Bandwidth 1/3, 1 or 2 Oct

Effects channel functions

Stereo effects channel	6 available configurable as Stereo or mono in, stereo out (8 in, 8 out for dynamic) Modulated delay effects Complex delay, reverbs
Effects channel GEQ	16 available in place of effects (above) 31 Bands, 1/3 Oct, Proportional Q Lo Pass Frequency 2kHz to 20kHz swept Slope 6dB/Oct or 12dB/Oct Hi Pass Frequency 20Hz to 500Hz swept Slope 6dB/Oct or 12dB/Oct

Appendix D: Troubleshooting

This appendix gives details of troubleshooting the PRO1 Live Audio System.

No audio

If you have set up the PRO1 and followed all of the instructions for obtaining audio, but you are not hearing anything through the speakers, check the following:

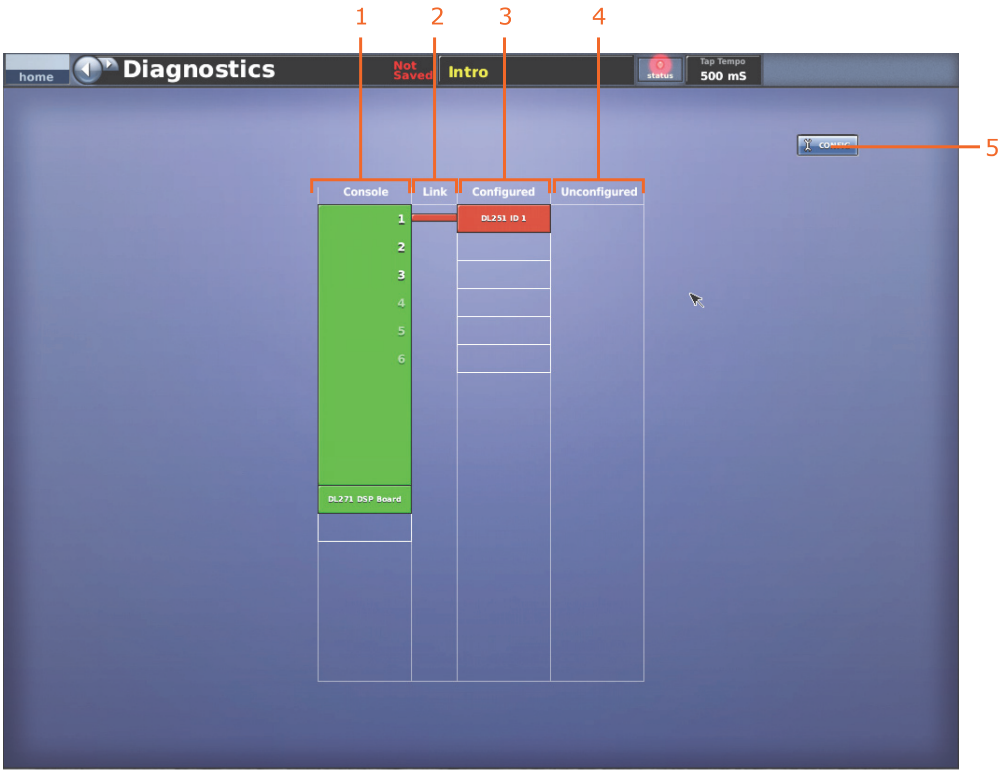
- Make sure the appropriate **ST** buttons in the channel fast strips are on.
- Make sure the appropriate **ST** buttons in the **source a**/**b** panels (monitors section of the output bay) are on.
- Make sure nothing is muted.
- Make sure no faders are set to minimum.
- Check that the VCA group and master faders are at unity gain.
- Use solo at selected points in the signal path to try and pinpoint where the signal is being lost.
- Check the **Preferences** screen for correct signal routing by making sure channel sources/destinations are correctly assigned.

If you still don't have any audio, contact Midas Technical Support.

Diagnostics

You can view the **Diagnostics** screen to get an overview of the current health and status of the system. The **Diagnostics** screen shows real-time connectivity of the system, the health of connected nodes and whether a device is configured or not.

The **status** LED at the top of the screen, which is constantly displayed while the control centre is switched on, is linked to the status of individual items on the **Diagnostics** screen. If there is a problem, you can click the **status** LED to open the **Diagnostics** screen and see what is causing the error.

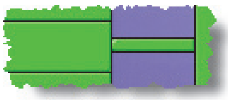
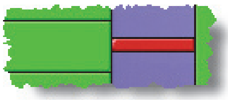
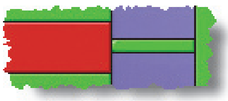
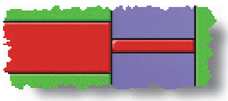


*An example of the **Diagnostics** screen*

Item	*Element*	*Description*
1	**Console** column	This column contains boxes that represent the internal processes of the PRO1 Control Centre, such as the control surface, master controller, DSP, Audio IO and GUI. If any of these develop an error condition the colour of the boxes will change to red. Clicking the **DL271 DSP Board** box will open a **Diagnostics Inspector** window (see "About the Diagnostics Inspector window" on page 331).
2	**Link** column	This column shows the health of the physical connection of each AES50-compatible device and the PRO1 Control Centre. Clicking a link will open a **Diagnostics Inspector** window (see "About the Diagnostics Inspector window" on page 331).
3	**Configured** column	This column shows any AES50 devices, such as a IDL251 Audio System I/O, that have been detected as connected to the system and are also configured for the system. Clicking one of these devices will open a **Diagnostics Inspector** window (see "About the Diagnostics Inspector window" on page 331).

Item	Element	Description
4	**Unconfirmed** column	This column shows any AES50 devices, such as a IDL251 Audio System I/O, that have been detected as connected to the system, but have not been configured during the patching procedure. Clicking one of these devices will open a **Diagnostics Inspector** window (see "About the Diagnostics Inspector window" below). To configure a device, see "Setting up the I/O rack devices" on page 39.
5	**CONFIG** button	Opens the **AES50 Device Configuration** window (see Figure 9 "Typical AES50 Device Configuration window" on page 39)

The colour of each device, together with its link (if applicable), indicates its current status, as shown in the following table.

State	Description	Unit status	Connection of active link	Connection of inactive link
	Both the unit and link are green	Good	Good	Good
	Unit is green and the link is red	Good	Bad	Not known
	Unit is red and the link is green	Malfunction	Good	Not known
	Both the unit and link are red	Not known	Bad	Bad

There is also an amber condition, which means that the item(s) is in error, but is not contributing to the audio.

About the Diagnostics Inspector window

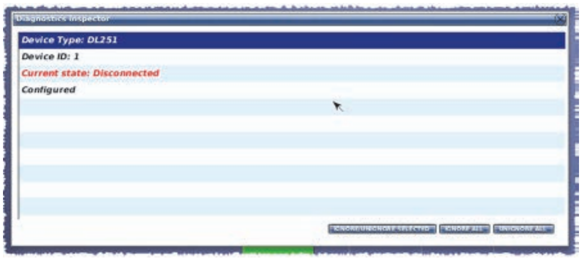
Clicking an item in the **Diagnostics** screen will open its **Diagnostics Inspector** window, which provides detailed information, particularly if the item has an error condition.

*Typical PRO Series **Diagnostics Inspector** window with the 'ignore' buttons at the lower right corner*

The 'ignore' buttons of the Diagnostics Inspector window let you configure the PRO Series to ignore errors on selected/all items. This is an important feature because there may be times when you are quite happy to work with a known error(s), but will want to know when a new error occurs.

Note: Diagnostic Inspector *windows are primarily for use by Midas service and software engineers. By providing useful information, such as device health and status, they aid fault diagnosis and rectification, and may help solve any problems that may arise. Apart from using the 'ignore' buttons, it is unlikely that operators of the DL251/ DL252 Audio System I/O will ever need to use this function.*

An example of DL271 DSP Board ***Diagnostics*** *screen*



An example of DL251 device ***Diagnostics*** *screen showing an error condition*

>> To ignore/unignore error condition

Click the desired item that is in an error condition and then click **IGNORE/UNIGNORE SELECTED.** Its text colour should change to black to show that it is being ignored. Click **IGNORE/UNIGNORE SELECTED** again to highlight the error condition.

To ignore all error conditions, click **IGNORE ALL.** Click **UNIGNORE ALL** to highlight all error conditions.

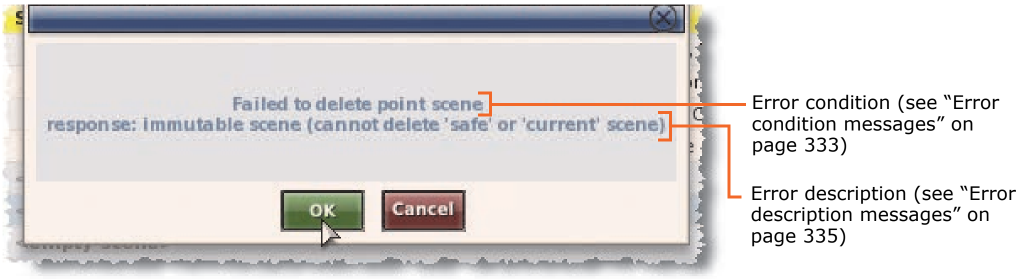
Troubleshooting automation

This section explains the error messages that you may see when using PRO1 automation.

Error messages

Error messages, which can appear when you are accessing the **Files** or **Automation** screens, provide useful information on the condition that triggered them. Due to the way the filing and automation systems interact with the internal processing system of the PRO1, not all error messages are indicative of a problem; some may appear due to the current state of the system and just require a retry of the operation.

Error messages comprise two components — a first line of text containing the error condition, followed on the next line by a description of the error (prefixed by the text "response:"). The error condition text indicates the operation that may have triggered the error message, while the error description text explains the reason for failure and, in some cases, also provides information that may be useful to service engineers. The following diagram shows a typical error message.



>> To proceed after an error message appears

1 Heed the error message.

2 Click **OK**.

3 Take the appropriate action for that particular error. Refer to Table 14 "List of error condition messages" on page 333 and Table 15 "List of error description messages" on page 335.

Automation error messages

An automation error message may be generated by any of the following:

- An attempt to perform a copy and paste is ignored.
- Attempting to assign notes to a scene. The text "There was an error setting the scene note" will be displayed.
- An attempt to assign MIDI data to a scene is ignored.
- An attempt to set the navigation mode, that is, switching rehearsal mode on or off, is ignored.
- An attempt to skip/unskip a scene or point scene; see "Rehearsals" on page 189.
- An attempt to assign the default store option is ignored.
- An attempt to assign the rehearsal mode state for all scenes is ignored. (This functionality is not available to the user.)
- An attempt to set the MIDI navigation mode is ignored. (This functionality is not available to the user.)
- An attempt to modify the scene list mode is ignored.

Error condition messages

The following table contains the possible error condition messages for both the file and automation systems. These messages comprise the first line of the error message as they appear on the GUI.

Table 14: List of error condition messages

System Type	*Error Message*	*Fault Condition*
File	Failed to copy file	Attempting to copy a file.
File	Failed to delete file	Attempting to delete a file.
File	Failed to rename file	Attempting to rename a file.
Automation	Failed to copy point scene to point scene	Attempting to copy one point scene to another.

System Type	*Error Message*	*Fault Condition*
Automation	Failed to create a new show	Attempting to create a new show.
Automation	Failed to delete point scene	Attempting to delete a point scene.
Automation	Failed to expand point scene range	Attempting to expand point scene range, that is, by inserting an extra 10 point scenes, for example, expanding scene 10.00 will add point scenes 10.10, 10.20 etc., up to 10.90.
Automation	Failed to initiate point scene storage	Attempting to initiate point scene storage, that is, when clicking **Store** on the GUI. (A successful outcome is to display the 'Store' window.)
Automation	Failed to insert point scene	Attempting to complete point scene storage by clicking **OK** after selecting "Insert before scene".
Automation	Failed to load show	Attempting to load a show file.
Automation	Failed to recall last scene	Attempting to recall the previous scene to the control surface.
Automation	Failed to recall Next scene	Attempting to recall the next scene to the control surface.
Automation	Failed to recall Now scene	Attempting to reload the current scene or the current jog scene (if any) to the control surface.
Automation	Failed to rename point scene	Attempting to rename a point scene.
Automation	Failed to save file	Attempting to save a currently loaded file.
Automation	Failed to save file to new name	Attempting to save a currently loaded show file to another file name, that is, by using the **Save As** button.
Automation	Failed to store point scene	Attempting to complete point scene storage by clicking **OK** after selecting "Store to empty scene", "Overwrite scene" or "Store to empty scene".
Automation	Failed to unexpand point scene range	Attempting to unexpand point scene range. This is the opposite of expanding the point scene range (immediately above) and can only be carried out if the 10 point scenes to be unexpanded are empty.

Error description messages

The following table contains the possible error description messages for both the file and automation systems, which will start on the second line of the error message. The "Error Message" column in the table contains the error message text that immediately follows the "response:" text. For ease of reference the table lists the error messages in alphabetical order.

Table 15: List of error description messages

Error Message	*System(s)*	*Problem*	*Solution*
hexadecimal number			
<error code in hexadecimal> unknown error code	File and Automation	Indication of a possible system error.	Note down the hexadecimal value of the error code and contact Midas Technical Support, giving them this value.
a			
artefact clone policy violation	File and Automation	The cloning of this artefact (file type) is not allowed.	Avoid using this type of operation.
artefact creation policy violation	File and Automation	The creation of this file type is not allowed.	Avoid using this type of operation.
artefact deletion policy violation	File and Automation	The deleting of this file type is not allowed.	Avoid using this type of operation.
artefact import violation	File and Automation	The importing of this file type is not allowed.	Avoid using this type of operation.
artefact load policy violation	File and Automation	The loading of this file type is not allowed.	Avoid using this type of operation.
artefact rename policy violation	File and Automation	The renaming of this file type is not allowed.	Avoid using this type of operation.
artefact replication policy violation	File and Automation	The replication of this file type is not allowed.	Avoid using this type of operation.
artefact save policy violation	File and Automation	The saving of this file type is not allowed.	Avoid using this type of operation.
attempt to overwrite existing data (overwrite not enabled)	File and Automation	The operation to save or copy to the existing file is not allowed, as files cannot be overwritten.	Avoid using this type of operation.
b			
bad device	File and Automation	Operation could not be carried out because the device, that is, the internal compact flash of the PRO1 or USB memory stick (if connected), does not contain the required directory structure.	• If the device is the internal compact flash of the PRO1, this could be an indication of a serious problem. Contact Midas Technical Support. • If the device is the USB memory stick, check that the device has not been disconnected from the control surface.

Error Message	System(s)	Problem	Solution
bad device ID	File and Automation	The device identifier has not been recognised.	If you are exporting a file to a USB memory stick, check that it has not been disconnected from the control surface.
bad directory	File and Automation	The file system path does not terminate in a directory.	This is highly unlikely to occur in practice, but is an indication of a serious error. Contact Midas Technical Support.
bad file	File and Automation	The file system path does not terminate in a file.	This is highly unlikely to occur in practice, but is an indication of a serious error. Contact Midas Technical Support.
bad file artefact	File and Automation	The file has been detected as not valid. Preferences, preset library and show files are validated by comparing their actual attributes against the corresponding fields stored in the header of the file, such as, file size, checksum etc.	Try again. If still unsuccessful, and if the file is a show file, try a backup file, if one is available.
bad file version	File and Automation	The preferences, preset library or show file could not be opened because its file header version field was not valid.	Try again. If still unsuccessful, and if the file is a show file, try a backup file, if one is available.
bad path	File and Automation	A file or directory is missing.	This is an indication of a serious error. Contact Midas Technical Support.
bad point scene ID	File and Automation	The scene's point scene ID cannot be found.	Try again.
c			
c-lib file error	File and Automation	Critical internal error.	This is an indication of a serious error. Contact Midas Technical Support.
c-lib error	File and Automation	Critical internal error.	This is an indication of a serious error. Contact Midas Technical Support.
c-lib process error	File and Automation	Critical internal error.	This is an indication of a serious error. Contact Midas Technical Support.
d			
device utilisation policy violation	File and Automation	The device is full.	If necessary, backup some files and delete them from the device to free up some memory.

Error Message	*System(s)*	*Problem*	*Solution*
e			
empty point scene index	File and Automation	An attempt was made to navigate an empty point scene index.	Try again. If unsuccessful, and if the file is a show file, try a backup file, if one is available.
event is already active	Automation	Critical internal error.	This is an indication of a serious error. Contact Midas Technical Support.
f			
failed to add to lock list	File	A device or file could not be 'locked' to prevent another task accessing it while it is in use.	Although, in practice, it is highly unlikely to occur, this error may indicate a serious system failure. Contact Midas Technical Support.
failed to add scene	File and Automation	The scene could not be added.	Try again.
failed to allocate memory	Automation	The MC was unable to allocate sufficient memory (RAM) to complete the task.	Switch off the PRO1 Live Audio System and switch it back on again. If the problem persists, it could be an indication of a serious error. Contact Midas Technical Support.
failed to configure scope mask	Automation	The 'copy and paste through scenes' operation failed.	Try again. If repeated attempts fail, contact Midas Technical Support.
failed to create show	Automation	A new show could not be created.	Try again. If repeated attempts fail, contact Midas Technical Support.
failed to deschedule event	Automation	Critical internal error.	This is an indication of a serious error. Contact Midas Technical Support.
failed to schedule event	Automation	Critical internal error.	This is an indication of a serious error. Contact Midas Technical Support.
i			
immutable scene (cannot delete 'safe' or 'current' scene)	File and Automation	The operation on the current or safe scene is not allowed. The safe scene cannot be edited or deleted and you cannot store to it. Also, you cannot delete the scene last recalled to the control surface. (Precludes the use of the **Now** button.)	Avoid using these types of operation.
j			
jog position is empty	Automation	The current scene is empty.	Avoid this type of operation on an empty scene.

Error Message	System(s)	Problem	Solution
m			
missing file	File and Automation	The required file cannot be found.	Try again.
missing navigation state	Automation	Critical internal error.	This is an indication of a serious error. Contact Midas Technical Support.
mtools lookup	File and Automation	Critical internal error.	This is an indication of a serious error. Contact Midas Technical Support.
n			
no CBMA access	Automation	Automation manager does not have access to the current control surface settings.	Try again or try switching off the PRO1 Live Audio System and then switching it back on again. If the problem persists, contact Midas Technical Support.
no next scene	Automation	There is no next scene relative to the current position in the scene list. This is generated by recalling the next scene when the current scene is the last in the scene list.	Avoid this operation on the last scene in the cue list.
no previous scene	Automation	There is no previous scene relative to the current position in the scene list. This is generated by recall the last scene when the current scene is 00.00, that is, the safe scene.	Avoid this operation on the safe scene.
no scene data	Automation	The scene contains no scene notes or MIDI data.	Only carry out this type of operation on a scene that contains scene notes or MIDI data.
no show loaded	Automation	There is no show loaded.	Only carry out this type of operation with a show loaded.
not in storing state	Automation	The Automation System was momentarily unable to store a scene.	Try again. If repeated attempts fail, contact Midas Technical Support.
null pointer	File and Automation	Critical internal error.	This is an indication of a serious error. Contact Midas Technical Support.
p			
persistent storage error	File and Automation	The GUI or one of its subsystems is out of date and cannot interpret the new failure modes.	Update the GUI and its subsystems. If the problem persists, contact Midas Technical Support.
point scene index continuity error	File and Automation	The show file being modified is damaged.	This is an indication of a serious error. Contact Midas Technical Support.

Error Message	System(s)	Problem	Solution
point scene index integrity error	File and Automation	The show file being modified is damaged.	This is an indication of a serious error. Contact Midas Technical Support.
point scene insert error	File and Automation	Failed to insert a point scene.	This is highly unlikely to occur in practice, but is an indication of a serious error. Contact Midas Technical Support.
portable scene format conversion error	Automation	The attempt to load a show, which was last saved by an MC built with a different enum version, failed during the scene conversion stage of the loading process.	Try again. If repeated attempts fail, contact Midas Technical Support.
r			
required device has files that are in use	File and Automation	Another task is currently accessing file(s) on the device, that is, the internal compact flash of the PRO1 or USB memory stick (if connected).	Try again.
required device is locked	File and Automation	Another task is currently accessing the device, that is, the internal compact flash of the PRO1 or USB memory stick (if connected).	Try again.
s			
scene capacity violation	File and Automation	The scene cannot be stored to the show file, as it already contains the maximum number of scenes allowed.	If necessary, delete one or more of the other scenes. The maximum capacity for a show file is 500 scenes for a 512MB master controller (MC) and 1000 scenes for a 1GB MC.
scene UID error	File and Automation	The file being modified is damaged.	This is an indication of a serious error. Contact Midas Technical Support.
shell command error	File and Automation	This is a critical internal error.	This is an indication of a serious error. Contact Midas Technical Support.
source point scene is empty	Automation	Specified source scene is an empty ‘slot’.	Only carry out this type of operation on a scene that is not empty.
specified file is already locked	File and Automation	Another task is currently accessing the file	Try again.

Error Message	*System(s)*	*Problem*	*Solution*
specified file was not found	File and Automation	The file could not be found on the specified device, that is, the internal compact flash of the PRO1 or USB memory stick (if connected).	• If the device is the internal compact flash of the PRO1, this could be an indication of a serious problem. Contact Midas Technical Support. • If the device is the USB memory stick, check that the device has not been disconnected from the control surface.
stdio stream error	File and Automation	This is a critical internal error.	This is an indication of a serious error. Contact Midas Technical Support.
stdio stream open error	File and Automation	This is a critical internal error.	This is an indication of a serious error. Contact Midas Technical Support.
stdio stream seek error	File and Automation	This is a critical internal error.	This is an indication of a serious error. Contact Midas Technical Support.
storage policy violation	File and Automation	There has been a 'storage policy' violation. (This is not necessarily a critical error.)	Ensure that all software components are up to date. If the problem persists Contact Midas Technical Support.
t			
the <file/ automation> manager is not registered	File and Automation	The System Manager is momentarily unavailable.	Try again.
u			
unknown parameter enum value	Automation	A parameter with a value that was not valid was supplied to the MC.	Try again. If this occurs again, contact Midas Technical Support.

Appendix E: Updating The PRO1 Host Software

This appendix shows you how to update the host software of the PRO1 and also its associated networked devices.

About the PRO1 updater

The PRO1 has an update facility that provides an easy and straightforward method of updating your system. It lets you install the latest version of host software on the PRO1 Control Centre and also any networked DLnnn or Klark Teknik DN9331 Rapide Graphic Controller units.

The updater file is copied onto a USB memory stick, which is plugged into the PRO1 Control Centre. The console will recognise the updater file and let you start the updater, which will detect all units in your system and let you selectively upgrade them to the new software.

By using the updater, you can install an earlier version of the host software on your system if you should ever need to.

About the updater screen

During installation the updater screen will appear. This screen lets you select the system devices you want to upgrade, start the update procedure and also shows you how the procedure is progressing.

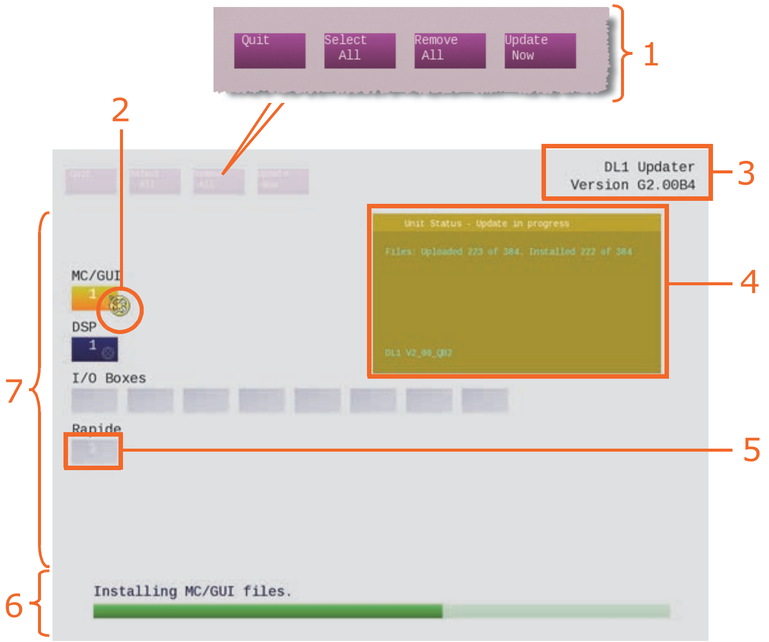


Figure 29: A typical updater display

Item	***Description***	***Function***
1	Updater menu	See "Updater menu" on page 343.
2	Pointer	The pointer, which has two possible icons (both translucent yellow) depending on the update status, changes to an arrowhead icon when the updater is ready for you to select the devices that you want to update. The pointer becomes a roundel icon during the update procedure; it rotates to show that the update is in progress.
3	PRO1 host software version	Shows the PRO1 host software version that the system will be updated to.
4	**Unit Status** window	See "Unit Status window" on page 343.
5	Device block	See "Device blocks" on page 343.
6	Update status bar	The green bar shows the progress of the update procedure, whether it is for the device currently being updated or for the whole procedure itself. The text immediately above shows what status the green bar is representing.
7	Device area	Shows possible/actual system devices.

Updater menu

The following table gives a description of the updater menu commands.

Command	*Function*
Quit	Exits the updater. After a power cycle, the control centre returns to the operating condition it was in when the updater command was selected from the GUI menu.
Select All	Selects all Midas devices connected (and detected) in the PRO1 Live Audio System.
Remove All	Deselects all Midas devices connected (and detected) in the PRO1 Live Audio System.
Update Now	Starts the update procedure.

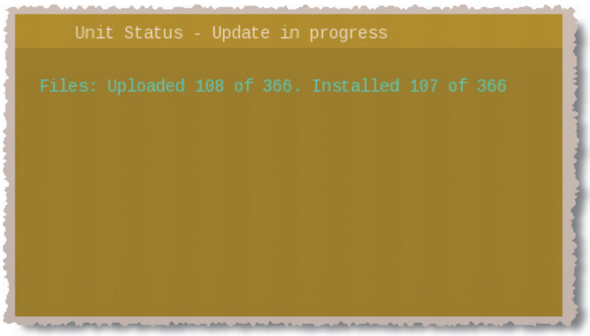
Device blocks

Each device block represents a possible or actual device connected in the system. The device block number is referenced to the device's ID. The colour of each device block indicates the update status of the device, which is shown in the following table.

State	*Description*
	Grey background — appears during the updater's 'triggering upgrade client' procedure. If its appearance doesn't change throughout the update procedure, either there is no device connected in this position or one has not been detected.
	Blue background without roundel — appears after the updater's 'triggering upgrade client' procedure has finished to show you that there is a device connected in this position.
	Blue background with roundel — appears after this device has been selected for update.
	Gold background with roundel — this device is currently being updated.
	Green background with tick — this device has successfully been updated.
	Red background with cross — this device's update has failed.

Unit Status window

You can get more detailed information on the update progress of a device by moving the pointer over it. A translucent window (a typical one is shown right) will open towards the upper-right corner of the screen. The window has a title bar at the top and information below.



The colour of the window matches the colour of its associated device block, which gives an indication of its update status, as described in the previous section.

Using the PRO1 updater

This section shows you how to update your PRO1 Live Audio System. However, before you begin there are a few things you will need and some things you must do.

What you will need

Before you begin, check that you have the following:

- **USB memory stick** The USB memory stick (flash drive) must have enough available memory to store any shows that you will need to backup, plus an additional 150MB of memory for the update package (that is, the file with a .tar extension). It should also preferably be of USB 2.0 specification.
- **Stable mains supply** If the power drops at a critical point during the update, it is possible — although unlikely — that this could cause some of the system components not to function. A warning window opens before you start the update procedure to remind you of this.

Preparation

Before you begin, we recommend that you do the following:

- **Backup your shows** It is likely that any shows will be erased when you power cycle the PRO1 following an update. We therefore recommend that you backup your shows onto the USB memory stick (see "Saving your show files to a USB memory stick" on page 88), and then copy them onto a PC.
- **Check that everything is connected** Make sure that everything on the system is correctly connected, configured and functioning properly.
- **Switch off speakers** During the update procedure the DSP and AES routing may perform a number of resets during which the audio may not be in a controlled state. We therefore recommend that you switch off any speakers connected to the system.
- **Make sure you have enough time** The update procedure may take quite a while to complete, so make sure you have at least 25 minutes free before you start. **We do not recommend carrying out an update just before a performance!**
- **Configure the USB memory stick** Create a folder at the top level (root directory) of your USB memory stick called "DL1upgrades". Then, copy the latest update file (DL1xxx.tar) into it.

Updating your system

! **UPGRADING YOUR SYSTEM WILL CAUSE THE CONTROL CENTRE TO LOSE SYNCHRONISATION, WHICH CAN RESULT IN LOUD NOISES FROM THE SYSTEM. ALWAYS MUTE THE PA AT THE AMPLIFIER/SPEAKER BEFORE UPDATING YOUR SYSTEM.**

! **Do not switch off the power to any of the system devices while the PRO1 is carrying out the installation of the host software.**

The installation process is carried out with the PRO1 Control Centre fully powered up and operational.

>> To update the PRO1

1. **Mute the PA at the amplifier/speaker.**

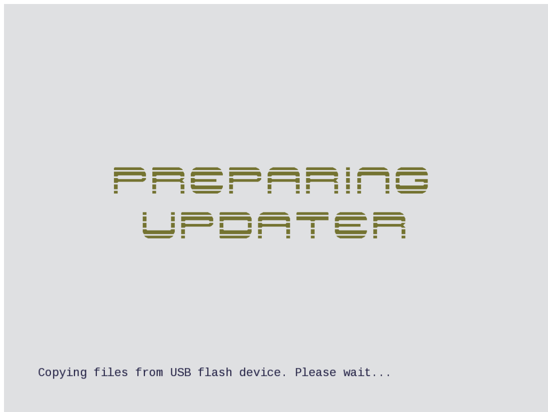
2. Plug the USB memory stick containing the .tar file into the **USB key** socket on the control surface of the PRO1 (see "Control surface" on page 246). The "Run upgrade utility?" window will open automatically.

   If there is more than one upgrade file on the USB memory stick, you will be instructed on how to select the desired one from the menu. Otherwise, a single upgrade file will be loaded automatically.

The following is the update procedure:

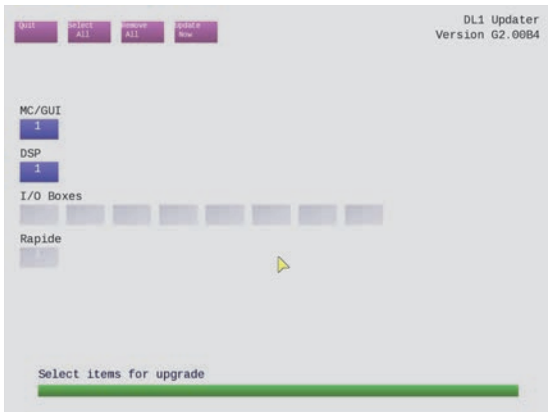
3. The "PREPARING UPDATER" screen opens. The updater makes an integrity check of all the files included in the update before launching fully.

   The text towards the bottom of the screen will inform you of the updater progress.



4. After the updater preparation has finished, the updater screen opens. It automatically runs through a sequence to detect all units in the system, during which the updater menu is unavailable. When it has finished, the text just above the green bar (shown right) will show "Select items for upgrade" and the devices available for upgrade will be coloured blue (see "Device blocks" on page 343).

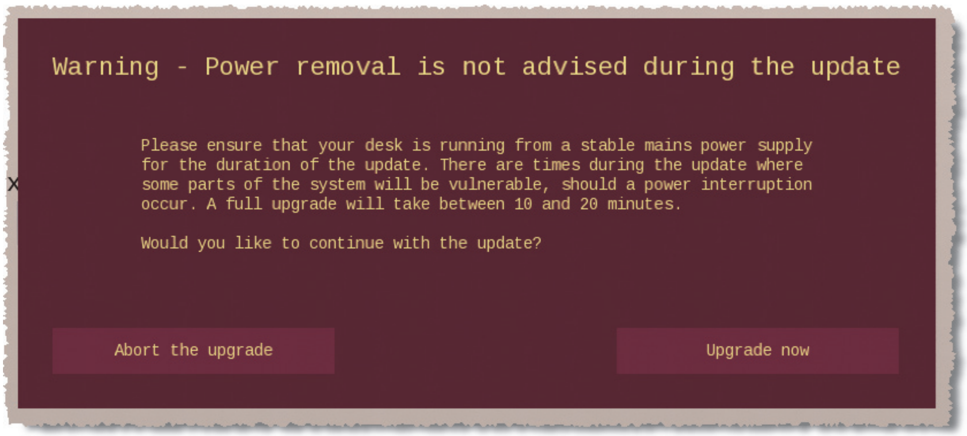
   Note: *In the unlikely event that the "Some units not detected" message appears, check that your system interconnections are good and click **OK** to continue.*



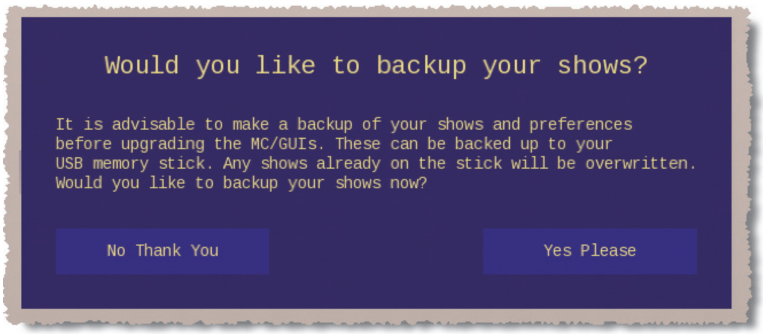
5. Click **Select All** to select all the devices for update. However, if you want to edit the selections, click on the device blocks to select/deselect individual ones. You can also click **Remove All** to deselect all of the ones selected.

   A Midas roundel will appear in the blocks of the devices selected for upgrade.

6 Click **Update Now.** A warning window will open (see below).



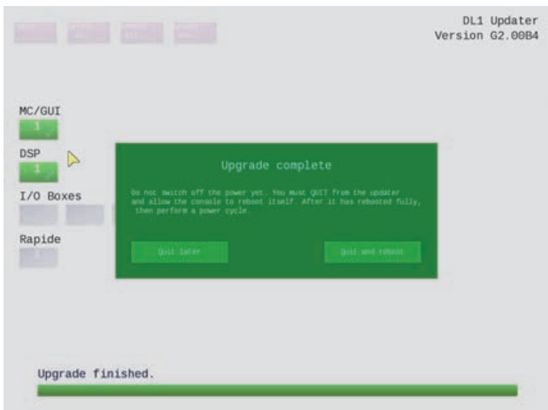
7 Click **Upgrade now.** A window will open asking if you would like to back up your shows.



8 Click **No Thank You** to start the upgrade. (If, at this point, you want to back up your shows and preferences, click **Yes Please.**) The device blocks will change colour according to their update status (see "Device blocks" on page 343). The green bar at the bottom will show the progress of the current action. A typical screen display during the update procedure is shown in Figure 29 "A typical updater display" on page 342.

During the upgrade, leave all of the system devices switched on.

9 When the upgrade procedure has finished, you will see the "Upgrade complete" message (shown right).



! **Do not switch off the power until the console has fully rebooted.**

10 Click **Quit and reboot** to exit the updater and automatically reboot the PRO1. (You can exit the updater later by clicking **Quit later.**)

The GUI will display the screen shown right.

Do not switch off the power until the console has fully rebooted.

11 When the PRO1 has fully rebooted, power cycle the system (including all I/O boxes) to restart using the new software. Do this by powering the system down and then powering it up again (see “Powering the PRO1 system” on page 30).

Appendix F: Parameters Affected By Scope

This appendix shows the parameters that are affected by scope.

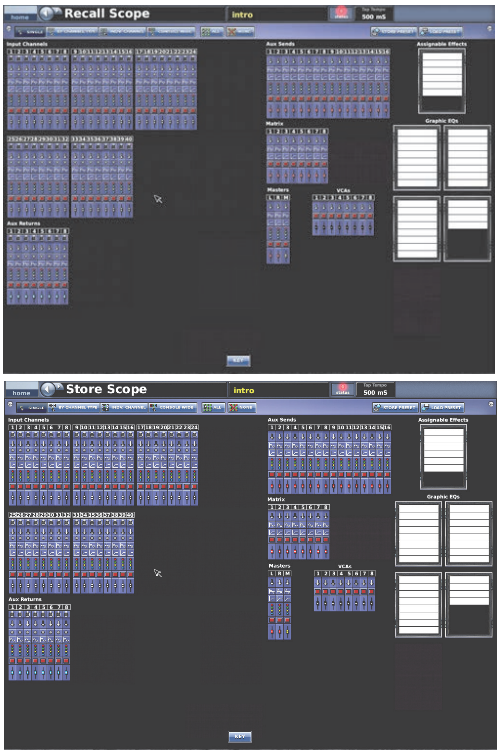
Note: *The parameter areas for the scopes (store and recall) and the safes are basically the same. However, the way they are presented in their respective appendices is different. This may provide you with a useful alternative when referring to this material, should you prefer one more than the other (see Appendix H "Parameters Protected By Safes" on page 397).*

Introduction

This appendix shows you which parameters are or are not scoped when you select the parameter sections in the channels, buses, groups, effects and GEQs on the **Store Scope** and **Recall Scope** screens.

There is a section for each area (channel, bus, group, effect and GEQ) on the scope screen, and these sections are then subdivided according to the processing areas in the channel strips on the control surface and GUI.

The following diagram shows the default **Store Scope** and **Recall Scope** screens, and the table to the right shows what the symbols in each section represent. In the table the orange letters in the **Ref.** column are used in the tables throughout this appendix to quickly identify the parameter sections.



Ref.	Parameter section	Symbol
A	Routing	
B	All	
C	Mic Amp	
D	EQ	
E	Dyn	
F	Busses	
G	Mute	
H	Fader	

In the parameter tables throughout this appendix, **Yes** = scoped, **No** = not scoped and N/A = not applicable.

The following table is intended as a reference guide to help you go quickly to the area you want.

Control area	*Input Channels*	*Aux Returns (Returns)*	*Aux Sends (Auxes)*	*Matrix (Matrices)*	*Masters*	*Graphic EQs*	*Assignable Effects*	*Variable Control Associations (VCAs)*
Patching	Page 351	Page 362	Page 368	Page 375	Page 381	Page 388	Page 390	N/A
Configuration	Page 352	Page 363	Page 369	Page 376	Page 382	N/A	N/A	N/A
Dynamics	Page 354	N/A	Page 370	Page 377	Page 383	N/A	N/A	N/A
Insert	Page 356	N/A	Page 371	Page 378	Page 384	N/A	N/A	N/A
EQ	Page 357	N/A	Page 372	Page 379	Page 385	N/A	N/A	N/A
Aux send	Page 358	N/A	N/A	N/A	N/A	N/A	N/A	N/A
Matrix send	Page 359	Page 366	Page 373	N/A	Page 386	N/A	N/A	N/A
Fader	Page 360	Page 367	Page 374	Page 380	Page 387	N/A	N/A	N/A
GEQ	N/A	N/A	N/A	N/A	N/A	Page 389	N/A	N/A
Effects	N/A	N/A	N/A	N/A	N/A	N/A	Page 391	N/A
Groups	N/A	N/A	N/A	N/A	N/A	N/A	N/A	Page 392

Note: *Throughout this appendix, when referring to the hardware controls in either fader bay the default channel assignments are used (that is, inputs in the channel fader bay and outputs in the mix fader bay). This is because any channel type can be assigned to either fader bay.*

Inputs

Each scope screen has 40 input channels in the **Input Channels** section.

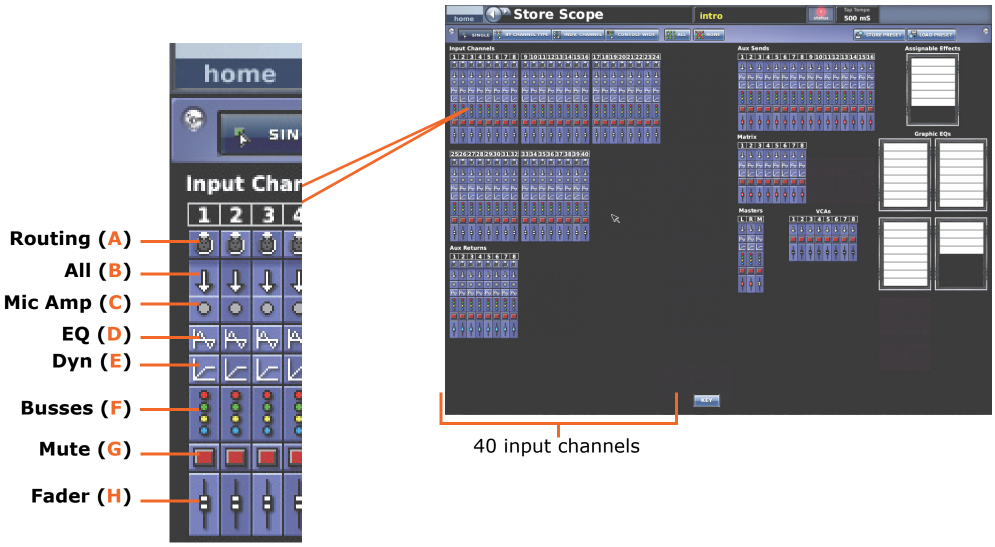
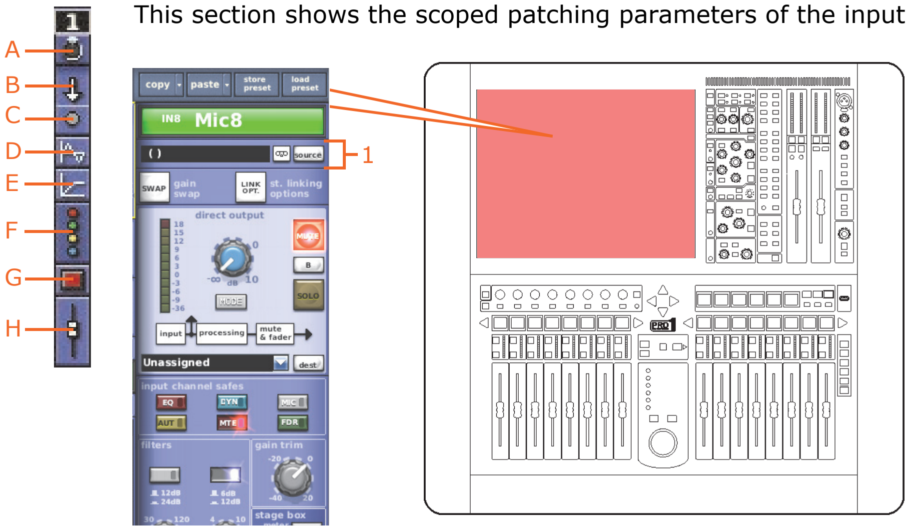


Figure 30: Parameter sections per input channel

Patching

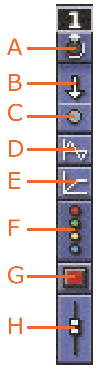
This section shows the scoped patching parameters of the input channels.



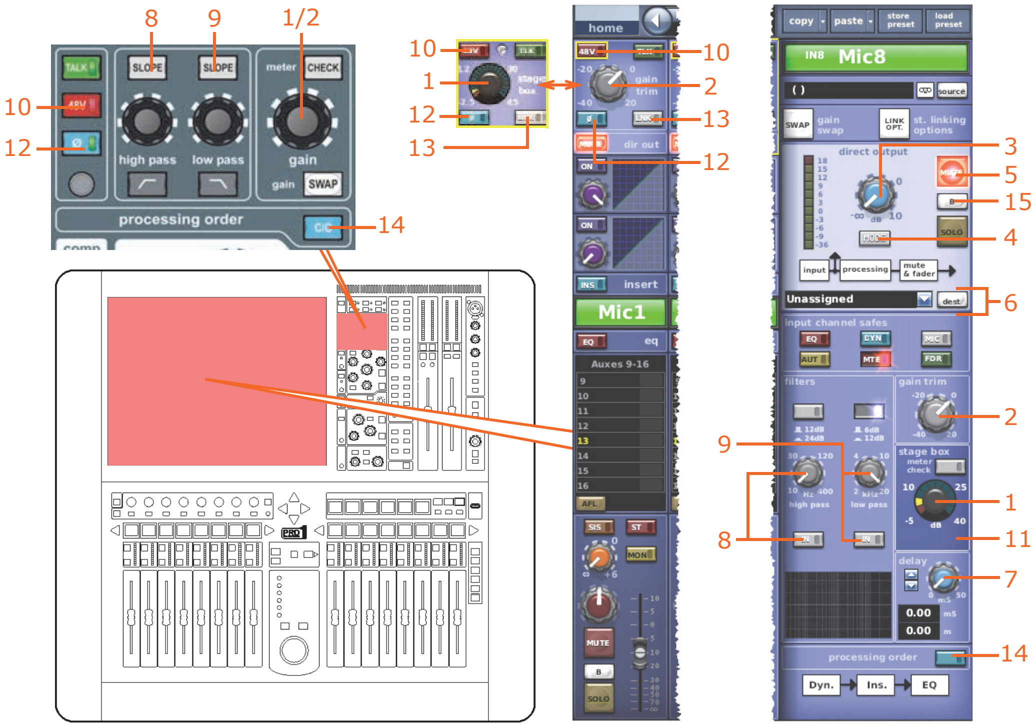
Item	Parameter	A	B	C	D	E	F	G	H
1	Input patching source	Yes*	Yes*	No	N/A	N/A	N/A	N/A	N/A

* Includes tape return and primary input sources.

Configuration



This section shows the scoped mic parameters of the input channels.



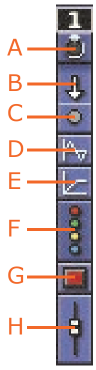
Item	*Parameter*	A	B	C	D	E	F	G	H
1	Mic gain*	N/A	Yes	Yes	N/A	N/A	N/A	N/A	N/A
2	Digital trim*	N/A	Yes	Yes	N/A	N/A	N/A	N/A	N/A
3	Direct output level	N/A	Yes	Yes	N/A	N/A	N/A	N/A	N/A
4	Direct output tap-off point	N/A	Yes	Yes	N/A	N/A	N/A	N/A	N/A
5	Direct output mute	N/A	Yes	No	N/A	N/A	N/A	Yes	N/A
6	Direct output patch destination	N/A	No	No	N/A	N/A	N/A	N/A	N/A
7	Input delay	N/A	Yes	Yes	N/A	N/A	N/A	N/A	N/A
8	Hi pass filter: slope in/out and rotary control	N/A	Yes	No	Yes	N/A	N/A	N/A	N/A
9	Low pass filter: slope in/out and rotary control	N/A	Yes	No	Yes	N/A	N/A	N/A	N/A
10	48V	N/A	Yes	Yes	N/A	N/A	N/A	N/A	N/A
11	30Hz filter** (not shown)	N/A	Yes	Yes	N/A	N/A	N/A	N/A	N/A
12	Input phase	N/A	Yes	Yes	N/A	N/A	N/A	N/A	N/A
13	Link	N/A	No	No	No	No	No	No	No

Item	Parameter	A	B	C	D	E	F	G	H
14	Processing order	N/A	**Yes**	N/A	**Yes**	N/A	N/A	N/A	N/A
15	Direct output solo B assignment	N/A	**No**	**No**	**No**	**No**	**No**	**No**	**No**

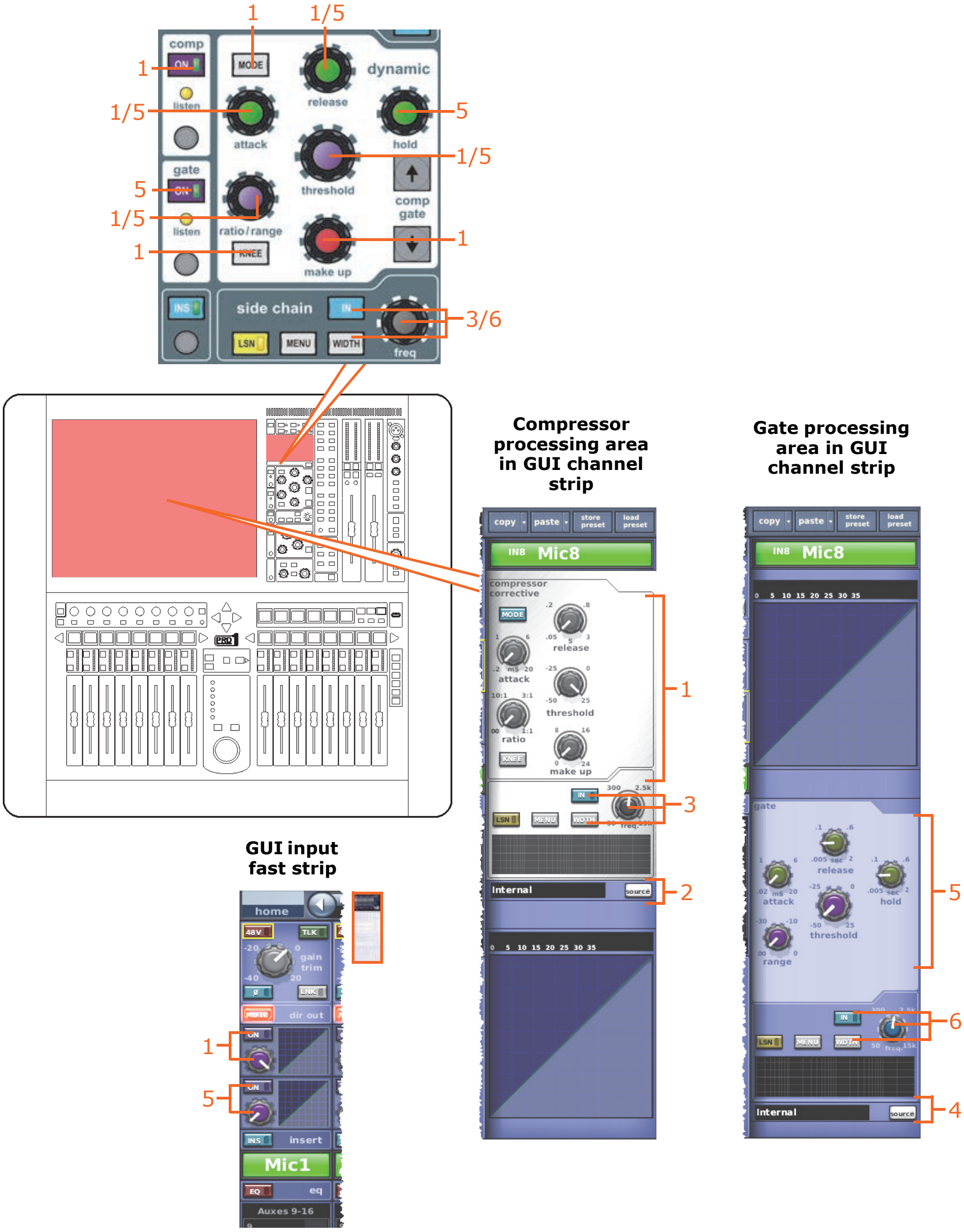
*   Depends on swap status.

** Only when sourced from a DL431 Mic Splitter.

Dynamics

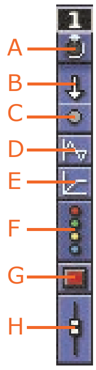


This section shows the scoped compressor and gate parameters of the input channels. Although only the corrective compressor is shown below, this is typically the same for the other compressor modes (adaptive, creative, vintage and shimmer).

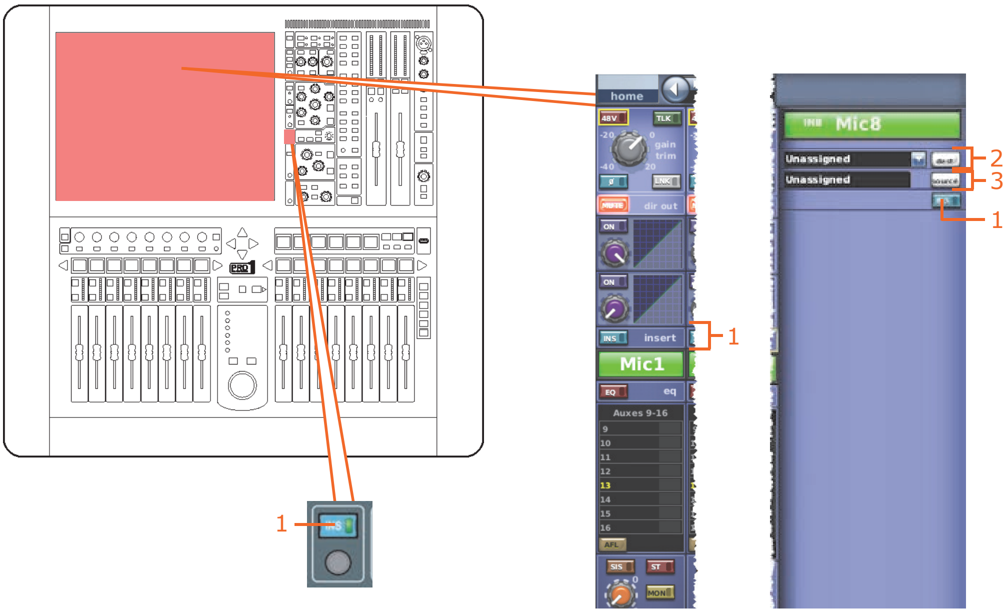


Item	*Parameter*	A	B	C	D	E	F	G	H
1	Compressor: attack, release, threshold, ratio, gain (make up), knee, mode and compressor on/off	N/A	**Yes**	N/A	N/A	**Yes**	N/A	N/A	N/A
2	Compressor sidechain source	**No**	N/A	N/A	N/A	N/A	N/A	N/A	N/A
3	Compressor sidechain: compressor sidechain in/out, frequency and width	N/A	**Yes**	N/A	N/A	**Yes**	N/A	N/A	N/A
4	Gate key in source	**No**	N/A	N/A	N/A	N/A	N/A	N/A	N/A
5	Gate: attack, release, hold, threshold, range and gate on/off	N/A	**Yes**	N/A	N/A	**Yes**	N/A	N/A	N/A
6	Gate sidechain: gate sidechain in/out, frequency and width	N/A	**Yes**	N/A	N/A	**Yes**	N/A	N/A	N/A

Insert

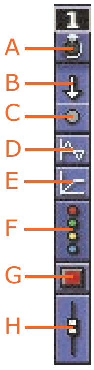


This section shows the scoped insert parameters of the input channels.

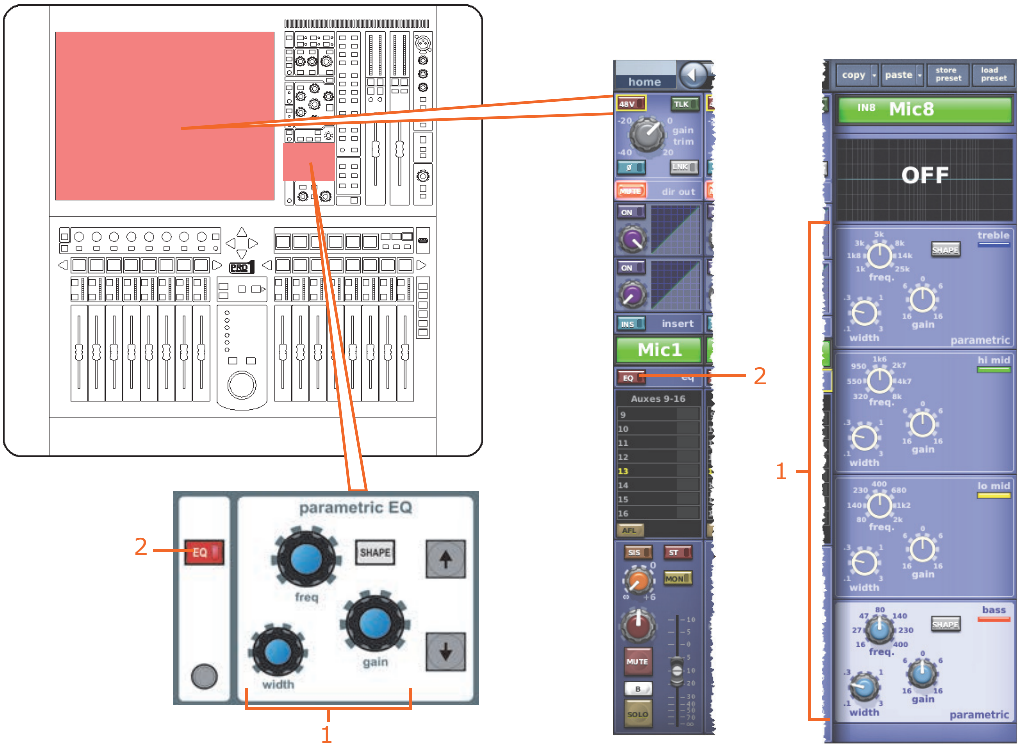


Item	*Parameter*	A	B	C	D	E	F	G	H
1	In/out	**Yes**	**Yes**	N/A	N/A	N/A	N/A	N/A	N/A
2	Insert send destination	N/A	**No**	N/A	N/A	N/A	N/A	N/A	N/A
3	Insert return source	N/A	**No**	N/A	N/A	N/A	N/A	N/A	N/A

EQ

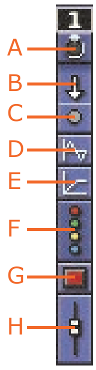


This section shows the scoped parametric EQ parameters of the input channels.

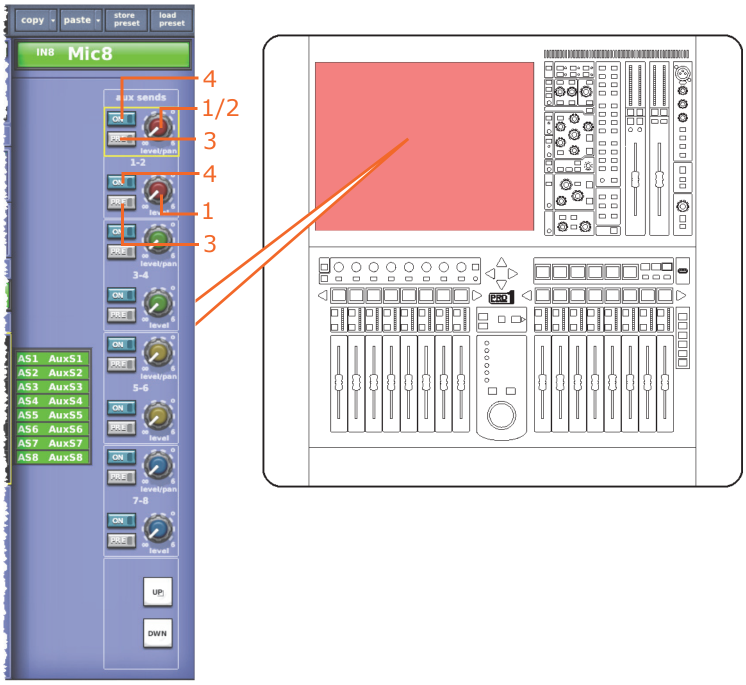


Item	*Parameter*	A	B	C	D	E	F	G	H
1	All filters: frequency, gain, width and shape (treble and bass only)	N/A	**Yes**	N/A	**Yes**	N/A	N/A	N/A	N/A
2	EQ in/out	N/A	**Yes**	N/A	**Yes**	N/A	N/A	N/A	N/A

Aux send



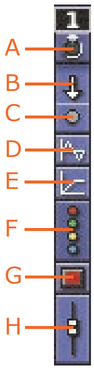
This section shows the scoped aux send parameters of the input channels. Although it only shows aux buses 1 to 8, it also applies to aux buses 9 to 16.



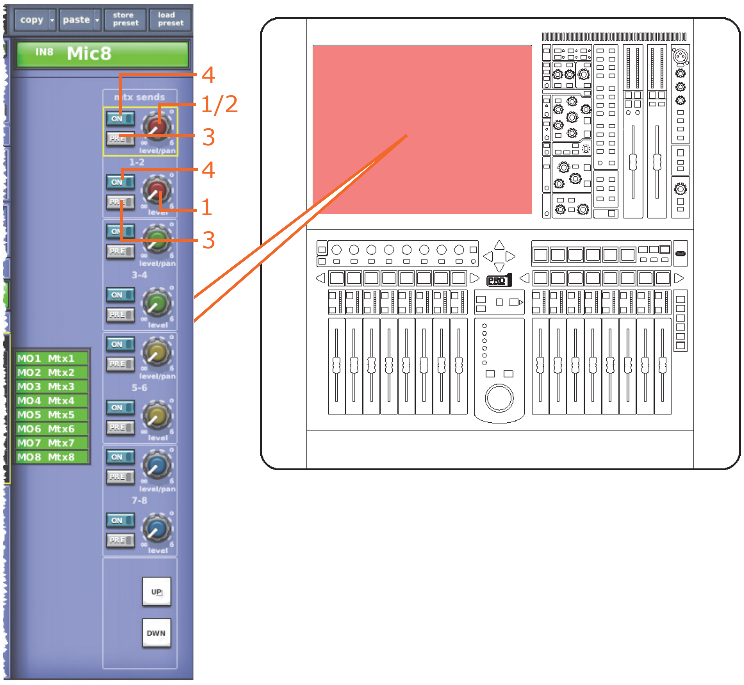
Item	*Parameter*	A	B	C	D	E	F	G	H
1	Send level	N/A	Yes	N/A	N/A	N/A	Yes	N/A	N/A
2	Send pan	N/A	Yes	N/A	N/A	N/A	Yes	N/A	N/A
3	Send pre-fader on/off	N/A	Yes	N/A	N/A	N/A	Yes	N/A	N/A
4	Send on/off	N/A	Yes	N/A	N/A	N/A	Yes	N/A	N/A

You can scope individual bus sends. In column B (**All**), all sends are affected, and in column F (**Busses**), individual sends can be scoped.

Matrix send



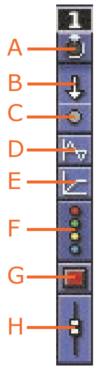
This section shows the scoped matrix send parameters of the input channels.



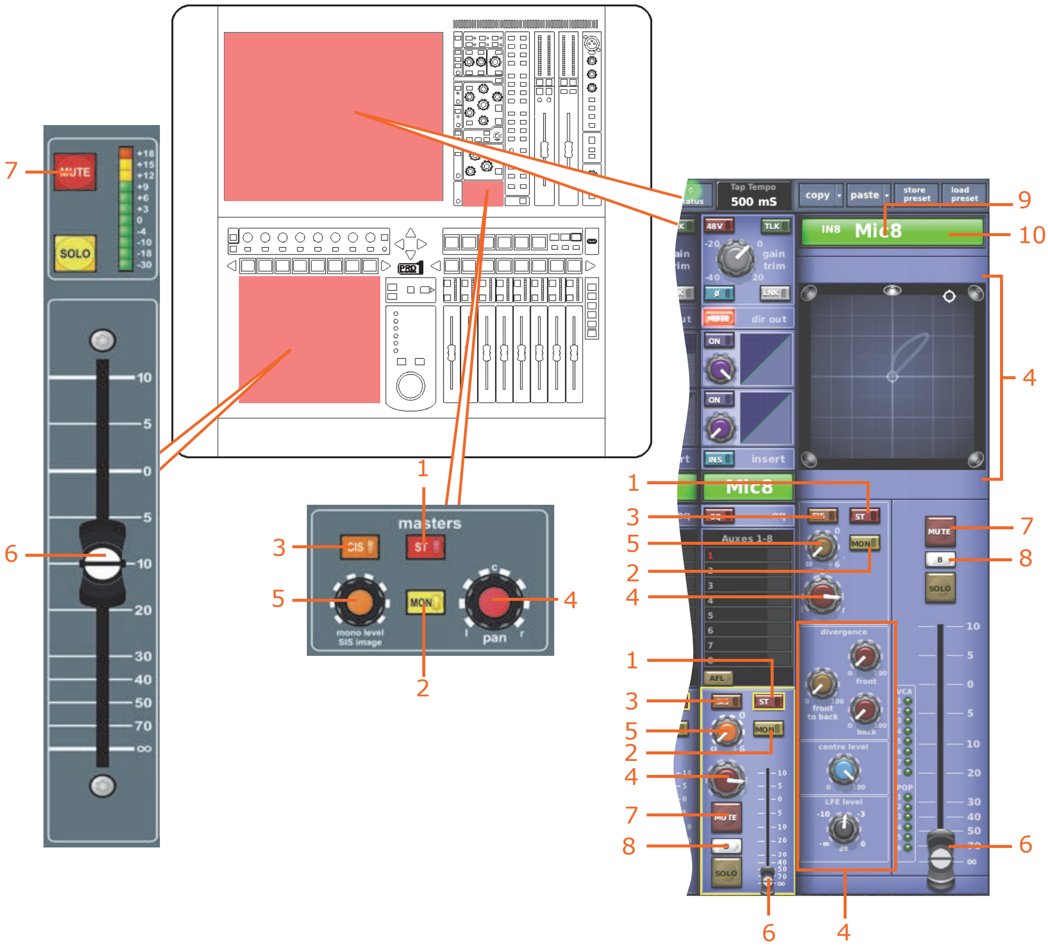
Item	*Parameter*	A	B	C	E	D	F	G	H
1	Send level	N/A	Yes	N/A	N/A	N/A	Yes	N/A	N/A
2	Send pan	N/A	Yes	N/A	N/A	N/A	Yes	N/A	N/A
3	Send pre-fader on/off	N/A	Yes	N/A	N/A	N/A	Yes	N/A	N/A
4	Send on/off	N/A	Yes	N/A	N/A	N/A	Yes	N/A	N/A

You can scope individual bus sends. In column B (**All**), all sends are affected, and in column F (**Busses**), individual sends can be scoped.

Fader



This section shows the scoped fader and master routing parameters of the input channels.

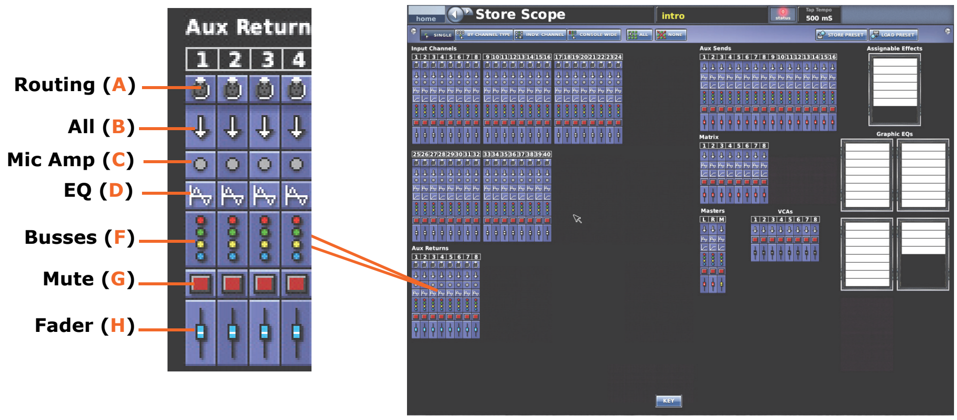


Item	Parameter	A	B	C	D	E	F	G	H
1	Stereo routing	N/A	Yes	N/A	N/A	N/A	N/A	N/A	Yes
2	Mono routing	N/A	Yes	N/A	N/A	N/A	N/A	N/A	Yes
3	SIS select (required for surround panning)	N/A	Yes	N/A	N/A	N/A	N/A	N/A	Yes
4	Pan (includes all surround sound parameters)	N/A	Yes	N/A	N/A	N/A	N/A	N/A	Yes
5	Mono level/SIS pan	N/A	Yes	N/A	N/A	N/A	N/A	N/A	Yes
6	Fader position	N/A	Yes	N/A	N/A	N/A	N/A	N/A	Yes
7	Channel mute	N/A	Yes	N/A	N/A	N/A	N/A	Yes	No
8	Solo B assignment	N/A	No	N/A	N/A	N/A	N/A	N/A	No

Item	Parameter	A	B	C	D	E	F	G	H
9	Channel name	N/A	Yes	N/A	N/A	N/A	N/A	N/A	No
10	Channel colour	N/A	Yes	N/A	N/A	N/A	N/A	N/A	No

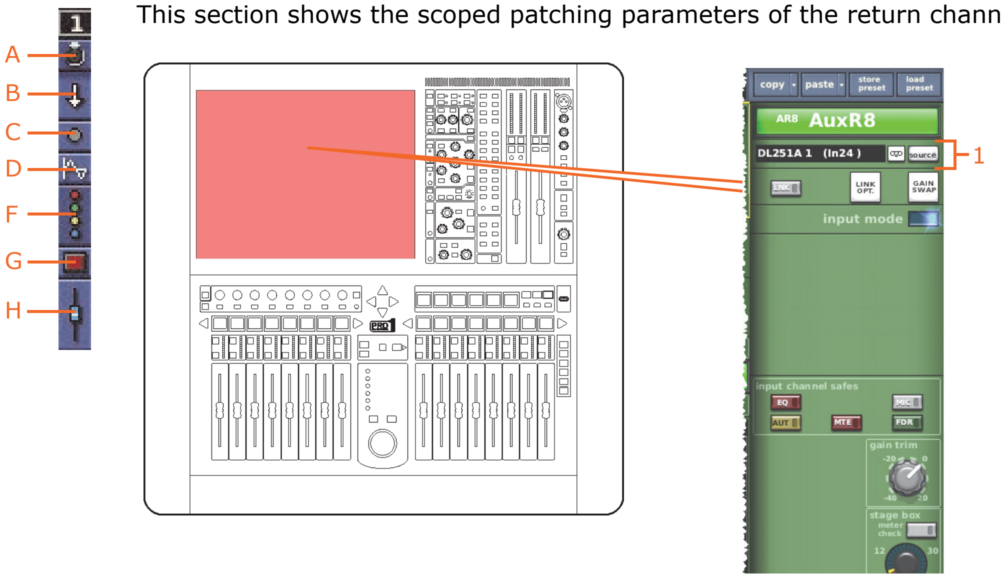
Returns (Aux Returns)

Each scope screen has eight returns in the **Aux Returns** section.



Patching

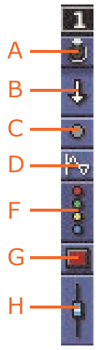
This section shows the scoped patching parameters of the return channels.



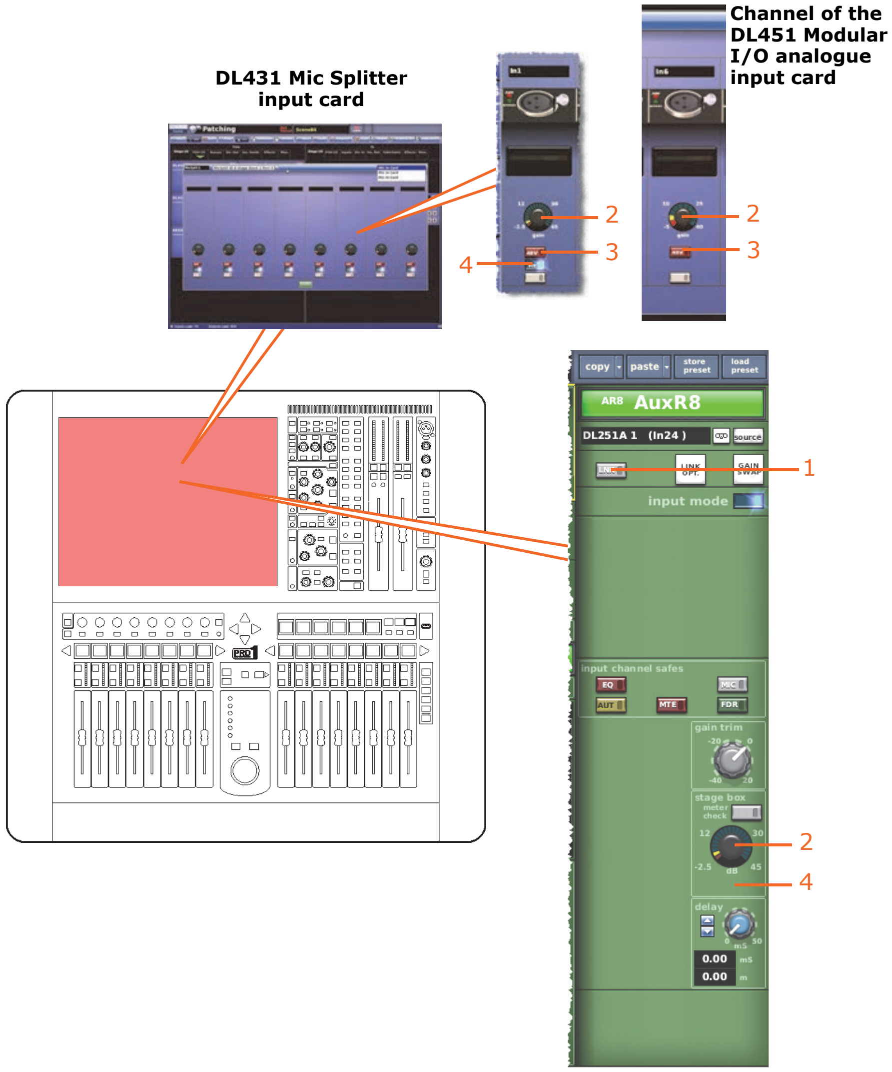
Item	*Parameter*	A	B	C	D	F	G	H
1	Input patching source	N/A	**Yes***	N/A	N/A	N/A	N/A	N/A

* Includes tape return and primary input sources.

Configuration



This section shows the scoped areas of the mic section in the return channels.



Item	*Parameter*	A	B	C	D	F	G	H
1	Link	N/A	No	No	No	No	No	No
2	Gain of remote amplifier	N/A	Yes	Yes	N/A	N/A	N/A	N/A
3	48V phantom gain	N/A	Yes	Yes	N/A	N/A	N/A	N/A
4	30Hz filter* (not shown)	N/A	Yes	Yes	N/A	N/A	N/A	N/A

* Only when sourced from a DL431 Mic Splitter.

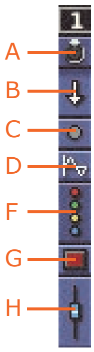
Dynamics

Not applicable.

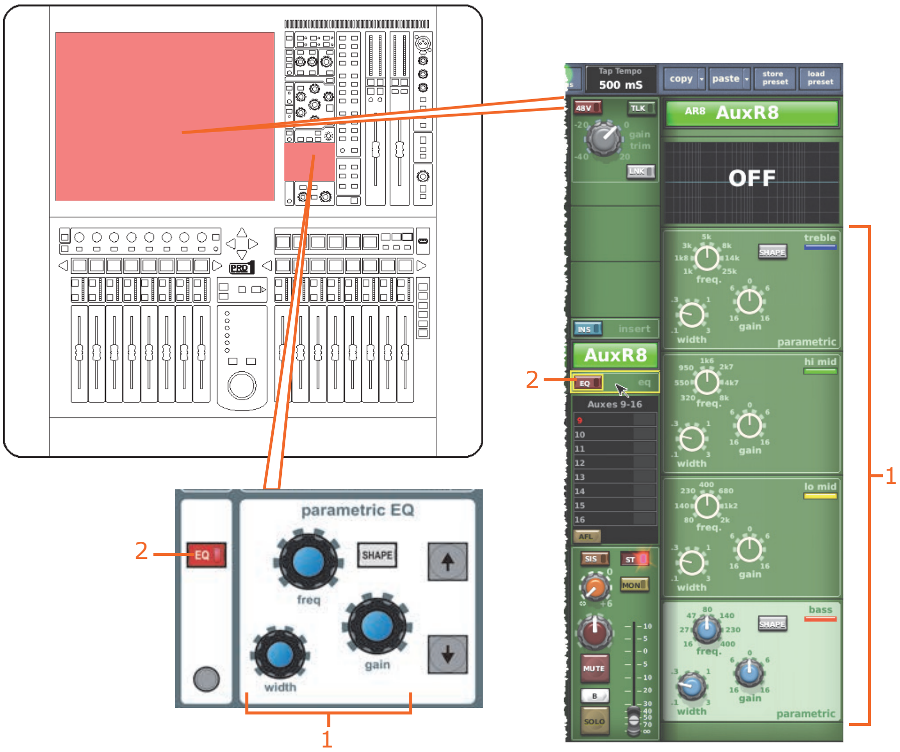
Insert

Not applicable.

EQ



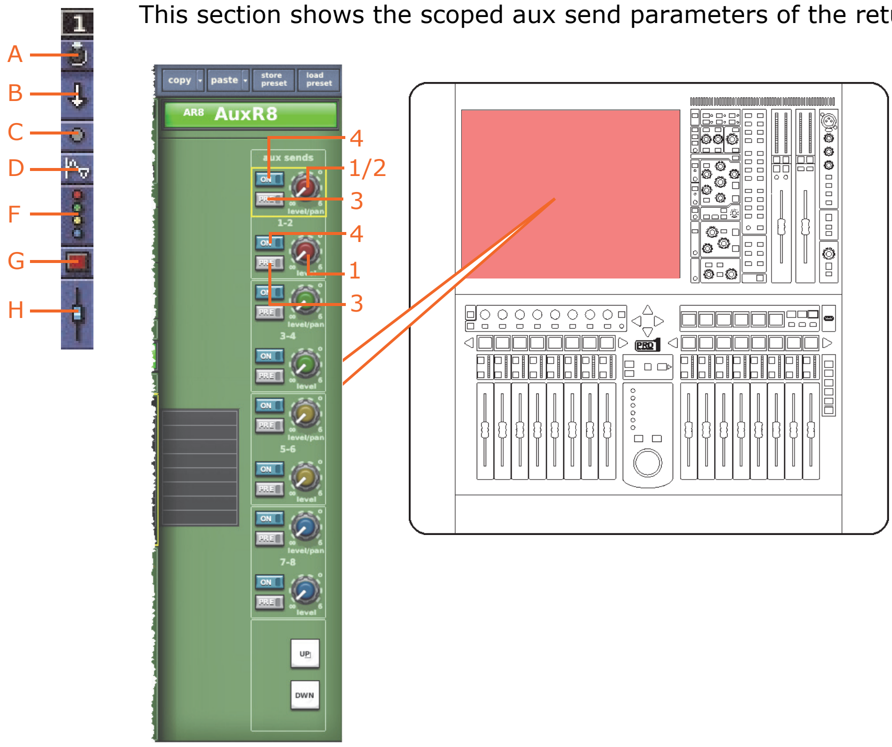
This section shows the scoped parametric EQ parameters of the input channels.



