Networking Audio Systems

LAWO

mc²56

Operators Manual

Version: 4.24/2

Edition: 23 October 2013

Table of Contents

Welcome

Welcome to the **mc^{2}56** mixing console - all the power of a mc^2 in a footprint ideal for mobile recording or compact studio environments.

This documentation covers the operation of both the classic and MKII revisions of the **mc^{2}56**. The manual uses MKII illustrations; notes for the classic console are added where necessary.

For more on installation, configuration or service/maintenance, please see the "mc^{2}56 Technical Manual".

You can access this and other information by registering at www.lawo.de (click on **Login**). By registering you will receive the latest news for your product, and can download software and documentation.

Getting Started

- Read the Overview and Getting Started to familiarise yourself with the basic operation. If you have limited time to learn the console, then these chapters are for you.
- The remaining chapters cover each area of operation in greater detail.
- You will find Appendices, Technical data, and a Glossary of terms at the end of the manual.

Marginal Notes

The following symbols are used to draw your attention to:

Points of clarification.

Useful tips and short cuts.

Warning

Warnings – alert you when an action should *always* be observed.

Changes to This Manual

- **Version 4.24/2** (Edition: October 23rd 2013) - valid from build **V4.24.x.x**. New features added:
  - o Support for the Nova73 Compact Core.
  - o Improved touch-screen operation (Central GUI PAGE button & context menus).
  - o Multi-row Metering on the Channel Display.
  - o The Machine Locators display.
  - o TONE to channel switching.
  - o Set Access option (**Signal List** display).
  - o ARIB standard supported by loudness metering.
  - o New **System Settings** options (extended channel mute; tiny channels for conference).
  - o New **Custom Functions** (bird beater aux; multi-row meter switching; fader R/W user button; test tone activate; test tone fader user button).
  - o System logfile time stamping.
- **Version 4.20/3** (Edition: March 21st 2013) - valid from build **V4.20.x.x**.

Important Safety Instructions

Warning

Exposure to excessive sound pressure levels can lead to impaired hearing and cause damage to the ear.

Please read and observe *ALL* of the following notes:

- Check all of the hardware devices for transport damage.
- Any devices showing signs of mechanical damage or damage from the spillage of liquids *MUST NOT* be connected to the mains supply or disconnected from the mains immediately by pulling out the power lead.
- All devices *MUST* be grounded. Grounding connectors are provided on all devices. In addition, all low-voltage devices external to the system must also be grounded before operation.
- For Scandinavian countries, *ALWAYS* use a grounded mains connection, to prevent the device from being grounded through Ethernet or other signal connections.
- All devices *MUST* be connected to the mains using the three-cord power leads supplied with the system. Only supply electrical interfaces with the voltages and signals described in these instructions.
- Do *NOT* use the system at extreme temperatures. Proper operation can only be guaranteed between temperatures of 10° C and 35° C and a maximum relative humidity of 85%.
- Neutrik PowerCon and Harting connectors must *NOT* be disconnected under load.
- Only Lawo service staff may replace batteries.
- Servicing of components inside a device MUST only be carried out by qualified service personnel according to the following guidelines:
  - We recommend switching off the loudspeakers before servicing.
  - Before removing parts of the casing, shields, etc. the device *MUST* be switched off and disconnected from all mains.
  - Before opening a device, the power supply capacitor *MUST* be discharged with a suitable resistor.
  - Components that carry heavy electrical loads, such as power transistors and resistors, should *NOT* be touched until cool to avoid burns.
- Servicing unprotected powered devices may only be carried out by qualified service personnel at their own risk. The following instructions *MUST* be observed:
  - *NEVER* touch bare wires or circuitry.
  - Use insulated tools *ONLY*.
  - *DO NOT* touch metal semi-conductor casings as they can bear high voltages.

Defective Parts/Modules

Warning

- The system components contain no user-serviceable parts. Therefore *DO NOT* open the devices other than to perform the procedures described in this manual.
- In the event of a hardware defect, please send the system component to your local service representative together with a detailed description of the fault. We would like to remind you to please check carefully whether the failure is caused by erroneous configuration, operation or connection before sending parts for repair. We recommend contacting our service department before sending parts for repair.

First Aid (in the case of electric shock)

Warning

- *DO NOT* touch the person or his/her clothing before power is turned off, otherwise you risk sustaining an electric shock yourself.
- Separate the person as quickly as possible from the electric power source as follows:
    - o Switch off the equipment.
    - o Unplug or disconnect the mains cable.
    - o Move the person away from the power source by using dry insulating material (such as wood or plastic).
- If the person is unconscious:
    - o Check their pulse and reanimate if their respiration is poor.
    - o Lay the body down and turn it to one side. Call for a doctor immediately.
- Having sustained an electric shock, *ALWAYS* consult a doctor.

Chapter 1: Overview

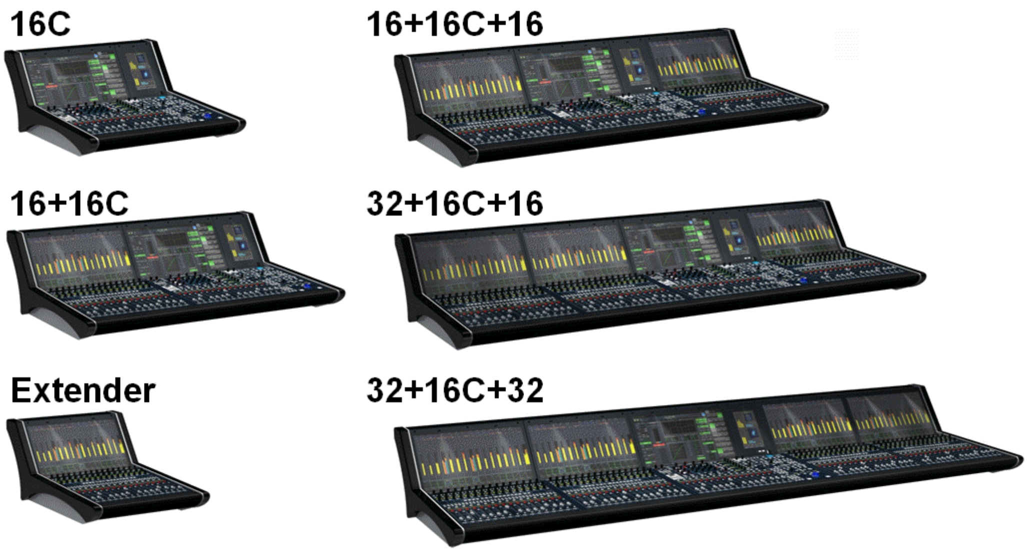
Introduction

This chapter provides an overview of the **mc²56** and its key features:

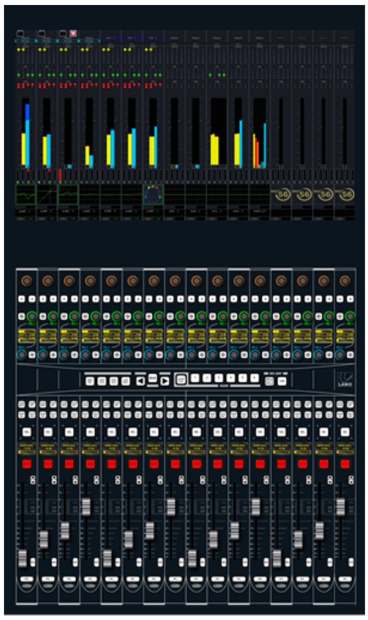
- Control Surface Overview
- System Overview
- Signal Flow
- The Power of Layering
- Mono, Stereo and Surround
- Comprehensive Control
- Flexible Metering
- Integrated Routing Matrix
- Console Reset
- mxGUI
- Timecode Automation
- Configuration
- Integration with the Outside World
- Classic vs MKII - the Differences

Control Surface Overview

The **mc²56** control surface is constructed in 16-fader sections, with frame sizes scaling from 16 faders up to 80 faders. You may add 16-fader extenders to expand the number of fader strips.



Channel Strips



Within each channel bay, you will find 16 dedicated fader strips providing level, mute, AFL/PFL monitoring, layer flip, fader selection and 4 user buttons.

Two assignable rotary controllers (Free Controls) offer local channel control to adjust EQ, Dynamics, auxiliary sends, etc.

A third upper controller is dedicated to input gain.

In addition, every channel bay houses a high resolution TFT (the Channel display), providing visual feedback and touch-screen operation of channel metering and bus/VCA assignments:

Centre Section

The centre section houses the Central GUI touch-screen, master controls and main fader strips.

The Central GUI provides access to a range of displays, and may be operated via the touch-screen, trackball, SCREEN CONTROL panel or console keyboard.

Space is available to the right of the GUI (MKII mc²56 only), for options such as the RTW goniometer or a Lawo User Panel. You will also find two USB ports, to connect the console keyboard or memory stick, and an XLR talkback mic connector.

Below is the Central Control Section offering direct control of *all* settings for the selected channel – input control, equalisation, dynamics processing, panning, auxiliary sends, etc.

On the right are a range of controls for monitoring, snapshot/sequence, production, banking and layering, bus and fader strip assignment, user buttons and SCREEN CONTROL navigation.

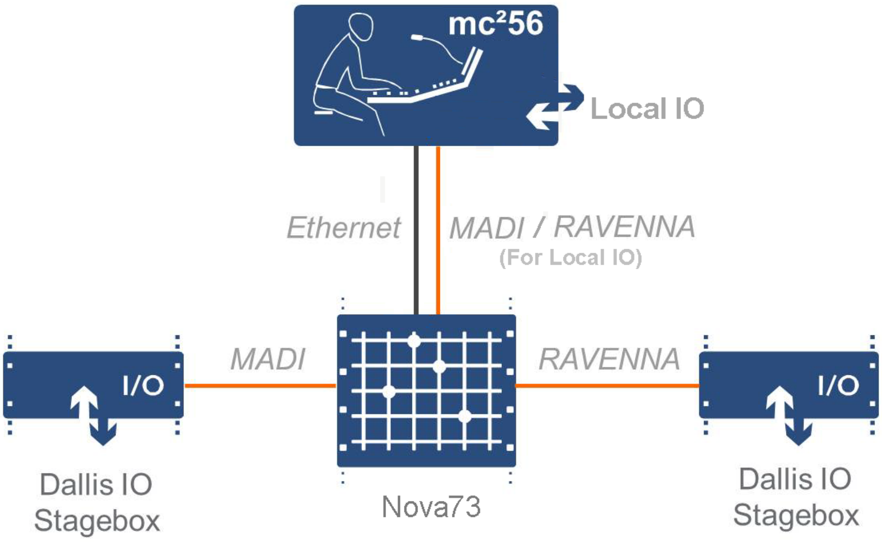
Below are the 16 additional main fader strips ideal for master VCAs, groups, etc.

Look along the front buffer of the console for the integrated headphone outputs and Ethernet connector (for an mxGUI or service/configuration computer).

System Overview

Unlike an analogue console, the **mc²56** consists of much more than just the control surface. For any single installation, there are three principal components:

- **Console control surface** - with integrated power supplies and local I/O connections.
- **Nova73** – with Router Modules, DSP boards and AES, MADI, ATM or RAVENNA I/O. Available in two sizes: **Nova73 HD** (10RU) or **Nova73 Compact** (7RU).
- **DALLIS I/O** – offering further I/O breakout options; connected to the Nova73 via MADI, ATM or RAVENNA I/O.



The exact hardware specification defines how many analogue and digital connections are available for external equipment, and how much DSP processing is available for input channels, monitor return channels, groups, sums and auxiliary sends. For a summary of the system capabilities, please see Technical Data.

> In the classic mc²56 there are no integrated local I/O cards, and so the only connection between the control surface and Nova73 is Ethernet.

Redundancy

Redundant power supplies may be fitted to the control surface, Nova73 and each DALLIS stagebox.

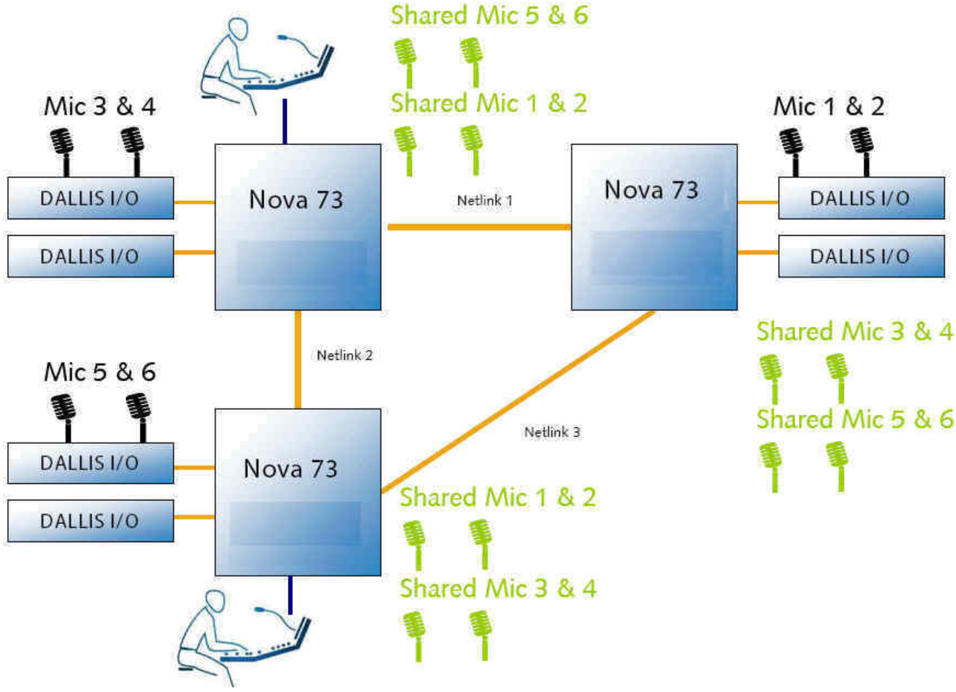
You may run any DSP board in standby for channel DSP redundancy. In addition, you may specify two Nova73 Router modules for full redundancy of the routing matrix, control system and all application software and user data. Combine this with redundant DALLIS master cards to create fully redundant audio signal paths.

For more details, see Redundancy.

System Networking

The **mc²56** is just one member of the mc² family of products, which utilise the same Nova73 and DALLIS architecture, and run on the same operating system and application software.

The Nova73 and DALLIS system is available in its own right as a stand alone routing matrix. Multiple systems may be networked to provide sharing of sources and destinations:



For more details, see Networking I/O Resources.

Signal Flow

The **mc²56** provides a pool of DSP resource which can be configured for input channels, monitor return channels, groups, sums (main mix outputs) and auxiliary sends. Each channel comes with either full signal processing or reduced signal processing (known as tiny channels). This enables EQ, Dynamics, Delay, etc. to be applied to both inputs and outputs.

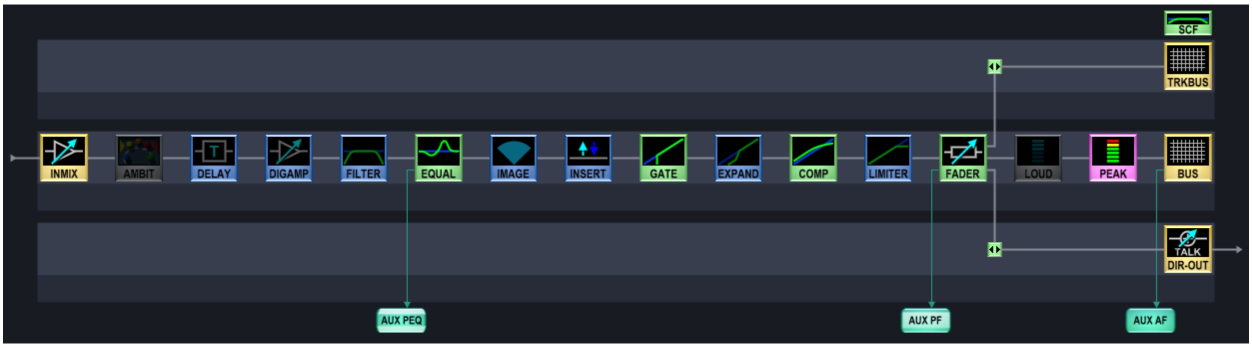
The number of input, monitor, group, sum and aux channels is determined by the number of channel DSP boards fitted to the Nova73 (up to 8); the sampling rate of the system (48/44.1kHz or 96/88.2kHz); and your choice of DSP configuration.

The DSP configuration is selected from a predefined list and stored when you save the production. DSP configurations are available in a choice of channel type:

- **Broadcast Channels** – provide twice as many channels per DSP board; each channel has a simplified signal flow (no track bus send, no independent filter section and simpler dynamics).
- **Recording Channels** – less channels per DSP board; each channel provides more processing and increased flexibility.

Input Channel (Broadcast)

Input Channel (Recording)



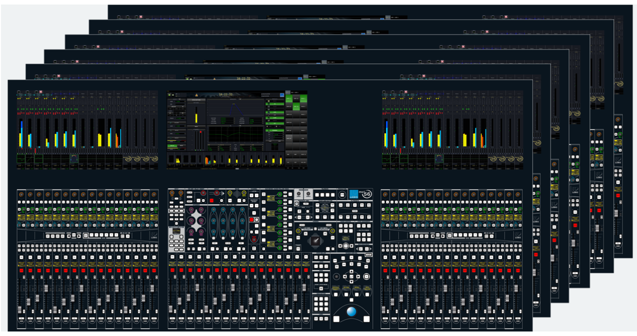
Each DSP configuration supports one channel type; you cannot mix Broadcast and Recording channels. To check that your system supports Broadcast channels, see Broadcast Channel Conditions.

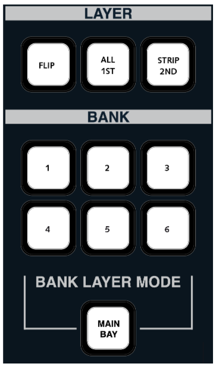
Once you have loaded a DSP configuration, you may modify the order of the processing modules (EQ, Delay, etc.) from the Channel Config display. This allows you to change the signal flow on a channel-by-channel basis.

The Power of Layering

The console's control surface includes both channel and main fader strips. Any fader strip may control any audio channel (input, monitor return, group, sum or aux), or any control channel (VCA, Surround VCA or GPC). This allows you to lay out your source channels, audio masters and control masters where you want them, see Fader Strip Assignment.

In addition, the physical size of the control surface does not restrict the number of audio processing channels. Additional channels may be added at any time by fitting more DSP boards to the Nova73; the extra channels are then accessed by paging the console's fader strips using banks and layers:





Banks and Layers

The console supports six control surface banks (1 to 6), each with two layers - Layer 1 and Layer 2.

Think of each bank as a separate console, with fast global or fader bay switching from one bank to another, see Bank Switching. Banks may be used to access different sets of channels (e.g. to switch from band 1 to band 2), or to switch between different fader strip layouts (e.g. to switch to an "effects" channel layout).

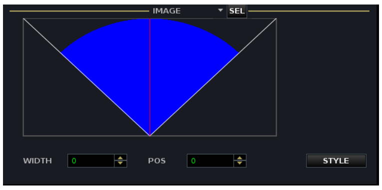
Within each bank, layers can be switched globally, within the fader bay, or individually, see Layer Switching. This makes layers ideal for related sources. For example, you could assign a presenter's input channel to Layer 1 with their mix minus aux master on Layer 2. Or, for multitrack recording, assign input channels to Layer 1 and monitor return channels to Layer 2.

If you wish, you may isolate individual fader strips so that they never switch bank or layer.

Or, isolate fader bays so, for example, they can be used by a second engineer.

Mono, Stereo and Surround

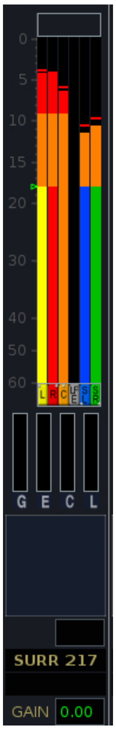
Any odd/even pair of input or output channels may be configured for stereo, and assigned to any fader strip bank or layer. Tools such as LR reverse, L to Both, R to Both, image width and positioning provide fast control of stereo signals from a single fader.



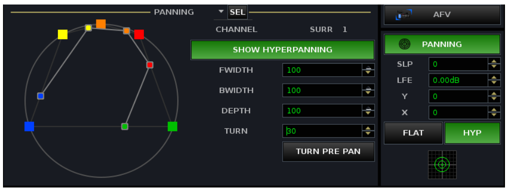
Similarly, multiple input or output channels may be configured for surround. A variety of multi-channel surround formats are supported up to 7.1. The surround format is set globally for each production from the **System Settings** display. This defines the format used for surround channels, pan laws and monitoring. For example, if you select Dolby Digital 5.1, then component channels 1 to 6 are configured as L, R, C, LFE, Ls and Rs.

Mono and stereo channels may be assigned onto any surround bus, and positioned using XY rotary controls or the console's motorised joystick.

A range of specialised tools provide easy management of surround channels:



- **Surround VCAs** - provide master control of the surround signal from a single fader strip. You can control the overall level, EQ, compression, etc. while metering all slave channels independently on the **Channel** display (shown opposite).
- **REVEAL** - temporarily assigns the individual surround slaves onto fader strips (within a pre-defined area or onto the optional REVEAL fader panel). This enables you to quickly offset fader levels and other relative parameters.
- **Hyper Panning** - provides an alternative to conventional XY panning. It is designed to help reposition surround sources within a surround field. For example, to turn a 5.1 source:



- **AMBIT** (AMBience IT) - is a special DSP module designed for upmix or spatialise processing:
  - o **Upmix** – a 2 in, 6 out upmixer which, using sophisticated algorithms, converts stereo signals into 5.1 surround.
  - o **Spatialise Only** – a 6 in, 6 out spatialiser which processes the surround left and right channels only, ideal for treating incoming 5.1 signals.

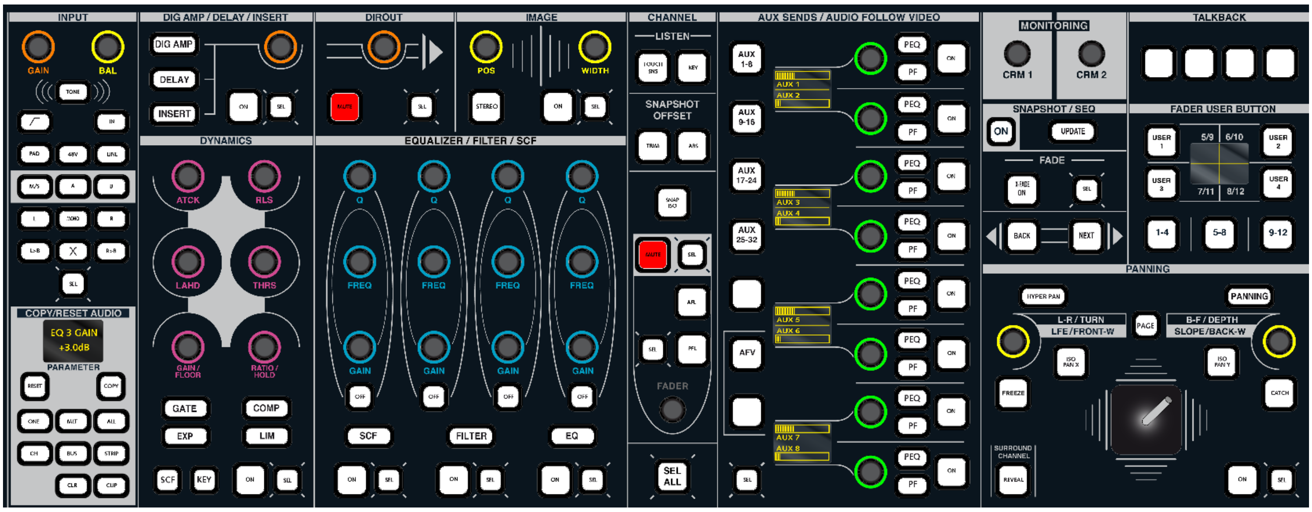
For more details, see Stereo Channels and Surround Channels.

Comprehensive Control

Central Control Section

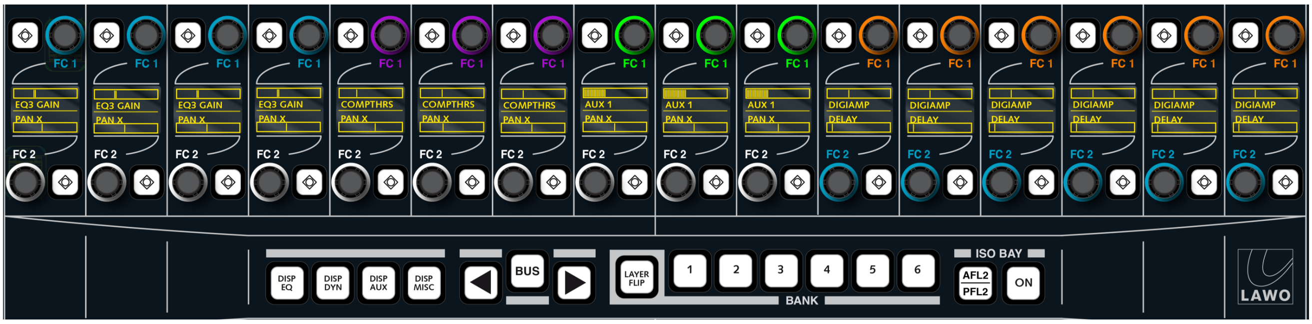
The Central Control Section provides master channel control for the channel in access - INPUT, DYNAMICS, EQUALIZER, etc.

Select a channel, by pressing its fader strip **SEL** button, and then reach out to control any parameter:



Channel Free Controls

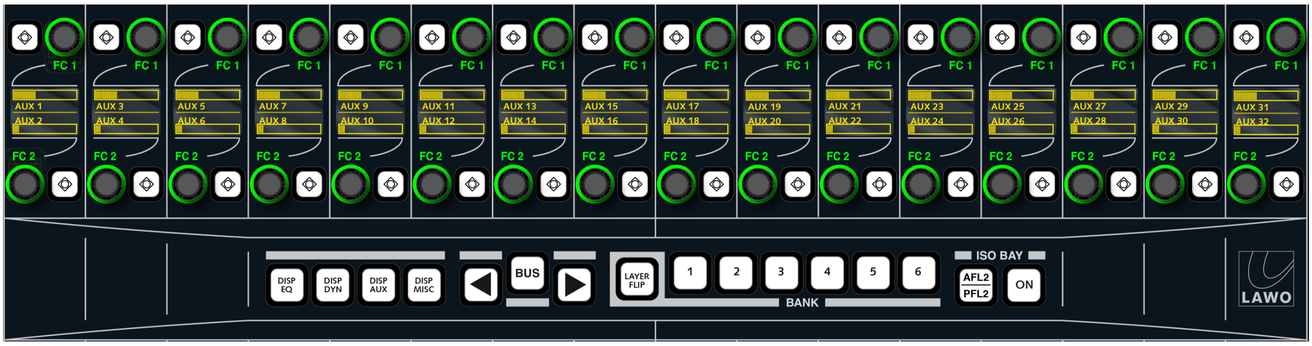
The two Free Controls on each channel fader strip may be assigned to key functions for the source. For example, on a presenter's channel you may want immediate access to the presenter's mix minus level and compressor threshold. Whereas, on a music replay channel, it is more important to access L/R Balance and Aux send level.



ISO Bay Operation

Alternatively, you can use ISO BAY as follows:

The ISO BAY **ON** and **DISP** buttons temporarily override the default Free Control assignments, so that all 32 Free Controls within a 16-fader bay can access multiple parameters for the selected channel (e.g. aux sends 1 to 32):



All channel DSP parameters (EQ, Dynamics, Aux sends, Delay, etc.) and bus assignments can be accessed in this manner. Note that local parameter control is *NOT* available for the classic mc²56.

Multiple Users

ISO BAY **ON** isolates the 16-fader section from the centre section's bank and layer switching. This allows a second engineer to independently bank/layer switch and control DSP settings within an isolated bay, while the main engineer has full control of the rest of the console.

Isolated bays can be excluded from snapshot loads. And, the AFL/PFL bus can be split to provide a second AFL/PFL output from the isolated bay(s) if desired.

Colour-coding

The control surface uses intelligent colour-coding to help distinguish different types of control:

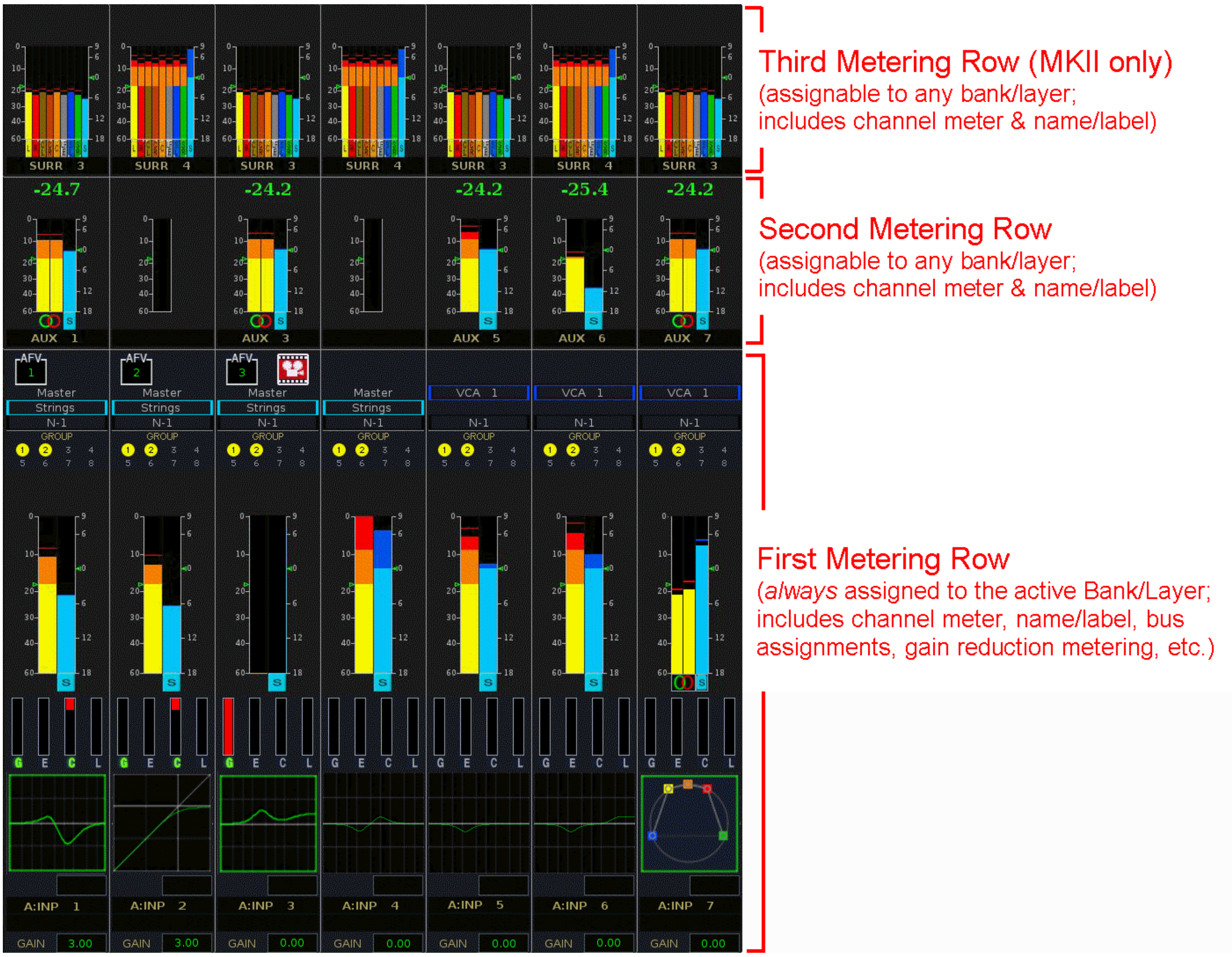
Colour coding is used within the Central Control Section and channel strip Free Controls so that EQ, Dynamics, Panning, etc. can be easily distinguished at a glance (MKII mc²56 only).

At the bottom of each fader strip, the **LAWO** backlight is colour-coded to indicate the channel type. This enables you to easily distinguish input channels (white) from groups (yellow), aux masters (green), VCAs (blue) and sums (red). Or, you may customise the channel colour coding - for example, music channels to be white, VTRs to be blue, presenter mics to be red and so on.

If you enable button-glow (MKII mc²56 only), then some fader strip buttons in their off state are dimly lit according to the channel colour. This makes channel identification even easier, especially in low-light conditions.

Flexible Metering

The Channel display provides metering, and other channel-related information, for up to three rows of channels:



For all on-screen meters you may choose to display peak metering, loudness metering, or a combination of both.

The peak bargraph meter may be switched to different points within the signal flow, and is mono, stereo or multi-channel according to the channel format. You may change the characteristics and scale for all peak meters across the console, and define colour coding to indicate a safe area (red), operating range (orange) and line up level (green arrow).

The loudness meter may be positioned independently from the peak meter. A single bargraph (blue) represents the average energy of the summed component channels: mono, stereo or surround. On summing channels, you may also start an integrated loudness measurement, displayed above the bargraph. This allows you to measure the loudness of summing channels over longer periods of time. One integrated loudness measurement, such as main programme, may be displayed in the title bar of the Central GUI. Loudness metering conforms to the ITU-R BS1770.

For more details, see Metering.

Integrated Routing Matrix

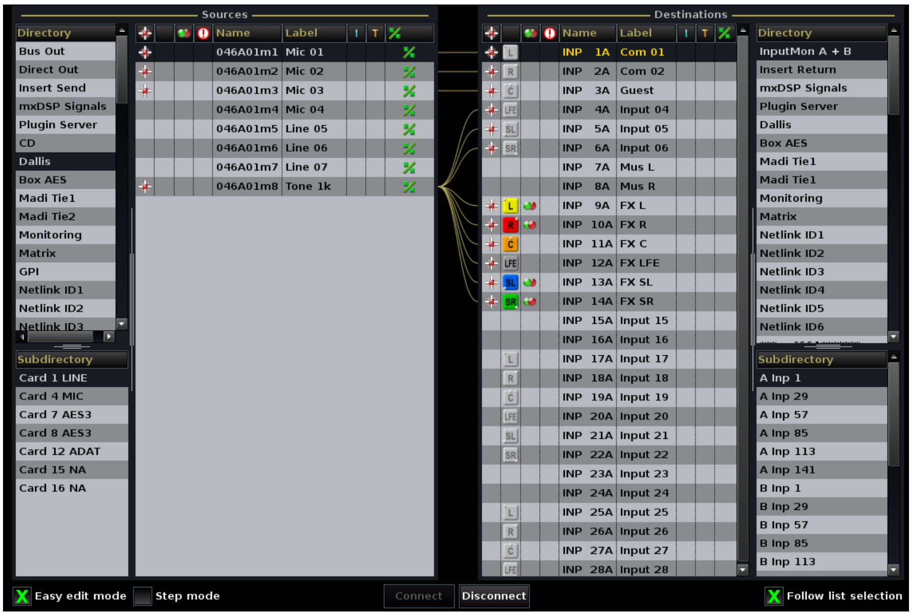
The **mc²56** includes an integrated digital routing matrix. Any source may be routed to any input or monitor channel, and any output bus or channel send routed to any destination. In addition, you may route sources directly to destinations, for example to feed a Mic/Line input to an AES output.

Multiple systems may also be networked in order to share I/O resources. For example, to share the same microphone input between two consoles.

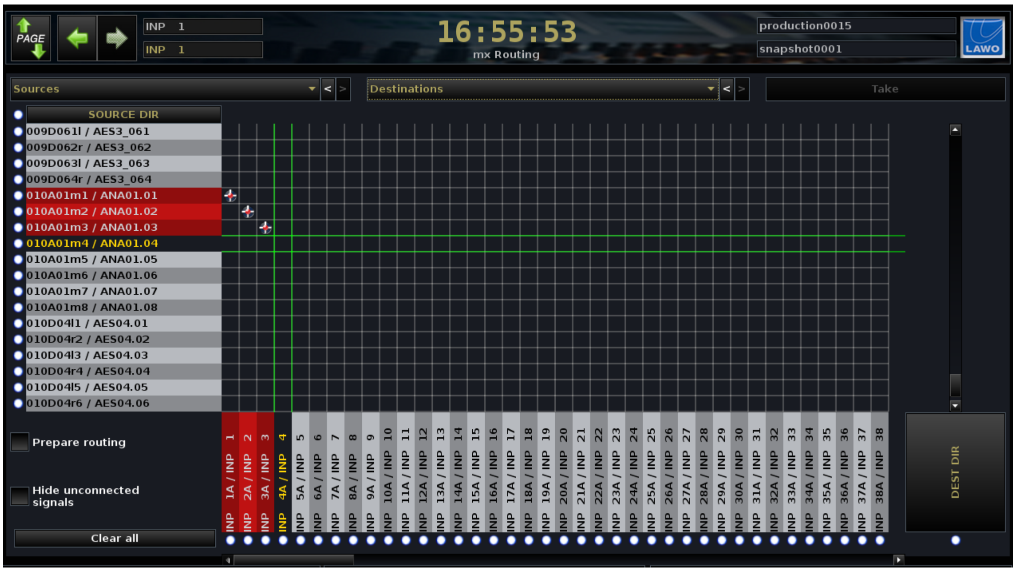
All routes are stored and recalled in productions and snapshots, reducing the amount of manual patching within the installation and saving hours of set up time!

Signal routing may be performed from either the Signal List or mx Routing displays:

Signal List Display



mx Routing Display



Console Reset

One of the major benefits of the **mc²56** is the ability to store and recall all the settings for a live show or type of application.

Productions

Productions form the top level for user data storage and store *all* the settings required for a production or type of job.

Productions store everything included in a snapshot, plus lower level settings such as the DSP configuration and system options. As a result, loading a production may cause a brief interruption to audio, and should *not* be used during a show. Instead, use snapshots to recall settings while live on-air.

Snapshots

Within each production, folders are created to store snapshots.

Snapshots store different mixes for recall before or during the show. For example, to recall a different mix for each band in a live entertainment show, or to recall scene changes during a live theatre production. To manage snapshot recall, snapshot isolate and filtering may be applied to protect channels or elements of the desk.

Sequences

Sequences are provided for convenient recall of snapshots during a live broadcast or theatre production.

A sequence is a list of snapshots which can be loaded in sequence during a live show. The transition between snapshots in a sequence can be cross faded if required. In addition, offsets can be applied to deal with last minute changes such as a change of artist. Note that the sequence itself does not store any settings, but simply creates a list of pointers to snapshots stored within the production folder.

Presets

Presets are stored independently of productions, and save and load settings for processing modules (EQ, Gate, Compressor, Panning, etc.) or for a complete channel. For example, you may wish to save your favourite Kick Drum EQ, or the complete settings for an announcer channel.

Transferring User Data

All user data is stored on the system's internal flashcard and may be imported or exported to a USB interface or mxGUI computer. In a networked installation, a central file server can be made accessible from each console within the network.

> User data is fully compatible with any mc² or Nova73, regardless of the hardware configuration. This enables the transfer of production data, snapshots, mixes or presets to and from any system (including any other mc²), in order to recall settings in a different studio.

mxGUI

mxGUI (Matrix GUI) is a software programme which runs on an external computer to provide offline setup or remote operation of any mc² system:

- **Offline Setup** - productions, snapshots, sequences, mixes and presets can be prepared and stored on the mxGUI computer, and then transferred to the system at a later date; thus saving valuable setup time before a show.
- **Remote Operation** - mxGUI can run online by connecting the mxGUI computer to the mc²56 Control System (via Ethernet). This provides additional screen displays or remote operation for a second engineer.

mxGUI runs an emulation of the mc² control system, providing identical displays to those found on the mc²56, 66 and 90 Central GUI. This enables the creation of a complete production offline, including signal routing, labels, fader strip assignments, processing settings, snapshots, sequences, etc.

For more details, see mxGUI.

Timecode Automation

The **mc²56** automation system automates console settings referenced to timecode, and is controlled from virtual automation panels (**VAP1** and **VAP2**) on the right of the Central GUI touch-screen:

Any channel type may be automated (inputs, groups, sums, auxes, VCA masters, surround VCA masters and GPCs). And automation may be enabled for any audio module (fader, mute, aux sends, EQ, bus routing, channel signal flow, etc.)

Automation data can be written with timecode rolling forwards, backwards and at any speed, providing fast and efficient mixing. The way in which data is written is governed by a number of modes, allowing you to write dynamic or static automation; step in or step out of write to make updates; trim existing moves; protect channels to prevent overwriting existing moves; and isolate channels to remove them from the automation system completely.

Each stream of automation data is recorded as a 'Pass', and multiple passes are stored within a 'Mix'. The 'Pass Tree' allows you to view the history and A/B between different passes within each mix. You can also edit mix passes in order to delete, copy, shift, insert or paste sections from different passes.

Multiple mixes may be created within each production; mixes are stored permanently on the system when you update or save a production.

Control of the playback machine may be programmed onto user buttons from the Custom Functions display, or handled from the optional Machine Control panel.

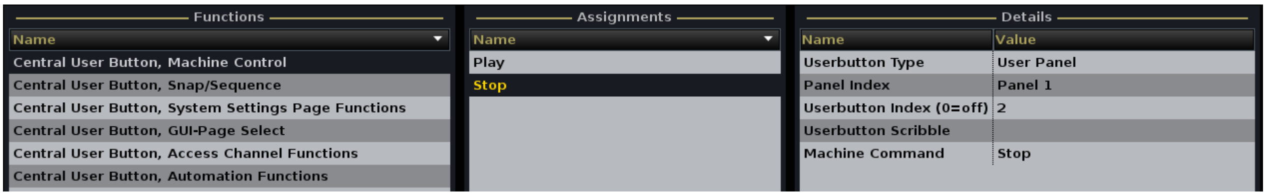
You may also use the Machine Locator display to store and recall cue points, and/or switch one of your console displays to a remote desktop in order to view and control a DAW.

Configuration

The **mc²56** can be customised by operators, technicians or Lawo personnel as follows:

Custom Functions

Functions such as user buttons can be re-assigned from the Central GUI using the Custom Functions display:



Custom functions are stored at a lower level to productions. This means that any changes will affect all users.

AdminHD

At a lower level (not accessible from the GUI) are a number of files which configure the system's hardware and define settings such as the sampling frequency, and the organisation of signals within the Directories and Subdirectories of the **Signal List** display. The AdminHD configuration is an essential part of the system. If a hardware component is not defined within the configuration, then it will not be visible to you even if it is powered and connected. In other words, the configuration is always the 'master' of the system, regardless of what physical components are added or removed.

The configuration is not designed to be changed by an operator, but can be edited by your systems engineer using a software application called AdminHD. For example, if a DALLIS stagebox is hired in for a production, then the unit must be added to the configuration and uploaded to the system before the signals and parameters become available to the operator.

For more details on the AdminHD configuration, see the "mc²56 Technical Manual".

TCL Functions

At a lower level than **AdminHD**, a number of other options may be factory-configured using TCL (Tool Command Language). TCL functions can only be programmed by Lawo personnel, and are designed to provide some flexibility at the specification stage. TCL allows the logical interlinking of GPIs, soft keys and events. For example, tally states, automated input allocation and fader starts can all be programmed using this protocol. Console monitoring is also handled by the TCL protocol.

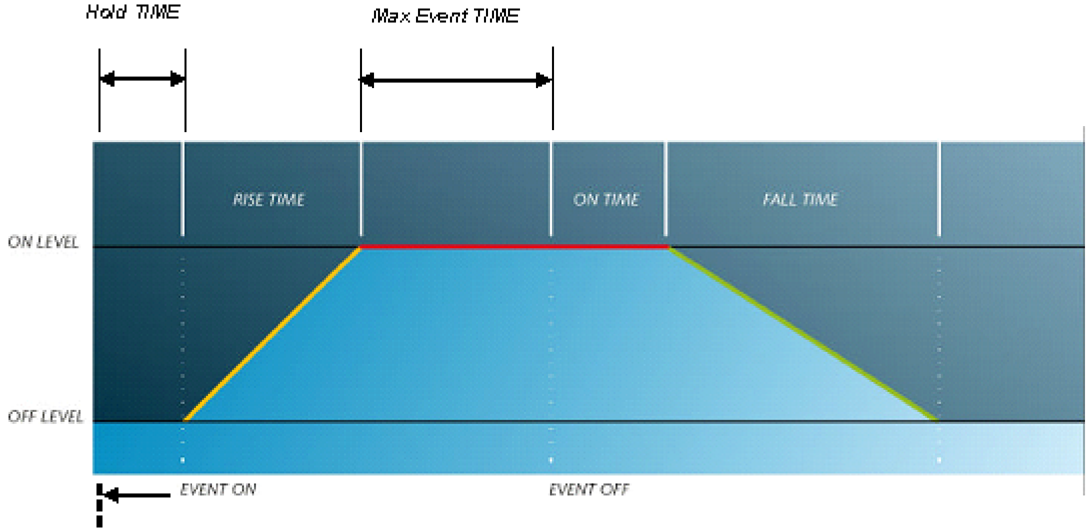
Integration with the Outside World

In modern production environments, communication between the individual components in an audio system takes on more and more importance. Here is an overview of some of the applications supported by the **mc²56**:

Audio Follow Video (AFV)

The **mc²56**'s Audio Follow Video provides the ability to open and close a channel or main fader from an external event, received via TCP/IP Ethernet (using Lawo's Remote MNOPL protocol) or GPIO. For example, during coverage of a live motor racing event, you may programme the audio channels associated with each camera to automatically open and close as the picture cuts between different shots.

Up to 128 events may be programmed, with each event corresponding to a different camera tally. An event can control an individual channel or a group of channels. Parameters for the Hold Time, Rise Time, Max Event Time, On Time and Fall Time control the envelope of the fade allowing smooth fades from one camera to another.

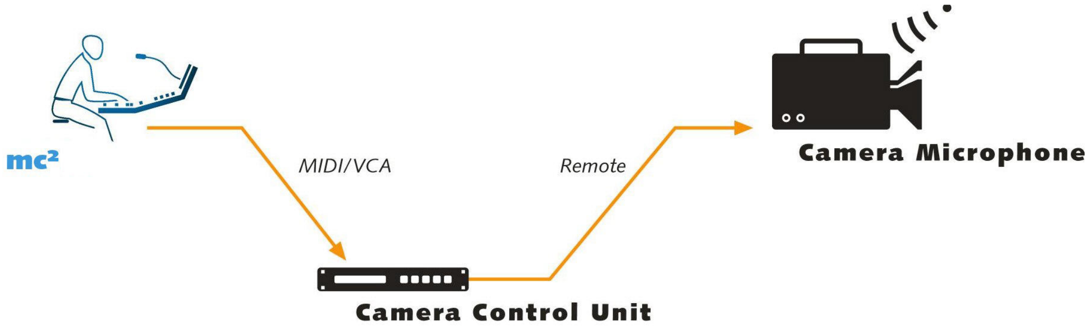


For more details, see Audio Follow Video.

General Purpose Channels (GPCs)

GPCs (General Purpose Channels) are control channels, assigned to any fader strip, which provide remote control of external devices (via MIDI). Typical applications include:

- Adjusting and storing camera microphone levels via MIDI to VCA converters:



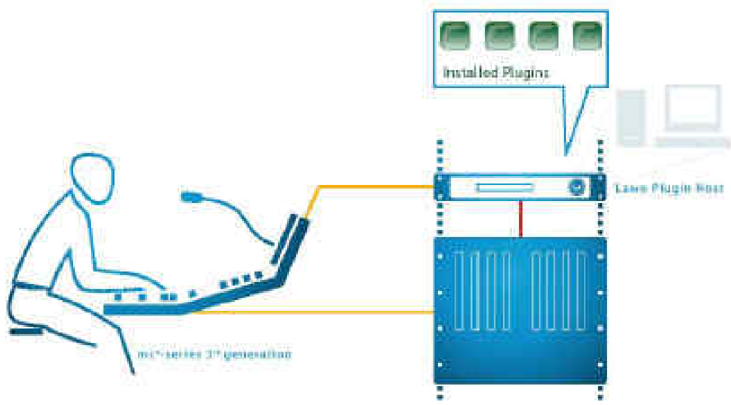
- Adjusting fader levels and other parameters within a digital audio workstation (DAW):

For more details, see General Purpose Channels.

Plugin Server

The LAWO Plugin Server allows plugins to be controlled from the mc² console, and all settings to be stored and recalled by a production or snapshot.

All Plugins are hosted on an external host, with audio connections to/from the Nova73 via 64-channel MADI and control connections via Ethernet:



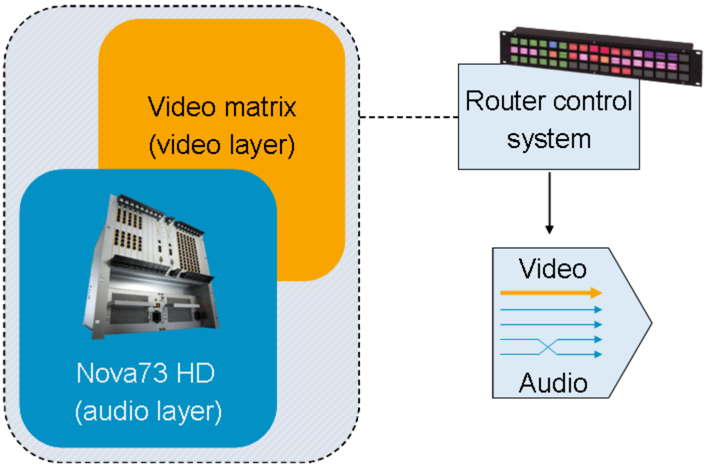
An APC UPS ensures that the Plugin Host shuts down automatically after switching off the power supply of the system.

Plugins are assigned and controlled from the **Plugin** display. Note that this display is only available if the Plugin server is configured for your system. Please refer to the separate "Lawo Plugin Collection Operators Manual" for more on operation, and the “Plugin Server Technical Documentation” guide for configuration.

Remote MNOPL Control

Lawo's Remote MNOPL protocol is a freely available Ethernet (TCP/IP) protocol providing control of virtually any system parameter from an external device.

A typical application is to provide third-party matrix control so that crosspoints within the **mc²56**'s routing matrix can be controlled by external control systems such as VSM, Evertz, Quartz, BFE, Pharos and others. (If your preferred supplier does not support the protocol, then please ask them to contact Lawo for further details):



Within your AdminHD configuration, each signal may be given a mapping address. Up to 16 different mapping tables can be defined so that different control systems can be supported simultaneously.

For details on implementation, please refer to the "mc²56 Technical Manual".

Remote Desktop

Any of the console's TFT displays (Channel or Central GUI) may be switched to a remote server in order to view and control other applications – for example, a VSM playback system or DAW.

The remote server connects to the Lawo control network via Ethernet. This function is programmed from the Custom Functions display.

Lawo Remote App

The Lawo Remote App is a free App which allows you to operate any fader of a mc² console, recall snapshots and control user-defined functions remotely from an iPhone, iPod or iPad. From the App you have access to the following:

- **Fader level, Mute and Metering** – for any fader assigned to the active Bank and Layer.
- **Snapshots** – load any Snapshot from any folder within the active production.
- **User Buttons** – a special page of buttons allow you to control user defined functions such as monitoring, GPI control, etc. The button assignments are made from the Custom Functions display and stored as part of the console configuration.

For more details, see the Lawo Remote App.

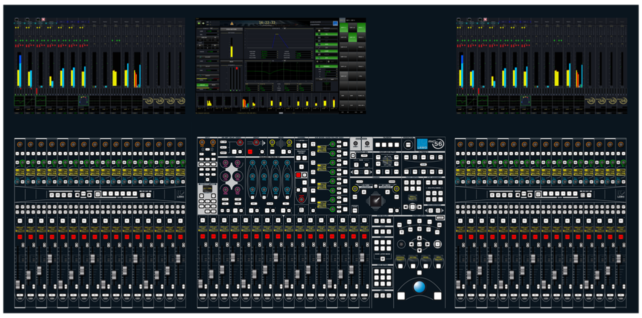
Machine Control

The optional Recording Com Kit provides Sony 9pin, LTC and MIDI connections to an external playback device. Machine control functions may be mapped onto user buttons from the Custom Functions display, or handled from the optional machine control panel, mounted externally from the console. The console's automation system slaves to timecode from the active port. For more details, see Timecode Automation.

Classic vs MKII - the Differences

Classic mc^{2}56

MKII mc^{2}56



The classic mc^{2}56 differs from the MKII as follows:

- External metering or user panels cannot be fitted to the right of the central GUI, see Overbridge Options.
- No local I/O cards or talkback mic preamp are integrated within the control surface.
- The Central Control Section DYNAMICS has less rotary controls.
- Rotary controls are not colour coded.
- Button-glow is not supported.
- Channel strip Free Controls provide rotary control only (there is no on/off function button).
- ISO Bay cannot be used to spread multiple DSP parameters across the isolated bay Free Controls.
- The fader strip user buttons cannot be paged (4 functions may be programmed, as opposed to 12).
- The classic mc^{2}56 supports two rows of multi-row metering (as opposed to three rows on the MKII).
- The centre section has 6 user buttons (as opposed to 9).
- The centre section includes physical buttons for ACCESS CHANNEL/ASSIGN (on the MKII, operation is from the touch-screen).
- The MKII control surface benefits from simpler construction and internal wiring, and reduced height, width and weight.

Due to the differences in control surface construction, you cannot fit classic mc^{2}56 panels into the MKII or vice versa, or add a MKII Extender bay to a classic control surface.

In all other respects, the systems are identical, allowing you to swap Nova73 and/or DALLIS components, and exchange all user data.

Chapter 2: Getting Started

Introduction

This chapter introduces the operating principles and guides you step-by-step through some common operations. The objective is not to teach every single detail, but to introduce the basics. For more in depth knowledge, please refer to the later chapters.

We are assuming that your console is fully commissioned such that a suitable AdminHD configuration has been transferred and all User Buttons are labelled.

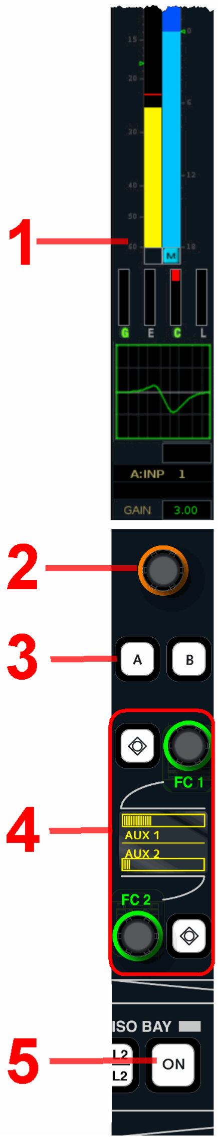
Topics covered in this chapter are:

- Fader Strip Quick Reference
- Centre Section Quick Reference
- The Central GUI
- SCREEN CONTROL Operation: including soft keys; trackball and console keyboard.
- ACCESS CHANNEL/ASSIGN
- Powering On
- Loading a production
- Interrogating the Fader Strips
- Adjusting Input Gain
- Monitoring Audio
- Creating Your Own Configuration: DSP Configuration; Signal List; Fader Strip and Bus Assign
- Saving, Transferring & Loading Settings
- Using Auxiliary Sends
- Creating a Mix Minus (N-1)
- Configuring Audio Sub Group Masters
- Using VCA Grouping
- Applying Signal Processing: EQ; Compressor; Delay; Inserts
- Using Free Controls
- Next Steps

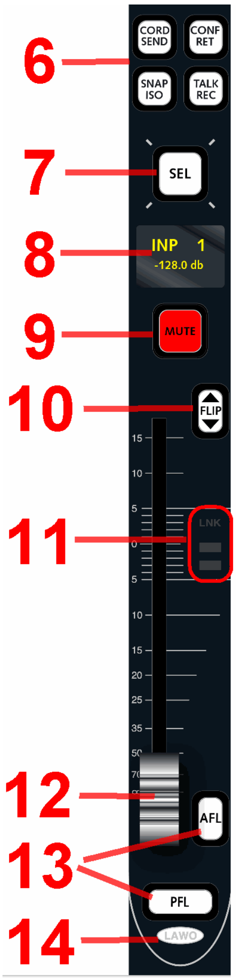
Fader Strip Quick Reference

Each channel fader strip on the **mc²56** provides:

1	Channel display	A high resolution touch-screen display providing metering, feedback on bus assignments and local parameter values. You can touch the screen to edit bus and VCA assignments, or change the meter pickup point and mode.
2	Input Gain	This control is dedicated to source gain (mic/line or digital). The amount of **GAIN** is shown on the Channel display.
3	A/B Input Switching	For any input channel, you may assign two sources (A and B) to provide a main and backup source for the channel. Press the input select buttons to switch between the two sources.
4	Free Controls	The Free Controls may be assigned to any DSP parameter, providing local control of key functions for each source. Controls are colour-coded (MKII only), making it easy to distinguish between Auxes (green), EQ (blue), etc. Free Controls may also be switched globally using FC PRESETS, and locally within the 16-fader bay to provide expanded parameter control.
5	ISO BAY panel	ISO BAY **ON** isolates the 16-fader bay from the main console. It can be used for multiple operators, and/or to provide expanded local parameter control for the selected channel.

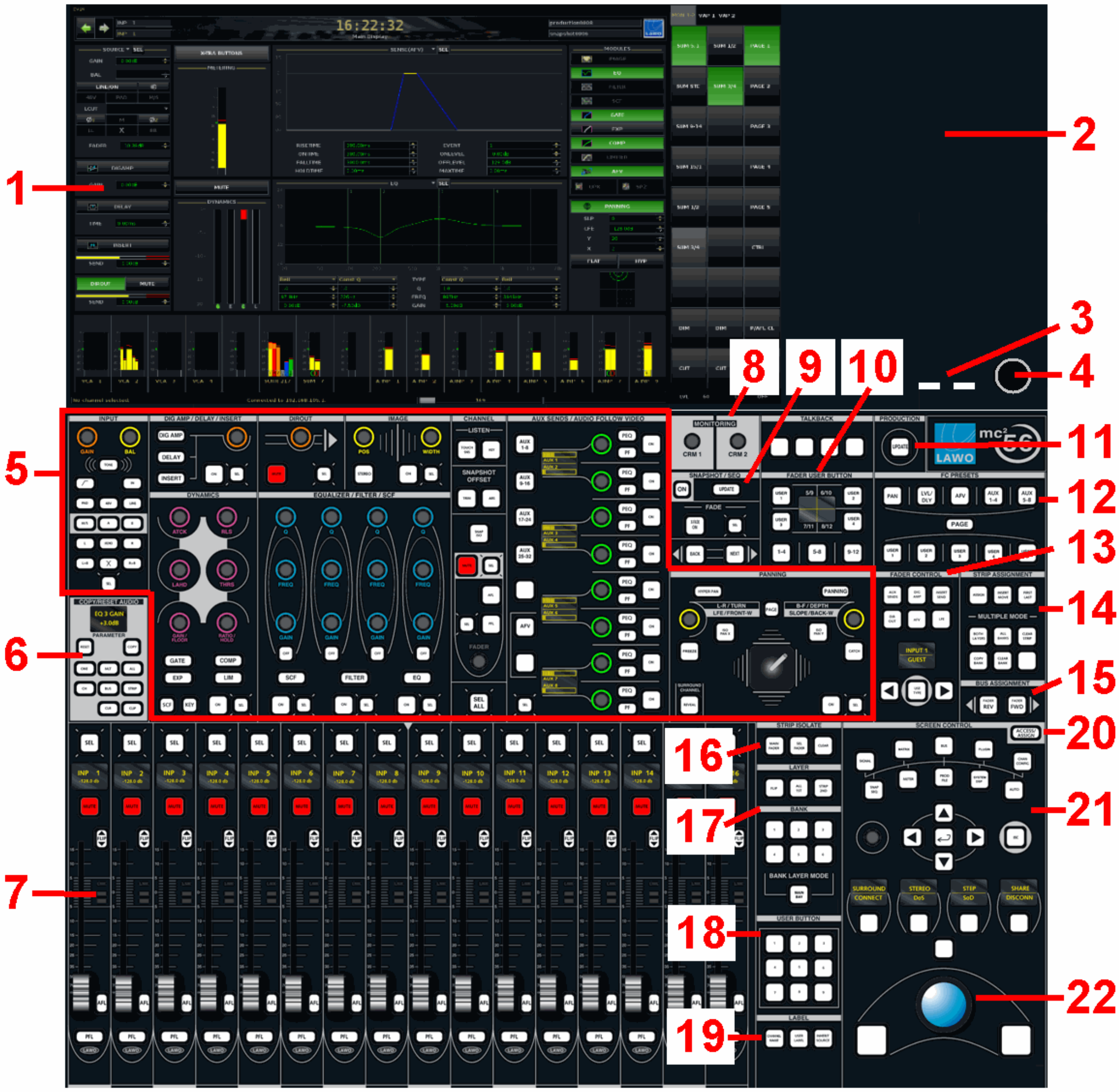


6	User Buttons	These buttons are programmed from the **Custom Functions** display. Applications include mix minus control, snapshot isolate and talkback.
7	SEL	Fader strip Select button. This button selects the channel for a variety of operations, including adjusting parameters from the Central Control Section, bus assign, etc.
8	Label	An 8-character display which shows the name or label of the channel assigned to the fader strip.
9	Mute	Press the **MUTE** button to mute (cut) the channel.
10	Layer FLIP	Press **FLIP** to switch the fader strip from Layer 1 to 2, or vice versa.
11	Status LEDs	**LNK** - lights if any processing modules within the channel are linked. **Signal Present** - these two LEDs light in different colours to show that signal is present. The LEDs always monitor the channel input, regardless of the peak meter pickup point.
12	Level	The fader is touch sensitive providing gain control from –128dB to +15dB.
13	AFL & PFL	Press **AFL** to listen to the post-fade channel signal. Press **PFL** to listen to the pre-fade channel signal. The listen busses may be switched to different outputs from the Monitoring Section.
14	LAWO backlight	The **LAWO** backlight is colour-coded to indicate the channel type. Colour codes may be customised, or set to the default (inputs = white, groups = yellow, sums = red, VCAs = blue, etc.)



For full details, see The Channel Display and The Channel Fader Strip.

Centre Section Quick Reference



1	Central GUI	A high resolution touch-screen providing access to a range of displays. The central GUI may be operated via the touch-screen, trackball, SCREEN CONTROL panel or console keyboard.
2	Overbridge Options	Space for optional RTW metering and/or a Lawo User Panel.
3	USB ports x 2	Connect the console keyboard for naming operations, or a memory stick for data import/export.
4	Talkback Connector (XLR)	Connect an external mic to feed the integrated talkback microphone preamplifier.
5	Central Control Section	Channel control - INPUT, DYNAMICS, EQUALIZER, etc. Select a channel, by pressing its fader strip **SEL** button, and then reach out to control any parameter.

6	Copy/Reset/Assign	Used to copy and reset channel parameters, or to assign parameters to the fader strip Free Controls.
7	Main Fader Strips	Identical to channel fader strips, except no input control, Free Controls or user buttons.
8	Monitor Level Controls	For the Control Room Monitor (CRM) outputs 1 and 2. Source selection, Monitor Dim, Cut, etc. are available from the touch-screen.
9	Sequence Controls	To play out a pre-prepared list of snapshots. Transitions may cross-fade and offsets may be applied.
10	Fader User Button Control	Switch the four fader strip user buttons through three pages of functions (User 1-4, 5-8 and 9-12).
11	Production Update	Press this button to store the current settings into the active production; the button flashes as a reminder to save.
12	Free Control Presets	Recall a preset to temporarily override the fader strip Free Controls, and access parameters globally across the console (e.g. Aux Sends).
13	Fader Control of Levels	Press a button to temporarily assign an aux send, direct output level, etc. onto the console's faders.
14	Fader Strip Assignment	Used to assign any channel type to a fader strip. You can make single or multiple assignments, across one or more banks and layers.
15	Bus Assign	Forward or Reverse assign, for routing channels onto mix busses (sums, groups, track busses, auxes) or VCAs.
16	Strip Isolate	To isolate fader strips from Bank switching.
17	Bank & Layer Switching	Global Bank and Layer switching (6 Banks, each with 2 Layers).
18	User Buttons	12 User Buttons programmed from the **Custom Functions** display.
19	LABEL Switching	Switch the fader strip labels between the channel system name, channel user label or inherited source label.
20	ACCESS/ASSIGN	Press this button to display the ACCESS CHANNEL/ASSIGN panel on the Central GUI touch-screen (in place of the monitoring buttons).
21	SCREEN CONTROL	Dedicated buttons to access all Central GUI displays, plus navigation controls and soft keys to select and adjust screen-based options.
22	Trackball & Left/ Right Select	Left-click to enable or disable an on-screen function. Right-click to view additional options.

For full details, see The Centre Section.

The Central GUI

The Central GUI may be operated via the touch-screen, trackball, SCREEN CONTROL panel or console keyboard, and is divided into the following areas:

1	**Title Bar**	Across the top you will always see the channel in access, the time (local time, timecode or integrated loudness), and the name of the current production and snapshot.
2	**SCREEN CONTROL displays**	The main area of the GUI works in conjunction with the SCREEN CONTROL panel. Here you can page through displays for Signal routing, Snapshots, productions, etc.
3	**Main Fader Metering**	Below are meters for the 16 main fader strips. Note that this area may be enabled or disabled from the **System Settings**.
4	**Status Bar**	The status bar provides feedback on the IP address of the system, the amount of used data storage space (%), the progress of operations such as load and save production, the software release version and the console PSU status.
5	**Touch-screen Buttons**	On the right are buttons for monitoring, automation and ACCESS/ASSIGN functions.

The title bar (1), main fader metering (3), status bar (4) and touch-screen buttons (5) remain visible at all times, regardless of the selected SCREEN CONTROL display (2).

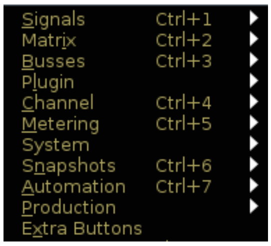
For clear feedback of information, there are no floating windows.

Title Bar (Headline)

The title bar contains some common elements:

➢ PAGE Menu

Select the **PAGE** button (top left) to access all the SCREEN CONTROL displays:



You can also use the SCREEN CONTROL panel or console keyboard for fast access to displays.

➢ Next/Previous Page Buttons

These on-screen buttons work just like the Forward and Back buttons on a web browser.

If you have viewed say the **DSP Configuration**, then the **Snapshots** list, and then the **Main** display, you can use the previous Page button to step backwards through this sequence of displays. The last 16 pages viewed are stored. If you reach the first or last page in the sequence, then the button turns grey indicating no further selections are available.

➢ Information

The title bar always shows:

- The name and user or source label of the channel in access – **INP 1**, **Kick**.
- The title of the selected display – **Signal List**.
- The name of the active production – **production0015** - and the current snapshot if loaded – **snapshot0014**.

You can edit the user label of a control channel, such as a VCA master, by clicking in the label field. (Note that the centre section LABEL buttons must be switched to **USER LABEL**.)

For DSP channels, such as an input channel (INP), labels are edited from the Signal List display.

➢ Time / Integrated Loudness

The headline in the title bar can show either **Timecode** or **Loudness**. Click on the headline to make your selection; the sub menu options update accordingly:

Having selected **Timecode display**, you can choose from:

- **Local** – displays the local system time in 24 hour clock.
- **Timecode** - displays SMPTE timecode from your selected timecode reference.
- **Offset TC** - displays SMPTE timecode + the Midnight offset.

Alternatively, select **Loudness metering** to display the integrated loudness measurement for a particular summing channel (in LUFS). Use the sub menu options to **Start/Stop** or **Reset** the integration:

➢ Warning Icons

You may also see:

- A yellow hazard warning flag, if there is a problem with the system status - see Diagnosing System Errors.
- The keyboard locked icon, if the console keyboard is locked.

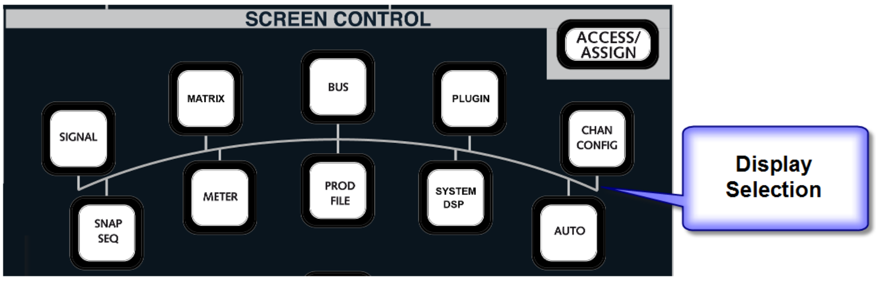
➢ The LAWO Logo

From V4.24 software onwards, click on the **LAWO** logo to manually timestamp the system logfile. This marks the **messages** file at a moment in time, and can assist Lawo's service department when diagnosing system behaviour. You can copy logfiles from the system via the File display.

SCREEN CONTROL Displays

The central working area may be paged to show different displays. One display is *always* active and its name is shown in the Title Bar - for example, the **Signal List** or **Main Display**:

To change display, use the SCREEN CONTROL panel, the Page menu or a keyboard "hot key".



Most buttons access more than one display, so keep pressing to cycle through the available pages:

Button	Display	Description
SIGNAL	Signal List	control signal routing.
"	Signal Settings	adjust I/O parameters, and check the system status.
MATRIX	mx Routing	crosspoint control of signal routing.
"	mxDSP	control DSP settings on the optional mxDSP modules.
"	Downmix	control downmix matrix parameters.
SNAP SEQ	Snapshots List	load, save and manage console snapshots.
"	Sequences	used to create and run real time sequence automation.
"	Snapshot Trim Sets	used to manage snapshot offset parameters.
AUTO	Mixes	load and manage timecode automation mixes.
"	Passes	used to manage passes of timecode automation within the active mix.
"	Machine Locators	create a cue list by storing, naming and recalling timecode locators.
METER	Meter 1 to 5	four pages of assignable meters, plus a fifth page which meters the main faders.
BUS	Bus Assign	view or change bus assignments from the channel in access.
"	Busses Reverse	interrogate bus assignments made to the channel in access.
PROD FILE	Productions	manage the console's productions.
"	File	import or export productions to/from USB or a network server.

Button	Display	Description
SYSTEM DSP	System Settings	set console options.
"	DSP Config	view or change the DSP Configuration.
"	Custom Functions	configure user buttons for custom functions.
PLUGIN	Plugin setup	access to the remote plugin server setup (optional).
"	Plugin Edit	access to plugin server editing (optional).
CHAN CONFIG	Main Display	view parameters for the channel in access.
"	Channel Config	adjust the signal flow for the channel in access.
n/a	Extra Buttons	access to touch-screen buttons for additional options. Note that there is no dedicated SCREEN CONTROL button to access this display. Instead, use the **Xtra** touch-screen button, or programme a centre section user button.

Main Fader Metering

Below the main display area, you may view metering for the main fader strips:

This mini display can be enabled or disabled from the **System Settings**, see Display Central Metering.

It shows peak metering, loudness metering, or a combination of both for each channel assigned to the main fader strips; the meters follow the same options as the Channel display.

To see more detail for the 16 main fader strips, such as bus assignments, use the dedicated Main Fader Metering display.

Touch-screen Buttons

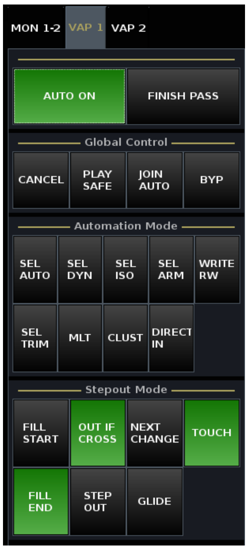
The 24 touch-screen buttons, on the right of the Central GUI, display either monitoring/automation functions or the ACCESS/ASSIGN control panel. In each case, touch a button to action a function; it turns green when selected.

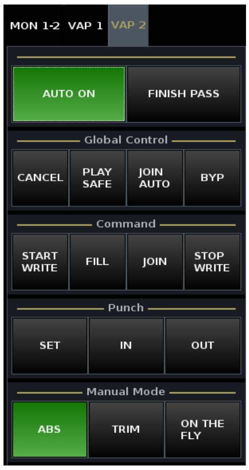
➢ Monitoring/Automation

The buttons default to monitoring and automation; use the tabs to page between:

- **MON 1-2** – monitoring functions.
- **VAP 1** – Virtual Automation Panel 1 (timecode automation functions).
- **VAP 2** - Virtual Automation Panel 2 (more timecode automation functions).

> Select the **X-tra** button, in the **MON 1-2** page for fast access to the Extra Buttons display.

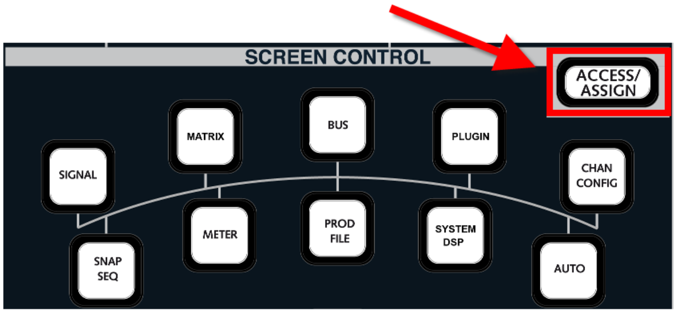




➢ **ACCESS/ASSIGN** (MKII mc²56 only)

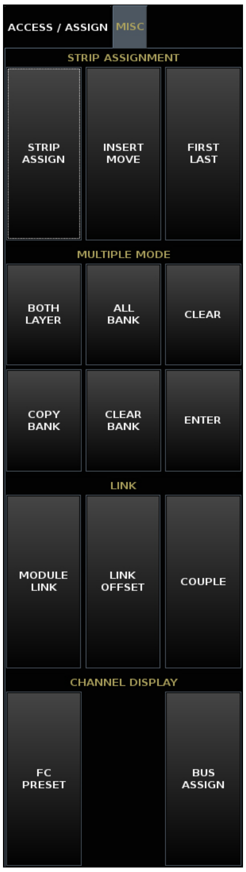
On the MKII mc²56, the ACCESS/ASSIGN panel also appears in this touch-screen area. To access the buttons:

1. On the SCREEN CONTROL front panel, press **ACCESS/ASSIGN**:



The touch-screen buttons update and you will see two new tabs:

- **ACCESS/ASSIGN** – the ACCESS CHANNEL/ASSIGN control panel.
- **MISC** – a range of functions, duplicated from the front panel and Extra Buttons display. These are provided for convenience, as they complement the ACCESS/ASSIGN selection.



2. Deselect **ACCESS/ASSIGN** (on the SCREEN CONTROL front panel) to return the touch-screen buttons to monitoring and automation.

SCREEN CONTROL Operation

The SCREEN CONTROL displays are divided into clearly defined areas – for example, in the **Snapshots** display, there are areas listing **Folders**, **Snapshots** and for entering a **Snapshot Memo**:

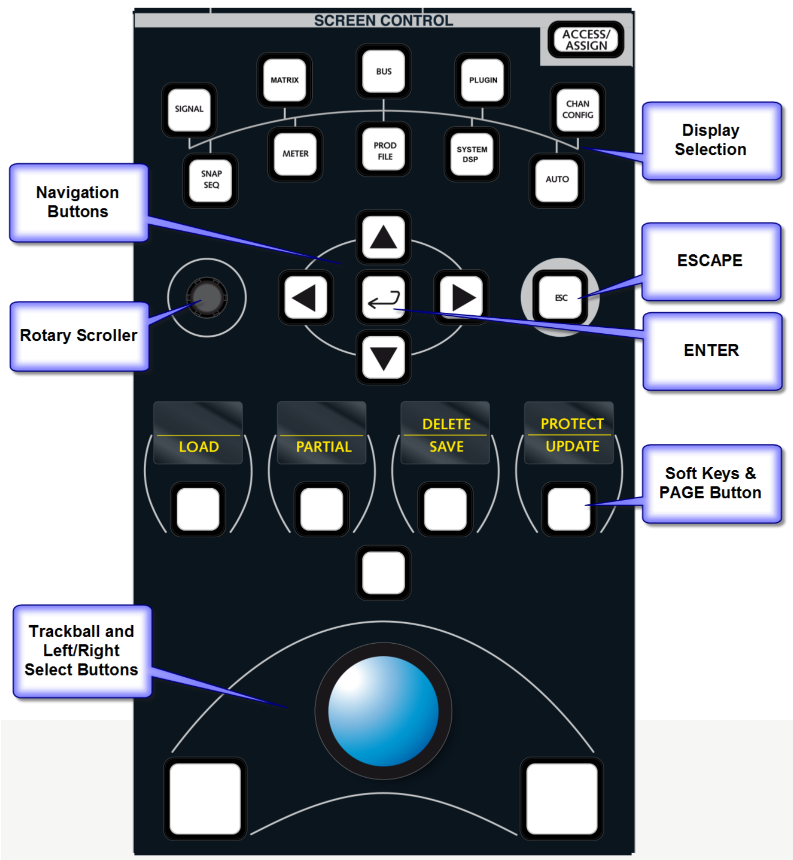
Within the **Folders** and **Snapshots** lists, selections are highlighted in black – our selected folder is **Music** and the selected snapshot is **Act 1 Scene 2**.

Screen buttons which perform an operation are always bevelled with white text – for example, **Save**, **Save partial**, **Load**, etc.

For most operations, you make a selection, or 'focus' on an area of the display, and then select one of the on-screen functions, or press a SCREEN CONTROL panel soft key.

For example, to load a snapshot:

1. Select the snapshot.
2. Then touch the on-screen **Load** button, or press the **LOAD** soft key.



Making Selections and Focussing the Display

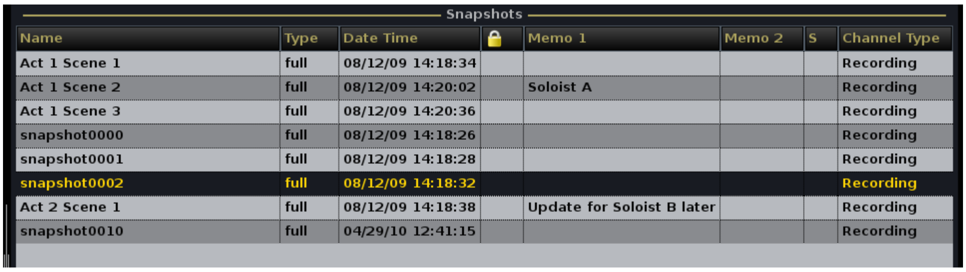
Within each display, there are four possible ways to make a selection:

➢ Using the Touch-Screen

Anything which is a button or menu option can be selected by touching the screen.

➢ Using the Trackball

To use the trackball, position the cursor above the name in the list and press the left select button. The selection – e.g. **snapshot0002** - highlights in black:

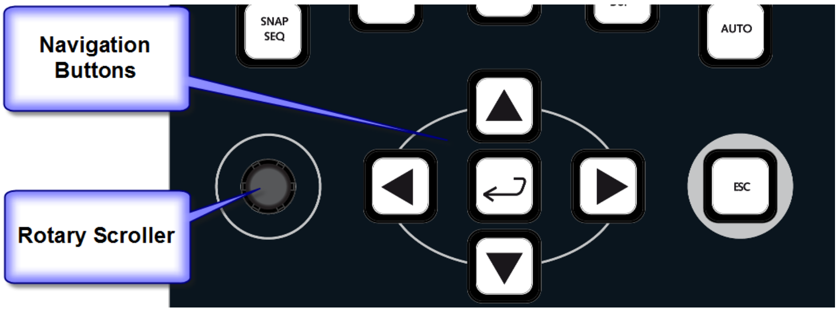


Snapshots

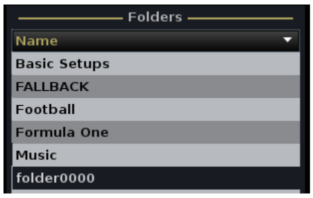
Name	Type	Date Time	🔒	Memo 1	Memo 2	S	Channel Type
Act 1 Scene 1	full	08/12/09 14:18:34					Recording
Act 1 Scene 2	full	08/12/09 14:20:02		Soloist A			Recording
Act 1 Scene 3	full	08/12/09 14:20:36					Recording
snapshot0000	full	08/12/09 14:18:26					Recording
snapshot0001	full	08/12/09 14:18:28					Recording
snapshot0002	full	08/12/09 14:18:32					Recording
Act 2 Scene 1	full	08/12/09 14:18:38		Update for Soloist B later			Recording
snapshot0010	full	04/29/10 12:41:15					Recording

➢ Using the Navigation Controls

Alternatively, use the rotary scroller or Up/Down navigation buttons:



1. In our example, turn the rotary control to scroll up or down the list of snapshots.
2. Or, press the Up or Down arrow buttons to step up or down the list.
3. The Left/Right navigation buttons change which part of the display is in focus.



For example, press the Left arrow button to move focus to the list of Folders. Now turn the rotary scroller, or press the Up/ Down arrows, to move through the Folders list.

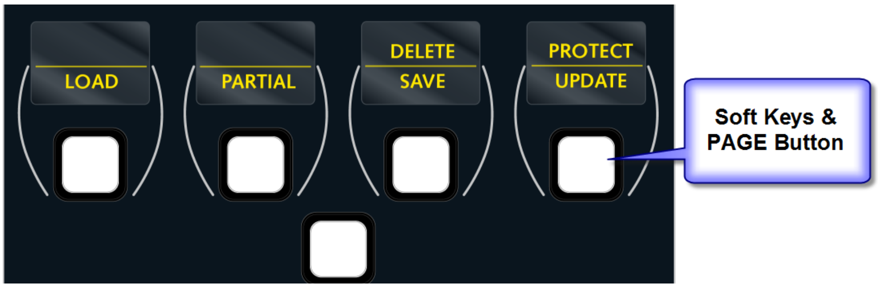
Note that the Right arrow cycles around the display in a clockwise manner, and the Left arrow in an anti-clockwise manner.

➢ Using the Console Keyboard

You may also use "hot keys" to make selections or change focus, see the Console Keyboard.

Soft Key Operations

Having made a selection, or focused on a new area of the display, the soft keys update to offer a variety of operations - in our example, to **LOAD** the selected snapshot, save a **PARTIAL** snapshot, etc:



1. To access the second level of functions – **DELETE** and **PROTECT** – press the **PAGE** button (this is the central button below the soft keys).

The displays update so that you can see which soft key to press for each operation.

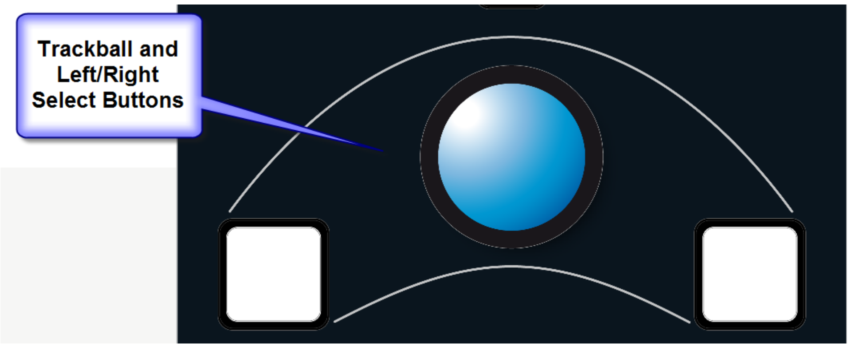
2. Deselect **PAGE** to go back to the first level.

The soft key functions vary depending on your choice of display and the area which is in focus. So, if you're struggling to find the correct soft key function, try focusing on a different area of the display.

Most soft key functions are duplicated on-screen, either as a dedicated touch-screen button or context menu option.

Context Menus (right-click)

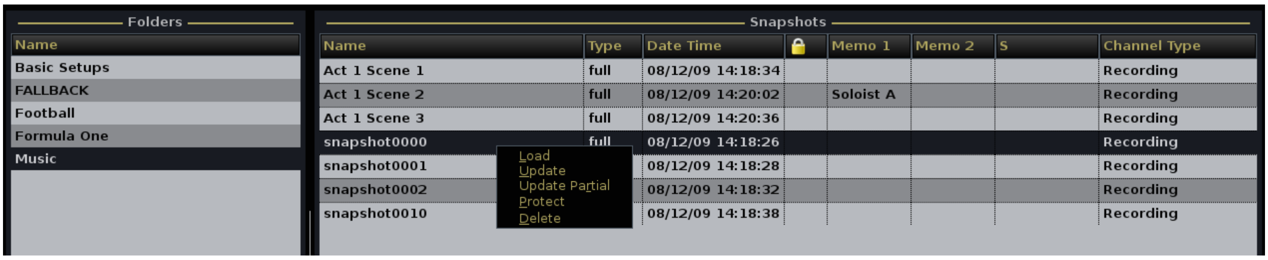
Many soft key functions appear on-screen when you right-click on a selection. Or, from V4.24 software onwards, touch the screen for a longer period of time:



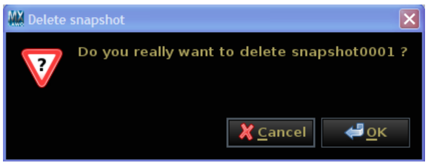
1. For example, select a snapshot and press the right select button. Or touch the snapshot name for about a second.

The snapshot context menu appears.

You can now **Load**, **Update**, **Protect** or **Delete** the snapshot:



You are usually asked for confirmation when performing screen-based operations:



Context menus appear larger on the Central GUI than on mxGUI to aid touch-screen operation.

Other Trackball Operations

There are some functions which can only be performed using the trackball.

1. Screen buttons are often used at the top of lists – for example, you can sort the **Snapshots** list differently by selecting **Name**, **Date Time**, etc:

2. You can resize a window by selecting and dragging the grey separator bar – for example, to widen the **Folders** list, position the cursor above the grey separator bar, then press and hold the left select button while dragging to the right; the **Folders** and **Snapshots** windows resize accordingly. Note that if there is no grey separator bar, then resizing is not possible.

3. You can also change the order of columns within a list – for example, to move the padlock (protection) column, position the cursor above the column title, then press and hold the left select button while dragging the column to the left or to the right. Release the left select button when you are happy with the new position of the column.

Note that any changes you make to window sizes and list orders will be reset after a console restart.

4. If information within a window is hidden, then left/right or up/down scroll bars will automatically appear. Select a scroll bar at the bottom to scroll left/right or up/down.

➢ Adjusting Parameter Values

On some displays, such as the **Main** display, you can use the trackball to change parameter values:

1. Click on the up or down arrows beside the parameter (e.g. **GAIN**) to adjust its value.

The Console Keyboard

Any available USB port may be used to connect the console keyboard. The keyboard is used to enter names. In addition, it can select a different display, make selections or adjust parameter values.The console keyboard can be disabled (and enabled) as follows:

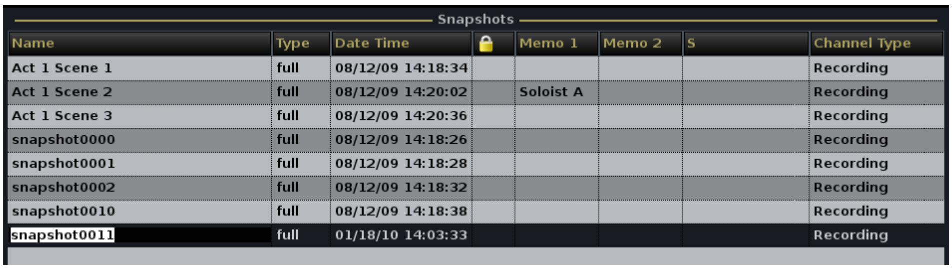
1. Press and hold **Fn** and then press **ON**.

When the keyboard is disabled, you will see "kbd locked" in the title bar of the Central GUI.

➢ Naming

1. First make your selection – e.g. select a snapshot.
2. Then do one of the following:

- Click once on the snapshot name using the trackball select button – all the existing text is selected (white) so that when you type you will automatically overwrite the existing name:

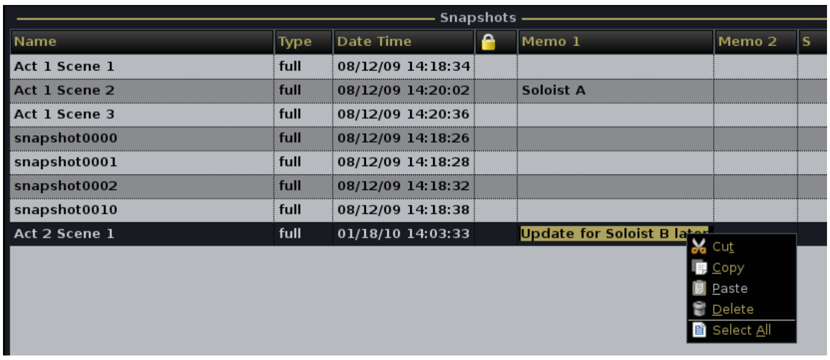


- Or, click twice to edit the existing name – a cursor appears at the end of the text (black) allowing you to easily append or modify the old name.

3. When you have finished, press the **Enter** button on the keyboard to confirm the new name.

4. Or, if you make a mistake and want to exit the naming mode without making any changes, press **ESC**.

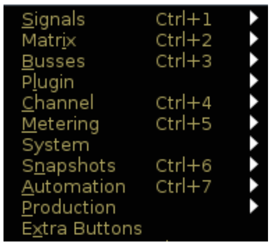
Note that if you right-select a text field, you will access **Cut**, **Copy**, **Paste**, **Delete** and **Select All**:



Use these options to copy and paste text from one field to another – for example, to copy and paste snapshot memo text.

Or press **CTRL+C** or **CTRL+V** to copy and paste selections.

➢ Selecting a Different Display



1. Press **ALT** + **P** to open the **Page** menu. Then press an underlined letter to select a display – for example, **S** to open **Signals**, **M** to open **Matrix**, etc.

2. Use any of the key combinations shown. For example, press [**CTRL**] + [**1**] to cycle through the available Signals displays: **Signal List** and **Signal Settings**.

3. Press [**ALT**] + [**Cursor Left**]/ [**Cursor Right**] to operate the next or previous Page buttons.

➢ Making Selections

1. Press **TAB** or **Shift**+**TAB** to change the focus area of the display – for example, to move from the list of **Snapshots** to **Folders** on the **Snapshots List** display:

Note that **TAB** cycles around the display in a clockwise manner, and **Shift**+**TAB** in an anti-clockwise manner.

2. Then use the Up and Down keyboard buttons to step through the entries in the list.

➢ Adjusting Parameter Values

On some displays, such as the **Main** display, you can use the keyboard to change parameter values:

1. Press **TAB** (or **Shift+TAB**) to focus on a parameter – for example, input **GAIN**.

2. Then use the Up and Down keyboard buttons to change the value, or type in a new value.

Keyboard Shortcuts (Hot Keys)

Below is a summary of all "hot key" functions. They can be used from the console keyboard, or from an external computer when operating mxGUI.

Global "Hot Keys":

- [**ALT**] + [**Cursor Left**]/ [**Cursor Right**] - operate the next or previous Page buttons.
- [**CTRL**] + [**1**] to [**8**] - cycle through the available SCREEN CONTROL displays.
- [**ALT**] + [**P**] - opens the Page menu. Then press [**S**] to open **Signals**, [**M**] to open **Matrix**, etc.
- [**TAB**] or [**SHIFT**] + [**TAB**] - change the focus area of the display.
- [**Cursor Up**]/ [**Cursor Down**] - step through entries in lists; if a parameter value is in focus, they adjust the value.
- [**CTRL**] + [**C**]/ [**V**] - when the contents of a text field are selected, these keys can be used to copy and paste.

Channel Config display:

- [**CTRL**] + [**Cursor Left**]/ [**Cursor Right**] - moves the selected audio module left or right within the channel signal path.
- [**CTRL**] + [**Cursor Up**]/ [**Cursor Down**] - moves the selected audio module between the track bus, channel and direct output path.

Signal List display:

- [**SHIFT**] + [**Enter**] - temporarily enables Easy Edit mode for fast labelling of consecutive signals.

mxGUI only:

- [**Strg**] + [**^**] - opens and closes the Access/Assign window (German QWERTZ keyboard layout only).

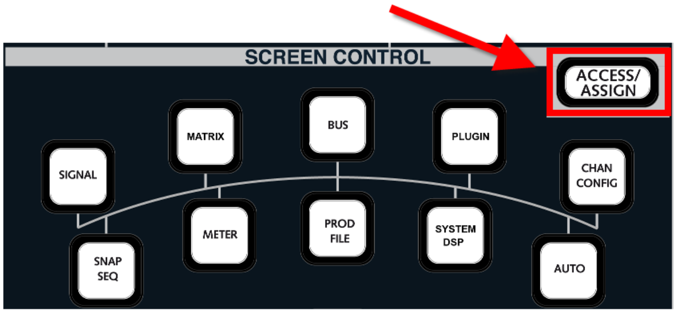
ACCESS CHANNEL/ASSIGN

The ACCESS CHANNEL/ASSIGN panel is used to modify the "channel in access". This is the channel which is assigned to the Central Control Section (for DSP parameter control). Also, to perform bus or fader strip assignments, the philosophy is to place a channel "in access" and then assign it directly to a destination. This provides fast configuration of the console without navigating through screen-based displays.

The NAME and LABEL of the "channel in access" are shown in the title bar of the Central GUI - for example: **INP 1**, **Kick**:

On the MKII mc²56, the ACCESS CHANNEL/ASSIGN panel appears on the Central GUI touch-screen (in place of the monitoring and automation buttons):

1. On the SCREEN CONTROL front panel, press **ACCESS/ASSIGN**:



The touch-screen buttons update and you will see two tabs:

- **ACCESS/ASSIGN**.
- **MISC** – other functions, see Touch-screen buttons for details.

The ACCESS/ASSIGN panel consists of:

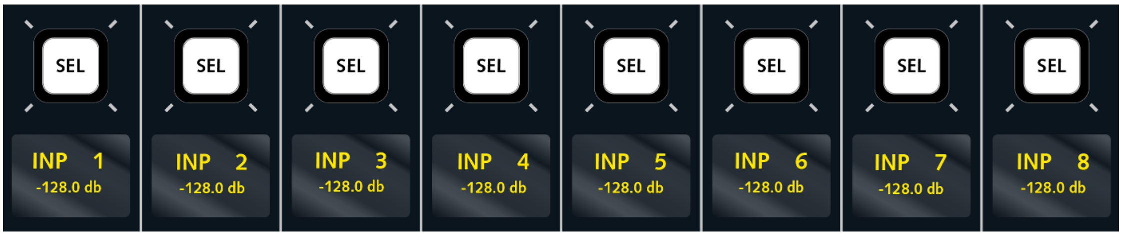
- Two 8-character NAME and LABEL displays.
- Channel type buttons – **INP**, **MON**, **AUX**, **GP-C**, **GRP**, **SUR**, **SUM** and **VCA**.
- A numeric keypad with Left/Right arrows.
- **BUS ASSIGN** – changes the operation of the panel to bus assign.
- **ESC** – can be used to exit any operation.
- Navigation buttons – **LEFT**, **RIGHT**, **NEXT** and **PREV**.
- **ENTER** - confirms an entry.

On the classic mc²56, the ACCESS CHANNEL/ASSIGN buttons are available on the front panel, to the right of the centre section.

Modifying the Channel in Access

There are three ways in which you can modify the channel in access:

1. Press a **SEL** button on a fader strip:



Your selection is shown in the title bar and on the ACCESS/ASSIGN panel - e.g. ***INP 1****.*

This is the simplest method for accessing channels which are already assigned to the control surface. To access channels not assigned to the surface, use method 2 or 3 as follows:

2. Enter the channel type and number:

Select a channel type by pressing one of the following buttons:

- **INP** – Input channels (up to 760).
- **MON** – Monitor channels/Track Busses (up to 96).
- **GRP** – Group masters (up to 64).
- **SUM** – Sum masters (up to 48).
- **AUX** – Auxiliary masters (up to 32).
- **VCA** – VCA masters (up to 128).
- **SUR** – Surround VCA masters (up to 64).
- **GPC** – General Purpose Channels (up to 256).

The channel type button flashes and the numeric keypad buttons illuminate; the flashing *TYPE NUM* message prompts you to enter a number:

- Select a number from the numeric keypad followed by **ENTER**. For example, press **1**, **2** and **ENTER** for the number twelve.
- Or, select a three digit number. For example, press **0**, **1**, and **2** will also enter the number twelve.

The channel type button stops flashing and your selection is shown in the title bar and on the ACCESS/ASSIGN panel.

If you enter an invalid selection, for example GRP 897, the NAME display tells you by flashing the letters **NOTAVAIL** for 'Not Available'. Press **ESC** to exit the operation and start again.

3. The third method is to scroll through the available channels:

- Press the **NEXT** or **PREV** buttons to increment or decrement the channel number by DSP type. For example, to scroll up or down through Input channels 1-760, Monitor channels 1-96, Groups 1-64, Sums 1-48, Auxes 1-32, VCA Masters 1-128, AFL/PFL Busses, General Purpose Channels (GPCs) 1-256 and Surround VCA Masters 1-64.
- Alternatively, press the **LEFT** or **RIGHT** buttons to select the next channel assigned to the control surface. For example, if INP 8 is currently in access and assigned to channel fader strip 8, pressing the **LEFT** button selects the channel assigned to fader strip 7.

The selected channel is shown in the title bar and on the ACCESS/ ASSIGN panel.

> The channel in access may be locked by pressing the Lock **ACC** button located on the Extra Buttons display. Therefore, if you cannot update the channel in access, check the status of this option.

Once you have selected the channel "in access", you may control its parameters from the Central Control Section, assign it to a fader strip or modify its bus assignments. We'll cover these operations later in this tutorial.

> When working in the **Signal List** display, you may also update the channel "in access" using the Set Access context menu option.

Powering On

To start the system, turn on the power to the control surface (mains connections at rear) and Nova73 (mains connections at front). The components may be powered in any order, but note that the control system resides within the Nova73. Therefore, the system boots when you turn on power to the Nova73.

You may switch on the power to other system components (e.g. DALLIS units) at any time.

The control system boots in a few seconds; during this time the Central GUI reports back on the boot-up progress.

By default, the warm start data is loaded at the end of boot-up. This means that the system comes back exactly as it was when you last shut down, ensuring fast recovery of all previous settings following a loss of power.

Depending on who was last using the console, you may be sat in front of a fully configured control surface with DSP settings or a series of blank fader strips! In either case, the fastest way to reset the console is to load a production.

> The control surface and Nova73 may be booted before DALLIS units. This enables you to prepare settings, including signal routing, before remote DALLIS stageboxes are connected or have received power.

Loading a Production

Productions form the top level for user data storage and store *all* the settings required for a production or type of job. Depending on the installation, you may have a number of setup productions or only one. Each should be clearly labeled – for example, **Basic Setups**.

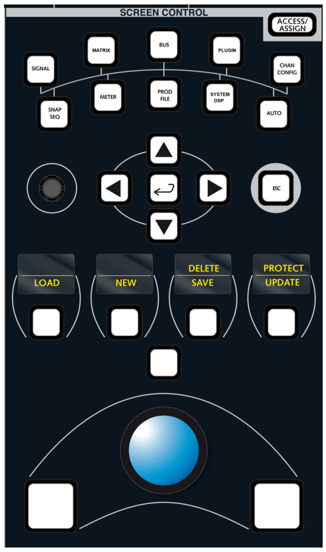
All setup productions should *always* be protected and *only* be modified by an authorised member of staff as they provide a common starting point for all users. Use the production to load a starting point; then save a new production to store your own settings.

Settings will vary from console to console, but generally a setup production should reset the following console elements:

- DSP configuration to a working default.
- Input and Output sample rate converter settings to match installed equipment.
- **System Settings** display options to a working default.
- **Metering** display pages to a working default.
- The assignment of channels to fader strips to a working default.
- DSP settings to flat.
- Basic signal routing and user labels – for example, routing to output distribution, monitoring and external metering.

To load a production:

1. Press the **PROD FILE** button, located on the SCREEN CONTROL panel, to view the **Productions** display:



The display is divided into two halves:

- **Productions** – lists all the productions stored on the internal user data flash card. This is where you can load, save, update rename, protect or delete a production.
- **Fallback** – lists any fallback productions stored in temporary memory. Fallback productions provide a level of undo in case you update or delete your production accidentally.

The active production (marked with an **A**) is also shown in the title bar of the Central GUI – in our example, **Olympic Games**. Therefore, you will *always* see the active production name across all displays.

To the right of each production name you will see the date and time when the production was last saved or updated, and the size of the production file. You may also see a padlock icon indicating that the production is protected.

If the list of **Productions** or **Fallback** Productions is longer than the available window space, focus on the list and use the rotary scroller on the SCREEN CONTROL panel to navigate up and down the list. You can also resize the windows and/or use the on-screen scroll bars.

2. Select the production you wish to load from the **Productions** list - for example Basic **Setups**.

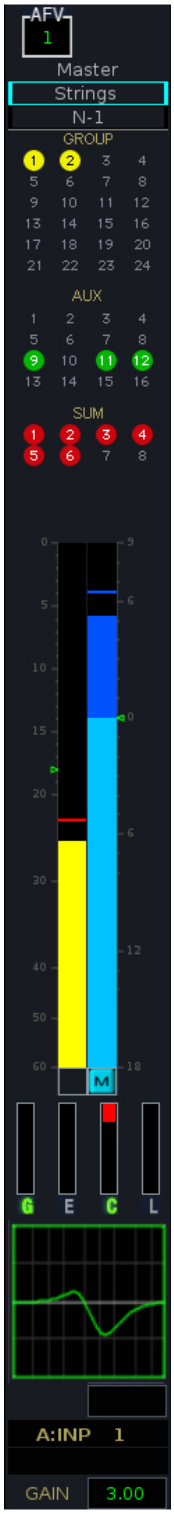
The selected production is highlighted in black.

3. Press the **LOAD** soft key, or right-click and select **Load**, to complete the operation.

*The console updates, and the title bar now shows that **Basic Setups** is the active production.*

For additional confirmation, watch the status bar at the bottom of the Central GUI; you should see a **loading…** message as the production data loads:

Interrogating the Fader Strips



Depending on the settings within the setup production, you may now be able to open the faders and monitor audio! Don't worry if this is not the case as we will look at how to modify the configuration shortly.

You can interrogate which channels have been assigned to the control surface by looking at the fader strip label displays (below), or the **Channel** display (opposite). You will see the channel name, channel label OR inherited source label depending on the centre section LABEL buttons:

The Channel display also shows metering, bus assignments, VCA or Link group masters, N-1 assignments, AFV, etc.

In addition, the LAWO backlights, at the bottom of each fader strip, are colour coded to provide quick channel indentification. The default colours are input and monitor channels (white); groups (yellow), aux masters (green), VCAs (blue) and sums (red):

If button-glow is enabled (MKII mc²56 only), then fader strip buttons in their off state are dimly lit according to the channel colour. This makes channel identification even easier, especially in low-light conditions.

Use the fader strip **FLIP** buttons, or the global BANK/LAYER buttons to interrogate other fader banks and layers, see Bank switching and Layer switching for details.

Adjusting Input Gain

Fader Strip: Input Gain

The upper rotary control on the channel fader strip is dedicated to source gain - either mic/line or digital depending on the channel's source. The amount of **GAIN** is shown on the Channel display.

Central Control Section: INPUT Control

For additional parameters, such as **48V** or the 20dB **PAD**, assign the channel to the Central Control Section by pressing its fader **SEL** button. For full details, see INPUT Control.

Monitoring Audio

The **mc²56** provides two monitor outputs:

- Control Room Monitor 1 (CRM 1) – up to 8-channel, as defined by the global surround format.
- Control Room Monitor 2 (CRM 2) – stereo.

Two stereo headphone outputs follow the control room monitor selectors with separate level adjustment.

The console may also support separate studio monitoring, external AFL/PFL loudspeakers and/or alternate speaker switching depending on the monitoring and I/O configuration.

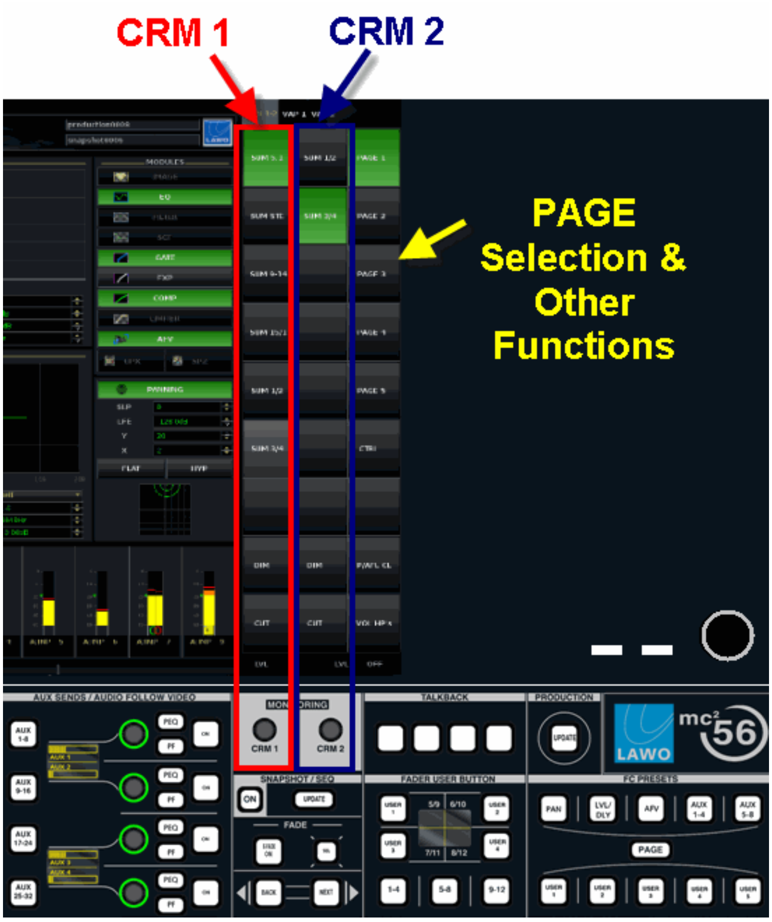
Level controls for CRM 1 and CRM 2 are located on the MONITORING panel. All other controls, including source selection, are programmed onto the Central GUI touch-screen monitoring buttons (displayed when ACCESS/ASSIGN is off).

Monitoring functions and I/O connections are programmed as part of the factory configuration (via TCL files). A description of the default configuration follows. However, you should refer to your system specification for full details.

On the MKII mc²56, the CRM 1 loudspeakers are usually connected to the analogue Line Out 1-8 on the rear panel, see Local I/O.

Monitor Source, Level, Cut & Dim

The touch-screen **MON 1-2** buttons are arranged into three columns. The first two columns select functions for CRM 1 and CRM 2, while the third column provides **PAGE** switching and access to other functions. Touch a button to action the function; it turns green when selected.



The default monitoring configuration provides the following functions:

1. Use the first two columns to select a source, and to **DIM** or **CUT**, the CRM outputs.
2. Use the dedicated rotary controls to adjust the CRM 1 or CRM 2 levels.

*The **LVL** is shown on the touch-screen display; the maximum level is defined by the configuration.*

3. Press the **PAGE** buttons (**SUM**, **AUX**, **GRP**, **PAGE 4** & **PAGE 5**) to access monitor sources.
4. Press **CRM1 ctrl** to access additional monitoring parameters.
5. Press the **X-tra** button to access the Extra Buttons display.
6. Press **P/AFL CL** to clear any AFL or PFL selections.
7. Press **VOL HP's** to adjust the headphone 1 & 2 levels from the CRM 1 & 2 controls.

Creating Your Own Configuration

Having loaded a setup production, you will want to modify the configuration to suit your particular show or mix. You can perform these operations in any order, but the most efficient way is as follows:

- Select a DSP configuration – this sets the number of input channels, monitor channels, groups, sums and auxes, and the channel type – Broadcast or Recording – for the production.
- Configure your channel formats and signal routing – from the **Signal List** display, you can label signals and configure signal routing. You can also choose which input channels, groups, sums, etc. need to be mono, stereo or surround.
- Assign your channels to fader strips – design your console layout by assigning your input channels, groups, sums, etc. where you want them.
- Assign channels to busses – configure your bus routing.

DSP Configuration

For the purposes of this tutorial, we are going to assume that your setup production loaded a DSP configuration with some input channels, groups, sums and auxes. To check this, or change the configuration, see DSP configurations.

Note that the DSP configuration determines the total number of mono channel paths; stereo channels use two paths; surround channels use up to eight paths depending on the surround format.

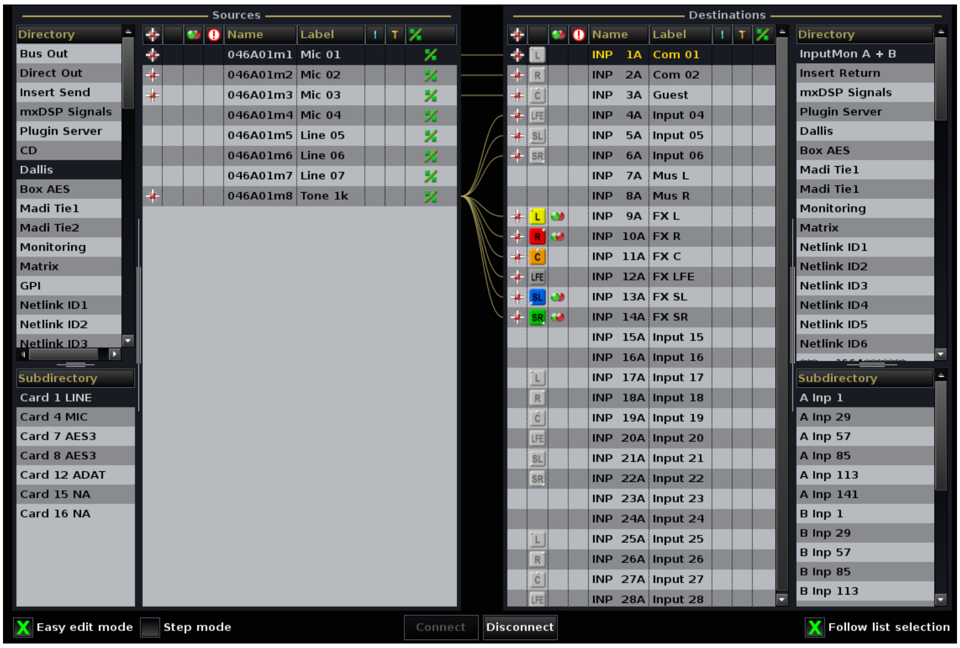
The Signal List

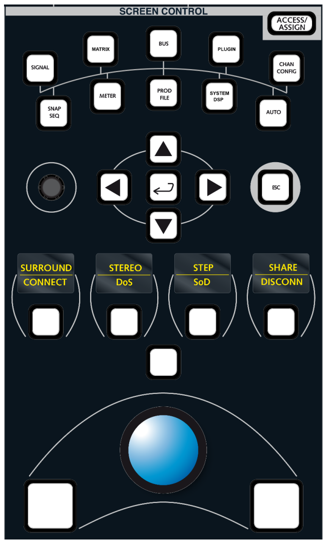
From the **Signal List** display, you can label signals and configure signal routing (input and output patching). You can also choose which input channels, groups, sums, etc. need to be mono, stereo or surround. For a detailed guide, see the Signal List display; here we will provide an introduction to basic routing and channel formats.

It is a good idea to configure mono, stereo and surround channels before making fader strip assignments, as the console will then distribute stereo faders automatically when making consecutive channel assignments.

Note that signal routing may also be performed from the mx Routing display (as a crosspoint matrix).

1. Press the **SIGNAL** button, located on the SCREEN CONTROL panel, to view the **Signal List** display:





Routing a Source to a Destination

To make a route - for example, to route a microphone source to an input channel:

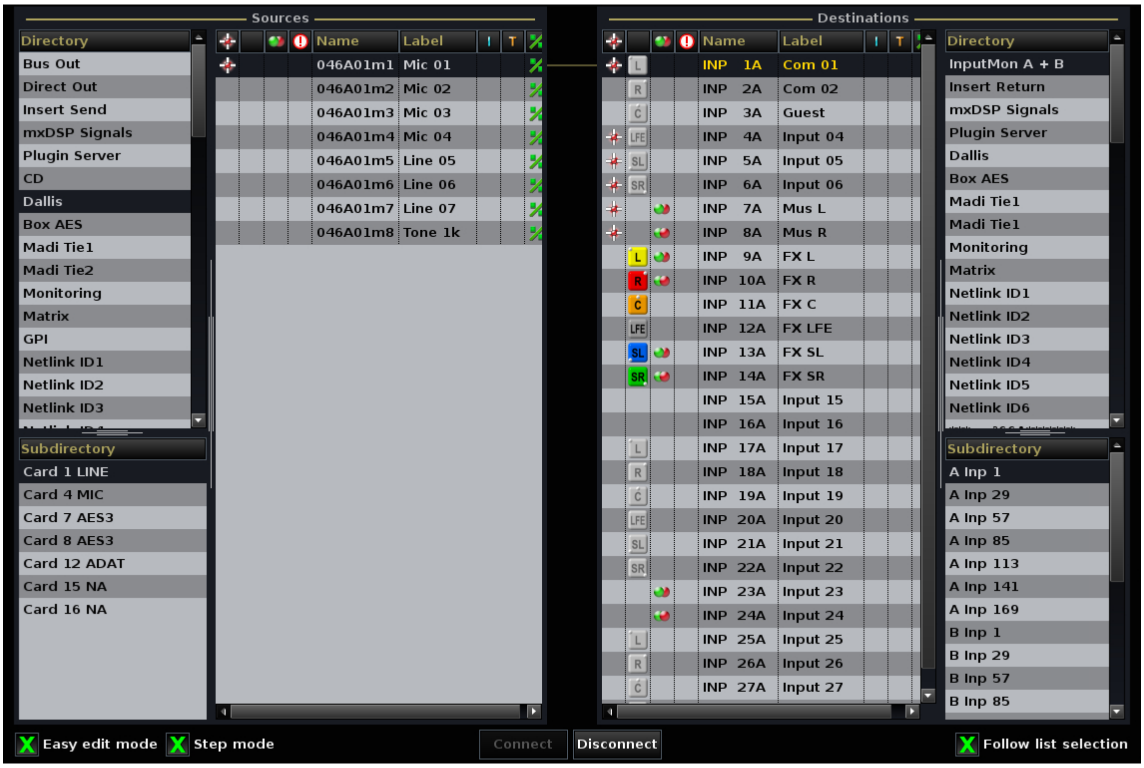
1. Select the source – for example, the source directory called **DALLIS**; subdirectory called **CARD 1 LINE**; and the source named **Mic 01**.

2. Select the destination – for example, the destination directory called **Input/Mon A + B**; subdirectory called **A Inp 1-28**; and destination called **INP 1A**.

Note that input and monitor channels support an A/B input switch. By selecting **INP 1A** as the destination, you will route to the A input of input channel 1.

3. Then press the on-screen **CONNECT** button, or SCREEN CONTROL soft key, to make the connection.

*The **Signal List** updates with a line between the source and destination:*



If the input channel is already assigned to a fader strip, and **INHERIT SOURCE** is selected (from the centre section LABEL buttons), then you will see the source label in the fader strip's label display. You will also see signal present beside the fader, and metering on the **Channel** display (according to the meter pickup point).

Routing Consecutive Sources to Destinations (Step Mode)

> To route consecutive sources to consecutive destinations, turn on **Step mode** to speed up the connection process.

1. Select the first source – for example, **Mic 01** – and the first destination – for example, **INP 1A**.

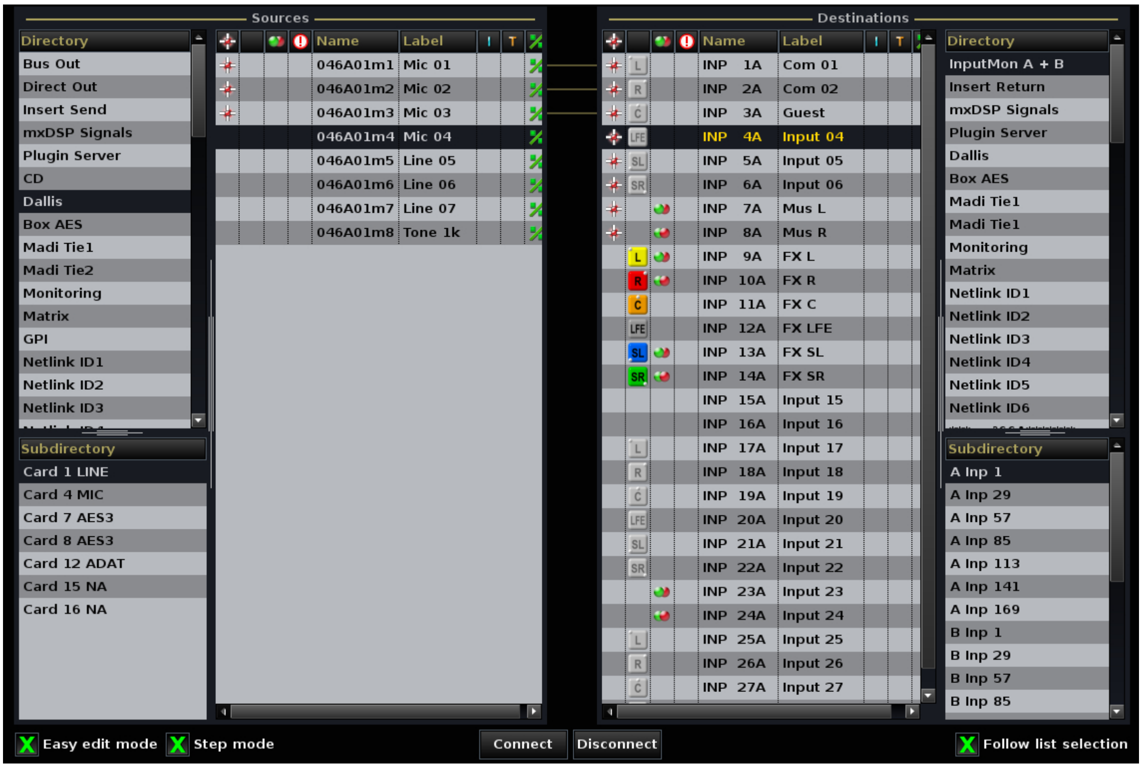
Your selected source and destination are highlighted in black.

2. *BEFORE* you press **CONNECT**, enable the on-screen **Step mode**, or select the **STEP** soft key.

3. Now press **CONNECT**.

The first route is made and the source and destination selections automatically step down to the next entries in the list:

4. Continue pressing **CONNECT** until all of your sources are connected to your destinations:



If the list of sources is shorter than the list of destinations, then when you reach the last source in the list, **Step mode** automatically scrolls back up to the first source in the list. This allows you to continue making routes from the sources to the remaining destinations, for example, to route microphones 1-16 to input channels 1-16, 17-32, etc.

Step mode can also be used with an offset between the starting source and destination: for example, to route Microphones 1-16 to Input Channels 17-32, repeat the above operation but set your first destination channel to be **INP 17** rather than **INP 1**.

Disconnect

To remove a route:

1. Select the destination (e.g. **INP 2A**).
2. And press the on-screen **DISCONNECT** button, or SCREEN CONTROL soft key.

The line between the source and destination disappears:

Turn on Step mode, select the first destination, and then keep pressing **DISCONNECT** to disconnect a range of destinations quickly and easily.

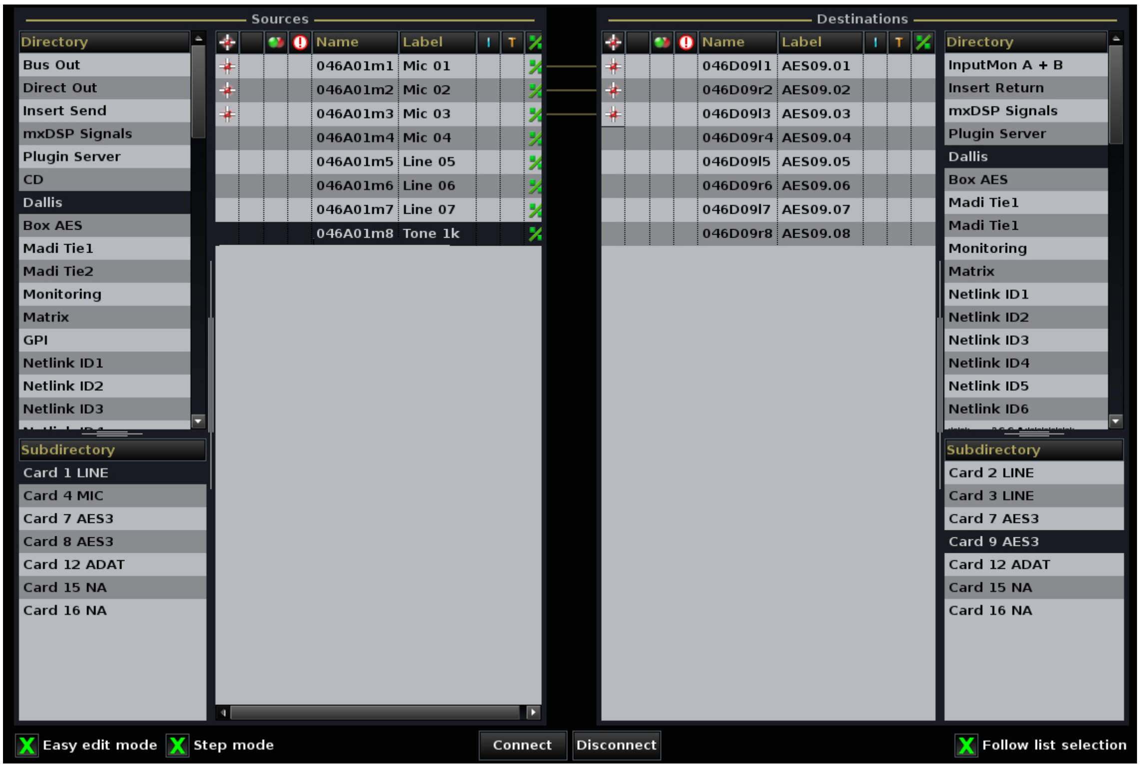
Note that if you route a source to a connected destination, then the previous source assignment is replaced; you don't have to disconnect the destination to assign a new source.

More Signal Routing Examples

The same steps may be used to connect any source to any destination. For example:

- To route a Sum bus to an output, select **Bus Out** -> **DOUT Sum 1** -> **Sum 1** as the source, and your external output as the destination:

- To route a microphone signal directly to an AES output, select the mic/line input as the source, and your AES output as the destination. This makes a direct route through the matrix, bypassing the console's channel DSP.



Creating Stereo or Surround Channels

Any odd/even pair of input or output channels may be configured for stereo and controlled from a single fader strip. Or, multiple channels may be configured for surround (up to 8-channel) and controlled from a single Surround VCA master.

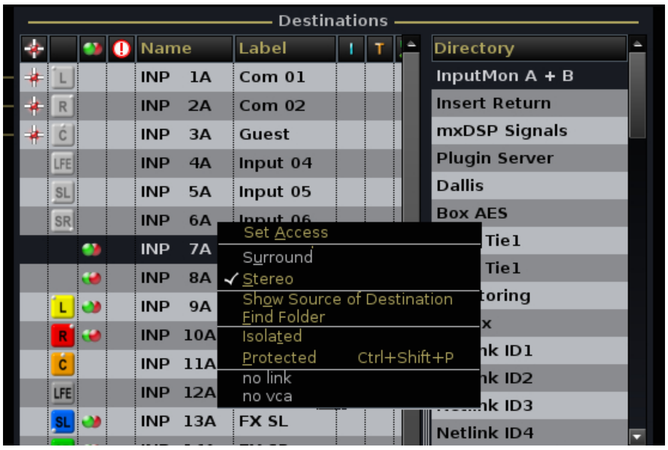
> The same procedure may be used on input, monitor, group, sum or aux DSP channels, allowing you to create stereo or surround input channels and output masters.

There are three ways to create a stereo channel and two ways to create a surround channel. Here we will deal only with the **Signal List** method, as this is the best approach when starting a production (as you can also label and route signals).

The other methods use the Channel Config display (stereo or surround), and the Central Control Section IMAGE panel (stereo only).

➢ To create a stereo input channel:

1. Select an odd numbered input channel from the **Destinations** list (e.g. **INP 7**).
2. Press the **STEREO** soft key, or right-click and select the **Stereo** option:

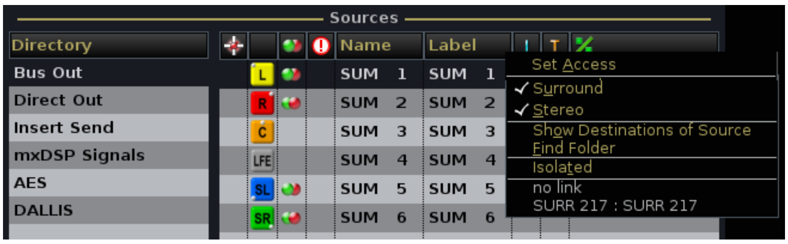


This links the selected channel to its adjacent DSP path. For example, INP 7 and INP 8.

You can link any odd/even pair of input or monitor channels using this method. Alternatively, select a **Bus Out** from the **Sources** list to create a stereo bus master.

➢ To create a surround sum:

1. Select the first sum for the surround output from the **Sources** list (e.g. **SUM 1**).
2. Press the **SURROUND** soft key, or right-click and select the **Surround** option:



*This links consecutive sums, according to the global surround format, and automatically assigns a Surround VCA - in our example, **SURR 217**.*

You can configure surround sums, groups or auxes using this method. Alternatively, select **InputMon** from the **Sources** list to configure surround input or monitor channels.

For surround inputs, panning is automatically reset so that INP 9 feeds SUM 1, INP 10 feeds SUM 2, etc. The best way to position a surround channel within the surround field is using Hyper Pan.

> Surround channels may only be created in 8-channel blocks, so you must select Sum 1, 9, 17, etc. You cannot select **Surround** if you right-click on an invalid channel number.

Note that the front and rear left/right pairs of a surround channel are automatically linked for stereo. This is for convenience when revealing the component channels. The stereo linking is only a default state; you can deselect the stereo link at any time.

Assigning Channels to Fader Strips

Any type of DSP or control channel - input, monitor, group, aux, sum, VCA, etc. - may be assigned to any fader strip.

Let's take the example of assigning input channels 1 to 24 across fader strips 1 to 24, and a single sum output channel (SUM 1) to a main fader strip.

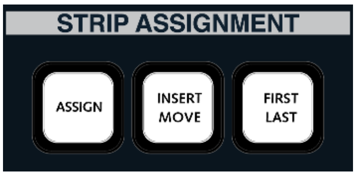
If you want to clear the current assignments to start from a series of blank fader strips, use CLEAR BANK.

Assigning a Single Channel

To assign SUM 1 to a fader strip:

1. Select the channel - either by pressing its fader **SEL** button or entering **SUM**, the number **1** and **Enter** from the ACCESS CHANNEL/ASSIGN panel.

2. Press the global **ASSIGN** button, located on the STRIP ASSIGNMENT panel:



*The fader **SEL** buttons across the console flash, in green:*

3. Press a fader **SEL** button to complete the assignment.

*The fader **SEL** stops flashing and changes colour, from green to red.*

*In addition, the fader strip controls update to show the settings for the new assignment - e.g. Fader Label = **SUM 1**; fader is set to 0dB (default level); and so on.*

You can assign the same channel to multiple fader strips by keeping the **ASSIGN** button selected - for example, to switch to a different bank or layer. Note that this assigns the *same* channel to multiple places, so if you choose **SEL** buttons on the same bank or layer, then you will have lots of faders controlling a single channel!

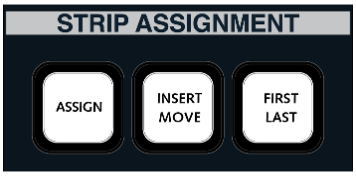
4. Deselect the **ASSIGN** button, or press **ESC** on the SCREEN CONTROL panel, to exit the fader strip assignment mode.

Assigning Consecutive Channels

To assign multiple fader strips in one operation:

1. Select the first channel in the range - either by pressing its fader **SEL** button or entering **INP**, the number **1** and **Enter** from the ACCESS CHANNEL/ASSIGN panel.

2. Press the **FIRST LAST** button, located on the STRIP ASSIGNMENT panel.



*This automatically selects global **ASSIGN**, and the fader **SEL** buttons across the console flash, in green:*

3. Press the fader **SEL** button on the first fader you wish to assign (e.g. strip 1) followed by the fader **SEL** button on the last fader (e.g. strip 24).

*The console incrementally assigns the input channels from the first selection (channel fader strip 1) to the last selection (channel fader strip 24), and cancels the **FIRST LAST** mode.*

If all the channels are mono, then you will have assigned INP 1 to 24 to fader strips 1 to 24.

If some channels are stereo, then they are automatically assigned to a single fader. For example, if INP 1&2 and INP 3&4 are stereo, then they are assigned to fader strips 1 and 2; INP 5 is assigned to fader strip 3, INP 6 to fader strip 4, and so on.

If some channels are surround, then it is the component channels which are assigned (e.g. L/R to fader strip 1, C to fader strip 2, LFE to fader strip 3, Ls/Rs to fader strip 4). You can control surround channels from a single master (called a Surround VCA), but this must be assigned to the control surface separately. See Surround VCAs for details.

The start and end of the range can be at any position across the control surface, and sources may be routed from left to right or from right to left by reversing the order of your first and last fader selection.

Note that **FIRST LAST** operations treat channel and main fader strips independently, allowing you to assign consecutive channel fader strips without affecting main fader strip assignments or vice versa.

4. Deselect the **ASSIGN** button, or press **ESC** on the SCREEN CONTROL panel, to exit the fader strip assignment mode.

Bus Assignment

There are several ways to assign a channel to a mix bus. However, the quickest method to route all 24 input channels onto our main output (SUM 1) is to use reverse assign. For details on other methods, see Bus Assign.

Reverse Assign

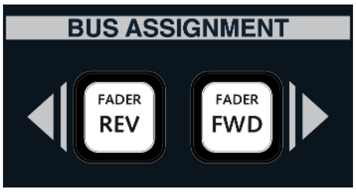
This method selects the bus first, and then the source channels. It is ideal for assigning a single bus *from* multiple channels (if the source channels are assigned to fader strips).

For example, to assign some input channels onto SUM 1:

1. Select the SUM 1 channel - either by pressing its fader **SEL** button or entering **SUM**, the number **1** and **Enter** from the ACCESS CHANNEL/ASSIGN panel.

> To select a Track bus as the destination, press the fader **SEL** button on the corresponding Monitor channel or enter MON x from the ACCESS CHANNEL/ASSIGN panel.

2. Press **FADER REV**, located on the BUS ASSIGNMENT panel:



*The fader **SEL** buttons, across the console, now indicate the status of bus assignments **to** the channel in access (SUM 1):*

- Steady state red = channel assigned to destination.
- Flashing green = channel not assigned to destination.
- **SEL** not lit = invalid destination (for example, you cannot assign another Sum channel onto SUM 1!)

3. Press the fader **SEL** buttons to modify the assignments.

For example, press the green fader **SEL** buttons on strips controlling INP 1, INP 2, etc. to assign these channels onto SUM 1. Or, press the red **SEL** buttons on INP 5, INP 6 and INP 7 to remove the existing assignments.

*The fader **SEL** buttons change state, and the Channel display updates.*

> If the bus is stereo or surround, then assignments onto the LR or surround channels are made in one operation, see Bus Assignments to a Surround Output.

4. Deselect the **FADER REV** button, or press **ESC** on the SCREEN CONTROL panel, to exit the bus assign mode.

Saving, Transferring & Loading Settings

One of the major benefits of the **mc²56** is the ability to store and recall all the settings for a live show or type of application.

Productions

Productions form the top level for user data storage and store *all* the settings required for a production or type of job.

Productions store everything included in a snapshot, plus lower level settings such as the DSP configuration and system options. As a result, loading a production may cause a brief interruption to audio, and should *not* be used during a show. Instead, use snapshots to recall settings while live on-air.

Snapshots

Within each production, folders are created to store snapshots.

Snapshots store different mixes for recall before or during the show. For example, to recall a different mix for each band in a live entertainment show, or to recall scene changes during a live theatre production. To manage snapshot recall, snapshot isolate and filtering may be applied to protect channels or elements of the desk.

Sequences

Sequences are provided for convenient recall of snapshots during a live broadcast or theatre production.

A sequence is a list of snapshots which can be loaded in sequence during a live show. The transition between snapshots in a sequence can be cross faded if required. In addition, offsets can be applied to deal with last minute changes such as a change of artist. Note that the sequence itself does not store any settings, but simply creates a list of pointers to snapshots stored within the production folder.

Presets

Presets are stored independently of productions, and save and load settings for processing modules (EQ, Gate, Compressor, Panning, etc.) or for a complete channel. For example, you may wish to save your favourite Kick Drum EQ, or the complete settings for an announcer channel.