Item	Parameter	A	B	C	D	F	G	H
1	All filters: frequency, gain, width and shape	N/A	Yes	N/A	Yes	N/A	N/A	N/A
2	EQ in/out	N/A	Yes	N/A	Yes	N/A	N/A	N/A

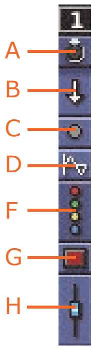
Aux send

This section shows the scoped aux send parameters of the return channels.

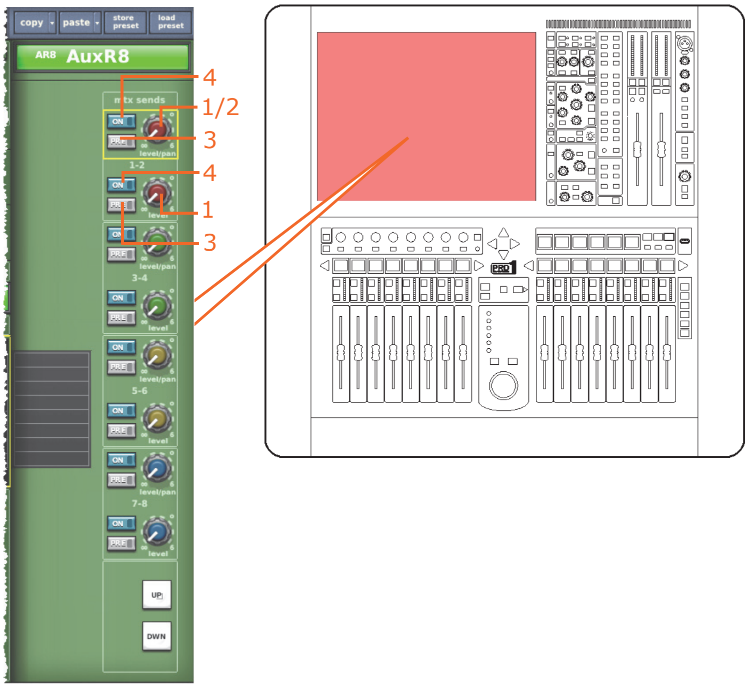


Item	Parameter	A	B	C	D	F	G	H
1	Send level	N/A	Yes	N/A	N/A	Yes	N/A	N/A
2	Send pan	N/A	Yes	N/A	N/A	Yes	N/A	N/A
3	Send pre-fader on/off	N/A	Yes	N/A	N/A	Yes	N/A	N/A
4	Send on/off	N/A	Yes	N/A	N/A	Yes	N/A	N/A

Matrix send

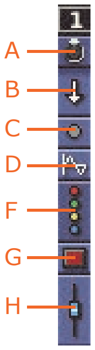


This section shows the scoped matrix send parameters in the return channels.

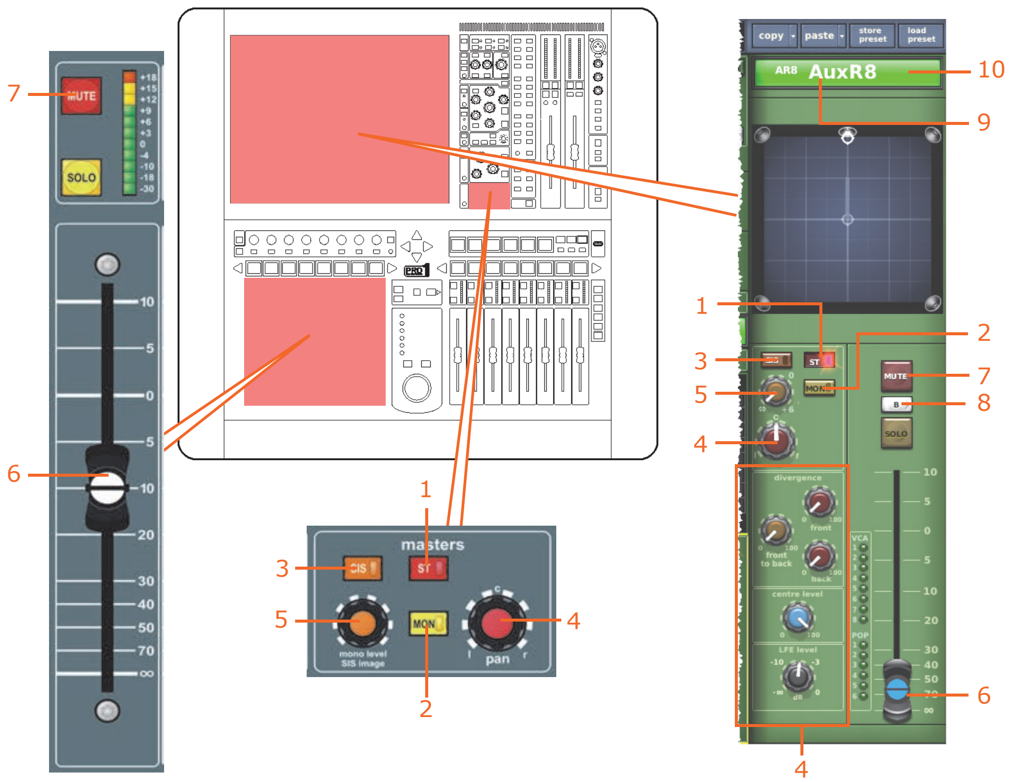


Item	Parameter	A	B	C	D	F	G	H
1	Send level	N/A	Yes	N/A	N/A	Yes	N/A	N/A
2	Send pan	N/A	Yes	N/A	N/A	Yes	N/A	N/A
3	Send pre-fader on/off	N/A	Yes	N/A	N/A	Yes	N/A	N/A
4	Send on/off	N/A	Yes	N/A	N/A	Yes	N/A	N/A

Fader



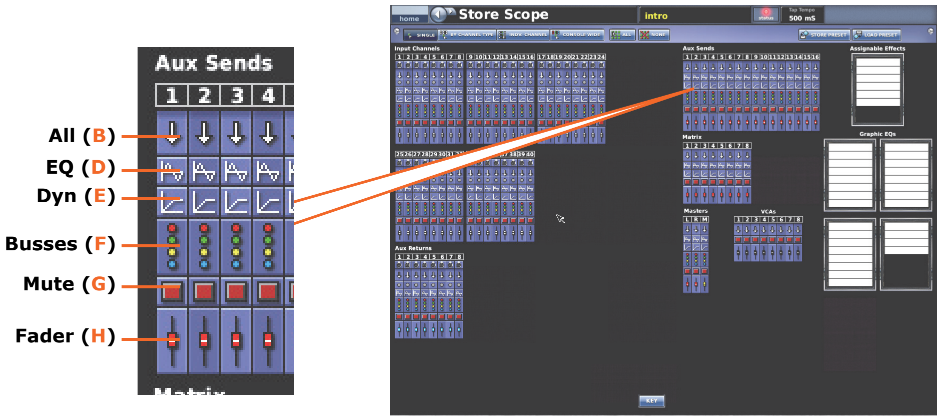
This section shows the scoped fader and master routing parameters of the return channels.



Item	Parameter	A	B	C	D	F	G	H
1	Stereo routing	N/A	Yes	N/A	N/A	N/A	N/A	Yes
2	Mono routing	N/A	Yes	N/A	N/A	N/A	N/A	Yes
3	SIS select (required for surround panning)	N/A	Yes	N/A	N/A	N/A	N/A	Yes
4	Pan (includes all surround sound parameters)	N/A	Yes	N/A	N/A	N/A	N/A	Yes
5	Mono level/SIS pan	N/A	Yes	N/A	N/A	N/A	N/A	Yes
6	Fader position	N/A	Yes	N/A	N/A	N/A	N/A	Yes
7	Channel mute	N/A	Yes	N/A	N/A	N/A	Yes	N/A
8	Solo B assignment	N/A	No	N/A	N/A	N/A	N/A	N/A
9	Channel name	N/A	Yes	N/A	N/A	N/A	N/A	N/A
10	Channel colour	N/A	Yes	N/A	N/A	N/A	N/A	N/A

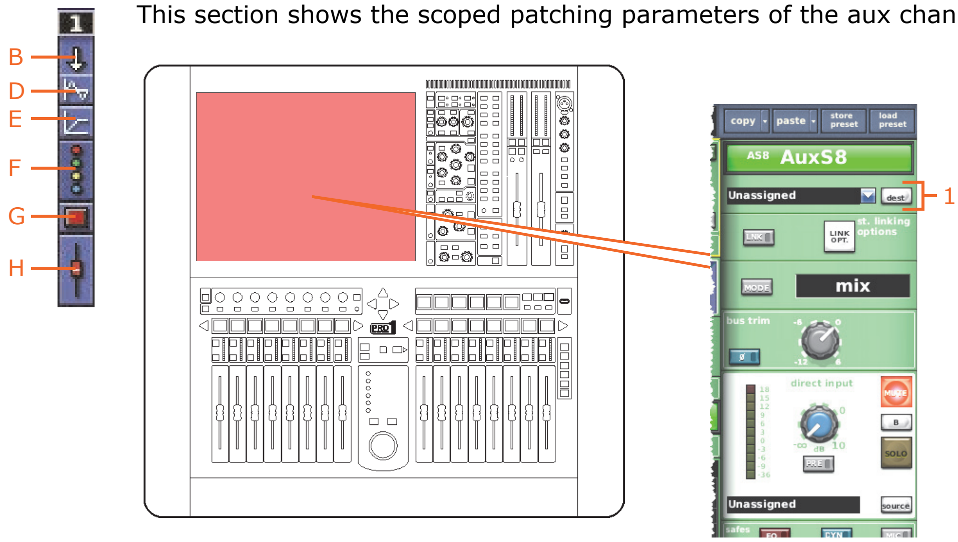
Auxes (Aux Sends)

Each scope screen has 16 auxes in the **Aux Sends** section.



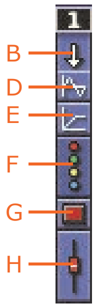
Patching

This section shows the scoped patching parameters of the aux channels.

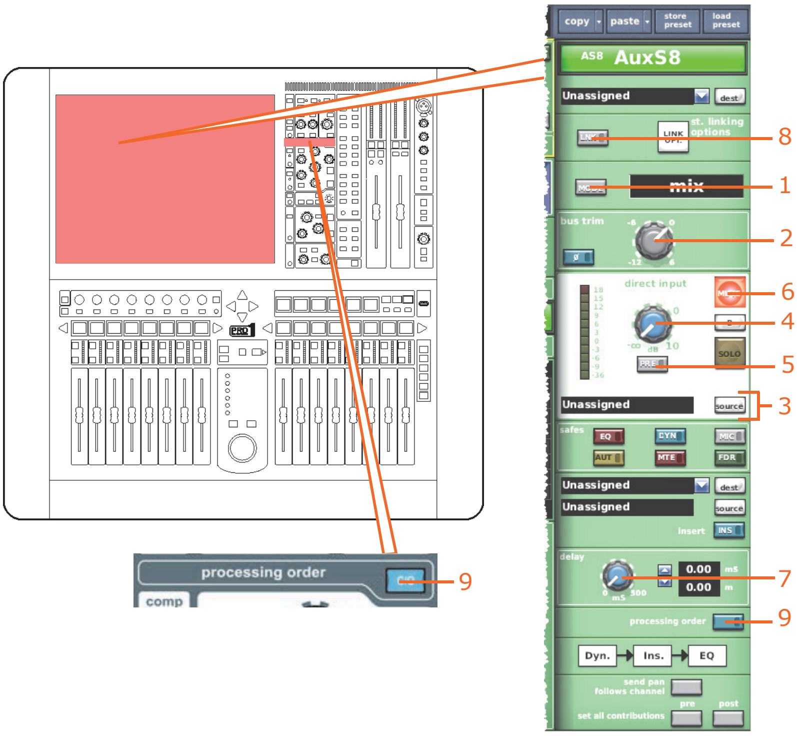


Item	Parameter	B	D	E	F	G	H
1	Output patching	No	N/A	N/A	N/A	N/A	N/A

Configuration



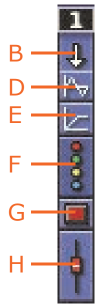
This section shows the scoped configuration and direct input parameters of the aux channels.



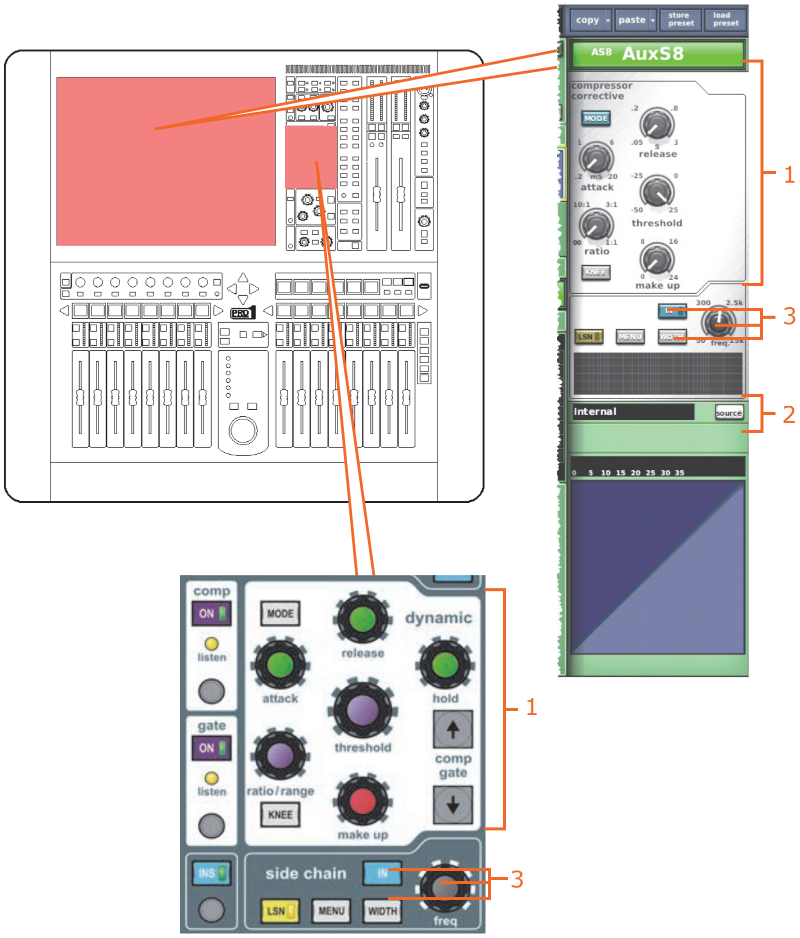
Item	Parameter	B	D	E	F	G	H
1	Bus mode	Yes	N/A	N/A	N/A	N/A	Yes
2	Bus trim	Yes	N/A	N/A	N/A	N/A	Yes
3	Direct input source	Yes*	N/A	N/A	N/A	N/A	N/A
4	Direct input level	Yes	N/A	N/A	N/A	N/A	N/A
5	Direct input pre-/post-	Yes	N/A	N/A	N/A	N/A	N/A
6	Direct input mute	Yes	N/A	N/A	N/A	Yes	N/A
7	Delay	Yes	N/A	N/A	N/A	N/A	Yes
8	Link	No	No	No	No	No	No
9	Processing order	Yes	Yes	N/A	N/A	N/A	N/A

* Only when automate patching is on.

Dynamics

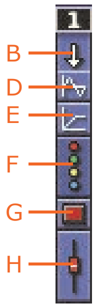


This section shows the scoped compressor parameters of the aux channels. Although only the corrective compressor is shown below, this is typically the same for the other compressor modes (adaptive, creative, vintage and shimmer).

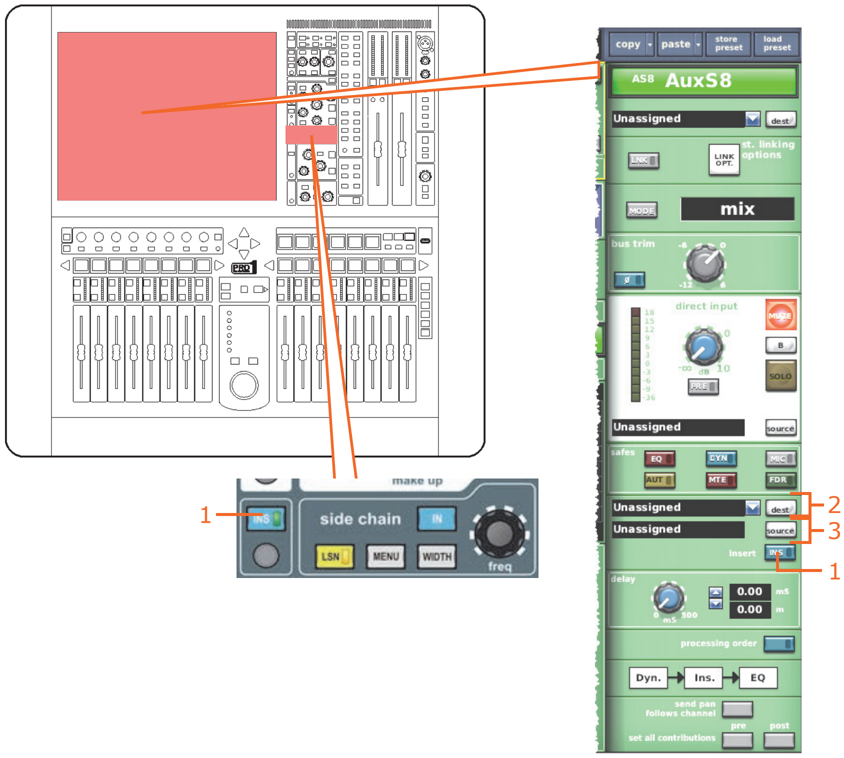


Item	Parameter	B	D	E	F	G	H
1	Compressor: attack, release, threshold, ratio, make up, knee and mode	Yes	N/A	Yes	N/A	N/A	N/A
2	Sidechain source	N/A	N/A	No	N/A	N/A	N/A
3	Compressor sidechain: compressor sidechain in/out, freq and width	Yes	N/A	Yes	N/A	N/A	N/A

Insert

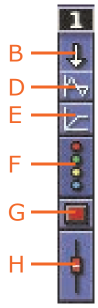


This section shows the scoped insert parameters of the aux channels.

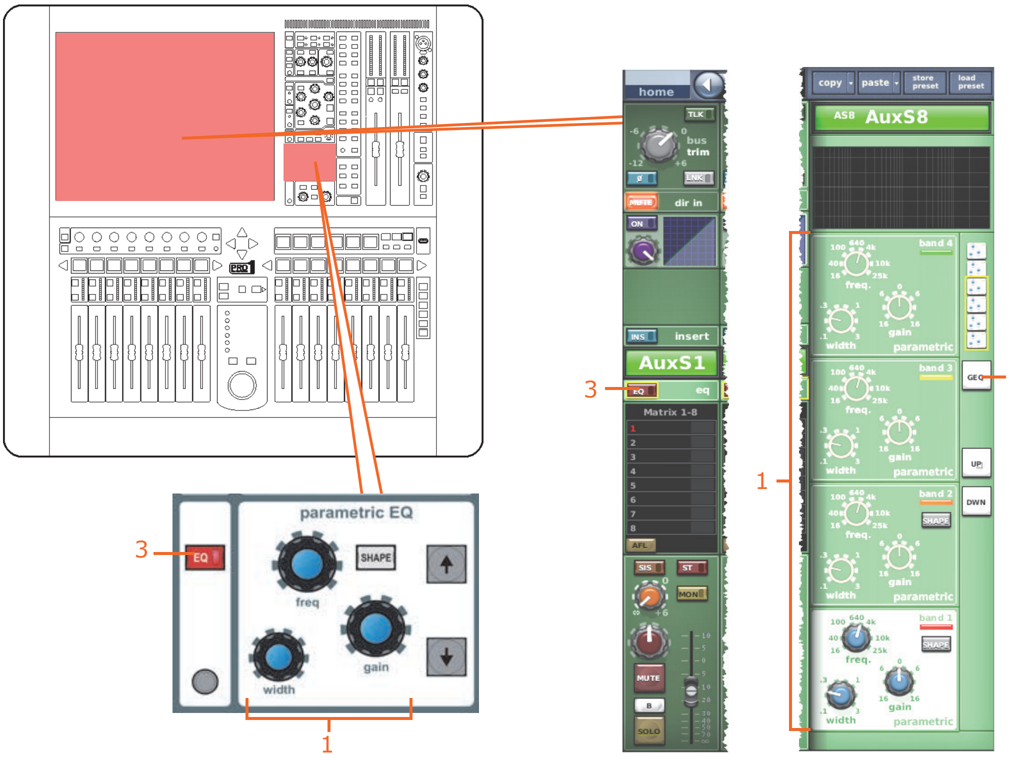


Item	*Parameter*	B	D	E	F	G	H
1	Insert in/out	**Yes**	N/A	N/A	N/A	N/A	N/A
2	Insert send destination	**No**	N/A	N/A	N/A	N/A	N/A
3	Insert return source	**No**	N/A	N/A	N/A	N/A	N/A

EQ



This section shows the scoped parameters in the EQ section of the aux channels.



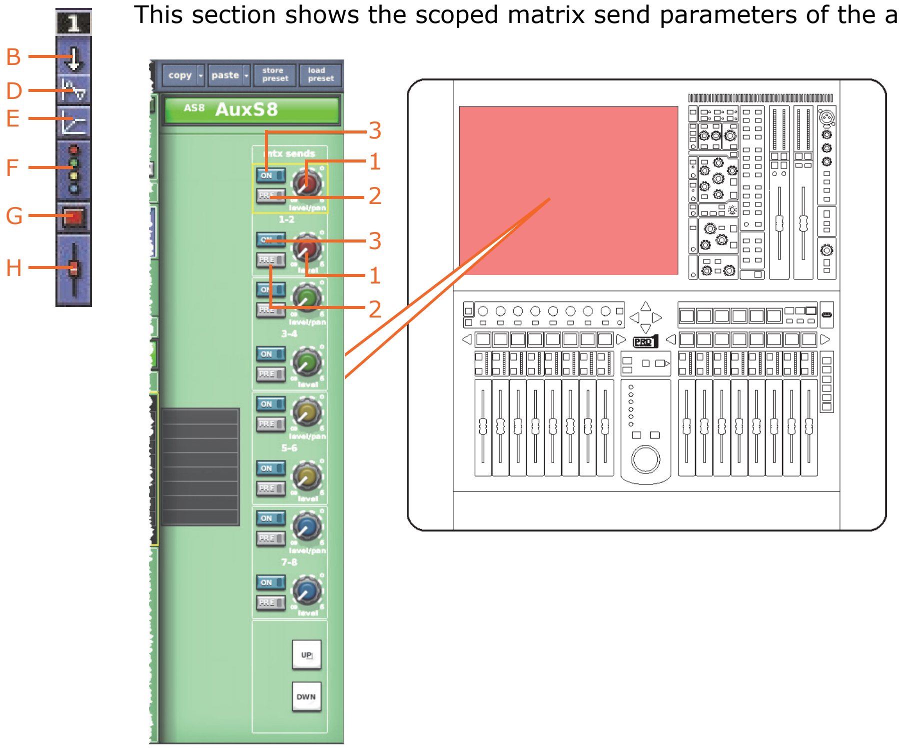
Item	Parameter	B	D	E	F	G	H
1	All PEQ filters (all six bands): frequency, gain, width and shape (bands 1, 2 and 6 only)	Yes	Yes	N/A	N/A	N/A	N/A
2	Parametric/Graphic type	No	No	N/A	N/A	N/A	N/A
3	EQ in/out	Yes	Yes	N/A	N/A	N/A	N/A

Aux send

Not applicable.

Matrix send

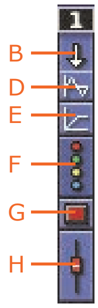
This section shows the scoped matrix send parameters of the aux channels.



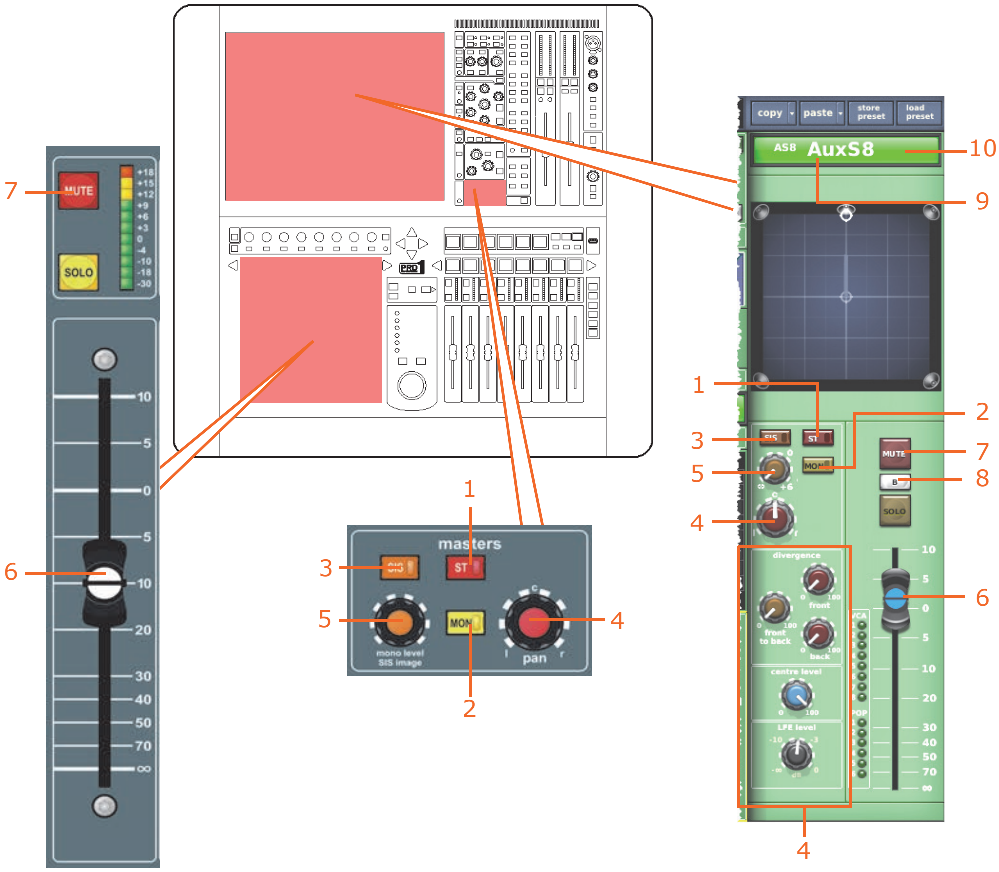
Item	Parameter	B	D	E	F	G	H
1	Send level	**Yes**	N/A	N/A	**Yes**	N/A	N/A
2	Send pre-fader on/off	**Yes**	N/A	N/A	**Yes**	N/A	N/A
3	Send on/off	**Yes**	N/A	N/A	**Yes**	N/A	N/A

You can scope individual bus sends. In column **B** (**All**), all sends are affected, and in column **F** (**Busses**), individual sends can be scoped.

Fader



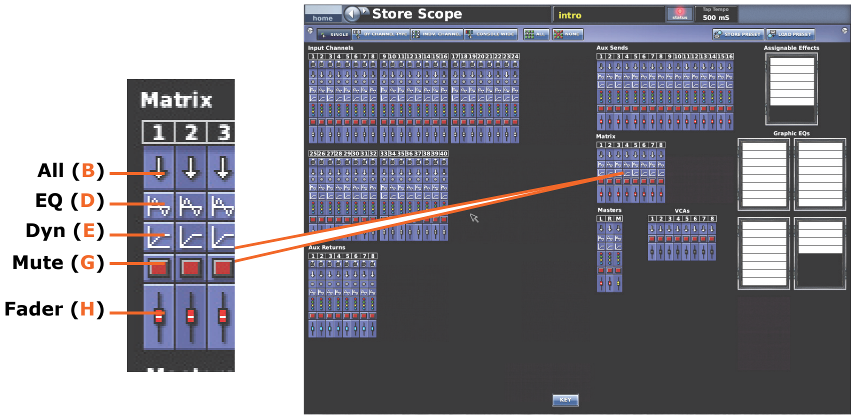
This section shows the scoped fader and master routing parameters of the aux channels.



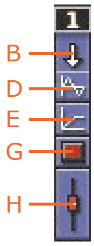
Item	Parameter	B	D	E	F	G	H
1	Stereo routing	Yes	N/A	N/A	N/A	N/A	Yes
2	Mono routing	Yes	N/A	N/A	N/A	N/A	Yes
3	SIS select (required for surround panning)	Yes	N/A	N/A	N/A	N/A	Yes
4	Pan (includes all surround sound parameters)	Yes	N/A	N/A	N/A	N/A	Yes
5	Mono level/SIS pan	Yes	N/A	N/A	N/A	N/A	Yes
6	Fader position	Yes	N/A	N/A	N/A	N/A	Yes
7	Channel mute	Yes	N/A	N/A	N/A	Yes	No
8	Solo B assignment	No	N/A	N/A	N/A	N/A	No
9	Channel name	Yes	N/A	N/A	N/A	N/A	No
10	Channel colour	Yes	N/A	N/A	N/A	N/A	No

Matrices

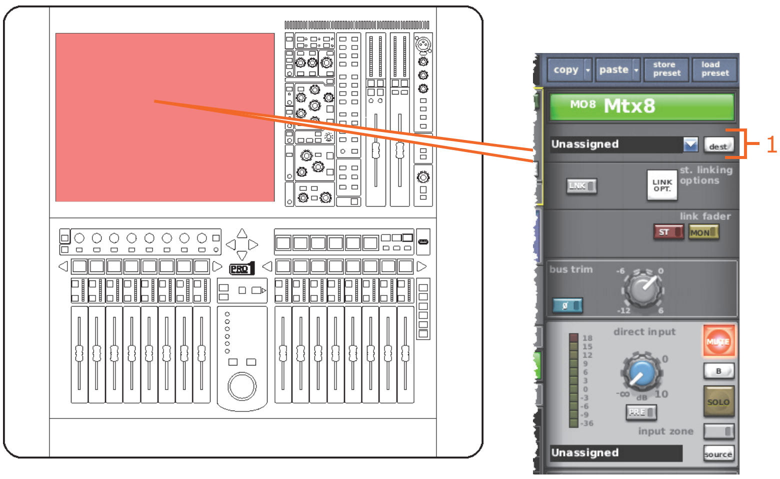
Each scope screen has eight matrices in the **Matrix** section.



Patching

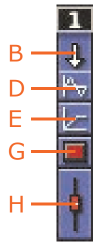


This section shows the scoped patching parameters of the matrix channels.

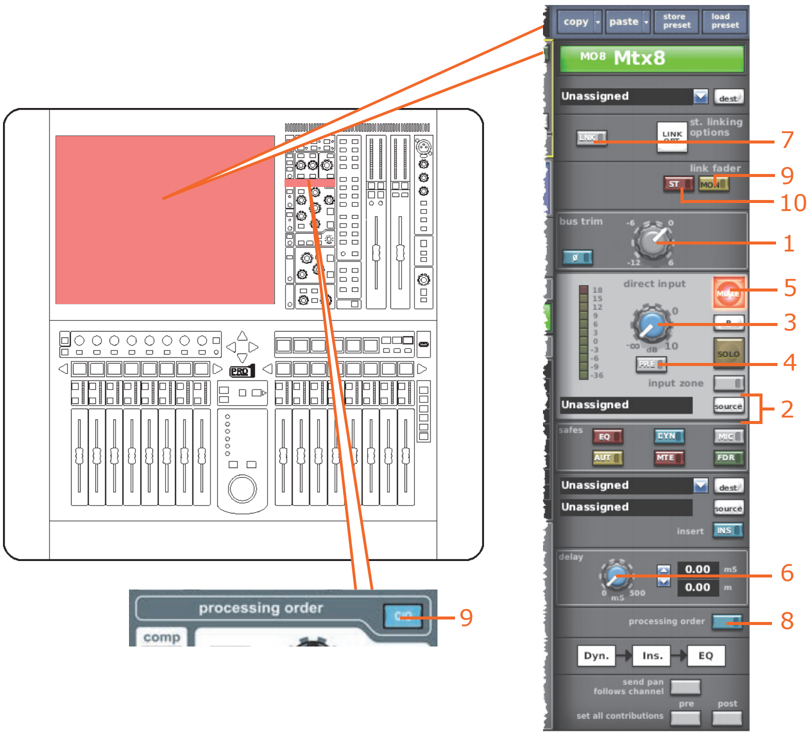


Item	Parameter	B	D	E	G	H
1	Output patching	No	N/A	N/A	N/A	N/A

Configuration



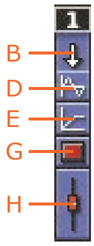
This section shows the scoped configuration and direct input parameters of the matrix channels.



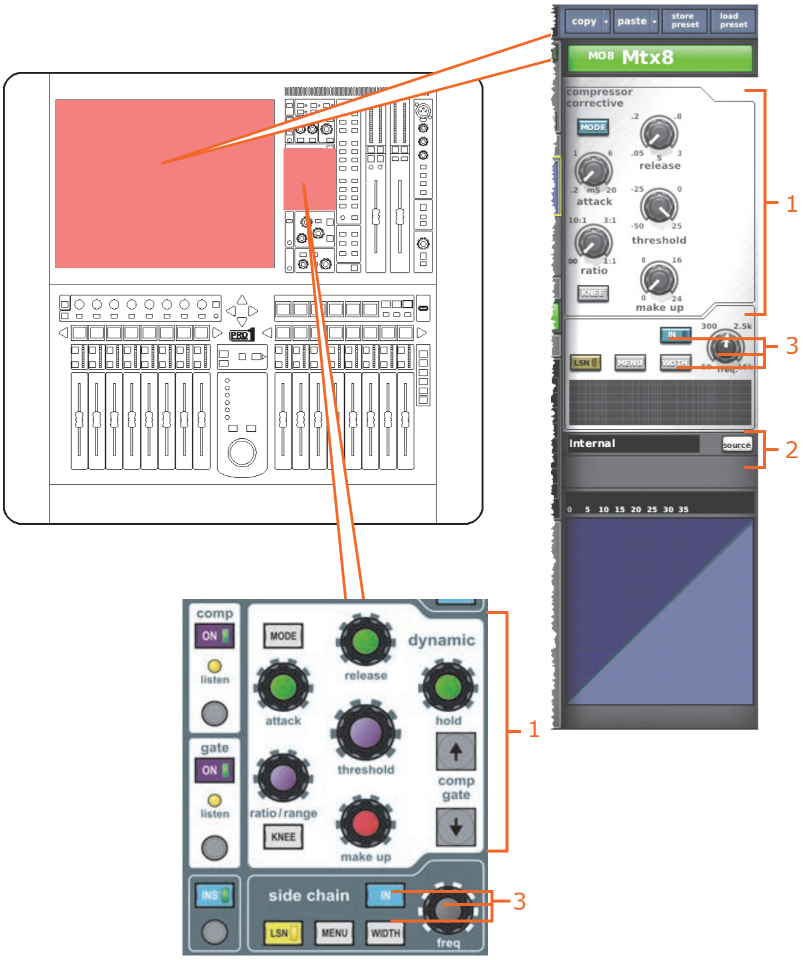
Item	Parameter	B	D	E	G	H
1	Bus trim	Yes	N/A	N/A	N/A	Yes
2	Direct input source	Yes	N/A	N/A	N/A	N/A
3	Direct input level	Yes	N/A	N/A	N/A	N/A
4	Direct input pre-/post-	Yes	N/A	N/A	N/A	N/A
5	Direct input mute	Yes	N/A	N/A	Yes	N/A
6	Delay	Yes	N/A	N/A	N/A	Yes
7	Link	No	No	No	No	No
8	Processing order	Yes	Yes	N/A	N/A	N/A
9	Link fader mono	Yes	N/A	N/A	N/A	Yes
10	Link fader stereo	Yes	N/A	N/A	N/A	Yes

* Only when automate patching is on.

Dynamics

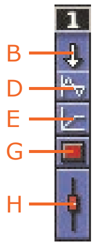


This section shows the scoped compressor parameters of the matrix channels. Although only the corrective compressor is shown below, this is typically the same for the other compressor modes (adaptive, creative, vintage and shimmer).

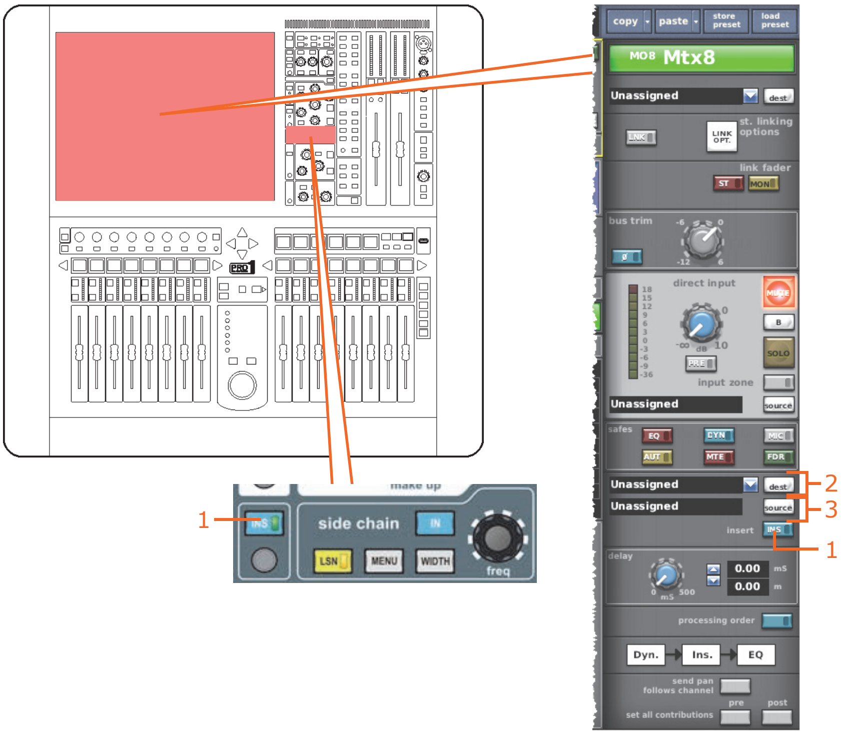


Item	Parameter	B	D	E	G	H
1	Compressor: attack, release, threshold, ratio, make up, knee and mode	Yes	N/A	Yes	N/A	N/A
2	Sidechain source	No	N/A	No	N/A	N/A
3	Compressor sidechain: compressor sidechain in/out, frequency and width	Yes	N/A	Yes	N/A	N/A

Insert

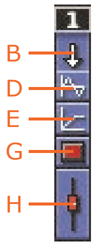


This section shows the scoped insert parameters of the matrix channels.

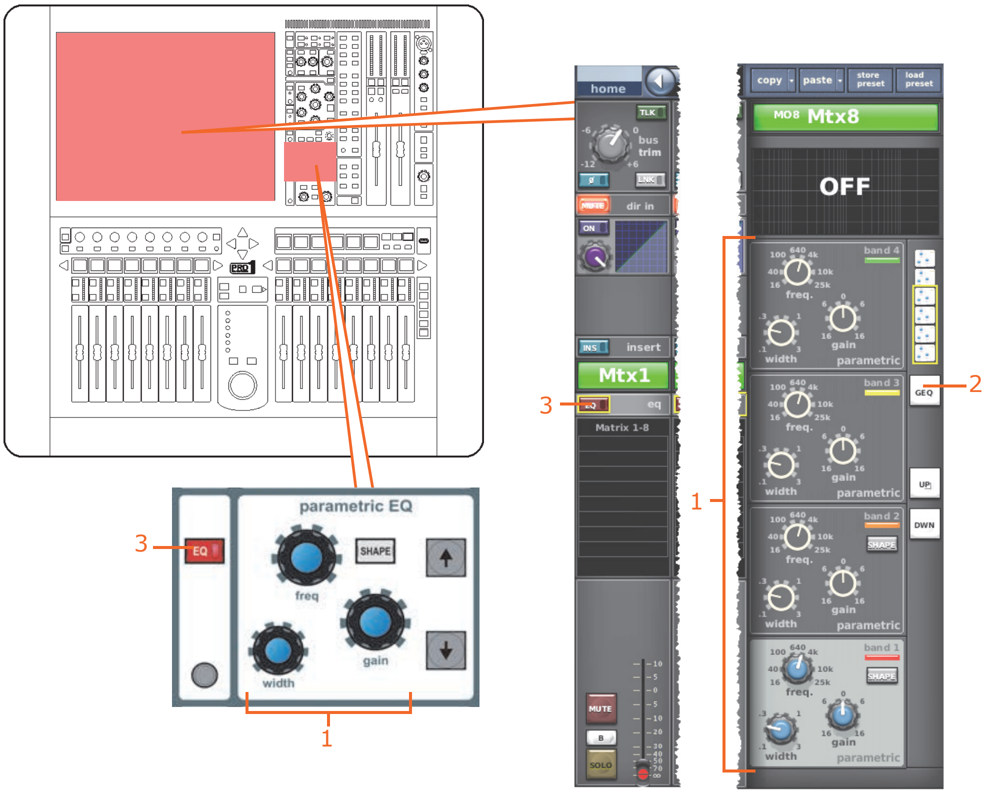


Item	Parameter	B	D	E	G	H
1	Insert in/out	Yes	N/A	N/A	N/A	N/A
2	Insert send destination	No	N/A	N/A	N/A	N/A
3	Insert return source	No	N/A	N/A	N/A	N/A

EQ



This section shows the scoped EQ parameters of the matrix channels.



Item	*Parameter*	B	D	E	G	H
1	All PEQ filters (all six bands): frequency, gain, width, shape (bands 1, 2 and 6 only)	Yes	Yes	N/A	N/A	N/A
2	Parametric/graphic type	No	No	N/A	N/A	N/A
3	EQ in/out	Yes	Yes	N/A	N/A	N/A

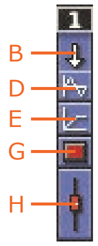
Aux send

Not applicable.

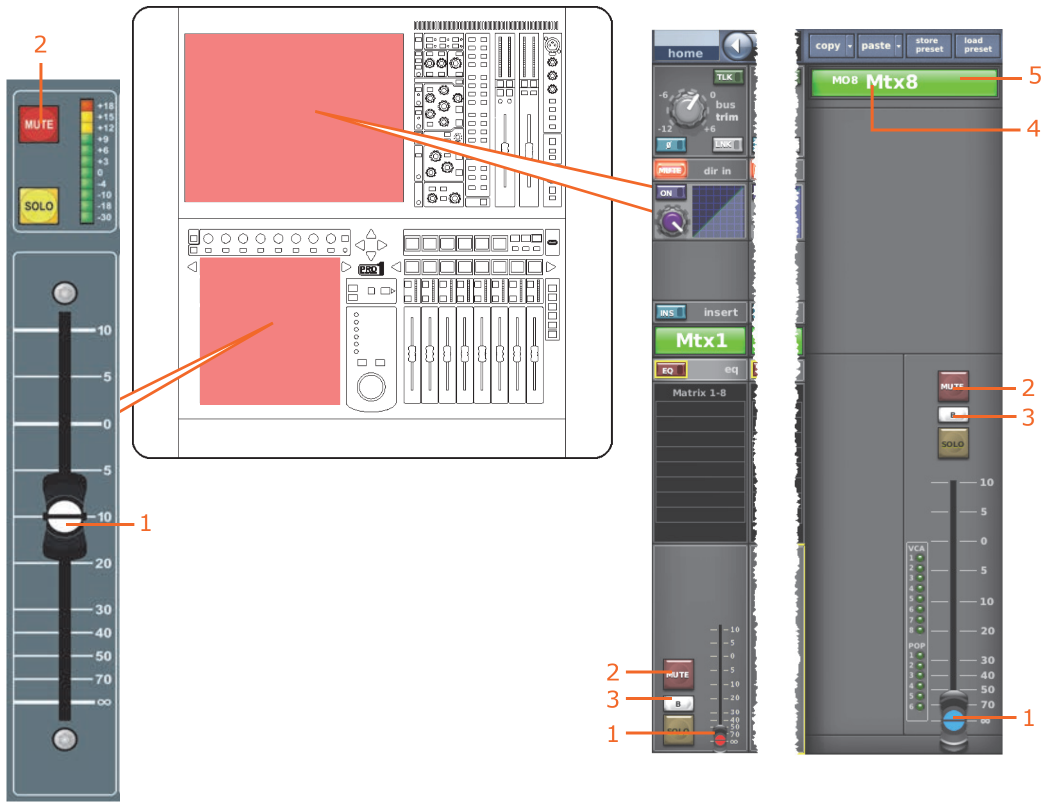
Matrix send

Not applicable.

Fader



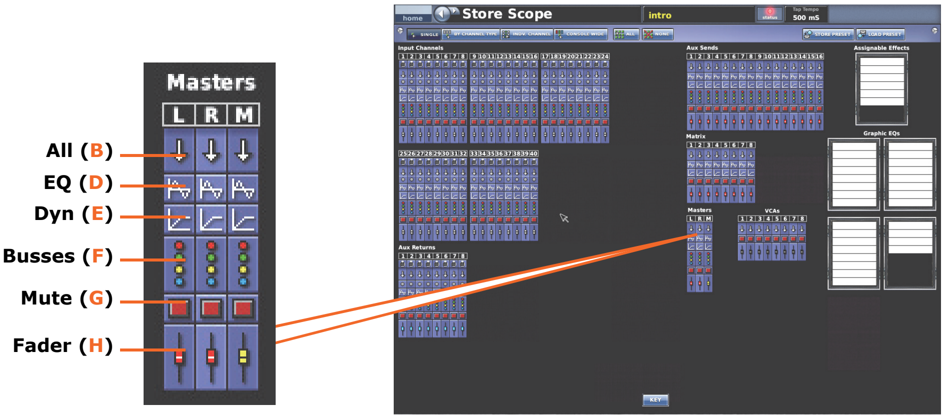
This section shows the scoped fader parameters of the matrix channels.



Item	*Parameter*	B	D	E	G	H
1	Fader position	Yes	N/A	N/A	N/A	Yes
2	Channel mute	Yes	N/A	N/A	Yes	No
3	Solo B assignment	No	N/A	N/A	N/A	No
4	Channel name	Yes	N/A	N/A	N/A	No
5	Channel colour	Yes	N/A	N/A	N/A	No

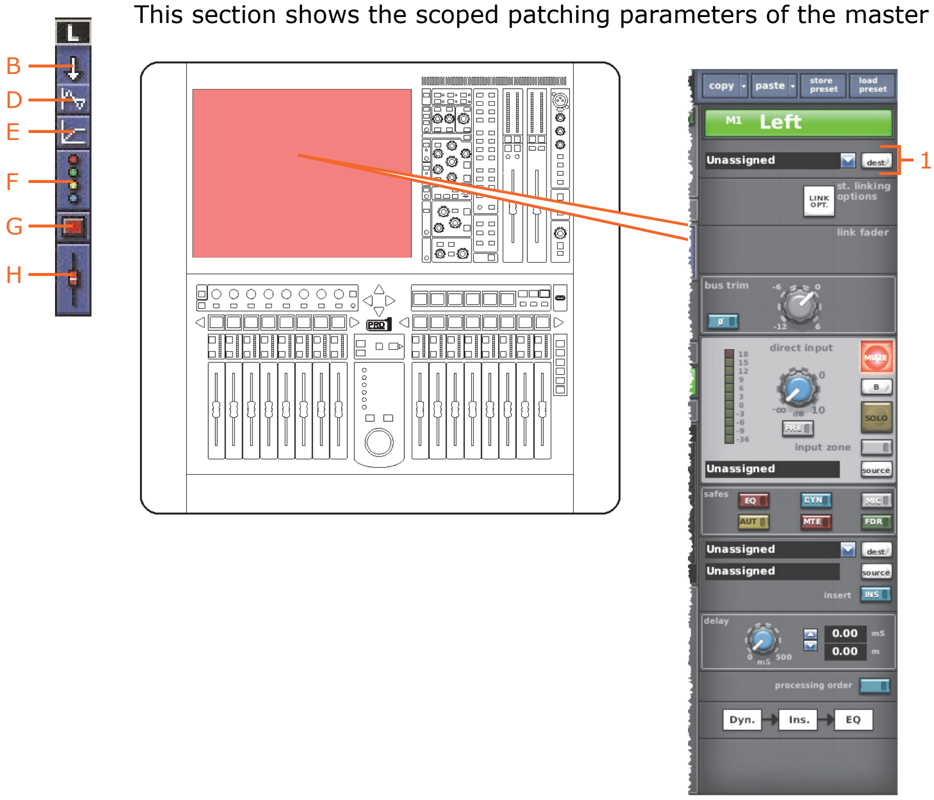
Masters

Each scope screen has three master channels (stereo left and right, and mono) in the **Masters** section.



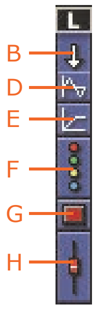
Patching

This section shows the scoped patching parameters of the master channels.

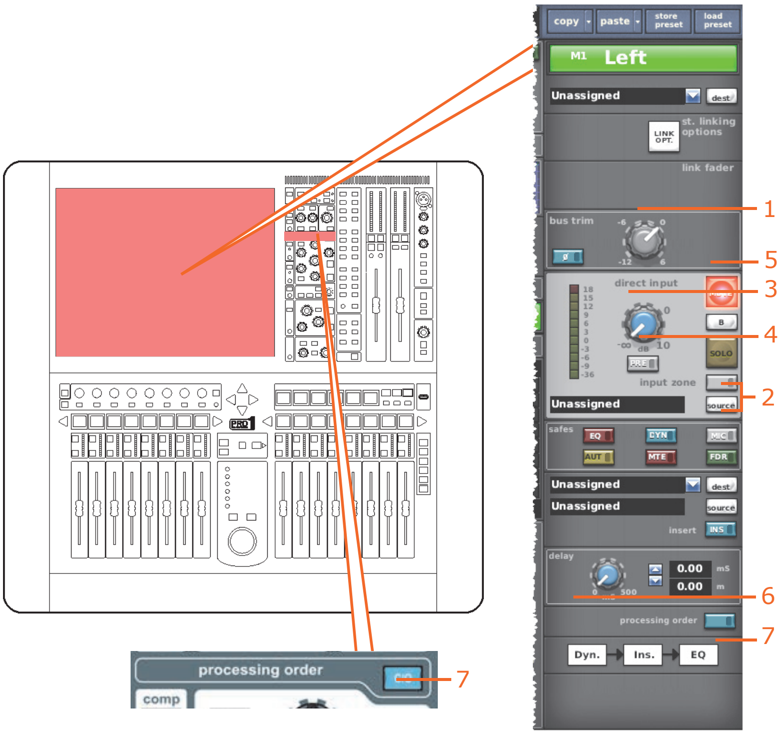


Item	*Parameter*	B	D	E	F	G	H
1	Output patching	**No**	N/A	N/A	N/A	N/A	N/A

Configuration



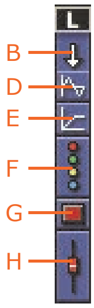
This section shows the scoped configuration and direct input section parameters of the master channels.



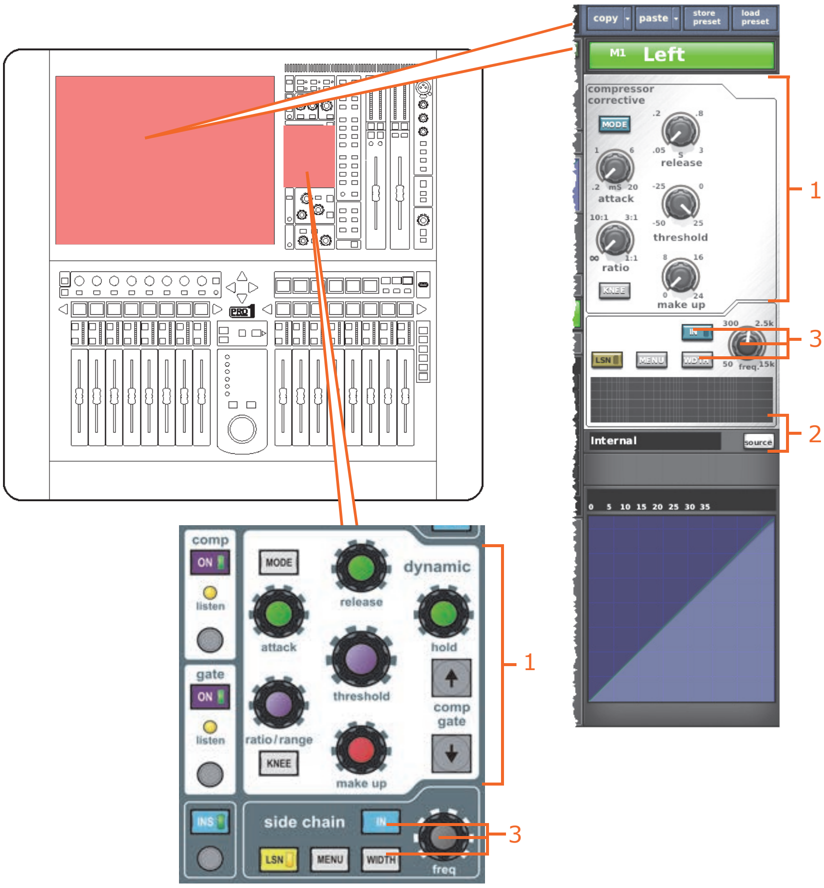
Item	Parameter	B	D	E	F	G	H
1	Bus trim	Yes	N/A	N/A	N/A	N/A	Yes
2	Direct input source	Yes*	N/A	N/A	N/A	N/A	N/A
3	Direct input level	Yes	N/A	N/A	N/A	N/A	N/A
4	Direct input pre-/post-	Yes	N/A	N/A	N/A	N/A	N/A
5	Direct input mute	Yes	N/A	N/A	N/A	Yes	N/A
6	Delay	Yes	N/A	N/A	N/A	N/A	Yes
7	Process order	Yes	Yes	N/A	N/A	N/A	N/A

* Only when automate patching is on.

Dynamics

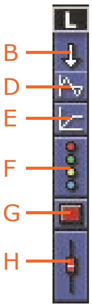


This section shows the scoped compressor parameters of the master channels. Although only the corrective compressor is shown below, this is typically the same for the other compressor modes (adaptive, creative, vintage and shimmer).

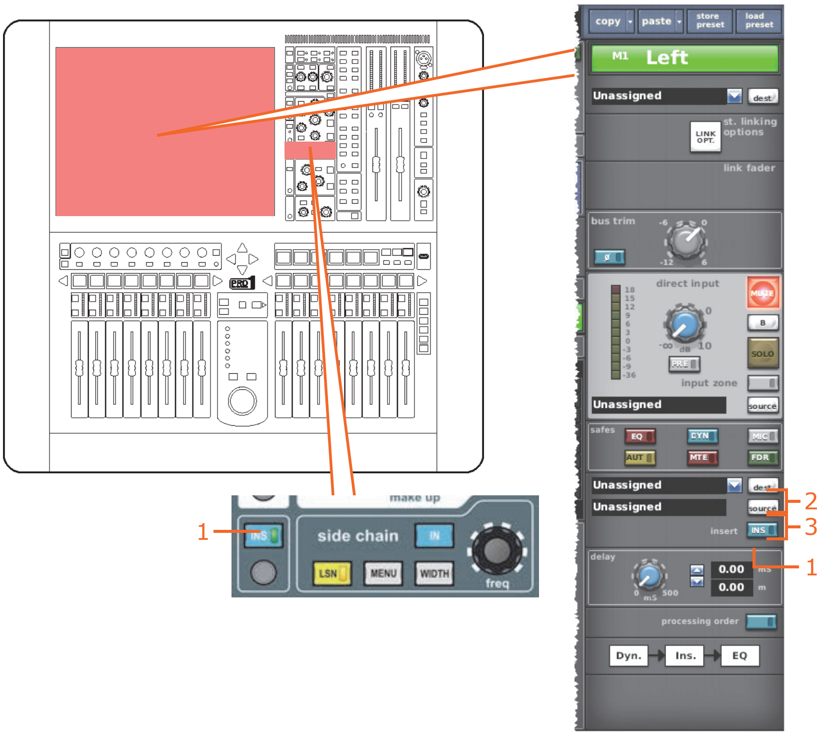


Item	Parameter	B	D	E	F	G	H
1	Compressor: attack, release, threshold, ratio, make up (gain), knee and mode	Yes	N/A	Yes	N/A	N/A	N/A
2	Sidechain source	N/A	N/A	No	N/A	N/A	N/A
3	Compressor sidechain: compressor sidechain in/out, frequency and width	Yes	N/A	Yes	N/A	N/A	N/A

Insert

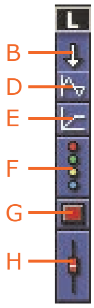


This section shows the scoped insert parameters of the master channels.

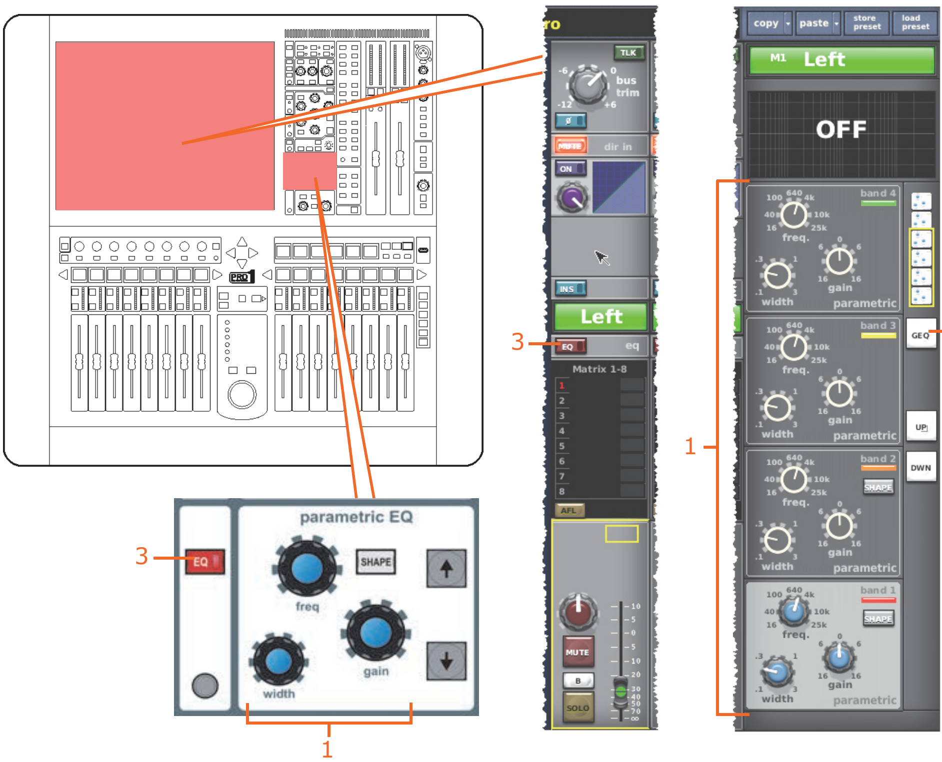


Item	Parameter	B	D	E	F	G	H
1	Insert in/out	Yes	N/A	N/A	N/A	N/A	N/A
2	Insert send destination	No	N/A	N/A	N/A	N/A	N/A
3	Insert return source	No	N/A	N/A	N/A	N/A	N/A

EQ



This section shows the scoped EQ parameters of the master channels.

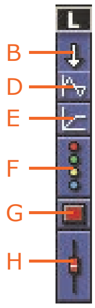


Item	Parameter	B	D	E	F	G	H
1	All PEQ filters (all six bands): frequency, gain, width and shape (bands 1, 2 and 6 only)	Yes	Yes	N/A	N/A	N/A	N/A
2	Parametric/Graphic type	No	No	N/A	N/A	N/A	N/A
3	EQ in/out	Yes	Yes	N/A	N/A	N/A	N/A

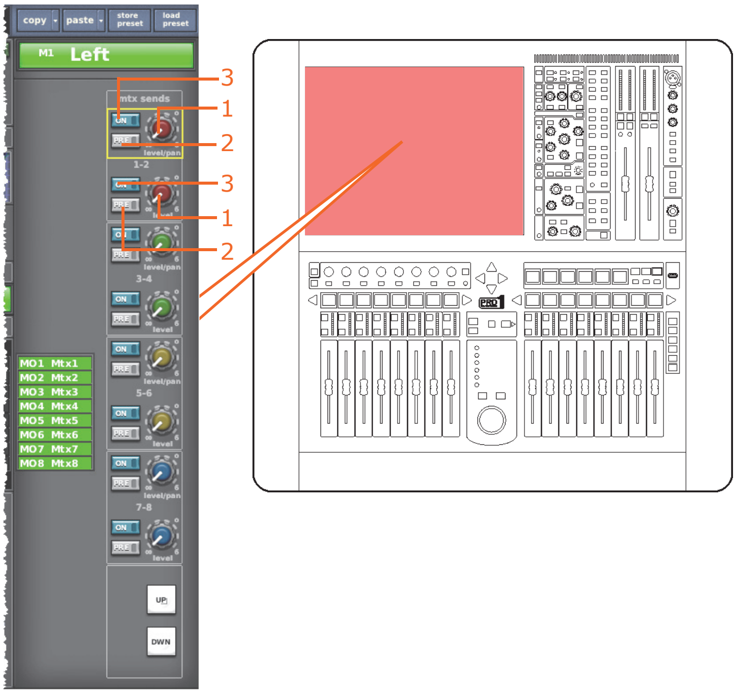
Aux send

Not applicable

Matrix send



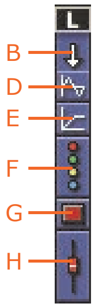
This section shows the scoped matrix send parameters of the master channels.



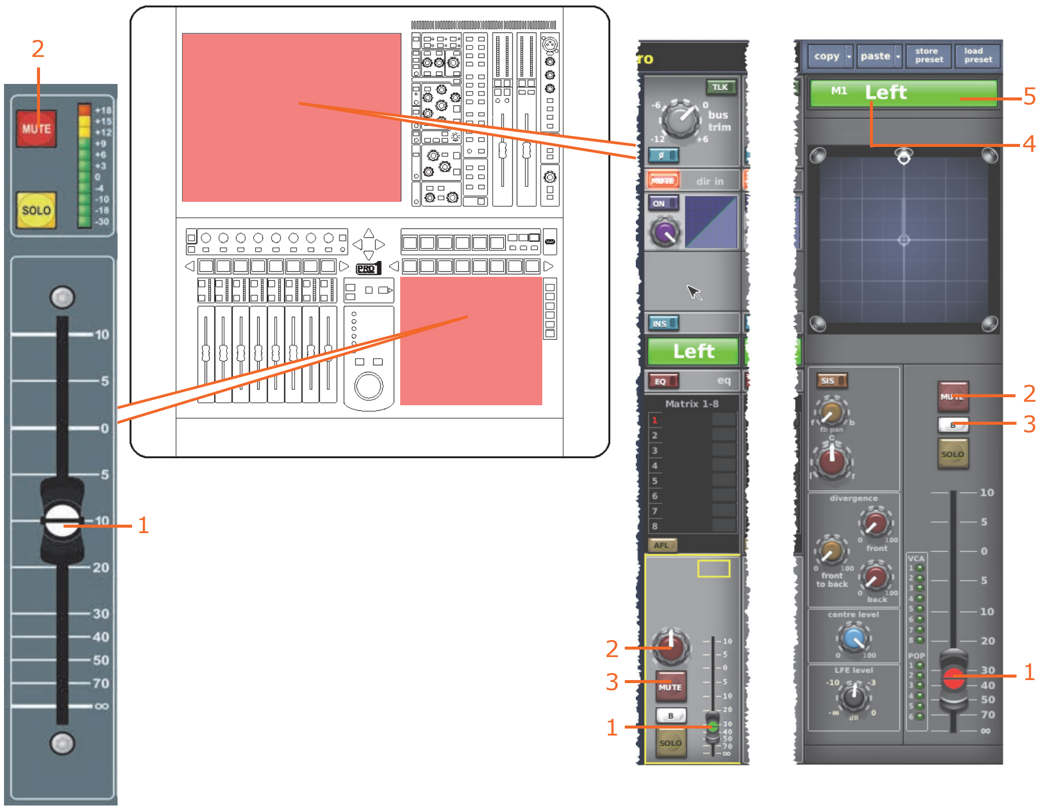
Item	*Parameter*	B	D	E	F	G	H
1	Send level	Yes	N/A	N/A	Yes	N/A	N/A
2	Send pre-fader on/off	Yes	N/A	N/A	Yes	N/A	N/A
3	Send on/off	Yes	N/A	N/A	Yes	N/A	N/A

You can scope individual bus sends. In column B (**All**), all sends are affected, and in column F (**Busses**), individual sends can be scoped.

Fader



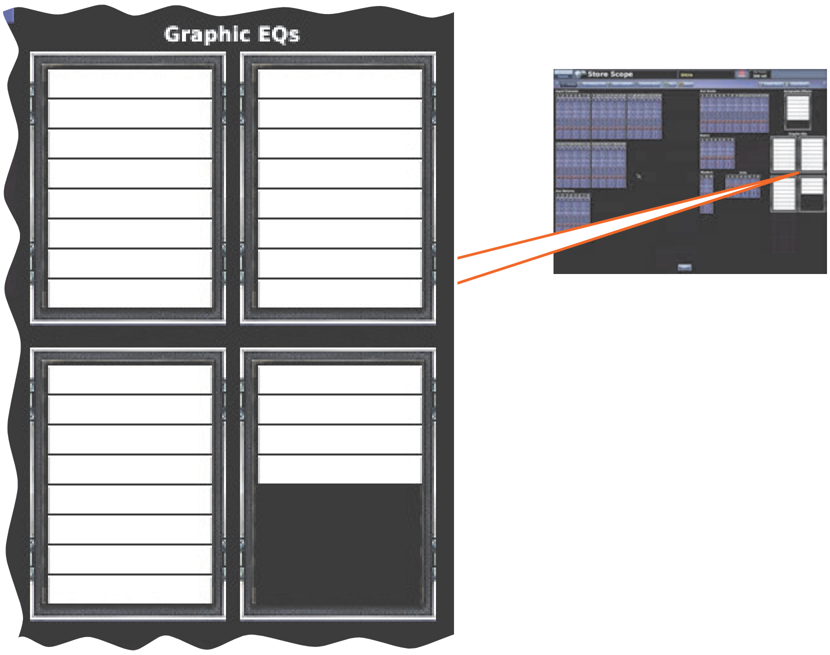
This section shows the scoped fader parameters of master channels.



Item	Parameter	B	D	E	F	G	H
1	Fader position	Yes	N/A	N/A	N/A	N/A	Yes
2	Channel mute	Yes	N/A	N/A	N/A	Yes	No
3	Solo B assignment	No	N/A	N/A	N/A	N/A	No
4	Channel name	Yes	N/A	N/A	N/A	N/A	No
5	Channel colour	Yes	N/A	N/A	N/A	N/A	No

GEQ rack

Each scope screen has a user-configurable rack of up to 28 GEQs in the **Graphic EQs** section.



Note: *A rack slot in the* **Graphic EQs** *section is equivalent to the* **All**  *scope area.*

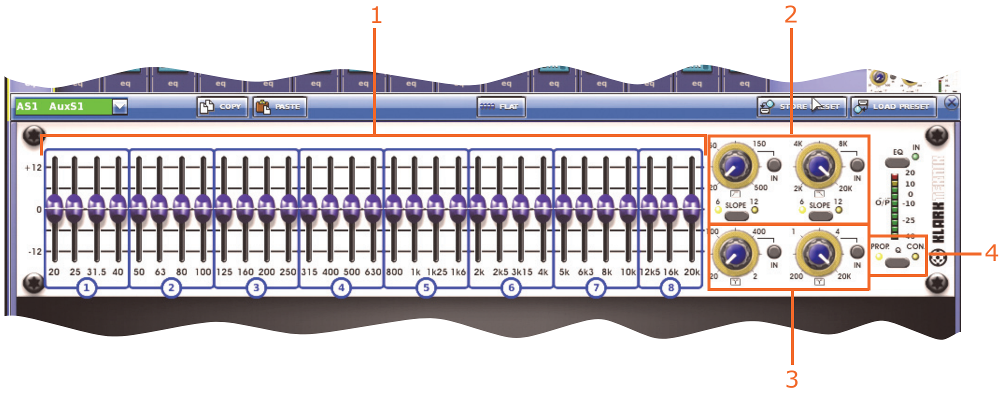
Patching

This section shows the scoped patching parameters of the GEQs, which are shown on the **Graphic GEQs** screen (below).

Item	*Parameter*	A	B	C	D	E	F	G	H
1	Bus assignment/type	N/A	**No**	N/A	N/A	N/A	N/A	N/A	N/A

GEQ

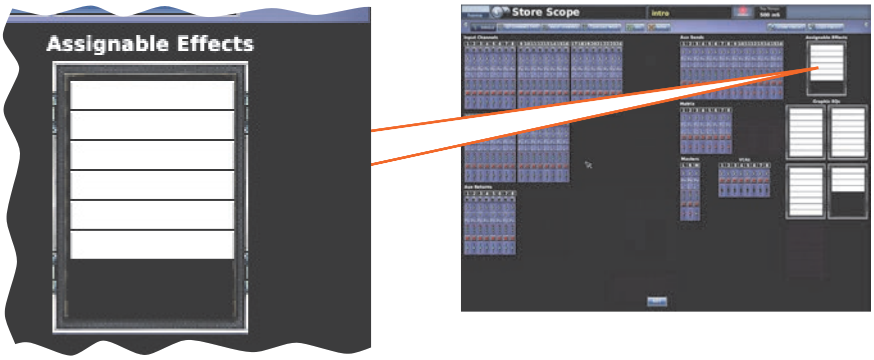
This section shows the scoped parameters of the GEQs,.



Item	*Parameter*	A	B	C	D	E	F	G	H
1	EQ band gains	N/A	**Yes**	N/A	N/A	N/A	N/A	N/A	N/A
2	HPF and LPF	N/A	**Yes**	N/A	N/A	N/A	N/A	N/A	N/A
3	Notch filters	N/A	**Yes**	N/A	N/A	N/A	N/A	N/A	N/A
4	GEQ mode (proportional/ constant Q)	N/A	**Yes**	N/A	N/A	N/A	N/A	N/A	N/A

Effects rack

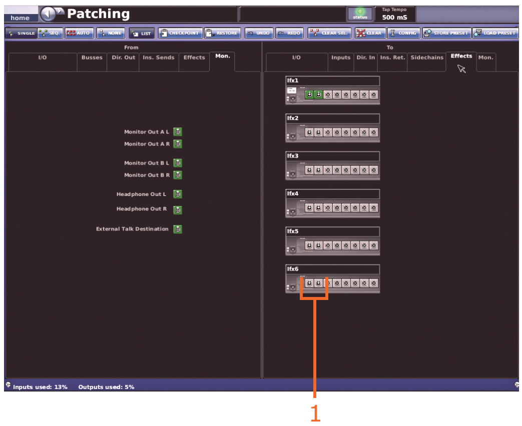
Each scope screen has a user-configurable rack of up to six effects in the **Assignable Effects** section.



Note:** A rack slot in the **Assignable Effects** section is equivalent to the **All  *scope area.*

Patching

The diagram right shows the scoped patching parameters of the effects, which are on the **Effects** tab of the **To** section of the **Patching** screen.



Item	*Parameter*	A	B	C	D	E	F	G	H
1	Input patching	N/A	**No**	N/A	N/A	N/A	N/A	N/A	N/A

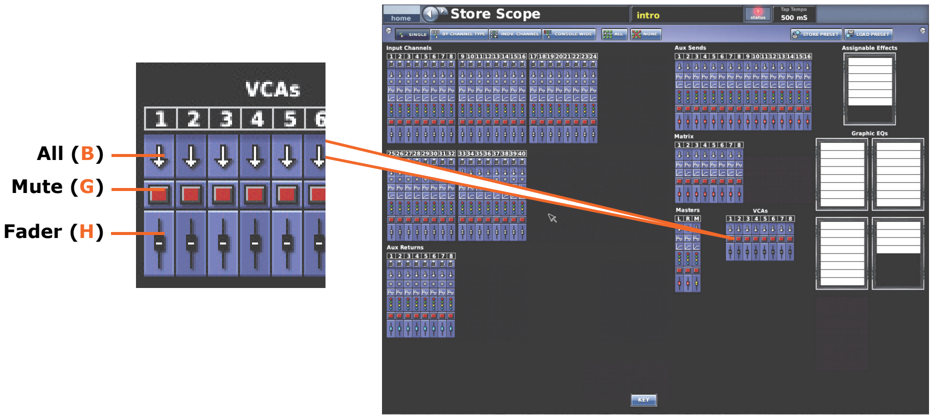
Effects

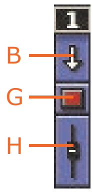
This section shows the scoped parameters of the effects.

Item	*Parameter*	A	B	C	D	E	F	G	H
1	Effect name	N/A	**Yes**	N/A	N/A	N/A	N/A	N/A	N/A
2	Effect colour	N/A	**Yes**	N/A	N/A	N/A	N/A	N/A	N/A
3	Effect type	N/A	**Yes**	N/A	N/A	N/A	N/A	N/A	N/A
4	All effect parameters	N/A	**Yes**	N/A	N/A	N/A	N/A	N/A	N/A

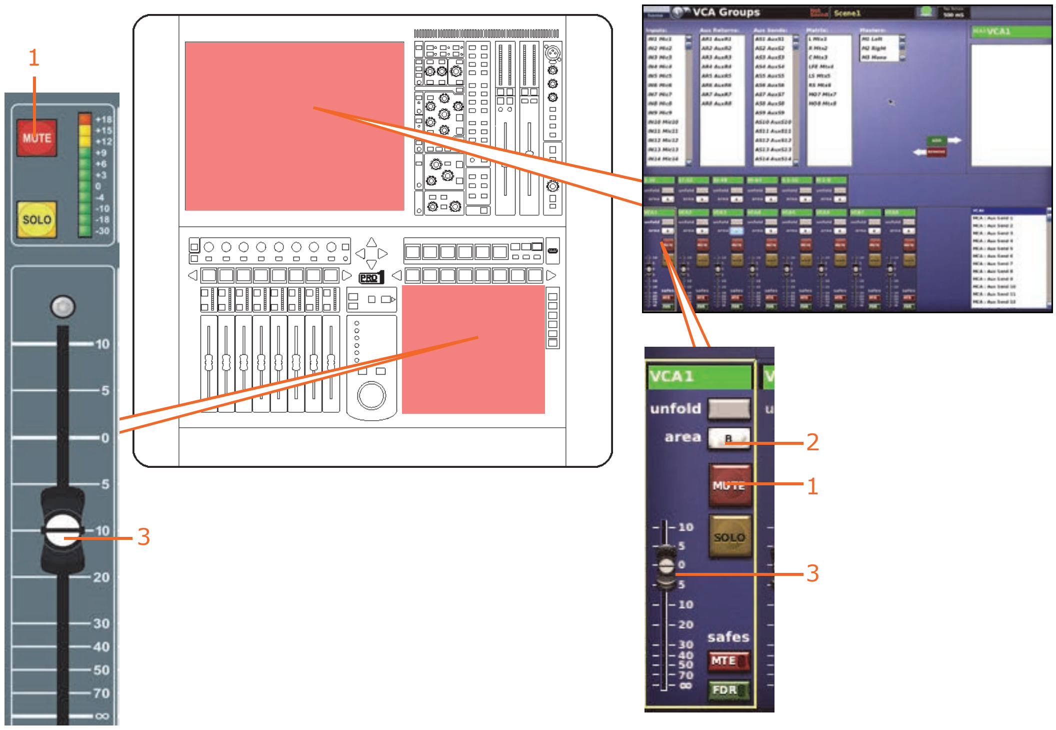
Groups

Each scope screen contains eight VCA channels in the **VCAs** section.





The following diagram shows the scoped parameters of the VCA group channels, and shows the **VCA Groups** screen on the GUI.



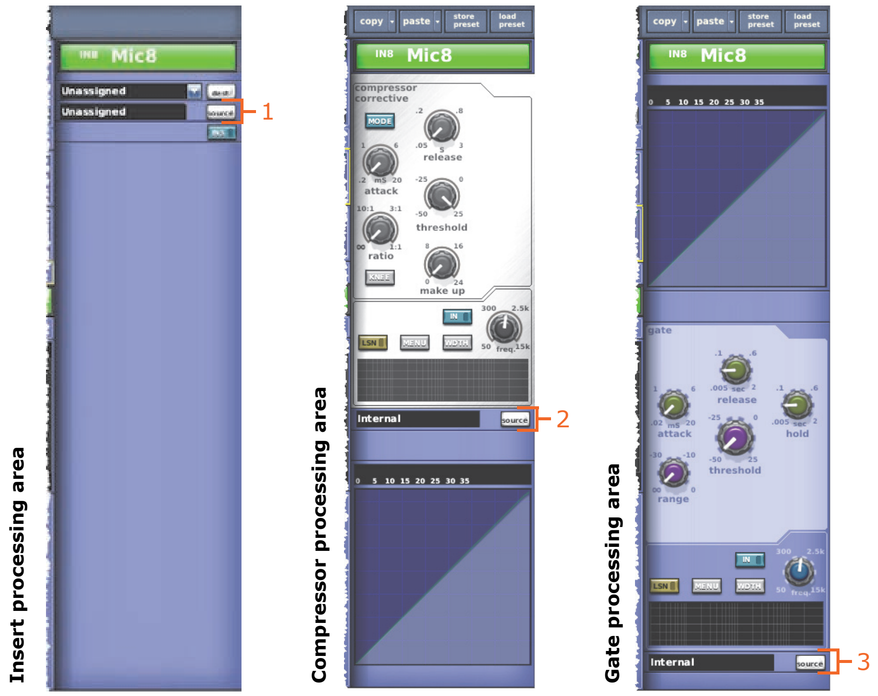
Item	*Parameter*	B	G	H
1	VCA mute	Yes	Yes	N/A
2	VCA area A/B	Yes	N/A	N/A
3	VCA fader level (required for surround panning)	Yes	N/A	Yes

Appendix G: Parameters Affected By Automate Patching

This appendix shows the patching parameters (sources) that can be changed on a per-scene basis in automation. These are only selectable when the **Automate Patching** option of the **Preferences** scene is selected (see "Using patching in automation" on page 187).

Inputs

The following input channel sources can be changed per scene.



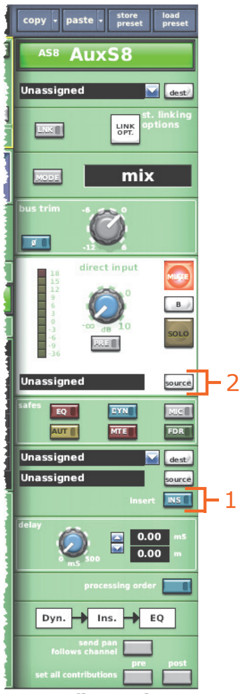
Item	Parameter
1	Insert return source
2	Compressor side chain source
3	Gate key source

Although the mic input and tape input sources are automatable, that is, they can be changed per scene, they are controlled by the input patching recall scope area switch (see Figure 30, "Parameter sections per input channel," on page 351). They are not affected by the **Automate Patching** function.

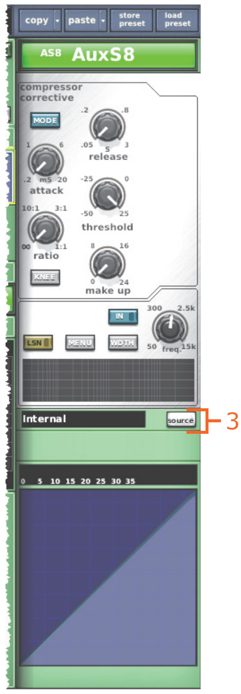
Auxes

The sources for each aux channel can be changed per scene.

Item	Parameter
1	Insert return source
2	Direct in source
3	Compressor side chain source



Configuration processing area



Compressor processing area

Matrices

The sources for each matrix channel can be changed per scene.

Item	Parameter
1	Insert return source
2	Direct in source
3	Compressor side chain source

Configuration processing area

Compressor processing area

Masters

The sources for each master channel can be changed per scene.

Item	*Parameter*
1	Insert return source
2	Direct in source
3	Compressor side chain source

Configuration processing area

Compressor processing area

Effects

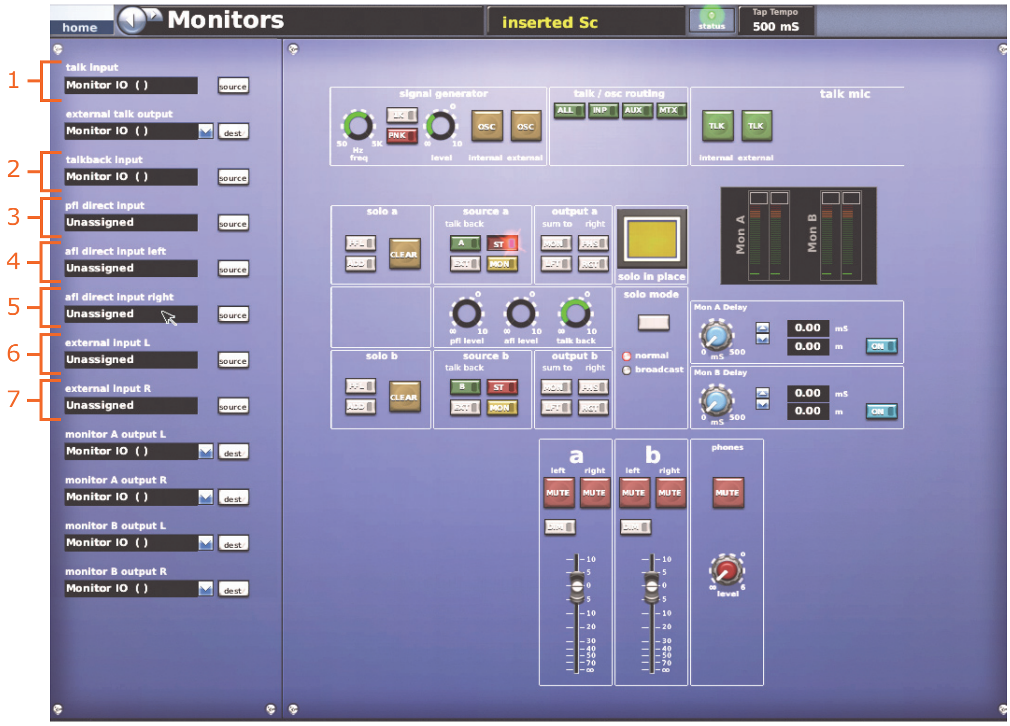
Effect input sources can be changed per scene via the **Effects** tab of the **To** section of the **Patching** screen (shown right).

System devices

Sources for the outputs of external devices, such as the DL451 Modular I/O and DN9696 Recorder, etc., can be changed per scene. The sources are selectable via the **Stage I/O** and **FOH I/O** tabs in **To** section of the **Patching** screen (see Chapter 8 "Patching" on page 45). However, this does not include the I/O card configuration.

Monitors

The following monitor sources can be changed per scene.



Item	Parameter
1	Talk input source
2	Taklback input source
3	PFL direct input source
4	AFL direct input left source
5	AFL direct input right source
6	External monitor input left source
7	External monitor input right source

Appendix H: Parameters Protected By Safes

This appendix shows the parameters affected/unaffected by each of the safe types (**EQ, DYN, MIC, AUTO, MUTE** and **FADER**).

Note: *The parameter areas for the scopes (store and recall) and the safes are, basically the same. However, the way they are presented in their respective appendices is different. This may provide you with a useful alternative when referring to this material, should you prefer one more than the other (see Appendix F "Parameters Affected By Scope" on page 349).*

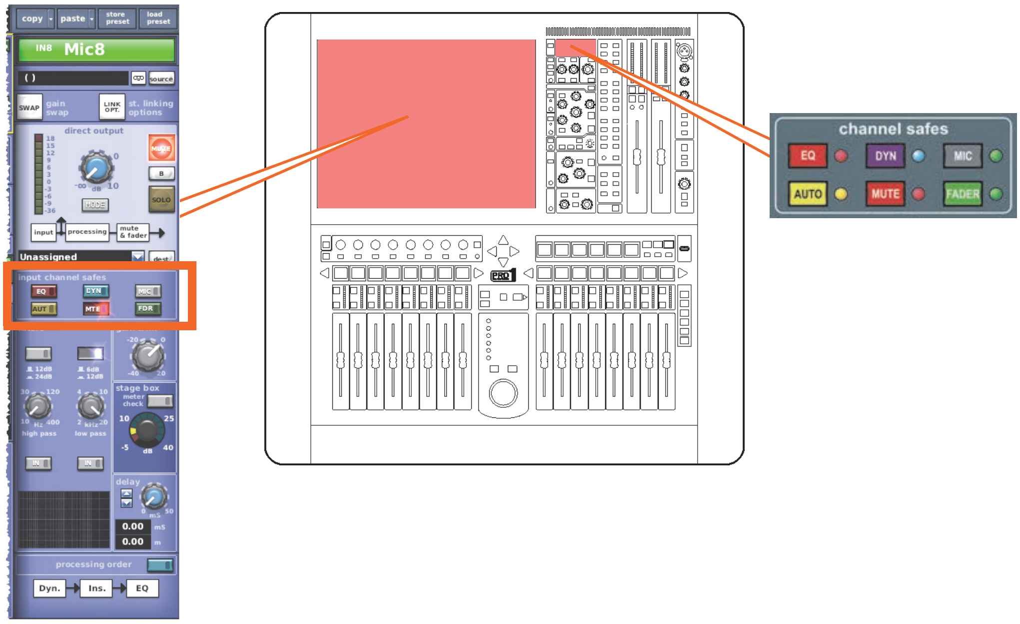
The following table is intended as a reference guide to help you go quickly to the area you want.

Channel or Group	***EQ safe***	***DYN safe***	***MIC safe***	***AUTO safe***	***MUTE safe***	***FADER safe***
Input channel	Page 400	Page 402	Page 404	Page 405	Page 406	Page 407
Aux	Page 410	Page 412	Page 413	Page 413	Page 414	Page 415
Return	N/A	N/A	Page 418	Page 418	Page 419	Page 420
Matrix	Page 423	Page 425	Page 426	Page 427	Page 427	Page 428
Master	Page 431	Page 433	Page 434	Page 434	Page 435	Page 436
VCA group	N/A	N/A	N/A	Page 438	Page 439	Page 439

Note: *Throughout this appendix, when referring to the hardware controls in either fader bay the default channel assignments are used (that is, inputs in the channel fader bay and outputs in the mix fader bay). This is because any channel type can be assigned to either fader bay.*

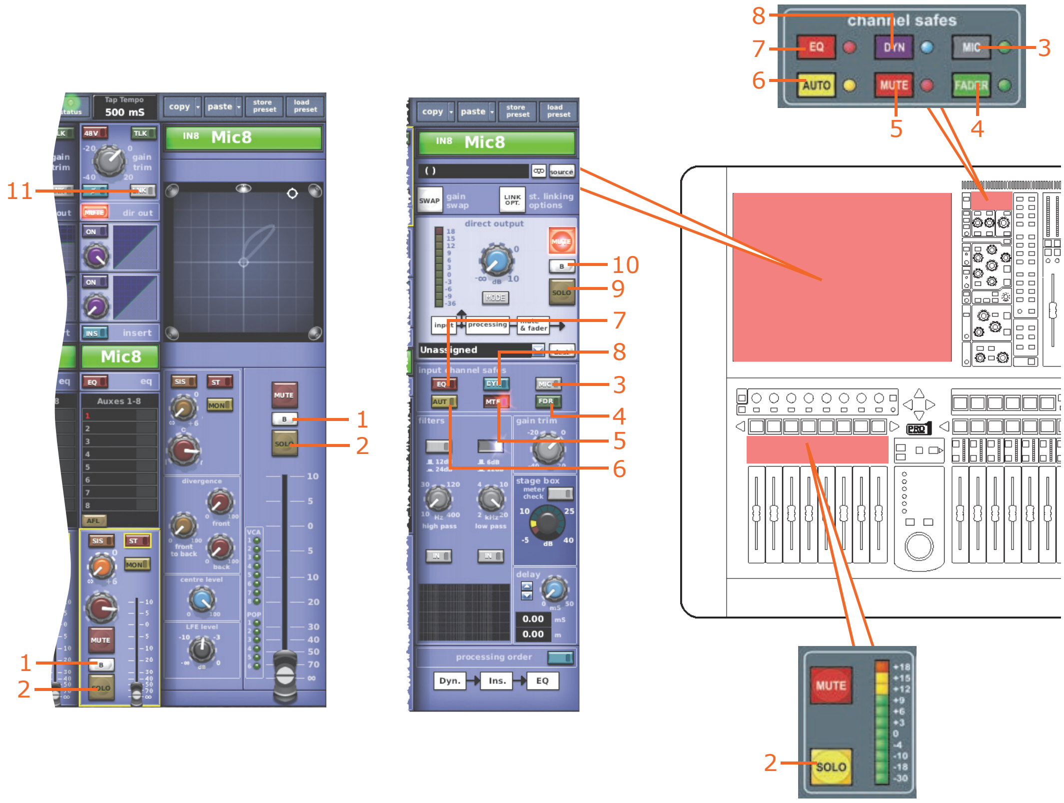
Inputs

The input safes are selected via the input channel safes section of the channel processing area of the configuration processing area of the GUI channel strip.



Input parameters not affected by the safes

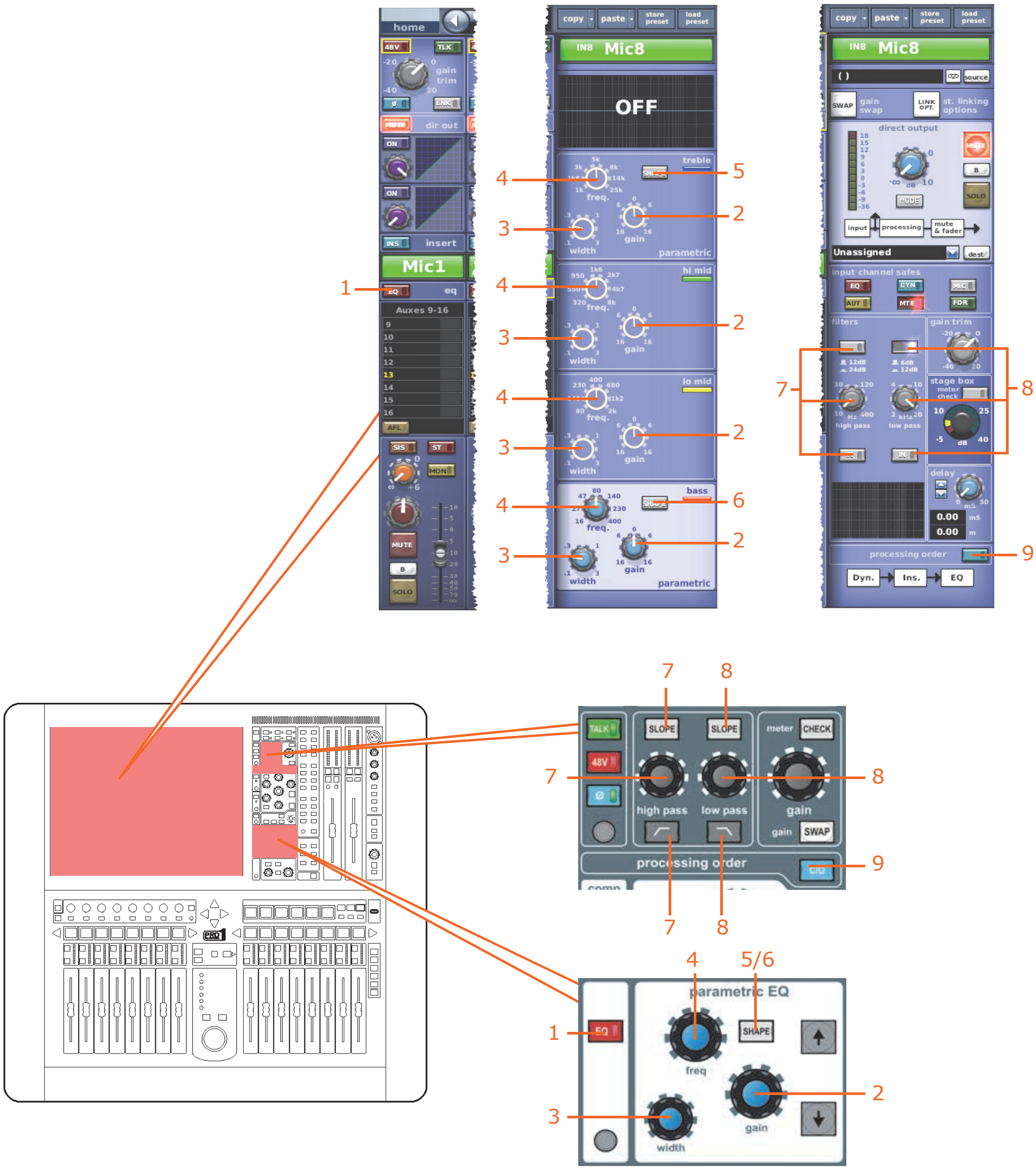
The following input channel parameters are *not* affected by any of the safes.



Item	Control	Parameter
1	**B** switch	Solo B on/off
2	**SOLO** switch	Solo on/off
3	**MIC** switch	Mic safe on/off
4	**FADER**/[**FDR**] switch	Fader safe on/off
5	**MUTE**/[**MTE**] switch	Mute safe on/off
6	**AUTO**/[**AUT**] switch	Automation safe on/off
7	**EQ** switch	EQ safe on/off
8	**DYN** switch	Dynamic safe on/off
9	**SOLO** switch	Direct output solo on/off
10	**B** switch	Direct output solo B on/off
11	**LNK** switch	Stereo linking on/off

EQ safe

This section shows the input parameters protected by the **EQ** safe.

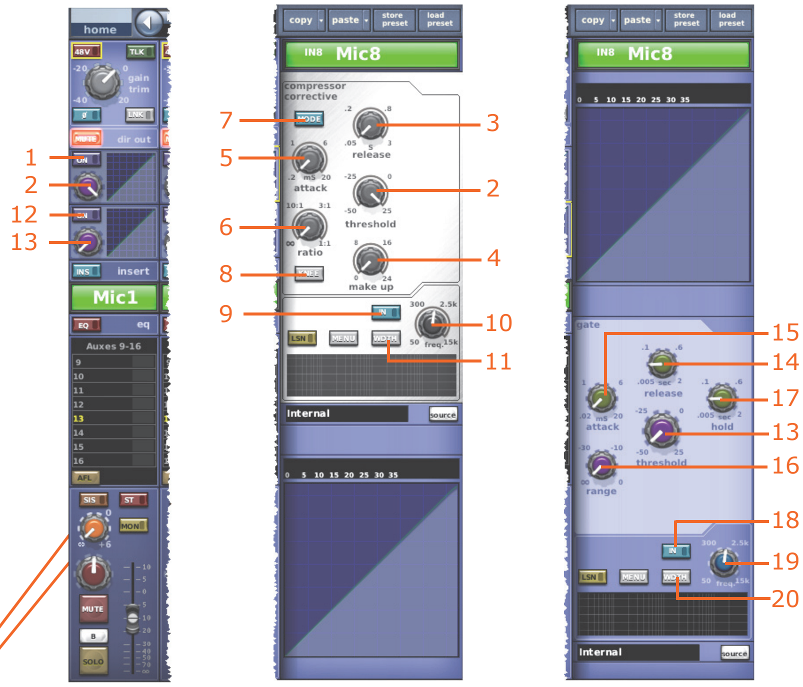


Item	Control(s)	Parameter
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch	Selects treble shelving mode: parametric, bright, classic or soft

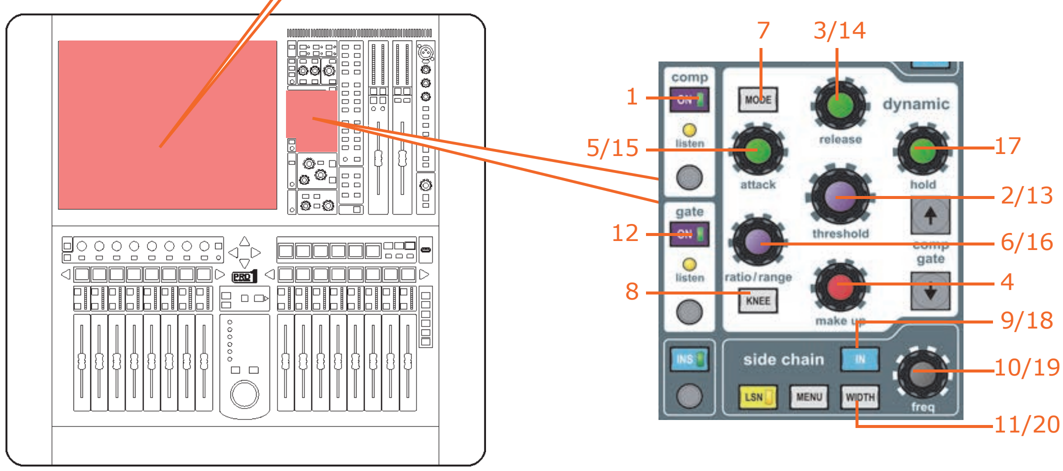
Item	*Control(s)*	*Parameter*
6	**SHAPE** switch	Selects bass shelving mode: parametric, deep, classic or warm
7	**SLOPE** switch, **high pass** control knob, / [**IN**] switch	High pass filter
8	**SLOPE** switch, **low pass** control knob, /[**IN**] switch	Low pass filter
9	**C/O** switch	Order of processing: **Dyn.**→**Ins.**→**EQ** or **EQ**→**Ins.**→**Dyn.**

DYN (dynamic) safe

This section shows the input parameters protected by **DYN** safe. Although only the corrective compressor is shown below, this is typically the same for the other compressor modes (adaptive, creative and vintage).



Input channel fast strip **Compressor processing area in channel strip** **Gate processing area in channel strip**

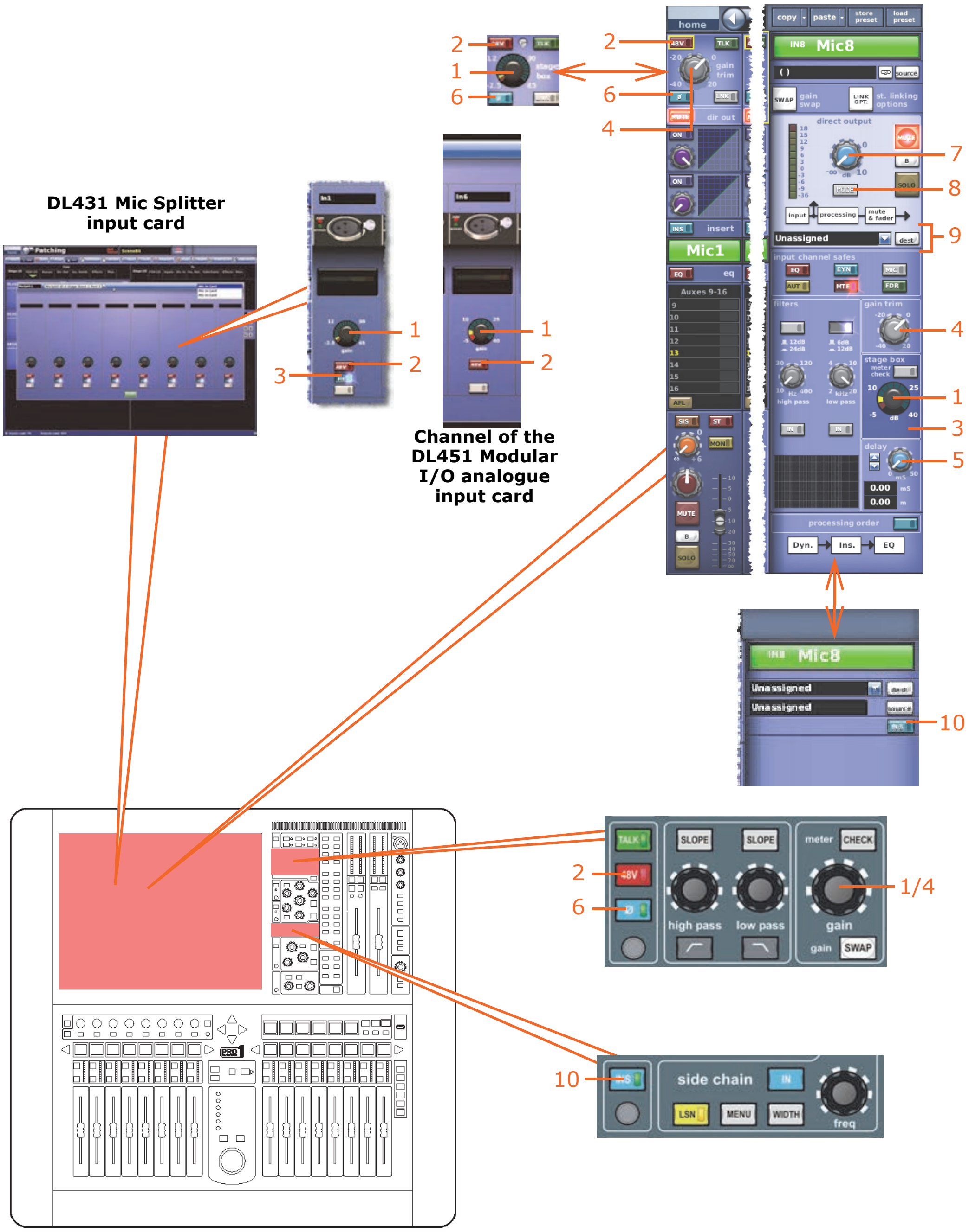


Item	*Control*	*Parameter*
1	**ON** switch	Compressor on/off
2	**threshold** control knob	Compressor threshold
3	**release** control knob	Compressor release
4	**make up** control knob	Compressor make up gain
5	**attack** control knob	Compressor attack
6	**ratio** control knob	Compressor ratio

Item	*Control*	*Parameter*
7	**MODE** pushbutton	Compressor mode: corrective (as shown), adaptive, creative or vintage
8	**KNEE** pushbutton	Compressor knee: hard, medium or soft
9	**IN** switch	Compressor sidechain in/out
10	**freq** control knob	Compressor sidechain frequency
11	**WIDTH** pushbutton	Compressor sidechain width: 2 Oct, 1 Oct or 0.3 Oct
12	**ON** switch	Gate on/off
13	**threshold** control knob	Gate threshold
14	**release** control knob	Gate release
15	**attack** control knob	Gate attack
16	**range** control knob	Gate range
17	**hold** control knob	Gate hold
18	**IN** switch	Gate sidechain in/out
19	**freq** control knob	Gate sidechain frequency
20	**WIDTH** pushbutton	Gate sidechain width: 2 Oct, 1 Oct or 0.3 Oct

MIC safe

This section shows the input parameters protected by **MIC** safe.



Item	Control	Parameter
1	**gain**/[**stage box**] control knob*	Gain of remote amplifier
2	**48V** switch	48V phantom gain
3	Filter switch**	30Hz filter
4	**gain**/[**gain trim**] control knob*	Digital input trim
5	**delay** control knob	Delay time

Item	*Control*	*Parameter*
6	**Ø** switch	Input phase invert on/off
7	Level control knob	Direct output level
8	**MODE** switch	Direct output tap-off point: "Post-fade and mute", "Pre-mute, pre-processing" or "Pre-mute, post-processing"
9	**dest** button, text field	Direct output tap-off point
10	**INS** switch	Insert in/out

* Depends on swap status.

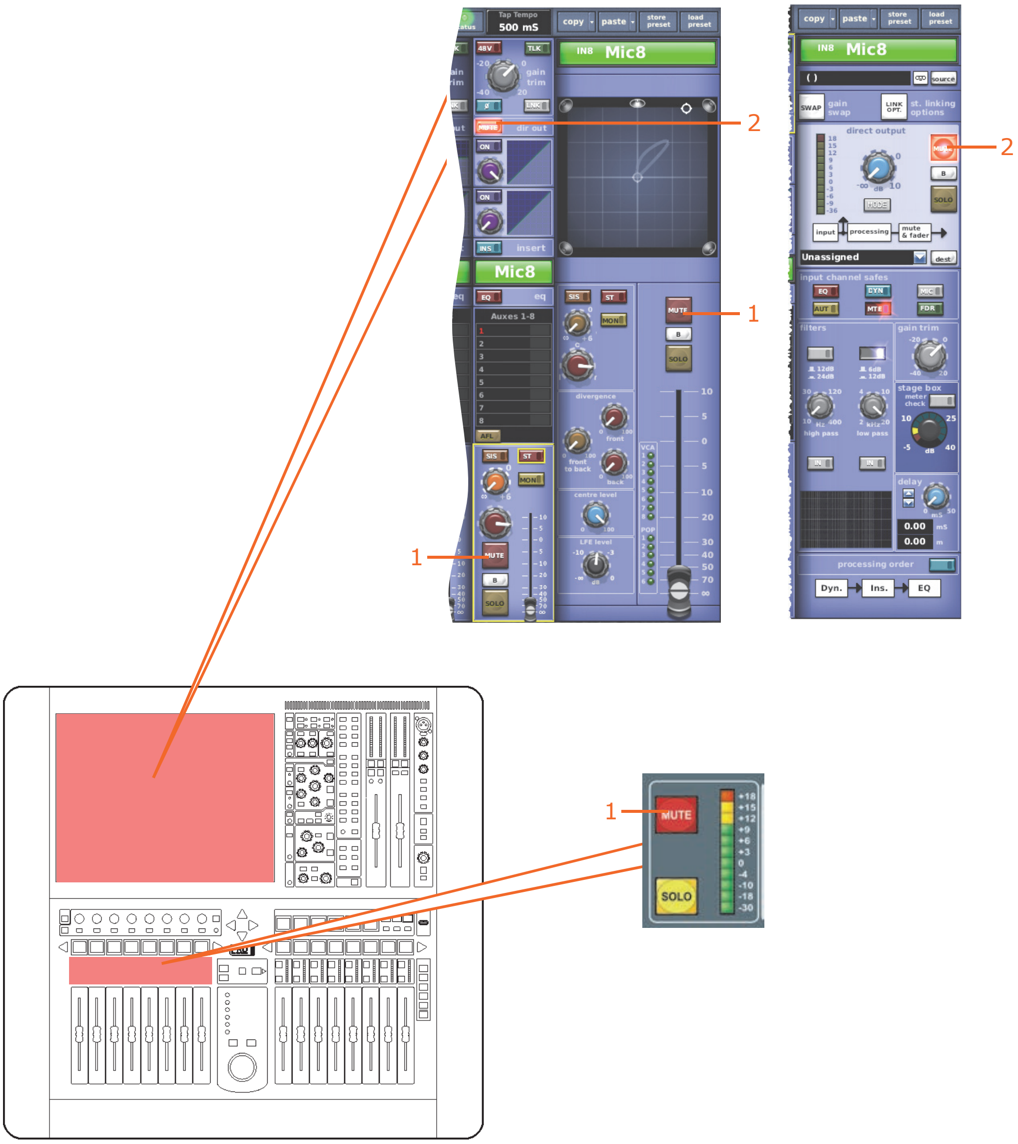
** Only when sourced from a DL431 Mic Splitter.

AUTO (automation) safe

All of the input channel parameters are protected by the **AUTO** safe — except, of course, for the ones unaffected by the safes (see "Input parameters not affected by the safes" on page 399).

MUTE safe

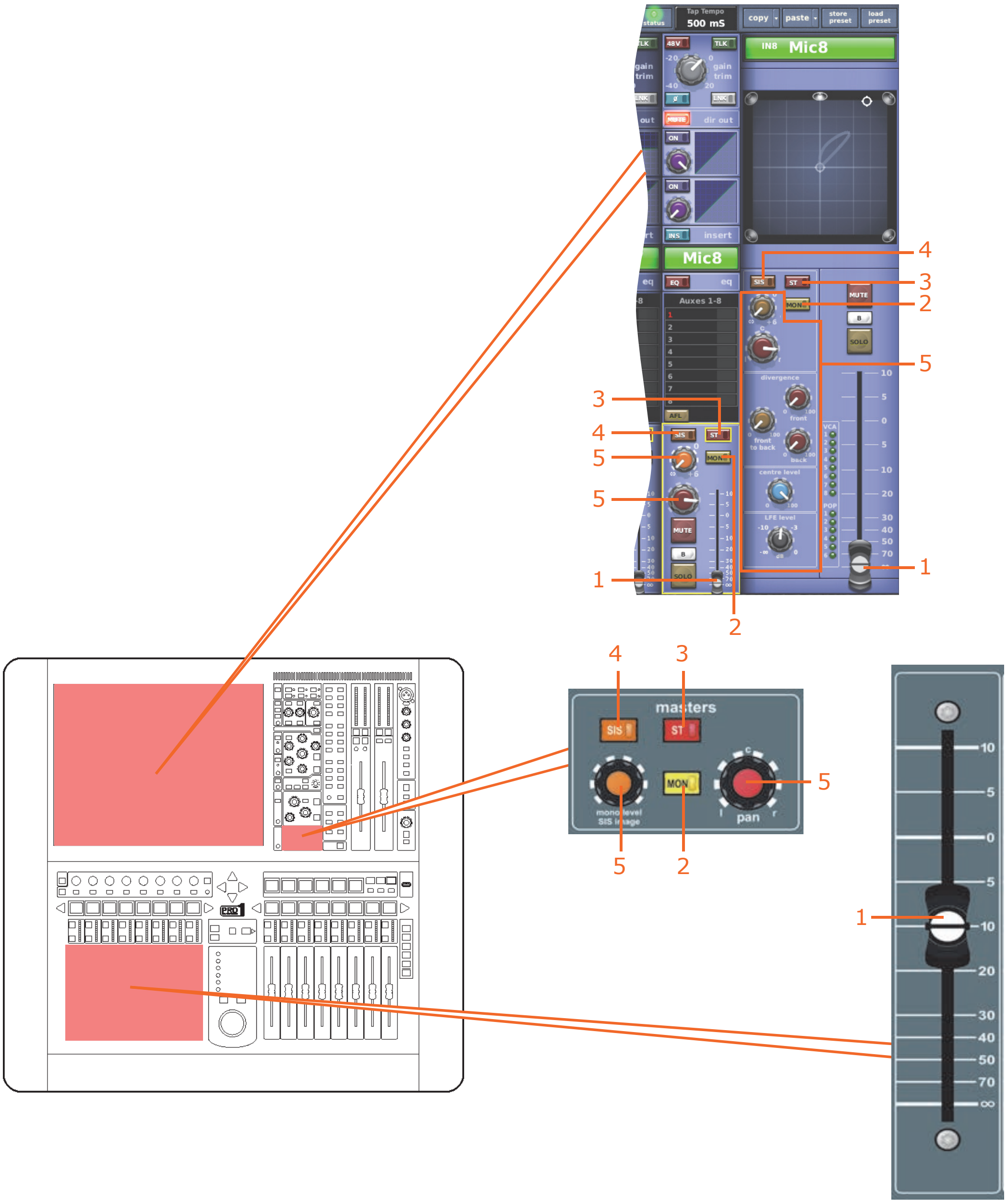
This section shows the input parameters protected by the **MUTE** safe.



Item	*Control*	*Parameter*
1	**MUTE** switch	Mute on/off
2	**MUTE** switch	Direct output mute on/off

FADER safe

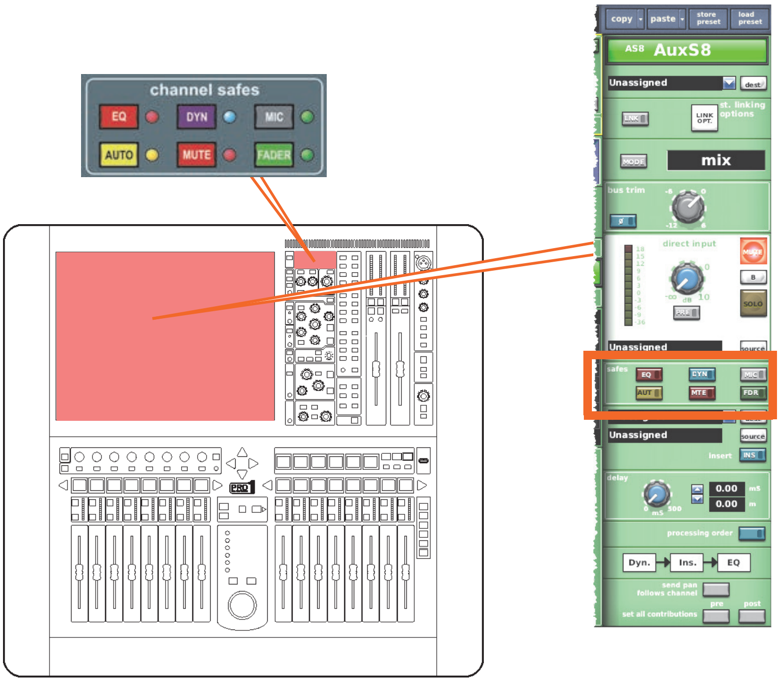
This section shows the input parameters protected by the **FADER** safe.



Item	Control(s)	Parameter
1	Fader	Fader level
2	**MON** switch	Mono routing on/off
3	**ST** switch	Stereo routing on/off
4	**SIS** switch	Spatial imaging system on/off
5	Panning control knobs	Surround panning (includes all surround sound parameters)

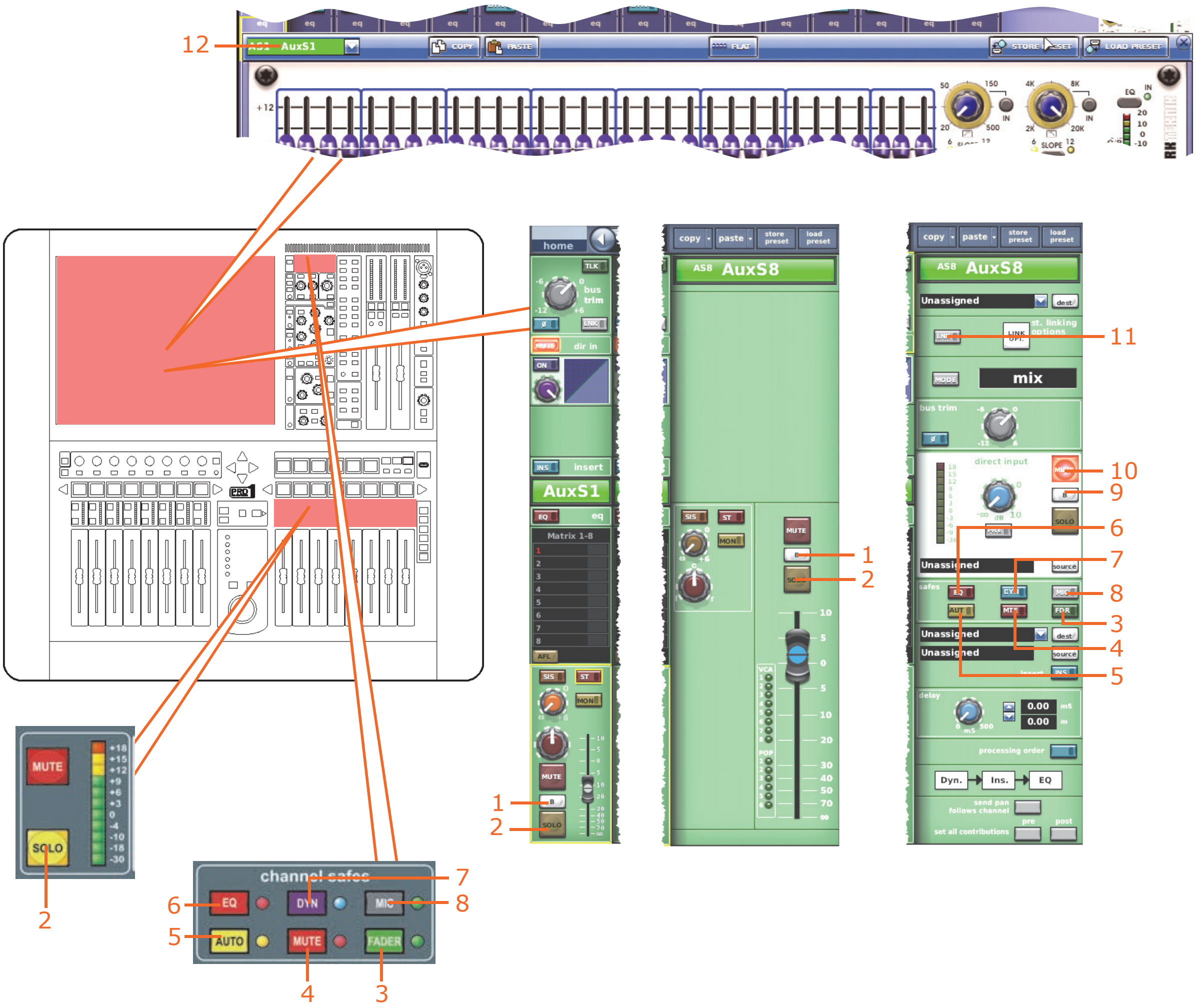
Auxes (Aux Sends)

The aux safes are selected via the output channel safes section of the channel processing area and the configuration processing area of the GUI channel strip.



Aux parameters not affected by the safes

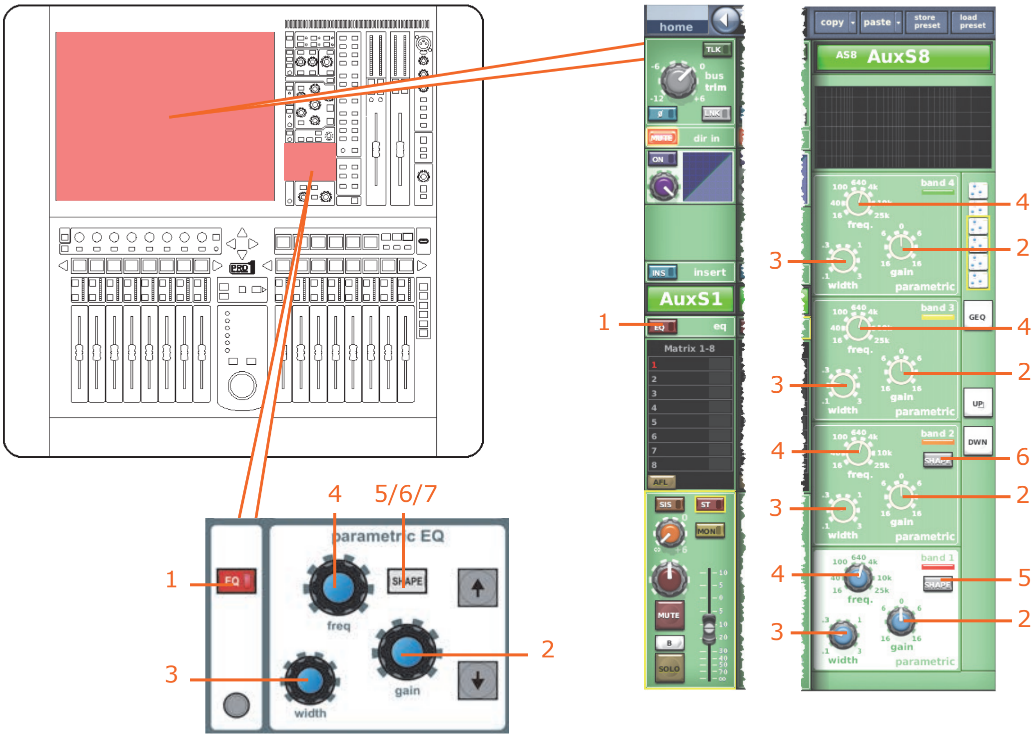
The following shows you which aux parameters are *not* affected by each safe.



Item	Control	Parameter
1	**B** switch	Solo B on/off
2	**SOLO** switch	Solo on/off
3	**FADER**/[**FDR**] switch	Fader safe on/off
4	**MUTE**/[**MTE**] switch	Mute safe on/off
5	**AUTO**/[**AUT**] switch	Automation safe on/off
6	**EQ** switch	EQ safe on/off
7	**DYN** switch	Dynamic safe on/off
8	**MIC** switch	Mic safe on/off
9	**SOLO** switch	Direct input solo on/off
10	**B** switch	Direct input solo B on/off
11	**LNK** switch	Stereo linking on/off
12	Field	GEQ assignment

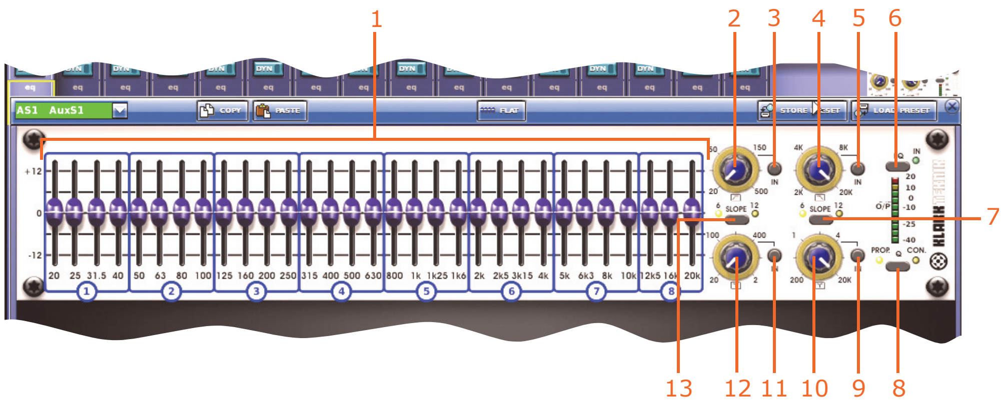
EQ safe

This section shows the aux parameters protected by the **EQ** safe.



Item	*Control*	*Parameter*
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch	Band 1 shelving mode: parametric, warm, high pass 6dB or high pass 12dB
6	**SHAPE** switch	Band 2 shelving mode: parametric or high pass 24dB
7	**SHAPE** switch (not shown in GUI channel strip above)	Band 6 shelving mode: parametric, soft, low pass 6dB or low pass 12dB

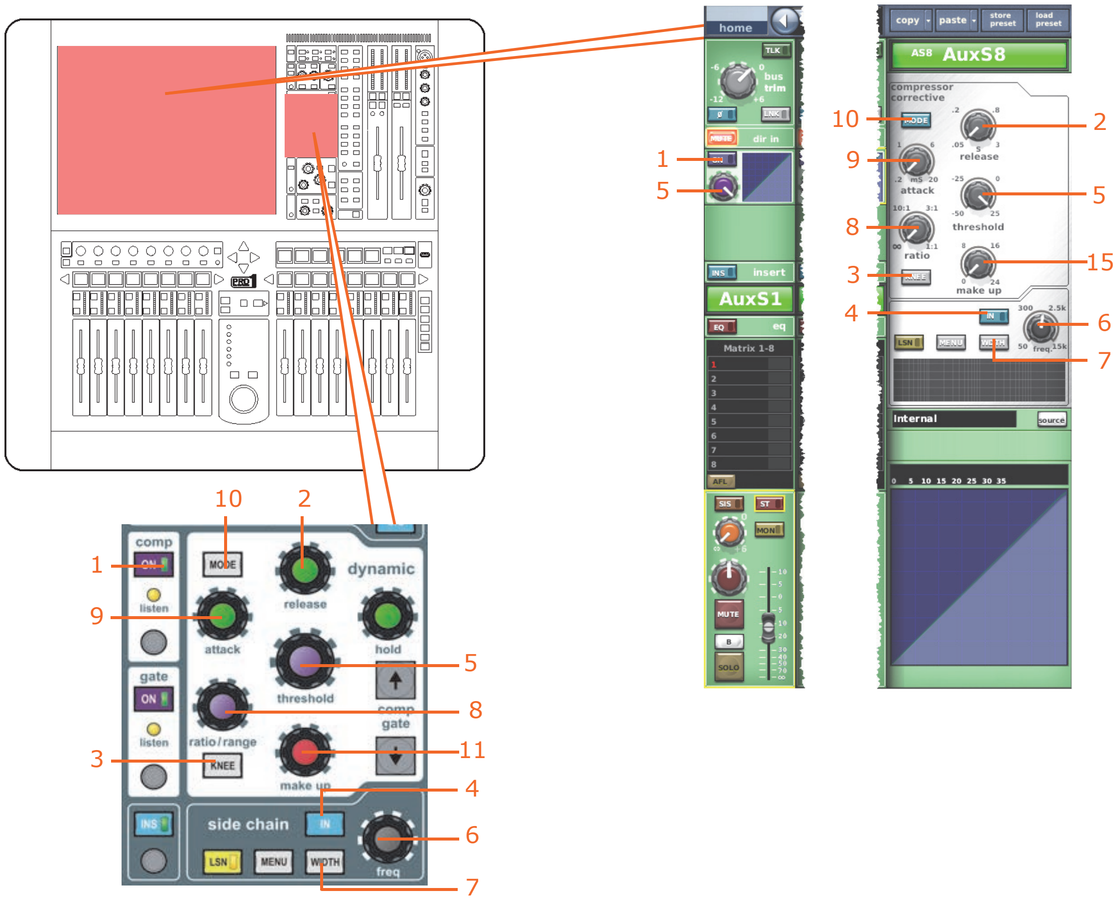
Note: *Although bands 5 and 6 are not shown above, the items in the table also apply. Both bands have items 2, 3 and 4, and band 6 also has item 7.*



Item	*Control*	*Parameter*
1	31 faders	Fader positions
2	High pass filter control knob	High pass filter cut off frequency
3	**IN** switch	High pass filter in/out
4	Low pass filter control knob	Low pass filter cut off frequency
5	**IN** switch	Low pass filter in/out
6	**EQ** switch	EQ in/out
7	**SLOPE** switch	Selects low pass filter as 6dB or 12dB
8	**Q** switch	Selects Q mode as proportional (**PROP.**) or constant (**CON.**)
9	**IN** switch	Switches 200Hz - 20kHz notch filter in/out
10	Notch filter control knob	200Hz - 20kHz notch filter frequency
11	**IN** switch	Switches 20Hz - 2kHz notch filter in/out
12	Notch filter control knob	20Hz - 2kHz notch filter frequency
13	**SLOPE** switch	Selects high pass filter as 6dB or 12dB

DYN (dynamic) safe

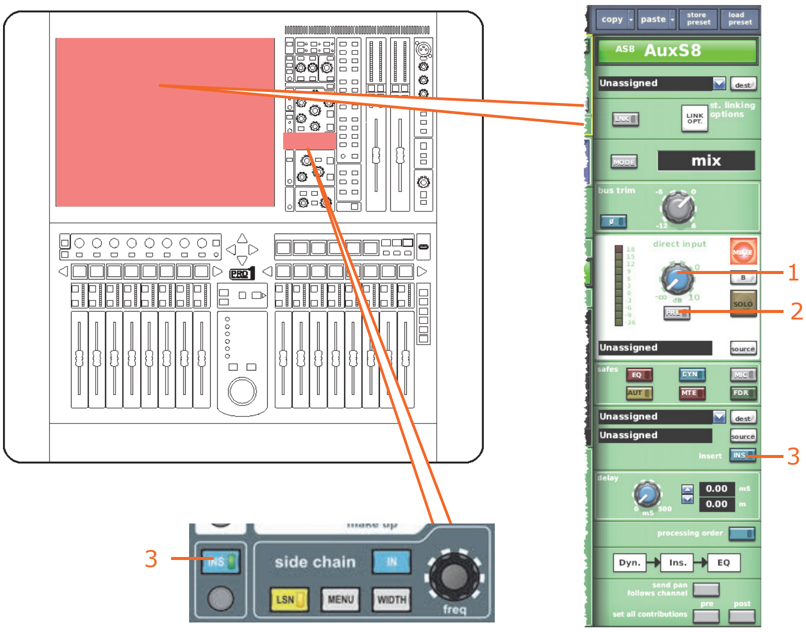
This section shows the aux parameters protected by the **DYN** safe. Only the corrective compressor is shown, but this is typically the same for the other compressor modes (adaptive, creative, vintage and shimmer).



Item	Control	Parameter
1	**ON** switch	Compressor on/off
2	**release** control knob	Compressor release
3	**KNEE** pushbutton	Compressor knee selector: hard, medium and soft
4	**IN** switch	Compressor sidechain in/out
5	**threshold** control knob	Compressor threshold
6	**freq** control knob	Compressor sidechain frequency
7	**WIDTH** pushbutton	Compressor sidechain width (unlabelled): 2 Oct, 1 Oct or 0.3 Oct
8	**ratio** control knob	Compressor ratio
9	**attack** control knob	Compressor attack
10	**MODE** pushbutton	Compressor mode — corrective, adaptive, creative, vintage or shimmer
11	**make up** control knob	Compressor gain

MIC safe

This section shows the aux parameters protected by the **MIC** safe.



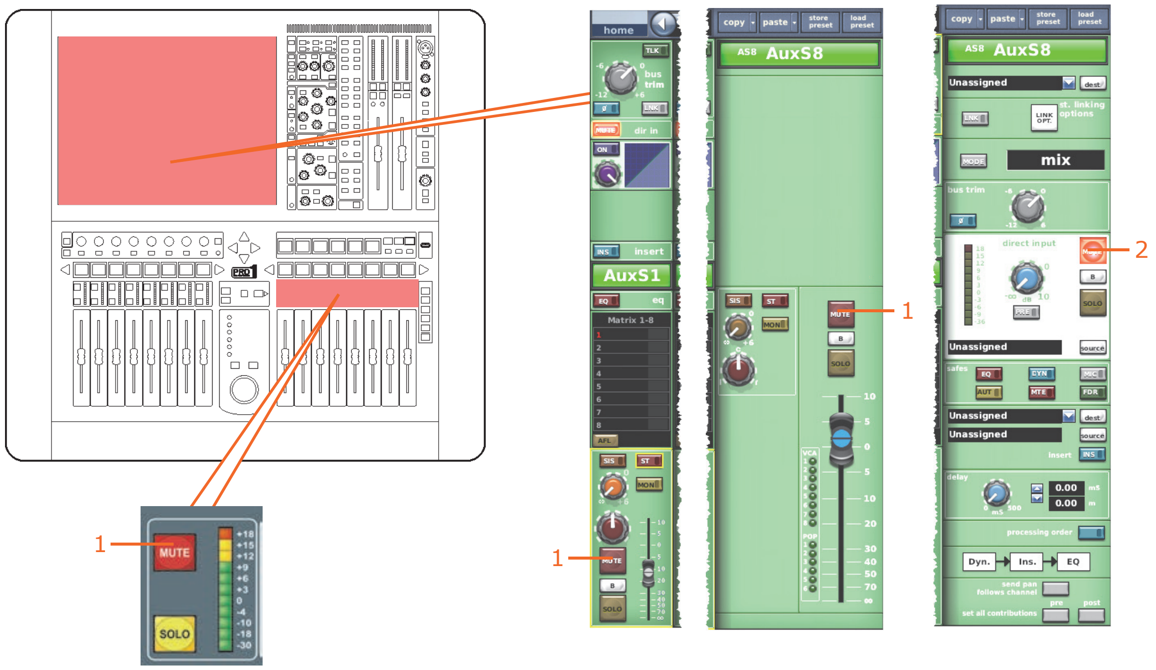
Item	Control	Parameter
1	Level control knob	Direct input level
2	**PRE** switch	Direct input pre- in/out
3	**INS** switch	Insert in/out

AUTO (automation) safe

All of the aux channel parameters are protected by the **AUTO** (automation) safe — except, of course, for the ones unaffected by the safes (see “Aux parameters not affected by the safes” on page 409).

MUTE safe

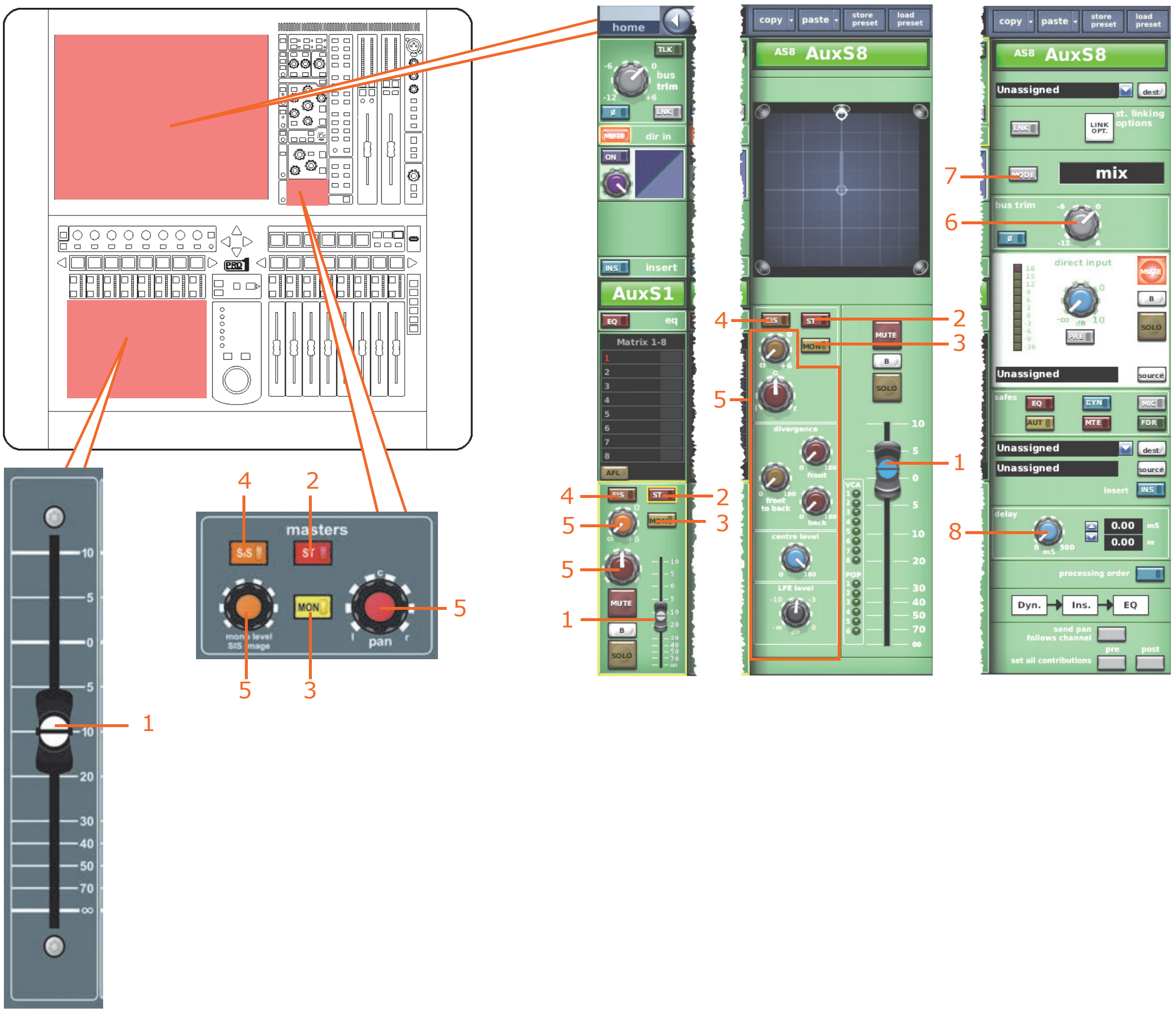
This section shows the aux parameters protected by the **MUTE** safe.



Item	*Control*	*Parameter*
1	**MUTE** switch	Mute on/off
2	**MUTE** switch	Direct input mute on/off

FADER safe

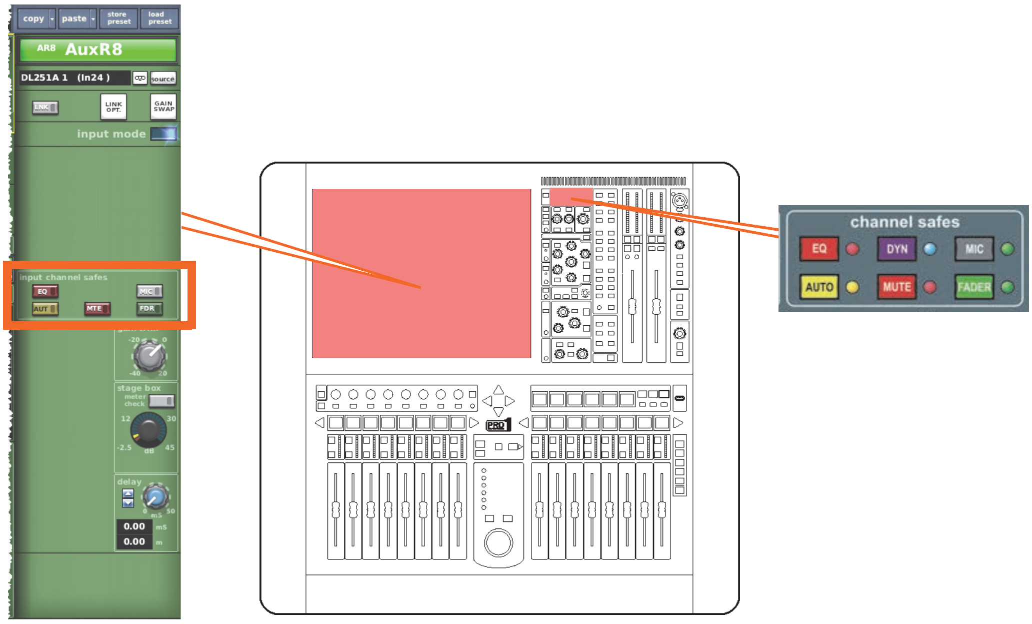
This section shows the aux parameters protected by the **FADER** safe.



Item	Control	Parameter
1	Fader	Fader level
2	**ST** switch	Stereo routing
3	**MON** switch	Mono routing
4	**SIS** switch	Spatial imaging system in/out
5	Panning control knobs	Surround sound panning (includes all surround parameters)
6	**bus trim** control knob	Bus trim level
7	**MODE** switch	Bus mode
8	**delay** control knob	Delay time

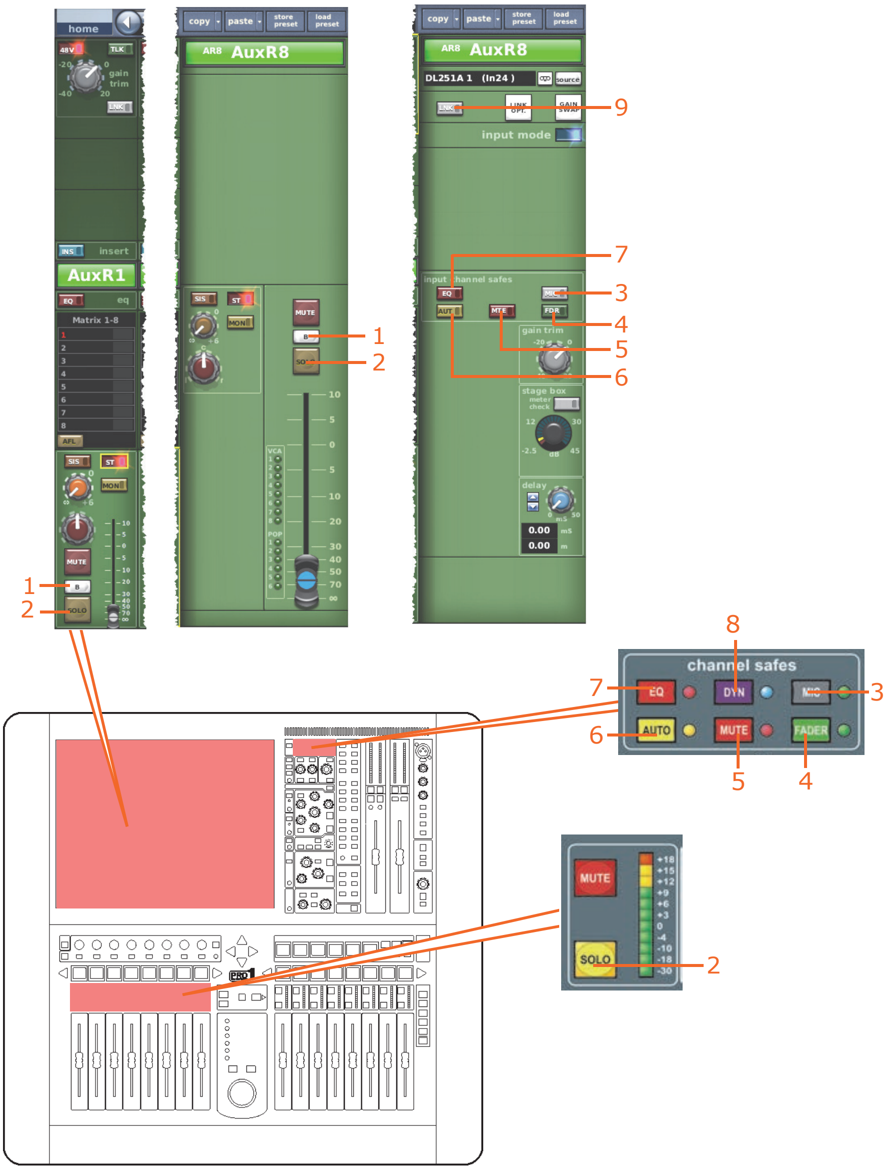
Returns (Aux Returns)

The return safes are selected via the input channel safes section of the channel processing area and the configuration processing area of the GUI channel strip.



Return parameters not affected by the safes

The following shows you which return parameters are *not* affected by each safe.



Item	*Control*	*Parameter*
1	**B** switch	Solo B on/off
2	**SOLO** switch	Solo on/off
3	**MIC** switch	Mic safe on/off
4	**FADER**/[**FDR**] switch	Fader safe on/off
5	**MUTE**/[**MTE**] switch	Mute safe on/off
6	**AUTO**/[**AUT**] switch	Automation safe on/off
7	**EQ** switch	EQ safe on/off
8	**DYN** switch	Dynamic safe on/off
9	**LNK** switch	Stereo linking on/off

EQ safe

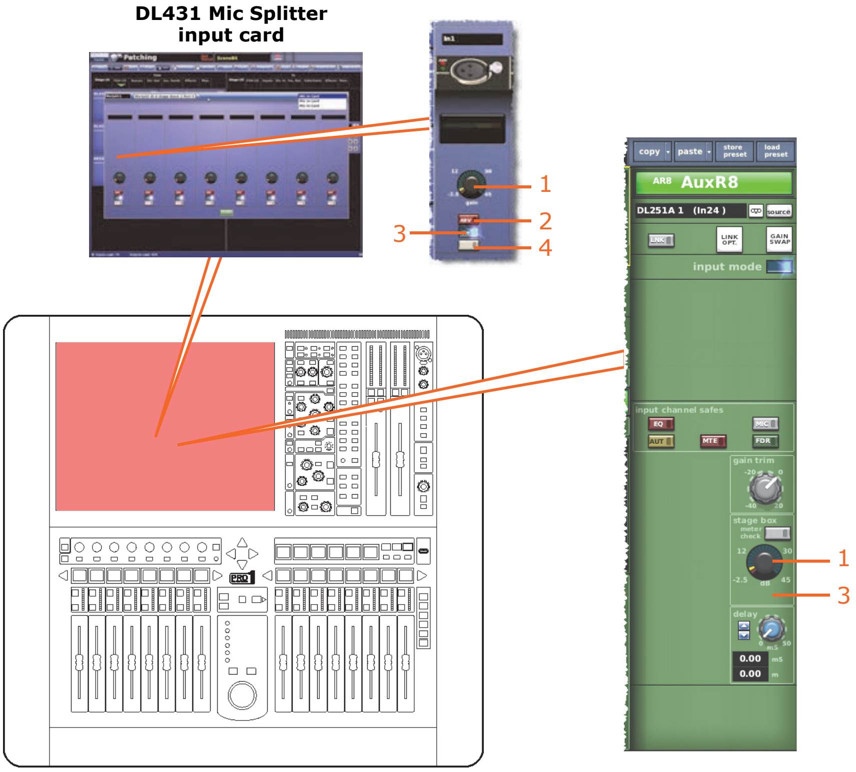
Not applicable.

DYN (dynamic) safe

Not applicable.

MIC safe

This section shows the return parameters protected by the **MIC** safe, which are accessible via the DL431 Mic Splitter configuration (see "Configuring the devices" on page 58).



Item	*Control*	*Parameter*
1	Stage box control knob	Mic gain
2	**48V** switch	48V phantom gain in/out
3	Filter switch* (not shown on GUI channel processing area)	30Hz filter in/out
4	Input zone switch	Input zone in/out

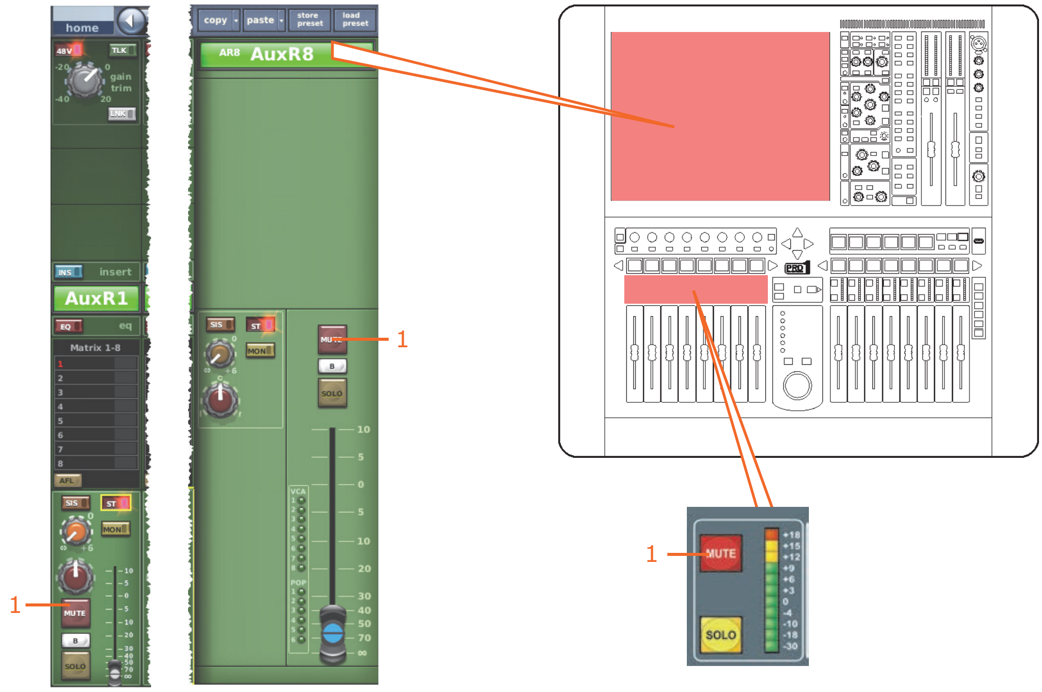
* Applies to tape and primary inputs.

AUTO (automation) safe

All of the return channel parameters are protected by **AUTO** (automation) safe — except, of course, for the ones unaffected by the safes (see "Return parameters not affected by the safes" on page 417).

MUTE safe

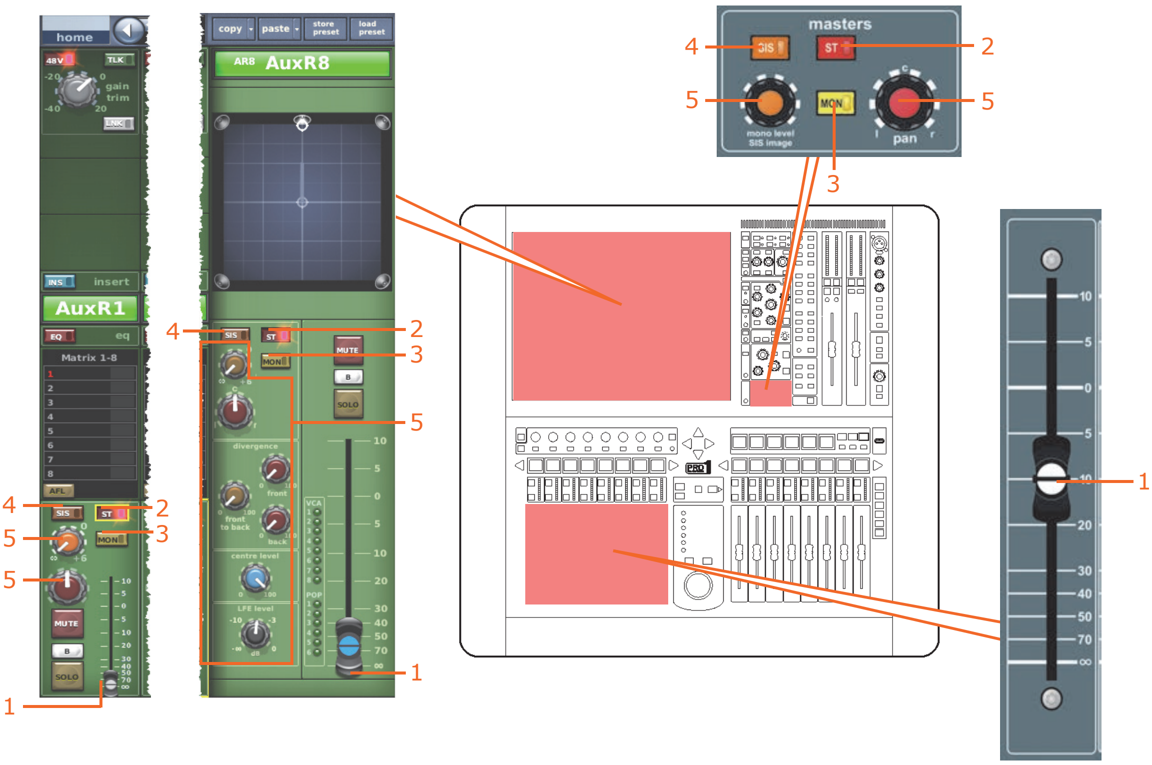
This section shows the return parameters protected by the **MUTE** safe.



Item	Control	Parameter
1	**MUTE** switch	Mute on/off

FADER safe

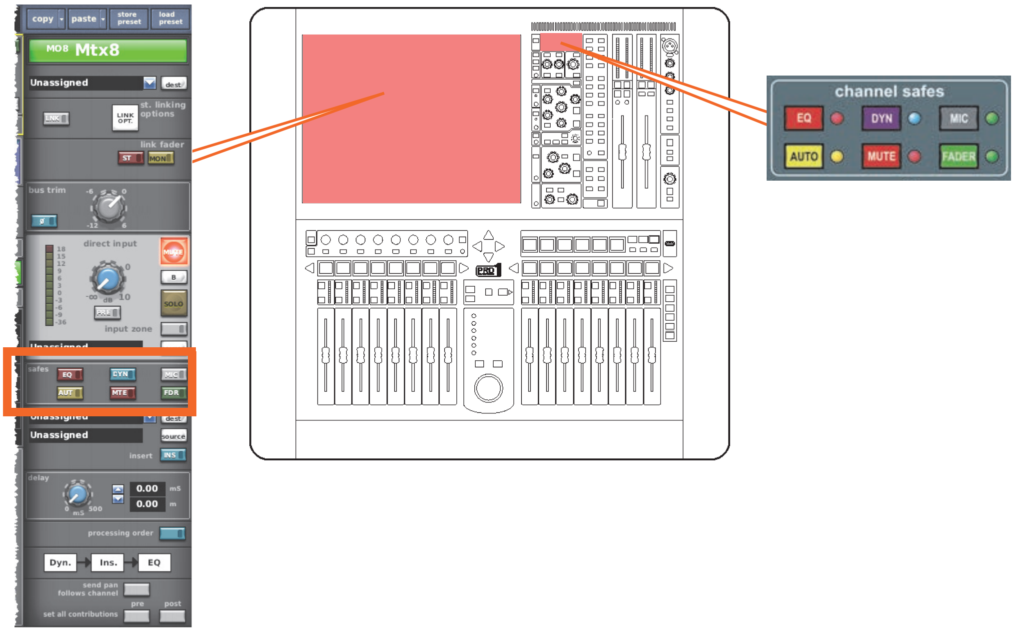
This section shows the return parameters protected by the **FADER** safe.



Item	*Control*	*Parameter*
1	Fader	Fader level
2	**ST** switch	Stereo routing
3	**MON** switch	Mono routing
4	**SIS** switch	Spatial imaging system in/out
5	Panning control knobs	Surround panning (includes all surround sound parameters)

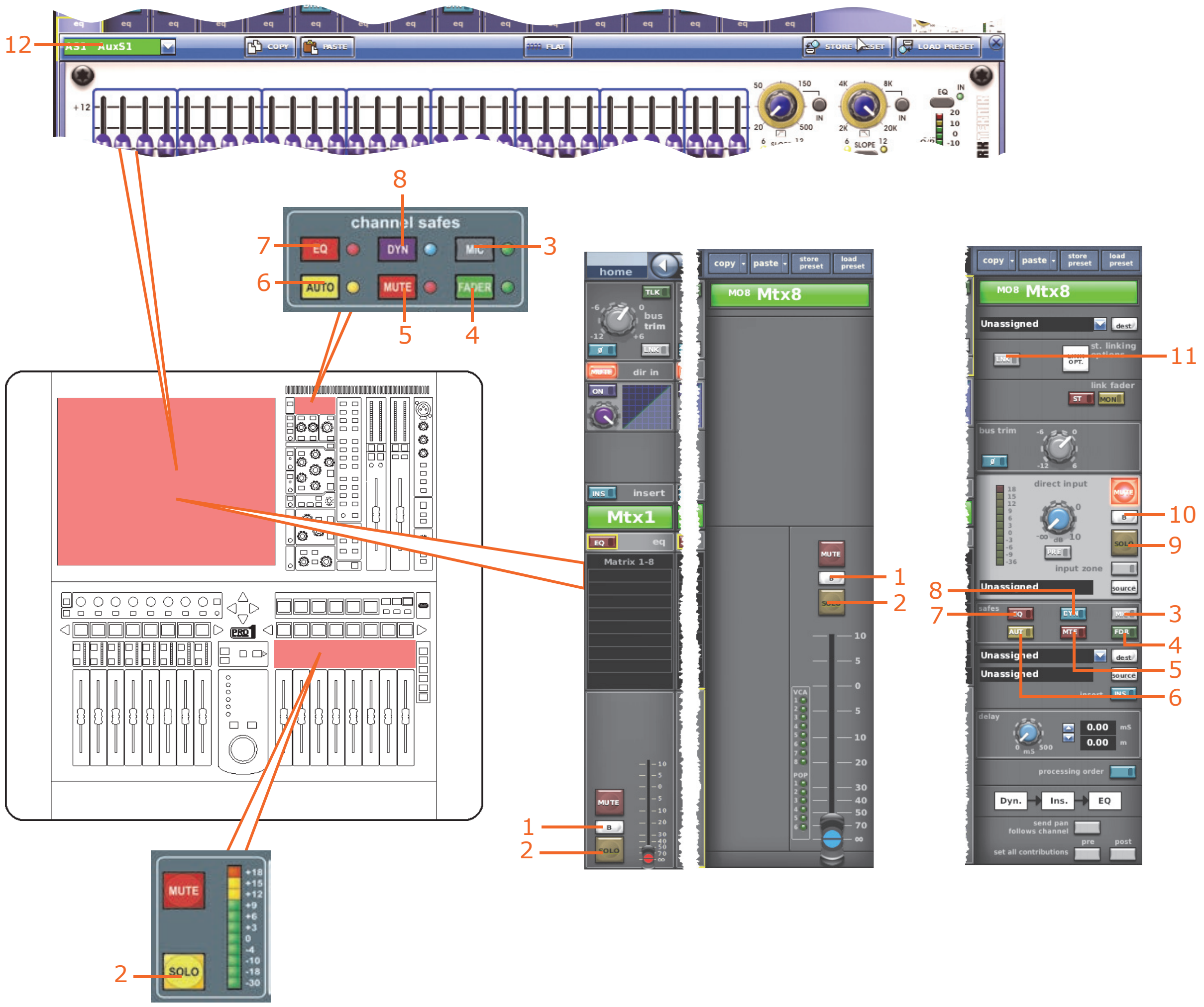
Matrices

The matrix safes are selected via the output channel safes section of the channel processing area and the configuration processing area of the GUI channel strip.