Transferring User Data

All user data is stored on the system's internal flashcard and may be imported or exported to a USB interface or mxGUI computer. In a networked installation, a central file server can be made accessible from each console within the network.

> User data is fully compatible with any mc² or Nova73, regardless of the hardware configuration. This enables the transfer of production data, snapshots, mixes or presets to and from any system (including any other mc²), in order to recall settings in a different studio.

Saving a New Production

You can save the current settings of the console into a new production using **SAVE**. (i.e. this operation performs a "Save As..".)

SAVE keeps all the current settings, including any snapshot/sequence folders and automation mixes, and saves them under a new production name. If you wish to clear the folders and mixes from memory, then see new production.

It is a good idea to save and organise your productions carefully.
Don't overwrite the studio's setup production with your own settings by using update! Instead, use the **SAVE** function to save into a new production.

To save a new production:

1. Select the on-screen **Save** button, or focus on the list of **Productions** and press the **SAVE** soft key.

The current settings are saved into a new production which is given a default name (e.g. ***production 0012****):*

The production is time and date stamped, and automatically becomes the active production (**A**) as indicated in the title bar.

For additional confirmation, watch the status bar at the bottom of the Central GUI; you should see a **saving...** message as the production data is saved.

Renaming a Production

1. Click on the production name:

Click once to select all the existing text (white) or twice (black cursor) to modify the existing name.

2. Enter a new name from the keyboard.

3. When you have finished, press the Enter button, on the keyboard, to confirm the new name (e.g. **Formula One**):

4. Or, if you make a mistake or want to exit without making any changes, press the **Esc** button on the keyboard.

Updating a Production

You can save the current settings of the console into an existing production using **UPDATE**.

Updating a production overwrites it. Therefore, make sure you select the correct production to update. If you do make a mistake, don't panic! When a production is updated, a backup of the "old" production is created in the **Fallback** list, see Fallback Productions.

> If a production is protected, then you can *not* update it. (Use **Protect** to safeguard any important productions which you do not want to accidentally overwrite).

There are two ways to update a production:

➢ The UPDATE button

This button *always* updates the active production, marked with an **A** and named in the title bar. (i.e. this operation performs a "Save".)

You can use the button at any time; the **Productions** display does not need to be selected. But, remember that a protected production can *not* be updated.

The button also flashes if you have snapshots or automation which have not been saved.

➢ Update in the Productions display

From the **Productions** display, you can update *any* existing production, not only the active one.

1. Select a production from the **Productions** list (e.g. **Formula One**):

2. Either press the **UPDATE** soft key, or right-click and select **Update**, to complete the operation. (Remember that a protected production can *not* be updated.)

   The selected production is overwritten with the current console settings. You can confirm this by looking at the new date and time stamp.

For additional confirmation, watch the status bar at the bottom of the Central GUI; you should see a **saving...** message as the production data is saved.

Using Auxiliary Sends

The **mc²56** supports 32 auxiliary sends which may be used for a variety of applications such as cue feeds, effects sends, mix minus (N-1) sends, etc.

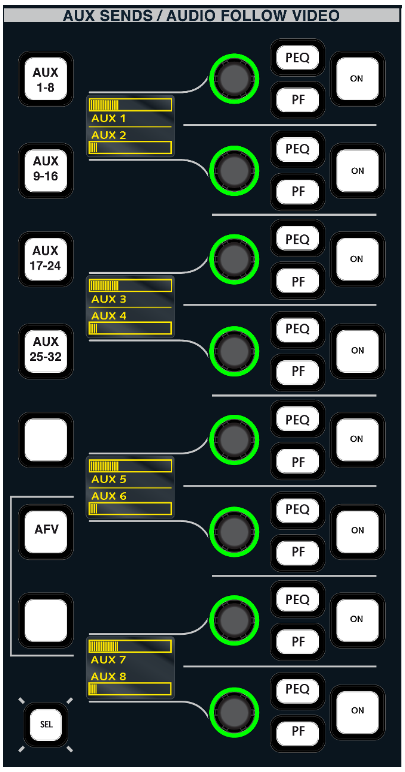
Any odd/even pair of mono sends may be linked for stereo operation. Or, you can create surround sends (up to 8-channel) from Auxes 1-8, 9-16, 17-24 or 25-32. This is handled in the same way as creating any other stereo or surround channel, see Creating Stereo/Surround Channels or Masters.

Aux sends may be controlled from the fader strip Free Controls or from the AUX SENDS panel in the Central Control Section. Alternatively, you can assign Aux send levels down onto the faders (see Fader Control of Levels), or adjust Aux on/off using any method of Bus Assign.

To get you started, let's use the AUX SENDS panel in the Central Control Section.

AUX SENDS

Each input, monitor or group channel may access up to 32 auxiliary sends. These are paged onto the eight rotary controls as follows:



1. Press **AUX 1..8** to assign the first eight auxiliary sends.

The name of the send (e.g. AUX 1 to AUX 8) appears in the alphanumeric display.

2. Press the **ON** button to activate the send.

The AUX bus assign boxes on the [Channel display](#) update to reflect your assignments:

3. Use the rotary control to adjust the send level.

The send level may be adjusted from -128dB to +15dB.

4. The send level defaults to be post fader. Press **PF** to switch the send pre fader or **PEQ** (Recording channels only) to switch to pre EQ.

The bus assign boxes are colour coded to reflect the different assignments:

- Post-fader – white writing on green (e.g. Aux 3).
- Pre-fader – black writing on white/green (e.g. Aux 5).
- Pre-EQ – white writing on green/white (e.g. Aux 8).

5. Press the **AUX 9..16**, **AUX 17..24** or **AUX 25..32** buttons to access the remaining auxiliary sends for the channel.

6. The [SEL button](#) is used to select the aux sends, in groups of 8, for operations such as copy or reset, channel linking, etc.

Note that the aux send options vary slightly between Recording and Broadcast channels:

Aux Send	Recording channels	Broadcast channels
Pre EQ	✓	×
Pre Fader	✓	✓
Post Fader	✓ (pre-bus)	✓ (after fader)

On Recording channels the pre EQ option follows any changes made to the position of EQ in the channel signal flow. This allows you to move the aux send to virtually any channel pickup position.

On Broadcast channels, the aux post fader send is a real post fader send, and not pre-bus as in a Recording channel. This means that you can position another module, for example delay, after the fader, and the delay will affect the main busses, but not the post fade aux send. See Changing the Signal Processing Order.

Creating a Mix Minus (N-1)

One of the most common functions required during a live production is the mix minus, or N-1, output. The **mc²56** may use any of its 32 auxiliary sends or 96 track busses (if available within the DSP configuration) to create mix minus sends.

The first step is to assign a mix minus bus to each source requiring a mix minus send.

Let's assume you have three microphone sources, each requiring an N-1 feed. The mic sources should be routed to three input channels and the input channels assigned to some fader strips.

To assign a mix minus bus to each source:

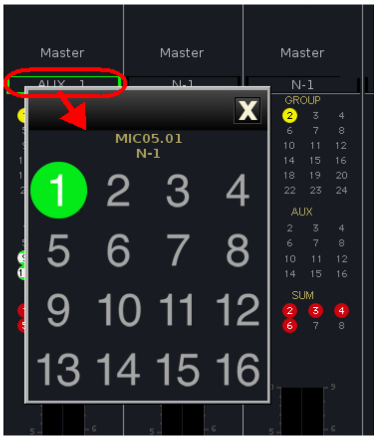
1. Touch the **N-1** text at the top of the fader strip's **Channel** display:

   An expanded pop-up window appears.

2. Touch a number to assign an aux as the N-1 bus for the source (the selection turns green).

3. Repeat for each source.

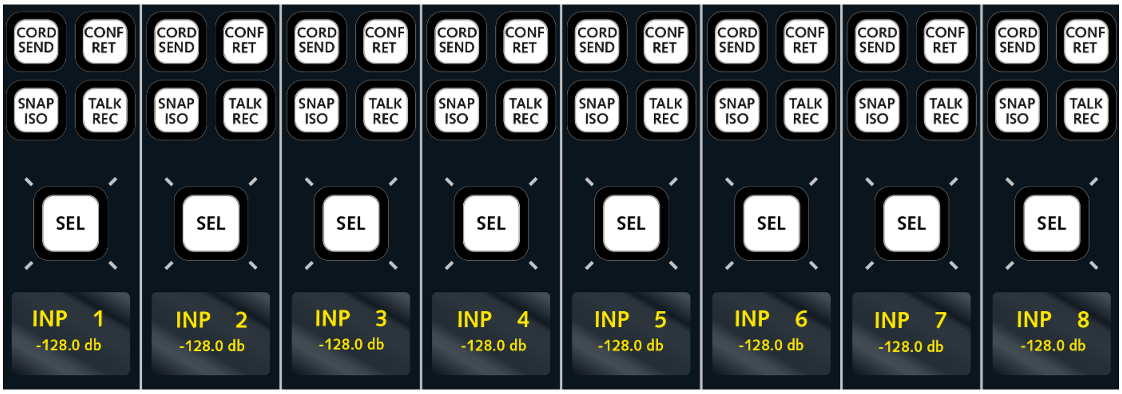
4. To close the pop-up, either touch the **X** in the top right corner, or touch twice in quick succession anywhere else on the display.



The mix minus bus names (e.g. **AUX 1**, **AUX 2**, **AUX 3**) are shown in the **N-1** field at the top of the **Channel** display. This provides feedback on which aux (or track bus) is assigned as the N-1 bus for each source/fader strip:

To activate and control the mix minus send:

1. Go to the fader strips controlling each source and press the **CONF** buttons on all three channels:



The mix minus is automatically activated for each of the three channels; you can see this reflected in the bus routing on the ***Channel*** *display. For example, fader strip 1 (mic 1) is assigned to all mix minus busses except its own (Aux 2 & 3); fader strip 2 (mic 2) is assigned all all mix minus busses except its own (Aux 1 & 3); fader strip 3 (mic 3) is assigned all all mix minus busses except its own (Aux 1 & 2):*

For more details, see Mix Minus (N-1) sends.

Configuring Audio Sub Group Masters

So far, we've routed our input channels directly to a main sum output. However, for many productions, you will want to use groups either to create independent mixes, like an international version, or to provide greater control over separate elements of the mix, for example to compress all of the music channels separately to the main presenter's microphones.

The number of groups is determined by your choice of DSP configuration; not all DSP configurations support groups. If you are unable to assign groups to the control surface, then check your DSP configuration.

To make your groups stereo or surround, use the **Signal List** display to configure the channel format. See Creating Stereo or Surround Channels, but select the **Bus Out** -> **DOUT Grps** directory to locate your group busses.

Then assign the group masters to some fader strips. See Fader Strip Assignment. You will need to put GRP x into access to make the assignment.

Next, you may wish to modify your bus assignments so that input channels are assigned to the groups (rather than directly to a sum), and the groups assign onto your sum. You can use any type of bus assign, but Reverse Bus assign is the fastest method. Put the sum into access (press its fader **SEL**), enable **REVERSE FADER**, and then deassign the input channels/assign the groups. Cancel **REVERSE FADER** and repeat but this time with the group in access. You can check your bus assignments using the Channel or Main Fader displays.

To monitor a group output, either press **AFL** (or **PFL**) on the fader strip controlling **GRP 1**. Or, use the CRM 1 source selection buttons to monitor the group directly. For more details, see Control Room Monitoring.

Finally, route your groups to their intended destinations (e.g. to an external recorder or feed). This is best done from the Signal List display.

Using VCA Grouping

A common application for the main fader strips is to use them as VCA masters. The console supports up to 128 VCA masters which may be controlled from main or channel fader strips. In addition, you may assign any type of channel to a VCA. This provides the ability not only to control input and monitor channels but also groups, sums, auxiliary and surround masters.

To create a VCA group:

1. First assign the VCA master to a fader strip in the usual manner - select VCA 1 from the ACCESS CHANNEL/ASSIGN panel, press the **ASSIGN** button, located on the STRIP ASSIGNMENT panel, and then press a fader **SEL.**

   Your VCA master is now assigned to the fader strip.

2. Then assign channels to the VCA using the same procedure as if assigning an audio bus. Put the VCA master into access (press its fader **SEL**), enable **REVERSE FADER**, and then assign the channels (press their fader **SEL**s). You will see your VCA bus assignments from the Channel or Main Fader displays.

VCA groups can use either moving or non-moving slave faders, defined by the Relative Slave Faders option in the **System Settings** display.

When working with non-moving slaves you can see and update slave fader positions even if the VCA master is closed, like an analogue VCA. Pay particular attention to the fader label displays: as you adjust the VCA master, the MAIN LEVEL on the slave channels updates - it is this value which indicates the real channel level and NOT the fader positions.

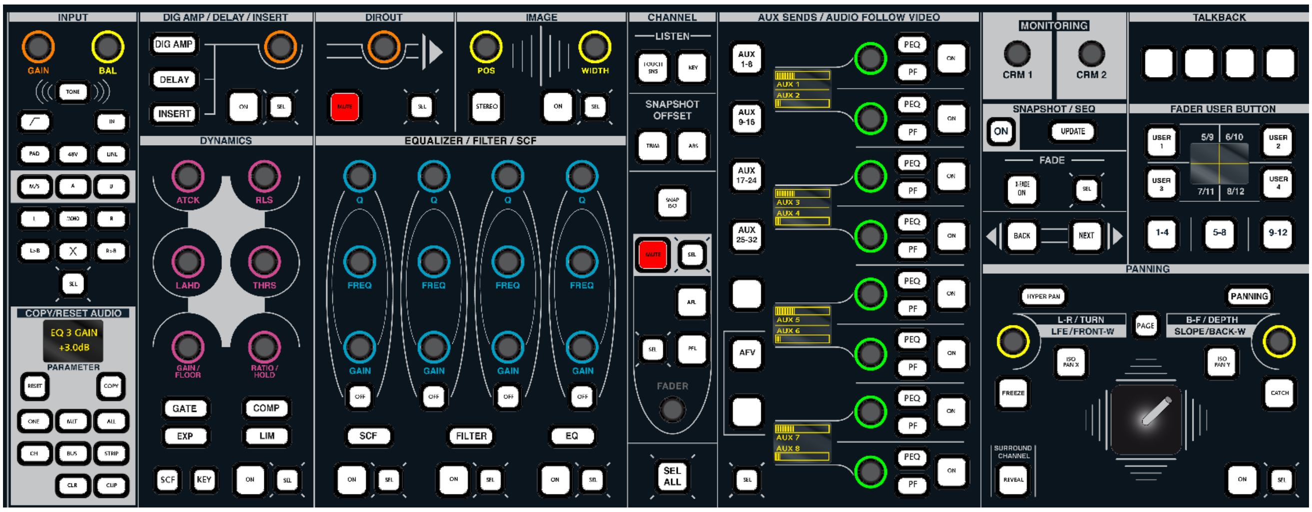
For more details, see VCA Grouping. For other methods of linking parameters, please see Link Groups and the Couple Group.

Applying Signal Processing

Depending on your choice of DSP configuration, full signal processing may be available on your input channels, monitor return channels, groups, sums and/or auxiliary masters. This allows you to compress a group output or EQ an auxiliary master in the same way you would apply signal processing to an input channel.

The following provides a step-by-step guide to some of the signal processing sections within the Central Control Section: EQ, Compressor, Delay and Insert. For a full tour, see the Central Control Section.

1. Press the **SEL** button on a fader strip to assign it to the Central Control Section:



Note that the controls are black (unlit) if a DSP module is not supported.

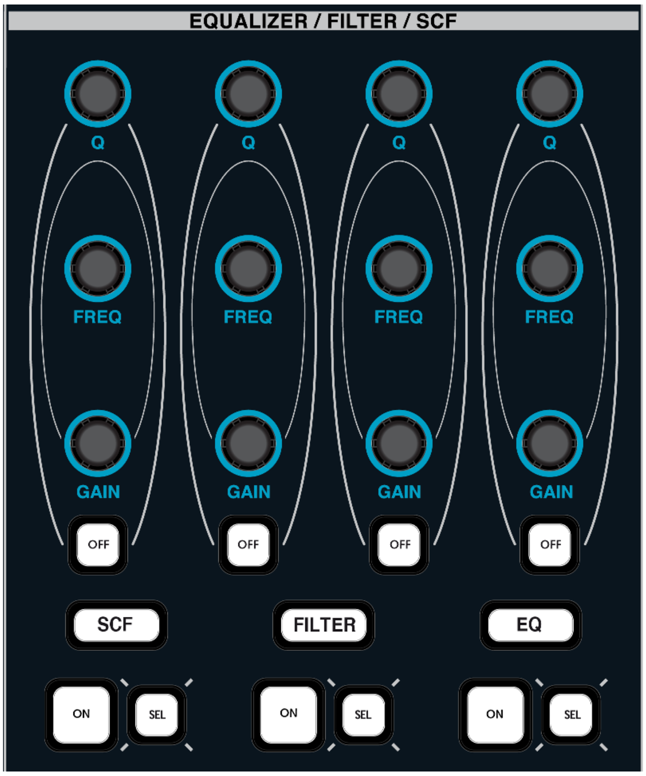
> Rotary controls are colour coded, making it easy to distinguish EQ from DYNAMICs, from AUX sends, etc. (MKII mc²56 only).
>
> All rotary controls are touch-sensitive; the controls default to provide fine parameter adjustment. For coarse adjustment (5 times faster), push down as you adjust the parameter.
>
> Remember to turn **ON** the DSP module to hear your adjustments!
>
> Select the Main Display for visual feedback on settings.

Using the 4-band Equaliser

Recording channels provide a 4-band equaliser (EQ) plus two 2-band high and low pass filter modules; one dedicated to the main channel (FILTER) and one dedicated to the dynamics sidechain (SCF).

Broadcast channels provide a single 4-band equaliser (EQ), and do not support separate filter or sidechain filter modules. However, the upper and lower bands of the equaliser can operate as a filter, shelf or parametric EQ.

The modules may be arranged in any order within the channel signal flow and are controlled from the EQUALIZER/FILTER/SCF control area:



Four sets of dedicated **GAIN**, **FREQ** and **Q** controls, with band **OFF** buttons are provided.

1. Switch the four sets of controls between sidechain filters, main channel filters and the 4-band equaliser using the **SCF**, **FILTER** and **EQ** buttons at the bottom of the panel:

Note that on Broadcast channels, you cannot select **SCF** or **FILTER**, as these DSP modules are not supported.

2. Press the **ON** buttons to turn each section on or off.
3. Now adjust the **GAIN**, **FREQ** and **Q** settings.

*The **Main Display** provides feedback on your parameter values. You can view the EQ, (and FILTER or SCF modules on Recording channels):*

All 4-bands of EQ (and 2-bands of filters on Recording channels) operate across the full frequency range (20Hz to 20kHz), and offer a variety of different EQ types. The frequency for each band is marked by a vertical line labeled 1, 2, 3 and 4 to show which band is acting at a particular frequency.

4. Press **OFF** to switch any individual band out of circuit.

5. Click on the EQ type touch-screen menu buttons to switch between bell, shelf and pass band filters for the high and low bands, and bell, constant Q and notch for the middle bands:

The filter and shelf parameters vary slightly between Recording and Broadcast channels:

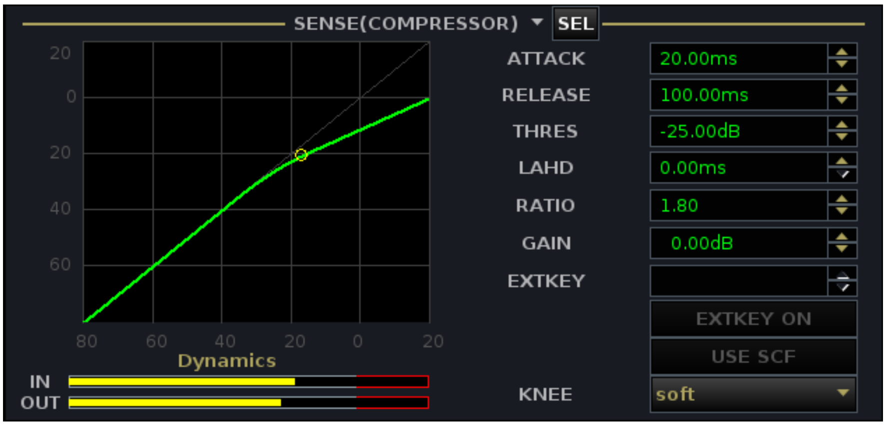
Recording channels	**Broadcast channels**
Max. 3rd order filter	Max. 2nd order filter
Max. 18dB/octave shelf	Max. 12dB/octave shelf

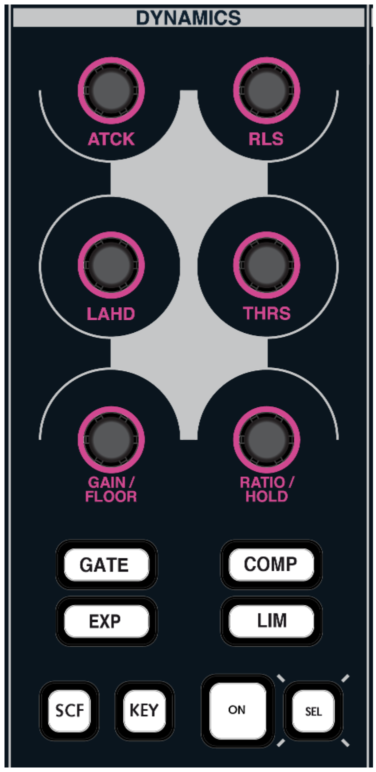
Note that if you load a Recording channel EQ setting to a Broadcast channel (e.g. using a Preset), and the stored parameter lies outside the range supported by Broadcast channels, then the closest available value is applied. For example, if the preset is attempting to load a 3rd order filter, then a 2nd order filter (the maximum) is applied.

Setting a Compressor

1. Press the **COMP** button to switch the DYNAMICS controls to the compressor section.
2. Press **ON** to switch on the compressor.
3. Use the six rotary controls to set the parameters.

The action is best described by looking at the **COMPRESSOR** graph on the **Main Display**:



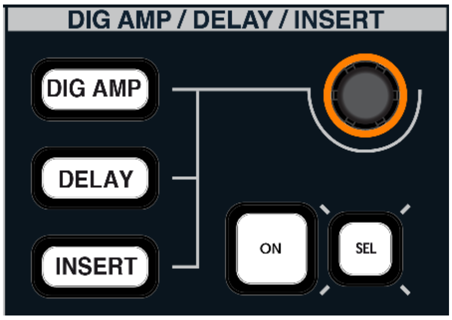


The Compressor parameters may be set as follows:

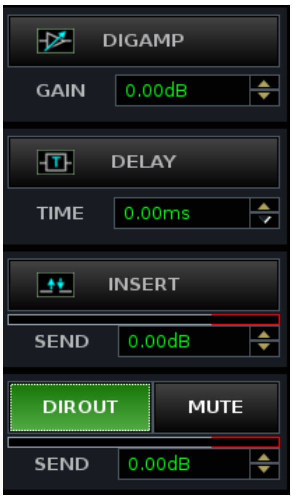
- Threshold Level – from -70dB to +20dB (must be at least 10dB higher than the Expander Threshold.)
- Ratio – from 1:1 to 10:1.
- Attack Time – from 0.29ms to 250ms.
- Release Time – from 40ms to 10s.
- Look Ahead Delay – from 0ms to 10ms (look ahead delay affects all three Combi-Dynamics modules).
- Gain - from -20dB to +20dB.
- Knee - hard or soft. This parameter is set from the **Main Display**. Use the trackball to set the **KNEE** option to either **hard** or **soft**.

Channel Delay (DELAY)

Central Control Section



Main Display

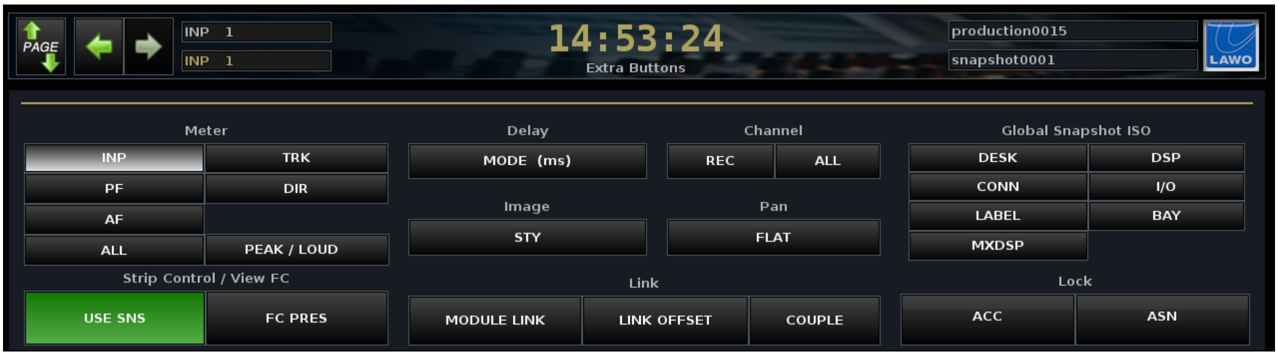


1. Press the **DELAY** button to switch the DIG AMP/DELAY/INSERT controls to the channel delay.
2. Press **ON** to switch the delay in and out of circuit
3. Move the rotary control to adjust the delay time.

*The amount of delay is displayed in the **TIME** box on the **Main Display**.*

> To enter a specific delay time, click on the **TIME** box on the **Main Display** and type in a value from the console keyboard.

4. You can change the delay mode from the Extra Buttons display. Touch the on-screen **MODE** button to cycle around the options – milliseconds (ms), frames (frms) or meters (m):



Set Delay in ms or frames when you are dealing with a specific time delay, for example, to delay the channel's audio relative to an incoming video feed.

Set Delay in meters when you are time aligning microphones positioned on the studio floor and know the distance between the microphones.

The available channel delay varies slightly between Recording and Broadcast channels:

Recording channels	Broadcast channels
Min. = 1 samples (0.02 ms)	Min. = 18 samples (0.38 ms)
Max. = 1.8 seconds	Max. = 1.3 seconds

Note that if you load a Recording channel delay to a Broadcast channel (e.g. using a Preset), and the stored parameter lies outside the range supported by Broadcast channels, then the closest available value is applied. For example, if the preset is attempting to load a delay of 5 samples, then 18 samples (the minimum) is applied.

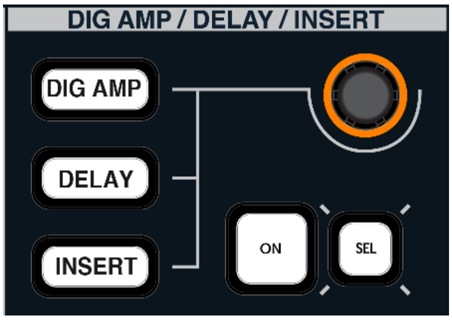
Depending on the hardware configuration of your console, an additional 48 delays may be available from the DSP Module 983-03. These are fixed time delays which may be inserted into any routing crosspoint and are programmed within the AdminHD configuration.

Inserting Outboard Processing

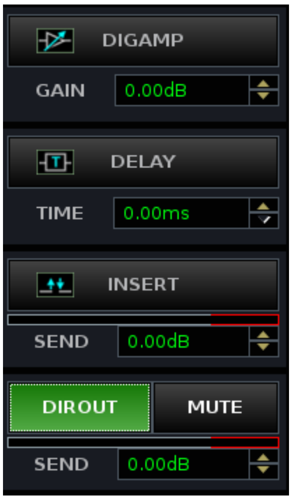
Routes to and from the channel insert send and return are made from the Signal List display. You should route the channel's insert send to the output feeding the insert device, and then route the output from the external device to the corresponding insert return.

The Central Control Section can then be used to control the insert on/off switching and send level:

Central Control Section



Main Display



1. Press the **INSERT** button to switch the DIG AMP/DELAY/INSERT controls to the channel insert.
2. Press **ON** to switch the insert return in and out of circuit.

If an insert return is not assigned, you will get silence when you switch the insert into circuit.

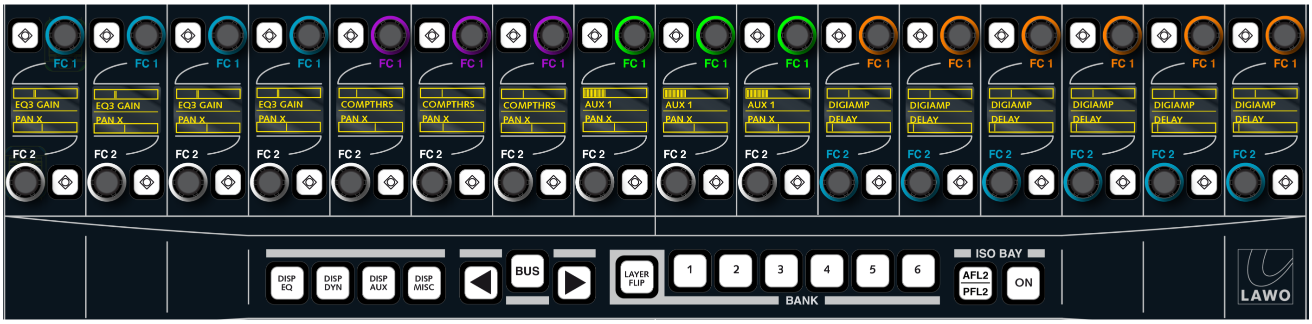
3. Adjust the rotary control to set the level of the insert send.

The SEND level is shown on the Main Display. It may be adjusted from -128dB to +15dB.

> The channel insert send is always active even when the return is not inserted. This allows the insert send to be used to generate an extra clean feed from the channel, with level control, which may be taken from any point in the channel signal flow, see Changing the Signal Processing Order.

Using Free Controls

The two Free Controls on each channel fader strip may be assigned to key functions for the source. For example, on a presenter's channel you may want immediate access to the presenter's mix minus level and compressor threshold. Whereas, on a music replay channel, it is more important to access L/R Balance and Aux send level.



There are several ways to assign parameters onto the Free Controls. To get you started, let's look at how to assign an individual parameter to a single control. For details on other methods, please see The Channel Fader Strip: Free Controls.

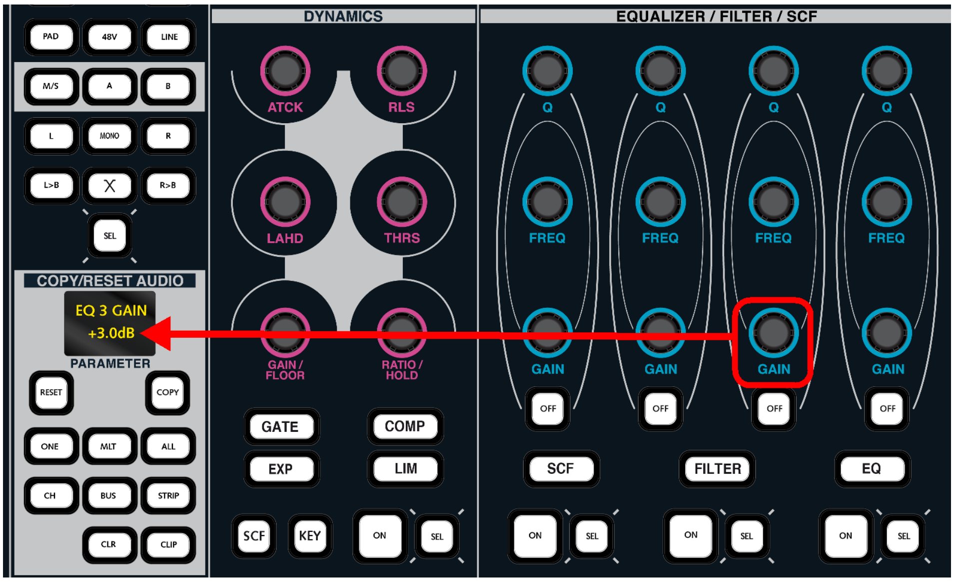
Assigning a Single Free Control

1. Press the **ONE** button, located on the COPY/RESET AUDIO panel, to activate a one-shot assignment.

*The **ONE** button flashes to indicate that parameter assign is now active.*

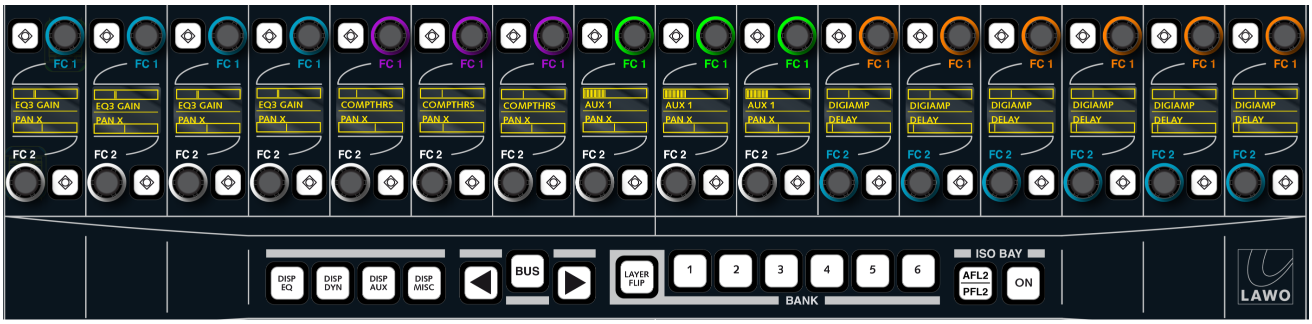
2. Select the parameter you wish to assign, by touching a rotary encoder on the Central Control Section - for example, touch the EQ Band 3 **GAIN** control.

The parameter is placed into the PARAMETER clipboard:



3. Now touch the Free Control on the destination channel strip.

*The assignment is made; the alphanumeric display below the FC updates; and the **ONE** button automatically cancels.*



Next Steps

Hopefully, you have enough information to being working with the console immediately. If you need more assistance, use the Index located at the back of the manual to locate information on a particular topic.

Otherwise keep reading to learn more about each area of the console's operation:

- Console Configuration - configuring signal flow and the control surface.
- Channel Control - from the channel fader strips and the Central Control Section.
- The Centre Section - centre section functions including monitoring, VCA, Link and Couple groups, the Extra Buttons display.
- Console Reset - productions, snapshots, sequences, presets and how to import/export data.
- Timecode Automation - dynamic automation referenced to timecode.
- Signal Routing/Settings - the Signal List, mx Routing, Signal Settings, mxDSP Settings and Downmix displays.
- System Configuration - the System Settings and Custom Functions displays, system components, redundancy, procedures for system shutdown and restart.
- mxGUI - offline setup or remote operation of any mc² system.
- Lawo Remote App - remote operation of an mc² console from iPhone, iPod or iPad.
- Technical Data - summary of the system specification.
- Appendices
- Glossary

Chapter 3: Console Configuration

Introduction

This chapter deals with configuring signal flow and the control surface.

Together these concepts allow you to define as many input channels, monitor return channels, track busses, groups, main sums and auxiliary sends as the production requires, and then assign these elements across the console's fader strips on any bank or layer.

Topics covered in this chapter are:

- Signal Flow Concepts
- DSP Configurations
- DSP Channel Types
- The Channel Config Display
- Control Surface Configuration
- Bank Switching
- Layer Switching
- Isolating Fader Strips from Bank Switching
- Isolating Fader Bays (ISO BAY)
- Fader Strip Assignment
- General Purpose Channels (GPCs)
- Monitor Channels

Signal Flow Concepts

The **mc²56** provides a pool of DSP resource which can be configured for input channels, monitor return channels, groups, sums (main mix outputs) and auxiliary sends. Each channel comes with either full signal processing or reduced signal processing (known as tiny channels). This enables EQ, Dynamics, Delay, etc. to be applied to both inputs and outputs.

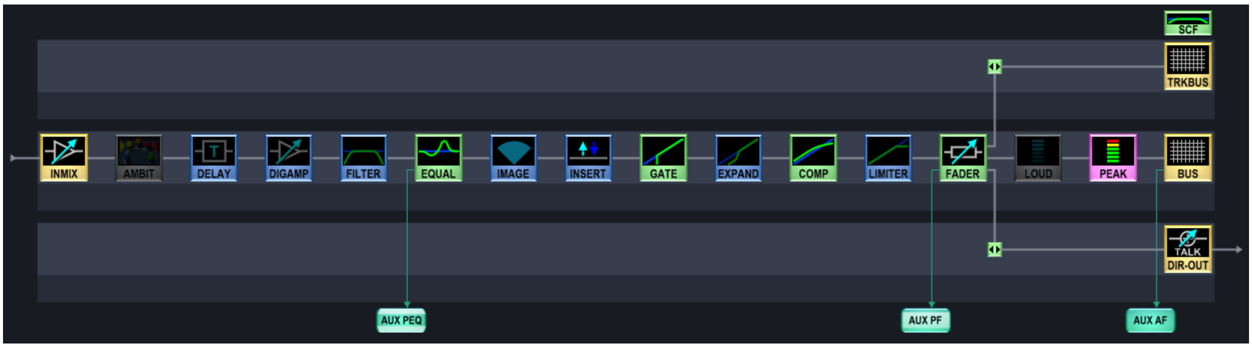
The number of input, monitor, group, sum and aux channels is determined by the number of channel DSP boards fitted to the Nova73 (up to 8); the sampling rate of the system (48/44.1kHz or 96/88.2kHz); and your choice of DSP configuration.

The DSP configuration is selected from a predefined list and stored when you save the production. DSP configurations are available in a choice of channel type:

- **Broadcast Channels** – provide twice as many channels per DSP board; each channel has a simplified signal flow (no track bus send, no independent filter section and simpler dynamics).
- **Recording Channels** – less channels per DSP board; each channel provides more processing and increased flexibility.

Input Channel (Broadcast)

Input Channel (Recording)



Each DSP configuration supports one channel type; you cannot mix Broadcast and Recording channels. To check that your system supports Broadcast channels, see Broadcast Channel Conditions.

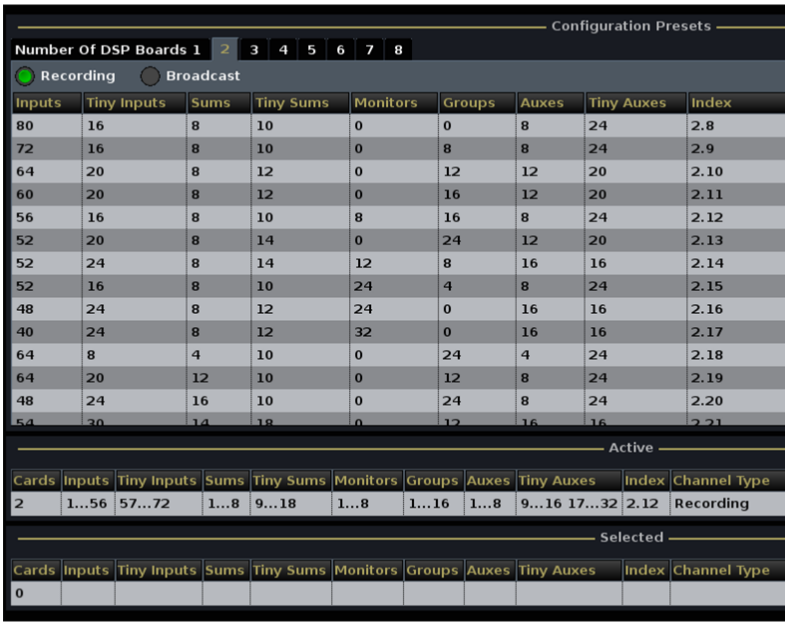
Once you have loaded a DSP configuration, you may modify the order of the processing modules (EQ, Delay, etc.) from the Channel Config display. This allows you to change the signal flow on a channel-by-channel basis.

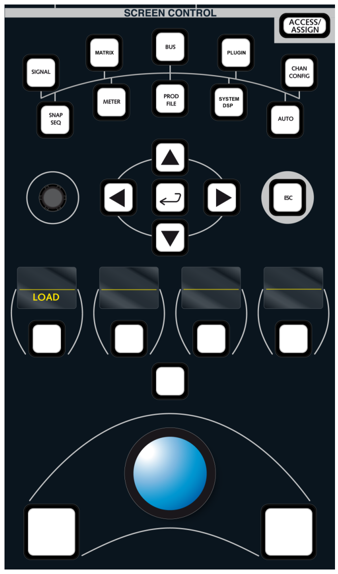
DSP Configurations

The DSP configuration defines the number of inputs, monitor returns, groups, sums and auxes; whether those channels use fully featured or reduced processing (full or tiny); and whether they are Broadcast or Recording channels. It is saved in the production, but not in snapshots (as changing the DSP configuration causes a brief interruption to audio).

The current configuration can be viewed on the **DSP Configuration** display:

1. Press the **SYSTEM DSP** button, located on the SCREEN CONTROL panel, to view this display:





The upper area lists the **Configuration Presets** available for different numbers of DSP boards; the number of boards is highlighted at the top – in our example, **2**. (Note that, even if your system is only fitted with 1 DSP board, you will be able to view the **Configuration Presets** for up to **8** boards – this tells you what could be available if you upgrade your system!)

The **Recording** and **Broadcast** radio buttons appear if your system supports Broadcast channels. Notice that the channel count effectively doubles when you select the **Broadcast** radio button.

The **Active** summary shows the details for your current DSP configuration. This is the configuration preset which is loaded.

The **Selected** summary provides similar details for the selected configuration. This allows you to interrogate an alternative configuration before making it active.

In each case you will find the following information:

- **Cards** – the number of DSP boards used by the DSP configuration.
- **Inputs** and **Tiny Inputs** – the number of fully featured and reduced processing input channels.
- **Sums** and **Tiny Sums** – the number of fully featured and reduced processing sum channels.
- **Monitors** – the number of monitor return channels. Note that monitor channels are always created using full audio processing, and are only available when using Recording channels.

- **Groups** – the number of group channels. Note that groups are always created using full audio processing.
- **Auxes** and **Tiny Auxes** – the number of fully featured and reduced processing auxiliary channels.
- **Index** – this is a unique reference number for the DSP configuration. You may be asked for this number when contacting Lawo for operational or technical support.
- **Channel Type** – shows whether you are using Broadcast or Recording channels.

All resources are displayed as mono channels. For example, a configuration with 24 inputs provides 24 mono input channels, or 12 stereo input channels, or any combination such as 16 mono plus 4 stereo input channels. Similarly, if you configure your main sum output for a surround format, this uses 4, 6, 7 or 8 of your available sum channels. For more details, see Stereo Channels and Surround Channels.

> The available channel count is affected by the number of DSP boards fitted to the Nova73 (up to 8), and the sampling rate of the system (48/44.1kHz or 96/88.2kHz). The sampling rate is defined by AdminHD and may not be modified by the user. Higher sample rates (e.g. 96kHz or 88.2kHz) use twice as much DSP resource as lower sample rates (e.g. 48kHz or 44.1kHz). Therefore you will see more input channels at lower sampling rates.

For more details on the differences between Broadcast and Recording, Full and Tiny, and Input, Monitor, Group, Sum and Aux channels, please see DSP Channel Types.

Changing the DSP Configuration

DSP configurations may be changed at any time, making it easy to modify the mix structure if, for example, the production requires some additional groups or inputs. Please note:

- Loading a new DSP configuration causes a brief interruption to audio. Therefore, it is not recommended to change DSP configuration while live on air!
- Changing from a Recording to Broadcast DSP configuration, or vice versa, midway through a production is not advised. This is because a mix started with Recording channels will not sound the same on Broadcast channels. Therefore, to avoid confusion, all channel DSP settings are reset to flat if you change the channel type.

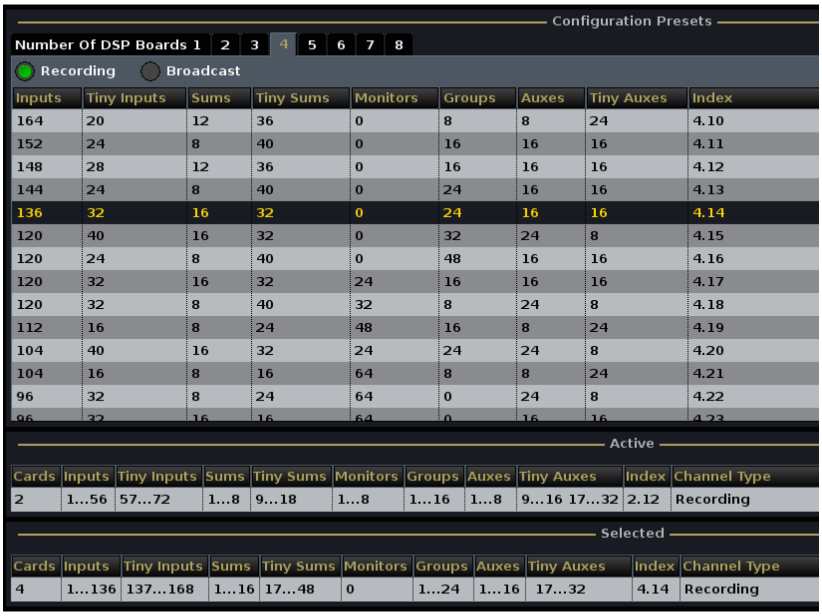
To change the DSP configuration:

1. Open the **DSP Configuration** display.
2. Select the **Number of DSP boards** fitted your system – in our example, **4**.
3. Select the channel type, **Recording** or **Broadcast**.

> These buttons are not visible if your system does not support Broadcast channels, see Broadcast Channel Conditions.

4. Then select one of the available **Configuration Presets** – in our example, **Index 4.14**.

The details are highlighted in black and are displayed in the **Selected** summary column allowing you to make a side-by-side comparison with your **Active** configuration:

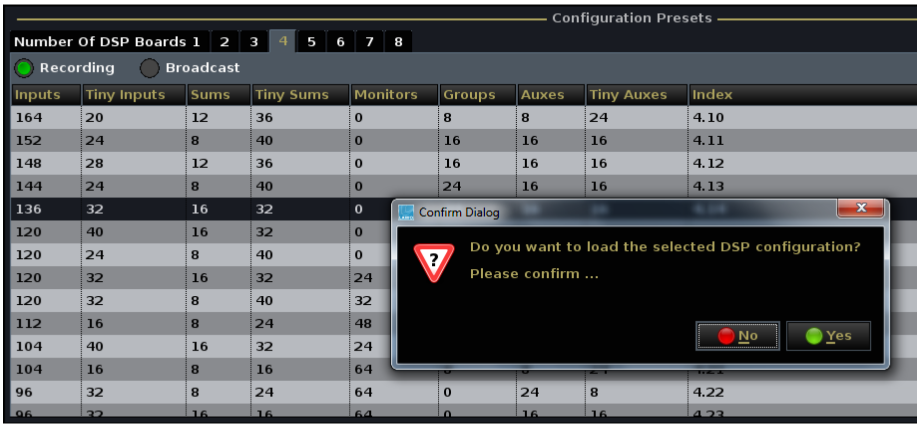


5. Right-click and select **Load** to continue.

One of two possible dialogue confirmation boxes appears...

Changing DSP Configuration (Same channel type)

If the selected DSP configuration uses the same channel type (Recording or Broadcast channels), then loading will cause a brief interruption to audio, but will not interfere with your DSP settings. In this case, you will see the following confirmation box:



Select **Yes** to proceed.

The console re-configures its processing, and the **Active** summary updates to reflect the new configuration.

> If the **Active** summary does not update, then the new DSP configuration could not be loaded. This can occur if you try to load an invalid selection – for example, a DSP configuration which requires more DSP boards than are physically available. See Transferring User Data for more details.

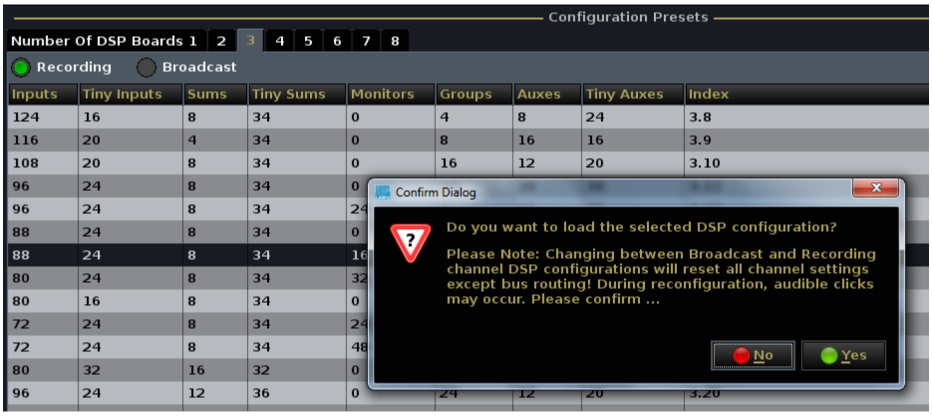
After a successful DSP configuration load, a number of things may happen to the DSP resource:

- If you have increased the amount of resource, for example you now have 8 groups rather than 4, any DSP settings applied to groups 1 to 4 remain intact, and the additional groups become available in the usual manner.
- If you have reduced the amount of full processing resource, the console will allocate tiny processing where possible. For example, you have reduced the number of Inputs from 24 to 20 but have 4 Tiny Input channels available - input channels 21 to 24 are automatically configured with tiny processing.
- If you have reduced the amount of full and tiny processing resource, then channels are removed but their settings remain in virtual memory. For example, if your input channel count has fallen some input channels are no longer available and are removed from the control surface. However, all settings for the previous configuration are stored. This means that if you recall the previous configuration, the settings for those channels are reinstated.

Changing DSP Configuration (Different channel type)

If the selected DSP configuration uses a different channel type (Recording to Broadcast, or Broadcast to Recording), then loading will significantly change the signal processing. As a result, all channel DSP settings (including EQ, Dynamics, Delay, Fader levels, etc.) are reset to their factory defaults, with the exception of bus assignments. In other words, any EQ parameters are reset to flat, faders to off, and so on.

To warn you that this is about to occur, the following dialogue box appears:



Select **Yes** to proceed.

The console re-configures its processing, including the channel type, and the **Active** summary updates to reflect the new configuration.

After a successful change of channel type, all DSP settings are reset to flat, with the exception of bus routing.

If you change the channel type in error, then don't panic! The system automatically saves a Fallback snapshot before each DSP configuration load. This provides a way of recovering settings if required.

Fallback Snapshots

The system automatically saves a fallback snapshot before each DSP configuration load. This provides a way of recovering settings should you change the channel type (Recording to Broadcast, or Broadcast to Recording) by accident.

To recover your settings:

1. Make a note of the time when you loaded the wrong DSP configuration, and also the correct channel type for your mix – Recording or Broadcast.

2. Then load a compatible DSP configuration - Recording or Broadcast - from the **DSP Configuration** display.

3. Press the **SNAP/SEQUENCE** button, on the SCREEN CONTROL panel, to view the **Snapshots List** display.

4. And select the **FALLBACK** folder:

A fallback snapshot is automatically saved every time a new DSP configuration is loaded. The **FALLBACK** folder holds 10 snapshots, providing 10 levels of undo before the oldest fallback snapshot is deleted.

The **Channel type** column shows whether the snapshot was saved when a Recording or Broadcast channel DSP configuration was active. In our example, we have been changing between channel types a lot!

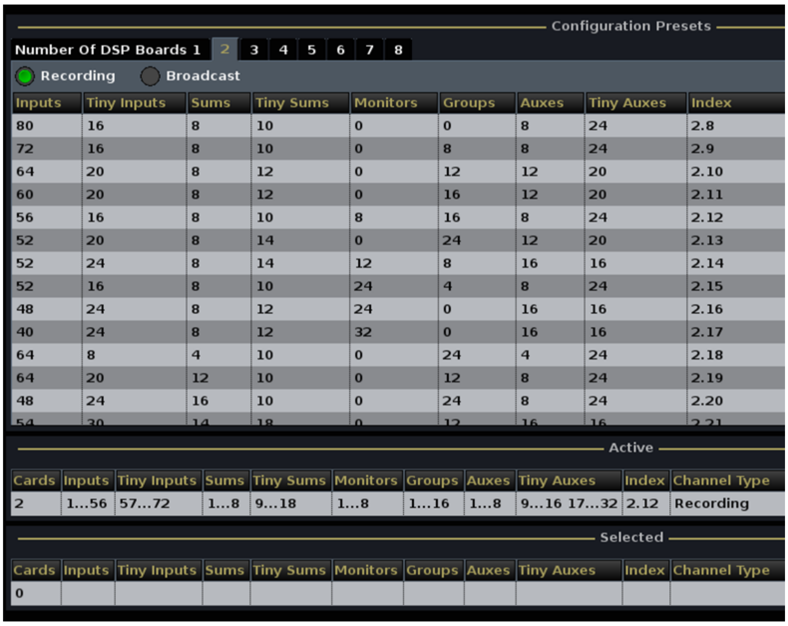
5. Load the correct fallback snapshot to match your chosen DSP configuration.

Your latest mix settings are reinstated.

Note that the 10 fallback snapshots are saved into the active production whenever you save or update the production.

Redundant DSP

Any number of DSP boards fitted to the Nova73 may be reserved to provide redundant processing in the unlikely event of a DSP board failure. You can investigate whether you have a redundant board from the **DSP Configuration** display:


Configuration Presets
Number Of DSP Boards 1 2 3 4 5 6 7 8
Recording Broadcast

Inputs	Tiny Inputs	Sums	Tiny Sums	Monitors	Groups	Auxes	Tiny Auxes	Index
80	16	8	10	0	0	8	24	2.8
72	16	8	10	0	8	8	24	2.9
64	20	8	12	0	12	12	20	2.10
60	20	8	12	0	16	12	20	2.11
56	16	8	10	8	16	8	24	2.12
52	20	8	14	0	24	12	20	2.13
52	24	8	14	12	8	16	16	2.14
52	16	8	10	24	4	8	24	2.15
48	24	8	12	24	0	16	16	2.16
40	24	8	12	32	0	16	16	2.17
64	8	4	10	0	24	4	24	2.18
64	20	12	10	0	12	8	24	2.19
48	24	16	10	0	24	8	24	2.20
54	30	14	18	0	12	16	16	2.21

Active

Cards	Inputs	Tiny Inputs	Sums	Tiny Sums	Monitors	Groups	Auxes	Tiny Auxes	Index	Channel Type
2	1...56	57...72	1...8	9...18	1...8	1...16	1...8	9...16 17...32	2.12	Recording

Selected

Cards	Inputs	Tiny Inputs	Sums	Tiny Sums	Monitors	Groups	Auxes	Tiny Auxes	Index	Channel Type
0										

If, for example, your Nova73 is fitted with 2 DSP boards and the **Active** configuration uses 1, then the second board will provide redundancy in the event of a failure.

This can also be seen on the Nova73 front panel where the spare board is shown in **STANDBY**:

> Note that the system uses boards from right to left across the front of the Nova73. So if board 8 is redundant, it is the DSP board on the left of the core.

In the unlikely event of a failure, the system automatically switches all DSP resources and settings from the faulty board to the spare; the faulty board may then be safely removed and replaced.

The replaced board now acts as the spare board either until the Nova73 is restarted or a new DSP configuration is loaded. Following the reconfiguration, boards are utilised from right to left across the Nova73, in our example slots 1 to 7 for main DSP resources and slot 8 in **STANDBY**.

Upgrading Your DSP Configuration

By now, your hardware specification will have been pre-determined, unless of course you are reading this prior to console purchase! However, you may upgrade your system by retrofitting additional DSP boards at any time in the future.

DSP Channel Types

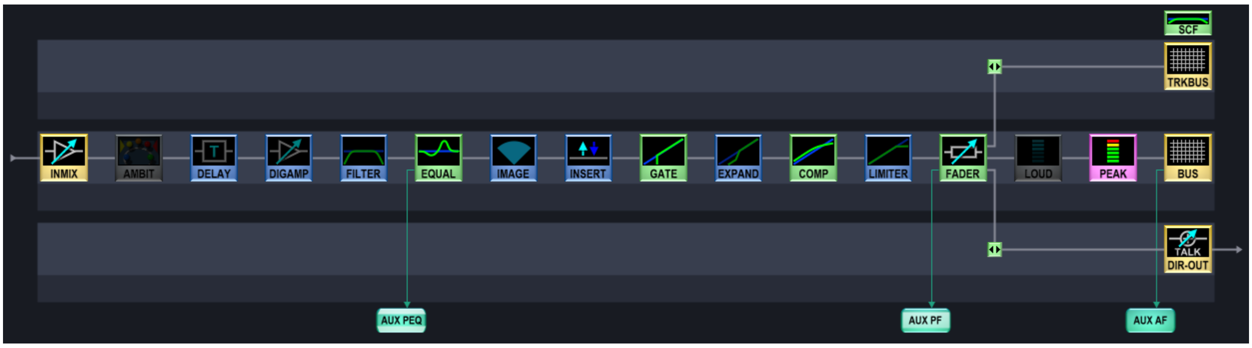
This section looks at the differences between:

- Recording and Broadcast channels.
- Other Channel Types - Input (INP), Group (GRP), Sum (SUM), Aux (AUX) and Monitor (MON) channels.
- Full and Reduced processing (Tiny) channels.

For details on interrogating or changing the channel types, see DSP Configurations.

Recording Channels

Every full processing input channel (INP), within a Recording DSP configuration, includes all of the following audio modules:



- **IN MIX** – channel input gain, phase and stereo input control.
- **AMBIT** – upmix and spatialise processing.
- **DELAY** – delay, adjusted in frames, ms or m.
- **DIGAMP** – digital gain trim.
- **FILTER** – 2-band filter/equaliser section.
- **EQUAL** – 4-band equaliser section offering a choice of characteristics.
- **IMAGE** – controls the image for a stereo channel. (Not active on mono channels.)
- **INSERT** – insert send and return for outboard processing. The insert send is always active providing an additional send.
- **GATE**, **EXPAND**, **COMP**, **LIMITER** – 4 independent dynamics.
- **FADER** – fader level, mute and AFL/PFL monitoring.
- **LOUD** – the channel's loudness meter pickup point.
- **PEAK** – the channel's peak meter pickup point.
- **AUX PEQ**, **PF**, **AF** – available pickup points for each aux send.
- **TRKBUS** – pickup point for assignments to track busses.
- **BUS** – main signal flow feed to group and sum busses.
- **DIR-OUT** – pickup point for the direct output.
- **SCF** – the channel's dynamics sidechain processing.

> When channels are defined for 5.1 surround with AMBIT processing active, the AMBIT module replaces the Delay, Filter, Image, Gate and Expander. When loudness metering is active, the LOUD module replaces your choice of DSP modules.

With the exception of the yellow INMIX and BUS sections, modules may be positioned in any order, see Channel Config.

Broadcast Channels

When a Broadcast channel DSP configuration is active, the signal flow of a full processing input channel (INP) is simplified:

The main differences to Recording Channels are that a Broadcast channel has no track bus send, no Filter section and simplified Dynamics (with a choice of Gate, Expander and Compressor, known as Combi-Dynamics, *or* Limiter. In addition, there are some restrictions on module positioning, and some limitations on DSP parameter values.

See Broadcast vs Recording channels for full details.

Choose Broadcast channels if you prefer a simpler channel, and wish to access more channels from the same DSP board resource.

Choose Recording channels if you wish to use track busses (and monitor channels), or if you require more complex signal processing: for example, to position the Gate, Expander, Compressor and Limiter independently.

Broadcast Channel Conditions

Broadcast channels are *NOT* supported if:

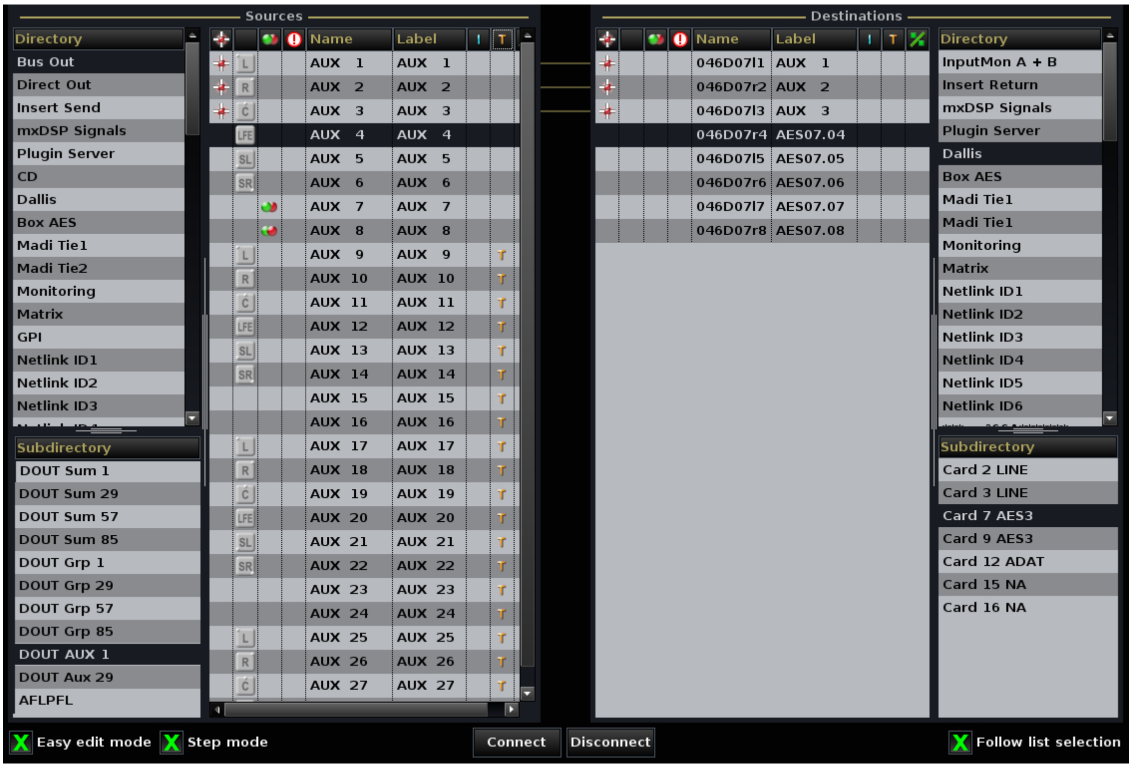
- Your system is running at higher sample rates such as 96kHz.
- Your system uses a 3K MkI Router module (used in some classic mc² systems).
- Your system has DSP 983/02 cards (Broadcast channels are only supported by DSP 983/03 cards).

If any of the above are true, then you will not see the **Recording** or **Broadcast** radio buttons on the DSP Configuration display. In such cases, the channel type is always the default (Recording channels).

Accessing the Additional Resources

If you have upgraded from an earlier software release (to V4.16), then you will need to update your **Signal List** configuration (gui_config.tcl), using AdminHD, in order to access the additional input channels, groups and sums.

Once updated, you will see the additional resources (sums and groups up to 96) within the **Signal List** display under **Bus Out**:



Please consult your technical department if this is not the case.

Broadcast vs Recording Channels

Each DSP configuration supports only one channel type; you cannot mix Broadcast and Recording channels. The differences between the channel types are:

DSP Module/Function	Broadcast Channels	Recording Channels
INMIX (Input Section)	✓	✓
DIGAMP (Digital Gain)	✓ Fixed position.	✓ Variable position.
IMAGE (stereo ch only)	✓ Fixed position.	✓ Variable position.
AMBIT (upmix)	✓ Suspends Dynamics, Delay and Insert.	✓ Suspends Dynamics, Delay and Insert.
DELAY	✓ Min. = 18 smpl / 0.38ms Max. = 1.3 s	✓ Min. = 1 smpl / 0.02ms Max. = 1.8 s
EQ **(4-band Filter/Shelf/Parametric)**	✓ Max. 2nd order filter	✓ Max. 3rd order filter
FILTER (2-band filters)	×	✓
SCF (2- band sidechain filters)	×	✓
INSERT	✓	✓
Dynamics: **GATE** **EXPANDER** **COMPRESSOR** **LIMITER**	Combi-Dyn OR Limiter ✓ × ✓ × ✓ × × ✓ In Combi-Dynamics, the GATE, EXP and COMP move together within the channel signal flow. Each section can be turned on/off separately, but thresholds cannot overlap; the 3 sections share one Look Ahead Delay value.	4 independent dynamics ✓ ✓ ✓ ✓ Each section can be positioned independently with separate on/off. There are no limitations on threshold values, and each section has its own Look Ahead Delay. You can also apply an external key and sidechain filtering.
FADER (Level, Mute, AFL, PFL)	✓	✓
LOUD (Loudness Meter)	✓ Suspends selected DSP modules.	✓ Suspends selected DSP modules.
PEAK (Peak Metering)	✓	✓
TRKBUS (Track Bus Send)	×	✓
BUS (Main Bus Send)	✓	✓
DIROUT (Direct Out)	✓	✓
Aux Sends: **Pre EQ** **Pre Fader** **Post Fader (AF)**	 × ✓ ✓ (after fader)	 ✓ ✓ ✓ (pre-bus)

For details on interrogating or changing the channel type, see DSP Configurations.

Other DSP Channel Types

Group Channels

A fully featured group (GRP) channel is identical to an input channel with the exception of no INMIX section:

Group Channel (Recording channel DSP Configuration)

This means that a group can be reassigned to another group or sum, it can feed auxiliary sends, and it has an independently configured insert point, direct output path (and track bus if using a Recording channel DSP configuration).

Sum and Aux Channels

A fully featured sum or aux channel is designed to be the final point in the signal chain. It features all signal processing modules, but cannot be reassigned to another bus (Sum, Group or Aux) and has no independent direct output path:

Sum Channel (Recording channel DSP Configuration)

Use the Insert Send to take an independent feed from a Sum or Aux channel at any point in the signal chain.

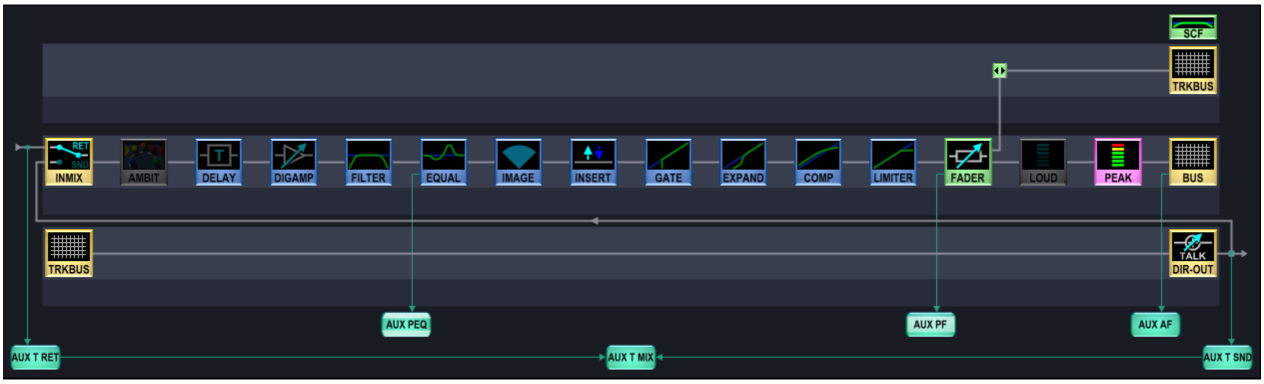
Monitor Channels and Track Busses

Monitor channels (MON) are designed for monitoring the send or return from a multitrack recorder.

A monitor channel is *always* associated with its corresponding track bus. So, for example, track bus 1 always feeds the send to monitor channel 1, track bus 2 feeds monitor channel send 2, etc. This means you can make track busses mono, stereo or surround by configuring the corresponding monitor channels to be mono, stereo or surround.

A full processing monitor channel is identical to an input channel with the exception of the INMIX section which features a send/return switch. The signal flow below shows the monitor channel path and its associated track bus:

Monitor Channel (Recording channel DSP Configuration)



Broadcast channel DSP configurations do *NOT* support monitor channels, so you must select a Recording channel DSP configuration if you wish to use this feature.

The number of monitor channels within the DSP configuration determines the number of track busses.

For more details on this application, see Monitor Channels.

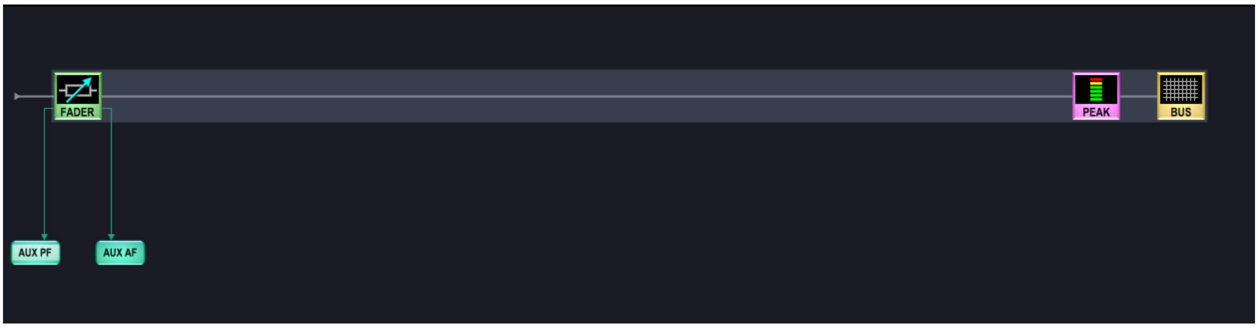
Reduced (Tiny) DSP Channels

Tiny channels have no signal processing modules (EQ, Dynamics, etc.) and, therefore, provide a channel with:

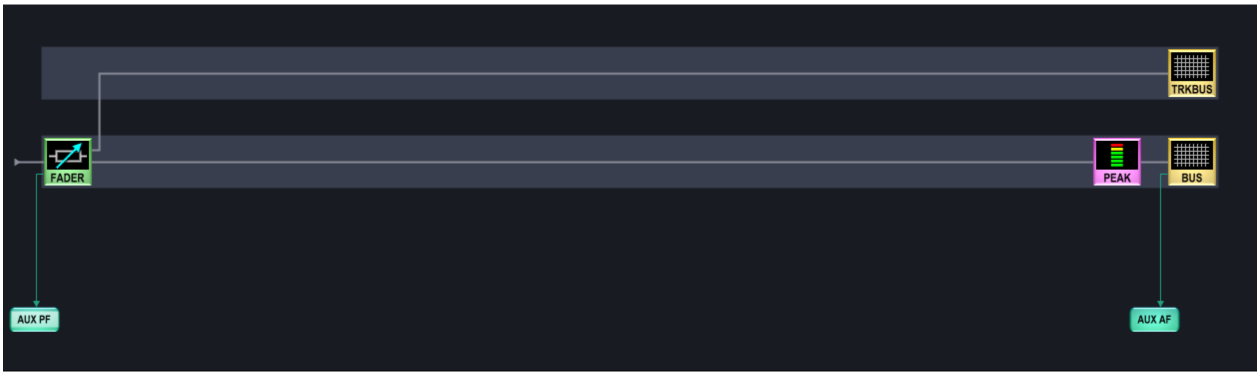
- **FADER** – fader level, mute and AFL/PFL monitoring.
- **BUS** – pickup point for group and sum bus assignments.
- **PEAK** – peak metering pickup point.
- **AUX SEND** – auxiliary sends which can be pre fader, or post fader for up to 32 auxiliary sends.
- **TRKBUS** – pickup point for track bus assignments (Recording Channel DSP configurations only).

Note that only input channels, auxiliaries and sums appear as tiny channels; groups and monitor channels are always configured with full audio processing.

Tiny Input Channel (Broadcast channel DSP Configuration)



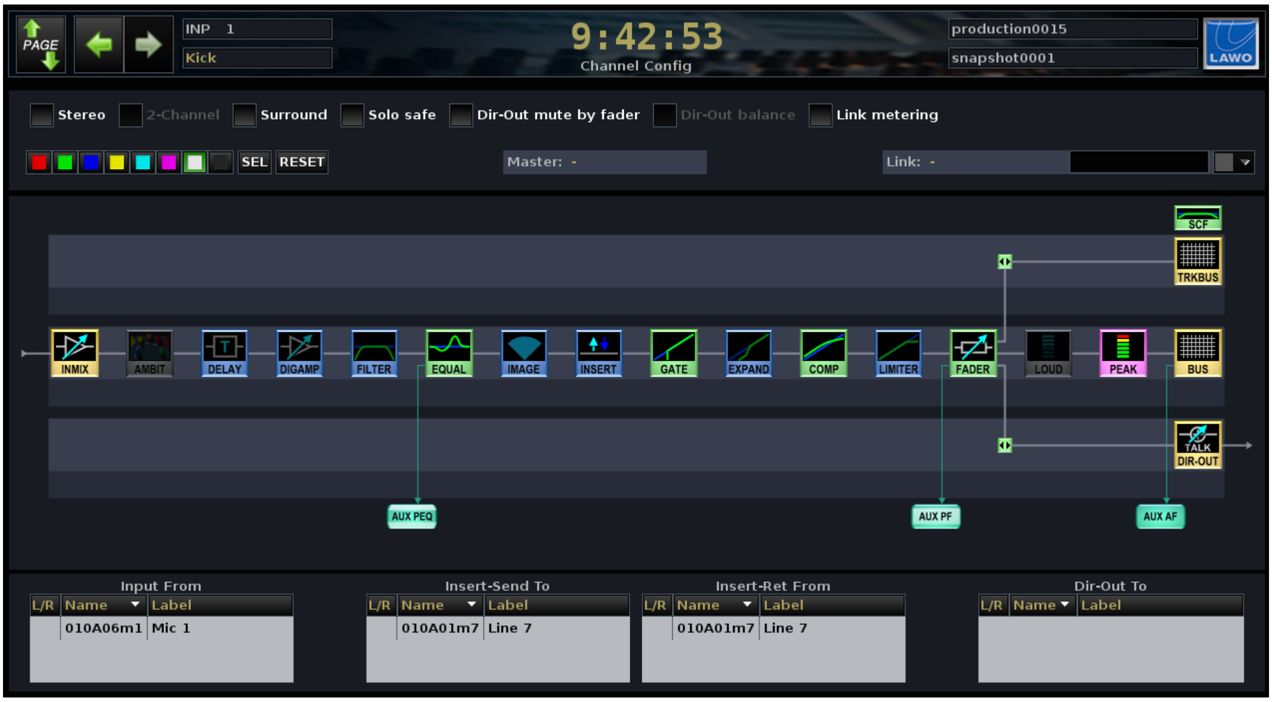
Tiny Input Channel (Recording channel DSP Configuration)



The Channel Config Display

Having loaded a DSP configuration, each channel's signal flow may be interrogated and modified from the **Channel Config** display.

1. Select the channel, by pressing its fader strip **SEL** button, or channel type and number from the ACCESS CHANNEL/ASSIGN control panel.
2. Press the **CHAN/CONFIG** button located on the SCREEN CONTROL panel:



At the top of the display are a number of channel configuration options (**Stereo**, **2-Channel**, **Surround**, etc.)

Below this you can apply colour coding to the selected DSP channel. And to the right, in the **Master** and **Link** fields, you will see the name of any VCA, Surround or Link masters (if assigned).

The main part of the display shows the signal flow for the channel in access – in our example, **INP 1**, a full processing input channel from a Recording channel DSP configuration. With the exception of the yellow INMIX and BUS assignment sections, audio modules may be positioned in any order in the chain. Audio modules coloured blue are switched off; those shown in green are switched on; those in grey are unavailable.

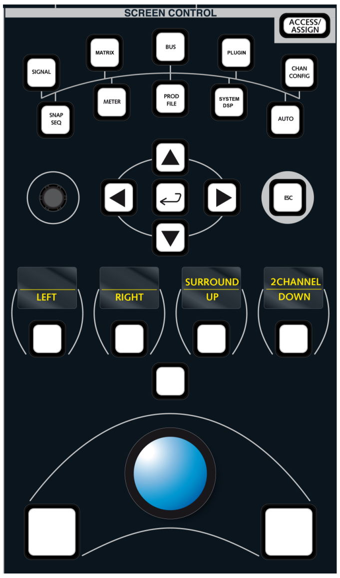
> When channels are defined for 5.1 surround with AMBIT processing active, the AMBIT module replaces the Delay, Filter, Image, Gate and Expander. When loudness metering is active, the LOUD module replaces your choice of DSP modules.

At the bottom of the display you will see the Names and Labels of any connections to and from the channel – the **Input**, **Insert Send**, **Insert Return** and **Direct Out**.

If a Broadcast channel DSP configuration is active, then the same principles apply but with the following differences:

- Broadcast channels have no **Filter** or **SCF** module.
- Broadcast channels have no **Track bus** or **pre-EQ** aux send.
- The **DIGAMP** and **IMAGE** modules always follow the **INMIX** section and cannot be moved independently.
- The **Channel model** defines the dynamics processing for the channel:
    - o **Combi-Dynamics** – a Gate, Expander and Compressor which can be moved as a single processing block anywhere within the signal flow.
    - o **Limiter** – a single Limiter module, which can be positioned anywhere within the signal flow.

Changing the Signal Processing Order



To change the signal processing order for the selected channel:

1. Using the trackball, select the audio processing module you wish to move.

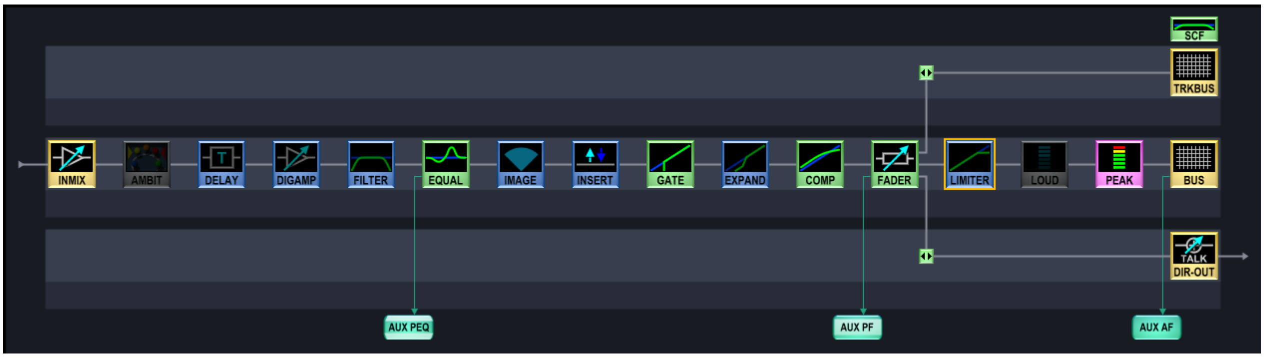
   The selected module is highlighted - in our example, the Limiter.

2. Use the soft keys or right-click to select:

- LEFT and RIGHT – moves the module left or right within the main channel signal path.
- UP and DOWN – moves the module into or out of the Track Bus, Channel or Direct Output path.

You can also press [**CTRL**] + [**Left/Right/Up/Down**] on the console keyboard to move the selected module.

The display updates to follow your changes:



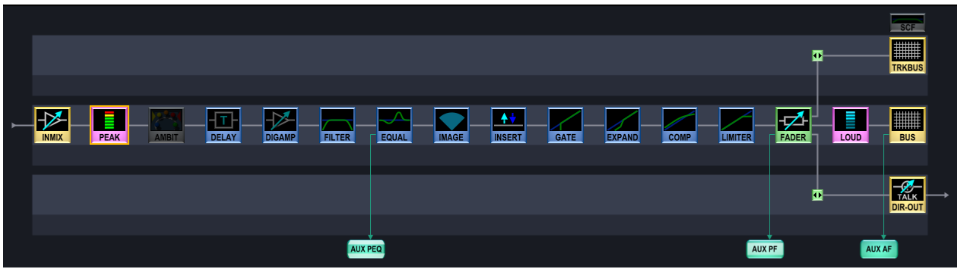
You cannot move the position of the **INMIX** or **BUS** modules.
When using Broadcast channels, you cannot select and move the **DIGAMP** or **IMAGE** modules.

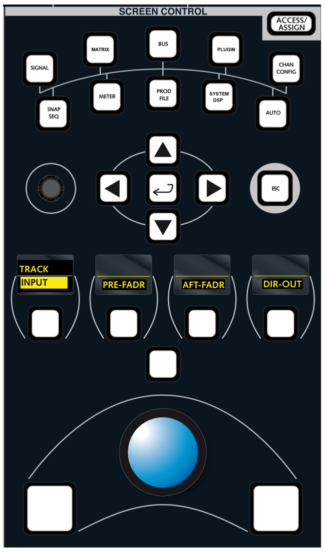
You can customise the signal processing order on a channel-by-channel basis for any input, sum, group, aux or monitor channel.
To adjust a range of channels, couple them and then change the processing order.
Alternatively, you can copy and paste the channel signal flow (**CH**) using the Parameter Copy/Assign panel.

The channel signal flow is saved in snapshots and productions.

Changing the Meter Pick up Point

1. If you select the **PEAK** audio module (pink), the soft key options allow you to change the channel peak metering point from a number of predefined options:





- **INPUT** – meters the channel input (post the INMIX section).
- **PRE FADR** – meters the pre-fader signal.
- **AFT FADR** – meters the input to the BUS assign module (after fader and processing).
- **DIR OUT** – meters the direct output.
- **TRACK** (Recording channels only) – meters the track bus output.

2. Select the **LOUD** audio module (pink when active), and the same soft key options allow you to select the pickup point for loudness metering.

You can select different channel pickup points for the **PEAK** and **LOUD** metering modules.

The meter pickup points may also be changed from the Extra Buttons display.

Changing the Dynamics Model (on Broadcast Channels)

This option is selected, using the trackball, from the **Channel model** drop-down menu when a full processing Broadcast channel is in access:

The **Channel model** option does not appear if a Recording channel DSP configuration is active.

For each full processing Broadcast channel, you can select:

- **Combi-Dynamics** – three modules: Gate, Expander and Compressor.
- **Limiter** – one module: a Limiter.

This allows you to use say Gating and Compression on one input channel, while applying a Limiter to another.

The **Limiter** or **Combi-Dynamics** may be placed anywhere within the channel signal flow, but note that the order within the **Combi-Dynamics** is *always* Gate, Expander, Compressor.

For surround channels, you may only select the **Channel model** on the surround VCA. This is because all surround slaves must be switched to the same model – either **Limiter** or **Combi-Dynamics**.

To set a range of channels to Combi-Dynamics, or Limiter, couple them and then change the **Channel model**.

Alternatively, you can copy and paste the channel signal flow (**CH**) using the Parameter Copy/Assign panel.

Limiter Model

With **Limiter** selected, the **LIM** module can be positioned anywhere within the channel signal flow in the usual manner:

Combi-Dynamics Model

With **Combi-Dynamics** selected, the **GATE**, **EXP** and **COMP** modules move together as one block:

1. Select any of the three modules, and you will see a green outline on the **Channel Config** display:

2. Use the soft keys or right-click to move the modules in the usual manner:

- LEFT and RIGHT – moves the module left or right within the main channel signal path.
- UP and DOWN – moves the module into or out of the Channel or Direct Output path.

> The order of sections within the Combi-Dynamics cannot be changed, and is always Gate, Expander and then the Compressor. If you wish to re-order dynamics modules, or have both a Limiter and Gate, Expander or Compressor, then switch to a DSP Configuration with Recording channels.

Each module can be turned on or off independently, and has separate threshold, ratio and other parameter values. For more details on operation, see Dynamics (Broadcast Channels): Combi-Dynamics.

Channel Config Options

The **Channel Config** display includes a number of other options for the channel in access:

Note that some options may be unavailable and are “greyed out” - for example, you cannot select **2-Channel** on input channels (only on sums, groups or auxes).

Note also that The fader strip **MUTE** buttons may be set to mute after the input mixer (pre-fader/pre-processing) or after the fader from the **System Settings** display, see Channel Mute.

➢ Stereo

Select this option to make the channel in access stereo.

The channel is automatically linked to its adjacent DSP path. For example, selecting **Stereo** on input channel 3 creates a stereo channel using inputs 3 and 4. Channels are always linked as an odd/even pair; you cannot make channels 4 & 5 stereo.

See Creating a Stereo Channel for more details.

➢ 2-Channel

Select this option to enable **2-Channel**, as an alternative to stereo. This provides independent fader strip control for the left and right sides of the output channel.

2-Channel is available for sum, group or aux channels only (not inputs or monitor channels). Channels are always linked as an odd/even pair; you cannot operate channels 4 & 5 in **2-Channel** mode.

See 2-Channel Mode for more details.

➢ Surround

Select this option to make the channel in access surround.

The channel is automatically linked to the next set of DSP paths. For example, selecting **Surround** on input channel 1 creates a surround channel using inputs 1 to 6 (for 5.1 surround). A Surround VCA is also automatically configured.

Note that there are certain restrictions on which channels can be linked for surround, so this option is greyed out unless you have a valid channel in access.

See Creating a Surround Channel for more details.

➢ Solo Safe

Select this option when Solo-in-place is enabled to prevent the channel being muted when a Solo is active. For example, you might select this option on your effects return channels so that you can hear both the source and the effect return when a channel is in Solo.

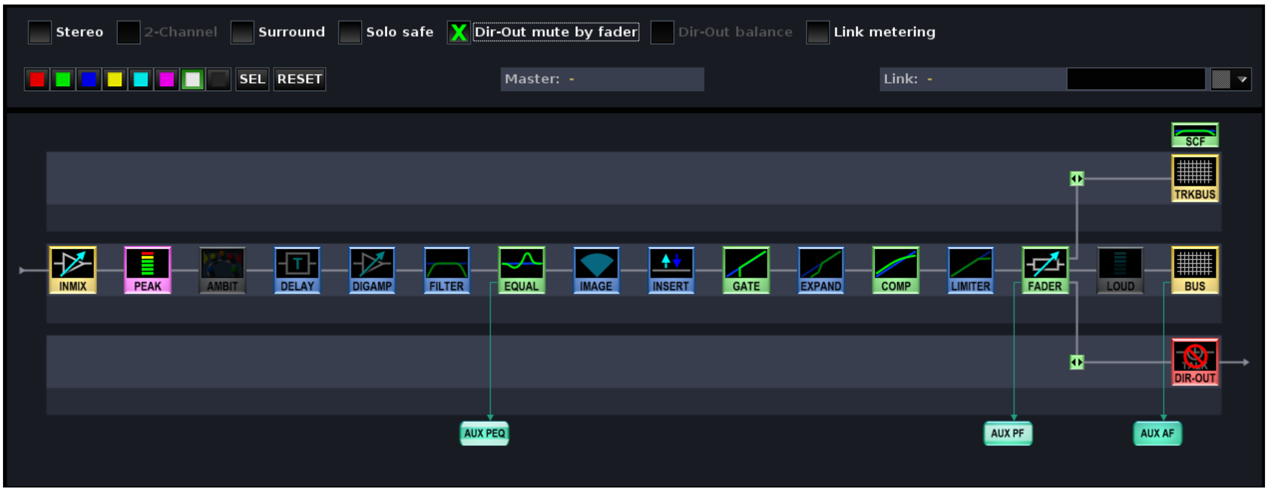
See Solo-in-Place for more details.

➢ Dir-Out mute by fader

This option sets the direct output to mute automatically when the channel fader opens.

It is designed for live broadcast applications where the direct out is positioned pre-fader to feed an intercom system, and the main programme feed is delayed (for example, when working with HD Cameras). By muting the intercom feed (direct out) when the channel sends to programme (fader open), echoes between the direct out and programme can be avoided.

Note that when the fader opens on the selected channel, the **DIR-OUT** module on the **Channel Config** display turns red to indicate the status of this option:



➢ Dir-Out Balance

This option determines whether the channel pan position affects the direct output on a stereo channel:

- **Dir-Out Balance disabled** (default) – the direct output does *NOT* follow the channel pan.
- **Dir-Out Balance enabled** – the direct output follows the channel pan position.

This is particularly useful for sum or aux masters as the left/right balance of the stereo master output can be readjusted using the channel pan control rather than having to use two mono faders.

Dir-Out Balance can only be enabled if the channel in access is stereo.

➢ Link Metering

This option affects the Channel display metering if the channel it is part of a link group.

When **Link Metering** is enabled, the first 8 linked channels are metered on any channel within the link group. This is useful if you want to leave only one channel on the surface and hide the remaining linked channels on a different bank or layer.

The option can only be enabled if the channel in access is part of a link group.

Channel Colour Coding

The **Channel Config** display can be used to colour code the selected DSP channel. For example, you might want to set all music channels to be white, VTRs to be blue, presenter mics to be red and so on.

The default colours are:

- Input and Monitor channels = white
- Groups = yellow
- Auxes = green
- VCAs = blue
- Sums = red

The colour coding affects the LAWO backlight at the bottom of each fader strip, the button-glow feature and the **Channel** display's fader sensing.

Colour code assignments are saved in snapshots and productions. This allows you to configure different colour coding for different snapshots during a show, or for different types of production.

1. To change the colour code of the channel in access, click on an option at the top of the **Channel Config** display - in our example, **INP 1** is set to white:

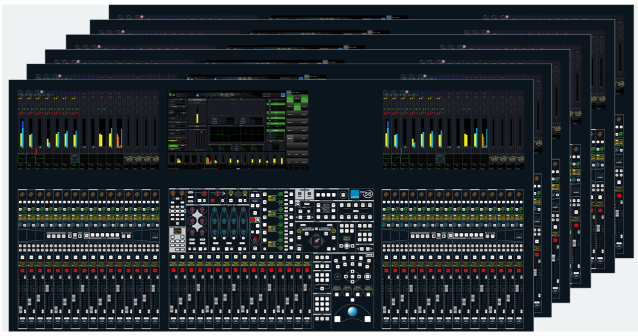
To assign a colour to a range of channels, couple them and then select the colour.

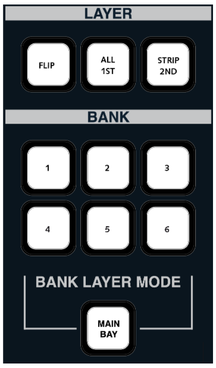
2. Or use the on-screen **SEL** button to copy the channel colour, as part of a copy and paste operation (see Parameter Copy/Assign).
3. Click on **RESET** to reset the colour code of an individual channel back to its system default.
4. Or, to reset all DSP channels to their default colour codes, select Reset colours (default) from the **System Settings** display.

Control Surface Configuration

The console's control surface includes both channel and main fader strips. Any fader strip may control any audio channel (input, monitor return, group, sum or aux), or any control channel (VCA, Surround VCA or GPC). This allows you to lay out your source channels, audio masters and control masters where you want them, see Fader Strip Assignment.

In addition, the physical size of the control surface does not restrict the number of audio processing channels. Additional channels may be added at any time by fitting more DSP boards to the Nova73; the extra channels are then accessed by paging the console's fader strips using banks and layers:





Banks and Layers

The console supports six control surface banks (1 to 6), each with two layers - Layer 1 and Layer 2.

Think of each bank as a separate console, with fast global or fader bay switching from one bank to another, see Bank Switching. Banks may be used to access different sets of channels (e.g. to switch from band 1 to band 2), or to switch between different fader strip layouts (e.g. to switch to an "effects" channel layout).

Within each bank, layers can be switched globally, within the fader bay, or individually, see Layer Switching. This makes layers ideal for related sources. For example, you could assign a presenter's input channel to Layer 1 with their mix minus aux master on Layer 2. Or, for multitrack recording, assign input channels to Layer 1 and monitor return channels to Layer 2.

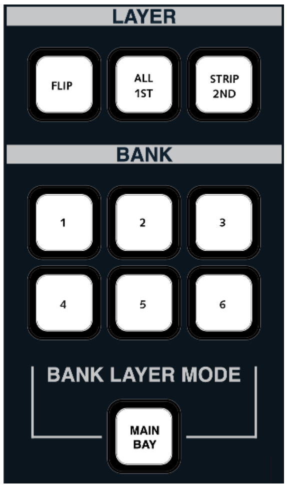
If you wish, you may isolate individual fader strips so that they never switch bank or layer.

Or, isolate fader bays so, for example, they can be used by a second engineer.

Bank Switching

You can switch between the 6 fader banks either globally across the whole surface, or locally within each 16-fader bay:

➢ Global BANK Switching



1. Locate the six BANK access buttons, numbered **1** to **6** in the centre section.

One of these buttons will be illuminated; this is your active fader bank.

2. With the **MAIN BAY** button turned off, you can switch *all* fader strips - channel and main - by pressing one of the BANK numbers **1** to **6**.

   *All fader labels, control positions and **Channel** displays update across the console to reflect the new settings. If there are no channels assigned to the bank, then you will switch to a series of blank fader strips.*

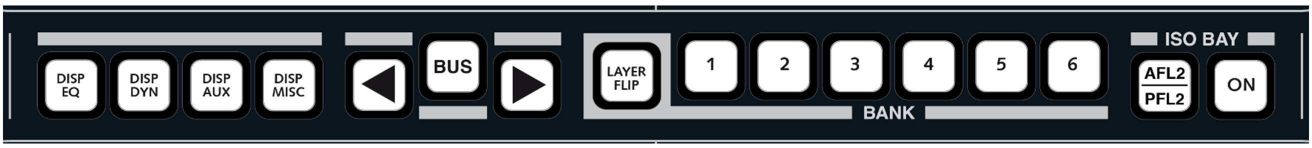
3. Turn on the **MAIN BAY** button to use BANK **1** to **6** to switch *only* the main fader strips.

(The **MAIN BAY** button does not exist on the classic mc²56. Instead you will find local BANK switching buttons, **1** to **6**, within the main fader bay.)

> If a fader strip or the fader bay is isolated, then the fader(s) are not switched by the global banking buttons. See Isolating Fader Strips and Isolating Fader Bays.

➢ Local BANK Switching

1. Locate the BANK **1** to **6** buttons on the ISO BAY panel:



One of these buttons will be illuminated; this is your active fader bank.

2. Press one of the BANK numbers (**1** to **6**) to switch to a different bank.

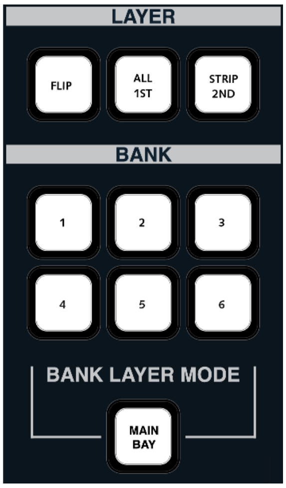
This time only the 16 fader strips within your local fader bay switch to the new bank.

> If a fader strip is isolated, then the fader(s) are not switched by the local banking buttons.
> If you press a global BANK button, this will reset the bank on the local bay (providing the bay is not isolated).

Layer Switching

Within each of the six banks, you can switch between layers 1 and 2 globally across the whole surface, locally within each 16-fader bay, or individually on each fader strip:

➢ Global LAYER Switching



1. Locate the LAYER access buttons in the centre section.

2. With the **MAIN BAY** button turned off, you can invert the layer of *all* fader strips - channel and main - by pressing **FLIP**.

   This inverts the individual fader strip **FLIP** buttons, allowing you to view all 'hidden' channels with one button push.

   *All fader labels, control positions and **Channel** displays update across the console to reflect the new settings. If there are no channels assigned to the alternate layer, then you will switch to a series of blank fader strips.*

3. Turn on the **MAIN BAY** button to use **FLIP** to invert *only* the main fader strips.

(The **MAIN BAY** button does not exist on the classic mc²56. Instead you will find a local LAYER **FLIP** button within the main fader bay.)

4. To temporarily switch *all* fader strips to either Layer 1 or Layer 2, press **ALL 1ST** or **ALL 2ND**.

   *This overrides the individual fader strip **FLIP** buttons. However, when you deselect **ALL 1ST** or **ALL 2ND**, the fader strip **FLIP**s are reinstated so that you return to your previous layer status.*

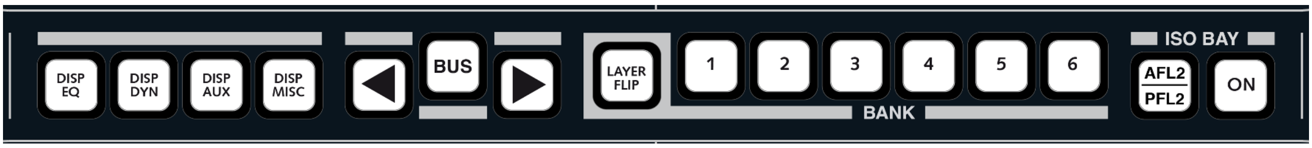
5. To permanently reset the layer of *all* fader strips, press and hold **ALL 1ST** or **ALL 2ND** for more than 3 seconds.

   *This resets the individual fader strip **FLIP** buttons to either all off or all on.*

If the fader bay is isolated, then the fader(s) are not switched by the global LAYER buttons. See Isolating Fader Bays.

➢ Local LAYER Switching

1. Press the **LAYER FLIP** button, on the ISO BAY panel, to invert *only* the fader strips within the 16-fader bay:



➢ Individual LAYER Switching

1. To invert the layer on a single fader, press the fader strip **FLIP** button.

*The fader's label, control positions and **Channel** display update to reflect the settings for the second layer. If there is nothing assigned to this layer, then you will switch to a blank fader strip.*

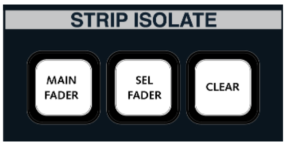
Isolating Fader Strips from Bank Switching

There will be times when you wish to keep a fader, for example your main presenter or sum master, on the control surface at all times.

The following method isolates fader strips from bank switching (but not layer switching).

If you wish to isolate a fader strip from both bank and layer switching, then an alternative approach is to assign the channel to the same position within every bank and layer using the ALL BANKS and BOTH LAYERS assignment options.

Alternatively, if you are working with more than one engineer and wish to provide separate 16-fader bays for the second engineer, you may use the ISO bay feature.



1. Press **MAIN FADER** to isolate *all* 16 main fader strips from bank switching.

If you switch banks and then deselect the **MAIN FADER** button, the fader strips update to reflect the settings for the selected bank.

2. Press **SEL FADER** to select individual fader strips to isolate.

*The fader **SEL** buttons across the console flash, in green:*

3. Press the **SEL** button(s) on the fader(s) you wish to isolate. You may select channel or main fader strips.

*The **SEL** buttons change colour from green to red to indicate that they are now isolated from bank switching operations.*

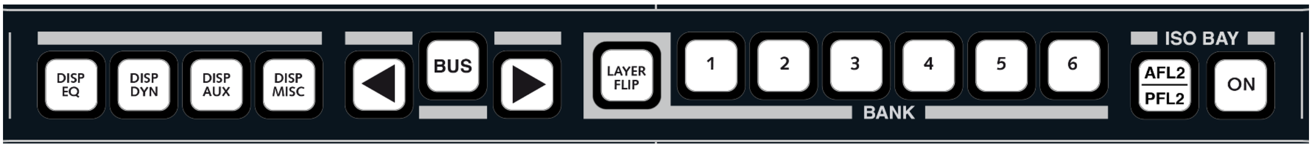
4. Deselect the STRIP ISOLATE **SEL FADER** button and now the selected faders remain isolated from bank switching.

5. To clear the bank switching protection, select **CLEAR**.

Isolating Fader Bays (ISO BAY)

To isolate a fader bay:

1. Press the **ON** button located on the ISO BAY access panel:



You can isolate multiple bays at any physical position.

Isolating fader bays has two principle applications:

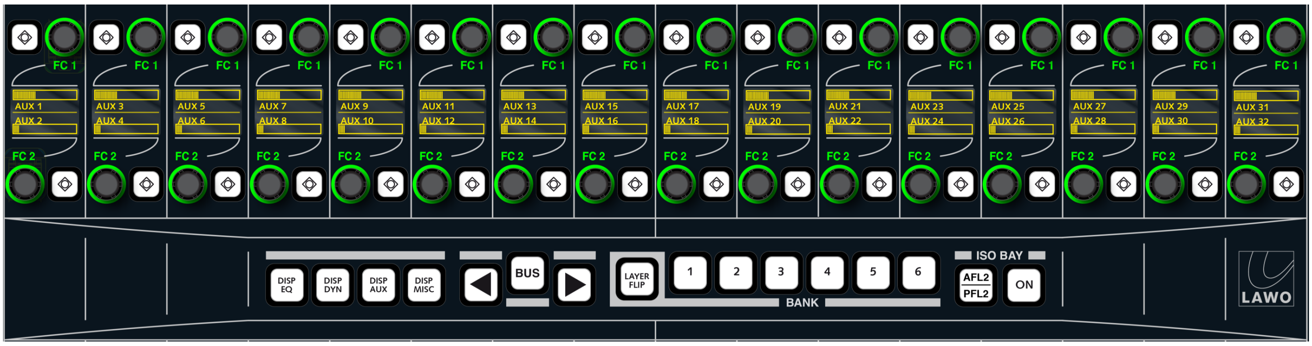
Multi-user Operation

ISO BAY **ON** isolates the 16-fader section from the centre section's bank and layer switching. This allows a second engineer to independently bank/layer switch and control DSP settings within an isolated bay, while the main engineer has full control of the rest of the console.

Isolated bays can be excluded from snapshot loads. And, the AFL/PFL bus can be split to provide a second AFL/PFL output from the isolated bay(s) if desired.

Local Parameter Control

The ISO BAY **ON** and **DISP** buttons temporarily override the default Free Control assignments, so that all 32 Free Controls within a 16-fader bay can access multiple parameters for the selected channel (e.g. aux sends 1 to 32):



All channel DSP parameters (EQ, Dynamics, Aux sends, Delay, etc.) and bus assignments can be accessed in this manner. Note that local parameter control is *NOT* available for the classic mc²56.

Bank and Layer Switching

On the isolated bay(s), you may now switch banks and layers independently from the main console using the local BANK **1** to **6** and **LAYER FLIP** buttons. See Bank Switching and Layer Switching for details.

Local bank and layer switching can be used, even if the fader bay is not isolated (ISO BAY **ON** disabled).

Independent AFL and PFL

AFL and PFL selections made within isolated bay(s) can be split away from the main console, and routed onto a second AFL and PFL bus, by enabling the ISO AFL2/PFL2 option in the **System Settings** display.

You can use this to provide the second engineer with independent headphone monitoring.

Global Snapshot ISO

To prevent snapshots recalled by the main engineer affecting the configuration of isolated fader bays, use the **BAY** Global Snapshot ISO option.

With Global Snapshot ISO **BAY** enabled, all settings on isolated bays are protected from a snapshot recall.

Fader Select (SEL)

The behaviour of the fader **SEL** button within the isolated bay is determined by the Select Isolate option in the **System Settings** display:

- **Select Isolate** (on) – the **SEL** buttons within isolated bays do NOT update the channel in access. Use this mode when you want isolated bays to work independently from the rest of the console. For example, when one engineer is working on an isolated fader bay and another with the rest of the console.
- **Select Isolate** (off) – the **SEL** buttons within isolated fader bays do update the channel in access. This mode is ideal for single operator use where you wish the channel in access to follow selections within isolated fader bays.

When working with **Select Isolate** on, note that if you deselect and then reselect the bay's ISO **ON** button, the console will remember the selected channel within the isolated bay so that you can return to adjusting its parameters easily.

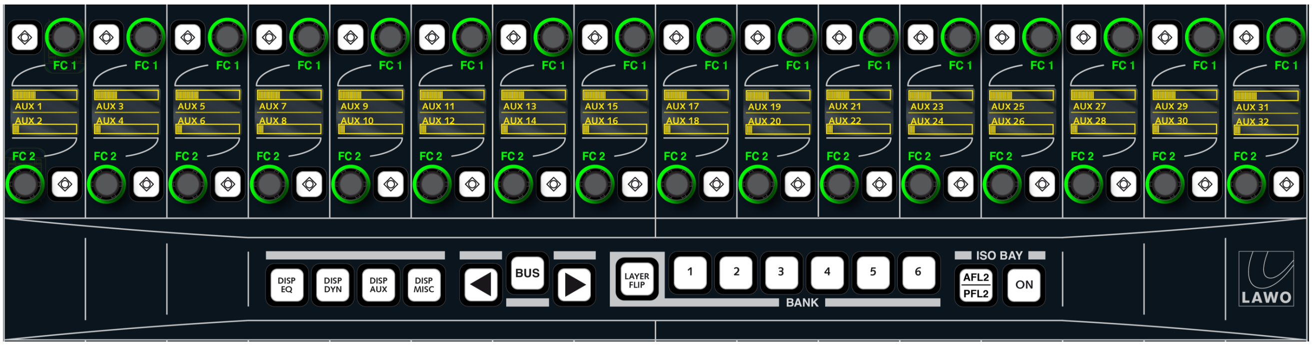
Local DSP Parameter Control

Within an isolated bay, the **DISP EQ**, **DISP DYN**, **DISP AUX** and **DISP MISC** buttons can be used to temporarily assign the 32 Free Controls to EQ, Dynamics, Aux or Miscellaneous parameters for the selected channel.

This function is *NOT* available on the classic mc²56.

1. First, if you haven't already done so, press the ISO BAY **ON** button to isolate the fader bay.
2. Next, select the parameters you wish to control – for example Aux sends by pressing **DISP AUX**.
3. Then select the channel you wish to adjust by pressing its fader **SEL** button:

The 32 free controls within the isolated bay update to show the AUX parameters for the selected channel:



4. Turn the controls to adjust each aux send from 1 to 32.
5. Press the button beside each send to turn the aux on or off.
6. Deselect **DISP AUX** to return the free controls to their default assignments.

All channel DSP parameters can be accessed in a similar manner as follows:

➢ DISP EQ

- EQ1 to EQ4 = Gain, Frequency, Q and EQ Type for the 4-band parametric EQ.
- FI1 and FI2 = Gain, Frequency, Q and EQ Type for the 2-band Filter section (Recording channels only).
- SC1 and SC2 = Gain, Frequency, Q and EQ Type for the 2-band dynamics sidechain filters (Recording channels only)

Press the button beside *any* of the EQ1 free controls to turn EQ band 1 on or off, and so on for the remaining bands/filters.

➢ DISP DYN

Threshold (THRS), ratio (RAT), attack time (ATT), release time (RLS), hold time (HOLD), floor level (FLR), make-up gain (GAIN) or look ahead delay (LAHD) for each of the dynamics sections: Gate, Expander, Compressor, Limiter.

Press the button beside any of the GATE free controls to turn the Gate on/off, and so on for the remaining sections.

If you are running a Broadcast channel DSP Configuration, then you will see either the Combi-Dynamics (Gate, Expander, Compressor) or Limiter parameters depending on the dynamics model of the selected channel.

➢ DISP AUX

Aux send level and on/off for auxes 1 to 32.

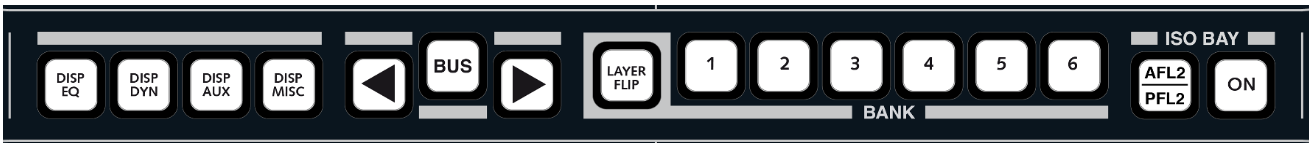
If an aux is stereo, then you can adjust the gain and pan, or gain and balance, for the stereo send.

➢ DISP MISC

- IN GAIN – channel input gain (INMIX gain).
- IN BAL – channel input balance (INMIX balance, if the input is stereo).
- DIGAMP – digital amplifier gain.
- DELAY – channel delay; press the button to switch in/out.
- INS SEND – insert send/return; press the button to switch in/out.
- DOUT LVL – direct output level; press the button to mute the direct output.
- PAN – panning parameters; press any of the PAN free control buttons to switch panning in/out of circuit.
- ON LEVEL, OFF LEVEL, etc. – Audio Follow Video (AFV) parameters.

Local Bus Routing

Within an isolated bay, the **BUS** button can be used to view or change bus assignments for the selected channel:

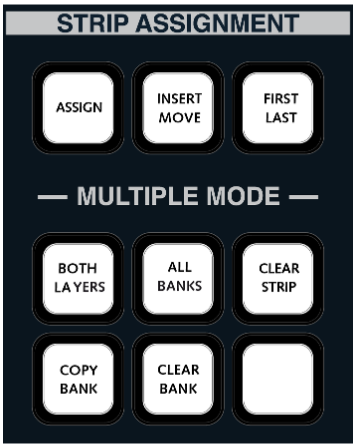


This function is *NOT* available on the classic mc²56.

1. First, if you haven't already done so, press the ISO BAY **ON** button to isolate the fader bay.
2. Select **BUS**, and the Free Control displays update to show bus assignments from the selected channel.
3. Use the left and right arrows to page through the different sets of bus outputs.
4. Press the Free Control button to enable or disable a bus assignment; the rotary control has no function in this mode.

Fader Strip Assignment

Fader strips are assigned using the STRIP ASSIGNMENT buttons in the centre section of the console:



Any audio processing or control channel may be assigned to any physical fader - channel or main fader strips.

If the assigned channel is stereo, then both the left and right sides are automatically controlled from a single fader strip.

For surround channels, you can assign the Surround VCA, for single fader strip control, and/or the individual component channels (e.g. L/R, C, LFE, Ls/Rs).

It is a good idea to configure mono, stereo and surround channels before making fader strip assignments, as the console will then distribute stereo faders automatically when making consecutive channel assignments.

If you want to clear the current assignments to start from a series of blank fader strips, use CLEAR BANK.

You can also insert or remove channels from a configuration, assign channels to multiple banks/layers, copy banks of fader strip assignments, or clear an individual fader strip.

The STRIP ASSIGNMENT buttons may be locked, to protect the existing console layout, by pressing the Lock **ACC** button located on the Extra Buttons display. Therefore, if you cannot assign to a fader strip, check the status of this option.

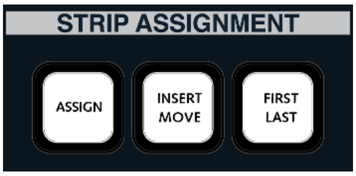
Fader strip assignments are stored and recalled in snapshots and productions. This allows you to store a single layout (in a production), or multiple layouts for recall during a show (using snapshots).

Assigning a Single Channel

To assign SUM 1 to a fader strip:

1. Select the channel - either by pressing its fader **SEL** button or entering **SUM**, the number **1** and **Enter** from the ACCESS CHANNEL/ASSIGN panel.

2. Press the global **ASSIGN** button, located on the STRIP ASSIGNMENT panel:



*The fader **SEL** buttons across the console flash, in green:*

3. Press a fader **SEL** button to complete the assignment.

*The fader **SEL** stops flashing and changes colour, from green to red.*

*In addition, the fader strip controls update to show the settings for the new assignment - e.g. Fader Label = **SUM 1**; fader is set to 0dB (default level); and so on.*

You can assign the same channel to multiple fader strips by keeping the **ASSIGN** button selected - for example, to switch to a different bank or layer. Note that this assigns the *same* channel to multiple places, so if you choose **SEL** buttons on the same bank or layer, then you will have lots of faders controlling a single channel!

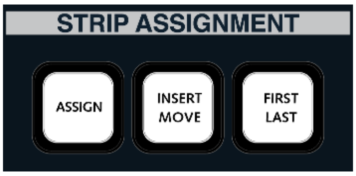
4. Deselect the **ASSIGN** button, or press **ESC** on the SCREEN CONTROL panel, to exit the fader strip assignment mode.

Assigning Consecutive Channels

To assign multiple fader strips in one operation:

1. Select the first channel in the range - either by pressing its fader **SEL** button or entering **INP**, the number **1** and **Enter** from the ACCESS CHANNEL/ASSIGN panel.

2. Press the **FIRST LAST** button, located on the STRIP ASSIGNMENT panel.



*This automatically selects global **ASSIGN**, and the fader **SEL** buttons across the console flash, in green:*

3. Press the fader **SEL** button on the first fader you wish to assign (e.g. strip 1) followed by the fader **SEL** button on the last fader (e.g. strip 24).

*The console incrementally assigns the input channels from the first selection (channel fader strip 1) to the last selection (channel fader strip 24), and cancels the **FIRST LAST** mode.*

If all the channels are mono, then you will have assigned INP 1 to 24 to fader strips 1 to 24.

If some channels are stereo, then they are automatically assigned to a single fader. For example, if INP 1&2 and INP 3&4 are stereo, then they are assigned to fader strips 1 and 2; INP 5 is assigned to fader strip 3, INP 6 to fader strip 4, and so on.

If some channels are surround, then it is the component channels which are assigned (e.g. L/R to fader strip 1, C to fader strip 2, LFE to fader strip 3, Ls/Rs to fader strip 4). You can control surround channels from a single master (called a Surround VCA), but this must be assigned to the control surface separately. See Surround VCAs for details.

The start and end of the range can be at any position across the control surface, and sources may be routed from left to right or from right to left by reversing the order of your first and last fader selection.

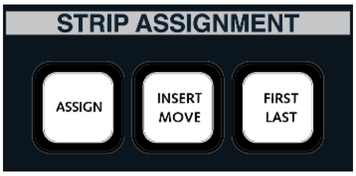
> Note that **FIRST LAST** operations treat channel and main fader strips independently, allowing you to assign consecutive channel fader strips without affecting main fader strip assignments or vice versa.

4. Deselect the **ASSIGN** button, or press **ESC** on the SCREEN CONTROL panel, to exit the fader strip assignment mode.

Inserting Channels

Normally, any existing fader strip assignments are replaced by the new selection. However, there are times when you may wish to keep your current assignments and insert a channel between two existing faders. For example, to insert an extra guest channel, onto fader strip 5:

1. Select the channel you wish to insert - either by pressing its fader **SEL** button or using the ACCESS CHANNEL/ASSIGN panel.
2. Press **INSERT MOVE**, located on the STRIP ASSIGNMENT panel.



*This automatically selects global **ASSIGN**, and the fader **SEL** buttons across the console flash, in green:*

3. Press the fader **SEL** button where you wish to insert the new channel - for example, on fader strip 5.

The replaced channel, and all channels to its right, move one step to the right across the control surface. And, the last channel assigned to the current bank/layer drops off the end of the console.

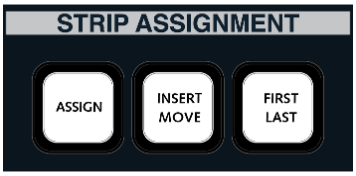
The settings for the end channel are stored as a virtual fader. For example, on a 48-fader control surface, the above operation would shift fader strip 48 up onto a virtual fader strip 49. This fader cannot be accessed, but remains in this location and will be added back onto the control surface if a fader strip assignment is removed.

Note that **INSERT MOVE** operations treat channel and main fader strips independently, allowing you to insert channel fader strips without affecting main fader strip assignments or vice versa.

Removing Channels

To remove a channel from a fader strip so that it does not leave a gap:

1. Select the channel by pressing its fader **SEL** button - this puts the channel into access.
2. Then press **INSERT MOVE**, located on the STRIP ASSIGNMENT panel.



*This automatically selects global **ASSIGN**. The fader **SEL** buttons flash, in green, except for the channel in access which should be red*

3. Press the red fader **SEL** button again to confirm the remove.

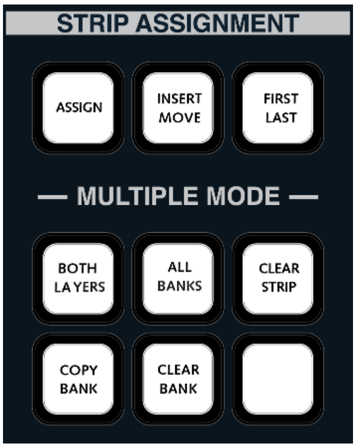
All channels to the right ripple down the control surface to fill in the gap.

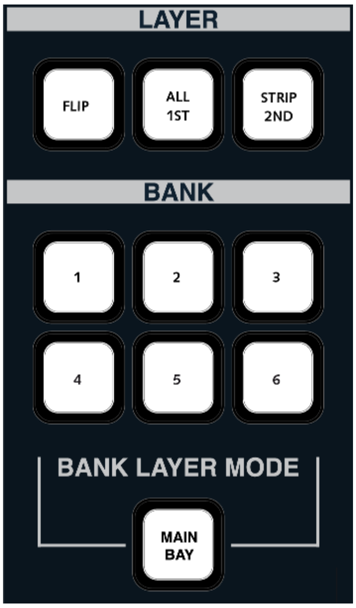
> Note that **INSERT MOVE** operations treat channel and main fader strips independently, allowing you to remove assignments from channel faders without affecting main faders or vice versa.

Assigning Channels to All Banks/Both Layers

You can assign channels to any bank or layer by selecting the bank or layer before you start the assignment process.

However, if you wish to assign a channel to the same fader strip across multiple banks or layers, then the STRIP ASSIGNMENT and BANK buttons can be used together to provide a number of short cuts:





For example, to assign input channel 1 to fader strip 1 across both layers of all six control surface banks:

1. Select **INP 1** - either by pressing its fader **SEL** button or using the ACCESS CHANNEL/ ASSIGN panel.
2. Press **ALL BANK** and **BOTH LAYERS**, located on the STRIP ASSIGNMENT panel.

*This automatically selects global **ASSIGN**, and the fader **SEL** buttons across the console flash, in green.*

*In addition the six BANK buttons **1** to **6** are illuminated.*

3. Deselect any numbers which you don't want to include in the assignment (you cannot deselect the current bank). For our example, keep all six BANK buttons lit.
4. Press a fader **SEL** button to complete the assignment - for example, on fader strip 1.

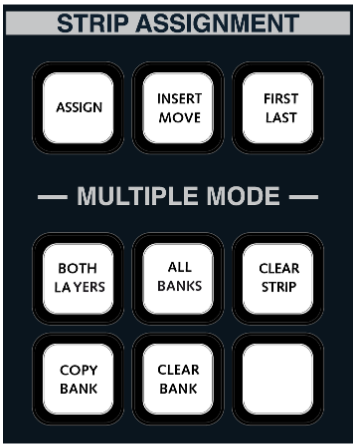
***INP 1** is assigned to fader strip 1 across both layers of all the selected banks.*

> You can select any combination of fader banks, with or without **BOTH LAYERS**, in order to achieve the desired fader strip assignment.

Clearing an Individual Fader Strip

To remove a channel so that it leaves a blank fader strip:

1. Press **CLEAR STRIP** on the STRIP ASSIGNMENT panel:



*This automatically selects global **ASSIGN**, and the fader **SEL** buttons across the console flash, in green:*

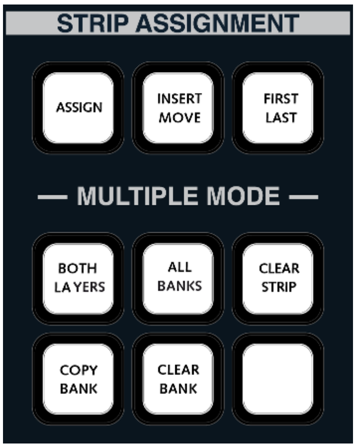
2. Press the fader **SEL** button(s) on any fader strips you wish to clear.

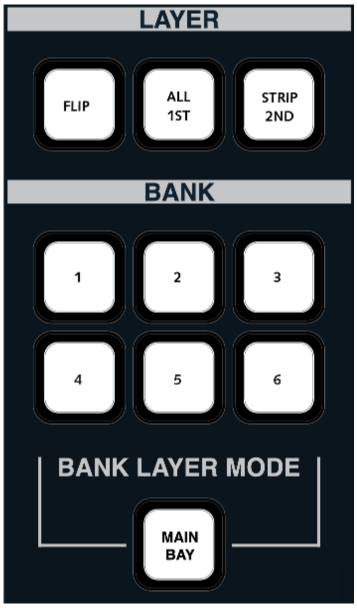
3. When you are finished, deselect **CLEAR STRIP** to prevent accidental changes to your configuration!

If you wish to remove a channel without leaving any gaps, use INSERT MOVE.

Clearing a Bank of Fader Strips

To clear a complete bank of fader strip assignments (including both layers):

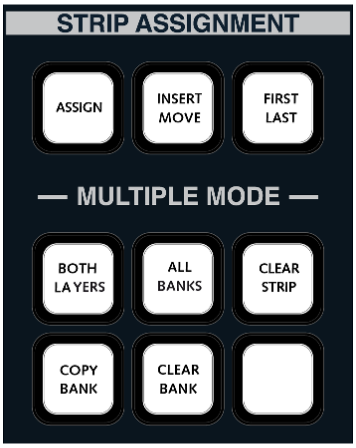


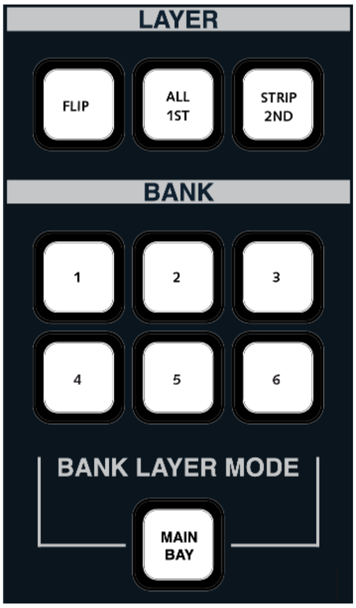


1. Press **CLEAR BANK** on the STRIP ASSIGNMENT panel.
2. Then select the bank or banks you wish to clear using the BANK **1** to **6** buttons.
3. Press Enter, located on the SCREEN CONTROL panel, to complete the operation.

Copying Banks

To copy the control surface configuration to a different bank or banks:



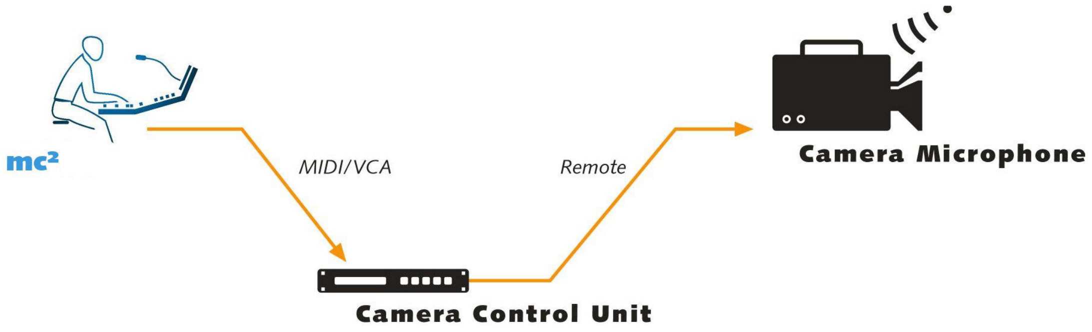


1. Select the bank you wish to copy using the BANK **1** to **6** buttons - the control surface updates accordingly.
2. Press **COPY BANK** on the STRIP ASSIGNMENT panel.
3. Then select the bank numbers to copy to, using the BANK **1** to **6** buttons.
4. Press **Enter**, located on the SCREEN CONTROL panel, to complete the operation.

General Purpose Channels (GPCs)

GPCs (General Purpose Channels) are control channels, assigned to any fader strip, which provide remote control of external devices (via MIDI). Typical applications include:

- Adjusting and storing camera microphone levels via MIDI to VCA converters:



- Adjusting fader levels and other parameters within a digital audio workstation (DAW):

The console supports up to 256 General Purpose Channels and each channel features the same control objects as a normal DSP channel – EQ, Delay, Compressor, Limiter, etc. Remember that these channels are for control only, and the parameters they adjust depend on the mapping within the MIDI protocol. For example, you may be using channel Delay to set the reverb time within an outboard effects unit. The assignment of channel objects to MIDI program changes and controller values is configured within the MIDI setup of the console. This is set within the factory configuration, so please refer to your console specification for details.

Fader Strip Assignment

GPCs may be assigned to any channel or main fader strip in the usual manner, by selecting **GPC** as the channel type.

Storing Settings

The settings for GPCs are stored in snapshots in exactly the same way as normal DSP channels.

Therefore, you may use SNAP ISO to isolate an individual GPC from snapshot recall, or protect all GPCs using the Global Snapshot ISO **DSP** option.

DSP Parameter to GPC Mapping

In addition to controlling external devices, GPCs may be used to control and automate DSP channel parameters. For example, to change a specific DSP parameter (e.g. Delay) from an Audio Follow Video event. By mapping the Delay parameter to a GPC fader, and then assigning the Audio Follow Video event to the GPC channel, the delay will be triggered when the AFV event is active.

This type of function is programmed from the Custom Functions display.

Monitor Channels

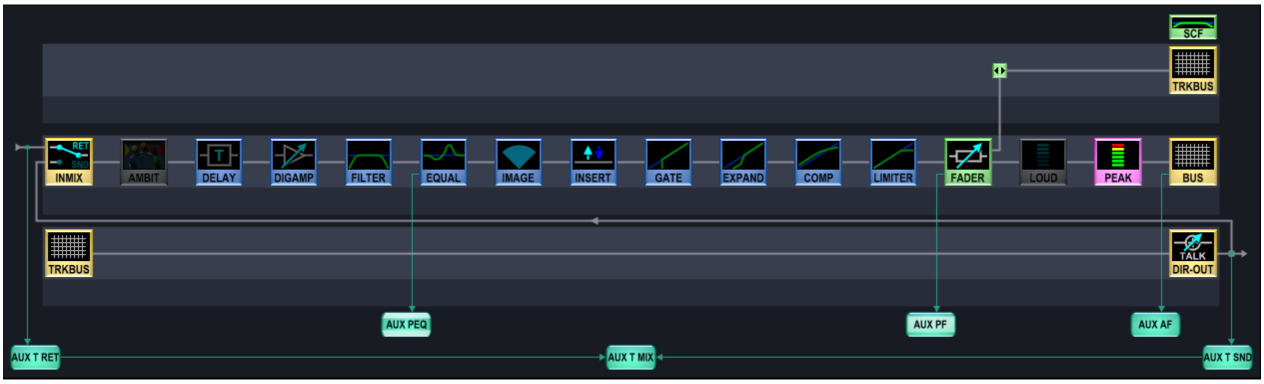
Monitor channels provide the ability to easily configure the **mc²56** for multitrack recording applications.

A monitor channel (MON) is identical to an input channel (INP) except for the following:

- Monitor channels feature a send/return switch in the INMIX section. Typically this is used to switch from monitoring the send to the return of the recorder.
- Monitor channels have no independent direct out (the direct out is used for the track bus).
- There are some special properties for auxiliary sends 17 to 32.

The signal flow below shows the monitor channel path and its associated track bus:

Monitor Channel (Recording channel DSP Configuration)



Broadcast channel DSP configurations do *NOT* support monitor channels, so you must select a Recording channel DSP configuration if you wish to use this feature.

The number of monitor channels within the DSP configuration determines the number of track busses.

Each monitor channel is *always* associated with its corresponding track bus. So, for example, track bus 1 always feeds the send to monitor channel 1, track bus 2 feeds monitor channel send 2, etc. This means you can make track busses mono, stereo or surround by configuring the corresponding monitor channels to be mono, stereo or surround.

In-Line Multitrack Recording

To simulate an in-line multitrack recording console:

1. Assign your input channels and monitor channels to the control surface, see Fader Strip Assignment.

> We recommend assigning input channels onto Layer 1 fader strips, and monitor return channels onto Layer 2, so that you can quickly switch between them using the fader strip **FLIP** buttons, see Layer switching.

2. Assign the sources you wish to record to your input channels using the Signal List display, and open your faders to set the record levels.

3. Assign the returns from the multitrack machine to your monitor channels and open your faders to set the monitoring levels.

4. And use either Forward or Reverse Bus Assign to route your input channels onto the track busses.

Note that a fixed relationship exists between each track bus and monitor channel. So, track bus 1 = monitor channel 1, track bus 2 = monitor channel 2, etc. Therefore, to route input channels onto track busses 1-24, you would select the monitor channels (MON 1 to 24) as your destinations.

> You can change the track bus pickup point from the Channel Config display.
>
> You can make track busses mono, stereo or surround by configuring the corresponding monitor channels to be mono, stereo or surround.
>
> You may also record other busses, such as a group, by assigning group channels rather than input channels to the monitor channels/track sends.

5. Now flip your monitor channels onto the layer 1 faders, and use the **SEND** and **RET** user buttons to switch the monitor channel input:

- **SEND** – the recorder send.
- **RET** – the recorder return.

Note that these functions must be programmed onto fader strip user buttons from the Custom Functions display.

> You can switch multiple channels using the Channel ALL function.

6. If machine control is configured, then you may have a **REC** user button to record arm the track.

7. Use the layer switching buttons to flip between your input channels (to control the send levels to the recorder) and monitor channels (to control the monitor mix).

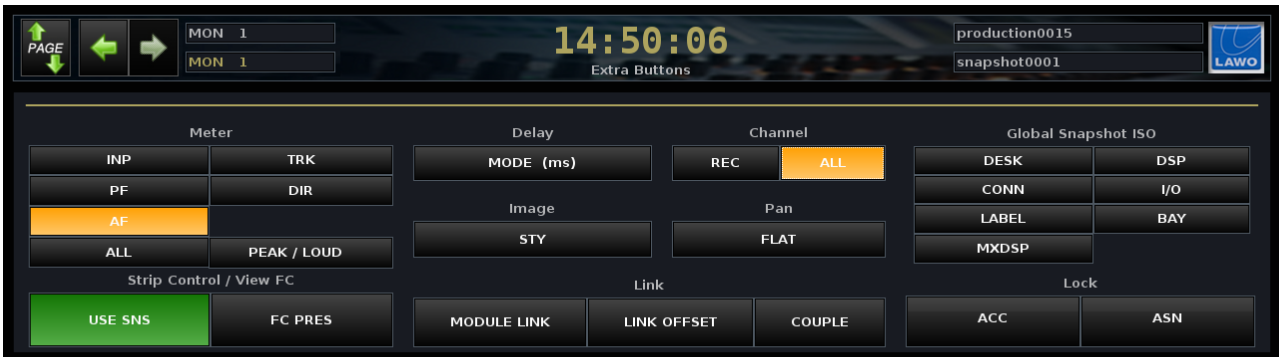
Normally for multitrack operation, a monitor channel cannot be assigned back to its associated track bus in order to prevent feedback. However, when using monitor channels for non-multitrack applications, you may override this feature using the Track Self Assign option in the **System Settings** display.

Switching the Input on Multiple Monitor Channels

The CHANNEL **ALL** button on the Extra Buttons display can be used to define a cluster of channels so that inputs are switched across multiple channels. For example, when recording, you may use this feature to switch all your monitor channels from send to return.

> This function is *only* available for monitor channels.

1. Press the **ALL** button, located in the **Channel** section:



*The **ALL** button flashes and the fader **SEL** buttons across the console flash, in green.*

2. Add channels to the cluster by pressing their fader **SEL** buttons.

*The fader **SEL** buttons turn red:*

3. Now press the **SEND** user button on any channel within the cluster.

   All channels within the cluster are switched to the send; channels not in the cluster are unaffected.

The **SEND** and **RET** input select buttons will continue to switch inputs within the cluster while the **ALL** button is lit.

(Note that **SEND** and **RET** must be programmed onto fader strip user buttons from the Custom Functions display.)

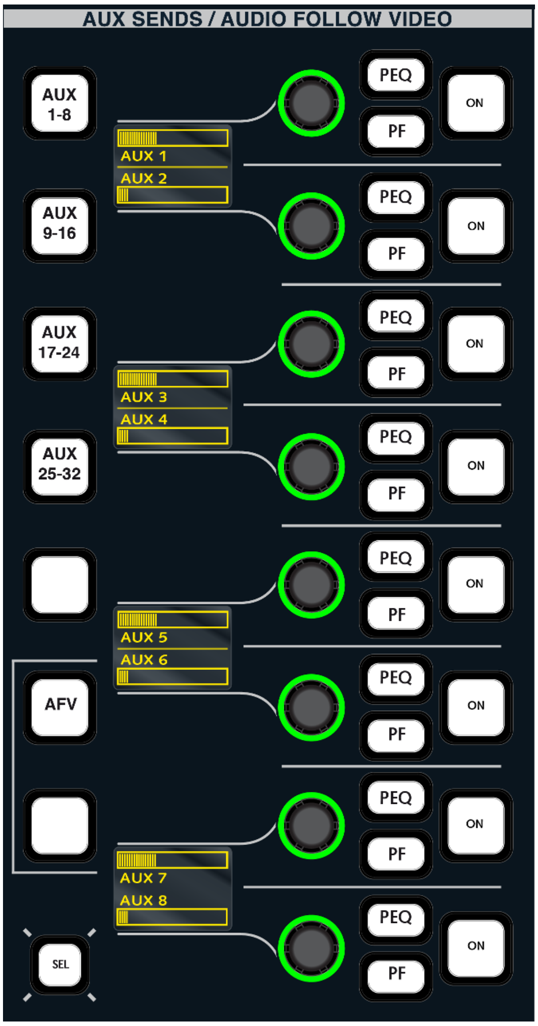
4. To return to individual monitor channel switching, deselect **ALL** on the **Extra Buttons** display.

Note that if you re-select the **ALL** button, the same cluster of channels as defined in step 2 will be reinstated.