Matrix parameters not affected by the safes

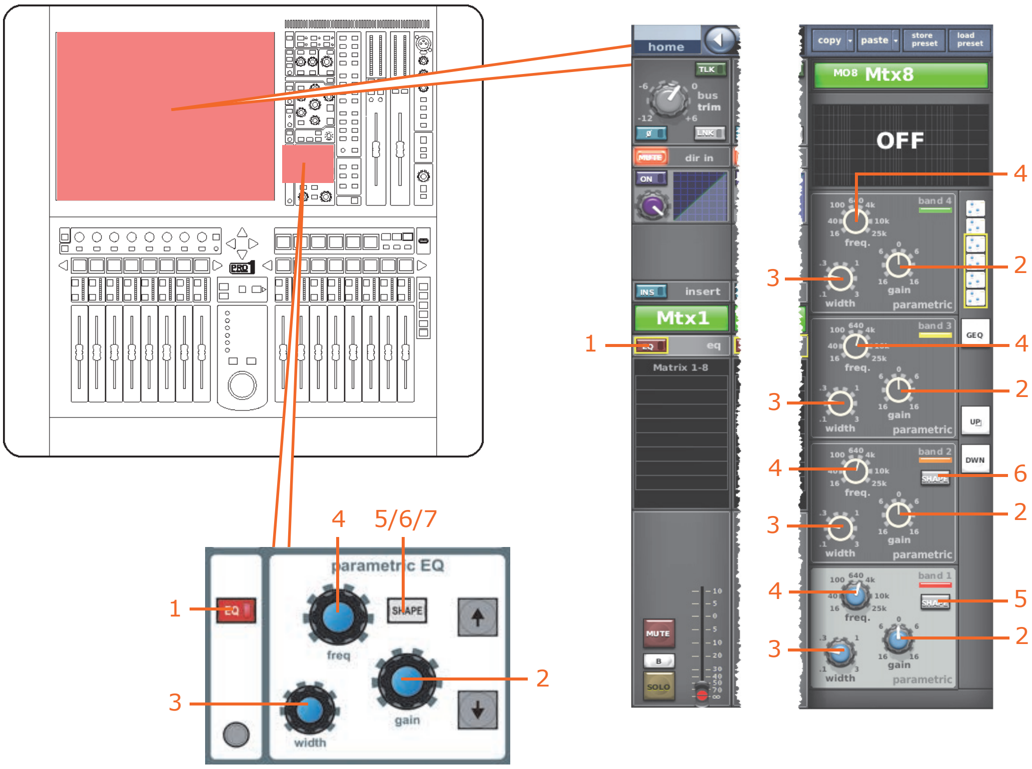
The following shows you which matrix parameters are *not* affected by each safe.



Item	Control	Parameter
1	**B** switch	Solo B on/off
2	**SOLO** switch	Solo on/off
3	**MIC** switch	Mic safe on/off
4	**FADER**/[**FDR**] switch	Fader safe on/off
5	**MUTE**/[**MTE**] switch	Mute safe on/off
6	**AUTO**/[**AUT**] switch	Automation safe on/off
7	**EQ** switch	EQ safe on/off
8	**DYN** switch	Dynamic safe on/off
9	**SOLO** switch	Direct input solo on/off
10	**B** switch	Direct input solo B on/off
11	**LNK** switch	Stereo linking on/off
12	Field	GEQ assignment

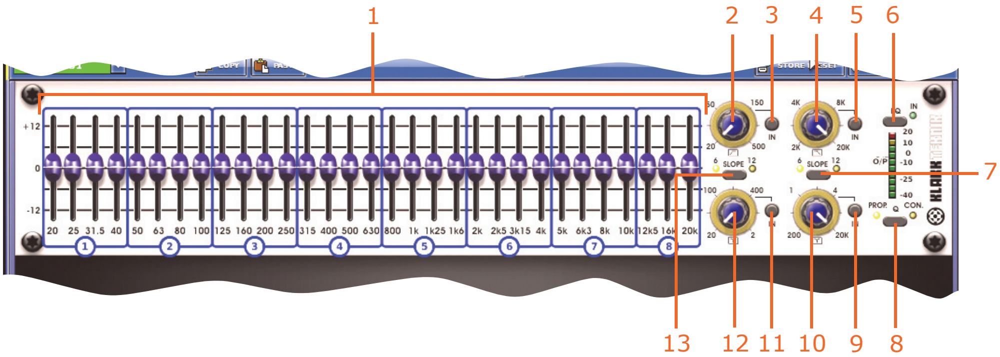
EQ safe

This section shows the matrix parameters protected by the **EQ** safe.



Item	Control	Parameter
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch	Selects band 1 shelving modes: parametric, warm, high pass 6dB or high pass 12dB
6	**SHAPE** switch	Selects band 2 shelving modes: parametric or high pass 24dB
7	**SHAPE** switch (not shown on diagram of GUI above)	Selects band 6 shelving modes: parametric, soft, low pass 6dB or low pass 12dB

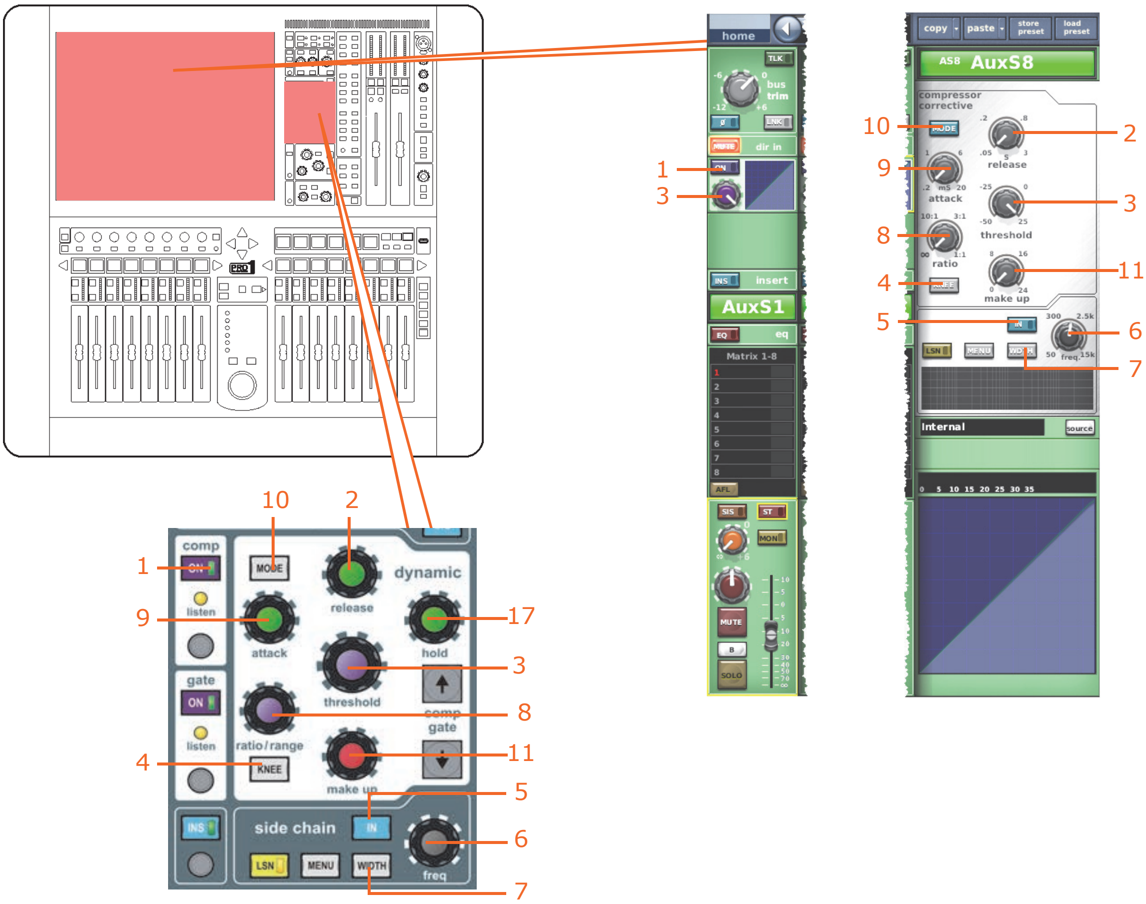
Note: *Although bands 5 and 6 are not shown above, the items in the table also apply. Both bands have items 2, 3 and 4, and band 6 also has item 7.*



Item	Control	Parameter
1	31 faders	Fader positions
2	High pass filter control knob	High pass filter cut off frequency
3	**IN** switch	High pass filter in/out
4	Low pass filter control knob	Low pass filter cut off frequency
5	**IN** switch	Low pass filter in/out
6	**EQ** switch	EQ in/out
7	**SLOPE** switch	Selects low pass filter as 6dB or 12dB
8	**Q** switch	Selects Q mode as proportional (**PROP.**) or constant (**CON.**)
9	**IN** switch	Switches 200Hz - 20kHz notch filter in/out
10	Notch filter control knob	200Hz - 20kHz notch filter frequency
11	**IN** switch	Switches 20Hz - 2kHz notch filter in/out
12	Notch filter control knob	20Hz - 2kHz notch filter frequency
13	**SLOPE** switch	Selects high pass filter as 6dB or 12dB

DYN (dynamic) safe

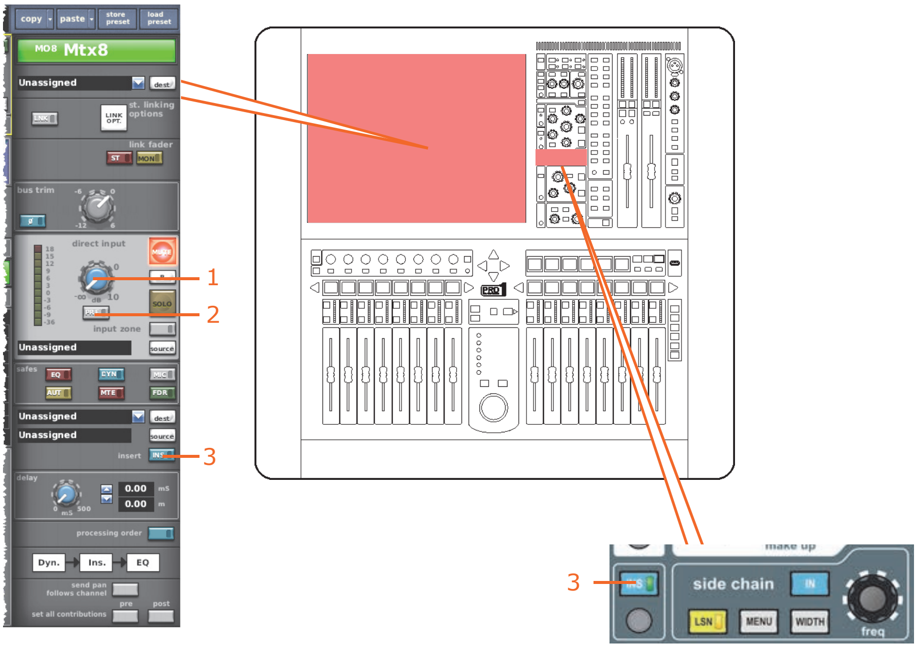
This section shows the matrix parameters protected by the **DYN** safe. Only the corrective compressor is shown, but this is typically the same for the other compressor modes (adaptive, creative, vintage and shimmer).



Item	Control	Parameter
1	**ON** switch	Compressor on/off
2	**release** control knob	Compressor release
3	**threshold** control knob	Compressor threshold
4	**KNEE** pushbutton	Compressor knee (unlabelled) selector: hard, medium and soft
5	**IN** switch	Compressor sidechain in/out
6	**freq** control knob	Compressor sidechain frequency
7	**WIDTH** pushbutton	Compressor sidechain width selector: 2 Oct, 1 Oct and 0.3 Oct
8	**ratio** control knob	Compressor ratio
9	**attack** control knob	Compressor attack
10	**MODE** pushbutton	Compressor mode selector — corrective, adaptive, creative, vintage or shimmer
11	**make up** control knob	Compressor gain

MIC safe

This section shows the matrix parameters protected by **MIC** safe.



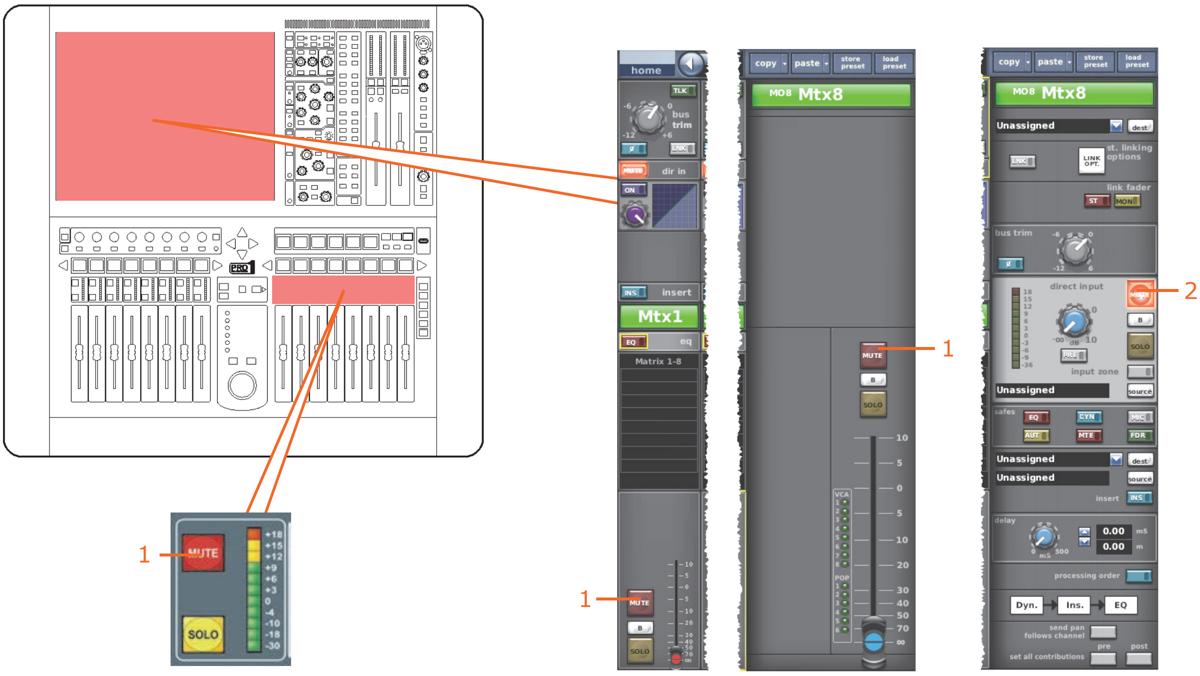
Item	Control	Parameter
1	Level control knob	Direct input level
2	**PRE** switch	Direct input pre- in/out
3	**INS** switch	Insert in/out

AUTO (automation) safe

All of the matrix channel parameters are protected by **AUTO** (automation) safe — except, of course, for the ones unaffected by the safes (see "Matrix parameters not affected by the safes" on page 422).

MUTE safe

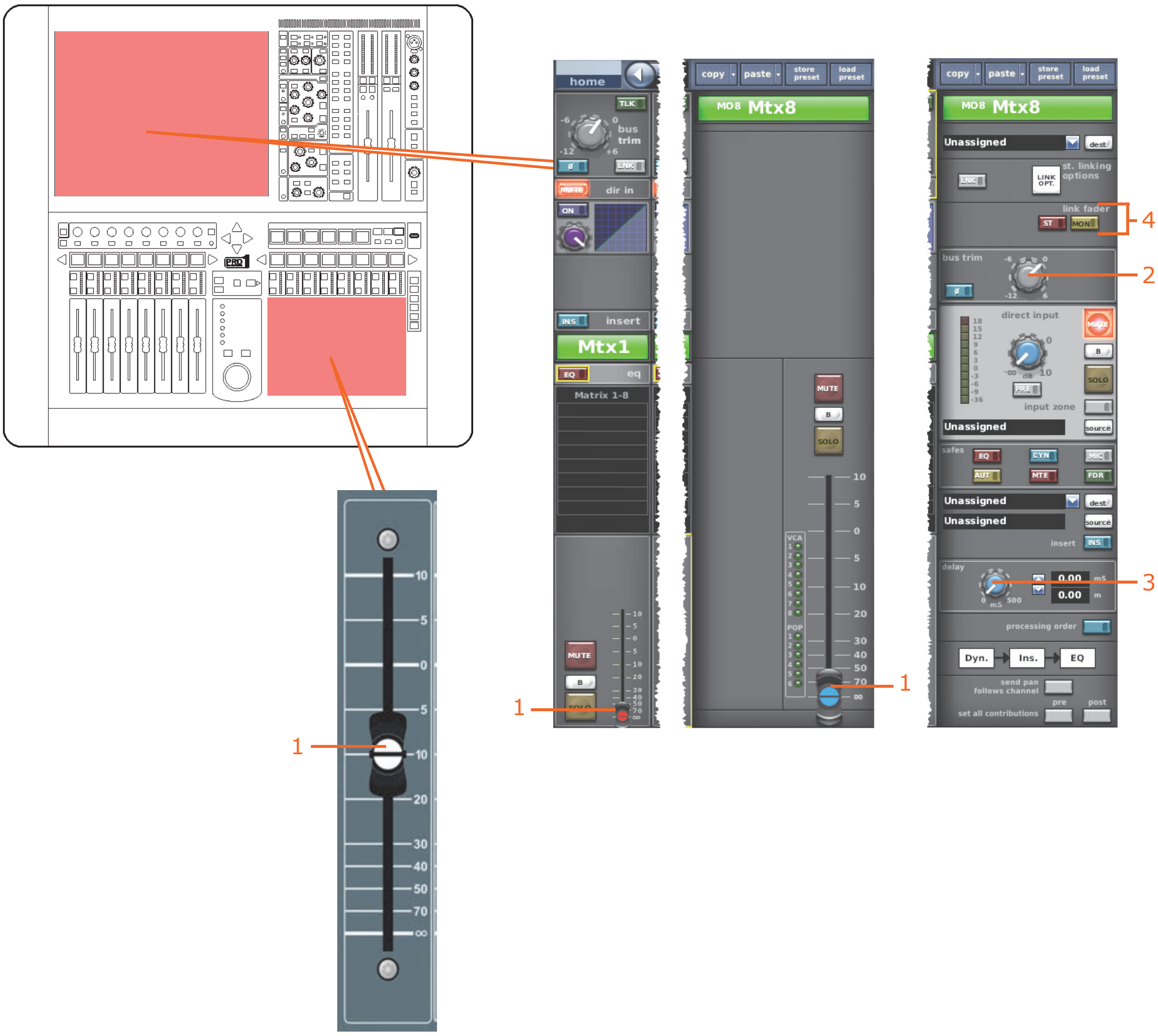
This section shows the matrix parameters protected by the **MUTE** safe.



Item	*Control*	*Parameter*
1	**MUTE** switch	Mute on/off
2	**MUTE** switch	Direct input mute on/off

FADER safe

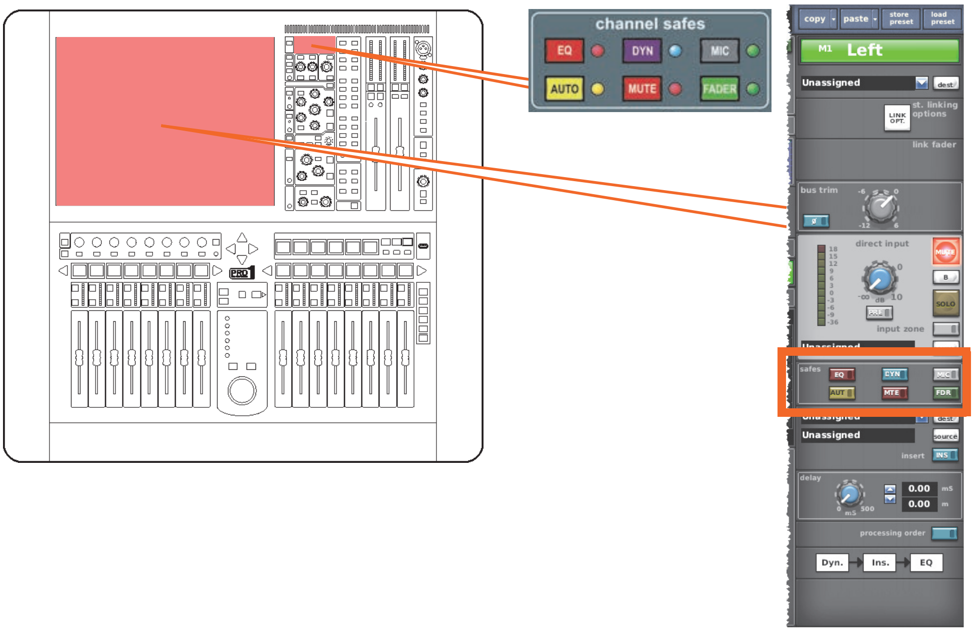
This section shows the matrix parameters protected by the **FADER** safe.



Item	Control(s)	Parameter
1	Fader	Fader level
2	**bus trim** control knob	Bus trim level
3	**delay** control knob	Delay time
4	**ST** switch, **MON** switch	Linking to stereo/mono master fader switch

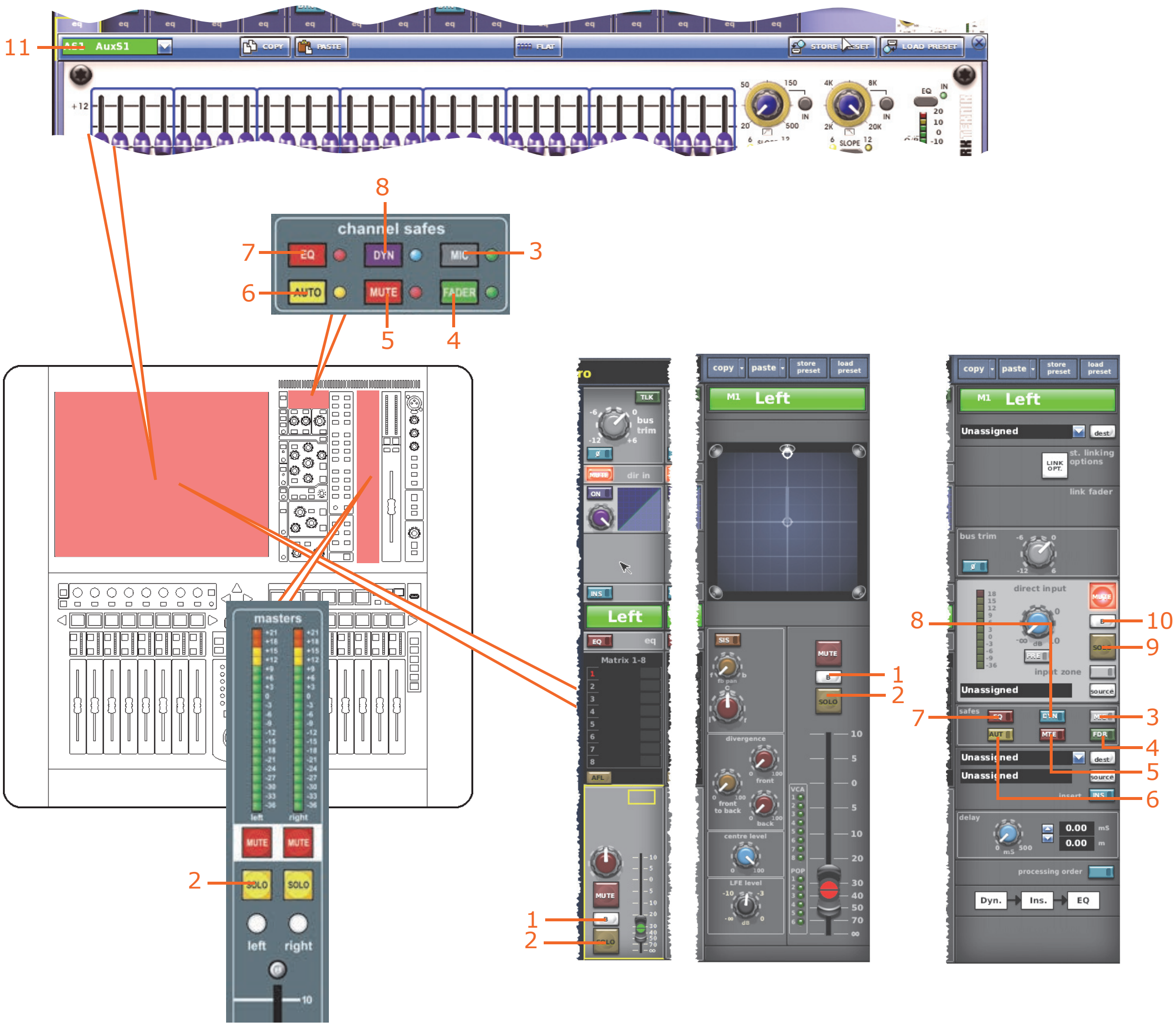
Masters

The master safes are selected via the output channel safes section of the channel processing area and the configuration processing area of the GUI channel strip.



Master parameters not affected by the safes

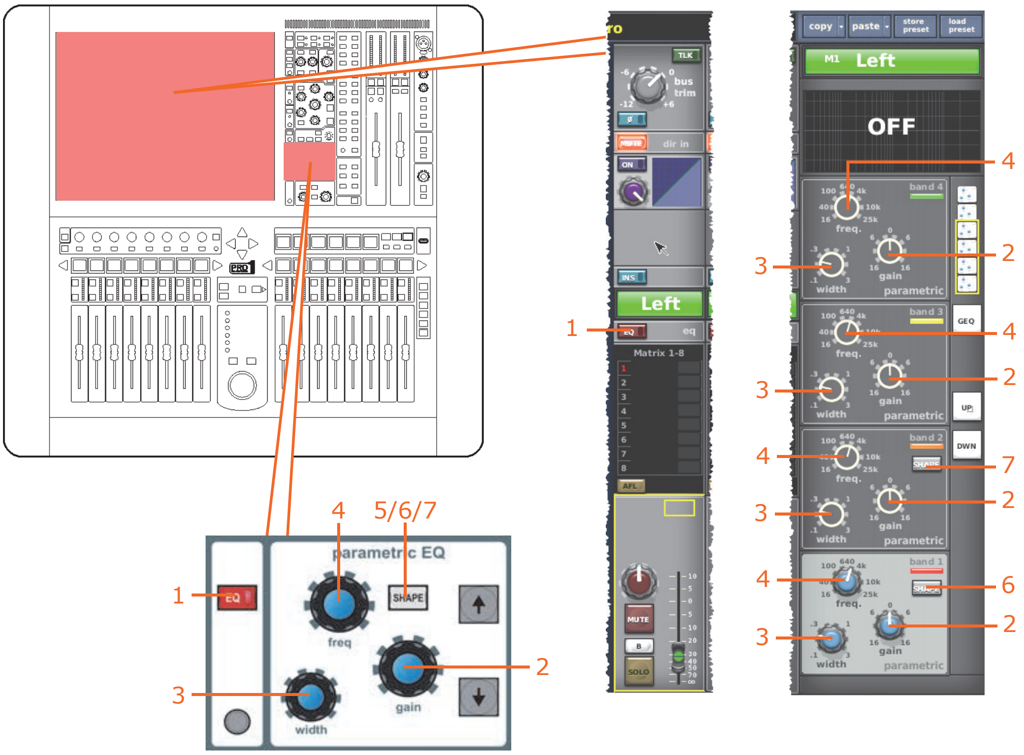
The following shows you which parameters are *not* affected by each safe.



Item	Control	Parameter
1	**B** switch	Solo B on/off
2	**SOLO** switch	Solo on/off
3	**MIC** switch	Mic safe on/off
4	**FADER**/[**FDR**] switch	Fader safe on/off
5	**MUTE**/[**MTE**] switch	Mute safe on/off
6	**AUTO**/[**AUT**] switch	Automation safe on/off
7	**EQ** switch	EQ safe on/off
8	**DYN** switch	Dynamic safe on/off
9	**SOLO** switch	Direct input solo on/off
10	**B** switch	Direct input solo B on/off
11	Field	GEQ assignment

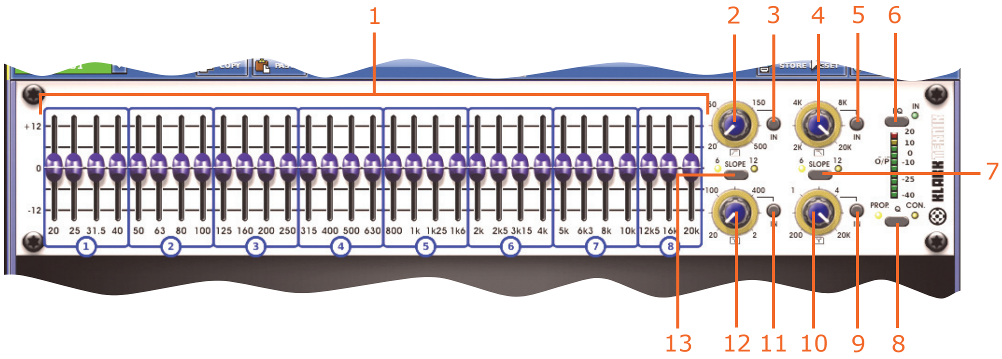
EQ safe

This section shows the master parameters protected by the **EQ** safe.



Item	Control	Parameter
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch (not shown in GUI diagram above)	Selects band 6 shelving modes: parametric, soft, low pass 6dB or low pass 12dB
6	**SHAPE** switch	Selects band 1 shelving modes: parametric, warm, high pass 6dB or high pass 12dB
7	**SHAPE** switch	Selects band 2 shelving modes: parametric or high pass 24dB

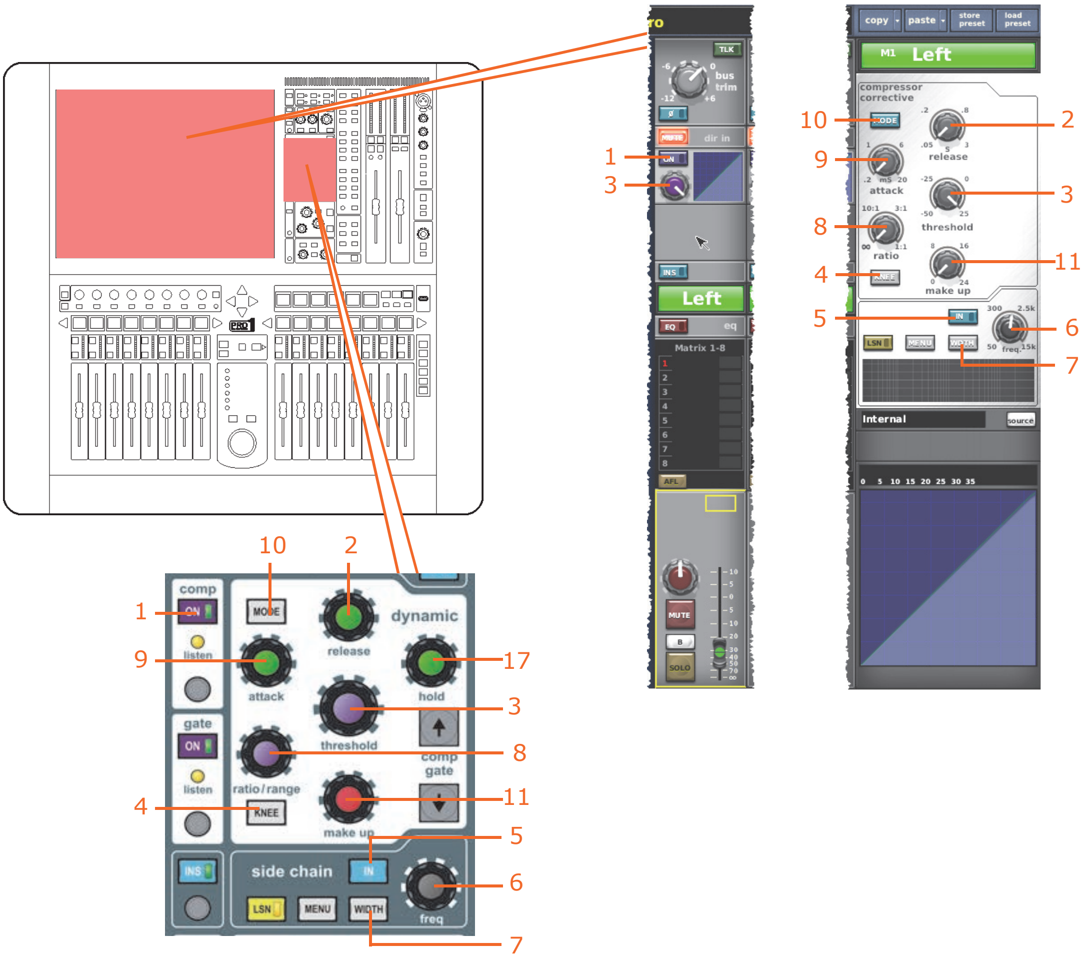
***Note:** Although bands 5 and 6 are not shown above, the items in the table also apply. Both bands have items 2, 3 and 4, and band 6 also has item 5.*



Item	Control	Parameter
1	31 faders	Fader positions
2	High pass filter control knob	High pass filter cut off frequency
3	**IN** switch	High pass filter in/out
4	Low pass filter control knob	Low pass filter cut off frequency
5	**IN** switch	Low pass filter in/out
6	**EQ** switch	EQ in/out
7	**SLOPE** switch	Selects low pass filter as 6dB or 12dB
8	**Q** switch	Selects Q mode as proportional (**PROP.**) or constant (**CON.**)
9	**IN** switch	Switches 200Hz - 20kHz notch filter in/out
10	Notch filter control knob	200Hz - 20kHz notch filter frequency
11	**IN** switch	Switches 20Hz - 2kHz notch filter in/out
12	Notch filter control knob	20Hz - 2kHz notch filter frequency
13	**SLOPE** switch	Selects high pass filter as 6dB or 12dB

DYN (dynamic) safe

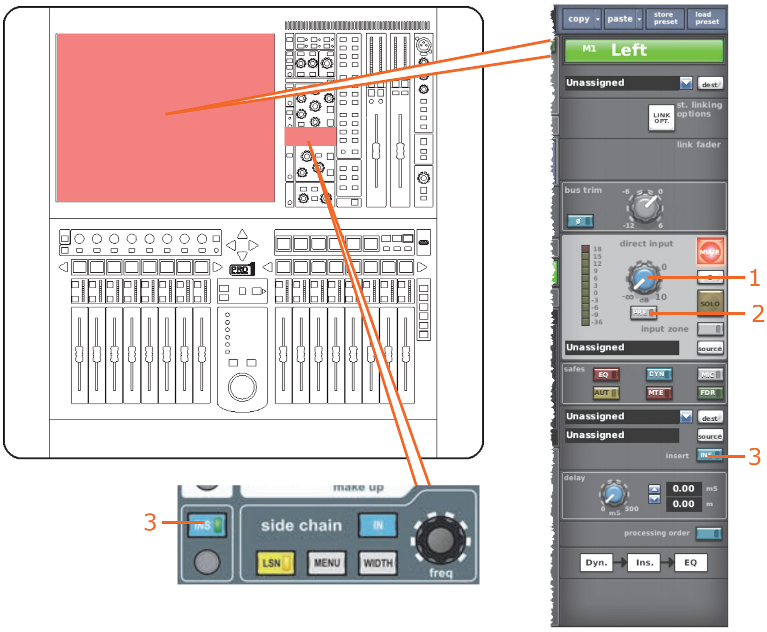
This section shows the master parameters protected by the **DYN** safe. Only the corrective compressor is shown, but this is typically the same for the other compressor modes (adaptive, creative, vintage and shimmer).



Item	*Control*	*Parameter*
1	**ON** switch	Compressor on/off
2	**release** control knob	Compressor release
3	**threshold** control knob	Compressor threshold
4	**KNEE** pushbutton	Compressor knee selector: hard, medium and soft
5	**IN** switch	Compressor sidechain in/out
6	**freq** control knob	Compressor sidechain frequency
7	**WIDTH** pushbutton	Compressor sidechain width: 2 Oct, 1 Oct or 0.3 Oct
8	**ratio** control knob	Compressor ratio
9	**attack** control knob	Compressor attack
10	**MODE** pushbutton	Compressor mode — corrective, adaptive, creative, vintage or shimmer
11	**make up** control knob	Compressor gain

MIC safe

This section shows the master parameters protected by **MIC** safe.



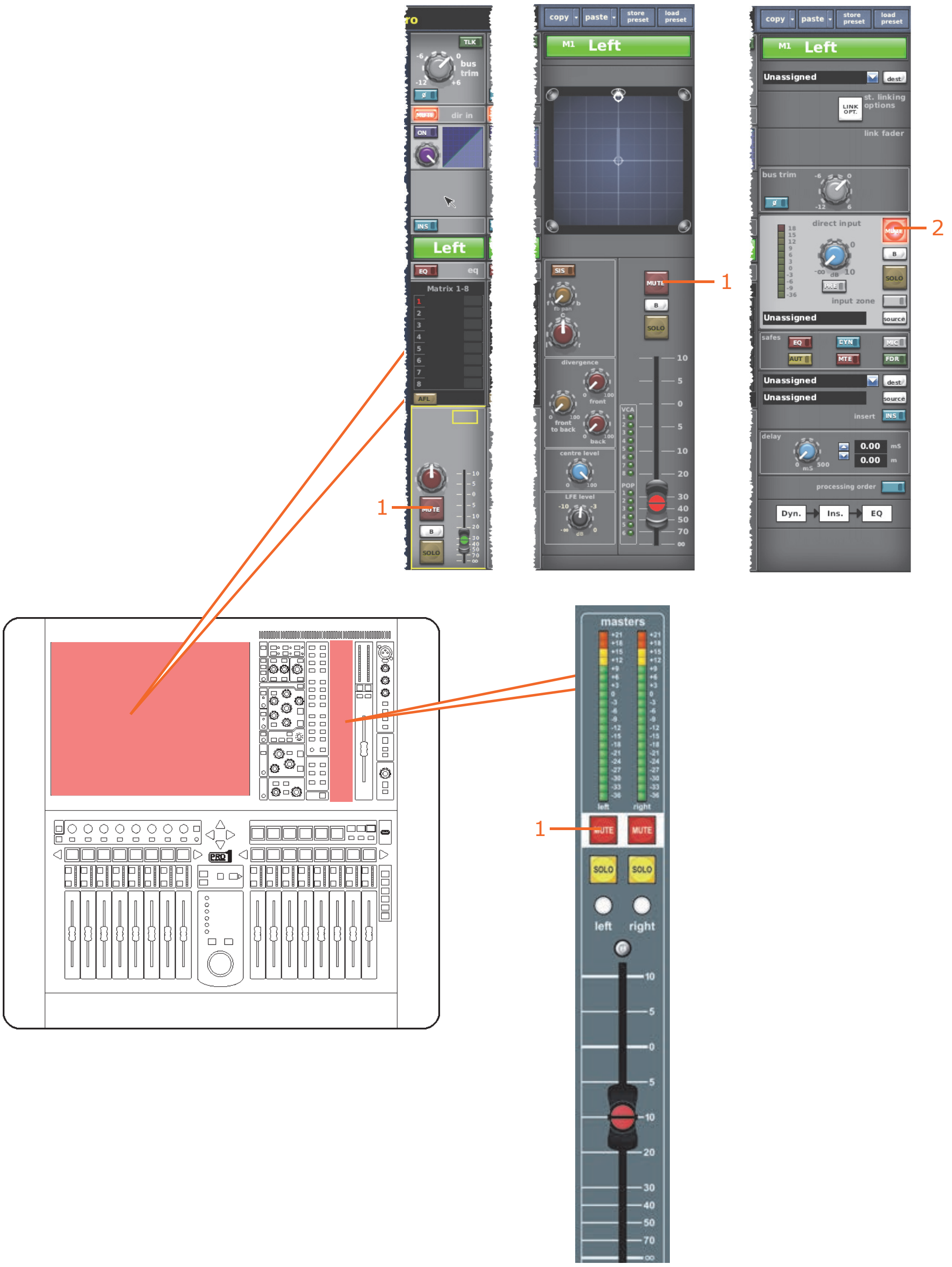
Item	Control	Parameter
1	Level control knob	Direct input level
2	**PRE** switch	Direct input pre- in/out
3	**INS** switch	Insert in/out

AUTO (automation) safe

All of the master channel parameters are protected by **AUTO** (automation) safe — except, of course, for the ones unaffected by the safes (see “Master parameters not affected by the safes” on page 430).

MUTE safe

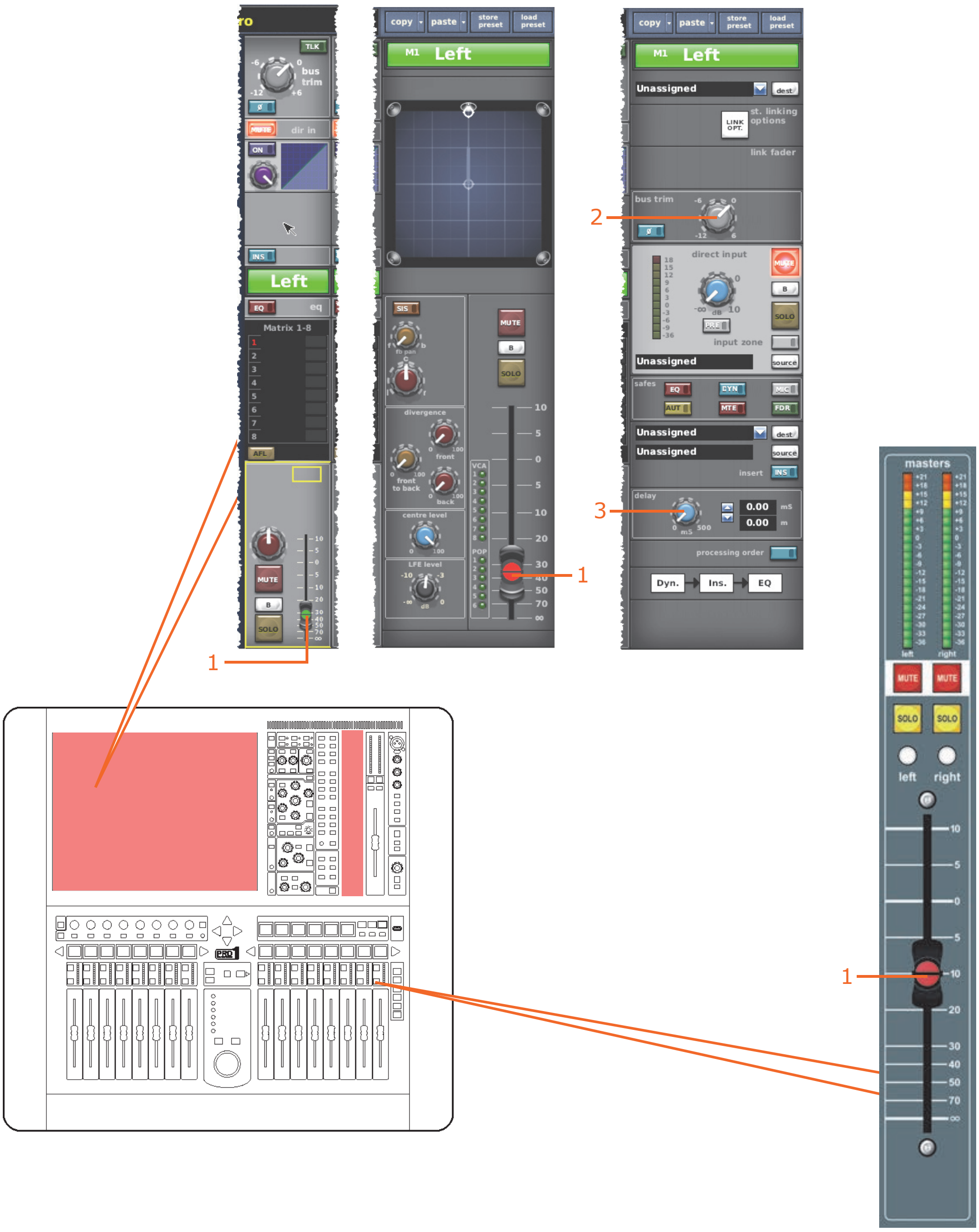
This section shows the master parameters protected by **MUTE** safe.



Item	Control	Parameter
1	**MUTE** switch	Mute on/off
2	**MUTE** switch	Direct input mute on/off

FADER safe

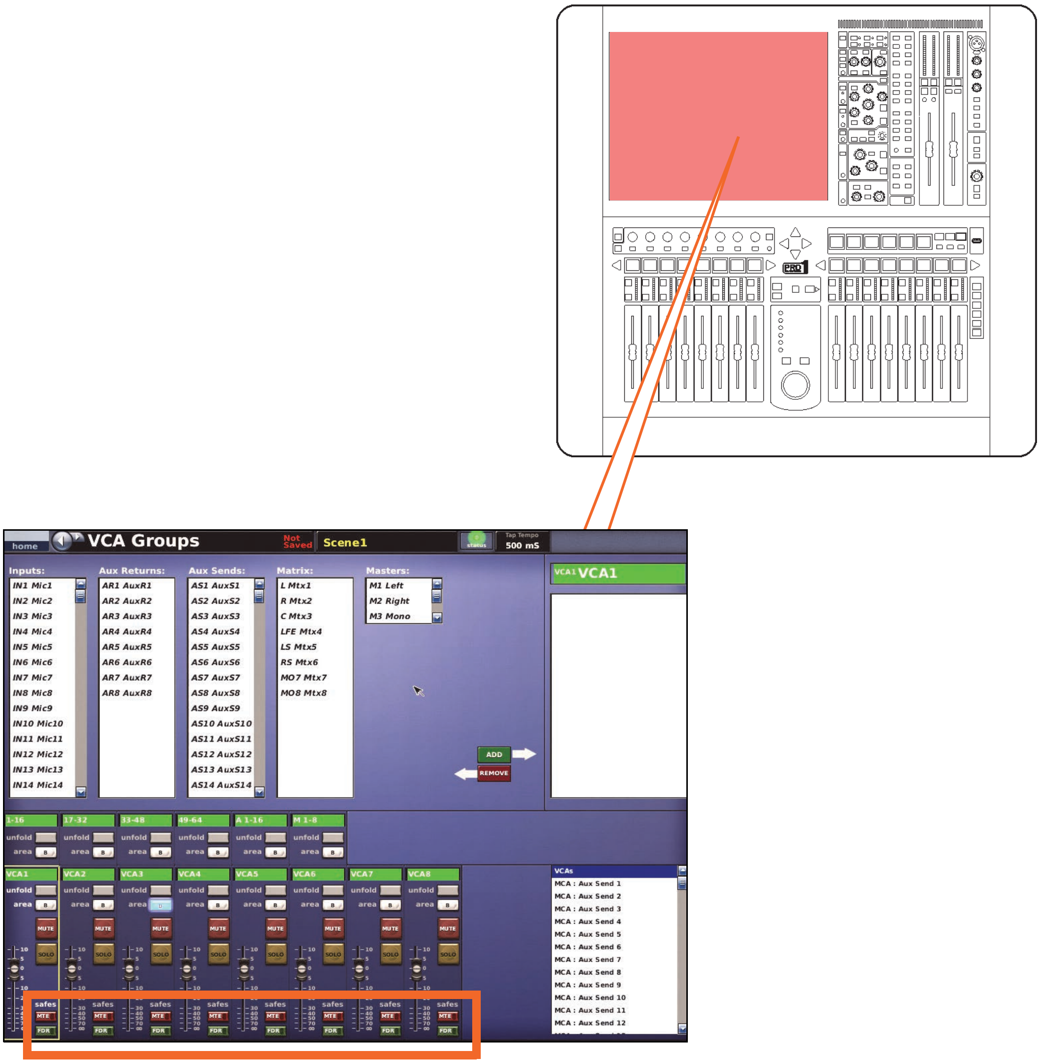
This section shows the master parameters protected by the **FADER** safe.



Item	Control	Parameter
1	Fader	Fader level
2	**bus trim** control knob	Bus trim level
3	**delay** control knob	Delay time

Groups

The group safes — mute and fader — are selected via the **VCA Groups** screen of the GUI (shown below).



EQ safe

Not applicable.

Dynamic safe

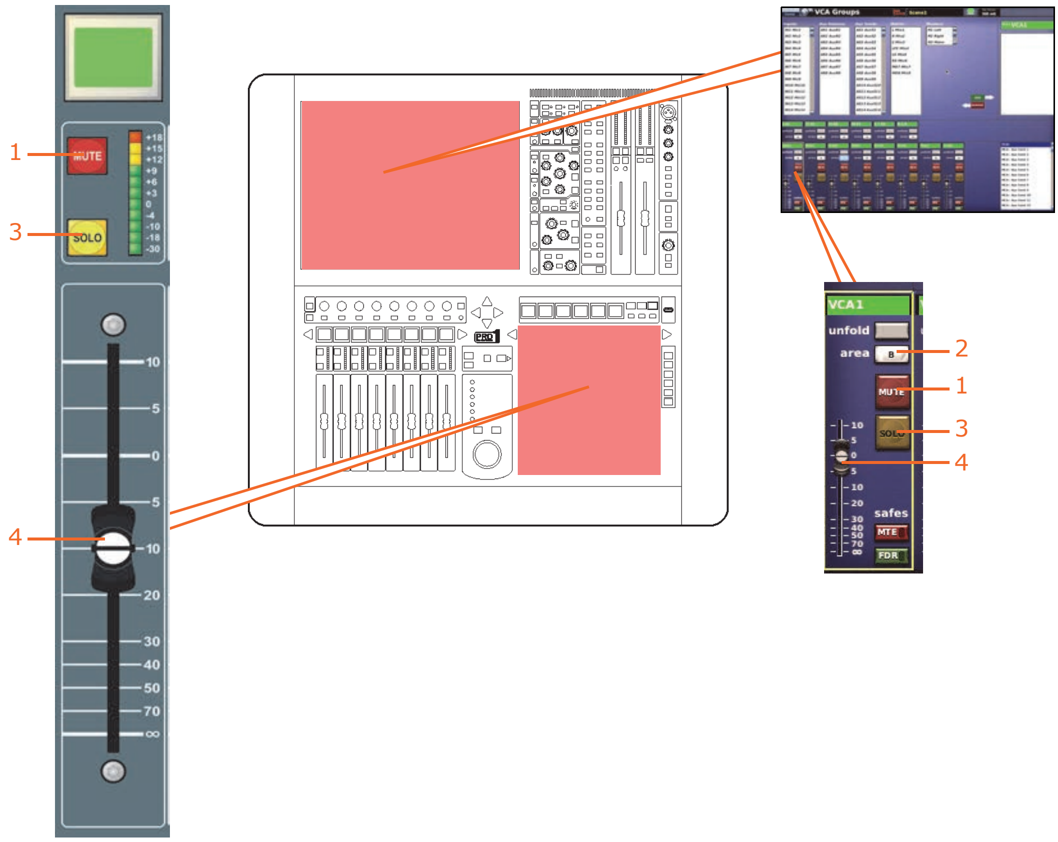
Not applicable.

Mic safe

Not applicable.

Automation safe

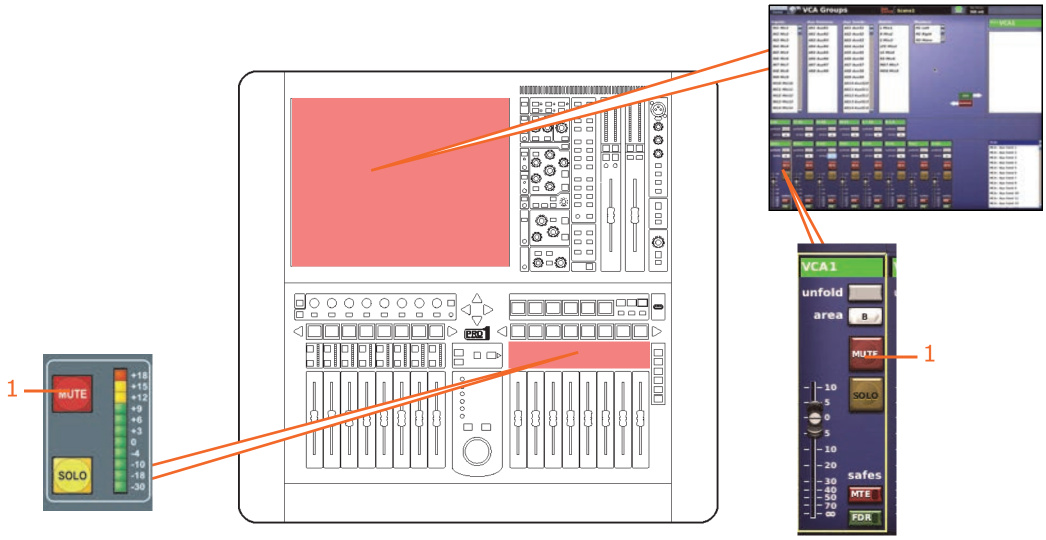
This section shows the VCA group parameters protected by the **AUT** (auto) safe.



Item	*Control*	*Parameter*
1	**MUTE** switch	Mute on/off
2	**B** switch	Solo B on/off
3	**SOLO** switch	Solo on/off
4	Fader	Fader level

Mute (MTE) safe

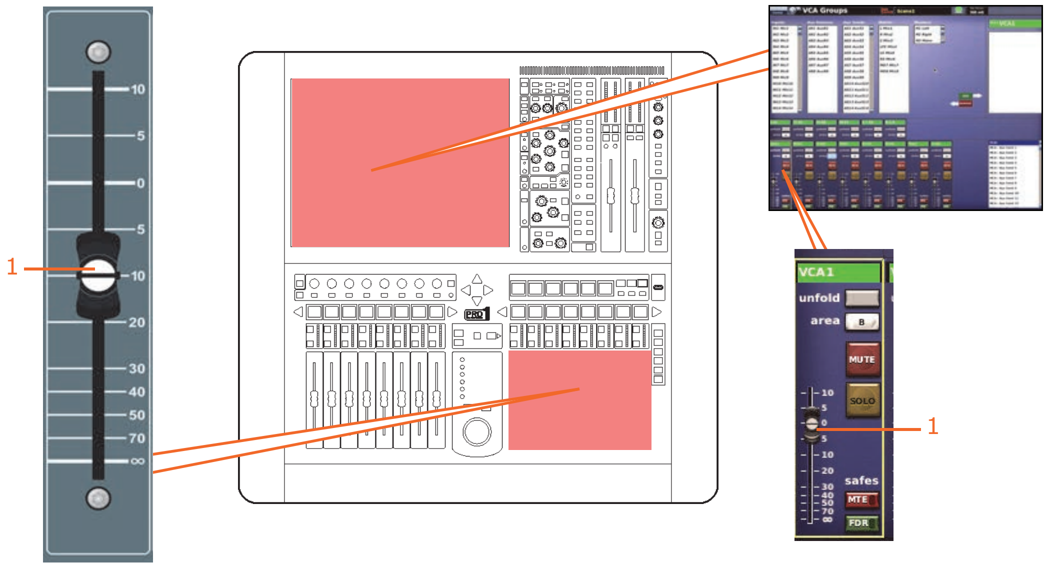
This section shows VCA group parameters protected by the **MTE** (mute) safe.



Item	*Control*	*Parameter*
1	**MUTE** switch	Mute on/off

Fader (FDR) safe

This section shows the VCA group parameters protected by the **FDR** (fader) safe.



Item	*Control*	*Parameter*
1	Fader	Fader level

Appendix I: Parameters Affected By Copy And Paste

This appendix shows the channel parameters affected by copy and paste operations, which are selected via the copy and paste buttons on the GUI (see "Using copy and paste" on page 84).

The following table provides a quick reference for finding the copy and paste parameters per control area for channel in this appendix.

The following table is intended as a reference guide to help you go quickly to the area you want.

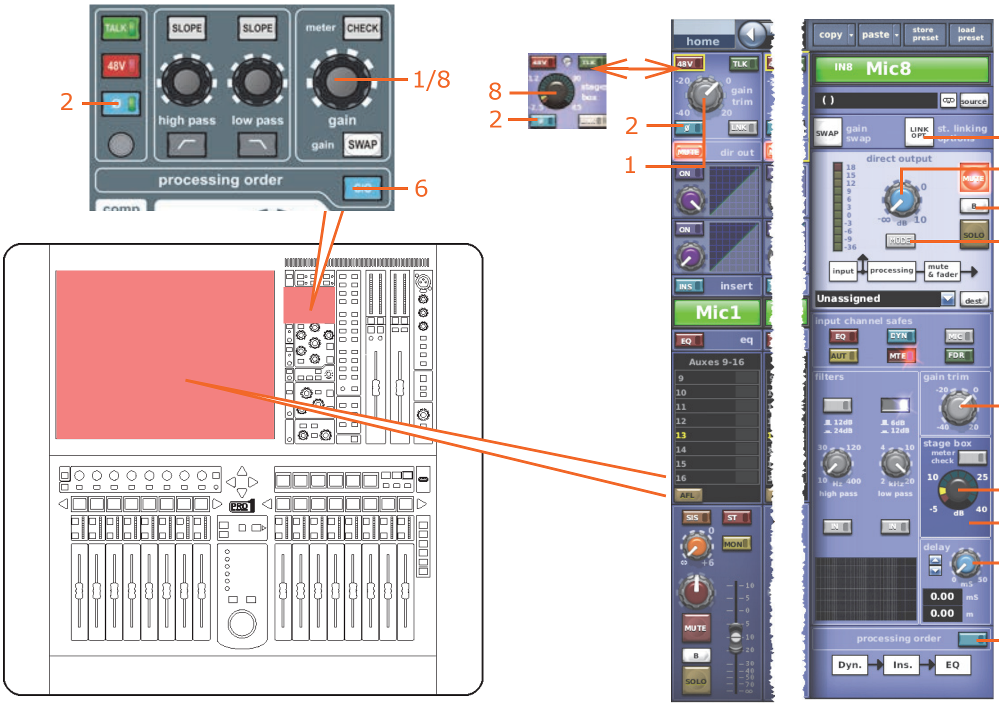
Control area	***Inputs***	***Auxes***	***Returns***	***Matrices***	***Masters***
Configuration	Page 442	Page 448	Page 454	Page 457	Page 462
Compressor	Page 443	Page 449	N/A	Page 458	Page 463
Gate	Page 444	N/A	N/A	N/A	N/A
EQ (GEQ)	Page 445	Page 450	N/A	Page 459	Page 464
Bus sends	Page 446	Page 452	Page 455	N/A	N/A
Master routing/fader section	Page 447	Page 453	Page 456	Page 461	Page 466

Inputs

This section shows you which input channel parameters are affected by copy and paste.

Configuration

This section shows the parameters of the configuration processing area affected by copy and paste.



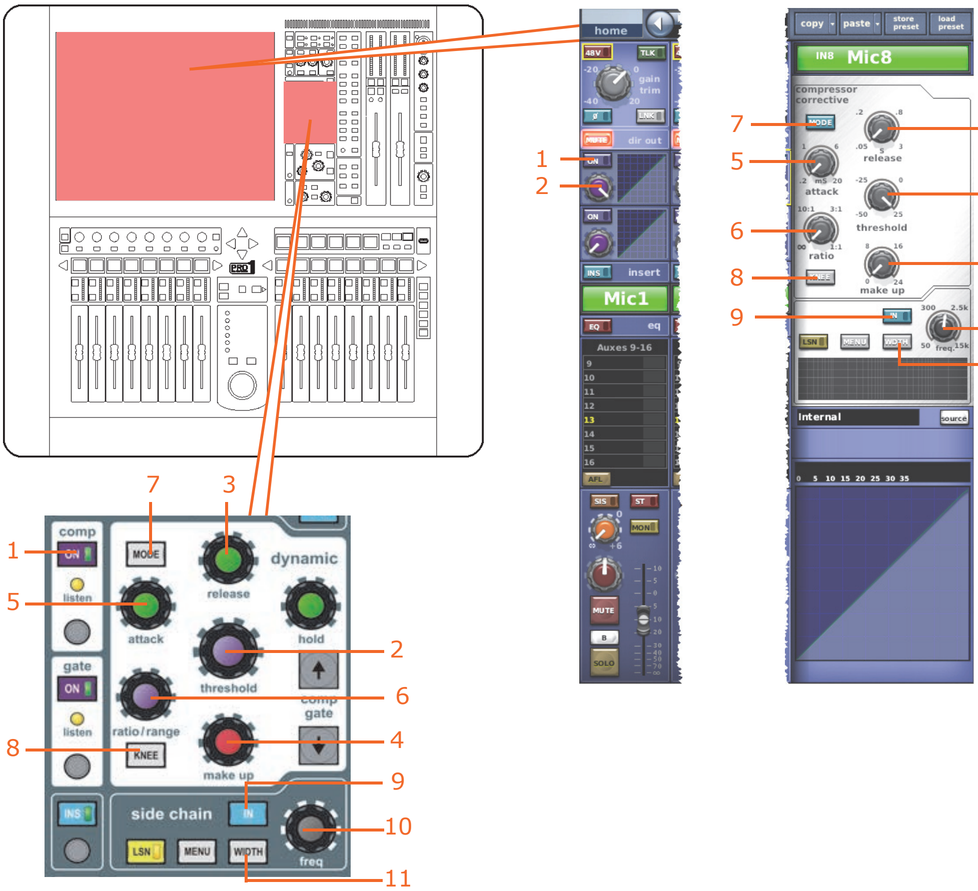
Item	Control	Parameter
1	**gain trim** control knob*	Digital trim
2	**Ø** switch	Phase invert on/off switch
3	Level control knob	Direct output level
4	**B** switch	Direct output solo B on/off
5	**MODE** switch	Direct output tap-off point: "Post-fade and mute", "Pre-mute, pre-processing" and "Pre-mute, post-processing"
6	**C/O** switch	Order of processing: **Dyn.→Ins.→EQ** or **EQ→Ins.→Dyn.**
7	**delay** control knob	Delay time
8	**stage box** control knob*	Gain of remote amplifier
9	Filter switch** (not shown)	30Hz filter in/out
10	**LINK OPT.** pushbutton	Stereo linking options

* Depends on swap status.

** Only when sourced from a DL431 Mic Splitter.

Compressor

This section shows the parameters of the compressor processing area affected by copy and paste.



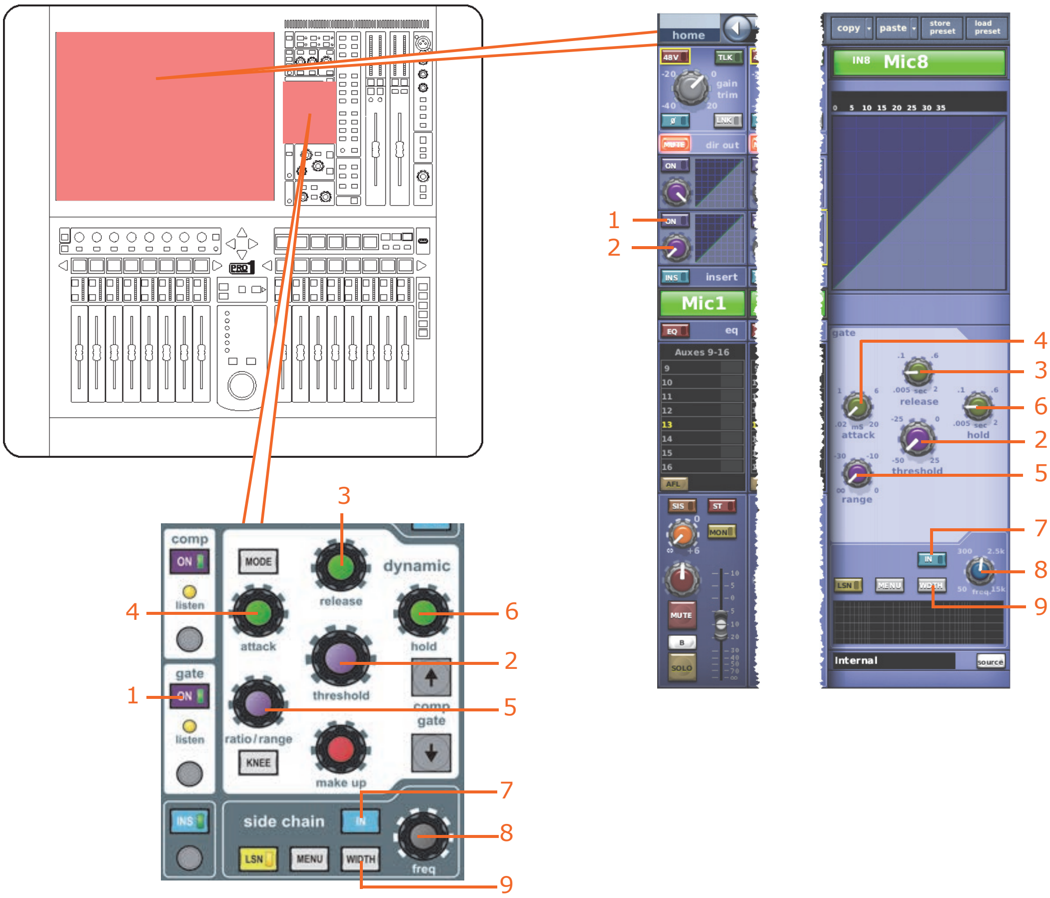
Note: *Only the corrective compressor is shown above, but this is typically the same for the other compressor modes (adaptive, creative and vintage).*

Item	*Control*	*Parameter*
1	**ON** switch	Compressor on/off
2	**threshold** control knob	Compressor threshold
3	**release** control knob	Compressor release
4	**make up** control knob	Compressor make up gain
5	**attack** control knob	Compressor attack
6	**ratio** control knob	Compressor ratio
7	**MODE** pushbutton	Compressor mode selector: corrective (shown above), adaptive, creative and vintage
8	**KNEE** pushbutton	Compressor knee: hard, medium or soft
9	**IN** switch	Compressor sidechain in/out

Item	Control	Parameter
10	**freq** control knob	Compressor sidechain frequency
11	**WIDTH** pushbutton	Compressor sidechain width: 2 Oct, 1 Oct or 0.3 Oct

Gate

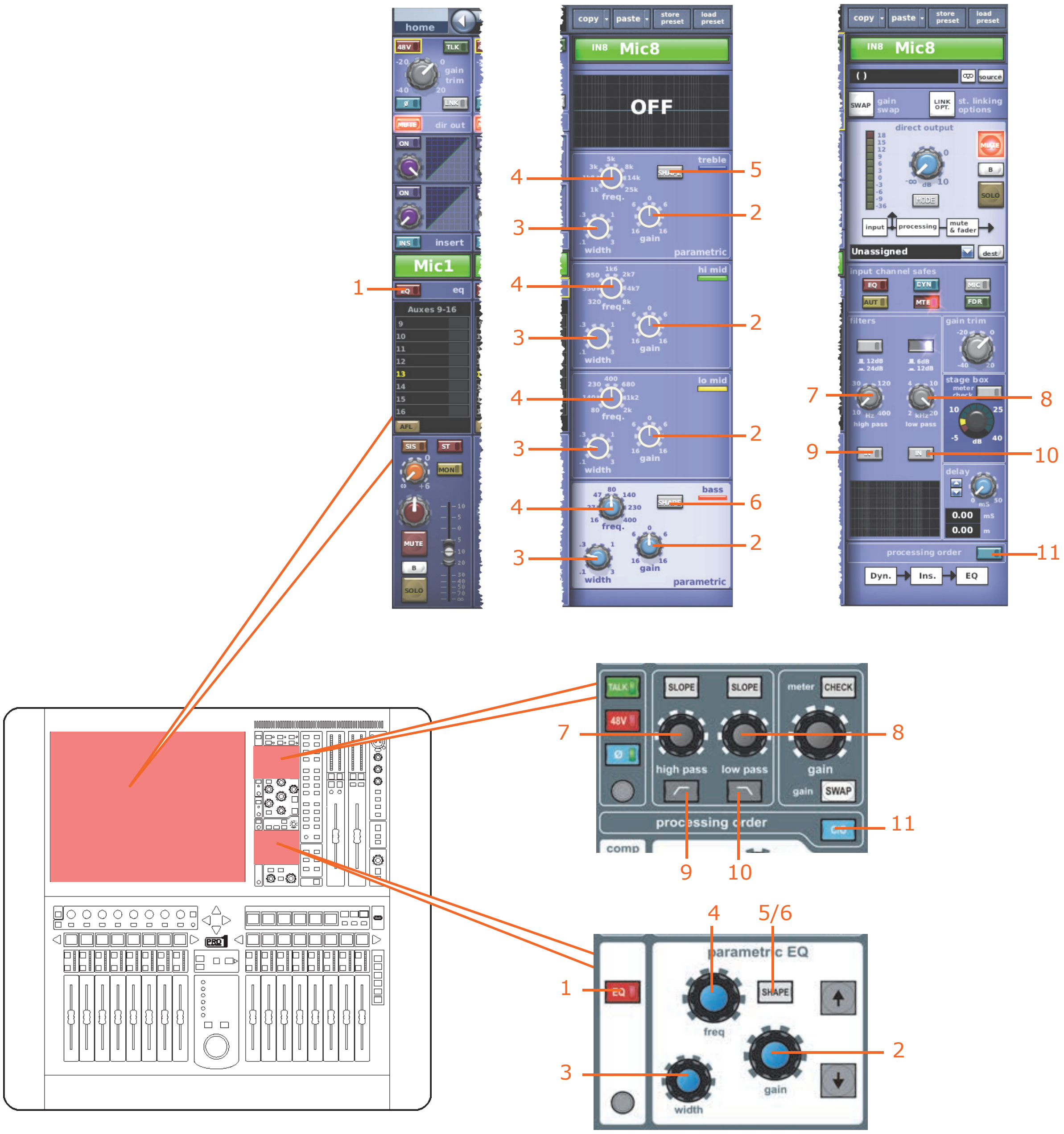
This section shows the parameters of the gate processing area affected by copy and paste.



Item	Control	Parameter
1	**ON** switch	Gate on/off
2	**threshold** control knob	Gate threshold
3	**release** control knob	Gate release
4	**attack** control knob	Gate attack
5	**range** control knob	Gate range
6	**hold** control knob	Gate hold
7	**IN** switch	Gate sidechain in/out
8	**freq** control knob	Gate sidechain frequency
9	**WIDTH** pushbutton	Gate sidechain width: 2 Oct, 1 Oct or 0.3 Oct

EQ

This section shows the parameters of the EQ processing area affected by copy and paste.

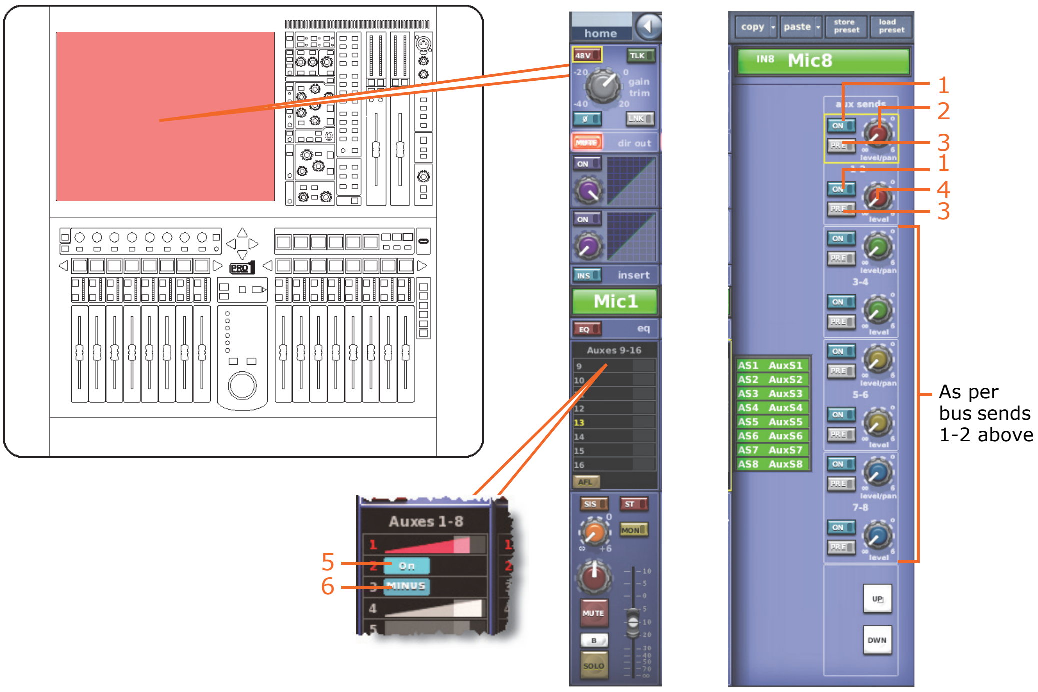


Item	Control	Parameter
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch	Treble shelving mode: parametric, bright, classic or soft
6	**SHAPE** switch	Bass shelving mode: parametric, deep, classic or warm
7	**high pass** control knob	High pass filter frequency

Item	*Control*	*Parameter*
8	**low pass** control knob	Low pass filter frequency
9	/[**IN**] switch	High pass filter in/out
10	/[**IN**] switch	Low pass filter in/out
11	**C/O** switch	To swap the order of processing

Bus sends

This section shows the parameters of the mix sends processing area affected by copy and paste.

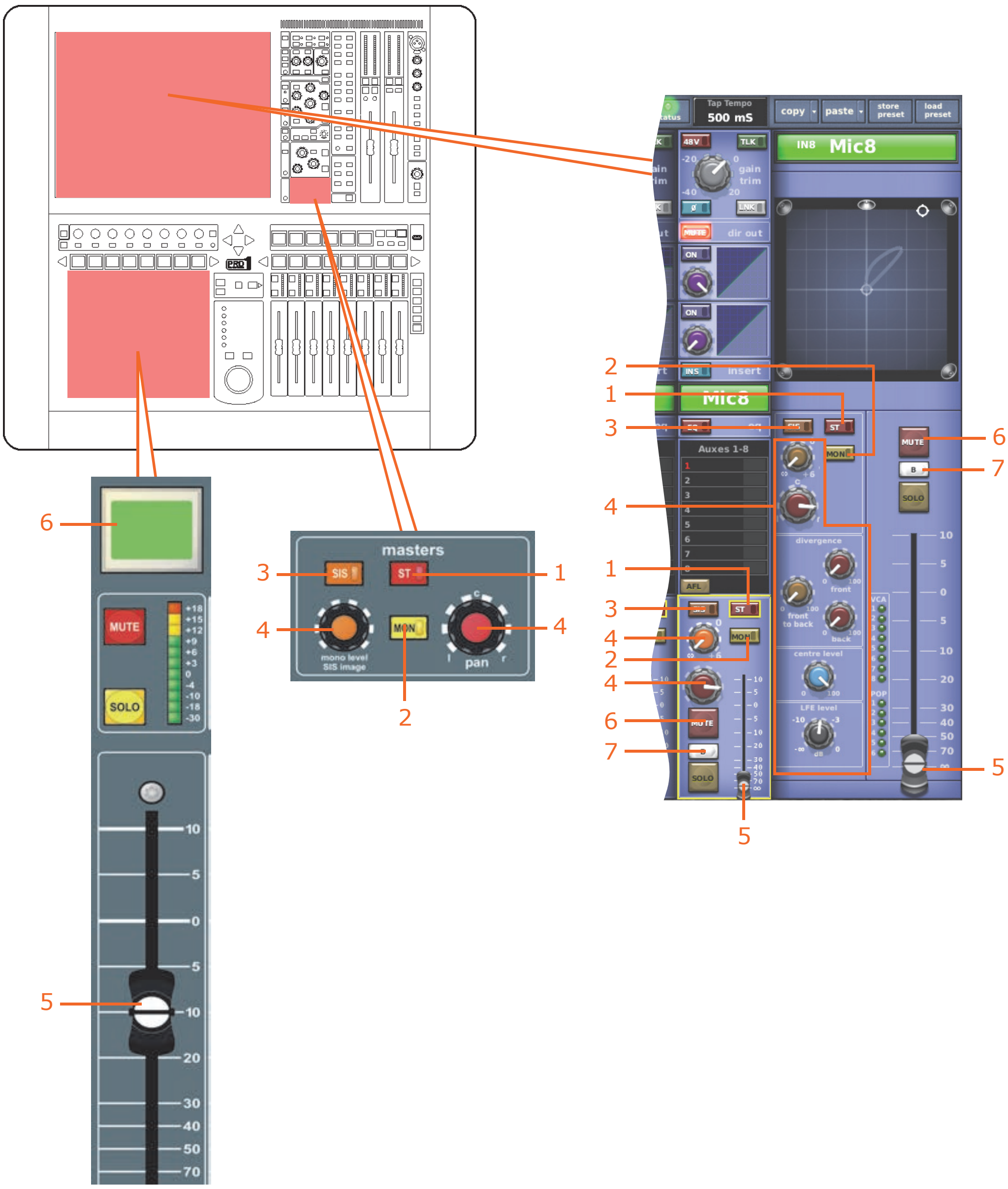


Note: *Only matrix sends 1 to 8 are shown above, but a copy/paste operation affects all aux sends as well.*

Item	*Control*	*Parameter*
1	**ON** switch	Bus send on/off
2	**level/pan** control knob	Bus level, or pan when bus is linked
3	**PRE** switch	Pre-fader on/off
4	**level** control knob	Bus level
5	**On** switch	Aux bus send on/off — only available when aux bus is in group mode
6	**MINUS** switch	Aux bus send mute on/off — only available when aux bus is in mix minus mode

Master routing

This section shows the parameters of the master routing processing area affected by copy and paste.



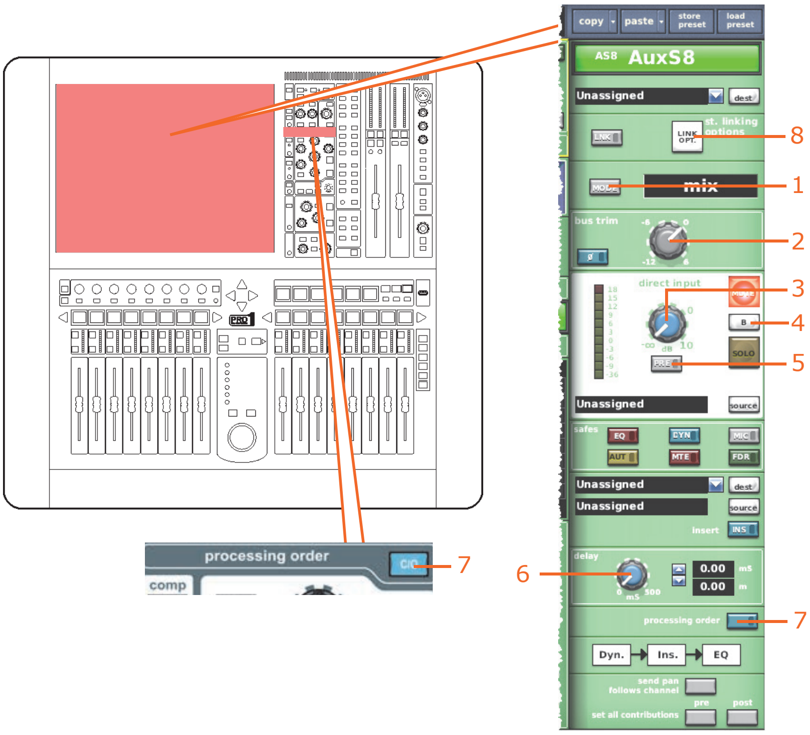
Item	Control	Parameter
1	**ST** switch	Stereo on/off
2	**MON** switch	Mono on/off
3	**SIS** switch	Spatial imaging system on/off
4	Panning control knobs	Surround panning (includes all surround sound parameters)
5	Fader	Level
6	**MUTE** switch	Mute on/off
7	**B** switch	Solo B on/off

Aux

This section shows you which aux channel parameters are affected by copy and paste.

Configuration

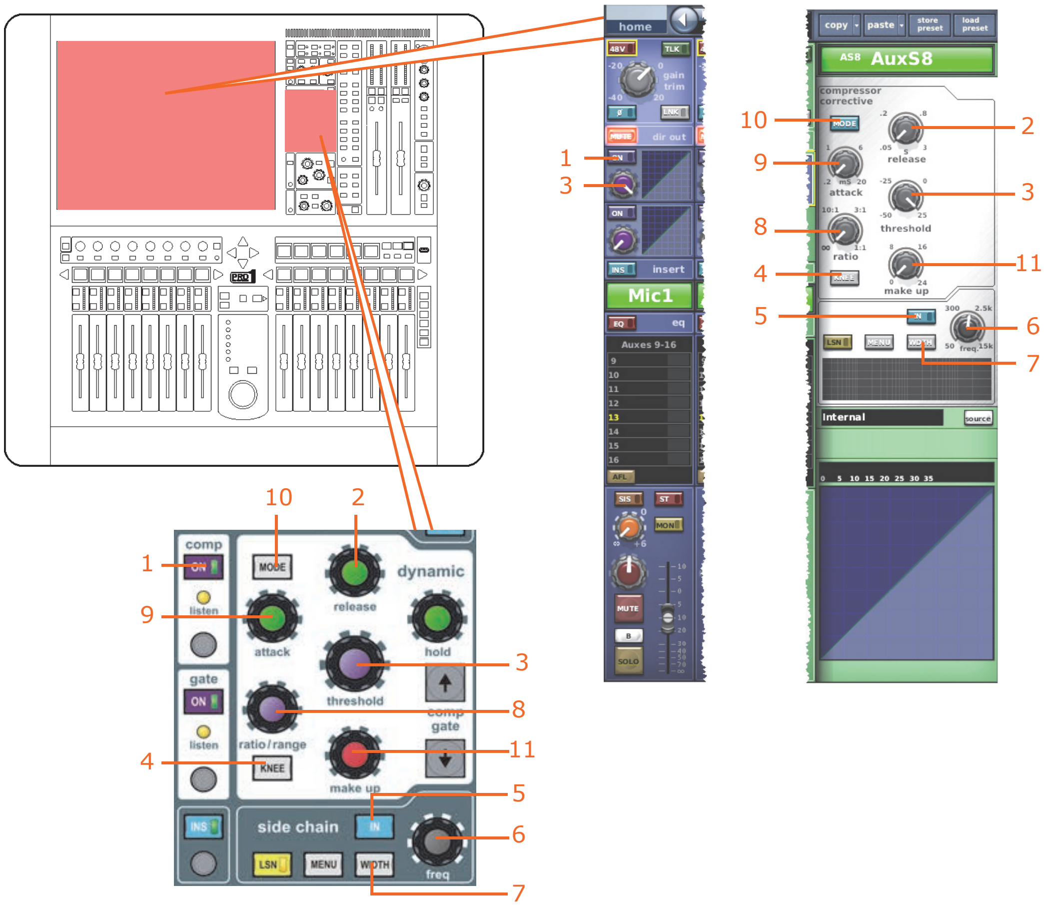
This section shows the parameters of the configuration processing area affected by copy and paste.



Item	Control	Parameter
1	**MODE** pushbutton	Bus mode: mix, group or mix minus
2	**bus trim** control knob	Bus trim level
3	Level control knob	Direct input level
4	**B** switch	Direct input solo B on/off
5	**PRE** switch	Direct input pre- in/out
6	**delay** control knob	Delay time
7	**C/O** switch	Order of processing: **Dyn.→Ins.→EQ** or **EQ→Ins.→Dyn.**
8	**LINK OPT.** button	Stereo linking options

Compressor

This section shows the parameters of the compressor processing area affected by copy and paste. Only corrective compressor shown below, but this is typically the same for the other compressor modes.



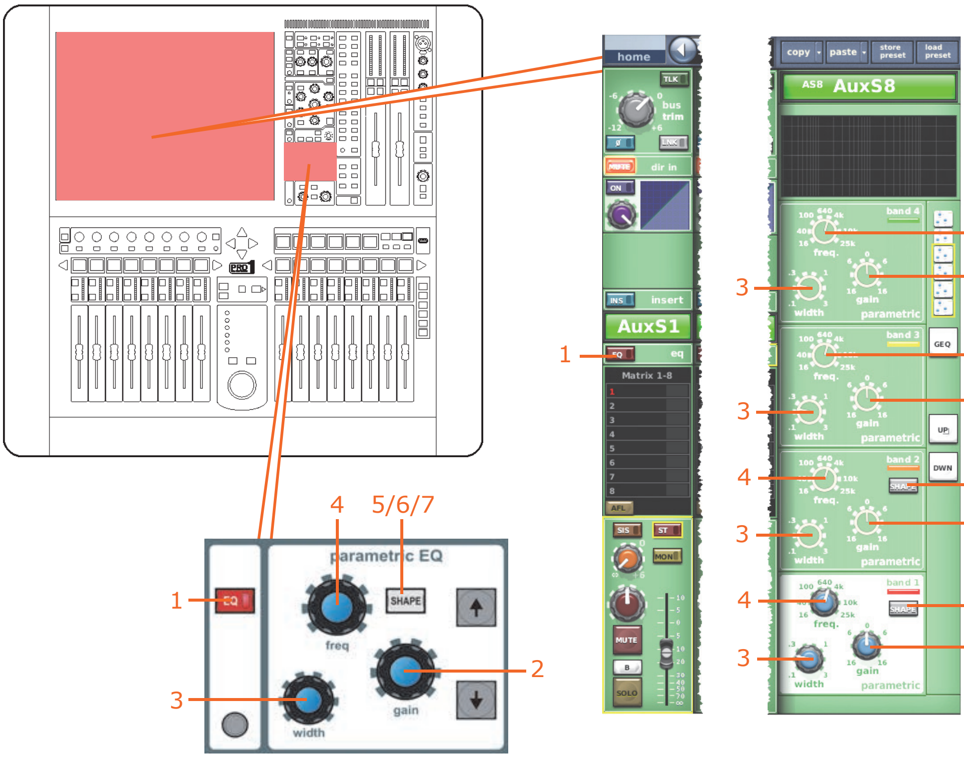
Item	*Control*	*Parameter*
1	**ON** switch	Compressor on/off
2	**release** control knob	Compressor release
3	**threshold** control knob	Compressor threshold
4	**KNEE** pushbutton	Compressor knee: hard, medium or soft
5	**IN** switch	Compressor sidechain in/out
6	**freq** control knob	Compressor sidechain frequency
7	**WIDTH** pushbutton	Compressor sidechain: 2 Oct, 1 Oct or 0.3 Oct
8	**ratio** control knob	Compressor ratio
9	**attack** control knob	Compressor attack
10	**MODE** pushbutton	Compressor mode: corrective, adaptive, creative, vintage or shimmer
11	**make up** control knob	Compressor gain

Gate

Not applicable.

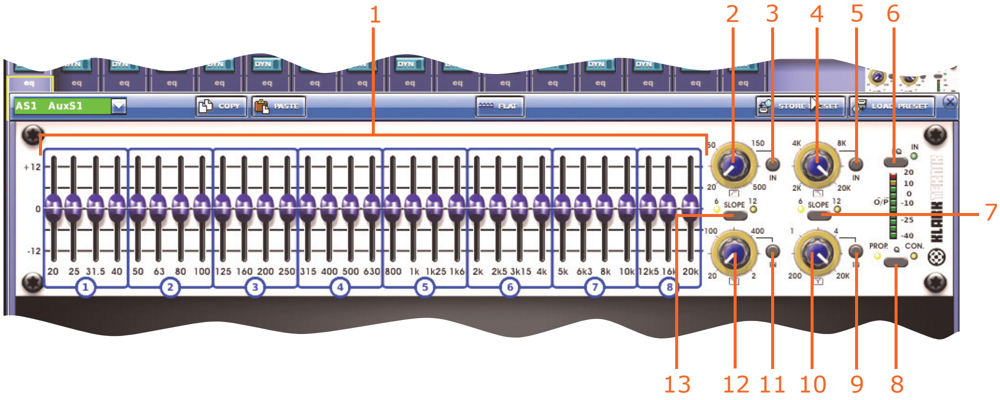
EQ (GEQ)

This section shows the parameters of the EQ processing area, including the GEQ, affected by copy and paste.



Item	Control	Parameter
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch	Band 1 shelving mode: bell, warm, high pass 6dB or high pass 12dB
6	**SHAPE** switch	Band 2 shelving mode: bell or high pass 24dB
7	**SHAPE** switch (not shown in diagram of GUI above)	Band 6 shelving mode: bell, soft, low pass 6dB or low pass 12dB

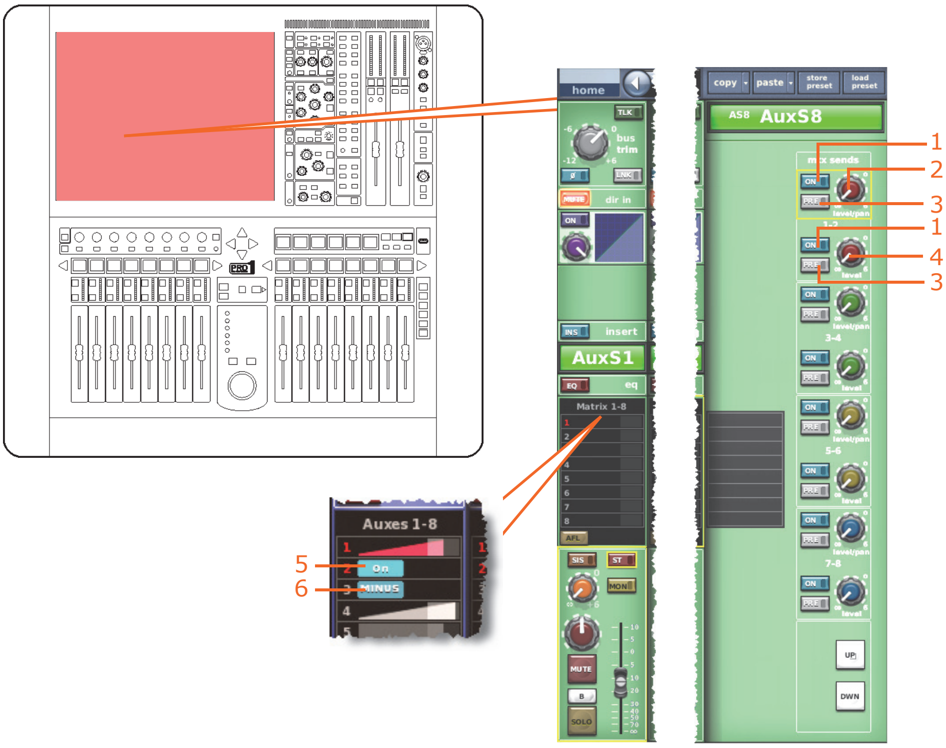
Note: *Although bands 5 and 6 are not shown above, the items in the table also apply. Both bands have items 2, 3 and 4, and band 6 also has item 7.*



Item	Control	Parameter
1	31 faders	Fader positions
2	High pass filter control knob	High pass filter cut off frequency
3	**IN** switch	High pass filter in/out
4	Low pass filter control knob	Low pass filter cut off frequency
5	**IN** switch	Low pass filter in/out
6	**EQ** switch	EQ in/out
7	**SLOPE** switch	Low pass filter: 6dB or 12dB
8	**Q** switch	Q mode as proportional (**PROP.**) or constant (**CON.**)
9	**IN** switch	200Hz - 20kHz notch filter in/out
10	Notch filter control knob	200Hz - 20kHz notch filter frequency
11	**IN** switch	20Hz - 2kHz notch filter in/out
12	Notch filter control knob	20Hz - 2kHz notch filter frequency
13	**SLOPE** switch	High pass filter: 6dB or 12dB

Bus sends

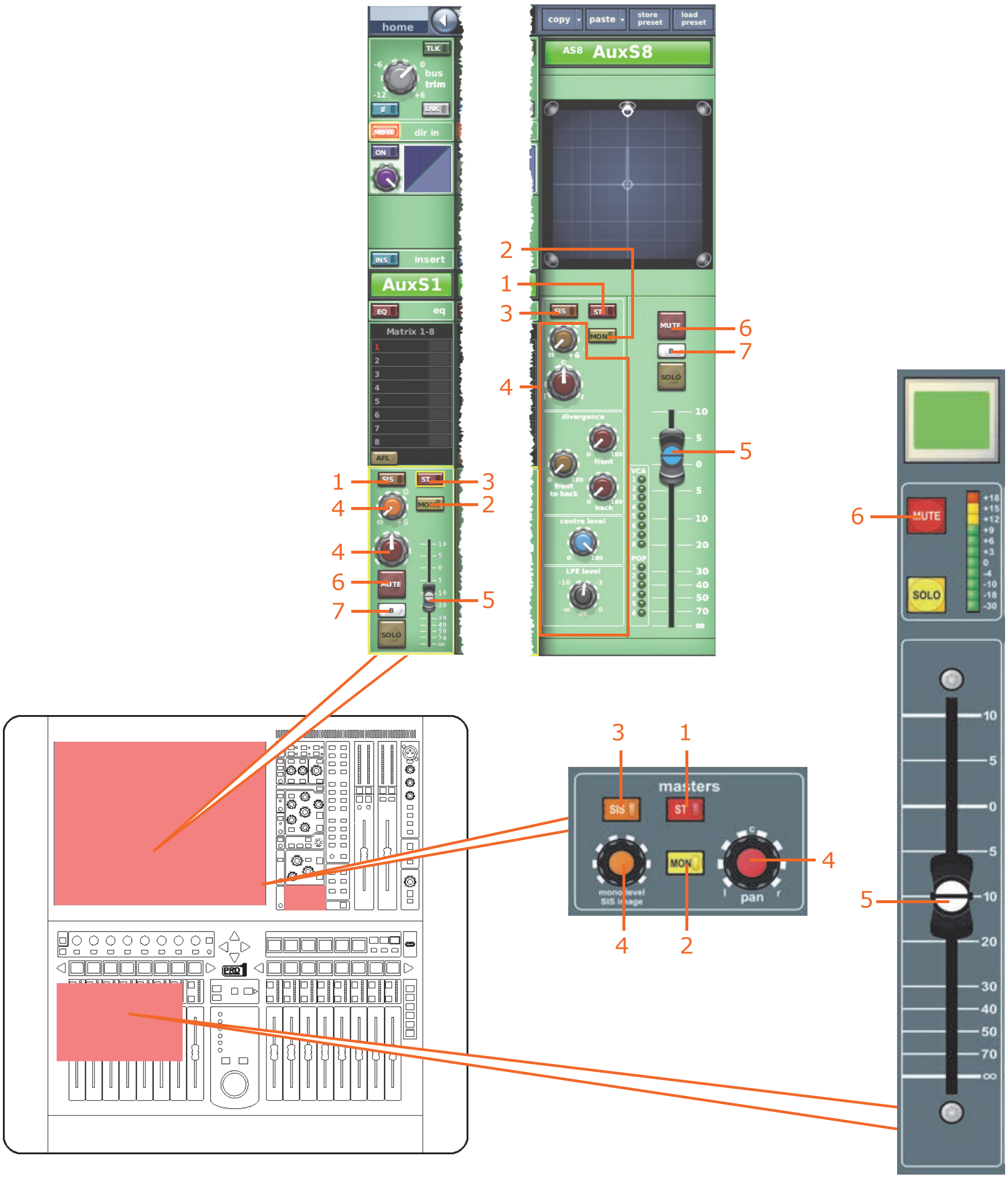
This section shows the parameters of the mix sends processing area affected by copy and paste.



Item	Control	Parameter
1	**ON** switch	Matrix bus send on/off
2	**level/pan** control knob	Bus level, or pan when bus is linked
3	**PRE** switch	Pre-fader on/off
4	**level** control knob	Bus level
5	**On** switch	Aux bus send on/off — only available when aux bus is in group mode
6	**MINUS** switch	Aux bus send mute on/off — only available when aux bus is in mix minus mode

Master routing

This section shows the parameters of the master routing processing area affected by copy and paste.



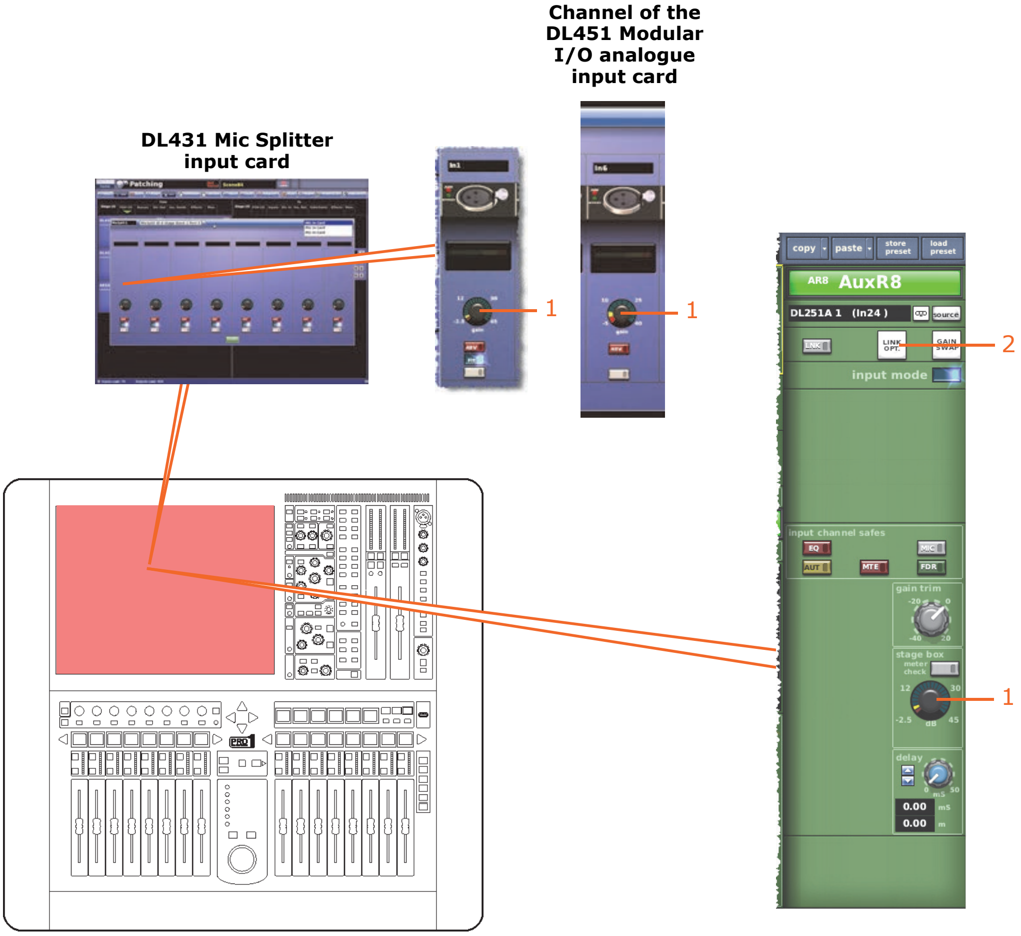
Item	Control	Parameter
1	**ST** switch	Stereo on/off
2	**MON** switch	Mono on/off
3	**SIS** switch	Spatial imaging system on/off
4	Panning control knobs	Surround panning (includes all surround sound parameters)
5	Fader	Level
6	**MUTE** switch	Mute on/off
7	**B** switch	Solo B on/off

Return

This section shows you which return channel parameters are affected by copy and paste.

Configuration

This section shows the parameters of the configuration processing area affected by copy and paste.



Item	Control	Parameter
1	Stage box control knob	Gain of remote amplifier
2	**LINK OPT.** button	Stereo linking options

Compressor

Not applicable.

Gate

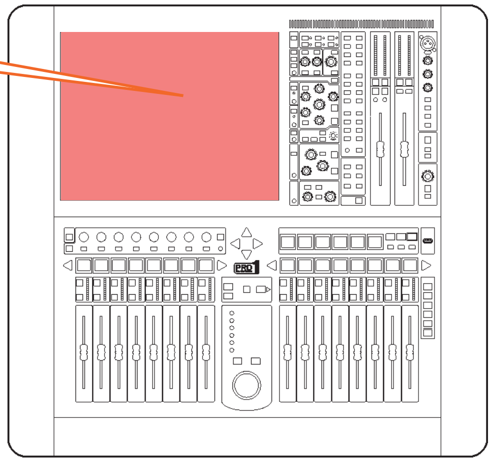
Not applicable.

EQ

Not applicable.

Bus sends

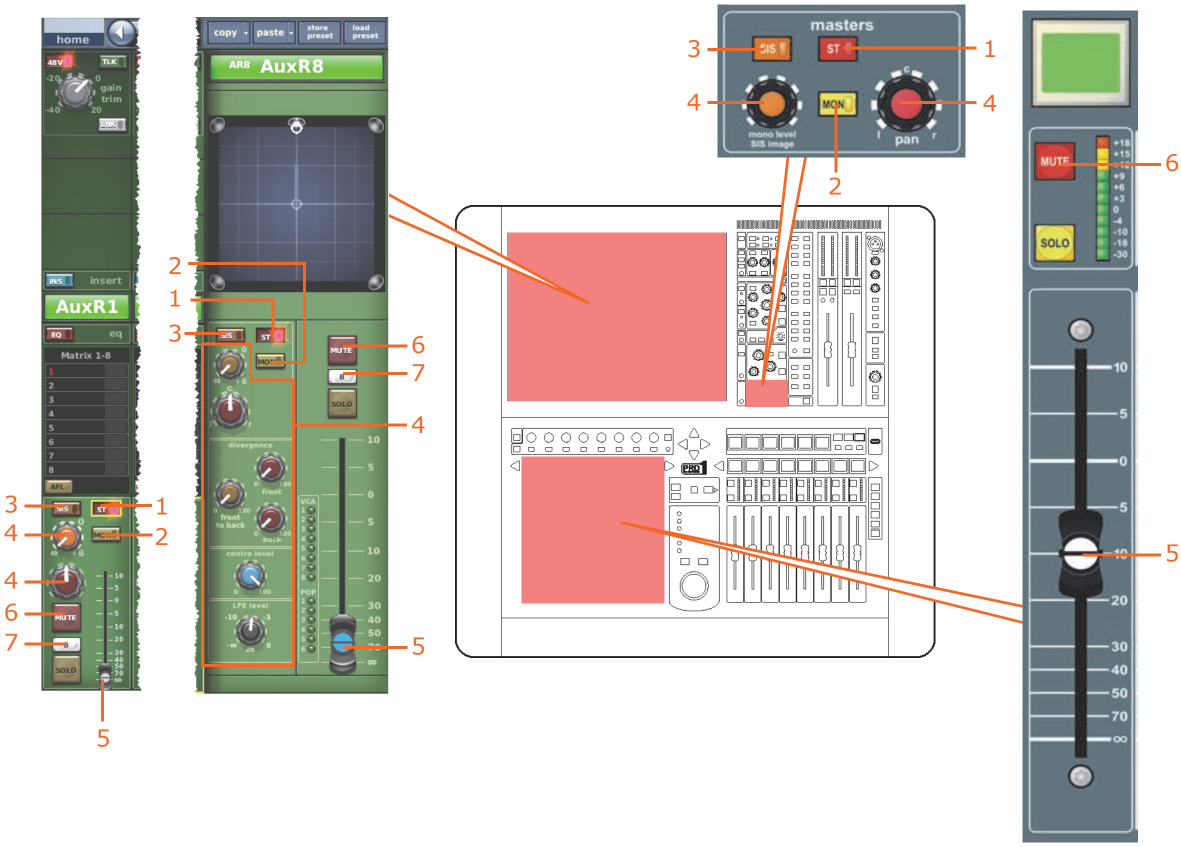
This section shows the parameters of the mix sends processing area affected by copy and paste.



Item	*Control*	*Parameter*
1	**ON** switch	Matrix bus send on/off
2	**level/pan** control knob	Bus level, or pan when bus is linked
3	**PRE** switch	Pre-fader on/off
4	**level** control knob	Bus level

Master routing

This section shows the parameters of the master routing processing area affected by copy and paste.



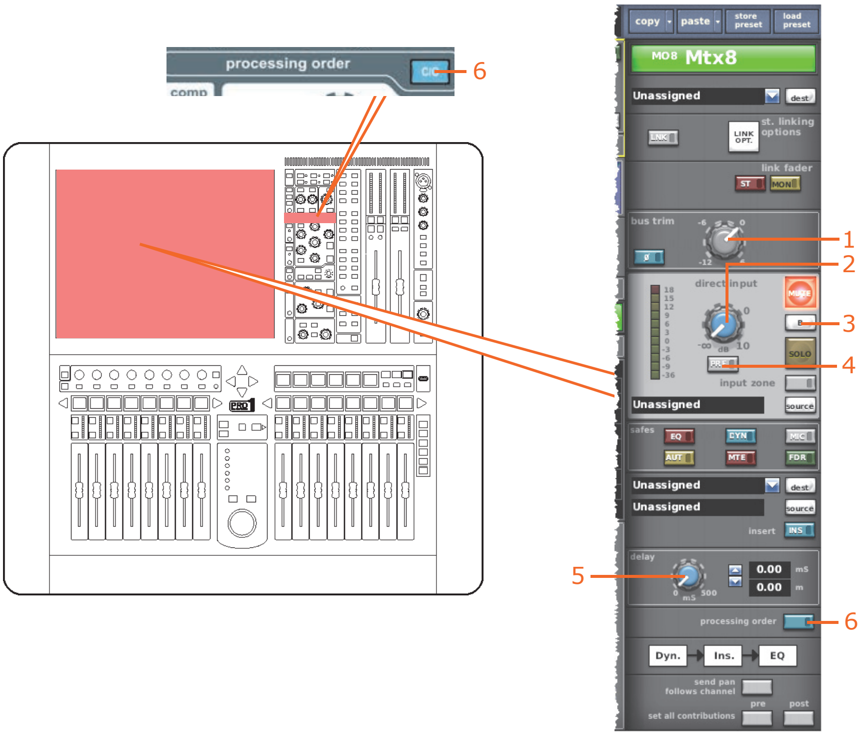
Item	Control	Parameter
1	**ST** switch	Stereo on/off
2	**MON** switch	Mono on/off
3	**SIS** switch	Spatial imaging system on/off
4	Panning control knobs	Surround panning (includes all surround sound parameters)
5	Fader	Level
6	**MUTE** switch	Mute on/off
7	**B** switch	Solo B on/off

Matrix

This section shows you which matrix channel parameters are affected by copy and paste.

Configuration

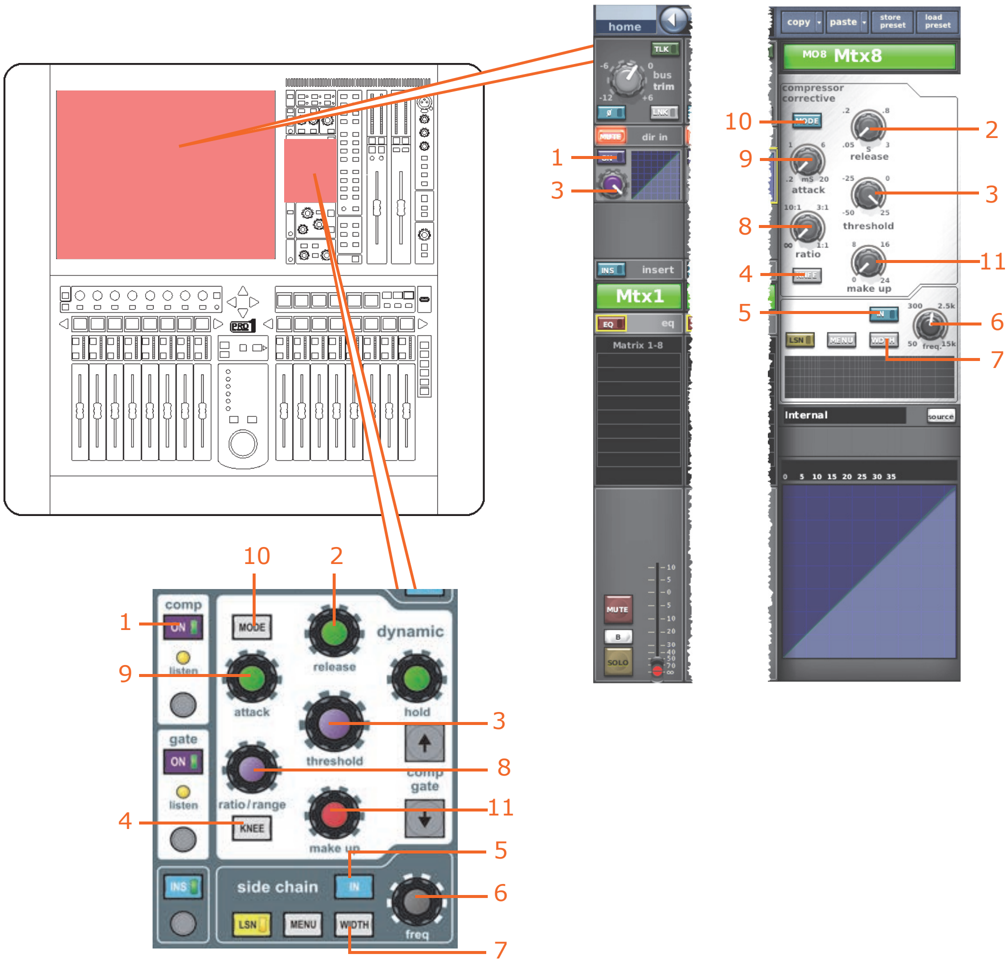
This section shows the parameters of the configuration processing area affected by copy and paste.



Item	*Control*	*Parameter*
1	**bus trim** control knob	Bus trim level
2	Level control knob	Direct input level
3	**B** switch	Direct input solo B on/off
4	**PRE** switch	Direct input pre- in/out
5	**delay** control knob	Delay time
6	**C/O** switch	Order of processing: **Dyn.**→**Ins.**→**EQ** or **EQ**→**Ins.**→**Dyn.**

Compressor

This section shows the parameters of the compressor processing area affected by copy and paste. Only corrective compressor shown below, but this is typically the same for the other compressor modes.



Item	Control	Parameter
1	**ON** switch	Compressor on/off
2	**release** control knob	Compressor release
3	**threshold** control knob	Compressor threshold
4	**KNEE** pushbutton	Compressor knee: hard, medium or soft
5	**IN** switch	Compressor sidechain in/out
6	**freq** control knob	Compressor sidechain frequency
7	**WIDTH** pushbutton	Compressor sidechain width: 2 Oct, 1 Oct or 0.3 Oct
8	**ratio** control knob	Compressor ratio
9	**attack** control knob	Compressor attack

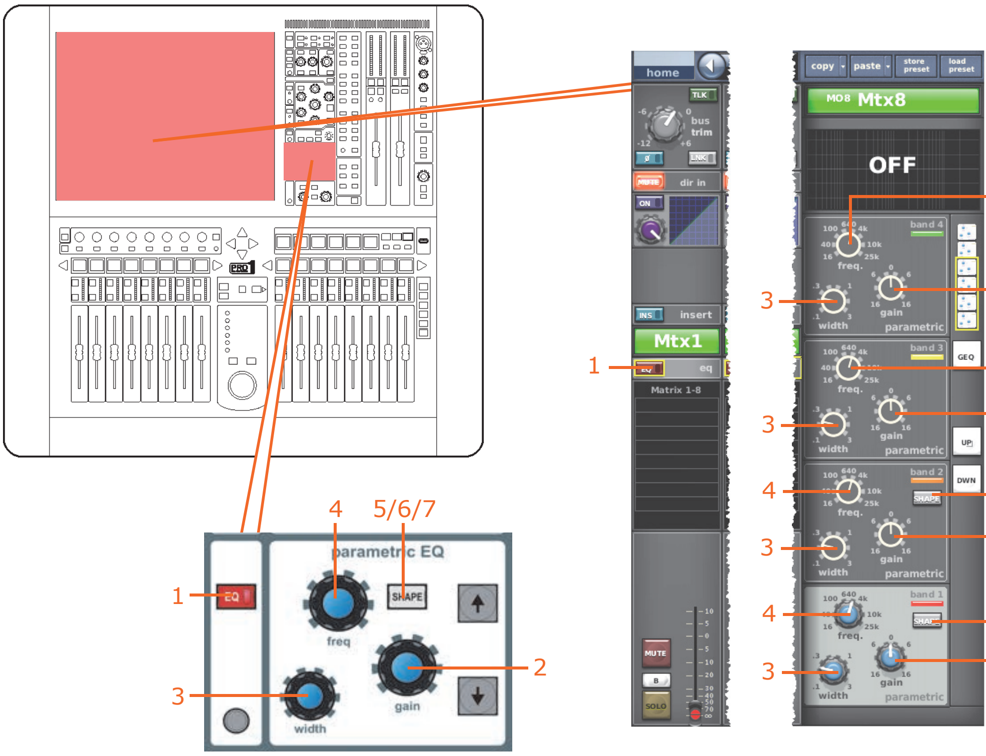
Item	*Control*	*Parameter*
10	**MODE** pushbutton	Compressor mode: corrective, adaptive, creative, vintage or shimmer
11	**make up** control knob	Compressor gain

Gate

Not applicable.

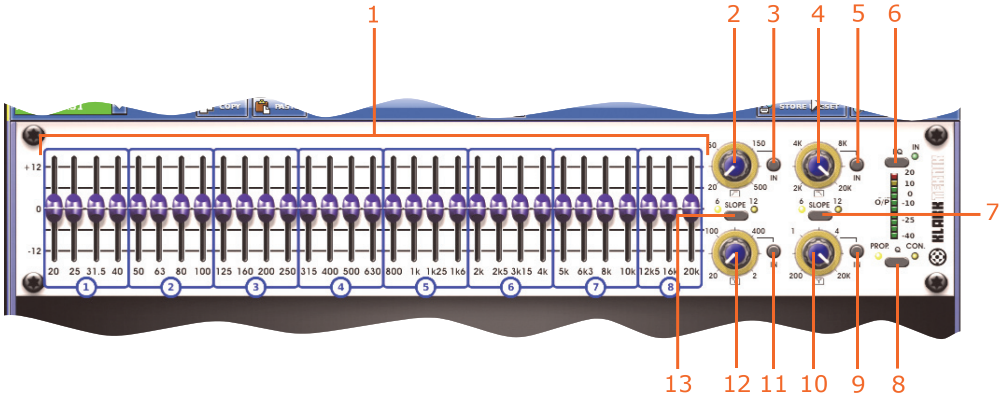
EQ (GEQ)

This section shows the parameters of the EQ processing area (including the GEQ) affected by copy and paste.



Item	*Control*	*Parameter*
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch (not shown in GUI diagram above)	Band 6 shelving mode: bell, soft, low pass 6dB or low pass 12dB
6	**SHAPE** switch	Band 2 shelving mode: bell or high pass 24dB
7	**SHAPE** switch	Band 1 shelving mode: bell, warm, high pass 6dB or high pass 12dB

***Note:** Although bands 5 and 6 are not shown above, the items in the table also apply. Both bands have items 2, 3 and 4, and band 6 also has item 5.*



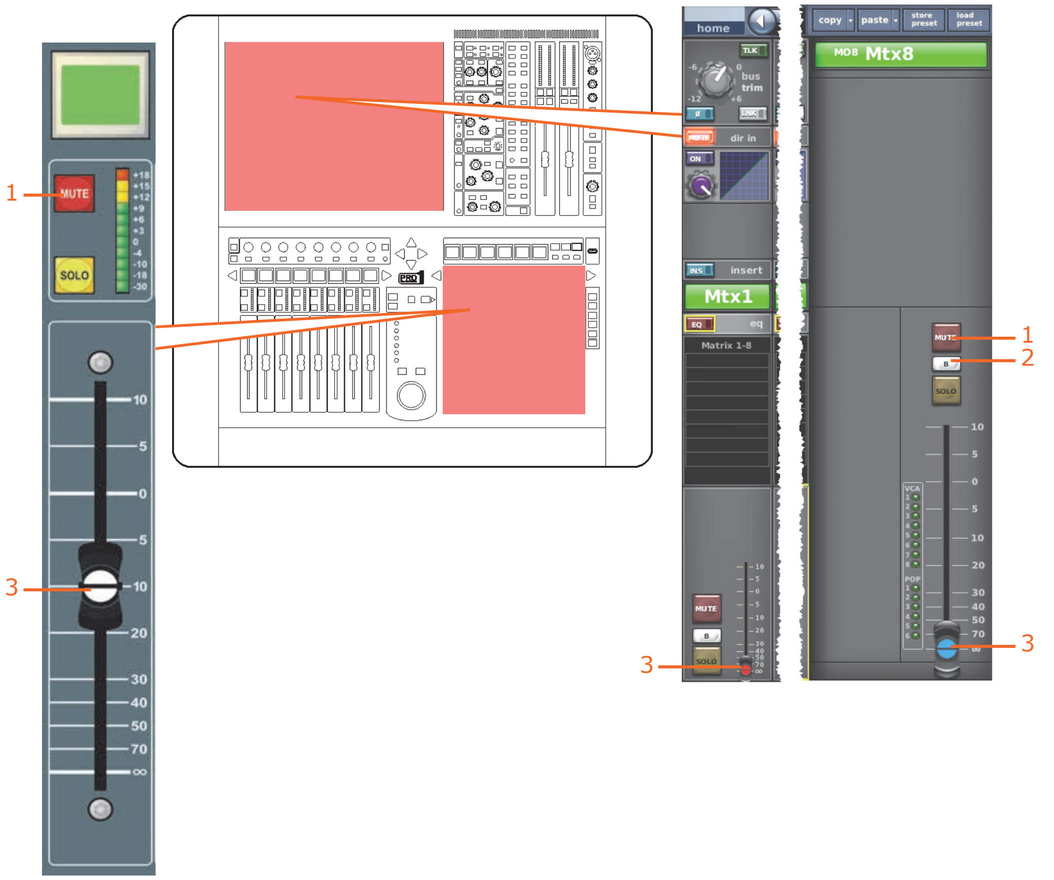
Item	Control	Parameter
1	31 faders	Fader positions
2	High pass filter control knob	High pass filter cut off frequency
3	**IN** switch	High pass filter in/out
4	Low pass filter control knob	Low pass filter cut off frequency
5	**IN** switch	Low pass filter in/out
6	**EQ** switch	EQ in/out
7	**SLOPE** switch	Low pass filter: 6dB or 12dB
8	**Q** switch	Q mode as proportional (**PROP.**) or constant (**CON.**)
9	**IN** switch	200Hz - 20kHz notch filter in/out
10	Notch filter control knob	200Hz - 20kHz notch filter frequency
11	**IN** switch	20Hz - 2kHz notch filter in/out
12	Notch filter control knob	20Hz - 2kHz notch filter frequency
13	**SLOPE** switch	High pass filter: 6dB or 12dB

Bus sends

Not applicable.

Fader section

This section shows the parameters of the master routing processing area affected by copy and paste.



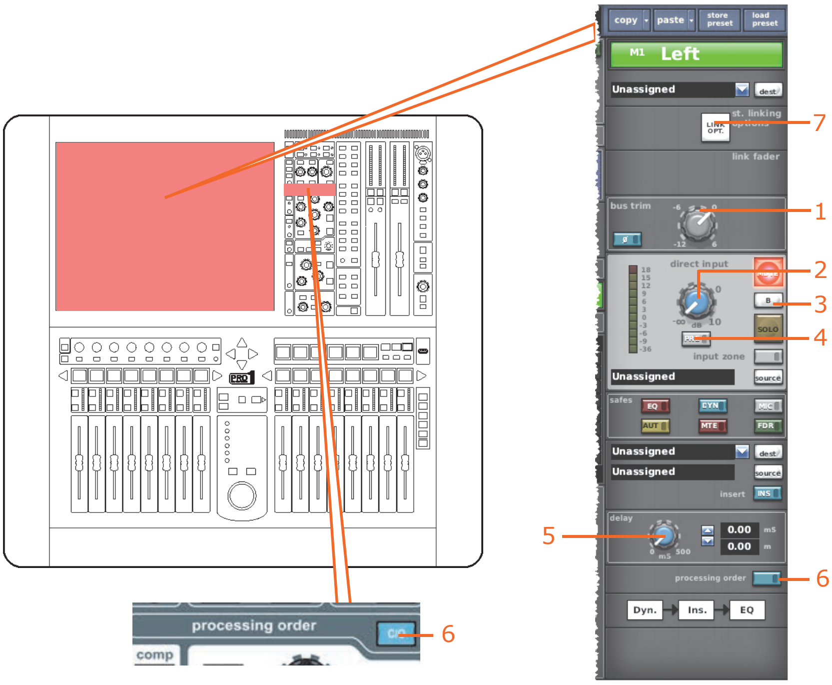
Item	Control	Parameter
1	**MUTE** switch	Mute on/off
2	**B** switch	Solo B on/off
3	Fader	Level

Master

This section shows you which master channel parameters are affected by copy and paste.

Configuration

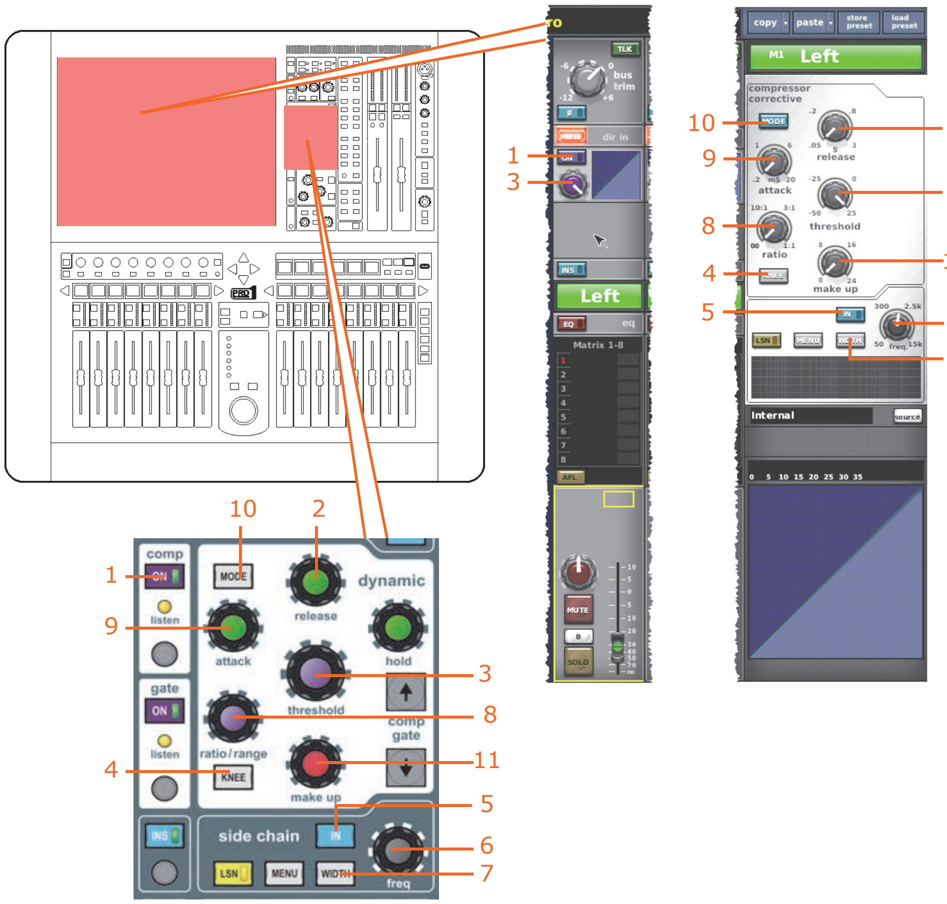
This section shows the parameters of the configuration processing area affected by copy and paste.



Item	Control	Parameter
1	**bus trim** control knob	Bus trim level
2	Level control knob	Direct input level
3	**B** switch	Direct input solo B on/off
4	**PRE** switch	Direct input pre- in/out
5	**delay** control knob	Delay level
6	**C/O** switch	Order of processing: **Dyn.→Ins.→EQ** or **EQ→Ins.→Dyn.**
7	**LINK OPT.** button	Stereo linking options

Compressor

This section shows the parameters of the compressor processing area affected by copy and paste. Only corrective compressor shown below, but typically the same for the other compressor modes.



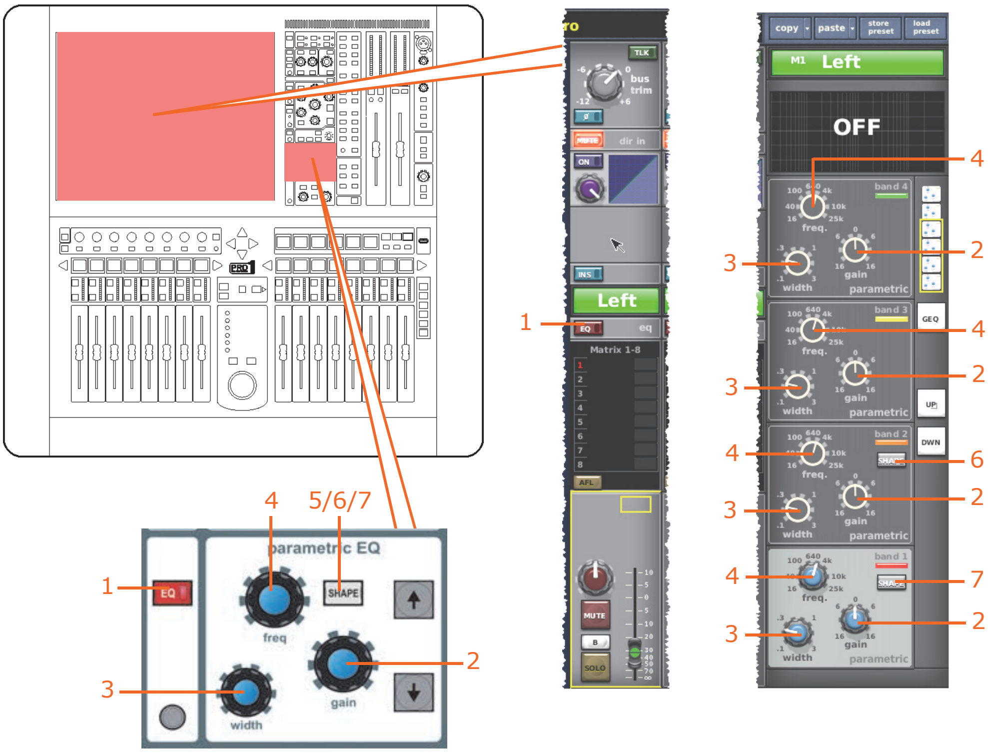
Item	*Control*	*Parameter*
1	**ON** switch	Compressor on/off
2	**release** control knob	Compressor release
3	**threshold** control knob	Compressor threshold
4	**KNEE** pushbutton	Compressor knee: hard, medium or soft
5	**IN** switch	Compressor sidechain in/out
6	**freq** control knob	Compressor sidechain frequency
7	**WIDTH** pushbutton	Compressor sidechain width: 2 Oct, 1 Oct or 0.3 Oct
8	**ratio** control knob	Compressor ratio
9	**attack** control knob	Compressor attack
10	**MODE** pushbutton	Compressor mode: corrective, adaptive, creative, vintage or shimmer
11	**make up** control knob	Compressor gain

Gate

Not applicable.

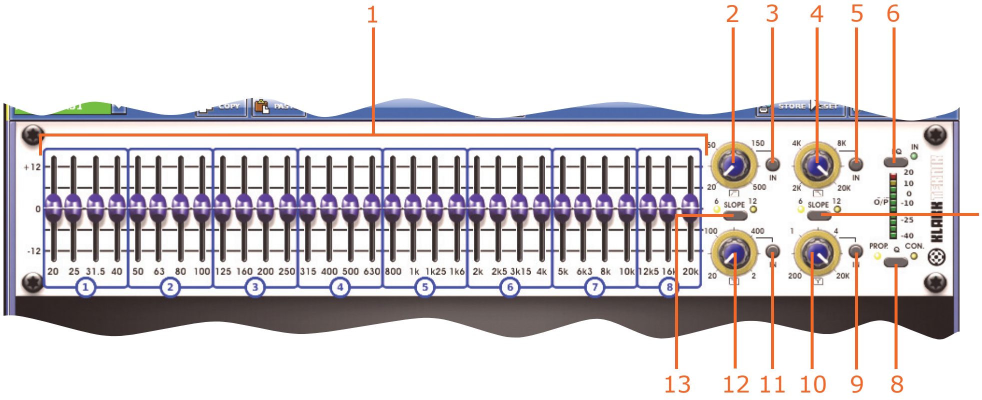
EQ (GEQ)

This section shows the parameters of the EQ detail are (including the GEQ) affected by copy and paste.



Item	Control	Parameter
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch (not shown on the GUI diagram above)	Band 6 shelving mode: bell, soft, low pass 6dB or low pass 12dB
6	**SHAPE** switch	Band 2 shelving mode: bell or high pass 24dB
7	**SHAPE** switch	Band 1 shelving mode: bell, warm, high pass 6dB or high pass 12dB

***Note:** Although bands 5 and 6 are not shown above, the items in the table also apply. Both bands have items 2, 3 and 4, and band 6 also has item 5.*



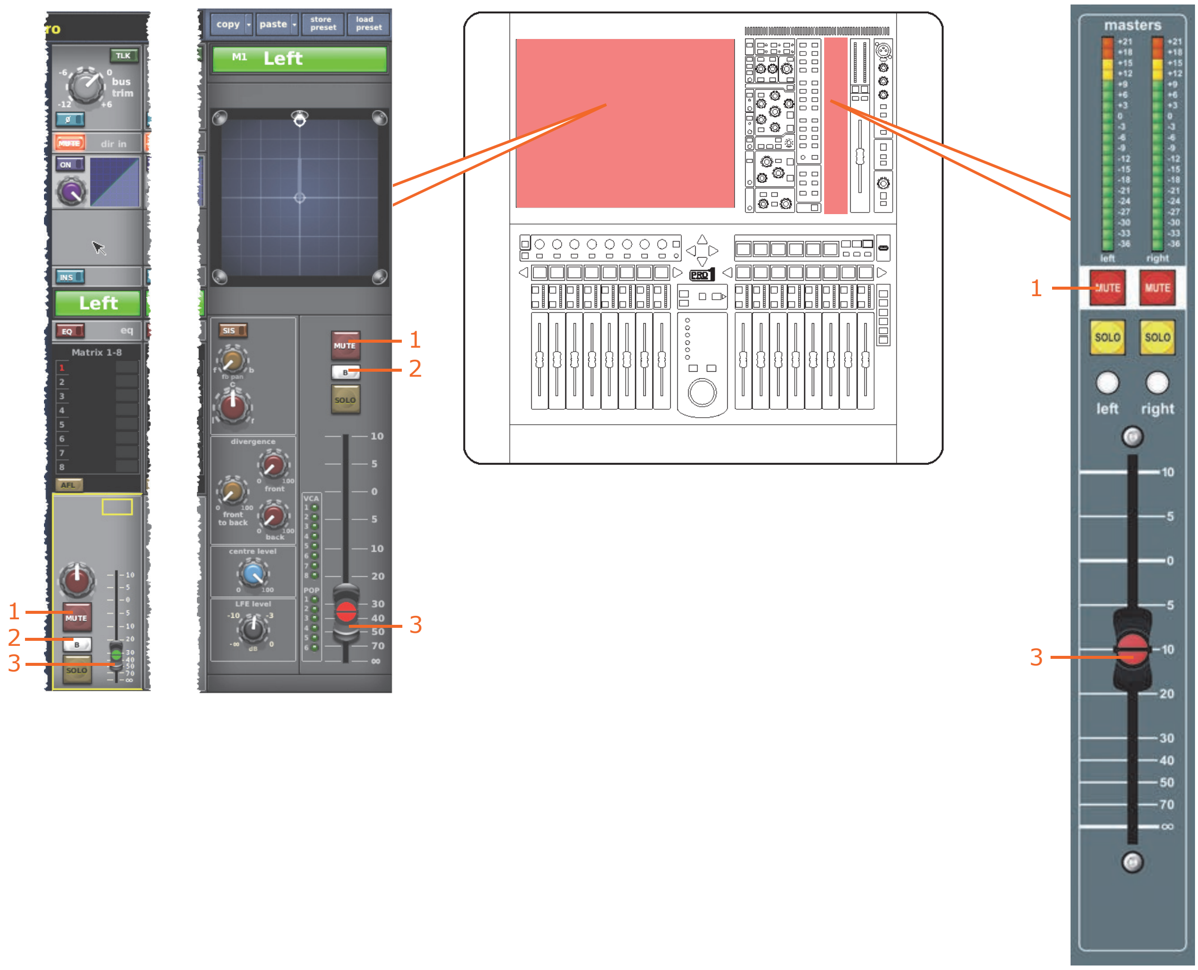
Item	Control	Parameter
1	31 faders	Fader positions
2	High pass filter control knob	High pass filter cut off frequency
3	**IN** switch	High pass filter in/out
4	Low pass filter control knob	Low pass filter cut off frequency
5	**IN** switch	Low pass filter in/out
6	**EQ** switch	EQ in/out
7	**SLOPE** switch	Low pass filter: 6dB or 12dB
8	**Q** switch	Q mode as proportional (**PROP.**) or constant (**CON.**)
9	**IN** switch	200Hz - 20kHz notch filter in/out
10	Notch filter control knob	200Hz - 20kHz notch filter frequency
11	**IN** switch	20Hz - 2kHz notch filter in/out
12	Notch filter control knob	20Hz - 2kHz notch filter frequency
13	**SLOPE** switch	High pass filter: 6dB or 12dB

Bus sends

Not applicable.

Master routing

This section shows the parameters of the master routing processing area affected by copy and paste.



Item	*Control*	*Parameter*
1	**MUTE** switch	Mute on/off
2	**B** switch	Solo B on/off
3	Fader	Level

Appendix J: Parameters Affected By Stereo Linking

This appendix shows which parameters per control area are shared by a pair of stereo linked channels. The control areas can be switched on/off globally for each channel type or per channel pair (see “Changing the linking options” on page 93).

The following table is intended as a reference guide to help you go quickly to the area you want.

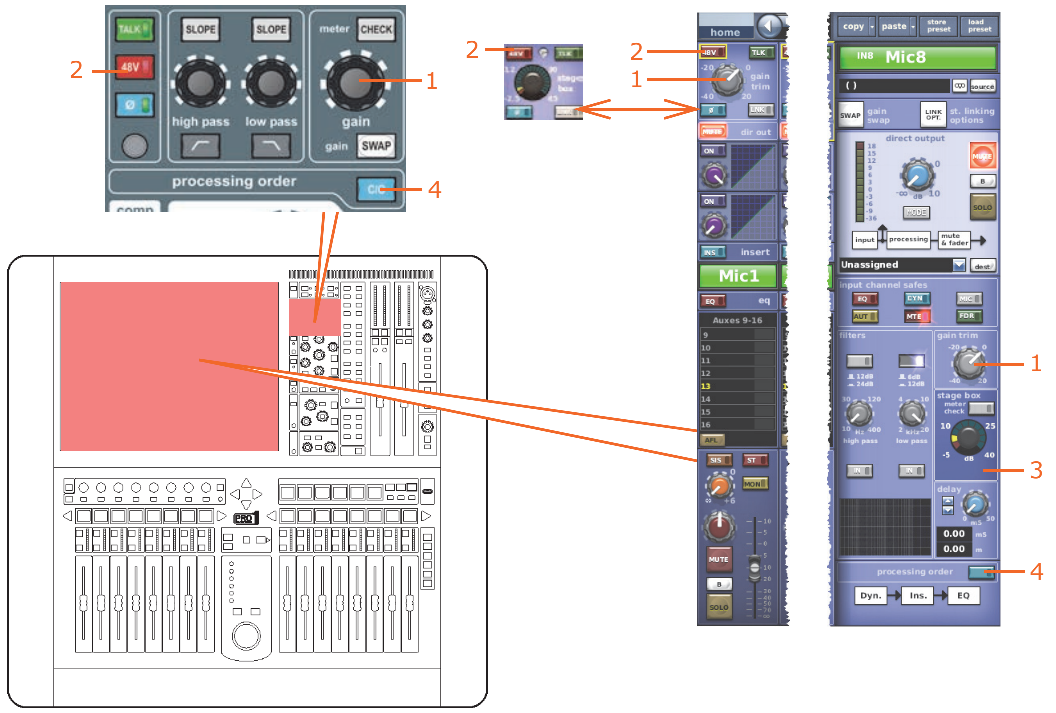
Control area	***Input Channels***	***Aux Returns (Returns)***	***Aux Sends (Auxes)***	***Matrix (Matrices)***	***Masters***
Input Controls	Page 468	Page 480	Page 488	Page 498	Page 506
Direct Output	Page 469	N/A	N/A	N/A	N/A
Direct Input	N/A	N/A	Page 489	Page 499	Page 507
Filters	Page 470	N/A	N/A	N/A	N/A
Dynamics	Page 471	N/A	Page 490	Page 500	Page 508
Insert	Page 473	Page 481	Page 491	Page 501	Page 509
EQ	Page 474	Page 482	Page 492	Page 502	Page 510
Bus Sends	Page 475	Page 483	Page 493	N/A	Page 511
Master Routing	Page 476	Page 484	Page 494	N/A	Page 512
Fader	Page 477	Page 485	Page 495	Page 503	Page 513
Delay	Page 478	Page 486	Page 496	Page 504	Page 514
Mute	Page 479	Page 487	Page 497	Page 505	Page 515

Input Channels

This section shows the linked parameters of the input channels.

Input controls

This section shows the input control parameters that are linked across a channel pair.

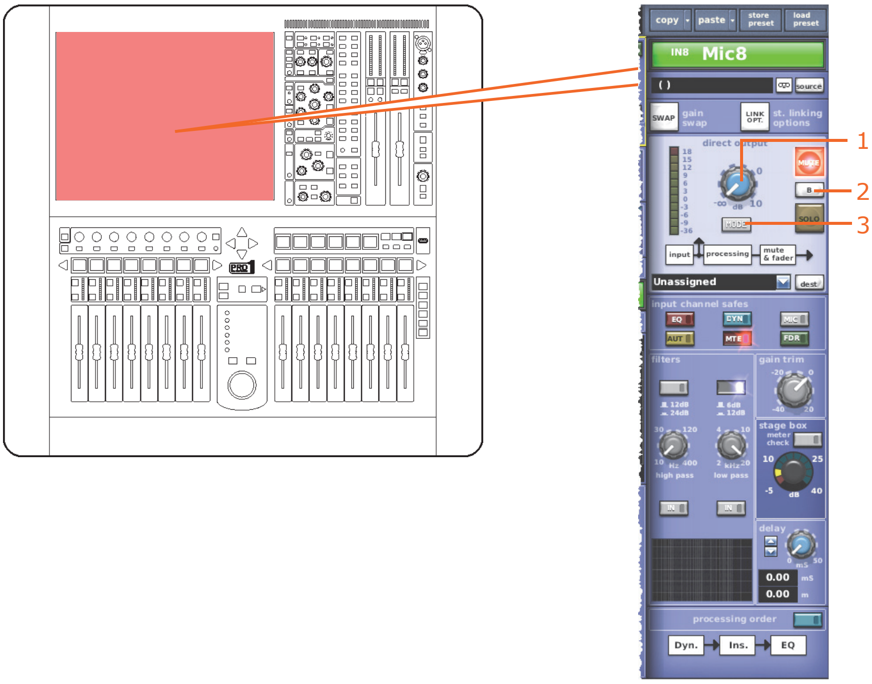


Item	Control	Parameter
1	**gain**/[**gain trim**] control knob	Digital trim level
2	**48V** switch*	48V phantom voltage on/off
3	Filter switch (not shown on GUI diagram above)	30Hz filter in/out
4	**C/O** switch*	Order of processing: **Dyn.→Ins.→EQ** or **EQ→Ins.→Dyn.**

* Applies to tape and primary inputs.

Direct output

This section shows the direct output control parameters that are linked across a channel pair.



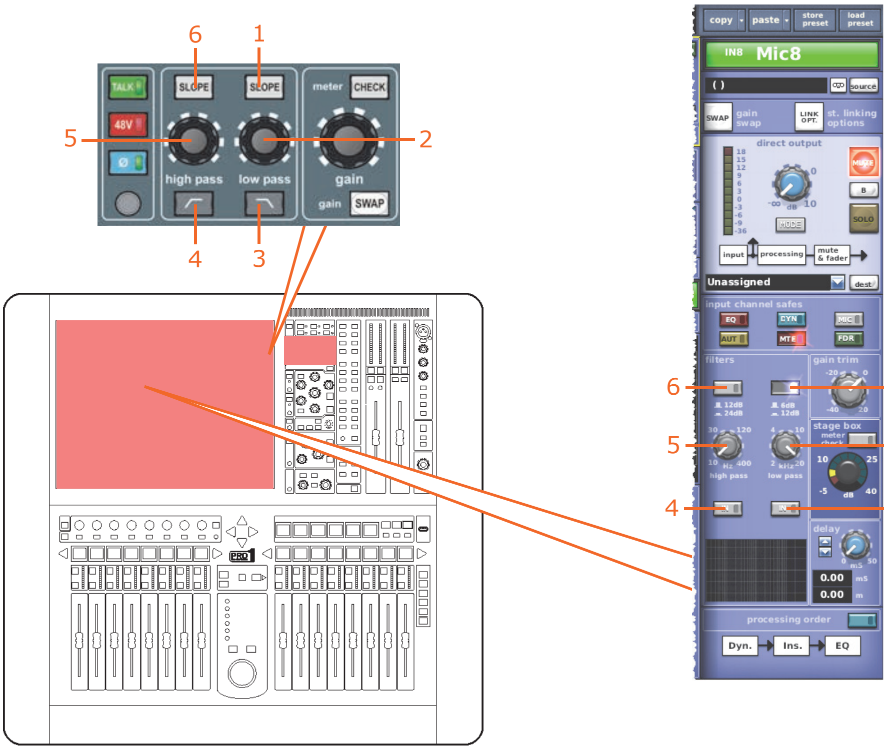
Item	*Control*	*Parameter*
1	Level control knob	Direct output level
2	**B** control knob	Direct output solo B in/out
3	**MODE** pushbutton	Direct output tap-off point: "Post-fade and mute", "Pre-mute, pre-processing" or "Pre-mute, post-processing"

Direct input

Not applicable.

Filters

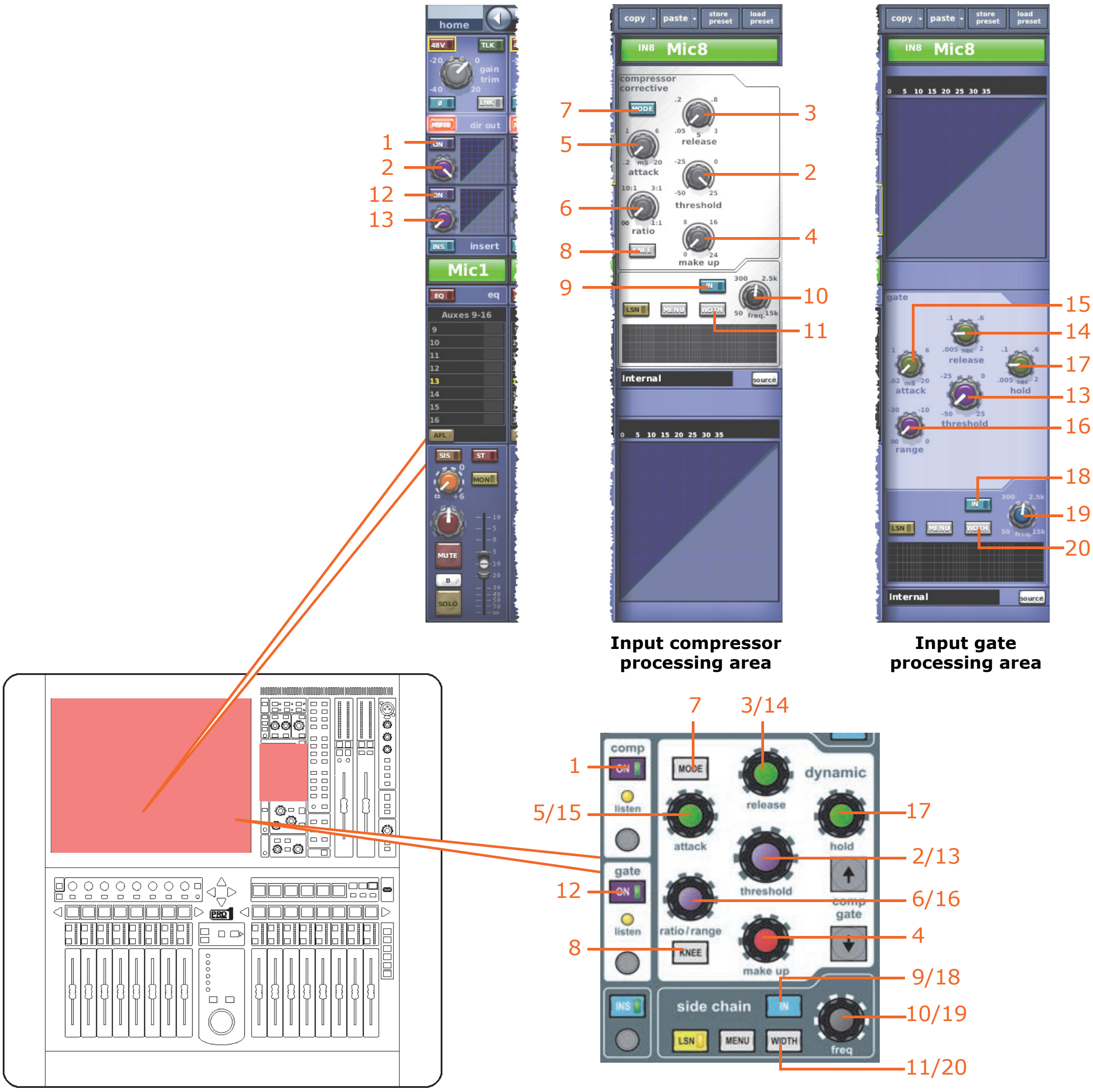
This section shows the filter control parameters that are linked across a channel pair.



Item	*Control*	*Parameter*
1	**SLOPE** pushbutton	Low pass filter slope 6dB or 12dB
2	**low pass** control knob	Low pass filter frequency
3	/[**IN**] switch	Low pass filter in/out
4	/[**IN**] switch	High pass filter in/out
5	**high pass** control knob	High pass filter frequency
6	**SLOPE** pushbutton	High pass filter slope 12dB or 24dB

Dynamics

This section shows the compressor and gate control parameters that are linked across a channel pair. Although only the corrective compressor is shown below, this is typically the same for the other compressor modes (adaptive, creative and vintage).



Input compressor processing area

Input gate processing area

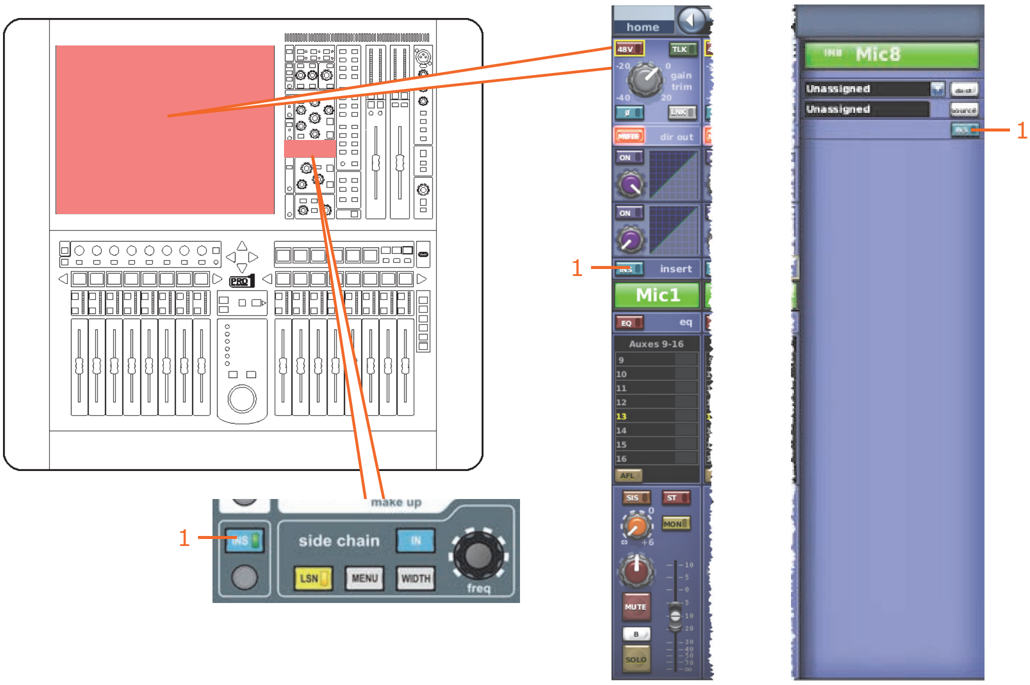
Item	Control	Parameter
1	**ON** switch	Compressor on/off
2	**threshold** control knob	Compressor threshold
3	**release** control knob	Compressor release
4	**make up** control knob	Compressor make up gain
5	**attack** control knob	Compressor attack
6	**ratio** control knob	Compressor ratio
7	**MODE** pushbutton	Compressor mode: corrective, adaptive, creative or vintage
8	**KNEE** pushbutton	Compressor knee: hard, medium or soft

Item	*Control*	*Parameter*
9	**IN** switch	Compressor sidechain in/out
10	**freq** control knob	Compressor sidechain frequency
11	**WIDTH** pushbutton	Compressor sidechain width: 2 Oct, 1 Oct or 0.3 Oct
12	**ON** switch	Gate on/off
13	**threshold** control knob	Gate threshold
14	**release** control knob	Gate release
15	**attack** control knob	Gate attack
16	**range** control knob	Gate range
17	**hold** control knob	Gate hold
18	**IN** switch	Gate sidechain in/out
19	**freq** control knob	Gate sidechain frequency
20	**WIDTH** pushbutton	Gate sidechain width (unlabelled) selector: 2 Oct, 1 Oct and 0.3 Oct

Note: *The compressor and gate sidechains of stereo paired channels are always linked such that they ensure the same amount of gain reduction is applied to both channels.*

Insert

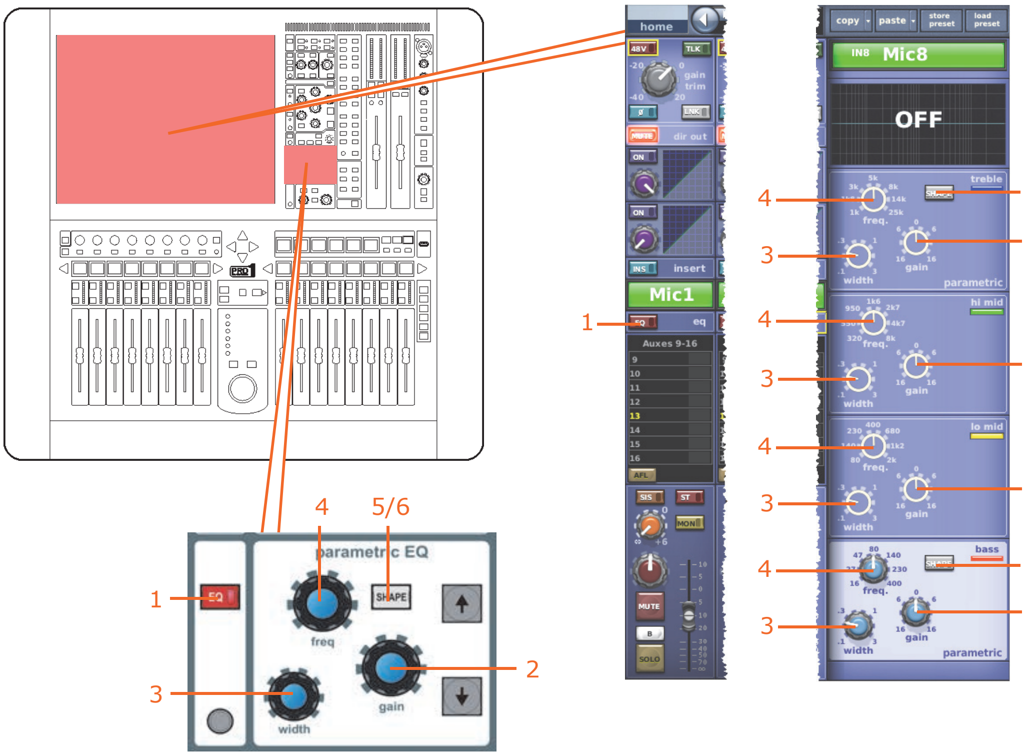
This section shows the insert control parameters that are linked across a channel pair.



Item	Control	Parameter
1	**INS** switch	Insert in/out

EQ

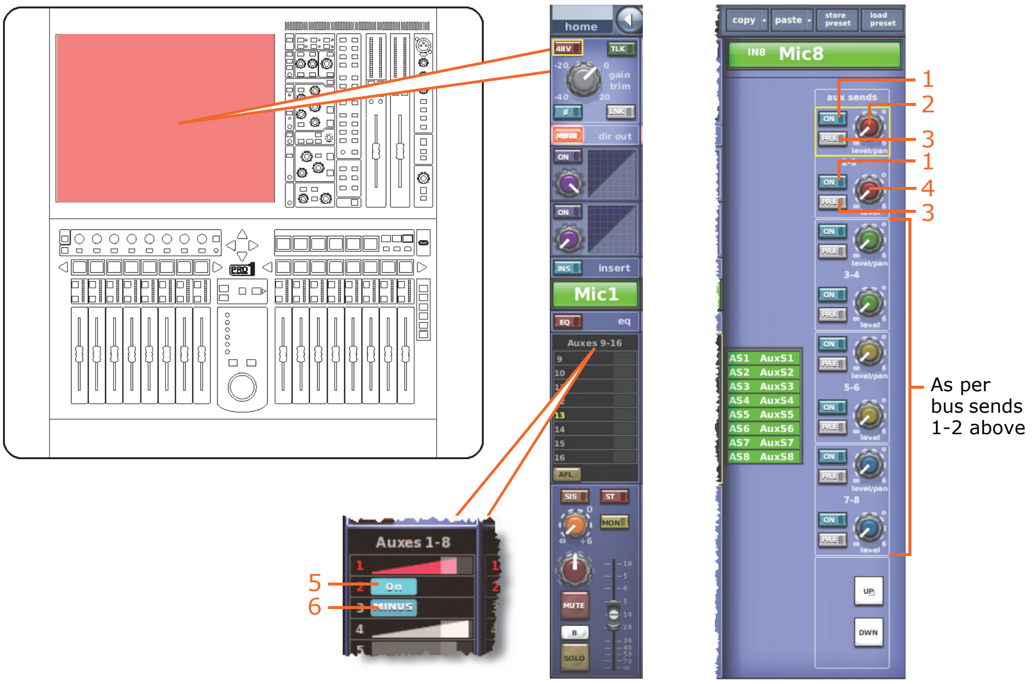
This section shows the EQ control parameters that are linked across a channel pair.