Monitor Channels and Auxiliary Sends

When auxiliary sends 17 to 32 are assigned from a monitor channel, their source can be switched between the monitor send and return from the AUX SENDS panel:

- Press the **PEQ** button to switch the monitor send signal to the auxiliary send.
- Press the **PF** button to switch the monitor return signal to the auxiliary send.



Note that this source selection occurs on auxiliary sends 17 to 32 from monitor channels *only*. It is designed for cue feeds when overdubbing a recording. All other aux sends behave in the usual manner.

To disable the automatic source selection for aux sends 17 to 32, use the Cue Aux Send/Return option in the **System Settings** display.

Non-Multitrack Applications

The only difference between input channels and monitor channels is that the monitor channel direct out is used to provide the track bus signal path. This means that you can also use monitor channels in non multitrack applications to handle any type of source with exactly the same processing facilities as an input channel, except there is no direct output.

When using monitor channels for non-multitrack applications, you may wish to adjust the Track Self Assign and Cue Aux Send/Return options in the **System Settings** display.

The track bus (from the monitor channel) can be used as a mix minus send.

Chapter 4: Channel Control

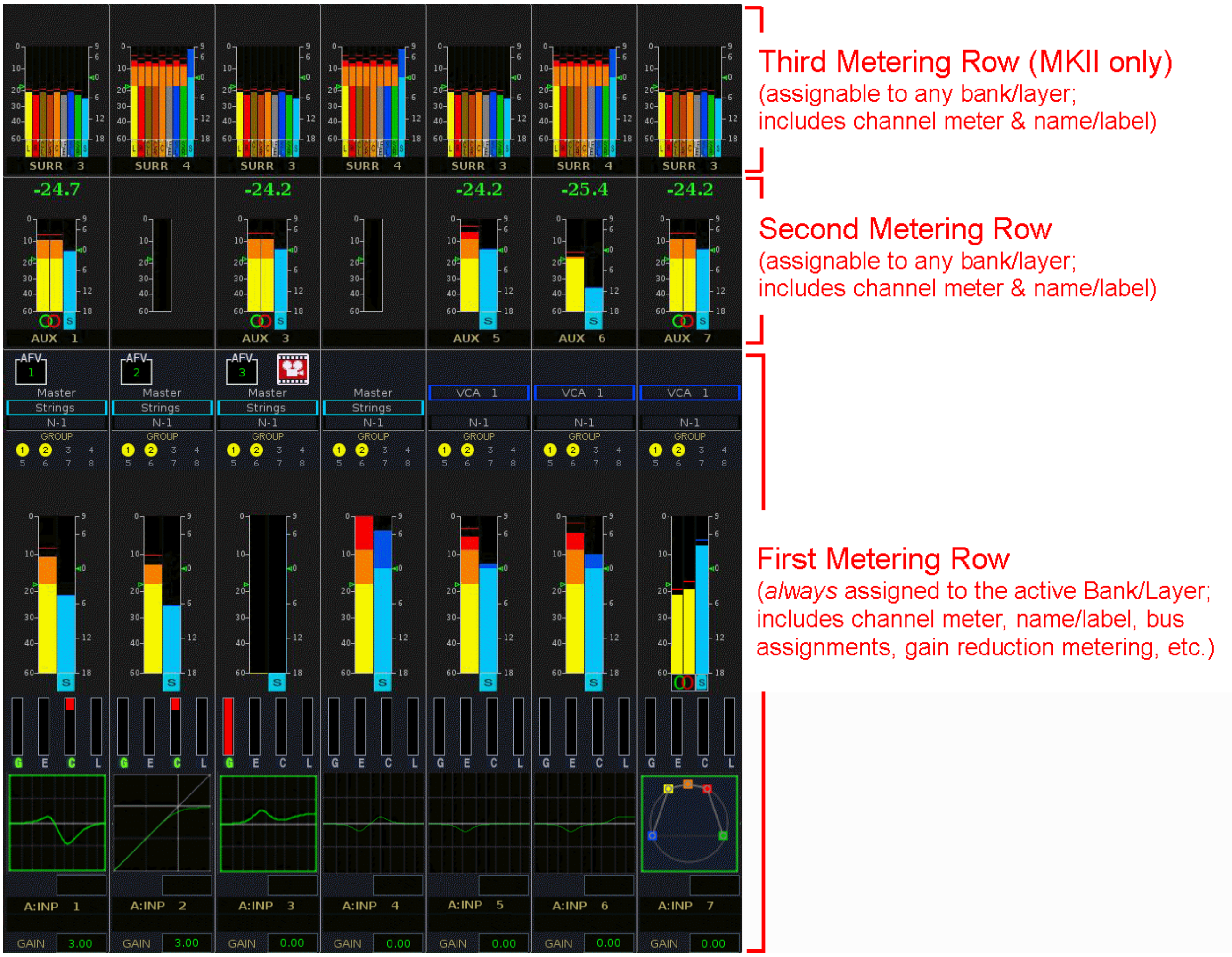
Introduction

In this chapter you will learn about the two main areas where you can adjust channel parameters - the console's channel fader strips and the Central Control Section. In addition, we will cover other channel-related topics such as bus assign and metering.

Topics covered in this chapter are:

The Channel Display

The **Channel** display provides metering, and other information, for up to three rows of channels:

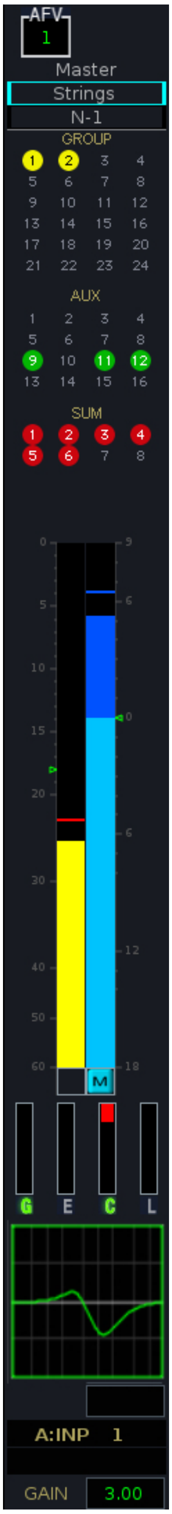


> The second and third metering rows are supported from Version 4.24 software onwards, and *only* by Router Module MKII systems.Due to the physical size of the displays, the classic mc²56 supports a maximum of two meter rows (as opposed to three on the MKII).

The first (lower) row *always* meters the active bank/layer, while the upper rows can be assigned to a "hidden" bank/layer of channels. Assignments can be made either "permanently" for the production or switched "on the fly" from user buttons. This allows you to enable and disable the upper metering rows, and/or meter different sets of channels. See Multi-row Metering Configuration for details.

For all on-screen metering; you may choose to display peak metering, loudness metering or both, see Bargraph Types.

> To help quickly identify the channel you are working on, you can enable the Show Fadersense mode from the **System Settings** display. When enabled, each time you touch a fader or free control, the corresponding channel within the **Channel** display is highlighted with a coloured outline matching the colour coding selected from the Channel Config display.



On the lower metering row (the active bank/layer), you will always see:

- **AFV** – the Audio Follow Video event number (if assigned) plus a camera icon (if the event is active).
- **Master** – the name and colour coding of VCA or Surround masters.
- **Link group Name** – the name and colour coding of link groups.
- **N-1** - the name and colour coding of the N-1 bus.
- **Bus assignments** onto the:
  - Group Busses
  - Track Busses (Recording channels only)
  - Aux Busses - with colour coding to indicate pre-fader, pre-EQ (Recording channels only) or post-fader assignments.
  - Sum Busses

The number of busses shown is set by the Bay Bus Count options in the **System Settings** display.

- **Channel Meter** – following the peak and loudness metering options set for the production, see Bargraph Types.
- **Dynamics gain reduction metering** for the:
  - **G** – Gate
  - **E** – Expander
  - **C** – Compressor
  - **L** – Limiter

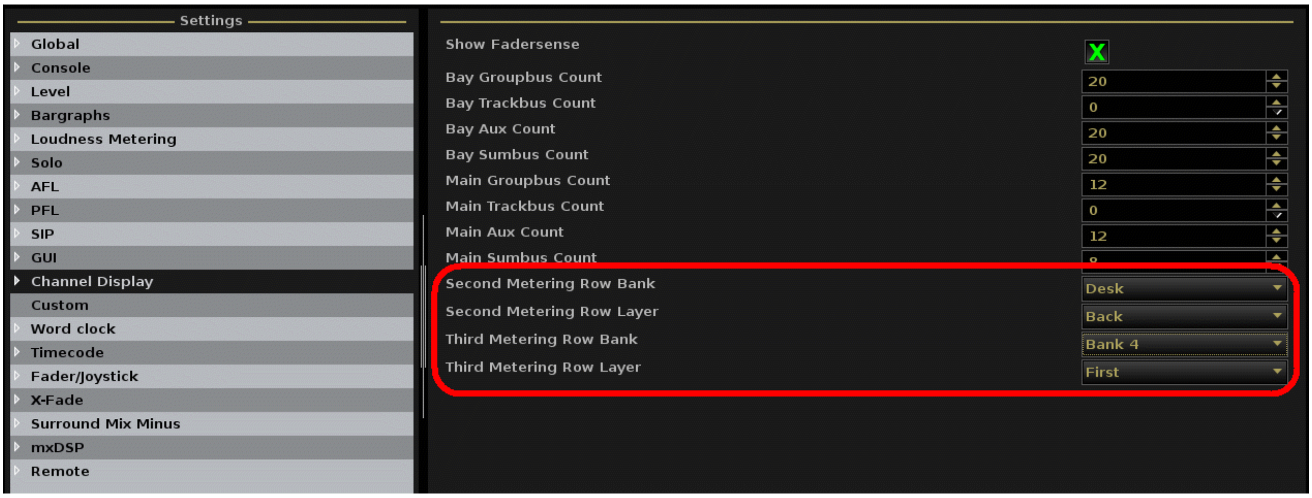
If the dynamics section is in circuit, then the **G**, **E**, **C** or **L** letters change from white to green.

- **Mini display** – graphical feedback on parameter values (e.g. EQ). If the processing module is turned on, then the outline of the mini display is green. If nothing is assigned to the fader strip, then the console logo appears.
- **Name or Label** - the two boxes display the name or label for the channels assigned to the 1st and 2nd layer fader strips. In our example, input channel 1 is assigned to layer 1 (INP 1) and there is nothing assigned to layer 2. You can choose to view the channel name, channel label or inherited source label from the centre section LABEL buttons.
- **Input GAIN** – this value displays the input gain for the channel. Note that this could be mic, line or digital gain depending on the source, see Input Control.

The second and third metering rows show the **Channel Meter** and **Label**.

Multi-row Metering Configuration

From V4.24 software onwards, a second and third metering row can be configured from the **System Settings** display:



Each row can be assigned to a specific bank and layer (e.g. Bank 4, Layer 1), or be set to follow the "desk" switching (for example, to meter the alternate Layer of the active Bank).

You can *only* assign a complete Bank and Layer to a metering row. Therefore, the channels you wish to meter *must* be assigned to either the first or second Layer of a fader strip Bank (1-6). The order of meters within the row follows the selected Bank/Layer fader strip assignments.

The settings affect all **Channel** displays across the console, and are saved and loaded by the production.

You can program user buttons, from the Custom Functions display, to switch the **Metering Row** options "on the fly". This allows you to enable and disable an upper metering row, and/or cycle the row through different banks/layers of channels.

➢ To Configure Multi-row Metering

1. From the System Settings display, select the **Channel Display** topic, and the **Second Metering Row Bank** option. Choose from:
   - **Bank 1** to **Bank 6** - assigns a specific bank.
   - **Desk** - the metering row follows the desk's Bank switching.
   - **None** - the metering row is disabled.
2. Select the **Second Metering Row Layer** option and choose from:
   - **First** or **Second** - assigns a specific layer.
   - **Front** - the metering row follows the desk's Layer switching, and displays the channels at the front (on the active layer).
   - **Back** - the metering row follows the desk's Layer switching, and display the channels at the back (on the inactive layer).

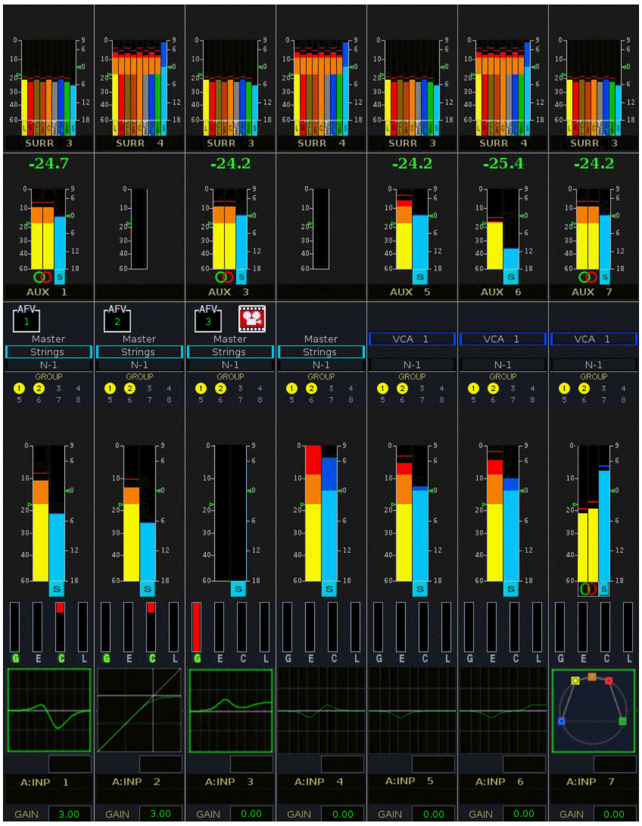
For example:

- to *always* meter channels assigned to the alternate Layer of the active Bank, you would select **Desk** + **Back**.
- to *always* meter channels assigned to the Bank 4, Layer 1, you would select **Bank 4** + **First**, and then assign the channels you wish to meter to Bank 4, Layer 1.
- to disable a metering row, select **None** from the **Metering Row Bank** option.

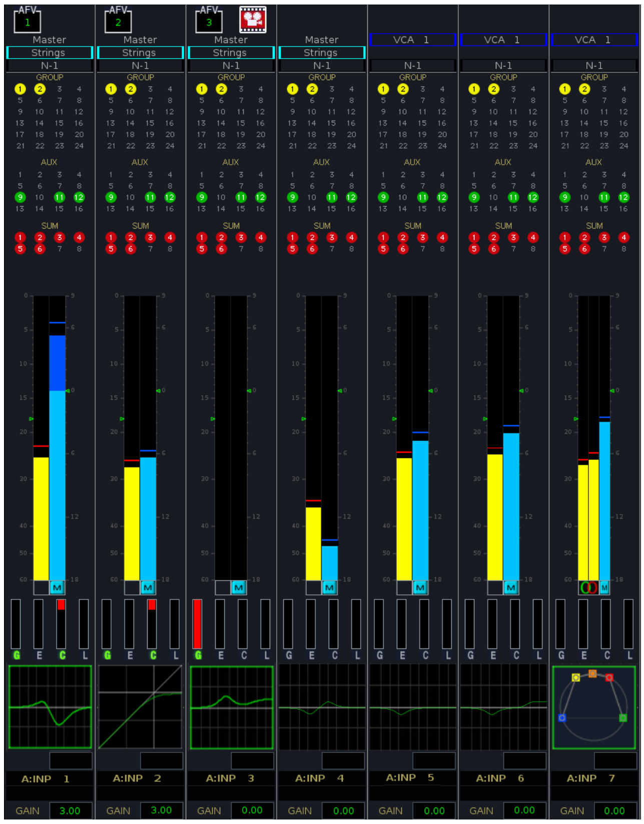
3. On the MKII mc²56, these steps can be repeated to configure a third metering row.

Once the **Metering Row Bank** option is set to anything other than **None**, the lower row resizes accordingly; the size of the channel meter and number of displayed bus assignments are the elements affected:

Multi-row Metering Enabled



Multi-row Metering Disabled

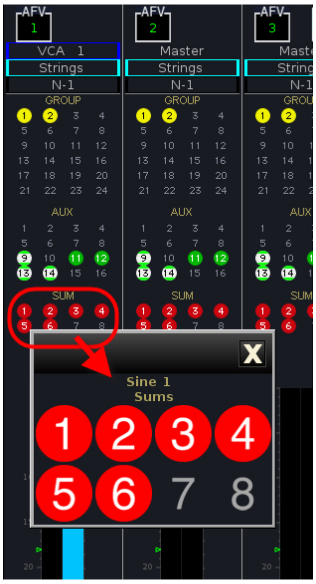


Use the Bay Bus Count options to adjust the number and type of displayed bus assignments. For example, to only display Group busses 1 to 8.

Touch-screen Functionality

You can touch the **Channel** display in order to adjust parameters:

> This provides a quick way to edit bus assignments, choose an N-1 bus or VCA master, or change the meter mode or pickup.
>
> To adjust a range of channels, couple them first and then edit any channel within the couple group.



For example, to edit the **SUM** bus assignments:

1. Touch the screen anywhere within the **SUM** bus area.

An expanded pop-up window appears on the display.

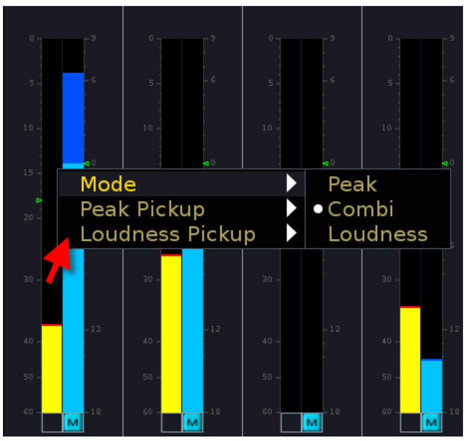
2. Now touch the number buttons to edit the assignments.

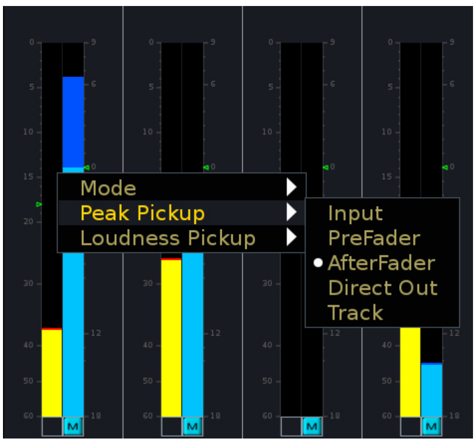
The pop-up window automatically closes after 3 seconds.

3. To manually close the pop-up, either touch the **X** in the top right corner, or touch twice in quick succession anywhere else on the display.

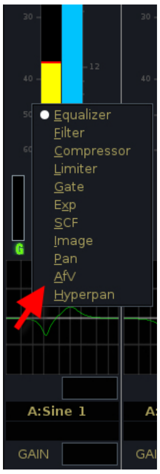
The following parameters may be edited in this way:

- **Bus assignments** - touch anywhere within the **GROUP**, **TRACK**, **AUX** or **SUM** areas to edit the bus assignments. The number of busses shown is set by the Bay Bus Count options in the **System Settings** display.
- **VCA assignments** - touch **Master** to select a VCA master (the first 32 VCA masters are displayed).
- **N-1 bus** - touch **N-1** to select the AUX to be used for the channel's mix minus bus (the first 16 auxes are displayed).
- **Meter Mode and Pickup** - touch the channel meter to select a different **Mode** or **Pickup** point for the peak or loudness meter, see Bargraph Types and Meter Pickup Points:





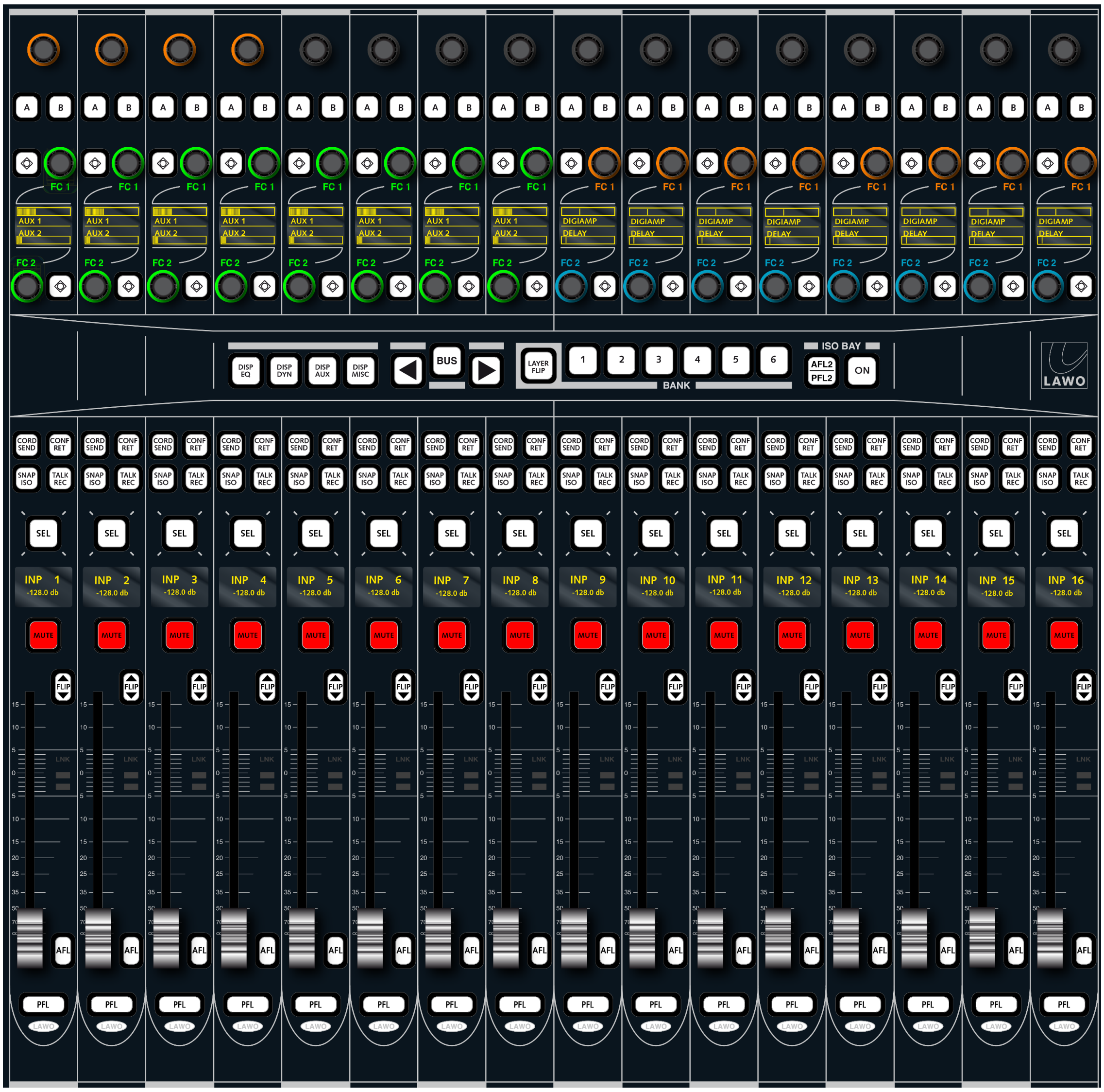
- **Mini display** - touch anywhere inside this area to select a different DSP module for the Mini display:



The Channel Fader Strip

Each of the **mc²56** channel bays are presented as a series of physical fader strips, each providing:

- Input Control
- Free Controls
- ISO Bay Panel
- User Buttons
- Fader SEL, Label, Mute, Flip, Level, AFL & PFL
- Status LEDs
- Colour Coding (LAWO Backlight & Button-Glow)



Input Control

Input Gain

The upper rotary control on the channel fader strip *always* adjusts input gain (source gain). The amount of **GAIN** is shown on the Channel display.

The gain will be either analogue or digital, depending on the type of source routed to the channel; the control is colour-coded to help identify this:

- **Mic/Line Analogue Source** (orange) - the control remotely adjusts the analogue mic preamp gain (before A-D conversion).
- **Digital Source** (unlit) - the control adjusts the digital I/O DSP gain (within the routing matrix).

The same parameters can be adjusted from the Central Control Section, see INPUT Control.

Note that two other gain elements are available within the channel's processing path (post source gain). These are the INMIX gain and DIGAMP. See INPUT Control and DIGAMP for details.

A/B Input Switching

For any input channel, you may assign two sources (A and B) to provide a main and backup source for the channel.

The sources are assigned from the **Signal List** display, see A/B Input Sources.

1. Use the **A** and **B** buttons to switch the input.

If there is no source assigned to the B input, then the **B** button cannot be selected.

2. Use the Input Gain control to set the source gain for the selected input.

Free Controls

The two Free Controls provide local fader strip control of any DSP parameter.

Free Controls are touch sensitive and colour-coded (MKII mc²56 only), making it easy to distinguish between Aux parameters (green), EQ (blue), Dynamics (purple), etc. Turn a control for fine adjustment; push down and turn for coarse adjustment.

Each control has its own dedicated display and push button (MKII mc²56 only). The display may be set to one of two modes from the Extra Buttons display:

- **USE SNS** (use touch sense) – is the default mode of working; the displays show the parameter function (e.g. AUX 1) and then the value (e.g. -4 dB) when touched.
- **FC VALUE** (show Free Control Value) – the displays show free control values (e.g. -4 dB) across the console. This button is a great way to see all the values for a parameter across the console.

The button function depends on the Free Control assignment. For example, on Aux sends, the button switches the send on/off.

Note that the controls are black (unlit) if a DSP module is not supported. This could be for a variety of reasons: for example, IMAGE is not available for mono channels; DSP modules are suspended if AMBIT or Loudness metering are active; not all DSP modules are supported on Broadcast channels.

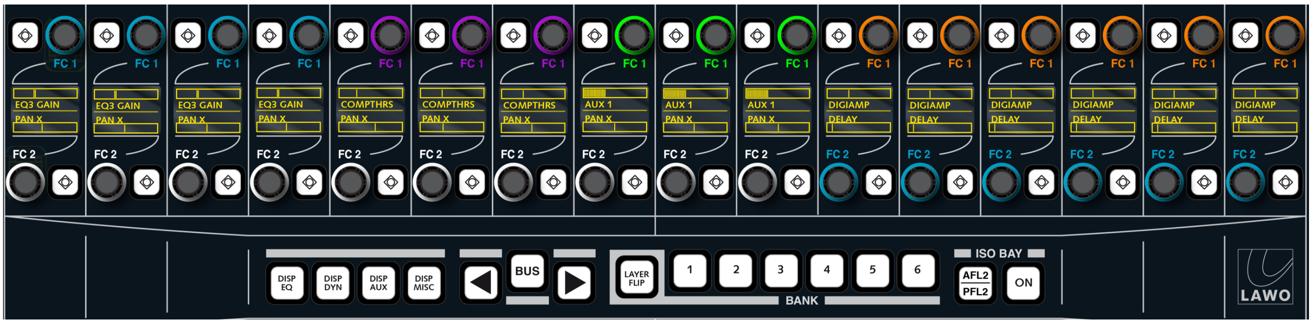
Assigning Parameters to Free Controls

There are three possible modes of operation for the Free Controls:

- Default Parameter Assignment - parameters are freely assigned, on an individual basis.
- FC PRESETS - these centre section buttons override the default assignments globally across the console.
- ISO BAY Local Parameter Control (MKII mc²56 only) - these buttons override the default assignments locally within the 16-fader bay. This mode provides expanded parameter control for the selected channel.

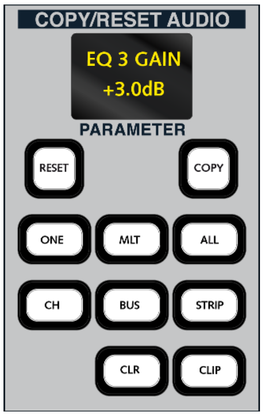
FC Default Parameter Assign

The two Free Controls on each channel fader strip may be assigned to key functions for the source. For example, on a presenter's channel you may want immediate access to the presenter's mix minus level and compressor threshold. Whereas, on a music replay channel, it is more important to access L/R Balance and Aux send level.



These default Free Control assignments are made from the COPY/ RESET AUDIO panel.

Controls may be assigned to any available channel parameter on an individual basis. These default assignments are then stored in snapshots and productions. Note that the Free Control assignments relate to the DSP channel (i.e. they move with the channel, if the channel is assigned to a different fader strip).



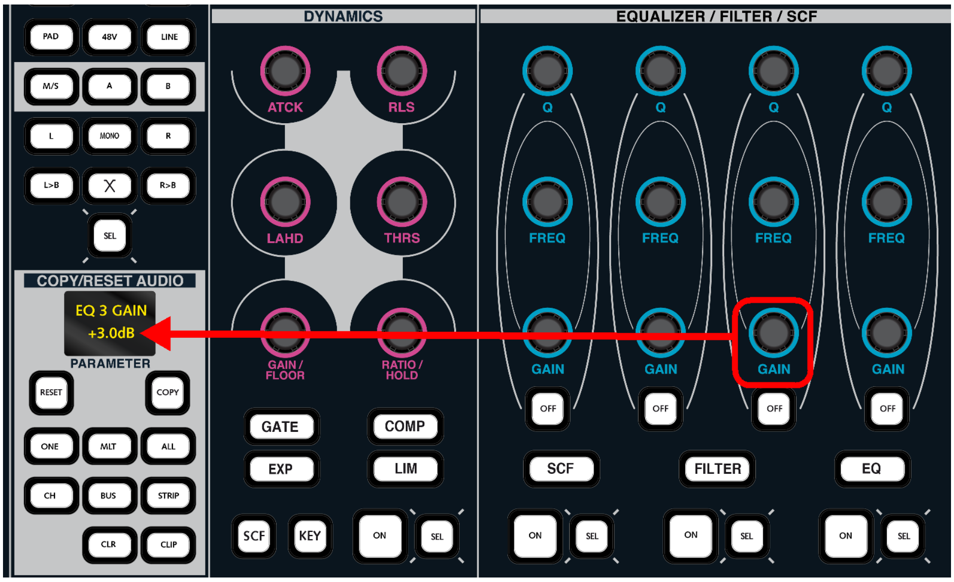
Assigning a Single Free Control

1. Press the **ONE** button, located on the COPY/RESET AUDIO panel, to activate a one-shot assignment.

*The **ONE** button flashes to indicate that parameter assign is now active.*

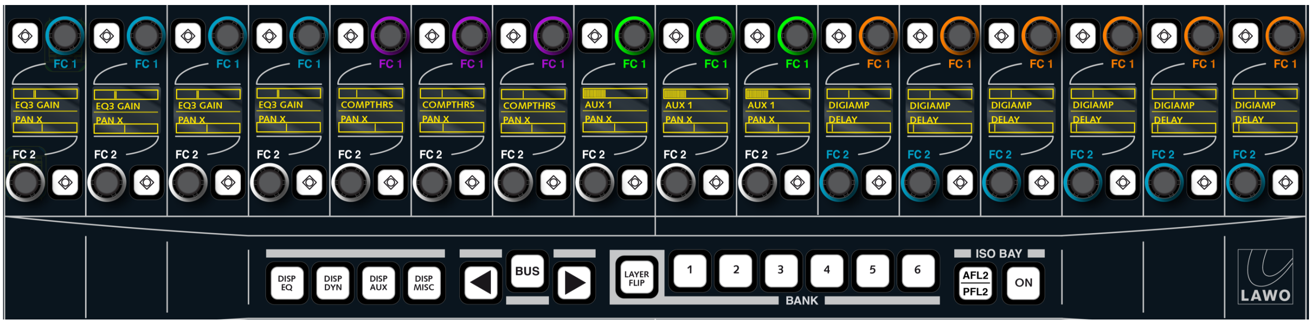
2. Select the parameter you wish to assign, by touching a rotary encoder on the Central Control Section - for example, touch the EQ Band 3 **GAIN** control.

The parameter is placed into the PARAMETER clipboard:



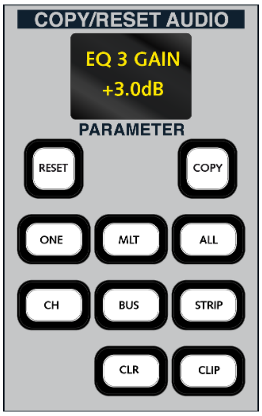
3. Now touch the Free Control on the destination channel strip.

*The assignment is made; the alphanumeric display below the FC updates; and the **ONE** button automatically cancels.*



Assigning Multiple Free Controls

To assign more than one Free Control at a time, you can use the **MLT** button to latch on the parameter assign mode. This saves you having to reselect the **ONE** button before each assignment.



1. Press the **MLT** button, located on the COPY/RESET AUDIO panel, to activate multi-assign mode.

   *The **MLT** button flashes to indicate that parameter assign is now active.*

2. Select the parameter you wish to assign, by touching a rotary encoder on the Central Control Section.

   The parameter is placed into the PARAMETER clipboard.

3. Now touch as many Free Controls across the surface as you wish.

   The clipboard parameter is assigned to each Free Control you touch.

4. Remember to deselect **MLT**, or press **ESC** on the SCREEN CONTROL panel, to exit the parameter assign mode. If you don't, then the next time you touch a channel free control, you will re-assign it!

Alternatively, you may assign a parameter to the same Free Control position across *all* fader strips for a certain channel type. For example:

1. Press the **ALL** button, located on the COPY/RESET AUDIO panel.

   *This automatically selects the **ONE** button for a one-shot assignment. (If you wish to make multiple **ALL** assignments press **MLT** instead of **ONE**.)*

2. Select the parameter you wish to assign, by touching a rotary encoder on the Central Control Section - for example, touch the L/R Pan control.

   PANX (Pan L/R) is placed into the PARAMETER clipboard.

3. Now choose the channel type you wish to assign to, from the ACCESS CHANNEL/ASSIGN panel – for example, select **INP**.

   You can select multiple channel types if you wish – e.g. select **INP**, **GRP** and **SUM** to make a Free Control assignment across all input, group and sum channels.

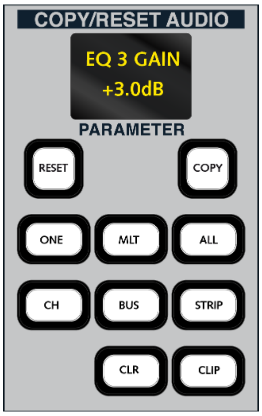
4. Now touch the Free Control destination on *any* input channel fader strip (INP).

   PANX is assigned to FC 1 across all input channels.

   Note that the assignment is made to *all* input channels, even those not assigned to a fader strip.

5. When working in **ONE** shot mode, parameter assign automatically cancels. (If you are working in multi-assign, remember to deselect **MLT** or press **ESC**, to exit the parameter assign mode.)

Clearing a Free Control Assignment



To clear a Free Control assignment so that it becomes inactive:

1. Press the **CLR** button located on the COPY/RESET AUDIO panel.

   *This automatically selects the **ONE** button for a one-shot operation. (If you wish to clear multiple assignments select **MLT** instead of **ONE**.)*

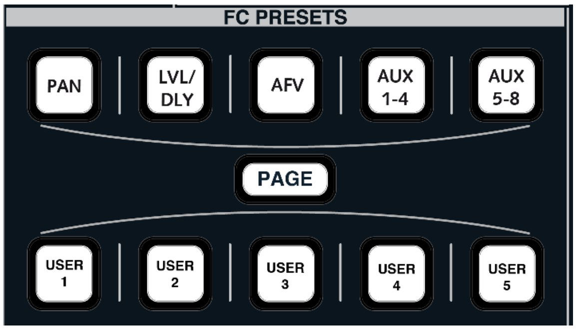
2. Select the Free Control you wish to clear by touching it on the fader strip.

   The assignment is cleared and Free Control display becomes blank.

3. If you have selected **MLT**, remember to deselect **MLT**, or press **ESC** on the SCREEN CONTROL panel, to exit the parameter assign mode.

FC Presets

You can temporarily override the default Free Control assignments, by recalling a preset from the centre section:



Each preset changes Free Controls globally across the console, so this is a great way to access say Aux Sends 1 and 2 across the console with a single button press. When you deselect the preset, the controls return to their default assignments.

On the MKII mc²56:

The upper row of FC PRESETS offer two pages of pre-defined functions - for example:

1. Press **PAN** (with **PAGE** off) to assign Pan L/R to FC 2 and Pan F/B to FC1.
2. Press **PAN** + **PAGE** to assign Pan Slope to FC 2 and LFE level to FC1.

The lower row of FC PRESETS offer two pages of user defined functions. These allow you to store and recall different parameter combinations as follows:

1. Assign the parameters you wish to store onto the Free Controls of any fader strip - for example, Pan L/R to FC 2 and Aux 1 Gain to FC 2.
2. Make sure that the fader strip is in access by pressing its fader **SEL** button.
3. Then press and hold one of the FC PRESET **USER** buttons until it flashes (for more than 3 seconds).

The Free Control assignments from the selected fader strip are stored.

4. Now press the FC PRESET **USER** button quickly, to recall its assignments globally across the console.

The Free Controls across all fader strips update to Pan L/R and Aux 1 Gain.

> On the classic mc²56, you will find two pages of user defined FREE CTRL PRESETS (8 x 2 = 16 FC PRESETS). These work in an identical manner to the **USER** presets on the MKII console. Note that there are no pre-defined presets on the classic mc²56.

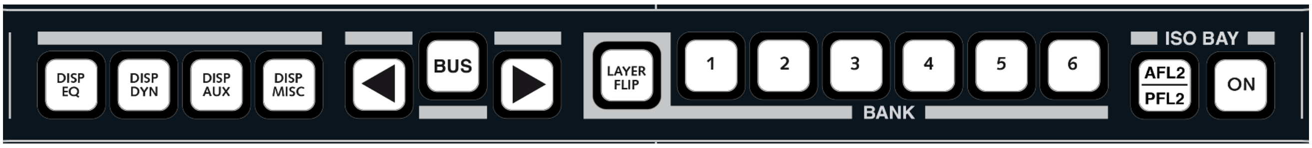
The behaviour of all FC PRESETS is summarised below. Note that following a cold start, the **USER** defined presets are pre-assigned to Aux sends 9 to 28.

MKII mc²56 FC PRESETS:

	PAN	LVL/DLY	AFV	AUX 1-4	AUX 5-8	USER 1	USER 2	USER 3	USER 4	USER 5
FC 1	Y (F/B Pan)	DELAY	OFF Level	AUX 2	AUX 6	AUX 10	AUX 14	AUX 18	AUX 22	AUX 26
FC 2	X (L/R Pan)	DIG AMP	ON Level	AUX 1	AUX 5	AUX 9	AUX 13	AUX 17	AUX 21	AUX 25

	+ PAGE	+ PAGE	+ PAGE	+ PAGE	+ PAGE	+ PAGE	+ PAGE	+ PAGE	+ PAGE	+ PAGE
FC 1	LFE Level	DIR OUT	FALL Time	AUX 4	AUX 8	AUX 12	AUX 16	AUX 20	AUX 24	AUX 28
FC 2	Pan SLOPE	INS SEND	RISE Time	AUX 3	AUX 7	AUX 11	AUX 15	AUX 19	AUX 23	AUX 27

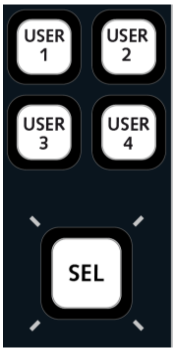
ISO Bay Panel

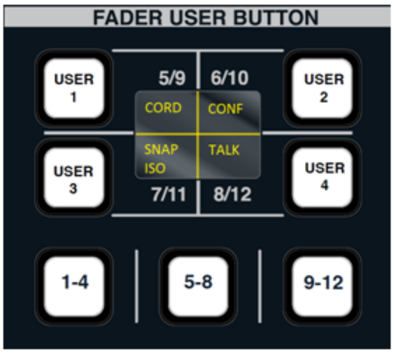


Please see Isolating Fader Bays.

User Buttons

The 12 fader strip USER buttons are programmed from the Custom Functions display. And, are switched through three pages of functions from the centre section FADER USER BUTTON panel:





This provides access to 12 user button functions per fader strip:

1. Press **1-4**, **5-8** or **9-12**, on the central FADER USER BUTTON panel, to switch between User 1-4, User 5-8 and User 9-12.

The fader strip user buttons are switched globally across the console.

The classic mc²56 does not include the central FADER USER BUTTON panel, and therefore supports a single page of functions, User 1-4.

On the fader strip, the key caps are engraved to reflect their function (see the examples below).

In the centre section, the FADER USER BUTTON panel includes an 8-character display which labels each button's function. This is especially useful if you intend to reprogramme user buttons, or use all three pages. The labels are programmed along with the button functions from the Custom Functions display.

User Button Functions

Functions are assigned to user buttons from the Custom Functions display. Note that these assignments are stored as part of the system configuration (and not in productions). This means that any changes will affect all users.

The following default functions are pre-configured for user buttons 1 to 4:

➢ **Mix Minus Control: CORD, CONF & TALK**

1. Press **CONF** to activate the mix minus (N-1) output for that channel.
2. Press **CORD** to activate a conference-style auxiliary send.
3. Press **TALK** to talk to the channel's N-1 bus. The talkback source is pre-configured to be the talkback mic input. However, this can be edited from the Custom Functions display.

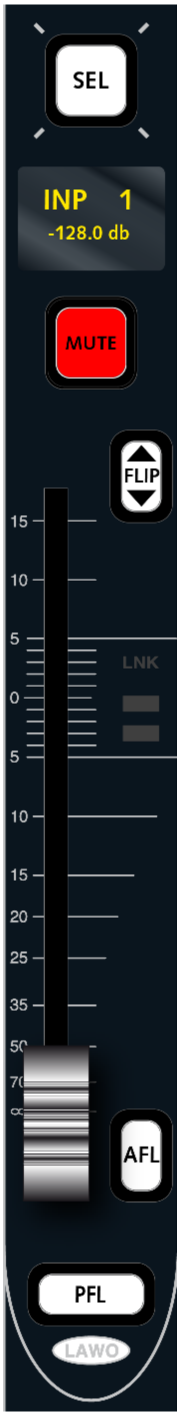
If a monitor channel is assigned to the fader strip, then the buttons control input switching and record arming (**SEND**, **RET** and **REC**).

➢ **Snapshot Isolate: SNAP ISO**

1. Enable **SNAP ISO** to isolate the channel strip from a snapshot recall.

Use **SNAP ISO** to isolate your main presenter channels and other key feeds when using snapshots to recall different mixes during a live production. See Snapshot Isolate for more details.

Fader SEL, Label, Mute, Flip, Level, AFL & PFL



SEL

This button selects the channel - for example, to assign it to the Central Control Section, or to select the channel during operations such as bus assign, etc.

Label

This display shows the name or label of the channel assigned to the fader strip. You can choose one of three options from the centre section LABEL buttons:

- **CHANNEL NAME** = the system name of the channel (e.g. INP 1).
- **USER LABEL** = the user label given to the channel (e.g. GUEST).
- **INHERIT SOURCE** = the user label given to the source which is routed to the channel (e.g. MIC 1).

Up to 8 characters may be displayed. Signal labels are edited from the Signal List display. Labels for control channels, such as VCA masters, may be edited from the Title Bar.

The display also shows the main channel level (in dB) when a fader is touched (if the Fader Display option in the **System Settings** display is enabled).

MUTE

Press the **MUTE** button to mute (cut) the channel.

The fader strip **MUTE** buttons may be set to mute after the input mixer (pre-fader/ pre-processing) or after the fader from the **System Settings** display, see Channel Mute.

MUTE buttons may be disabled, to prevent accidental selection, using the Mute option in the **System Settings** display.

Layer FLIP

Press **FLIP** to switch the fader strip from Layer 1 to 2, or vice versa. See Layer Switching for more details.

Level

The fader is touch sensitive, providing gain control from –128dB to +15dB. As you adjust the fader, you will see the level in dB on the Label display (providing the Fader Display option is enabled).

You may customise the feel of the faders, add a notch at a particular level (e.g. 0dB) or activate the fader backstop using the Fader/Joystick options in the **System Settings** display.

Note that if the Label display is flashing, then the fader is controlling a different channel parameter – for example, you may assign your aux send levels onto the faders. See Fader Control of Levels for details.

AFL & PFL

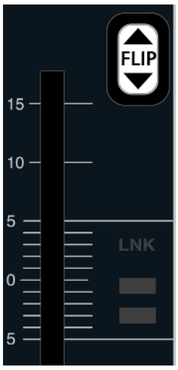
Press **AFL** to listen to the post-fade channel signal, or **PFL** to listen to the pre-fade channel signal. The AFL and PFL busses may be switched to different outputs from the Monitoring Section. A variety of AFL, PFL and Solo button options are available from the **System Settings** display. Or, AFL may be switched to operate as Solo-in-place.

Status LEDs

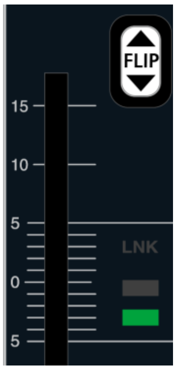
Beside the fader you will find a number of status LEDs:

- **LNK** - lights if any processing modules within the channel are linked, see Link Groups.
- **Signal Present** - these two LEDs light in different colours to show that signal is present; there are five possible states:

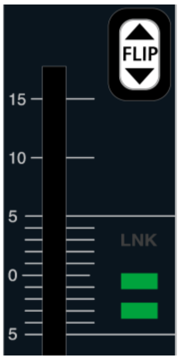
Signal Level < -60dB



Signal Level > -60dB

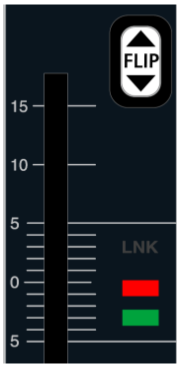


Signal Level > -30dB



Signal Level > -15dB

Signal Level > OVR



The signal present LEDs *always* monitor the channel input level (the output of the INMIX audio module), regardless of the peak meter pickup point.

Note that the OVR state may be adjusted using the Overload Threshold option in the **System Settings** display.

> The classic mc²56 provides a confidence level meter beside the fader, rather than Signal Present LEDs. It also includes a LINK and OVR LED which work as described above.

Colour Coding (LAWO Backlight)

LAWO Backlight

At the bottom of each fader strip, the **LAWO** backlight is colour-coded to indicate the channel type. This enables you to easily distinguish input channels (white) from groups (yellow), aux masters (green), VCAs (blue) and sums (red). Or, you may customise the channel colour coding - for example, music channels to be white, VTRs to be blue, presenter mics to be red and so on.

Button-Glow (MKII mc²56 only)

If you enable the **System Settings** Button-Glow option, then fader strip buttons in their off state are dimly lit according to the channel colour. This makes channel identification even easier, especially in low-light conditions. The fader strip buttons affected are A/B input switching, Free Control on/off buttons, the four channel user buttons, AFL and PFL.

Note that this function is not supported by the classic mc²56.

Source Routing (Input Patching)

Any source connected to the routing matrix may be routed to any input or monitor channel. In addition, you may assign a backup source so that the channel can be switched quickly from A to B.

Source routing can be controlled either from the **Signal List** display, or **mx Routing** display (as a crosspoint matrix). Please see Signal Routing/Settings for details.

Bus Assign (Routing)

The mc²56 offers several methods for assigning channels onto mix busses:

- Forward or Reverse assign (from the BUS ASSIGNMENT panel).
- The ISO BAY panel (within each Channel bay).
- The touch-screen **Channel** display (within each Channel bay).
- The Bus Assign or Busses Reverse displays on the Central GUI.

Note that:

- Input channels (INP) can assign to any bus - Track bus, Aux, Group or Sum.
- Monitor channels (MON) can assign to any Aux, Group or Sum.
- Group channels (GRP) can assign to any Aux, another Group or any Sum.
- Sum and Aux channels (SUM and AUX) are designed to be the final point in the signal chain, and cannot be reassigned to another bus. (To do this, you would need to route the Sum or Aux output back into an input channel).

Any bus may be configured as mono, stereo or surround, such that the corresponding pan law is applied. See Panning.

You can use any of the bus assignment methods to adjust Aux on/off, as an alternative to the Auxiliary Send panel.

Or to assign channels to a VCA master, see VCA Grouping.

The Forward and Reverse BUS ASSIGNMENT buttons may be locked, to protect existing bus assignments, using the **Lock ASN** button located on the Extra Buttons display.

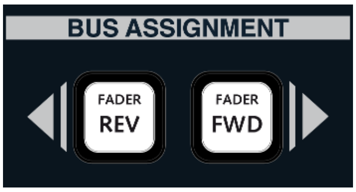
Forward Assign

This method is ideal for assigning a single channel *to* multiple bus outputs (if the output channels are assigned to fader strips).

For example, to assign an input channel (INP 1) to some groups and sums:

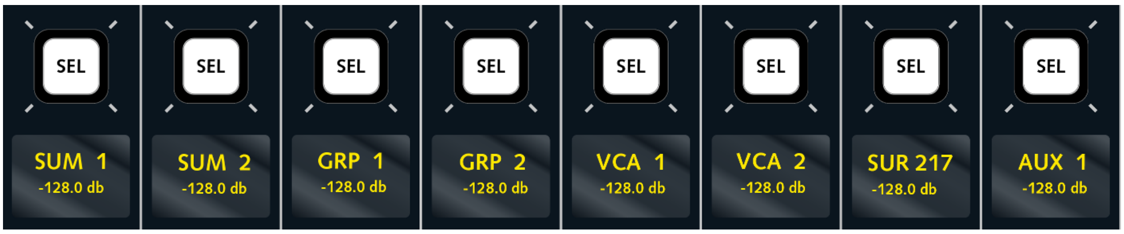
1. Select the input channel - either by pressing its fader **SEL** button or entering **INP**, the number **1** and **Enter** from the ACCESS CHANNEL/ASSIGN panel.

2. Press **FADER FWD**, located on the BUS ASSIGNMENT panel:



*The fader **SEL** buttons, across the console, now indicate the status of bus assignments **from** the channel in access (INP 1):*

- Steady state red = channel assigned to destination.
- Flashing green = channel not assigned to destination.
- **SEL** not lit = invalid destination (for example, you cannot assign INP 1 onto another input channel!)



3. Press the fader **SEL** buttons to modify the assignments.

For example, press the green fader **SEL** buttons on strips controlling SUM 1, SUM 2, GRP 2, etc. to assign INP 1 onto these busses. Or, press red fader **SEL** buttons to remove existing assignments. (To assign an input channel to a Track bus, press the fader **SEL** buttons on the corresponding Monitor channels).

*The fader **SEL** buttons change state, and the Channel display updates.*

If the bus is stereo or surround, then assignments onto the LR or surround channels are made in one operation, see Bus Assignments to a Surround Output.

If the bus you wish to assign to is not available on a fader strp, then either switch banks or layers, or use the ACCESS CHANNEL/ASSIGN panel.

4. Deselect the **FADER FWD** button, or press **ESC** on the SCREEN CONTROL panel, to exit the bus assign mode.

Reverse Assign

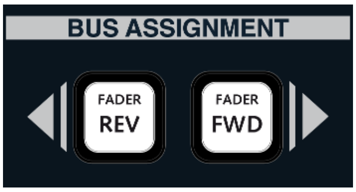
This method selects the bus first, and then the source channels. It is ideal for assigning a single bus *from* multiple channels (if the source channels are assigned to fader strips).

For example, to assign some input channels onto SUM 1:

1. Select the SUM 1 channel - either by pressing its fader **SEL** button or entering **SUM**, the number **1** and **Enter** from the ACCESS CHANNEL/ASSIGN panel.

> To select a Track bus as the destination, press the fader **SEL** button on the corresponding Monitor channel or enter MON x from the ACCESS CHANNEL/ASSIGN panel.

2. Press **FADER REV**, located on the BUS ASSIGNMENT panel:



*The fader **SEL** buttons, across the console, now indicate the status of bus assignments **to** the channel in access (SUM 1):*

- Steady state red = channel assigned to destination.
- Flashing green = channel not assigned to destination.
- **SEL** not lit = invalid destination (for example, you cannot assign another Sum channel onto SUM 1!)

3. Press the fader **SEL** buttons to modify the assignments.

For example, press the green fader **SEL** buttons on strips controlling INP 1, INP 2, etc. to assign these channels onto SUM 1. Or, press the red **SEL** buttons on INP 5, INP 6 and INP 7 to remove the existing assignments.

*The fader **SEL** buttons change state, and the Channel display updates.*

> If the bus is stereo or surround, then assignments onto the LR or surround channels are made in one operation, see Bus Assignments to a Surround Output.

4. Deselect the **FADER REV** button, or press **ESC** on the SCREEN CONTROL panel, to exit the bus assign mode.

Bus Assign from the ACCESS CHANNEL/ASSIGN panel

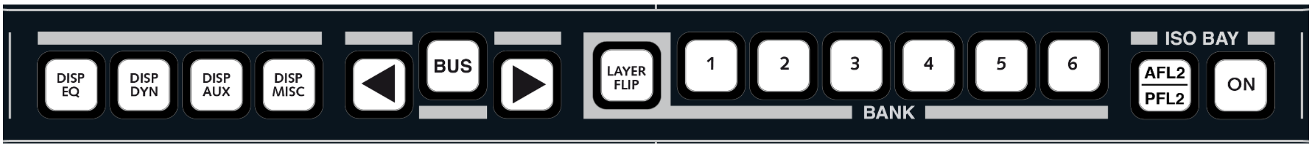
If the bus you wish to access is not assigned to the control surface, then you can use the **BUS ASSIGN** button to change the operation of the ACCESS CHANNEL/ASSIGN panel. This method routes onto Track, Group, Sum or Aux busses (up to 30):

1. First select the channel you wish to assign (e.g. INP 1) - either by pressing its fader **SEL** button or entering the channel type and number from the ACCESS CHANNEL/ASSIGN panel.
2. Then press **BUS ASSIGN** on the ACCESS CHANNEL/ASSIGN panel.
3. Select the bus you want to assign to by choosing a channel type:
   - **MON** – Track busses.
   - **GRP** – Groups.
   - **SUM** – Sum.
   - **AUX** – Auxiliary sends.
4. And then a number:
   - For busses 1 to 9, press **1** to **9**.
   - For bus 10, press **0**.
   - To access busses 11 to 20, press the left arrow (+10) button.
   - To access busses 21 to 30, press the right arrow (+20) button.

The channel in access (INP 1) is assigned onto the selected bus; the numeric keypad illuminates to show the assignment.

5. Exit bus assign by deselecting the **BUS ASSIGN** button.

Local Bus Routing (ISO BAY)



Please see Local Bus Routing.

Bus Assign from the Channel Display

Please see Touch-screen functionality.

The Bus Assign display

The **Bus Assign** display works in a similar manner to Forward assign, and provides touch-screen control of all assignments from the channel in access.

It also provides a way to edit stereo or surround bus assignments. For example, if Sum 1 and Sum 2 are linked for stereo operation, then Forward and Reverse assign (from the front panel) automatically assigns onto both Left and Right sums in one operation. To assign a channel to the Right bus only (e.g. Sum 2), then you should use the **Bus Assign** display.

1. Press the **BUS** button, located on the SCREEN CONTROL panel, to view the **Bus Assign** display.

The display shows all bus assignments from the channel in access (e.g. from INP 1).

If the channel is assigned to a bus, then the buttons are colour-coded with groups shown in yellow; track sends in blue; auxes in green; and sums in red.

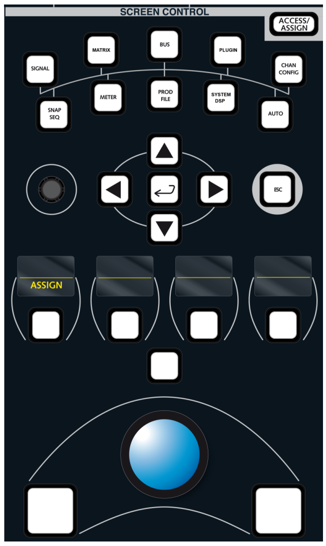
The number and type of busses available is defined by the DSP configuration.

For convenience, VCA and link group assignments are also shown at the bottom of the display (VCAs outlined in blue; link groups outlined in the link group colour). Note that you cannot change VCA or link grouping from this display.

To change the bus assignments from the channel in access, use the touch-screen buttons.

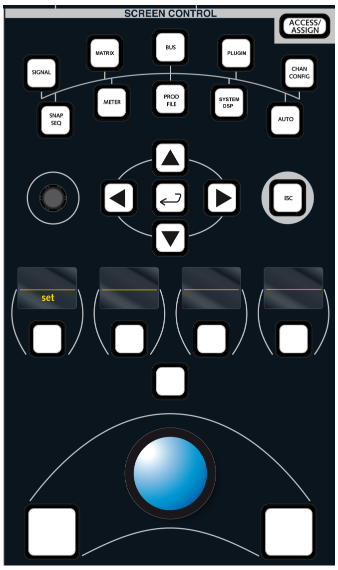
Or use the SCREEN CONTROL trackball/navigation controls as follows:

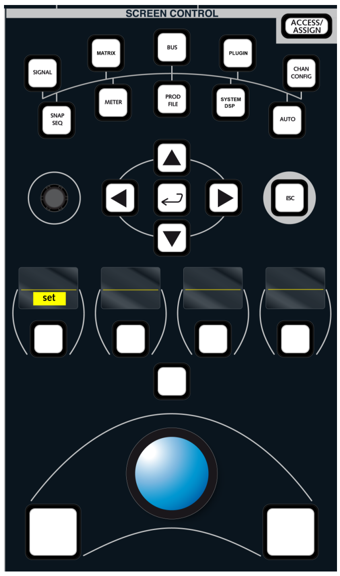
1. Select a bus.
2. Press the **ASSIGN** soft key to make, or unmake, the bus assignment:



If the bus is an aux, then you may also adjust the level of the send:

3. Press the **SET** soft key - it highlights:





4. Turn the rotary control to adjust the aux send level.
5. Or click on the up/down arrows beside the send level.
6. Or, type in an aux send value using the console keyboard.

For stereo aux sends, you may use the same method to adjust aux pan/balance.

The Busses Reverse display

The **Busses Reverse** display works in a similar manner to Reverse assign, and provides touch-screen control of assignments from the channel in access.

This is a great way to view all the channels assigned to a group, sum, aux or track bus.
It is also the *only* way to mute an individual channel feed to a bus.

1. Press the **BUS** button, located on the SCREEN CONTROL panel, to view the **Busses Reverse** display.

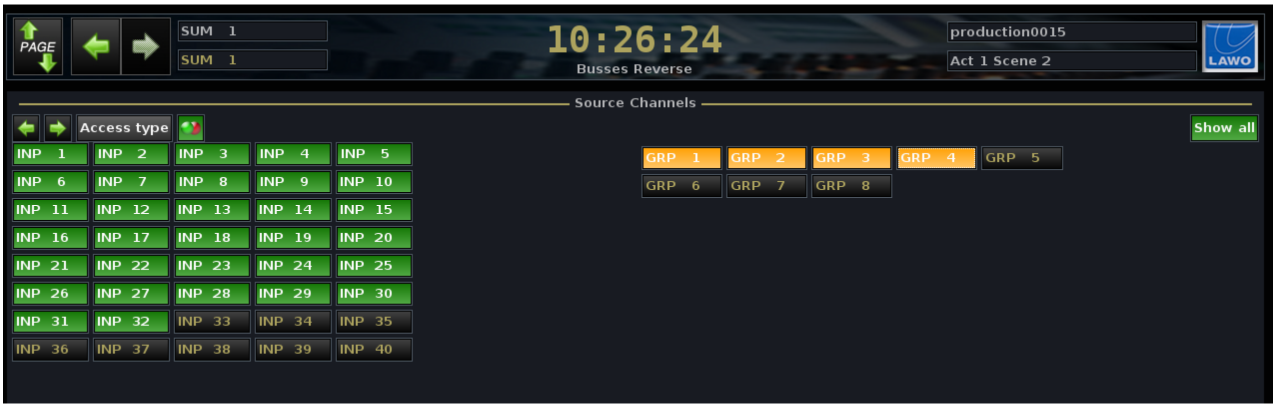
If something is assigned to the channel in access, then you will see buttons for the assigned channels on the display - colour-coded green for input channels; yellow for groups; etc.

In our example, we can see all the input channels currently assigned to SUM 1:

2. To view all available channels, select **Show all**:

The display updates to show all available source channels within your DSP configuration. Note that you will only see the channels which can be assigned to the channel in access – in our example, inputs and groups can be assigned to Sum1:

To change the bus assignments to the channel in access, use the touch-screen buttons, or left-click using the trackball and left-select key:



You can also change the channel in access from the **Busses Reverse** display as follows:

1. Click on the left/right arrows beside **Access Type** to increment or decrement the channel in access.
2. Click on **Access Type** to cycle through the different DSP channel types: inputs, monitors (track busses), groups, sums and auxes.
3. Right-click on one of the source channels (e.g INP 1) and select **Access** to change to Input 1:

You can also change the mono/stereo status of the channel in access, by selecting (or unselecting) the green/red circle icon - in our example, SUM 1 is set to stereo.

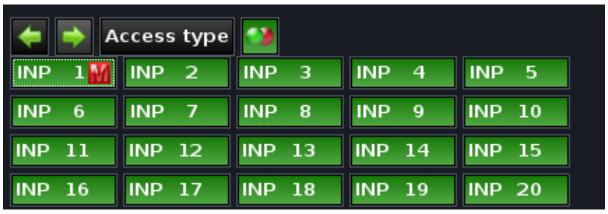
Bus Assign Mute

The **Busses Reverse** display also allows you to mute individual channel feeds to the bus in access. This provides an alternative to removing the bus assignment (useful for temporary overrides), or using the channel's MUTE (which will mute all bus assignments from the channel).

1. Use the right select button to right-click on the channel you wish to mute:

2. And select the **Mute** option.

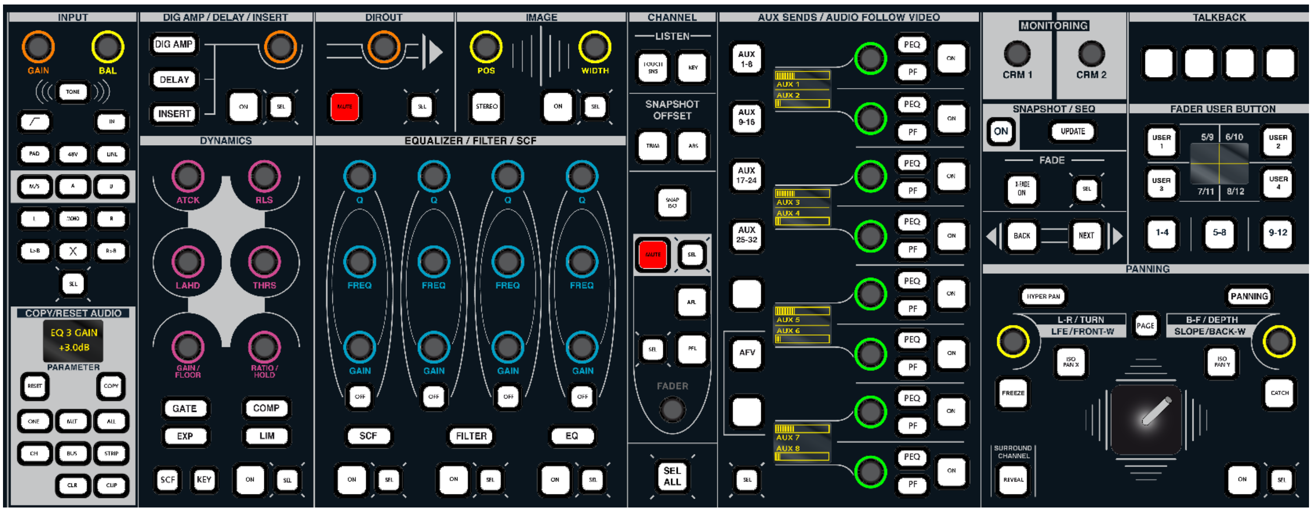
The display updates showing that the bus assignment from Input 1 is still made but is now in a muted state:



The Central Control Section

The Central Control Section provides master channel control for the channel in access - INPUT, DYNAMICS, EQUALIZER, etc.

Select a channel, by pressing its fader strip **SEL** button, and then reach out to control any parameter:



Assigning a Channel to the Central Control Section

The channel in access is *always* assigned to the Central Control Section. So, either press a fader strip **SEL** button, or use the ACCESS/CHANNEL ASSIGN panel, to change to a different channel.

You may use the ACCESS/CHANNEL ASSIGN panel to select a channel which is not assigned to a fader strip. For example, your Sum master or a communications input.

The channel in access can be locked, so that the Central Control Section is *always* assigned to a specific channel, using the **Lock ACC** option on the Extra Buttons display.

Central Control Section Modules

Controls are divided into clearly defined sections covered later in this chapter, see Channel Processing Modules.

Note that the controls are black (unlit) if a DSP module is not supported. This could be for a variety of reasons: for example, IMAGE is not available for mono channels; DSP modules are suspended if AMBIT or Loudness metering are active; not all DSP modules are supported on Broadcast channels.

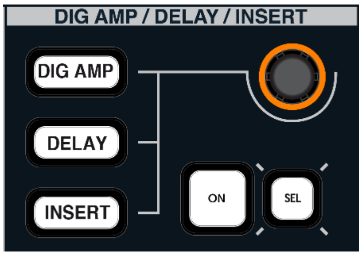
Rotary controls are colour coded, making it easy to distinguish EQ from DYNAMICs, from AUX sends, etc. (MKII mc²56 only).

All rotary controls are touch-sensitive; the controls default to provide fine parameter adjustment. For coarse adjustment (5 times faster), push down and turn.

Remember to turn **ON** the DSP module to hear your adjustments!

Select the Main Display for visual feedback on settings.

SEL Buttons



Every module includes a **SEL** (Selection) button. This is used to:

- COPY/RESET AUDIO – copy or reset channel parameters.
- LISTEN – AFL an individual processing section.
- LINK – link selected modules between channels.
- SNAPSHOT/SEQUENCE – select modules for snapshot cross fades when running a sequence.
- AUTOMATION – select modules for timecode automation.

SEL ALL, below the FADER level control, selects, or deselects, all modules.

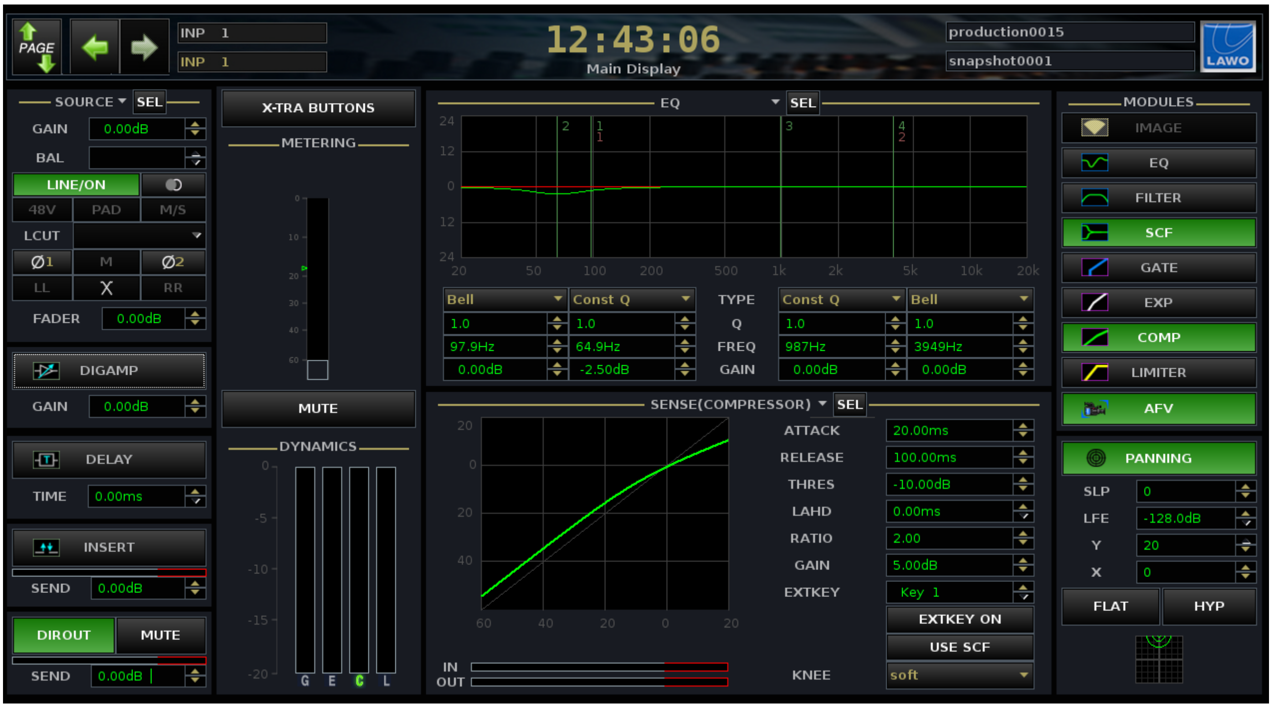
See Selecting Channel Parameters for more details.

The Main Display

The **Main Display** provides a useful visual accompaniment to the Central Control Section.

1. Press the **CHAN/CONFIG** button, located on the SCREEN CONTROL panel, to view this display.

You will see an overview of parameters for the channel in access – e.g. for INP 1:



The **Main Display** is divided into the same sub sections as the Central Control Section front panel. And, as you adjust controls, the display updates to reflect your settings. In addition, you can change parameters from any of the screen buttons.

Green buttons indicate that a section or option is active (ON); on the right of the display, you can see the on/off status for all MODULES on the channel.

Any modules not supported by the selected DSP channel are greyed out. This could be for a variety of reasons: for example, IMAGE is not available for mono channels; DSP modules are suspended if AMBIT or Loudness metering are active; not all DSP modules are supported on Broadcast channels.

The **Main Display** *always* shows the following sections:

- **SOURCE/INMIX** – source and inmix parameters.
- **FADER** – main channel level.
- **DIGAMP** – digital amplifier.
- **DELAY** – channel delay.
- **INSERT** – insert return switching and send level.
- **DIROUT** – direct output.
- **METERING** – channel meter.
- **MUTE** – channel mute.
- **DYNAMICS** – gain reduction metering for the Gate, Expander, Compressor and Limiter sections. Note that if the section is switched on, the **G**, **E**, **C** or **L** is green.
- **MODULES** – on/off status for all processing modules.
- **PANNING** – X/Y pan parameters.

In addition, the two central areas can be assigned to other processing sections. Click on the drop-down menu, at the top of each area, to select an audio module:

If you select **SENSE**, then the area automatically follows the last control touched, so that you will see the processing section that you are working on.

Note that some of the **Main Display** sections include an on-screen **SEL** button. This mimics the operation of the Central Control Section SEL button and can be used to select the module for copy, reset, and other operations.

Channel Processing Modules

The next series of sections cover the different modules available to a fully-featured DSP channel. We will concentrate on operation from the Central Control Section as this offers maximum control.

> The **mc²56** supports two channel types – Recording and Broadcast. Broadcast channels do not support all the DSP modules found within a Recording channel, and there are also some operational differences. This section deals with the operation of both channel types. For details on their signal flow, see DSP Channel Types.

Topics covered are:

- INPUT - input gain, microphone preamplifier settings, stereo input balance/controls and tone.
- DIG AMP - digital amplifier (gain).
- DELAY - channel delay.
- INSERT - insert switching.
- DIROUT – channel direct output.
- IMAGE - width and positioning for a stereo channel.
- DYNAMICS: GATE, EXPANDER, COMPRESSOR and LIMITER.
- EQUALIZER/FILTER/SCF - 4-band equaliser + 2-band filter + sidechain filter sections.
- CHANNEL Buttons - various other channel functions including snapshot isolate (SNAP ISO), AFL for individual modules (LISTEN), etc.
- PANNING - multi-channel panning onto the channel's mix bus outputs.
- AUX SENDS - 32 aux sends.
- AUDIO FOLLOW VIDEO - audio follows video.

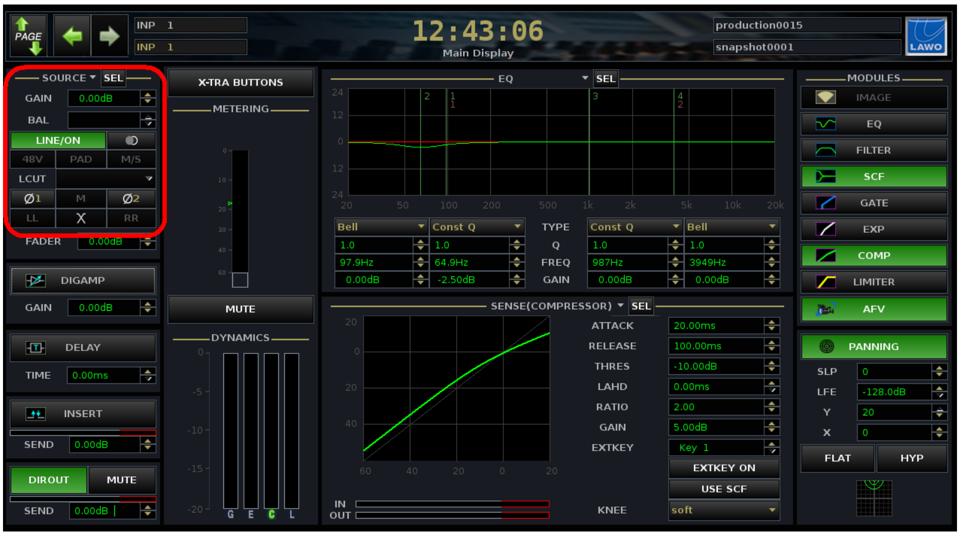
INPUT Control

The INPUT panel provides access to *all* available SOURCE and INMIX parameters:

- **SOURCE** – parameters applied to the source.
- **INMIX** – parameters applied to the INMIX channel DSP module.

Parameters are displayed on the SOURCE/INMIX area of the **Main Display**:

Main Display



Central Control Section

The available parameters depend on the type of source routed to the channel (mic/line or digital) and the channel format (mono or stereo). Any parameters which are not available for your input are greyed out on the **Main Display** and cannot be selected from the front panel.

The INPUT panel also provides access to TONE switching. When **TONE** is enabled, all other INPUT controls are temporarily disabled.

Note that the **IN** button is reserved for future implementation.

SOURCE & INMIX Modes

The default mode for the INPUT panel is to control **SOURCE** parameters. However, it can be switched to **INMIX** mode from the **Main Display** as follows:

1. Using the trackball select either **INMIX** or **SOURCE** from the drop-down menu.

Alternatively, you can programme a user button to switch the mode from the Custom Functions display.

The choice of mode affects GAIN, BALANCE and sometimes phase controls depending on the type of source. It also affects the operation of the on-screen **SEL** button; select **SOURCE** to select source gain or **INMIX** to select channel gain.

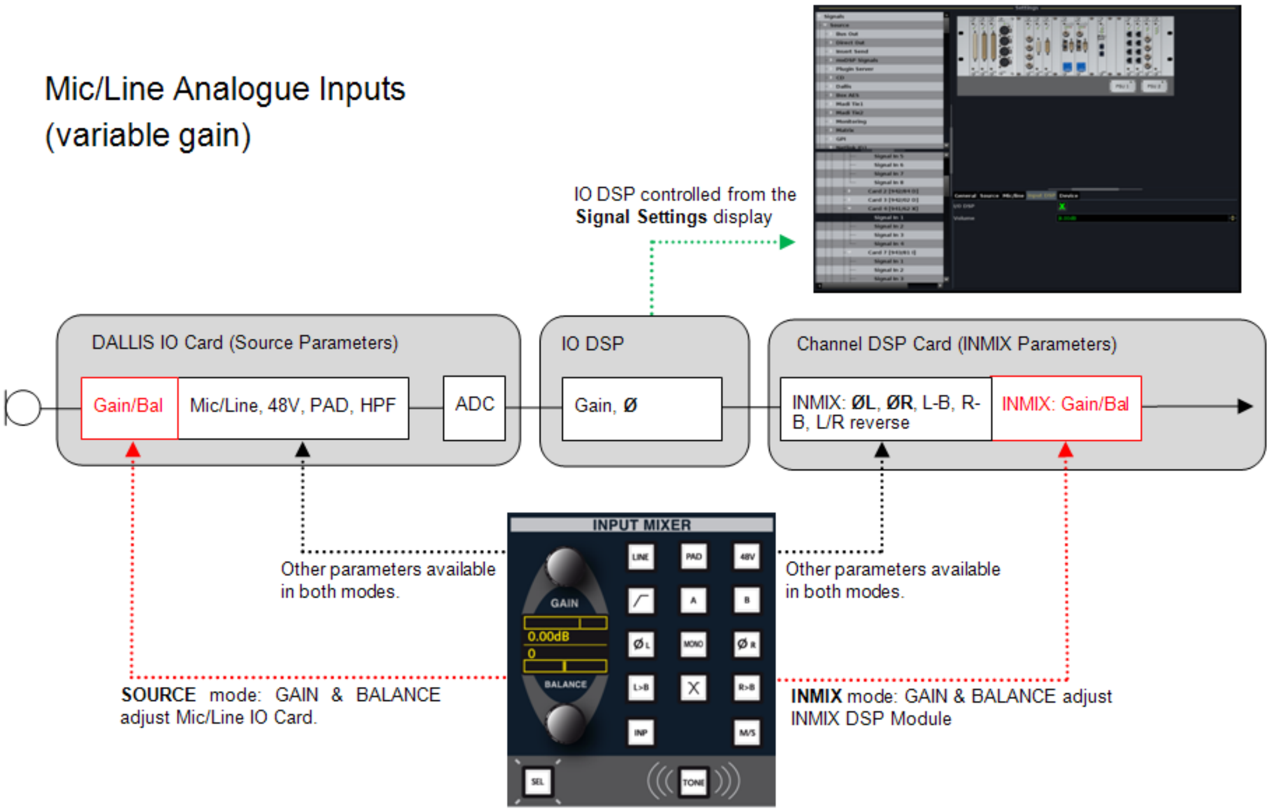
We recommend working in **SOURCE** mode most of the time to ensure that GAIN is applied to the source. Switch to **INMIX** mode if you need to adjust the channel input gain. For example, to access additional input gain for a digital microphone channel.

Changes made to **SOURCE** parameters affect *all* destinations routed from the source. Changes to **INMIX** parameters affect *only* the selected channel.

Mic/Line Input Control

Channels routed from a mic/line analogue input card (with variable gain) have access to:

- **I/O Card Parameters** (SOURCE mode) – mic/line switching, gain, balance, 48V, PAD and a high pass filter are applied in the analogue domain before analogue-to-digital conversion.
- **I/O DSP** – volume and phase can also be applied by DSP on the I/O card. These parameters are adjusted from the Signal Settings display.
- **INMIX DSP Parameters** (INMIX mode) – gain, balance, phase and stereo input control are applied within the channel's INMIX DSP.



Note that gain may be applied from the:

- INPUT panel (**SOURCE** mode) – GAIN adjusts the analogue mic preamp gain.
- **Signal Settings** display – the I/O DSP Volume parameter adjusts gain within the routing matrix.
- INPUT panel (**INMIX** mode) – GAIN adjusts the INMIX channel input gain.

➢ Mic Level Signals

1. Make sure that the INPUT panel is switched to **SOURCE** mode (the default).

*The **Main Display** should show **SOURCE**.*

2. For a mic level signal, deselect the **LINE/ON** button.

3. Use the **GAIN** control to remotely set the mic preamp gain within the analogue domain prior to A-D conversion.

The gain range is normally adjusted from -20dB to +70dB, although this may vary depending on your hardware configuration. Please refer to the DALLIS I/O data sheets for details.

4. Press the **48V** button to select 48V phantom power.

5. Press **PAD** to insert the PAD.

6. Press the high pass filter button to insert an analogue subsonic filter prior to the A-D conversion.

7. Toggle the high pass filter button to cycle through the froll-off frequency options: **Off**, **40Hz**, **80Hz** and **140Hz**.

*The **Main Display** shows the status of the settings.*

➢ Line Level Signals

1. Select the **LINE/ON** button.

*The **LINE/ON** indicator lights on the **Main Display**, and **48V** and **PAD** are cancelled; they cannot be selected for a line input.*

2. Use the **GAIN** control to remotely set the input gain within the analogue domain prior to the A-D conversion.

3. Set the high pass filter as described above.

➢ INMIX Parameters

1. To adjust the channel input gain, switch to **INMIX** mode from the **Main Display**.

*The **Main Display** should show **INMIX**.*

2. Use the **GAIN** control to adjust the INMIX gain within the digital domain.

Gain may be adjusted from -128dB to +70dB.

➢ Phase Reverse

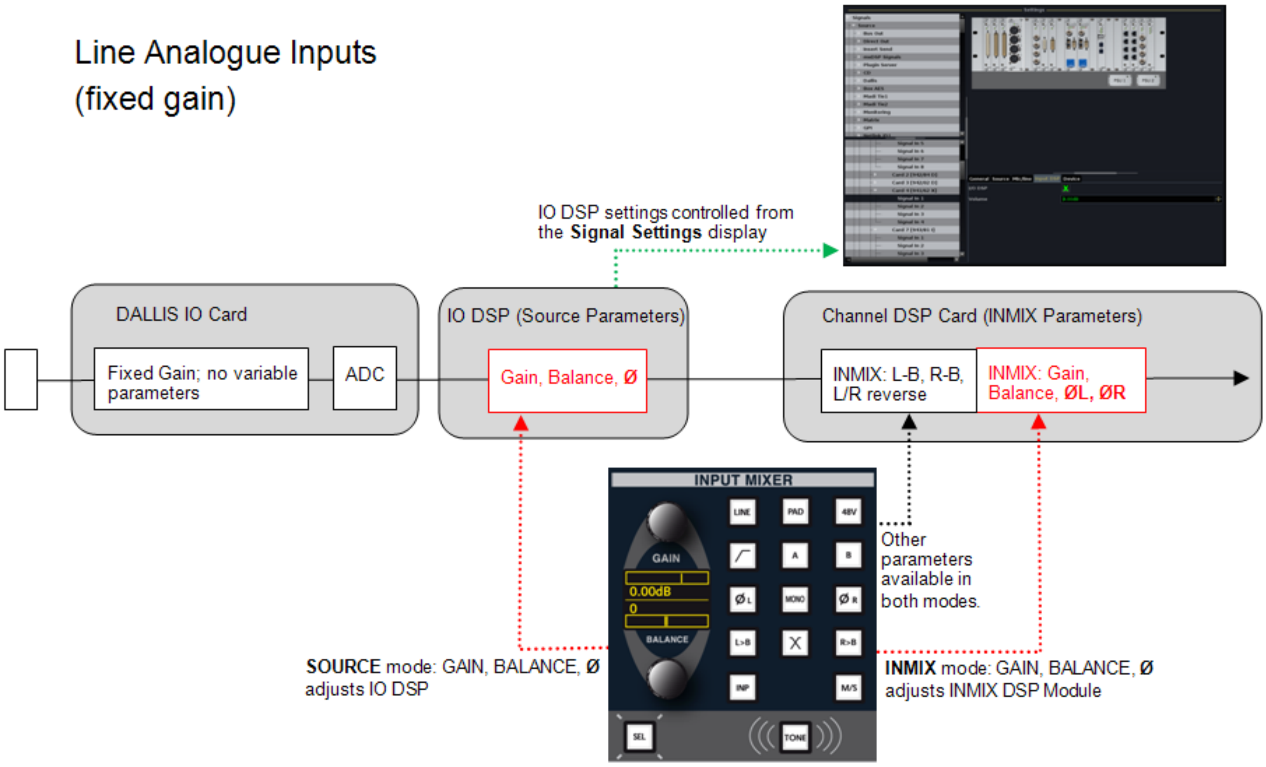
1. On a mono channel, press the **Ø L** button to reverse the phase.

For a mic/line input, the **Ø** buttons adjust the phase within the INMIX DSP module. Therefore, this reverses the phase of the channel and not the source. If you wish to reverse the phase of the source, then use the I/O DSP **Ø** which can be controlled from the Signal Settings display.

Analogue Input Control (Fixed Gain)

Channels routed from a line level analogue input card (with fixed gain) have no variable I/O card parameters. Therefore, the I/O DSP is used to provide source parameter control. Channels have access to:

- **I/O DSP Parameters** (SOURCE mode) – digital gain, balance and phase are applied by DSP on the I/O card.
- **INMIX DSP Parameters** (INMIX mode) – gain, balance, phase and stereo input control are applied within the channel's INMIX DSP module.



➢ SOURCE Parameters (I/O DSP)

1. Make sure that the INPUT panel is switched to **SOURCE** mode (the default).

*The **Main Display** should show **SOURCE**.*

2. Select the **LINE/ON** button - this button turns the I/O DSP on or off:

- **LINE/ON** lit = I/O DSP On
- **LINE/ON** unlit = I/O DSP Off

3. Use the **GAIN** control to adjust the I/O DSP gain (volume).

Gain may be adjusted from -128dB to +15dB.

4. Press the **Ø L** button to reverse the phase of the mono source.

*The Ø 1 indicator lights on the **Main Display**.*

Note that any changes you make to the I/O DSP will also appear on the Signal Settings display.

➢ INMIX Parameters

1. To adjust the channel input gain, switch to **INMIX** mode from the **Main Display**.

*The **Main Display** should show **INMIX**.*

2. Use the **GAIN** control to adjust the INMIX gain within the digital domain.

Gain may be adjusted from -128dB to +70dB.

3. Press the **Ø L** button to reverse the phase of the channel (INMIX phase).

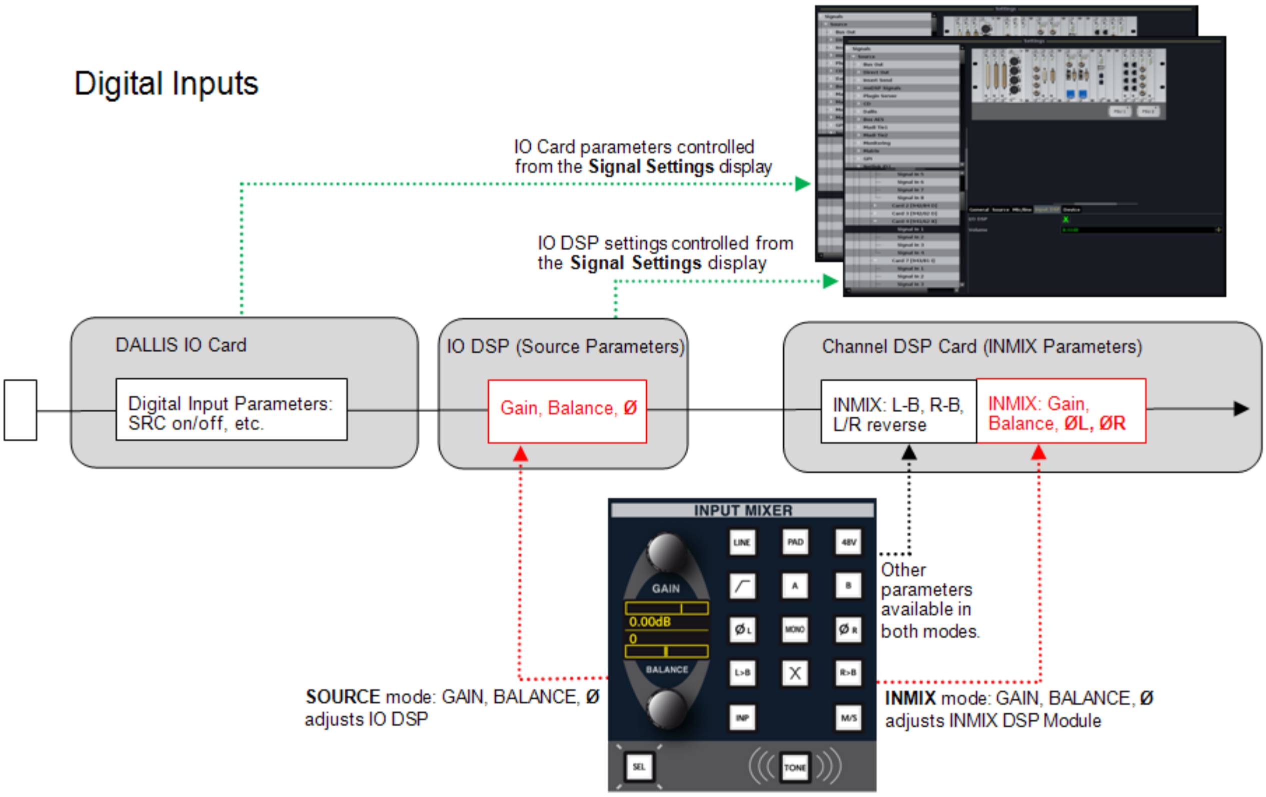
*The **Ø 1** indicator lights on the Main Display.*

Note that the **48V**, **PAD** and analogue filter buttons are inactive.

Digital Input Control

Channels routed from a digital input card may have some I/O card parameters, such as SRC on/off, which are adjusted from the **Signal Settings** display. The I/O DSP is used to provide source parameters. Therefore, channels have access to:

- **I/O Card Parameters** – for example, SRC on/off. These parameters are adjusted from the **Signal Settings** display, see AES/EBU Input Parameters.
- **I/O DSP Parameters** (SOURCE mode) – digital gain, balance and phase are applied by DSP on the I/O card.
- **INMIX DSP Parameters** (INMIX mode) – gain, balance, phase and stereo input control are applied within the channel's INMIX DSP module.



Parameters are controlled in an identical manner to a fixed gain analogue input, please see the previous section.

Internal Signals

Input channels routed from an internal signal such as a summing bus have no I/O card or I/O DSP parameters. This leaves the INMIX parameters:

➢ **INMIX Parameters**

1. To adjust the channel input gain, switch to **INMIX** mode from the **Main Display**.

*The **Main Display** should show **INMIX**.*

2. Use the **GAIN** control to adjust the INMIX gain within the digital domain.

Gain may be adjusted from -128dB to +70dB.

3. Press the **Ø L** button to reverse the phase of the channel (INMIX phase).

*The **Ø 1** indicator lights on the Main Display.*

Note that the **48V**, **PAD** and analogue filter buttons are inactive.

A/B Input Switching

For any input channel, you may assign two sources (A and B) to provide a main and backup source for the channel.

The sources are assigned from the **Signal List** display, see A/B Input Sources.

1. Use the **A** and **B** buttons to switch the input.

If there is no source assigned to the B input, then the **B** button cannot be selected.

2. Use the **GAIN** control (in **SOURCE** mode) to set an independent gain value for source A and source B. Depending on the type of input, you may be adjusting the mic/line gain before A-D conversion, or digital I/O DSP gain.

3. Use the **GAIN** control (in **INMIX** mode) to adjust the channel input gain after the A/B input switch.

Stereo Input Control

When an input channel is stereo, a number of additional controls become available: **BALANCE**, **Ø R** and stereo input management.

Note that **GAIN**, **BALANCE** and **Ø** may be applied to the source or channel depending on the SOURCE/INMIX mode.

1. With the INPUT panel switched to **SOURCE** mode (the default), use the **GAIN** control to adjust source gain - the gain range depends on the type of input (mic/line or analogue fixed gain/digital).

   *The gain for left and right inputs is adjusted in parallel; any offsets are retained and represented by a positive or negative **BALANCE** value.*

   To adjust source gain independently for the left and right inputs, you can use the Mic/Line Gain or I/O DSP Volume parameters from the **Signal Settings** display.

2. Use the **BALANCE** control to set the Left/Right input balance for the stereo input.

3. Press the **MONO** button to sum the Left and Right inputs.

4. Press the button to reverse the Left and Right inputs.

5. Press the **Ø L** or **Ø R** buttons to reverse the phase.

6. Press either **L>B** (Left to Both) or **R>B** (Right to Both) to route either the left or right source to both sides of the stereo channel.

7. Select **M/S** for sources recorded using sum and difference coding.

8. Switch the INPUT panel to **INMIX** mode if you wish to adjust the GAIN and BALANCE for the channel.

*The status of all settings is indicated on the **Main Display**.*

TONE to Channel

From V4.24 software onwards, the **TONE** button switches test tone to the channel.

Note that this function may *only* be selected on Input or Monitor channels, and temporarily replaces the channel's source.

By default, the first internal tone generator signal (sine 1) is used as the test tone source. However, you can use the Activate Test Tone Button, programmed from the **Custom Functions** display, to specify a different tone source - for example, to use one of other internal generator signals (sine 2, white noise, pink noise) or an external source.

When using the internal tone generator, select the signal (e.g. **sine 1**), in the Signal Settings display, to adjust the level and frequency of the tone.

When **TONE** is enabled, all other INPUT controls are temporarily disabled.

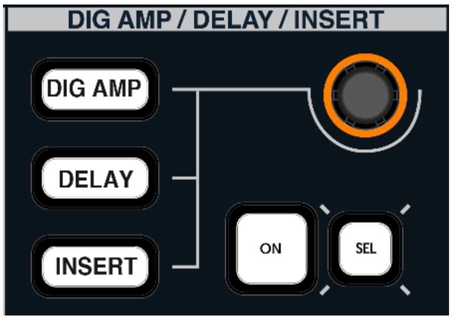
Switch **TONE** off to return the channel to its assigned source (displayed in the **Signal List** / **mx Routing** displays).

On systems upgraded to V4.24, you must use the Activate Test Tone Button, programmed from the **Custom Functions** display, to activate the test tone source. On all new systems (>V4.24) or those updated by creating new CF cards, tone is automatically activate.

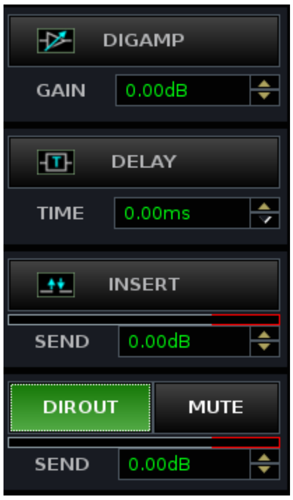
You can program a fader strip user button, from the Custom Functions display, to emulate the **TONE** button. This can be used to provide fast, direct access to tone switching across the console.

Digital Amplifier (DIG AMP)

Central Control Section



Main Display

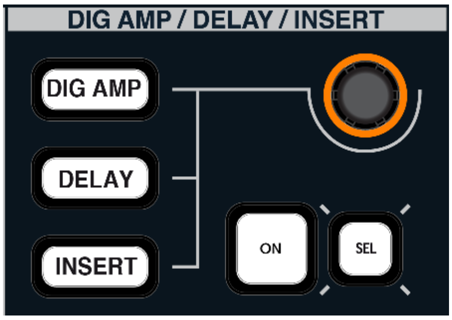


1. Press the **DIG AMP** button to switch the DIG AMP/DELAY/INSERT controls to the digital amplifier.
2. Press **ON** to switch the gain element in and out of circuit.
3. Move the rotary control to adjust the digital gain - gain may be adjusted from -128dB to +15dB.

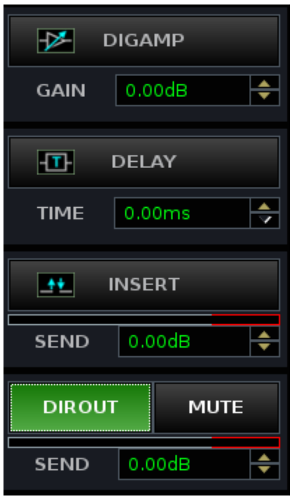
*The gain value is displayed in the **DIGAMP GAIN** box on the **Main Display**.*

Channel Delay (DELAY)

Central Control Section



Main Display

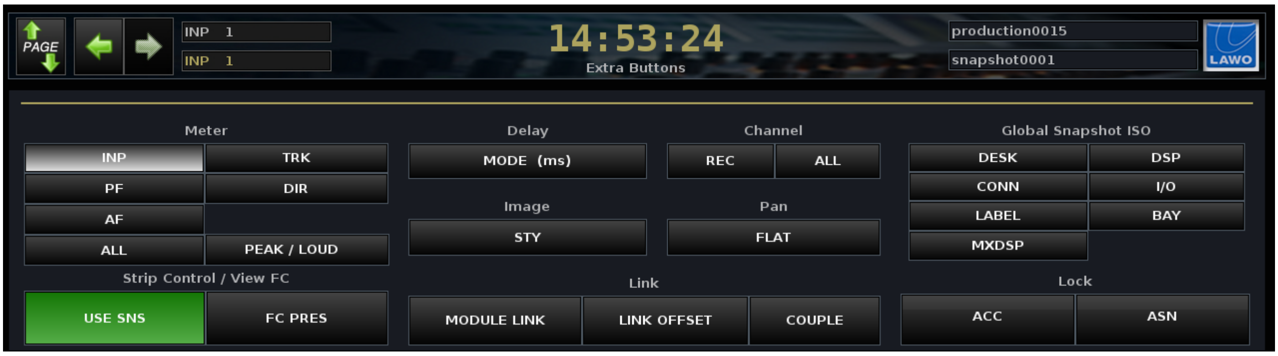


1. Press the **DELAY** button to switch the DIG AMP/DELAY/INSERT controls to the channel delay.
2. Press **ON** to switch the delay in and out of circuit
3. Move the rotary control to adjust the delay time.

*The amount of delay is displayed in the **TIME** box on the **Main Display**.*

> To enter a specific delay time, click on the **TIME** box on the **Main Display** and type in a value from the console keyboard.

4. You can change the delay mode from the Extra Buttons display. Touch the on-screen **MODE** button to cycle around the options – milliseconds (ms), frames (frms) or meters (m):



Set Delay in ms or frames when you are dealing with a specific time delay, for example, to delay the channel's audio relative to an incoming video feed.

Set Delay in meters when you are time aligning microphones positioned on the studio floor and know the distance between the microphones.

The available channel delay varies slightly between Recording and Broadcast channels:

Recording channels	Broadcast channels
Min. = 1 samples (0.02 ms)	Min. = 18 samples (0.38 ms)
Max. = 1.8 seconds	Max. = 1.3 seconds

Note that if you load a Recording channel delay to a Broadcast channel (e.g. using a Preset), and the stored parameter lies outside the range supported by Broadcast channels, then the closest available value is applied. For example, if the preset is attempting to load a delay of 5 samples, then 18 samples (the minimum) is applied.

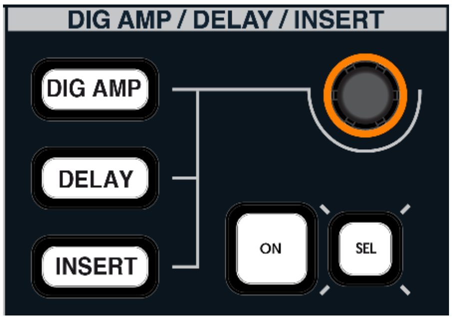
Depending on the hardware configuration of your console, an additional 48 delays may be available from the DSP Module 983-03. These are fixed time delays which may be inserted into any routing crosspoint and are programmed within the AdminHD configuration.

Channel Insert (INSERT)

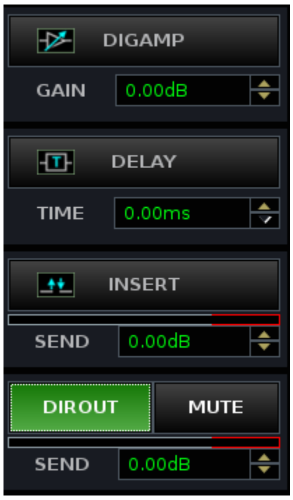
Routes to and from the channel insert send and return are made from the Signal List display. You should route the channel's insert send to the output feeding the insert device, and then route the output from the external device to the corresponding insert return.

The Central Control Section can then be used to control the insert on/off switching and send level:

Central Control Section



Main Display



1. Press the **INSERT** button to switch the DIG AMP/DELAY/INSERT controls to the channel insert.
2. Press **ON** to switch the insert return in and out of circuit.

If an insert return is not assigned, you will get silence when you switch the insert into circuit.

3. Adjust the rotary control to set the level of the insert send.

The SEND level is shown on the Main Display. It may be adjusted from -128dB to +15dB.

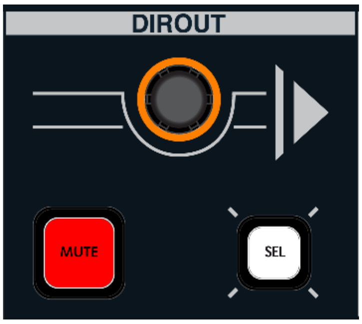
The channel insert send is always active even when the return is not inserted. This allows the insert send to be used to generate an extra clean feed from the channel, with level control, which may be taken from any point in the channel signal flow, see Changing the Signal Processing Order.

Direct Output (DIROUT)

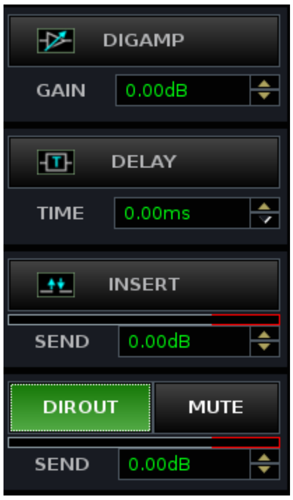
Routing from the channel direct output is made from the Signal List display.

The Central Control Section can then be used to control the direct output level:

Central Control Section



Main Display



1. Locate the DIR OUT controls in the Central Control Section.
2. Move the rotary control to adjust the direct output send level.

*The **SEND** level is displayed on the **Main Display**. It may be adjusted from -128dB to +15dB.*

3. Press **MUTE** to disable the direct output.

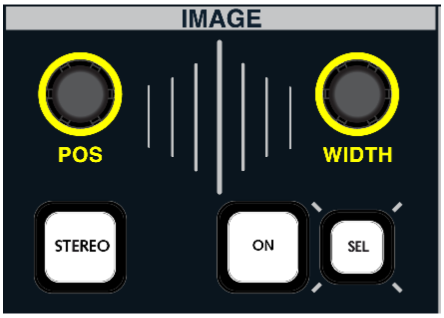
You can also set the direct out to mute when the channel fader opens, see Dir-Out mute by fader.

On a stereo channel, the direct out can be set to follow the channel pan position, see Dir-Out Balance.

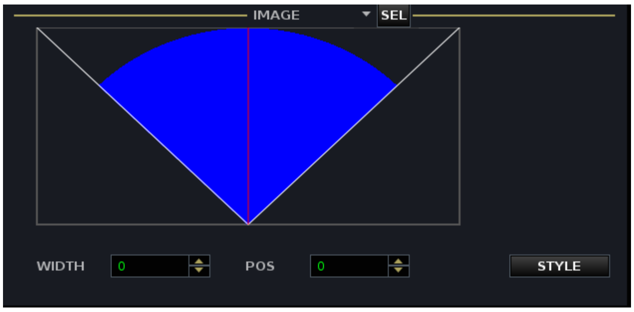
The direct output may be fed from any position in the channel signal flow. For example, you could use the direct output to create a pre fader send to feed a multitrack recorder, while using the post fader output for the live production mix. See Changing the Signal Processing Order.

IMAGE

Central Control Section



Main Display



The IMAGE controls deal with image positioning and width on stereo channels. Note that the controls are blank (unlit) if the Central Control Section is assigned to a mono channel.

1. Press the **STEREO** button to make (or unmake) a stereo channel from two adjacent mono DSP channel paths, see Creating a Stereo Channel for details.

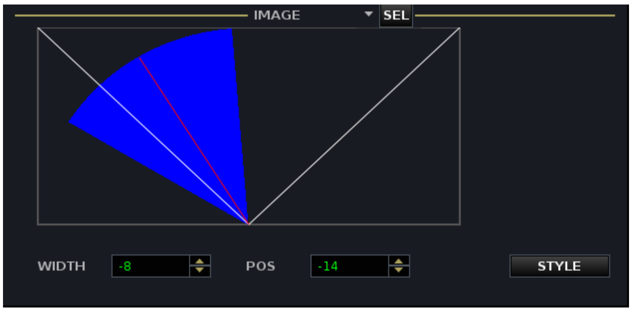
With **STEREO** enabled, you may then adjust the IMAGE controls:

2. Select **ON** to switch the Image section into circuit.
3. Choose the style, using the on-screen **STYLE** button, on the **Main Display**:

- **STYLE** off (default) - retains the width of the stereo image and offsets its position within the stereo field.
- **STYLE** on - collapses the width of the stereo image as you adjust the left/right position.

3. Use the **WIDTH** control to widen or narrow the stereo image.
4. Adjust the **POS** control to move the narrowed or widened image within the stereo field.

In our example, the image width is retained, and the red line on the Image graph moves as you adjust this control to represent the direction of the image control:



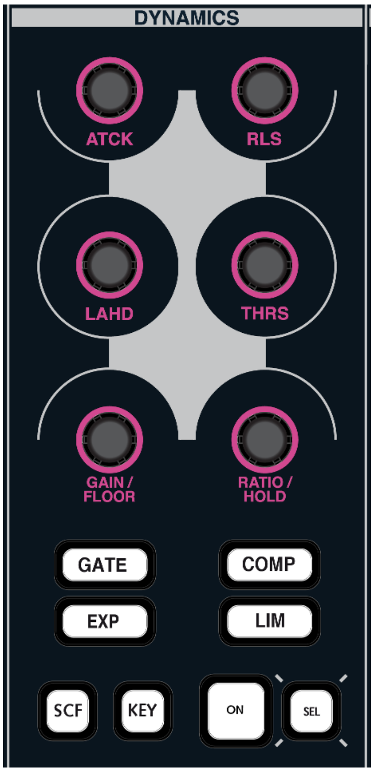
Be careful not to widen the stereo image too far. If you do so, you may create phasing problems.

DYNAMICS

The console's dynamics processing varies between Broadcast and Recording channels. Therefore, this section covers the two channel types separately:

- DYNAMICS (Recording channels)
- DYNAMICS (Broadcast channels)

DYNAMICS (Recording channels)



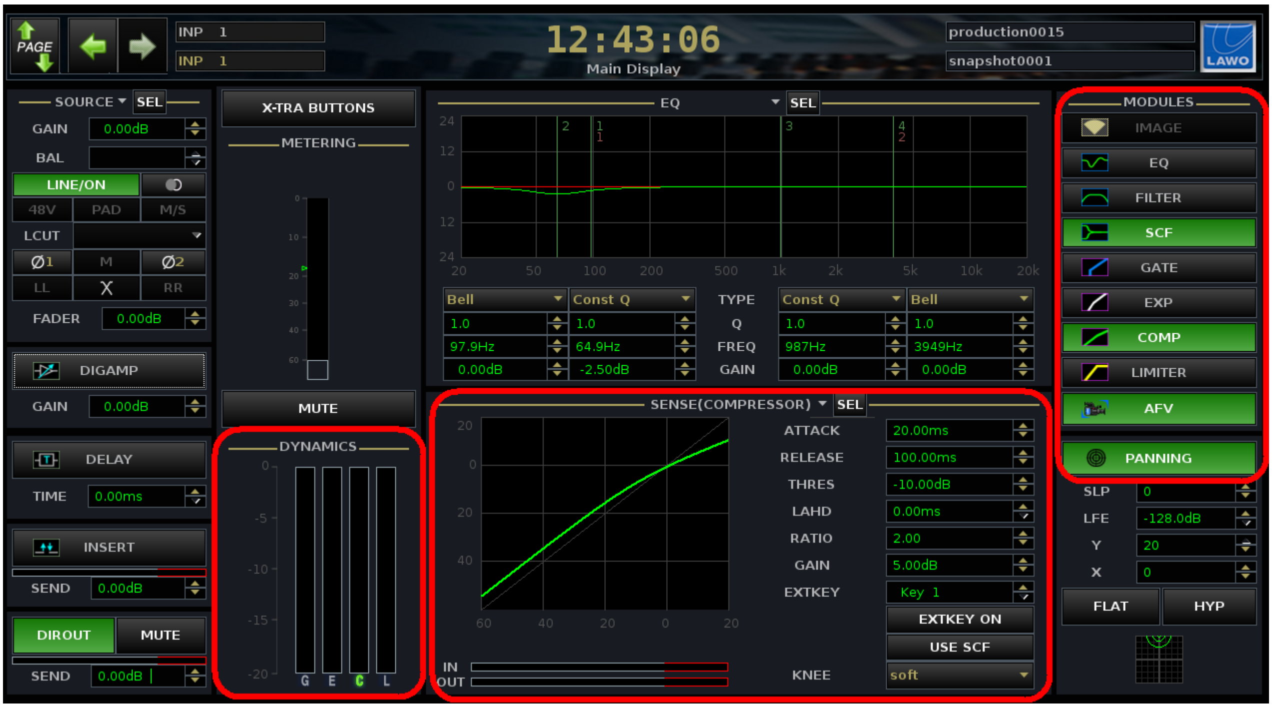
On Recording channels, each full processing channel contains four independent blocks of dynamics processing (Gate, Expander, Compressor and Limiter).

Any of the four sections may be placed anywhere within the channel signal flow. For example, to gate pre EQ and compress post EQ, or to limit the channel signal post fader while compressing the feed to the direct output. See Changing the Signal Processing Order.

In addition, a dedicated 2-band filter section may be inserted into the sidechain of the compressor or gate, see Filtering the Dynamics Sidechain.

The **KEY** (External Key) button can be used to trigger the gate and compressor sections from an external dynamics key.

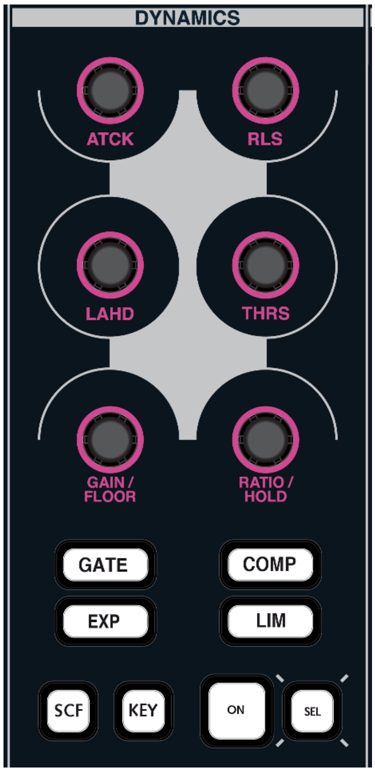
The **Main Display** always shows gain reduction metering (**DYNAMICS**) and the on/off status (**MODULES**) for all 4 sections. In addition, the current Gate, Expander, Compressor or Limiter parameters can be assigned to the display:



Note that the **IN** and **OUT** meters show the levels to and from module. The **DYNAMICS** metering shows the amount of gain reduction; the **G, E, C** and **L** light in green if the module is turned on.

Setting a Gate

1. Press the **GATE** button to switch the DYNAMICS controls to the gate section.
2. Press **ON** to switch on the gate.
3. Use the six rotary controls to set the parameters.



The action is best described by looking at the **GATE** graph on the **Main Display**:

The gate parameters may be set as follows:

- Threshold Level – from -80dB to 0dB.
- Floor Level – from 0dB to -128dB.
- Attack Time – from 0.10ms to 250ms.
- Release Time – from 40ms to 10s.
- Hold Time – from 0ms to 500ms.
- Look Ahead Delay – from 0ms to 10ms.

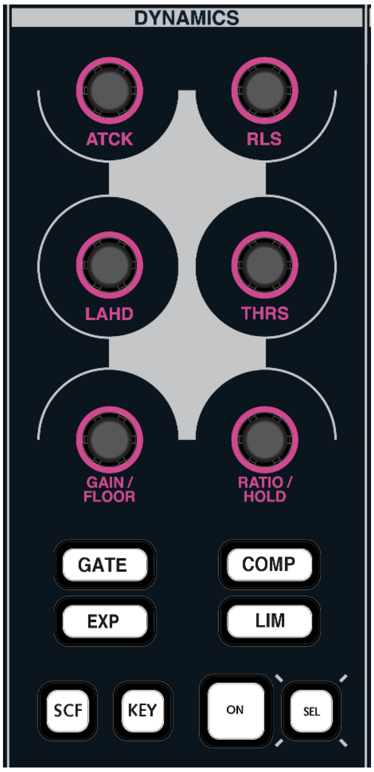
Note that the **DYNAMICS** gain reduction metering follows the attack and release settings for each dynamics section. So, if you have a very fast gate attack, the metering will reflect this.

Using the Expander

1. Press the **EXP** button to switch the DYNAMICS controls to the expander section.

2. Press **ON** to switch the expander into circuit

3. Use the six rotary controls to set the parameters.

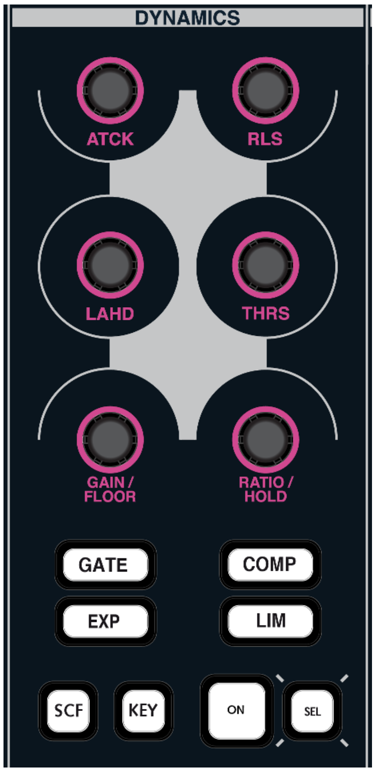


The action is best described by looking at the **EXPANDER** graph on the **Main Display**:

The expander parameters may be set as follows:

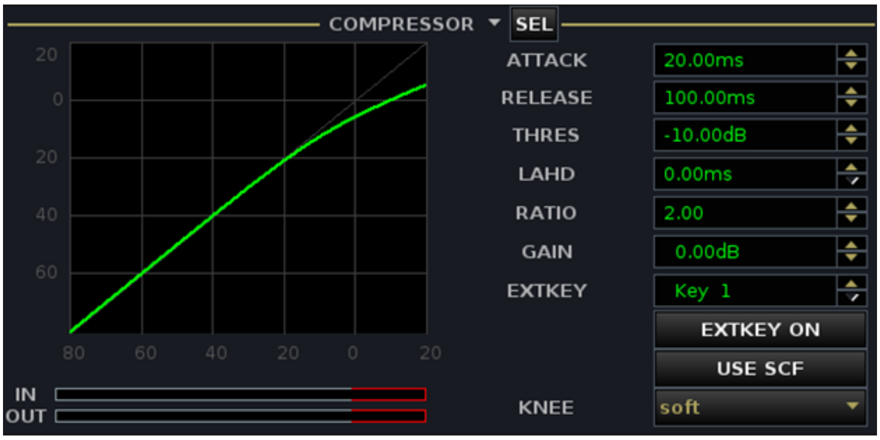
- Threshold Level – from -80dB to 0dB.
- Ratio – from 0.1:1 to 1:1.
- Floor Level – from 0dB to -40dB.
- Attack Time – from 0.10ms to 250ms.
- Release Time – from 40ms to 10s.
- Look Ahead Delay – from 0ms to 10ms.

Setting a Compressor



1. Press the **COMP** button to switch the DYNAMICS controls to the compressor section.
2. Press **ON** to switch on the compressor.
3. Use the six rotary controls to set the parameters.

The action is best described by looking at the **COMPRESSOR** graph on the **Main Display**:

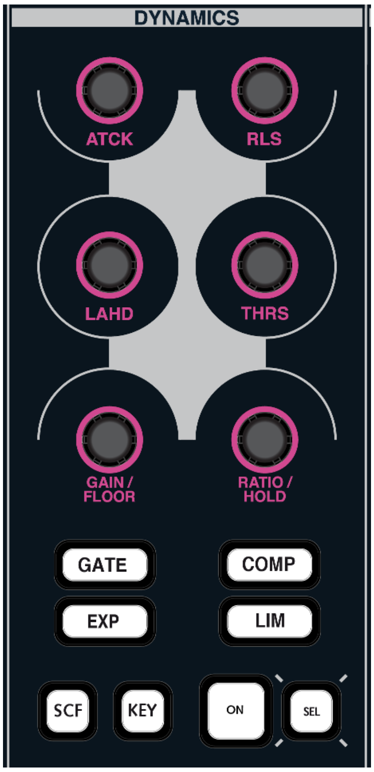


The compressor parameters may be set as follows:

- Threshold Level – from -70dB to +20dB.
- Ratio – from 1:1 to 10:1.
- Attack Time – from 0.29ms to 250ms.
- Release Time – from 40ms to 10s.
- Look Ahead Delay – from 0ms to 10ms.
- Gain – from -20dB to +20dB.
- Knee – hard or soft. This parameter is set from the **Main Display**. Use the trackball to set the **KNEE** option to either **hard** or **soft**.

For a smoother compressor, assign the 2-band sidechain filter to the compressor and set -10dB gain for an 18dB/octave low shelf at around 125Hz to remove unwanted low frequencies.

Setting a Limiter



1. Press the **LIM** button to switch the DYNAMICS controls to the limiter section.
2. Press **ON** to switch on the limiter.
3. Use the six rotary controls to set the parameters.

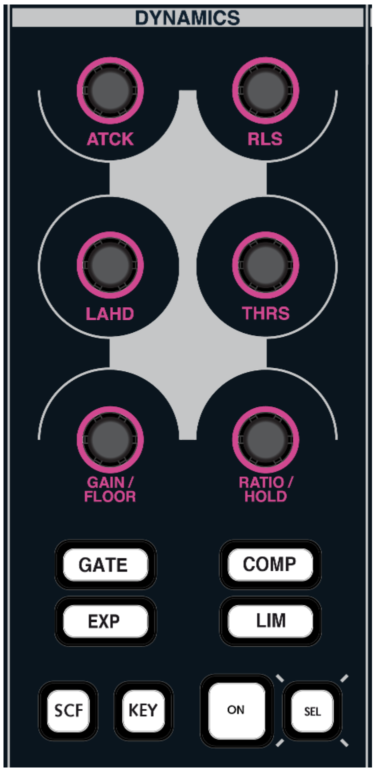
The action is best described by looking at the **LIMITER** graph on the **Main Display**:

The limiter parameters may be set as follows:

- Threshold Level – from -40dB to +20dB.
- Attack Time – from 0.29ms to 20ms.
- Release Time – from 40ms to 10s.
- Hold Time – from 0ms to 500ms.
- Look Ahead Delay – from 0ms to 10ms.
- Knee – hard or soft. This parameter is set from the **Main Display**. Use the trackball to set the **KNEE** option to either **hard** or **soft**.

For best results you should give the limiter the chance to ‘see’ signal peaks in advance by setting a look ahead delay of 5ms.

Filtering the Dynamics Sidechain



On the DYNAMICS panel you will find a **SCF** (Sidechain Filters) button:

1. Press the **SCF** button to key either the gate or the compressor from the filtered sidechain.

Note that you may not select sidechain filtering for more than one section of dynamics processing, and filtering may only be applied to the gate or compressor (not expander or limiter).

2. To view the sidechain filter settings on the **Main Display**, assign the **SCF** module to the display, or if **SENSE** is already assigned, touch a sidechain filter control to update the **SENSE** area:

3. Now move across to the EQUALIZER/FILTER/SCF section and use the **SCF** (Sidechain Filter) controls to process the sidechain signal.

You may audition the sidechain signal by using the CHANNEL: LISTEN function.

Dynamics External Key Inputs

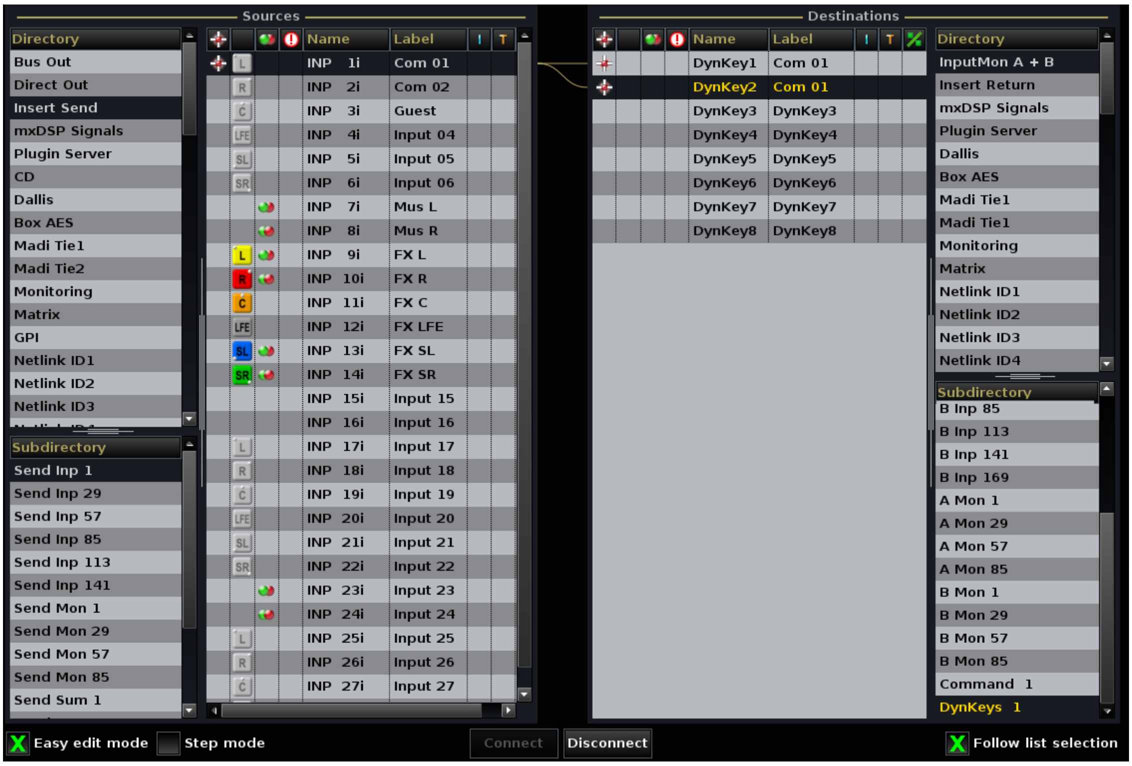
On the DYNAMICS panel, you will find the **KEY** (External Key) button.

The console supports eight external key inputs which can be assigned to any Gate or Compressor section. Each dynamics key may be routed from any source and each key may be assigned to one or more dynamics sections.

➢ Routing the External Key Source

Any source can be routed to a dynamics key signal from the Signal List display:

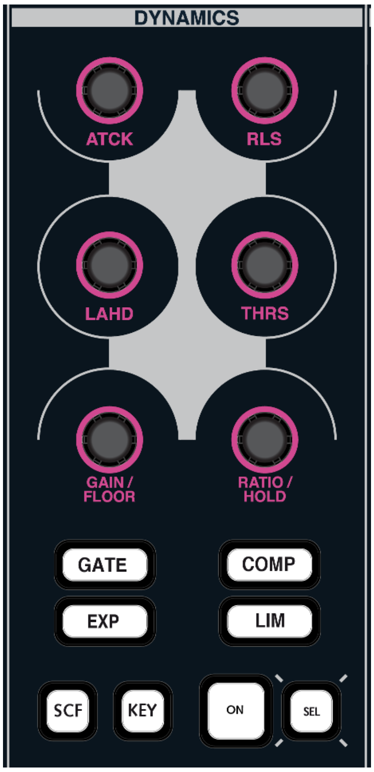
1. Select the **Input/Mon A+B** directory as your destination to reveal the **DynKeys 1-8** subdirectory:



2. Select the subdirectory and connect sources to each dynamics key in the usual manner.

You can assign a physical input or internal signals such as a mix bus, insert send or direct out. For example, if you wish to trigger a gate from another channel, choose the channel insert send as the source for the key signal.

Assigning the Key to a Gate/Compressor



Each of the eight dynamics key signals can be assigned to any Gate or Compressor section from the **Main** display:

1. Enter the number (**Key 1** to **Key 8**) of the signal you wish to assign into the **EXTKEY** field:

2. Activate the key signal by enabling **EXTKEY ON** or pressing the **KEY** button on the front panel.

Note that you can assign the same key signal to several dynamics sections if you wish.

DYNAMICS (Broadcast channels)

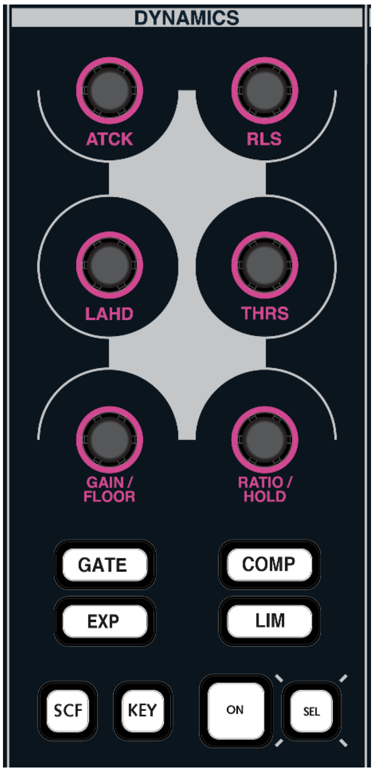
On Broadcast channels, each full processing channel has dynamics which can operate using one of two models:

- **Combi-Dynamics** – three modules: Gate, Expander and Compressor.
- **Limiter** – one module: a Limiter.

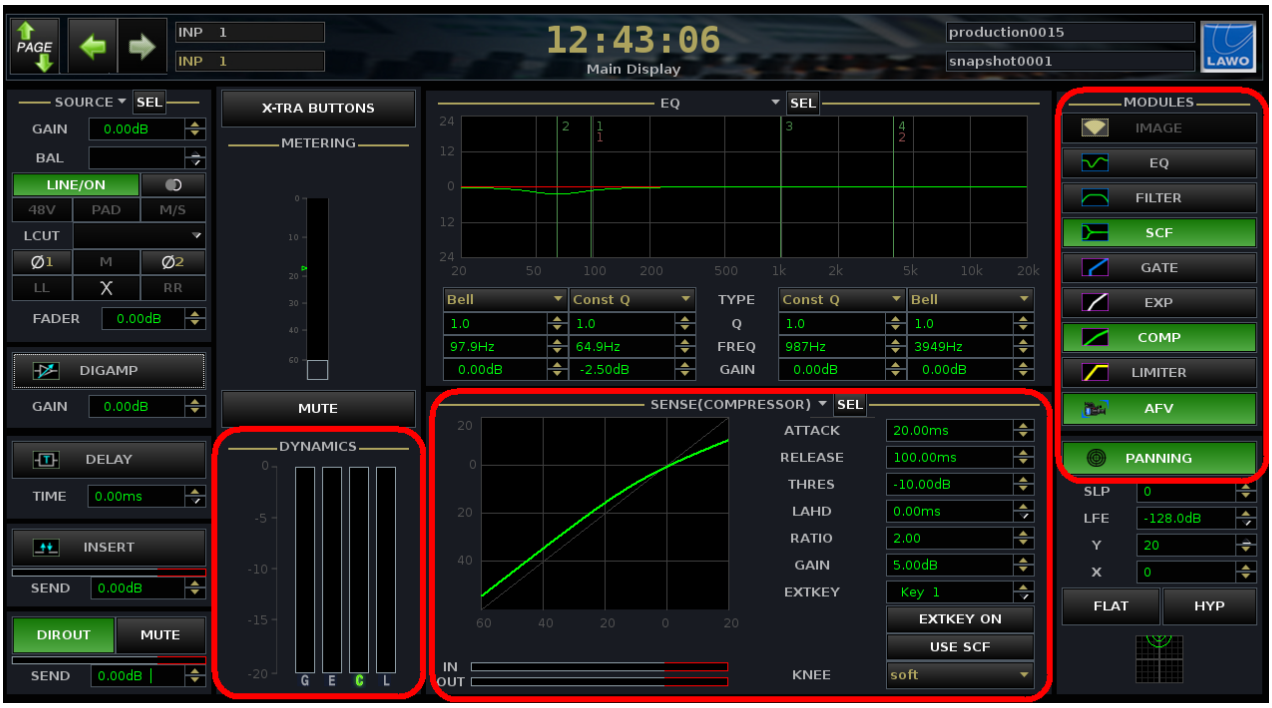
This option is selected from the **Channel Config** display, see Changing the Dynamics Model.

Depending on the choice of model, you may be able to switch the DYNAMICS controls to **GATE**, **COMP** and **EXP**, or only the **LIM**.

Broadcast channels do not support sidechain filtering or external key inputs.

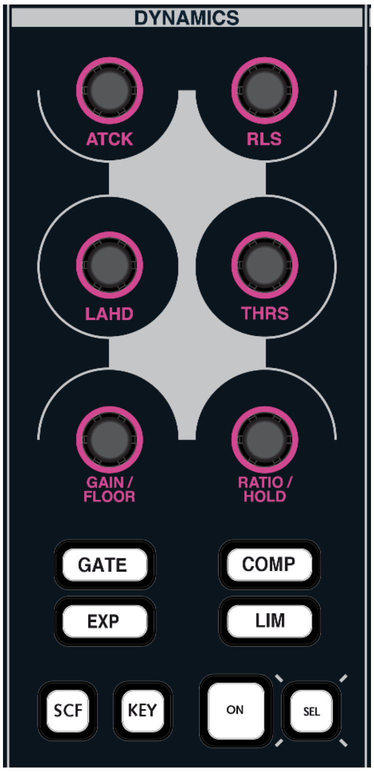


The **Main Display** always shows gain reduction metering (**DYNAMICS**) and the on/off status (**MODULES**) for all 4 sections. In addition, the current Gate, Expander, Compressor or Limiter parameters can be assigned to the display:



Note that the **IN** and **OUT** meters show the levels to and from module. The **DYNAMICS** metering shows the amount of gain reduction; the **G, E, C** and **L** light in green if the module is turned on.

Limiter Model



When the **Limiter** model is selected:

1. Press the **LIM** button to switch the DYNAMICS controls to the limiter section.

Note that if you select **GATE**, **EXP** or **COMP** the controls go blank (unlit) as the other dynamics modules do not exist.

2. Press **ON** to switch on the limiter.

3. Use the six rotary controls to set the parameters.

The action is best described by looking at the **LIMITER** graph on the **Main Display**:

The limiter parameters may be set as follows:

- Threshold Level – from -40dB to +20dB.
- Attack Time – from 0.29ms to 20ms.
- Release Time – from 40ms to 10s.
- Hold Time – from 0ms to 500ms.
- Look Ahead Delay – from 0ms to 10ms.
- Knee – hard or soft. This parameter is set from the **Main Display**. Use the trackball to set the **KNEE** option to either **hard** or **soft**.

For best results you should give the limiter the chance to 'see' signal peaks in advance by setting a look ahead delay of 5ms.

Combi-Dynamics Model

When the **Combi-Dynamics** model is selected, three dynamics modules can be adjusted: GATE, EXPANDER and COMPRESSOR.

Each module can be turned on or off independently, and has separate threshold, ratio and other parameter values. However, because the **Combi-Dynamics** works as a single block of processing, the following restrictions apply:

- The thresholds of the Gate, Expander and Compressor cannot overlap:
  - o The Gate Threshold must be equal to or lower than the Expander Threshold.
  - o The Expander Threshold must be at least 10dB lower than the Compressor Threshold (due to the soft knee operation of the compressor).

If you move a threshold outside of these limits, then the corresponding thresholds move up or down accordingly. For example, with the Compressor Threshold set to -20dB, and the Gate Threshold to -40dB, if you move the Gate Threshold above -30dB, the Compressor Threshold is also raised, to maintain the 10dB gap:

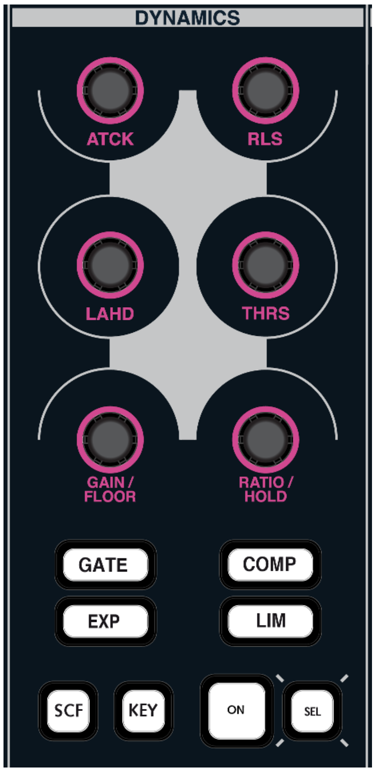
- There is one look ahead delay (**LAHD**) for the **Combi-Dynamics** sidechain. In otherwords, you cannot delay the Gate sidechain independently from the Compressor.