Item	*Control*	*Parameter*
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch	Treble shelving mode: parametric, bright, classic or soft
6	**SHAPE** switch	Bass shelving mode: parametric, deep, classic or warm

Bus sends

This section shows the mix send control parameters that are linked across a channel pair.

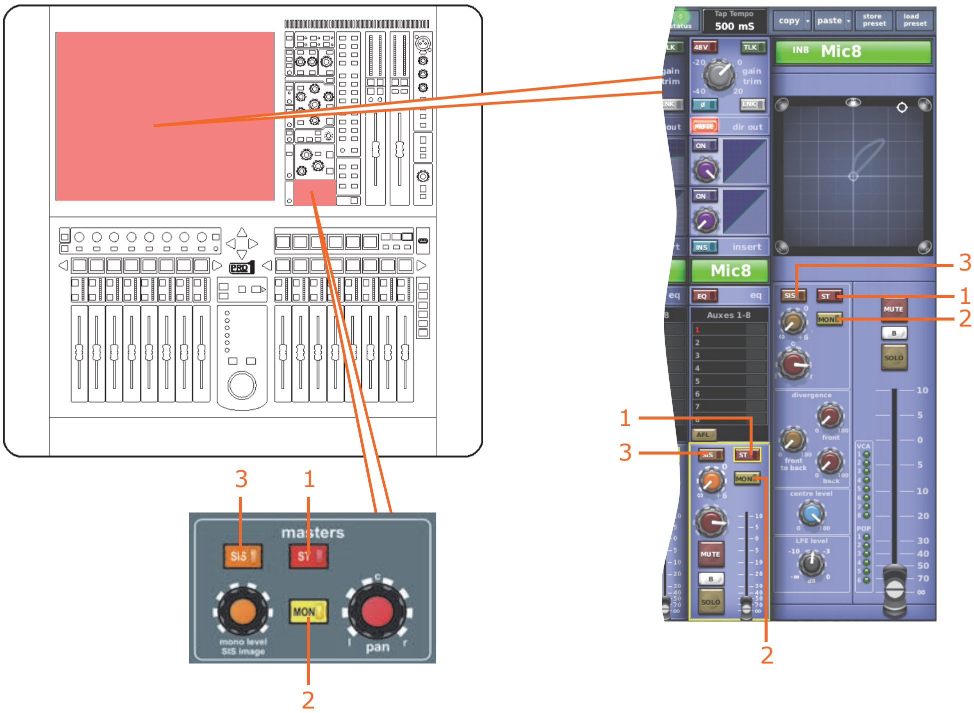


Note: *Only aux sends 1 to 8 are shown above, but this is typically the same for all bus sends (eight matrices and 16 auxes).*

Item	*Control*	*Parameter*
1	**ON** switch	Bus send on/off
2	**level/pan** control knob	Bus level, or pan when bus is linked. (When sending onto a stereo bus the send pan controls are not linked.)
3	**PRE** switch	Pre-fader on/off
4	**level** control knob	Bus level
5	**On** switch	Aux bus send on/off — only available when aux bus is in group mode
6	**MINUS** switch	Aux bus send mute on/off — only available when aux bus is in mix minus mode

Master routing

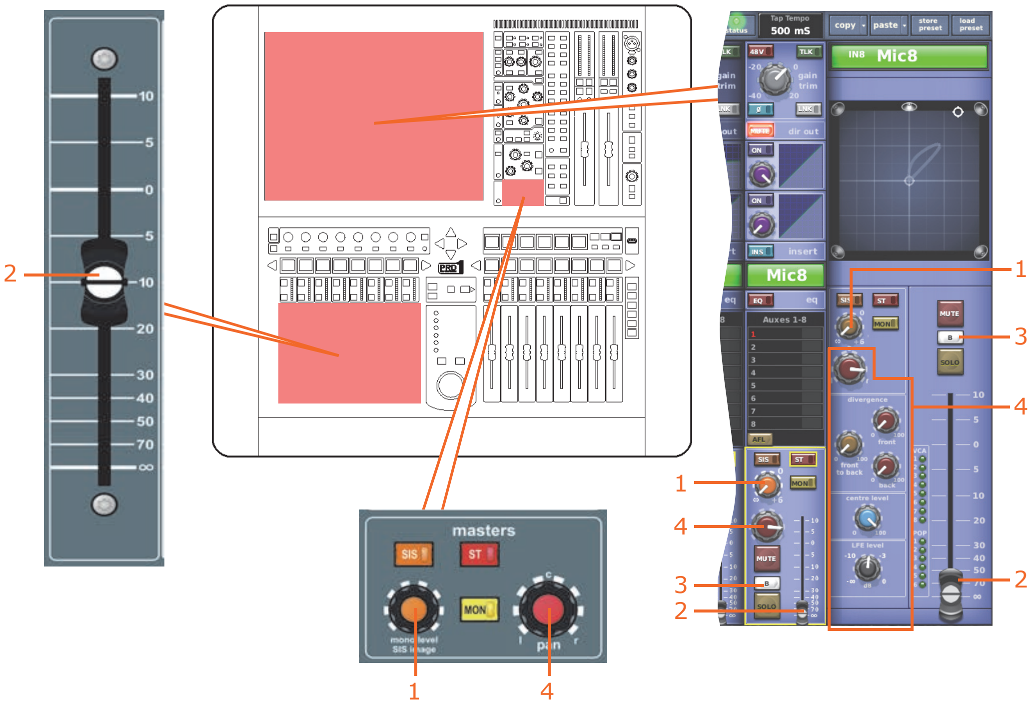
This section shows the master routing control parameters that are linked across a channel pair.



Item	*Control*	*Parameter*
1	**ST** switch	Stereo on/off
2	**MON** switch	Mono on/off
3	**SIS** switch	Spatial imaging system on/off

Fader

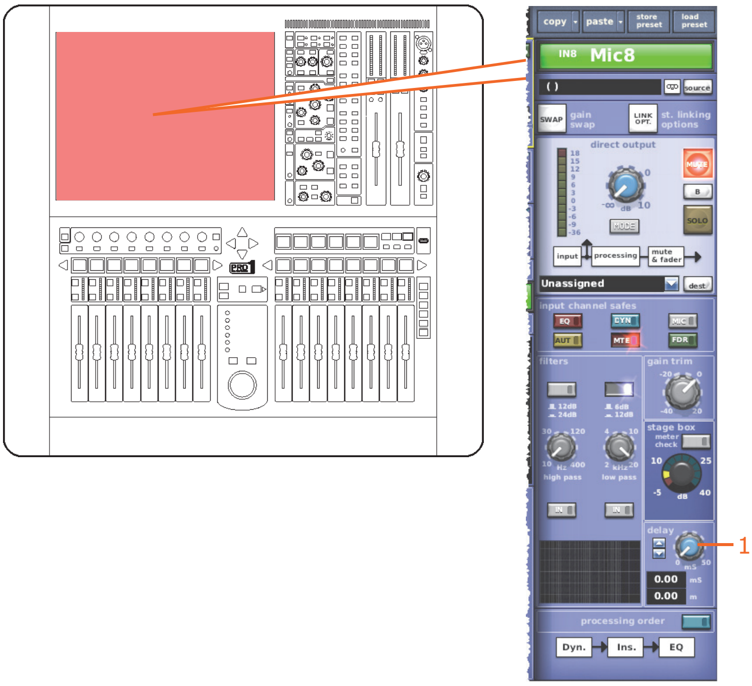
This section shows the fader control parameters that are linked across a channel pair.



Item	Control	Parameter
1	**mono level/SIS image** control knob	Mono send level. (Only linked when SIS is out on both channels and surround mode is not selected.)
2	Fader	Fader level
3	**B** switch	Solo B in/out
4	Panning control knobs	Surround panning, which includes all surround sound parameters, provided **Surround Mode** is not configured as **None** (see “Selecting the surround mode” on page 225)

Delay

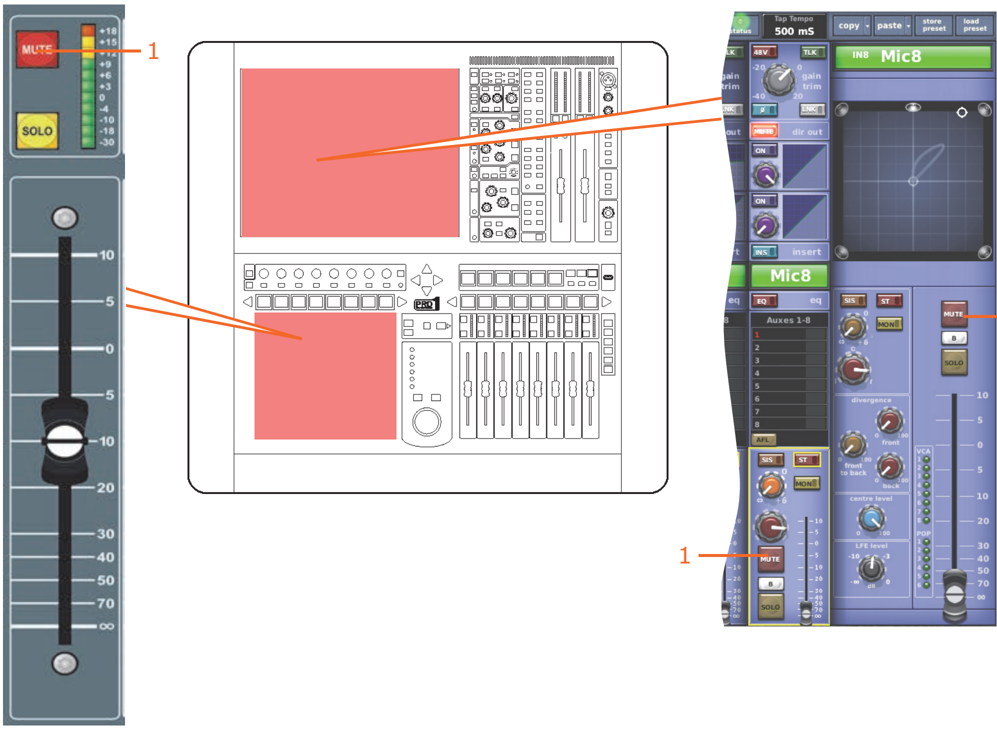
This section shows the delay control parameters that are linked across a channel pair.



Item	*Control*	*Parameter*
1	**delay** control knob	Delay time

Mute

This section shows the mute control parameters that are linked across a channel pair.



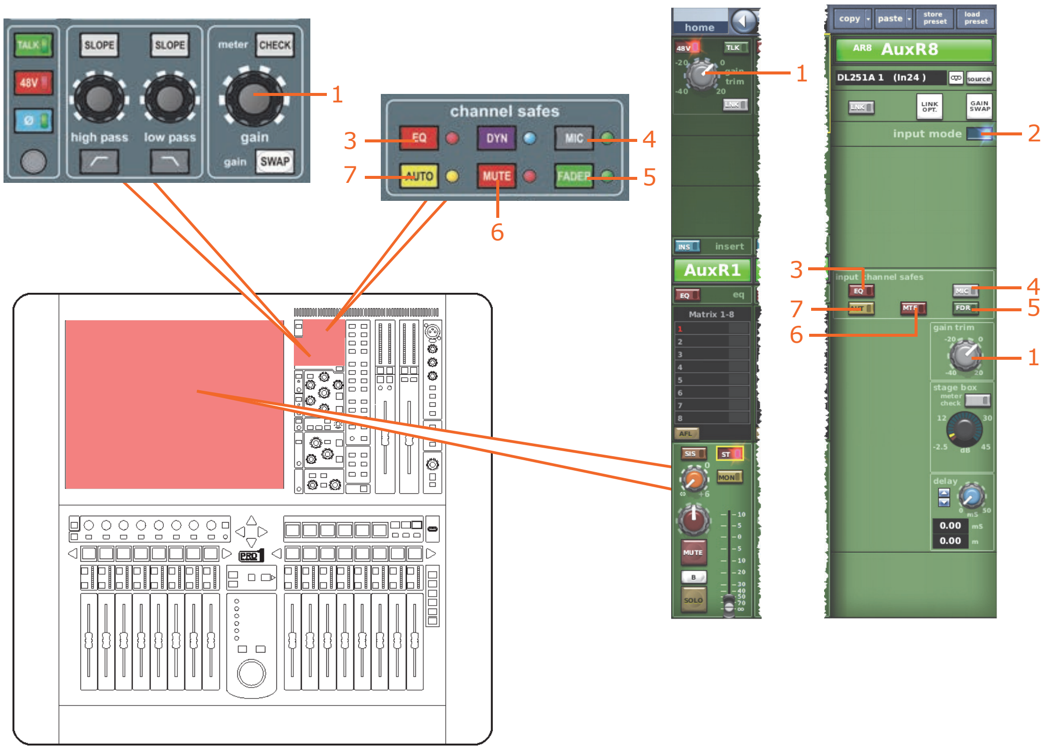
Item	*Control*	*Parameter*
1	**MUTE** button	Mutes the input channel signal

Aux Returns

This section shows the linked parameters of the return channels.

Input controls

This section shows the input control parameters that are linked across a channel pair.



Item	*Control*	*Parameter*
1	**gain**/[**gain trim**] control knob	Digital trim level
2	**input mode** button	For time aligning the return channel with the input channels
3	**EQ** pushbutton	EQ safe on/off
4	**MIC** pushbutton	Mic safe on/off
5	**FADER**/[**FDR**] switch	Fader safe on/off
6	**MUTE**/[**MTE**] switch	Mute safe on/off
7	**AUTO**/[**AUT**] switch	Auto safe on/off

Direct output

Not applicable.

Direct input

Not applicable.

Filters

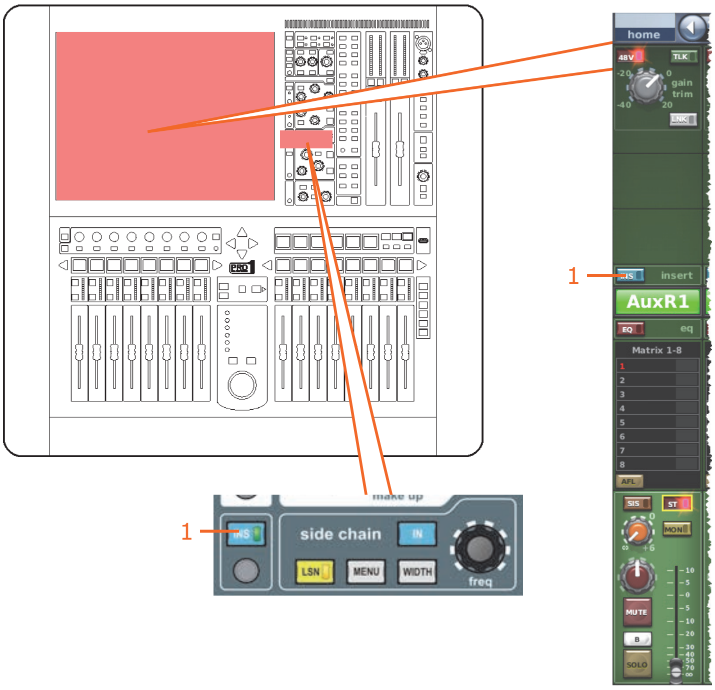
Not applicable.

Dynamics

Not applicable.

Insert

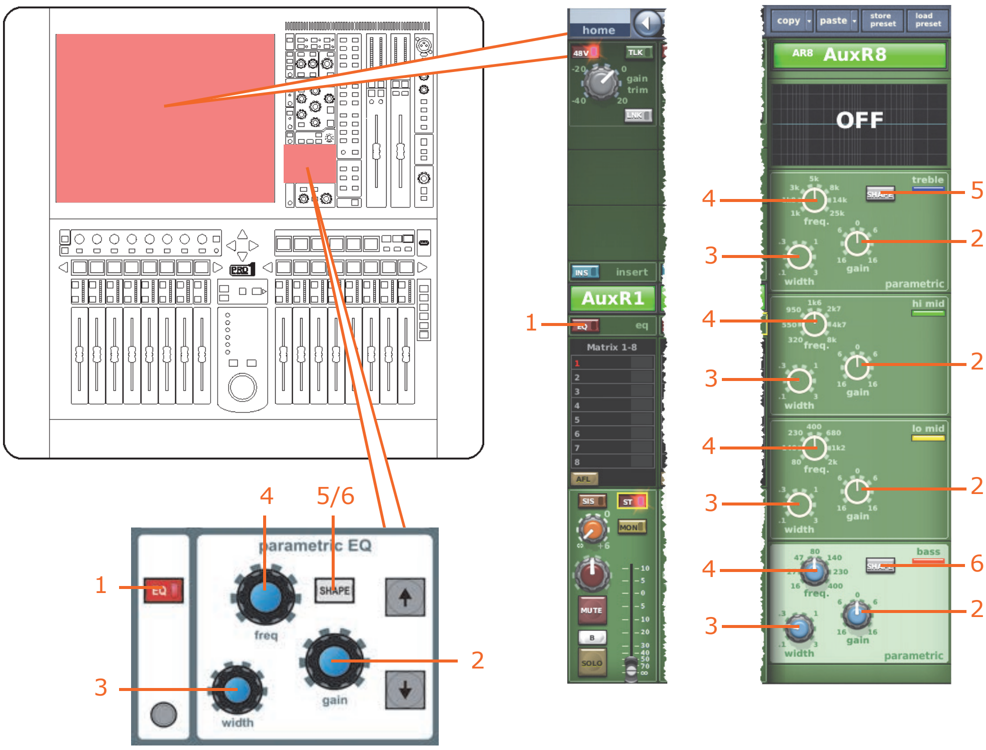
This section shows the insert control parameters that are linked across a channel pair.



Item	Control	Parameter
1	**INS** switch	Insert in/out

EQ

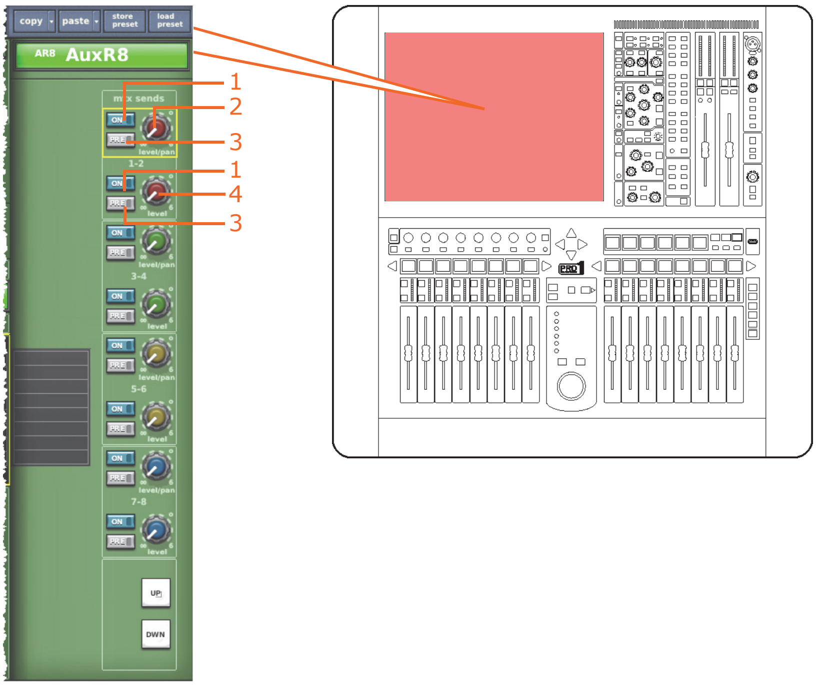
This section shows the parametric EQ control parameters that are linked across a channel pair.



Item	*Control*	*Parameter*
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch	Treble shelving mode: parametric, bright, classic or soft
6	**SHAPE** switch	Bass shelving mode: parametric, deep, classic or warm

Bus sends

This section shows the mix send control parameters that are linked across a channel pair.

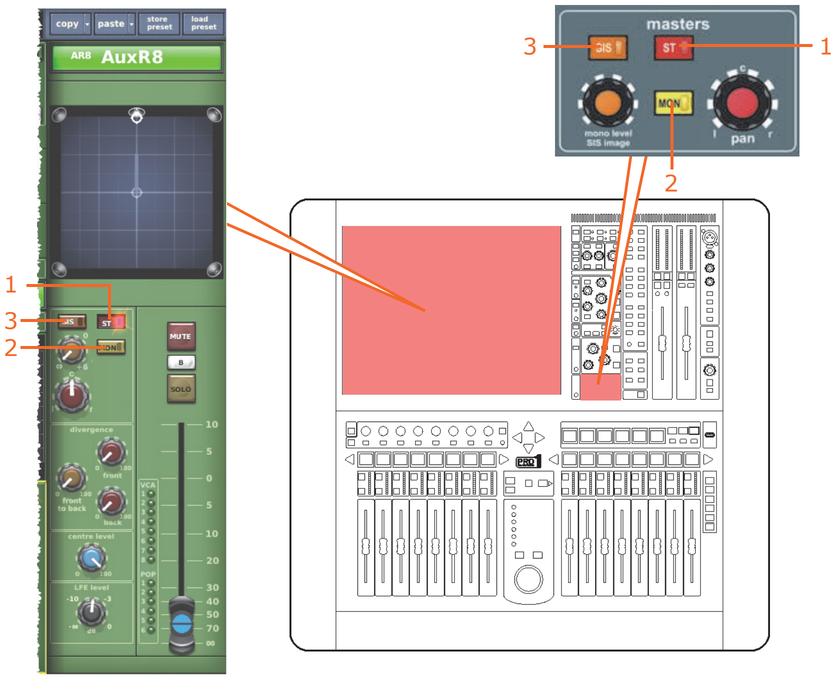


Note: *Although only matrix sends 1-2 are referenced above, this also applies to all eight matrix sends.*

Item	***Control***	***Parameter***
1	**ON** switch	Matrix bus send on/off
2	**level/pan** control knob	Bus level, or pan when bus is linked. (The pans are not linked, only the send levels are linked.)
3	**PRE** switch	Pre-fader on/off
4	**level** control knob	Bus level

Master routing

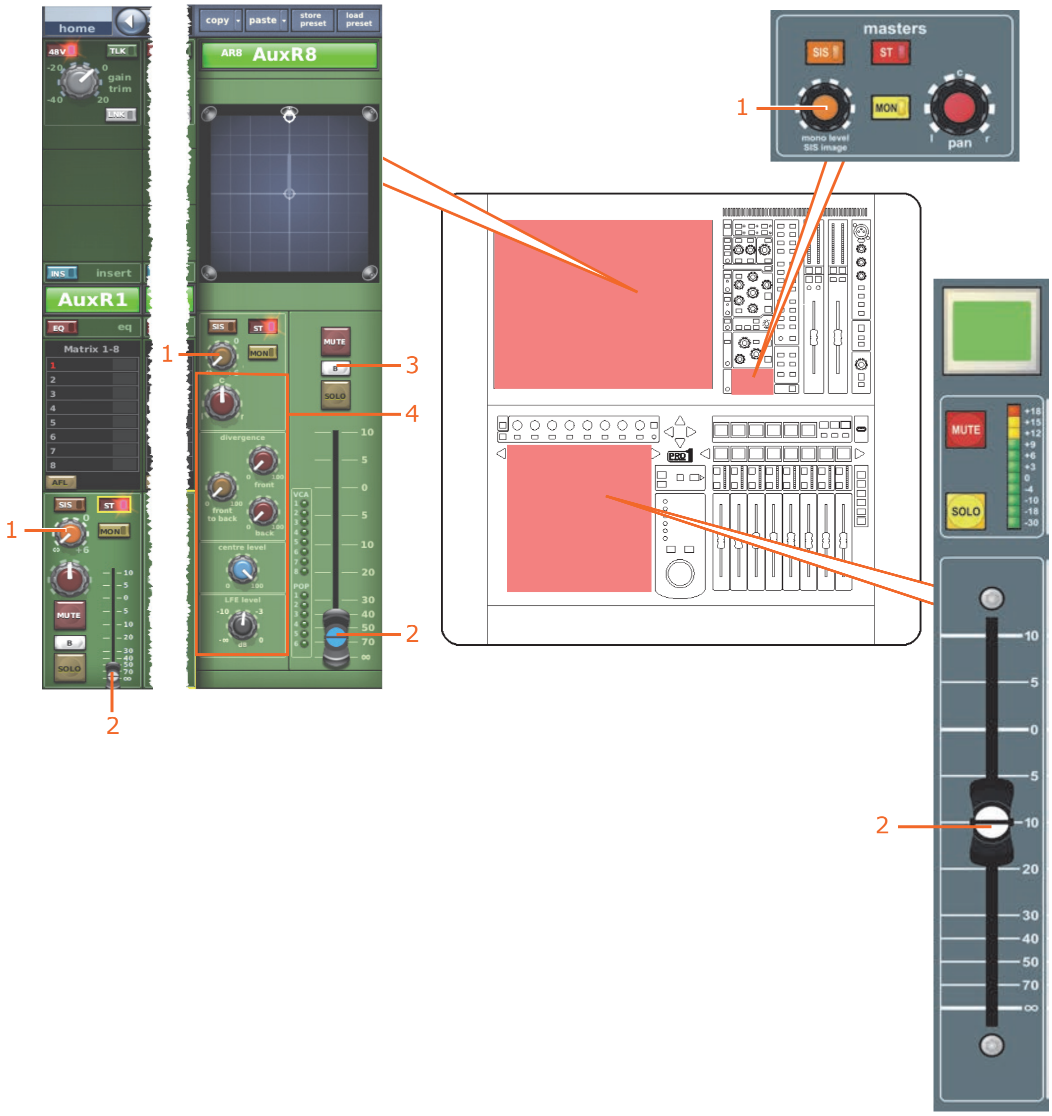
This section shows the master routing control parameters that are linked across a channel pair.



Item	Control	Parameter
1	**ST** switch	Stereo on/off
2	**MON** switch	Mono on/off
3	**SIS** switch	Spatial imaging system on/off

Fader

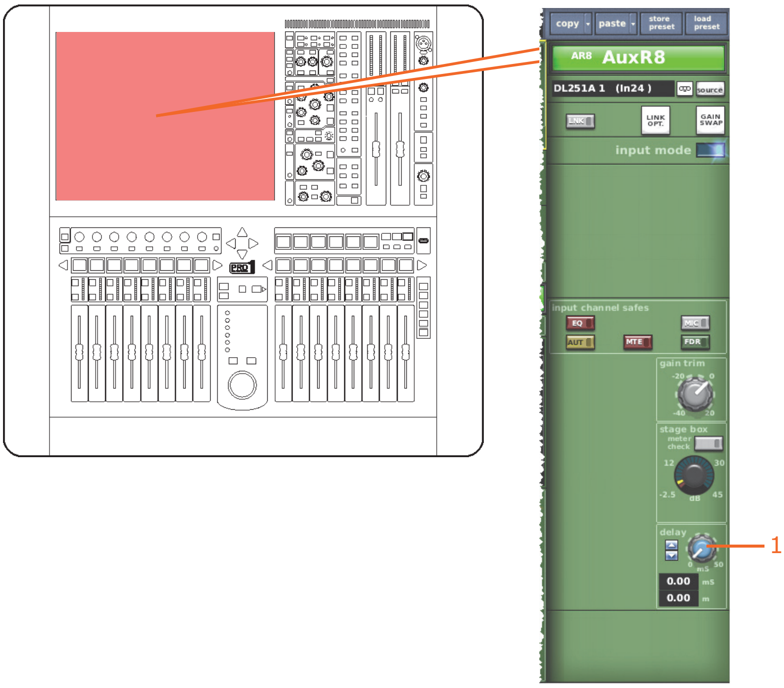
This section shows the fader control parameters that are linked across a channel pair.



Item	Control	Parameter
1	**mono level/SIS image** control knob	Mono send level. (Only linked when **SIS** is out on both channels and surround mode is off.)
2	Fader	Level
3	**B** switch	Solo B on/off
4	Surround control knobs	Surround panning levels

Delay

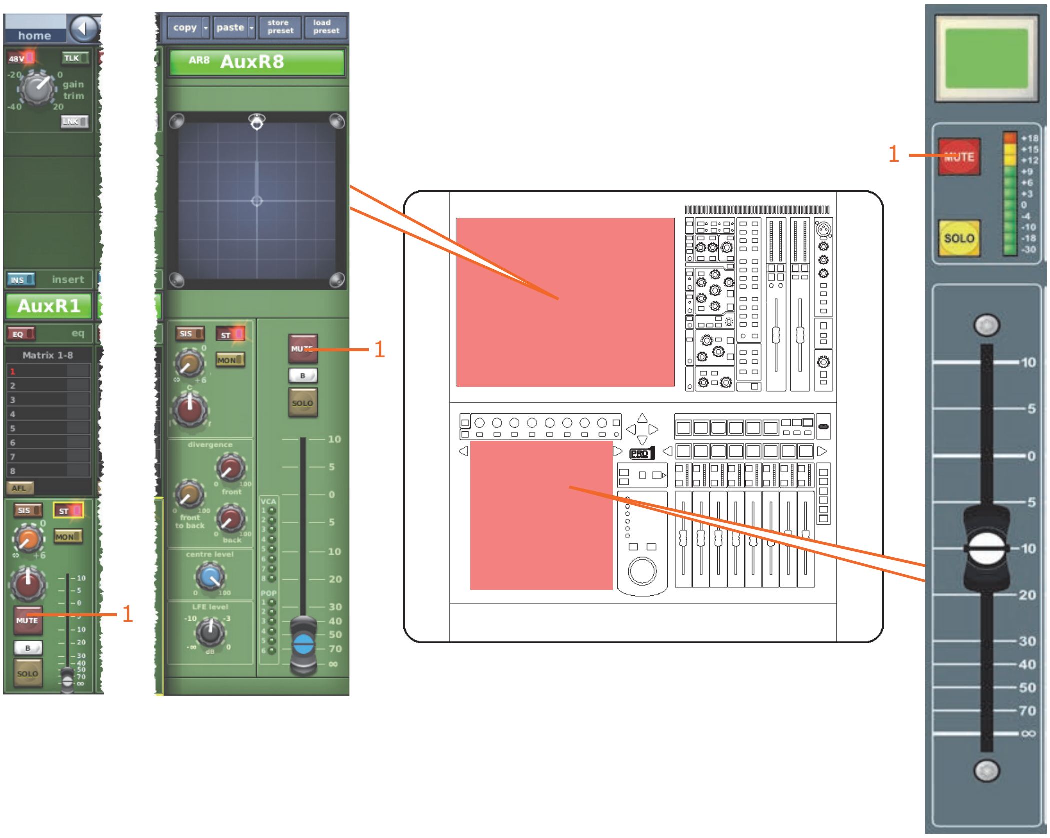
This section shows the delay control parameters that are linked across a channel pair.



Item	*Control*	*Parameter*
1	**delay** control knob	Delay time

Mute

This section shows the mute control parameters that are linked across a channel pair.



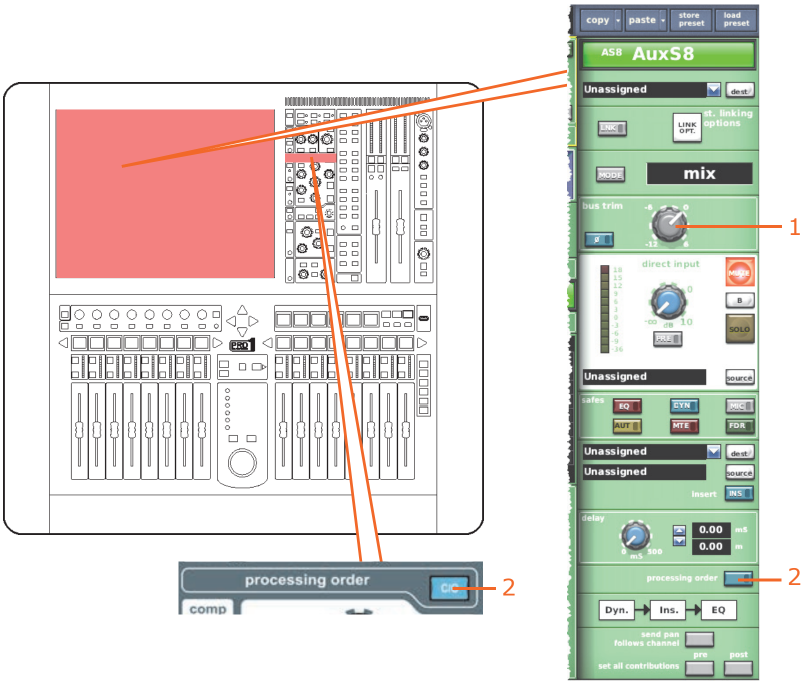
Item	Control	Parameter
1	**MUTE** button	Mutes the aux send signal

Aux Sends

This section shows the linked parameters of the aux channels.

Input controls

This section shows the input control parameters that are linked across a channel pair.



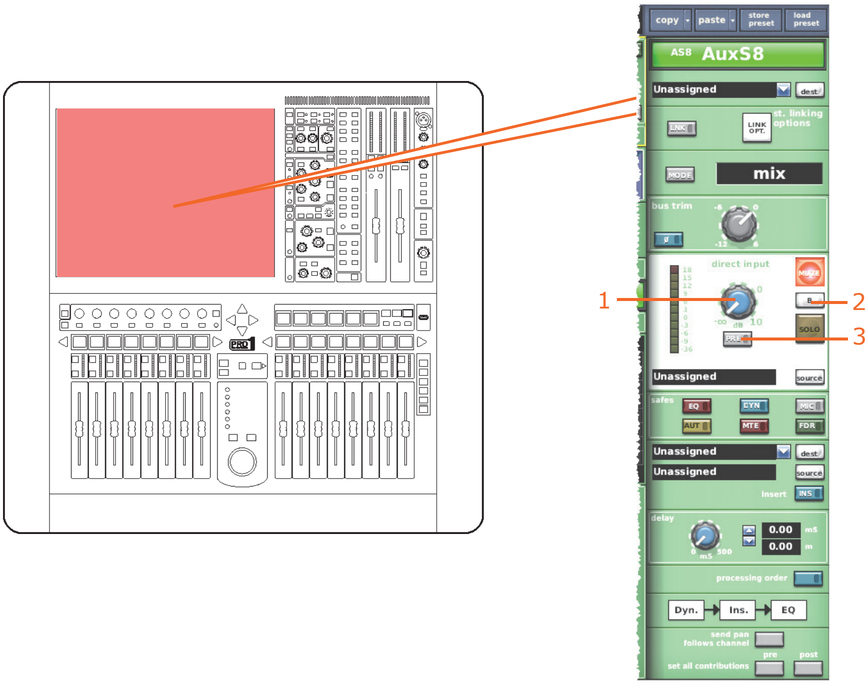
Item	*Control*	*Parameter*
1	**bus trim** control knob	Bus trim level
2	**C/O** switch	Order of processing: **Dyn.→Ins.→EQ** or **EQ→Ins.→Dyn.**

Direct output

Not applicable.

Direct input

This section shows the direct input parameters that are linked across the aux channel pair.



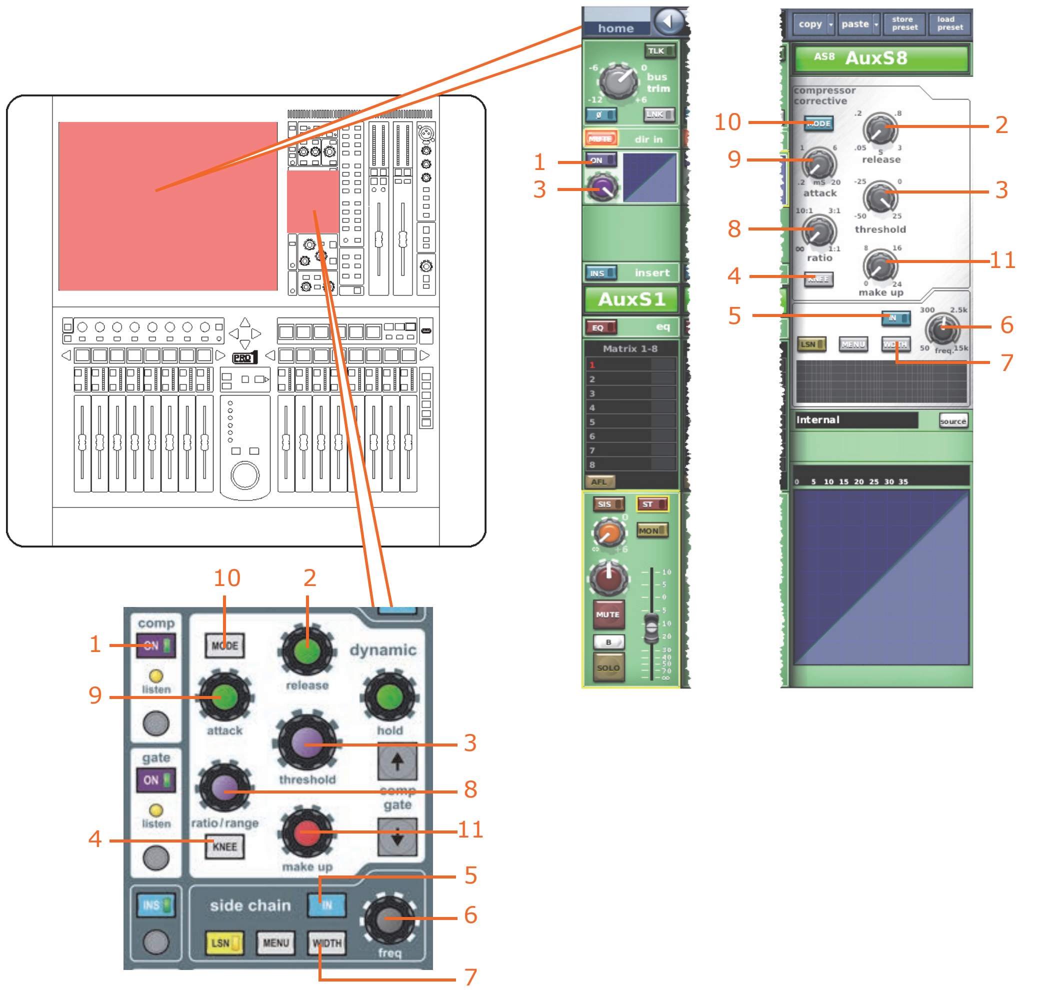
Item	Control	Parameter
1	Level control knob	Direct input level
2	**B** switch	Direct input solo B on/off
3	**PRE** switch	Direct input pre- in/out

Filters

Not applicable.

Dynamics

This section shows the compressor control parameters that are linked across a channel pair. Only corrective compressor shown below, but this is typically the same for the other compressor modes.

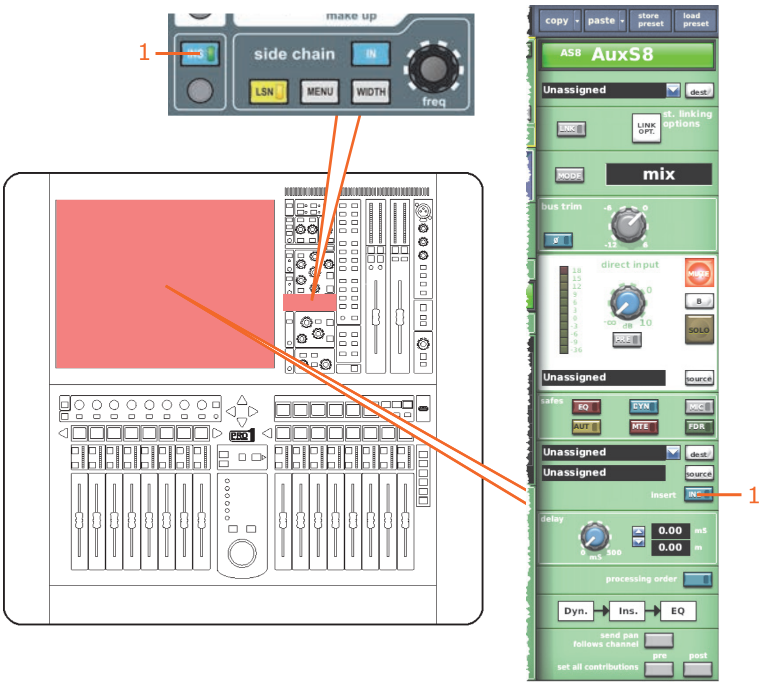


Item	*Control*	*Parameter*
1	**ON** switch	Compressor on/off
2	**release** control knob	Compressor release
3	**threshold** control knob	Compressor threshold
4	**KNEE** pushbutton	Compressor knee: hard, medium or soft
5	**IN** switch	Compressor sidechain in/out
6	**freq** control knob	Compressor sidechain frequency
7	**WIDTH** pushbutton	Compressor sidechain width: 2 Oct, 1 Oct or 0.3 Oct
8	**ratio** control knob	Compressor ratio
9	**attack** control knob	Compressor attack

Item	Control	Parameter
10	**MODE** pushbutton	Compressor mode: corrective, adaptive, creative, vintage or shimmer
11	**make up** control knob	Compressor gain

Insert

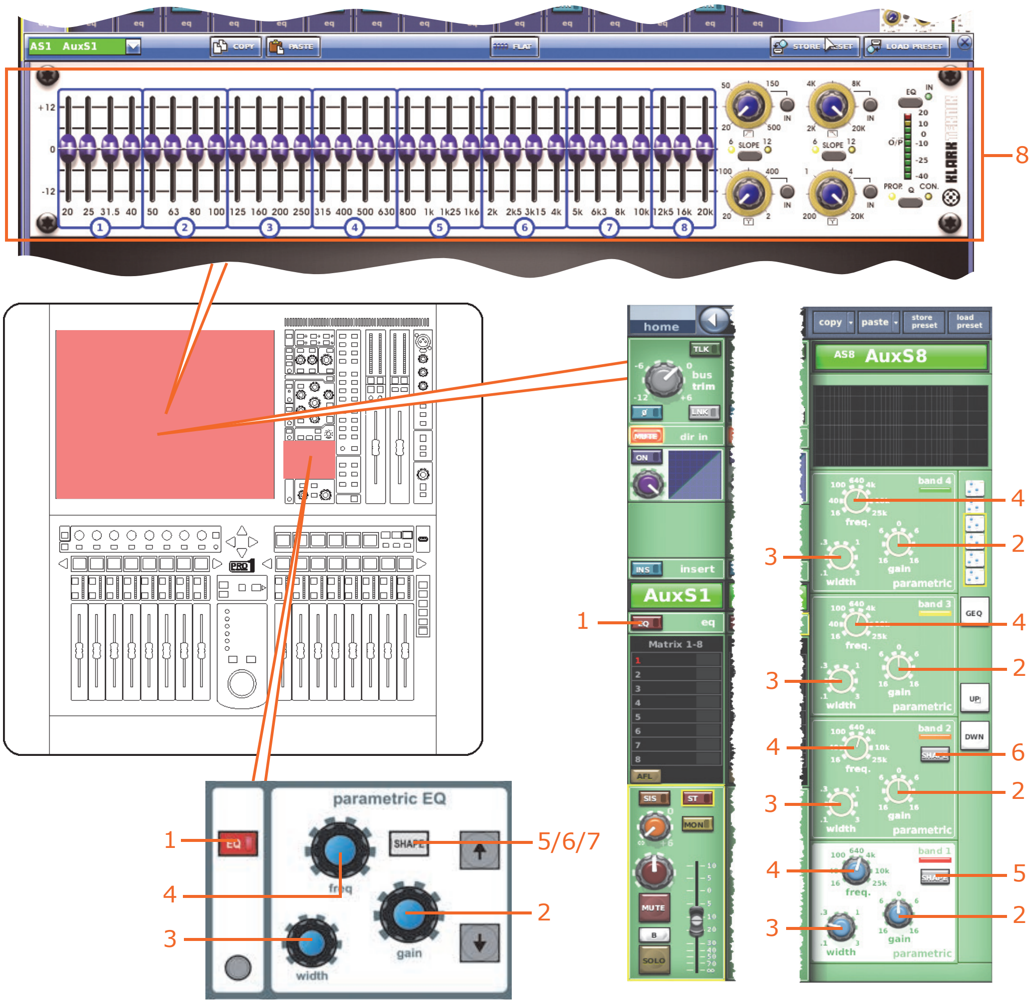
This section shows the insert control parameters that are linked across a channel pair.



Item	Control	Parameter
1	**INS** switch	Insert in/out

EQ

This section shows the EQ (and GEQ) control parameters that are linked across a channel pair.

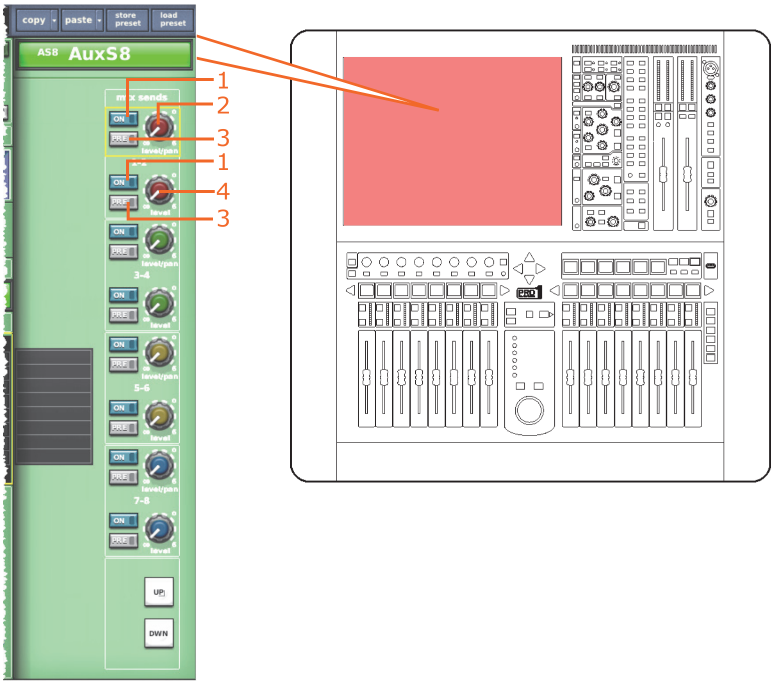


Item	*Control*	*Parameter*
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch	Band 1 shelving mode: bell, warm, high pass 6dB or high pass 12dB
6	**SHAPE** switch	Band 2 shelving mode: bell or high pass 24dB
7	**SHAPE** switch (not shown on GUI above)	Band 6 shelving modes: bell, soft, low pass 6dB or low pass 12dB
8	GEQ	All GEQ parameters. (GEQ parameters linked when both linked channels have a GEQ assigned to them.)

Note: *Although bands 5 and 6 are not shown above, the items in the table also apply. Both bands have items 2, 3 and 4, and band 6 also has item 7.*

Bus sends

This section shows the mix send control parameters that are linked across a channel pair.

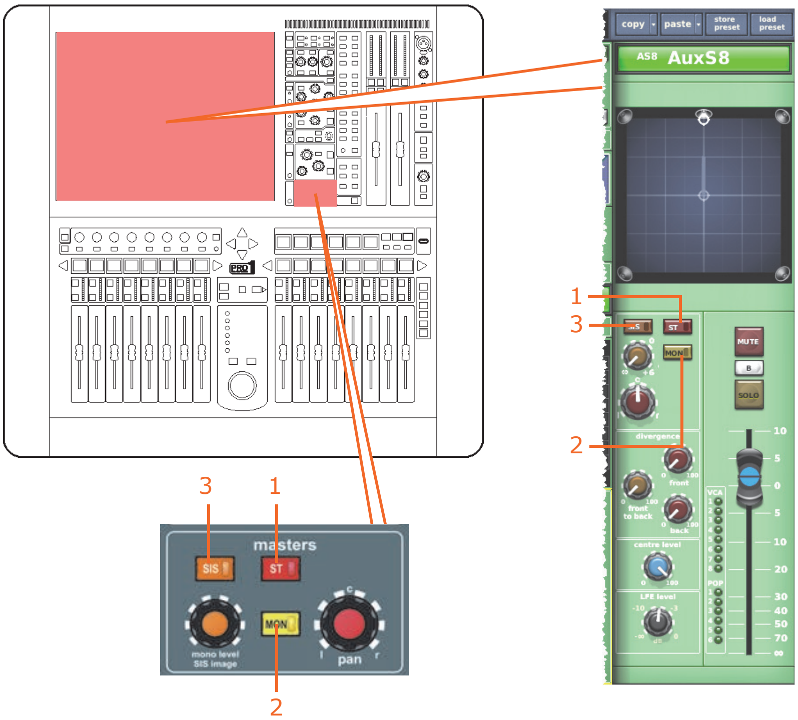


Note: *Although only matrix sends 1-2 are referenced above, this also applies to all eight matrix sends.*

Item	***Control***	***Parameter***
1	**ON** switch	Matrix bus send on/off
2	**level/pan** control knob	Bus level, or pan when bus is linked. (The pans are not linked, only the send levels are linked.)
3	**PRE** switch	Pre-fader on/off
4	**level** control knob	Bus level

Master routing

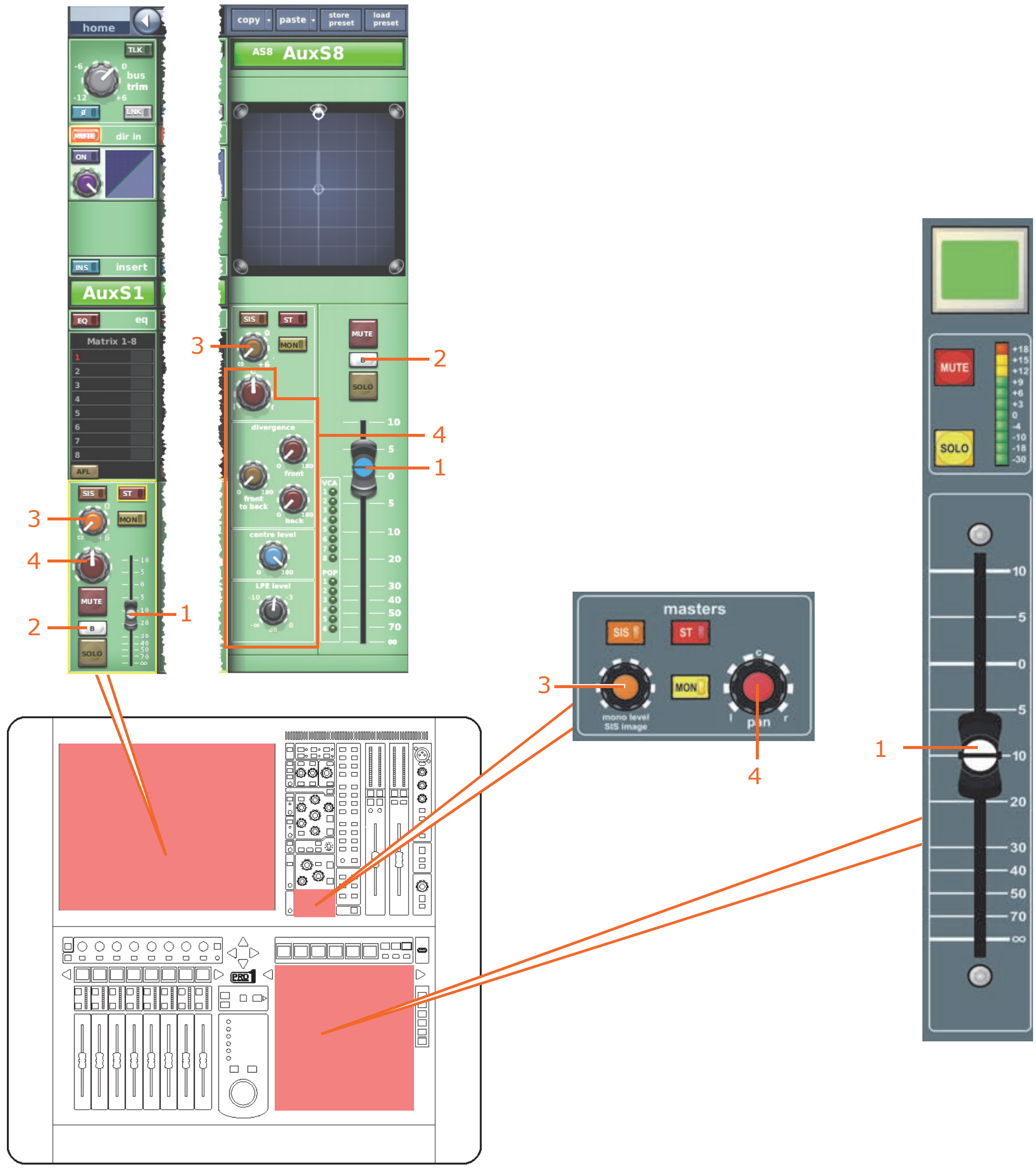
This section shows the master routing control parameters that are linked across a channel pair.



Item	*Control*	*Parameter*
1	**ST** switch	Stereo on/off
2	**MON** switch	Mono on/off
3	**SIS** switch	Spatial imaging system on/off

Fader

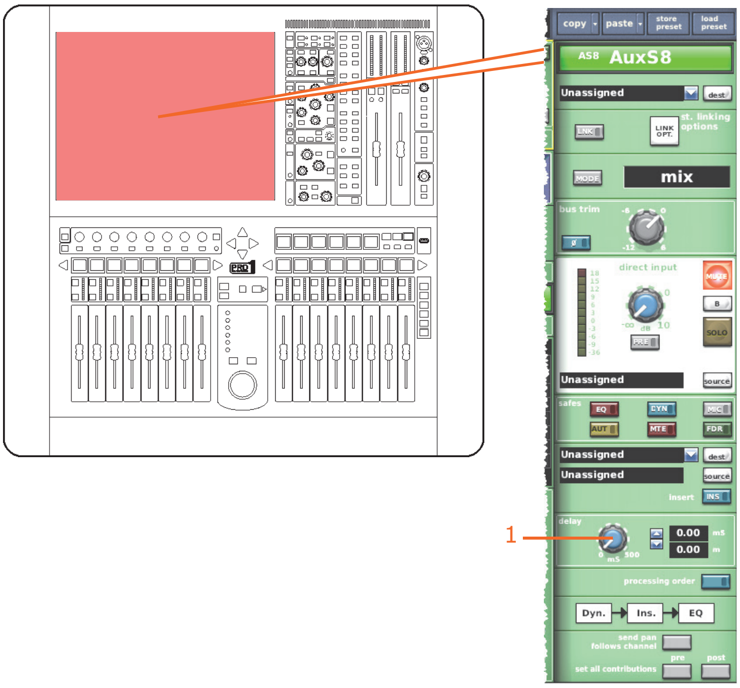
This section shows the fader control parameters that are linked across a channel pair.



Item	Control	Parameter
1	Fader	Level
2	**B** switch	Solo B on/off
3	**mono level/SIS image** control knob	Mono send level. (Only linked when SIS is out on both channels and surround mode is not selected.)
4	Panning control knobs	Surround panning, which includes all surround sound parameters, provided **Surround Mode** is not configured as **None** (see "Selecting the surround mode" on page 225)

Delay

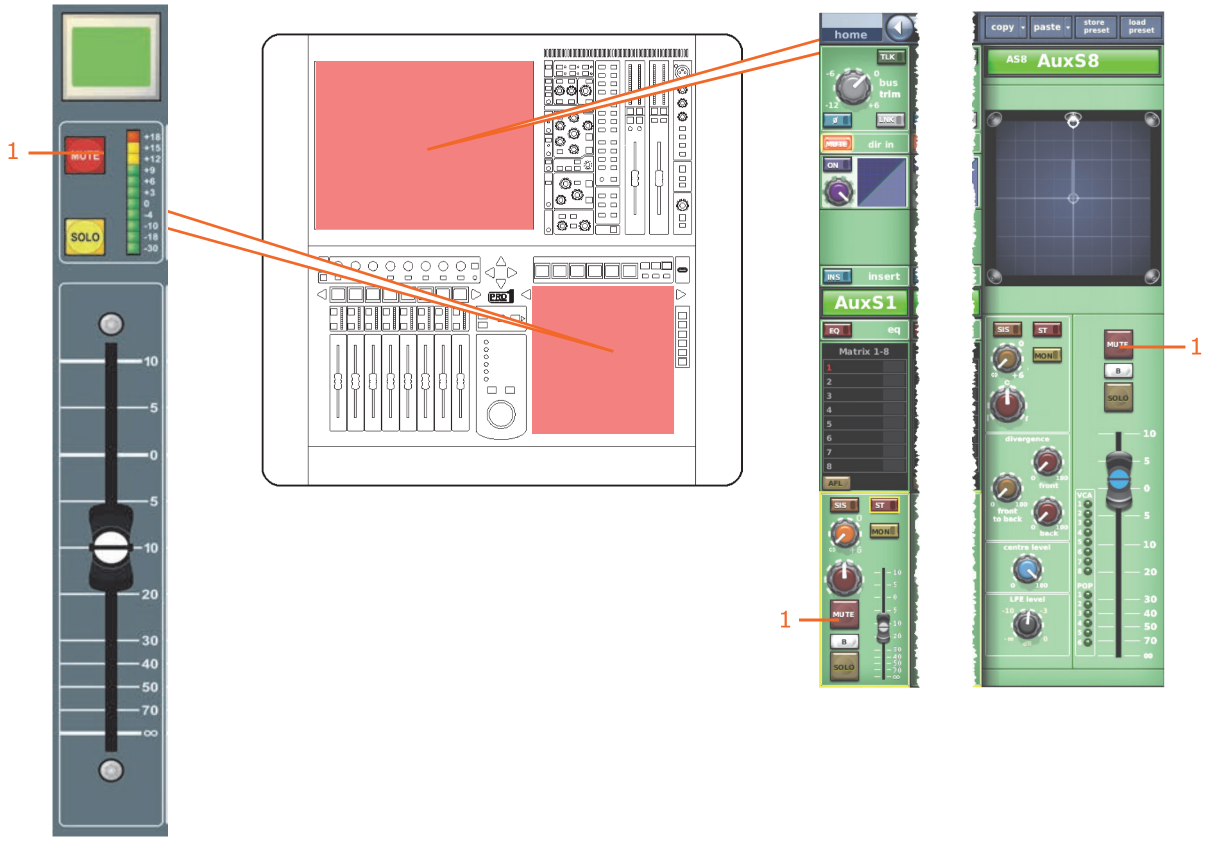
This section shows the delay control parameters that are linked across a channel pair.



Item	*Control*	*Parameter*
1	**delay** control knob	Delay time

Mute

This section shows the mute control parameters that are linked across a channel pair.



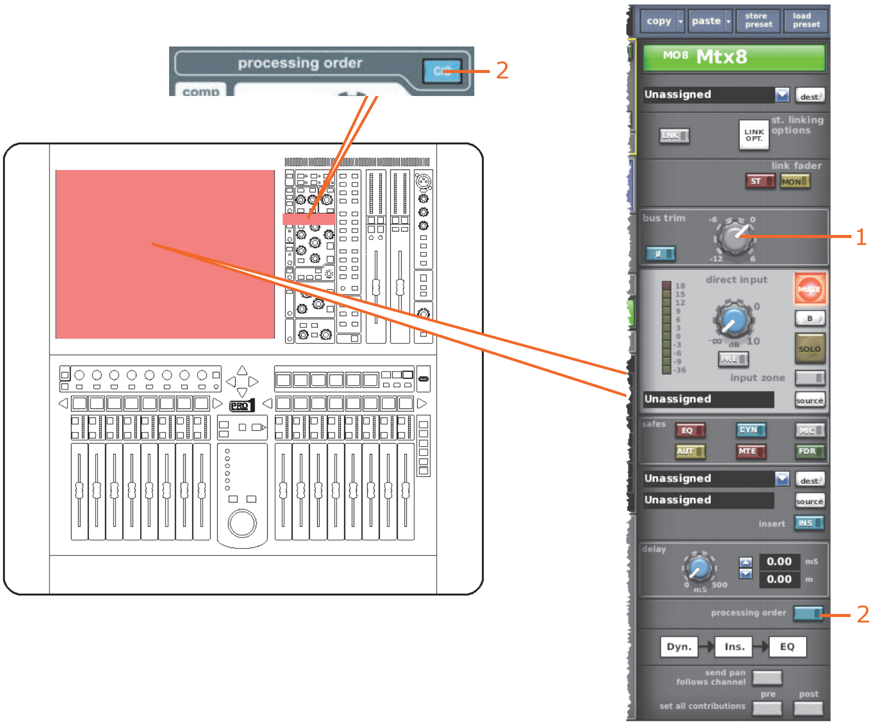
Item	*Control*	*Parameter*
1	**MUTE** button	Mutes the aux send signal

Matrix

This section shows the linked parameters of the matrix channels.

Input controls

This section shows the input control parameters that are linked across a channel pair.



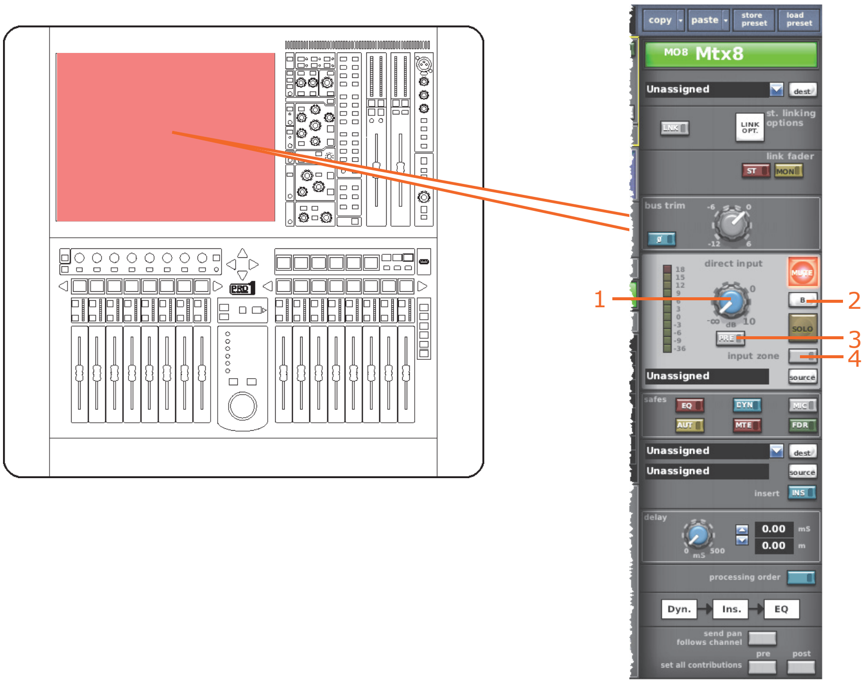
Item	*Control*	*Parameter*
1	**bus trim** control knob	Bus trim level
2	**C/O** switch	Order of processing: **Dyn.**→**Ins.**→**EQ** or **EQ**→**Ins.**→**Dyn.**

Direct output

Not applicable.

Direct input

This section shows the direct input control parameters that are linked across a channel pair.



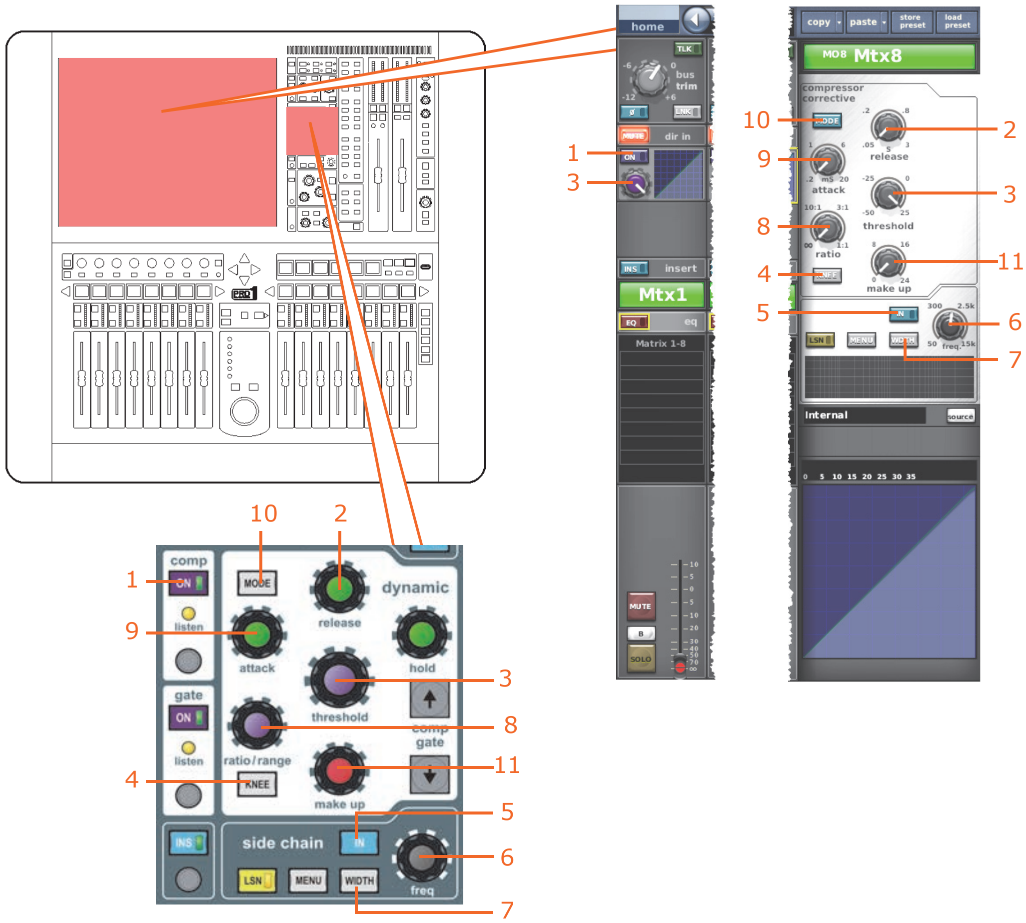
Item	Control	Parameter
1	Level control knob	Direct input level
2	**B** switch	Direct input solo B on/off
3	**PRE** switch	Direct input pre- in/out
4	**input zone** switch	Input zone in/out

Filters

Not applicable.

Dynamics

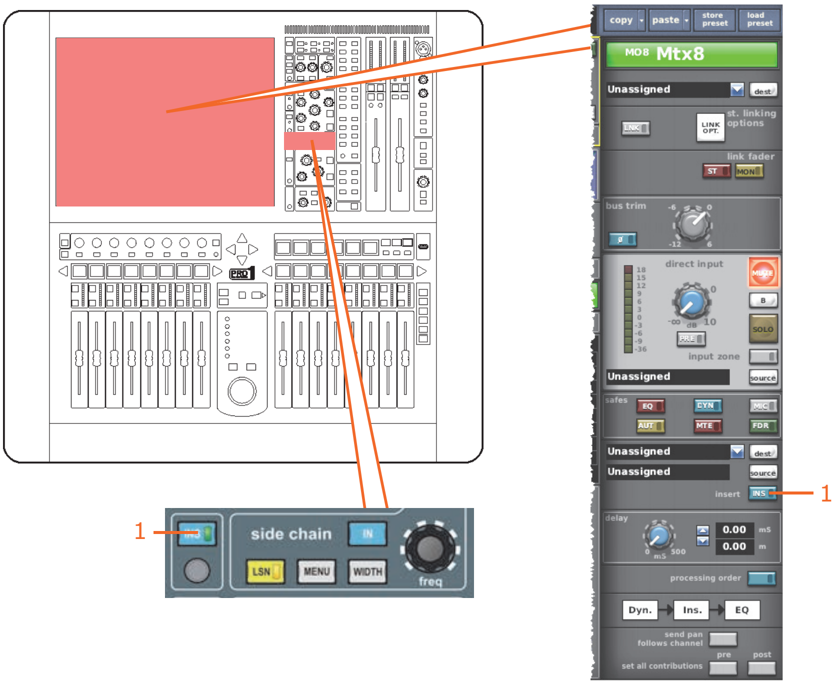
This section shows compressor control parameters that are linked across a channel pair. Only corrective compressor shown below, but this is typically the same for the other compressor modes.



Item	*Control*	*Parameter*
1	**ON** switch	Compressor on/off
2	**release** control knob	Compressor release
3	**threshold** control knob	Compressor threshold
4	**KNEE** pushbutton	Compressor knee: hard, medium or soft
5	**IN** switch	Compressor sidechain in/out
6	**freq** control knob	Compressor sidechain frequency
7	**WIDTH** pushbutton	Compressor sidechain width: 2 Oct, 1 Oct or 0.3 Oct
8	**ratio** control knob	Compressor ratio
9	**attack** control knob	Compressor attack
10	**MODE** pushbutton	Compressor mode: corrective, adaptive, creative, vintage or shimmer
11	**make up** control knob	Compressor gain

Insert

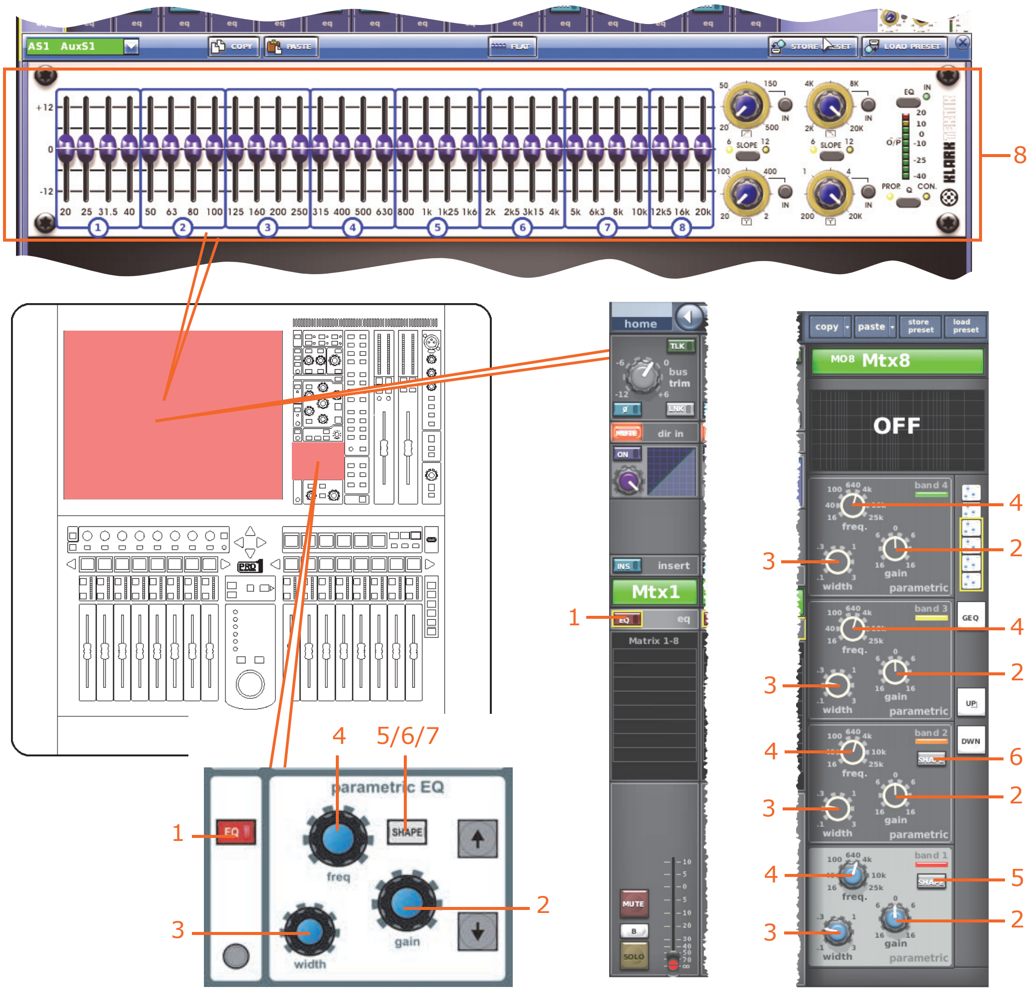
This section shows the insert control parameters that are linked across a channel pair.



Item	Control	Parameter
1	**INS** switch	Insert in/out

EQ

This section shows the EQ (and GEQ) control parameters that are linked across a channel pair.



Item	Control	Parameter
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch	Band 1 shelving mode: bell, warm, high pass 6dB or high pass 12dB
6	**SHAPE** switch	Band 2 shelving mode: bell or high pass 24dB
7	**SHAPE** switch (not shown on GUI above)	Band 6 shelving mode: bell, soft, low pass 6dB or low pass 12dB
8	GEQ	All GEQ parameters. (GEQ parameters linked when both linked channels have a GEQ assigned to them.)

***Note:** Although bands 5 and 6 are not shown above, the items in the table also apply. Both bands have items 2, 3 and 4, and band 6 also has item 7.*

Bus sends

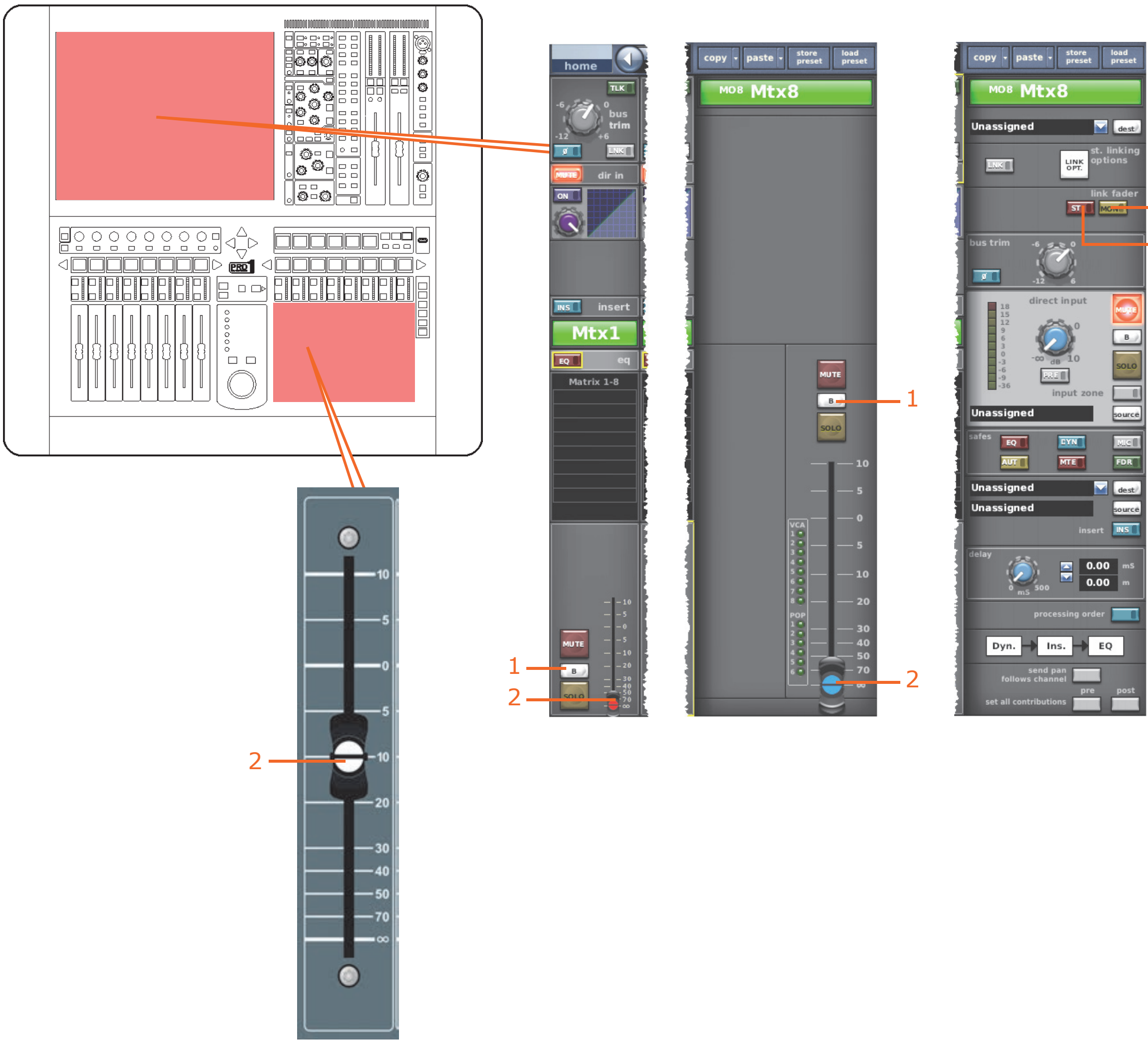
Not applicable.

Master routing

Not applicable.

Fader

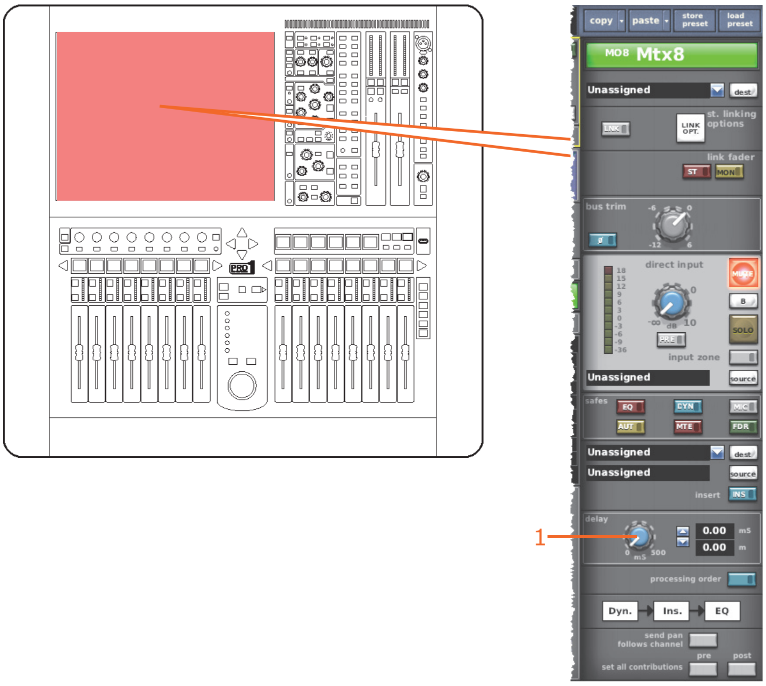
This section shows the fader control parameters that are linked across a channel pair.



Item	Control	Parameter
1	**B** switch	Solo B on/off
2	Fader	Level
3	**MON** switch	Link to mono master fader
4	**ST** switch	Link to stereo master fader

Delay

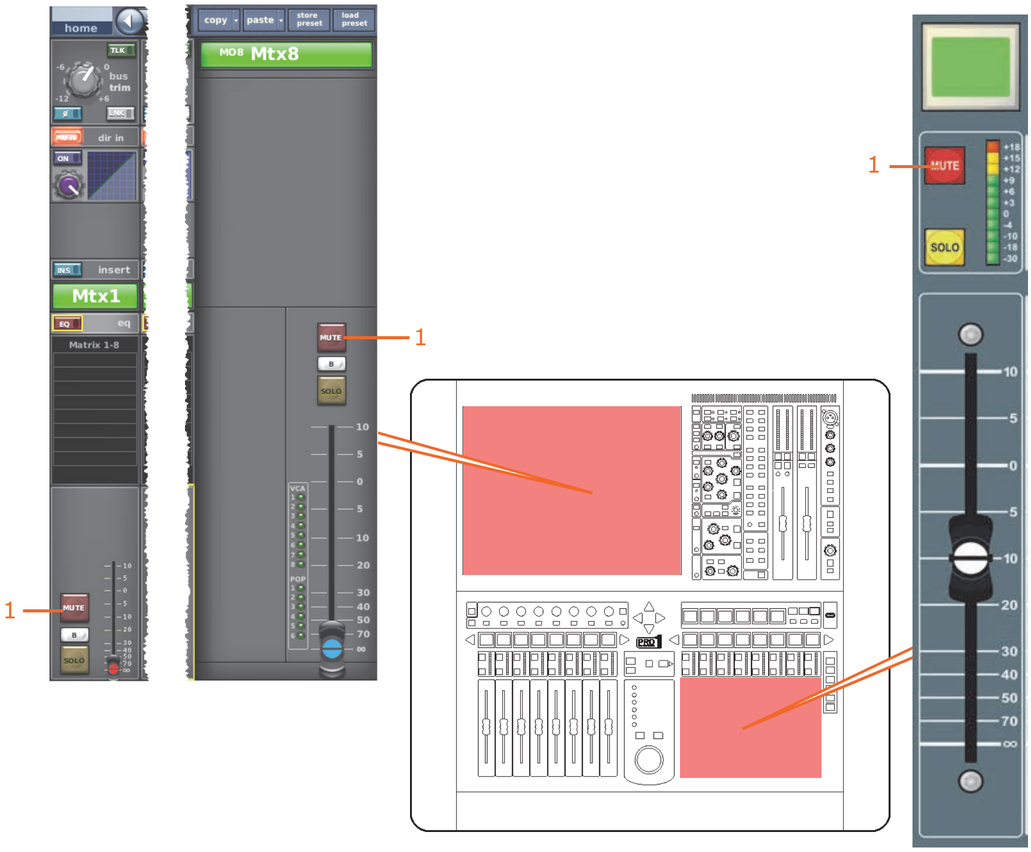
This section shows the delay control parameters that are linked across a channel pair.



Item	*Control*	*Parameter*
1	**delay** control knob	Delay time

Mute

This section shows the mute control parameters that are linked across a channel pair.



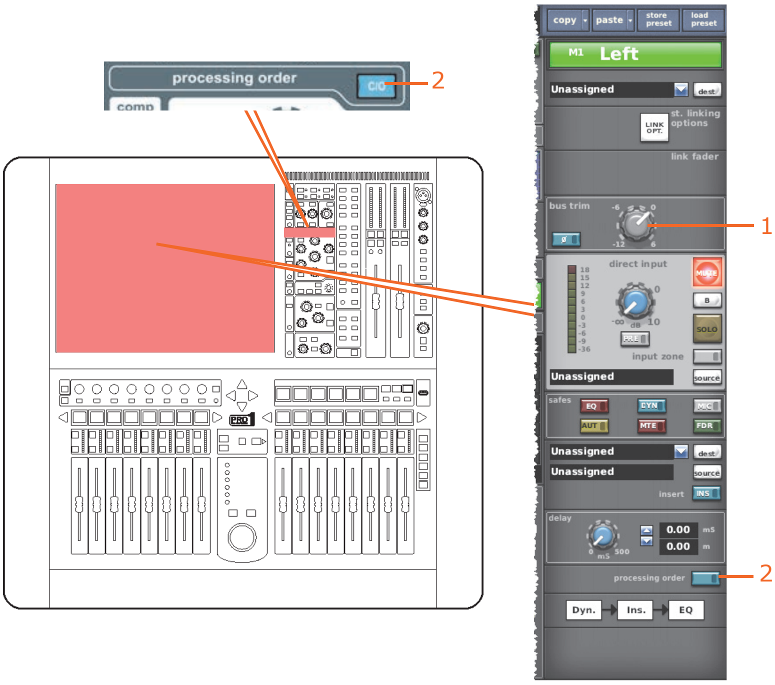
Item	*Control*	*Parameter*
1	**MUTE** button	Mutes the aux send signal

Masters

This section shows the linked parameters of the master channels.

Input controls

This section shows the input control parameters that are linked across a channel pair.



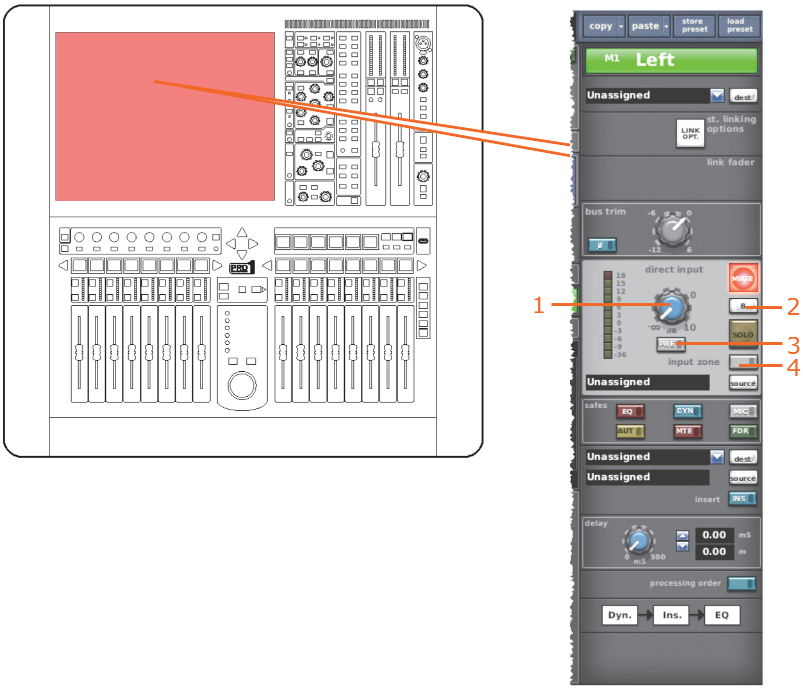
Item	*Control*	*Parameter*
1	**bus trim** control knob	Bus trim level
2	**C/O** switch	Order of processing: **Dyn.→Ins.→EQ** or **EQ→Ins.→Dyn.**

Direct output

Not applicable.

Direct input

This section shows the direct input control parameters that are linked across a channel pair.



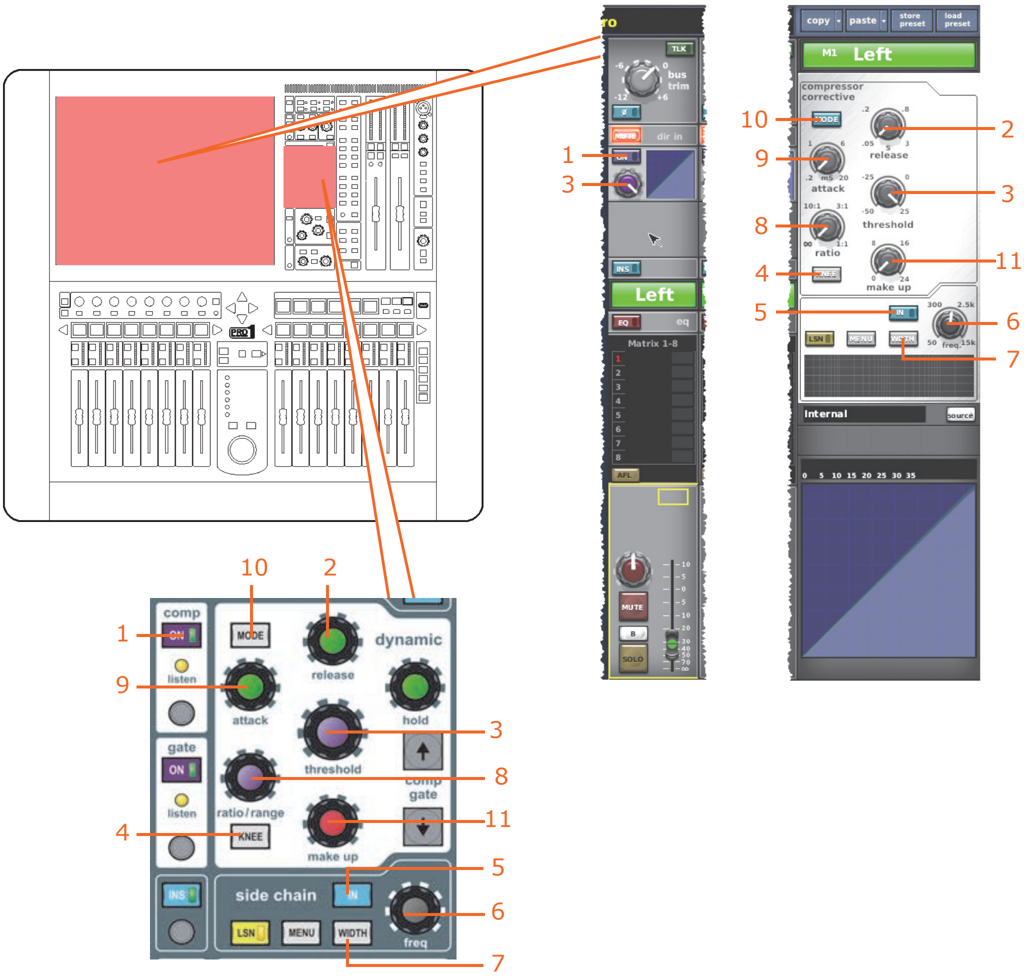
Item	*Control*	*Parameter*
1	Level control knob	Direct input level
2	**B** switch	Direct input solo B on/off
3	**PRE** switch	Direct input pre- in/out
3	**input zone** switch	Input zone in/out

Filters

Not applicable.

Dynamics

This section shows the compressor control parameters that are linked across a channel pair. Only corrective compressor shown below, but typically the same for the other compressor modes.

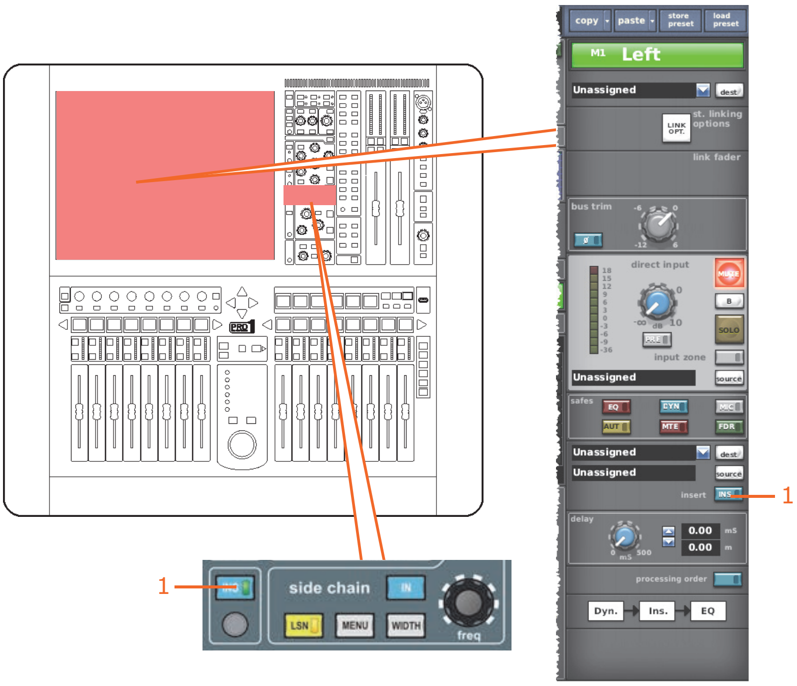


Item	Control	Parameter
1	**ON** switch	Compressor on/off
2	**release** control knob	Compressor release
3	**threshold** control knob	Compressor threshold
4	**KNEE** pushbutton	Compressor knee: hard, medium or soft
5	**IN** switch	Compressor sidechain in/out
6	**freq** control knob	Compressor sidechain frequency
7	**WIDTH** pushbutton	Compressor sidechain width: 2 Oct, 1 Oct or 0.3 Oct
8	**ratio** control knob	Compressor ratio
9	**attack** control knob	Compressor attack

Item	*Control*	*Parameter*
10	**MODE** pushbutton	Compressor mode: corrective, adaptive, creative, vintage or shimmer
11	**make up** control knob	Compressor gain

Insert

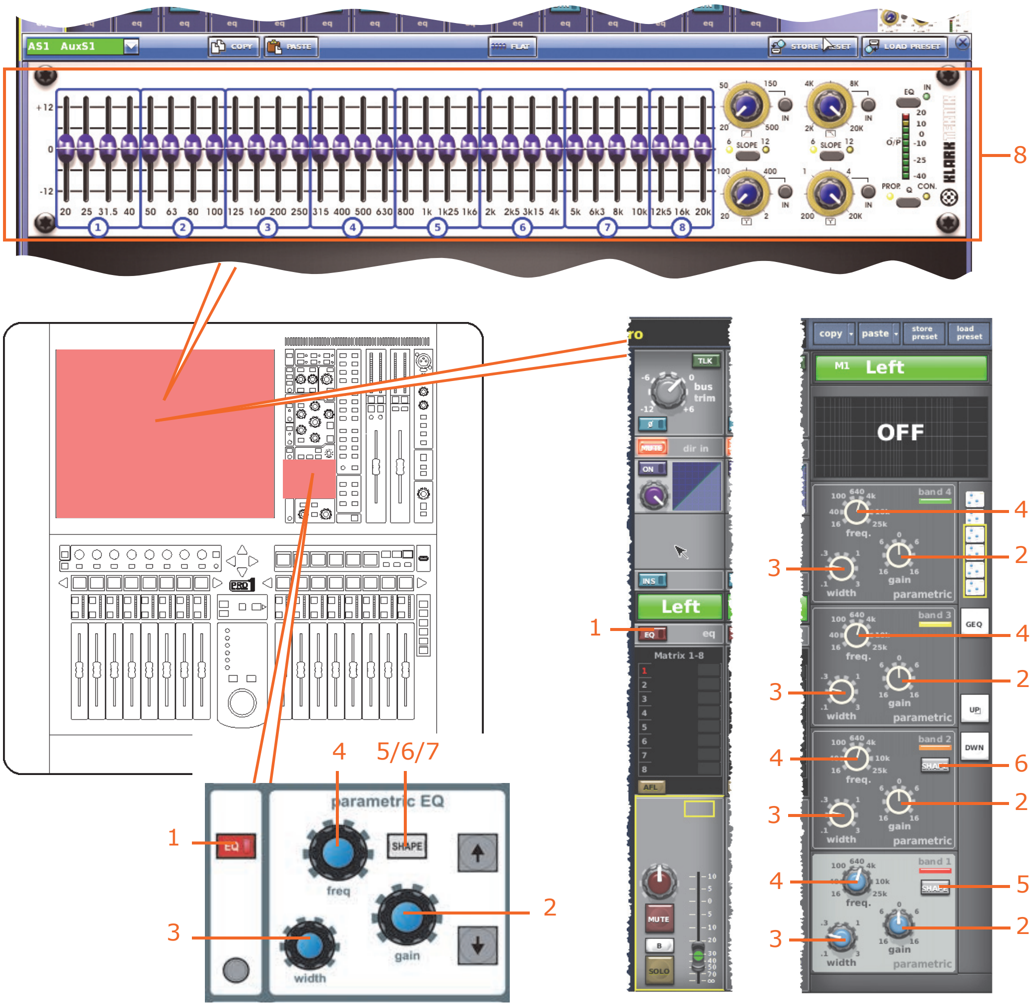
This section shows the insert control parameters that are linked across a channel pair.



Item	*Control*	*Parameter*
1	**INS** switch	Insert in/out

EQ

This section shows the EQ (and GEQ) control parameters that are linked across a channel pair.

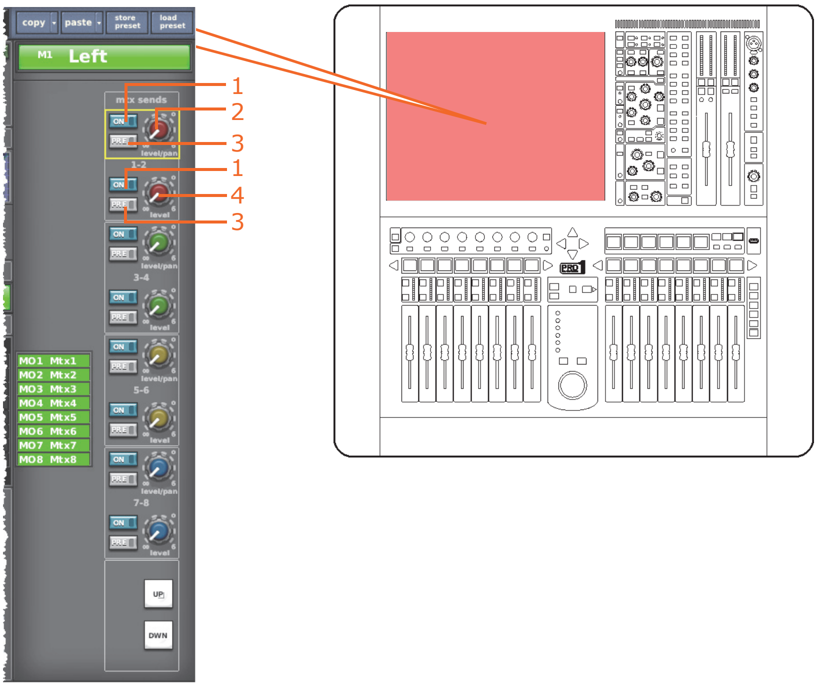


Item	Control	Parameter
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch	Band 1 shelving mode: bell, warm, high pass 6dB or high pass 12dB
6	**SHAPE** switch	Band 2 shelving mode: bell or high pass 24dB
7	**SHAPE** switch (not shown on GUI above)	Band 6 shelving mode: bell, soft, low pass 6dB or low pass 12dB
8	GEQ	All GEQ parameters. (GEQ parameters linked when both linked channels have a GEQ assigned to them.)

Note: *Although bands 5 and 6 are not shown above, the items in the table also apply. Both bands have items 2, 3 and 4, band 6 also has item 7.*

Bus sends

This section shows the mix send control parameters that are linked across a channel pair.

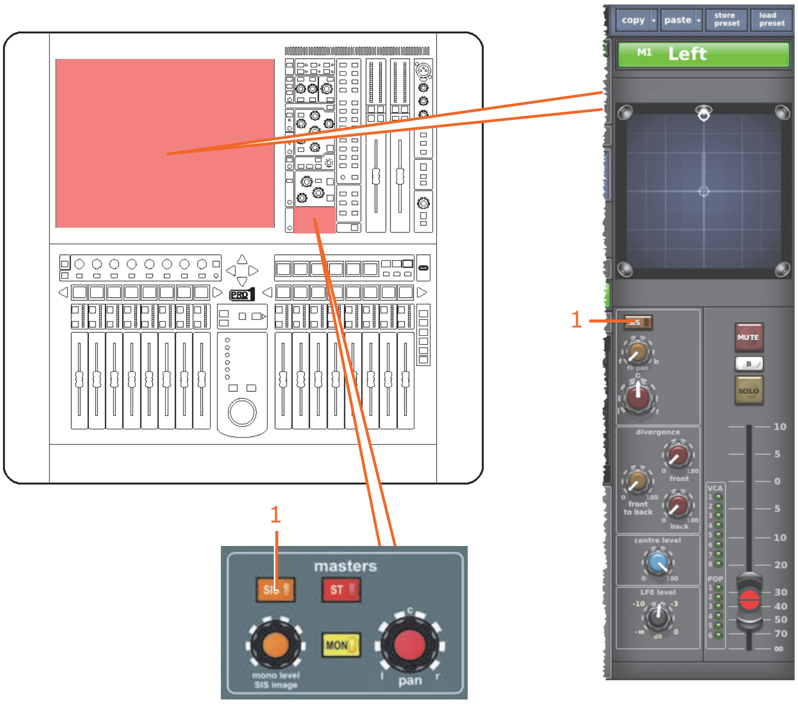


Note: *Although only matrix sends 1-2 are referenced above, this also applies to all eight matrix sends.*

Item	*Control*	*Parameter*
1	**ON** switch	Matrix bus send on/off
2	**level/pan** control knob	Bus level, or pan when bus is linked. (The pans are not linked, only the send levels are linked.)
3	**PRE** switch	Pre-fader on/off
4	**level** control knob	Bus level

Master routing

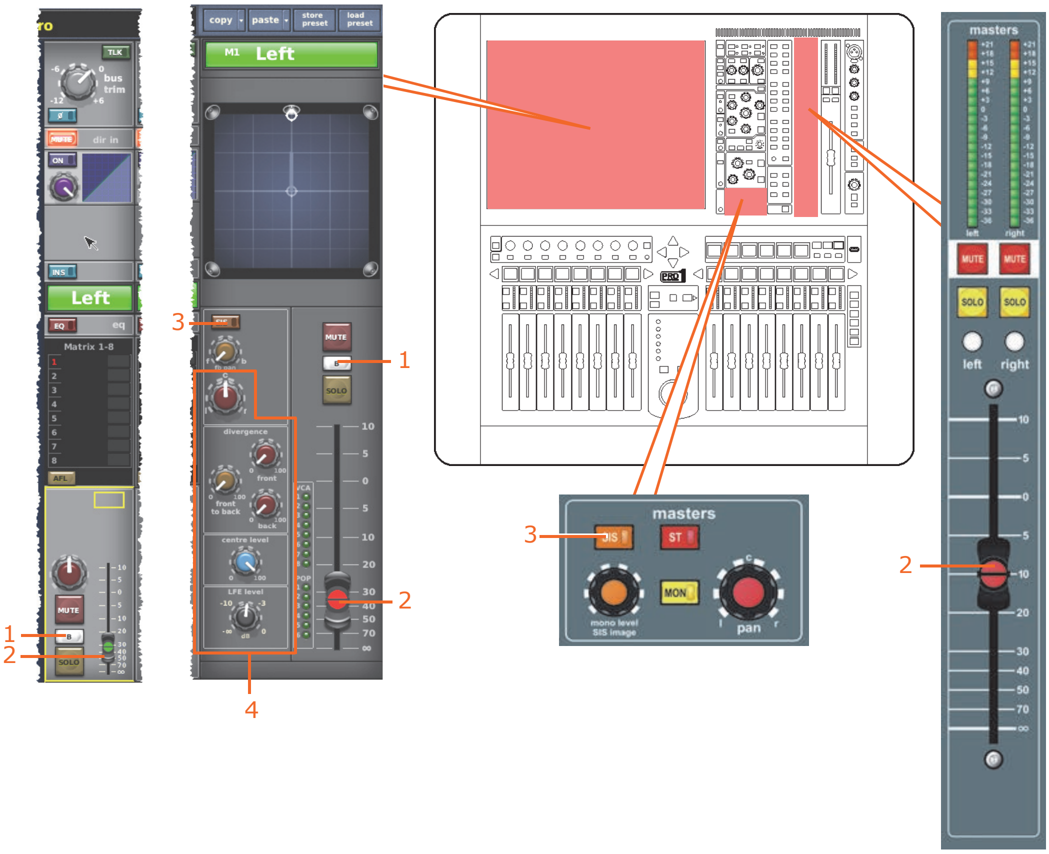
This section shows the master routing control parameters that are linked across a channel pair.



Item	Control	Parameter
1	**SIS** switch	Spatial imaging system on/off. (Only available if one of the surround modes is selected.)

Fader

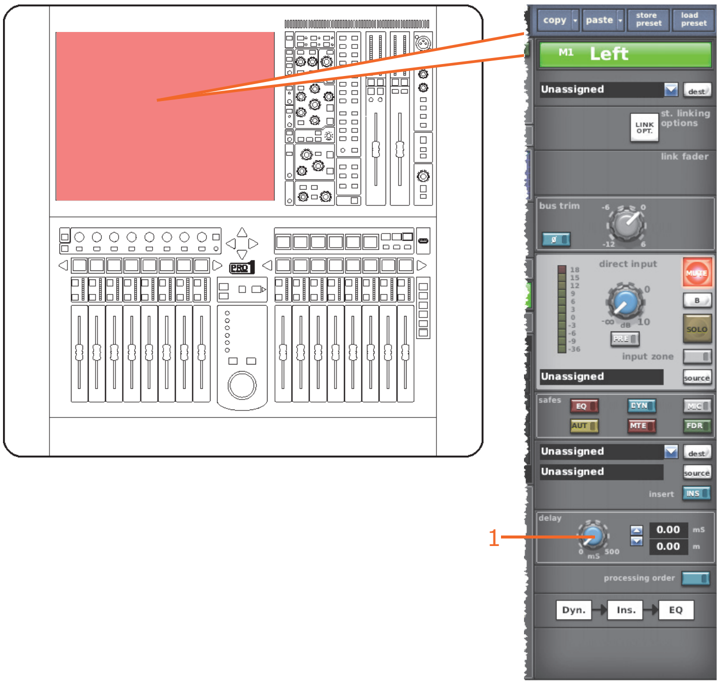
This section shows the fader control parameters that are linked across a channel pair.



Item	*Control*	*Parameter*
1	**B** switch	Solo B on/off
2	Fader	Level
3	**SIS** switch	Route to surround on/off
4	Surround control knobs	Surround panning levels

Delay

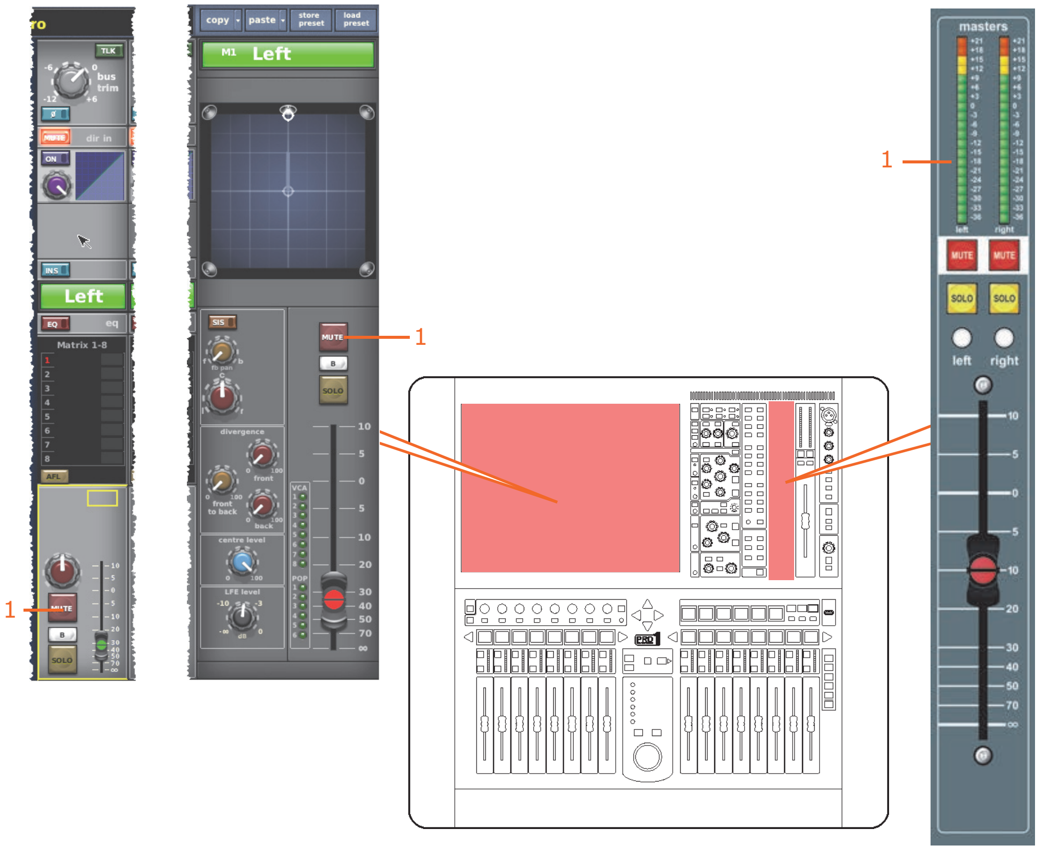
This section shows the delay control parameters that are linked across a channel pair.



Item	*Control*	*Parameter*
1	**delay** control knob	Delay time

Mute

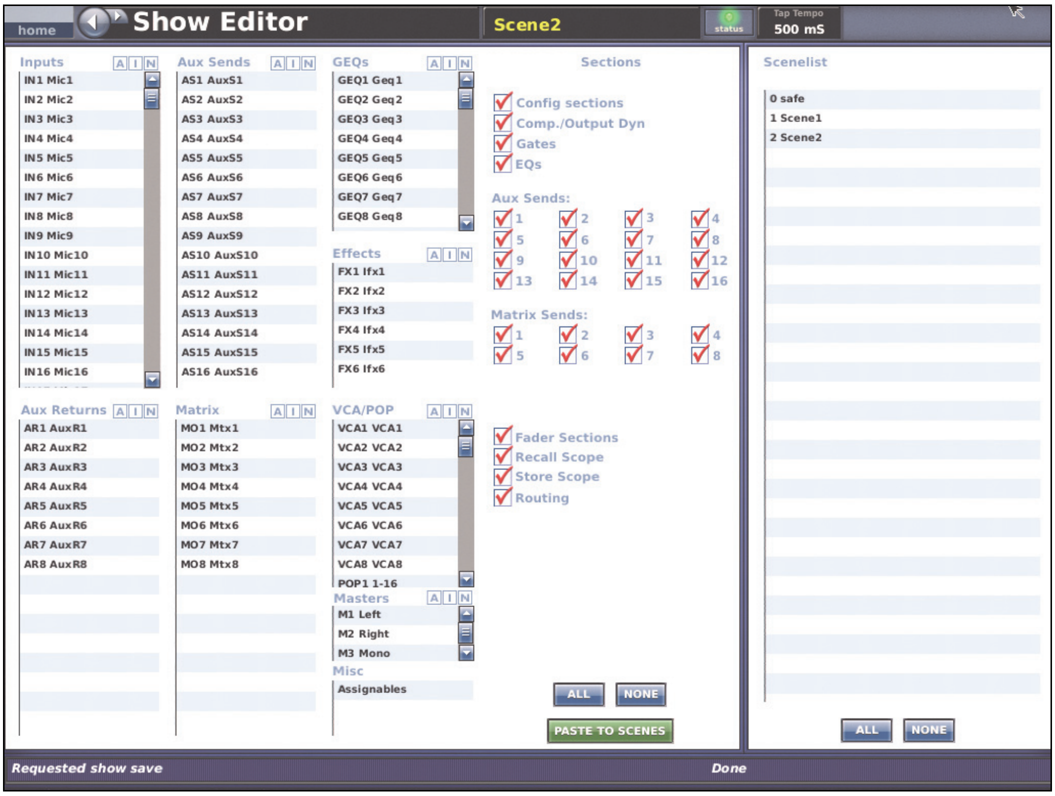
This section shows the mute control parameters that are linked across a channel pair.



Item	Control	Parameter
1	**MUTE** button	Mutes the aux send signal

Appendix K: Parameters Copied Through Scenes

This appendix shows the parameters per section — selected from the **Sections** panel in the **Show Editor** screen (shown below) — that can be copied through scenes.



The following table is intended as a reference guide to help you go quickly to the channel control area you want.

Control area	*Inputs (Input Channels)*	*Aux Returns (Returns)*	*Aux Sends (Auxes)*	*Matrix (Matrices)*	*Masters*
Config sections	Page 519	Page 527	Page 533	Page 539	Page 545
Comp./ Output Dyn	Page 520	N/A	Page 534	Page 540	Page 546
Gates	Page 521	N/A	N/A	N/A	N/A
EQs	Page 522	Page 528	Page 535	Page 541	Page 547
Aux Sends	Page 523	Page 529	N/A	N/A	N/A
Matrix Sends	Page 524	Page 530	Page 536	N/A	Page 548
Fader Sections	Page 525	Page 531	Page 537	Page 542	Page 549
Recall Scope	Page 525	Page 531	Page 538	Page 543	Page 549
Store Scope	Page 525	Page 531	Page 538	Page 543	Page 549
Routing	Page 526	Page 532	Page 538	Page 543	Page 550

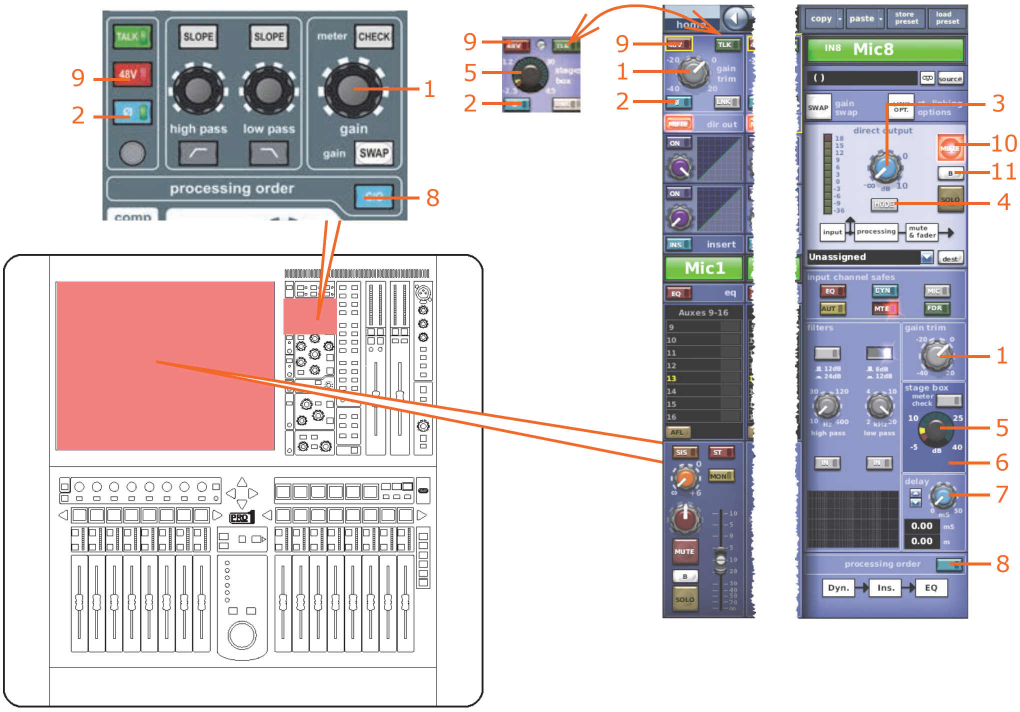
For details of the GEQ, effect, group and assignable control parameters copied through scenes, see “GEQs” on page 544, “Effects” on page 544, “VCA/POPulation (groups)” on page 545 and “Misc (miscellaneous)” on page 550, respectively.

Inputs (input channels)

This section shows you which input channel parameters are affected by copy through scenes.

Config sections

This section shows the configuration processing area parameters copied through scenes.

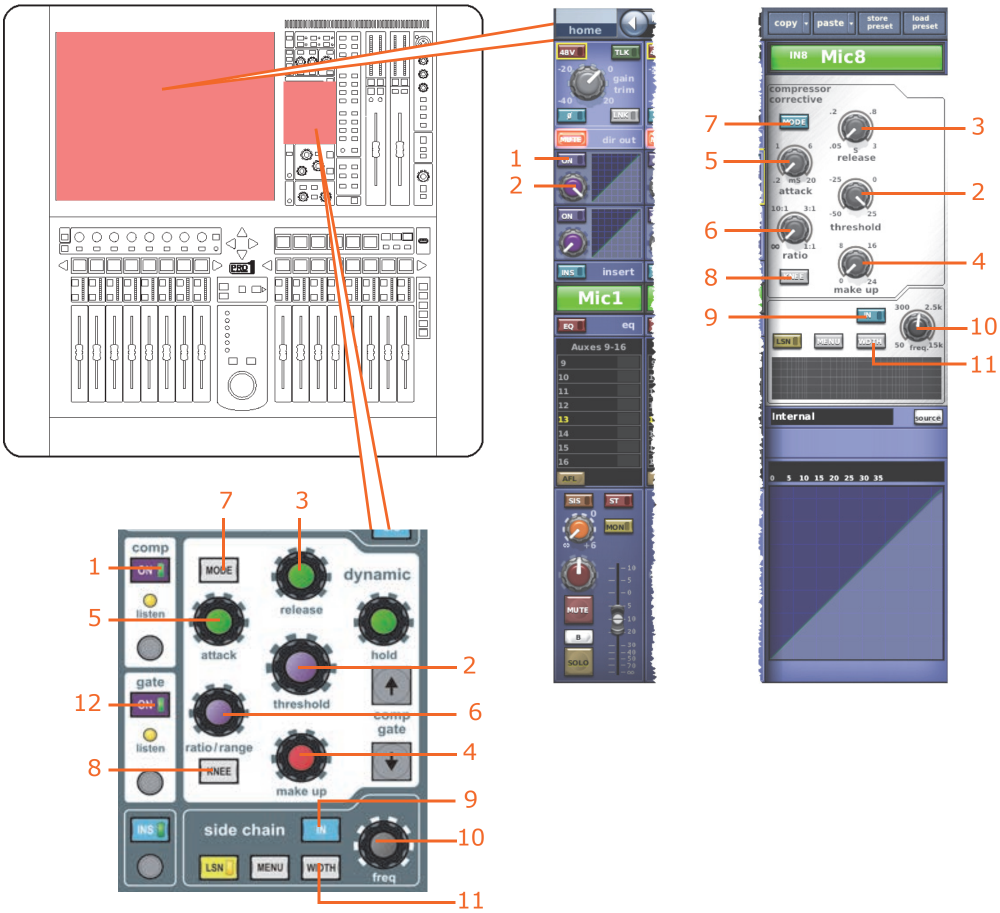


Item	Control	Parameter
1	**gain**/[**gain trim**] control knob	Digital trim level
2	**Ø** switch	Phase on/off switch
3	Level control knob	Direct output level
4	**MODE** switch	Direct output tap-off point: "Post-fade and mute", "Pre-mute, pre-processing" and "Pre-mute, post-processing"
5	**stage box** control knob*	Remote amplifier level
6	Filter switch* (not shown on the GUI above)	30Hz filter on/off
7	**delay** control knob	Delay time
8	**C/O** switch	Order of processing: **Dyn.→Ins.→EQ** or **EQ→Ins.→Dyn.**
9	**48V** switch*	48V phantom voltage on/off
10	**MUTE** switch	Direct output mute on/off
11	**B** switch	Direct output solo B on/off

* Applies to primary and tape inputs.

Comp./Output Dyn

This section shows the compressor processing area parameters copied through scenes.



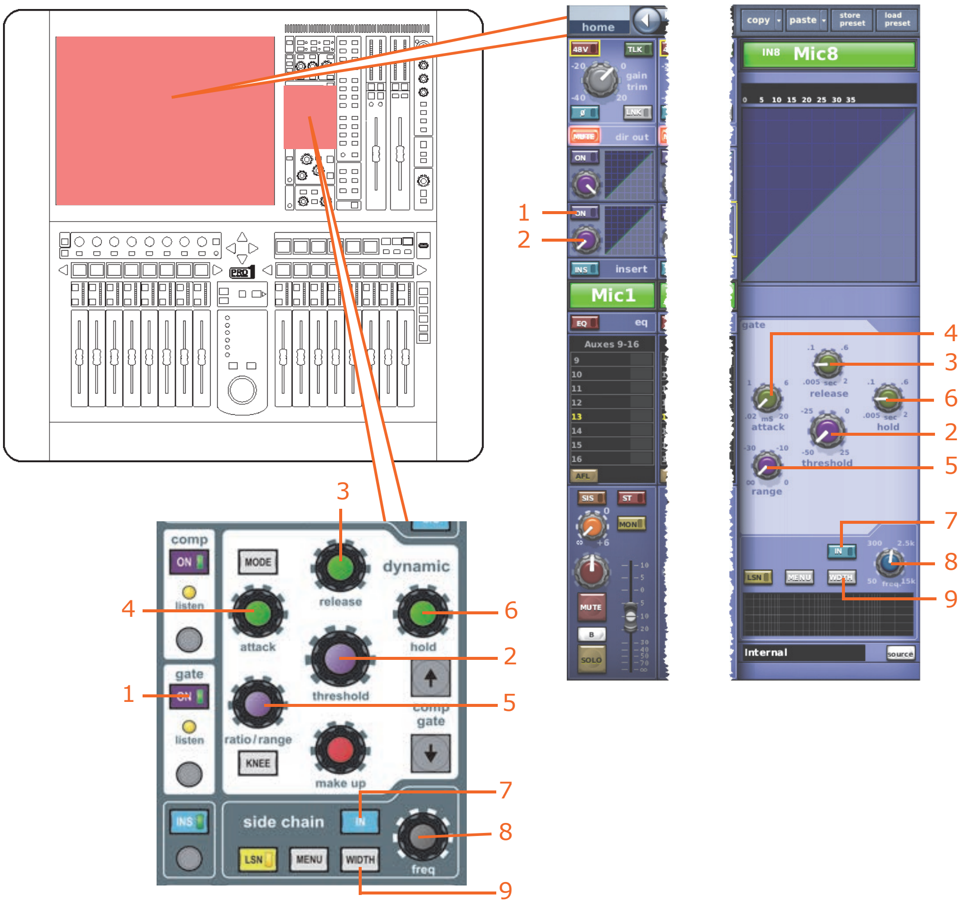
Note: *Only the corrective compressor is shown above, but this is typically the same for the other compressor modes (adaptive, creative and vintage).*

Item	***Control***	***Parameter***
1	**ON** switch	Compressor on/off
2	**threshold** control knob	Compressor threshold
3	**release** control knob	Compressor release
4	**make up** control knob	Compressor make up gain
5	**attack** control knob	Compressor attack
6	**ratio** control knob	Compressor ratio
7	**MODE** pushbutton	Compressor mode: corrective, adaptive, creative or vintage
8	**KNEE** pushbutton	Compressor knee: hard, medium or soft
9	**IN** switch	Compressor sidechain in/out

Item	Control	Parameter
10	**freq** control knob	Compressor sidechain frequency
11	**WIDTH** pushbutton	Compressor sidechain: 2 Oct, 1 Oct or 0.3 Oct

Gates

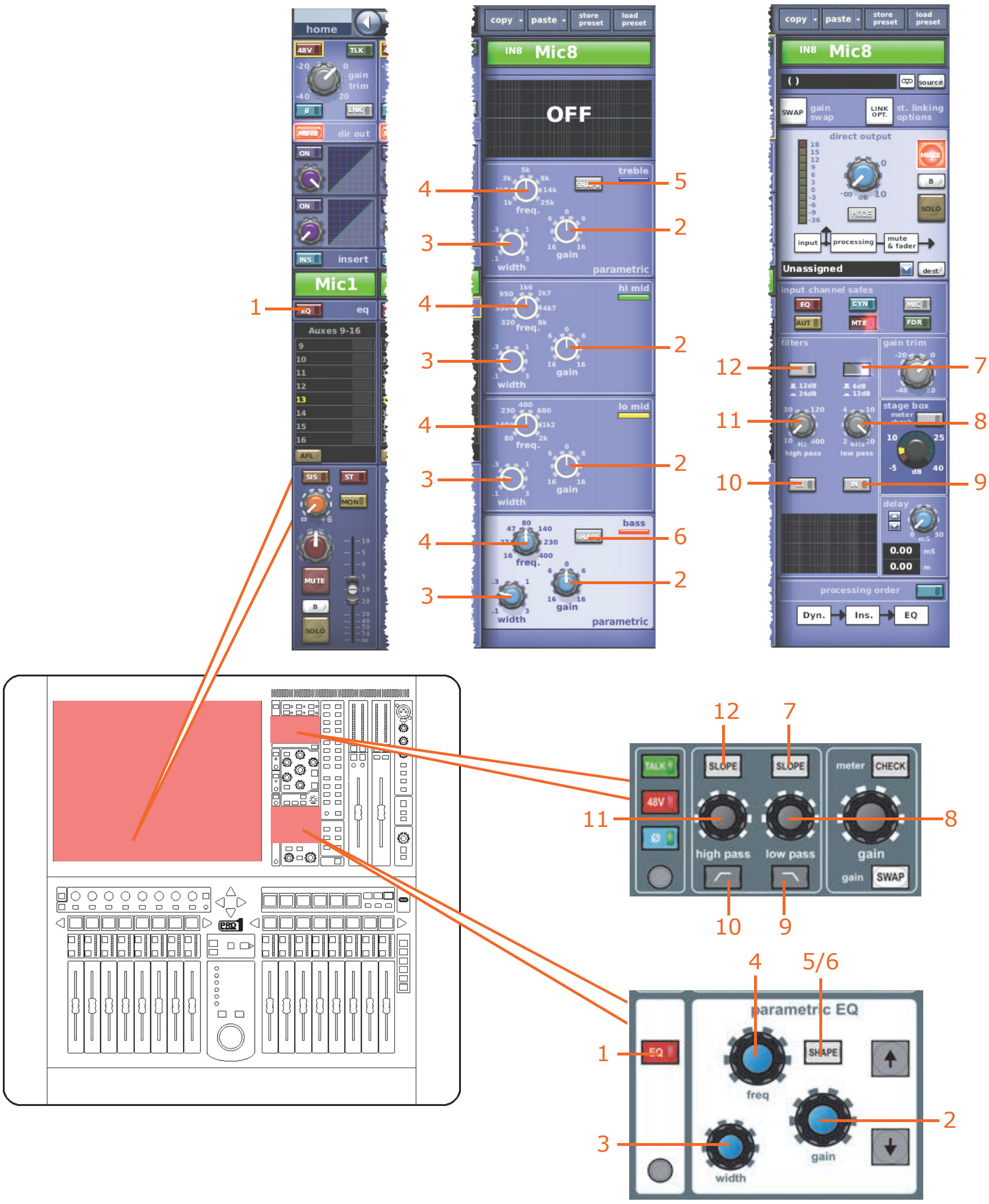
This section shows the gate processing area parameters copied through scenes.



Item	Control	Parameter
1	**ON** switch	Gate on/off
2	**threshold** control knob	Gate threshold
3	**release** control knob	Gate release
4	**attack** control knob	Gate attack
5	**range** control knob	Gate range
6	**hold** control knob	Gate hold
7	**IN** switch	Gate sidechain in/out
8	**freq** control knob	Gate sidechain frequency
9	**WIDTH** pushbutton	Gate sidechain width: 2 Oct, 1 Oct or 0.3 Oct

EQs

This section shows the EQ (parametric) processing area parameters copied through scenes.

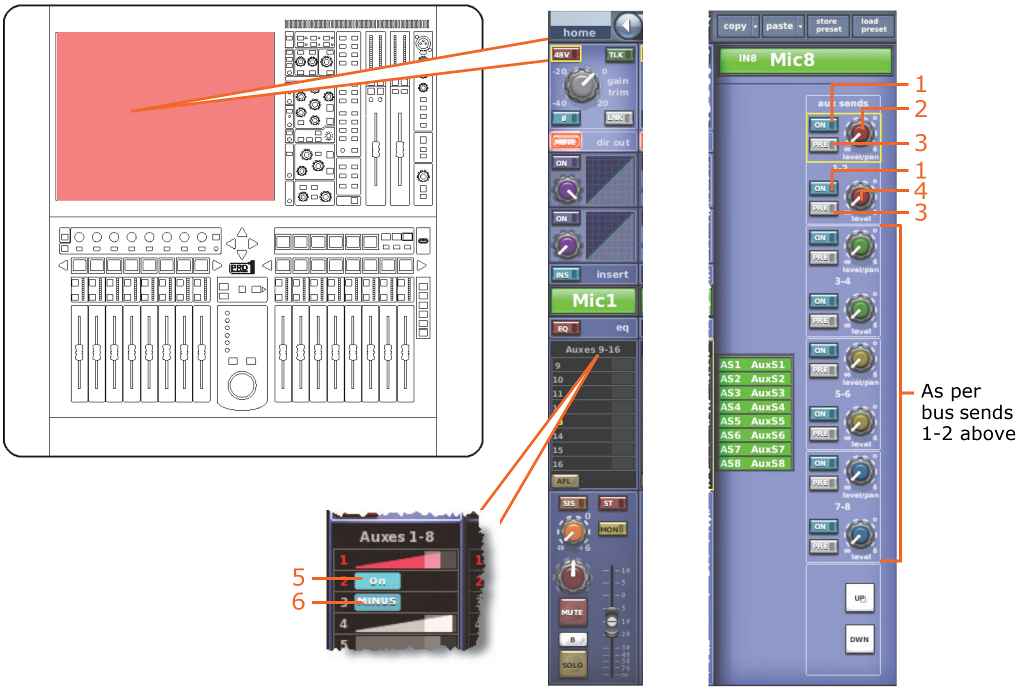


Item	*Control*	*Parameter*
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch	Treble: parametric, bright, classic or soft
6	**SHAPE** switch	Bass: parametric, deep, classic or warm

Item	Control	Parameter
7	**SLOPE** pushbutton	Low pass filter slope 6dB or 12dB
8	**low pass** control knob	Low pass filter frequency
9	/[**IN**] switch	Low pass filter in/out
10	/[**IN**] switch	High pass filter in/out
11	**high pass** control knob	High pass filter frequency
12	**SLOPE** pushbutton	High pass filter slope 12dB or 24dB

Aux Sends (1 to 16)

This section shows the aux send parameters copied through scenes.

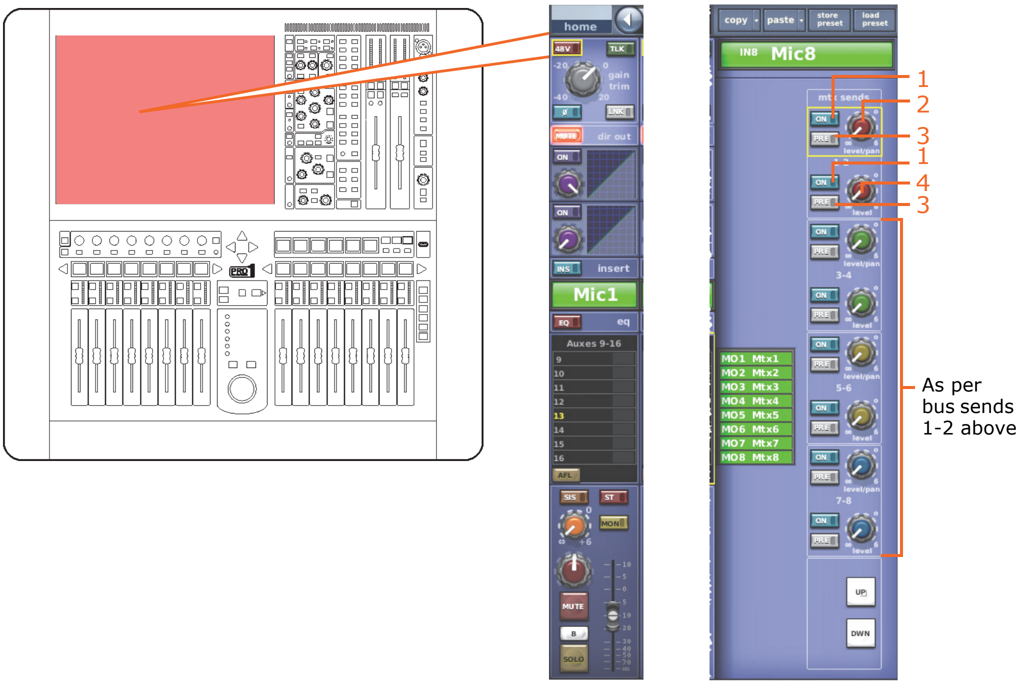


Note: *Only aux sends 1 to 8 are shown above, but this is typically the same for all of the 16 aux sends.*

Item	Control	Parameter
1	**ON** switch	Bus send on/off
2	**level/pan** control knob	Bus level, or pan when bus is linked
3	**PRE** switch	Pre-fader on/off
4	**level** control knob	Bus level
5	**On** switch	Aux bus send on/off — only available when aux bus is in group mode
6	**MINUS** switch	Aux bus send mute on/off — only available when aux bus is in mix minus mode

Matrix Sends (1 to 8)

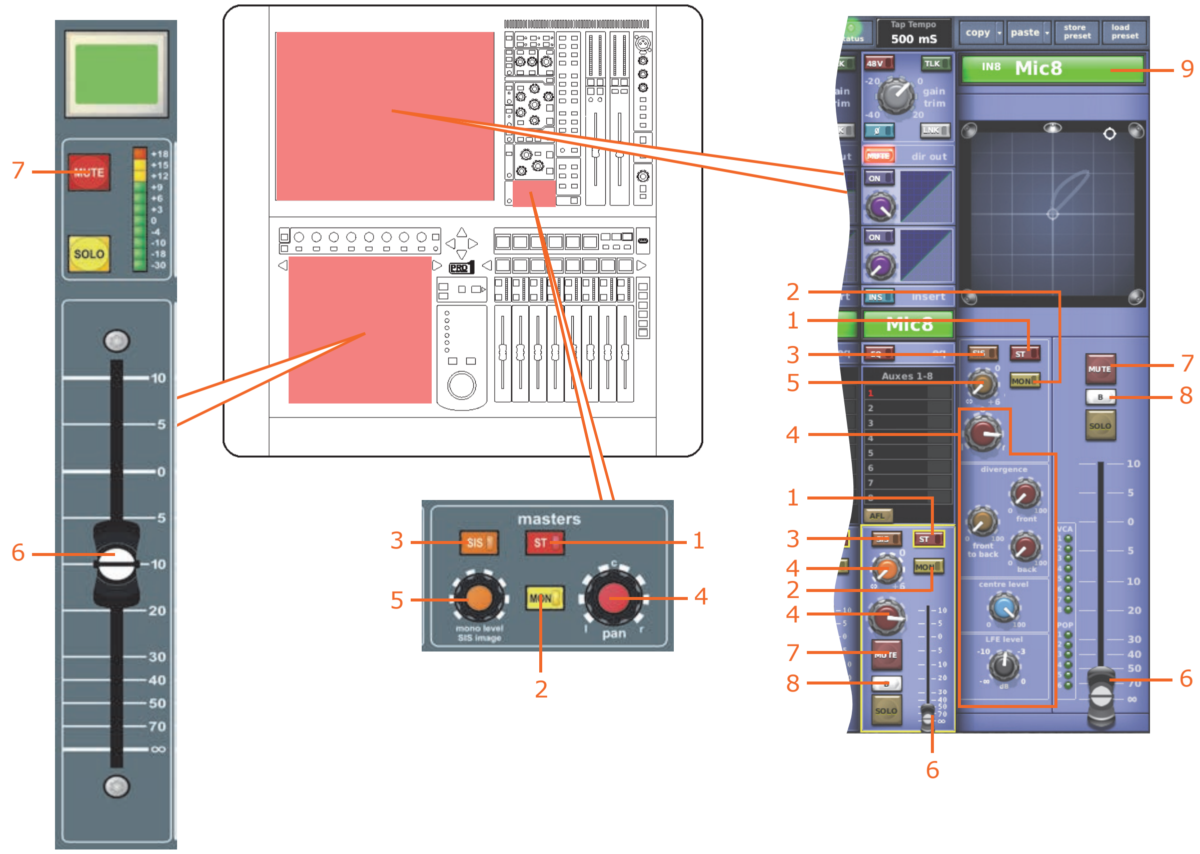
This section shows the matrix send parameters copied through scenes.



Item	Control	Parameter
1	**ON** switch	Bus send on/off
2	**level/pan** control knob	Bus level, or pan when bus is linked
3	**PRE** switch	Pre-fader on/off
4	**level** control knob	Bus level

Fader Sections

This section shows the master routing processing area parameters that are copied through scenes.



Item	Control	Parameter
1	**ST** switch	Stereo on/off
2	**MON** switch	Mono on/off
3	**SIS** switch	Spatial imaging system on/off
4	Panning control knobs	Surround panning (includes all surround sound parameters)
5	**mono level/SIS image** control knob	Mono level (SIS off) or SIS image (SIS on)
6	Fader	Level
7	**MUTE** switch	Mute on/off
8	**B** button	Solo B in/out
9	Field	Channel name

Recall Scope

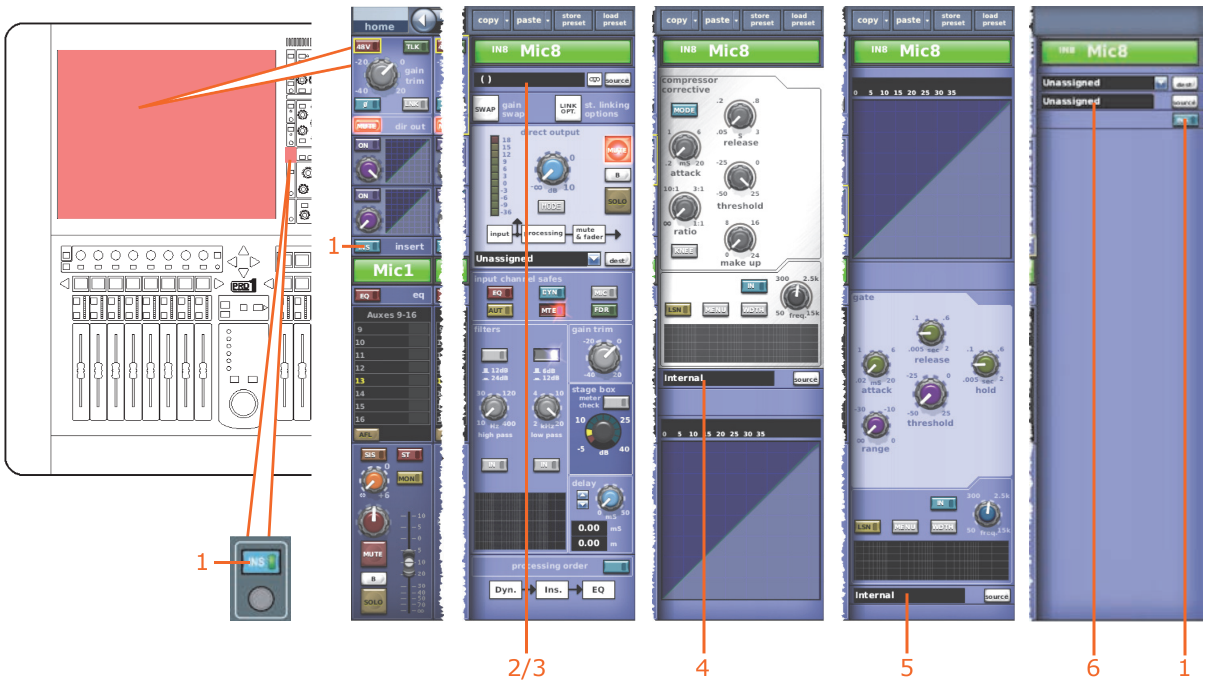
See “Inputs” on page 351.

Store Scope

See “Inputs” on page 351.

Routing

This section shows the routing parameters copied through scenes.



Item	*Control*	*Parameter*
1	**INS** switch	Insert in/out
2	Field	Primary input source
3	Field	Tape input source
4	Field	Compressor sidechain source*
5	Field	Gate key source*
6	Field	Insert return source*

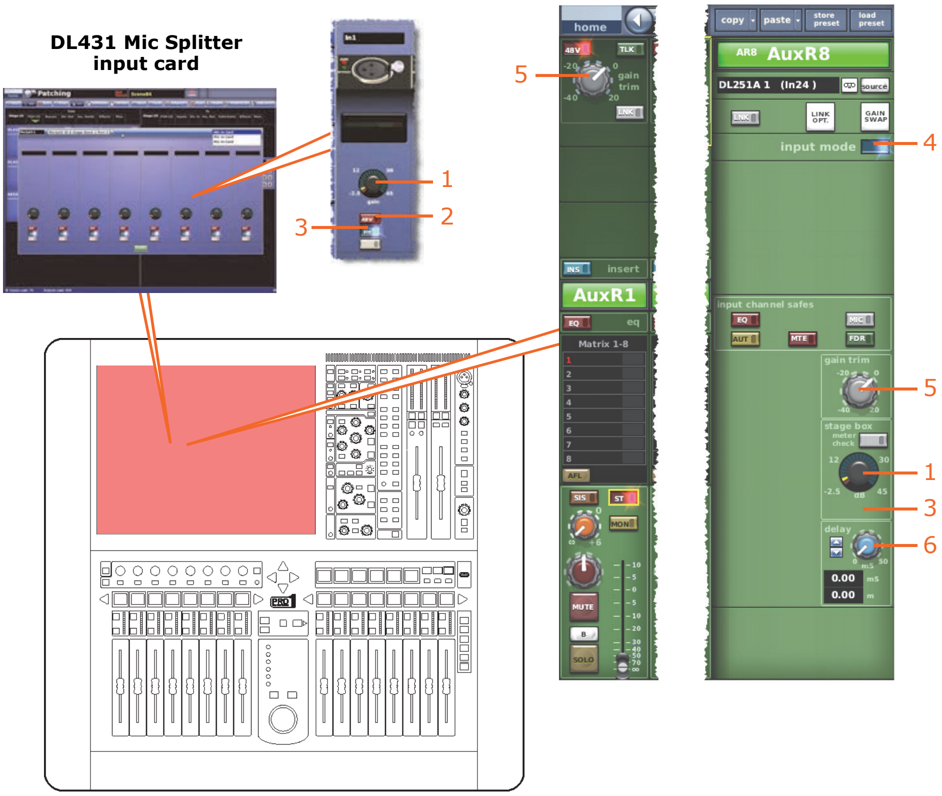
* Only automated when automate patching is on.

Aux Returns (return channels)

This section shows you which return channel parameters are affected by copy through scenes.

Config sections

This section shows the configuration processing area parameters copied through scenes.



Item	Control	Parameter
1	**stage box** control knob*	Remote amplifier level
2	**48V** switch**	48V phantom voltage on/off
3	Filter switch (not shown in GUI diagram above)**	30Hz filter in/out
4	**input zone** switch	Input zone in/out
5	**gain trim** control knob*	Digital input trim
6	**delay** control knob	Delay time

*   Depends on swap status.

** Only when sourced from a DL431 Mic Splitter.

Comp./Output Dyn

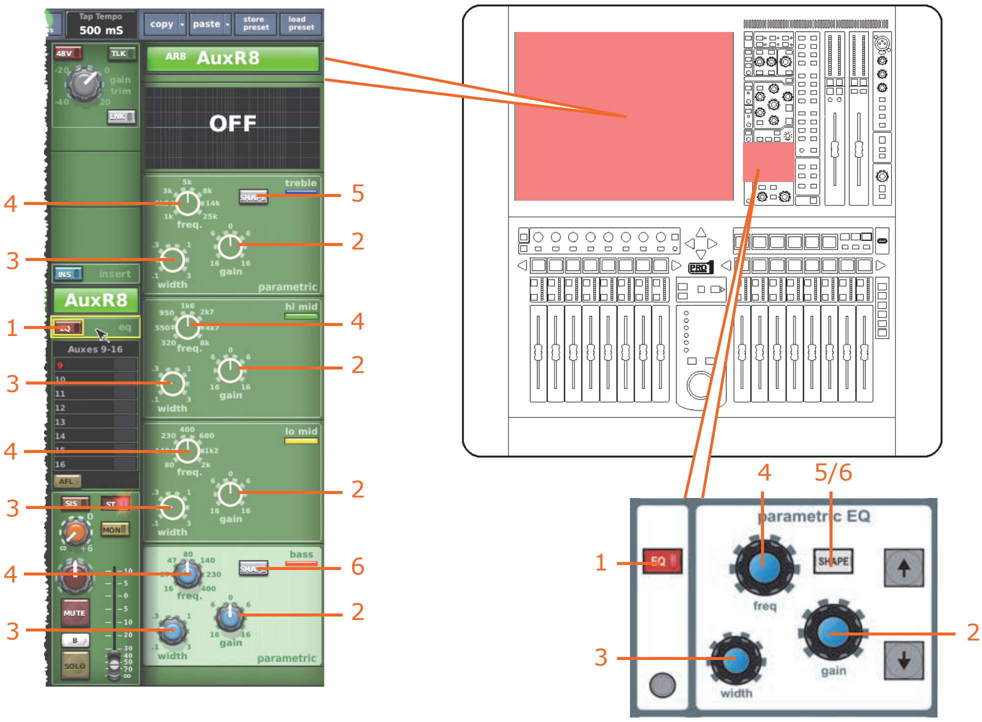
Not applicable.

Gates

Not applicable.

EQs

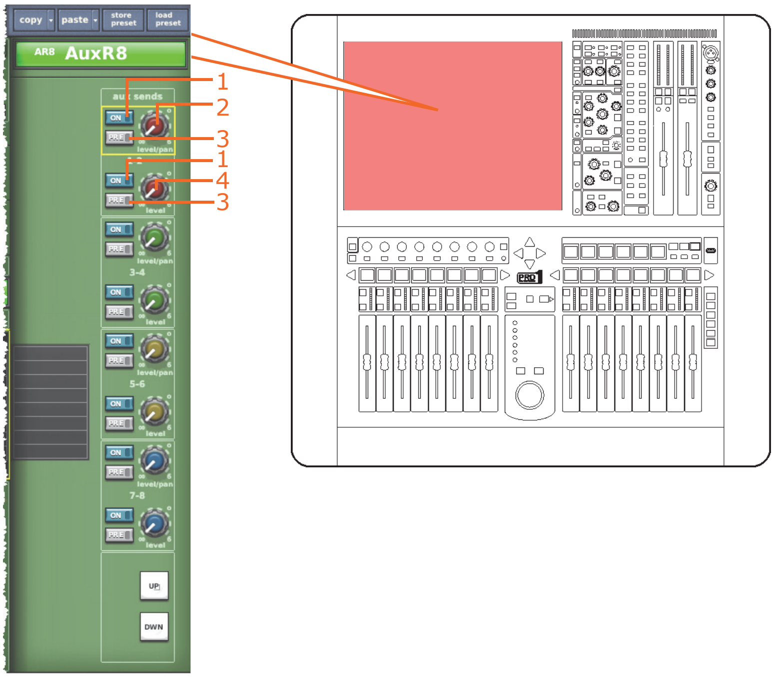
This section shows the EQ (parametric) processing area parameters copied through scenes.



Item	*Control*	*Parameter*
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch	Treble: parametric, bright, classic or soft
6	**SHAPE** switch	Bass: parametric, deep, classic or warm

Aux Sends (1 to 16)

This section shows the aux send parameters copied through scenes.

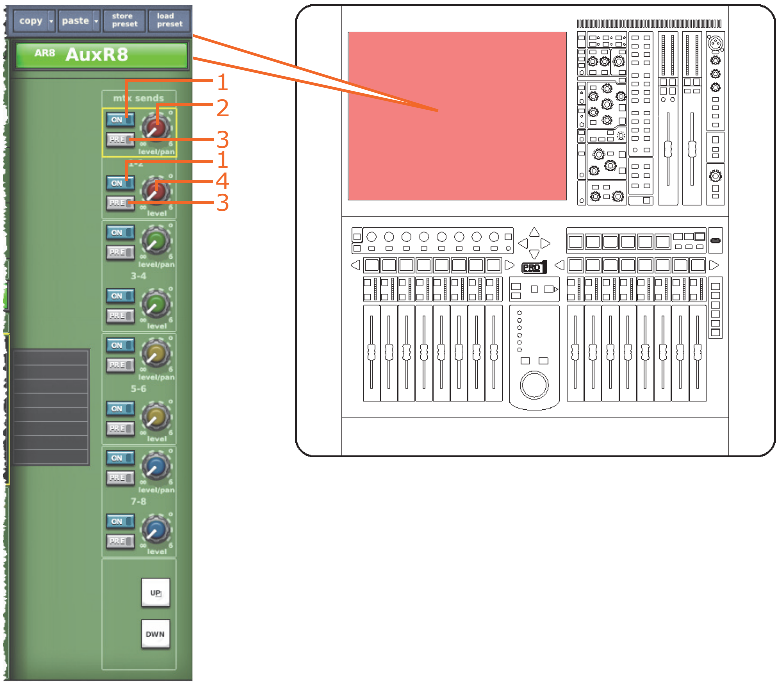


Note: *Although only aux sends 1-2 are referenced above, this also applies to all 16 aux sends.*

Item	***Control***	***Parameter***
1	**ON** switch	Matrix bus send on/off
2	**level/pan** control knob	Bus level, or pan when bus is linked
3	**PRE** switch	Pre-fader on/off
4	**level** control knob	Bus level

Matrix Sends (1 to 8)

This section shows the matrix send parameters copied through scenes.

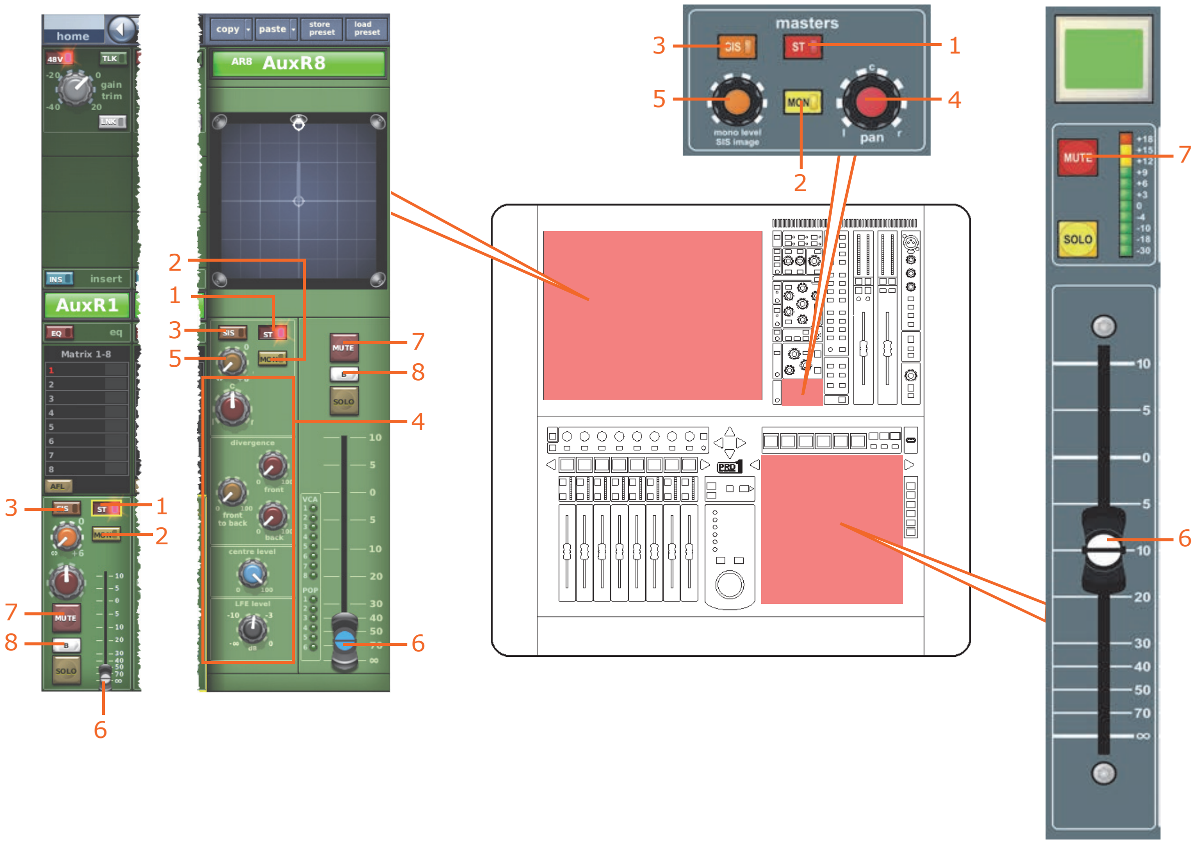


Note: *Although only matrix sends 1-2 are referenced above, this also applies to all eight matrix sends.*

Item	***Control***	***Parameter***
1	**ON** switch	Matrix bus send on/off
2	**level/pan** control knob	Bus level, or pan when bus is linked
3	**PRE** switch	Pre-fader on/off
4	**level** control knob	Bus level

Fader Sections

This section shows the fader and master routing parameters copied through scenes.



Item	Control	Parameter
1	**ST** switch	Stereo on/off
2	**MON** switch	Mono on/off
3	**SIS** switch	Spatial imaging system on/off
4	Panning control knobs	Surround panning (includes all surround sound parameters)
5	**mono level/SIS image** control knob	Mono level (SIS off) or SIS image (SIS on)
6	Fader	Level
7	**MUTE** switch	Mute on/off
8	**B** switch	Solo B in/out

Recall Scope

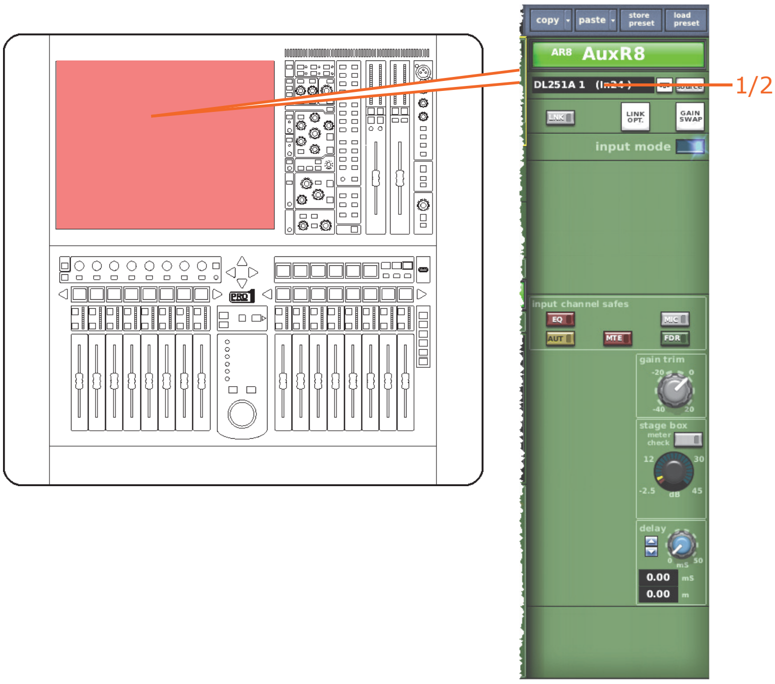
For details, see "Returns (Aux Returns)" on page 362.

Store Scope

For details, see "Returns (Aux Returns)" on page 362.

Routing

This section shows the routing parameters copied through scenes.



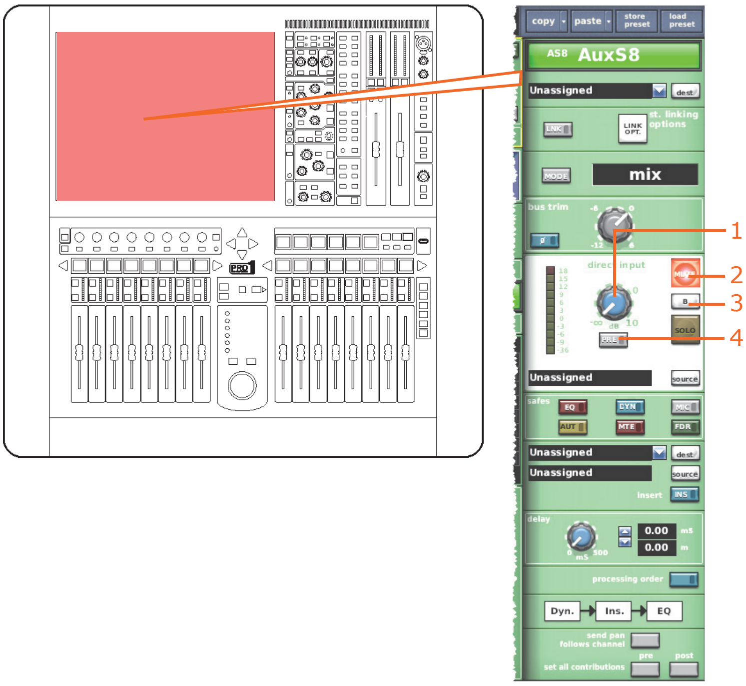
Item	Control	Parameter
1	Field	Primary input source
2	Field	Tape input source

Aux Sends (aux channels)

This section shows you which aux send parameters are affected by copy through scenes.