If you adjust the **LAHD** control on the Gate, then you will see the **LAHD** value on the Compressor and Expander follow, and vice versa:

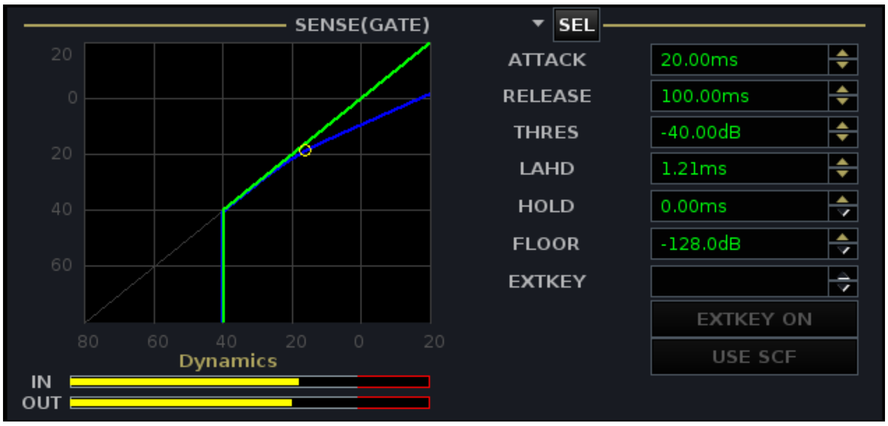
- The **IN** and **OUT** meters on the **Main Display** represent the levels to and from the complete **Combi-Dynamics**. In other words, the **IN** meter shows the level at the input to the Gate, and the **OUT** meter shows the level at the output from the Compressor.
- Each of the **Main Display** graphs reflects the combined result of the **Combi-Dynamics**: the green line shows the parameter curve for the sensed or selected section; the blue line shows the resultant curve of the active dynamics.
- When you pre-listen any of the **Combi-Dynamics** modules, you are switching the output of the **Combi-Dynamics** to the AFL bus. In other words you are listening to the combined result of the Gate, Expander and Compressor. See CHANNEL: LISTEN.

Setting a Gate

1. Press the **GATE** button to switch the DYNAMICS controls to the gate section.
2. Press **ON** to switch on the gate.
3. Use the six rotary controls to set the parameters..



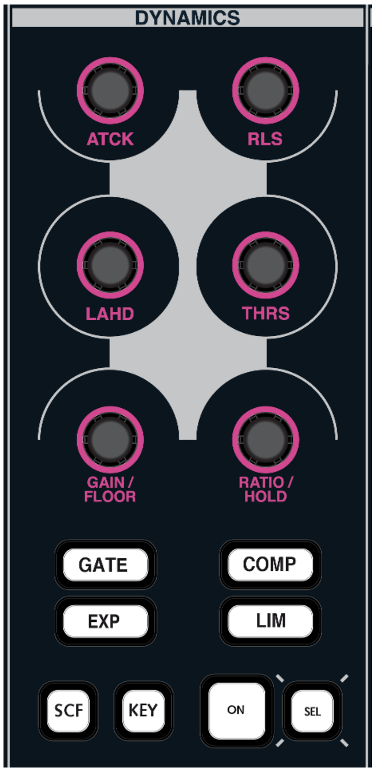
The action is best described by looking at the **GATE** graph on the **Main Display**:



The Gate parameters may be set as follows:

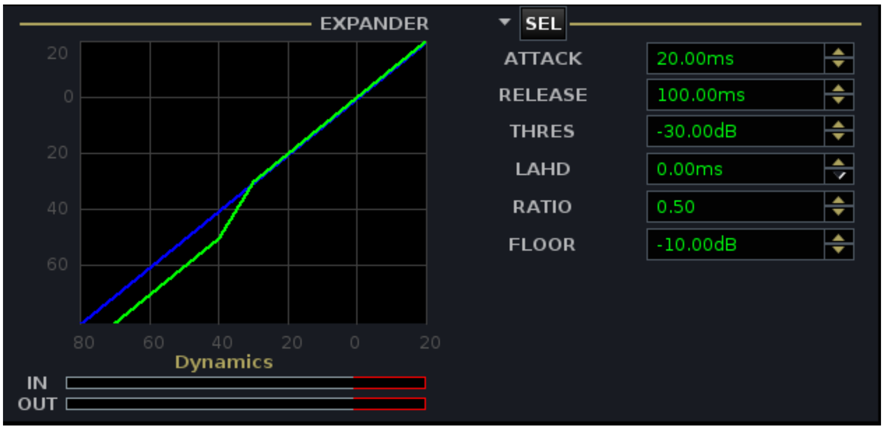
- Threshold Level – from -80dB to 0dB (must be equal to or lower than the Expander Threshold.)
- Floor Level – from 0dB to -128dB.
- Attack Time – from 0.10ms to 250ms.
- Release Time – from 40ms to 10s.
- Hold Time – from 0ms to 500ms.
- Look Ahead Delay – from 0ms to 10ms (look ahead delay affects all three Combi-Dynamics modules).

Using the Expander



1. Press the **EXP** button to switch the DYNAMICS controls to the expander section.
2. Press **ON** to switch the expander into circuit
3. Use the six rotary controls to set the parameters.

The action is best described by looking at the **EXPANDER** graph on the **Main Display**:



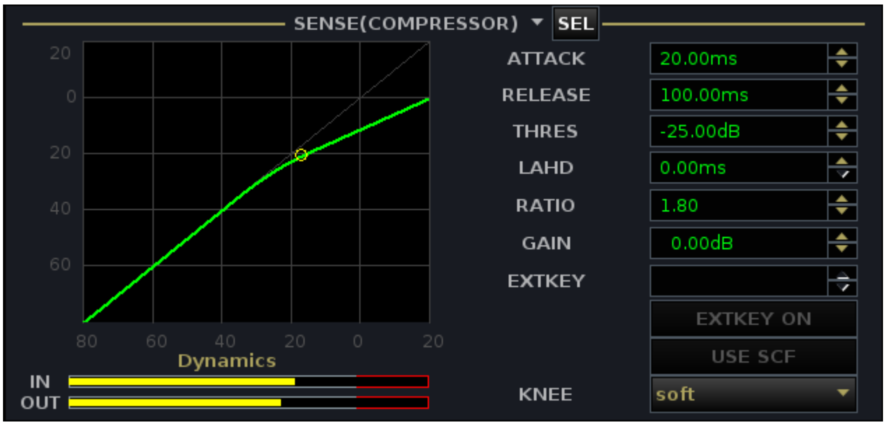
The Expander parameters may be set as follows:

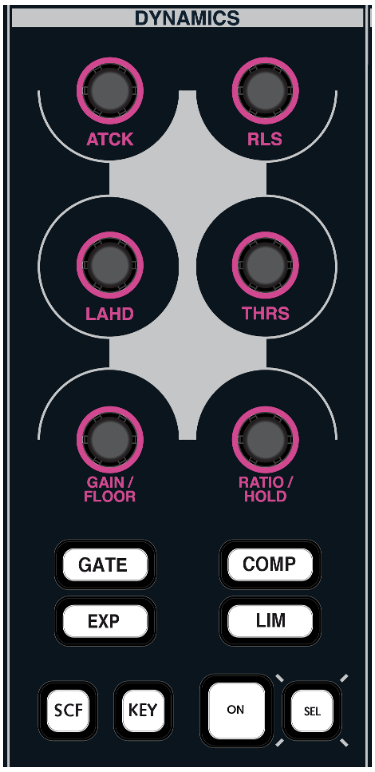
- Threshold Level – from -80dB to 0dB (must be equal to or higher than the Gate Threshold, and at least 10dB lower than the Compressor Threshold.)
- Ratio – from 0.1:1 to 1:1.
- Floor Level – from 0dB to -40dB.
- Attack Time – from 0.10ms to 250ms.
- Release Time – from 40ms to 10s.
- Look Ahead Delay – from 0ms to 10ms (look ahead delay affects all three Combi-Dynamics modules).

Setting a Compressor

1. Press the **COMP** button to switch the DYNAMICS controls to the compressor section.
2. Press **ON** to switch on the compressor.
3. Use the six rotary controls to set the parameters.

The action is best described by looking at the **COMPRESSOR** graph on the **Main Display**:

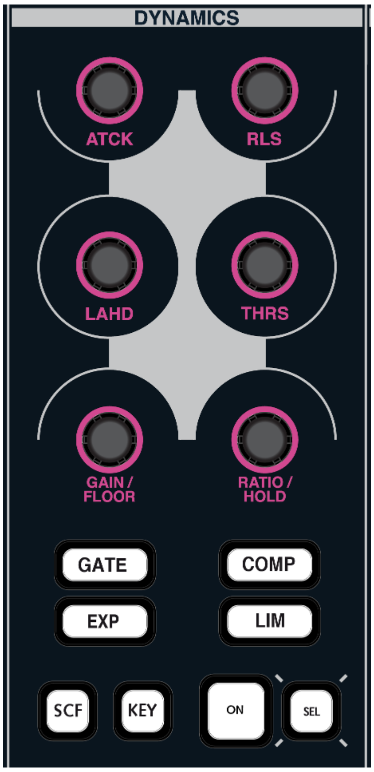




The Compressor parameters may be set as follows:

- Threshold Level – from -70dB to +20dB (must be at least 10dB higher than the Expander Threshold.)
- Ratio – from 1:1 to 10:1.
- Attack Time – from 0.29ms to 250ms.
- Release Time – from 40ms to 10s.
- Look Ahead Delay – from 0ms to 10ms (look ahead delay affects all three Combi-Dynamics modules).
- Gain - from -20dB to +20dB.
- Knee - hard or soft. This parameter is set from the **Main Display**. Use the trackball to set the **KNEE** option to either **hard** or **soft**.

SCF and KEY



Broadcast channels do not support sidechain filtering or external key inputs. Therefore:

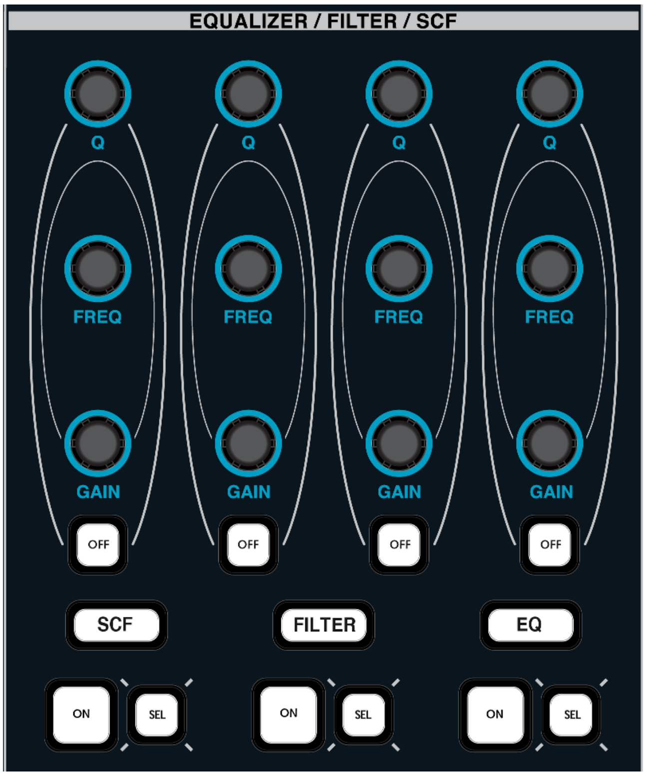
- The **SCF** key cannot be selected to switch filters into the dynamics sidechain.
- The **KEY** button cannot be used to turn on an external key input. You will find the eight external key inputs remain within the **Signal List**, but they cannot be assigned to a dynamics module.

EQUALIZER/FILTER/SCF

Recording channels provide a 4-band equaliser (EQ) plus two 2-band high and low pass filter modules; one dedicated to the main channel (FILTER) and one dedicated to the dynamics sidechain (SCF).

Broadcast channels provide a single 4-band equaliser (EQ), and do not support separate filter or sidechain filter modules. However, the upper and lower bands of the equaliser can operate as a filter, shelf or parametric EQ.

The modules may be arranged in any order within the channel signal flow and are controlled from the EQUALIZER/FILTER/SCF control area:



Four sets of dedicated **GAIN**, **FREQ** and **Q** controls, with band **OFF** buttons are provided.

1. Switch the four sets of controls between sidechain filters, main channel filters and the 4-band equaliser using the **SCF**, **FILTER** and **EQ** buttons at the bottom of the panel:

Note that on Broadcast channels, you cannot select **SCF** or **FILTER**, as these DSP modules are not supported.

2. Press the **ON** buttons to turn each section on or off.

3. Now adjust the **GAIN**, **FREQ** and **Q** settings.

*The **Main Display** provides feedback on your parameter values. You can view the EQ, (and FILTER or SCF modules on Recording channels):*



All 4-bands of EQ (and 2-bands of filters on Recording channels) operate across the full frequency range (20Hz to 20kHz), and offer a variety of different EQ types. The frequency for each band is marked by a vertical line labeled 1, 2, 3 and 4 to show which band is acting at a particular frequency.

4. Press **OFF** to switch any individual band out of circuit.

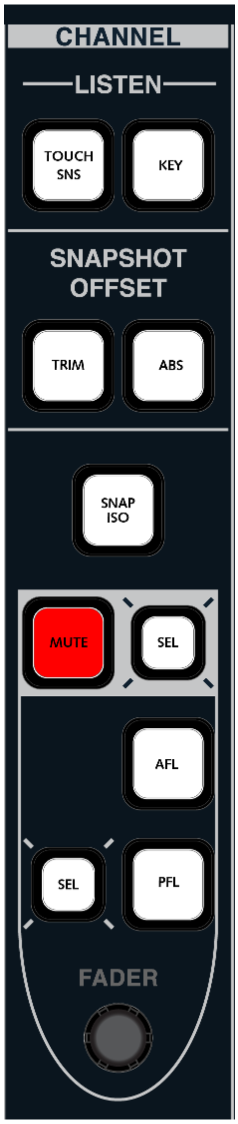
5. Click on the EQ type touch-screen menu buttons to switch between bell, shelf and pass band filters for the high and low bands, and bell, constant Q and notch for the middle bands:

The filter and shelf parameters vary slightly between Recording and Broadcast channels:

Recording channels	Broadcast channels
Max. 3rd order filter	Max. 2nd order filter
Max. 18dB/octave shelf	Max. 12dB/octave shelf

Note that if you load a Recording channel EQ setting to a Broadcast channel (e.g. using a Preset), and the stored parameter lies outside the range supported by Broadcast channels, then the closest available value is applied. For example, if the preset is attempting to load a 3rd order filter, then a 2nd order filter (the maximum) is applied.

CHANNEL Buttons



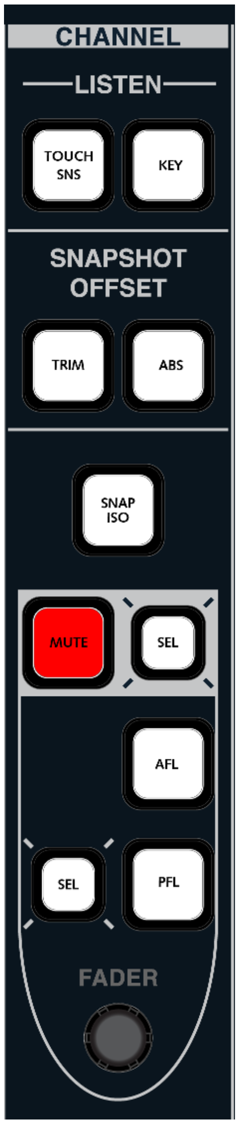
The CHANNEL section includes:

- LISTEN - these buttons can be used to AFL an individual processing module, see LISTEN controls.
- SNAPSHOT OFFSET - these buttons can be used used to select parameters for snapshot trim sets.
- **SNAP ISO** - enable this button to isolate the channel from a snapshot load, see Snapshot Isolate.
- FADER, **MUTE**, **AFL** and **PFL** - these controls duplicate the fader level, **MUTE**, **AFL** and **PFL** on the fader strip. Independent SEL buttons are provided for the **MUTE** and FADER level, so that they may be selected independently (for copy/reset, link groups, etc.) You will also see the FADER level and **MUTE** status on the **Main Display**:

LISTEN Controls

The **TOUCH SNS** and **KEY** buttons are used to provide AFL monitoring for individual audio modules within the Central Control Section.

Note that the Key mode in the **System Settings** display sets latching or momentary operation.



➢ **TOUCH SENSE**

This function can be used to turn each module's **SEL** button into its own AFL enable. For example to AFL the channel post EQ:

1. Press **TOUCH SNS**.

The button flashes.

2. Now touch an Equaliser control.

*You are now monitoring the output of the Equaliser section on your AFL bus; the EQUALISER **SEL** button illuminates to indicate this.*

If the Key mode is sensing, the AFL automatically cancels when you stop touching the control.

3. Alternatively, if the Key mode is latching, deselect **TOUCH SNS**, press **ESC** or press **CLEAR AFL/PFL** to cancel AFL monitoring.

Note that you may listen to any audio module within the Central Control Section except DIG AMP and DELAY. You cannot listen to a module, if AFL on a fader is already selected!

➢ **KEY (available on Recording Channels only)**

This function allows you to AFL the dynamics sidechain on a Recording channel. For example, to audition sidechain filtering applied to a compressor or AFL a channel pre the dynamics section:

1. Select **KEY** from the LISTEN buttons.

The button flashes.

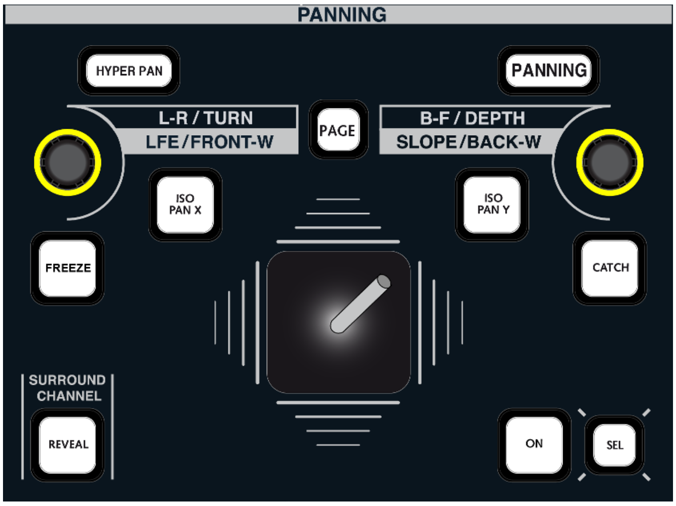
2. Touch any of the compressor controls.

*You are now monitoring the compressor sidechain on your AFL bus; the Sidechain Filter **SEL** button illuminates to indicate this.*

If the Key mode is sensing, the AFL automatically cancels when you stop touching the control.

3. Alternatively, if the Key mode is latching, deselect **TOUCH SNS**, press **ESC** or press **CLEAR AFL/PFL** to cancel AFL monitoring.

PANNING



The PANNING controls provide stereo or surround panning onto Group, Sum, Aux and Track busses assigned from the channel. Whether stereo or surround panning is applied depends on the format of the bus masters.

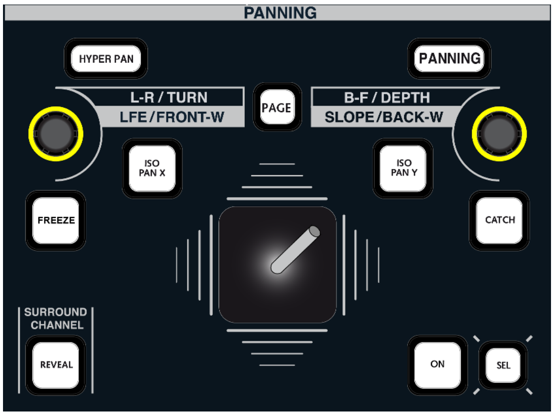
The controls operate in one of two modes:

- PANNING – conventional X/Y pan positioning.
- HYPER PAN – a new mode, ideal for adjusting the position of a surround source within a surround field.

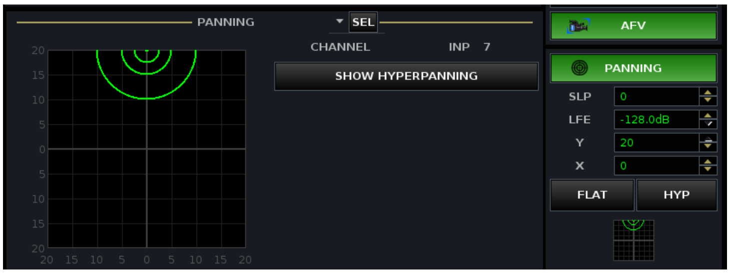
The panel also houses the SURROUND CHANNEL **REVEAL** button. See Revealing Surround/VCA Slaves for details.

X/Y Panning

Central Control Section



Main Display



1. Switch the panning section into circuit by pressing **ON**.

> The channel must be assigned to either a stereo or surround bus for panning to be active. If your channel is *only* assigned to a mono bus, then panning cannot be turned on.

*The status of the PANNING module is always shown on the **Main Display**. You can also view a graphical representation of the pan position by assigning PANNING to one of the central display areas.*

Note that if you are using the FREEZE function to lock the joystick to a particular channel, then the Central Control channel and the panning channel may be different. Therefore, the PANNING CHANNEL name is always shown on the **Main Display** – in our example, **INP 7**.

2. To pan in conventional X/Y mode, panning must be enabled with Hyper Pan off.

Either press the **HYPER PAN** button and check that the **ON** button is off. Then return to X/Y pan mode by pressing **PANNING**.

Or, on the **Main Display**, check that **PANNING** is on (green) and **HYP** is off (grey).

➢ **Panning onto a stereo bus:**

3. Check that the **PAGE** button is off and use the **L-R** rotary control to adjust the left/right pan position. (Or left/right balance if panning from a stereo channel).

➢ **Panning onto a surround bus:**

4. Use the **L-R** control to adjust the left/centre/right pan position.

5. Use the **B-F** control to adjust the Back to Front pan position.

6. Turn on the **PAGE** button and use the **LFE** control to adjust the level to the Low Frequency Effect channel.

The LFE level may be set from -128dB to +15dB.

7. Use the **SLOPE** control to adjust the slope.

Slope

The **SLOPE** adjusts signals feeding to the discrete centre channel within the surround field, and may be used to adjust the balance between discrete and phantom centre signals.

To see the effect, position your channel to front centre, and adjust the **SLOPE** control to a setting of +20; the signal feeds only the discrete centre channel. Move the **SLOPE** control anticlockwise to a setting of -20; the signal now feeds only the left and right channels (phantom centre). Please refer to the Appendix: Pan Slope for more details on the effect of the **SLOPE** control.

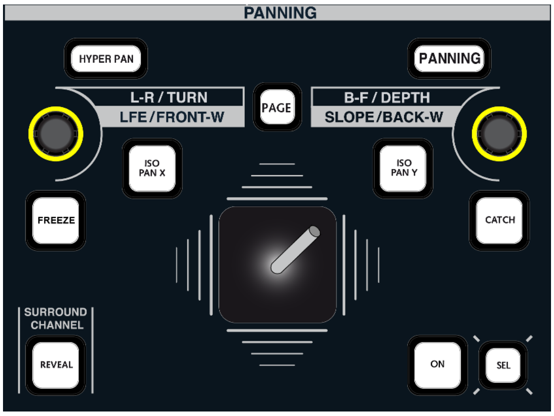
Flat

The level of signal feeding the centre channel is also affected by the selection of the **FLAT** button. You can change this from the **Main Display** or Extra Buttons display.

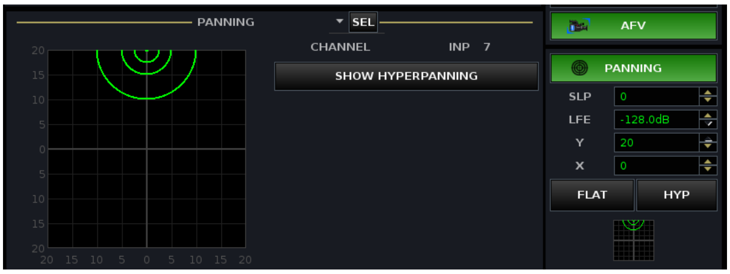
The default (**FLAT** off) is to apply level compensation as you pan across left, centre, right channels. Select **FLAT** on if you wish the level feeding the centre channel to remain constant as you pan across LCR.

The Joystick

Central Control Section



Main Display



In X/Y PANNING mode, the joystick provides another method of controlling the channel's XY pan position. The joystick follows the channel in access unless you use the **FREEZE** function.

➢ Freezing the Joystick

1. First make sure that the **FREEZE** button is deselected.
2. Update the channel in access, either by pressing a **SEL** button on a fader strip or using the ACCESS CHANNEL/ASSIGN panel.

   The channel is assigned to the Central Control Section (and the joystick).
3. Select **FREEZE** to lock the assignment.

The joystick now remains 'locked' to the assigned channel until you deselect **FREEZE**.

Note that when **FREEZE** is active, the channel in access may be different to that assigned to the joystick. Therefore, the channel name is always shown in the PANNING module on the **Main Display** – in our example, **INP 7**.

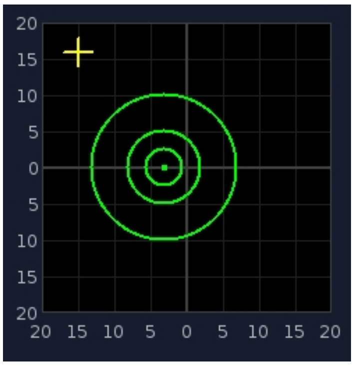
➢ Using the Joystick

Once assigned, you can use the joystick to control the channel's X/Y pan position:

1. Make sure that the panning section is switched into circuit by pressing **ON**.

- If the joystick is motorised, then it will automatically move to the current pan position of the channel, and the **CATCH** button will light. Move the joystick to change the pan position.
- If the joystick is not motorised, then it may be in a different position to the current pan position. You can either:
  - o Press **CATCH** to change the pan position to the position of the joystick.
  - o Or, move the joystick to 'catch' the channel; nothing happens until you move the joystick through the current pan position. At this point, the **CATCH** button lights to indicate that you now have control and are changing the panning.

If the joystick position is different from the current pan position, then this is indicated within the **Main Display**. A yellow cross indicates the joystick position. Once you move the joystick through the current pan position and 'catch' the channel, the yellow cross disappears:



2. Move the joystick left or right to control the Left-Right pan position, or left/right balance if panning from a stereo channel.
3. Move the joystick up or down to control Front-Back (Y-axis).

In either of these two modes, you can restrict the joystick to provide more control for a particular axis by selecting:

- **X** - this isolates the X-axis so that any left-centre-right movements are ignored.
- **Y** - this isolates the Y-axis so that any up/down (Front-Back) movements are ignored.

You may customise the feel of the joystick or add a notch at a particular position (e.g. Front Center) using the Fader/Joystick options in the **System Settings** display.

The joystick motors are enabled or disabled from the Joystick Motor option in the **System Settings** display and saved within the production.

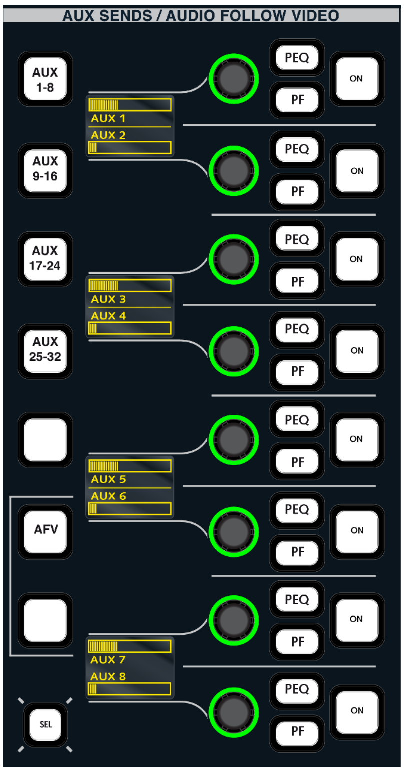
Note that the joystick motors cannot be enabled on US systems.

Hyper Pan

HYPER PAN is an alternate mode of panning ideal for positioning a surround source within a surround field. It can be used on mono, stereo or surround channels but is designed with surround sources in mind. Therefore, this topic is covered later in the manual, see Surround Channels: Hyper Pan.

AUX SENDS

Each input, monitor or group channel may access up to 32 auxiliary sends. These are paged onto the eight rotary controls as follows:



1. Press **AUX 1..8** to assign the first eight auxiliary sends.

The name of the send (e.g. AUX 1 to AUX 8) appears in the alphanumeric display.

2. Press the **ON** button to activate the send.

The AUX bus assign boxes on the Channel display update to reflect your assignments:

3. Use the rotary control to adjust the send level.

The send level may be adjusted from -128dB to +15dB.

4. The send level defaults to be post fader. Press **PF** to switch the send pre fader or **PEQ** (Recording channels only) to switch to pre EQ.

The bus assign boxes are colour coded to reflect the different assignments:

- Post-fader – white writing on green (e.g. Aux 3).
- Pre-fader – black writing on white/green (e.g. Aux 5).
- Pre-EQ – white writing on green/white (e.g. Aux 8).

5. Press the **AUX 9..16**, **AUX 17..24** or **AUX 25..32** buttons to access the remaining auxiliary sends for the channel.

6. The SEL button is used to select the aux sends, in groups of 8, for operations such as copy or reset, channel linking, etc.

Note that the aux send options vary slightly between Recording and Broadcast channels:

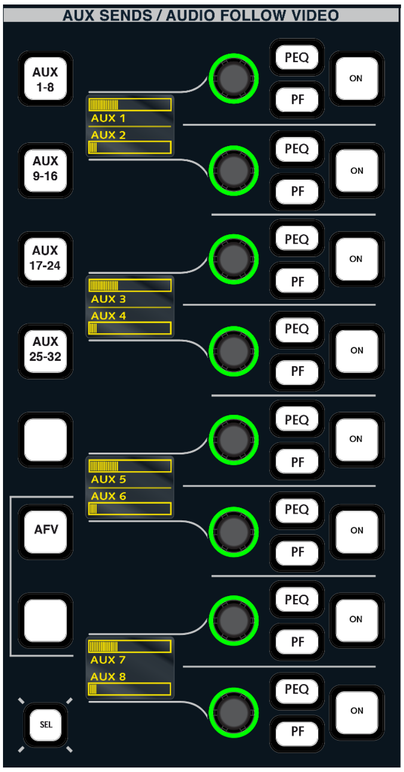
Aux Send	Recording channels	Broadcast channels
Pre EQ	✓	×
Pre Fader	✓	✓
Post Fader	✓ (pre-bus)	✓ (after fader)

On Recording channels the pre EQ option follows any changes made to the position of EQ in the channel signal flow. This allows you to move the aux send to virtually any channel pickup position.

On Broadcast channels, the aux post fader send is a real post fader send, and not pre-bus as in a Recording channel. This means that you can position another module, for example delay, after the fader, and the delay will affect the main busses, but not the post fade aux send. See Changing the Signal Processing Order.

Stereo & Surround Auxiliary Sends

Any odd/even pair of mono sends may be linked for stereo operation. Or, you can create surround sends (up to 8-channel) from Auxes 1-8, 9-16, 17-24 or 25-32. This is handled in the same way as creating any other stereo or surround channel, see Stereo Channels and Surround Channels.



➢ **When an Aux is Stereo:**

1. Press either of the linked Aux **ON** buttons (e.g. on Aux 1 or Aux 2) to activate the send.
2. Use the upper rotary control to set the level (Gain) of the aux send.
3. Use the lower rotary control to adjust the pan position, or stereo balance if routed from a stereo input channel.

By default, the pan (balance) onto a stereo aux send is linked to the channel XY pan position. You can disable this using the Delta Panpot to Aux Sends option in the **System Settings** display.

➢ **When an Aux is Surround:**

1. Press any of the linked Aux **ON** buttons (e.g. on Aux 1 to 6) to activate the send.

   All Aux send channels are enabled.

Note that you can deselect individual **ON** buttons to switch off particular elements of the surround send – for example, to turn off the send to the LFE channel.

2. Use the first Aux (e.g. Aux 1) to adjust the level of *all* surround Aux sends.

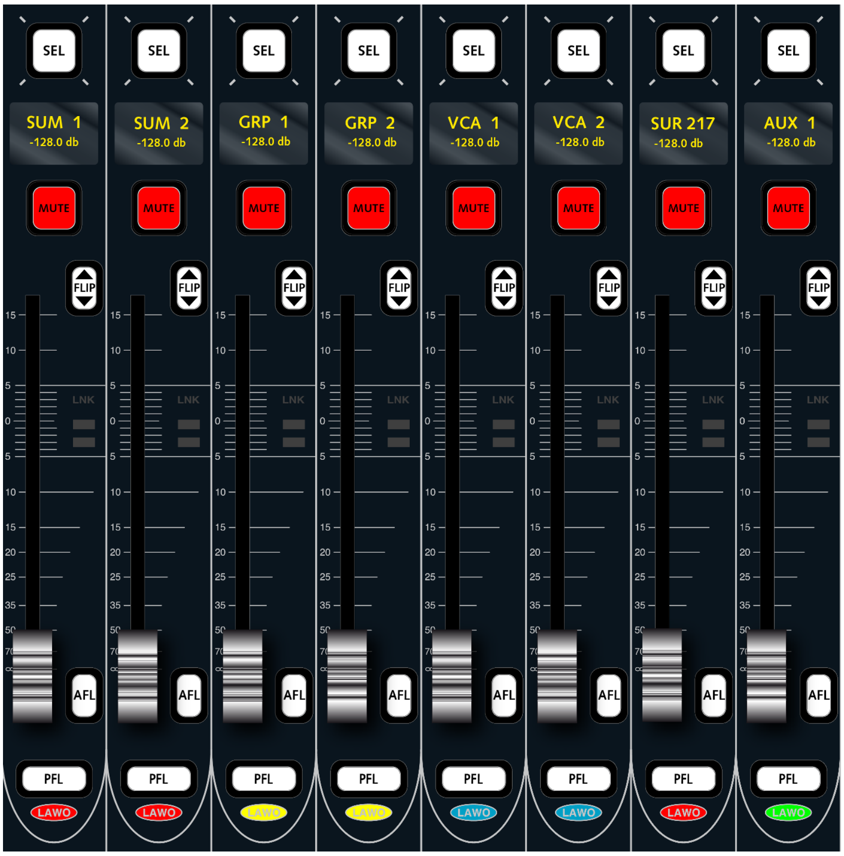
Note that the panning onto a surround aux send is *always* linked to the channel pan position.

Aux Masters

Each Aux channel may be assigned onto any fader strip, in the same way as you would assign an input channel to a fader, and may provide full or tiny signal processing depending on your DSP configuration.

To control your Aux masters:

1. Assign AUX 1 to 32 to fader strips in the usual manner, see Fader Strip Assignment.



If you assign all Aux masters 1 to 32 to a lower fader bank (e.g. Bank 6), you can get instant access using Bank Switching.

2. Control the level, mute, AFL and PFL and assign free controls as you would for an input channel, see The Channel Fader Strip.

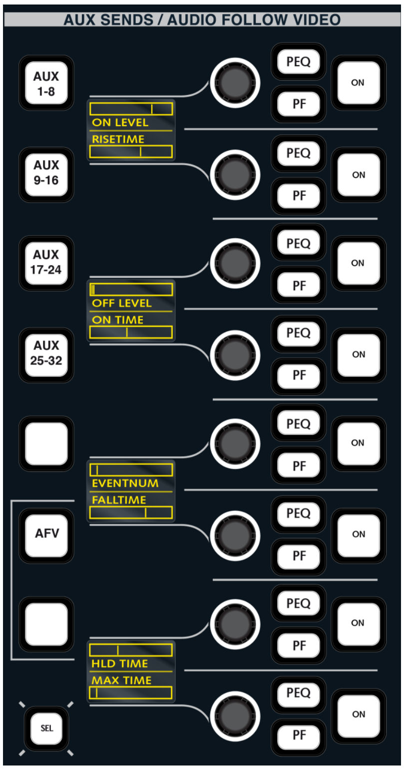
3. Apply signal processing, if available, using the Central Control Section.

AUDIO FOLLOW VIDEO

The **mc²56**'s Audio Follow Video provides the ability to open and close a channel or main fader from an external event, received via TCP/IP Ethernet (using Lawo's Remote MNOPL protocol) or GPIO. For example, during coverage of a live motor racing event, you may programme the audio channels associated with each camera to automatically open and close as the picture cuts between different shots.

Up to 128 events may be programmed, with each event corresponding to a different camera tally. An event can control an individual channel or a group of channels. Parameters for the Hold Time, Rise Time, Max Event Time, On Time and Fall Time control the envelope of the fade allowing smooth fades from one camera to another.

To configure the AFV parameters:

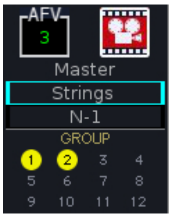


1. Press **AFV** on the AUX SENDS/AUDIO FOLLOW VIDEO panel.

   The eight rotary controls switch to AFV functions as indicated on the alphanumeric displays.

2. Turn the **EVENTNUM** control to assign the external event. Active events are numbered from 1 to 128. Select 0 for no event.

   *The event number appears beside the control and on the **Main** and **Channel** displays:*

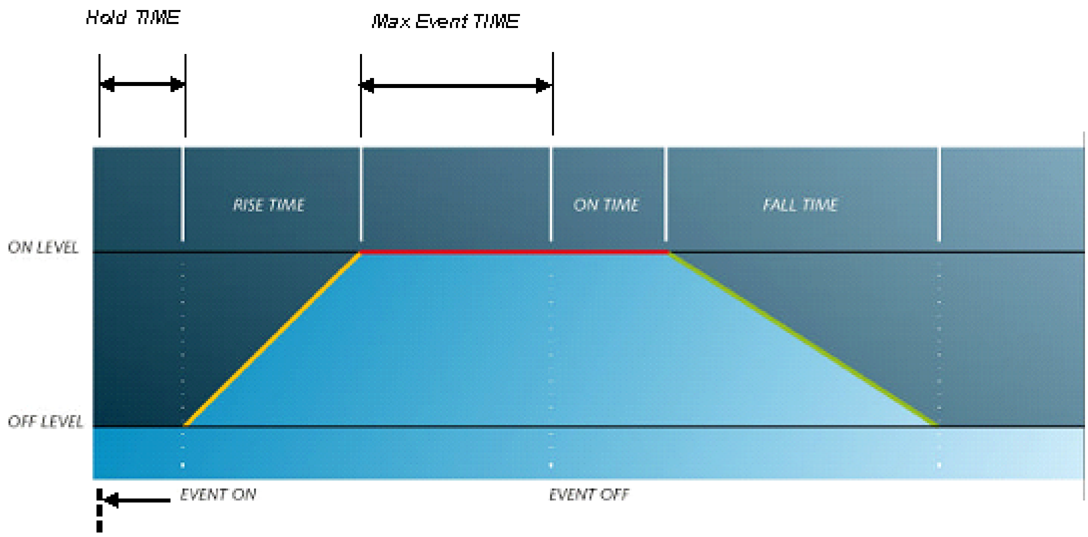


3. Enable AFV by selecting the **ON** button beside the **ON LEVEL** control.

4. To action the event locally from the console, press the **ON** button beside the **EVENTNUM** control.

The fader opens (and closes) according to the AFV parameters.

5. Use the remaining rotary controls to adjust the AFV behaviour:



- **ON LEVEL** – the level which the fader opens to when the event if switched on.
- **OFF LEVEL** - the level which the fader closes to when the event is switched off.
- **HOLD TIME** – use this control to delay the opening of the fader after the event on trigger.
- **RISE TIME** – the time taken for the fader to move from off to on level after the hold time has expired.
- **MAX TIME** – the maximum amount of time the fader stays open (if no event off).
- **ON TIME** – the amount of time the fader stays at the on level, after the event is switched off.
- **FALL TIME** - the time taken for the fader to move from on to off level.

6. Once you have connected and configured the external controller, sit back and watch your fader(s) open and close from the AFV event controller.

*The **Main Display** shows the progress of the AFV event: the line for rise time is red as the fader rises; the line for on time is red while the event is on; the line for fall time is red as the fader falls.*

You may override a fader at any time, for example, to adjust the level of an open camera mic.

In addition, if you touch the fader while the camera is cut (AFV switched off), the fader remains open. This allows you to perform a manual fade out.

You may link as many channel or main faders to external events as you wish, providing flexible AFV control for individual and groups of faders.

If channels are grouped to a VCA master which is controlled by Audio Follow Video, then the slave faders will be automated by the AFV master event. If you do not wish this to occur, deselect the AFV enable (**ON** button beside the **On Level** control) on the slave channels.

7. To set Audio Follow Video either on or off for all channels, use the All AFV on/off option in the **System Settings** display.

Mix Minus (N-1) Sends

The **mc²56** may use any of its 32 auxiliary sends (or 96 track busses in Recording Channels mode) to create mix minus feeds. There are several applications including an N-1 send back to a commentator; an N-many send to multiple guests in a studio; or a pre-talk conference send where participants can chat off-air prior to their on-air contribution.

The only difference between using an aux and a track bus is that track busses do not offer a send control. If you need to adjust the mix minus send level from a channel, then you can create a track bus send level by moving the DIGAMP module into the track bus path (from the Channel Config display).

Any number of aux sends and/or track busses may be configured for mix minus operation, and controlled directly from the fader strip by programming the **CORD** and **CONF** functions onto the fader strip user buttons.

Broadcast Channel DSP configurations do not support track busses. Therefore, you *MUST* use auxiliary busses for mix minus sends when running in this mode.

To create a mix minus from tiny channels, you must enable the Tiny Channels for Conference option in the **System Settings** display.

When using a Recording Channel DSP configuration, and creating a mix minus from tiny channels, you *MUST* use auxiliary busses for mix minus sends (as track bus conference facilities are not supported from tiny DSP channels).

Assigning the Mix Minus Busses

The first step is to assign a mix minus bus to each source requiring a mix minus send.

To generate N-1 sends, you should assign a different bus to each source. To generate an N-many send, assign the same bus to multiple sources.

> The mix minus bus assignment is linked to the source routed to a channel (and not the DSP channel itself). This means that if you route the source to a different channel, the mix minus bus and its controls follow.

There are two ways to assign a mix minus bus to a source: from the **Channel** display touch-screen or from the Signal Settings display. Here we will use the **Channel** display, as this is the quickest method to assign an aux from 1 to 16. To assign auxes 17 to 32 or track busses you must use the Signal Settings display.

Let's assume you have three microphone sources, each requiring an N-1 feed. The mic sources should be routed to three input channels and the input channels assigned to some fader strips.

To assign a mix minus bus to each source:

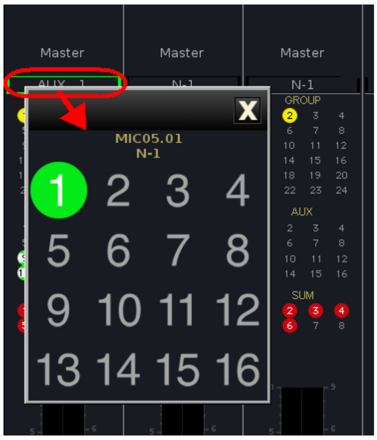
1. Touch the **N-1** text at the top of the fader strip's **Channel** display:

   An expanded pop-up window appears.

2. Touch a number to assign an aux as the N-1 bus for the source (the selection turns green).

3. Repeat for each source.

4. To close the pop-up, either touch the **X** in the top right corner, or touch twice in quick succession anywhere else on the display.

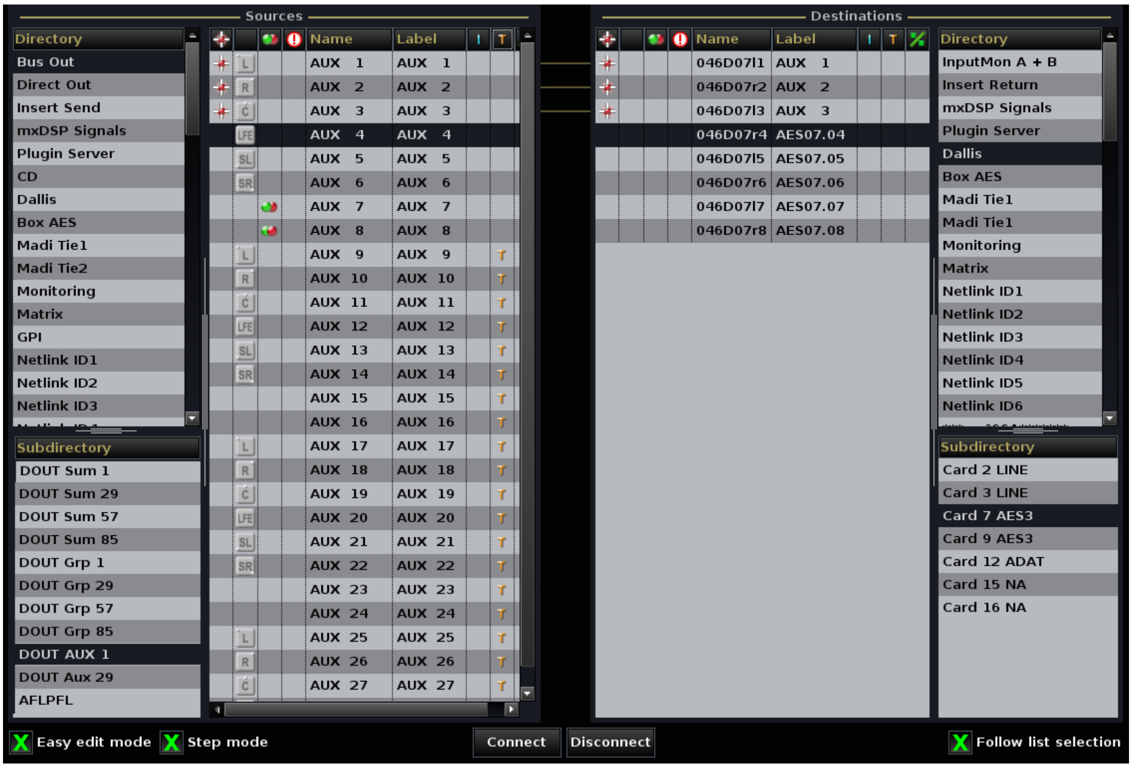


The mix minus bus names (e.g. **AUX 1**, **AUX 2**, **AUX 3**) are shown in the **N-1** field at the top of the **Channel** display. This provides feedback on which aux (or track bus) is assigned as the N-1 bus for each source/fader strip:

Routing the Mix Minus to its Destination

Next, route each mix minus bus back to its destination - for example, route Aux 1 to the earpiece for Mic 1, etc.

You can make these routes either from the Signal List or mx Routing displays. When using the **Signal List** display, you will find the Aux outputs under the **Bus Out** Source Directory:

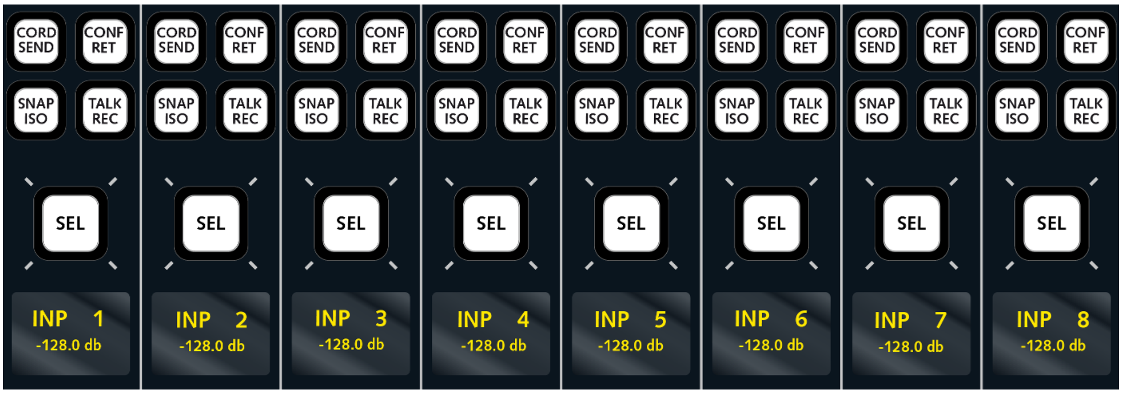


Controlling the Mix Minus Sends

Having assigned a mix minus bus to each source, you can now activate and control the sends from the fader strip using the **CONF** buttons.

To activate a mix minus, the **CONF** function *MUST* be programmed onto a fader strip user button from the Custom Functions display.

1. Go to the fader strips controlling each source and press the **CONF** buttons on all three channels:



The mix minus is automatically activated for each of the three channels; you can see this reflected in the bus routing on the ***Channel*** *display. For example, fader strip 1 (mic 1) is assigned to all mix minus busses except its own (Aux 2 & 3); fader strip 2 (mic 2) is assigned all all mix minus busses except its own (Aux 1 & 3); fader strip 3 (mic 3) is assigned all all mix minus busses except its own (Aux 1 & 2):*

The automatic assignments onto each aux are made with a send level of 0dB. You can adjust the individual channel send levels by assigning channels to the Central Control Section and using the Aux send level controls.

2. To add channels not within the conference group to the mix minus, press their **CONF** buttons.

The channels are routed onto all mix minus sends (e.g. Aux 1, 2 & 3).

3. To control the output level or AFL/PFL a mix minus send, assign the Aux master channels onto fader strips and use the fader, **AFL** and **PFL**.

4. To meter the mix minus sends, look at the Channel display metering on the Aux master channels. Or, assign the Aux masters to the Metering display.

5. To talk to the mix minus send, press the **TALK** user button. (Note that this function *must* be programmed from the Custom Functions display.)

> If you activate **CONF** or **CORD** on a stereo channel, then Left+Right feeds the mix minus bus.
>
> If you activate **CONF** or **CORD** on a surround VCA channel then you can choose which of the surround slaves will feed the mix minus bus from the **System Settings** display (see the Surround Mix Minus options).

Conference (Pre-Talk) Mix Minus Sends

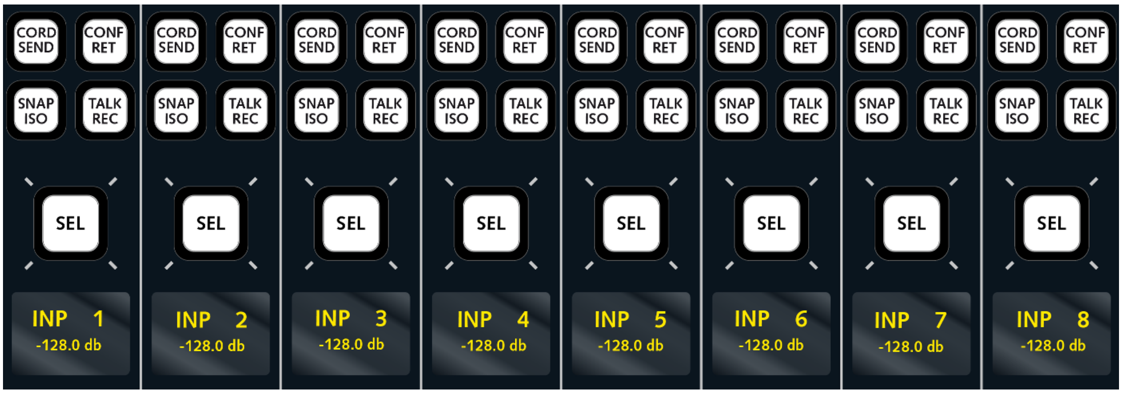
The **CORD** button changes the mix minus from an N-1 into a pre-talk conference send.

This is a great facility for enabling guests and presenters to talk to each other while off-air. As long as their channel fader is closed, each conference bus receives a pre-fader mix of all **CORD** contributors minus themselves. As soon as the channel fader is opened, and they are on-air, the pre-fader bus reverts to a post fader mix minus.

To use this function, both **CONF** and **CORD** functions *MUST* be programme onto the fader strip user buttons either from the Custom Functions display or the factory configuration. Please refer to your system specification for details.

1. Go to the fader strips controlling each source and press the **CONF** and **CORD** buttons on the channels you wish to act a pre-talk sends.

Note you will only be able to active **CORD** if the channel fader is closed.



You can select a mixture of buttons across channels to configure pre-talk sends (**CONF** plus **CORD**) for some presenters and post-fader mix minus sends (**CONF** only) for others. For example, your guest in New York may wish to talk to the studio presenters, but not to the guest in Australia!

Stereo Channels

Any odd/even pair of input or output channels may be configured for stereo and controlled from a single fader strip.

The same procedure may be used on input, monitor, group, sum or aux DSP channels, allowing you to create stereo input channels and output masters. (To create a stereo track bus, link the corresponding monitor channels).

The operation of a stereo channel is identical to that of a mono channel, with the following additional features:

- Stereo Balance and Input Control
- Image Width and Positioning
- Panning from a Stereo Channel

All other processing (EQ, Dynamics, Delay, etc.) is applied equally to both left and right sides.

Additional notes for stereo output channels:

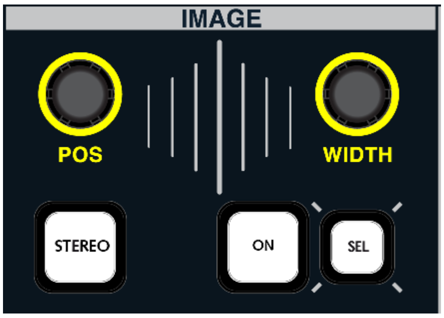
- Any odd/even pair of output channels (group, sum or aux) may be configured for 2-Channel mode, as an alternative to stereo. This provides independent fader strip control for the left and right sides of the output master.
- If you create a stereo output channel (group, sum or aux), then Forward and Reverse bus assign (from the front panel) automatically routes channels to both Left and Right in one operation. To assign a channel to the Right bus only, then you should use the Bus Assign display.

Creating a Stereo Channel

There are three ways to create a stereo channel. Here we will deal with the Central Control Section IMAGE panel. The other methods use the Signal List display and Channel Config display.

1. Assign the odd channel, of the odd/even pair, to the Central Control Section by pressing its fader strip **SEL** button or using the ACCESS CHANNEL/ASSIGN panel.

2. Locate the IMAGE controls on the Central Control Section:



3. Press the **STEREO** button.

*This links the selected channel to its adjacent DSP path. For example, pressing **STEREO** on INP 7 creates a stereo channel from INP 7 and INP 8.*

If the two mono channels used to create a stereo channel are present on the control surface, then the right hand channel (e.g. input 8) will disappear from the surface leaving a blank fader strip. If you unmake the stereo channel, and the blank fader strip is still available, then input 8 will return to the active surface. However, if you have assigned another channel to its old position, you will need to reassign input 8 to a different fader strip location.

To indicate that the channel is stereo, you will see stereo metering and the stereo red/green circles on the Channel display.

4. To change a stereo channel back to mono, deselect the **STEREO** button on the IMAGE panel.

The channel metering reverts to a mono bargraph.

When a stereo channel is created, settings from the left channel are copied to the right and the two sides are automatically panned left and right for stereo operation. This means that if the stereo link is removed, the resulting left and right mono channels have identical settings and are panned centre.

Stereo Balance & Input Control

When an input channel is stereo, a number of additional controls become available: **BALANCE**, **Ø R** and stereo input management.

Note that **GAIN**, **BALANCE** and **Ø** may be applied to the source or channel depending on the SOURCE/INMIX mode.

1. With the INPUT panel switched to **SOURCE** mode (the default), use the **GAIN** control to adjust source gain - the gain range depends on the type of input (mic/line or analogue fixed gain/digital).

The gain for left and right inputs is adjusted in parallel; any offsets are retained and represented by a positive or negative ***BALANCE*** *value.*

To adjust source gain independently for the left and right inputs, you can use the Mic/Line Gain or I/O DSP Volume parameters from the **Signal Settings** display.

2. Use the **BALANCE** control to set the Left/Right input balance for the stereo input.

3. Press the **MONO** button to sum the Left and Right inputs.

4. Press the button to reverse the Left and Right inputs.

5. Press the **Ø L** or **Ø R** buttons to reverse the phase.

6. Press either **L>B** (Left to Both) or **R>B** (Right to Both) to route either the left or right source to both sides of the stereo channel.

7. Select **M/S** for sources recorded using sum and difference coding.

8. Switch the INPUT panel to **INMIX** mode if you wish to adjust the GAIN and BALANCE for the channel.

The status of all settings is indicated on the ***Main Display****.*

Image Width & Positioning

The IMAGE controls may be used to adjust the stereo width and positioning. Please see the IMAGE panel.

Panning from a Stereo Channel

The channel panning controls on a stereo channel behave in a very similar fashion to a mono channel, allowing you to pan a stereo channel in surround when routed to a surround bus destination. The only difference in control is that the L/R pan control now adjusts the left/right balance of the stereo channel.

Similarly, the X-axis movement of the joystick adjusts the left/right balance when panning a stereo channel.

See X/Y PANNING for details.

2-Channel Mode

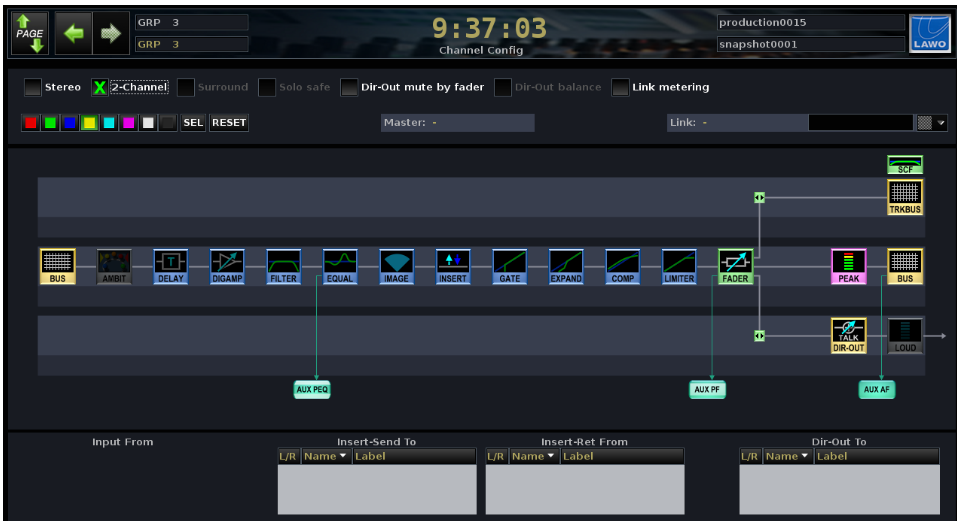
Any odd/even pair of sum, group or aux channels may be configured as 2-channel as an alternative to stereo. This provides independent fader strip control for the left and right sides of the output channel.

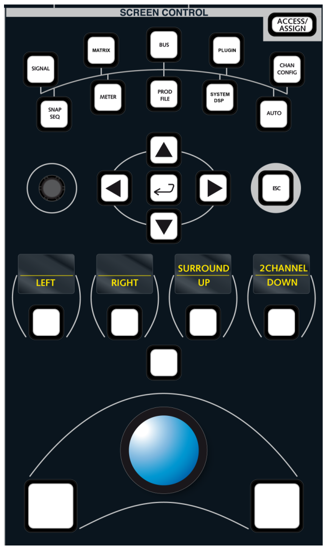
1. Assign the odd channel, of the odd/even pair, to the Central Control Section by pressing its fader strip **SEL** button or using the ACCESS CHANNEL/ASSIGN panel.

2. Press the **CHAN/CONFIG** button, located on the SCREEN CONTROL panel, to view the **Channel Config** display.

3. Press the **2-CHANNEL** soft key or select the **2-Channel** screen option.

This configures the selected channel and its adjacent DSP path for 2-Channel operation. In our example, GRP 1 and GRP 2:





If you now bus assign an input channel onto GRP 1, then the input will be assigned to GRP 1 and 2, and panned with a left/right pan law, just as for a stereo group.

The difference from stereo operation is that you can assign GRP 1 and GRP 2 independently to the console surface. This allows you to adjust the left and right sides of the 2-channel output independently.

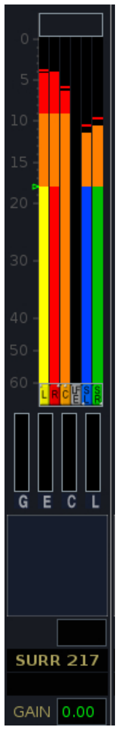
4. To undo the 2-channel configuration, put GRP 1 back into access and deselect the **2-CHANNEL** soft key.

Surround Channels

Multiple input or output channels may be configured for surround.

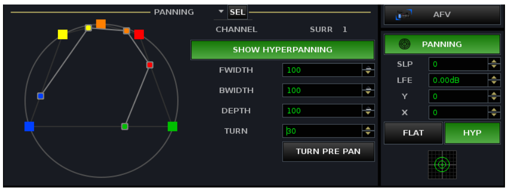
> The same procedure may be used on input, monitor, group, sum or aux DSP channels, allowing you to create surround input channels and output masters. (To create a surround track bus, link the corresponding monitor channels).

A variety of multi-channel surround formats are supported up to 7.1. The surround format is set globally for each production from the **System Settings** display. This defines the format used for surround channels, pan laws and monitoring. For example, if you select Dolby Digital 5.1, then component channels 1 to 6 are configured as L, R, C, LFE, Ls and Rs.



A range of specialised tools provide easy management of surround channels:

- **Surround VCAs** - provide master control of the surround signal from a single fader strip. You can control the overall level, EQ, compression, etc. while metering all slave channels independently on the **Channel** display (shown opposite).
- **REVEAL** - temporarily assigns the individual surround slaves onto fader strips (within a pre-defined area or onto the optional REVEAL fader panel). This enables you to quickly offset fader levels and other relative parameters.
- **Hyper Panning** - provides an alternative to conventional XY panning. It is designed to help reposition surround sources within a surround field. For example, to turn a 5.1 source:



- **AMBIT** (AMBience IT) - is a special DSP module designed for upmix or spatialise processing:
  - o **Upmix** – a 2 in, 6 out upmixer which, using sophisticated algorithms, converts stereo signals into 5.1 surround.
  - o **Spatialise Only** – a 6 in, 6 out spatialiser which processes the surround left and right channels only, ideal for treating incoming 5.1 signals.

Additional notes for surround output channels:

- If you create a surround output channel (group, sum or aux), then Forward and Reverse bus assign (from the front panel) automatically routes channels onto all components of the surround bus in one operation. When you *deselect* bus assignments, they are deselected *one by one* allowing you to edit the routing. See Bus Assignments to a Surround Output.

Topics covered in this section are:

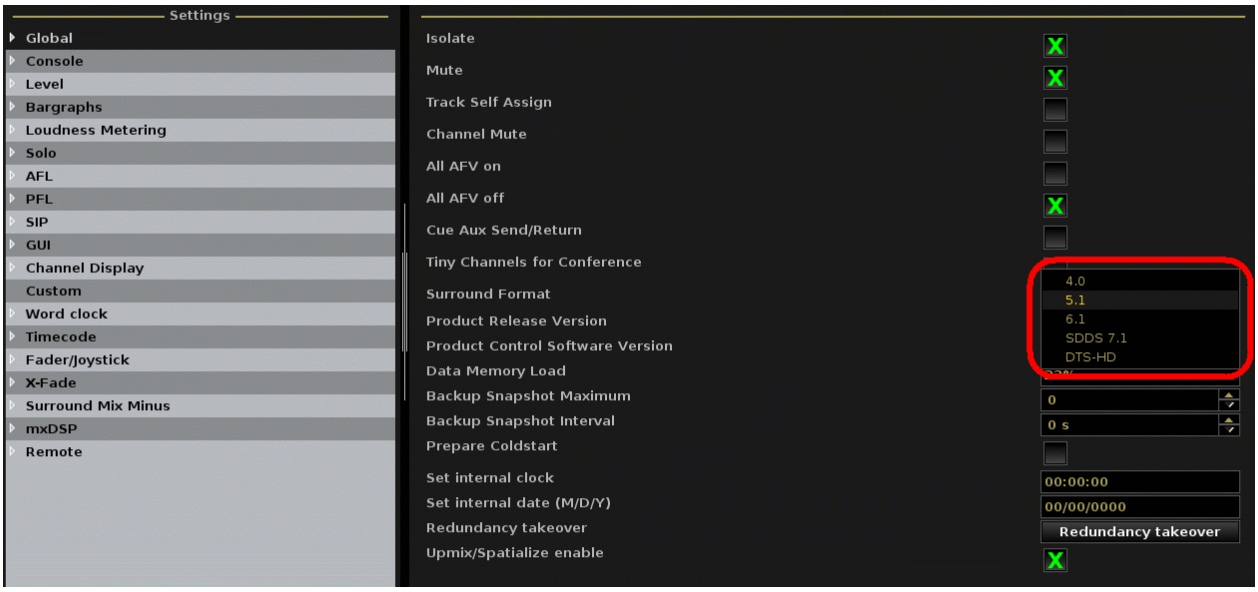
- Defining the Global Surround Format
- Creating a Surround Channel
- Bus assignments to a Surround Output
- Monitoring in Surround
- Surround VCAs – master control of a surround channel.
- Revealing the Surround/VCA Slaves
- Hyper Pan - ideal for positioning a surround source within the surround field.
- AMBIT - Upmix and Spatialise processing.

Defining the Global Surround Format

This option defines the global surround format used for surround channels, pan laws and monitoring. For example, if you select Dolby Digital 5.1, then component channels 1 to 6 are configured as L, R, C, LFE, Ls and Rs.

To select the global surround format:

1. Press the **SYSTEM DSP** button, located on the SCREEN CONTROL panel, to view the System Settings display.
2. And navigate to the **Global** topic.
3. Select the **Surround Format** option, and use the drop-down menu to make your selection:



- **4.0** – L, R, C, S for Dolby ProLogic.
- **5.1** – L, R, C, LFE, Ls, Rs for Dolby Digital and DTS.
- **6.1** – L, R, C, LFE, Ls, Rs, Cs for Dolby Digital EX and DTS ES.
- **SDDS** – L, R, Lc, Rc, C, LFE, Ls, Rs for 7.1 SDDS.
- **7.1** – L, R, C, LFE, Lm, Rm, Ls, Rs for DTS-HD.

Creating a Surround Channel

Surround channels are *always* created in 8-channel blocks, even if the surround format uses less channels. For example, to create a surround sum, the first component *MUST* be SUM 1, 9, 17, etc. If the format is Dolby Digital 5.1, then this creates a 6-channel surround channel as follows:

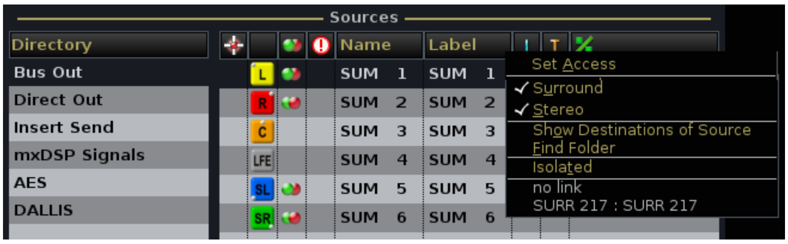
- Sum 1, 9, 17 = Front Left
- Sum 2, 10, 18 = Front Right
- Sum 3, 11, 19 = Front Centre
- Sum 4, 12, 20 = LFE
- Sum 5, 13, 21 = Surround Left
- Sum 6, 14, 22 = Surround Right
- Sums 7 & 8, 15 & 16, 23 & 24 – are free to be configured as mono or stereo.

It is a good idea to bear this in mind while choosing a DSP Configuration.
Note that you can configure channels as surround, even if they use tiny processing.

There are two ways to create a surround channel. Here we will deal with the Signal List display. See the Channel Config display for the alternative method.

➢ **To create a surround sum:**

1. Select the first sum for the surround output from the **Sources** list (e.g. **SUM 1**).
2. Press the **SURROUND** soft key, or right-click and select the **Surround** option:



*This links consecutive sums, according to the global surround format, and automatically assigns a Surround VCA - in our example, **SURR 217**.*

You can configure surround sums, groups or auxes using this method. Alternatively, select **InputMon** from the **Sources** list to configure surround input or monitor channels.

For surround inputs, panning is automatically reset so that INP 9 feeds SUM 1, INP 10 feeds SUM 2, etc. The best way to position a surround channel within the surround field is using Hyper Pan.

Surround channels may only be created in 8-channel blocks, so you must select Sum 1, 9, 17, etc. You cannot select **Surround** if you right-click on an invalid channel number.

Note that the front and rear left/right pairs of a surround channel are automatically linked for stereo. This is for convenience when revealing the component channels. The stereo linking is only a default state; you can deselect the stereo link at any time.

Bus Assignments to a Surround Output

Having configured a surround output, bus assignments from your source channels can be made using any of the usual bus assign methods. However, there are some additional points relating to surround bus assignments.

When you assign a mono, stereo or surround channel onto a surround bus, using either Forward or Reverse bus assign, the console assumes that you wish to make the assignment onto all of the busses within the multi-channel output:

For example, let's say SUM 1 to 6 have been configured as a 5.1 surround output.

1. Assign an input channel to SUM 1 using either Forward or Reverse bus assign.

The console assigns the channel to SUMs 1, 2, 3, 4, 5 and 6 in one operation.

2. Having made the assignment, you can now edit it. So, for example, if you wish to remove the channel from SUM 4 (LFE), you can do so by deselecting the assignment to SUM 4.

To summarise:

- When you route *onto* a surround output, assignments are made onto *all* busses within the output.
- When you *deselect* routes from a surround output, they are deselected *one by one* allowing you to edit a surround assignment.

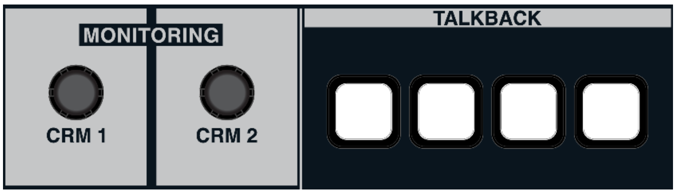
Note that if you use the Bus Assign display, then assignments are always made to surround component channels individually.

Monitoring in Surround

Control Room Monitor 1 (**CRM 1**) usually provides monitor source selection and level control in surround.

In our example, the **SUM 1-6** button on **PAGE 1** switches the 5.1 surround sum to the CRM 1 monitor output.

Use the **CRM 1** level control to adjust the monitor level:



Note that these functions are programmed by the factory configuration, so please refer to your system specification for details.

See also Control Room Monitoring.

Surround VCAs

Surround VCAs provide master control of a surround channel from a single fader strip.

A surround VCA is automatically designated each time you configure a surround channel. For example, Surround VCA 217 (**SUR 217**) provides master control of the first surround sum (SUM 1-x).

By assigning the surround VCA to a fader strip, you can control the overall level of the surround channel and adjust master parameters such as EQ, compression, AMBIT upmixing and spatialise, etc.

You will also be able to meter all the surround component channels on the Channel display:

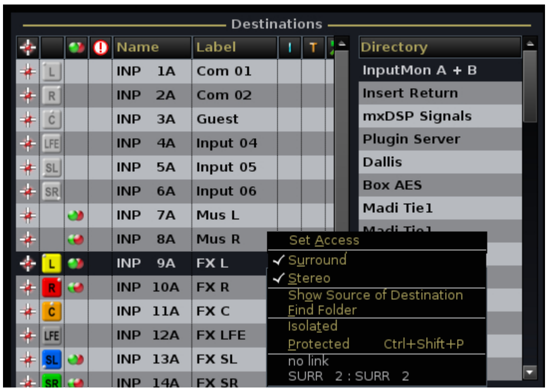
Interrogating the Surround VCA Number

Before you can assign a surround VCA to a fader strip, you will need to know its number. This is shown on the **Signal List** display and on the **Channel Config** display when you create a surround channel.

From the **Signal List** display:

1. Right-click on any surround component channel - in our example, on **INP 9**.

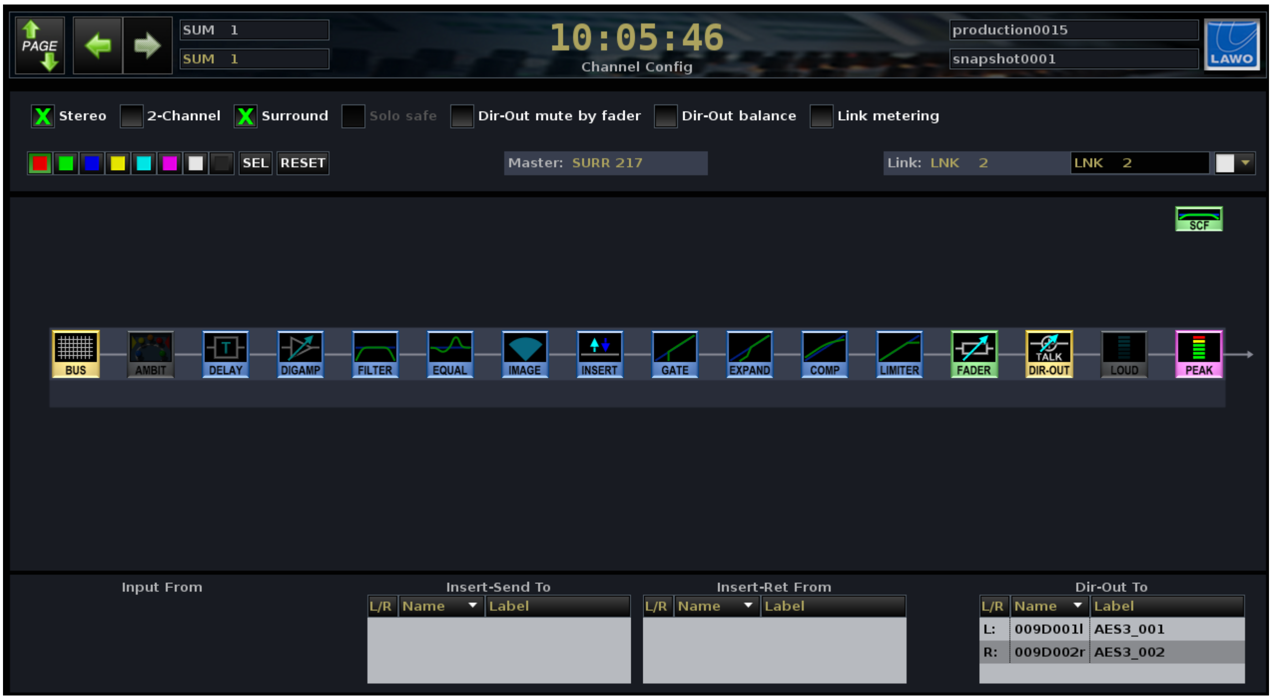
*The surround VCA number is shown at the bottom of the drop-down list - **SURR 2**:*



Or to use the **Channel Config** display:

1. Put one of the surround component channels into access, by pressing its fader strip **SEL** button or using the ACCESS CHANNEL/ASSIGN panel.

*The surround VCA number is shown in the **Master** field - in our example, **SURR 217**:*



The numbers used for surround VCAs are always the same on every mc² system. For example, **SURR 1** is *always* the master for INP 1-8; **SURR 193** is the master of GRP 1-8; **SURR 217** is the master of SUM 1-8, and so on.

Working with Surround VCAs

1. Use the surround VCA number (e.g. **SURR 217**) to assign it to a fader strip - channel or main. See Fader Strip assignment.

   The fader strip label updates, and you will see metering for the surround component channels on the Channel display.

2. You can now adjust the master level of the surround channel from the fader, and control master parameters from the Free Controls or Central Control Section.

The master/slave behaviour varies depending on the parameter. For example, main level and input gain are controlled relatively so that you can offset the slave positions; EQ frequency and Q are *always* set by the master (absolute), so that any change is inherited by all slaves; the MUTE is switched ON from a Surround VCA master but not OFF. For full details on all parameters, see the Appendices.

Renaming a Surround VCA

The user label of the surround VCA may be edited from the Title Bar when the surround VCA is in access.

Revealing the Surround/VCA Slaves

The **REVEAL** button provides a quick way to temporarily bring surround or VCA slaves onto fader strips so that you can offset fader levels and other relative parameters.

The faders used to "reveal" slaves may be any bay of fader strips or the optional Reveal Surround Fader user panel. When using the latter, note that you can *only* reveal surround VCAs (and not the slaves of a normal VCA group).

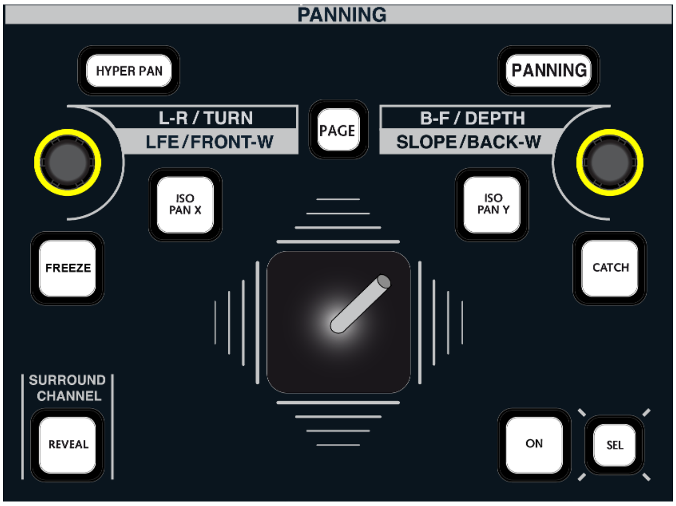
Reveal on Fader Strips

This method may be used to reveal Surround VCAs or normal VCAs.

Before using **REVEAL**, you should check where the slave faders will appear, using the Reveal options in the **System Settings** display. You can also use these options to disable **REVEAL** if you wish.

To reveal the slaves of a VCA master:

1. Put the surround or VCA master channel into access by selecting its fader **SEL** button.
2. Now press the SURROUND CHANNEL **REVEAL** button located on the PANNING panel:



The slave channels automatically appear, in their predetermined position, on the control surface.

This position is determined by the Reveal options in the **System Settings** display, so if the slave channels do not appear, check the options.

3. You may now adjust the slave fader positions and other relative parameters. (For details on which parameters are relative, see the Appendices).

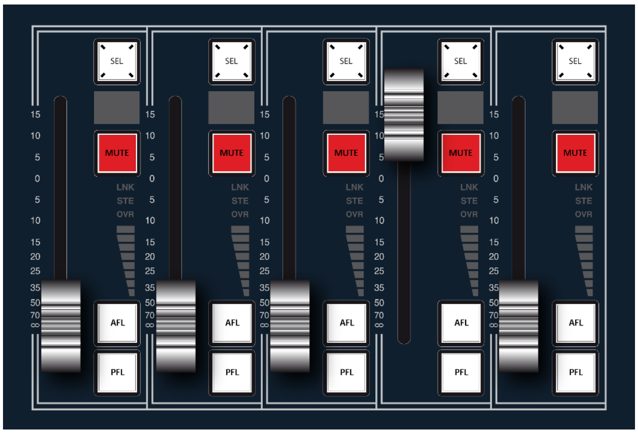
For Surround VCA masters, faders are *always* moving. Therefore, you will need to open the surround VCA master fader in order to offset the slaves.

Note also that when surround channels are created, the front and rear left/right pairs of the surround channel are automatically linked for stereo. This means that when you reveal the slaves, the front L/R and rear L/R components appear on two stereo fader strips. If you wish to control the Left and Right independently, then unmake the stereo link – press **SEL** on the L/R slave channel and deselect the **STEREO** button from the IMAGE panel.

4. Deselect **REVEAL** to revert the fader strips to their previous assignments.

A VCA can be master of a surround VCA. If this is the case, select the VCA master's fader **SEL** button and press **REVEAL** to reveal the VCA group slaves; then press the fader **SEL** on the surround VCA master to reveal the surround slaves.

The Reveal Fader Surround User Panel



This optional panel may be fitted to the right of the Central GUI, see Overbridge optionswithin the central user panel area.

It provides five dedicated faders for controlling surround VCA slaves; the main differences to revealing on normal fader strips are:

- The user panel is dedicated to surround VCA slaves and does not reveal normal VCAs.
- The faders are *always* active; the last selected surround VCA remains assigned to the reveal faders even if you select a different channel type.

1. Put a Surround VCA master channel into access by selecting its fader **SEL** button.

The slaves appear on the five faders; you will see the name of the component channel (e.g. Lr, C, LF, etc.) in the fader display.

2. Adjust the fader and **MUTE** buttons as required. You can also **AFL**, **PFL** or select (**SEL**) each slave.

3. Select a different surround VCA to reveal its slaves.

Note that the last selected surround VCA remains active even if you press ***SEL*** *on a different channel type.*

The layout of the slave channels varies depending on the surround format, and whether channels are linked for stereo. For example, if the global surround format is **5.1**, you will see:

- Fader 1 = Lr (front LR linked for stereo)
- Fader 2 = C
- Fader 3 = LF (LFE)
- Fader 4 = Su (surround LR linked for stereo)
- Fader 5 = blank

If you remove the stereo linking for both the front LR and surround LR slaves you will have 6 fader levels (too many for the user panel). To access the additional channels, a central user button may be factory-configured to switch say fader 3 between Centre and LFE.

You can reveal to both console fader strips and the user panel if you wish.
Or, set the Reveal bay count option to **0** to disable reveal on normal fader strips.

Hyper Pan

The console's Hyper Pan section is designed to help reposition surround sources within a surround field. For example, you may have to deal with a surround microphone where the left and right inputs are out of phase, or maybe you wish to rotate the surround source around the sweet spot axis.

To adjust the Hyper Panning of a surround channel:

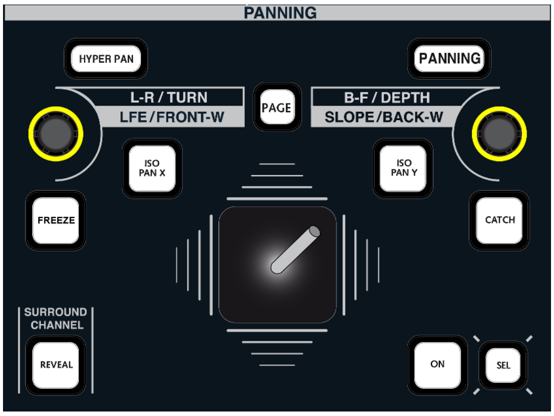
1. Put the surround VCA master channel (e.g. **SUR 1**) into access by pressing its fader **SEL** button.

2. Select the Central Control Section **PANNING** button. Check that **ON** is enabled (this switches panning into circuit), and the current XY pan position - the default starting point should be the sweet spot.

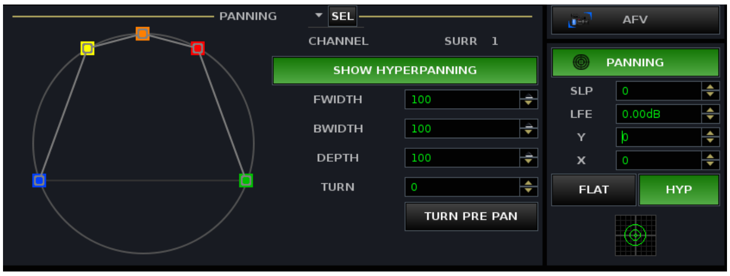
3. Then select **HYPER PAN** and enable its **ON** button (this switches panning from XY to Hyper Pan mode).

*You can double-check all your selections from the **Main Display** - both **PANNING** and **HYP** should be on (green), and the XY position should be at the sweet spot (**X = 0** and **Y = 0**):*

Central Control Section



Main Display



If you are unsure about any of the current settings, then you can reset panning using the RESET function.

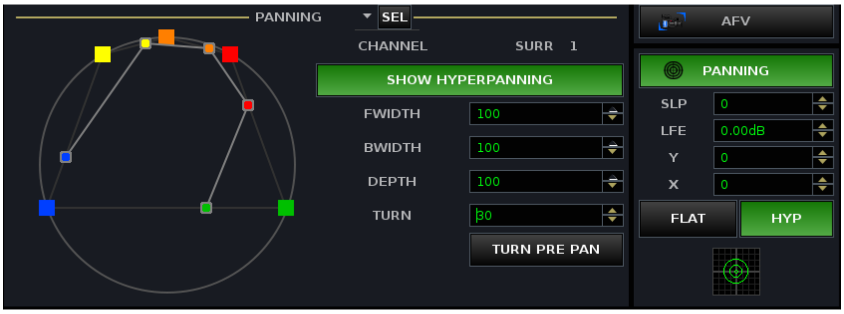
Hyper Pan is best explained by looking at the **Main Display**. The current positions of each node are colour coded according to the surround format (as defined by the AES). In our 5.1 example, the colours are:

- Yellow = Front Left
- Orange = Front Centre
- Red = Front Right
- Blue = Surround Left
- Green = Surround Right

4. Use the rotary controls (plus the **PAGE** button) to adjust each Hyper Pan parameter. In the examples which follow, we have reset each control before adjusting the next to show the affect of each parameter. However, you may combine parameters as you wish.

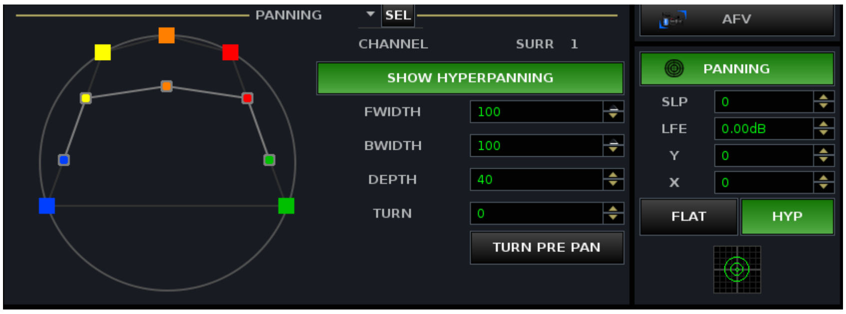
➢ TURN

This parameter rotates the surround source within the surround field. It can be adjusted from 0 degrees to +180 or -180 degrees:



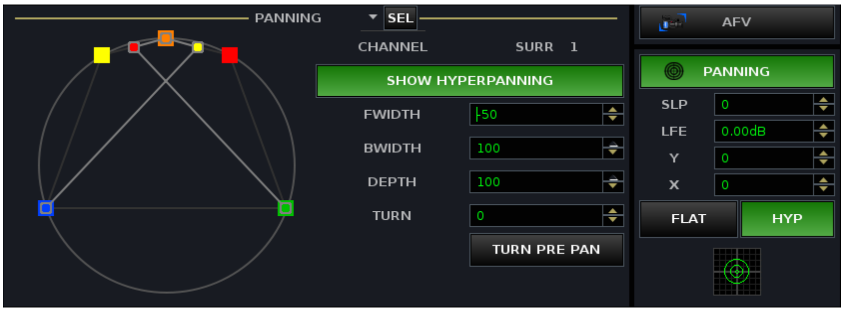
➢ DEPTH

This parameter reduces the depth of the surround source with respect to the sweet spot. It can be adjusted from +100% through 0% (all nodes are aligned at the sweet spot) to -100% (front and rear nodes are reversed) – our example shows the depth reduced to +40%:



➢ FWIDTH (Front Width)

This parameter adjusts the width of the front channels. It can be adjusted from +100% (full width) through 0% (all channels centered) to -100% (left and right channels are reversed):

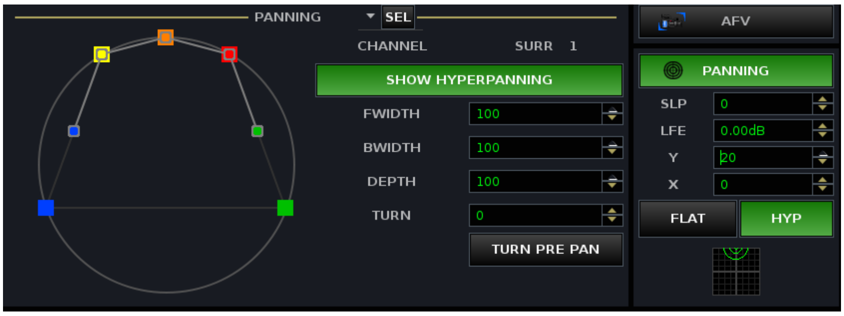


➢ BWIDTH (Back Width)

This parameter adjusts the width of the rear channels. It can be adjusted from +100% (full width) through 0% (all channels centered) to -100% (left and right channels are reversed) - our example shows Back Width set to +20%:

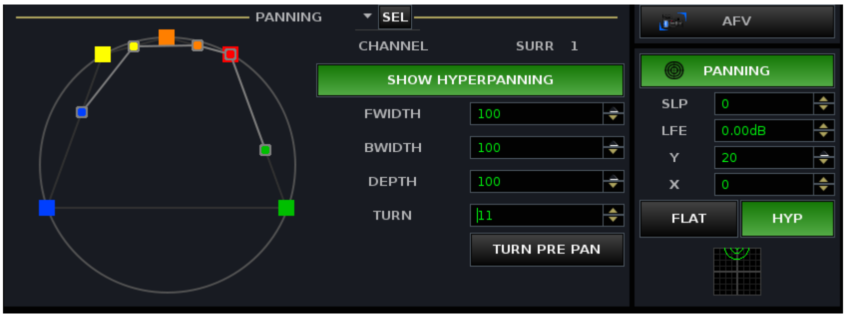
➢ Adjusting the Sweet Spot

You may use the joystick to reposition the sweet spot – the example below shows all parameters set to their defaults, but with the joystick position set forward, effectively bringing the surround channels closer to the front field:



➢ Combining Parameters

If you now adjust the **TURN** control, you will find that the surround source rotates around front centre (the current joystick position):

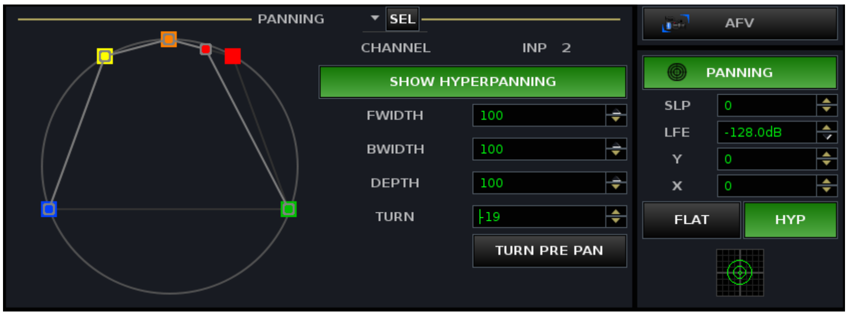


By enabling the **TURN PRE PAN** button (on-screen), you can turn the surround source and then position the rotated source using the joystick.

Hyper Pan on Surround Slave Channels

You can use Hyper Pan on individual surround slaves to adjust the relative position of an individual channel – for example, if a surround source is offset slightly to the right, then put the right channel (e.g. INP 18) into access and use the Hyper Pan **TURN** control to adjust the offset independently from the other slaves. Remember to undo the stereo linking for inputs 1 and 2 first!

Main Display

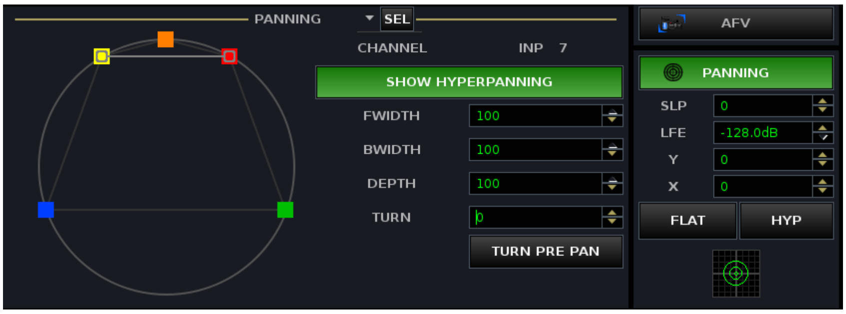


Hyper Pan on Mono or Stereo Channels

The Hyper Pan controls may also be used on mono or stereo channels which are assigned to a surround output.

The example below shows the default position of a stereo source when working in Hyper Pan mode:

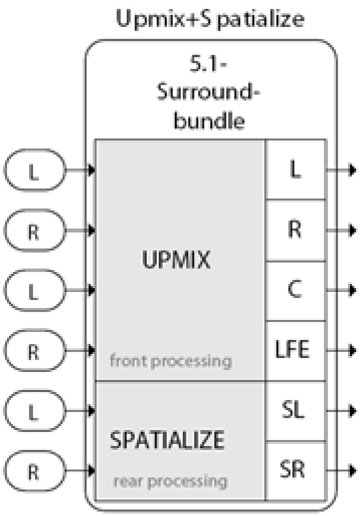
Main Display

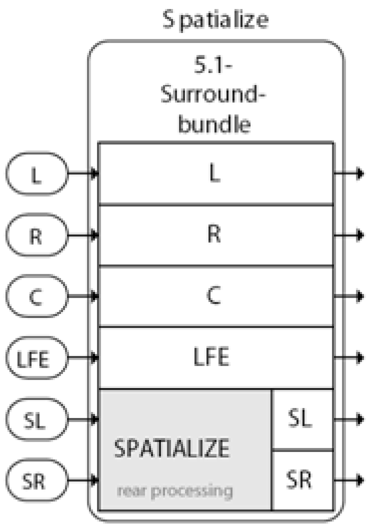


Set the **Y** pan position (front/rear) to **0** and then use the **TURN** control to rotate the stereo source within the surround field.

AMBIT

The Lawo AMBIT (AMBience IT) is a DSP module specifically designed for 5.1 surround channels providing upmix and spatialise processing. It may operate in one of two modes:





- **Upmix (& Spatialise)** – a 2 in, 6 out upmixer which, using sophisticated algorithms, converts stereo signals into 5.1 surround.
- **Spatialise Only** – a 6 in, 6 out spatialiser which processes the surround left and right channels only, ideal for treating incoming 5.1 signals.

The module is available in all full processing channels configured for 5.1 surround, except auxes. In other words, AMBIT may be applied to 5.1 input channels, monitor channels, groups and sums.

Please note:

- AMBIT processing can be applied to Inputs, Monitor channels, Groups or Sums, but not to Auxes.
- The global surround format must be 5.1. If a different format is selected, the AMBIT module is not available.
- AMBIT processing must be enabled from the **System Settings** display (using the Upmix/ Spatialize Enable option).
- Once an AMBIT module is active, the 5.1 component channels lose some of their other DSP. This is necessary to support the extra processing required for the AMBIT algorithm:
  - o On Recording channels, the Delay, Filter, Image, Gate and Expander modules are suspended.
  - o On Broadcast channels, the Delay, Insert and Dynamics modules are suspended.
- All AMBIT parameters are stored in productions, snapshots and presets.
- AMBIT processing is fully compatible with any downmix.
- The following functions are *NOT* supported by the AMBIT module - Link Groups; COUPLE; LISTEN; Snapshot Cross Fades; Timecode Automation.

How AMBIT Works

The Upmix & Spatialise mode uses sophisticated algorithms to convert 2-channel stereo signals into 5.1 surround. It can be used on an Input or Monitor channel, or on an output Group or Sum.

Applications

Here are some examples of when you might apply AMBIT processing to different channel types:

- Inputs – you could use the AMBIT module within a 5.1 Input channel to create a surround upmix from a stereo ambience microphone.
- Sums – you can create a 5.1 upmix from your stereo master by applying the AMBIT processing to a Sum.
- Groups – to upmix some stereo sources but not others, then apply the AMBIT processing to a 5.1 Group which feeds a 5.1 Sum. Route all the stereo sources you wish to upmix to the Group. Then route any channels you wish to bus and pan manually onto the 5.1 Sum.

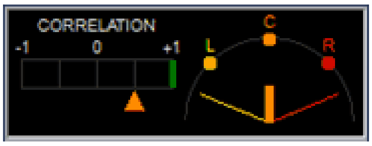
Parameters

Imagine that you are sitting in a virtual room, listening to the source from a pair of stereo loudspeakers. You are the target. Using AMBIT you can define:

- **The Virtual Room** – the size of the room and how it handles reflections.
- **The Source position** – the position and width of the source playback loudspeakers.
- **The Target position** – your listening position.

Having defined how the source signal is "heard", you can then determine how the 5.1 output is processed:

- **Front Processing** – these parameters define how much correlated signal (mono signal) feeds the discrete centre channel, as opposed to left and right (phantom centre). There are two modes in which you can work:
  - o **Auto-centre** – in this mode the AMBIT module decides automatically how much correlated signal feeds the discrete centre channel versus left and right (phantom centre), based on the correlation threshold and time. The algorithm works dynamically, according to changes in the correlated signal level, producing a stable front image for any content:



  - o **Manual centre** – with Auto-centre turned off, the correlated signal feeding the centre channel is set manually. You can adjust the left/right width (Basewidth) and discrete centre channel level (Centering). You can also choose to link Centering and Basewidth in order to maintain a consistent ratio.
- **Rear Processing** – these parameters define the processing applied to the surround left and right channels. Parameters are available to control the left/right width (Basewidth), high pass filtering and the virtual room simulation:

In Spatiliase mode, only the rear processing is applied to the incoming surround left and surround right channels.

Turning On AMBIT

The AMBIT DSP module is enabled, disabled and controlled from the surround VCA master of a 5.1 channel.

> AMBIT processing can be applied to Inputs, Monitor channels, Groups or Sums, but not to Auxes.
>
> The global surround format must be 5.1. If a different format is selected, the AMBIT module is not available.
>
> AMBIT processing must be enabled from the **System Settings** display (using the Upmix/ Spatialize Enable option).

1. Select the Surround VCA master by pressing its fader **SEL** button.

2. Press the **CHAN/CONFIG** button, located on the SCREEN CONTROL panel, to view the **Main Display**:

*With the surround VCA in access, the buttons to control the AMBIT module appear at the bottom of the **MODULES** list on the right of the display:*

If you cannot see the Upmix (**UPX**) and Spatialise (**SPZ**) buttons, then check the following:

- Is the surround VCA in access?
- Is the global surround format set to 5.1?
- Is the Upmix/Spatialize Enable option turned off within the **System Settings** display?

3. Turn on the Upmix & Spatialise mode (2 in: 6 out) by selecting the **UPX** touch-screen button.

*Both **UPX** and **SPZ** are enabled (green).*

4. Alternatively, turn on Spatialise mode (6 in: 6 out, rear processing only) by selecting **SPZ** only.

When AMBIT processing is turned on, then the component channels lose some of their other DSP:

- Delay, Filter, Image, Gate and Expander (Recording channels).
- Delay, Insert and Dynamics (Broadcast channels).

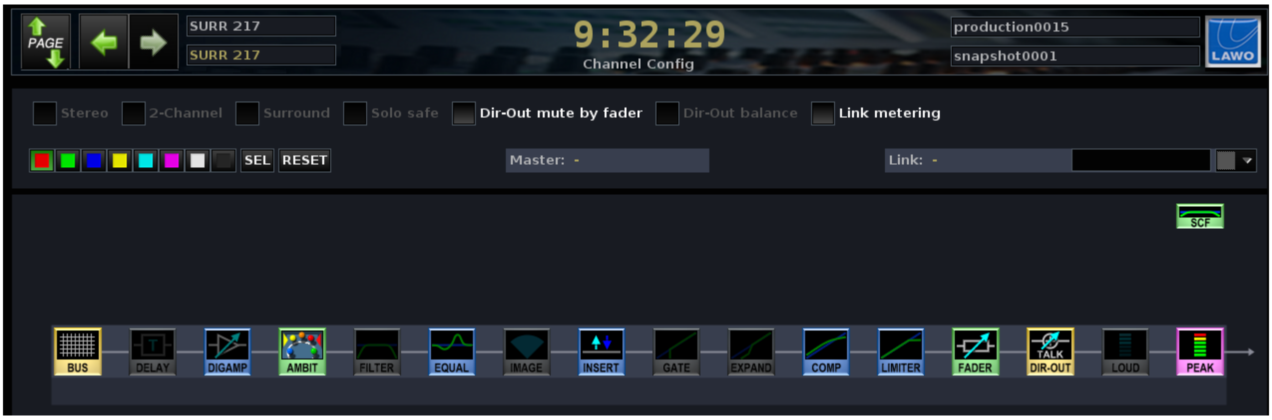
This is reflected by the greyed-out module icons on the **Main Display**.

5. To reinstate the suspended DSP modules, you must turn off the AMBIT processing - deselect **UPX** and **SPZ**.

Changing the Signal Flow

The AMBIT module can be moved within the channel signal flow in the same manner as other DSP modules.

1. Put the surround VCA into access – press **SEL** on its fader strip.
2. Open the **Channel Config** display.
3. Select the AMBIT module and adjust its position using either the **LEFT**/**RIGHT** or **UP**/**DOWN** soft keys:

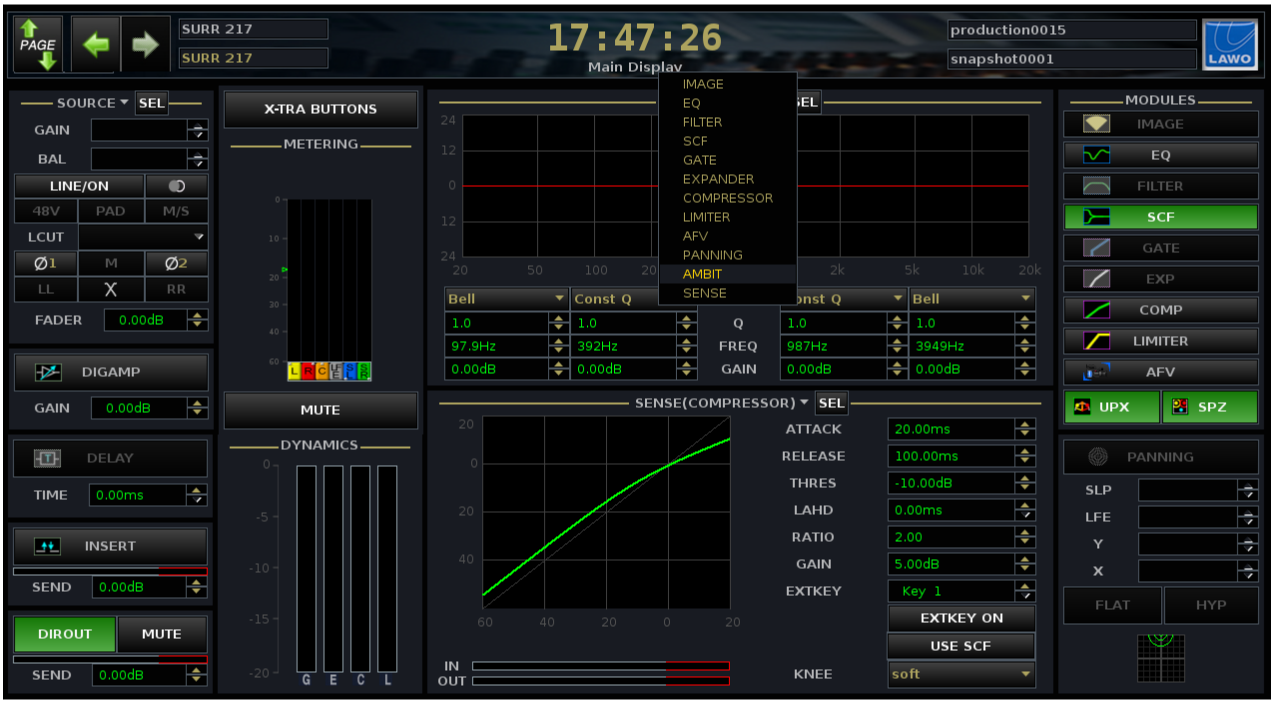


For more details, see Changing the Signal Processing Order.

Controlling AMBIT Processing

All AMBIT parameters are adjusted from the **Main Display**:

1. Make sure that the surround VCA is in access and assign the **AMBIT** module to one of the assignable views on the Main Display:



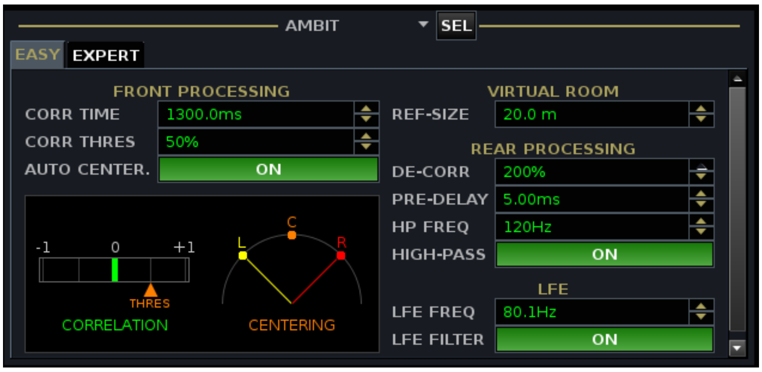
3. Make sure that the AMBIT module is turned on – either **UPX** and **SPZ**, or **SPZ** only.

4. Adjust the parameters from the on-screen buttons. You have a choice of two modes:

- Easy Setup
- Expert Setup

AMBIT Easy Setup

The **Easy** tab allows you to configure AMBIT processing using just 10 parameter options:



1. First set the **REF-SIZE** for the VIRTUAL ROOM.

This defines the virtual room size in metres (m).

2. Then determine the FRONT PROCESSING parameters as follows:

In Easy setup keep AUTO CENTER turned **ON**. In this mode the AMBIT module decides automatically how much correlated signal (mono source signal) feeds the discrete centre channel versus left and right (phantom centre) according to the Correlation Threshold and Time. The algorithm works dynamically, following changes in the correlated signal level, producing a stable front image for any type of content.

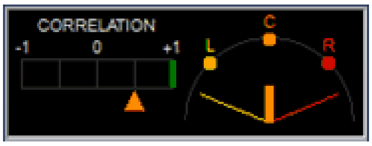
The correlated signal threshold and time determine the working point for the auto-centering algorithm:

- **CORR THRES** – sets the correlated signal threshold (100% = +1, 0% = 0).

Correlated signals above the threshold feed the centre channel and those below feed left and right equally (phantom centre).

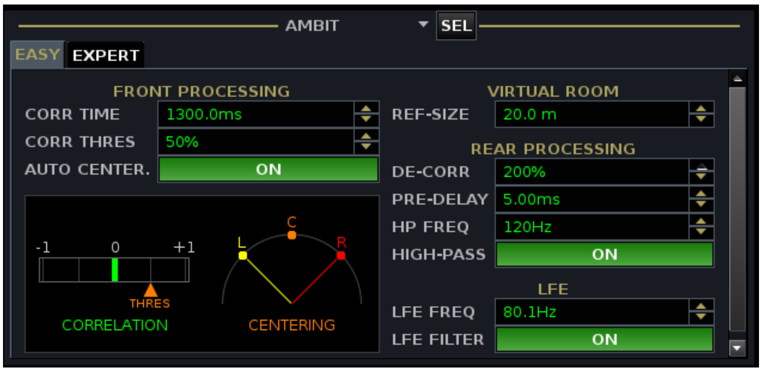
- **CORR TIME** – sets how quickly the auto-centering reacts to correlated signals falling above/ below the threshold.

You can monitor the affect of the auto-centering algorithm using the on-screen graphics:



If you wish to control the front processing parameters manually, then use Expert setup.

3. Next adjust the REAR PROCESSING parameters:



- **DE-CORR** – sets the amount of de-correlated signal applied to the rear. In other words, the impact of mono source content on the rear channels.

100% is the default value. You can increase it to 200% (only de-correlated signal) or reduce it to 0% (only correlated signal).

- **PRE-DELAY** – sets the amount of pre-delay in milliseconds (ms) applied to the rear channels.

The bigger the pre-delay, the more reflective the virtual room will appear.

- **HP FREQ** – sets the roll-off frequency for the high pass filter in Hz (see below).
- **HIGH-PASS** – turns the high pass filter on or off.

The rear processing high pass filter is a 2nd order (12dB/octave) filter which can be applied to the rear channels.

4. Finally adjust the LFE low pass filter parameters:

- **LFE FREQ** – sets the roll-off frequency for the low pass filter in Hz (see below).
- **LFE FILTER** – turns the LFE filter on or off.

The LFE low pass filter is a 4th order (24dB/octave) filter which can be applied to the Low Frequency Effect (subwoofer) channel.

Use the REVEAL function to assign the surround component channels to the surface, in order to apply offsets to Left, Right, Centre, Surround Left, Surround Right or the LFE.

AMBIT Expert Setup

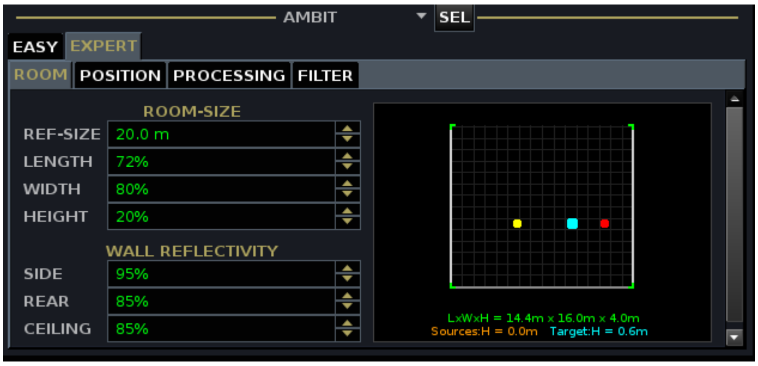
The **Expert** tab provides access to more advanced parameters.

When working in **Expert** setup, assign the AMBIT module to both assignable views within the **Main Display**. This allows you to view different tabs simultaneously.

For example, it makes sense to view the Reflection Patterns for the Surround Left/Right channels (**POSITION** tab) while adjusting **ROOM** and then **PROCESSING** parameters, as both have an effect on the reflection patterns:

>> ROOM

1. Select **ROOM** to define the virtual room:



2. Use the ROOM-SIZE parameters to define the shape and size of the virtual room:

- **REF-SIZE** - Room size in metres (m). Also available in Easy setup.
- **LENGTH** - Length as a % of the room size.
- **WIDTH** - Width as a % of the room size.
- **HEIGHT** – Height as a % of the room size.

Any changes are represented by the on-screen graphics.

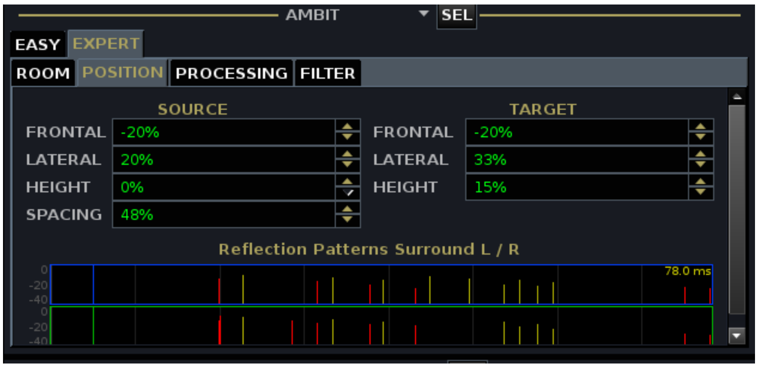
3. Use the WALL REFLECTIVITY parameters to define the reflectivity of the surfaces within the virtual room.

100% = very reflective; 0% = not reflective:

- **SIDE** - Reflectivity of the side walls (left/right).
- **REAR** - Reflectivity of the rear wall.
- **CEILING** - Reflectivity of the ceiling.

>> POSITION

1. Select **POSITION** to define the source and target positions:



2. Use the SOURCE parameters to define the position and spacing of the stereo source loudspeakers:

- **FRONTAL** - front/rear speaker position: 100% = front; 0% = middle of the room; -100% = rear.
- **LATERAL** - left/right speaker position: 100% = right; 0% = centre; -100% = left.
- **HEIGHT** - height of speaker position: 100% = top; 0% = middle; -100% = bottom.
- **SPACING** – left/right spaving: 100% = full width; 0% = mono.

3. Use the TARGET parameters to define the position of the listening target:

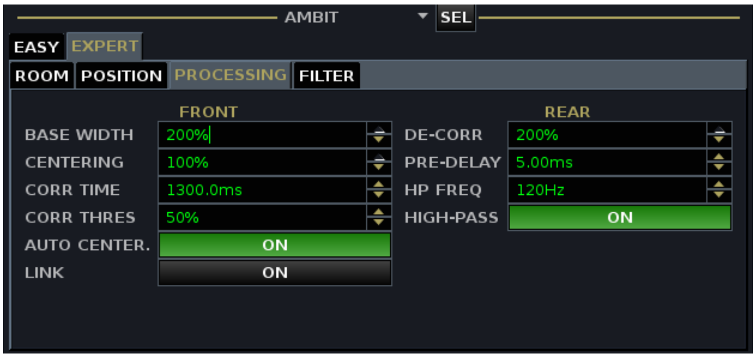
- **FRONTAL** - front/rear target position: 100% = front; 0% = middle of the room; -100% = rear.
- **LATERAL** - left/right target position: 100% = default.
- **HEIGHT** - height of target position: 100% = default.

The on-screen graphic provides a visualization of the resulting reflection pattern.

4. If not already visible, select the **ROOM** tab and you will see that the source and target positions are represented by the yellow (left channel), red (right channel) and turquoise (target) dots.

>> PROCESSING

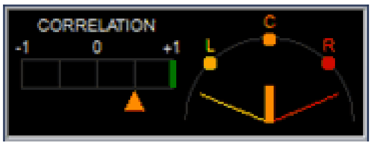
1. Select **PROCESSING** to define the front and rear processing parameters.



2. When dealing with the FRONT processing parameters, it is best to work with **AUTO CENTER** either on or off:

- **AUTO CENTER ON** - use the correlated signal threshold and time as described for Easy Setup.
- **AUTO CENTER OFF** – control the front upmix processing manually:
  - o **BASEWIDTH** – sets the left/right width: 100% = default; 200% = overwidth; 0% = mono.
  - o **CENTERING** – sets the amount of correlated signal feeding the discrete centre channel: 100% = discrete centre only; 0% = phantom centre, no discrete.
  - o **LINK** - turn on this option to link **BASEWIDTH** and **CENTERING**. This ensures that the correlated signal level remains constant and that there is an equal distribution of power between the three front channels.

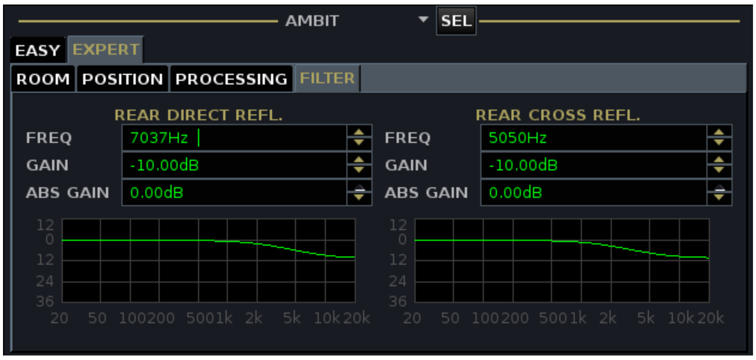
The results are best represented by the on-screen graphic within the Easy tab:



3. The REAR processing parameters are identical to those in Easy setup.

>> FILTER

1. Select **FILTER** to define the remaining parameters:



2. Use the REAR DIRECT REFL. and REAR CROSS REFL. parameters to adjust the virtual room reflections applied by the Spatialiser to surround left and right.

Note that these parameters affect the room simulation, and are completely separate from the rear processing high pass filter (controlled from the PROCESSING tab).

Direct reflections describe reflections from the closest wall; cross reflections come from an opposing wall. So, if a signal eminates from the left source loudspeaker, then direct reflections come from the left and cross reflections from the right.

For each pattern, you can apply a shelving filter with parameters for:

- **FREQ** - the roll-off frequency of the shelving EQ.
- **GAIN** – the gain of the shelving EQ.
- **ABS GAIN** – offsets the resultant shelving pattern.

Save, Load, Select and Copy/Reset

➢ Saving and Loading AMBIT Parameters

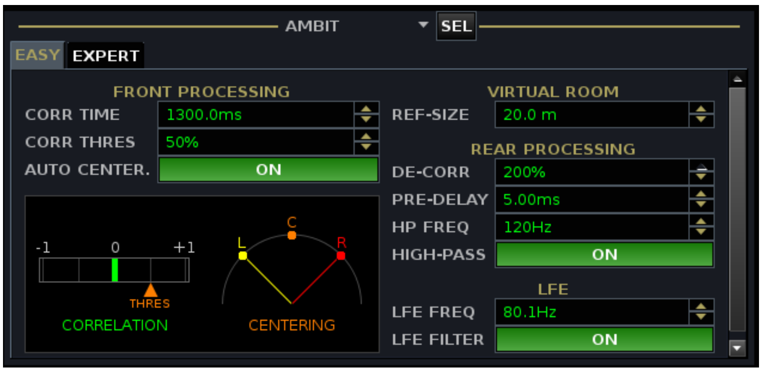
All AMBIT parameters are stored in productions and snapshots. Therefore, you can easily store and recall upmix processing as part of your mix.

AMBIT parameters may also be stored as presets in order to save and load favourite settings. To save or load a preset:

1. Right-click on either the **UPX** or **SPZ** button on the right of the **Main Display** and select **Load** or **Save Preset**:

➢ Selecting the AMBIT Module

Some operations, such as copy channel, require you to select the AMBIT module. This is achieved by using the on-screen **SEL** button within the **Main Display**:



The **SEL** button turns green when selected.

Note that the AMBIT module is automatically selected by pressing SEL ALL on the Central Control Section.

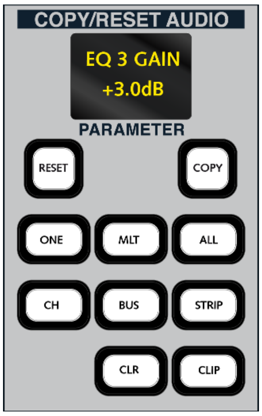
➢ Copy and Reset for AMBIT

- **Copy Parameters** - AMBIT parameters can be copied between surround VCAs. You cannot copy AMBIT parameters to other channel types.
- **Reset Parameters** - AMBIT parameters can be reset.

See Copy & Reset for more details.

Copy & Reset

The COPY/RESET AUDIO panel, located within the Central Control Section, may be used to copy and reset DSP parameters, bus assignments, channel signal flow, channel colour coding and fader strip free control assignments.



Individual or groups of settings may be copied or reset. When performing a copy, you may copy to single or multiple destinations.

Note that this panel is also used to make Free Control assignments The **CLR** button is used to clear a Free Control, and has no function when combined with **COPY** or **RESET**.

Copying to a Single Channel

To copy parameters from one channel to another:

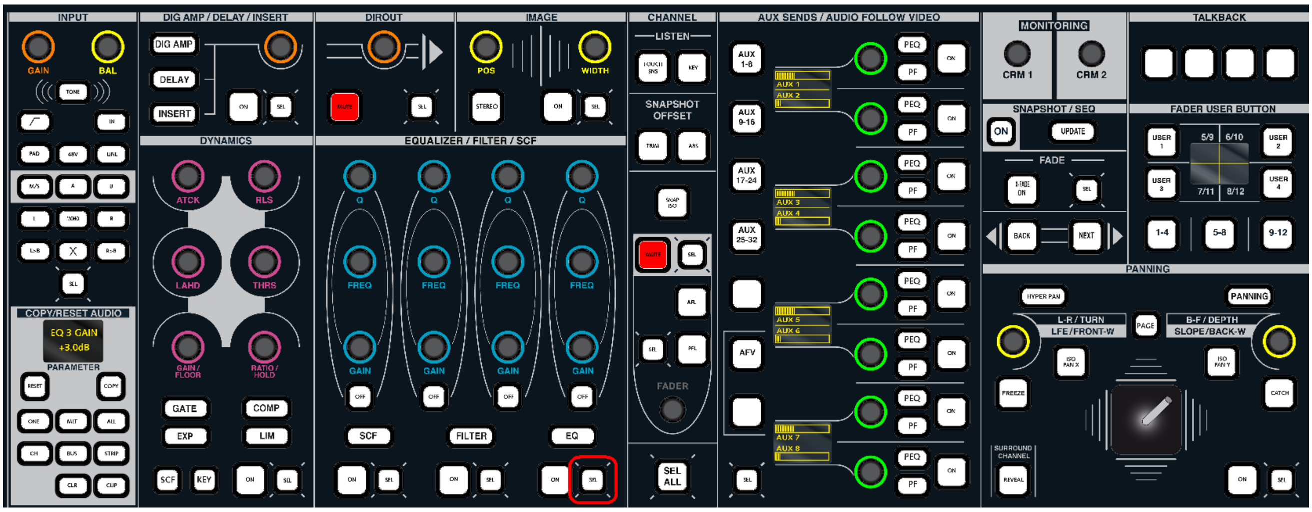
1. Assign the source channel to the Central Control Section, by pressing its fader **SEL** button or using the ACCESS CHANNEL/ASSIGN panel.
2. Press the **COPY** and **ONE** buttons, located on the COPY/RESET AUDIO panel, to activate a one-shot assignment.

   *The fader **SEL** buttons across the console flash, in green:*

3. Select the audio module(s) you wish to copy, by enabling the **SEL** buttons on the Central Control Section.

> To clear down any existing selections, toggle the **SEL ALL** button (this selects and then deselects all modules). This ensures that there no "hidden" selections.

Then turn on the **SEL** button(s) to make your selections. In our example, the EQ **SEL** button selects all parameters from the 4-band EQ module:



You can choose multiple **SEL** buttons and other channel parameters if you wish, see Selecting Channel Parameters.

4. Choose the destination channel by pressing its fader **SEL** button.

   *The selected parameters are copied, and the **COPY** and **ONE** buttons cancel.*

If you wish to exit without copying any settings, just deselect the **COPY** button or press **ESC**, on the SCREEN CONTROL panel, at any point during the operation.

Copying to Multiple Channels

Copy to Multiple

You can copy the selected parameters to multiple channels by selecting **COPY** and **MLT** during step 2 (rather than **COPY** and **ONE**).

This activates the multi-assign mode so that in step 4, you can choose multiple destination channels:

Remember to deselect the copy mode, by turning off **COPY** and **MLT** or pressing **ESC**, when you have finished choosing the destinations.

Copy to ALL

Alternatively, to copy parameters to *all* channels of a particular type, select **COPY** and **ALL** during step 2 (rather than **COPY** and **ONE**). Then for step 4:

4. Choose the channel type you wish to assign to, from the ACCESS CHANNEL/ASSIGN panel – for example, select **INP**.

You can select multiple channel types if you wish – e.g. select **INP**, **GRP** and **SUM** to copy a parameter across all input, group and sum channels.

5. Press the Enter button on the ACCESS CHANNEL/ASSIGN control panel to complete the copy operation.

The selected parameters are copied to all input channels.

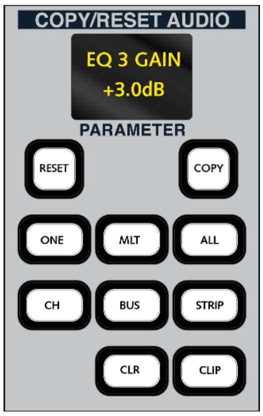
Selecting Channel Parameters

Audio Modules

To select all the DSP parameters from an audio module, use the **SEL** buttons within the Central Control Section:

- INPUT – the **SEL** button selects *either* the source *or* channel input parameters, depending on the current SOURCE/INMIX mode.
- DIG AMP, DELAY, INSERT, DIROUT – you can enable the **SEL** button for each section independently, to select the digital amplifier gain, channel delay, insert send or direct output parameters.
- IMAGE – selects the stereo image and position.
- DYNAMICS – you can enable the **SEL** button for each section independently, to select the gate, expander, compressor or limiter parameters.
- SCF – selects the sidechain filter parameters (Recording channels only).
- FILTER – selects the 2-band filter parameters (Recording channels only).
- EQ – selects the 4-band EQ parameters.
- MUTE **SEL** – selects the status of the mute button.
- Fader **SEL** – selects the main channel level.
- AUX SENDS/ AUDIO FOLLOW VIDEO – you can enable the **SEL** button for each page of aux sends and AFV independently. This allows you to select eight aux sends at a time (Aux 1-8, Aux 9-16, Aux 17-24 or Aux 25-32).
- PANNING – selects the stereo/surround pan parameters.

Bus Assignments, Channel Signal Flow and Free Controls



You can select other channel parameters using the buttons on the COPY/RESET AUDIO panel:

- **CH** - selects the channel signal processing order as defined from the Channel Config display. (On Broadcast channels, this includes the dynamics model).
- **BUS** - selects the channel's bus assignments to groups, track busses and sums. (Note that aux assignments are not included; you should use the AUX SENDS panel).
- **STRIP** - selects the fader strip's Free Control assignments.

Channel Colour Coding

In addition, you can select the channel's colour code using the on-screen **SEL** button on the Channel Config display:

Select All/Clearing Selections

Every time you re-enter the copy or reset mode, any previous selections are retained.

SEL ALL, below the FADER level control, will select, or deselect, all available channel parameters (including **CH**, **BUS** and **STRIP**.)

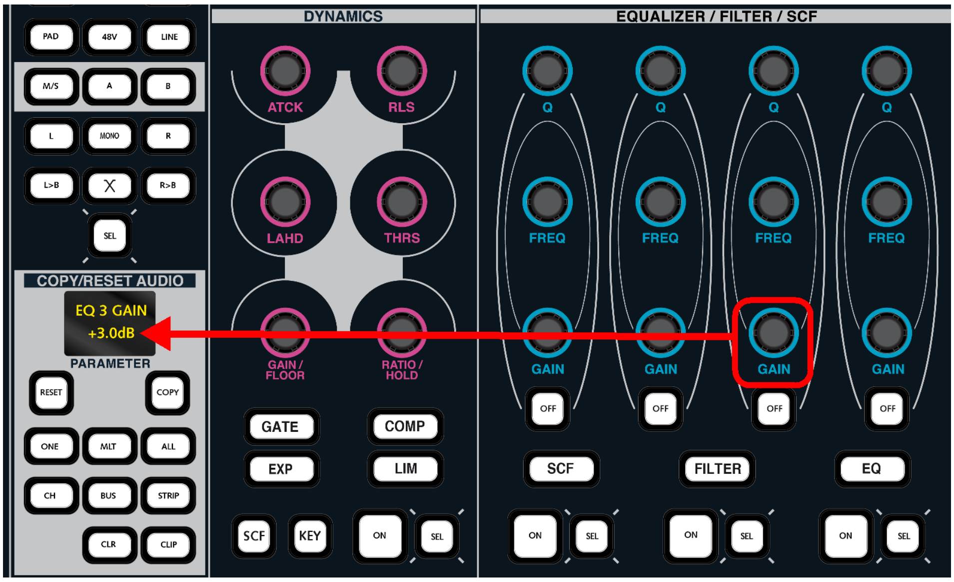
Copying an Individual Parameter

To copy just one parameter from an audio module – for example **EQ 3 Gain** – use the CLIPBOARD and **CLIP** button as follows:

1. Assign the source channel to the Central Control Section, by pressing its fader **SEL** button or using the ACCESS CHANNEL/ASSIGN panel.

2. Select the parameter you wish to copy, by touching a rotary encoder on the Central Control Section - for our example, touch the EQ Band 3 **GAIN** control.

The parameter is placed into the PARAMETER clipboard:



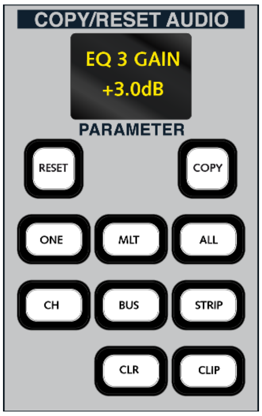
3. Press the **COPY** and **CLIP** buttons on the COPY/RESET AUDIO panel.

*This automatically selects the **ONE** button for a one-shot copy. (If you wish to copy the parameter to multiple channels, press **MLT** instead of **ONE**.)*

4. Choose the destination channel(s) by pressing the fader **SEL** button(s).

The EQ 3 Gain parameter is copied to the destination channel(s); all other EQ parameters are unchanged.

Resetting Channel Parameters



The **RESET** button, located on the COPY/RESET AUDIO panel, may be used to reset channel parameters to their default values.

This works in a similar manner to copying parameters. However, as you are not copying from a source to a destination channel, any channel may be assigned to the Central Control Section.

1. Press **RESET** plus **ONE**, **MLT** or **ALL** depending on whether you wish to reset parameters on a single or multiple channels.

2. Select the audio module(s) you wish to reset from the Central Control Section, see Selecting Channel Parameters.

3. Choose the channels to reset by pressing the fader strip **SEL** button(s), or entering the channel type (if using **ALL**).

Metering

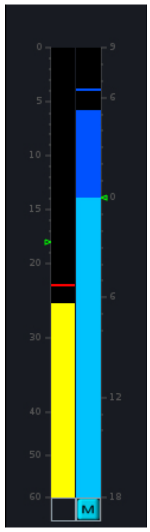
There are several places where signals are metered:

- The Channel display provides dedicated metering for every channel fader strip.
- The signal present LEDs provide a confidence indicator beside every fader.
- The Main Display on the Central GUI includes a meter which follows the same options as applied to the **Channel** display, and also meters signals at other points such as the insert send, direct out and dynamics modules.
- The Metering display on the Central GUI contains four pages of assignable meters.
- The Overbridge may be fiitted with RTW external metering.

This section deals with:

- Bargraph Types - selecting peak metering, loudness metering or both.
- Peak Metering - options and characteristics.
- Loudness Metering - options and characteristics.
- Meter Pickup Points - for peak and loudness metering.
- The Metering Display - assignable metering.
- The Main Faders Display - metering the 16 main fader strips.

Bargraph Types



For all on-screen meters you may choose to display peak metering, loudness metering, or a combination of both.

Note that the bargraph type affects all on-screen meters, including the **Channel**, **Main** and **Metering** displays.

Also note that loudness metering must be active, before any loudness measurements are displayed.

The default option can be set independently for input channels and summing channels (groups, sums, auxes) from the System Settings display.

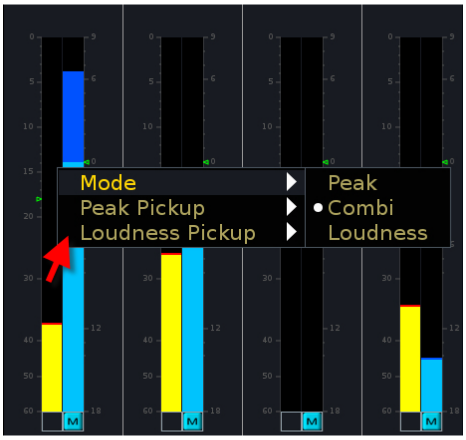
In each case, you may choose from:

- **Combi** – peak and loudness metering side by side.
- **Peak** – peak metering only.
- **Loudness** – loudness metering only.

You may then edit the bargraph type, individually for each channel, either from the Channel display or Main display.

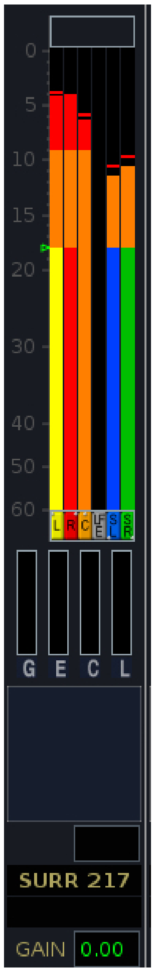
Click anywhere in the METERING area and select **Peak**, **Combi** or **Loudness** from the pop-up window:

Channel Display



Main Display on Central GUI

Peak Metering



Peak metering bargraphs are mono, stereo or multi-channel according to the format of the channel.

Pickup Point

The meter may be switched to different points within the channel signal flow by adjusting the peak meter pickup point.

Peak Meter Scales and Characteristics

A range of meter scales and characteristics are available from the **System Settings** display, see Bargraph options.

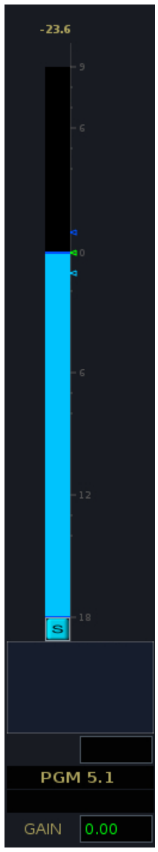
For ITU compliant operation, you should choose **True Peak** as the Full Channel Mode characteristic, and **dBFS** as the Scale mode. Then set the Reference Level equal to your maximum Analogue Level and the Headroom to 0dB. This ensures that the dBFS metering across the console matches any external AES metering you may have. You may use the Safe Area, Operation Range and Line Up Level options to help manage your own headroom.

Peak Hold

This function is also enabled from the **System Settings** display, see Peakhold options.

When enabled, the system monitors and marks the peak level reached on each meter across the console. You can set the peak hold indicator to clear automatically after a certain time period, or manually using the **CLEAR** peak hold soft key. You can also set the colour for the peak hold indicator.

Loudness Metering



The **mc²56** provides loudness metering conforming to the ITU-R BS1770.

Loudness Metering Bargraphs

A single bargraph (blue) represents the average energy of the summed component channels: mono, stereo or surround. The colour indicates whether loudness is above or below the Target Level:

- **Light Blue** = equal to, or below, the Target Level.
- **Dark Blue** = above the Target Level.

The dark and light blue scale markers indicate a tolerance of +/- 1 LU/LK.

The **M** or **S** at the bottom of the bargraph represents the integration time for the measurement:

- **M** = Momentary integration time (400ms sliding window)
- **S** = Short term integration time (3s sliding window)

This and other options are defined within the **System Settings** display, see Measurement Mode for Input/Summing Channels.