Config sections

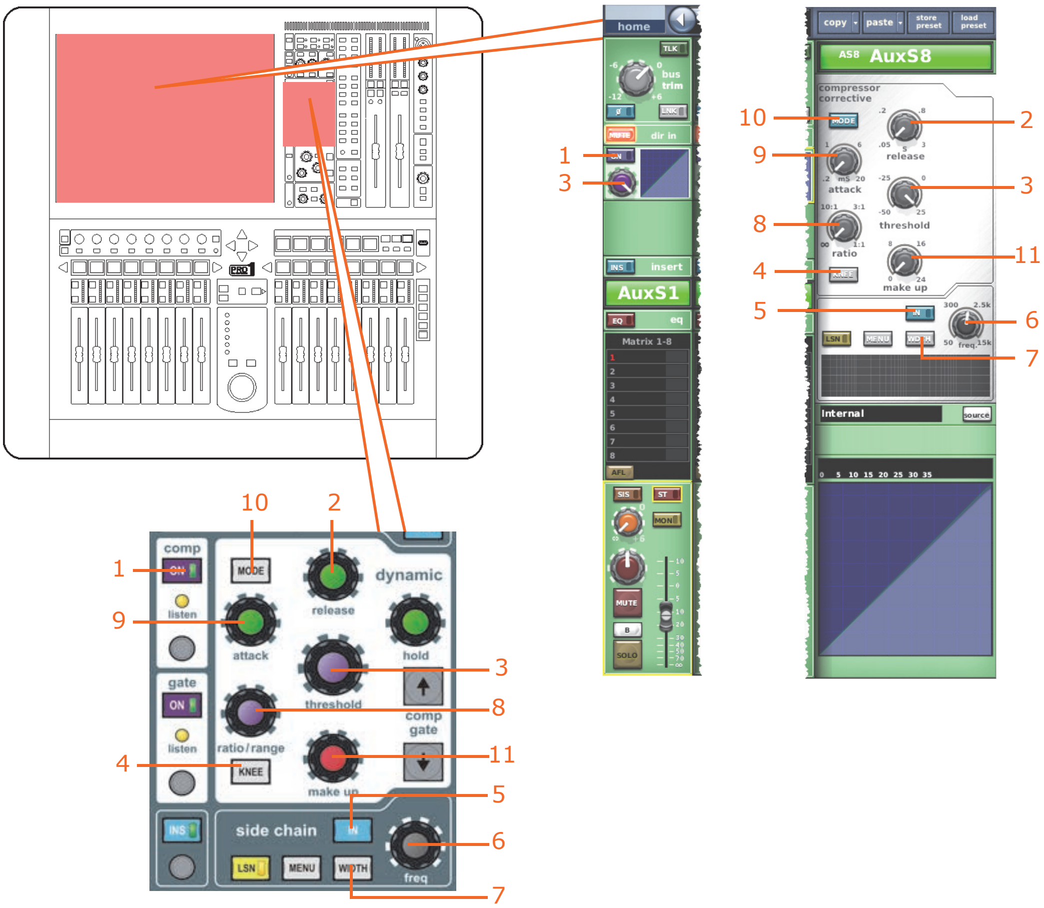
This section shows the configuration parameters copied through scenes.



Item	*Control*	*Parameter*
1	Level control knob	Direct input level
2	**MUTE** switch	Direct input mute on/off
3	**B** switch	Direct input solo B in/out
4	**PRE** switch	Direct input pre- in/out

Comp./Output Dyn

This section shows the compressor parameters copied through scenes. Only corrective compressor shown below, but typically the same for the other compressor modes.



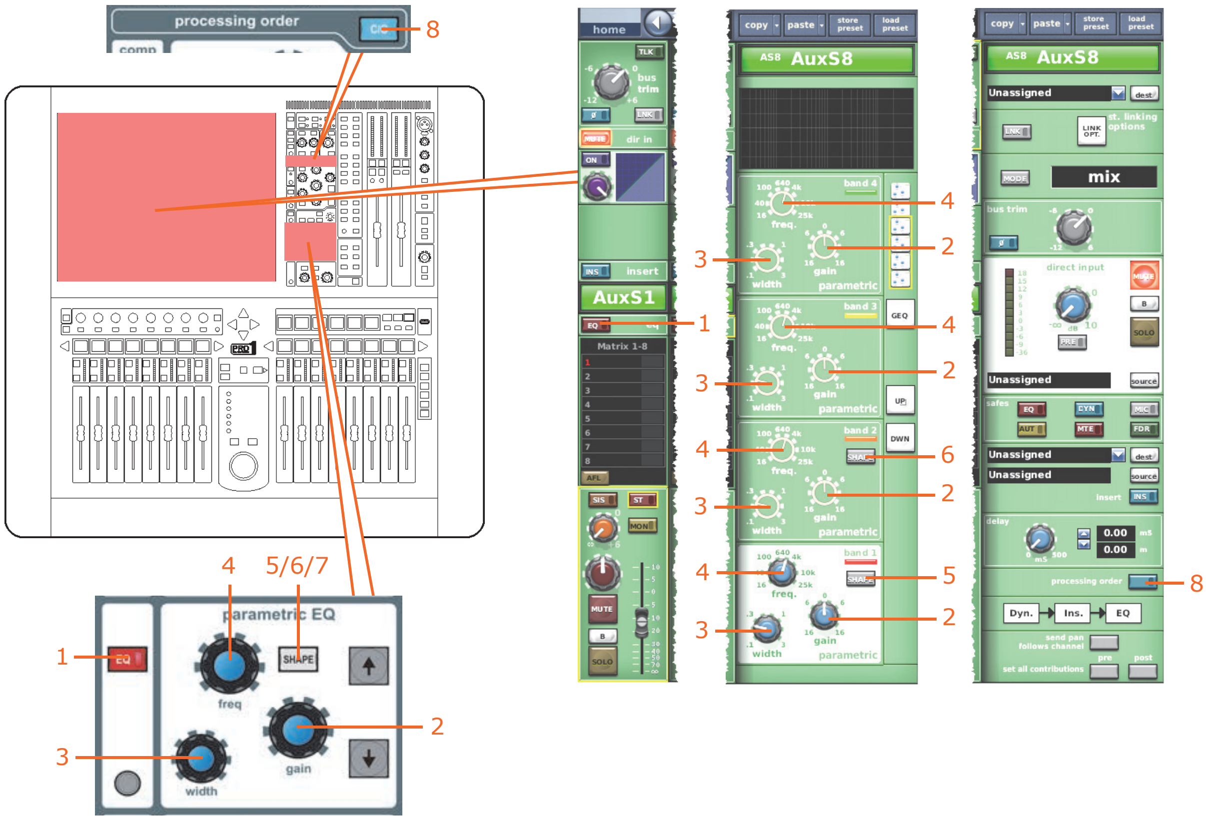
Item	*Control*	*Parameter*
1	**ON** switch	Compressor on/off
2	**release** control knob	Compressor release
3	**threshold** control knob	Compressor threshold
4	**KNEE** pushbutton	Compressor knee: hard, medium or soft
5	**IN** switch	Compressor sidechain in/out
6	**freq** control knob	Compressor sidechain frequency
7	**WIDTH** pushbutton	Compressor sidechain width: 2 Oct, 1 Oct or 0.3 Oct
8	**ratio** control knob	Compressor ratio
9	**attack** control knob	Compressor attack
10	**MODE** pushbutton	Compressor mode: corrective, adaptive, creative, vintage or shimmer
11	**make up** control knob	Compressor gain

Gates

Not applicable.

EQs

This section shows the EQ parameters copied through scenes.



Item	Control	Parameter
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch	Band 1 shelving mode: parametric, warm, high pass 6dB or high pass 12dB. (This band is not available if band 2 is selected as high pass 24dB.)
6	**SHAPE** switch	Band 2 shelving mode: parametric or high pass 24dB
7	**SHAPE** switch (not shown on GUI above)	Band 6 shelving mode: parametric, soft, low pass 6dB or low pass 12dB
8	**C/O** switch	Order of processing: **Dyn.→Ins.→EQ** or **EQ→Ins.→Dyn.**

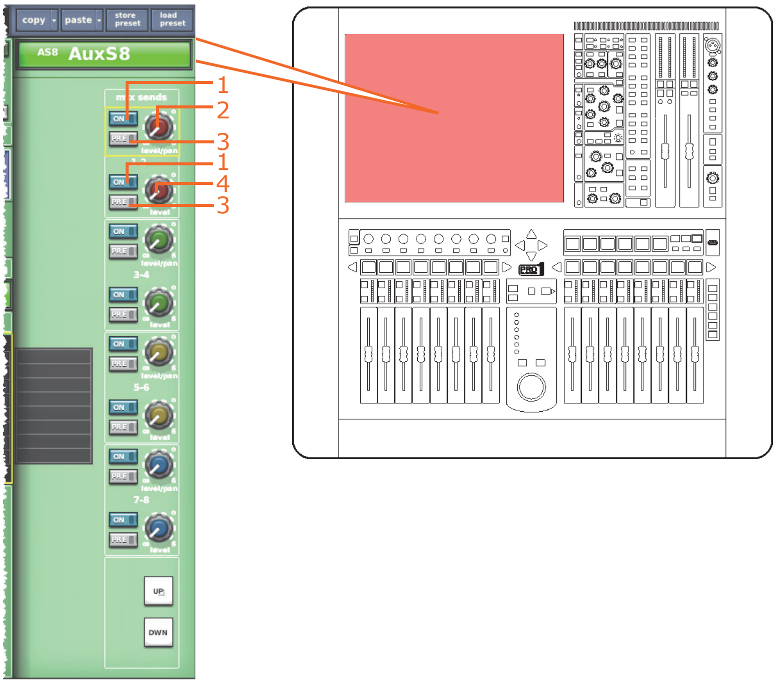
***Note:** Although bands 5 and 6 are not shown above, the items in the table also apply. Both bands have items 2, 3 and 4, and band 6 also has item 7.*

Aux Sends (1 to 16)

Not applicable.

Matrix Sends (1 to 8)

This section shows the matrix send parameters copied through scenes.

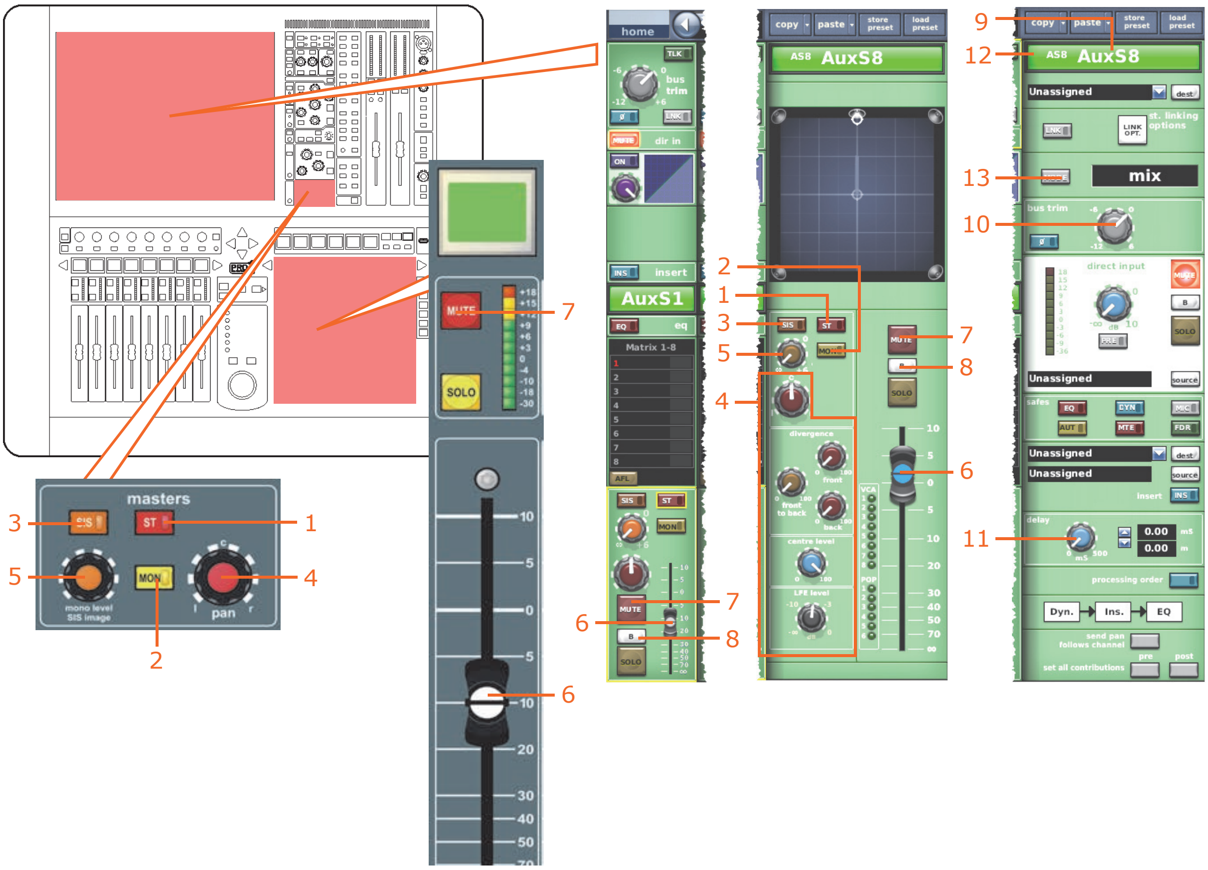


***Note:** Although only matrix sends 1-2 are referenced above, this also applies to all eight matrix sends.*

Item	***Control***	***Parameter***
1	**ON** switch	Matrix bus send on/off
2	**level/pan** control knob	Bus level, or pan when bus is linked
3	**PRE** switch	Pre-fader on/off
4	**level** control knob	Bus level

Fader Sections

This section shows the fader and master routing parameters copied through scenes.



Item	*Control*	*Parameter*
1	**ST** switch	Stereo on/off
2	**MON** switch	Mono on/off
3	**SIS** switch	Spatial imaging system on/off
4	Panning control knobs	Surround panning (includes all surround sound parameters)
5	**mono level/SIS image** control knob	Mono level (SIS off) or SIS image (SIS on)
6	Fader	Level
7	**MUTE** switch	Mute on/off
8	**B** switch	Solo B in/out
9	Field	Channel name
10	**bus trim** control knob	Bus trim level
11	**delay** control knob	Delay time
12	Field	Channel colour
13	**MODE** button	Bus mode

Recall Scope

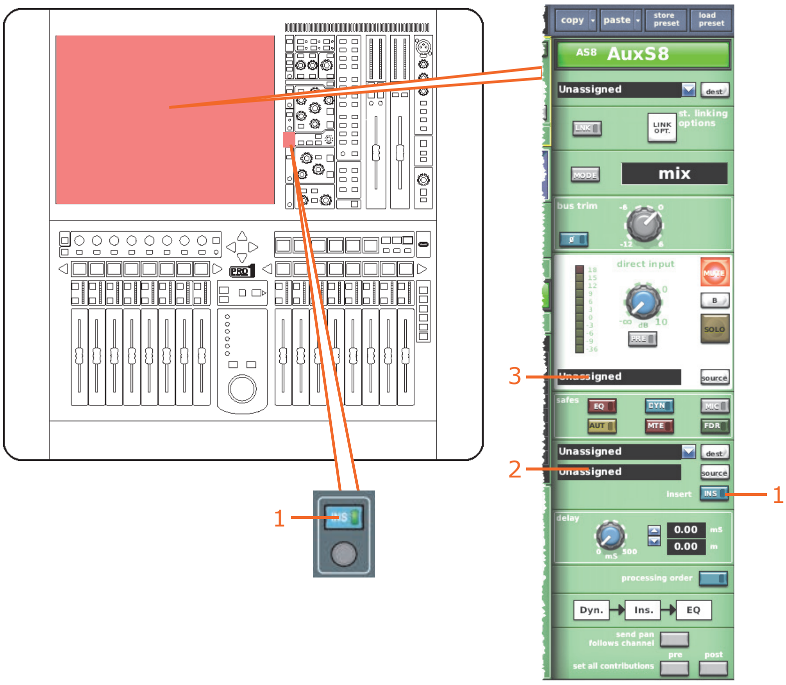
For details, see "Auxes (Aux Sends)" on page 368.

Store Scope

For details, see "Auxes (Aux Sends)" on page 368.

Routing

This section shows the routing parameters copied through scenes.



Item	Control	Parameter
1	**INS** switch	Insert in/out
2	Field	Insert return source*
3	Field	Direct input source*

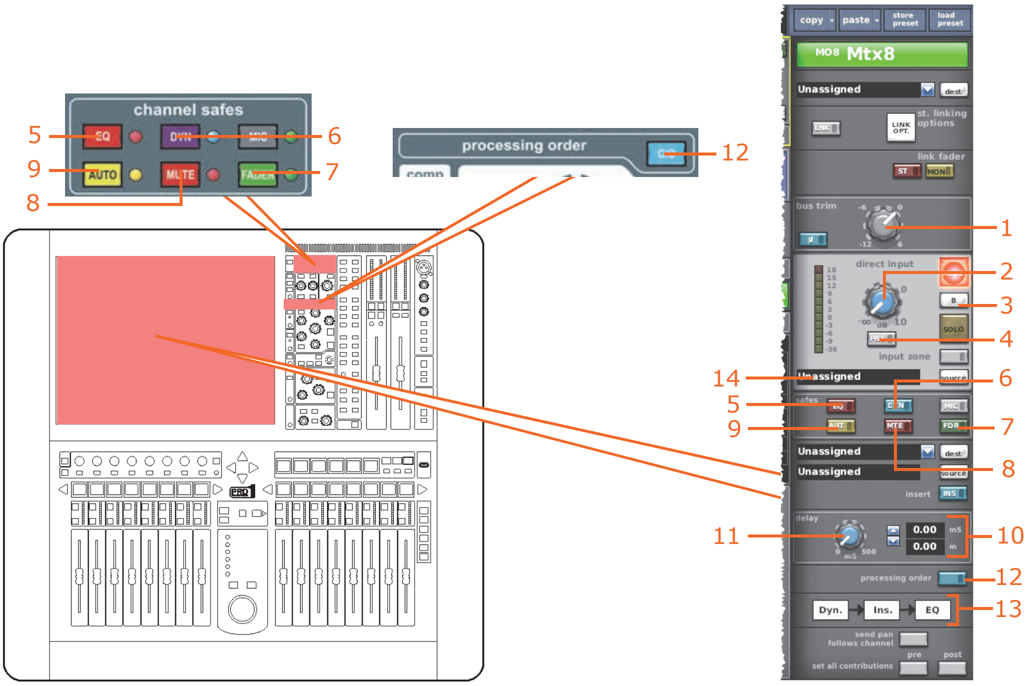
* Only automated when automate patching is on.

Matrix (matrix channels)

This section shows you which matrix channel parameters are affected by copy through scenes.

Config sections

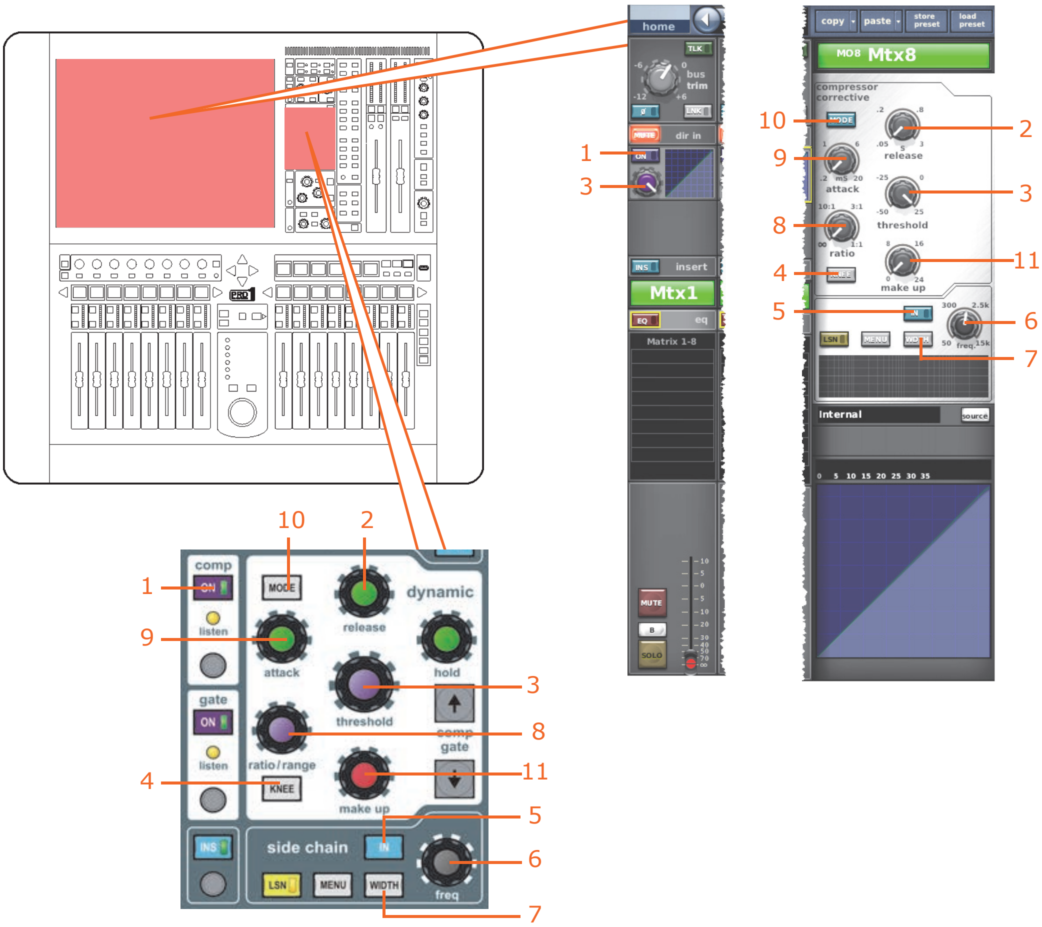
This section shows the configuration parameters copied through scenes.



Item	Control	Parameter
1	**bus trim** control knob	Bus trim level
2	Level control knob	Direct input level
3	**B** switch	Direct input solo B on/off
4	**PRE** switch	Direct input pre- in/out
5	**EQ** pushbutton	EQ safe on/off
6	**DYN** pushbutton	Dynamic safe on/off
7	**FADER**/[**FDR**] switch	Fader safe on/off
8	**MUTE**/[**MTE**] switch	Mute safe on/off
9	**AUTO**/[**AUT**] switch	Auto safe on/off
10	Delay field	Delay in milliseconds (ms) and metres (m)
11	**delay** control knob	Delay level
12	**C/O** switch	Order of processing: **Dyn.**→**Ins.**→**EQ** or **EQ**→**Ins.**→**Dyn.**
13	Graphic	Order of processing
14	Field	Direct input source

Comp./Output Dyn

This section shows the compressor parameters copied through scenes. Only corrective compressor shown below, but typically the same for the other compressor modes.



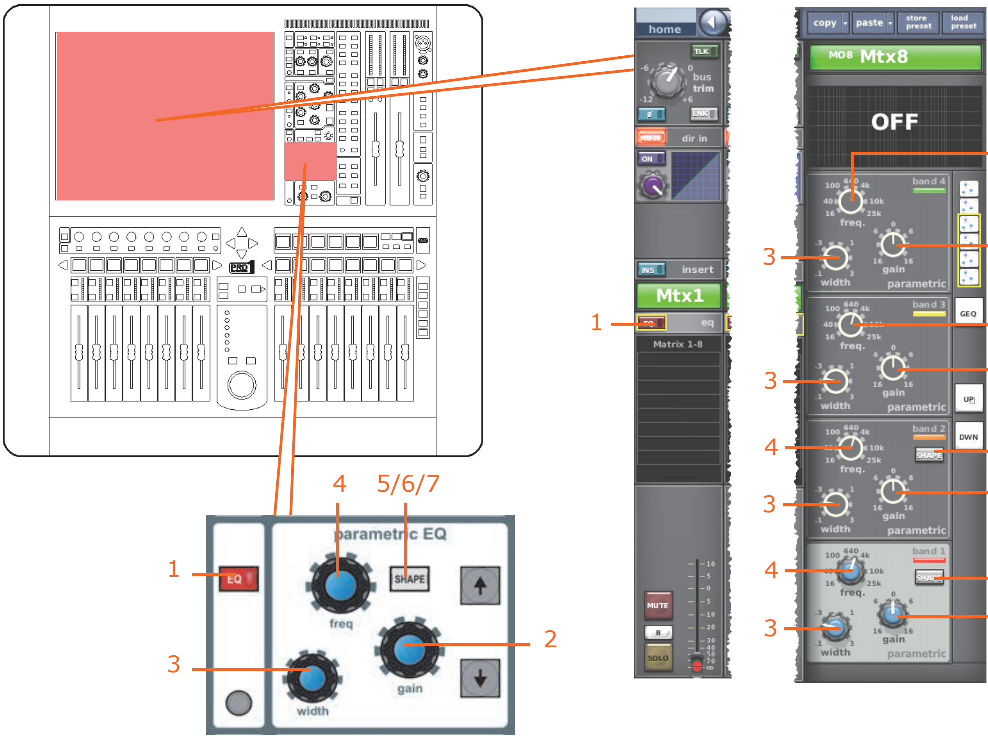
Item	*Control*	*Parameter*
1	**ON** switch	Compressor on/off
2	**release** control knob	Compressor release
3	**threshold** knob	Compressor threshold
4	**KNEE** pushbutton	Compressor knee: hard, medium or soft
5	**IN** switch	Compressor sidechain in/out
6	**freq** control knob	Compressor sidechain frequency
7	**WIDTH** pushbutton	Compressor sidechain width: 2 Oct, 1 Oct or 0.3 Oct
8	**ratio** control knob	Compressor ratio
9	**attack** control knob	Compressor attack
10	**MODE** pushbutton	Compressor mode: corrective, adaptive, creative, vintage or shimmer
11	**make up** control knob	Compressor gain

Gates

Not applicable.

EQs

This section shows the EQ parameters copied through scenes.



Item	Control	Parameter
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch (not shown on GUI above)	Band 6 shelving mode: parametric, soft, low pass 6dB or low pass 12dB
6	**SHAPE** switch	Band 2 shelving mode: parametric or high pass 24dB
7	**SHAPE** switch	Band 1 shelving mode: parametric, warm, high pass 6dB or high pass 12dB. (This band is not available if band is selected as high pass 24dB.)

Note: *Although bands 5 and 6 are not shown above, the items in the table also apply. Both bands have items 2, 3 and 4, and band 6 also has item 5.*

Aux Sends (1 to 16)

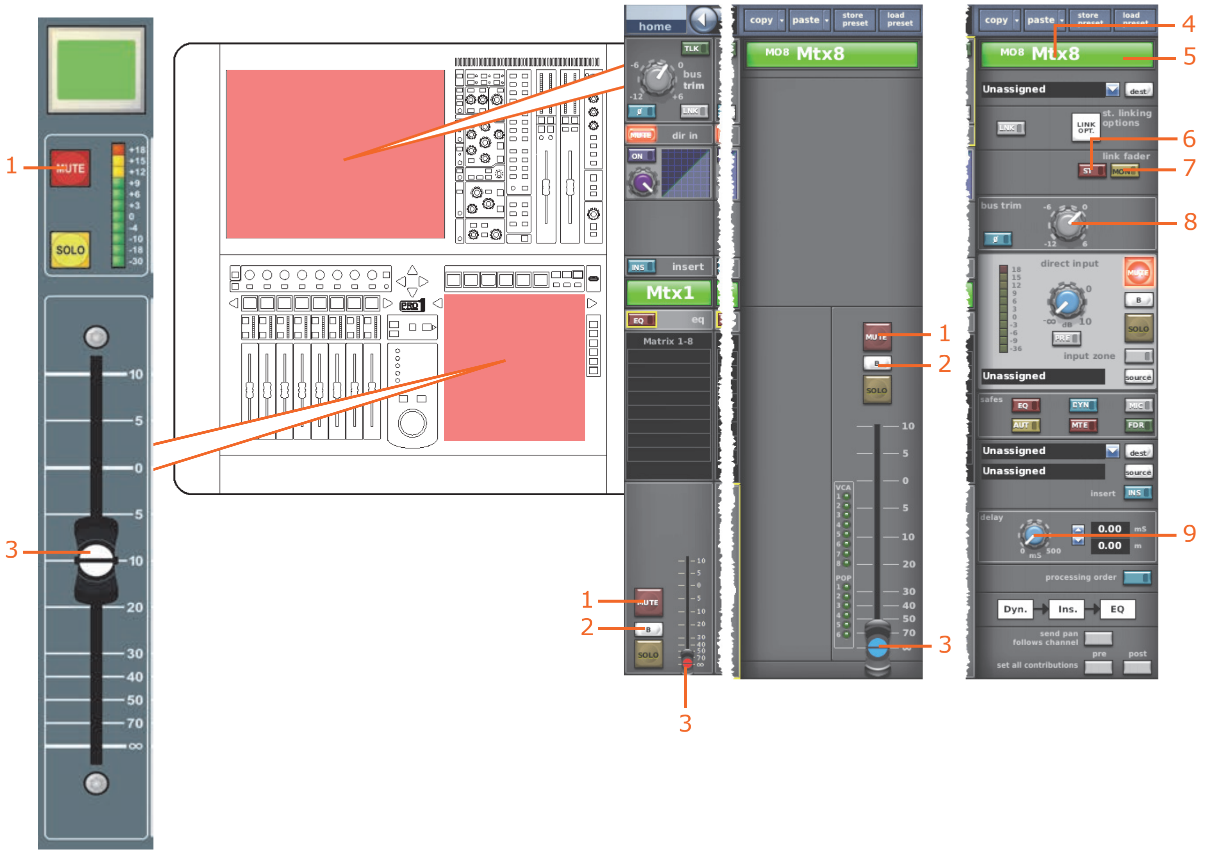
Not applicable.

Matrix Sends (1 to 8)

Not applicable.

Fader Sections

This section shows the fader parameters copied through scenes.



Item	*Control*	*Parameter*
1	**MUTE** switch	Mute on/off
2	**B** switch	Solo B in/out
3	Fader	Level
4	Field	Channel name
5	Field	Channel colour
6	**ST** switch	Link to stereo master fader
7	**MON** switch	Link to mono master fader
8	**bus trim** control knob	Bus trim level
9	**delay** control knob	Delay time

Recall Scope

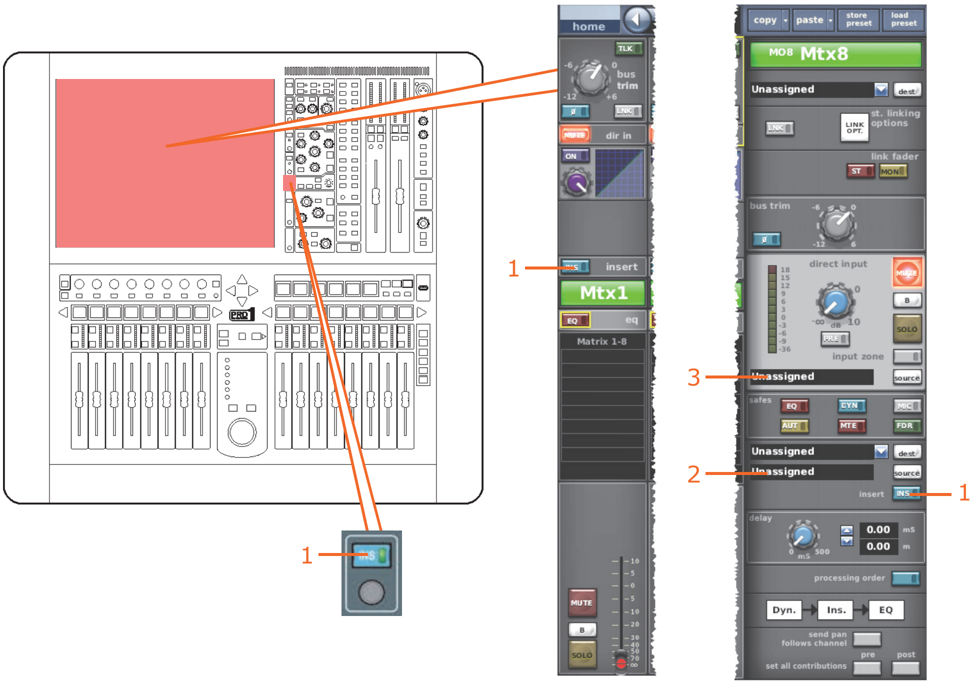
For details, see "Matrices" on page 375.

Store Scope

For details, see "Matrices" on page 375.

Routing

This section shows the routing parameters copied through scenes.



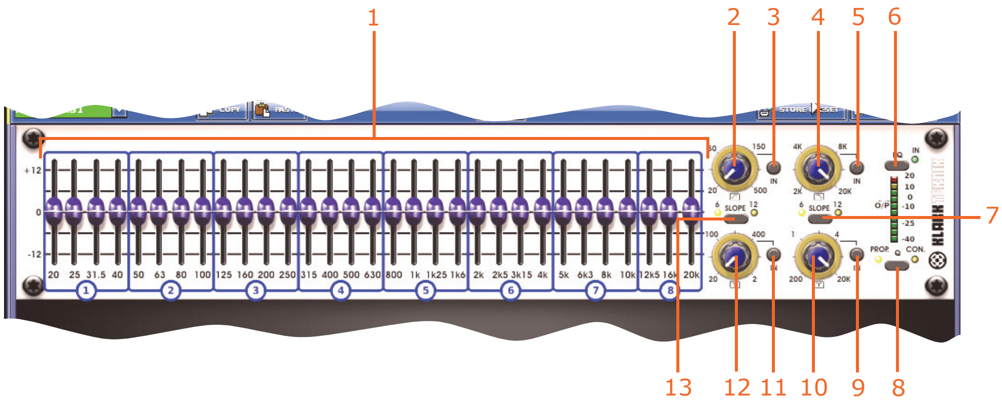
Item	Control	Parameter
1	**INS** switch	Insert in/out
2	Field	Insert return source*
3	Field	Direct input source*

* Only automated when automate patching is on.

GEQs

You can copy the control settings of any of the GEQs through scenes.

Only the **Recall Scope** and **Store Scope** options in the **Sections** area of the **Show Editor** screen are applicable to this option.



Item	*Control*	*Parameter*
1	31 faders	Fader positions
2	High pass filter control knob	High pass filter cut off frequency
3	**IN** switch	High pass filter in/out
4	Low pass filter control knob	Low pass filter cut off frequency
5	**IN** switch	Low pass filter in/out
6	**EQ** switch	EQ in/out
7	**SLOPE** switch	Low pass filter: 6dB or 12dB
8	**Q** switch	Q mode as proportional (**PROP.**) or constant (**CON.**)
9	**IN** switch	200Hz - 20kHz notch filter in/out
10	Notch filter control knob	200Hz - 20kHz notch filter frequency
11	**IN** switch	20Hz - 2kHz notch filter in/out
12	Notch filter control knob	20Hz - 2kHz notch filter frequency
13	**SLOPE** switch	High pass filter: 6dB or 12dB

Effects

With only the internal effects unit(s) selected, the unit type and control settings of the desired effect(s) are copied through scenes. In addition, if the **Routing** option (**Sections** area of the **Show Editor** screen) is selected, the input patching is also copied through scenes.

VCA/POPulation (groups)

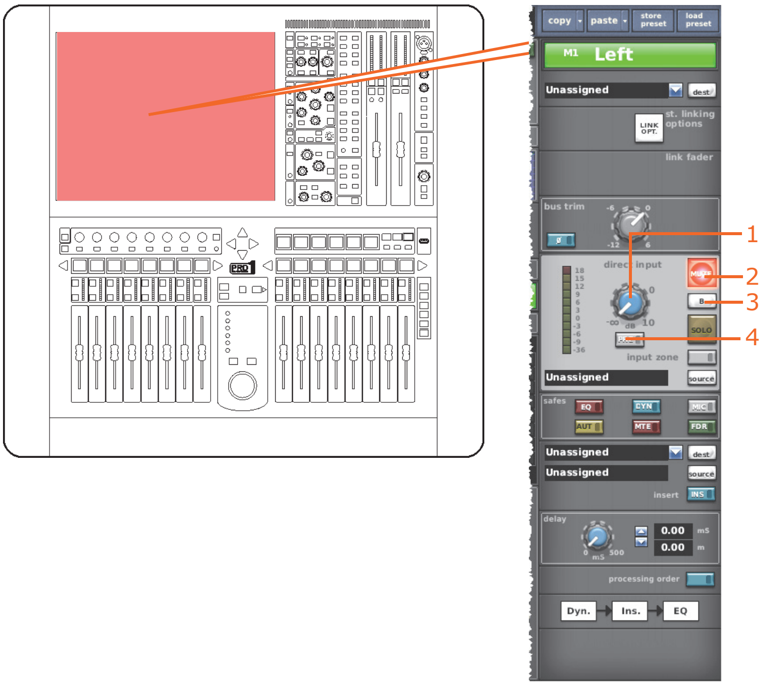
You can copy the group membership allocation of the VCA and Groups through scenes. However, none of the options in the **Sections** area are applicable to this option.

Masters (master channels)

This section shows you which parameters for each of the three master channels (mono and stereo left and stereo right) are affected by copy through scenes.

Config sections

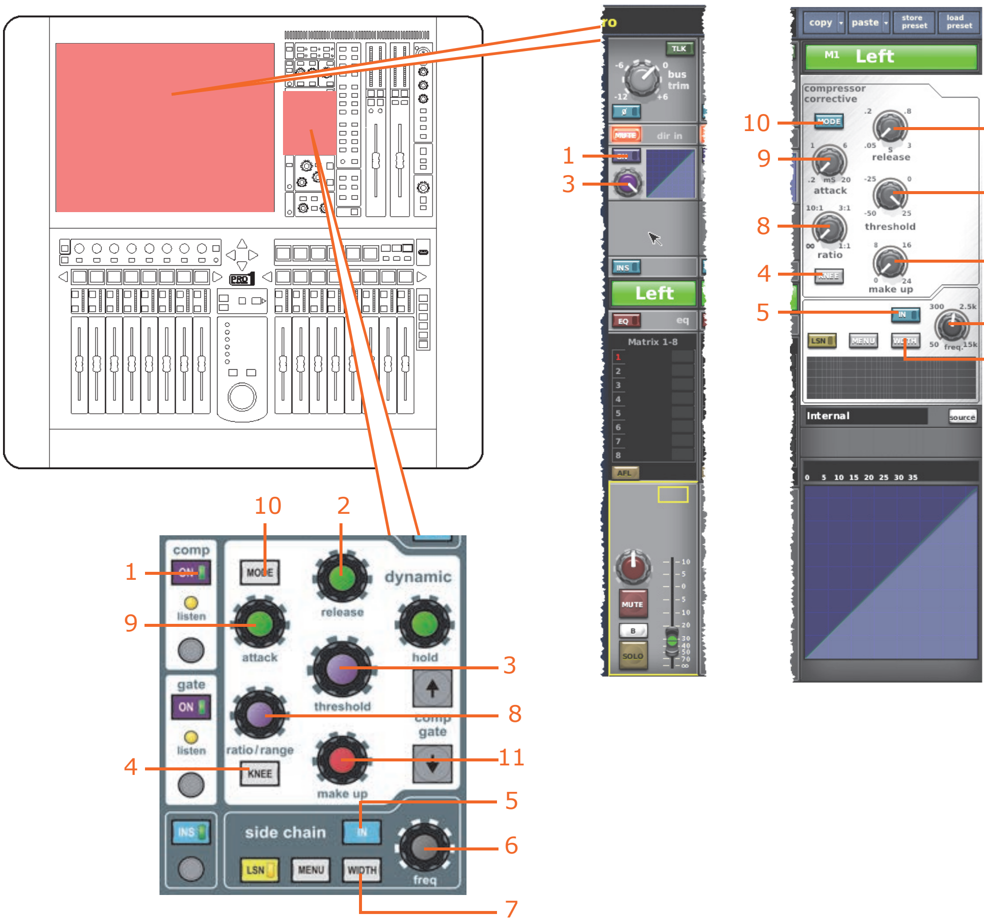
This section shows the configuration parameters copied through scenes.



Item	Control	Parameter
1	Level control knob	Direct input level
2	**MUTE** switch	Direct input mute on/off
3	**B** switch	Direct input solo B on/off
4	**PRE** switch	Direct input pre- in/out

Comp./Output Dyn

This section shows the compressor parameters copied through scenes. Only corrective compressor shown below, but typically the same for the other compressor modes.



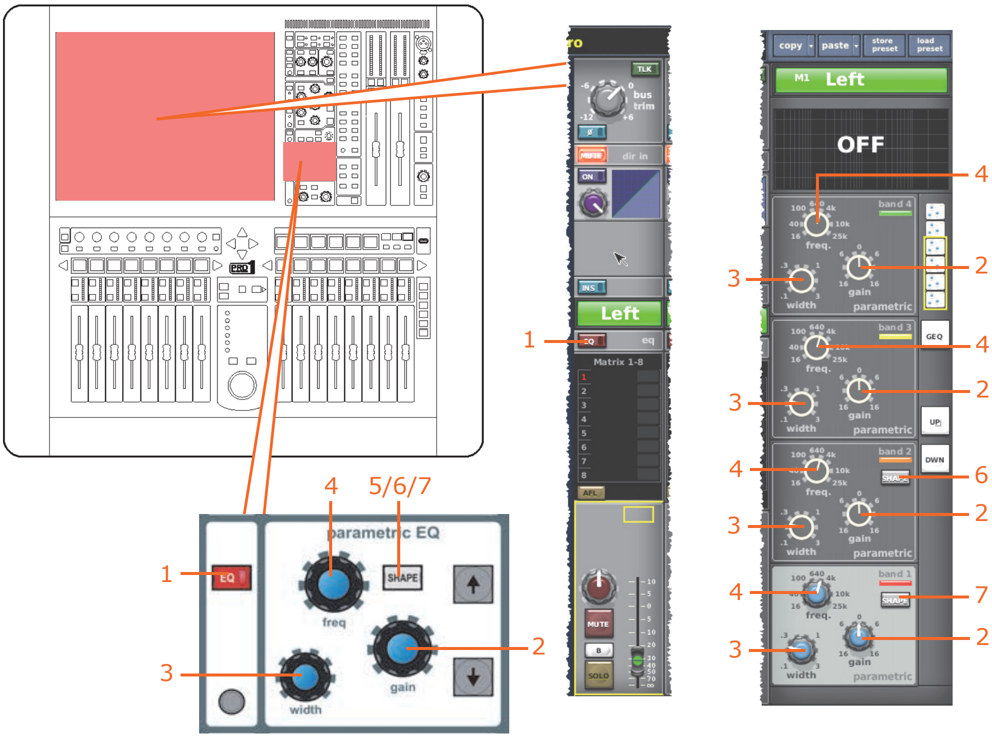
Item	*Control*	*Parameter*
1	**ON** switch	Compressor on/off
2	**release** control knob	Compressor release
3	**threshold** control knob	Compressor threshold
4	**KNEE** pushbutton	Compressor knee: hard, medium or soft
5	**IN** switch	Compressor sidechain in/out
6	**freq** control knob	Compressor sidechain frequency
7	**WIDTH** pushbutton	Compressor sidechain width: 2 Oct, 1 Oct or 0.3 Oct
8	**ratio** control knob	Compressor ratio
9	**attack** control knob	Compressor attack
10	**MODE** pushbutton	Compressor mode: corrective, adaptive, creative, vintage or shimmer
11	**make up** control knob	Compressor gain

Gates

Not applicable.

EQs

This section shows the EQ parameters copied through scenes.



Item	Control	Parameter
1	**EQ** switch	EQ on/off
2	**gain** control knob	EQ gain level
3	**width** control knob	EQ width
4	**freq** control knob	EQ frequency
5	**SHAPE** switch (not shown on GUI above)	Band 6 shelving mode: bell, soft, low pass 6dB or low pass 12dB
6	**SHAPE** switch	Band 2 shelving mode: bell or high pass 24dB
7	**SHAPE** switch	Band 1 shelving mode: bell, warm, high pass 6dB or high pass 12dB

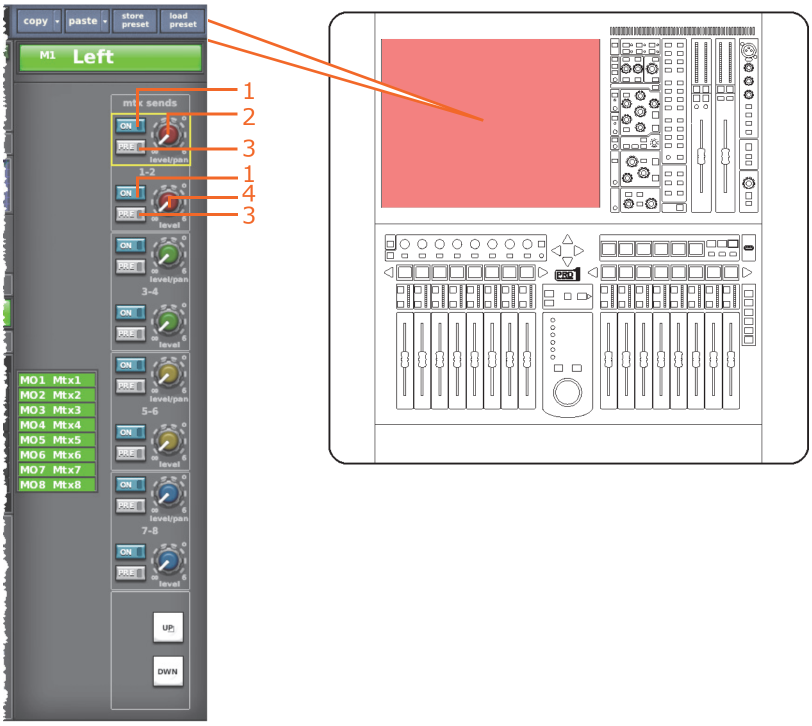
Note: *Although bands 5 and 6 are not shown above, the items in the table also apply. Both bands have items 2, 3 and 4, and band 6 also has item 7.*

Aux Sends (1 to 16)

Not applicable.

Matrix Sends (1 to 8)

This section shows the matrix send parameters copied through scenes.

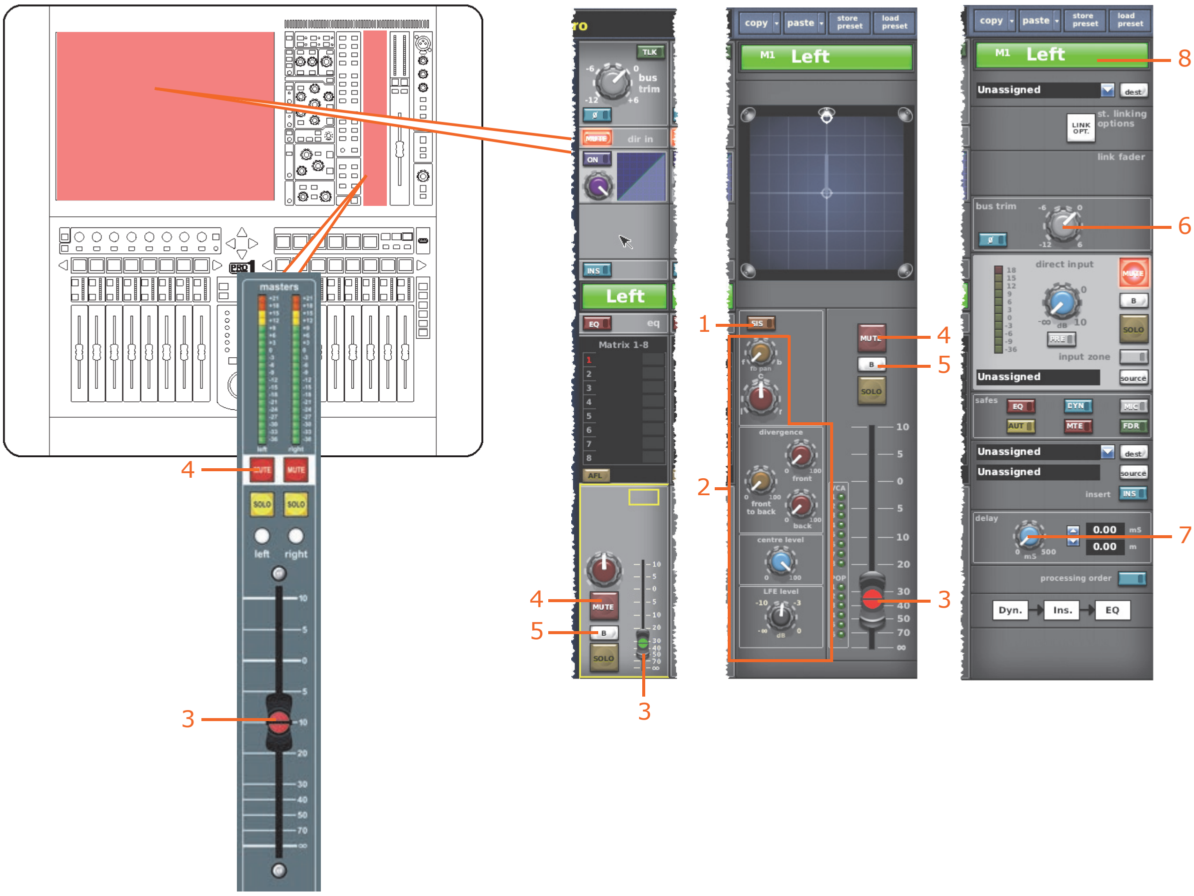


Item	*Control*	*Parameter*
1	**ON** switch	Matrix bus send on/off
2	**level/pan** control knob	Bus level, or pan when bus is linked
3	**PRE** switch	Pre-fader on/off
4	**level** control knob	Bus level

Note: *Although only matrix sends 1-2 are referenced above, this also applies to all eight matrix sends.*

Fader Sections

This section shows the fader parameters copied through scenes.



Item	Control	Parameter
1	**SIS** switch	Spatial imaging system on/off
2	Panning control knobs	Surround panning (includes all surround sound parameters)
3	Fader	Level
4	**MUTE** switch	Mute on/off
5	**B** switch	Solo B in/out
6	**bus trim** control knob	Bus trim level
7	**delay** control knob	Delay time
8	Field	Channel colour

Recall Scope

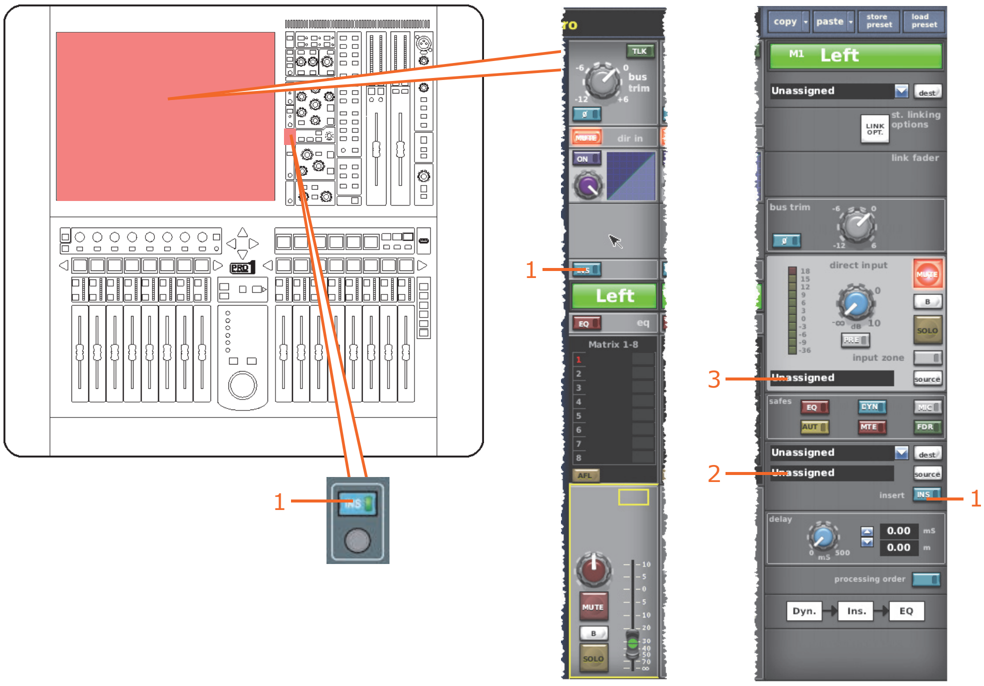
For details, see "Masters" on page 381.

Store Scope

For details, see "Masters" on page 381.

Routing

This section shows the routing parameters copied through scenes.



Item	*Control*	*Parameter*
1	**INS** switch	Insert in/out
2	Field	Insert return source*
3	Field	Direct input source*

* Only automated when automate patching is on.

Misc (miscellaneous)

The **Misc** section has an **Assignables** option, which lets you copy the current control assignments of the **Assignable Controls** window through scenes. (The levels are not copied.) However, none of the options in the **Sections** area are applicable to this option.

For more information on assignable controls, see Chapter 19 "Assignable Controls" on page 169.

Appendix L: Service Information

The service manual for this equipment is available for purchase. Please contact your local distributor for details.

Electrostatic discharge (ESD) precautions

Observe full electrostatic discharge (ESD) — also known as “anti-static” — precautions when carrying out procedures in this manual that are accompanied by the ESD Susceptibility Symbol (shown above). This caution symbol shows you that ESD damage may be caused to items unless proper ESD precautions are taken, which include the following practices:

- Keep the work area free from plastic, vinyl or styrofoam.
- Wear an anti-static wrist strap.
- Discharge personal static before handling devices.
- Ground the work surface.
- Avoid touching ESD-sensitive devices.

Routine maintenance

To help keep your PRO1 Control Centre unit in good working order and to make sure it gives you optimum performance, we recommend that you carry out the following about once every month.

- Clean the control centre, as detailed in “Cleaning the control centre” below.
- Check controls for freedom of operation. As the controls are ‘self-cleaning’, this operation will help to prevent them from sticking.
- Check functionality of all controls, that is, control knobs, faders, pushbuttons, LEDs etc.
- Check functionality of equipment.

Cleaning the control centre

Switch off the control centre and electrically isolate it from the mains before cleaning.

Clean the control centre using a dry, lint-free cloth. Do not use harsh abrasives or solvents. When cleaning the unit, take great care not to damage faders, pushbuttons etc.

Cleaning the GUI screen

Switch off the control centre and electrically isolate it from the mains before cleaning.

Carefully wipe the surface of the GUI screen with a soft, lint-free cloth or screen wipe specially designed for the purpose. When cleaning the GUI screen please take the following precautions:

- Avoid putting pressure on the screen.
- Don’t use harsh abrasives, for example, paper towels.
- Don’t apply liquids directly to the screen.
- Don’t use ammonia-based cleaners and solvents, such as acetone.

If you are in doubt have any queries about cleaning the GUI screens, please don’t hesitate to contact us.

Equipment disposal

When this equipment has come to the end of it useful life, its disposal may come under the DIRECTIVE 2002/96/EC OF THE EUROPEAN PARLIAMENT AND OF THE COUNCIL of 27 January 2003 on waste electrical and electronic equipment (WEEE).

Hazardous substances in WEEE contaminate water, soil and air and ultimately put at risk our environment and health. The directive aims to minimize the impacts of WEEE on the environment during their life times and when they become waste.

The WEEE directive addresses the disposal of products when they have reached the end of their life and contributes to the reduction of wasteful consumption of natural resources. This will help to reduce pollution, and protect the environment and ourselves.

If this equipment carries a ‘crossed-out wheelie bin’ (shown left), please do not dispose of WEEE as unsorted municipal waste but collect and dispose of in accordance with local WEEE legislation. The horizontal bar underneath indicates that the product was placed on the EU market after 13th August 2005.

For WEEE disposal; see our website at www.midasconsoles.com for information.

Glossary

This section provides an explanation of some of the symbols, terms and abbreviations used in this manual.

5.1 surround: A surround sound system created from six channels that form a discrete signal, which is played back over a speaker system comprising five speakers (three front and two rear) and a subwoofer (which is the ".1" or LFE channel). See *LFE.*

µ: Micro- prefix symbol that represents 10^{-6} or one millionth.

A

A/D: Abbreviation for "analogue to digital". The conversion of a continuous signal into a numeric discrete sample sequence.

AC: Abbreviation for "alternating current".

Acoustic feedback: A sound loop existing between an audio input and audio output that is amplified on each cycle. For example, a mic input signal is amplified and passed to a loudspeaker. The output from the loudspeaker is picked up by the mic, which amplifies it again and passes it back to the loudspeaker, and so on.

AES/EBU: Abbreviation for "Audio Engineering Society/European Broadcasting Union"; see *AES3*.

AES3: Also known as "AES/EBU", this is a serial interface for transferring digital audio between devices.

AES50: AES digital audio engineering standard. AES50 is a high resolution, multi-channel audio interconnection (HRMAI). Rather than a network, it is a high-performance, point-to-point audio interconnection, although the auxiliary data may operate as a true network, independently of the audio. HRMAI provides a professional multi-channel audio interconnection that uses Cat 5e data cable and is compatible with Ethernet networks.

AFL: Abbreviation for "after fader listen". A function that allows the signal to be monitored post-fader, that is, after it has been acted upon by the fader.

Algorithm: In computing, a set of instructions for accomplishing a specific task.

amp (A): Abbreviation for "ampere". A unit of current.

Anti-aliasing: When referring to digital images, a technique that avoids poor pixelation.

Area A: Primary input control area.

Area B: Secondary input control area.

Assignable controls: Any controls, such as select buttons (LCD switches), that can be set up by the user to control any function required.

Auto safe: Prevents channel from accepting scene recall.

Auto-mute: A function that automatically mutes the channel's signal under certain conditions.

Auto-mute group: A function that automatically mutes a number of selected channels under certain conditions.

Automation: 1. Memorization and playback of changes made to mixer settings. 2. An area on the output bay that controls these.

Aux: Abbreviation for "auxiliary send" or "aux send". A designation for extra buses, typically used for sending signal to effects, headphone amps and other destinations. See *Bus*.

Aux send: See *Aux*.

B

Balanced audio: A type of audio connection that uses the three leads in a cable, connector and jack as part of a phase-cancelling arrangement to boost the signal and reduce noise.

Band: In EQ, a range of frequencies.

Bandwidth: In EQ, the width of a band, that is, the number of frequencies that will be boosted/cut above and below a centre frequency.

Bank: A fixed number of channels.

Bass: Lower frequencies in a signal.

Bay: One of the five main control centre modules, which contains a control surface and a GUI screen.

Bus: A pathway down which one or more signals can travel.

C

Cat 5e: A specification for a type of cable used typically for Ethernet computer networks.

Channel: Single path taken by an audio signal (input or output) through the control centre.

Channel strip: Row of controls in traditional analogue layout used for the shaping of a signal.

Checkpoint: A patching data store point, created by clicking **CHECKPOINT**. See *Patching.*

Click: A method of GUI operation, mainly for button operation and selection purposes.

CMR: Abbreviation for "common mode rejection". A measure of how well a differential amplifier rejects a signal that appears simultaneously and in-phase at both input terminals. CMR is usually stated as a dB ratio at a given frequency.

Comb filtering: Removal of signal components at a number of regularly spaced frequencies.

Compressor: A dynamics processor that reduces the level of any signal exceeding a specified threshold volume.

Condenser microphone: A high quality mic that uses a capacitor to detect changes in the ambient air pressure, which it then converts into an electrical signal. This type of mic requires power from a battery or external source.

Control centre: The console of the PRO1, comprising control surface and GUI.

Control surface: Area on the control centre that houses all of the user's hardware controls, such as pushbuttons, control knobs, switches etc.

Crossfade: To combine signals such that one channel or source fades out while another fades in, but maintaining an essentially constant programme volume. Also known as "X-fade" or "xfade".

Cursor: Generally, used to describe the "I"-shaped pointer on the GUI that indicates a text insertion point. See *Pointer.*

D

D zone: Section in the input channel strip for controlling dynamic parameters.

D/A: Abbreviation for "digital to analogue". The conversion of digital data to analogue audio.

DARS: Abbreviation for "digital audio reference signal".

Dashboard: A standard GUI screen display - usually on the output bay - that shows all channel meters (inputs, auxes, returns, masters etc.) all of the time.

dB: Symbol for "decibel". A unit of measurement of the loudness of sound. See *dBu*.

dBu: A unit of measurement of sound used in professional audio. Derived from the decibel, where the "u" stands for unloaded, this unit is an RMS measurement of voltage based on $0.775V_{RMS}$, which is the voltage at which you get 1mV of power in a 600 ohm resistor. This used to be the standard impedance in most professional audio circuits.

DC: Abbreviation for "direct current".

Delay: An effect by which a reproduction of a signal is played back later then its original.

Destination: In patching, the patch connector to which a signal is routed. See *Patching.*

Detail area: see *Processing area.*

Device: A diagram(s) on the I/O tabs (GUI patching) representing a physical rack unit, such as a line I/O, mic splitter, DN9696, AES50 etc. See *Patching.*

DHCP: Abbreviation for "dynamic host configuration protocol". A network configuration protocol for IP network hosts.

DI: Abbreviation for "direct inject" or "direct injection". Signal is plugged directly into the audio chain without using a microphone.

DI box: Device for matching the signal level impedance of a source to mixer input.

Divergence: The spreading of sound waves from a source in a free field environment, that is, one with no reflections. Causes the sound pressure levels in the far field of the source to decrease as the distance from it increases.

Drag: A method of GUI operation, mainly for control adjustment. Also used for selecting blocks of patch connectors during patching.

DSP: Abbreviation for "digital signal processing" or "digital signal processor". Any signal processing done after an analogue audio signal has been converted into digital audio. Can be used to create, for example, compression, equalization etc., of a digital signal. A digital signal processor is a piece of equipment specifically designed for carrying out signal processing.

E

E zone: Section in the input channel strip for controlling EQ parameters.

Effect: One of a number of audio processes that can be applied to a signal to modify it, such as reverb, flanging, phasing, delay etc.

Effects rack: On the PRO1 GUI, a virtual rack of internal processors. See *Virtual rack*.

Envelope: 1. How a sound or audio signal varies in intensity over time. 2. The visual representation of such, usually shown on a graph in a GUI channel strip.

EQ: Abbreviation for "equaliser" or "equalisation".

Equalisation: Adjusting the frequency response so that the levels of all frequencies are equal or the same. Bass and treble controls are equalization controls.

EtherCon®: A cable connector for data transfer interconnections, which is more robust than the basic RJ45.

F

Fader: Slider-type device for precise adjustment of signal level or volume of a channel.

Fast strip: A channel strip in one of the fast zones. See *Fast zone*.

Fast zone: An area on a bay that contains quick controls. See *Fast strip*.

FB: Abbreviation for "front-back". A term used in surround panning.

Feedback: See *Acoustic feedback*.

Filter: A device for removing frequencies above or below certain levels.

FOH: Abbreviation for "front of house". The area in a theatre used by the public. Used to describe a console/control centre being used to control the sound that the audience will hear (and not the performers' monitor system).

Frequency: The number of times that a sound wave's cycle repeats within one second.

Fricative: A consonant, such as "f" or "s", produced by the forcing of breath through a constricted passage.

From section: The leftmost area of the patching screen that contains the source patch connectors. See *Patching.*

FX: Abbreviation for "sound effects".

G

Gain: Another term for signal level.

Gain reduction (compressor): Decrease in gain when input signal is above threshold. See *Gain.*

GEQ: Abbreviation for "graphic equaliser". See *Graphic EQ*.

GEQ rack: A virtual rack of GEQs. See *Virtual rack*.

Glide pad: Device for controlling the GUI. See *Navigation zone*, *Touchpad* and *Trackball*.

Granularity: A measure of the size of components or a description of the components comprising a system.

Graphic EQ: A form of EQ that has a number of faders for controlling the gain of the audio signal. The faders are set at frequency bands that are evenly-spaced according to octaves.

GUI: Abbreviation for "graphical user interface".

GUI channel strip: Rightmost section of a GUI screen that represents the processing area of the input or output channel strip selected to the control surface.

GUI menu: A menu selectable at either GUI screen by clicking the **home** button (upper-left corner).

GUI screen: One of the PRO1's five screens, which comprise the GUI.

H

HPF: Abbreviation for "high pass filter". A filter that removes lower frequencies from a signal, leaving the higher frequencies unaffected.

Hum: Undesirable low frequency tone present in a signal due to grounding problems or proximity to a power source.

Hz: Symbol for "Hertz". A unit of frequency equal to one cycle of a sound wave per second.

I

I zone: Area on the output bay that contains the operator-assignable controls.

I/O: Abbreviation for "input/output".

ID: Abbreviation for "identification".

Ident: Scale marking, or gradation, around a control knob to help indicate the current setting and to assist in accurate adjustment.

Impedance (Z): Opposition to the flow of alternating current in a circuit, measured in ohms.

Input: 1. The signal being received by a device. 2. The physical location of where a device receives a signal. 3. Concerning the input bays on the PRO1 control surface.

IP: Abbreviation for "internet protocol". Principal network communications protocol.

K

k: Kilo- prefix symbol that represents 10^3 or one thousand, for example, kHz (one thousand Hz) or km (one thousand metres).

Kernel: For computers, the kernel is the central component of most operating systems.

L

LCD: Abbreviation for "liquid crystal display".

LCR: Abbreviation for "left-centre-right", referring to speakers.

LED: Abbreviation for "light emitting diode".

Level: General term for volume or amplitude.

LF: Abbreviation for "low frequency".

LFE: Abbreviation for "low frequency effects". A discrete channel intended for playback through a subwoofer speaker, although it can be fed to any speaker that can handle low frequency signals. LFE is generally used to enhance sound effects in films and is, typically, the ".1" in "5.1 surround". LFE is not full bandwidth (range). Examples of LFEs are: thunder, explosions and other bass effects.

LHS: Abbreviation for "left-hand side".

Limiter: An extreme form of compressor that only affects signals above a selected threshold level (dB).

Linux: Also known as "Linux kernel". Operating system kernel used by a family of Unix-like operating systems. See *kernel.*

LPF: Abbreviation for "low pass filter". A filter that removes higher frequencies from a signal, leaving the lower frequencies unaffected.

LS: Abbreviation for "left surround". The left rear speaker in a 5.1 surround system.

M

m: 1. Prefix symbol for "metre(s)", for example, as in "200 m" (200 metres); please note the intermediate space. 2. Prefix symbol for milli-, which represents 10^{-3} or one thousandth, for example, as in "2ms" (2 milliseconds); there is no intermediate space.

MADI: Abbreviation for "multi-channel audio digital interface". An AES standard for digital interconnection between multi-track recorders and mixing consoles.

Main bus: A type of bus; see *Bus*.

Masters: The three master channels (mono and stereo left and right) in the output bay.

MB: Abbreviation for "megabyte".

MC: Abbreviation for "master controller".

MCA: Abbreviation for "mix control associate".

Meter: Visual device to indicate the level of a signal.

Mic: Abbreviation for "microphone".

Microphone: Device for converting sound waves into audio signals.

MIDI: Acronym for "musical instrument digital interface". A digital signal system standard that facilitates integration of musical instruments, such as synthesizers and guitars, with computers.

Mix: 1. A signal that contains a combination of signals, such as a pair of stereo signals with numerous effects. 2. The act of creating such a combination. 3. A type of bus (see *Bus*).
4. Concerning the mix bay on the PRO1 control surface.

Mixer: 1. A console or other device that blends input signals into composite signals for output.
2. An engineer/technician who carries this out, especially during a live performance.

mm: Symbol for "millimetre" (one thousandth of a metre).

MON: Abbreviation for "monitor", used to describe a console/control centre being used to mix the signals sent to the stage monitor speakers.

Monitor: 1. Speaker(s) used for listening to a mix or live audio. 2. The act of listening to a mix or live audio. 3. Concerning the primary or secondary monitor bus system, A or B respectively.

Monitors: Control area on the master bay for monitoring the A and B signal paths.

Mono: A single signal.

Mute: Function that allows a channel's signal to be silenced.

Mute safe: Function that means a mute cannot be controlled by scene recall or auto-mutes.

N

N/A: Abbreviation for "not applicable".

Navigation: The act of directing channels or buses to the control surface for selection, mixing, processing etc.

Navigation zone: Area on the control surface concerning navigation.

nm: Symbol for nanometre (one billionth of a metre).

Normalisation: An automatic process whereby the gain of all program material is adjusted so that the peak level will just arrive at 0dB.

Normalise: To boost the amplitude of a digital sound so that it is as high as it can be without clipping (0dB).

Normalised connection: Also known as "normalled connection". A connection that allows a signal to pass through it when no plug is inserted in it, but breaks the connection when a plug is inserted.

Normalising: The process of making audio files the same volume.

NVRAM: Abbreviation for "Non-volatile random access memory". this is the general name used to describe any type of RAM that retains its information when power is switched off. For example, flash memory.

O

O/B: Abbreviation for "outside broadcast".

Oct: Abbreviation for "octave".

Octave: A difference in pitch where one tone has a frequency that is double or half of the frequency of another tone.

ohm (Ω): Unit of electrical resistance.

OpticalCon®: A cable connector for fibre optic cables.

OS: Abbreviation for "operating system".

OSC: Abbreviation for "oscillator" or "oscillation".

Out of phase: 1. A signal, being similar to another in amplitude, frequency and wave shape, but offset in time by part of a cycle. 2. 180° out of phase or having opposite polarity. See *Phase*.

Outboard: External, as in an "external device".

Outboard equipment: External equipment used with the DL251/DL252 Audio System I/O, but that is not part of it.

Output: 1. The signal put out by a device. 2. The physical location of where a device sends out a signal. 3. Concerning the output bay on the PRO1 control surface.

Overload: A condition where the signal level is too high.

P

Pan: To move from one side to another or up and down.

Panning: The left/right positioning of a signal across a stereo image.

Parameter: A setting whose value can be altered by the user.

Parametric EQ: A type of EQ that allows all of the parameters of equalisation to be changed, including centre frequency, boost/cut in gain, and bandwidth.

Patch: A temporary connection (physical or virtual) made between two audio devices or inside one.

Patch connector: Any tab patching point, for example, an XLR connector, bus, sidechain compressor etc. See *Patching.*

Patching: Also known as "soft patching". The process of routing a channel/signal from a source to a destination(s).

PCB: Abbreviation for "printed circuit board".

PEQ: Abbreviation for "parametric equaliser". See *Parametric*.

PFL: Abbreviation for "pre-fade listen". A function that allows the signal to be monitored pre-fader, that is, before it reaches the fader.

Phantom power: The power required for the operation of a condenser microphone when it is not supplied by internal batteries or a separate power supply. This is supplied by the PRO1 Live Audio System itself.

Phase: A measurement (in degrees) of the time difference between two waveforms.

Pitch: A continuous frequency over time. Musical interpretation of an audio frequency.

Pitch shift: Alteration of pitch or frequency, but without adjusting tempo.

Point scene: Subdivision of a scene. See *Scene*.

Pointer: 1. On the GUI, the pointer is the arrow-shaped object on the screen that moves when the user moves the trackball or external mouse. 2. On a control knob, it is the marking that, when used in conjunction with the ident around edge of control knob, helps to indicate the setting.

POPulation group: A number of channels assigned to a group that has unfold and area B controls. Provides an easy and quick method of manipulating and controlling the numerous channels available on the PRO1 Live Audio System.

Post-: The point for accessing audio just after it leaves a specific channel component, for example, "post-fader", where the audio is tapped from just after it leaves the channel's main level control.

Pre-: The point for accessing audio just before it reaches a specific module, for example, "pre-EQ", where the audio is tapped from just before it gets to a channel strip's EQ.

Processing area: Also known as "detail area", a control section in the channel strips.

Pschycoacoustic noise: Noise that affects the physiology of the listener.

PSU: Abbreviation for "power supply unit".

Psychoacoustics: The study of the perception of sound, that is, how we listen, our psychological responses and the physiological effects on the human nervous system.

Q

Quick access button: Button for quickly selecting its associated strip section.

R

RAM: Abbreviation for "Random access memory".

Return: Auxiliary return or aux return. An extra input used for receiving a signal from the output of an internal or external effect processor. See *Bus*.

Reverb: An effect where the ambience of a physical space is simulated. This is done by copying a signal and replaying at regular intervals at ever decreasing levels. The intervals are so close that each copy is not heard individually.

RHS: Abbreviation for "right-hand side".

RMS: Abbreviation for "root-mean-square". The square root of the mean of the sum of the squares. Commonly used as the effective value of measuring a sine wave's electrical power. A standard in amplifier measurements. The effective average value of an AC waveform.

RS: Abbreviation for "right surround". The right-hand rear speaker in a 5.1 surround system.

S

s: Symbol for "second"; a unit of time.

Scene: In automation, a set of mix settings for a particular part of a play or song.

Sibilance: Energy from a voice, centred around 7kHz, caused by pronouncing "s", "sh" or "ch" sounds.

Side chain: A special circuit that diverts a proportion of the main signal so that it can be processed, as required. Compressors use the side chain to derive their control signals.

Signal flow: The path of a signal from one place to another.

SIP™: Abbreviation for "solo in place".

SIS: Abbreviation for "spatial imaging system". Combines a central loudspeaker cluster with a left-right system to form three discrete sound channels.

Snapshot: A captured group of mixer settings that reflect the state of the mixer at a particular moment within a performance. This snapshot can then be recalled at the required moment in the performance/playback.

Solo: During monitoring, the isolation of one signal by silencing all other signals.

Source: The patch connector from which a signal is patched. See *Patching.*

SPL: Abbreviation for "sound pressure level". Given in decibels (dB), SPL is an expression of loudness or volume.

Splash screens: The GUI display during power up.

SRC: Abbreviation for "sample rate converter".

SSD: Abbreviation for "solid-state disk". Data storage device that uses non-volatile memory to store data. Quicker than the conventional hard disk and less susceptible to the failures associated with hard disk drives.

Status indicator: A device specifically designed to show the condition of something. For example, an LED that shows whether a pushbutton is on or off, or a meter showing the level of a signal.

Stereo: Two separate channels, left and right, used to give the listener the perception of where the noise is coming from. Usually used with music to give a fuller, more natural sound.

Stereo image: The perception of the different sound sources coming from far left, far right or anywhere in between.

Surround: Audio that has more than two speaker locations and, therefore, more than two channels. Also commonly termed "surround sound".

Sync: Abbreviation for "synchronisation".

Synchronisation: Coordination of timing between devices.

T

Tab: A 'sheet' in the **From** and **To** sections of the **Patching** screen (GUI) that contains a specific group of patch connectors. See *Patching.*

Talk output panel: Routing system for talking out from the console.

TFT: Abbreviation for "thin film transistor".

Threshold: Level at which dynamics processing will begin to operate.

Tie line: A dedicated connection between two systems, typically between FOH and MON positions.

To section: The rightmost area of the Patching screen that contains the destination patch connectors. See *Patching.*

Tooltip: The information box that appears next to the cursor on some GUI screens when it is on or near certain items, such as the channels on the **Dashboard** screen or patch connectors on the **Patching** screen.

Touchpad: Also known as "trackpad". An input device on a laptop PC for controlling the on-screen pointer. See *Glide pad* and *Trackball*.

Track: Single stream of recorded audio data.

Trackball: Device for controlling the GUI. See *Navigation zone*, *Glide pad* and *Trackball*.

Treble: Higher frequencies in a signal.

Tunnelling Ethernet: Mechanism by which to transport a foreign protocol across a network that normally wouldn't support it.

TW: Abbreviation for "twin wire".

U

Unbalanced audio: A type of audio connection that utilises only two of the leads of a cable, connector and jack.

Unfold: Navigates the input channels of a group to the input bays.

USB: Abbreviation for "universal serial bus". A 'plug and play' interface that provides a fast connection between a computer and peripherals, such as keyboards, printers, scanners, digital cameras etc.

V

VCA: Abbreviation for "variable control association", as in VCA group (also "voltage controlled amplifier").

VCA fader: The fader control of a VCA group.

VCA group: A group of channels that are controlled globally, such as via their group's fader and other controls. Provides an easy and quick method of manipulating and controlling the numerous channels available on the PRO1 Live Audio System. See *VCA.*

VGA: Abbreviation for "video graphics array". A graphics display system for PCs developed by IBM.

Virtual rack: A traditional 19" rack, represented on the GUI. A virtual rack will, typically, contain internal devices, such as effects and GEQs.

volt (V): A unit of electrical potential differential or electromotive force.

Volume: General term for a signal's loudness.

VU meter: Abbreviation for "volume unit meter".

W

Window: A small self-contained panel that appears on the GUI, usually after selection of a specific control. Typically, contains a number of user-selectable options or information in the form of a message or prompt.

X

X-fade: Abbreviation for "crossfade".

X-over: Abbreviation for "crossover".

XLR connector: High-quality three-pin audio connector, which is also used for AES/EBU digital audio connections.

Other important information

1 **Register online.** Please register your new Midas equipment right after you purchase it by visiting www.midasconsoles.com. Registering your purchase using our simple online form helps us to process your repair claims more quickly and efficiently. Also, read the terms and conditions of our warranty, if applicable.

2 **Malfunction.** Should your MUSIC Group Authorized Reseller not be located in your vicinity, you may contact the MUSIC Group Authorized Fulfiller for your country at www.midasconsoles.com. If your country is not listed please contact the "United Kingdom (Midas/KT main office)" located under "Service ↳Service/Repairs" on the www.midasconsoles.com website. Alternatively, please submit the online warranty return form found under "Service ↳Warranty Registration" on www.midasconsoles.com BEFORE returning the product. All enquires must be accompanied by the description of the problem and the serial number of the product. The warranty eligibility will be verified from the original sales receipt.

3 **Power Connections.** Before plugging the unit into a power socket, please make sure you are using the correct mains voltage for your particular model. Faulty fuses must be replaced with fuses of the same type and rating without exception.

FEDERAL COMMUNICATIONS COMMISSION COMPLIANCE INFORMATION

MUSIC Group Research UK Limited
PRO1 Live Audio System

Responsible party name:	**MUSIC Group Research UK Limited**
Address:	**Klark Industrial Park, Walter Nash Road, Kidderminster. Worcestershire. DY11 7HJ. England.**
Phone/Fax No:	**Phone: +44 1562 741515** **Fax: +44 1562 745371**

PRO1 Live Audio System

complies with the FCC rules as mentioned in the following paragraph:

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Important information:

Changes or modifications to the equipment not expressly approved by MUSIC Group can void the user's authority to use the equipment.

Thank you for reading through this Addendum. We hope you found it useful.

Please feel free to send us your comments. Our contact details and website address can be found at the front of this document.

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