Integrated Loudness Measurement

On summing channels, you may also start an integrated loudness measurement. The result is displayed above the bargraph. In our example, **PGM 5.1** is reading **-23.6** LUFS (Loudness Units Full Scale).

The integrated measurement provides a very useful tool for measuring loudness over long periods of time. For example to measure the loudness of a complete programme transmission.

Pickup Point

The loudness meter may be positioned independently from the peak meter by adjusting the loudness meter pickup point.

Presets and Options

All options for loudness metering are adjusted from the **System Settings** display, see Loudness Metering Options. A choice of **Active Presets** recall the default settings specified by the EBU R128 or ATSC A/85 & ARIB.

You can find more information on loudness metering, and the international standards, in a white paper titled “Loudness Metering” available from the Lawo website:

English: http://www.lawo.de/en/products/mixing-consoles/loudness-metering.html

German: http://www.lawo.de/de/produkte/mischpulte/loudness-metering.html

Configuring Loudness Metering

When loudness metering is activated you must disable (suspend) some DSP from the processing channel in order to provide resources for the metering algorithm.

You can choose which DSP modules you would like to suspend on a channel by channel basis. The choice of suspended DSP module(s) is saved in the production.

> If you choose to suspend the EQ DSP module, then on Recording channels, you will lose the pre-EQ Aux send. This is due to the fact that the send is taken from the input to the EQ module.
>
> If you wish to activate loudness metering and AMBIT (upmix processing):
>
> - On Recording channels, you must choose DSP modules which do not include the Delay, Filter, Image, Gate or Expander (disabled when AMBIT is active).
> - On Broadcast channels, you must suspend the EQ (the only module not suspended by AMBIT processing).

The default suspended DSP module can be set independently for input channels and summing channels (groups, sums, auxes) from the System Settings display.

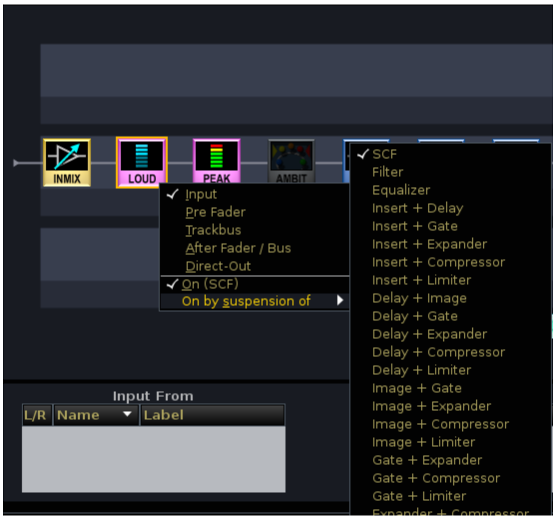
To modify the suspended DSP module on an individual channel:

1. Select the channel you wish to modify by pressing its fader **SEL** button.
2. Press the CHAN CONFIG button, on the SCREEN CONTROL panel, to open the **Channel Config** display.
3. Right-click on the **LOUD** DSP module to access the loudness metering options.

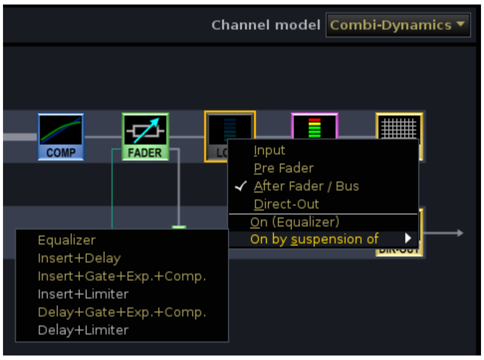
*In our example, the suspended module will be **SCF** as indicated at the bottom of the drop-down menu: **On (SCF)**:*

4. Select **On by suspension of** to open a second drop-down menu where you may alter the selection:

Options for Recording Channels



Options for Broadcast Channels



For example, when working with Recording channels, select **Delay + Image** and the display updates accordingly:

Selecting an option also activates loudness metering. This is indicated by the pink **LOUD** module, and the greyed out suspended modules (e.g. **DELAY** and **IMAGE**).

Note that if you subsequently change the Default Module Suspend Set option (from the **System Settings** display), then this will reset any individual channel modifications.

Activating the Loudness Meter Bargraphs

Before activating loudness metering, you should check that you have:

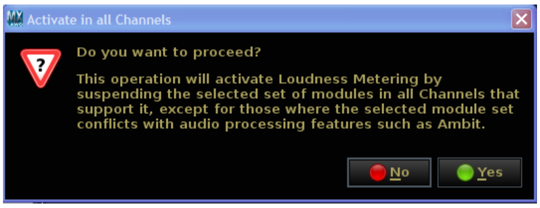
- Configured input and/or summing channels to display either **Loudness** or **Combi** metering, see Bargraph Types.
- Recalled an Active Preset (either EBU R128 or ATSC A/85 / ARIB) and adjusted the loudness metering options.

You can then activate loudness metering globally for all channels or for individual channels as follows:

➢ **Activate Loudness Metering (Global)**

1. From the **System Settings** display, enable the Activate In All Channels **Loudness Metering** option:

You will be presented with a confirmation pop-up:



2. Select **Yes** to proceed.

Loudness metering is activated for all channels that support it across the console. This could be for all input channels, all summing channels or both according to your Bargraph Type.

➢ Activate Loudness Metering (Single Channel)

1. From the **Channel Config** display, right-click on the **LOUD** DSP module to access the loudness metering options:

2. Select the **On (xxx)** option to enable or disable the loudness metering DSP.

*Loudness metering is off when the **LOUD** module is grey (as above).*

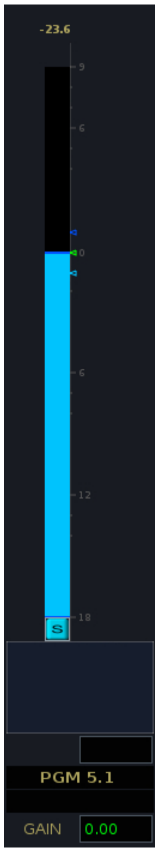
*Loudness metering is on when the **LOUD** module is pink and the suspended DSP modules are in grey (e.g. **SCF**):*

➢ To Disable Loudness Metering (and reinstate any suspended DSP modules):

1. Right-click on the **LOUD** DSP module and deselect the **On (xxx)** option so that it becomes unticked.

*Loudness metering is off when the **LOUD** module is grey.*

Integrated Loudness Measurement

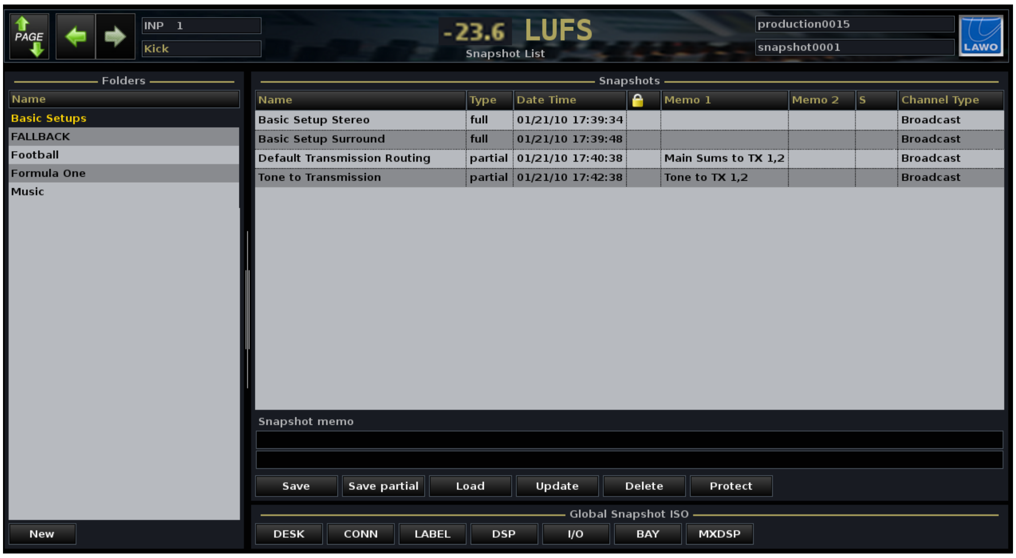


On any summing or monitor channel you may start and stop an integrated loudness measurement. This allows you to measure the loudness of channels, such as main programme or a clean feed, from start to finish. You can even pause the measurement during any unwanted periods such as an advert break.

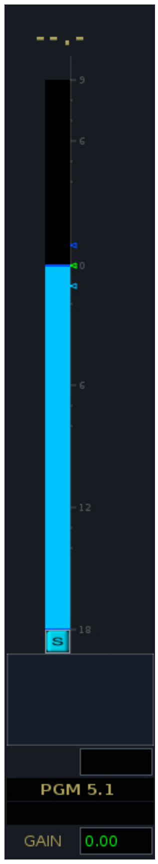
Note that the integrated loudness measurement is only available on summing channels (Sums, Groups, Auxes) and monitor channels. It is not available for input channels.

The result of the integrated measurement is displayed above the loudness bargraph either in LU/LUFS or LK/LKFS according to the choice of EBU or ATSC/ ARIB presets. In our example, **PGM 5.1** is reading **-23.6** LUFS.

In addition, you can display the integrated loudness measurement for a particular summing channel, such as main programme, in the title bar of the central GUI. This allows you to keep track of its loudness while working in other displays, or selecting different channels:



>> Starting the Integrated Loudness Measurement:



When you first activate loudness metering, the integrated loudness measurement remains blank, as shown opposite. This indicates that either the integration has not been started, or that there is no signal to measure.

Note that to comply with the ITU standard, the signal's loudness must be greater than -70 LUFS before an integrated measurement is registered.

To start the measurement:

1. Select the summing channel by pressing its fader **SEL** button, or using the ACCESS CHANNEL/ASSIGN panel.

Note that that on a surround channel, you must select the Surround VCA (e.g. **SURR 217**), and *NOT* one of the component channels.

2. Press the **CHAN/CONFIG** button, located on the SCREEN CONTROL panel, to view the **Main Display**.

3. Click anywhere in the METERING area and select **Start integration** from the pop-up window:

Providing that there is signal > -70 LUFS at the loudness meter pickup point, the integrated loudness reading updates. This figure represents the integrated loudness over time, and continually updates during your transmission.

The measurement is displayed either as an absolute value in LUFS, or relative to the Target Level in LU, as defined in the System Settings display.

Note that when you start the integration, the channel's **SNAP ISO** button may also be enabled. This protects the summing channel from snapshot recalls which may destroy the integrated loudness measurement. This default option can be modified from the System Settings display.

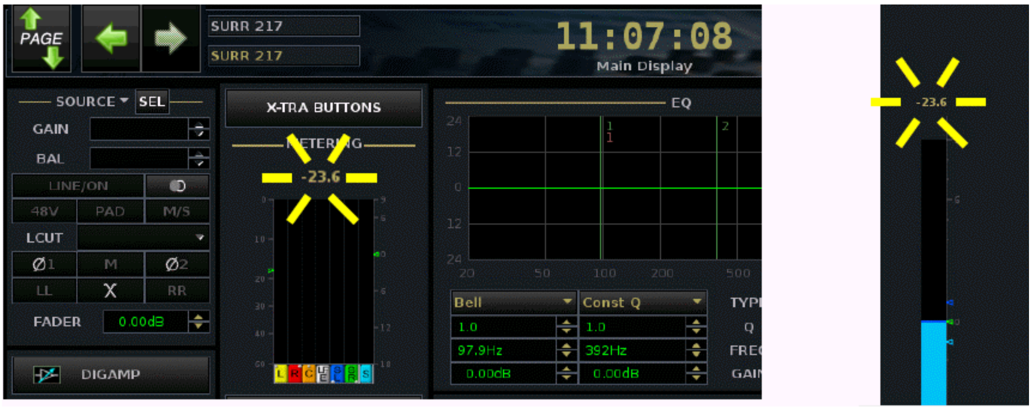
>> Stop (Pause) and Reset

To pause the integration:

1. Make sure that the correct channel is in access (**SEL** lit).

2. Click in the METERING area on the **Main** display and select **Pause integration** from the pop-up window.

As long as the integrated loudness measurement is paused, the reading flashes on the metering displays:



This value represents the average loudness of the channel since you started the measurement.

3. To restart the measurement, select the channel, click in the METERING area of the **Main** display, and select **Start integration**:

The integrated loudness measurement restarts, continuing from before the pause. The readings stop flashing to indicate that integration is active.

> You can pause and restart the integrated loudness measurement as many times as you wish. For example, you may exclude any advert breaks from the programme loudness measurement.

4. If you wish to clear and restart the measurement, then select **Reset integration**.

This clears the current reading, and starts a new integrated loudness measurement.

> As an alternative to using the on-screen METERING pop-up window, you can programme user buttons to start, pause and reset the integrated loudness measurement. These functions are available from the Custom Functions display.

>> Displaying Integrated Loudness in the Title Bar

The integrated loudness measurement for a particular channel, such as main programme, can be displayed in the title bar of the central GUI. This allows the Loudness measurement for a particular channel to remain in view at all times, regardless of which display or which channel is selected.

To change the title bar display:

1. Click on the headline and select **Loudness metering display** from the pop-up window:

The measurement is displayed either as an absolute value in LUFS, or relative to the Target Level in LU, as defined in the System Settings display.

Next assign the summing channel you wish to meter:

1. Select the channel by pressing its fader **SEL** button – in our example, we have selected **SURR 217**, the Surround VCA master for our 5.1 programme.
2. Press the **CHAN/CONFIG** button to view the **Main Display**.
3. Click anywhere in the METERING area and select **Show in Title** from the pop-up window:

The channel is assigned, and if integration has been started, you will see the value update.

You can confirm the assignment by hovering the cursor over the reading; a message appears stating the name of the assigned channel.

Once assigned, you can start, pause or reset the integrated loudness measurement from the title bar:

Meter Pickup Points

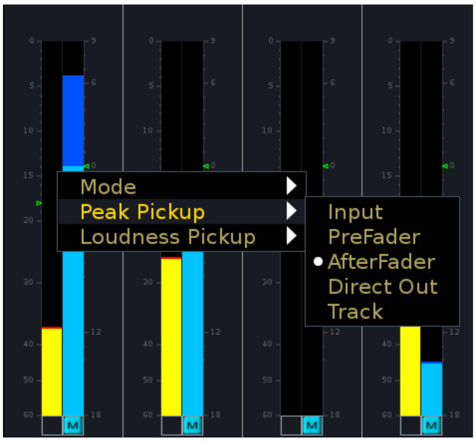
The meter pickup point may be selected independently for the peak and loudness meters. There are several methods you can use (see below). In each case, the pickup options are:

- **INP** – meters the channel input (post the INMIX section).
- **PF** – meters the pre fader signal.
- **AF** – meters the post fader signal.
- **TRK** – meters the track bus output.
- **DIR** – meters the direct output.

➤

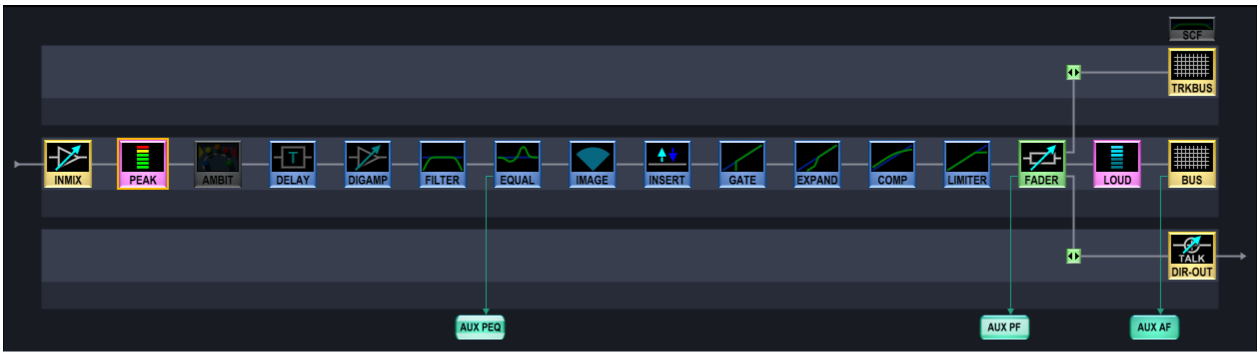
➤ Using the Channel display

Touch the meter on the Channel display and select an option for the **Peak** or **Loudness Pickup**:



➤ Using the Channel Config display

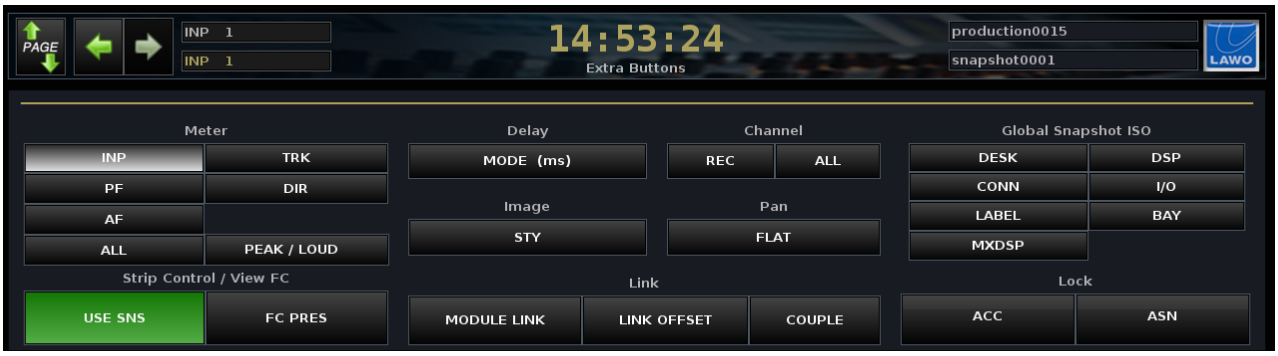
Select the channel you wish to adjust, and open the Channel Config display. Then select either the **PEAK** or **LOUD** modules to move their position:



➢ Using the Extra Buttons display

Select the channel you wish to adjust, and open the Extra Buttons display. Then use the **Meter** select buttons to adjust the pickup. Note that the **PEAK/LOUD** button determines whether you are choosing a pickup point for the peak or loudness meter.

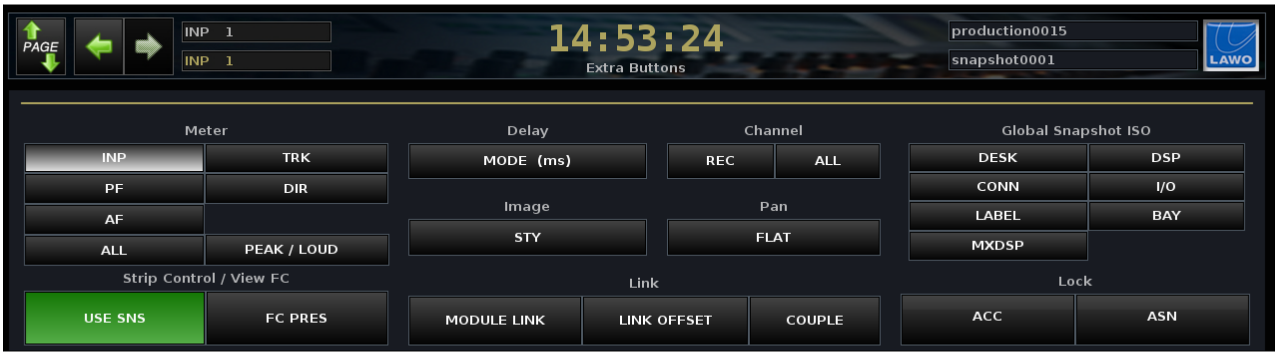
The **ALL** button can be used to switch the metering point for a range of channels.



Switching the Meter Point for Multiple Channels

The METER **ALL** button on the Extra Buttons display can be used to define a cluster of channels so that the meter point is switched across multiple channels.

1. Press the **ALL** button, located on the **Meter** section:



*The **ALL** button flashes and the fader **SEL** buttons across the console flash, in green.*

2. Add channels to the cluster by pressing their fader **SEL** buttons.

*The fader **SEL** buttons turn red:*

3. Now switch the meter point for all channels in the cluster, by selecting a touch-screen button – for example, press **INP**.

The channels are switched to meter the input; channels not in the cluster are unaffected.

The **Meter** touch-screen buttons will continue to switch metering for the cluster while the **ALL** button is lit.

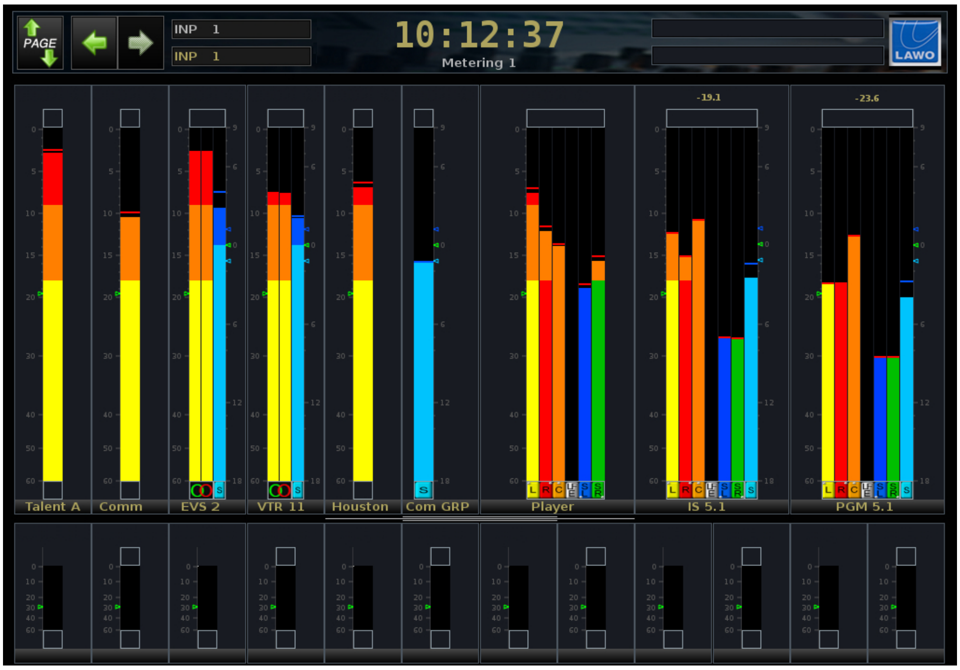
4. To return to individual channel meter switching, deselect **ALL**.

Note that if you re-select the **ALL** button, the same cluster of channels as defined in step 2 will be reinstated.

The Metering Display

The **Metering** display contains four pages of assignable meters.

1. Press the **METER** button, located on the SCREEN CONTROL panel, to view the **Metering** display. Keep pressing to cycle through the four pages - the current page is always named at the top – in our example, **Metering 1**:



2. Select the meter you wish to assign - it is highlighted.

3. Choose the channel you wish to meter by placing it in access (from the ACCESS CHANNEL/ ASSIGN panel).

4. Press the **ASSIGN** soft key.

The selected channel is assigned to the meter; its label updates and the display automatically selects the next meter. This makes it easy to make multiple assignments quickly.

Note that the format of the meter depends upon the channel format - mono, stereo or surround.

5. To remove an assignment, select the meter and press the **CLEAR** soft key.

The **Metering display** assignments are saved within productions, but not in snapshots.

Main Fader Metering

On the **mc²56**, a fifth Metering page provides metering for the main fader strips.

1. Press the **METER** button, located on the SCREEN CONTROL panel, to cycle through:

- **Metering 1** to **Metering 4** – assignable meters.
- **Main Faders** – dedicated metering for the main fader strips:

This display includes the same features as the Channel display.

Chapter 5: The Centre Section

Introduction

This chapter deals with centre section functions, including those available from the **Extra Buttons** display.

Note that several control areas, such as FREE CONTROL PRESETS, are dealt with in other chapters of the manual; please follow the links from the Centre Section Quick Reference guide.

Topics covered in this chapter are:

- Control Room Monitoring
- Talkback
- Overbridge Options
- Main Fader Strips
- VCA Grouping
- Link Groups
- The Couple Group
- Grouping Hierarchy
- Fader Control of Levels
- Labels
- Central User Buttons
- The Extra Buttons display

Control Room Monitoring

The **mc²56** provides two monitor outputs:

- Control Room Monitor 1 (CRM 1) – up to 8-channel, as defined by the global surround format.
- Control Room Monitor 2 (CRM 2) – stereo.

Two stereo headphone outputs follow the control room monitor selectors with separate level adjustment.

The console may also support separate studio monitoring, external AFL/PFL loudspeakers and/or alternate speaker switching depending on the monitoring and I/O configuration.

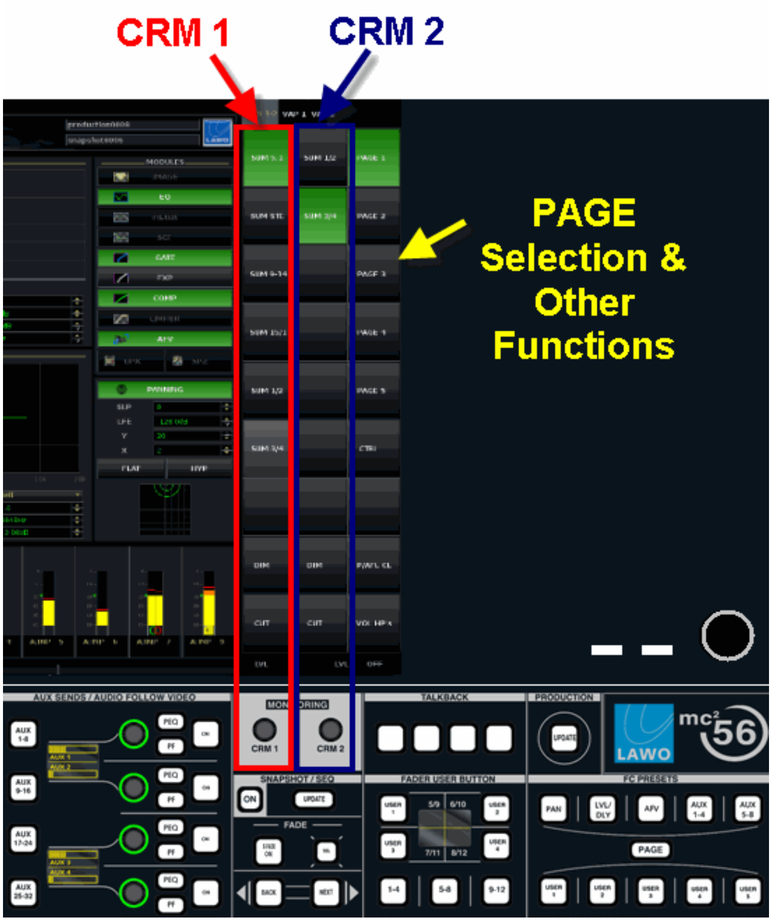
Level controls for CRM 1 and CRM 2 are located on the MONITORING panel. All other controls, including source selection, are programmed onto the Central GUI touch-screen monitoring buttons (displayed when ACCESS/ASSIGN is off).

Monitoring functions and I/O connections are programmed as part of the factory configuration (via TCL files). A description of the default configuration follows. However, you should refer to your system specification for full details.

On the MKII mc²56, the CRM 1 loudspeakers are usually connected to the analogue Line Out 1-8 on the rear panel, see Local I/O.

Monitor Source, Level, Cut & Dim

The touch-screen **MON 1-2** buttons are arranged into three columns. The first two columns select functions for CRM 1 and CRM 2, while the third column provides **PAGE** switching and access to other functions. Touch a button to action the function; it turns green when selected.



The default monitoring configuration provides the following functions:

1. Use the first two columns to select a source, and to **DIM** or **CUT**, the CRM outputs.
2. Use the dedicated rotary controls to adjust the CRM 1 or CRM 2 levels.

*The **LVL** is shown on the touch-screen display; the maximum level is defined by the configuration.*

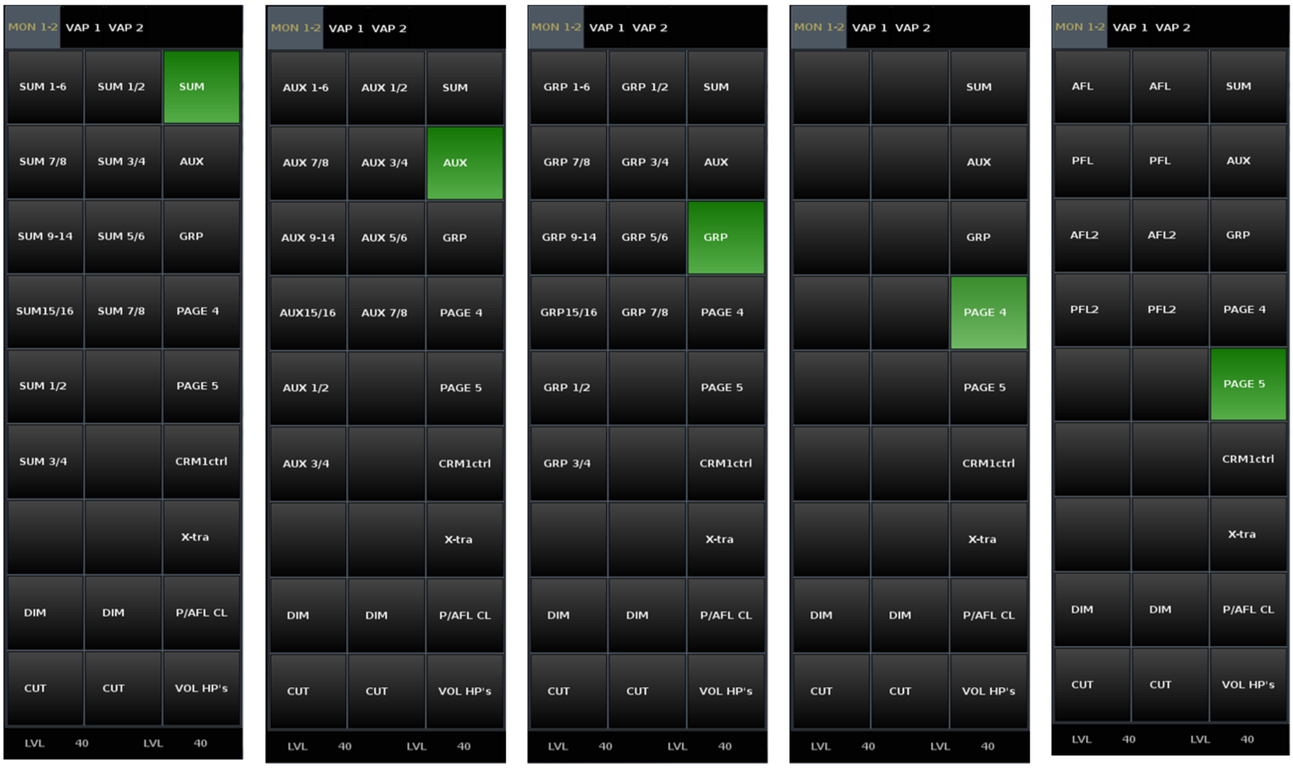
3. Press the **PAGE** buttons (**SUM**, **AUX**, **GRP**, **PAGE 4** & **PAGE 5**) to access monitor sources.
4. Press **CRM1 ctrl** to access additional monitoring parameters.
5. Press the **X-tra** button to access the Extra Buttons display.
6. Press **P/AFL CL** to clear any AFL or PFL selections.
7. Press **VOL HP's** to adjust the headphone 1 & 2 levels from the CRM 1 & 2 controls.

Monitor Sources

The default monitoring configuration provides five pages of monitor sources.

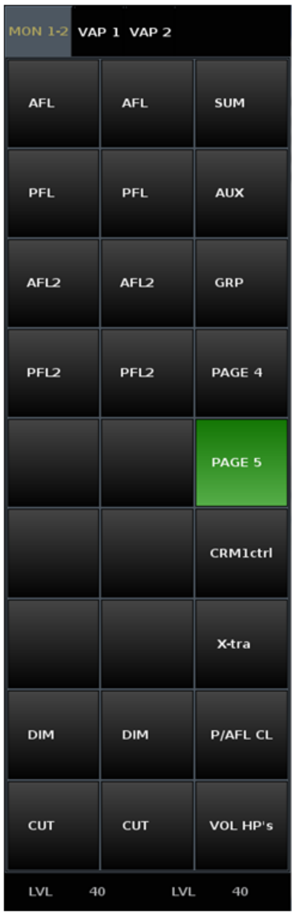
The first three pages provide "hard-wired" access to sums, auxes and groups; page 4 is reserved for external inputs (these will vary from one installation to another); page 5 provides options to monitor AFL and PFL on the CRM 1 and CRM 2 loudspeakers.

> Within page 4, it can be a good idea to have some buttons (labelled **GUI 1-6**, **GUI 1/2**, etc.) which can be accessed from the Signal List display. This allows you to route any matrix source to a monitor source selector button. For more details, please contact your local Lawo representative or email service@lawo.de.



AFL and PFL Monitoring

➢ AFL & PFL to CRM 1/2



In the default monitoring configuration, **PAGE 5** provides options to monitor AFL and PFL on the CRM 1 and CRM 2 loudspeakers. These options define where the listen busses appear, when an AFL or PFL button is active.

Note that the AFL and PFL busses can be split, providing a second output (**AFL2** and **PFL2**) from isolated fader bays. This allows a second engineer to have independent monitoring from the main console in a multi user situation. See Isolating Fader Bays.

For example:

1. Select **AFL** in column 1 to monitor the main AFL bus on CRM 1.
2. Select **PFL2** in column 2 to monitor PFL from isolated bays on CRM 2/Headphones 2.

Note that ISO AFL2/PFL2 (in the **System Settings** display) must be active to split the listen busses.

Note that AFL 1 is a surround bus (up to 8-channels), while AFL 2, PFL 1 and PFL 2 are stereo busses.

➢ AFL & PFL to External

To route AFL or PFL to an external output, use the Signal List display. You will find all the listen bus outputs under the **Bus Out** Source Directory:

➢ AFL & PFL Levels/Metering

To change the level of a listen bus, or to assign a listen bus to a meter, the AFL and PFL masters can be put into access. You will find the AFL and PFL busses after VCA channel 128 in the access channel sequence.

➢ AFL & PFL Options

A variety of AFL, PFL and Solo button options are available from the **System Settings** display. Or, AFL may operate as Solo-in-place.

Headphones

The MKII mc²56 provides two stereo headphone outputs supported by the local I/O board. These are wired to the stereo phones connectors on the front buffer of the console. On larger frames, two additional phones connectors (HP3 & HP4) operate in parallel with HP1 & HP2.

The default monitoring configuration sets HP 1 to follow the CRM 1 monitor source selector, and HP 2 to follow CRM 2.

1. Press the **VOL HP's** touch-screen button to adjust the headphone levels from the CRM 1 & 2 rotary controls.

CTRL Parameters

The **CRM1 ctrl** page provides access to additional monitoring parameters. For example, to mute individual loudspeakers, check mono compatibility, set the monitor dim level, etc.

In each case:

1. Touch a button to action its function - for example, press **L** to mute the left speaker.
2. To adjust a parameter value:
   - Press **SET** to enter the "set parameter" mode, and select a function (e.g. **DIM**).

   The touch-screen updates to show the parameter you are adjusting - for example, Dim LVL.

   - Use the CRM rotary control to adjust the parameter value.
   - Remember to deselect the **SET** button to exit the "set parameter" mode after each operation.

All possible parameters are described over the next few pages. Note that, depending on your monitoring configuration, not all may be available.

➢ Dim Level

The console features two independent monitor dim settings:

- Monitor Dim – actioned by pressing the **DIM** button.
- Talkback Monitor Dim – actioned by a user button or external GPI trigger (defined in the monitoring configuration). This can be used to automatically dim the main monitoring when you press a Talkback button.

To adjust the amount of dim:

1. Press **SET** and **DIM** - the touch-screen displays **DIM LVL**. Use the CRM control to adjust the monitor dim level.
2. Press **SET** again to set the amount of talkback dim - the touch-screen displays **TB DIM LVL**. Use the CRM control to adjust the talkback dim level.

➢ Mono Left/Right

1. Press the **MONO** button to mono the Left and Right monitor outputs to both speakers. This automatically applies a 3dB reduction to the left and right channels to compensate for the mono sum.
2. To adjust the mono gain reduction, press **SET** and **MONO** - the touch-screen displays **TRIM MONO**. Use the CRM control to adjust the mono trim level.

➢ Stereo Monitoring Functions

1. Press **LtoB** to monitor the Left CRM output on both left and right speakers.
2. Press **RtoB** to monitor the Right CRM output on both left and right speakers.
3. Press **PH L** to reverse the phase of the Left CRM output.
4. Press **PH R** to reverse the phase of the Right CRM output.

Note that both phase left and phase right buttons are available to deal with phasing issues on either speaker.

➢ Left/Right Monitor Balance

The monitor balance control allows you to offset the Left and Right CRM levels to compensate for poorly aligned stereo speakers.

1. Press **SET** - the touch-screen displays **BALANCE**. Use the CRM control to adjust the left/ right balance.

Balance may be adjusted from -20dB to +20dB.

➢ Individual Loudspeaker Mutes

The **L**, **C**, **R**, **SL**, **SR** and **LFE** touch-screen buttons are used to mute the individual surround speakers and select parameters for setting balance and volume trim settings.

1. To mute a speaker, touch the corresponding mute button - the mute button turns red when selected.

Note that not all mute buttons may be active depending on your choice of surround format.

➢ Individual Loudspeaker Level Trims

Each of the CRM outputs may be individually trimmed to help align your surround loudspeakers.

1. Press **SET** and one of the speaker mute buttons (**L**, **C**, **R**, **SL**, **SR** or **LFE**) - the touch-screen updates to display the speaker trim level (e.g. **TRIM FL** for Trim Front Left). Use the CRM control to trim the speaker level.

   Levels may be trimmed between -128 and +15dBr.

   For more details on the available surround formats and how they correspond to the front panel mute buttons, see the Appendix: Surround Levels.

➢ Individual Loudspeaker Solos

The **L**, **C**, **R**, **SL**, **SR**, and **LFE** touch-screen buttons may be used to solo individual surround loudspeakers. The solos are additive.

1. To solo a speaker, touch the **SOLO** button followed by the corresponding mute button - the mute button turns green.

   Note that if a speaker mute button was activated before the **SOLO** mode, then if you try and solo the same speaker its LED turns orange to indicate that you are now attempting to solo a muted speaker!

➢ Alternate Loudspeaker Switching

1. Press the **ALT** touch-screen button to cut the main speakers and switch the CRM1 monitoring output to an alternate set of speakers.

Talkback

The MKII **mc²56** includes an integrated talkback microphone preamplifier and four programmable TALKBACK user buttons:

Depending on your system specification, talkback may be connected in one of three ways:

- To the integrated talkback mic preamp - described below.
- To the optional INTERCOM user panel (962/16) - with integrated talkback mic preamp and return talkback loudspeaker.
- Externally, to any matrix source - for example, to connect talkback from an external communications system.

The factory default is to use the integrated talkback mic preamp. For details on other options, see Local I/O: Jumper Switch Positions.

The Integrated Talkback Mic Preamp

The female XLR connector, shown above, feeds the integrated talkback mic preamp mounted inside the control surface. This, in turn, feeds **Line input 16** of the local I/O (according to the local I/O board jumper switch positions).

The XLR socket is wired directly to the microphone preamplifier, and provides 48V phantom power.

The mic preamp gain is adjusted by a trim potentiometer; the trimmer is accessible via a small access hole next to the XLR connector.

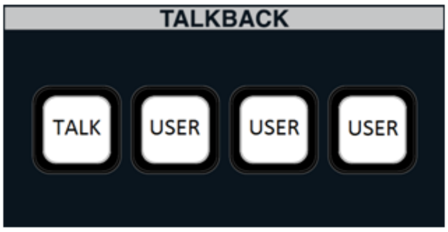
The mic preamp contains a compressor/limiter; the output gain of the limiter is fixed to +15dBu.

Note that if your system's operating levels are set for a **Maximum Analogue Level** > +15dBu, then the output level from the talkback mic preamp can seem low (due to the analogue limiter). If this is the case, increase the level by adjusting the I/O DSP **Volume** for Line input 16 of the local I/O.

A line level output from the mic preamp, prior to A-D conversion, is provided via the **TBK** connector on the control surface rear panel.

Talkback Switching

Once connected, talkback switching is programmed from the Custom Functions display. This allows you to switch talkback from the fader strip, from the Central User Buttons or from the TALKBACK panel:



Note that the four TALKBACK buttons can be assigned to *any* user button function, not only talkback switching.

By default, the first button (**TALK**) is programmed to switch your talkback source to the N-1 bus of the channel in access.

Overbridge Options

Space is available in the Overbridge to fit either RTW metering (shown above) and/or a Lawo User Panel.

The permitted variations are:

Part Number	RTW	User Panel	Fitted
958/90	No	No	Blank Panel
958/91	Yes	No	TM 9 (shown above)
958/92	Yes	Yes	TM 7 + User Panel
958/93	No	Yes	Blank Panel + User Panel

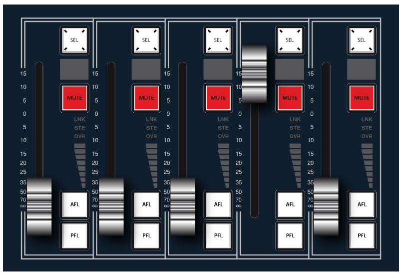
When the RTW TM 7 or TM 9 are fitted, they connect to the AES3 in/out 5-8 of the local I/O.

The default configuration usually sets the RTW to automatically follow the CRM 1 monitor source selector.

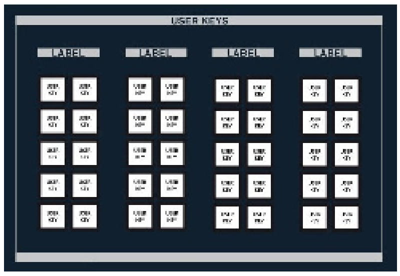
The Overbridge User Panel options are:

Part Number	User Panel	Description
962/29	REVEAL FADER	5 dedicated faders for revealing surround slaves.
962/14	USER KEYS	40 user buttons configured from the Custom Functions display.
962/16	INTERCOM	integrated loudspeaker and internal talkback microphone, see Local I/O Wiring.
962/18	AUTOMATION	timecode automation controls.
962/15	USER CONTROLS	8 rotary controls defined by the factory configuration.

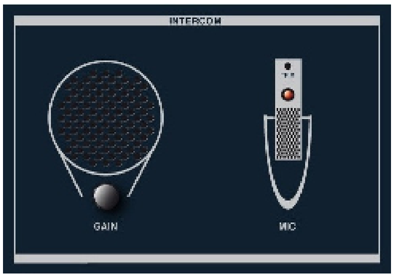
REVEAL FADER



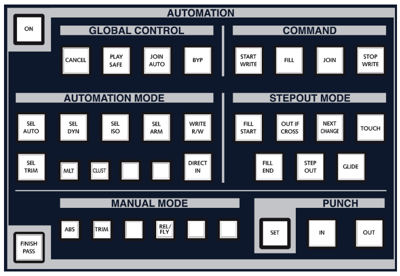
USER KEYS



INTERCOM



AUTOMATION



USER CONTROLS

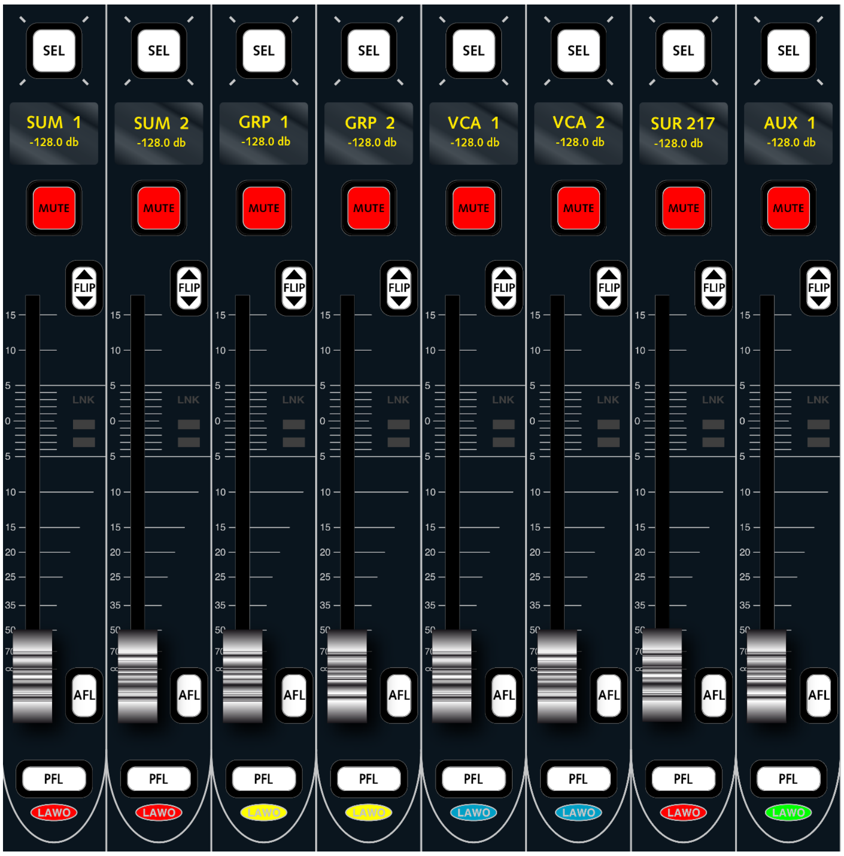
Main Fader Strips

The main fader strips in the centre section may be used to control any channel type – input channels, monitor return channels, groups, sums, auxiliaries, VCA masters, surround masters – in exactly the same way as a channel fader strip.

The only differences in operation are that main fader strips do not have dedicated input control, Free Controls, user buttons or **Channel** display metering. However, metering is provided by the mini main faders display, or the Main Fader Metering page.

You may also independently Bank or Layer switch the main fader strips, by using the **MAIN BAY** button, see Bank and Layer switching.

Typical applications for the main fader strips include providing master channel control for sum, group, VCA or aux masters:



VCA Grouping

The console supports up to 128 VCA masters.

You may assign any number of channel or main fader strips to each VCA. This provides the ability not only to control input channels but also groups, sums, aux masters, GPCs and surround VCAs from a single fader strip.

VCA assignments are stored within both snapshots and productions.

The channels assigned to a VCA can be on any Bank or Layer. This allows you to have a single VCA master controlling a number of slave channels on a "hidden" Bank or Layer.

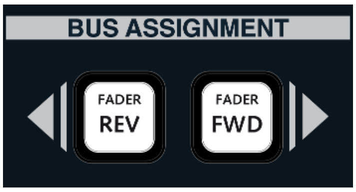
Note that:

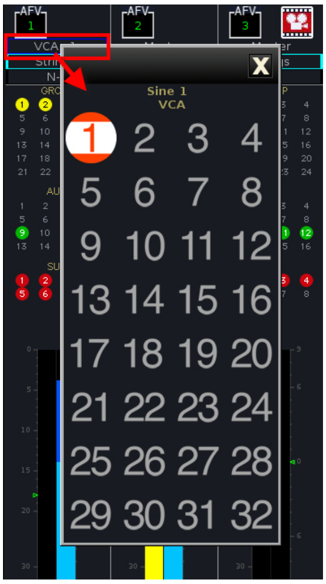
- VCA groups may use moving or non-moving slave faders, defined by the Relative Slave Faders option in the System Settings display.
- The master/slave behaviour varies depending on the parameter, see the Appendices.
- A channel may only ever be assigned to a *single* VCA.
- A VCA master *cannot* be assigned to another VCA.

You can assign a VCA master to a Link group. This can be used to create groups within groups if required. See Grouping Hierarchy.

Creating a VCA Group

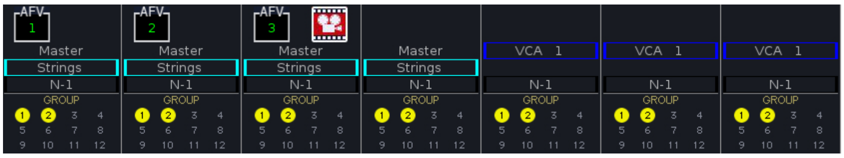
You may use any one of the following bus assign methods to assign channels to a VCA master:





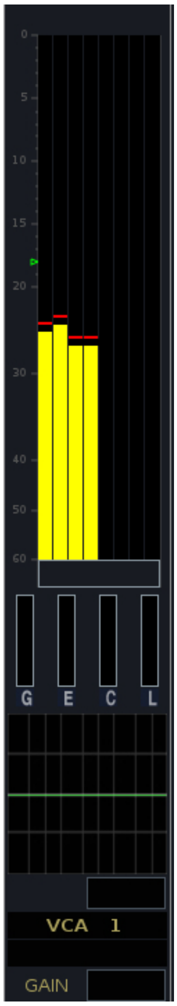
- Forward Assign - put the slave channel into access; press **FADER FWD** and then select the VCA master.
- Reverse Assign - put the VCA master into access; press **FADER REV** and then select the slave channels.
- Channel Display - touch **Master** at the top of the slave channel's meter, and select a number (the first 32 VCA masters are displayed).

*VCA assignments are shown in the **Master** field at the top of the Channel Display - in our example, the last three channels are assigned to **VCA 1**:*

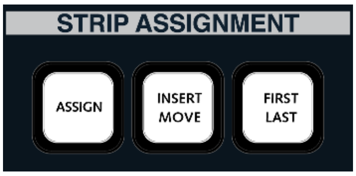


Working with VCAs

Assign the VCA master to a fader strip in the usual manner:

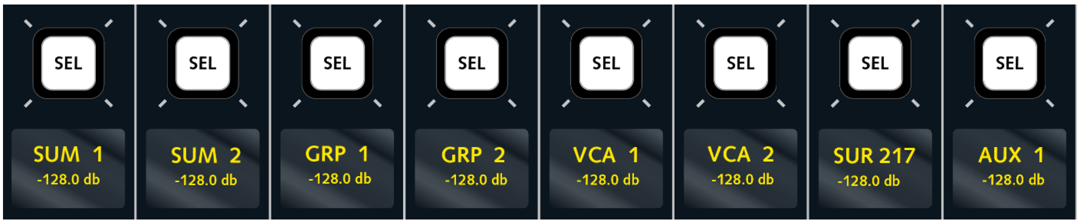


1. Select **VCA** and **1** from the ACCESS CHANNEL/ASSIGN panel.
2. Press **ASSIGN** located on the STRIP ASSIGNMENT panel:



3. Then press the fader **SEL** on a channel or main fader strip.

The fader strip updates, and you will see metering for the first 8 slave channels on the Channel display.



Note that the meters display the lowest to highest fader strip slave from left to right. For example, if the VCA master is controlling fader strips 1, 3, 5 and 6, then you will see the slave channels in that order on the VCA master meters.

Also note that the pickup point for the slave channel metering is set from the VCA master. Therefore, make sure you have the VCA master is in access when changing the meter pickup point.

4. You can adjust the level of the slave channels from the VCA master fader, and other parameters from the Free Controls or Central Control Section.

If the Relative Slave Faders option is set to non-moving faders, then the slave faders remain stationary when the master moves; you will see the level change applied to each slave channel in its fader label display.

The master/slave behaviour varies depending on the parameter. For example, main level and input gain are controlled relatively so that you can offset the slave positions; EQ frequency and Q are *always* set by the master (absolute), so that any change is inherited by all slaves; the MUTE is switched ON from a Surround VCA master but not OFF. For full details on all parameters, see the Appendices.

Renaming a VCA Master

The user label of the VCA master may be edited from the Title Bar when the VCA master is in access.

Revealing the VCA Slaves

Use the REVEAL function to temporarily reveal VCA slaves onto fader strips.

This is particularly useful if the slave channels are on a hidden Bank or Layer, as you can use Reveal to quickly access the slaves to offset fader levels and other relative parameters.

Link Groups

In addition to VCA grouping, the console supports link groups.

The key differences are:

- Every channel within a link group is a master. For example, moving any of the 8 faders within a link group adjusts the level of all 8 channels.
- The link can apply to all channel parameters or to individual processing sections – for example, to link EQ sections but not faders.

An unlimited number of link groups may be created.

Any number of channels may be assigned to each link group, including channels of a different DSP type (e.g. inputs and groups).

Link groups are stored within both snapshots and productions.

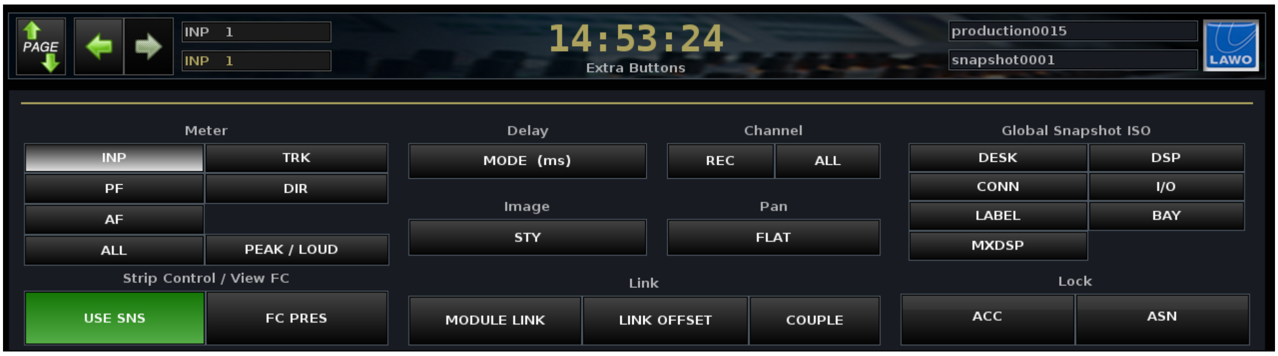
The channels assigned to a link group can be on any Bank or Layer. This allows you to assign one member of the link group to your working Bank/Layer, and have it control other channels on a "hidden" Bank or Layer.

Note that:

- When working with Link groups, faders are *always* moving.
- The master/slave behaviour varies depending on the parameter, see the Appendices.
- A channel may be assigned to both a link group and a VCA master, see Grouping Hierarchy.

Creating a Link Group

1. Select one of the channels you wish to link - either by pressing its fader **SEL** button or using the ACCESS CHANNEL/ASSIGN panel.
2. Open the Extra Buttons display, and select the on-screen **MODULE LINK** button:



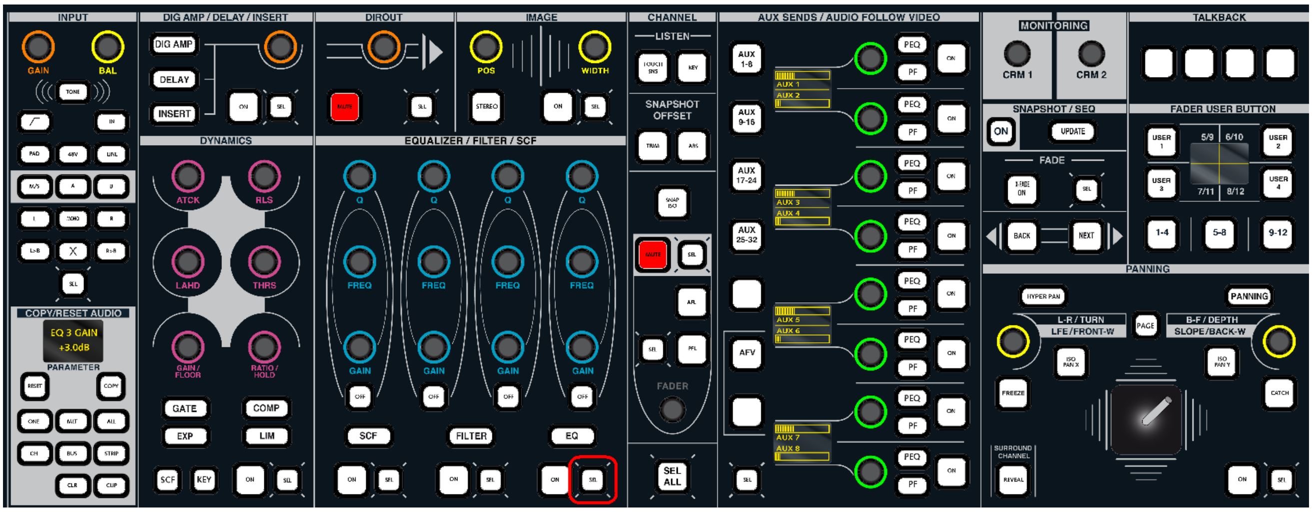
*The fader **SEL** buttons across the console flash, in green.*

3. *BEFORE* selecting any faders, first select the audio module(s) you wish to link, by enabling the **SEL** buttons on the Central Control Section.

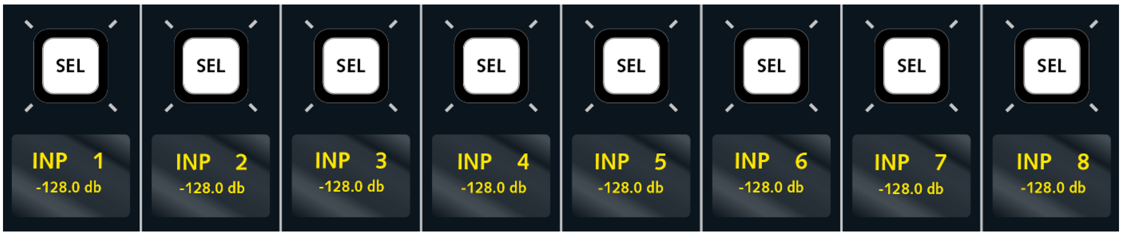
Note that you may select any audio module on the Central Control Section using the **SEL** buttons. You may also link bus assignments by selecting **BUS** on the COPY/RESET AUDIO panel. See Selecting Channel Parameters.

To clear down any existing selections, toggle the **SEL ALL** button (this selects and then deselects all modules). This ensures that there no "hidden" selections.

Then turn on the **SEL** button(s) to make your selections. For example, press **SEL** beside the EQ section to link EQ:

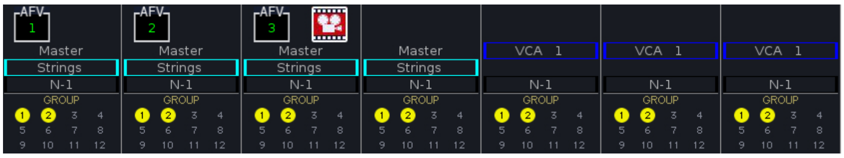


4. Now press the fader **SEL** buttons on the channels you wish to link:



*The selected audio modules across the selected channels are linked; the fader **SEL** buttons stop flashing and change colour, from green to red; the **LNK** status LED lights on the fader strip; and the Channel display updates:*

In our example, the first four channels are part of the link group named "Strings":

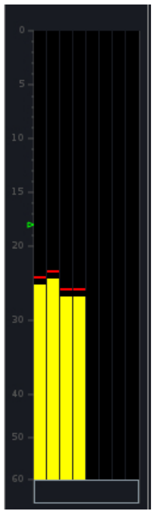


5. Deselect **MODULE LINK** to complete the operation.

6. Repeat the steps to create additional link groups.

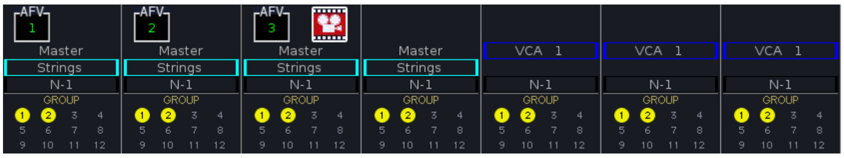
7. To edit an existing link group, select any channel within the link, press **MODULE LINK**, and adjust the Central Control Section and/or fader **SEL** buttons. (To clear the link group, deselect all the fader **SEL**s within the group.)

Link Group Options



Link groups are indicated at the top of the Channel display where you will see:

- The link group name – e.g. **Strings**.
- The link group colour code – e.g. turquoise.



In addition, you can choose to meter the first 8 linked channels on each of the grouped channels. This is particularly useful if you want to assign one member of the link group to your working Bank/Layer, and move the other members of the link to a "hidden" Bank or Layer.

Each of these options is edited from the **Channel Config** display:

1. Select one of the linked channels, either by pressing its fader **SEL** button or using the ACCESS CHANNEL/ASSIGN panel.
2. Press the **CHAN/CONFIG** button, located on the SCREEN CONTROL panel, to view the **Channel Config** display.

Our example shows that the channel in access (INP 1) is assigned to Link group 1:

3. To edit the link group name, click within the **LINK 1** field and type a new name – e.g. **Strings**.
4. To assign a colour, use the drop-down menu:

5. To enable multi-channel metering on each of the link group channels, select the **Link metering** option:

Working with Link Groups

Once a link group is created, any channel within the link group may be used as the master; moving a control adjusts the parameter across all the linked channels. The master/slave behaviour varies depending on the parameter, see the Appendices.

Any offsets which were present when the link was created are retained. To adjust offsets, there are two methods:

Touch-Sense

1. Hold the first control so that its touch-sense is active – e.g. fader 1.
2. While holding fader 1, adjust another control within the link group – e.g. fader 5.

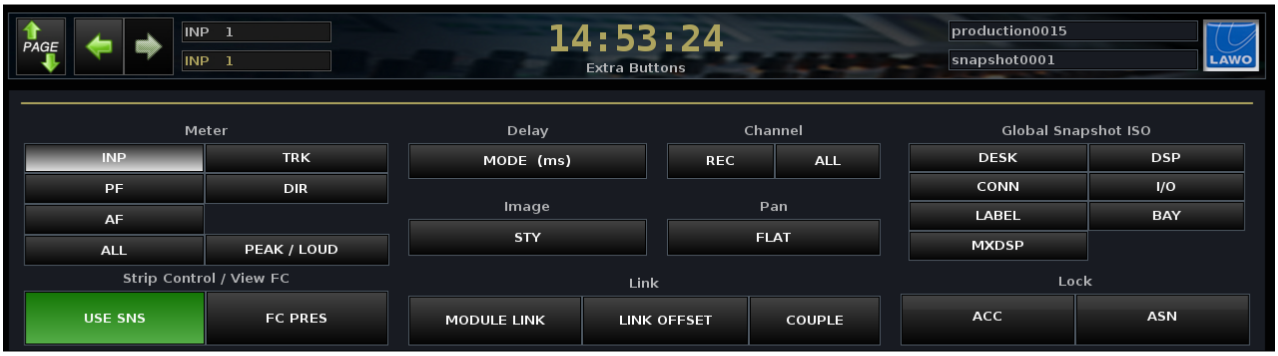
You are now adjusting the offset position of fader 5 relative to the rest of the link group.

You can use this method for any touch-sensitive control: fader or rotary encoder.

Link Offset

Alternatively, if you want to change the offsets for lots of controls it is better to use **LINK OFFSET**. This temporarily suspends the link group to allow adjustments to individual control positions:

1. Open the Extra Buttons display, and select the on-screen **LINK OFFSET** button:



The button flashes to indicate that it is active.

2. Now adjust the position of your controls.

While **LINK OFFSET** is active, any link groups are temporarily suspended. This allows you to completely change the balance within a group quickly and easily.

3. When you are happy, deselect **LINK OFFSET** or press **ESC** on the SCREEN CONTROL panel.

The link groups now return to their normal “grouped” mode of operation.

Managing the Link Group Numbering

Every time you link channel parameters, a link group number is automatically assigned by the system. Thus, the first set of linked parameters form link group 1, the second link group 2, and so on.

For most operations this is fine. However, there are a few instances when you may need to select a different link group number. One example is if you wish to recall a snapshot or automation with stored links to one part of the console, while retaining an existing link group (with their channels in SNAP ISO).

If the link group on the console is Link 1, and the stored snapshot already uses Link 1 then there will be a conflict and the link groups will not operate correctly. The solution is to change the link group currently in use on the console. You can do this as follows:

1. Press the fader **SEL** button on any channel within the link group.
2. Select **MODULE LINK** (from the Extra Buttons display).

This puts the link group into edit mode.

3. Now locate the **BUS ASSIGN** button on the ACCESS CHANNEL/ ASSIGN panel.
4. Press the **BUS ASSIGN** button until it flashes.

In this mode, the NAME and LABEL display shows the current link group number – e.g. LINK 1.

5. Press the **NEXT** or **PREV** buttons to increase or decrease the link group number.
6. When you are finished, deselect **MODULE LINK** to exit the edit mode.

You can verify the link group number from the **Link** field in the **Channel Config** display:

The Couple Group

In addition to VCA and link groups, the console supports a single couple group which is ideal for adjusting a parameter across a range of channels.

The key difference between link groups and the couple group is their application:

- **Link Groups** – are ideal for "permanent" grouping where channels need to remain linked throughout a scene or production.
- **The Couple Group** – is ideal for temporary operations such as adjusting the mic gain across a range of channels.

The couple group is very similar to link groups but has some important differences:

- There is only *one* couple group – you cannot create multiple couples. (Use links whenever you need multiple groups.)
- Channels assigned to the couple group *must* be on adjacent fader strips – i.e. you cannot couple non-consecutive faders.
- The couple group links *all* channel parameters.

Using the couple group is often faster than copying channel parameters. For example, to apply an EQ setting across a range of channels; assign the channels to the couple group; adjust the EQ parameters on any of the coupled channels, then dissolve the couple.

The coupled channels can be of any channel type, as long as they are assigned to adjacent faders.

Note that:

- When working with a couple group, faders are *always* moving.
- The master/slave behaviour varies depending on the parameter, see the Appendices.
- Whenever a channel is part of the couple, all other groups are temporarily suspended. This means that the couple can be used at any time and across all types of channels and groupings. See Grouping Hierarchy.

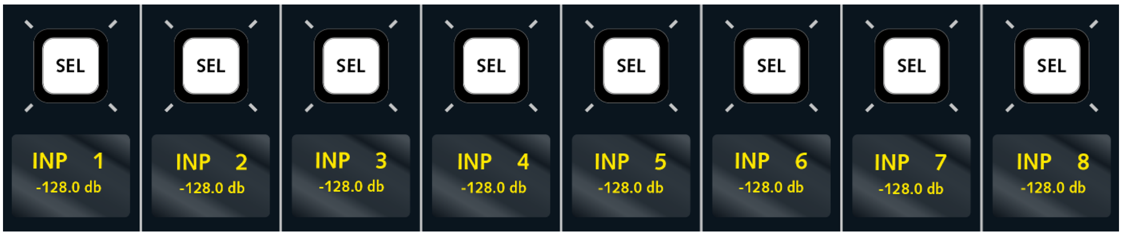
Creating a Couple

To create a couple, there are two methods:

Fader SEL

This method can only be used if the Direct Couple option is enabled in the **System Settings** display. By default, this option is turned on. With the option enabled:

1. Press and hold the fader **SEL** button on the first channel you wish to couple:



2. Then press the fader **SEL** button on the last channel.

*All channels within the range, including the first and last, are assigned to the couple group. Their fader **SEL** buttons turn red to indicate that the couple is active.*

3. Now adjust a parameter – for example, turn the EQ gain control on any channel within the couple.

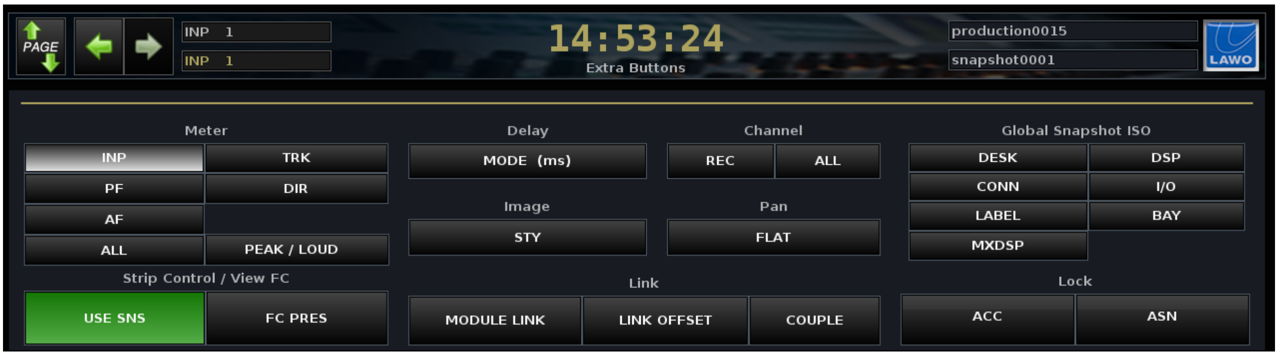
All EQ gains across the couple group are adjusted; any offsets are retained.

4. To dissolve the couple, press the fader **SEL** button on any fader strip outside the couple group range.

The COUPLE Button

Alternatively, if you prefer not to use the method above:

1. Open the Extra Buttons display, and select the on-screen **COUPLE** button:



The button flashes to indicate that it is active.

2. Press and hold the fader **SEL** button on the first channel you wish to couple:

3. Then press the fader **SEL** button on the last channel.

All channels within the range are assigned to the couple group.

4. Make your parameter adjustment – for example, turning on Aux 5 will enable aux 5 for all channels in the couple.

5. To dissolve the couple group, deselect the **COUPLE** button.

Grouping Hierarchy

The hierarchy of the different group types in the system is as follows:

- Couple
- Link
- VCA
- Surround VCA

This means that:

- Whenever a channel is part of a couple, all other groups are temporarily suspended. This means that the couple can be used at any time and across all types of channels and groupings.
- A channel can be assigned to *both* a link group and a VCA. For example, the vocal soloists within a choir can be controlled by a link group for the choir, and by a separate VCA for the soloists; the link group takes overall priority.
- You can assign a surround VCA to a VCA master, and a VCA master to a Link group. This can be used to nest groups if required.

Fader Control of Levels

The FADER CONTROL buttons temporarily switch the faders to other objects than channel level. For example, to control auxiliary send levels from the channel faders to set up a quick headphone balance.

Note that the buttons switch all faders - channel and main - globally across the console.

There are two different assignment methods:

- Select and scroll through the available level objects - for example, to scroll through Aux Sends 1 to 32.
- Pre-select a level object, by copying it to the FADER CONTROL clipboard, and then assign it to the faders. This allows you to switch directly to say Aux Send 28 without scrolling.

➢ To scroll through the level objects:

1. Select a level object by pressing:

- **AUX SENDS** – Send levels for the console's 32 auxiliaries.
- **DIG AMP** – Digital amplifier gain.
- **INSERT SEND** – Insert send level.
- **DIR OUT** – Direct output level.
- **AFV** – Audio Follow Video On level.
- **LFE** – Low Frequency Effect (Subwoofer) level, see Panning.

The name of the selected object (e.g. ***AUX 1****) flashes in the clipboard display and the faders move to reflect the current Aux send values.*

The fader label displays also update to show the parameter name; the name flashes to warn you that you are now controlling something other than channel level!

2. If you have selected **AUX SENDS**, then use the Left or Right arrow buttons to scroll up or down through the 32 sends.

Each time a new send or level object is selected, the faders across the console update accordingly.

3. Move the faders to adjust the levels.

4. When you have finished, deselect the level object button to return the faders to their normal operation.

➢ **To pre-select a level object:**

1. Select the level object by touching its rotary control on the Central Control Section. For example, touch Aux send 1 gain to place AUX 1 into the FADER CONTROL clipboard.

2. When you are ready to switch the object to the faders, press **USE TYPE**.

The faders are now controlling the clipboard object.

3. To switch back to normal fader level operation, deselect **USE TYPE**.

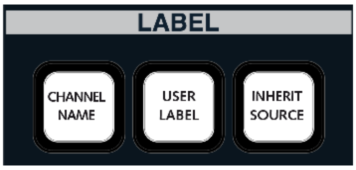
Note that the FADER CONTROL and COPY/RESET AUDIO CLIPBOARD are one and the same. Therefore, be aware that if you update the clipboard to assign say Aux send 3 to a free control, this also puts Aux send 3 into the FADER CONTROL clipboard.

You may only assign valid level objects to the faders; if you try to select EQ gain, for example, then the **USE TYPE** button cannot be enabled.

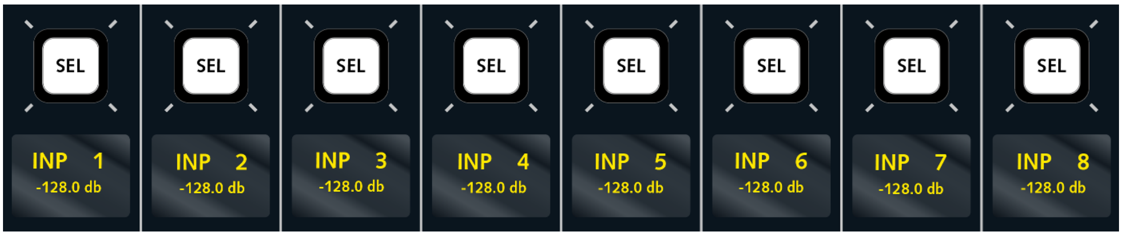
You may Bank and Layer switch while in the FADER CONTROL mode to gain access to all assigned levels across multiple banks and layers of channels.

To quickly set *all* aux sends from a channel to the same level (e.g. unity gain), switch **AUX 1** to the faders and set the fader to 0dB. Keep touching the fader and scroll through the aux sends using the Left and Right arrow buttons – each aux send is set in turn to 0dB.

Labels



The LABEL buttons change what is viewed on the fader strip labels (and the label fields in the Title Bar, Channel display and Signal List display):



You can choose one of three options; each switches the labels globally across the console:

- **CHANNEL NAME** = the system name of the channel (e.g. INP 1).
- **USER LABEL** = the user label given to the channel (e.g. GUEST).
- **INHERIT SOURCE** = the user label given to the source which is routed to the channel (e.g. MIC 1).

Select **CHANNEL NAME** while preparing the console. This enables you to easily view where you are assigning your input channels, group masters, auxiliary masters, VCA masters, etc.

Select **INHERIT SOURCE** once your console fader strips are configured, to confirm that the correct signal routing is made.

For example, if you have used the names **Com1**, **Com2** and **Guest** as channel user labels, and **Mic1**, **Mic2** and **Mic3** as source labels for microphone inputs, then when you begin to route sources to channels, you can use the Fader Label displays to confirm that the correct routes have been made - if you look across the console and see **Mic1**, **Mic2** and then **Guest**, you know that first two mic channels are correctly routed, but the **Guest** channel has no source.

Note that if no source is assigned to an input or monitor channel, then the **USER LABEL** is displayed.

Similarly, if the source label does not apply to the channel - for example, you cannot route signals to groups, sums, auxes, VCAs, Surround VCAs or GPCs - then you will see the **USER LABEL** even if **INHERIT SOURCE** is selected.

Also note that for an individual source, the inherit function may be inhibited. This must be performed within the factory configuration. Please consult your console specification for details.

The following provides further information on each of the LABEL options and how they can be edited:

➢ CHANNEL NAME

This is the fixed system name for the DSP or control channel assigned to the fader strip. For example, **INP 1** for input channel 1, **GRP 4** for group channel 4, **VCA 7** for VCA master 7, etc.

You cannot edit this name.

Note that for DSP channels (inputs, monitors, groups, sums and auxes), this is the same as what you see in the **Name** field on the Signal List display.

➢ USER & SOURCE LABELS

Both of these labels may be edited.

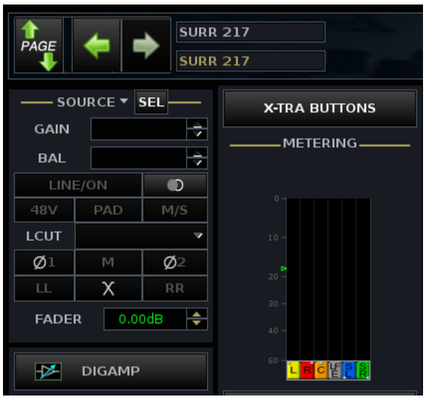
The user label names the channel (e.g. **GUEST**), while the source label names the signal routed to the channel (e.g. **MIC 1**).

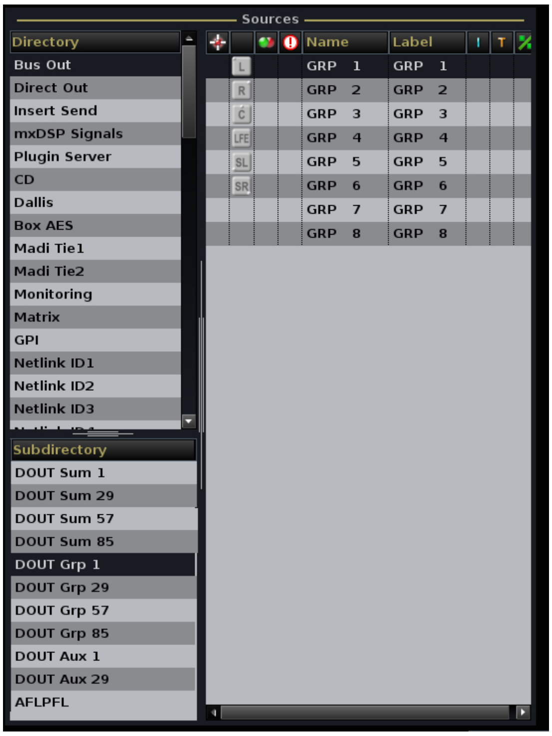
For input and monitor DSP channels, use the Source **Label** field (in the Signal List display) to edit your source labels, and the Destination **Label** field to edit the channel user labels:

For groups, sums, auxes, VCAs, Surround VCAs and GPCs, the only relevant label is the user label (as you cannot route signals directly to these channels).

For groups, sums and auxes, use the Source **Label** field (in the Signal List display) to edit the channel user label.

For control channels (VCAs, Surround VCAs and GPCs), click in the label field in the Title Bar to edit the channel user label. (Note that the centre section LABEL buttons must be switched to **USER LABEL**).





Central User Buttons

The centre section includes 9 user buttons:

Their functions are programmed from the Custom Functions display.

The classic mc²56 supports 6, as opposed to 9 Central User Buttons.

The Extra Buttons Display

The **Extra Buttons** display contains touch-screen functions which you may only need from time to time.

There are several ways to access the display:

1. From the Page menu – select **Pages** -> **Extra Buttons**.

2. From the Central GUI - select **X-TRA** from the **MON 1-2** touch-screen buttons on the right of display, or select **X-TRA BUTTONS** (above METERING) on the Main display:

3. From a central USER BUTTON:

Each function is described in detail elsewhere in this manual, so please follow the links for full details:

- **Meter** – selects the meter pick-up point for the channel in access.
- **Strip Control/View FC** – sets whether the Free Control displays show parameter functions (e.g. Aux 5) or values (e.g. +4dB).
- **Delay MODE** – cycles the delay mode for the channel in access between milliseconds, frames and meters.
- **Image STY** – switches the Image section for the channel in access between new and old styles.
- **Channel REC and ALL** – global send/return switching for monitor channels.
- **Pan FLAT** – affects level compensation applied to the centre channel when panning across the front surround channels.
- **Link** – used for link groups and coupling.
- **Global Snapshot ISO** – these buttons allow you to isolate different console elements from a snapshot recall.
- **Lock ACC** - protects the channel in access.
- **Lock ASN** - locks the STRIP ASSIGNMENT and Forward and Reverse BUS ASSIGNMENT buttons.
- **EQ** – changes the EQ type for the EQ, Filter or Sidechain filter sections.
- **Aux 1-8** – changes the aux send pickup point. Eight aux sends are displayed at a time, and follow the front panel **AUX 1..8**, **AUX 9..16**, **AUX 17..24** and **AUX 25..32** buttons.

Chapter 6: Console Reset

Introduction

This chapter explains the reset capabilities of the console and covers the operation of productions, snapshots, sequences, presets and how to import/export data.

Topics covered in this chapter are:

User Data: Overview

One of the major benefits of the **mc²56** is the ability to store and recall all the settings for a live show or type of application.

Productions

Productions form the top level for user data storage and store *all* the settings required for a production or type of job.

Productions store everything included in a snapshot, plus lower level settings such as the DSP configuration and system options. As a result, loading a production may cause a brief interruption to audio, and should *not* be used during a show. Instead, use snapshots to recall settings while live on-air.

Snapshots

Within each production, folders are created to store snapshots.

Snapshots store different mixes for recall before or during the show. For example, to recall a different mix for each band in a live entertainment show, or to recall scene changes during a live theatre production. To manage snapshot recall, snapshot isolate and filtering may be applied to protect channels or elements of the desk.

Sequences

Sequences are provided for convenient recall of snapshots during a live broadcast or theatre production.

A sequence is a list of snapshots which can be loaded in sequence during a live show. The transition between snapshots in a sequence can be cross faded if required. In addition, offsets can be applied to deal with last minute changes such as a change of artist. Note that the sequence itself does not store any settings, but simply creates a list of pointers to snapshots stored within the production folder.

Presets

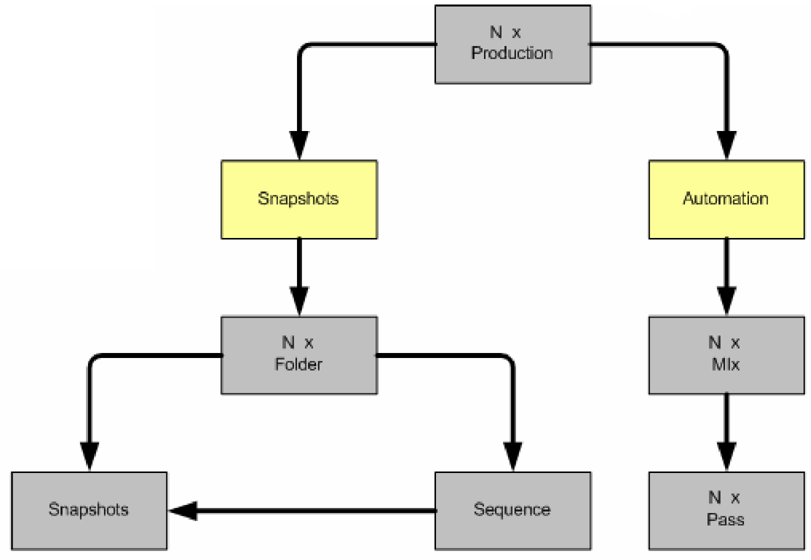
Presets are stored independently of productions, and save and load settings for processing modules (EQ, Gate, Compressor, Panning, etc.) or for a complete channel. For example, you may wish to save your favourite Kick Drum EQ, or the complete settings for an announcer channel.

Transferring User Data

All user data is stored on the system's internal flashcard and may be imported or exported to a USB interface or mxGUI computer. In a networked installation, a central file server can be made accessible from each console within the network.

User data is fully compatible with any mc² or Nova73, regardless of the hardware configuration. This enables the transfer of production data, snapshots, mixes or presets to and from any system (including any other mc²), in order to recall settings in a different studio.

What's Stored in a Production?



Each production holds multiple folders to store snapshots and sequences. Each production also holds multiple automation mixes, each one with its own Pass Tree.

In addition, the production snapshot stores everything included in a snapshot plus the following settings:

- DSP configuration
- Input and Output sample rate converter settings, see I/O Parameters.
- System Settings display - the status of *all* system options including Levels, Metering, the surround format, etc.
- Metering display setup.
- ISO BAY status for each fader bay.

What's Stored in a Snapshot?

The console offers two types of snapshot:

Full Snapshots

These are one-shot memories which may be used to recall settings during a live show. Every full snapshot stores all of the following settings:

- **DESK**: the Console Configuration for the main desk; the assignment of channels to fader strips across all banks and layers, and the current status of bank and layer switching.
- **DSP**: all channel DSP settings including analogue input control.
- **CONN**: signal routing connections for all sources and destinations (via the **Signal List** or **mx Routing** displays).
- **I/O**: remote mic preamp and router I/O settings such as router level and word length.
- **LABEL**: User and Source Labels.
- **BAY**: the Console Configuration for any isolated (ISO) bays; the assignment of channels to fader strips across all banks and layers, and the current status of bank and layer switching for isolated fader bays.
- **MXDSP**: all settings for the optional mxDSP modules.

Partial Snapshots

A "partial snapshot" stores selected routing crosspoints only. For example, you could use a partial snapshot to route tone to all transmission feeds for a line check without affecting other aspects of the mix.

In this chapter we will deal with full snapshots. For more details on partial snapshots, see the mx Routing display.

Working with Productions and Snapshots

You should create a production for each client or type of work. For example, to store the low level settings required for a series of shows.

Within this production, you may then create a number of folders to store and recall snapshots to bring back different signal routing, DSP settings and console layouts, for each show transmission or while you are live on-air.

Warning

Productions store and recall low level settings which may cause a brief interruption to audio. Therefore, do *not* load a production during a live show. Instead, use snapshots to recall settings while live on-air.

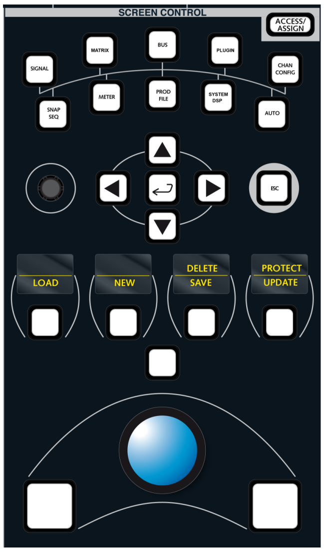
Snapshots are *only* written onto the user data flash card, once you save or update a production, so remember to save your work regularly.

For details on importing and exporting user data, see File Import/Export.

Productions

Productions are managed from the **Production** display.

1. Press the **PROD FILE** button, located on the SCREEN CONTROL panel, to view the **Productions** display:



The display is divided into two halves:

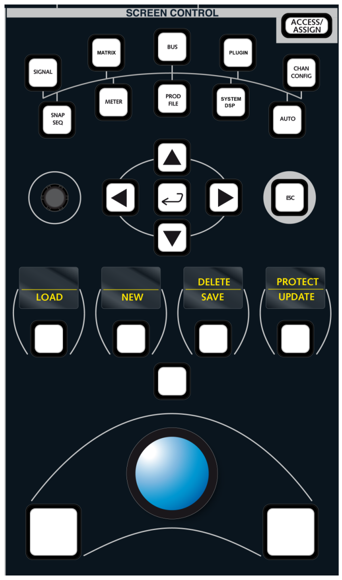
- **Productions** – lists all the productions stored on the internal user data flash card. This is where you can load, save, update rename, protect or delete a production.
- **Fallback** – lists any fallback productions stored in temporary memory. Fallback productions provide a level of undo in case you update or delete your production accidentally.

The active production (marked with an **A**) is also shown in the title bar of the Central GUI – in our example, **Olympic Games**. Therefore, you will *always* see the active production name across all displays.

To the right of each production name you will see the date and time when the production was last saved or updated, and the size of the production file. You may also see a padlock icon indicating that the production is protected.

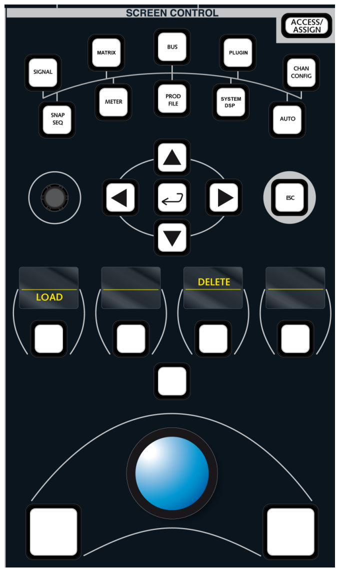
If the list of **Productions** or **Fallback** Productions is longer than the available window space, focus on the list and use the rotary scroller on the SCREEN CONTROL panel to navigate up and down the list. You can also resize the windows and/or use the on-screen scroll bars.

2. Focus on the **Productions** list and the following soft key functions become available:



- LOAD – loads the selected production settings to the console.
- NEW – clears any existing snapshot/sequence folders and mixes from memory, to create a new empty production.
- SAVE – creates a new production by saving the current console settings, including any snapshot/sequence folders and mixes. ("Save As...")
- UPDATE– saves the current console settings into an existing production. ("Save")
- DELETE – deletes the selected production.
- PROTECT – protects the selected production.

3. Now focus on the **Fallback** Productions list:



The soft keys update to:

- LOAD – loads the selected fallback production.
- DELETE - deletes the selected fallback production.

The same functions are available either as on-screen buttons.

Or by right-clicking on a production:

Loading a Production

You can load stored settings to the console at any time by loading a production.

When you load a production you will reset the console, so make sure you save the current settings if you wish to retrieve them later. If you do make a mistake, don't panic! When a production is loaded, a backup of the current settings is created in the **Fallback** list, see Fallback Productions.

> **Warning**
>
> Productions store and recall low level settings which may cause a brief interruption to audio. Therefore, do *not* load a production during a live show. Instead, use snapshots to recall settings while live on-air.

1. Select a production from the **Productions** list (e.g. **Football**):

2. Either press the **LOAD** soft key, or right-click and select **Load**, to complete the operation.

*The console status updates, and the title bar shows that **Football** is now the active production:*

For additional confirmation, watch the status bar at the bottom of the Central GUI; you should see a **loading…** message as the production data loads:

Saving a New Production

You can save the current settings of the console into a new production using **SAVE**. (i.e. this operation performs a "Save As..".)

SAVE keeps all the current settings, including any snapshot/sequence folders and automation mixes, and saves them under a new production name. If you wish to clear the folders and mixes from memory, then see new production.

> It is a good idea to save and organise your productions carefully.
> Don't overwrite the studio's setup production with your own settings by using update! Instead, use the **SAVE** function to save into a new production.

To save a new production:

1. Select the on-screen **Save** button, or focus on the list of **Productions** and press the **SAVE** soft key.

The current settings are saved into a new production which is given a default name (e.g. ***production 0012****):*

The production is time and date stamped, and automatically becomes the active production (**A**) as indicated in the title bar.

For additional confirmation, watch the status bar at the bottom of the Central GUI; you should see a **saving...** message as the production data is saved.

Renaming a Production

1. Click on the production name:

Click once to select all the existing text (white) or twice (black cursor) to modify the existing name.

2. Enter a new name from the keyboard.

3. When you have finished, press the Enter button, on the keyboard, to confirm the new name (e.g. **Formula One**):

4. Or, if you make a mistake or want to exit without making any changes, press the **Esc** button on the keyboard.

Updating a Production

You can save the current settings of the console into an existing production using **UPDATE**.

Updating a production overwrites it. Therefore, make sure you select the correct production to update. If you do make a mistake, don't panic! When a production is updated, a backup of the "old" production is created in the **Fallback** list, see Fallback Productions.

> If a production is protected, then you can *not* update it. (Use **Protect** to safeguard any important productions which you do not want to accidentally overwrite).

There are two ways to update a production:

➢ The UPDATE button

This button *always* updates the active production, marked with an **A** and named in the title bar. (i.e. this operation performs a "Save".)

You can use the button at any time; the **Productions** display does not need to be selected. But, remember that a protected production can *not* be updated.

The button also flashes if you have snapshots or automation which have not been saved.

➢ Update in the Productions display

From the **Productions** display, you can update *any* existing production, not only the active one.

1. Select a production from the **Productions** list (e.g. **Formula One**):

2. Either press the **UPDATE** soft key, or right-click and select **Update**, to complete the operation. (Remember that a protected production can *not* be updated.)

   The selected production is overwritten with the current console settings. You can confirm this by looking at the new date and time stamp.

For additional confirmation, watch the status bar at the bottom of the Central GUI; you should see a **saving...** message as the production data is saved.

New Production

NEW clears any existing snapshot/sequence folders and mixes from memory, so that when you save a production you can start from an empty Folders or Mixes list.

Note that **NEW** only clears folders and mixes; it does *not* alter the current state of the console and it does not save any data.

1. Select the on-screen **New** button, or focus on the list of **Productions** and press the **NEW** soft key.

The snapshot/sequence folders and mixes are cleared, and you will see an empty active production name in the title bar at the top of the display:

You can now save the current state of the console using either Save or Update.

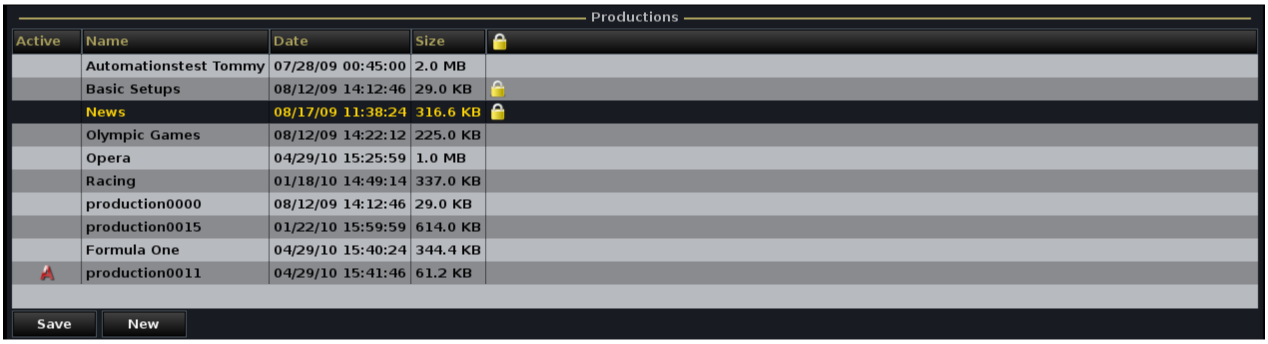
Protect & Delete

Protect

A protected production cannot be updated or deleted. You can use this safeguard any important productions which you do not want to accidentally overwrite or delete.

1. Select a production from the **Productions** list (e.g.**News**).
2. Either press the **PROTECT** soft key, or right-click and select **Protect**, to complete the operation.

The padlock icon indicates that the production is now protected:



Delete

Delete removes a production and all of its contents – snapshots, sequences and mixes - from the internal user data flash card.

To prevent accidental deletion, protected productions may not be deleted.

1. Select a production from the **Productions** list (e.g. **Football**).
2. Either press the **DELETE** soft key, or right-click and select **Delete**, to complete the operation:

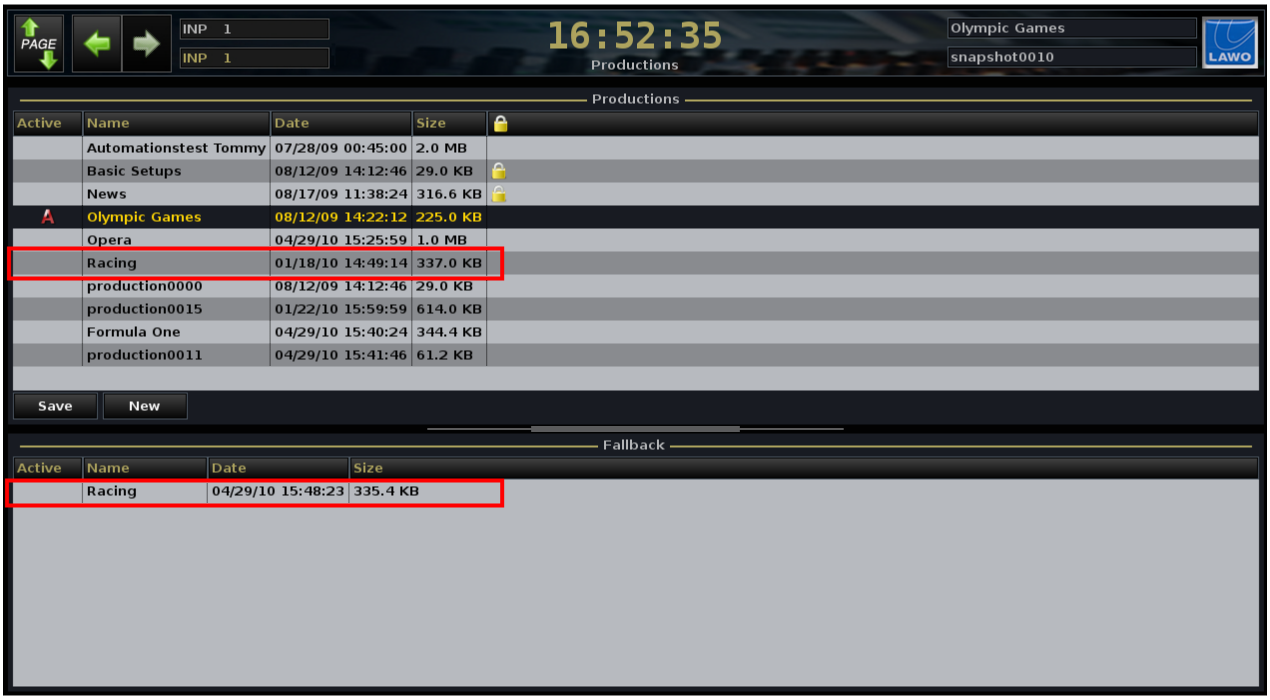
Fallback Productions

When a production is loaded, updated, deleted or cleared (using NEW production), a temporary copy of the current console settings or overwritten/deleted production is created in the fallback productions memory.

Five fallback productions are stored providing five levels of undo.

For example, whilst setting up for **Racing**, the operator forgets to update the production. He/she decides to load a different production to check the settings for **Olympic Games**. In the background, before the load is performed, the console automatically stores the current settings into a fallback production.

The name of the fallback production is taken from the active production when the mistake was made, in our example **Racing**. However, note that the fallback is *not* a copy **Racing** but a backup of the unsaved settings before the load operation was performed. You can see this from the different time and date stamp date between productions called **Racing** in the Productions and Fallback lists:



The operator then realises their mistake.

To recover the unsaved settings...

1. Select the **Racing** production from the **Fallback** list.

2. Either press the **LOAD** soft key, or right-click and select **Load**, to complete the operation.

The fallback production settings are loaded into the console and the operator's settings are restored!

3. To save these settings permanently, you must now save the settings into a new production, or update the original **Racing** production.

Warning

If you do not save or update the settings into a permanent production, then they may be lost forever. The fallback productions memory is a first-in first-out memory holding a maximum of 5 fallback productions.

Importing and Exporting Productions

A complete production, or elements of a production (such as a folder, snapshot or mix) may be imported or exported to a USB interface, mxGUI computer or network drive. This allows you to archive or transfer productions between systems. In addition, you can use this function to copy elements from one production to another. See File Import/Export for details.

Folders

Folders are used to organise the snapshots and sequences within a production. Each production may contain any number of folders, and within each folder you may store multiple snapshots and sequences.

Note that when using sequences, it's important to consider how you organise data within your folders. A sequence may reference any snapshot contained within the same folder, but not snapshots from a different folder. For example, you can't include a snapshot stored in 'Music' if the sequence is stored in 'Football'.

If necessary, you can use File Import/Export to copy snapshots between folders.

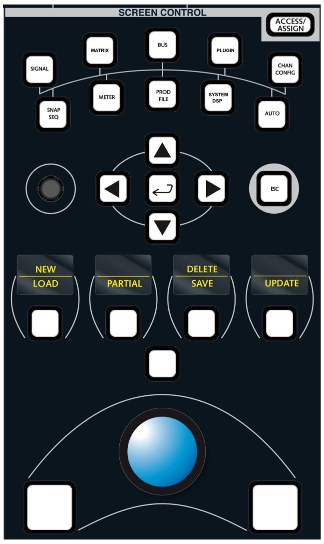
Note that the Folders list includes two special folders:

- **FALLBACK** - contains fallback snapshots, which provide a way of recovering settings should you change the DSP Configuration (Recording to Broadcast, or Broadcast to Recording) by accident. This folder cannot be renamed or deleted.
- **BACKUP** - contains backup snapshots, which provide levels of undo by periodically saving snapshots. This folder appears when backup snapshots are enabled.

Creating a New Folder

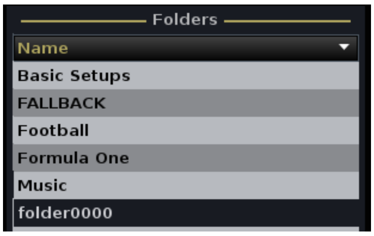
You can create a new folder either from the Snapshots or Sequences display. Here, we will use the **Snapshots List** display.

1. Press the **SNAP/SEQ** button, located on the SCREEN CONTROL panel, to view the **Snapshots** display:



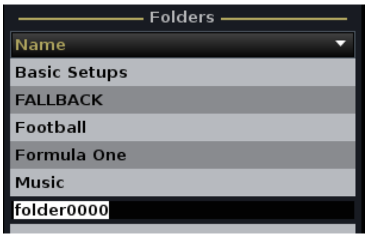
2. Focus on the list of **Folders** on the left hand side of the display, and either select the on-screen **New** button or press the **NEW** soft key.

A new folder appears in the Folders column with a default name (e.g. ***folder 0000****):*



Renaming a Folder

1. Click on the folder name:



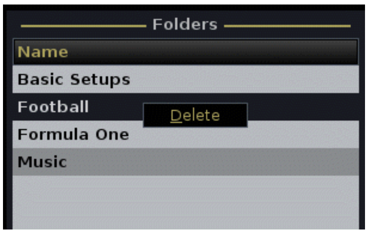
Click once to select all the existing text (white) or twice (black cursor) to modify the existing name.

2. Enter a new name from the keyboard.
3. When you have finished, press the Enter button, on the keyboard, to confirm the new name.
4. Or, if you make a mistake or want to exit without making any changes, press the **Esc** button on the keyboard.

Deleting a Folder

To prevent accidental deletion of snapshots only empty folders may be deleted. Therefore, first delete any snapshots contained within the folder before attempting this operation.

1. Select the folder you wish to delete from the **Folders** list.
2. Either press the **DELETE** soft key, or right-click and select **Delete**, to complete the operation:



Importing and Exporting Folders

A complete folder may be imported and exported to a USB interface, mxGUI computer or network drive. This allows you to archive or transfer folders between systems. See File Import/Export for details.

Snapshots

Snapshots may be used to store different mixes/setups for recall before or during a live show. For example, to recall a different mix for each band in a live entertainment show, or to recall scene changes during a live theatre production.

Note that a production stores the same settings as a snapshot plus other lower level settings. Therefore, you only need to use snapshots to save and load *different* settings within the same production, or to recall settings while live on-air. See What's Stored in a Production.

In addition, there are two types of snapshot - **Full** and **Partial** - which can be used to store all snapshot settings, or only routing crosspoints. See What's Stored in a Snapshot.

In this section we will be dealing with full snapshots. However, the same principles of load, save, delete, etc. may be applied to partial snapshots.

Note that snapshots are *only* written to the user data flash card, once you save or update a production, so remember to save your work regularly.

To help manage the reset, you may isolate individual channels, individual signals, or global console elements from a snapshot load.

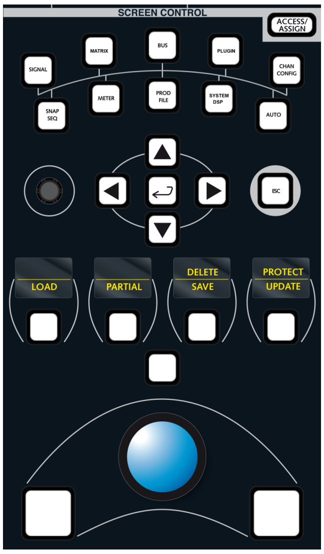
The Snapshots Display

Snapshots are managed from the **Snapshots** display.

1. Press the **SNAP/SEQ** button, located on the SCREEN CONTROL panel, to view the **Snapshots** display.

2. Select a folder from the **Folders** list (e.g. **Music**).

You will see all snapshots contained within the selected folder:



If the list of **Folders** or **Snapshots** is longer than the available window space, focus on the list and use the rotary scroller on the SCREEN CONTROL panel to navigate up and down the list. You can also resize the windows and/or use the on-screen scroll bars.

The **Snapshots** area lists all the snapshots by name together with the following information:

- **Type** - indicates whether it is a **full** or **partial** snapshot, see What's Stored in a Snapshot.
- **Date/Time** - the date and time when the snapshot was saved or last updated.
- **Padlock Icon** - identifies snapshots which have been write-protected.
- **Memo 1 & 2** - a summary of any notes added to the snapshot.
- **Channel Type** - indicates the DSP channel type which was active when the snapshot was saved or last updated - either **Broadcast** or **Recording**. This is important as you cannot load a **Broadcast** snapshot to **Recording** channels, or vice versa. See Transferring User Data for more details.

The **Snapshot memo** area may be used to add notes to a snapshot.

The on-screen buttons provide access to snapshot operations: Save, Load, Update, Delete and Protect. These functions are also available from the SCREEN CONTROL soft keys when you are focussed on the **Snapshots** list. Or, if you select a snapshot and right-click.

The Global Snapshot ISO buttons can be used to isolate global elements from a snapshot load.

> The name of the last snapshot saved or loaded is *always* shown in the title bar of the Central GUI across all displays – in our example, **Act 1 Scene 2**.

Loading a Snapshot

You can load stored settings to the console at any time, even while on-air, by loading a snapshot.

If snapshot offsets are active, then they are applied to the loaded parameters.

Any isolated objects are *not* reset by the snapshot load. You can isolate individual channels, individual signals, or global console elements, see Snapshot Isolate.

1. Select a snapshot from the **Snapshots** list (e.g. **Act 1 Scene 2**):

2. Select the on-screen **Load** button, or focus on the list of **Snapshots** and press the **LOAD** soft key to complete the operation.

*The console instantly updates and the title bar shows that **Act 1 Scene 2** is now the active snapshot:*

Snapshot Isolate

Before loading a snapshot you may isolate certain objects, so that they are *not* reset by the snapshot load. You can choose to isolate individual channels, individual signals, or global console elements. Snapshot isolates are stored and recalled by productions.

Isolating Channels (SNAP ISO)

To isolate a complete channel, select its **SNAP ISO** button. This isolates all the channel's DSP settings including analogue input control, see Channel processing.

SNAP ISO may be programmed onto a fader strip user button, or selected from the Central Control Section CHANNEL buttons.

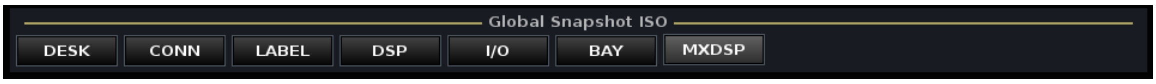
Note that the Isolate option, on the **System Settings** display, prohibits the selection of **SNAP ISO** buttons across the console.

Isolating Signals

To isolate an individual source or destination, select the Isolate function on the **Signal List** display.

Global Snapshot ISO

These buttons appear on both the Snapshots and Extra Buttons displays. They isolate global console elements from a snapshot recall:



Select the elements you do *NOT* want to recall *before* loading the snapshot. The buttons apply globally across the system; for example, select **DESK** to protect the layout of your fader strips, but still recall all of your DSP settings, signal routing, etc.

- **DESK**: the Console Configuration for the main desk; the assignment of channels to fader strips across all banks and layers, and the current status of bank and layer switching.
- **DSP**: all channel DSP settings including analogue input control.
- **CONN**: signal routing connections for all sources and destinations (via the **Signal List** or **mx Routing** displays).
- **I/O**: remote mic preamp and router I/O settings such as router level and word length.
- **LABEL**: User and Source Labels.
- **BAY**: the Console Configuration for any isolated (ISO) bays; the assignment of channels to fader strips across all banks and layers, and the current status of bank and layer switching for isolated fader bays.
- **MXDSP**: all settings for the optional mxDSP modules.

Saving a Snapshot

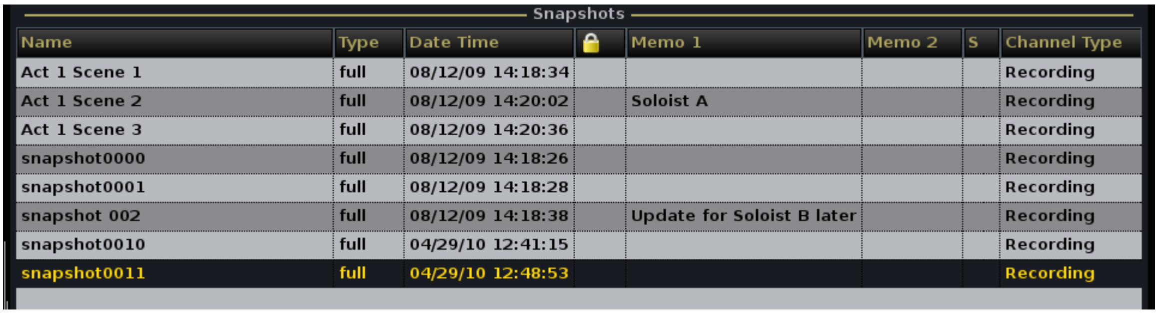
You can save the current settings of the console into a new snapshot using **SAVE**.

All settings are always saved into a snapshot regardless of the snapshot isolate status.

1. Select the folder you wish to save into on the left of the **Snapshots** display.

2. Then select the on-screen **Save** button, or focus on the list of **Snapshots** and press the **SAVE** soft key.

The current settings are saved into a new snapshot which is given a default name (e.g. ***snapshot 0011****):*



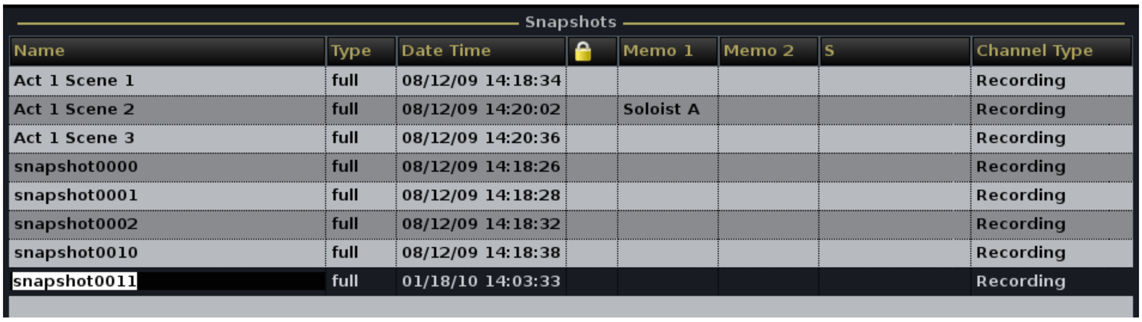
Name	Type	Date Time		Memo 1	Memo 2	S	Channel Type
Act 1 Scene 1	full	08/12/09 14:18:34					Recording
Act 1 Scene 2	full	08/12/09 14:20:02		Soloist A			Recording
Act 1 Scene 3	full	08/12/09 14:20:36					Recording
snapshot0000	full	08/12/09 14:18:26					Recording
snapshot0001	full	08/12/09 14:18:28					Recording
snapshot 002	full	08/12/09 14:18:38		Update for Soloist B later			Recording
snapshot0010	full	04/29/10 12:41:15					Recording
snapshot0011	full	04/29/10 12:48:53					Recording

The snapshot is time and date stamped, marked as a **full** snapshot, and automatically becomes the active snapshot as indicated in the title bar.

Note that the **Save Partial** screen button or **PARTIAL** soft key is used to save a partial snapshot.

Renaming a Snapshot

1. Click on the snapshot name:


Snapshots

Name	Type	Date Time		Memo 1	Memo 2	S	Channel Type
Act 1 Scene 1	full	08/12/09 14:18:34					Recording
Act 1 Scene 2	full	08/12/09 14:20:02		Soloist A			Recording
Act 1 Scene 3	full	08/12/09 14:20:36					Recording
snapshot0000	full	08/12/09 14:18:26					Recording
snapshot0001	full	08/12/09 14:18:28					Recording
snapshot0002	full	08/12/09 14:18:32					Recording
snapshot0010	full	08/12/09 14:18:38					Recording
snapshot0011	full	01/18/10 14:03:33					Recording

Click once to select all the existing text (white) or twice (black cursor) to modify the existing name.

2. Enter a new name from the keyboard.

3. When you have finished, press the Enter button, on the keyboard, to confirm the new name.

4. Or, if you make a mistake or want to exit without making any changes, press the **Esc** button on the keyboard.

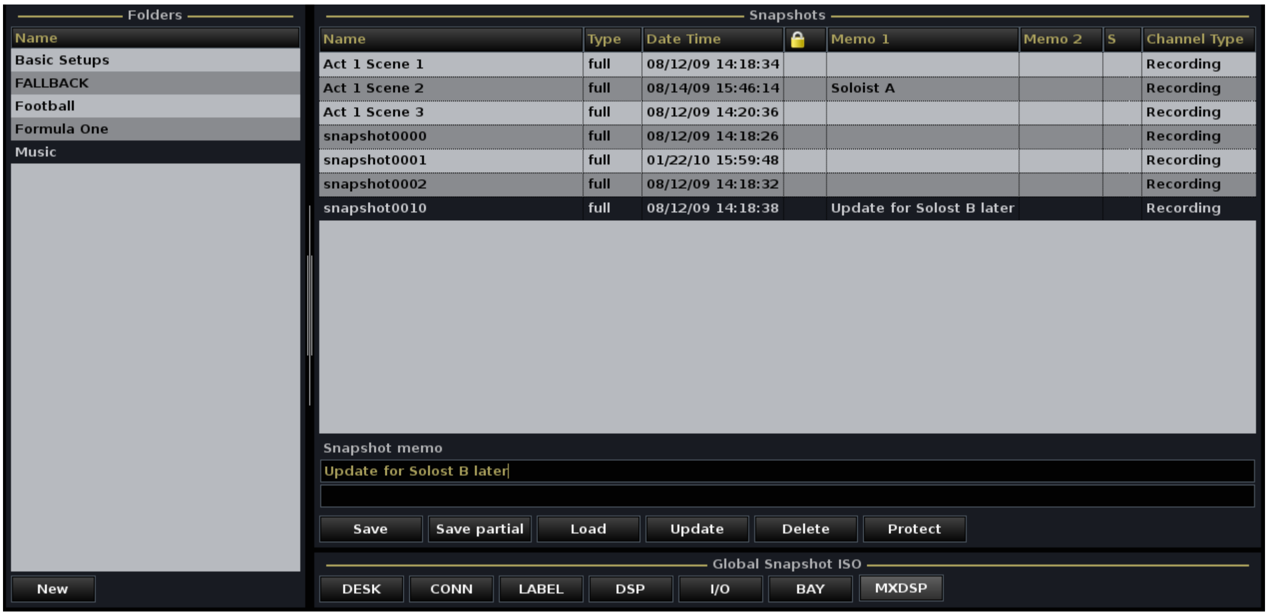
Adding a Memo

You may use the two **Snapshot Memo** lines to add memo information. For example, you may wish to remind yourself about the artist's position on stage for a particular snapshot.

1. Select the snapshot and then select a line in the **Snapshot Memo** field.

A black cursor appears.

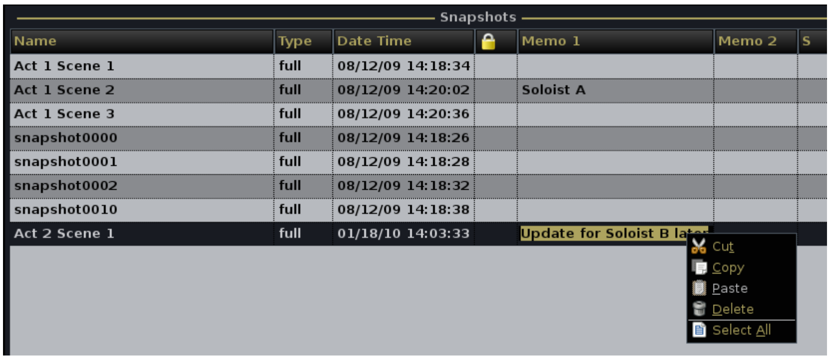
2. You may now type to enter your information from the keyboard:



The first and second memo lines appear beside the snapshot name in the **Snapshots** list. You can enter as many characters as you wish in each line; the list will automatically resize to fit.

If you cannot enter any memo text, check that the snapshot is not protected.

Right-click on a snapshot memo to **Copy** and **Paste** the text to another snapshot:



You can also drag and drop the **Memo** columns to change their position on the display.

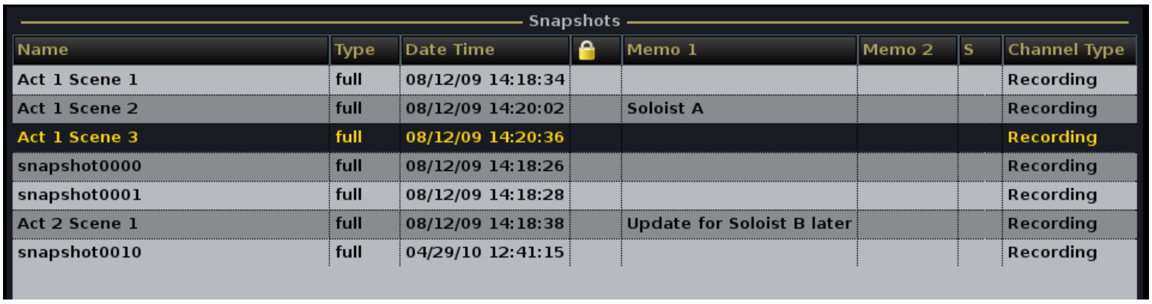
Updating a Snapshot

You can save the current settings of the console into an existing snapshot using **UPDATE**.

Updating a snapshot overwrites it. Therefore, make sure you select the correct snapshot to update; there is no undo from this operation!

> If a snapshot is protected, then you can *not* update it. (Use **Protect** to safeguard any important snapshots which you do not want to accidentally overwrite).

1. Select the snapshot you wish to update from the **Snapshots** list (e.g. **Act 1 Scene 3**):



Snapshots

Name	Type	Date Time		Memo 1	Memo 2	S	Channel Type
Act 1 Scene 1	full	08/12/09 14:18:34					Recording
Act 1 Scene 2	full	08/12/09 14:20:02		Soloist A			Recording
Act 1 Scene 3	full	08/12/09 14:20:36					Recording
snapshot0000	full	08/12/09 14:18:26					Recording
snapshot0001	full	08/12/09 14:18:28					Recording
Act 2 Scene 1	full	08/12/09 14:18:38		Update for Soloist B later			Recording
snapshot0010	full	04/29/10 12:41:15					Recording

2. Select the on-screen **Update** button, or focus on the list of **Snapshots** and press the **UPDATE** soft key.

The selected snapshot is overwritten with the current settings. You can confirm this by looking at the new date and time stamp.

Protect & Delete

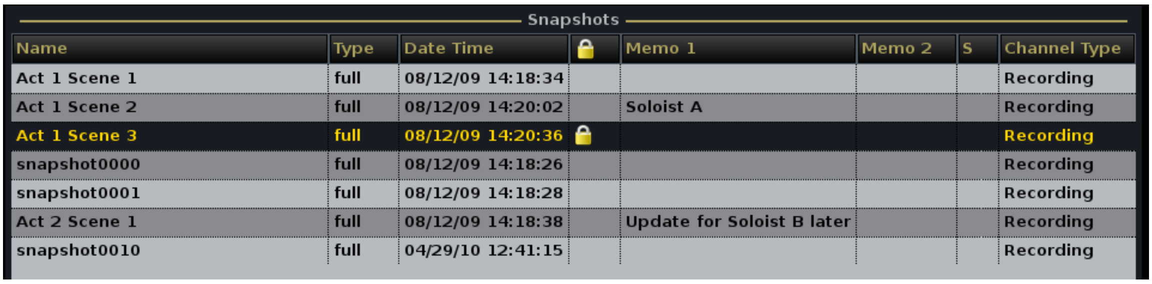
Protect

A protected snapshot cannot be updated or deleted. You can use this safeguard any important snapshots which you do not want to accidentally overwrite or delete.

1. Select a snapshot from the **Snapshots** list.

2. Select the on-screen **Protect** button, or focus on the list of **Snapshots** and press the **PROTECT** soft key.

The padlock icon indicates that the snapshot is now protected:



Snapshots

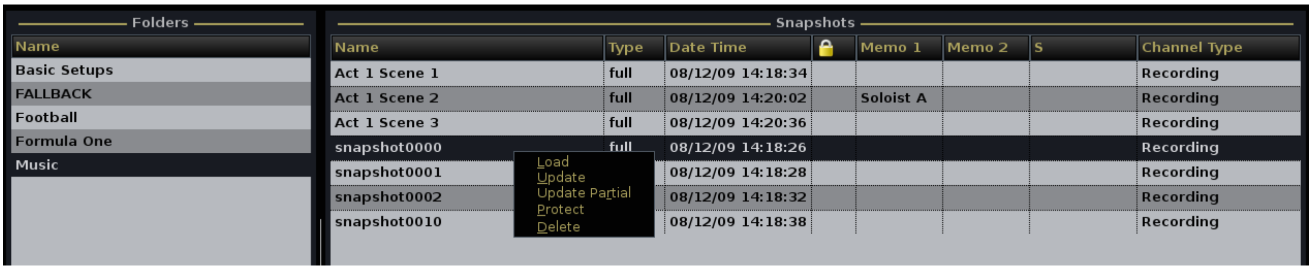
Name	Type	Date Time	🔒	Memo 1	Memo 2	S	Channel Type
Act 1 Scene 1	full	08/12/09 14:18:34					Recording
Act 1 Scene 2	full	08/12/09 14:20:02		Soloist A			Recording
Act 1 Scene 3	full	08/12/09 14:20:36	🔒				Recording
snapshot0000	full	08/12/09 14:18:26					Recording
snapshot0001	full	08/12/09 14:18:28					Recording
Act 2 Scene 1	full	08/12/09 14:18:38		Update for Soloist B later			Recording
snapshot0010	full	04/29/10 12:41:15					Recording

Delete

Delete removes the snapshot from the internal memory.

1. Select a snapshot from the **Snapshots** list.

2. Select the on-screen **Delete** button, or focus on the list of **Snapshots** and press the **DELETE** soft key, or right-click and select **Delete**:



Folders

Name
Basic Setups
FALLBACK
Football
Formula One
Music

Snapshots

Name	Type	Date Time	🔒	Memo 1	Memo 2	S	Channel Type
Act 1 Scene 1	full	08/12/09 14:18:34					Recording
Act 1 Scene 2	full	08/12/09 14:20:02		Soloist A			Recording
Act 1 Scene 3	full	08/12/09 14:20:36					Recording
snapshot0000	full	08/12/09 14:18:26					Recording
snapshot0001		08/12/09 14:18:28					Recording
snapshot0002		08/12/09 14:18:32					Recording
snapshot0010		08/12/09 14:18:38					Recording

Backup Snapshots

Backup snapshots may be used to provide levels of undo.

You can set how often the backup snapshots are stored and how many are held in memory from the **System Settings** display, using the Backup Snapshot options.

For example, you may set the backup snapshot interval to every 5 minutes, and limit the number to 12 backup snapshots giving yourself a 1 hour 'undo' window.

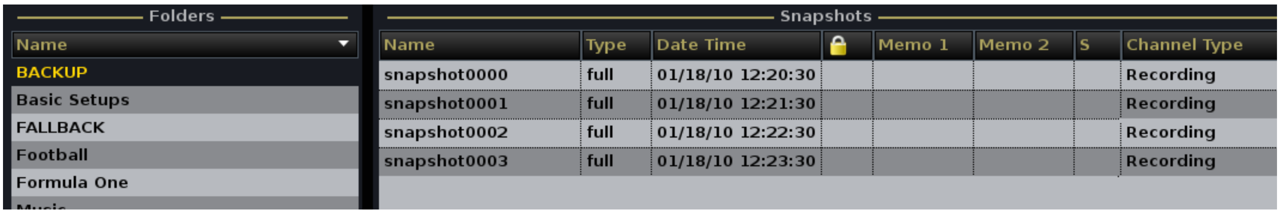
Note that you can disable backup snapshots by setting the number to 0. This can be a good idea during on-air operation, as each automatic save causes a brief interruption to console control.

All the backup snapshots for a particular production are stored within a special folder named **BACKUP**; this folder cannot be deleted. However, you can rename the folder if you wish to keep the last set of backup snapshots. After the next backup snapshot interval, a new **BACKUP** folder will be created.

At any time, you can use the backup snapshot system to revert to an earlier configuration:

1. Select the **BACKUP** folder on the left of the **Snapshots** display.

*The **Snapshots** list now shows all the backup snapshots, each one time and date stamped:*



2. Select a backup snapshot from the list, and select **Load**.

The console updates to the backup snapshot settings.

Note that the backup snapshot load works in the same manner as a normal snapshot load. Therefore, any snapshot isolates will be applied.

Importing and Exporting Snapshots

Individual snapshots can be imported and exported to a USB interface, mxGUI computer or network drive. This allows you to archive or transfer snapshots between systems. See File Import/Export for details.

Sequences

For convenient recall of snapshots during a live broadcast or theatre production, the **mc²56** provides real time sequence automation. A sequence is a list of snapshots which can be loaded in sequence during a live show. Note that the sequence itself does not store any settings, but simply creates a list of pointers to snapshots stored within the production folder.

Multiple sequences may be created within each folder to deal with different versions of the show. For example, when rehearsing a live theatre production, the running order may vary. By creating multiple sequences, you have the option to play out any variation.

You may also reference a single snapshot several times within each sequence. For example, in an entertainment show you may reference the same snapshot to return to presenter links between each music act. This means that any updates to the snapshot are carried through all occurrences within the sequence.

Note that a sequence may reference any snapshot contained within the same folder, but not snapshots from a different folder. For example, you can't add a snapshot stored in a folder called 'Music' to a sequence stored in 'Football'.

The Sequences display is used to create, edit and play out sequence automation:

- You can create a sequence and save or update snapshots as you work through a rehearsal. This approach works well when the rehearsal runs in the same order as the show, as you can save snapshots and add them to the sequence in one operation.
- Or you can add existing snapshots into a sequence offline. For example, if the rehearsal is unlikely to follow the same running order as the show, then it's best to save your snapshots into the production folder from the **Snapshots** display, and create the sequence at a later time.
- During play out, you may load sequence snapshots and choose to skip snapshots or revert back to the previous snapshot at the touch of a button. In addition, you can cross fade between snapshots to fade automatically from one scene to the next.

Use the SNAPSHOT/SEQUENCE front panel buttons to access the most important play out functions.

You may apply offsets to each snapshot load, using the Snapshot Trim Sets display. This is great if there are last minute changes you wish to make to the whole show.

The Sequences Display

The **Sequences** display is used to create, edit and play out sequences.

1. Press the **SNAP/SEQ** button, located on the SCREEN CONTROL panel, to view the **Sequences** display:

The display is divided into five areas:

- **Folders** – lists the folders stored within the current production; one will be selected, in our example, **Mozart**.
- **Sequences** - lists the sequences contained within the selected folder.
- **ACTIVE SEQUENCE** - shows the name and file path of the active sequence - e.g. **Mozart/ Magic Flute**. This is the sequence which will play out when you enable automation using the **Sequence on** button.
- **Sequence Contents** – shows the name and file path of the selected sequence and the snapshots within it. In our example, this happens be the **active** sequence. Snapshots will play out in the order shown; the **Status** column indicates:
  - o **N** - the Next snapshot to be loaded.
  - o **C** - the Current snapshot (i.e. the last snapshot loaded).
  - o **B** - the 'Back' snapshot. This is the last snapshot loaded before the current one.
- **Snapshots** - lists all the snapshots contained within the selected folder. Snapshots which are used within a sequence are marked with an **S**. You can add any of these snapshots into the sequence using the add function.

Beside the name of each snapshot you will see memo information, the date and time when the snapshot was saved (or updated), and details of the cross-fade parameters.

If information is hidden, then click on the left/right or up/down scroll bars and the additional columns will appear. Or, resize the areas by clicking and dragging the grey separator bars.

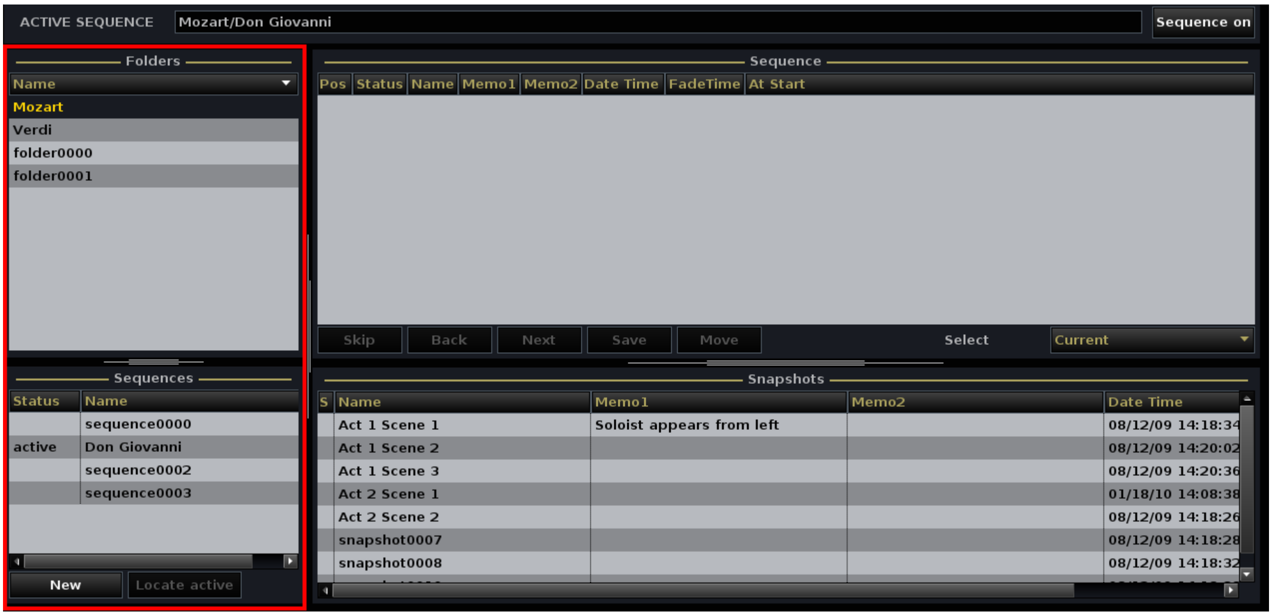
Sequences and Folders

Sequences are stored in folders within your current production. Use the **Folders** and **Sequences** lists on the left of the Sequences display to manage these areas as follows:

To change to a different sequence:

1. First select a folder from the **Folders** list (e.g. **Mozart**).

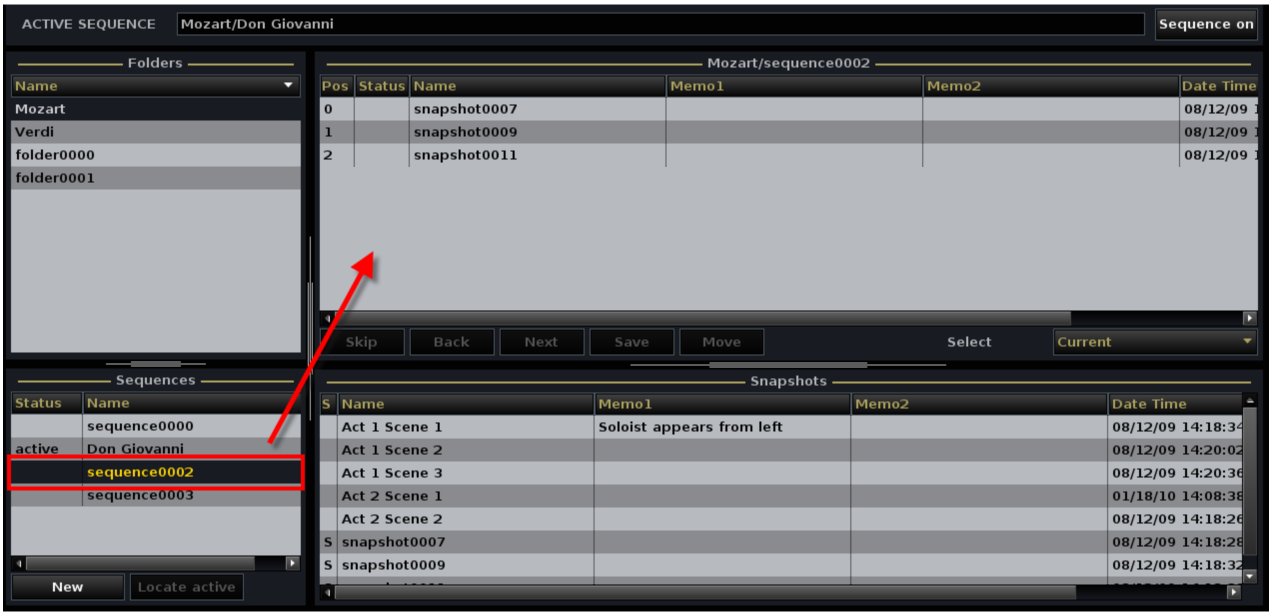
*The **Sequences** and **Snapshots** lists update to show what is contained within the folder:*



While you are focused on the **Folders** list you can create a new folder and edit the folder name in the usual manner. However, you cannot delete folders from the **Sequences** display; you must do this from the Snapshots display.

2. Now select a sequence from the **Sequences** list - e.g. ***sequence0002***

The selected sequence is highlighted and you will see its file path and contents – our example includes three snapshots:



While you are focused on the **Sequences** list, a series of functions for managing sequences become available on the SCREEN CONTROL soft keys:

- NEW – creates a new sequence.
- LOAD - sets the selected sequence to active.
- ACTIVE - navigates back to the active sequence if you have selected a different folder.
- NEXT - loads the Next snapshot in the sequence.
- DELETE – deletes the sequence from the folder.

The same functions are available either as on-screen buttons, or by right-clicking on a sequence:

New, Rename & Delete

New

To create a new sequence:

1. Select the on-screen **New** button, or focus on the **Sequences** list and press the **NEW** soft key.

*A new sequence is added to the list and given a default name – for example, **Sequence0004**.*

Rename

1. Click on the folder name using the trackball.

Click once to select all the existing text (white) or twice (black cursor) to modify the existing name.

2. Enter a new name from the keyboard.

3. When you have finished, press the Enter button, on the keyboard, to confirm the new name.

4. Or, if you make a mistake or want to exit without making any changes, press the **Esc** button on the keyboard.

Delete

1. Either right-click on the sequence and select **Delete**, or focus on the **Sequences** list and press the **DELETE** soft key.

The sequence is removed from the list.

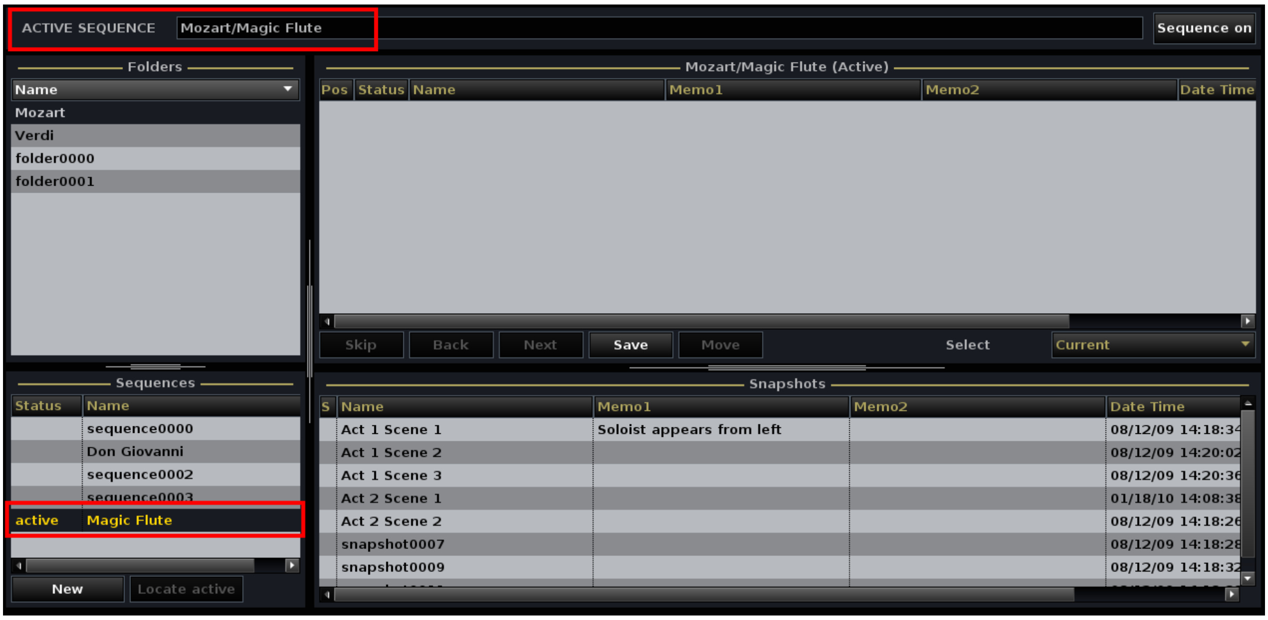
Note that this deletes the sequence, but does not delete any snapshots from the system.

Setting the Sequence to Active

Before snapshots can be added to a sequence or the sequence played out in real time, it must be set to active:

1. Select the sequence you wish to make active from the **Sequences** list (e.g. **Magic Flute**).
2. Either right-click and select **Load**, or focus on the **Sequences** list and press the **LOAD** soft key.

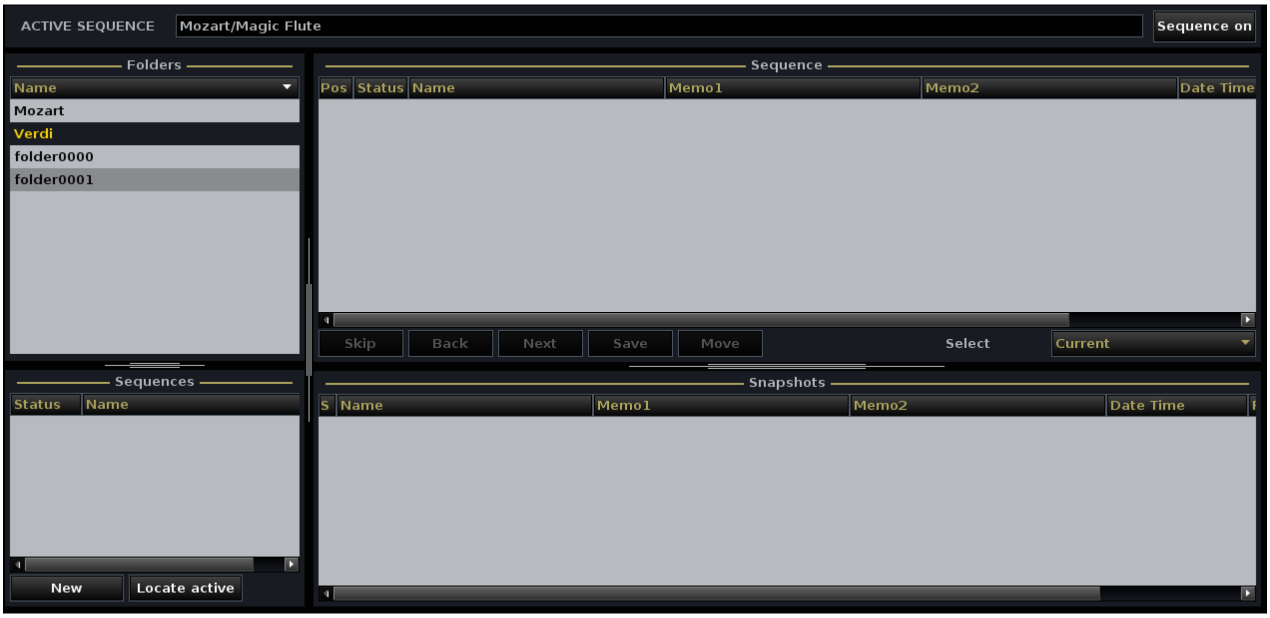
*The sequence status updates to active, and its name appears in the **ACTIVE SEQUENCE** box at the top of the display:*



The sequence is now ready for preparation or play out.

Navigating to the Active Sequence

If you have been viewing a different folder, then the active sequence may not be visible within the **Sequences** list:



To quickly locate the active sequence:

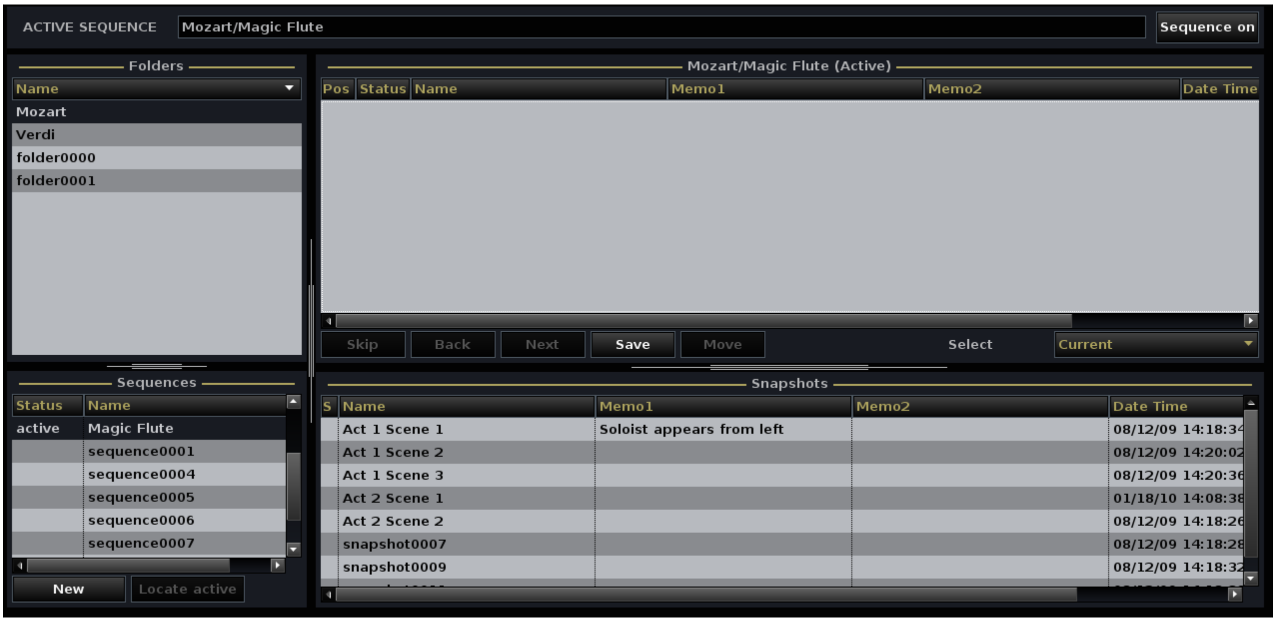
1. Select the on-screen **Locate active** button, or focus on the **Sequences** list and press the **ACTIVE** soft key.

The display updates to reveal the active sequence.

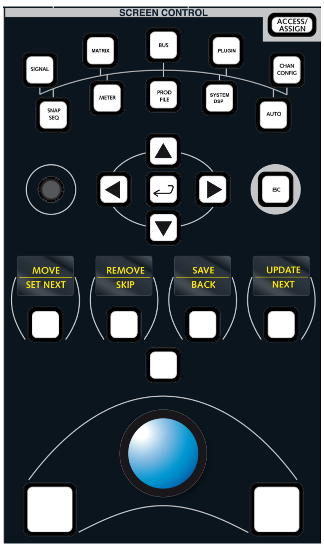
The Sequence Contents

Now change the focus of the Sequences display to the **Sequence Contents** area – this lists all the snapshots contained within the sequence.

If this is a new sequence, then the list will be empty:



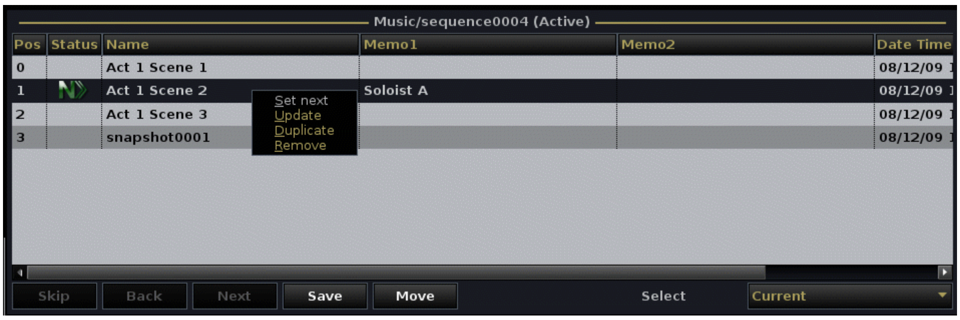
While you are focused on this area, a series of functions for saving, editing and playing out sequence snapshots become available on the SCREEN CONTROL soft keys:



Here we will deal with sequence preparation functions:

- MOVE – moves the selected snapshot position.
- REMOVE – removes the selected snapshot from the sequence.
- SAVE – saves a new snapshot and enters it in the sequence.
- UPDATE – updates the selected snapshot.

The same functions are available either as on-screen buttons, or by right-clicking on a snapshot within the sequence:



Saving Snapshots

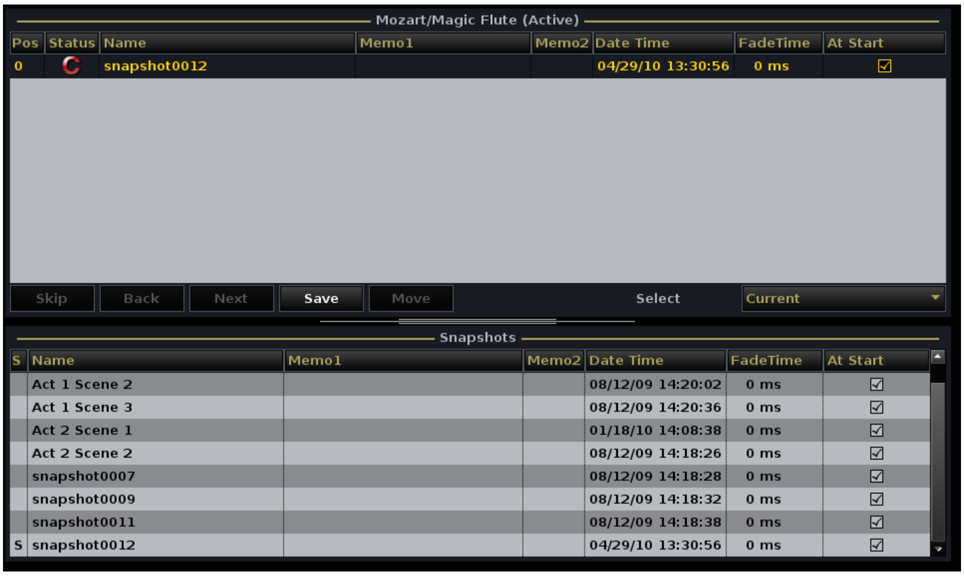
If the rehearsal is running more or less in the same order as the show, it makes sense to save snapshots directly from the **Sequences** display as this allows you to save snapshots and add them to the sequence in one operation.

> You cannot save snapshots into a sequence unless the sequence is active - the name and file path should appear in the **ACTIVE SEQUENCE** box at the top of the display.

To save a snapshot into the current production folder, and add it to the active sequence:

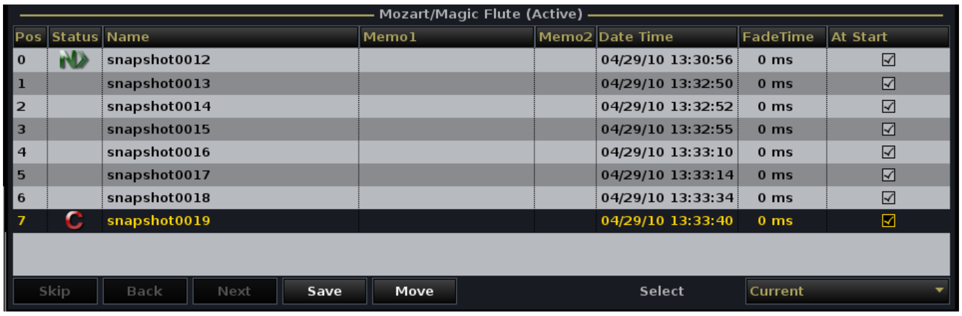
1. Either select the on-screen **Save** button, or focus the display on the **Sequence Contents** area and press the **SAVE** soft key.

   A new snapshot appears below the current selection. If this is the first snapshot to be added to the sequence, then it will appear at the top of the list!



Just as when saving snapshots from the Snapshots display, the snapshot is given a default name, and date/time stamped.

2. Press **SAVE** a few times to save a number of snapshots to the active sequence:

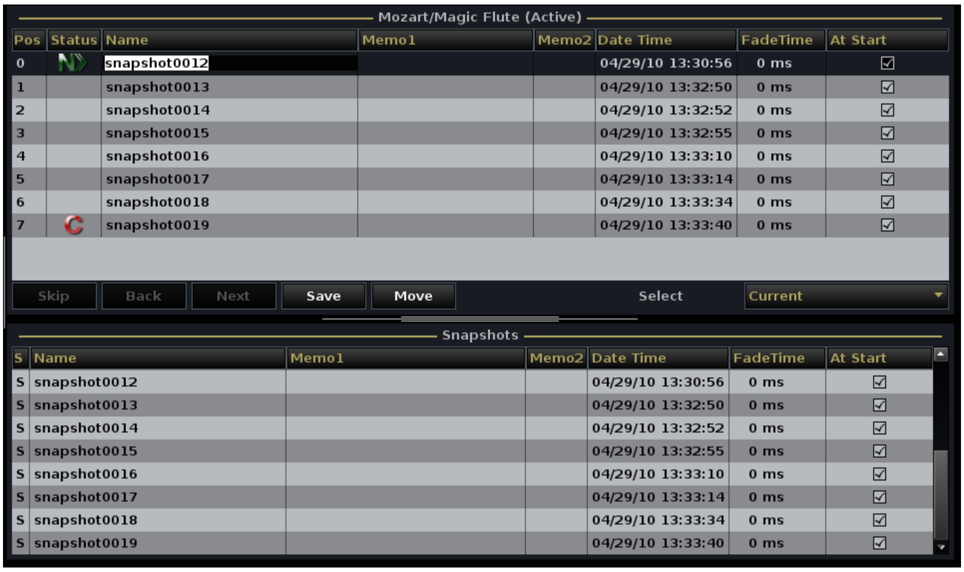


Naming a Snapshot & Adding a Memo

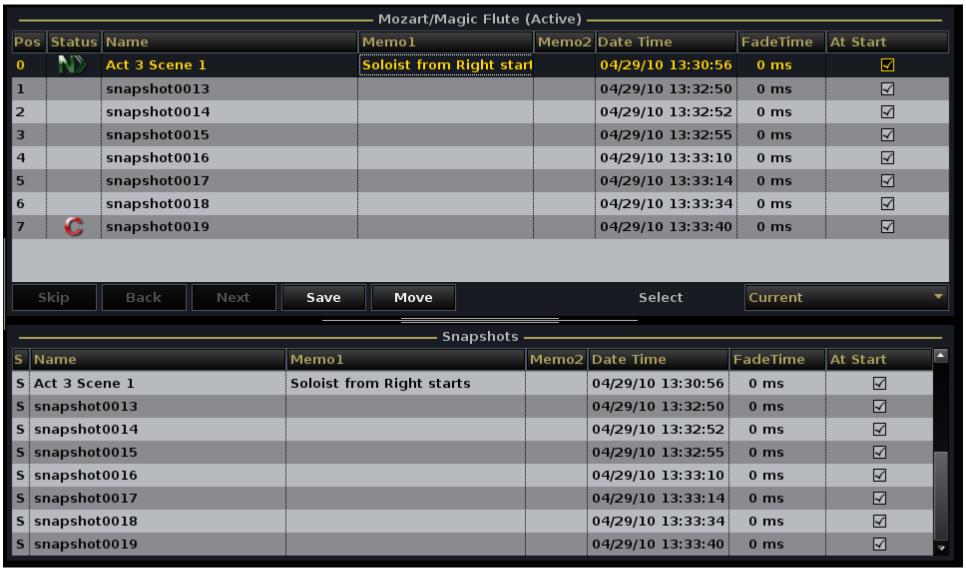
For convenience, you can rename a snapshot or add memo information from the **Sequences** display.

Both of these operations work in a similar manner to the **Snapshots** display. See Renaming a Snapshot and Adding a Memo for details.

Renaming a Snapshot from the Sequences display:

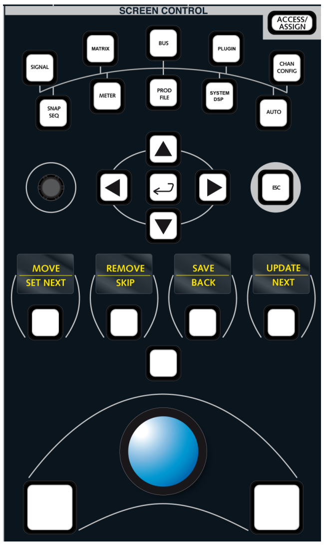


Adding Memo Text to a Snapshot from the Sequences display:



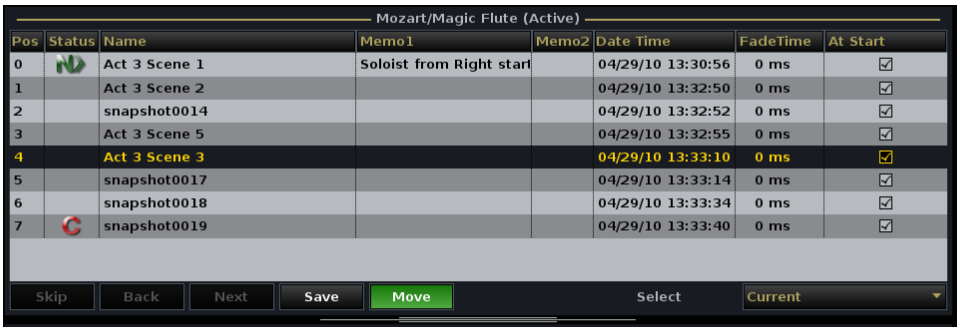
Changing the Sequence Order (Move)

Having added some snapshots to the sequence, you can change their order as follows:



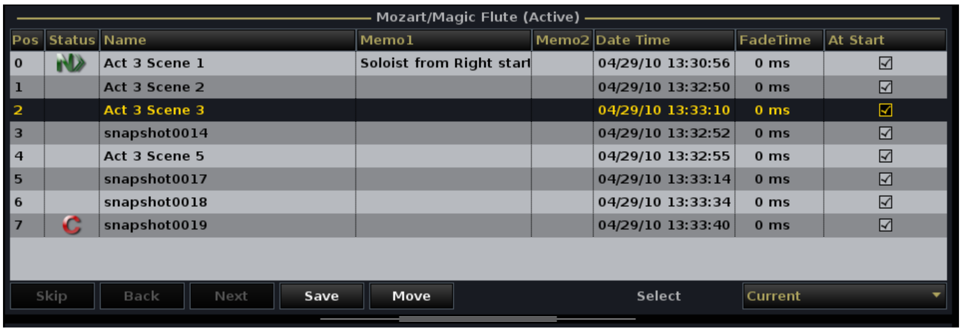
1. Select the snapshot you wish to move and turn on the **Move** button.

The button turns green when active:



2. Now use the navigation buttons or rotary scroller on the SCREEN CONTROL panel, or the up/down arrows on the your console keyboard, to nudge the position of the selected snapshot, up or down the list.

3. When you are happy with the order, deselect **MOVE**:

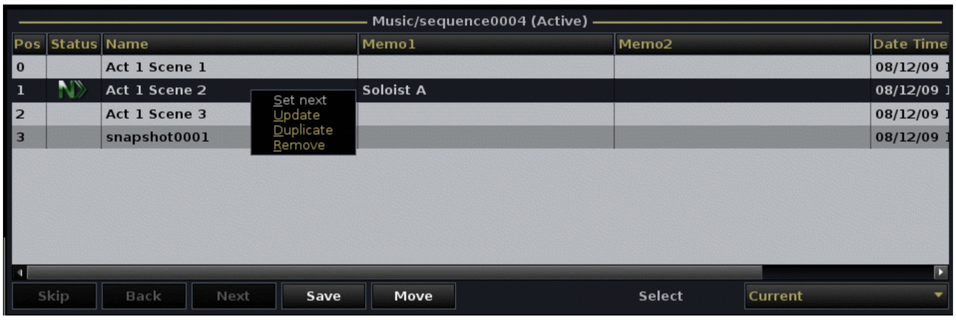


Removing Snapshots

You can remove a snapshot from the sequence as follows.

Note that this operation only removes the snapshot from the sequence, and does not delete the snapshot from the production folder. To delete a snapshot from the system completely, delete it from the Snapshots display.

1. Select the snapshot you wish to remove.
2. Either right-click and select **Remove**, or press the **REMOVE** soft key:



The snapshot is now removed from the active sequence.

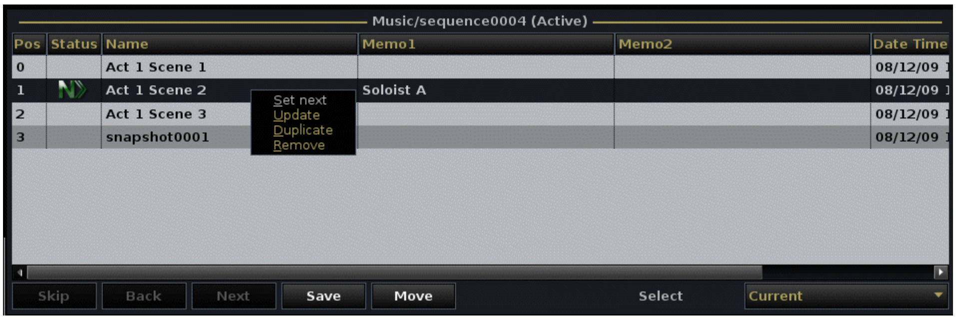
Updating Snapshots

To avoid having to revert to the **Snapshots** display during a rehearsal, you may update a snapshot within the active sequence from the **Sequences** display. This operation will update the snapshot contents with the current console settings.

> Note that this updates the snapshot. Therefore, if the snapshot is used multiple times, either within the same sequence or different sequences, then all occurrences are updated.
>
> If you wish to use the same snapshot within a sequence, but keep all occurrences independent, then use duplicate to copy the snapshot.
>
> If the snapshot is protected then it cannot be updated.

To update a snapshot:

1. Select the snapshot you wish to update.
2. Either right-click and select **Update**, or press the **UPDATE** soft key:



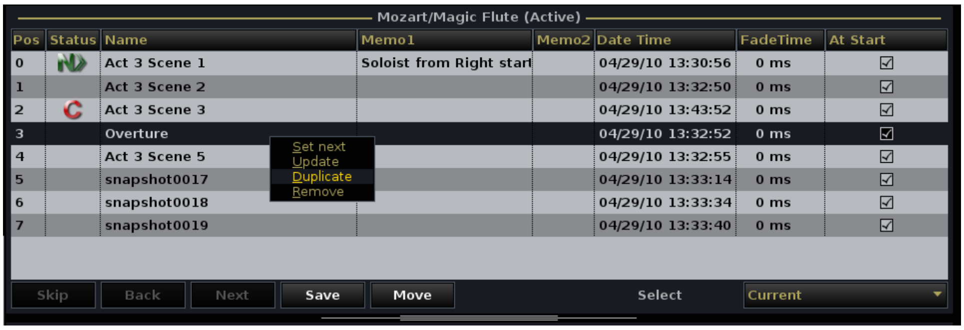
The snapshot is overwritten with the current console settings. You can confirm this by looking at the new date and time stamp.

Duplicating Snapshots in a Sequence

The console provides the ability to either reference the same snapshot multiple times throughout a sequence, or duplicate a snapshot to keep each occurrence independent.

Note that this function is only available from the touch-screen.

1. Select the snapshot you wish to duplicate.
2. Then right-click and select **Duplicate**:



A copy of the snapshot is created with a new name.

The name given is taken from the original snapshot name. For example, duplicating a snapshot named **Overture** will create a new snapshot called **Overture(1)**.

The Snapshots List

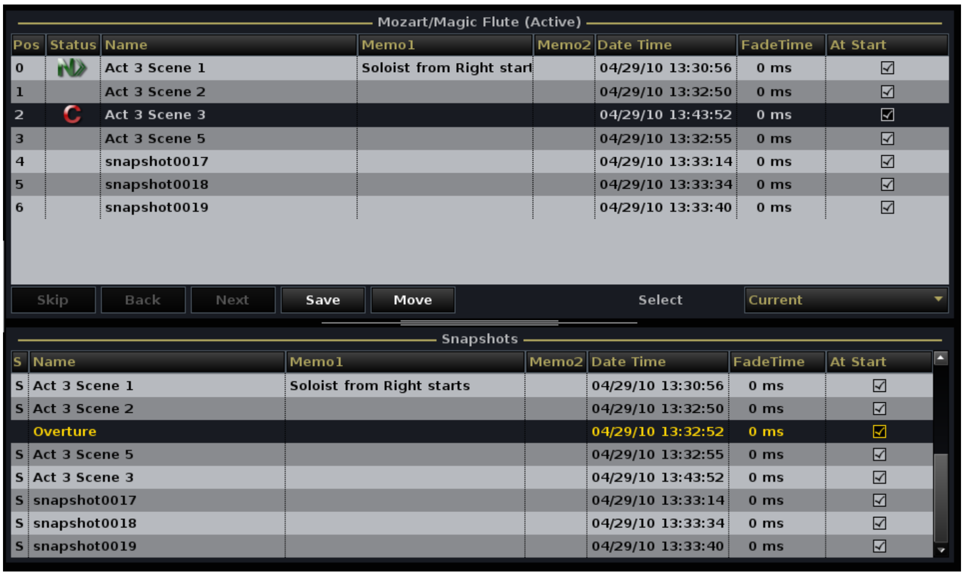
Now change the focus of the Sequences display to the **Snapshots** list – this lists all the snapshots contained within the selected folder, and may be used to add an existing snapshot to a sequence. Any snapshots already used in a Sequence are marked in the **S** column:

Adding Existing Snapshots to a Sequence

You can add any snapshot contained within your production folder to the active sequence.

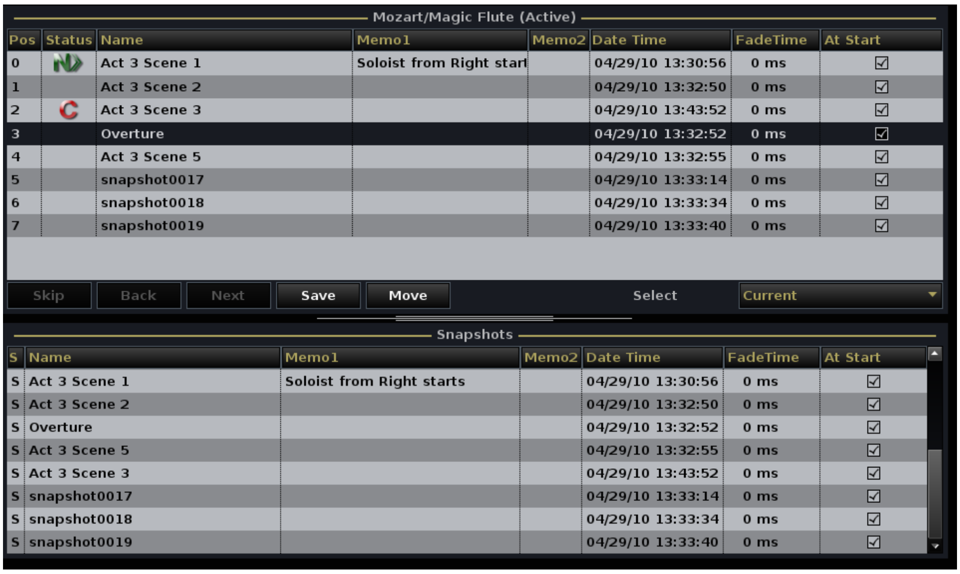
> Note that you cannot add snapshots located in a different folder. Although you could always load the snapshot to the console, change folders and then resave the snapshot to achieve the same result.

1. Select the snapshot you wish to add from the **Snapshots** list – for example, **Overture**:



2. Either press the **ADD** soft key, or right-click and select **add**.

*The snapshot is added to the sequence, below the selected snapshot – in our example, below **Act 3 Scene 3**:*

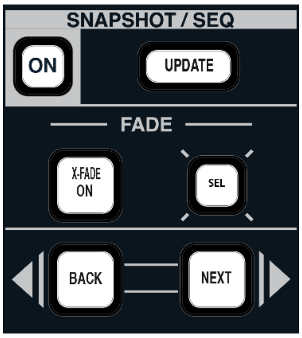


> You can add the same snapshot multiple times if you wish, creating several pointers to a single snapshot.

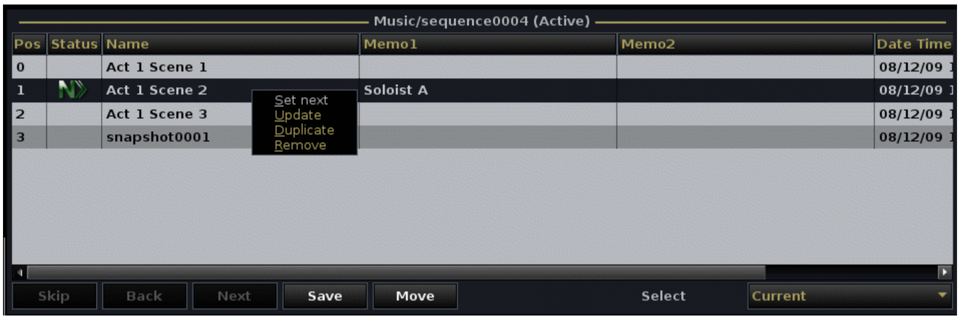
Running a Sequence

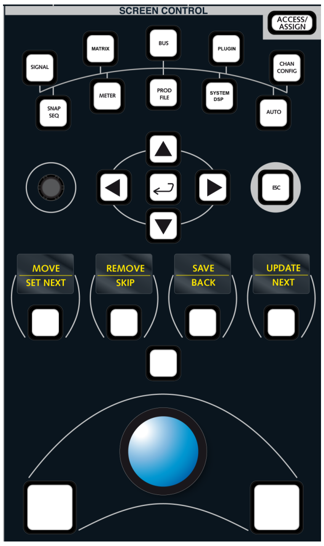
Once you have created a sequence and either saved or added snapshots to it, you may run the sequence in real time.

You can use the SNAPSHOT/SEQUENCE front panel, SCREEN CONTROL soft keys or on-screen buttons to play out the sequence.



If you want to use the soft keys, then focus on the **Sequence** list to update the soft key functions. The same functions are available either as on-screen buttons, or by right-clicking on a snapshot within the sequence:



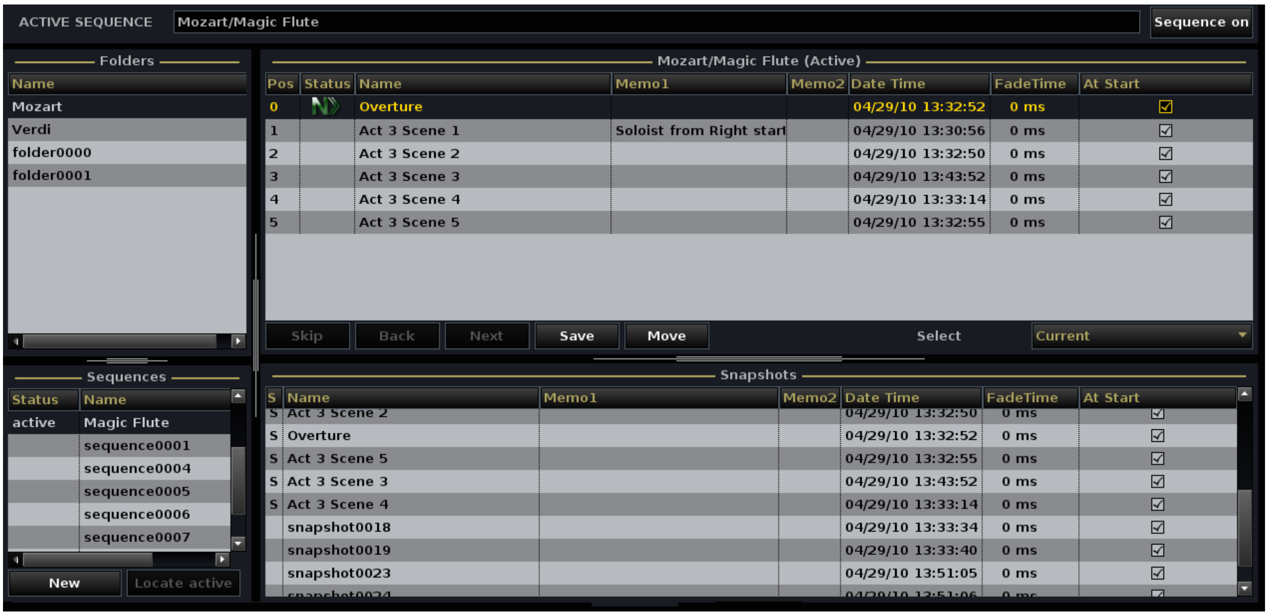


- SET NEXT – sets the selected snapshot to be the Next snapshot to be loaded in the sequence.
- SKIP – skips a snapshot.
- BACK – loads the ‘Back’ snapshot.
- NEXT – loads the Next snapshot.

Preparing for Sequence Play Out

1. Snapshots play out from the active sequence.

The name and file path appears in the ***ACTIVE SEQUENCE*** *box at the top of the* ***Sequences*** *display:*



If you have been viewing a different sequence, use the ACTIVE function to navigate back to the active sequence.

You can resize the display areas, by clicking and dragging the grey separator bars, so that only the active sequence is visible.

2. Then enable sequence automation, either by pressing the front panel **ON** button, or selecting **Sequence on**.

The ***Sequence on*** *button turns green; you are now ready to run the sequence:*

Running the Sequence in Order

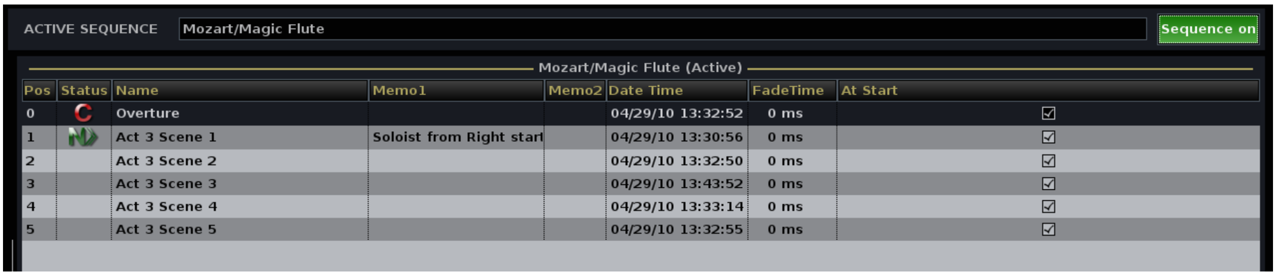
Usually you will want to play out the snapshots from the **Sequence** list in order – i.e. from top to bottom.

If this is the first time you've played out the sequence then the **N** flag will appear against the snapshot located at the top of the list. This tells you that this snapshot will be the next to play out when you press the **NEXT** button:

Pos	Status	Name	Memo1	Memo2	Date Time	FadeTime	At Start
0	N	Overture			04/29/10 13:32:52	0 ms	☑
1		Act 3 Scene 1	Soloist from Right start		04/29/10 13:30:56	0 ms	☑
2		Act 3 Scene 2			04/29/10 13:32:50	0 ms	☑
3		Act 3 Scene 3			04/29/10 13:43:52	0 ms	☑
4		Act 3 Scene 4			04/29/10 13:33:14	0 ms	☑
5		Act 3 Scene 5			04/29/10 13:32:55	0 ms	☑

1. Press **NEXT** to start the play out.

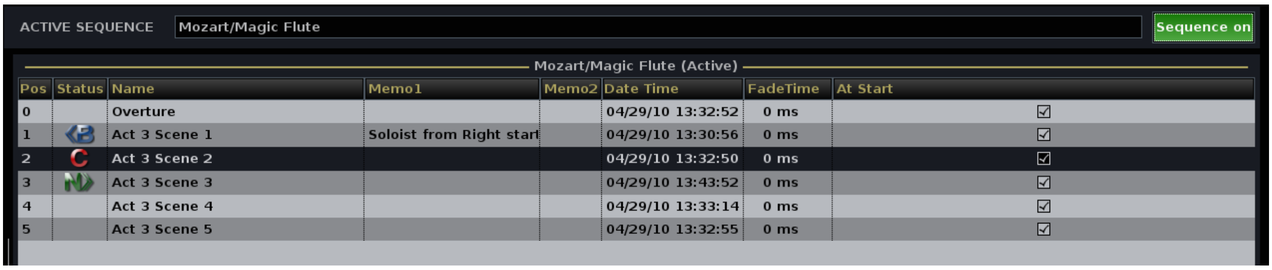
The snapshot is loaded according to any snapshot isolates and offsets you have applied. In addition, the loaded snapshot is marked with a ***C*** *for Current:*



Pos	Status	Name	Memo1	Memo2	Date Time	FadeTime	At Start
0	C	Overture			04/29/10 13:32:52	0 ms	☑
1	N	Act 3 Scene 1	Soloist from Right start		04/29/10 13:30:56	0 ms	☑
2		Act 3 Scene 2			04/29/10 13:32:50	0 ms	☑
3		Act 3 Scene 3			04/29/10 13:43:52	0 ms	☑
4		Act 3 Scene 4			04/29/10 13:33:14	0 ms	☑
5		Act 3 Scene 5			04/29/10 13:32:55	0 ms	☑

If the snapshot does not load, check your snapshot isolate settings.

2. Keep pressing **NEXT** to step down through the sequence, loading each snapshot in turn:



Pos	Status	Name	Memo1	Memo2	Date Time	FadeTime	At Start
0		Overture			04/29/10 13:32:52	0 ms	☑
1	B	Act 3 Scene 1	Soloist from Right start		04/29/10 13:30:56	0 ms	☑
2	C	Act 3 Scene 2			04/29/10 13:32:50	0 ms	☑
3	N	Act 3 Scene 3			04/29/10 13:43:52	0 ms	☑
4		Act 3 Scene 4			04/29/10 13:33:14	0 ms	☑
5		Act 3 Scene 5			04/29/10 13:32:55	0 ms	☑

Notice how the **N**, **C** and **B** indicators update:

- **N** indicates the Next snapshot to be loaded and therefore shows you exactly which snapshot you will recall when you press the **NEXT** button.
- **C** marks the Current snapshot. This is always the last snapshot loaded.
- **B** indicates the 'Back' snapshot. This is snapshot loaded before the current one.

3. If you want to play out the sequence from beginning to end, then keep pressing **NEXT** repeatedly.

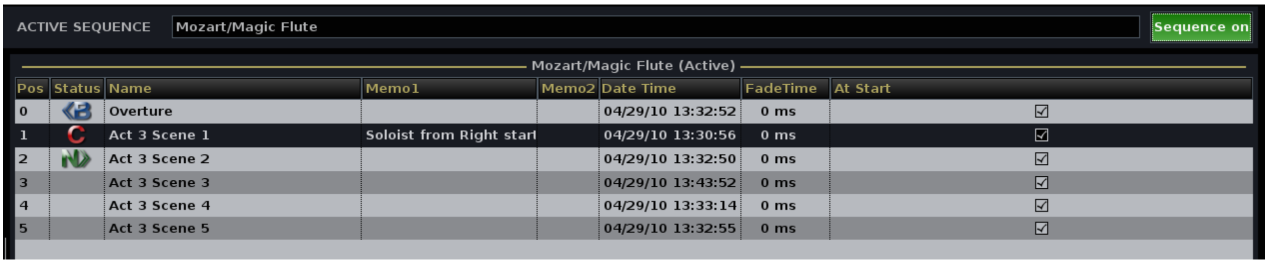
When you reach the last snapshot in the list, the sequence ends and the **NEXT** button performs no further function.

Loading the Back Snapshot

At any time, you can load the back snapshot to return to the previous snapshot's settings. For example, if an artist misses their cue you may need to quickly revert to the previous snapshot.

1. Press **BACK** to load the Back snapshot:

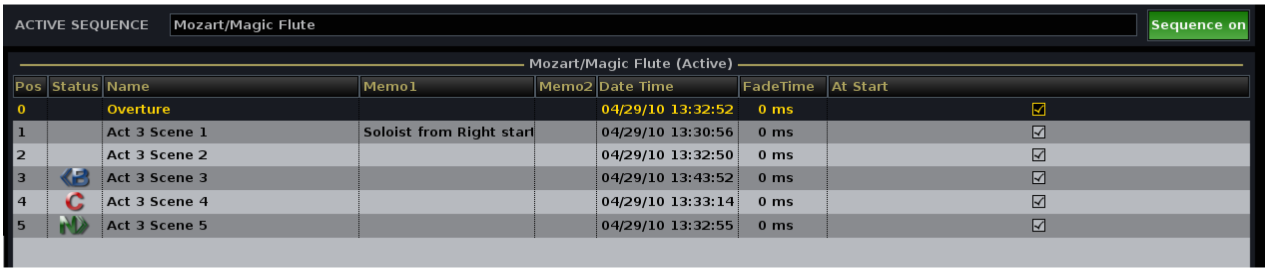
The Back snapshot loads and the Next, Current and Back indicators update accordingly:



Resetting the Sequence

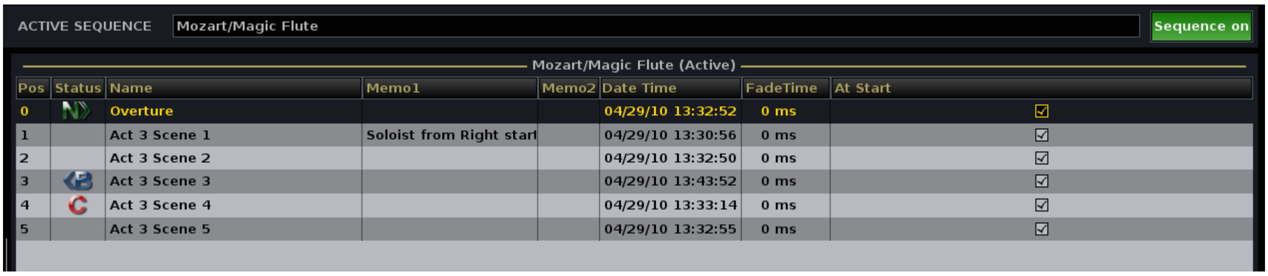
To reset the sequence, or start from a different snapshot position:

1. Select the snapshot you wish to return to, from the **Sequence** list:



2. Press **SET NEXT** to set the snapshot as the **N**ext snapshot.

The display updates accordingly:



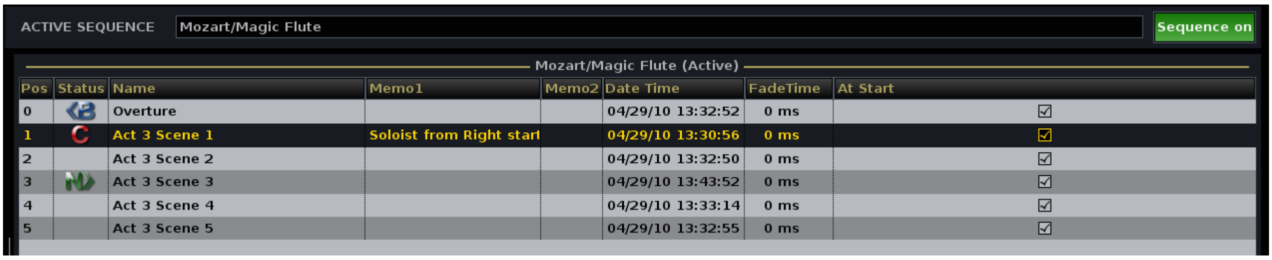
*The sequence will restart from this position when you press **NEXT**.*

Skipping a Snapshot or Snapshots

You can skip a single snapshot or multiple snapshots in the sequence if, for example, an act is cut from the performance.

1. Press **SKIP** to skip the next snapshot in the sequence list.

*The **Next** indicator moves one position down the list:*



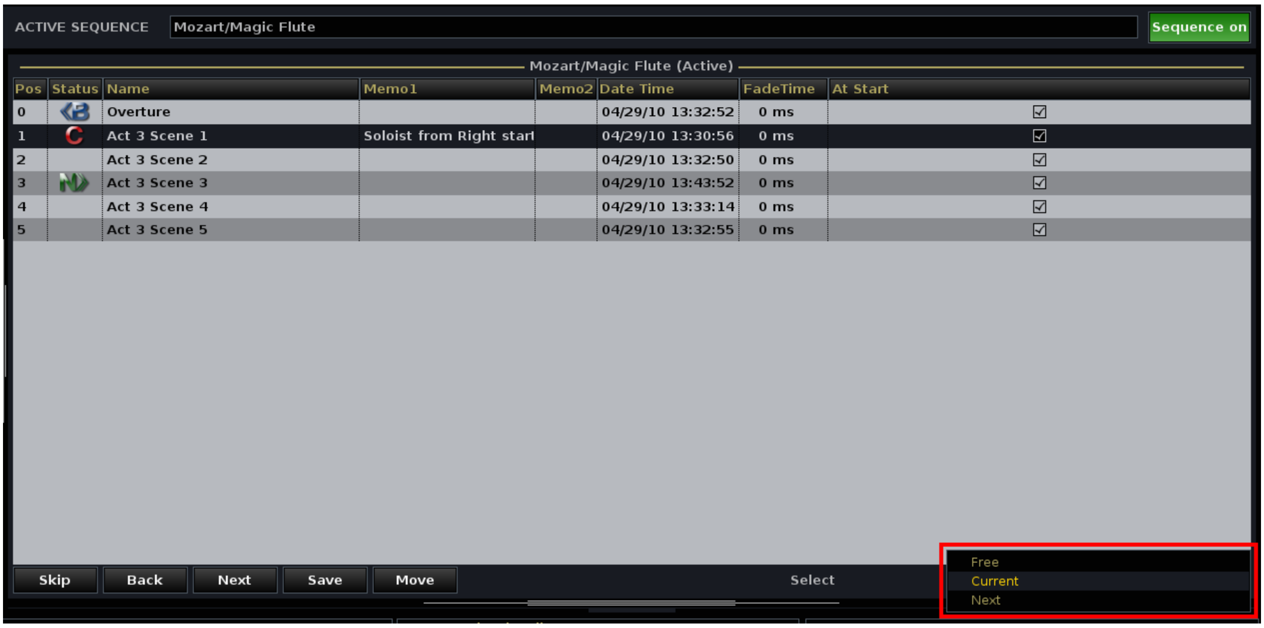
By pressing **SKIP** multiple times, you can skip more than one snapshot.

2. Now press **NEXT** to load the next snapshot in the sequence.

Selecting Snapshots in the Sequence List

The **Select** option determines how snapshot selections within the **Sequence** list behave and how the list scrolls. For example, during a dress rehearsal, you may always want the selected snapshot to be your current snapshot so that it is easy to update. While at other times, you want to select snapshots independently of the **NEXT** playout button.

1. Click on the drop-down options beside **Select** at the bottom right hand corner of the **Sequence** list. You can choose from:



- **Free** – allows free selection of snapshots from the **Sequence** list. Choose this mode so that the display will stay fixed on the selected snapshot and not revert to the Current or Next snapshot after pressing **NEXT**.
- **Current** – sets the **Select** mode to follow the Current snapshot. This forces the selected snapshot to revert to the current snapshot each time you press the **NEXT** button.
- **Next** - sets the **Select** mode to follow the Next snapshot. This forces the selected snapshot to revert to the next snapshot each time you press the **NEXT** button.

Snapshot Cross Fade (X-Fade)

When playing out snapshots from a sequence, you may choose to cross fade from one snapshot to another.

For each snapshot, you can decide which modules will cross fade (e.g. faders, mutes, EQ, etc.), set the cross fade time, and whether switched functions, such as mutes, change state at the start or the end of the cross fade. These parameters are saved with each snapshot whenever it is saved or updated.

When the sequence plays out, with X-Fade enabled, then the cross fade parameters for the snapshot you are fading *to* are applied.

For example, if Snapshot 0001 has a cross fade time of 1 second and Snapshot 0002 a cross fade time of 2 seconds:

- If you step from Snapshot 0001 to 0002, using the **NEXT** button, a cross fade time of 2 seconds is applied.
- If you then step back from Snapshot 0002 to Snapshot 0001, using the **BACK** button, a cross fade time of 1 second is applied.

Note that cross fade parameters are only applied when you play out snapshots from a sequence, and not when you load a snapshot from the Snapshots display.

Preparing a Sequence/Default X-Fade Parameters

The most efficient way to prepare a sequence, for snapshot crossfades, is as follows:

1. First, create a new sequence for the show from the **Sequences** display.

2. Make this sequence active.

3. Then *BEFORE* you save any snapshots, set the default cross fade parameters from the System Settings display.

This will save time editing the individual snapshot cross fade parameters later.

4. Now set up the console for the show. When you are ready, save a snapshot for each new console setting.

Each snapshot is saved with the default cross fade parameters you set in step 3.

5. Move the snapshots so they are in the correct order for play out.

6. Edit the cross fade parameters for individual snapshots as required.

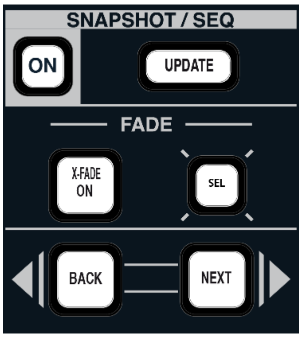
Selecting Modules to Cross Fade

By default, when snapshots are saved or updated, they are set so that no modules will cross fade. Therefore, the next step is to select, for each snapshot, which modules you wish to fade.

To do this, you will need to select the modules to fade, and then store these settings with the snapshot by updating it. The most efficient work flow is as follows:

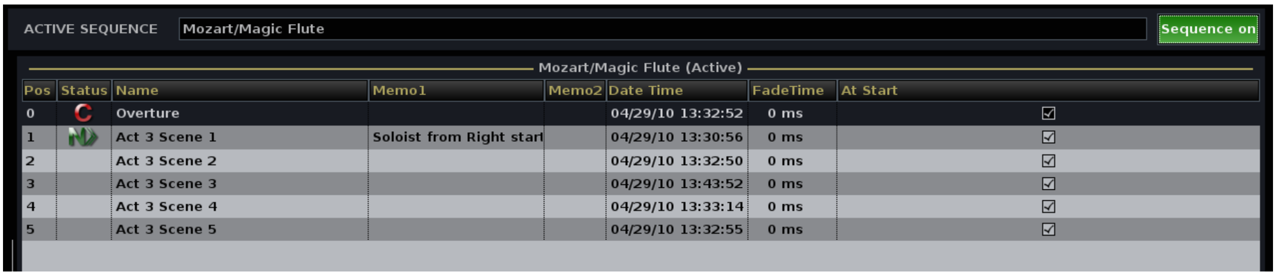
1. Select the **Sequences** display so that you are viewing your active sequence.

2. Enable sequence automation by pressing the **ON** button on the SNAPSHOT/SEQUENCE front panel:

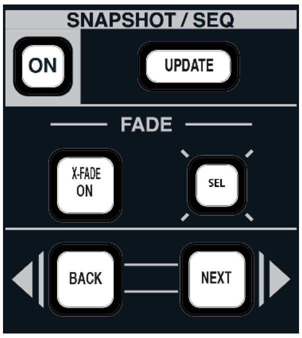


3. Press **NEXT** to load the next snapshot.

The next snapshot is loaded into the console; the console should update; and the snapshot is marked as Current.



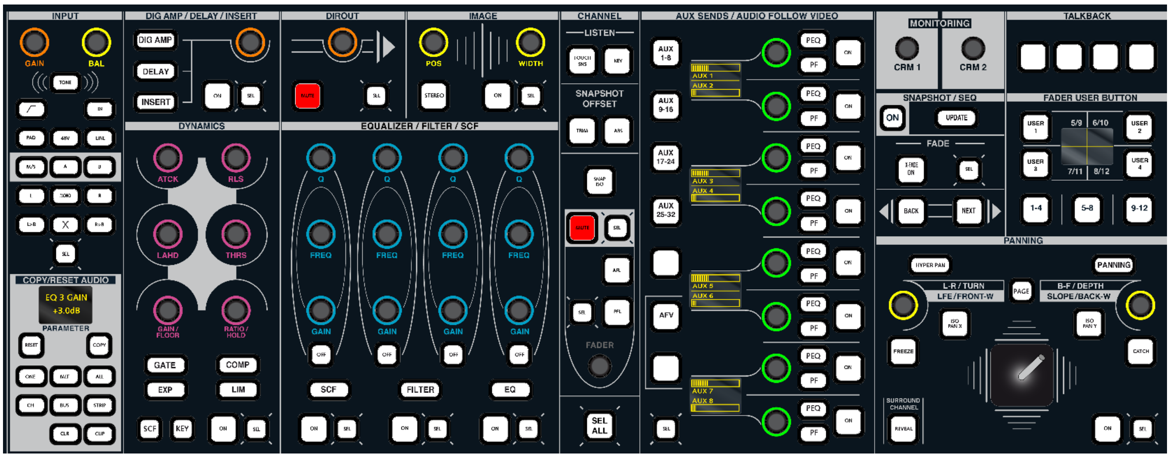
4. Now turn on the FADE **SEL** button on the SNAPSHOT/SEQUENCE panel - the button flashes (red) when active:



5. And select the audio module(s) you wish to cross fade, by enabling the **SEL** buttons on the Central Control Section, see Selecting Channel Parameters.

To clear down any existing selections, toggle the **SEL ALL** button (this selects and then deselects all modules). This ensures that there no "hidden" selections.

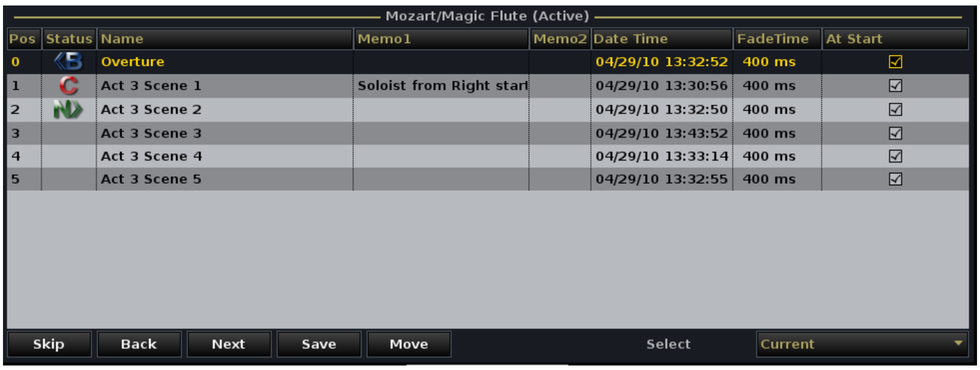
For example, to cross fade the fader and mute, press **SEL** beside the fader, and mute sections:



6. When you have made your selections, turn off the FADE **SEL** button.

7. Finally, store your selections by updating the snapshot.

The X-Fade module selections are stored with the snapshot:


Mozart/Magic Flute (Active)

Pos	Status	Name	Memo1	Memo2	Date Time	FadeTime	At Start
0		Overture			04/29/10 13:32:52	400 ms	☑
1		Act 3 Scene 1	Soloist from Right start		04/29/10 13:30:56	400 ms	☑
2		Act 3 Scene 2			04/29/10 13:32:50	400 ms	☑
3		Act 3 Scene 3			04/29/10 13:43:52	400 ms	☑
4		Act 3 Scene 4			04/29/10 13:33:14	400 ms	☑
5		Act 3 Scene 5			04/29/10 13:32:55	400 ms	☑

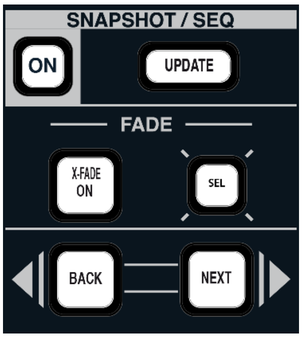
Skip | Back | Next | Save | Move | Select | Current

8. Press **NEXT** and repeat steps 4 to 7 for each of the snapshots in the sequence.

Running the Sequence with Cross Fades

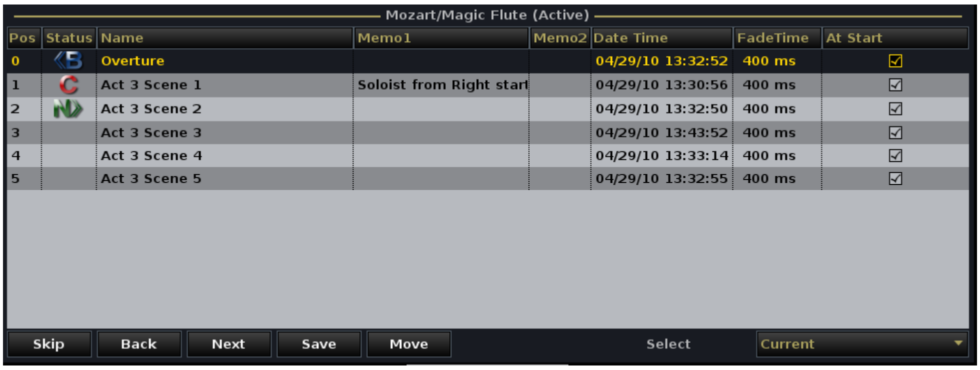
To play out the sequence with cross fades enabled:

1. Turn on the **XFADE ON** button on the SNAPSHOT/SEQUENCE panel - the button illuminates when active:



Note that you cannot enable the button unless SEQUENCE **ON** is already active.

2. Now run the sequence as you would normally. In our example, press **NEXT** and the console will cross fade from **Act 3 Scene 1** to **Act 3 Scene 2**:



The crossfade parameters are defined by the snapshot you are fading to:

- Continuous parameters (e.g. fader level) fade from one setting to the other over the Fadetime – in our example, 400 ms.
- Switched parameters (e.g. mute) will change state either at the start or end of the fade – in our example, at the start of the fade.

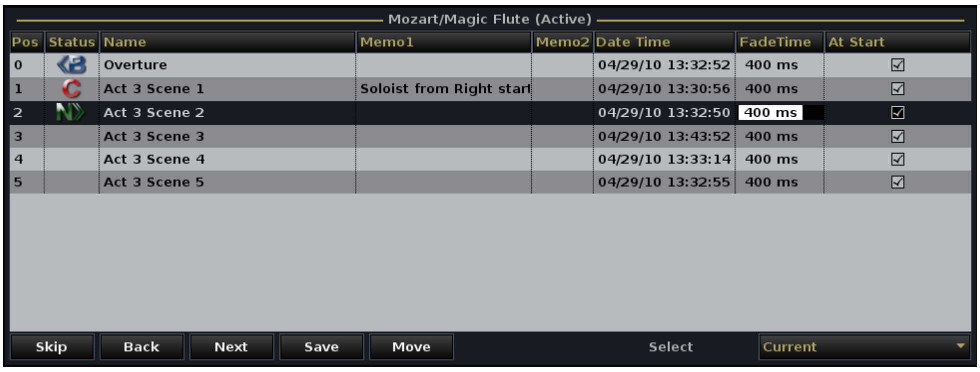
If nothing happens when you recall your snapshot, check the following:

- Have you selected and stored the modules to cross fade?
- Are those modules protected from snapshot recall, either using snapshot isolate?
- Have you entered a cross fade time greater than 0ms?!
- Make sure that you're not cross fading between snapshots with the same parameters!

Editing Snapshot Cross Fade Parameters

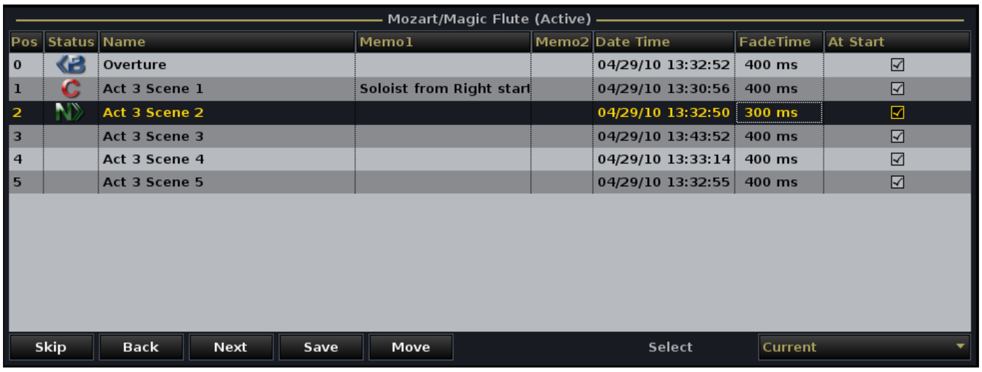
You may edit the X-Fade Switch Mode and/or Fade Time parameters for an individual snapshot. However, note that if you update the snapshot, then the default crossfade parameters are applied, overwriting any individual edits.

1. From the **Sequences** display, focus on the snapshot you wish to edit and click on the **Fade Time (ms)**:



Click once to select all the existing text (white) or twice (black cursor) to modify the existing name.

2. Enter the new fade time in ms and press Enter:

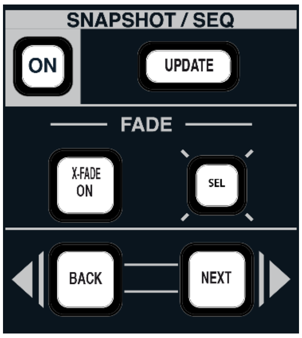


3. To change the switch mode, check or uncheck the **At Start** column:

- **at start** checked – switched functions change state at the start of the cross fade.
- **at start** unchecked – switched functions change state at the end of the cross fade.

SNAPSHOT/SEQUENCE Front Panel Summary

The SNAPSHOT/SEQUENCE panel provides quick access to the following functions:



- **ON** - press to enable (or disable) sequence automation.
- **UPDATE** - press to update the current snapshot. Note that the button *only* updates if sequence automation is enabled (**ON**).
- **FADE: XFADE ON** - press to enable (or disable) snapshot crossfades.
- **FADE: SEL** - used to select audio module(s) for snapshot crossfade.
- **BACK** - press to load the back snapshot.
- **NEXT** - press the load the next snapshot.

Snapshot Offsets

Whenever a snapshot is recalled, either from the **Snapshots** or **Sequences** display, it may be recalled with offset parameters.

For example, if you are running an opera where different soloists will perform on different nights, you can store a basic set of snapshots for the show, and then apply offset parameters for soloist A, soloist B, etc. without affecting the original snapshot values.

Any number of offsets may be applied, and may include a mixture of absolute and trim values:

- Use an absolute offset when you want a new static value throughout the sequence – for example, to apply a new EQ setting for soloist B.
- Use a trim offset when you want to keep the relative changes from the snapshots within the sequence – for example, to make soloist B's fader level +3dB louder throughout the show.

> Whilst snapshot offsets are designed for sequence automation play out, active offsets are applied to *any* snapshot load.

The active snapshot offsets are known as the Current Trim Set. This is a temporary buffer which you can update at any time allowing you to modify offset parameters during a show. For example, if soloist B sings louder than during rehearsal you may wish to adjust their trim offset!

You can also store offsets by saving the contents of the **Current Trim Set** into memories called Oversnaps. Each oversnap can store any number of offset parameters, and different combinations of oversnaps may be added to the Current Trim Set. This allows you to make any combination of offsets active – for example, to combine the offsets for soloist A with those for trombonist B.

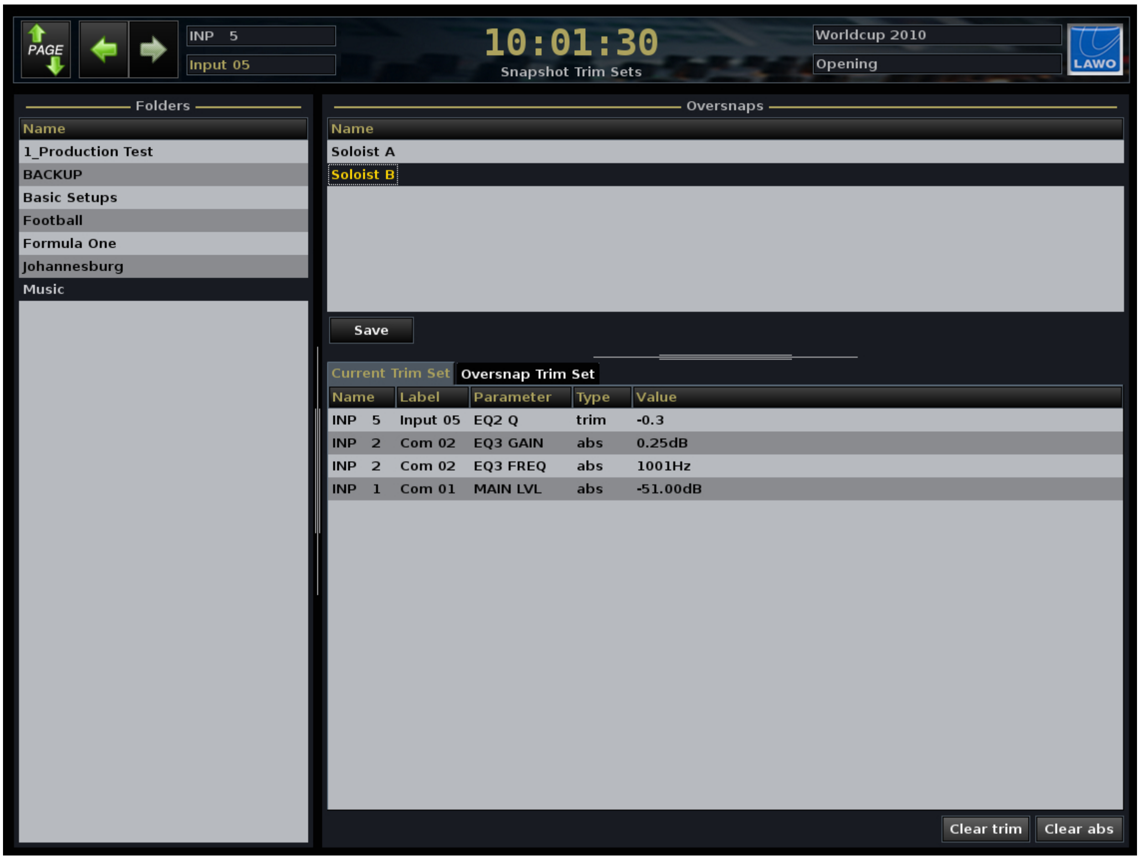
Oversnaps are stored separately from snapshots within your production folder; you may save up to 999 oversnaps per folder.

> When trimming input GAIN, you may only trim the SOURCE gain for mic/line inputs, and not for fixed gain or digital sources. In other words, you cannot trim the I/O DSP gain (Volume).
>
> For any type of input, you can apply trim to the INMIX channel input gain.

The Snapshot Trim Sets Display

Snapshot offsets are managed from the **Snapshot Trim Sets** display.

1. Press the **SNAP/SEQ** button, located on the SCREEN CONTROL panel, to view the **Snapshot Trim Sets** display:



On the left, you will see a list of the Folders within the current production.

The upper part of the display shows the names of any **Oversnaps** stored within the selected folder.

The lower part of the display shows either the **Current Trim Set** or **Oversnap Trim Set** – click on the headings to toggle between the options:

- **Current Trim Set** - lists the active snapshot offset parameters. If the list is empty, then a snapshot will load with its original values. If the list contains offsets, then the offset values will be applied. Use the **Current Trim Set** to update the active offset parameters. This can be done live from the console, or by loading an oversnap.
- **Oversnap Trim Set** – lists the offset parameters which are stored in the selected oversnap. This allows you to view offset parameters before you load the oversnap.

During a live show, keep the lower part of the display on **Current Trim Set**. This way you can be sure that you are viewing the active offset parameters which will be applied to your next snapshot load.

Adding Snapshot Offsets to the Current Trim Set

The **Current Trim Set** lists the active snapshot offset parameters. Offsets listed here are applied to *all* snapshot loads.

To update the list, you can either add offset parameters live from the console, or load a stored oversnap. Here we will look at adding offsets to the **Current Trim Set** from the console.

> We are going to assume that you are offsetting snapshots within a prepared sequence. However, the contents of the **Current Trim Set** are applied to *all* snapshot loads, not just those from a sequence play out.

When adding offsets, the console compares the current desk position to the value stored in the last loaded snapshot. Therefore, it's a good idea to start by loading the snapshot you want to use as a reference point for the comparison. (In our workflow, this will be the last snapshot played out from the sequence.)

1. Load a snapshot from your sequence.

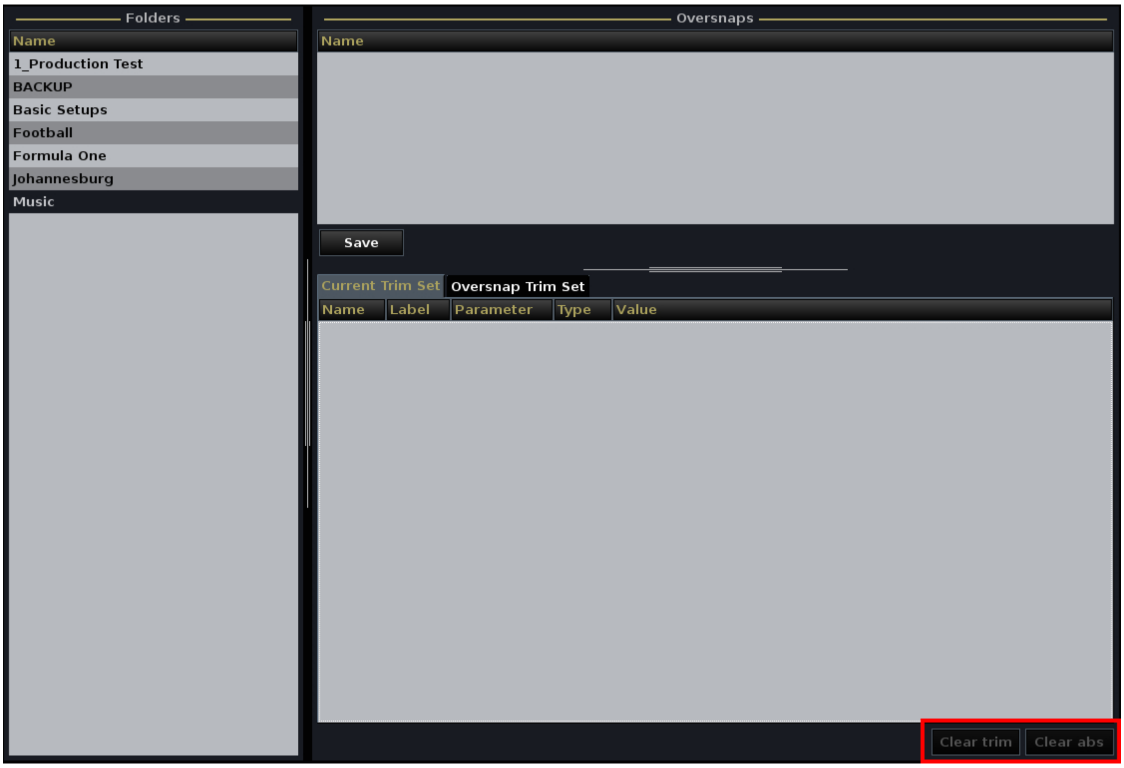
The console updates to the stored positions.

2. Open the Snapshot Trim Sets display.

3. Click on **Current Trim Set** to view any active snapshot offsets.

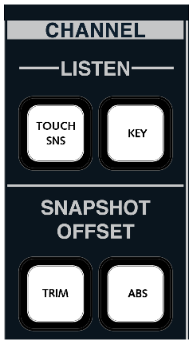
*The **Current Trim Set** will be empty, unless you have already been working with snapshot offsets.*

4. You can clear the **Current Trim Set** by clicking on the **clear abs** and **clear trim** buttons:



5. Now adjust the console parameters you wish to offset – for example, some fader levels and an EQ setting.

Let's assume that we want the new EQ setting to be static for the whole show (an absolute offset), but that the level changes should be relative (trim offsets).

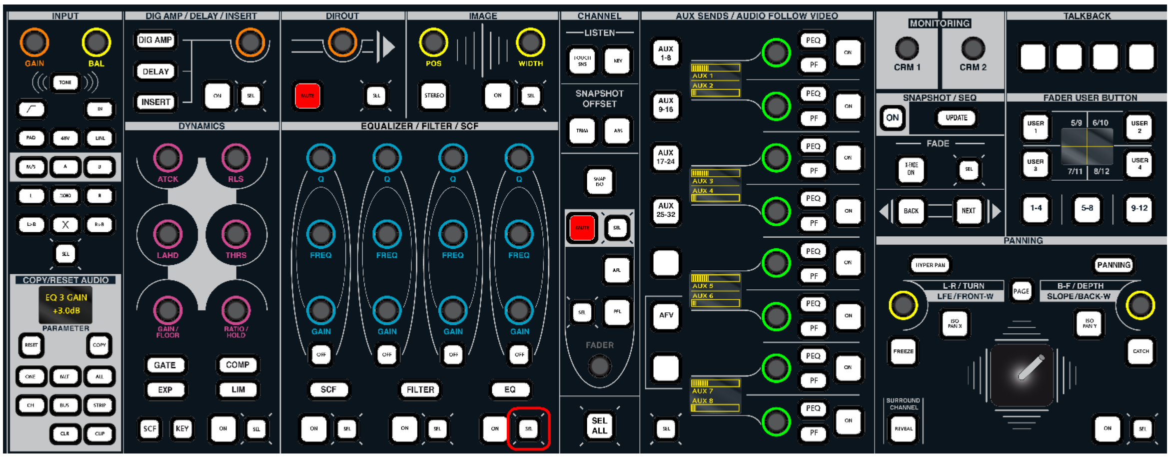


6. Press the **ABS** button, on the SNAPSHOT OFFSET panel, to activate the absolute offset parameter selection.

*The **ABS** button flashes.*

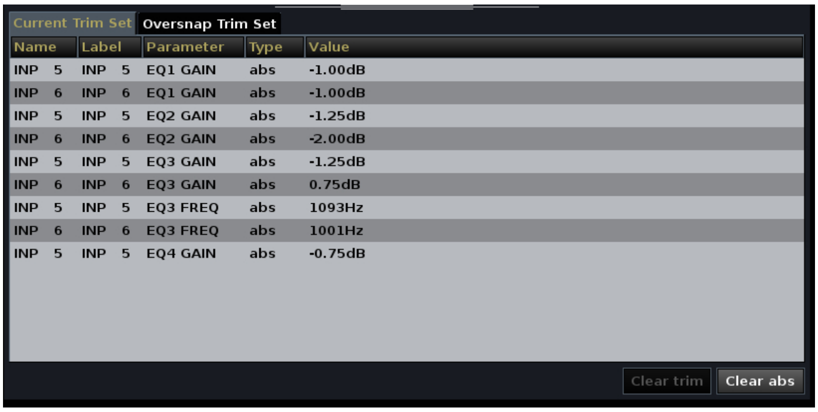
7. Assign the channel with the EQ setting to the Central Control Section, by pressing its fader **SEL** button:

*Any audio modules which have a different setting to that stored in the last loaded snapshot are displayed with green **SEL** buttons – in our example, the **SEL** button on the EQ section:*



8. To add the new EQ setting to the **Current Trim Set**, press the green EQ **SEL** button.

*The **SEL** button turns red and each modified EQ parameter is added to the **Current Trim Set** as an absolute (**ABS**) offset on the **Snapshot Trim Sets** display:*



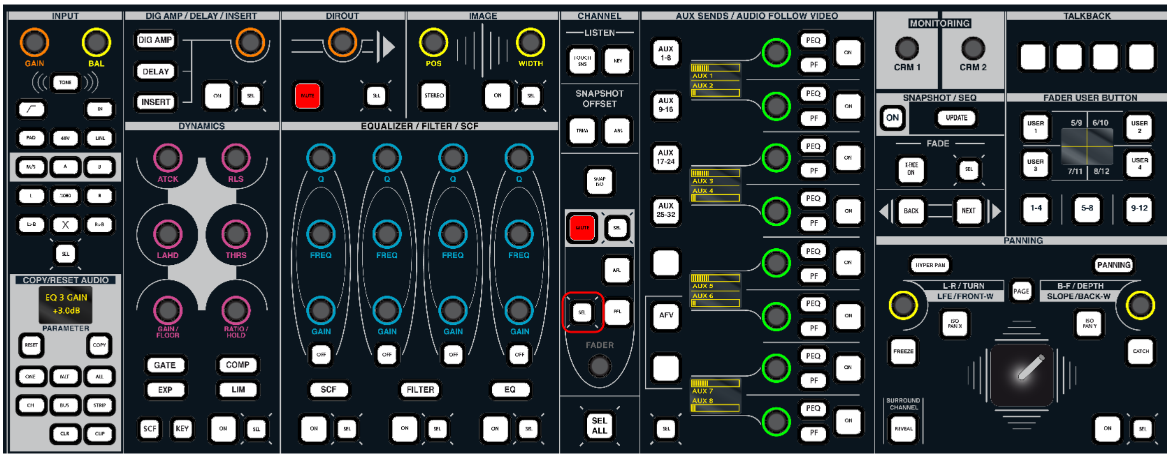
Name	Label	Parameter	Type	Value
INP 5	INP 5	EQ1 GAIN	abs	-1.00dB
INP 6	INP 6	EQ1 GAIN	abs	-1.00dB
INP 5	INP 5	EQ2 GAIN	abs	-1.25dB
INP 6	INP 6	EQ2 GAIN	abs	-2.00dB
INP 5	INP 5	EQ3 GAIN	abs	-1.25dB
INP 6	INP 6	EQ3 GAIN	abs	0.75dB
INP 5	INP 5	EQ3 FREQ	abs	1093Hz
INP 6	INP 6	EQ3 FREQ	abs	1001Hz
INP 5	INP 5	EQ4 GAIN	abs	-0.75dB

9. Now repeat the last three steps, but select the SNAPSHOT OFFSET **TRIM** button.

*The **TRIM** button flashes to show that you are now selecting trim offset parameters.*

10. Assign one of the channels with a new fader level to the Central Control Section.

*This time the fader **SEL** button lights (green) to indicate that the level has changed from that stored in the last loaded snapshot:*

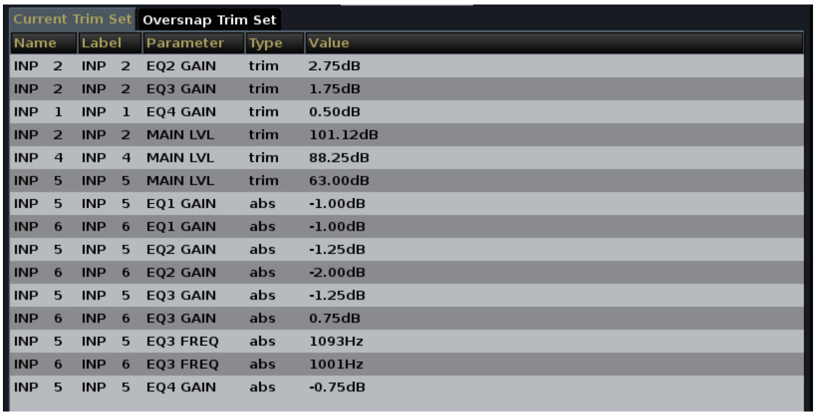


11. Press the green **SEL** button to add the trimmed fader level to the **Current Trim Set**.

*The **SEL** button turns red and the trimmed fader level is added to the **Current Trim Set** on the **Snapshot Trim Sets** display.*

12. Repeat for each new fader level, by assigning the channel to the Central Control Section and then pressing the green parameter **SEL** buttons.

*As each offset parameter is selected, it is added to the **Current Trim Set** on the **Snapshot Trim Sets** display:*


Current Trim Set | Oversnap Trim Set

Name	Label	Parameter	Type	Value
INP 2	INP 2	EQ2 GAIN	trim	2.75dB
INP 2	INP 2	EQ3 GAIN	trim	1.75dB
INP 1	INP 1	EQ4 GAIN	trim	0.50dB
INP 2	INP 2	MAIN LVL	trim	101.12dB
INP 4	INP 4	MAIN LVL	trim	88.25dB
INP 5	INP 5	MAIN LVL	trim	63.00dB
INP 5	INP 5	EQ1 GAIN	abs	-1.00dB
INP 6	INP 6	EQ1 GAIN	abs	-1.00dB
INP 5	INP 5	EQ2 GAIN	abs	-1.25dB
INP 6	INP 6	EQ2 GAIN	abs	-2.00dB
INP 5	INP 5	EQ3 GAIN	abs	-1.25dB
INP 6	INP 6	EQ3 GAIN	abs	0.75dB
INP 5	INP 5	EQ3 FREQ	abs	1093Hz
INP 6	INP 6	EQ3 FREQ	abs	1001Hz
INP 5	INP 5	EQ4 GAIN	abs	-0.75dB

Note that the trim offset is the difference in level between the current fader position and the level stored in the last loaded snapshot – for example, if the snapshot loads a main fader level of -6dB, and you have moved the fader to +4dB, then the trim offset is +10dB.

13. When you have finished selecting offset parameters, turn off the SNAPSHOT OFFSET **ABS** and **TRIM** buttons.

Note that the **Current Trim Set** is a temporary buffer and its contents are not saved other than in the system's warm start data. To save your offsets so that they may be recalled at a later date, save an oversnap.

Recalling Snapshots with Offsets

As soon as you have added offset parameters to the **Current Trim Set**, these offsets are active. This means that *any* snapshot loaded from this point on, either from a sequence or from a snapshot load, will have the **Current Trim Set** offsets applied:

Current Trim Set | Oversnap Trim Set

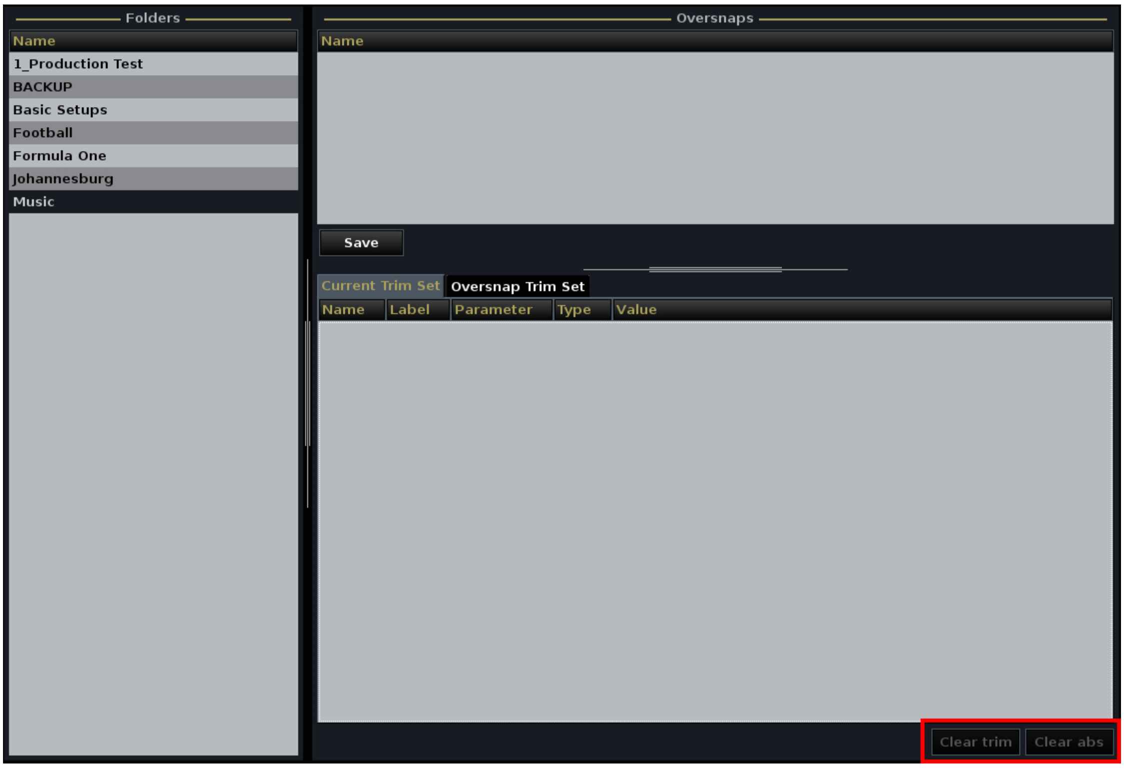
Name	Label	Parameter	Type	Value
INP 2	INP 2	EQ2 GAIN	trim	2.75dB
INP 2	INP 2	EQ3 GAIN	trim	1.75dB
INP 1	INP 1	EQ4 GAIN	trim	0.50dB
INP 2	INP 2	MAIN LVL	trim	101.12dB
INP 4	INP 4	MAIN LVL	trim	88.25dB
INP 5	INP 5	MAIN LVL	trim	63.00dB
INP 5	INP 5	EQ1 GAIN	abs	-1.00dB
INP 6	INP 6	EQ1 GAIN	abs	-1.00dB
INP 5	INP 5	EQ2 GAIN	abs	-1.25dB
INP 6	INP 6	EQ2 GAIN	abs	-2.00dB
INP 5	INP 5	EQ3 GAIN	abs	-1.25dB
INP 6	INP 6	EQ3 GAIN	abs	0.75dB
INP 5	INP 5	EQ3 FREQ	abs	1093Hz
INP 6	INP 6	EQ3 FREQ	abs	1001Hz
INP 5	INP 5	EQ4 GAIN	abs	-0.75dB

See Running a Sequence and Loading a Snapshot for details.

Clearing Snapshot Offsets

To disable snapshot offsets, you must clear the **Current Trim Set**.

1. Click on the **clear abs** button to clear all absolute snapshot offsets.
2. And click on **clear trim** to clear all trim offsets:



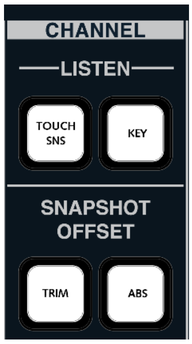
Once the **Current Trim Set** list is empty, snapshots will be loaded with their original values.

You may update the snapshot offsets in order to clear or modify a single offset parameter.

Updating Snapshot Offsets

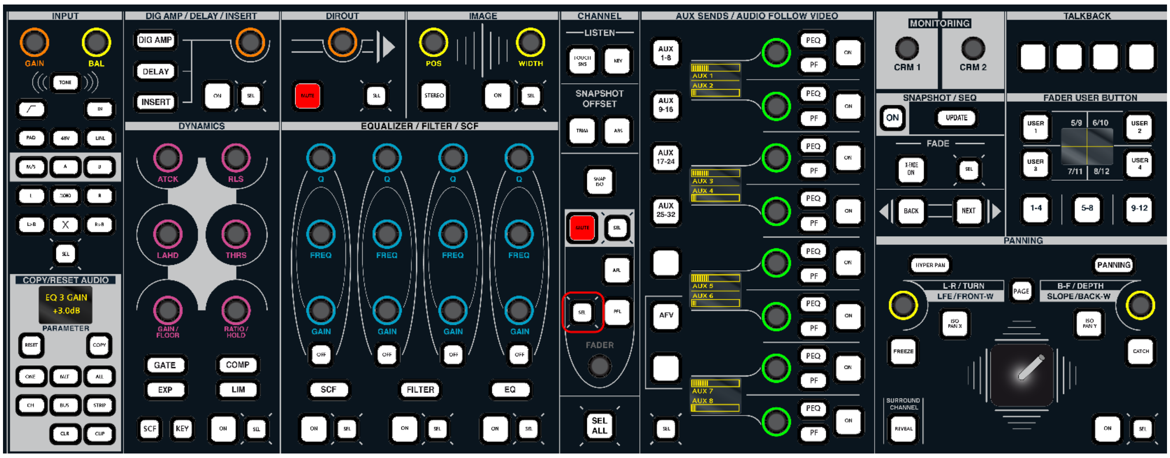
To update a snapshot offset - for example, if Soloist B sings louder than in rehearsal:

1. Press the SNAPSHOT OFFSET **TRIM** button.



2. Assign the Soloist B channel to the Central Control Section:

*The fader **SEL** button will be red as this parameter already has an active offset:*



3. Adjust the fader level to the new setting.

*The parameter **SEL** button turns orange.*

4. Press the orange parameter **SEL** button to confirm the new setting.

*The **SEL** button turns red and the trim offset updates within the **Current Trim Set***

Current Trim Set | Oversnap Trim Set

Name	Label	Parameter	Type	Value
INP 2	INP 2	EQ2 GAIN	trim	2.75dB
INP 2	INP 2	EQ3 GAIN	trim	1.75dB
INP 1	INP 1	EQ4 GAIN	trim	0.50dB
INP 2	INP 2	MAIN LVL	trim	101.12dB
INP 4	INP 4	MAIN LVL	trim	88.25dB
INP 5	INP 5	MAIN LVL	trim	63.00dB
INP 5	INP 5	EQ1 GAIN	abs	-1.00dB
INP 6	INP 6	EQ1 GAIN	abs	-1.00dB
INP 5	INP 5	EQ2 GAIN	abs	-1.25dB
INP 6	INP 6	EQ2 GAIN	abs	-2.00dB
INP 5	INP 5	EQ3 GAIN	abs	-1.25dB
INP 6	INP 6	EQ3 GAIN	abs	0.75dB
INP 5	INP 5	EQ3 FREQ	abs	1093Hz
INP 6	INP 6	EQ3 FREQ	abs	1001Hz
INP 5	INP 5	EQ4 GAIN	abs	-0.75dB

5. Alternatively, to remove the trim offset altogether, press the red parameter **SEL** button.

*The **SEL** button returns to green and the Main LVL offset is removed from the **Current Trim Set**.*

6. Remember to deselect the SNAPSHOT OFFSET **ABS** and **TRIM** buttons when you have finished updating offsets.

If you don't, and adjust a parameter with an active offset, then you will update the offset!

Oversnaps

At any time, you may save the contents of the Current Trim Set into a memory called an **Oversnap**. This allows you to recall offset parameters at a later date.

Each oversnap may store any number of offset parameters. And, different combinations of oversnaps may be loaded back to the **Current Trim Set**. This allows you to make a combination of offsets active – for example, to combine the offsets for soloist A with those for trombonist B.

Oversnaps are stored within your production folder; you may save up to 999 oversnaps per folder.

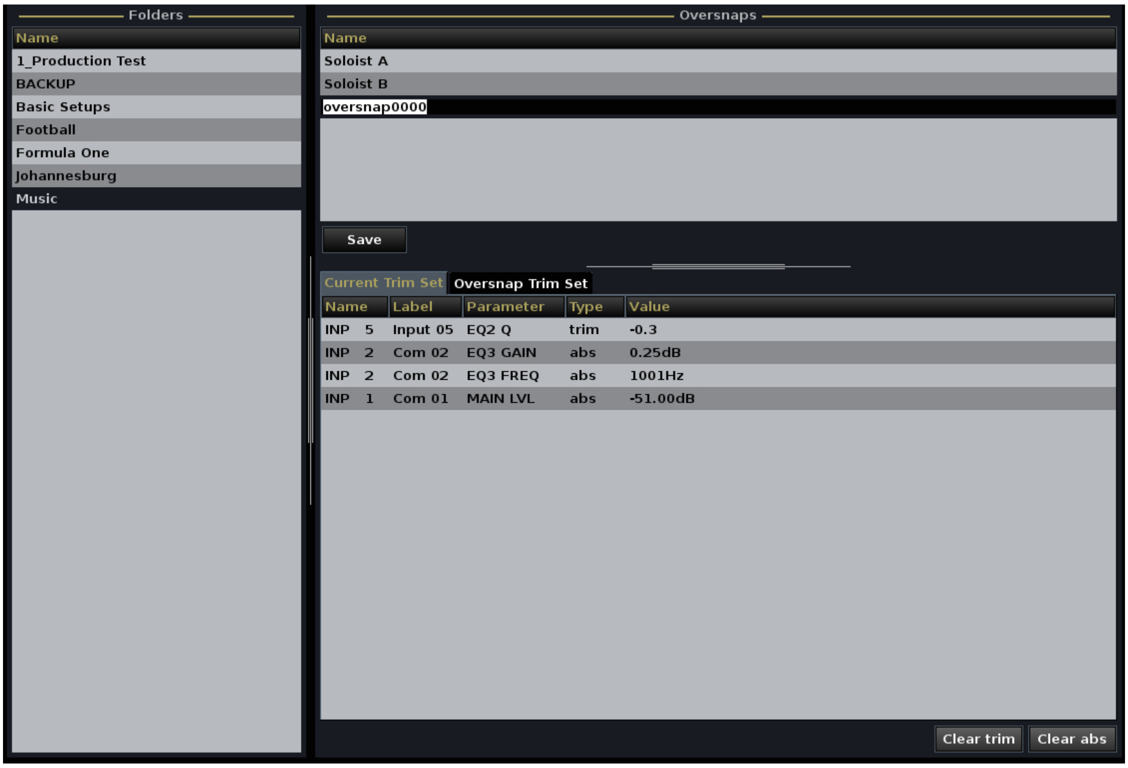
> Note that you must use oversnaps to store and recall snapshot offset parameters. (The **Current Trim Set** is a temporary buffer which is saved in the system's warm start data to protect you from a system restart. However, if you clear the **Current Trim Set**, or change production, then any active snapshot offsets will be lost.)

Saving an Oversnap

To save the contents of the **Current Trim Set**:

1. Click on the **save** button at the bottom of the **Oversnaps** list.

*A new oversnap is saved into the current Folder (e.g. **Music**) and appears at the bottom of the list with a default name (e.g. **oversnap0000**):*

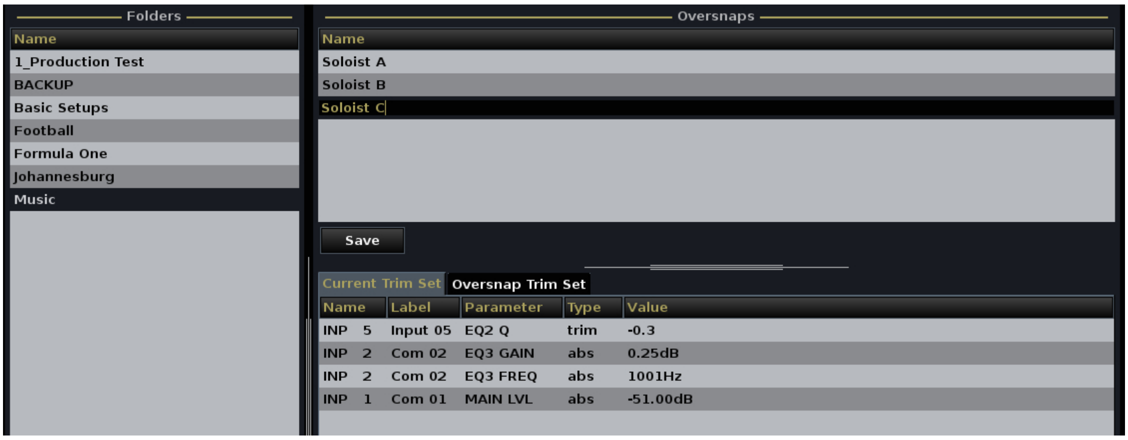


Renaming an Oversnap

1. Click on the oversnap name using the trackball.

> Click once to select all the existing text (white) or twice (black cursor) to modify the existing name.

2. Enter a new name from the keyboard.
3. When you have finished, press the Enter button, on the keyboard, to confirm the new name:



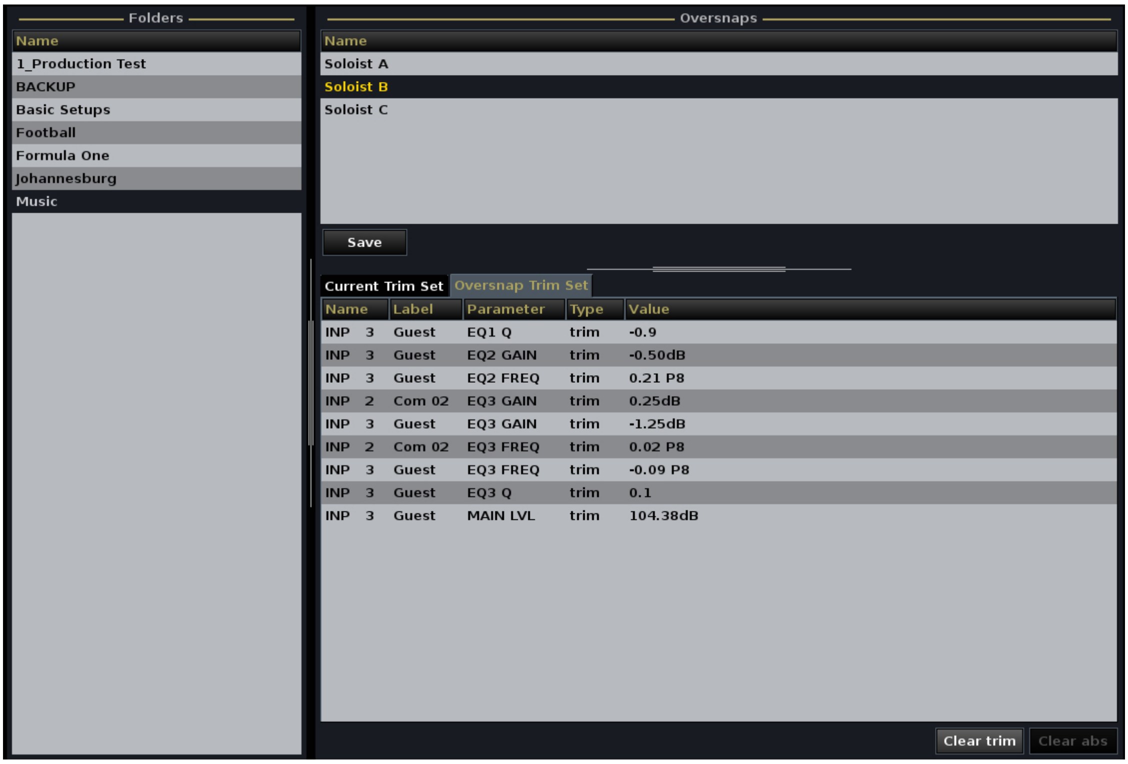
4. Or, if you make a mistake or want to exit without making any changes, press the **Esc** button on the keyboard.

Checking the Contents of an Oversnap

You may check what offsets are stored in an oversnap as follows:

1. Select the oversnap you wish to interrogate.
2. Click on the **Oversnap Trim Set** heading in the lower half of the display.

The trim set updates to show the contents of the selected oversnap:



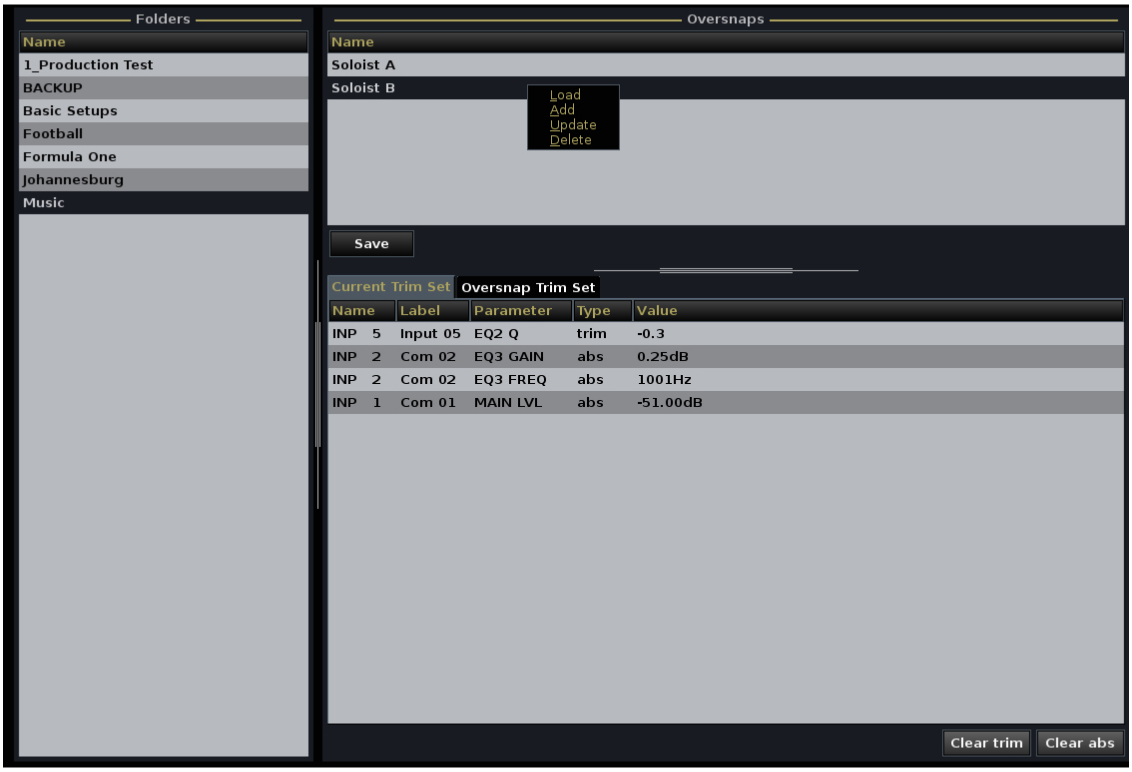
3. When you have finished interrogating stored oversnaps, it is a good idea to switch back to the **Current Trim Set**. This way you can be sure that you are viewing the active offset parameters which will be applied to your next snapshot load.

Recalling an Oversnap

When recalling snapshot offsets, you can choose to either load or add an oversnap to the **Current Trim Set**. Any offsets listed within the **Current Trim Set** will then be applied to subsequent snapshot loads.

1. Right-click on the oversnap, and select either **Load** or **Add**:

- **Load** – replaces the contents of the **Current Trim Set** with the stored offsets.
- **Add** – adds the stored offsets to the existing parameters within the **Current Trim Set**. For example, to combine the snapshot offset parameters saved for different artists.



*The contents of the **Current Trim Set** update accordingly.*

If an added oversnap contains parameters for an identical audio module to that in the existing **Current Trim Set**, then the added parameter replaces the existing one.

Updating an Oversnap

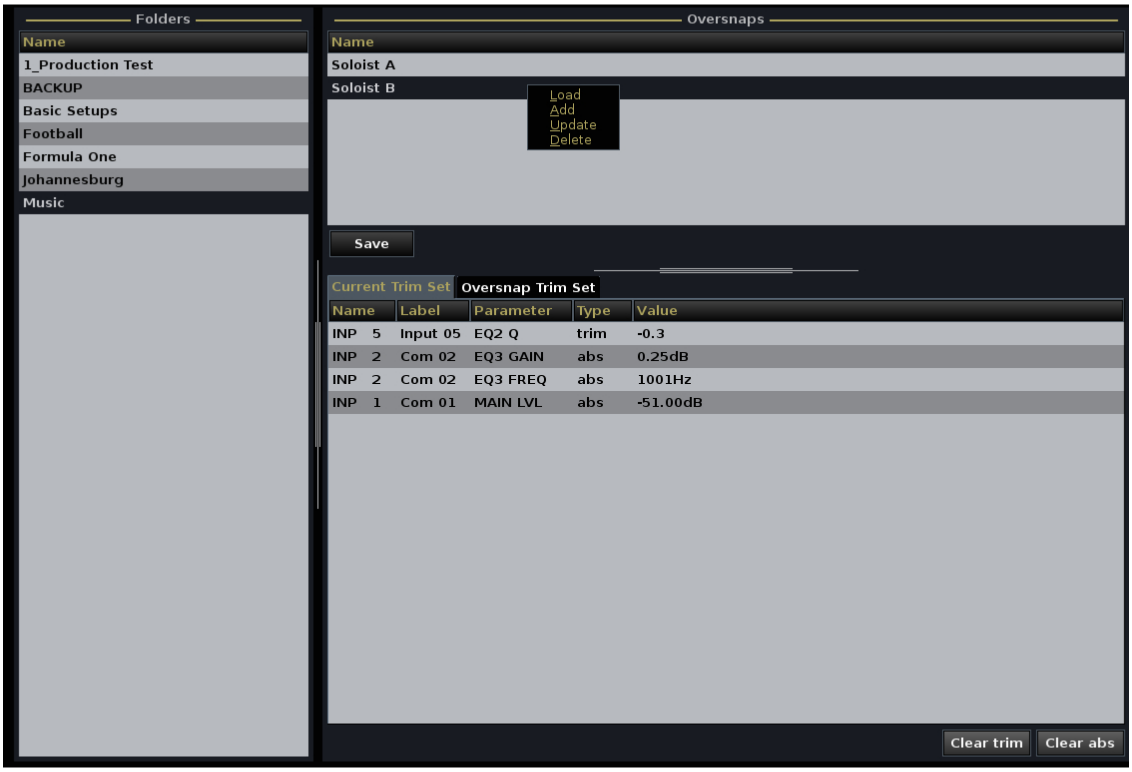
You can overwrite the contents of an oversnap, with the **Current Trim Set**, by using the **update** button:

1. Add the offset parameters you wish to store to the **Current Trim Set**.

> To edit the contents of an existing oversnap: load the oversnap first; then adjust the snapshot offsets.

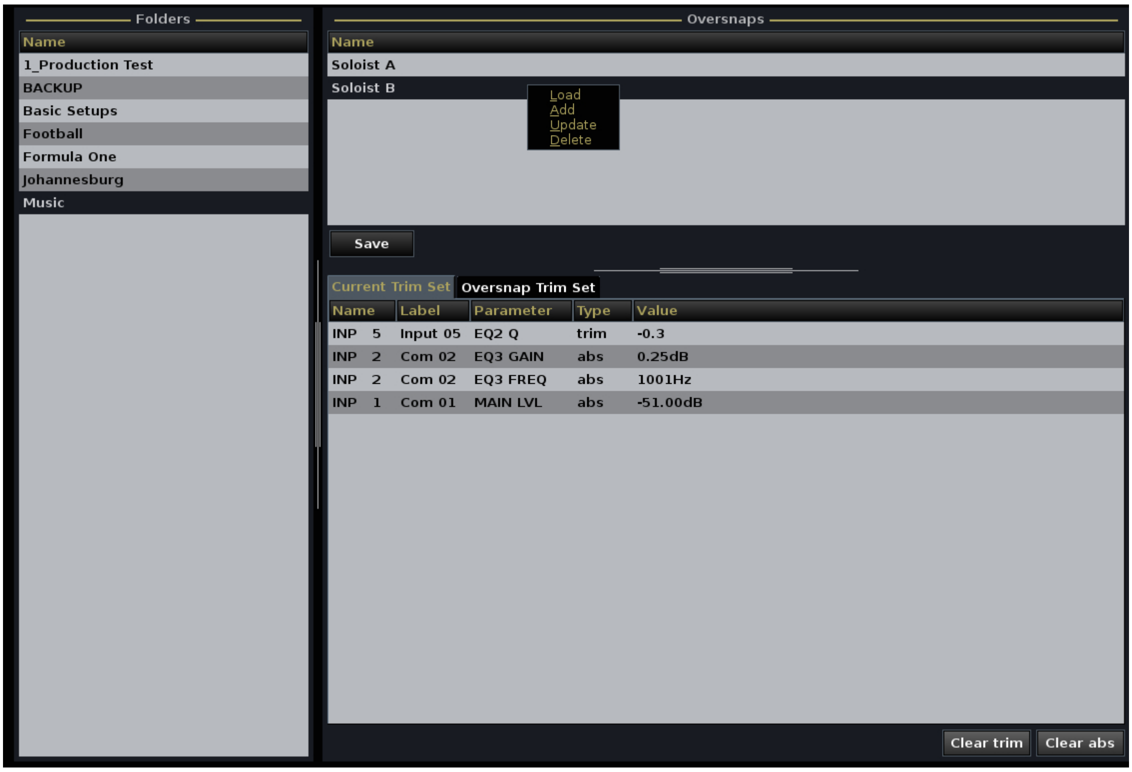
3. Then right-click on the oversnap and select **update**.

*The oversnap is overwritten with the contents of the **Current Trim Set**:*



Deleting an Oversnap

1. Right-click on the oversnap, and select **Delete**:



The oversnap is deleted from the production folder.

Importing and Exporting Oversnaps

Oversnaps are stored separately from snapshots within the production folder. While oversnaps cannot be exported individually, you can export oversnaps as part of the complete folder.

1. Copy the folder to a USB interface or network drive using the File Export function from the **File** display.
2. Connect your USB interface or network drive to the destination console.
3. And import the folder into the current production using File Import from the **File** display.

If you now go to the **Snapshot Trim Sets** display and select the imported Folder, you can access the oversnaps.

Presets

Presets provide a way of saving and loading settings for individual modules – EQ, Gate, Compressor, Panning, etc. – or for a complete channel. For example, you may wish to save your favourite Kick Drum EQ, or the complete settings for an announcer channel.

Presets are stored independently of the production, and therefore, you can load back a preset to any channel within any production. They can also be transferred between consoles, allowing you to recall processing prepared on say a mc²56 to a mc²66 or mc²90.

Note that it is possible to load a preset saved on a Broadcast channel to a Recording channel, or vice versa. If you do so, all matching parameter values are recalled. However, this may exclude other important parameters, and the result may not sound the same. For example, if you attempt to load a 3rd order filter setting from a Recording channel preset to a Broadcast channel, then a 2nd order filter (the maximum) is applied.

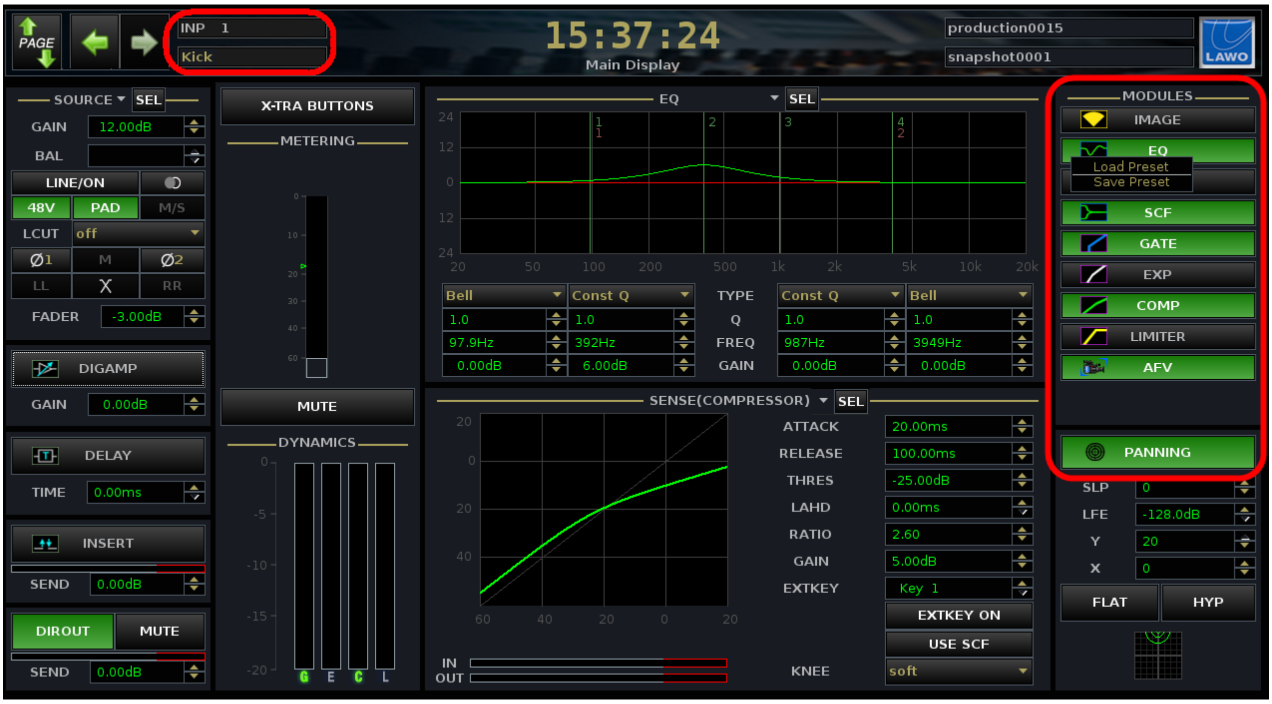
Two different types of preset can be stored:

- **Module presets** – these store settings for individual processing modules: Image, EQ, Filters, Sidechain Filters, Gate, Expander, Compressor, Limiter, AFV settings, Panning and AMBIT. (Note that module presets *cannot* be stored for the input mixer, digamp, delay, insert, direct out or fader level. AMBIT module presets may only be saved and loaded to/from surround VCAs.)
- **Channel presets** – store settings for the complete channel. This includes all the processing modules listed above plus the input mixer, digamp, delay, insert, direct out and fader level. The only settings *NOT* stored by a channel preset are bus routing assignments.

Both types of preset are saved and loaded from the **Main** display:

1. Press the **CHANNEL** button, located on the SCREEN CONTROL panel, to open the **Main Display**:

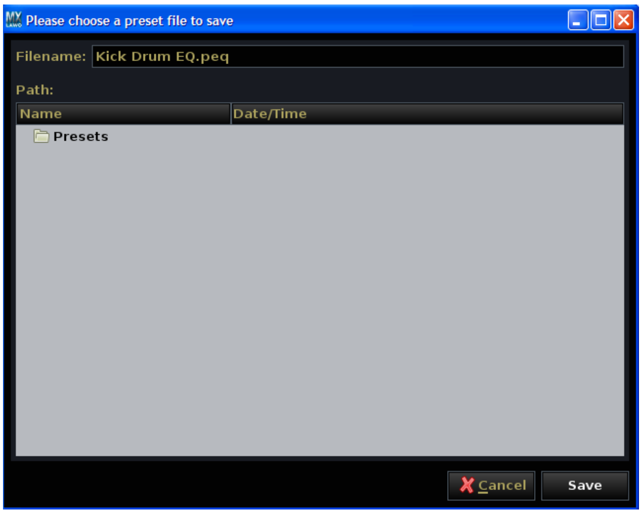
Presets are saved and loaded from the module on/off buttons on the right of the display. You are always saving from and loading to the channel in access – in our example, **INP 1**:



Saving a Module Preset

1. Put the channel you wish to save from into access, either by pressing its fader **SEL** button or using the ACCESS CHANNEL/ASSIGN panel.

2. Using the trackball, right-click on the module you wish to save (e.g. **EQ**) and select **Save Preset** - the 'Preset File' window appears:



3. Select a folder (if one has been created - e.g. **Presets**).

4. Type in a filename (e.g. **Kick Drum EQ**) and select **Save.**

The EQ module settings are saved as a preset into the selected folder on the console's user data flashcard.

5. Repeat these steps to save settings for other modules by right-clicking on the appropriate module on/off button.

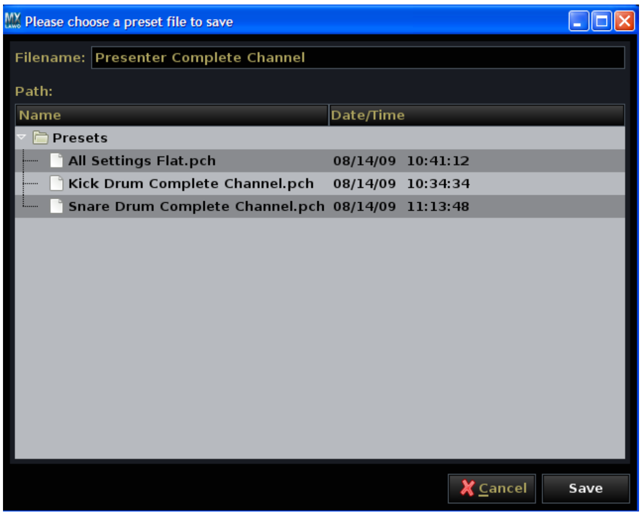
You can save presets for Image, EQ, Filters, Sidechain Filters, Gate, Expander, Compressor, Limiter, AFV settings and Panning modules, but *NOT* for the input mixer, digamp, delay, insert, direct out or fader level.

To save an AMBIT module preset, you must have the surround VCA channel in access.

Presets are stored as different file types to help distinguish EQ presets (**.peq**) from Compressor presets (**.pco**) and so on.

Saving a Channel Preset

1. To save a preset for the complete channel, right-click on the word **MODULES** and select **Save Preset**:



2. Select a folder (if one has been created - e.g. **Presets**).
3. Type in a filename (e.g. **Presenter Complete Channel**) and select **Save.**

The complete channel settings are stored.

A channel preset stores *all* processing modules including the input mixer, digamp, delay, insert, direct out or fader level. The only channel settings *not* stored by a channel preset are bus assignments.

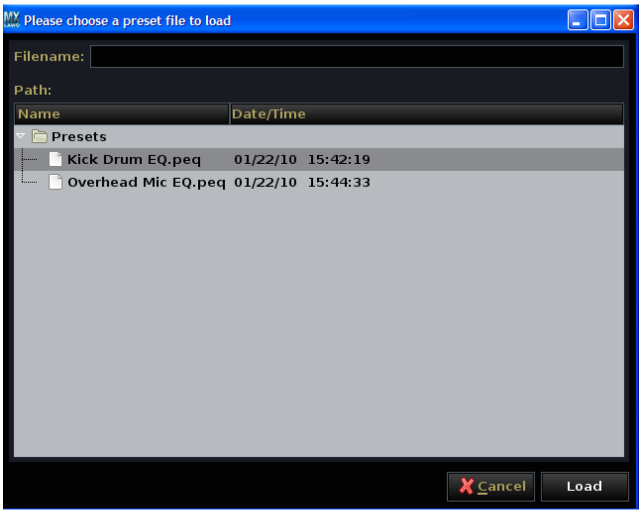
Channel presets have a **.pch** suffix to identify their file type.

Loading a Preset

Presets are stored independently of the production, and can be loaded to *any* channel within *any* production.

1. Put the channel you wish to load to into access, either by pressing its fader **SEL** button or using the ACCESS CHANNEL/ASSIGN panel.

2. Using the trackball, right-click on the module you wish to load (e.g. **EQ**), or right-click on the word **MODULES** to load a channel preset, and select **Load Preset** - the 'Preset File' window opens:



3. Click on the arrows to open up your folder (if one has been created), select a preset and then select **Load**.

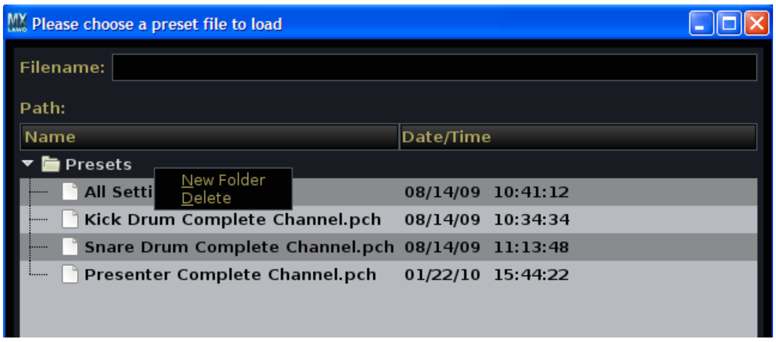
Note that you will only see presets applicable to the selected module - in our example, **.peq** files. This avoids you accidentally loading say a compressor preset to an EQ module!

The preset is loaded to the EQ module, resetting all parameters including the status of the module on/off button.

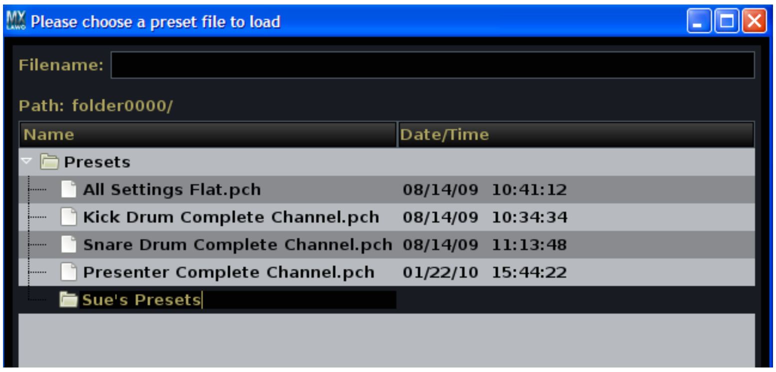
Organising Presets in Folders

Folders (and sub folders) may be created to help organise presets on the system. For example, you may wish to create a separate folder for each user.

1. Open the 'Preset File' window by right-clicking on any audio module and selecting **Load Preset**.
2. Right-click at the top directory level, or on an existing folder, and select **New Folder**:



3. Type in a name for the folder and press Enter to confirm - the new folder is created:



4. Select **Cancel** to exit the 'Preset File' window.

When you next save or load a preset, you will see the new folder.

> You may create sub folders within folders if you wish.
>
> Note that you *cannot* move presets between folders from the 'Preset File' window. If you wish to move the locations of existing presets, then use the File display (on the console) or File Transfer display (on mxGUI) to copy presets to/from folders.

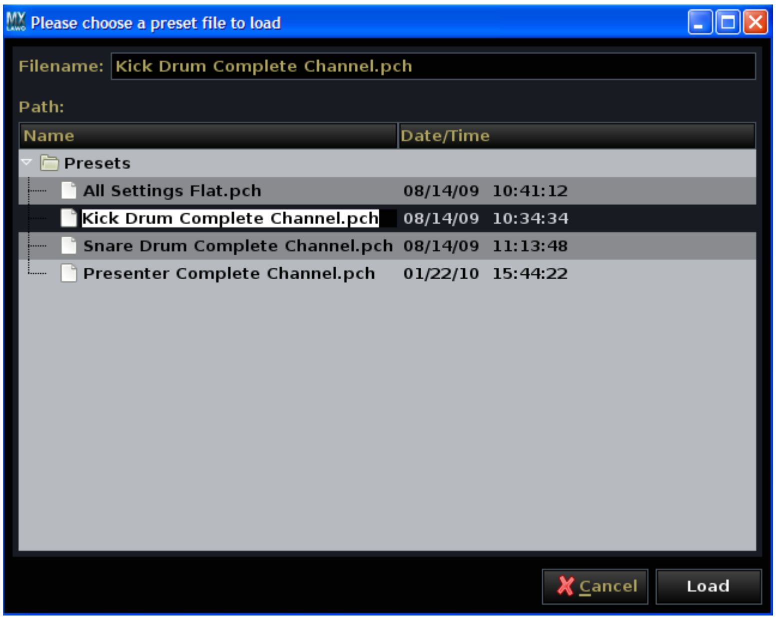
Renaming a Preset

To rename a preset (or preset folder):

1. Open the 'Preset File' window by right-clicking on the appropriate audio module and selecting **Load Preset**.

Note that you will only see presets applicable to the selected module. So, to rename an EQ preset, right-click on the EQ module.

2. Select the preset (or folder) you wish to rename:



Click once to select all the existing text (white) or twice (black cursor) to modify the existing name.

3. Enter a new name from the keyboard.

4. When you have finished, press the Enter button, on the keyboard, to confirm the new name.

5. Or, if you make a mistake or want to exit without making any changes, press the **Esc** button on the keyboard.

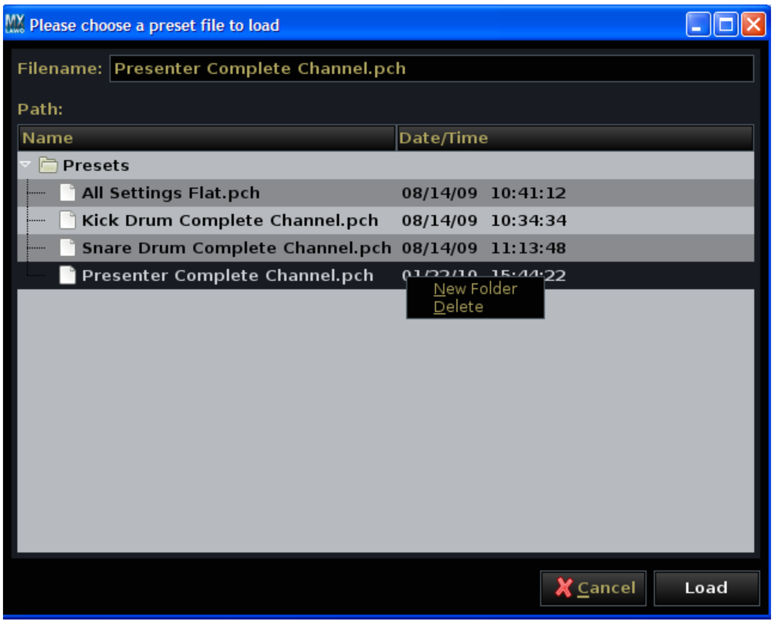
Deleting a Preset

To delete a preset:

1. Open the 'Preset File' window by right-clicking on the appropriate audio module and selecting **Load Preset**.

Note that you will only see presets applicable to the selected module. So, to delete an EQ preset, right-click on the EQ module.

2. Select the preset (or folder) you wish to delete, right-click and select **Delete**:



Note that to delete a folder, you must first delete all presets contained within it.

Importing and Exporting Presets

Individual presets and folders can be imported and exported to a USB interface, mxGUI computer or network drive. This allows you to archive or transfer presets between systems. See File Import/ Export for details.

File Import/Export

The console's file import/export functions can be used for a number of applications:

- To archive or transfer user data between systems.
- To archive or transfer system logfiles for servicing purposes.
- To copy elements within the console's user data flashcard. For example, to copy a snapshot to a different production folder.

User data includes complete productions or elements of a production (such as a folder, snapshot or automation mix) and presets.

From the console, the File display is used to transfer this data to/from a USB interface or network drive.

(Note that you may also transfer data to an external computer using mxGUI, via the File Transfer display.)

User data is fully compatible with any mc² or Nova73, regardless of the hardware configuration. This enables the transfer of production data, snapshots, mixes or presets to and from any system (including any other mc²), in order to recall settings in a different studio.

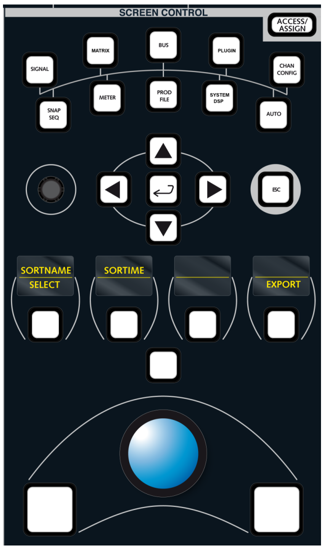
You may need to take care when moving productions to a system with fewer DSP boards, and be aware that the channel DSP settings saved in snapshots from Recording channels cannot be loaded to Broadcast channels. See Transferring User Data for more details.

The File Display

The **File** display transfers user data to/from a USB interface or network drive.

> Lawo cannot guarantee compatibility with all available USB interfaces. Therefore, please check the compatibility of your USB interface on your system.

1. Press the **PROD FILE** button, located on the SCREEN CONTROL panel, to view this display:



The display is divided into two halves:

- **Production Files** - on the left you are always viewing files or folders on the console's internal data card.
- **File Browser** - on the right you can view files or folders on one of the following storage devices:
  - o **Internal** - the internal data card.
  - o **USB** - a mounted USB device.
  - o **Net** - a network drive (pre-configured within AdminHD).

At the bottom of the display you will the amount of free space (in MB) on your selected device.

For each file, you can see its name, type, the date and time when the file was last updated and the file size in Kb.

Open or close folders by double-clicking on the folder name (or click on the arrow beside the name).

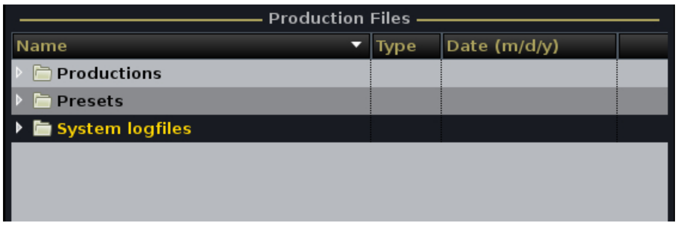
Right-click on a file or folder and select:

- **EXPORT** - to transfer from left to right (internal to internal, USB or network drive).
- **IMPORT** - to transfer from right to left (internal, USB or network drive to the internal data card).

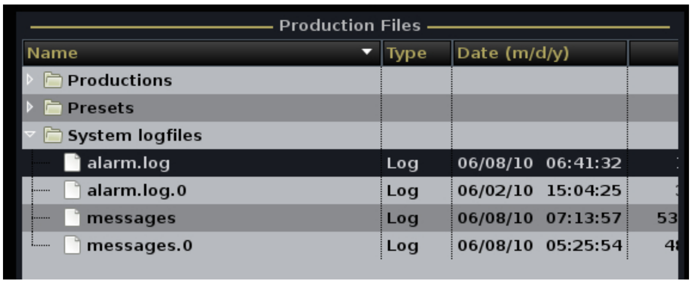
The Production Files List

1. Double-click, or use the arrows, to close the folders in the **Production files** list until you reach the top level of the internal data card.

*You should see three folders – **Productions**, **Presets** and **System logfiles**:*



You can open the **System logfiles** if you need to access message files or the alarm logfile - these are diagnostics files which you may need to copy to USB and email to your service engineer should you encounter a system problem:

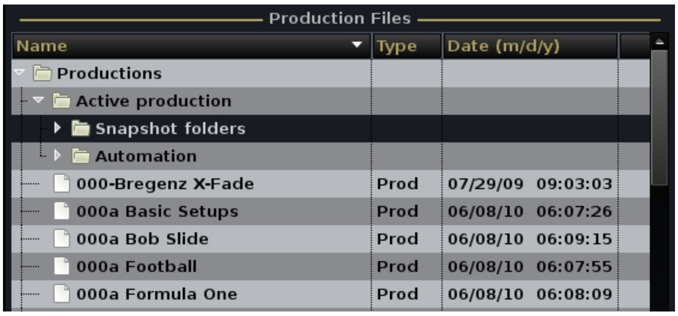


More commonly, you will be dealing with the **Productions** folder in order to copy or export a production, folder or snapshot.

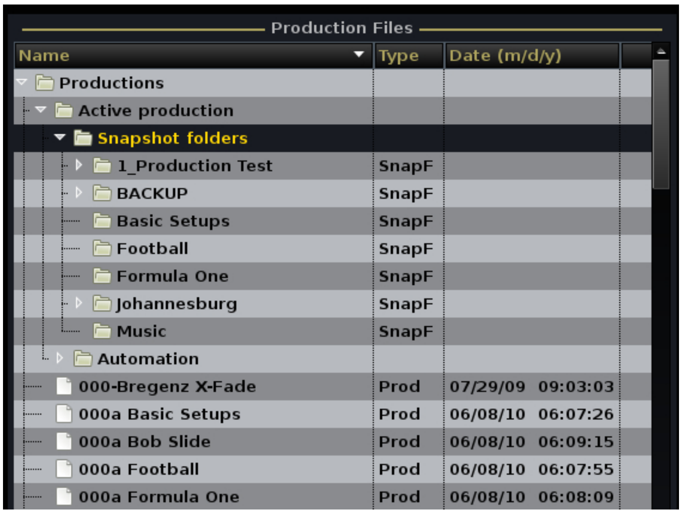
2. Open **Productions** and the display will update to show all the productions stored on your system.

At this level, the productions you see are zipped. They can only be selected as a complete file, and cannot be opened to view or individual elements. The only entry which can be opened further is the **Active production** as this is not zipped.

3. Open the **Active production** to reveal two further directories: **Snapshot folders** and **Automation**:



4. Open **Snapshot folders** to access any Folders stored within the **Active production**:



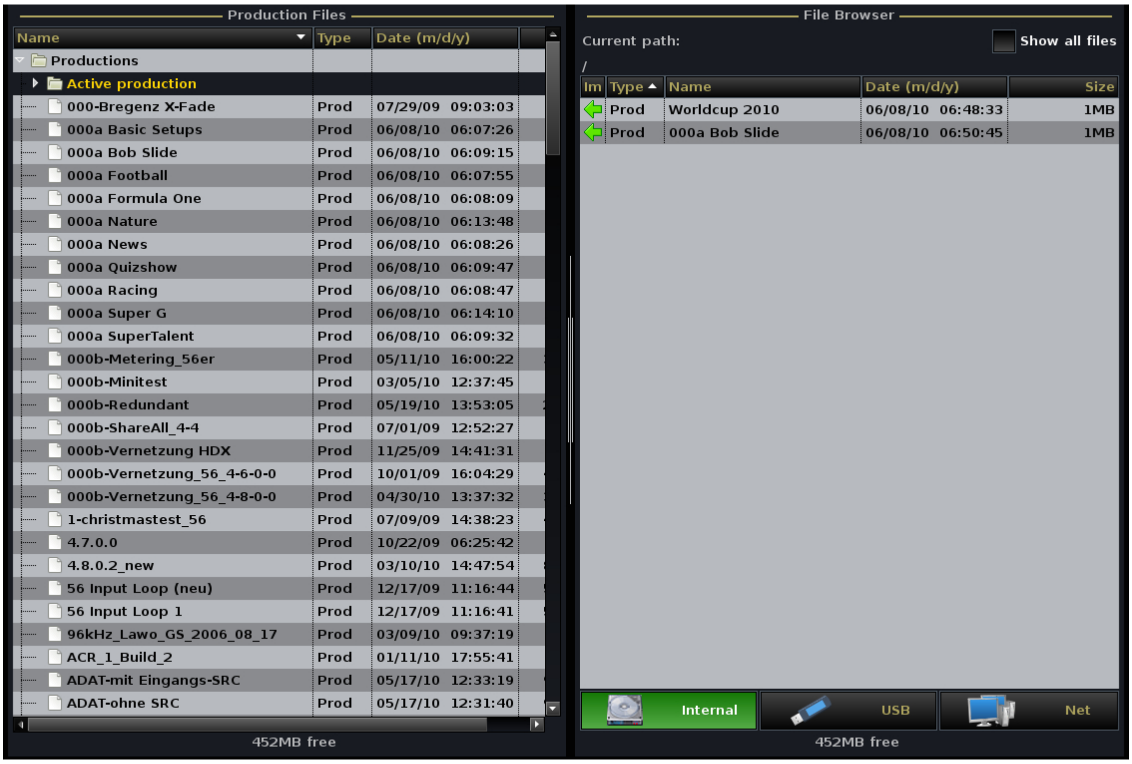
5. And open a Folder to access the individual snapshot files.

6. Remember that at any time, you can go back one level by closing the folder – double-click on the folder name, or click on the arrow beside the name.

The File Browser

Selecting and navigating within the **File Browser** varies slightly from the **Productions list**.

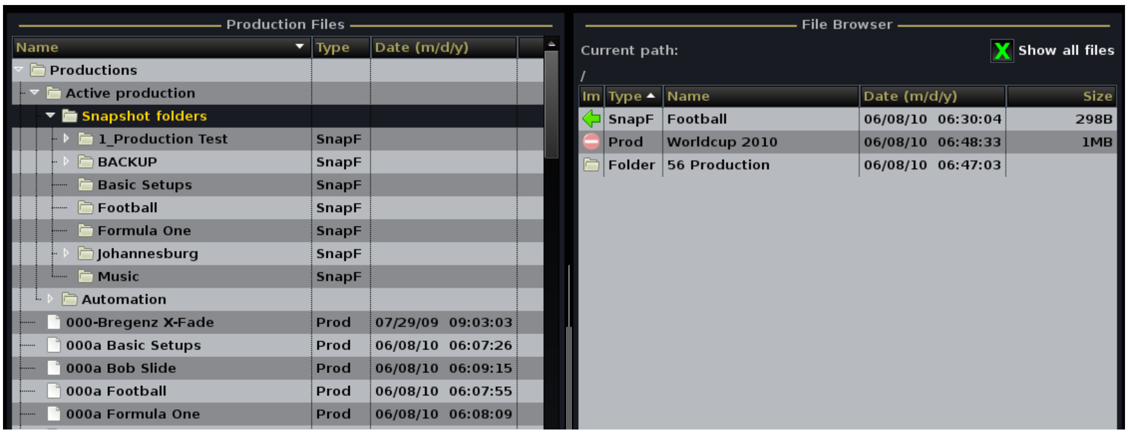
This is because the files you see within the **File Browser** are dependent on the directory level of the **Productions list**. For example, if you are viewing zipped productions within the **Productions list**, then you will only see zipped productions in the **File Browser**. This prevents you from copying files to ‘illegal’ locations:



1. To see all files from the **File Browser** regardless of their compatibility, select the **show all files** option.

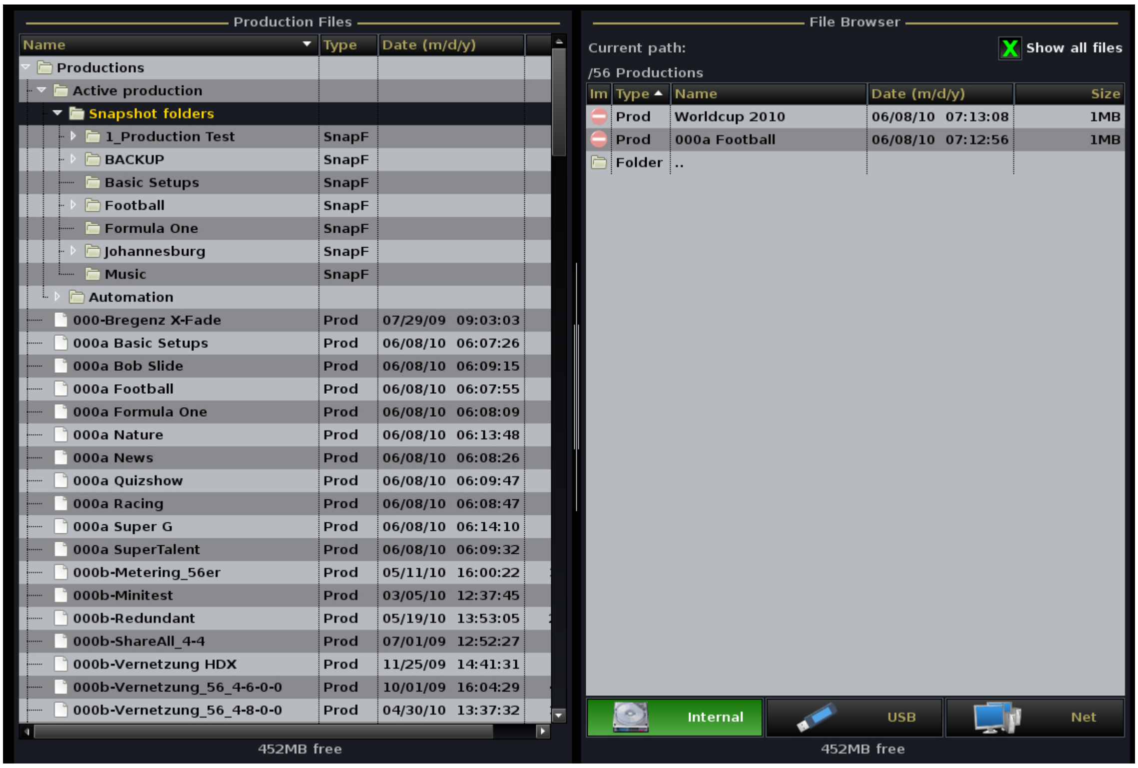
*The **File Browser** updates to list all files on the selected device; the **Type** column shows whether they are compatible for import.*

In our example, the Snapshot folder called **Football** is compatible for import:



2. If your selected storage device contains folders, then you can open a folder by double-clicking on the folder name.

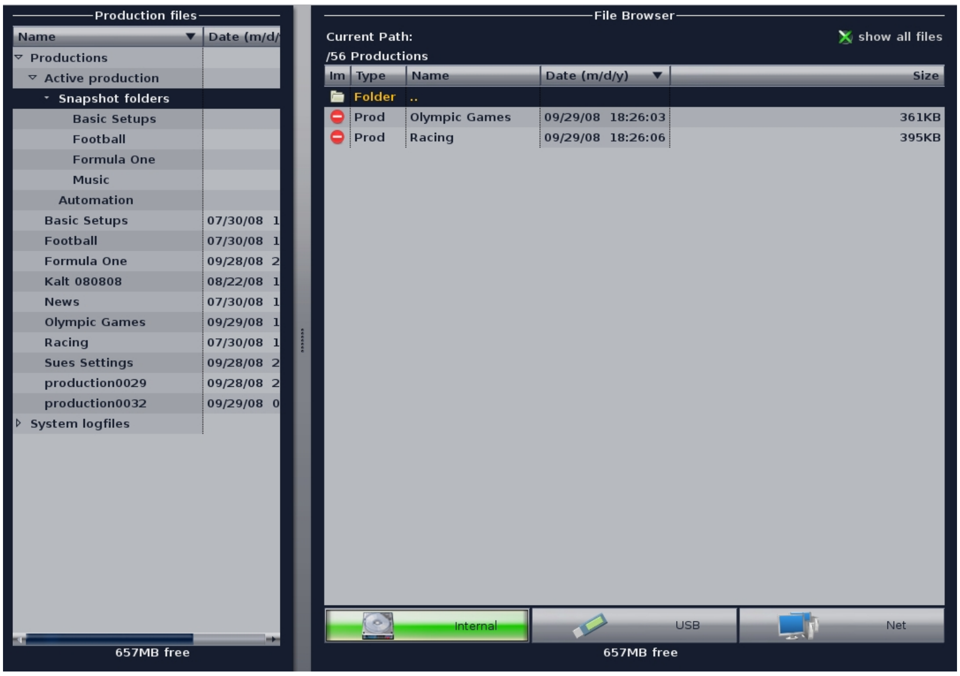
The file path is shown at the top of the ***File Browser*** *– in our example,* ***/56 productions****:*

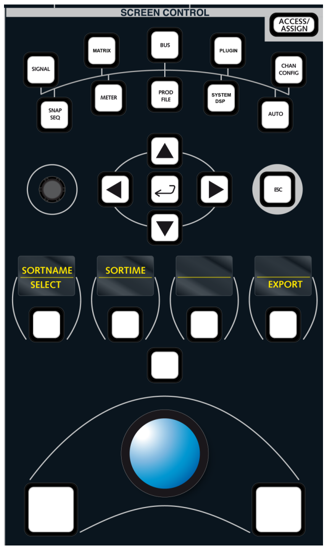


3. To close the folder and move back to the top level directory, double-click on **Folder…**

Resizing and Sorting

1. You can resize the **Production files** and **File Browser** areas by clicking and dragging on the grey separator bar:





2. You can sort files by name, date, size, type, etc. by clicking on the column headers.

Or press the **SORT NAME** or **SORT TIME** soft keys.

The **SELECT** soft key provides another method for opening or closing a folder.

File Types & Extensions

The **File** display can be used to export any the following files from the internal data card:

- The **Active Production**. The active production can be exported in full, or opened in order to select individual elements such as a folder, snapshot or automation mix.
- **Prod** – zipped production files. These are zipped files which cannot be opened. They can be exported as a file to the external storage device, imported on another console, and then unzipped within that console to access their individual elements.
- **SnapF** – an individual Folder within the Active production. By selecting a Folder, you can easily export all the snapshots for a particular show.
- **Snap** – an individual Snapshot within the Active production.
- **Mix** – an individual Automation Mix within the Active production.
- **Presets** – an individual channel or processing module preset.
- **Log** – a message file (system log file).

Note that you can also export these files to an external computer running mxGUI, see the File Transfer display.

Warning

You can view and rename user data files on an external computer. However, if you edit the contents, or modify the file extension, you may corrupt the file and lose data!

The following extensions must be intact to permit a valid file import:

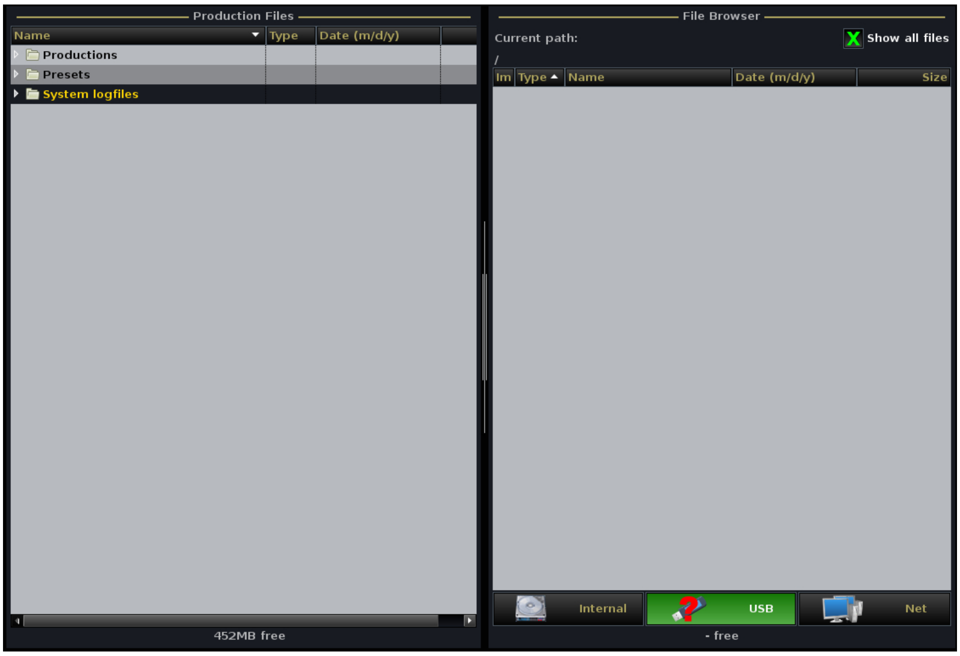
- **.lsn** – a snapshot
- **.lpn** – production
- **.lau** – automation mix
- **.pch** – channel preset
- **.peq**, etc. – EQ preset, Compressor preset, etc.
- **.lcf** – complete configuration (mxGUI only)
- **.lco** – core configuration: config.tcl (mxGUI only)
- **.lsl** – signal list: gui_config.tcl (mxGUI only)

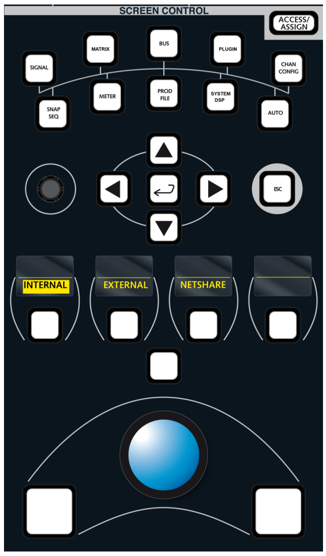
Exporting to USB

1. First, connect your USB interface to one of the console's USB ports.

2. Click on the **USB** interface icon at the bottom right of the display, or press the **EXTERNAL** soft key:

- If the USB interface is mounted, then its icon will turn green and the File Browser will show any files or folders already stored on the device.
- If the USB interface is not mounted, then you will see the following:





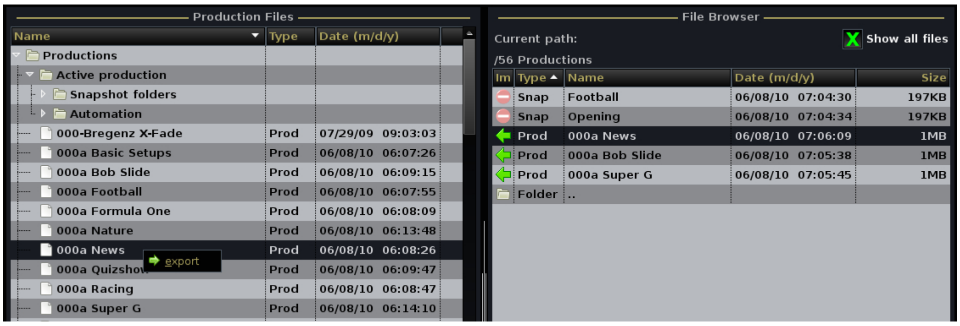
3. Try refreshing the **USB** selection - select **Internal** and then back to **USB.**

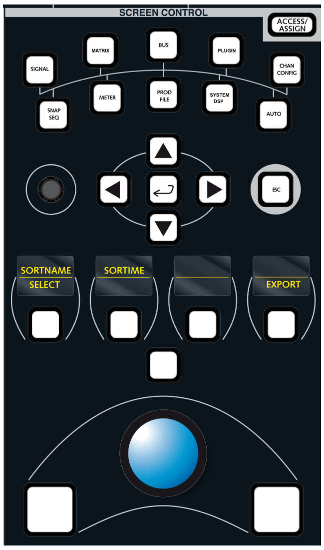
The device should now show as ready. If not, check your connection or try a different USB device.

4. Select the destination folder on your USB from the File Browser. (You can create folders to help organise files.)

5. Then right-click on the file you wish to export from the Production files list and select **Export**, or press the **EXPORT** soft key.

*The file is copied from the internal data card onto your USB storage device - in our example, we have exported the production named **000a News**:*





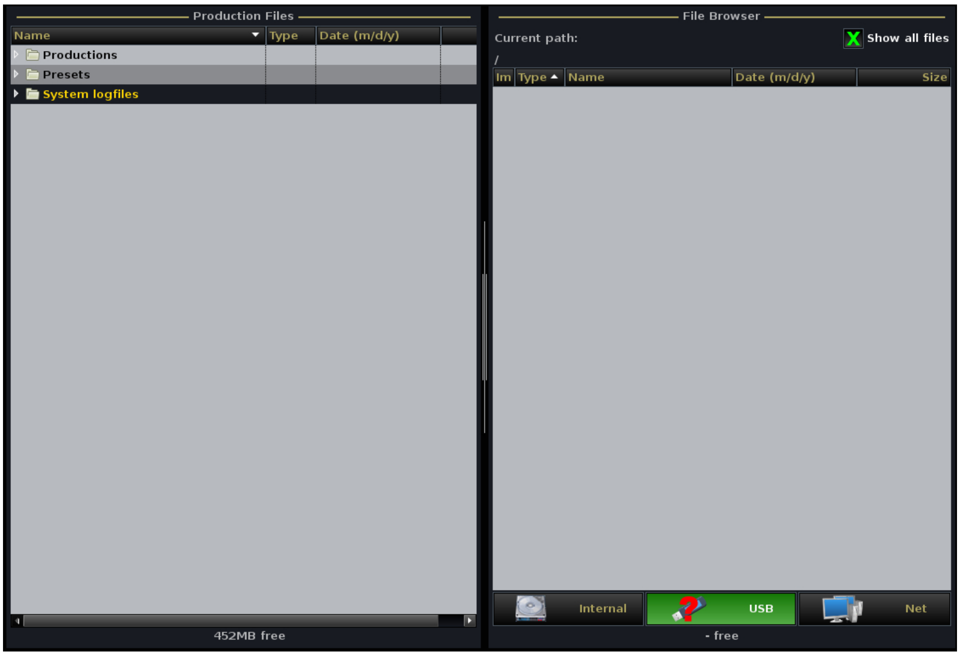
6. Once the data has finished transferring, you can unplug the USB device.

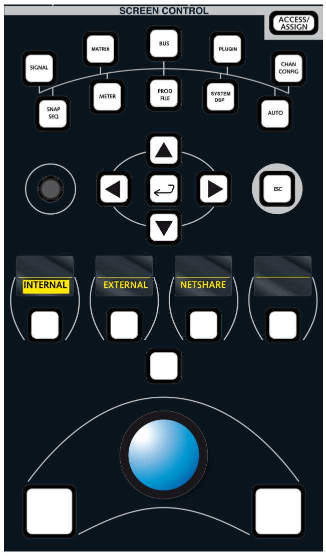
Note that there is no need to dismount the USB interface before you remove it. However, DO NOT unplug the USB interface while data is transferring as this may result in loss of data.

Exporting to a Network Drive

Follow the same steps to export a file or directory to a network drive. Note that the drive must be configured by your system administrator using the AdminHD configuration software for it be available.

1. Once configured, you can mount the drive by selecting the **Net** icon or pressing the **NETSHARE** soft key:





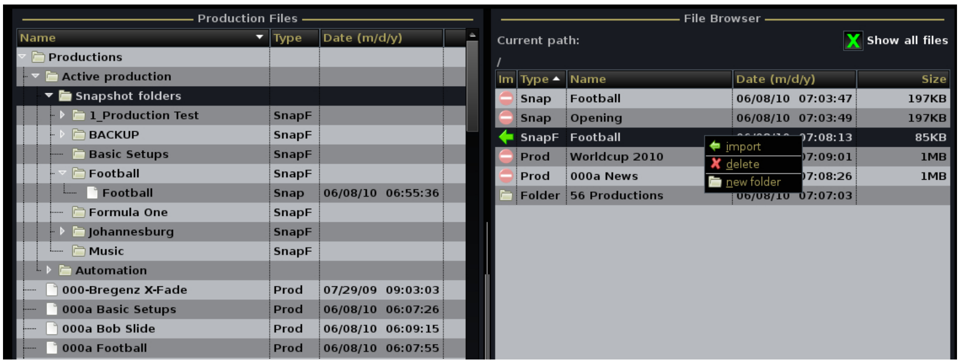
> Note that the contents of the **File Browser** will not automatically update if changes are made from another console or computer. So, to see any changes, refresh the **Net** selection - select **Internal** and then back to **Net.**

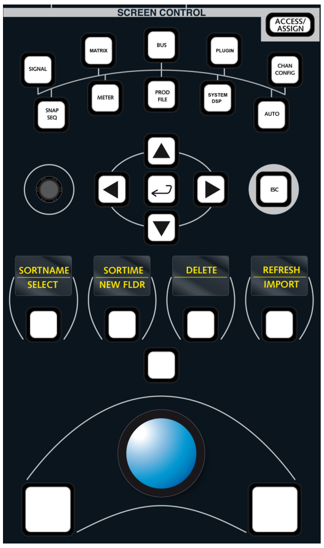
Importing a File

You can import files from a USB interface or networked drive in a very similar manner to File Export, but this time:

1. Select the file you wish to import from the File Browser.
2. Select the correct destination level from the Production Files list.
3. Then right-click on the File Browser file and select **Import**, or press the **IMPORT** soft key.

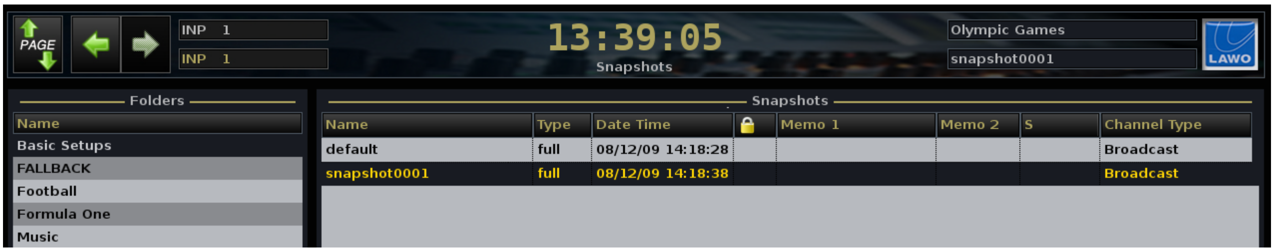
In our example, we have imported the snapshot folder called **Football** into the **Snapshot folders** of the Active Production:





Note that if a file or folder of the same name already exists, then the file will be copied with an appended name – for example, **Football (0001)**.

4. If you now select the **Snapshots List** display, you will find the imported Folder in the Folders list on the left of the display:



Copying Files Internally within the System

The **File** display may also be used to copy files internally within the system. For example, if you wish to copy a snapshot or folder from one production to another.

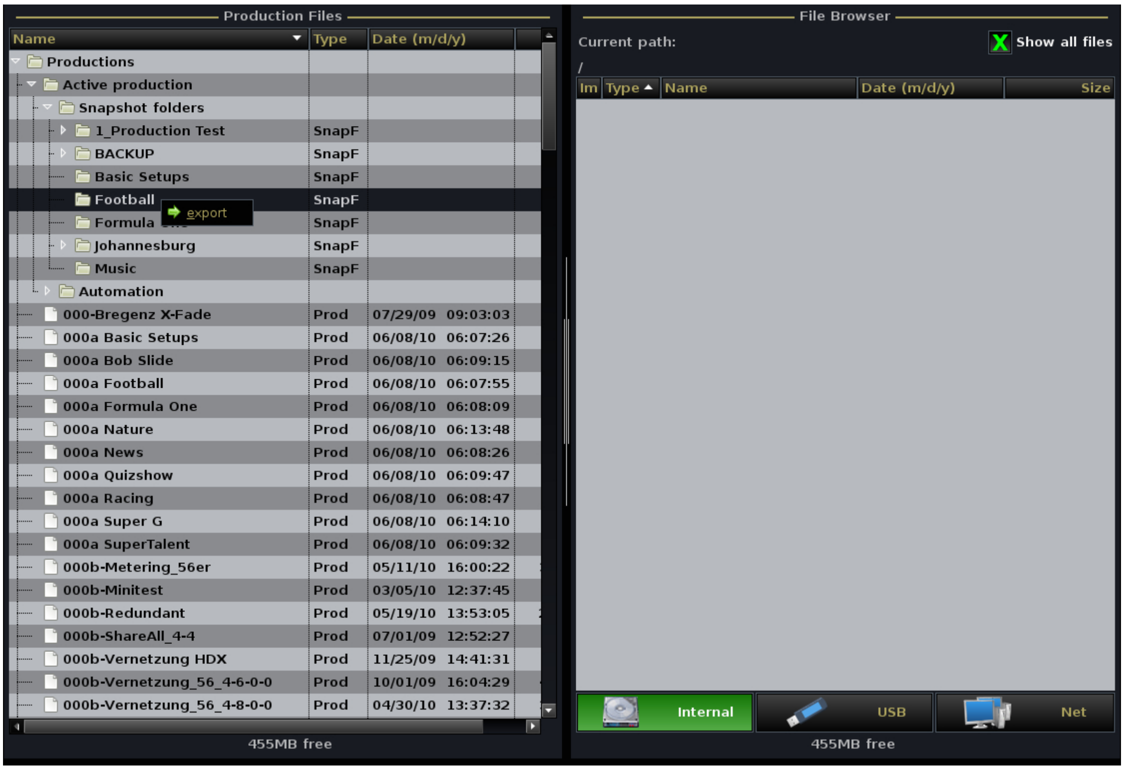
As you cannot open up a zipped production to select individual elements, you will need to perform this operation in several stages:

1. First, load the production which contains the snapshot you wish to copy from the **Productions** display.

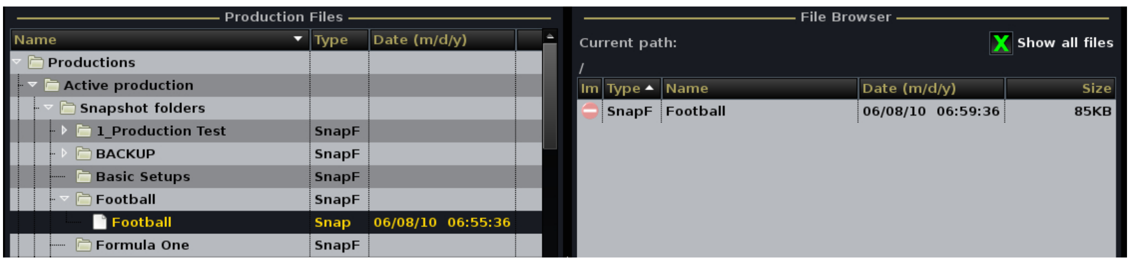
2. Now, go to the **File** display and within the File Browser, click the **Internal** drive icon.

You will see the contents of a temporary directory within the internal drive as your destination. This directory may be empty, or it may contain previously copied files.

3. Now, from the Production files list, open up the **Active production**, the **Snapshot folders** directory and select the folder to copy – in our example, **Football**:



4. Select **EXPORT** to export the folder to the temporary directory:

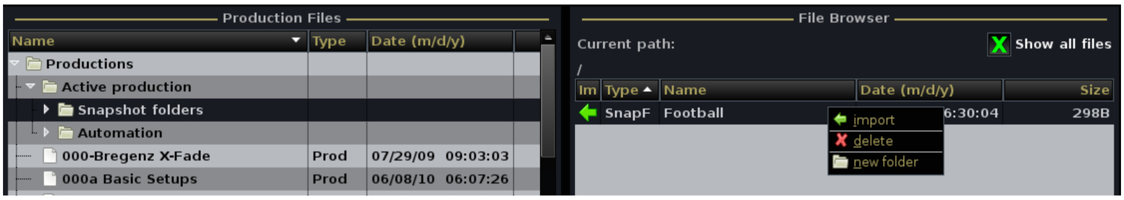


5. Next, return to the **Productions** display and load the production you wish to copy into - this now becomes the Active Production.

6. Select the **File** display.

7. From the **Production files** list open up the **Active production** and select the **Snapshot folders** directory.

8. And, on the right hand side, select the folder you copied earlier – **Football**:



9. Select **IMPORT** to import the snapshot to the Active Production Folder.

10. If you now select the **Snapshots List** display, you will find the imported Folder.

If you are using this operation to copy a lot of files, then it is a good idea to delete files from the temporary directory.

Remember the File Browser may only show files which can be imported to your selected destination. For example, if you have selected a snapshot folder, you will only see snapshots; if you have selected the **Productions** directory, then you will see productions. To see all files, turn on the **show all files** option.

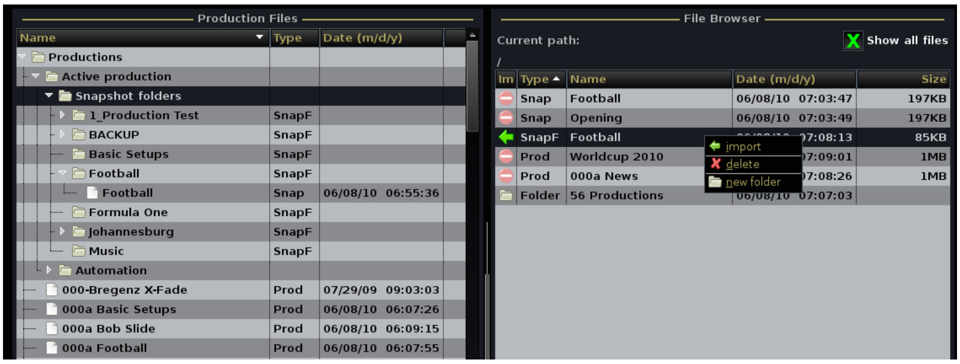
File Management

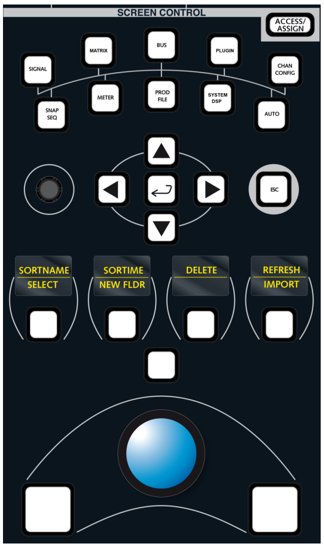
To help manage your data on your external USB interface or networked drive, the **File** display enables you to create a new folder or delete a file or folder in the File Browser (on the right of the display).

These functions are designed to give you the basic tools to manage your exported data. However, to reorganise the data structure on your storage device, connect it to your PC!

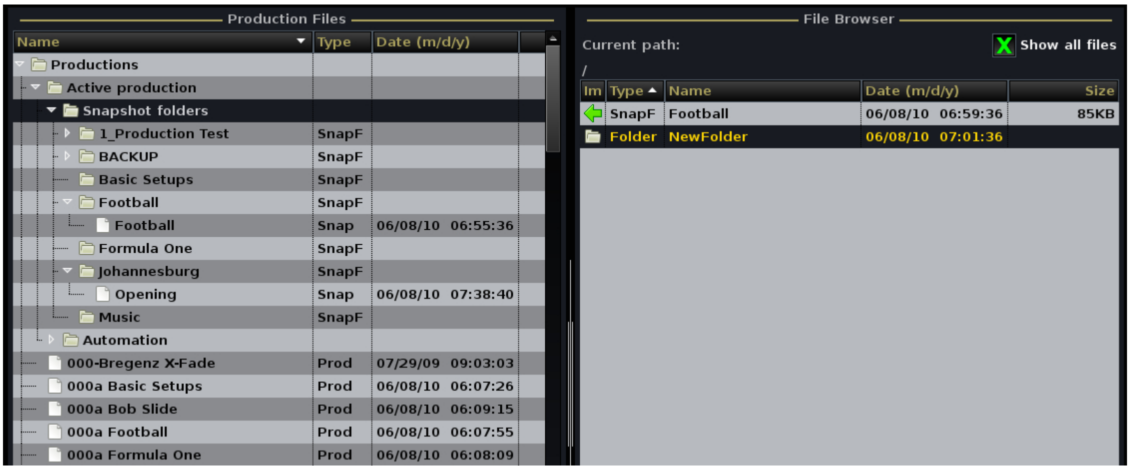
➢ To Create a New Folder

1. Select **Internal**, **USB** or **Net** to select your interface.
2. Right-click on the **File Browser** and select **New Folder**, or press the **NEW FLDER** soft key.





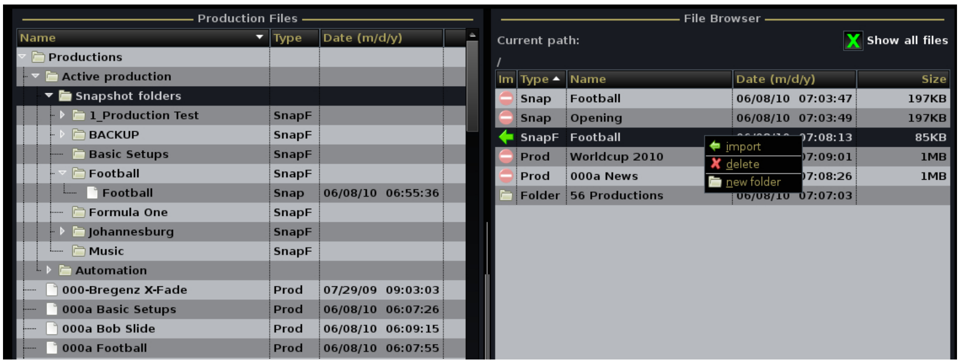
A new folder is created with a default name:

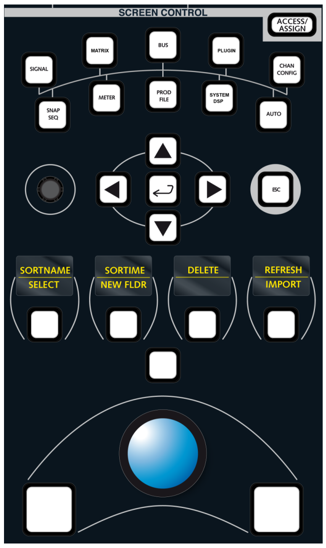


3. You can rename the folder by clicking on the folder name and using the console keyboard.

➢ To Delete Files or Folders

1. Select **Internal**, **USB** or **Net** to select your interface.
2. Right-click on the file or folder from the **File Browser** and select **delete**, or press the **DELETE** soft key.





You can delete files or folders within the temporary folder on the internal drive, or on your external USB interface or network drive.

Note that you cannot delete files or folders from the **Production files** list. (To perform these data management functions, go to the Snapshots, Mixes or Productions displays.)

Transferring User Data

All user data is fully compatible with any mc² system. However, you should note the following if you are transferring data between systems:

Productions

The DSP configuration is saved and loaded as part of the production. Normally, you only need to save or update the production to ensure all settings are recalled when you later load the production back.

However, if you move a production to a console with fewer DSP boards, then the DSP configuration will not load (as it is looking for more physical cards). And, if the current channel type is not compatible with the production snapshot, your DSP settings will not load either.

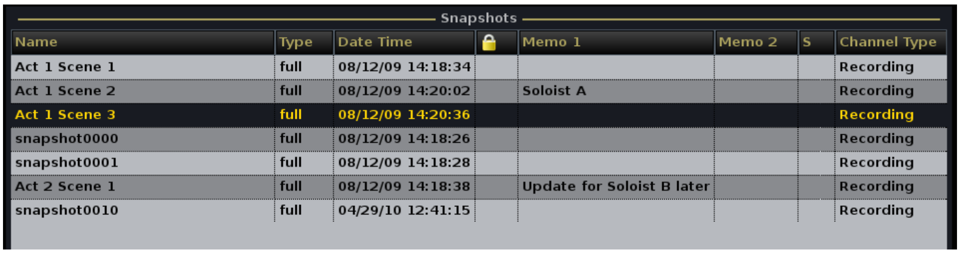
To overcome this, save a snapshot on the original console in addition to saving the production. When you move the production to the new console, load the production, then manually load a DSP configuration with a compatible channel type. Now load the snapshot. Your settings will be recalled to all available DSP channels.

Snapshots

Snapshots do *NOT* store the DSP configuration (to avoid an interruption to audio from a snapshot load). And, you cannot load a Broadcast snapshot to a console running Recording channels, or vice versa. To help manage this, the **Snapshots List** includes a channel type column:

1. Press the **SNAP/SEQUENCE** button, on the SCREEN CONTROL panel, to view the **Snapshots List** display.

2. Scroll to the right to view the **Channel type** column. This shows which channel type was active when the snapshot was saved:


Snapshots

Name	Type	Date Time	🔒	Memo 1	Memo 2	S	Channel Type
Act 1 Scene 1	full	08/12/09 14:18:34					Recording
Act 1 Scene 2	full	08/12/09 14:20:02		Soloist A			Recording
Act 1 Scene 3	full	08/12/09 14:20:36					Recording
snapshot0000	full	08/12/09 14:18:26					Recording
snapshot0001	full	08/12/09 14:18:28					Recording
Act 2 Scene 1	full	08/12/09 14:18:38		Update for Soloist B later			Recording
snapshot0010	full	04/29/10 12:41:15					Recording

To keep things simple, always choose a DSP configuration which matches the snapshots for the production. When this is the case, settings load as normal, and you can load snapshots from one console to another, even if the number of DSP boards or DSP configuration varies. For example, if a snapshot saved with settings for 192 Broadcast input channels, is loaded onto a console running 96 Broadcast input channels, then settings are recalled to the matching available input channels (1 to 96).

If you do try and load a Recording snapshot to a console running Broadcast channels, then the following will happen:

- Incompatible channel DSP settings (e.g. EQ, Dynamics, Fader levels, etc.) cannot be loaded. The only exception to this is bus assignments.
- All other parameters – signal routing, I/O settings, desk configuration, etc. – are loaded as normal.

If you really need to transport a snapshot from one channel type to another, then this can be achieved using the dynamic automation (see Mixes).

Presets

Unlike snapshots, it is possible to load a preset saved on a Broadcast channel to a Recording channel, or vice versa. If you do so, all matching parameter values are recalled. However, this may exclude other important parameters, and the result may not sound the same.

For example, if you attempt to load a 3rd order filter setting from a Recording channel preset to a Broadcast channel, then a 2nd order filter (the maximum) is applied.

Mixes

You can also load a mix created with Broadcast channels, to a DSP configuration running Recording channels, or vice versa.

Any matching parameter values, such as fader levels, are recalled. However, when it comes to signal processing modules, the recall may exclude other important parameters, and the result may not sound the same.

This is a way of transporting snapshots from one channel type to another:

- Enable dynamic automation and recall the snapshot you wish to transfer.
- Turn off the automation.
- Then change the DSP configuration channel type (all channel DSP settings reset.)
- Turn on the automation to recall the compatible parameter values.
- Now save a new snapshot which matches the DSP configuration channel type.

Chapter 7: Timecode Automation

Introduction

This chapter explains the operation of the timecode automation system, including remote machine control and locators (cue points).

Your system must be specified with the Recording Com Kit (958/80) to provide Sony 9pin, LTC and/or MIDI connections to an external playback device. Please consult your system specification for details.

Topics covered in this chapter are:

Automation: Overview

The **mc²56** automation system automates console settings referenced to timecode, and is controlled from virtual automation panels (**VAP1** and **VAP2**) on the right of the Central GUI touch-screen:

Any channel type may be automated (inputs, groups, sums, auxes, VCA masters, surround VCA masters and GPCs). And automation may be enabled for any audio module (fader, mute, aux sends, EQ, bus routing, channel signal flow, etc.)

Automation data can be written with timecode rolling forwards, backwards and at any speed, providing fast and efficient mixing. The way in which data is written is governed by a number of modes, allowing you to write dynamic or static automation; step in or step out of write to make updates; trim existing moves; protect channels to prevent overwriting existing moves; and isolate channels to remove them from the automation system completely.

Each stream of automation data is recorded as a 'Pass', and multiple passes are stored within a 'Mix'. The 'Pass Tree' allows you to view the history and A/B between different passes within each mix. You can also edit mix passes in order to delete, copy, shift, insert or paste sections from different passes.

Multiple mixes may be created within each production; mixes are stored permanently on the system when you update or save a production.

Control of the playback machine may be programmed onto user buttons from the Custom Functions display, or handled from the optional Machine Control panel.

You may also use the Machine Locator display to store and recall cue points, and/or switch one of your console displays to a remote desktop in order to view and control a DAW.

Before You Mix

Let's assume that you have created a new production and have a basic setup with levels, panning, EQ, etc. Before enabling the automation system, there are a few basic checks to perform:

1. Select the timecode reference for the automation system using the Timecode/Frame Rate options in the **System Settings** display.

> Control of the playback machine may be programmed onto user buttons from the Custom Functions display, or handled from the optional Machine Control panel.
>
> Use the Machine Locator display to store and recall locators (create a cue list).

2. Change the Central GUI headline to display timecode rather than local time or loudness.

This option can be set using the Time Display option in the **System Settings**, or by clicking in the headline at the top of the title bar:

Press **PLAY** on your machine and check that the timecode follows.

> Choose **Offset Timecode** from the headline options, and set a timecode offset from the **Passes** display, if your mix starts at an odd timecode value.

3. If necessary, set a pre-roll tolerance for your playback machine.

This option is set by the Pre-roll window option in the **Passes** display.

4. Check the **Mixes** display and create a new mix to store automation. Details follow on the next page.

5. Select the channels/modules you want to automate. Details follow shortly.

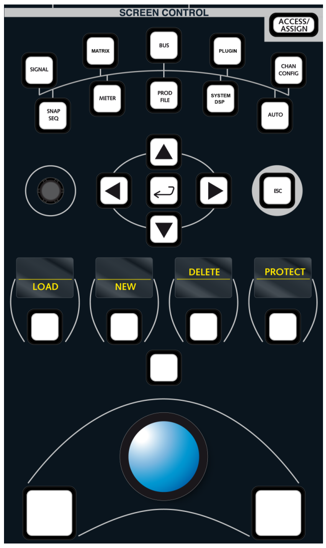
> You may wish to enable Solo-in-Place from the **System Settings** display.
>
> On the mc²56, we recommend programming a **R/W** user button for faders from the Custom Functions display. This will allow you to step in and out of write, and view the status of fader automation, across multiple channels.

Checking the Active Mix

When you enable automation, data from the **Active Mix** is loaded to the console. Therefore, to make sure you don't lose your current settings, you should check the **Mixes** display.

1. Press the **AUTO** button, located on the SCREEN CONTROL panel, to view the **Mixes** display.

*The **Mixes** list shows all the mixes in memory:*



If the list is longer than the available window space, focus on the list and use the rotary scroller on the SCREEN CONTROL panel to navigate up and down the list, and/or use the on-screen scroll bars.

The name of the **Active Mix** is shown at the top of the display. It is the passes from this mix which appear in the Pass Tree, and its Play pass which loads when you enable automation.

The columns beside each mix name show the date and time stamp, the number of passes within the Pass Tree, whether the mix is protected (padlock icon) and the size of the mix. You may drag and drop columns to change their order.

At the bottom of the display, the **Mix Memo** box may be used to add notes to a particular mix.

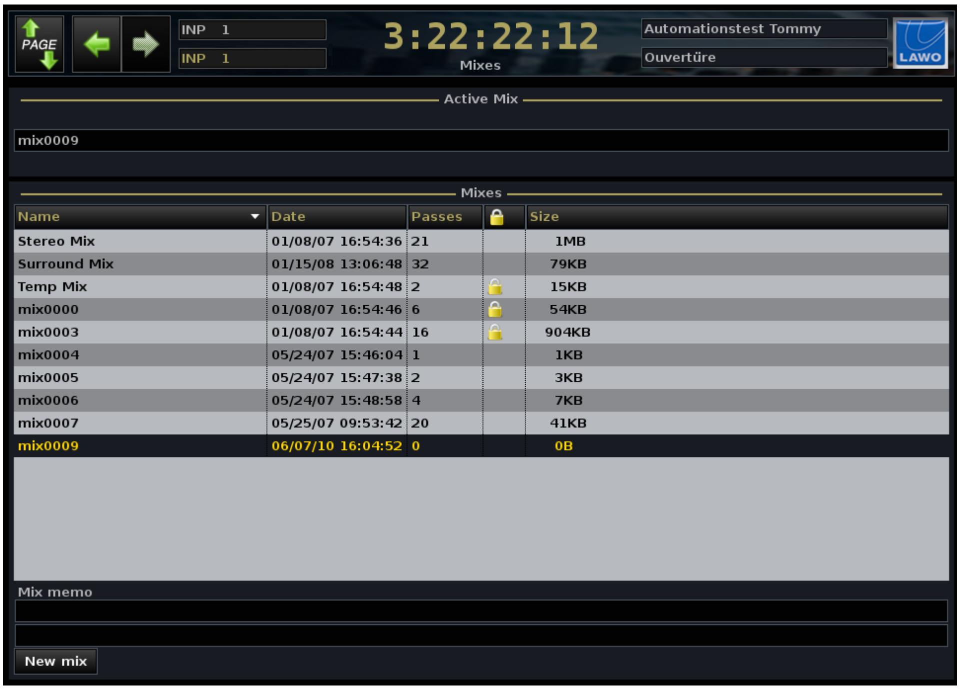
The on-screen buttons and SCREEN CONTROL soft keys provide access to Load, New, Protect and Delete operations. These functions are also available if you select a mix and right-click.

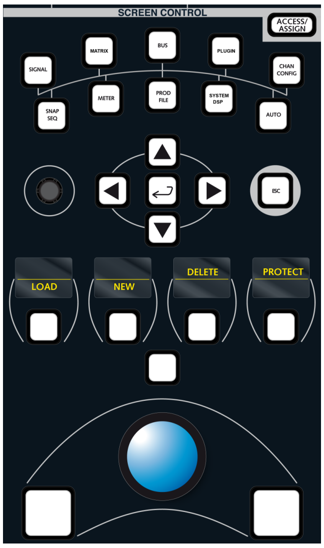
- If the **Active Mix** box is empty, you can skip straight onto selecting the channels and modules you want to automate. The first time you enable the automation, a new mix and Record pass are created automatically.
- However, if an **Active Mix** already exists (loaded from the production), you should create a new mix before proceeding. Otherwise, when you press the **AUTO ON** button, the system will load the Play pass from the **Active Mix**, thereby resetting your existing settings.

Creating a New Mix

1. Press the **NEW** soft key, or on-screen **New mix** button, to create a new mix.

*An empty mix appears at the bottom of the **Mixes** list and automatically becomes the **Active Mix**. It is given a default name (e.g. **mix0009**), and is date and time stamped:*





Selecting Channels/Modules for Automation

The **SEL AUTO** function selects which modules within each channel are enabled (or disabled) for automation.

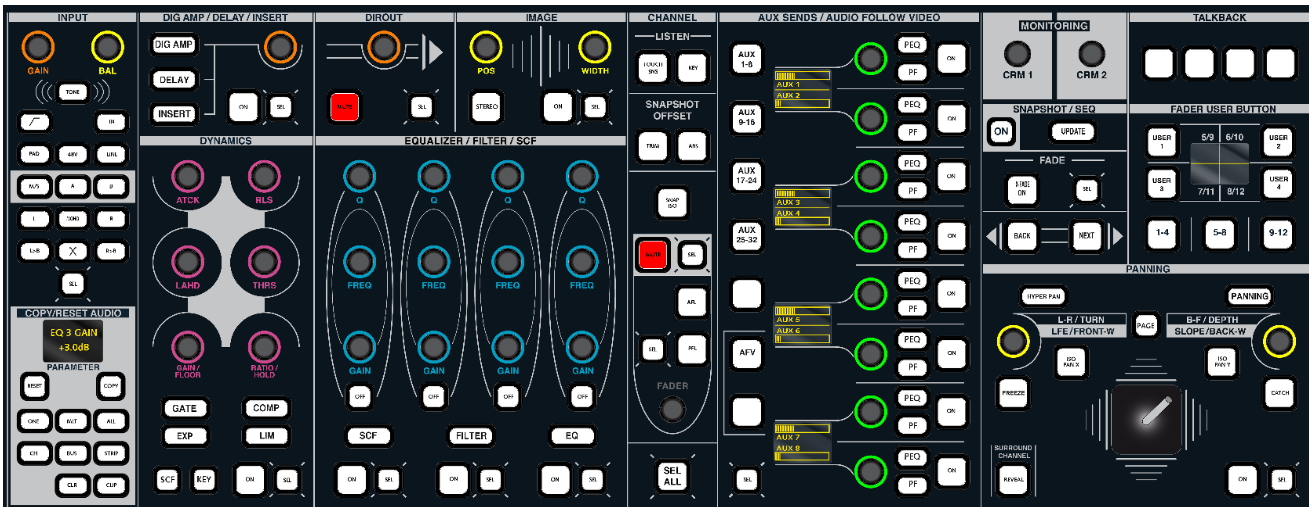
> The first time you enable a module for automation, it defaults to dynamic automation mode and is armed (ready to read and write automation) in absolute.

This operation uses the virtual automation panel (**VAP 1**), available from the touch-screen buttons on the right of the Central GUI.

1. Assign one of the channels you want to automate to the Central Control Section, either by pressing its fader **SEL** button or entering the channel type and number from the ACCESS CHANNEL/ASSIGN control panel.
2. Press the **SEL AUTO** touch-screen button (on VAP 1) - the button flashes to show it is active:

On the Central Control Section, the current status of each module is shown by the select buttons:

- **SEL** button lit (orange) = automation enabled.
- **SEL** button off = automation disabled.



3. Press the Central Control Section **SEL** buttons to enable, or disable automation, for each audio module.

You may select any audio module(s), plus the channel signal processing (**CH**), bus routing (**BUS**), fader strip assignment (**STRIP**) and channel colour coding, see Selecting Channel Parameters for details.

Use **SEL ALL** to enable, or disable, automation for all channel parameters.

For our example, toggle **SEL ALL** to deselect all channel parameters, and then press the **SEL** beside the fader control so that it lights. This enables fader automation on the channel in access.

4. Next press the **MLT** touch-screen button (on VAP 1) to apply this setting across more than one channel:

*The multiple **SEL AUTO** mode is active, and all the fader **SEL** buttons across the console flash, in green.*

5. Press the fader **SEL** buttons on the fader strips you wish to automate.

*The fader **SEL** buttons change from green to red when selected:*

You have now enabled automation for the fader modules on the selected channels.

6. Deselect **SEL AUTO**, or press **ESC** on the SCREEN CONTROL panel, to exit the **SEL AUTO** mode.

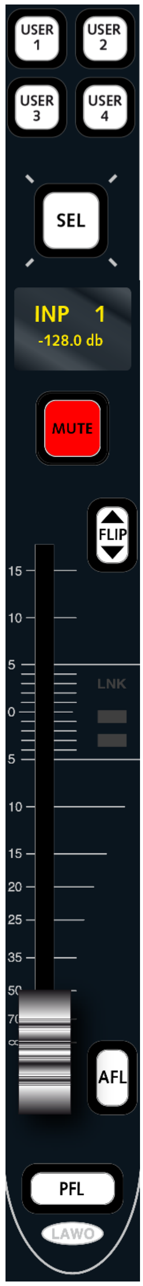
If you wish to select faders on hidden banks or layers, then bring each bank or layer to the surface and press the fader **SEL** buttons during step 5.

To change what is automated at a later date:

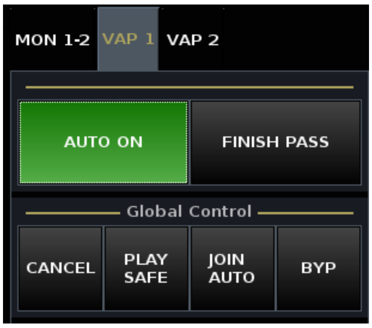
- Repeat steps 1 to 3 to enable, or disable, automation on the channel in access.
- Repeat steps 4 and 5 to apply the new module selections to multiple channels - note that you will need to refresh the fader **SEL** buttons in step 5 (turn them off and back on) to update existing selections.

Writing Automation: First Steps

Having dealt with the preparation steps, you are now ready to turn on the automation and write your first pass.

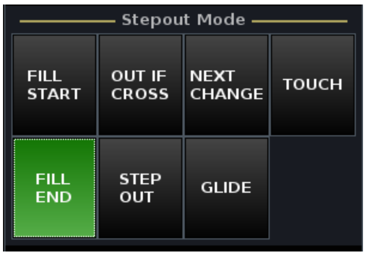


1. Rewind your playback machine to the start of the mix, and turn on automation by enabling the **AUTO ON** button (on VAP 1):



The fader ***R/W*** *user buttons, programmed from the Custom Functions display, turn green. If they don't, then the fader is NOT selected for automation or it may be disarmed.*

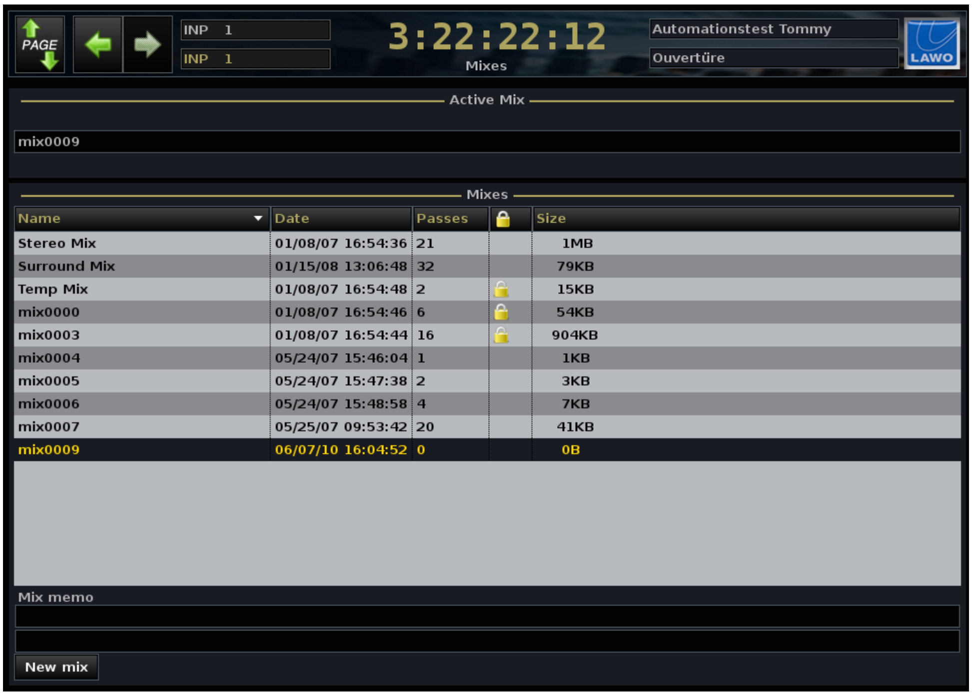
2. Check that **FILL END** is selected as the 'Stepout Mode' (on VAP 1):



The **FILL END** 'Stepout mode' is great for writing early passes where you are working through the song or production chronologically. Each time you stop and finish a pass, any values in write are written through to the end of the mix. This means that you don't have to play through the whole song just to write a fader level to the end of the mix. See Step Out Automation Modes for more details.

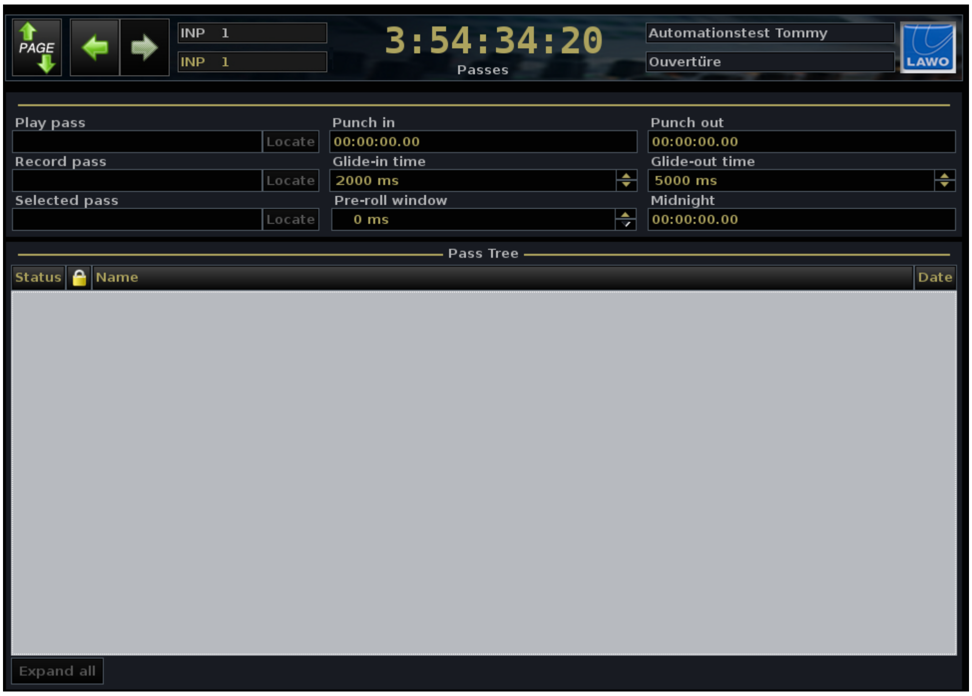
3. Press the **AUTO** button, located on the SCREEN CONTROL panel, to view the **Mixes** display.

*The **Active Mix** is shown at the top of the display - in our example, **mix0009**:*



4. Press the **AUTO** button again to page to the **Passes** display.

The **Pass tree** should be empty as we have not yet written any data:



5. Now press play and write some dynamic fader moves as the timecode rolls forwards.

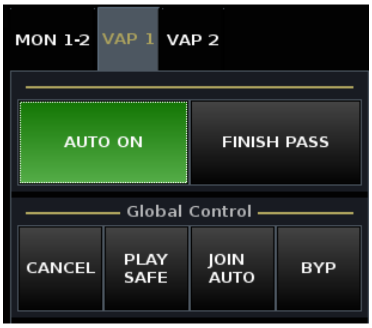
*As soon as you touch a fader, its **R/W** user button changes from green to red to indicate that you are writing new data.*

You will see that a **Record pass** is created – as this is the first pass, it is named **pass0000**:

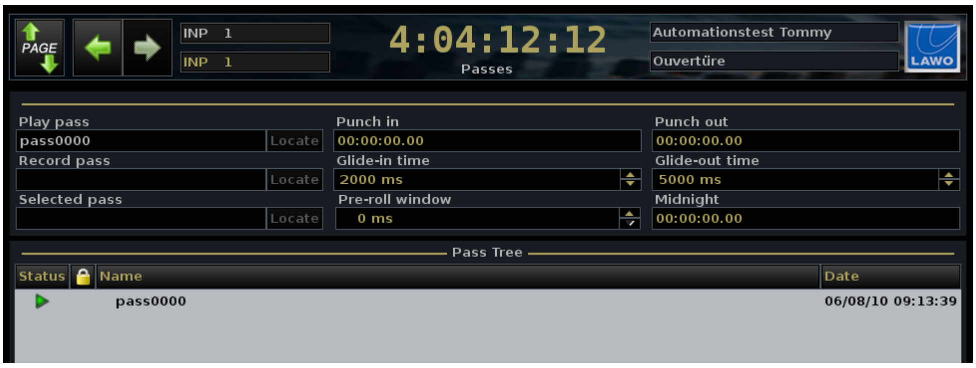
> You cannot create a new **Record pass** if the mix is protected.
>
> You will not be able to write dynamic automation if the fader has been disarmed or is running in static automation mode.

6. When you are ready, finish the pass in one of two ways:

- Press rewind or locate backwards; the change of timecode direction causes the pass to finish automatically.
- Press the **FINISH PASS** button (on VAP 1) to finish the pass manually:



*The stream of automation data is recorded in **pass0000** which moves to the **Play pass** box in the **Passes** display:*



*In addition, all fader **R/W** user buttons return to green indicating that they are back in read mode.*

7. Locate back to the beginning of the mix and press play.

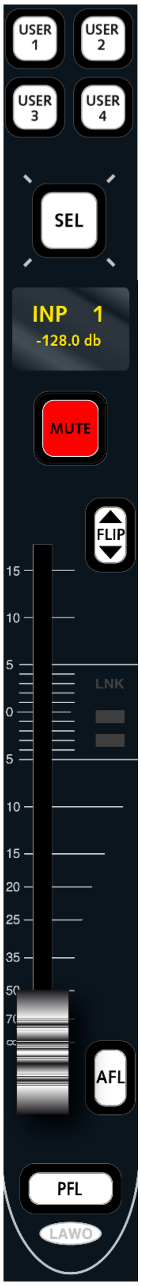
Watch your recorded moves play back against timecode!

A normal automation day starts at 00:00:00:00 and ends at 23:59:59:xx, meaning that the maximum mix pass length is 24 hours!

If your audio starts or crosses 00:00:00:00, then you should offset the timecode from the playback device to avoid the 23:59:59:xx/00:00:00:00 change of day.

Updating a Pass

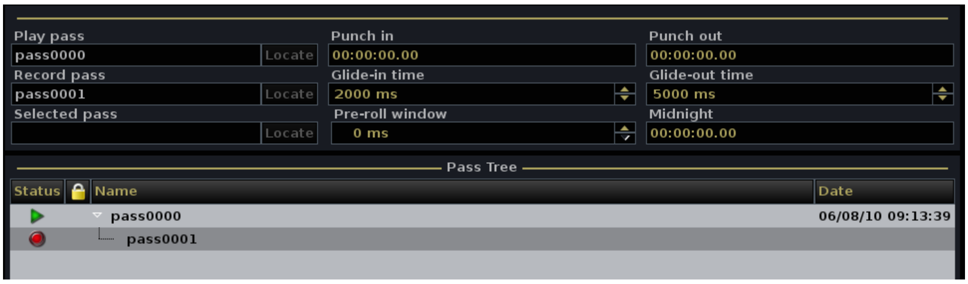
To update the moves in **pass0000**:



1. Press play and touch the faders you want to update.

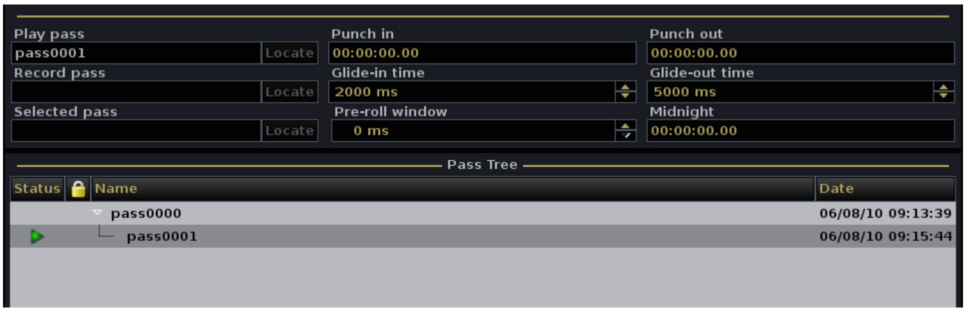
   *The fader **R/W** user buttons, programmed from the Custom Functions display, turn red to show that they are back in write. The **R/W** buttons on untouched faders remain green and play back the moves from **pass0000**.*

   Having written some moves, a new **Record pass – pass0001 –** is created:



2. Finish the pass, either by pressing **FINISH PASS** or locating backwards.

   ***Pass0001** now becomes the current **Play pass** ready for further updates:*



If **FILL END** is still selected as the 'Stepout mode' (on VAP 1), any levels in write when you finish the pass are written to the end of the mix. For alternatives, see Step Out Automation Modes.

3. Continue updating the mix as above.

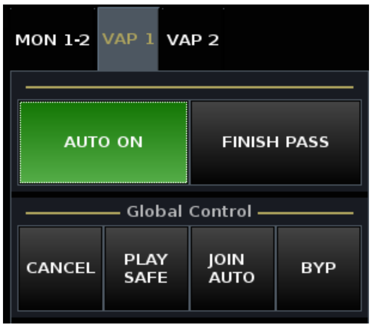
Each time you update the current **Play pass**, a new **Record pass** is created with a new unique reference number – **pass0002**, **pass0003**, etc.

The passes are kept within the **Pass Tree** showing the history of each update.

Cancelling a Pass

If you start a new **Record pass** and make a mistake, you can throw away the data before finishing the pass:

1. Press the **CANCEL** touch-screen button (on VAP 1):



The next time you begin a **Record pass**, it takes the next unique pass number. For example, if you cancel **pass0001**, then the next **Record pass** is named **pass0002**.

Reverting to an Earlier Pass

If you've made a mistake and have already finished the pass, then you can go back to an earlier Play pass using the **Pass tree**:

1. Select the pass to revert to, for example **pass0002**.

*The name of the pass is shown in the **Selected pass** box.*

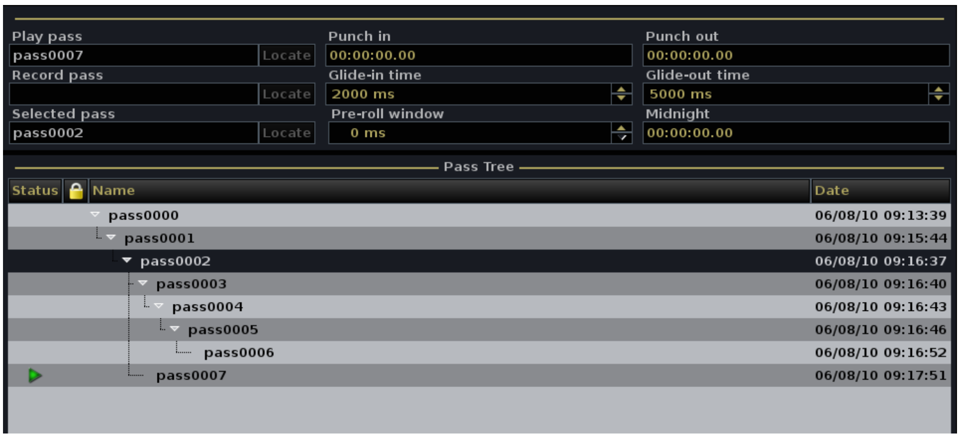
2. Press the **PLAY** soft key, or right-click and select **Play** from the drop-down menu options.

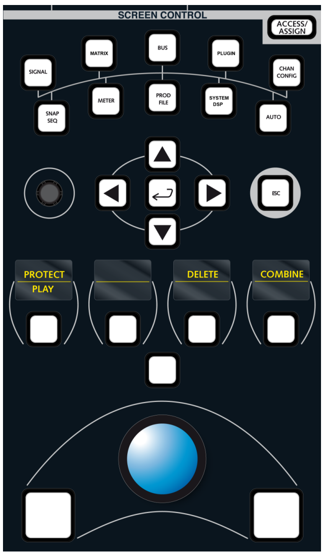
***Pass0002** becomes the current **Play pass** and the console settings update to reflect the new replay data.*

3. Now go into play and write some fader updates.

4. Finish the pass, either by pressing **FINISH PASS** or locating backwards.

*The newly created pass takes the next unique pass reference number, in our example **pass0007**, and appears as a new branch in the **Pass tree**:*





The **Pass tree** provides a history for every pass created within the active mix. You can use the **Pass tree** to A/B between different mixes or to write different versions of automation for a chorus or scene. Passes may be loaded, renamed, deleted and edited.

For more details, see the Passes display.

Saving & Loading Automation Data

Every time you finish a pass, you create a new pass which is stored within the active mix memory. At any time, you may create a new mix and store any number of passes within it.

Note that *all* this data remains in temporary memory, until you either update (or save) a production. The system then stores all the mixes in memory, and the passes within them, into the production on the user data flash card. See What's Stored in a Production.

In addition, the system stores which mix is active, and which pass is the Play pass for each mix. This means that when you load back a production, you will always get back to the last mix and pass you were working on.

So:

- to save your automation data, either update or save a production.
- to load a mix stored within the same production, simply load a mix and turn on Automation.
- to load a mix from a different production, load the production.

You can use a single production to store multiple mixes for, say, all the songs on an album.

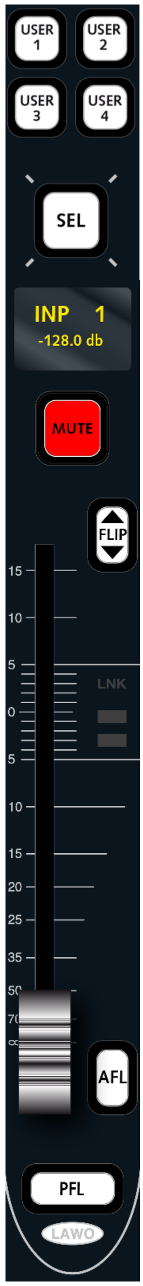
Within each mix (song), create different passes to manage your mix variations - for example, vocals higher, rhythm section lower, etc.

Individual mixes or productions may be imported and exported to a USB interface, mxGUI computer or network drive. This allows you to archive or transfer automation data between systems. See File Import/Export for details.

Updating Fader Moves

So far we have updated fader moves by touching the faders to step into write, and finishing the pass to step out of write. By using the **FILL END** 'Stepout mode', any levels in write when the pass is finished are written through to the end of the mix.

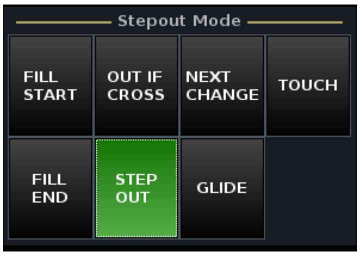
To go back and correct moves earlier in the song or production, we need to change from **FILL END** to **STEP OUT**. In addition, you can make mixing more efficient by using the fader **R/W** user buttons, programmed from the Custom Functions display,, or **TOUCH**, to step out of write while in play.



Step Out

Writing in **STEP OUT** means that any parameter in write reverts to the play pass data when you step out of write. This allows you to write a new move early on in the song or production, step out of write and keep all the moves which follow from the previous Play pass.

1. Select the **STEP OUT** touch-screen button (on VAP 1), to change from 'fill to end' to 'step out' automation:



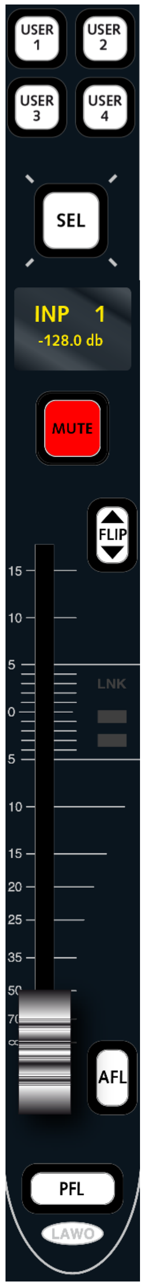
2. Press Play and touch the faders you want to update.

*The **R/W** user buttons on these faders turn red to show that they are in write.*

3. Finish the pass, either by pressing **FINISH PASS** or locating backwards.

Now play back the pass and you should see your new fader moves followed by moves from the previous Play pass.

Using the Fader R/W Buttons



To make mixing more efficient you can use the fader **R/W** user buttons, programmed from the Custom Functions display, to step in and out of write while in play. This allows you to step in and out of write several times during a single pass.

1. First select the **WRITE R/W** touch-screen button (on VAP 1):

2. To step in to write, you can now either touch the fader or press its **R/W** user button.

*The **R/W** button turns red to show that the fader is in write.*

Note that when **WRITE R/W** is not enabled, the **R/W** user button disarms the fader.

3. To step out of write, press the fader **R/W** button again.

*The fader jumps back to the Play pass position, and its **R/W** button turns green to indicate that the fader is now reading the Play pass.*

4. Continue stepping in and out of write on as many faders as you wish, and throughout the pass.

5. Finish the pass, either by pressing **FINISH PASS** or locating backwards.

All the updates you have made are recorded in the new Play pass.

You can combine **STEP OUT** and **WRITE R/W** with other modes such as GLIDE, OUT IF CROSS or NEXT CHANGE.

You can also step in and out of write globally (for all automated parameters) using the START and STOP WRITE buttons. Or, for a cluster of channels, using the CLUST button.

Touch

If you would like the faders to step out of write when you release them, then turn on the **TOUCH** button (on VAP 1):

1. Step into write by touching the fader.

*The fader **R/W** user button turns red to show that the fader is in write.*

2. Keep touching the fader and when you wish to step out of write, release the fader.

*The fader jumps back to its previous pass position and its **R/W** user button turns green to indicate that the fader is now reading back the Play pass.*

You can combine **TOUCH** with GLIDE if you wish the faders to glide back to the Play pass on release.

TOUCH applies to any variable control so you can use it on touch sensitive rotary controls such as panning, aux sends, etc.

To offset existing fader moves, use TRIM automation.

Writing Automation on Controls and Switches

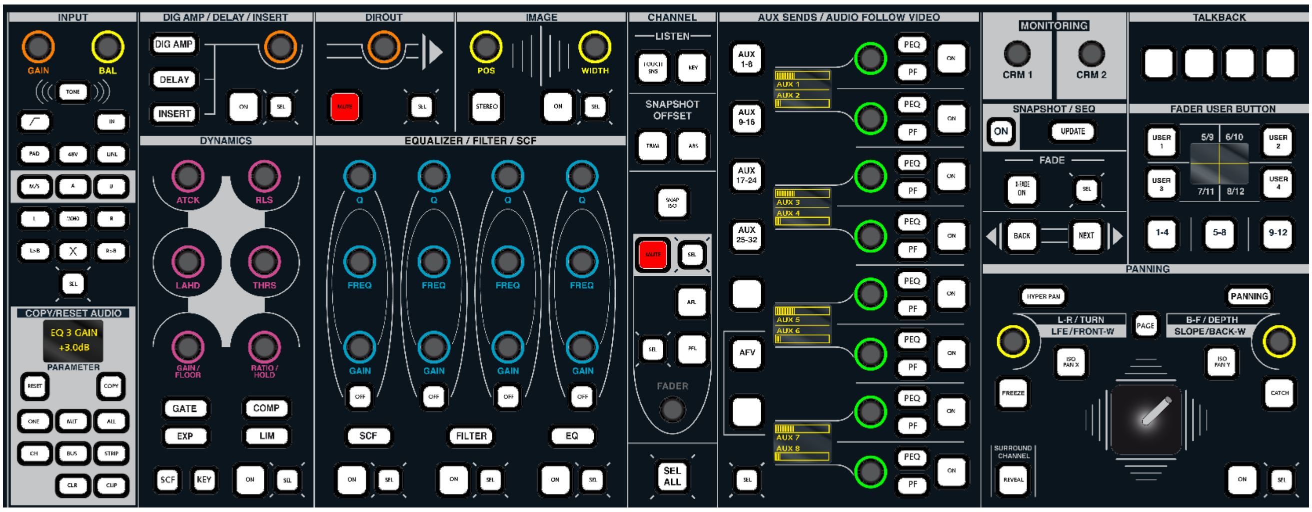
In addition to faders, you can write automation for any other channel control, for example, mutes, panning, EQ, even functions such as signal processing order and bus routing.

Return to the SEL AUTO mode, but this time enable automation for say the fader, mute and pan.

Faders and rotary controls are touch sensitive which allows them to step into write whenever you touch a fader or control. Switches step into write whenever you change the state of the switch or force a step in using the WRITE R/W mode.

You can check which parameters are in read or write on an individual channel, by assigning the channel to the Central Control Section, and pressing **SEL ARM** (on VAP 1).

The **SEL** button on each audio module now lights as follows:



- **SEL** button lit (green) = the complete audio module is in read.
- **SEL** button lit (orange) = at least one parameter within a module is in write. For example, the EQ1 Gain.
- **SEL** button lit (red) = the complete audio module is in write. For example, the 4-band EQ section.
- **SEL** button off = the complete audio module is disarmed (read only).

Let's look at some examples - writing automated mutes and a dynamic pan move.

Writing Switch Automation

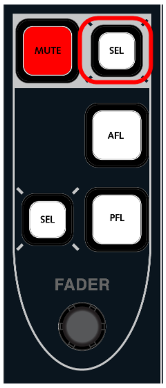
Switches step into write whenever you change the state of the switch or force a step in using the WRITE R/W mode. Assuming that you have enabled automation for the mute buttons, you should be ready to go.

> You will not be able to write dynamic automation if the switch has been disarmed or is running in static automation mode.

1. With automation enabled, go into play and press the **MUTE** button either on the fader strip or Central Control Section to write your changes.

2. To check that you are writing automation, assign the channel to the Central Control Section, and press **SEL ARM** (on VAP 1):

*The mute **SEL** button turns orange when in write:*



3. Finish the pass, either by pressing **FINISH PASS** or locating backwards, and play back the automation.

*The mute **SEL** button turns green when in replay.*

Updating Switch Automation

You can update switch automation by rewinding and rewriting the switch change. However, if you want to remove a switch change you will want to step in and out of write while in play.

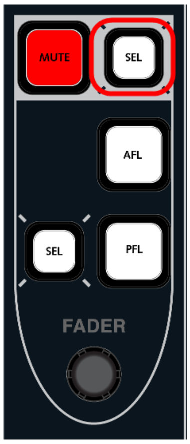
1. Assign the channel you want to automate to the Central Control Section.

2. Select the **WRITE R/W** and **SEL ARM** touch-screen buttons (on VAP 1):

*The **SEL** buttons beside each Central Control Section module now allow you to step in and step out of write for that module. (i.e. they behave like the fader **R/W** buttons described earlier).*

3. Rewind before the switch change you want to remove.

4. Step in to write by pressing the **SEL** button beside the switch on the Central Control Section - for example:



*The switch goes into write in its current state – i.e. mute off. The **SEL** button turns red to show that the complete **MUTE** section is in write.*

5. Locate or play past the end of the unwanted mute.

6. To step out of write, press the ***SEL*** button again.

*The **SEL** button turns green to indicate it is now in replay and the mute button reverts to the data from the Play pass.*

7. Finish the pass, either by pressing **FINISH PASS** or locating backwards.

The updates you have made are recorded in the new Play pass.

Another great way to update switch automation is to combine **STEP OUT** with NEXT CHANGE.

Writing Rotary Control Automation

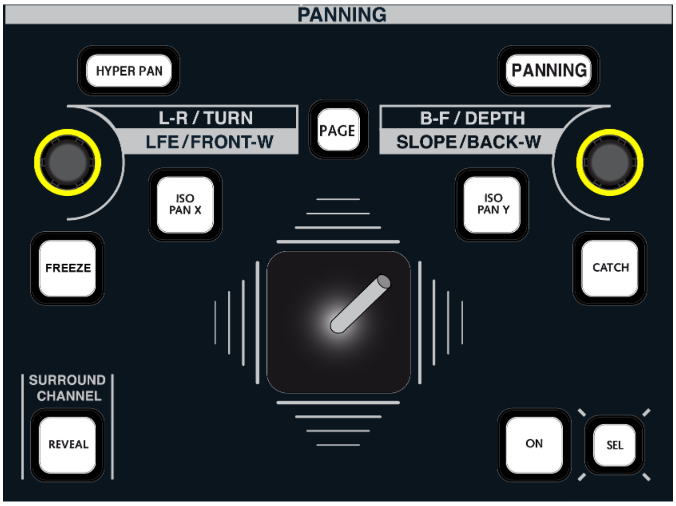
Rotary controls, like faders, are touch sensitive and go into write when you touch them or force a step in using the WRITE R/W mode. Let's write a dynamic pan left/right pan move. Assuming that you have enabled automation for the pan module, you should be ready to go.

> You will not be able to write dynamic automation if the control has been disarmed or is running in static automation mode.

1. With automation on, go into play and move the left/right pan control either from a free control or the Central Control Section to write your changes.

2. To check that you are writing automation, assign the channel to the Central Control Section, and press **SEL ARM** (on VAP 1).

*The panning **SEL** button turns orange when an individual control is in write:*



3. Finish the pass, either by pressing **FINISH PASS** or locating backwards, and play back the automation.

*The panning **SEL** button turns green when the module is in replay.*

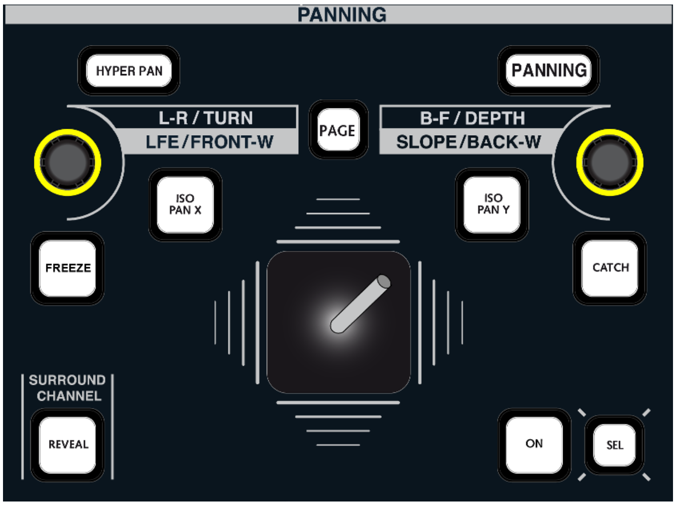
Updating Rotary Control Automation

You can update the pan move by rewinding and rewriting the move. However, if the move starts too early or you want to write other automation on other parameters you will want to step in and out of write while in play.

1. Assign the channel you want to automate to the Central Control Section.
2. Select the **WRITE R/W** and **SEL ARM** touch-screen buttons (on VAP 1):

*The **SEL** buttons beside each Central Control Section module now allow you to step in and step out of write for the complete module.*

3. Rewind before the start of the pan move you want to update.
4. Step in to write by pressing the **SEL** button beside the pan module on the Central Control Section:



*The pan module goes into write in its current state – i.e. left/right pan at its starting position. The **SEL** button turns red to show that the complete module is in write.*

5. Now go into play and move the left/right pan control to rewrite your move at the correct timecode.
6. To step out of write, press the **SEL** button again.

*The **SEL** button turns green to indicate that the pan module is now in replay and the left/right pan control reverts to the data from the Play pass.*

7. Finish the pass, either by pressing **FINISH PASS** or locating backwards.

The updates you have made are recorded in the new Play pass.

Another great way to update rotary control automation is to combine **STEP OUT** with OUT IF CROSS.

Or, if you would like the controls to step out of write when you release them, then turn on TOUCH.

Updating Automation on Individual Controls

When using the **WRITE R/W** and **SEL ARM** mode, you have the option to step in either on the complete audio module, or on an individual control as follows:

1. Select the **WRITE R/W** and **SEL ARM** touch-screen buttons (on VAP 1):

2. Assign the channel you are automating to the Central Control Section.

3. With automation on, go into play.

The distinction between whether you write an individual control, for example EQ1 Gain, or the complete module is made as follows:

4. Move the EQ1 Gain rotary control to step in to write just on the one control.

*The EQ **SEL** button turns orange indicating that only part of the module is in write.*

5. Alternatively, press the EQ ***SEL*** button to force the module into write in its current state.

*The EQ **SEL** button turns red indicating that all controls and switches within the EQ module are in write.*

6. Make your changes and step out either by pressing the EQ ***SEL*** button or finishing the pass in the usual manner.

Automation Modes

The way in which automation is written is governed by three primary modes:

- Dynamic or Static automation
- The STEP OUT mode
- Absolute or Trim

Dynamic or Static Automation

The **SEL DYN** function selects which modules within each channel write dynamic or static automation.

For example, you may wish to emulate an analogue console's automation system by writing dynamic fader and mute changes, while keeping all EQ, Compression, etc. at one static setting for the entire mix.

In theory, it is not strictly necessary to select static automation if you want to save a single EQ setting for an entire mix. If you select only faders and mutes for automation, then all other modules remain at their current settings (in "manual") while running automation.

However, if you then disable automation, load a different snapshot and re-enable automation. Because the EQ and other settings were not stored in the mix pass, you will not get back those settings simply by enabling the automation. To work in this way, you will need to make sure that you have saved a snapshot for all settings outside of the automation pass.

For this reason, we recommend selecting *all* modules for automation. You can then use static or dynamic automation modes to control whether settings are written dynamically or not.

You can select dynamic or static automation for any number of parameters on any number of channels. The first time modules are selected for automation, they default to dynamic automation.

To change the mode, use the **SEL DYN** touch-screen button (on VAP 1):

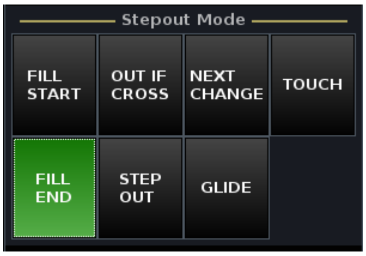
The selection process works in a similar manner to **SEL AUTO**, see Selecting Channels/Modules for Automation. Note that the Central Control Section select buttons light as follows:

- **SEL** button lit = static automation mode.
- **SEL** button off = dynamic automation mode.

Remember to use **MLT**, to apply selections to multiple channels, and refresh the fader **SEL** buttons if you are updating existing selections.

Step Out Automation Modes

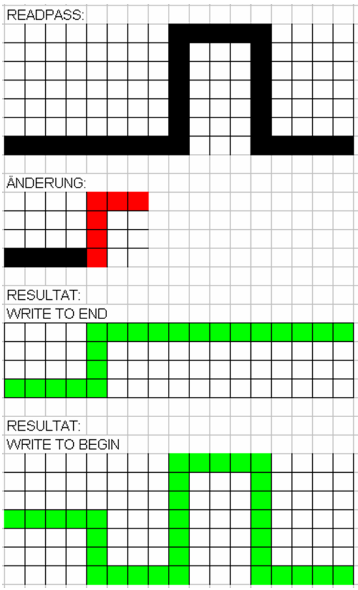
These modes affect what happens when you step out of write, and are selected from the 'Stepout Mode' touch-screen buttons (on VAP 1):



There is a choice of several modes, many of which can be used in combination with each other to achieve different results.

It is the mode selected when you finish the pass which is applied. For example, if you are in **STEP OUT** mode while playing through the chorus but then decide you wish to write the updated values to the beginning of the mix, you can stop, change to **FILL START** and then press **FINISH PASS** to finish the pass.

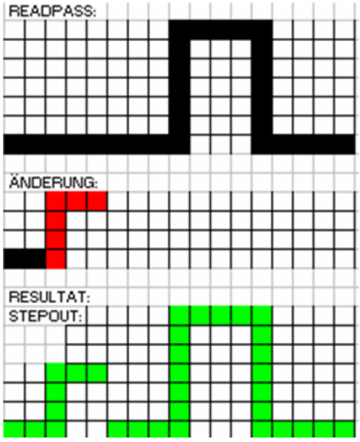
FILL START and FILL END



- **FILL END** – this automation mode is great for writing your first passes where you are working through the song or production chronologically. Each time you stop and finish a pass, any values in write are written through to the end of the mix. This means that you don't have to play through the whole song just to write a fader level to the end of the mix.
- **FILL START** – using this mode, any values in write are written back to the start of the mix.
- **FILL END** plus **FILL START** – with both modes selected, any values in write are written as a static value for the whole mix.

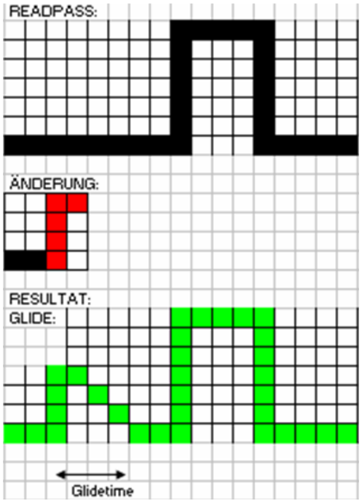
You may also write static values between specific timecode points using the **Punch In** and **Punch Out** times in conjunction with the COMMAND FILL button.

STEP OUT



Writing in **STEP OUT** mode means that any parameter in write reverts to the Play pass data when you step out of write. This allows you to write a new move early on in the song or production, step out of write and keep all your following moves from the previous Play pass.

GLIDE



Combine **STEP OUT** with **GLIDE** to create a glide back to the Play pass when you step out of write.

GLIDE is applied to all variable parameters – fader levels, panning, aux send levels, etc. – and can be used in conjunction with **TOUCH** such that controls will step out of write on release and glide back to the Play pass.

The glide time is set by the Glide-out time at the top of the **Passes** display.

OUT IF CROSS

This mode can be combined with **STEP OUT**, **FILL END** or **FILL START** and is a great mode for updating variable parameters such as fader levels.

When selected, any values in write will automatically step out when their value crosses the read pass.

For example, you may use this mode if you wish to update a fader level before a fantastic move you have just written! Go back and update your level; when the read pass crosses the new level, the automation automatically steps out and replays your fantastic move from earlier!

NEXT CHANGE

This mode can be combined with **STEP OUT**, **FILL END** or **FILL START** and is great mode for updating switched parameters such as mutes.

When selected, any values in write will automatically step out when a parameter change occurs in the read pass:

For example, let's say that you have written some mute automation and now wish to update a section earlier in the mix. Go back and update your mute automation and leave your mute switch in write. When the next change of mute position occurs in the read pass, then the mute will automatically step out of write.

TOUCH

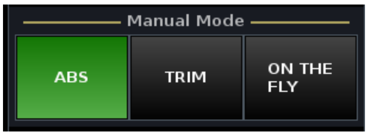
If you would like faders or rotary controls to step out of write when you release them, then turn on **TOUCH**.

You can combine **TOUCH** with GLIDE if you wish the faders or controls to glide back to the Play pass on release.

Note that **TOUCH** applies to any variable control so you can use it on touch sensitive rotary controls such as panning and aux sends as well as faders!

Absolute and Trim

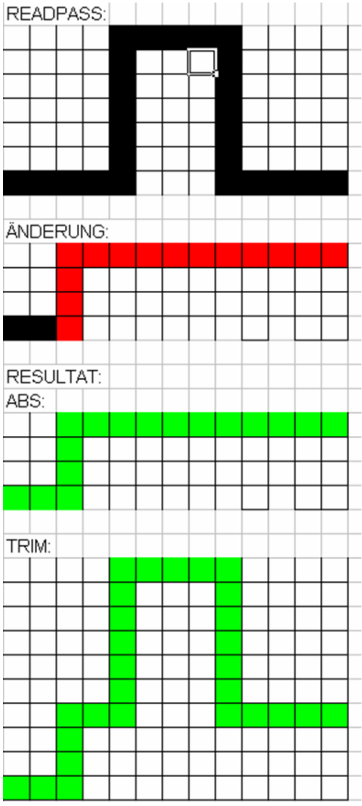
The **ABS** and **TRIM** touch-screen buttons (on VAP 2) determine how automation data is written:



So far all our automation data has been written as Absolute data. In other words, when you put a control into write you are overwriting its absolute value.

Trim mode may be used to offset existing values. For example, you may have written some good fader moves for the chorus, and now you'd like to trim the moves up or down in level as you mix.

Trim works by offsetting the absolute data by a trim value. When you finish the pass, either by rolling back in time or pressing **FINISH PASS**, the trim data is automatically combined with the original Play pass to create a new absolute Play pass. The diagrams below show the affect of an update (ANDERUNG) to the Play pass (READPASS) when written in absolute, and when written in trim:



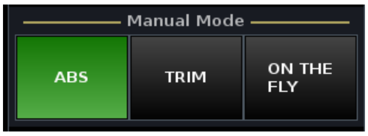
Note that Trim can be used to offset dB parameters such as fader level and EQ gain, or ranges such as Pan L/R. Other parameters, such as EQ frequency, cannot be trimmed and will always update in absolute.

Also note that Trim can be selected either globally across the console or selectively for specific controls or channels.

Global Trim

The simplest way to use Trim is to activate **Trim** as a global automation mode.

This selection is made from the 'Manual Mode' buttons (on VAP 2):



1. Select **TRIM** to activate trim.
2. Select **ABS** to return to absolute.

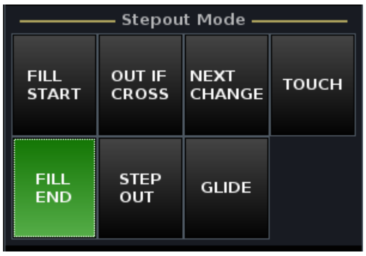
Note that if both buttons are off (unlit), then some controls are selected for trim while others remain in absolute. See Selective Trim for details.

Note also that certain parameters, such as frequency, cannot be trimmed, and will always update in absolute regardless of the **ABS**/**TRIM** mode.

Trim Modes

Once **TRIM** is enabled, you have the choice of two different Trim modes -Trim On the Fly or Trim Relative. Both modes may be used for any trimmable parameter, but to explain the modes, let's trim a fader.

Note that, in each case, trim is applied according to the Stepout Mode, so check the status of these buttons before performing your update:



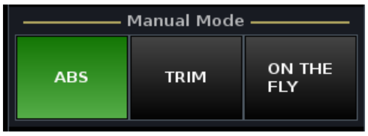
For example:

- To trim a control to the end of the mix, select **FILL END**.
- To trim a section of the mix, you could use **STEP OUT** (steps out of write when you finish the pass), or **TOUCH** (steps out of write when you let go of the control).

>> Trim On the Fly

Trim On the Fly is great if you wish to keep a sense of the underlying Play pass from the physical fader positions, as the faders replay the Play pass, and only stop moving when you touch them.

1. Select **TRIM** and **ON THE FLY** (lit) from VAP 2:



2. Select the Stepout Mode, for example, **TOUCH** from VAP 1.
3. While automation is playing back, touch the fader to update its position.

The fader stops moving allowing you to change its position.

Any level changes are written as a trim offset; the amount of trim is shown in the Fader Label Display, temporarily replacing the Play pass level.

4. Let go of the fader to step out of write.

As soon as you let go, the fader returns to replay. Fader moves from the current Play pass are replayed and the Fader Label Display returns to the Play pass value.

By enabling **GLIDE** you can have your fader automatically glide back to the Play pass when you let go.

As an alternative to **TOUCH** you could use **STEP OUT** with **WRITE R/W**. Move the fader to step into write and apply your trim offset. Let go of the fader and moves replay from the Play pass. The fader remains in write until you finish the pass. This method of working means that you don't have to keep touching the fader for the duration of the trim update.

>> Trim Relative

Trim Relative is great if you wish to use the physical position of the fader to show the amount of trim offset.

1. Select **TRIM** and deselect **ON THE FLY** (unlit) from VAP 2.

As soon as you enter Trim Relative mode, all faders selected for trim move to a default position (0dB).

2. Select the Stepout Mode, for example, **STEP OUT** from VAP 1.
3. Press **PLAY** to replay the Play Pass.

In Trim Relative, the faders do not move so use the Fader Label Displays to view any changes in level applied by the Play pass.

4. Touch the fader to step into write.

Any level changes are written as a trim offset; the amount of trim is shown in the Fader Label Display, temporarily replacing the Play pass level.

5. Because you selected the **STEP OUT** mode, you can let go of the fader and it remains in write (trim).

Note that as soon as you let go, the Fader Label Display returns to the automation values from the Play pass. The fader position represents the trim offset (from 0dB).

6. When you want to step out of write, finish the pass.

The fader returns to replay.

Selective Trim

If you wish to update some controls or channels in Trim while others update in Absolute, then you can:

- Define a channel User Button to select Trim or Absolute on a channel-by-channel basis. (This function must be programmed from the **Custom Functions** display, see Fader User Buttons, Channel Functions.)
- Use **SEL TRIM** as described below.

Note that automation must be enabled (**AUTO ON** lit), and any selections you make are temporary. So, if you turn automation off and back on, all parameters are reset to **ABS**.

You can select any number of parameters on any number of channels to be in Trim, using the **SEL TRIM** touch-screen button (on VAP 1):

The selection process works in a similar manner to **SEL AUTO**, see Selecting Channels/Modules for Automation. Note that not all controls can be trimmed. The Central Control Section select buttons light as follows:

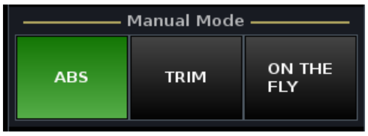
- **SEL** button lit = trim.
- **SEL** button off = absolute.

Remember to use **MLT**, to apply selections to multiple channels, and refresh the fader **SEL** buttons if you are updating existing selections.

> You may also use the Cluster function to select trim for a cluster of channels.

If you have selected a mixture of Abs and Trim statuses, then this is indicated on the 'Manual Mode' panel (on VAP 2) where you will see both **ABS** and **TRIM** buttons are off (unlit).

1. To reset all controls and channels to Absolute, press the global **ABS** button:

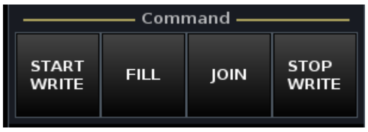


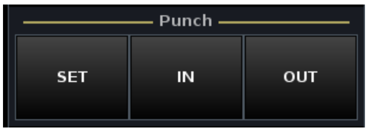
2. Or, to reset all controls and channels to Trim, press global **TRIM**.

Command Functions

Earlier we used the Central Control Section **SEL** buttons to step in and out of write on individual channel parameters (using WRITE R/W).

However, there are a number of 'Command' functions (on VAP 2) which you can use to step in or out of write across multiple channels, or to set an automatic step in/step out between timecodes (Punch In/Punch Out):





Note that the 'Command' functions only affect which elements of the console step in or out of write. The way in which automation data is written is still governed by the automation mode.

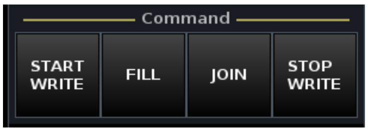
For example, if you use **START WRITE** in combination with **FILL END**:

1. Press Play on your machine so that timecode is rolling.
2. Press **START WRITE** and all parameters enabled for automation step into write at their current positions.
3. What happens next depends on your choice of operation:

- If you locate backwards to finish the pass, then the **FILL END** stepout mode is applied. In other words, the values in write will be written to the end of the mix.
- However, if instead of finishing the pass, you press **STOP WRITE**, all your parameters will step out back into replay. In other words, you have achieved a step in and step out, without having to change automation mode!
- If you combine the above with **GLIDE**, then rather than an instant step out, variable parameters will glide back to their replay positions.

Global Step In/Step Out

The **START WRITE** and **STOP WRITE** touch-screen buttons (on VAP 2) allow you to step in and out of write globally across all automated parameters on the console:



1. Select **START WRITE** to step into write across the console.

All parameters and modules which have been selected for automation step into write.

2. Press **STOP WRITE** to step out of write across the console.

Any parameters in write step out back to the Play pass.

Use **STOP WRITE** to step out on all parameters at a section change such as the end of a chorus.
Combine **STOP WRITE** with **GLIDE** to glide back to the Play pass values.

Cluster Step In/Step Out

You can use the cluster function to step in or out of write across multiple channels.

The cluster works like a group but just for automation parameters. First define which channels you wish to cluster. Then when you step into write on say the EQ on one channel, all EQ sections within the cluster also step into write.

➢ **To define the cluster:**

1. Select the **CLUST** touch-screen button (on VAP 1):

*The button flashes, and all the fader **SEL** buttons across the console flash, in green.*

2. Press the fader **SEL** buttons to add channels to the cluster - you can select any number of channels, from any bank or layer.

*The fader **SEL** buttons change from green to red when selected:*

3. Deselect the **CLUST** button to complete this part of the operation.

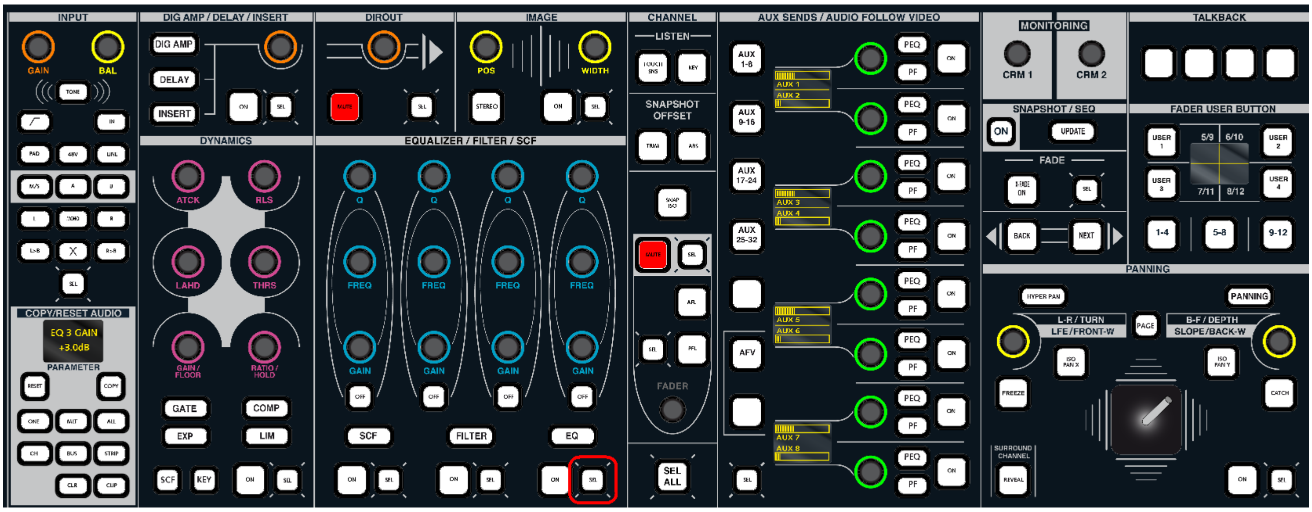
The cluster is now defined.

Note that the cluster remains active until you repeat the steps above and remove all channels from the cluster.

➢ To step in and out of write using the Cluster

1. Assign any channel within the cluster to the Central Control Section, by pressing its fader **SEL** button.

2. Step into write on the EQ module using the **SEL** button:



All EQ modules within the cluster step into write at their current values.

Note that parameters will only step into write if they have been selected for automation and are armed.

➢ Using the Cluster for other Functions

The cluster may also be used to arm or disarm modules, or select Trim automation, across the clustered channels.

Define the channel cluster. Then with one of the clustered channels 'in access', use either **SEL ARM** or **SEL TRIM** and make your module selections - the selections are applied to all channels within the cluster.

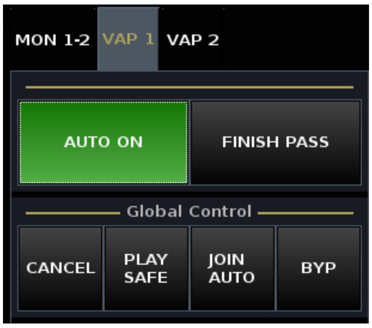
Join

The **JOIN** and **JOIN AUTO** buttons also allow you to step into write across a selection of parameters. However, the parameters which are 'joined' are automatically defined for you and are the parameters which were in write when you finished your last pass.

These functions are especially useful when working on a section of the mix, such as the Chorus of a song, where you are constantly rewinding to make updates.

Auto Join

1. Select the **JOIN AUTO** touch-screen button (available on both VAP 1 & 2):



2. Locate to the beginning of the Chorus and press Play.
3. During the Chorus write some fader and control moves.

*You will now have a selection of parameters in write as indicated by the red **SEL** buttons on the Central Control Section (if the channel is in access).*

4. Locate back to the beginning of the Chorus and press play to play back the pass.

Your moves replay and at the timecode where you located backwards (or finished your last pass), all the parameters which were in write in step 3 automatically step into write at their current value (this is called an auto join).

So, by working in **JOIN AUTO**, you can be constantly rewinding to make updates without having to pay attention to the step out point or to which parameters you updated.

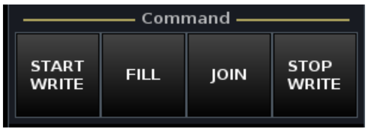
Command Join

This function is similar to Auto Join but allows you to join controls manually. This can be useful for overwriting a move you didn't like. For example:

1. Locate to the beginning of the Chorus and press Play.
2. During the Chorus write some fader and control moves.

You will now have a selection of parameters in write. However, let's say that you liked the first series of moves but not the latter.

3. Locate back to the beginning of the Chorus and press play.
4. Watch your moves replay and at the point where you wish to step back into write, press the **JOIN** touch-screen button (on VAP 2):



All the parameters which were in write in step 2 now step into write (join) at their current value. If you keep playing you will now overwrite your unwanted moves.

Punch In/Punch Out Automation

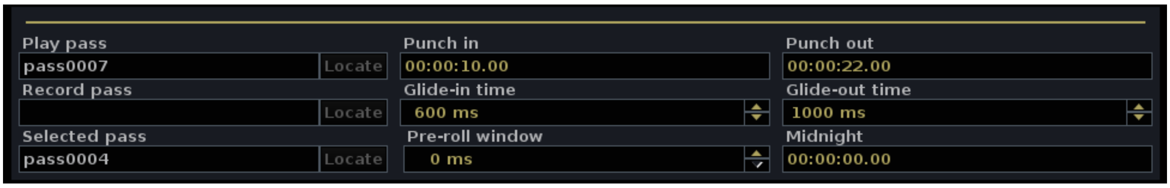
The **Punch In** and **Punch Out** times can be used for two different applications:

- To automatically punch in and punch out of write, so that you do not accidentally update automation outside of a specified timecode window.
- To apply parameter values to a region of the mix. For example, to write values for the whole of a Chorus or scene.

In either case, first you need to set the punch in and out times as follows:

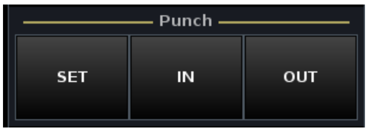
1. Press the **AUTO** button, located on the SCREEN CONTROL panel, to view the **Passes** display.

*The **Punch In** and **Punch Out** times are shown at the top of the display:*



2. Play or locate your timecode to the required punch in time.

3. And on VAP 2, press **SET** (it will flash) followed by **IN**:



*The current timecode position is entered in the **Punch in** time box on the **Passes** display.*

4. Now play or locate your machine to the required punch out time, and press **SET** followed by **OUT**.

*The current timecode position is entered in the **Punch out** time box on the **Passes** display.*

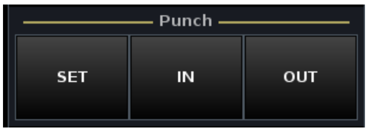
6. Deselect the flashing **SET** button to complete this part of the operation.

You may use **SET** and **IN**/**OUT** while in play to enter Punch In and Punch Out times 'on the fly'.
Click in **Punch In** or **Punch Out** time boxes to enter a timecode manually from the console keyboard.

Automatic Punch In and Out

To use the punch in and out times to automatically step in and out of write.

1. Make sure that the **SET** button (on VAP 2) is off:



2. Turn on both the **IN** and **OUT** buttons to make the punch in and punch out times active. (Or, select the **IN** and **OUT** buttons independently if you wish to only step in or only step out.)

The buttons turn blue when active.

3. Now rewind before the punch in timecode and press Play.

*At the **Punch in** time, all parameters and modules which have been selected for automation step into write at their current values.*

4. You can now write new moves into the automation.

*When you pass through the **Punch out** time, all parameters and modules step out of write back to the Play pass.*

5. When you have finished mixing that section, remember to deselect the **IN** and **OUT** buttons to deactivate the automatic punch in/punch out mode.

Fill Region

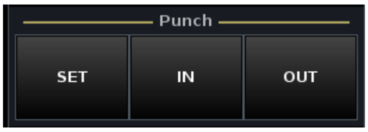
To apply parameter values to a region of the mix. For example, to write values for the whole of a Chorus or scene:

1. Set the **Punch In** and **Punch Out** times, as described earlier, to define the start and end of the region.

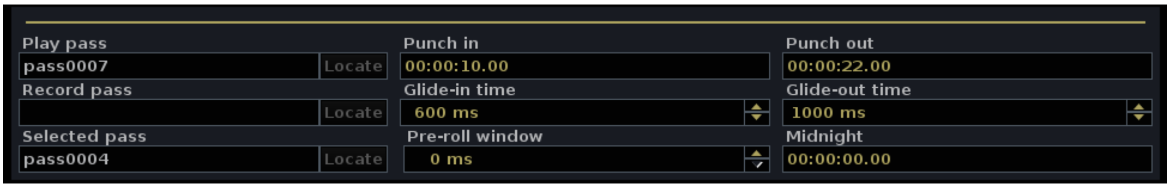
2. Now play through the section of the mix and adjust any parameters to the values you wish to write for the region.

You will now have a selection of parameters in write.

3. Before you rewind or finish the pass, press the **FILL** button (on VAP 2):



*Any parameters in write are written at their current value between the **Punch In** and the **Punch Out** times:*



Protecting Automation Data

Having written automation, you may wish to play back your mix data but protect it from being overwritten. There are a number of options available:

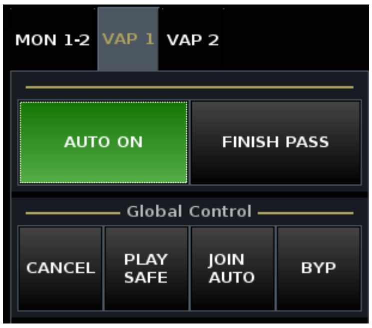
- PLAY SAFE – all channels read automation data from the Play pass but cannot write new data. If you adjust a parameter you will *NOT* hear any change in audio.
- BYP – identical to ‘Play Safe’, except that if you adjust a parameter you *WILL* hear the change in audio. If you like the new parameter value, you can step the control into write using the DIRECT IN button.
- SEL ARM – can be used to protect individual parameters. Armed controls are armed for reading and writing automation data. Disarmed controls will read automation but cannot write new data.

Play Safe

In this mode, all channels read automation data from the Play pass but cannot write new data.

In addition, if you adjust a parameter, you will *NOT* hear any change in the audio. 'Play Safe' applies globally to all channels and parameters, and is a great mode to use when laying back your mix.

1. Select the **PLAY SAFE** touch-screen button (available on both VAP 1 & 2):



Once selected, all channels enabled for automation will read data from the Play pass but not write new data if touched or changed.

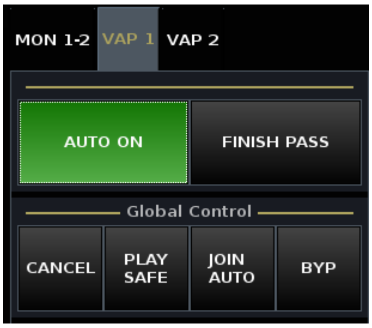
2. If you adjust a parameter, you will *NOT* hear any change in the audio. When you let go of the control, the parameter reverts to its Play pass position.

Bypass

This mode is identical to ‘Play Safe’, except that if you adjust a parameter you *WILL* hear the change in audio.

'Bypass' applies globally to all channels and parameters, and is a great mode to use when auditioning your mix.

1. Select the **BYP** touch-screen button (available on both VAP 1 & 2):



Once selected, all channels enabled for automation will read data from the Play pass but not write new data.

2. If you adjust a parameter, you will hear the change. You then have two options:

- If you Stop, Rewind and press Play, the parameter will revert to its Play pass position.
- If you like the new parameter value, you can step into write at the new value using the **DIRECT IN** button:

3. Press the **DIRECT IN** button (on VAP 1):

Any parameters which have been altered from the Play pass position step into write at the current value. If you now finish the pass, automation is written according to your choice of Stepout mode.

The **DIRECT IN** button is *only* active when running in Bypass mode.

Arm and Disarm

The **SEL ARM** function allows you to protect automation data on individual modules within each channel. Armed modules are armed for reading and writing automation data. Disarmed modules will read automation but cannot write new data.

You can arm or disarm any number of parameters on any number of channels. The first time modules are selected for automation, they default to armed.

To change the mode, use the **SEL ARM** touch-screen button (on VAP 1):

The selection process works in a similar manner to **SEL AUTO**, see Selecting Channels/Modules for Automation. Note that the Central Control Section **SEL** buttons light as follows:

- **SEL** button lit (green) = module is armed (read and write).
- **SEL** button off = module is disarmed (read only).

Remember to use **MLT**, to apply selections to multiple channels, and refresh the fader **SEL** buttons if you are updating existing selections.

> You may also use the Cluster function to arm or disarm modules for a cluster of channels.

Recalling a Snapshot or Sequence

You may recall snapshots or play out a sequence, with or without crossfades, while running the automation system. For example, to record a complete scene/section change against timecode.

The system behaves as if every control was touched and therefore allows you to step in and out of write as if you had manually updated the controls.

The snapshot recall will respond to Snapshot Isolate in the usual way.

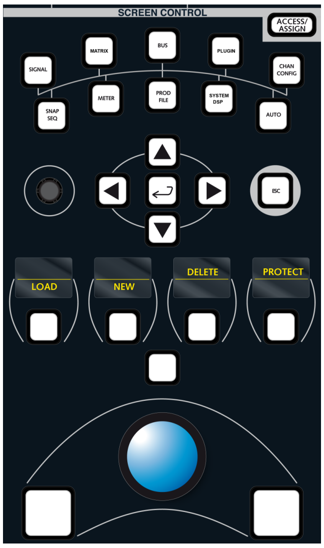
> Depending on the size of the mix, and the number of changes actioned by the snapshot, there may be a slight delay when recalling the snapshot.

The Mixes Display

Mixes are managed from the **Mixes** display.

1. Press the **AUTO** button, located on the SCREEN CONTROL panel, to view the **Mixes** display.

*The **Mixes** list shows all the mixes in memory:*



If the list is longer than the available window space, focus on the list and use the rotary scroller on the SCREEN CONTROL panel to navigate up and down the list, and/or use the on-screen scroll bars.

The name of the **Active Mix** is shown at the top of the display. It is the passes from this mix which appear in the Pass Tree, and its Play pass which loads when you enable automation.

The columns beside each mix name show the date and time stamp, the number of passes within the Pass Tree, whether the mix is protected (padlock icon) and the size of the mix. You may drag and drop columns to change their order.

At the bottom of the display, the **Mix Memo** box may be used to add notes to a particular mix.

The on-screen buttons and SCREEN CONTROL soft keys provide access to Load, New, Protect and Delete operations. These functions are also available if you select a mix and right-click.

Loading a Mix

Loading a mix recalls all the passes stored within the mix to the Pass Tree, including the Play pass. Therefore, if automation is enabled, you will see your automated parameters reset. This provides quick access to any mix stored in the **Mixes** list.

Note that when you load a mix, any passes created within the previous mix are held in temporary memory. This allows you to quickly change between mixes (and passes) without losing data.

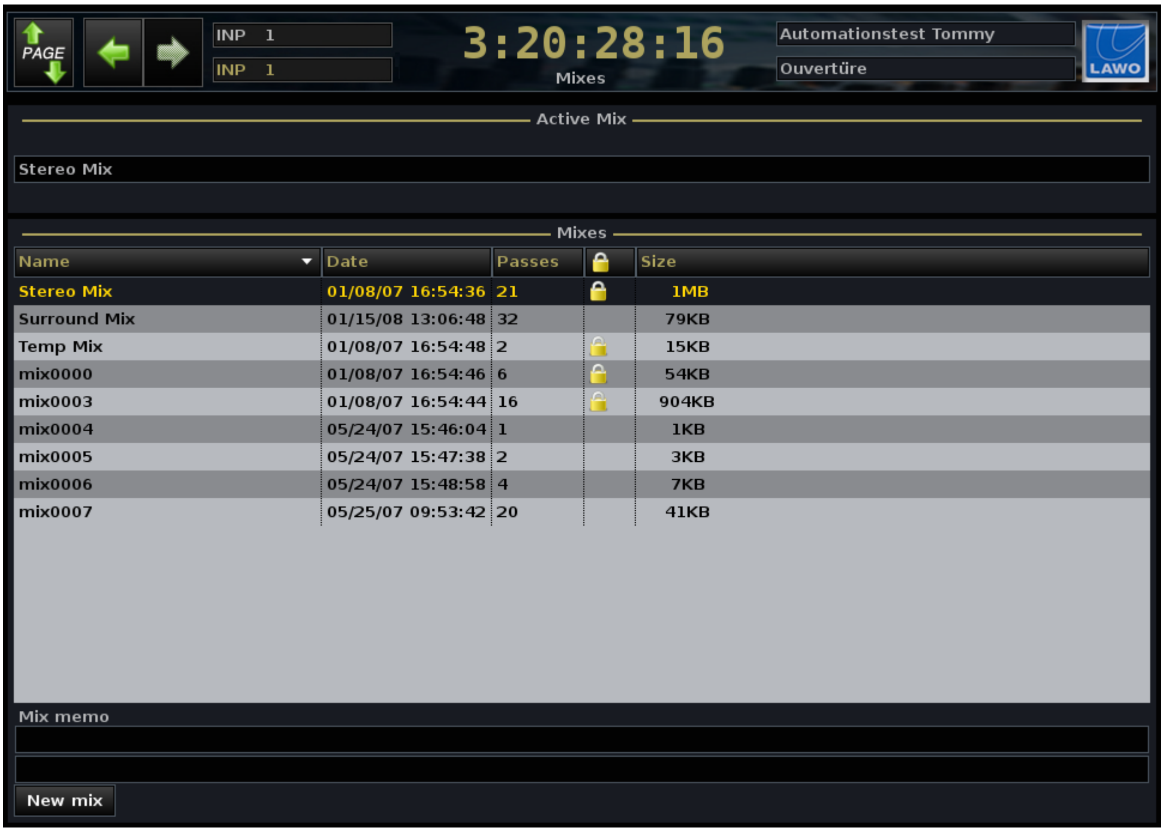
However, note that if you load a *different* production, the memory is cleared and a new set of mixes is loaded into the **Mixes** list.

Therefore, always update (or save) a production, before loading a different production, to safeguard your automation data.

To load a mix:

1. Select a mix from the **Mixes** list (e.g. **Stereo Mix**).
2. Right-click and select **Load**, or press the **LOAD** soft key, to complete the operation.

*The loaded mix becomes the **Active Mix** shown at the top of the display:*



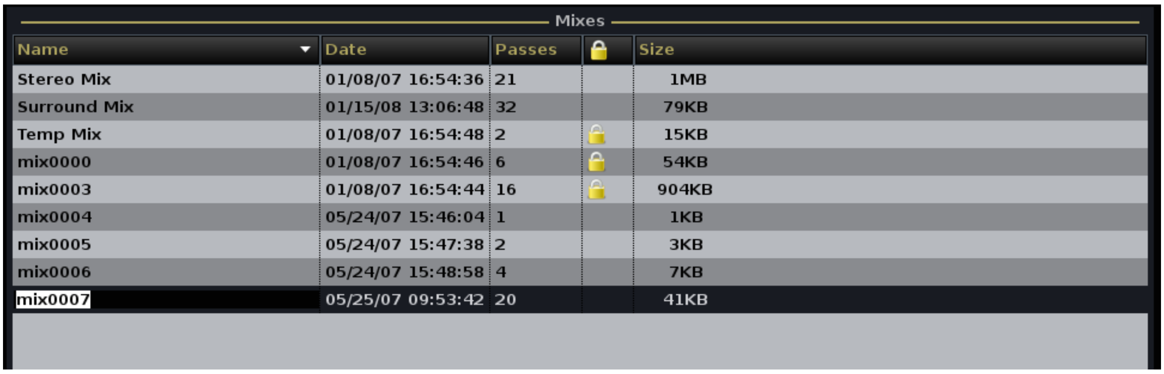
If automation is enabled, then you will see your automated parameters immediately reset (to the values stored in the Play pass).

3. Alternatively, enable automation, using the **AUTO ON** button (on VAP 1).
4. You can now play back and update your mix.

Note that the system stores the current Play pass for each mix. This means that when you load a different mix, you will always get back to the last pass you were working on.

Renaming a Mix

1. Click on the mix name using the trackball:



Mixes

Name	Date	Passes	🔒	Size
Stereo Mix	01/08/07 16:54:36	21		1MB
Surround Mix	01/15/08 13:06:48	32		79KB
Temp Mix	01/08/07 16:54:48	2	🔒	15KB
mix0000	01/08/07 16:54:46	6	🔒	54KB
mix0003	01/08/07 16:54:44	16	🔒	904KB
mix0004	05/24/07 15:46:04	1		1KB
mix0005	05/24/07 15:47:38	2		3KB
mix0006	05/24/07 15:48:58	4		7KB
mix0007	05/25/07 09:53:42	20		41KB

Click once to select all the existing text (white) or twice (black cursor) to modify the existing name.

2. Enter a new name from the keyboard.

3. When you have finished, press the Enter button, on the keyboard, to confirm the new name.

4. Or, if you make a mistake or want to exit without making any changes, press the **Esc** button on the keyboard.

Adding a Memo

You may use the two **Mix memo** lines to add memo information. For example, you may wish to remind yourself about the details of the mix.

1. Select the mix and then select a line in the **Mix memo** field.

A black cursor appears.

2. You may now type to enter your information from the console keyboard:

You can enter as many characters as you wish in each line; the list will automatically resize to fit.

If you cannot enter any memo text, check that the mix is not protected.

Right-click on a mix memo to **Copy** and **Paste** the text to another snapshot.

You can also drag and drop the **Memo** columns to change their position on the display.

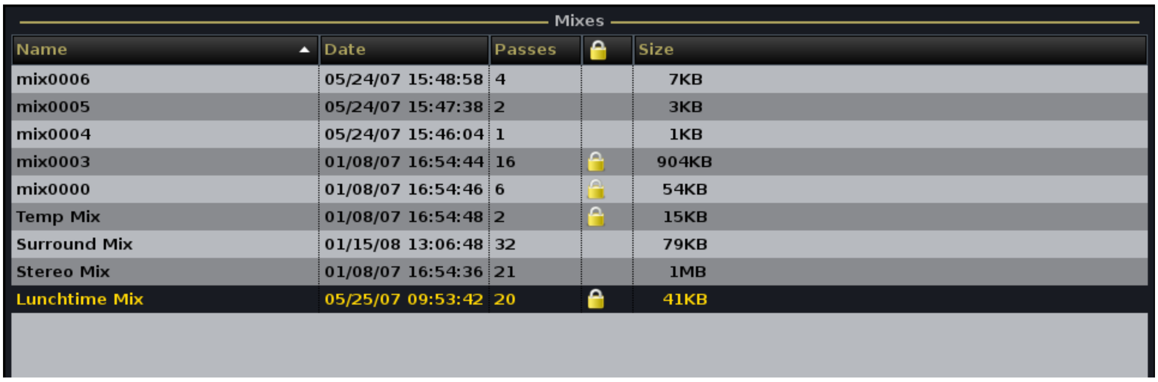
Protect & Delete

Protect

A protected mix cannot be deleted. And you *cannot* create a new Record pass within a protected mix. You can use this safeguard any important mixes which you do not want to accidentally overwrite or delete.

1. Select a mix from the **Mixes** list.
2. Right-click and select **Protect**, or press the **PROTECT** soft key.

The padlock icon indicates that the mix is now protected:



Delete

Delete removes the mix from the internal memory.

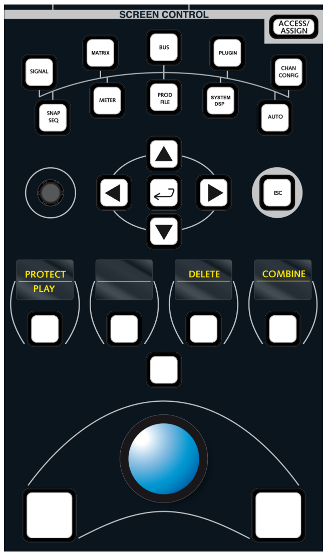
1. Select a mix from the **Mixes** list.
2. Right-click and select **Delete**, or press the **DELETE** soft key, to complete the operation.

Note that you cannot delete a protected mix.

The Passes Display

Passes are managed from the **Passes** display.

1. Press the **AUTO** button, located on the SCREEN CONTROL panel, to view the **Passes** display:



At the top of the display are various fields for:

- **Play**, **Record and Selected pass** - display the current Play and Record passes (for writing new automation) and Selected pass (for editing automation). The **Locate** buttons automatically reveal the pass, in the **Pass Tree**, if it has been hidden by closing a branch.
- **Punch-in** and **Punch-out** times - define the punch-in/out timecode window.
- **Glide-in** and **Glide-out** times - define the glide times.
- **Pre-roll window** - defines the pre-roll tolerance for machines which pre-roll when going into Play.
- **Midnight** - defines the timecode offset.

The **Pass Tree** displays all the finished passes in memory.

Each time you revert to an earlier pass and then make updates, you start a new branch within the tree. Click on the arrows beside each branch to open or close. Or, click on **Expand all** at the bottom of the display to open up all branches of the **Pass tree**.

To avoid mixes become too large, a maximum of 10 passes are stored within each branch of the **Pass tree**; after the tenth pass, the first pass is deleted to make space for new data, and so on. To keep a specific pass indefinitely you should protect it; it will then be retained as one of the 10 passes with the branch.

The columns beside each pass name show its date and time stamp and whether the pass is protected (padlock icon). You may also rename or delete a pass; the **Status** field marks the current Play and Record pass with icons.

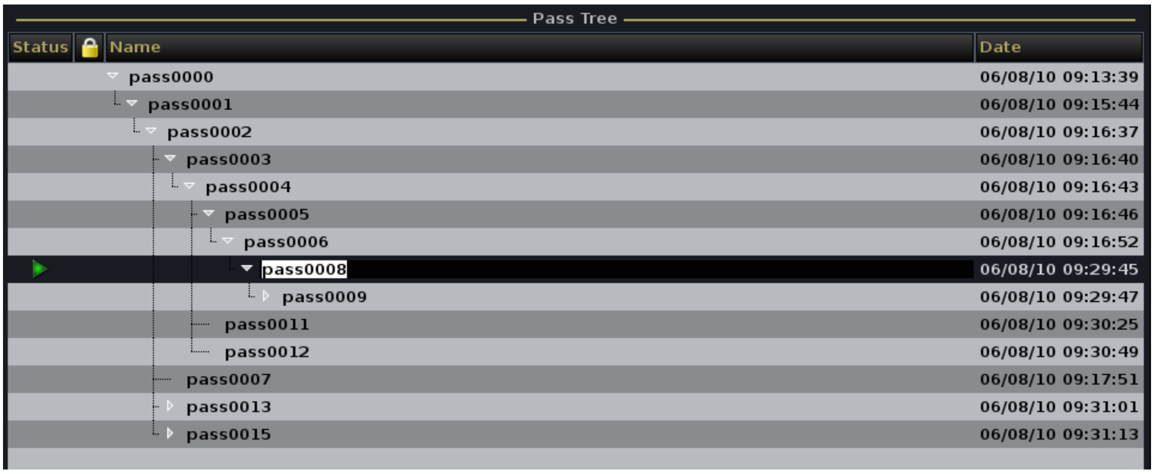
All passes are stored inside the **Active Mix** when you update or save a production. Or, if you load a different mix or production, then the **Pass Tree** updates accordingly.

You can use the **Pass Tree** to change the Play pass at a any time. For example, to A/B between two different versions of automation for a chorus or scene. See Reverting to an Earlier Pass.

You can also edit mix passes in order to combine, delete, copy, shift, insert or paste sections of automation data.

Renaming a Pass

1. Click on the pass name using the trackball:



Click once to select all the existing text (white) or twice (black cursor) to modify the existing name.

2. Enter a new name from the keyboard.

3. When you have finished, press the Enter button, on the keyboard, to confirm the new name.

4. Or, if you make a mistake or want to exit without making any changes, press the **Esc** button on the keyboard.

Protect & Delete

Protect

A protected pass cannot be deleted manually, or automatically by the system (when it reaches the 10 pass per branch limit).

1. Select a pass from the **Pass Tree**.
2. Right-click and select **Protect**, or press the **PROTECT** soft key.

A padlock icon indicates that the pass is now protected.

Delete

Delete removes the pass from the internal memory.

1. Select a pass from the **Pass tree**.
2. Right-click and select **Delete**, or press the **DELETE** soft key, to complete the operation.

Note that you cannot delete a protected pass or the current **Play pass**.

Setting an Offset Timecode

The **Midnight** field at the top of the **Passes** display is used to offset the internal timecode of the automation system. For example, if your mix starts at an odd timecode value and you wish to view it as starting at 00:00:00:00.

1. First, set the Central GUI headline to show **Timecode display** and **Offset Timecode**. You can do this by clicking on the headline, or selecting the Time Display option from the **System Settings** display:

2. Now click in the **Midnight** field, at the top of the **Passes** display, to enter the offset timecode:

3. Use the console keyboard to enter the timecode which you wish to correspond to midnight (00:00:00:00).

For example, you could locate to the beginning of your mix (e.g. **21:00:20:15**) and enter this value as midnight.

4. Press Enter and you will see the start of your mix as 00:00:00:00 in the Central GUI time display.
5. To clear an offset, click in the **Midnight** field and enter **00:00:00:00**.

Setting Pre-roll

The **Pre-roll** window at the top of the **Passes** display is used to set a pre-roll tolerance time for machines which pre-roll slightly when going into Play – for example, a tape machine.

Any small rewind in timecode causes the automation system to finish the pass. Often this is undesirable, as it prevents you from putting controls into write while in Stop, and then writing these values forwards on entering Play.

To avoid this problem:

1. Use the console keyboard to enter a value in ms into the **Pre-roll window** – for example, 50ms.

   The automation system now requires a rewind of more than 50ms to finish a pass, and therefore tolerates the machine's pre-roll when entering Play.

2. Test your entry by putting some controls into write while in Stop and pressing Play.

   The controls should remain in write when you go into Play.

If not, adjust the **Pre-roll window** to a longer time accordingly.

Note that the **Pre-roll window** affects how a pass can be finished; you must rewind by more than the **Pre-roll window** time in order to finish a pass.

Glide Time

The **Glide-in time** and **Glide-out time** fields, at the top of the **Passes** display, are used in conjunction with a number of functions:

- **Glide-in time** - used when performing mix pass edits.
- **Glide-out time** - used when performing mix pass edits, and when stepping out of automation using GLIDE.

To adjust the glide times:

1. Either click within the **Glide-in/out time** box and type in a value from the console keyboard.
2. Or use the up/down arrows to adjust the time in 100 ms steps.

Both values may be adjusted from 0 to 60000ms (60 seconds).

Mix Pass Editing

The **Passes** display provides a number of functions for mix pass editing including combine, delete, copy, shift, insert and paste.

The Combine function combines the automation data from the **Selected pass** into the **Play pass**, while all the other Edit operations are applied to the current **Play pass**.

Combine

Combines the automation data from the **Selected pass** into the **Play pass**, between the **Punch in** and **Punch out** times.

1. First set the punch in and out times.

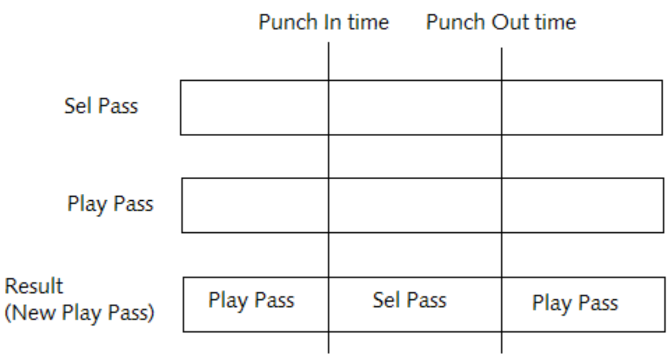
2. Make the pass you wish to add data into the current **Play pass** - select it and press the **PLAY** soft key, or right-click and select **Play**.

3. Then select the pass you wish to combine from.

*In our example, we are combining from **Chorus 2 Better Vocals** into **Good Vocals**:*

4. Press the **COMBINE** soft key, or right-click and select **Combine with Play pass**.

A new pass is created as shown below:



Edit Operations

All other edits are applied to the current **Play Pass** and are performed as follows:

1. Make the pass you wish to edit the current **Play pass** - select it and press the **PLAY** soft key, or right-click and select **Play**.
2. Then, right-click on the **Play Pass** and select either **Edit** -> **Pass** or **Edit** -> **Access**:

- **Edit** -> **Pass** – edits the complete mix pass (all channels).
- **Edit** -> **Access** – edits only the channel in access. This option will leave automation data on other channels intact.

*A range of sub operations are revealed (the same options are available for **Edit** -> **Pass** and **Edit** -> **Access**):*

If any of the options are greyed out, then check the following:

- To perform an edit, timecode automation must be enabled so make sure **AUTO ON** is selected.
- If a **Record Pass** is active, then you cannot perform an edit. Finish the pass and then select the edit.
- Most edits require a valid timecode “window” which is defined by the **Punch in** and **Punch out** times at the top of the **Passes** display. The “window” must be greater than zero for **Delete**, **Cut**, **Copy**, **Clear** and **Shift**. See Punch in and out times.

3. Select an operation from the drop-down menu – for example, **Delete**.

*The edit is performed and a new **Play Pass** is created.*

4. To undo the edit, revert to the previous Play Pass.

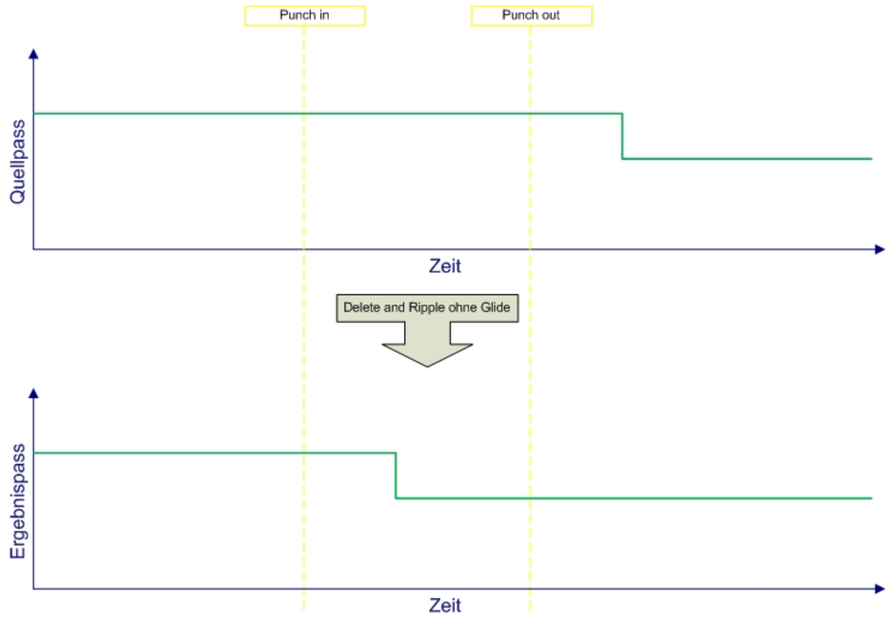
You may apply glide times to each edit by entering a value in the Glide-in and Glide-out time fields; the result is specific to each edit. But take care to avoid glide times longer than the **Punch in** to **Punch out** timecode window, otherwise you may experience some strange results!

Note that it is possible to copy data from a complete mix pass (via **Edit** -> **Pass**) and insert or paste it into the channel in access (via **Edit** -> **Access**). However, the reverse is not possible.

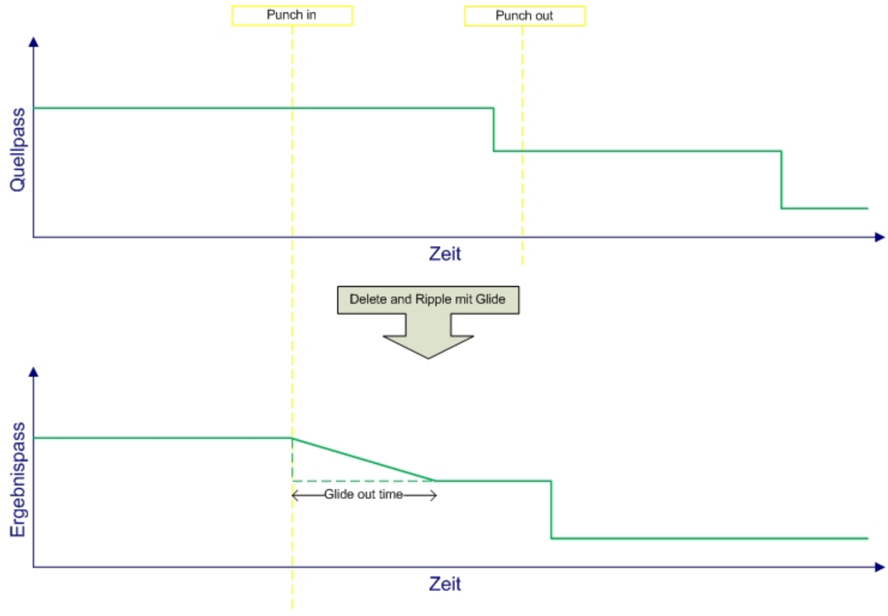
Delete

This edit performs a "delete and ripple". You might use it to remove the automation for a section of the mix. For example, if a chorus has been deleted from the song.

Automation between the **Punch in** and **Punch out** timecode values is deleted, and all data after the **Punch out** time ripples up to the **Punch in** time:



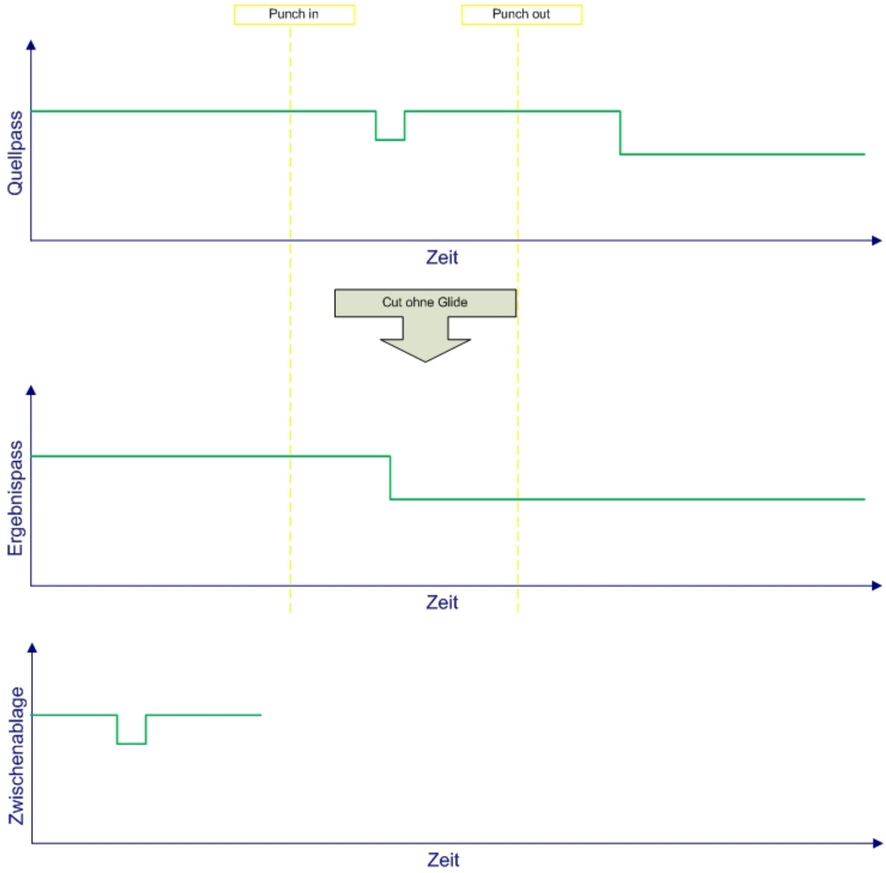
The **Glide-out time** is applied at the Punch in point; **Glide-in time** has no affect on this edit:



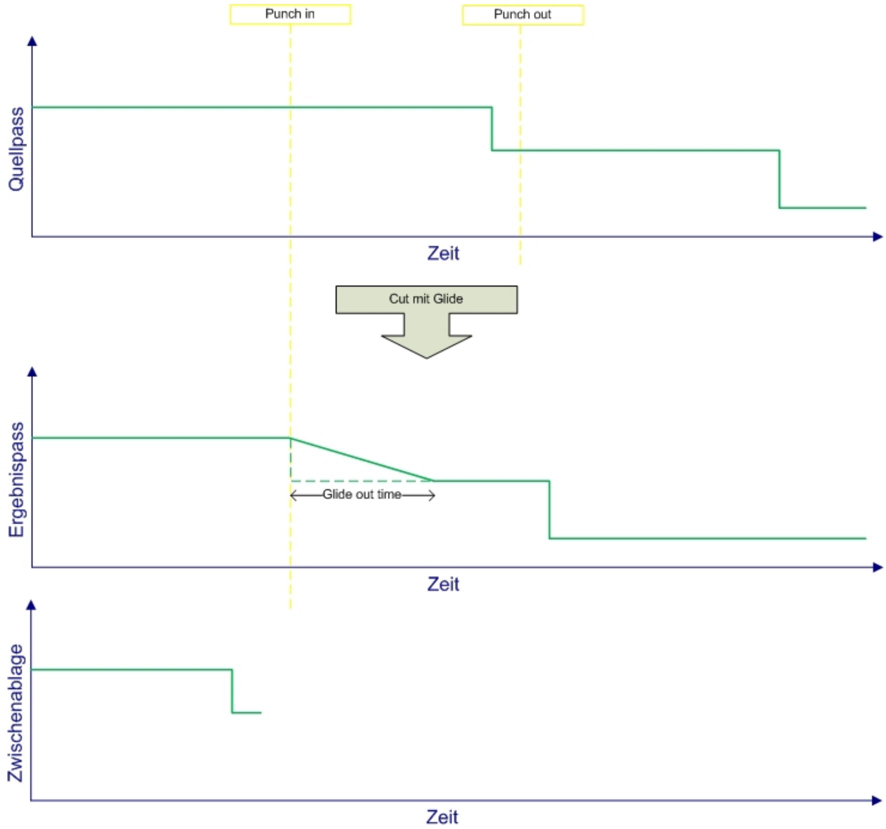
Cut

This edit cuts out a section of the pass and copies it to the clipboard. Its affect on the current **Play Pass** is identical to a Delete. However, you would use this edit if you wish to paste or insert the clipboard data to another location. For example, to move the position of a chorus in the song.

Automation between the **Punch in** and **Punch out** timecode values is deleted and copied to the clipboard (Zwischenablage); all data after the **Punch out** time ripples up to the **Punch in** time:



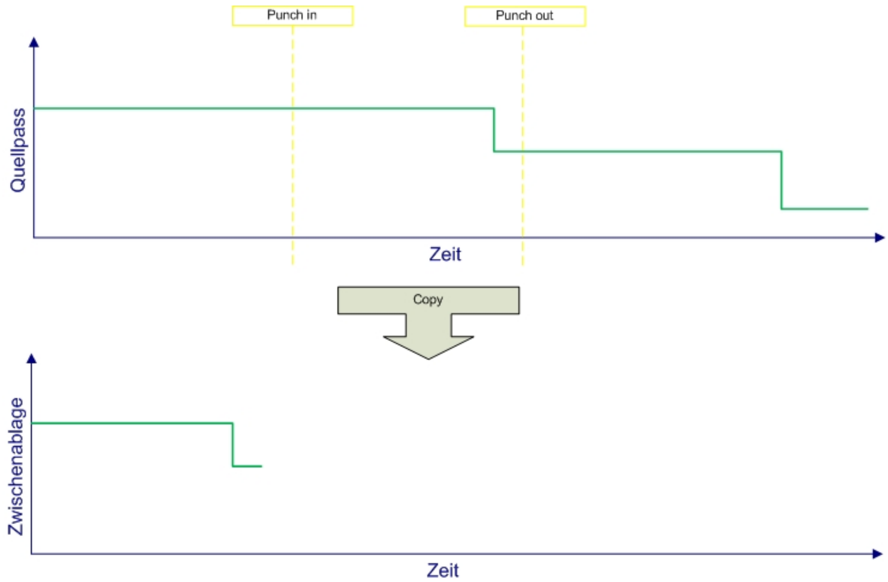
The affect of **Glide-in** and **Glide-out** times is identical to a Delete.



Copy

This is a non-destructive edit which copies a section of the mix to the clipboard. It has no affect on the **Play Pass**. You would use this edit if you wish to paste or insert the clipboard data to another location. For example, to copy a chorus to another location.

Automation between the **Punch in** and **Punch out** timecode values is copied to the clipboard (Zwischenablage):

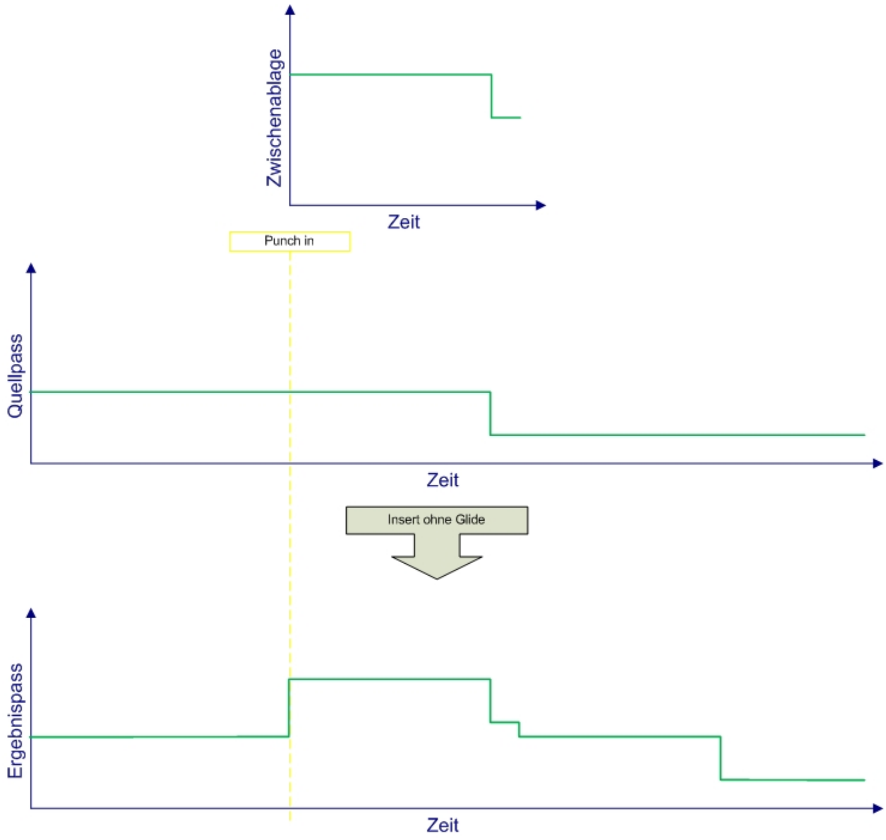


The **Glide-in** and **Glide-out times** have no affect on this edit.

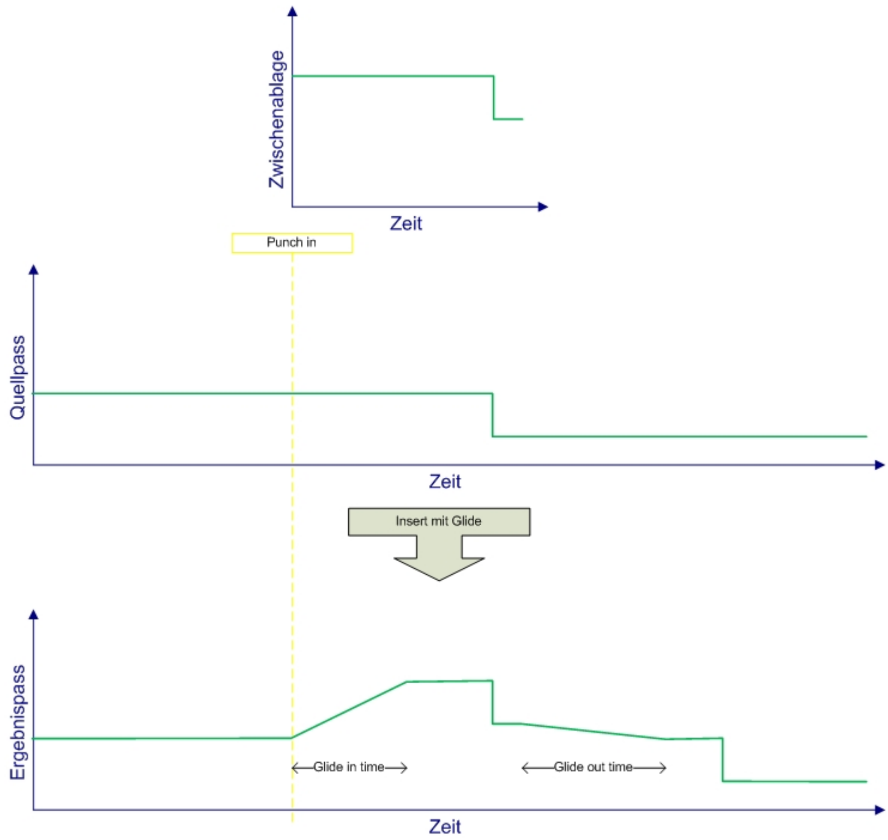
Insert

Having cut or copied data to the automation clipboard, it can be inserted into the **Play Pass**. You might use this edit to insert automation for a Chorus when you wish to keep the existing structure of the song intact. In other words, the song gets longer by one Chorus!

This edit inserts the clipboard data at the **Punch in** time. It is different to a Paste in that the existing **Play Pass** automation ripples down and is tagged onto the end of the insert. Note that the **Punch out** time has no affect on this edit.



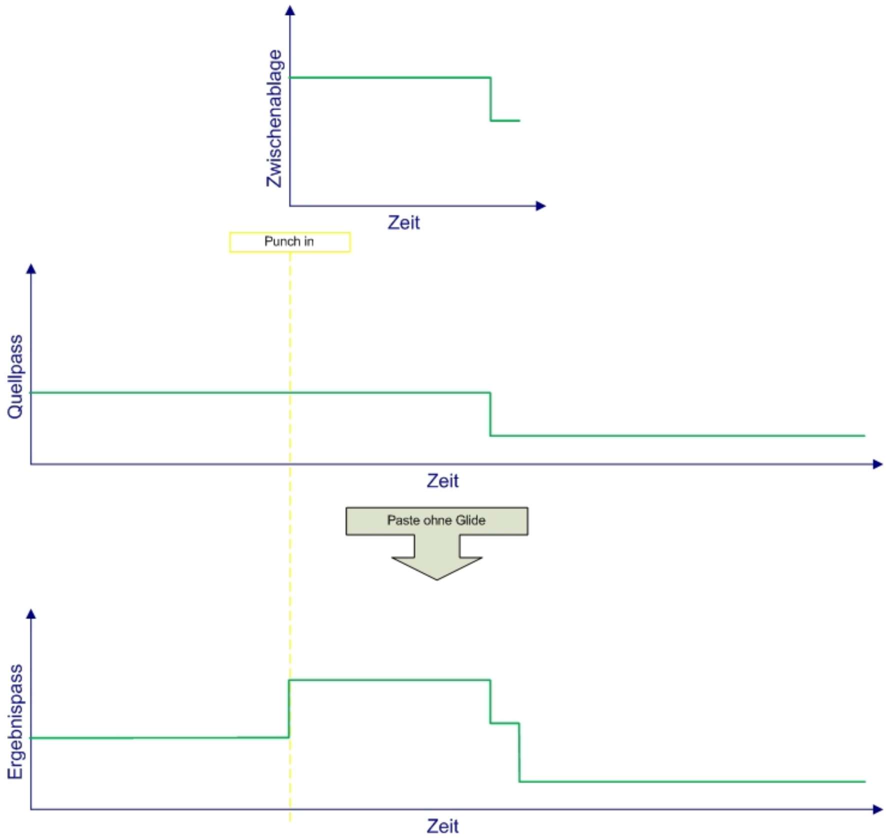
The **Glide-in time** is applied at the **Punch in** point; the **Glide-out time** is applied at the end of inserted clipboard:



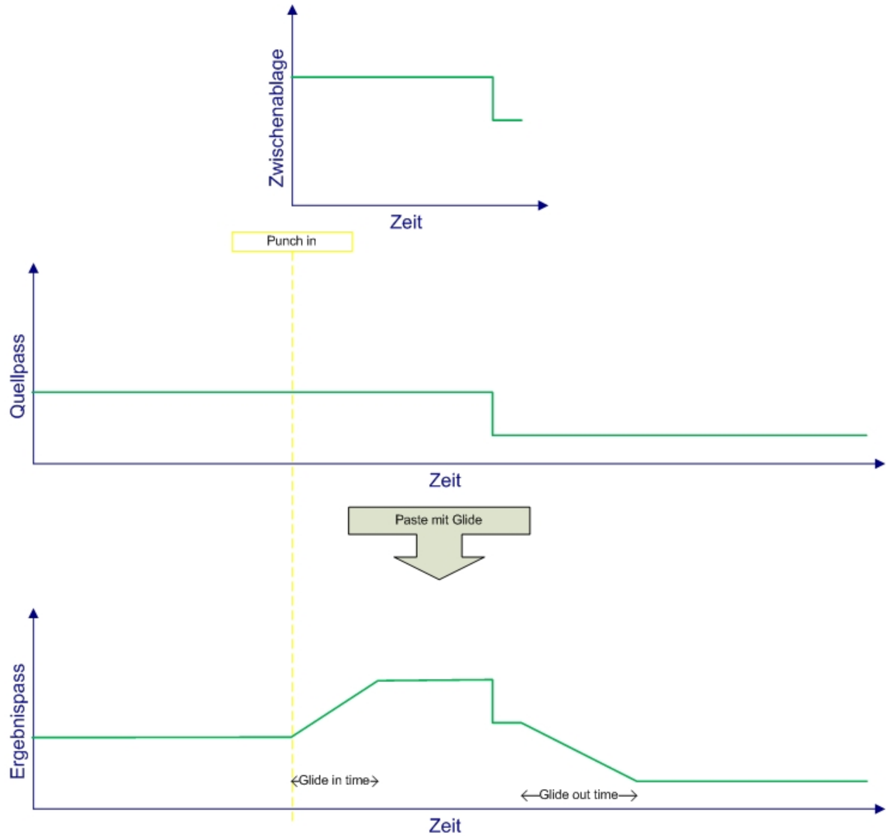
Paste

Having cut or copied data to the automation clipboard, it can be pasted into the **Play Pass**. You might use this edit to replace the automation for a Chorus with a newer pass. In other words, the clipboard replaces the existing Play Pass.

This edit pastes the clipboard data at the **Punch in** time. It is different to an Insert in that the existing **Play Pass** automation is replaced. Note that the **Punch out** time has no affect on this edit.



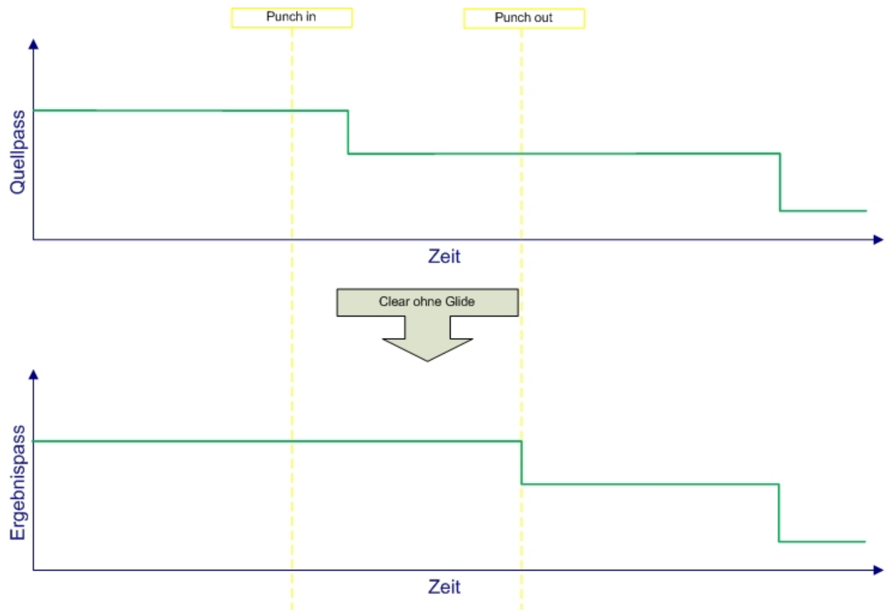
The **Glide-in time** is applied at the **Punch in** point; the **Glide-out time** is applied at the end of pasted clipboard:



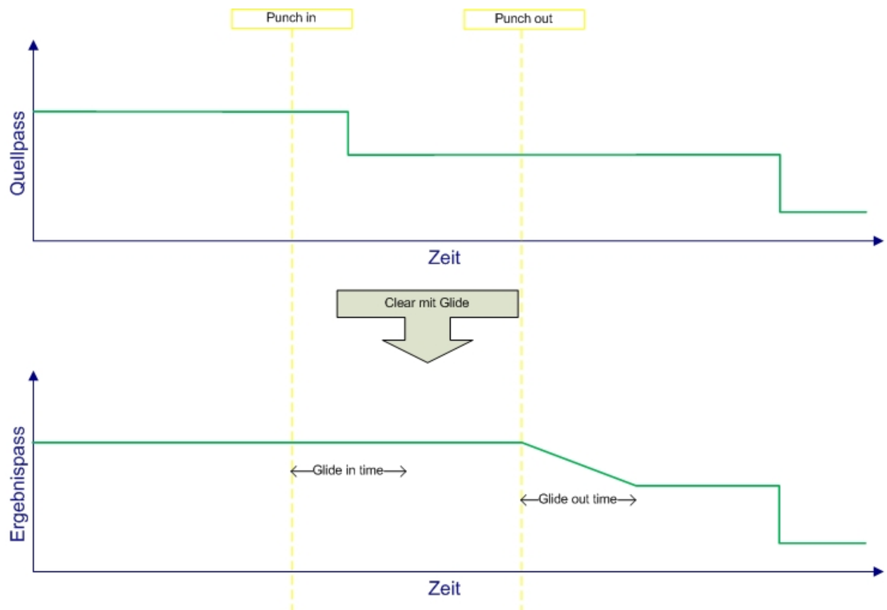
Clear

This edit clears a section of the mix. You might use it if the order of a song changes and you want to write new automation data in the cleared section. It is different to a Shift in that the existing **Play Pass** automation is replaced.

Automation data between the **Punch in** and **Punch out** times is cleared by extending the values from the **Punch in** time through to the **Punch out** time. The overall length of the mix remains intact:



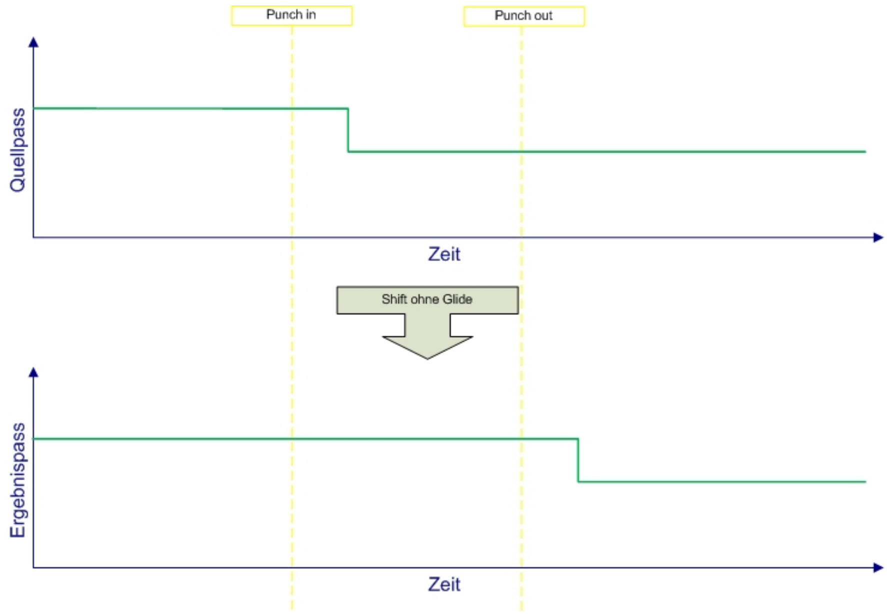
The **Glide-in time** is applied at the **Punch in** point; the **Glide-out time** is applied at the **Punch in** point:



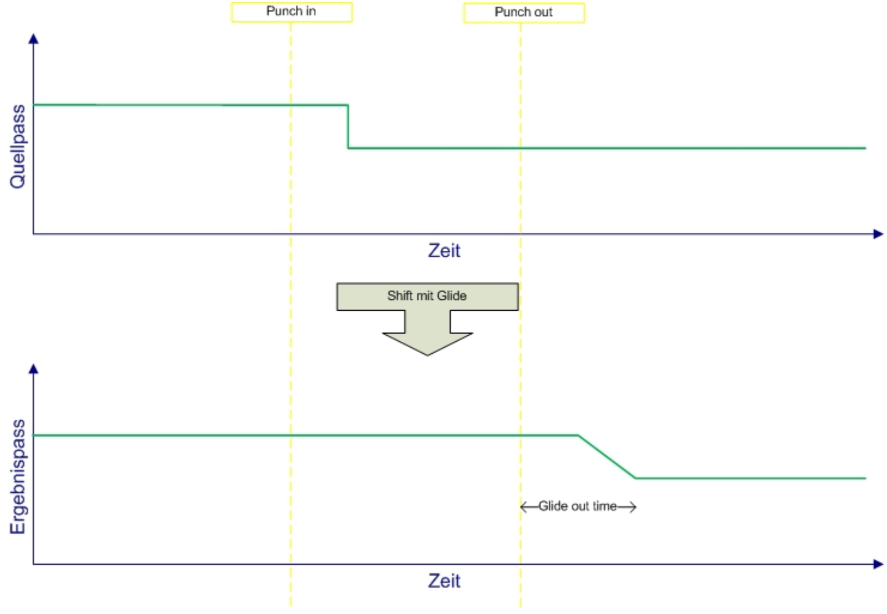
Shift

This edit shifts or moves a section of the mix. You might use it if you want to keep all your existing automation but write new data for an Instrumental which has been added to the song.

Automation data between the **Punch in** and **Punch out** times is shifted to the **Punch Out** time. Values at the **Punch in** time are then extended to the **Punch out** time to fill in the cleared section:



The **Glide-out time** is applied at the Punch in point; **Glide-in time** has no affect on this edit:



Advanced Editing Options

When performing mix pass edits, *only* the controls selected and armed for automation are copied, pasted, inserted, etc. This allows you to copy data for all channels and controls, and then selectively insert, paste, etc.

For example, to copy and paste just the automation data for the vocal channels during a Chorus:

1. Copy the Chorus data from the **Play Pass**.

Automation data for all channels and controls in replay is copied to the clipboard.

2. *BEFORE* performing the **Paste**, deselect any channels or controls which you do not want to include in the paste by disarming them – in our example, disarm all channels except the main and backing vocals.

3. Now perform the Paste.

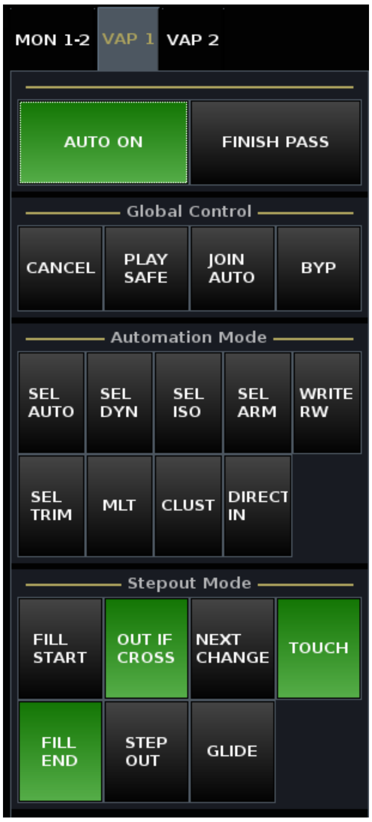
*The **Paste** is applied only to the armed channels.*

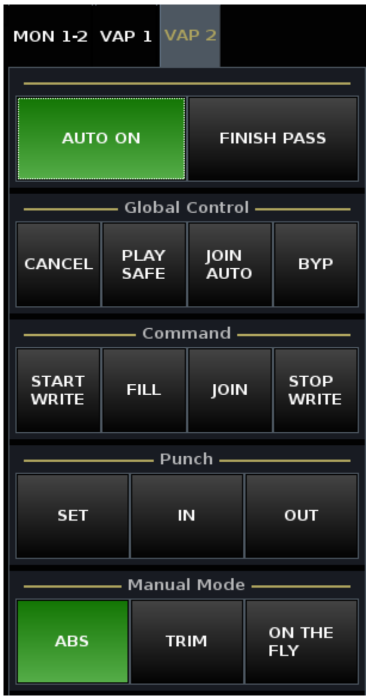
4. Then re-enable arm for all channels (back to replay).

5. Press Play to play back your edited mix pass!

VAP Summary

The Virtual Automation Panels (**VAP 1** and **VAP 2)** provide access to the following functions. Both panels are available from the touch-screen button area on the right of the Central GUI:





Global Control (VAP 1 and 2)

- AUTO ON – turns the automation system on or off. When on, automation replays from the **Play pass** within the **Active Mix**.
- FINISH PASS – finishes the pass without having to stop and rewind.
- CANCEL – cancels the Record pass. Use this button to discard moves you have just written.
- PLAY SAFE – parameters read automation data from the **Play pass** but cannot write new data. In addition, if you adjust a parameter value, you will *NOT* hear any change in the audio.
- JOIN AUTO – use this button if you are going to review and update a section of the mix.
- BYP – identical to 'Play Safe'; parameters read automation data from the **Play pass** but cannot write new data. However, if you adjust a parameter, you *WILL* hear the change in the audio.

Automation Mode (VAP 1)

- SEL AUTO – selects modules to be enabled or disabled for automation, using the Central Control Section **SEL** buttons.
- SEL DYN – selects modules to write in dynamic or static automation mode, using the Central Control Section **SEL** buttons.
- **SEL ISO** – reserved for future implementation.
- SEL ARM – selects modules to be armed (read & write) or disarmed (read only), using the Central Control Section **SEL** buttons.
- WRITE R/W – turns the fader strip **R/W** user buttons and Central Control Section **SEL** buttons into step in and step out of write controls.
- SEL TRIM – selects modules to write in absolute or trim mode, using the Central Control Section **SEL** buttons. Selections are cleared by pressing either the **ABS** or **TRIM** 'Manual Mode' button.
- MLT – used with **SEL AUTO**, **SEL DYN** and **SEL TRIM** to apply selections across multiple channels.
- CLUST – allows you step in and out of write on a cluster of channels.
- DIRECT IN – allows you to step a control into write when running in Bypass.

Stepout Mode (VAP 1)

These modes define what happens when you step out of write:

- FILL START – selects the fill to start automation mode.
- OUT IF CROSS – selects the out if cross automation mode.
- NEXT CHANGE – selects the next change automation mode.
- TOUCH – when enabled, faders and variable controls will automatically step out of write on release.
- FILL END– selects the fill to end automation mode.
- STEP OUT – selects the step out automation mode.
- GLIDE – when enabled, variable parameters will glide back to the **Play pass**. The glide time can be set from 0 to 60,000 ms.

Command (VAP 2)

- START WRITE – press this button to step all parameters into write.
- FILL (Fill Region) – use this button to write any parameter values in write between the **Punch In** and **Punch out** times.
- JOIN – use this button if you are going to be reviewing and updating a section of the mix.
- STOP WRITE – press this button to step all parameters out of write.

Punch (VAP 2)

Used to set the punch in and out times, or activate automatic punch in/out.

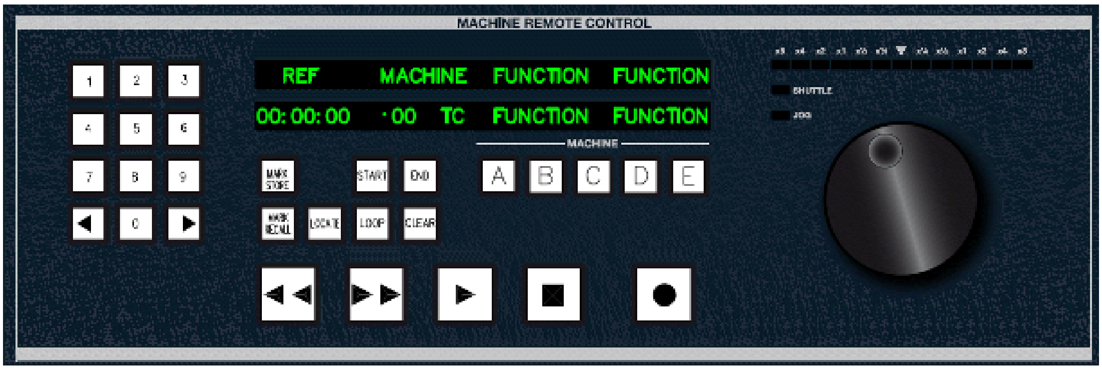
- **SET** – press to set the punch in or out times.
- **IN** – press to enable automatic step into write at the punch in time. (If **SET** is active, press to set the punch in time.)
- **OUT** – press to enable automatic step out of write at the punch out time. (If **SET** is active, press to set the punch out time.)

Manual Mode (VAP 2)

- ABS – selects Absolute automation mode.
- TRIM – selects Trim automation mode..
- ON THE FLY – selects Trim relative (unlit) or Trim on the fly (lit).
- FINISH PASS – press to finish a mix pass manually.

Machine Control

Control of the playback machine may be programmed onto user buttons from the Custom Functions display, or handled from the optional MACHINE REMOTE CONTROL user panel:



The panel provides remote control of one of three external machines. Sony 9pin (A) and Midi Machine Control (C) ports are supported; one port can be active at a time. When active, the console's automation system slaves to timecode from the active port.

> Your system must be specified with the Recording Com Kit (958/80) to provide Sony 9pin, LTC and/or MIDI connections to an external playback device. Please consult your system specification for details.

Transport Control

1. To control one of the three machines, select a port enable button – **A**, **B** or **C** – and use the RW, PLAY, FW and STOP transport controls.

   *The first line of the display shows the active port (**A** to **C**) and the type of machine (e.g. **DA-88**). The second line shows the current timecode position of the machine (on the left), and timecode entered in the temporary buffer (on the right).*

The temporary buffer is used when storing and recalling marks or setting up a loop.

If there is no connection between the console and the machine, then the display shows **NO MACHINE**. If there is no tape in the machine, then the display shows **NO TAPE**.

2. To change the jog wheel between jog and shuttle modes, press down on the jog wheel.
3. To punch in and out of record while a machine is in play:

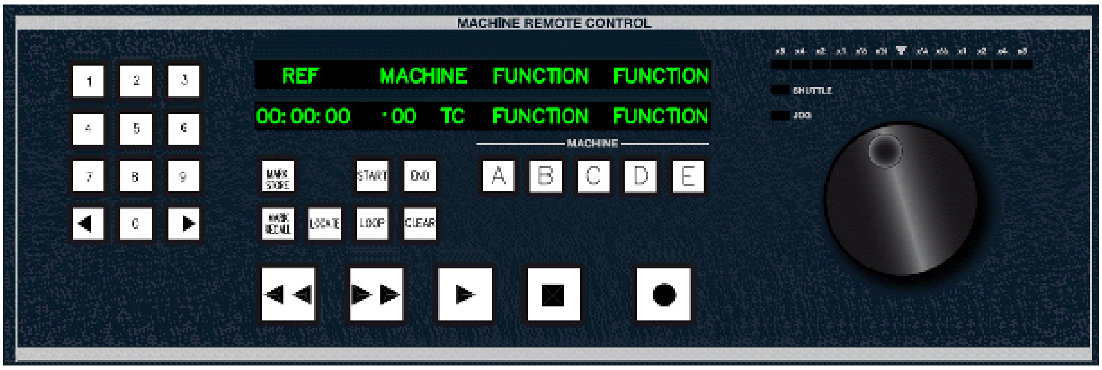
- Press RECORD to punch in (while in play).
- Press PLAY to punch out (while in record).

Locating

You can locate the machine to a particular timecode either by manually typing in a timecode position, or recalling a stored mark.

To manually locate to a timecode:

1. Type the timecode position using the **0** to **9** buttons:



The timecode should be entered in the following format:

HH:MM:SS:FF

You *must* enter *all* fields, including frames, for the timecode value to be accepted.

The timecode appears in the temporary buffer on the right of the display.

You can use the left and right arrow buttons to navigate through the timecode characters. If you make a mistake, use the **CLEAR** button as follows:

- A short press (for less than 3 seconds) will delete one character.
- A long press (for more than 3 seconds) will delete the entire timecode value.

2. Now press **LOCATE**.

The machine locates to the temporary buffer timecode position; once the locate point has been reached, the machine goes into Play.

If you make a mistake and want to stop the machine locating, press **CLEAR**.

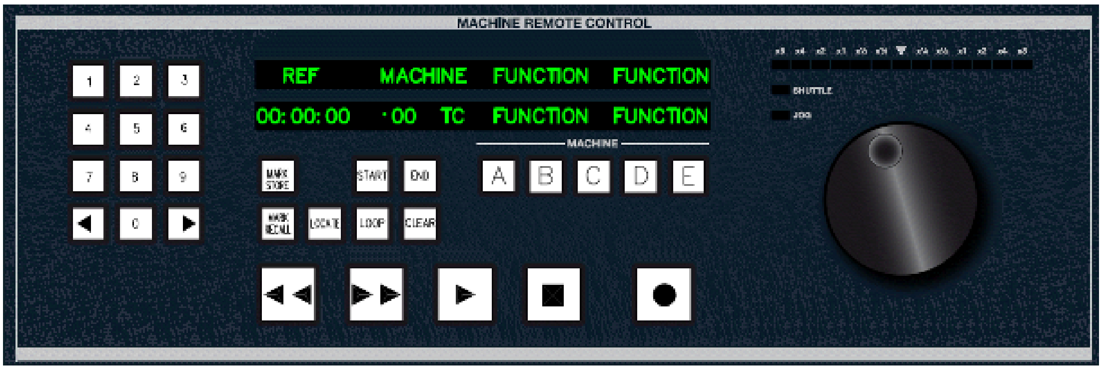
Storing and Recalling Marks

Marks can be used to store and recall up to 10 timecode positions so that you may use them as locate points.

The 10 mark buttons can also be assigned to locators from the Machine Locators display; this display provides an unlimited number of locators and a memo field to name each locate point.

➢ **To store a mark:**

1. Press the **STORE** button:



*The current timecode position is stored into the temporary buffer and the buttons – **0** to **9** – start to flash in green.*

Note that any buttons which are not flashing and are red already have a timecode stored.

2. Press one of the **0** to **9** buttons to select a location.

The timecode from the temporary buffer is stored into the selected location.

Alternatively, to store a particular timecode, for example, 01:00:00:

1. Press the **STORE** button.
2. Type in the timecode position using the **0** to **9** buttons:

The timecode in the temporary buffer updates.

3. Then press one of the **0** to **9** buttons to select a location.

➢ **To recall a mark:**

1. Press the **MARK RECALL** button.

*Any buttons – **0** to **9** – which contain a mark start to flash.*

2. Press the mark you wish to recall - **0** to **9.**

The stored timecode is recalled into the temporary buffer. It may now be used with the LOCATE or START/END functions.

➢ **To clear a mark so that the memory becomes inactive:**

1. Press the **CLEAR** button.

*Any buttons – **0** to **9** – which contain a mark start to flash.*

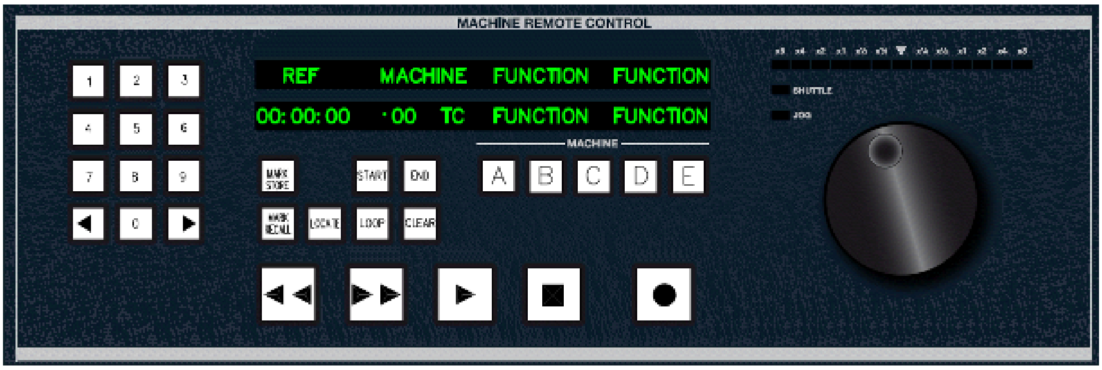
2. Press the mark or marks you wish to clear - **0** to **9.**

Setting Up a Loop (Cycle)

You can set up a loop so that the machine will cycle between a start and end timecode.

1. First, enter the timecode you wish to use as the start point into the temporary buffer.

You can do this by typing in a timecode position, recalling a mark or by pressing the **STORE** button to enter the current timecode position.



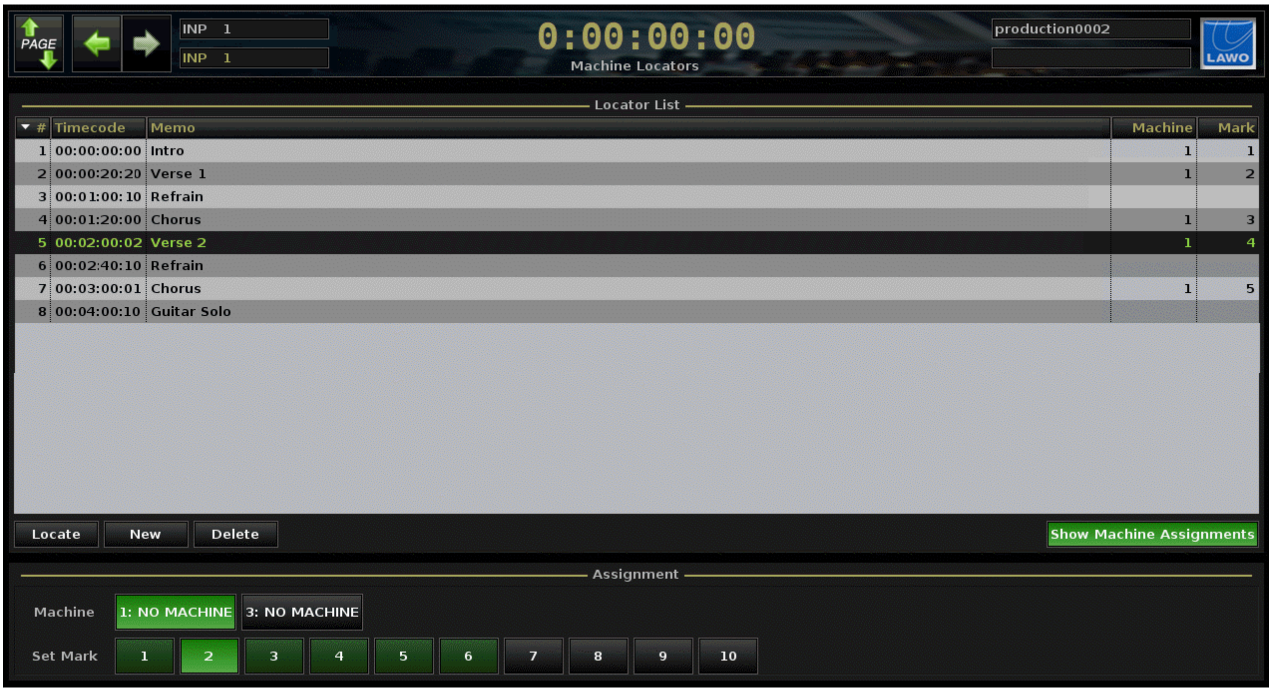
2. Press **START** to store temporary buffer timecode as the start point for the loop.
3. Repeat steps 1 and 2 but this time press **END** to store the loop end point.
4. Press **LOOP** to activate the loop.

*The machine will rewind to the **START** timecode, go into Play and when it reaches the **END** timecode repeat the loop.*

5. Turn off **LOOP** to stop the cycle.

Machine Locators (Cue List)

1. From V4.24 software onwards, press the **AUTO** button, located on the SCREEN CONTROL panel, to view the **Machine Locators** display:



This display provides a cue list for storing and recalling timecode positions.

You may store an unlimited number of locators, each with its own **Timecode** stamp and **Memo** field.

If your console is fitted with the optional Machine Control user panel, then the 10 mark buttons on this panel can be assigned to locators - the assignment is indicated in the **Machine** (machine number) and **Mark** (button number) fields, and in the **Show Machine Assignments** area at the bottom of the display. See Assigning Locators to the MRC panel.

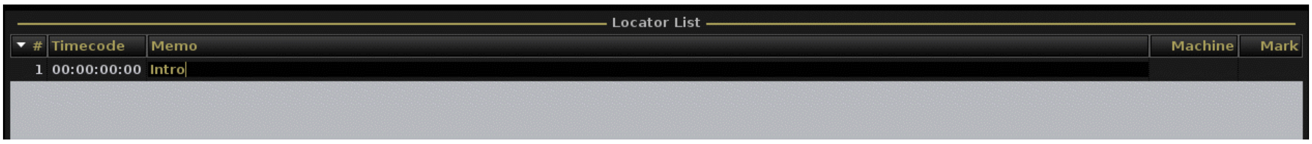
Storing and Naming Locators

1. Select the on-screen **New** button, or press the **Create New Locator** user button (programmed from the Custom Functions display). You can store locators while timecode is running at any speed: in Stop, Play, Fast forward, etc.

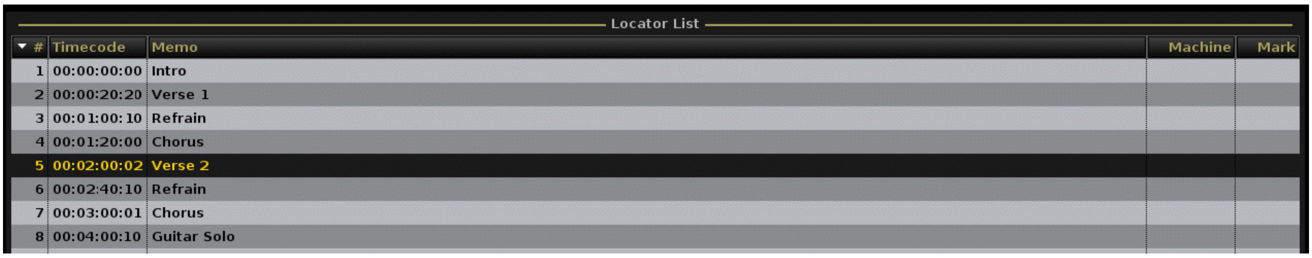
   *The current timecode position is saved into the next available locator ID - in our example, ID **1**.*

2. Type into the **Memo** field (up to 256 characters) to name the locator and press Enter.

   Note that the **Memo** field is automatically active after selecting **New**. This allows you to immediately type your text entry:



3. Repeat these steps to store more locators - for example:

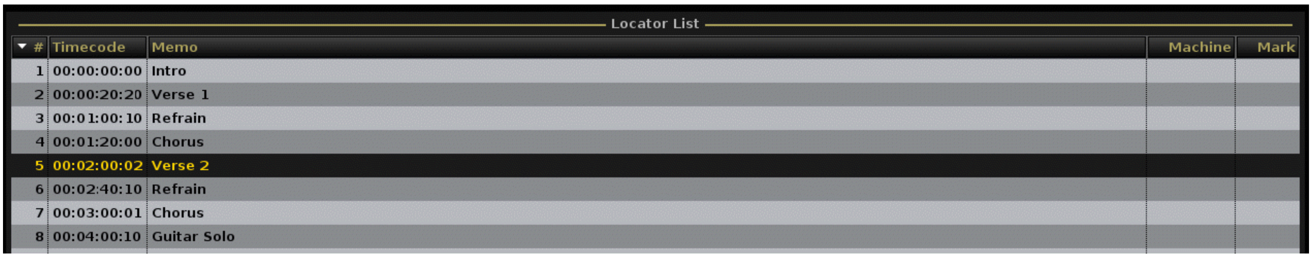


The first 10 locators (IDs 1 to 10) can be recalled from the **Goto Locator** user buttons programmed from the Custom Functions display.

Recalling a Locator

1. Double-click on an entry from the list, or select an entry and then click the on-screen **Locate** button.

The stored timecode position is recalled - e.g. ***Verse 2****:*

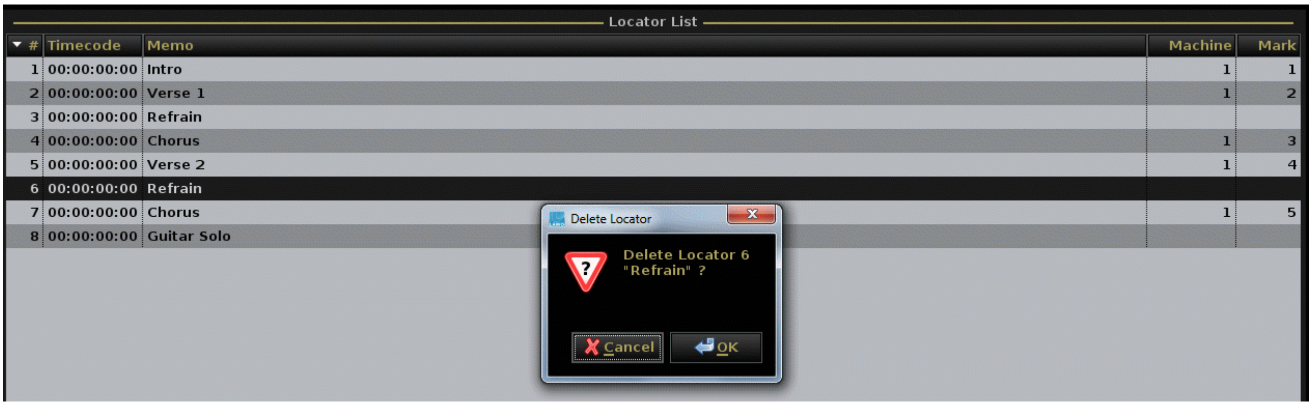

Locator List

#	Timecode	Memo	Machine	Mark
1	00:00:00:00	Intro		
2	00:00:20:20	Verse 1		
3	00:01:00:10	Refrain		
4	00:01:20:00	Chorus		
5	00:02:00:02	Verse 2		
6	00:02:40:10	Refrain		
7	00:03:00:01	Chorus		
8	00:04:00:10	Guitar Solo		

You can program up to 10 **Goto Locator** user buttons from the Custom Functions display to recall locators from a single button press.

Deleting a Locator

1. Select an entry and then click the on-screen **Delete** button.
2. Confirm the delete by selecting **Yes**:

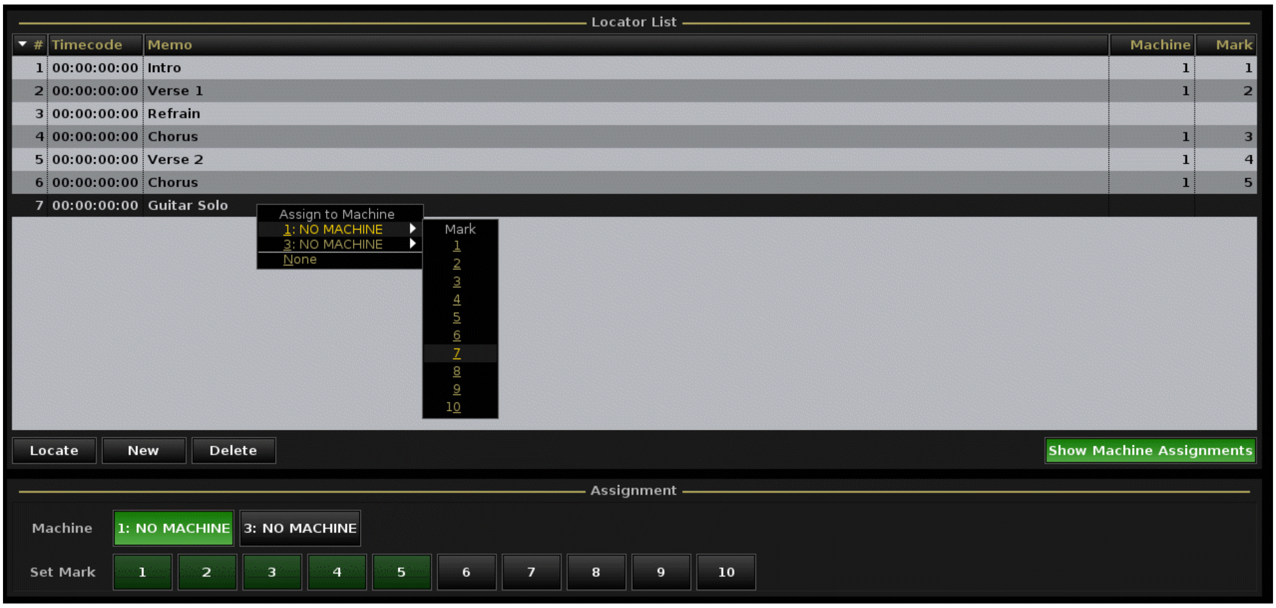


The locator is deleted from the list.

Assigning Locators to the MRC Panel

If your console is fitted with the optional Machine Control user panel, then the 10 mark buttons on this panel can be assigned to any locator entry in the list.

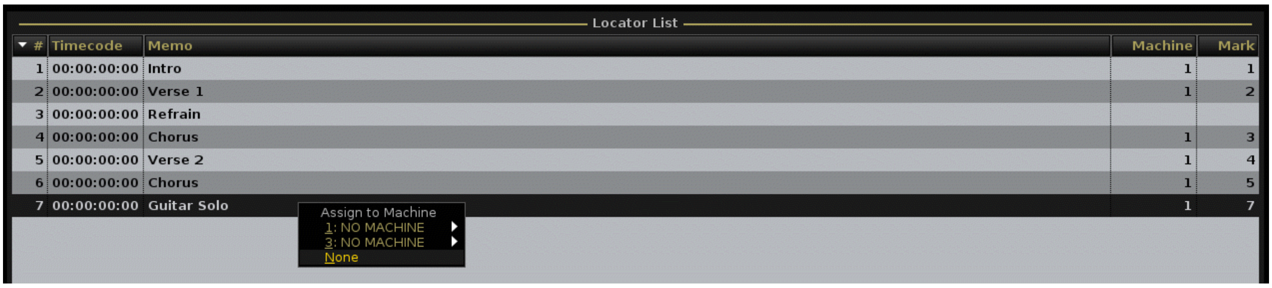
1. To make an assignment, right-click on the locator and select one of the drop-down options - first select the machine (in our example, **1** or **3**), followed by the physical Mark button (from **1** to **10**):



*Once you have made your selection, the **Machine** and **Mark** fields, on the right of the display, update.*

2. For an overview of assignments, select the on-screen **Show Machine Assignments** button, and select a machine (e.g. Machine **1**) - green buttons are assigned; grey buttons are available.

3. To remove an assignment, right-click on the locator and select **None**:



Chapter 8: Signal Routing/Settings

Introduction

This chapter covers the operation of the **Signal List**, **mx Routing**, **Signal Settings**, **mxDSP Settings** and **Downmix** displays.

Topics covered are:

- The Signal List and mx Routing displays - input and output routing may be handled from either of these displays. The **Signal List** presents lists of Sources and Destinations, whereas the **mx Routing** display provides a crosspoint overview. In addition, the **Signal List** is used to edit labels and define channel formats (mono, stereo, surround); the **mx Routing** display is used to create partial snapshots for recalling selective routes.
- The Signal Settings display - handles input and output parameters such as gain, sample rate conversion, etc. In addition, the display provides graphical feedback on system components, and serves as a system diagnostics tool.
- The mxDSP Settings display - if your system is configured with one or more mxDSP modules (optional), then settings within each DSP chain can be controlled from this display.
- The Downmix display - if your system is configured with downmix DSP resources (optional), then the matrix can be controlled from this display.

> mxDSP module(s) and Downmix DSP are configured by AdminHD. If your system configuration does not support these resources, then the displays appear empty.

mxGUI users should ignore any references to front panel operation; instead use the on-screen buttons or right-click context menus to action a function.

Signal Routing: Overview

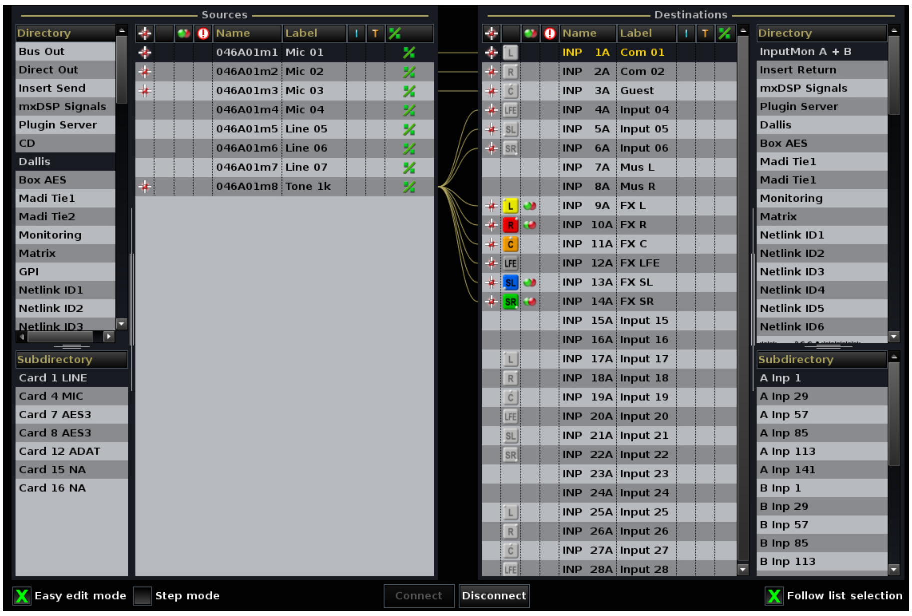
The **mc²56** includes an integrated digital routing matrix. Any source may be routed to any input or monitor channel, and any output bus or channel send routed to any destination. In addition, you may route sources directly to destinations, for example to feed a Mic/Line input to an AES output.

Multiple systems may also be networked in order to share I/O resources. For example, to share the same microphone input between two consoles.

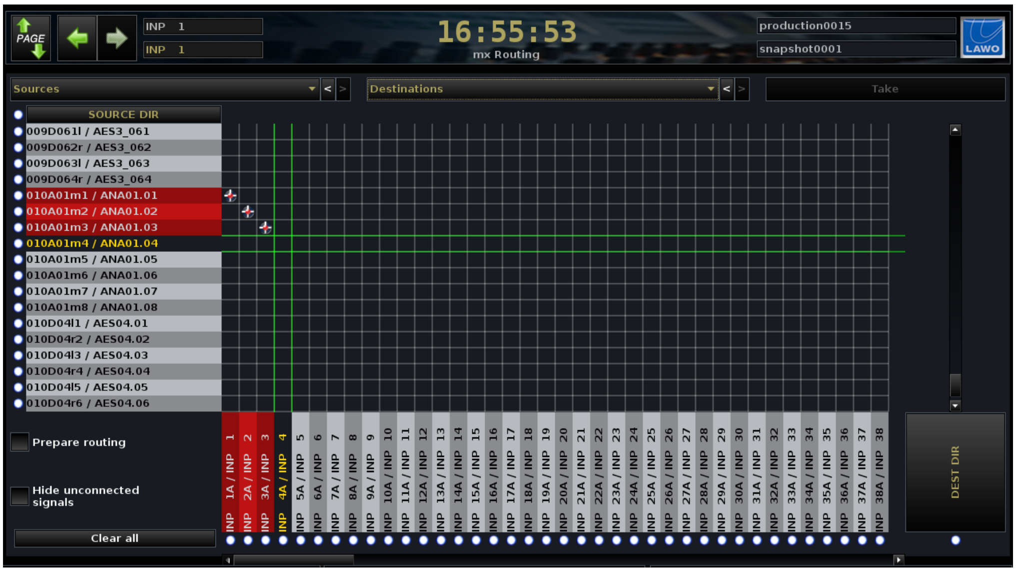
All routes are stored and recalled in productions and snapshots, reducing the amount of manual patching within the installation and saving hours of set up time!

Signal routing may be performed from either the Signal List or mx Routing displays:

Signal List Display

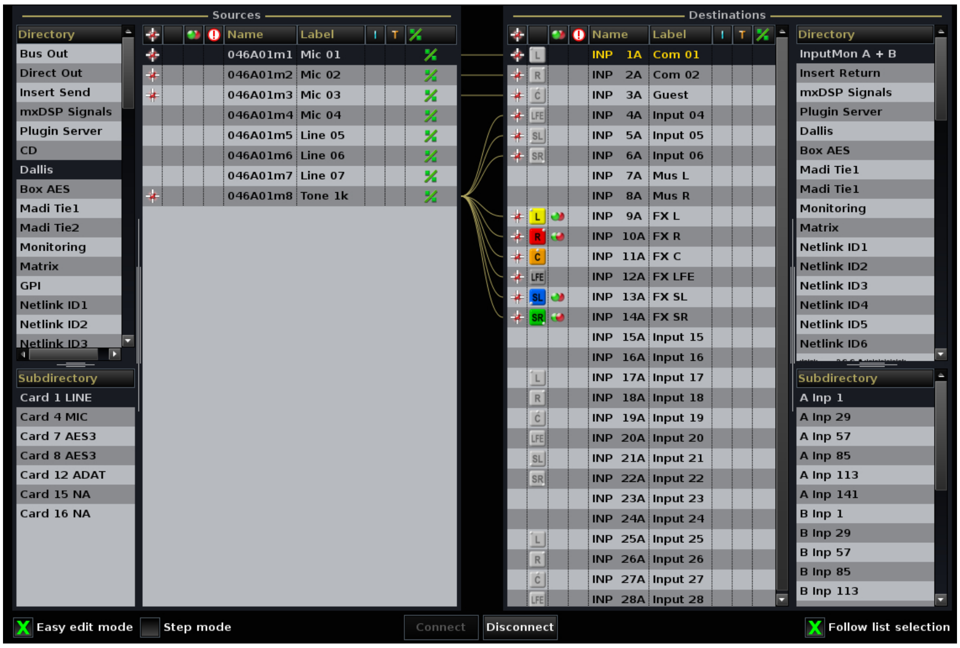


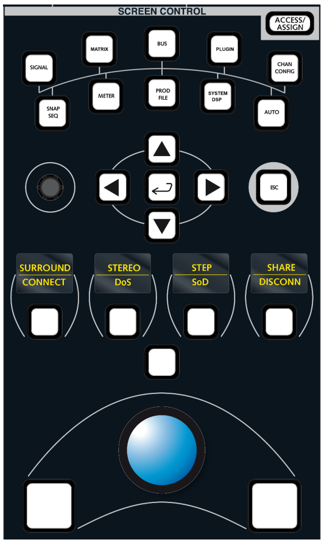
mx Routing Display



The Signal List Display

1. Press the **SIGNAL** button, located on the SCREEN CONTROL panel, to view the **Signal List** display:





The display shows connections from **Sources** (on the left) to **Destinations** (on the right). In order to keep the list manageable, sources and destinations are divided into Directories and Subdirectories.

Open a **Directory** or **Subdirectory** by double-clicking on the directory name, or using the arrows beside the name. You can use the SCREEN CONTROL navigation buttons and rotary scroller to focus on different areas of the display and scroll up/down the **Directory**, **Subdirectory** and **Sources** or **Destinations** lists. You can also resize the windows and/or use the on-screen scroll bars.

If a source or destination is connected, then you will see a red and white cross in the connection column. In addition, if the source and destination are both in view, then a line appears to show the connection. In our example, we can see that the first three input channels (**INP 1** to **INP 3**) are routed from microphones (**Mic 01** to **Mic 03**), while other input channels are routed from **Tone 1k**.

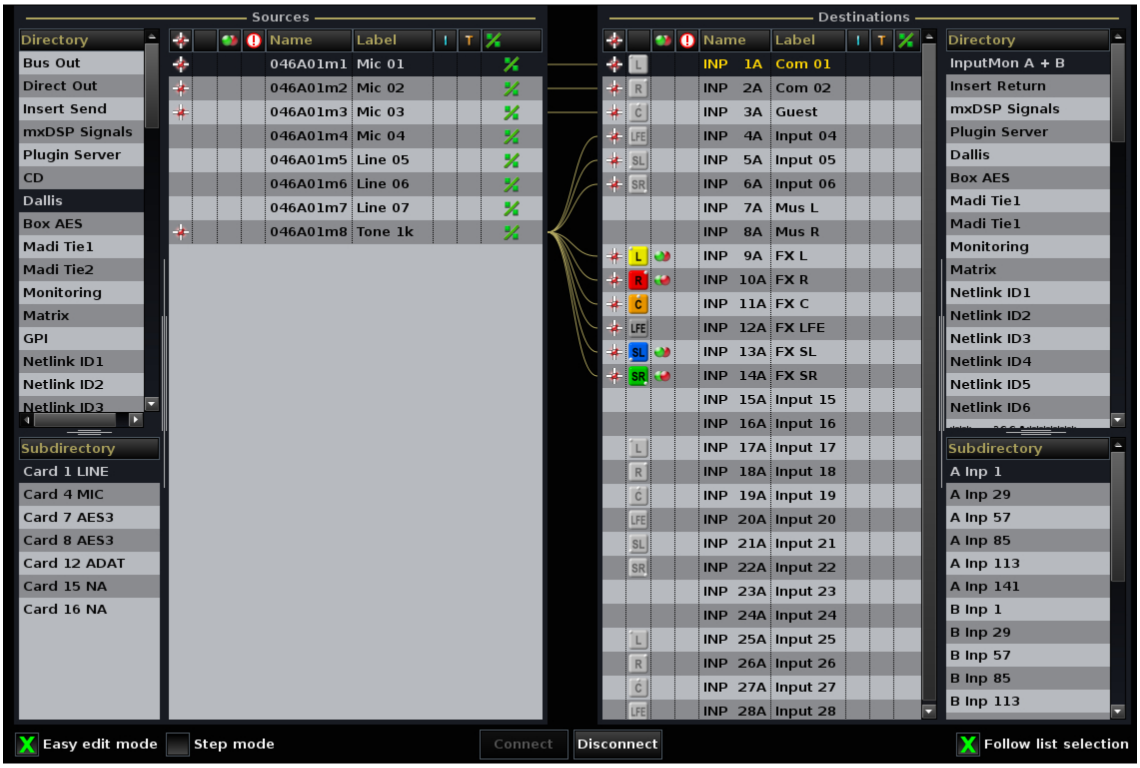
The **Connect** and **Disconnect** on-screen buttons are used to make or unmake routes to/from the selected source and destination. If you enable **Step mode**, then you can quickly step down the list to quickly make or unmake a series of connections.

The **Label** column is used to edit the user label for sources or destinations. **Easy edit mode** will carry text forwards when editing a range of signals.

A number of other functions are available from the SCREEN CONTROL soft keys, or by right-clicking on a source or destination. They include defining the channel format (mono, stereo or surround), reverse interrogation of routing, isolating or protecting individual signals, and placing the selected DSP channel in access.

When running mxGUI offline, all signals appear as unavailable.

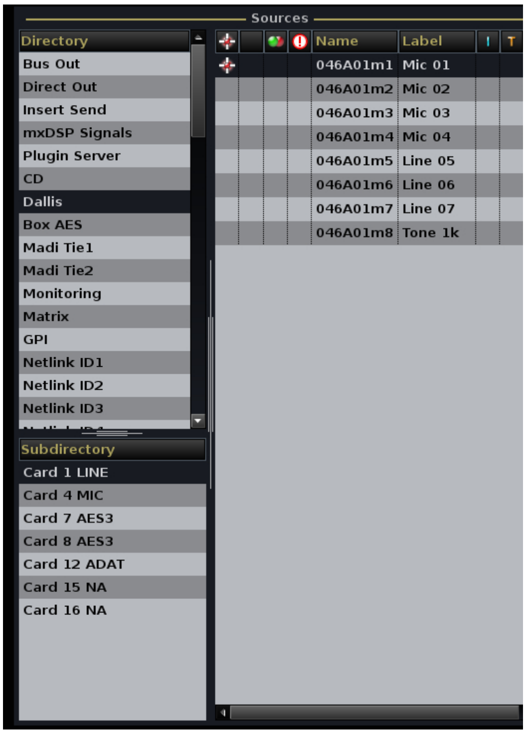
Signal List Columns

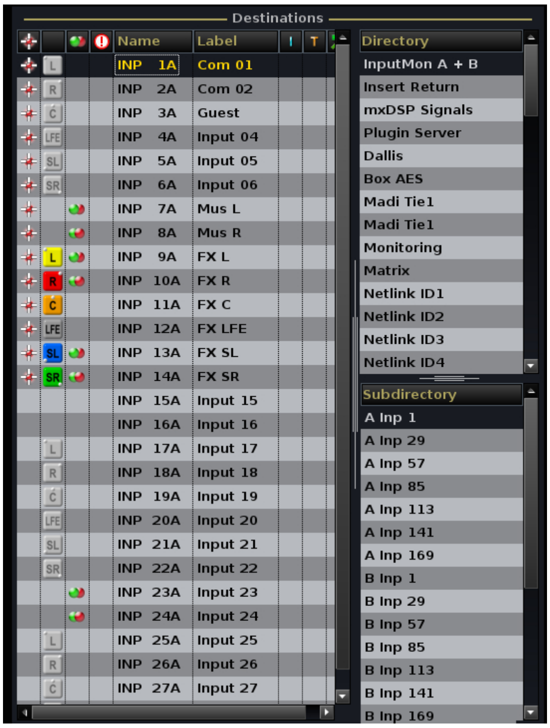


The columns beside each signal provide the following information:

- **Connection** – a red and white cross appears when a source or destination is connected. If a destination is protected, then you will also see a padlock icon.
- **Surround** – you will see colour coded channel definitions if a source or destination is surround. (The greyed out definitions show which blocks of channels may be configured for surround).
- **Stereo** – interlocking red and green circles appear when a source or destination is stereo.
- **Unavailable** – a warning symbol appears beside signals which are not available.
- **Name** – this is the system name for the signal (defined by the AdminHD configuration).
- **Label** – this is the user label for the signal. You can rename signal labels from this column.
- **I** – indicates if a signal is Isolated from snapshot recall.
- **T** - indicates a Tiny (reduced processing) channel.
- **%** - indicates 'Shared' or 'Imported' sources within a networked installation.

Directories & Subdirectories





Every mc² console supports a number of common source and destination **Directories** which are supported by all systems: **Bus Out**, **Direct Out**, etc. (Note that these **Directories** cannot be renamed or reorganised by AdminHD, and *always* appear as the first in the list.)

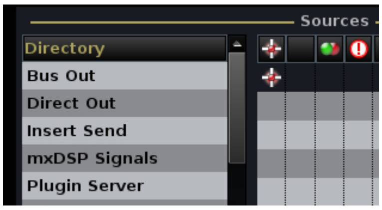
External signals such as mic/line, AES, MADI, etc. vary depending on the input and output cards and type fitted to your system. Therefore, AdminHD is used to place these signals within custom-named **Directories** and **Subdirectories**, and give them a system **Name** and default user **Label**.

In our system we have some custom source **Directories** named **CD**, **DALLIS**, **Box AES**, etc. Within the **DALLIS** directory, the **Subdirectories** are named **Card 1 LINE**, **Card 4 MIC**, **Card 7 AES3**, etc. And within each **Subdirectory**, we have access to our signals.

In your system, you will have different **Directories**, **Subdirectories** and signal **Names** in order to easily identify the location/application of each signal.

> From the Central GUI, you *cannot* change the **Directory**, **Subdirectory** or signal **Names.** You *can* edit user Labels, and save and recall them with snapshots or a production.

Common Source Directories



The source **Directories**, supported by all systems, are as follows. Note that the number of sums, groups, etc. depends on your choice of DSP configuration:

- **Bus Out**
  - **DOUT Sum** - sum bus outputs.
  - **DOUT Grp** - group bus outputs.
  - **DOUT Aux** - aux bus outputs.
  - **AFL/PFL** - AFL and PFL bus outputs.
- **Direct Out**
  - **DOUT Inp** - input channel direct outputs.
  - **DOUT Mon** - monitor channel direct outputs.
- **Insert Send**
  - **Send Inp** - input channel insert sends.
  - **Send Mon** - monitor channel insert sends.
  - **Send Sum** - sum insert sends.
  - **Send Grp** - group insert sends.
  - **Send Aux** - aux insert sends.
- **mx DSP Signals** (optional) - this directory only appears if you have an mxDSP module configured. It provides access to the mxDSP source signals.
- **Downmix Matrix** (optional) - this directory only appears if you have a Downmix matrix configured. It provides access to the downmix matrix source signals.
- **Plugin Server** (optional) - this directory only appears if you have a Plugin Server. It provides access to the plug-in server source signals.

Common Destination Directories

The destination **Directories**, supported by all systems, are as follows. Note that the number of inputs, monitors, etc. depends on your choice of DSP configuration:

- **InputMon A + B**
  - o **A Inp** - input channels (A inputs).
  - o **B Inp** - input channels (B inputs).
  - o **A Mon** - monitor channels (A inputs).
  - o **B Mon** - monitor channels (B inputs).
  - o **Command 1-8** - the 8 talkback bus sources (used by talkback user buttons configured from the Custom Functions display.)
  - o **DynKey 1-8** - the 8 dynamics key inputs (used by the dynamics processing, if you enable the External Key.)
- **Insert Return**
  - o **Ret Inp** - input channel insert returns.
  - o **Ret Mon** - monitor channel insert returns.
  - o **Ret Sum** - sum insert returns.
  - o **Ret Grp** - group insert returns.
  - o **Ret Aux** - aux insert returns.
- **mx DSP Signals** (optional) - this directory only appears if you have an mxDSP module configured. It provides access to the mxDSP destination signals.
- **Downmix Matrix** (optional) - this directory only appears if you have a Downmix matrix configured. It provides access to the downmix matrix input signals.
- **Plugin Server** (optional) - this directory appears if you have a Plugin Server. It provides access to the plug-in server destination signals.

Routing a Source to a Destination

To make a route - for example, to route a microphone source to an input channel:

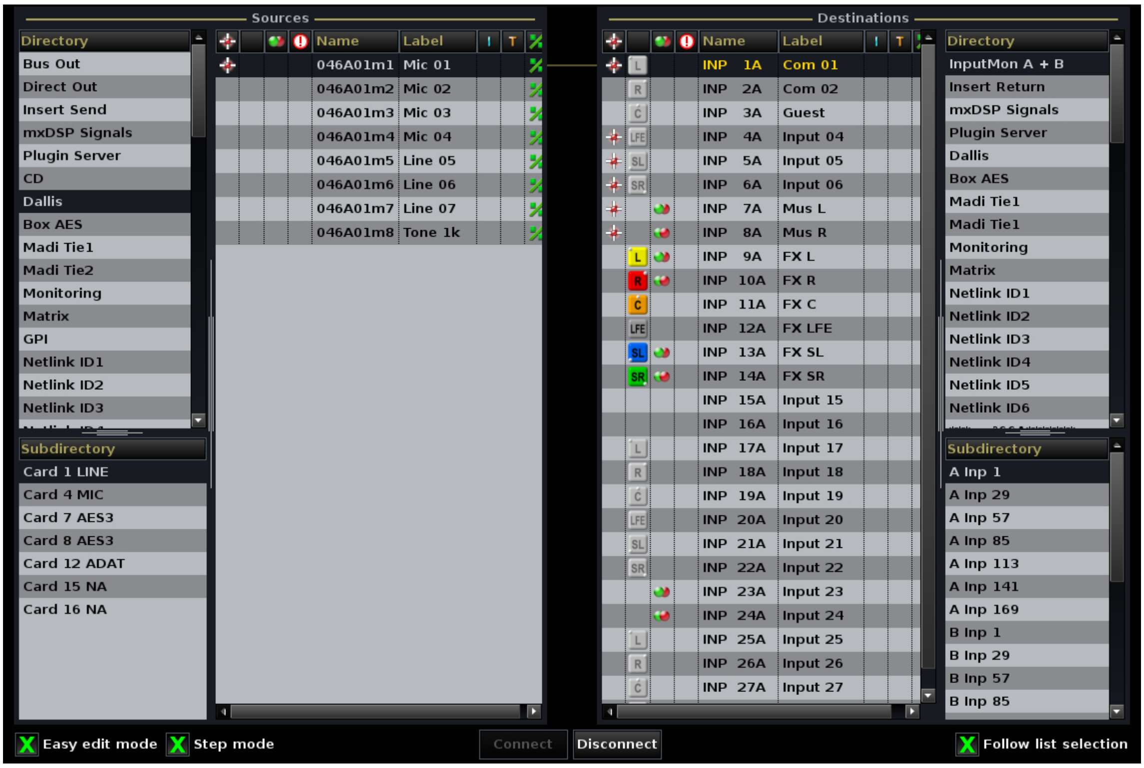
1. Select the source – for example, the source directory called **DALLIS**; subdirectory called **CARD 1 LINE**; and the source named **Mic 01**.

2. Select the destination – for example, the destination directory called **Input/Mon A + B**; subdirectory called **A Inp 1-28**; and destination called **INP 1A**.

Note that input and monitor channels support an A/B input switch. By selecting **INP 1A** as the destination, you will route to the A input of input channel 1.

3. Then press the on-screen **CONNECT** button, or SCREEN CONTROL soft key, to make the connection.

*The **Signal List** updates with a line between the source and destination:*



If the input channel is already assigned to a fader strip, and **INHERIT SOURCE** is selected (from the centre section LABEL buttons), then you will see the source label in the fader strip's label display. You will also see signal present beside the fader, and metering on the **Channel** display (according to the meter pickup point).

Routing Consecutive Sources to Destinations (Step Mode)

To route consecutive sources to consecutive destinations, turn on **Step mode** to speed up the connection process.

1. Select the first source – for example, **Mic 01** – and the first destination – for example, **INP 1A**.

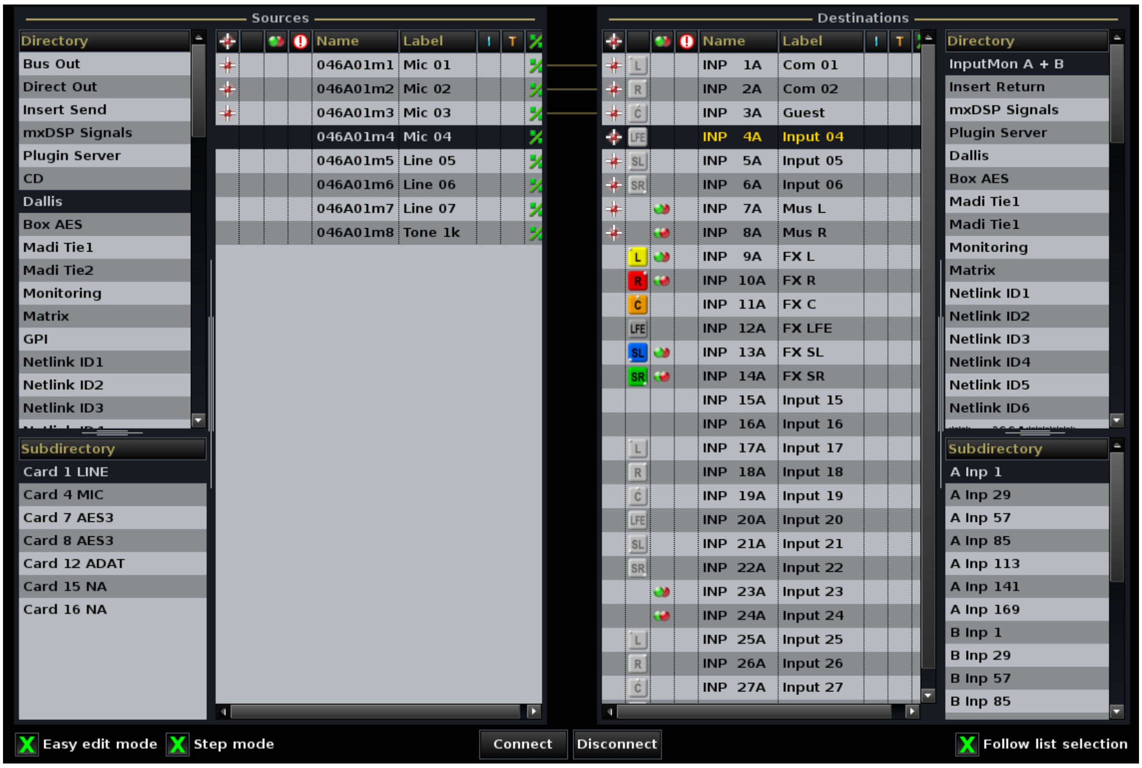
Your selected source and destination are highlighted in black.

2. *BEFORE* you press **CONNECT**, enable the on-screen **Step mode**, or select the **STEP** soft key.

3. Now press **CONNECT**.

The first route is made and the source and destination selections automatically step down to the next entries in the list:

4. Continue pressing **CONNECT** until all of your sources are connected to your destinations:



If the list of sources is shorter than the list of destinations, then when you reach the last source in the list, **Step mode** automatically scrolls back up to the first source in the list. This allows you to continue making routes from the sources to the remaining destinations, for example, to route microphones 1-16 to input channels 1-16, 17-32, etc.

Step mode can also be used with an offset between the starting source and destination: for example, to route Microphones 1-16 to Input Channels 17-32, repeat the above operation but set your first destination channel to be **INP 17** rather than **INP 1**.

Disconnect

To remove a route:

1. Select the destination (e.g. **INP 2A**).
2. And press the on-screen **DISCONNECT** button, or SCREEN CONTROL soft key.

The line between the source and destination disappears:

Turn on Step mode, select the first destination, and then keep pressing **DISCONNECT** to disconnect a range of destinations quickly and easily.

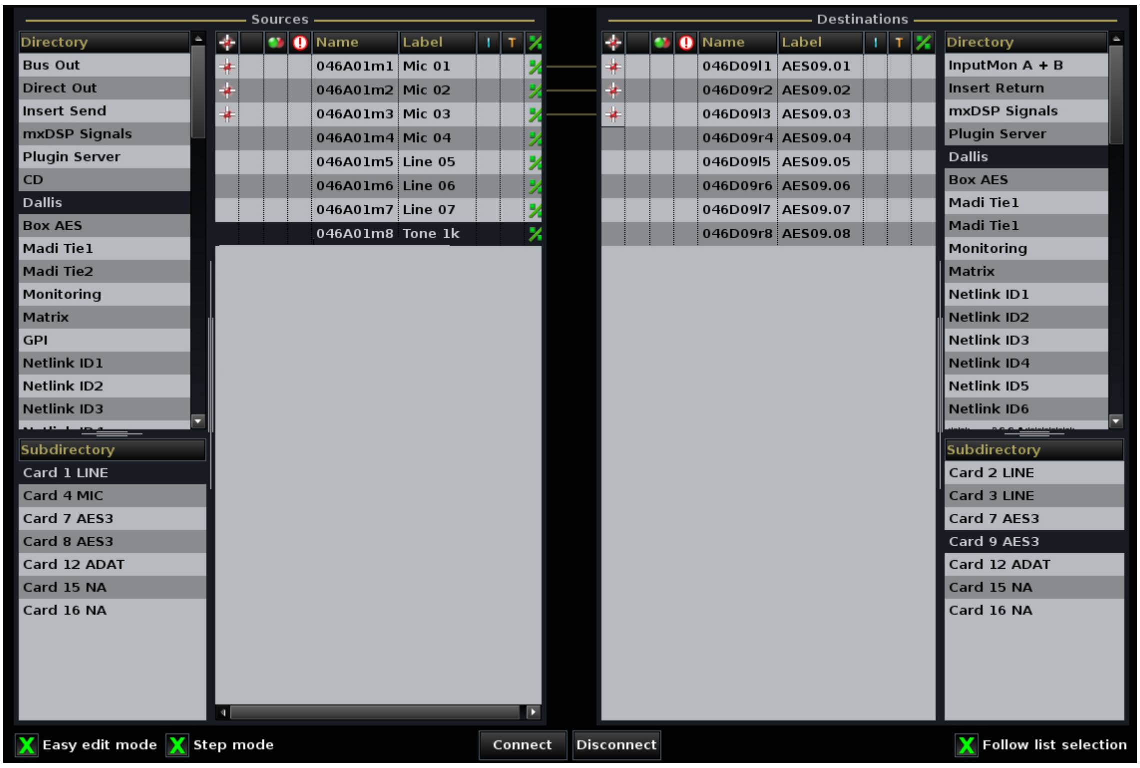
Note that if you route a source to a connected destination, then the previous source assignment is replaced; you don't have to disconnect the destination to assign a new source.

More Signal Routing Examples

The same steps may be used to connect any source to any destination. For example:

- To route a Sum bus to an output, select **Bus Out** -> **DOUT Sum 1** -> **Sum 1** as the source, and your external output as the destination:

- To route a microphone signal directly to an AES output, select the mic/line input as the source, and your AES output as the destination. This makes a direct route through the matrix, bypassing the console's channel DSP.

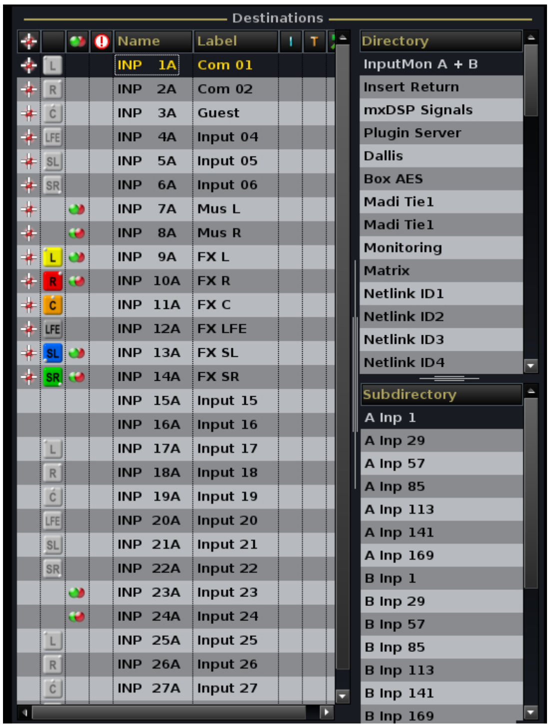


A/ B Input Sources

For any input or monitor DSP channel, you may assign two sources (A and B) to provide a main and backup source for the channel.

A/B input switching is available from the fader strip, or the Input Control section.

The A and B input sources are assigned from the **Signal List** display:



- Select **A Inp** or **A Mon** to assign a source to the A input of a channel.
- Select **B Inp** or **B Mon** to assign a source to the B input of a channel.

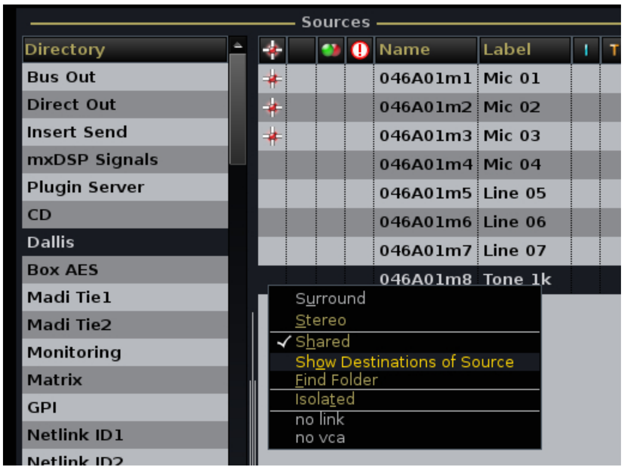
Reverse Interrogation of Signal Routing

Reverse interrogation provides a quick way of viewing all the sources feeding a particular destination, or all destinations routed from a particular source.

➢ To view all the destinations fed from a source:

1. Select the source you wish to interrogate on the left of the display – e.g. **Tone1**.

2. Then right-click and select **Show Destinations of source**, or press the **DoS** (Destinations of Source) soft key:



*A list of all current destinations for the selected source appears in the **Destinations** list:*

Note that if the source is routed to an input or monitor channel, then for each channel assignment you will see three routes: Source to Input; Source to Input A; Source to Input B.

➢ **To find the source which feeds a destination, reverse the procedure:**

1. Select the destination you wish to interrogate on the right of the display.
2. Then right-click and select **Show Source of destination**, or press the **SoD** (Source of Destination) soft key:

*The source assigned to the selected destination appears in the **Sources** list.*

Find Folder

If you are unsure which directory or subdirectory this source (or destination) belongs to, then you can use **Find folder** as follows:

1. Right-click on the destination (or source), and select **Find folder**.

*The **Directory** and **Subdirectory** update to reveal the correct folder for the selected destination:*

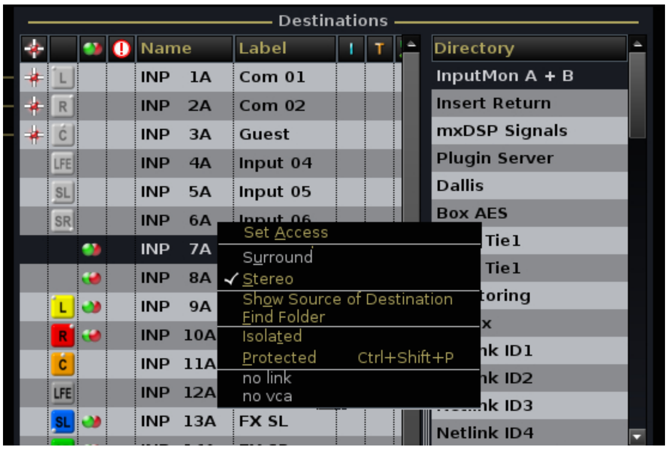
Creating Stereo or Surround Channels/Busses

While making routes from the **Signal List** display, you can also configure whether your channels and busses are mono, stereo or surround. For example:

➢ **To create a stereo input channel:**

1. Select an odd numbered input channel from the **Destinations** list (e.g. **INP 7**).

2. Press the **STEREO** soft key, or right-click and select the **Stereo** option:



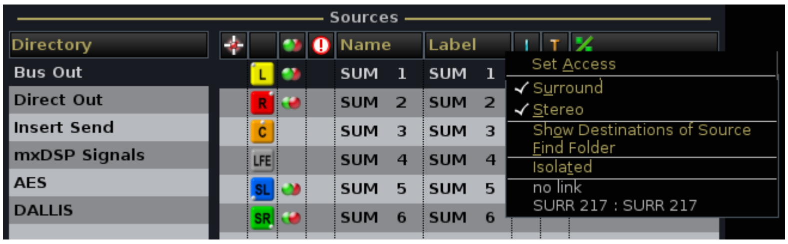
This links the selected channel to its adjacent DSP path. For example, INP 7 and INP 8.

You can link any odd/even pair of input or monitor channels using this method. Alternatively, select a **Bus Out** from the **Sources** list to create a stereo bus master.

➢ **To create a surround sum:**

1. Select the first sum for the surround output from the **Sources** list (e.g. **SUM 1**).

2. Press the **SURROUND** soft key, or right-click and select the **Surround** option:



*This links consecutive sums, according to the global surround format, and automatically assigns a Surround VCA - in our example, **SURR 217**.*

You can configure surround sums, groups or auxes using this method. Alternatively, select **InputMon** from the **Sources** list to configure surround input or monitor channels.

For surround inputs, panning is automatically reset so that INP 9 feeds SUM 1, INP 10 feeds SUM 2, etc. The best way to position a surround channel within the surround field is using Hyper Pan.

Surround channels may only be created in 8-channel blocks, so you must select Sum 1, 9, 17, etc. You cannot select **Surround** if you right-click on an invalid channel number.

Note that the front and rear left/right pairs of a surround channel are automatically linked for stereo. This is for convenience when revealing the component channels. The stereo linking is only a default state; you can deselect the stereo link at any time.

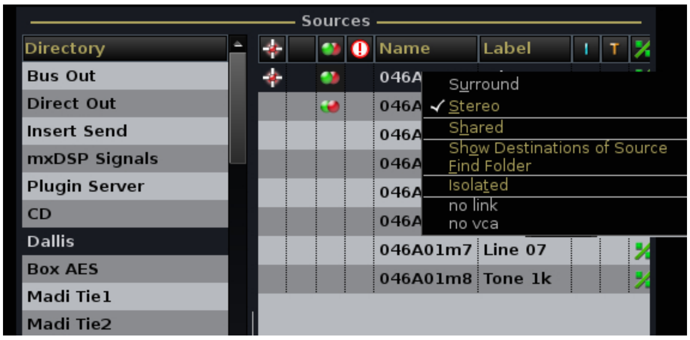
Even if channels are configured for stereo or surround, signal routing is still handled individually. This allows you to route non-consecutive sources to the inputs of a stereo or surround channel.

Stereo Signals

You may link external signals as **Stereo**. This affects the behaviour of the signal's I/O DSP, but signal routing is still handled independently.

For example, if you link two microphone signals, they can be routed to destinations independently, but their I/O DSP operates in stereo.

1. Select the odd numbered source you wish to link.
2. Then right-click and select **Stereo**:



The red and green circles in the Stereo column reflect the status .

Editing Source and Destination User Labels

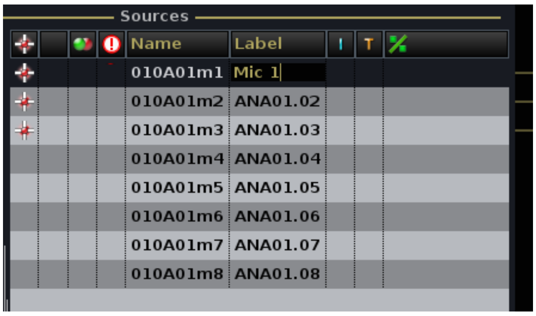
The user **Label** for each source and destination may be edited from the **Signal List**, and is stored and recalled by both snapshots and productions:

Note that the source and destination **Name** is defined by the AdminHD configuration, and cannot be edited from the Central GUI. This provides a fixed **Name**, relevant to the installation, which remains consistent for all users.

In addition to labelling signals in the **Signal List** display, source and channel labels may be viewed on the fader strip label display, the Title Bar and Channel display. The centre section LABEL buttons control what is displayed. For input and monitor DSP channels, use the Source **Label** field to edit your source labels, and the Destination **Label** field to edit the channel user labels. See Labels for further advice on how to use and switch between the different label types.

➢ **To edit a single label:**

1. Click on the source or destination label:



Click once to select all the existing text (white) or twice (black cursor) to modify the existing name.

2. Enter a new name from the keyboard.

3. When you have finished, press the Enter button, on the keyboard, to confirm the new name.

4. Or, if you make a mistake or want to exit without making any changes, press the **Esc** button on the keyboard.

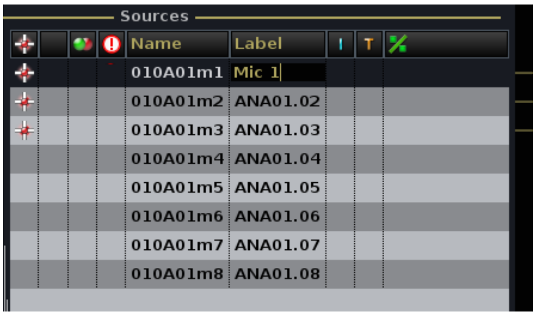
Easy Edit Mode

To edit labels for consecutive sources or destinations, turn on **Easy Edit** to speed up the labelling process.

Easy Edit carries forward your text, so that you can quickly enter the same label for multiple signals. Or, if the label ends with a number, then the number will increment.

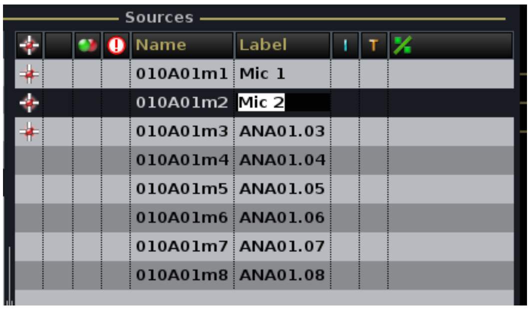
1. *BEFORE* editing your first source or destination label, turn on **Easy Edit** at the bottom of the **Signal List** display:

2. Then select and edit your first source or destination label in the usual manner - e.g. **Mic 1**:

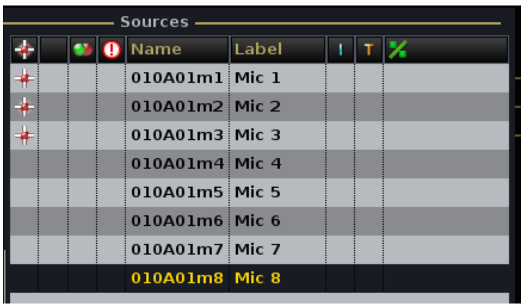


3. Press Enter, on the keyboard, to confirm.

*With **Easy edit** enabled, the system automatically steps down to the next signal in the list. The text label is copied, and if the text ends with a number, then the number increments:*



4. Keep pressing Enter to label all the signals in the list:



5. When you have entered the last label, press the **Esc** button on the keyboard, to exit the labelling mode.

If the label does not end with a number, then the same text is carried into the next label field.

For temporary **Easy Edit**, use the console keyboard **SHIFT** button as follows:

1. Turn off the **Easy Edit** checkbox.
2. Select the first signal label and enter a new label in the usual manner.
3. Press and hold **SHIFT** and then press **Enter** on the keyboard.

*Holding down **SHIFT** temporarily enables **Easy edit**, so the system automatically steps down to the next signal in the list. The text label is copied, and if the text ends with a number, then the number increments.*

4. Keep holding **SHIFT** and pressing **Enter** until you have labelled all the required fields.
5. Press the **Esc** button on the keyboard, to exit the labelling mode.

Storing and Recalling User Labels

User labels are stored in both snapshots and productions.

Saving snapshots with different labels allows you to easily recall new labels for a different part of a show.

You may use the **LABEL** Global Snapshot Isolate option to protect user labels from a snapshot reset.

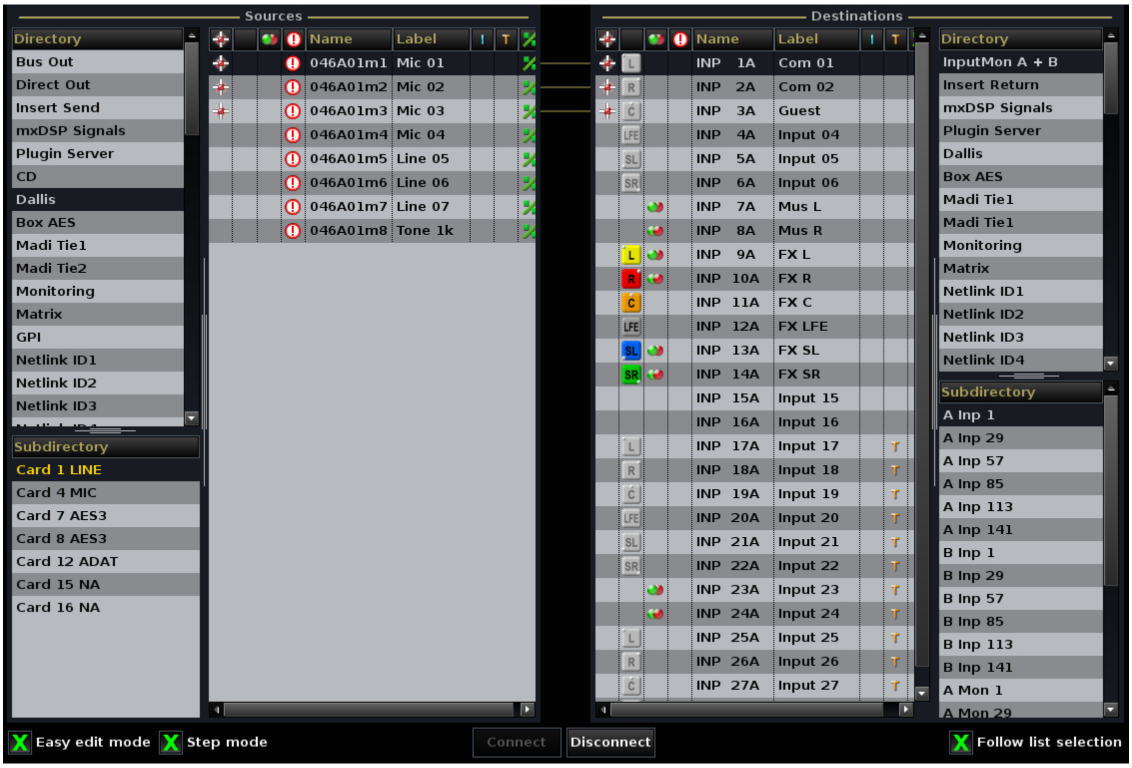
Not Available Signals

If a warning flag is present within the unavailable column, then a signal is currently unavailable.

> When running mxGUI offline, all signals appear as unavailable.

The warning flags can be useful for fault finding and reassurance. For example, in an outside broadcast vehicle, you may have a number of remote DALLIS Stageboxes. During the setup for the broadcast, you can make routes from microphone sources which connect to these Stageboxes, even if the Stagebox is not yet connected.

The warning flag indicates that the signal is currently unavailable. However, you can continue to label the signal and make routes to/from it as normal:



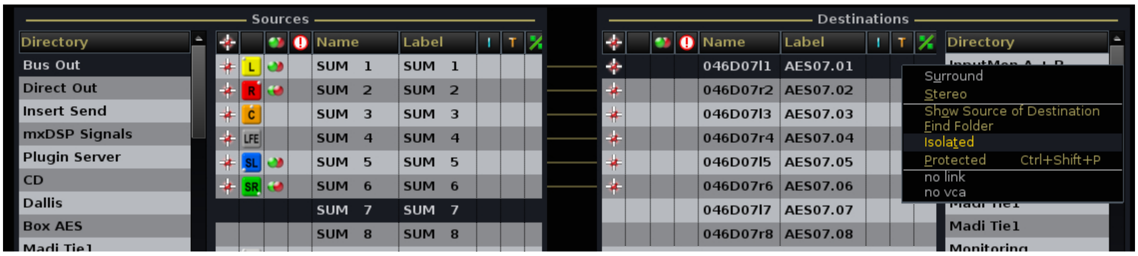
When the Stagebox is connected to the system the column updates accordingly and the warning flag disappears.

Isolated Signals

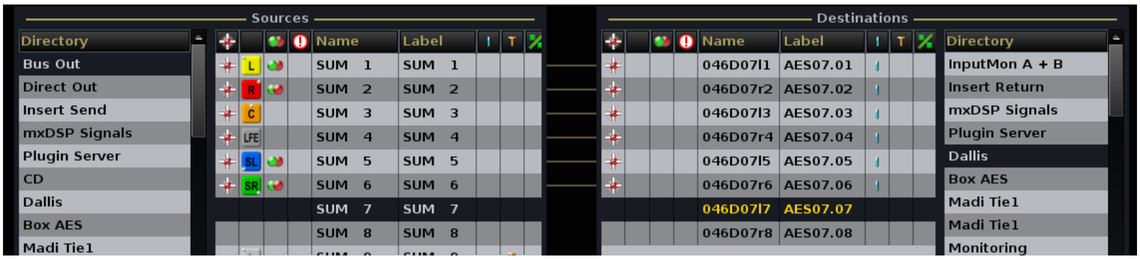
The **I** column indicates if a signal is isolated from a snapshot recall. For example, you may wish to protect important signals, such as main sum distribution or monitor feeds, from accidental reset.

On sources, only the source parameters are isolated. On destinations, the destination parameters and any routes made to the destination are isolated. Therefore, to isolate matrix crosspoints, select the destination.

1. Right-click on the source or destination and select the **Isolated** option:



*The **I** column updates to identify all isolated signals:*



Note that the isolate function does not prevent routes from being stored when a snapshot is saved or updated; Isolate only applies when settings are loaded back from a snapshot.

Snapshot isolates are stored and recalled by productions.

Individual signals can be isolated or protected, at a lower level, by using a Custom Function - see Snap Iso List. Or within the factory configuration (via a tcl file) - please check your system specification.

Protected Signals

To apply more comprehensive protection to a matrix destination, then it can be protected so that nothing can alter its connection.

Once protected, nothing can alter the connection to the destination – not the **Signal List** or **mx Routing** displays, not snapshots, productions, mxGUI or remote MNOPL. This is ideal for critical signals, such as mains distribution.

Note that only destinations can be protected.

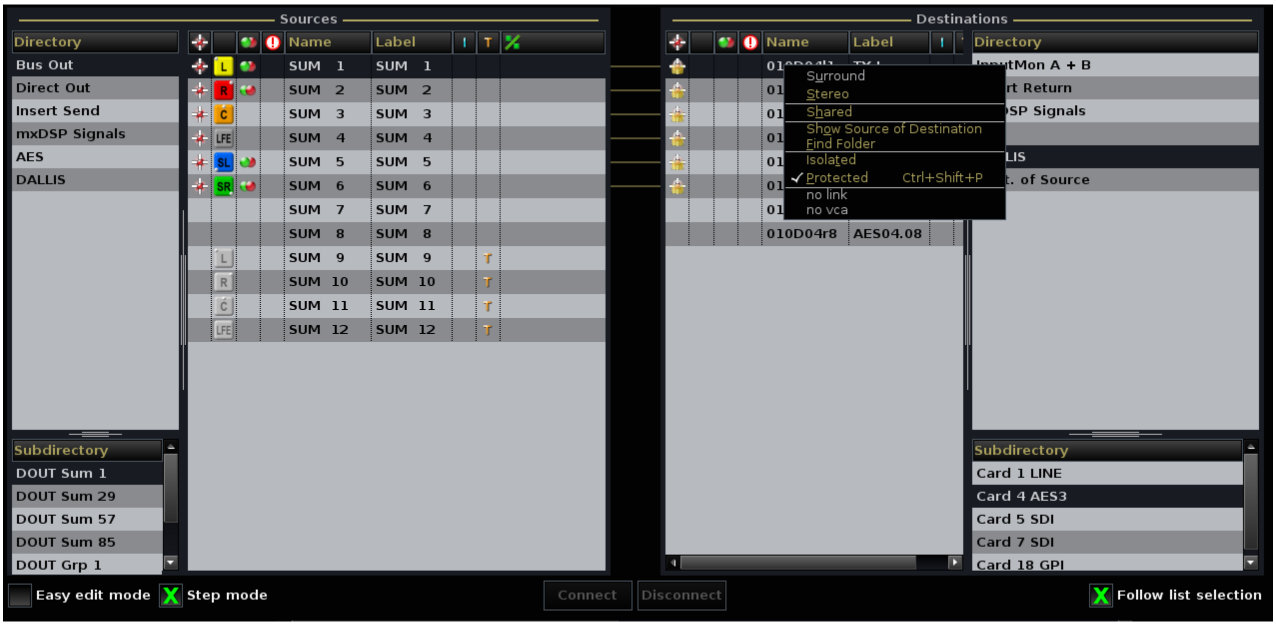
The state of protected signals is not saved or loaded by productions, snapshots or automation. Therefore, any changes are permanent, and affect all users, unless you deselect the **Protected** option.

> Individual signals can be isolated or protected, at a lower level, by using a Custom Function - see Snap Iso List. Or within the factory configuration (via a tcl file) - please check your system specification.

➢ To protect a destination signal:

1. Select the destination you wish to protect.
2. Right-click and select the **Protected** option.

Protected destinations are displayed with a padlock icon in the connection column:

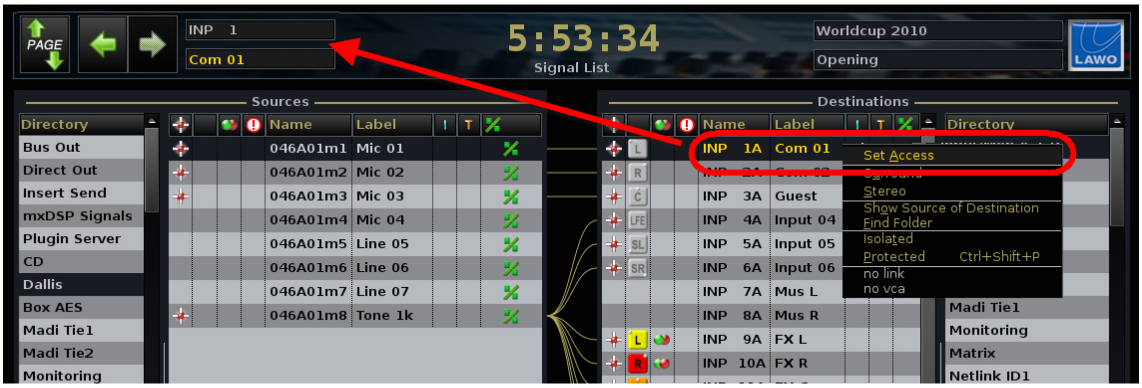


3. To change the route to a protected destination, you must first turn off the **Protected** option.

Set Access

From V4.24 software onwards, for any type of DSP channel, you may quickly place the selected channel in access as follows:

1. Right-click on the source or destination channel and select **Set Access**:

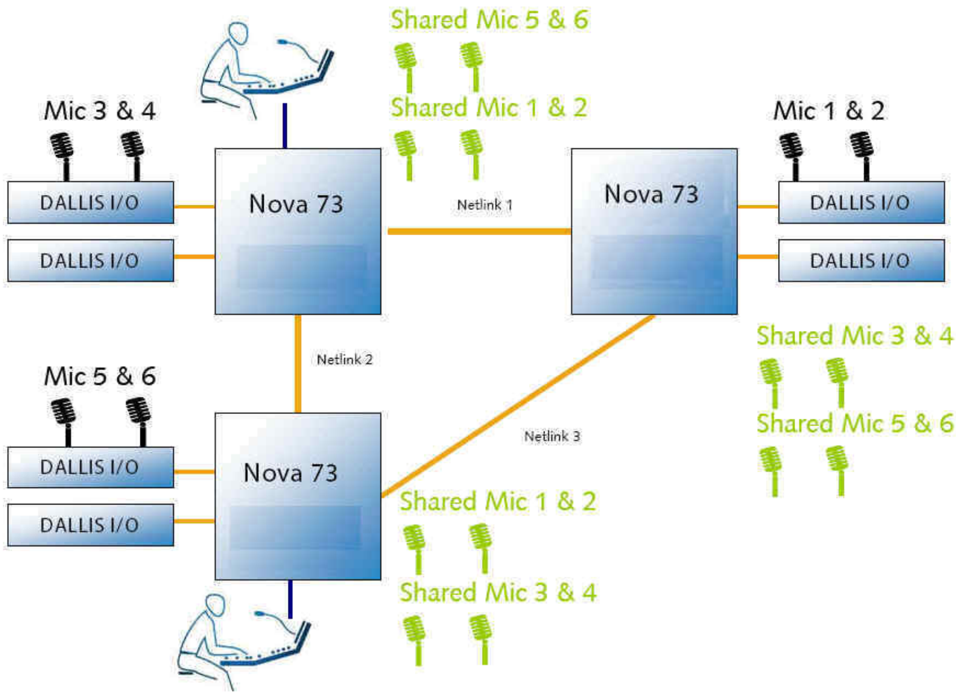


The **Set Access** context menu option *only* appears when a DSP channel is selected; you cannot place an input or output signal into access.

Networking I/O Resources

The **mc²56** is just one member of the mc² family of products, which utilise the same Nova73 and DALLIS architecture, and run on the same operating system and application software.

The Nova73 and DALLIS system is available in its own right as a stand alone routing matrix. Multiple systems may be networked to provide sharing of sources and destinations:



In the example above, mics are physically connected, via a DALLIS, to each system. Signals are transferred between systems via 'Netlinks', providing the ability to share any mic input.

Each 'Netlink' is an audio connection which may be MADI, RAVENNA, ATM, AES or analogue audio, and signals are dynamically allocated as each operator makes routes from the **Signal List** display.

Any number of sources may be distributed depending on the physical limitations of your network. Please consult your system specification for details.

On any system within the network, you can view which sources are distributed from the **%** column on the **Signal List** display:

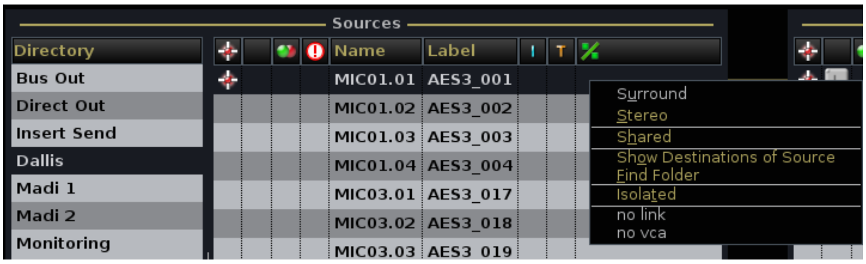
-  indicates that a source is connected locally to this console, and is 'Shared' (made available) to other consoles within the network.
-  indicates that a source is 'Imported'. In other words, it is not connected locally to this console.

> From version 4.0.2.2 onwards, all Lawo products have adopted a consistent software release numbering system to indicate compatibility. In each case, the first three digits of the software version *must* match.
>
> So, for example, a mc²66 console running version **4.20.2.0** can be networked to a mc²56, mc²90 or Nova73 running **4.20.2.x**. You can check the software version of your mc² system from the Global Options in the **System Settings** display.

Sharing Sources

On the system which is distributing the signals – in our example, console A - you can select which sources are shared from the **Signal List** display.

1. Select the source you wish to share (e.g. **Mic1**).
2. Right-click and select the **Shared** option, or press the **SHARE** soft key:



*An icon appears in the **%** column to indicate that the source is now shared.*

3. Press **SHARE** again to unshare the source.

> You *cannot* unshare a source if it has been routed as an imported source within another console. For example, if console B has made a route using the Mic 1 signal, then console A cannot unshare the Mic 1 source until console B's route is removed. This protects one console from removing routes which are in use by another within the network.

If you wish to share a number of sources, then enable Step mode to step through and **SHARE** a number of sources.

Once the source has been shared from console A, then other consoles within the network may access this source from their **Signal List** display. An icon appears in the **%** column to indicate that the source is imported.

> Console B will *only* be able to access the source if its AdminHD configuration has been programmed to do so – i.e. a location for the imported source must have been created within **Directory** and **Subdirectory**. Please consult your technical department for further details.

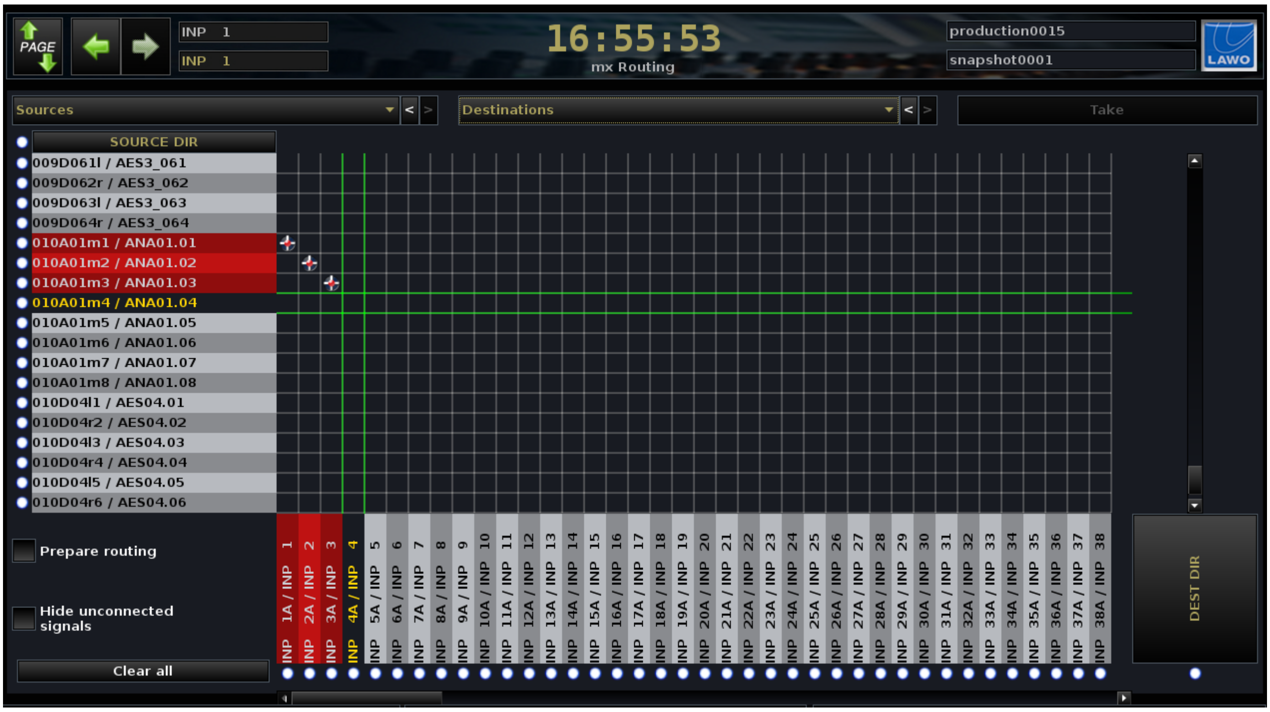
Once console B can 'see' the imported source, then making a route or changing parameters is done in exactly the same way as if the source were local to the console.

All consoles within the network have access to the source parameters, and the last console to make a change wins. In our example, consoles A and B both have access to mic pre-amp control for mics 1 and 2. Similarly for a shared digital destination, both consoles may change parameters like SRC, etc.

This philosophy extends to snapshots. So if both console A and B are using the Mic 1 signal, parameter settings like mic gain, etc. can be reset from snapshots from either console. To control which console resets the mic parameters, use the **I/O** Global Snapshot Isolate option to prevent recall of I/O settings. Alternatively, you may employ a third party system, such as VSM, to manage control priorities.

The mx Routing Display

1. Press the **MATRIX** button, located on the SCREEN CONTROL panel, to view the **mx Routing** display:



This display provides a crosspoint overview of signal routing, with sources running down the left hand side, and destinations running across the bottom. The names of the source and destination directories are shown at the top of the display – in our example, all **Sources** and all **Destinations**.

If a source or destination is connected, then it is highlighted in red. If the source and destination are both in view, you will see a red and white cross on the grid to show the crosspoint connection. If a destination is protected, you will see a padlock icon.

The **mx Routing** display can view or change signal routing, and access many of the same options as the Signal List. Any changes are reflected in the **Signal List** display, and vice versa.

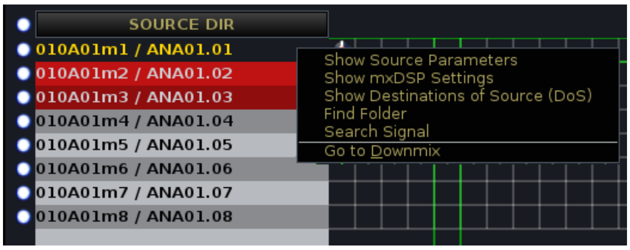
This section concentrates on operations which *cannot* be performed from other displays. They are:

- Signal routing - via the crosspoint matrix.
- Search signal - to locate a signal by name or label.
- Preparing signal routing - to prepare a set of connections and then action them simultaneously.
- Partial snapshots - to store and recall selected routing crosspoints.

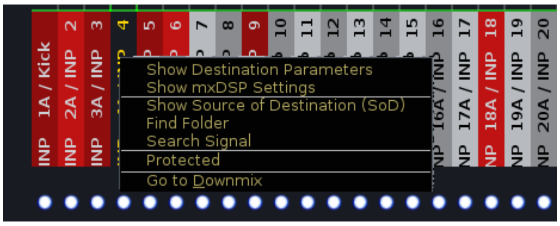
Right-click Functions

If you right-click on a signal, then the following functions become available. Note that most of these are "duplicate" functions, so please follow the links below for more details:

Right-click on Source signal



Right-click on Destination signal



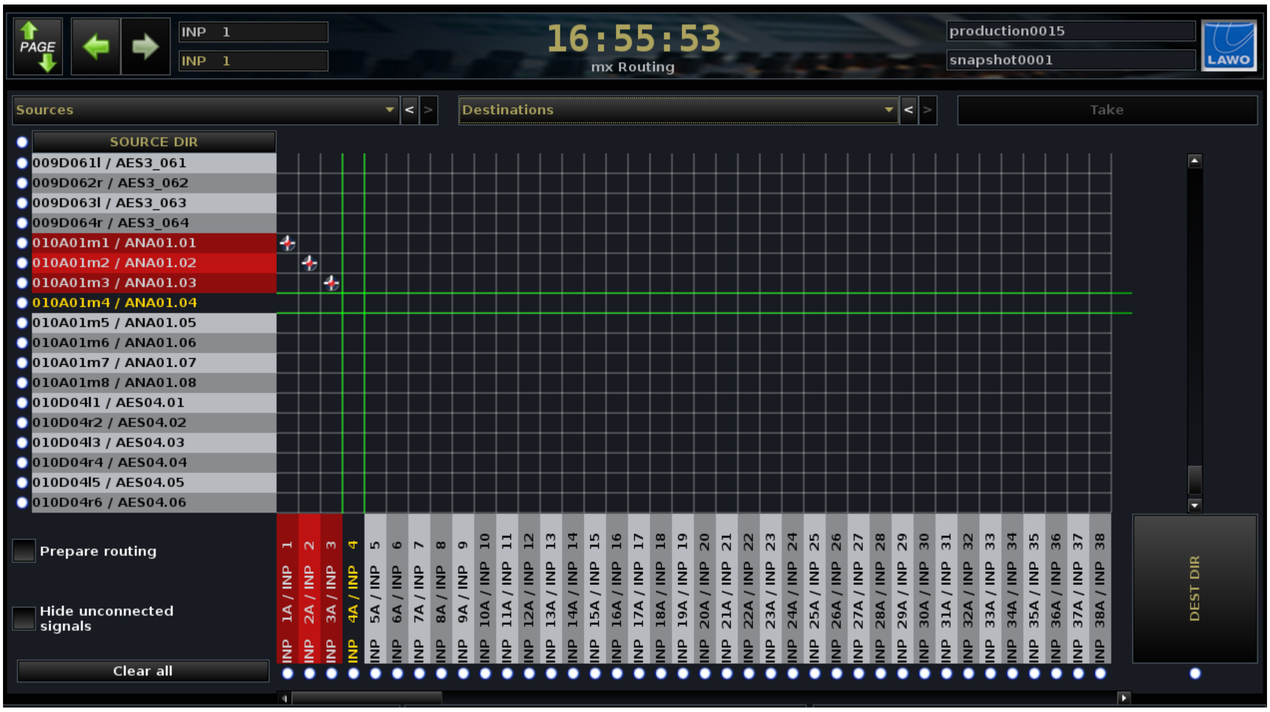
- **Show Source/Destination Parameters** - opens a pop-up window where you can adjust parameters for the selected signal. These options are identical to those found on the Signal Settings display.
- **Show mxDSP Settings** - opens a pop-up window where you can adjust mxDSP parameters. These options are identical to those found on the mxDSP Settings display.
- **Show Dest of Source/Source of Dest** - provides reverse interrogation of signal routing, and works in a similar manner to the Signal List display.
- **Find Folder** - reveals the signal's folder, and works in a similar manner to the Signal List display.
- **Search Signal** - covered later in this section. Note that this operation is *only* available from the **mx Routing** display.
- **Protected** - protects the selected destination, and works in a similar manner to the Signal List display.
- **Go to Downmix** - if the selected signal is an input or output to a downmix matrix, then this option automatically opens the Downmix display.

Signal Routing from the mx Routing Display

To make a route from the **mx Routing** display:

1. Position the cursor to select a source and a destination.

The crosspoint is highlighted in green:



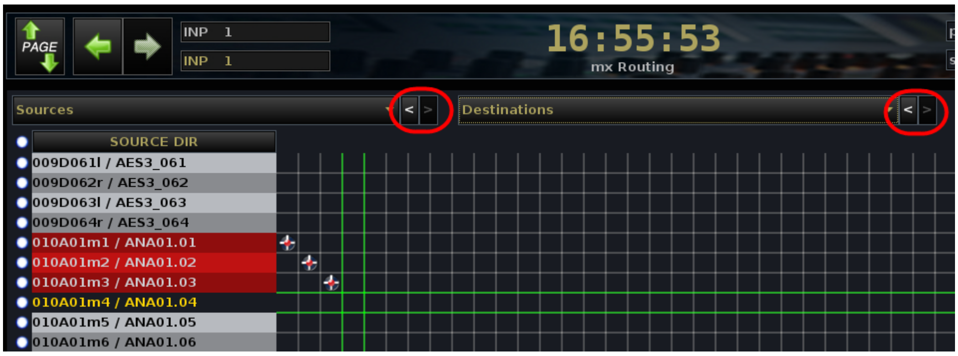
2. And press the left select button to make (or unmake) the connection.

The route is made as indicated by a red and white cross.

3. You can choose to display *only* connected signals by selecting the **Hide Unconnected Signals** checkbox.

4. You can choose to view a particular source or destination **Directory**, by clicking on the drop-down **Sources** (or **Destinations**) list - the available Directories and Subdirectories are identical to those found in the Signal List:

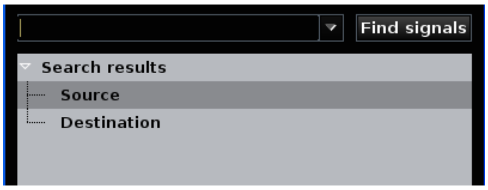
Use the on-screen next and previous directory buttons to quickly navigate to recent directory selections:



Search Signal

This function is *only* available from the **mx Routing** display (it is not available from the **Signal List**) and allows you to search for a signal by name or label. For example, you may suspect that a CD player is connected to the system but do not know its directory:

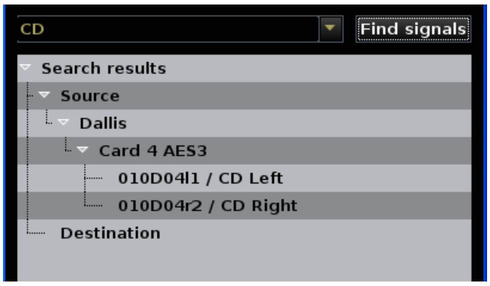
1. Right-click anywhere within the matrix grid, and select **Search Signal** to open the **Signal find** pop-up window:



2. Type in the name or user label of the source (or destination) you wish to locate – in our example, **CD**.

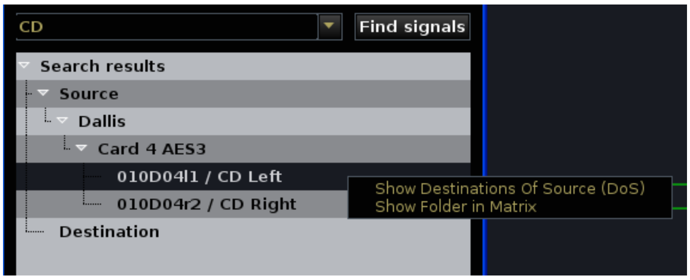
3. Then select **find signals**.

The system searches the system name and user label for all matching text strings – in our example two sources named CD Left and CD Right have been found:



4. Now select one of the results and right-click:

- Use **Show Destinations of Source (DoS)** to view all connections made from the source.
- Or, **Show Folder in Matrix** to open the source directory.



Preparing Signal Routing (the Take Button)

The **mx Routing** display allows you to prepare a set of connections and then action them simultaneously – for example, to route 8 returns from a digital effects unit to 8 channels all from one button press.

1. *BEFORE* you make or unmake any connections, select the **Prepare Routing** checkbox on the left of the display.

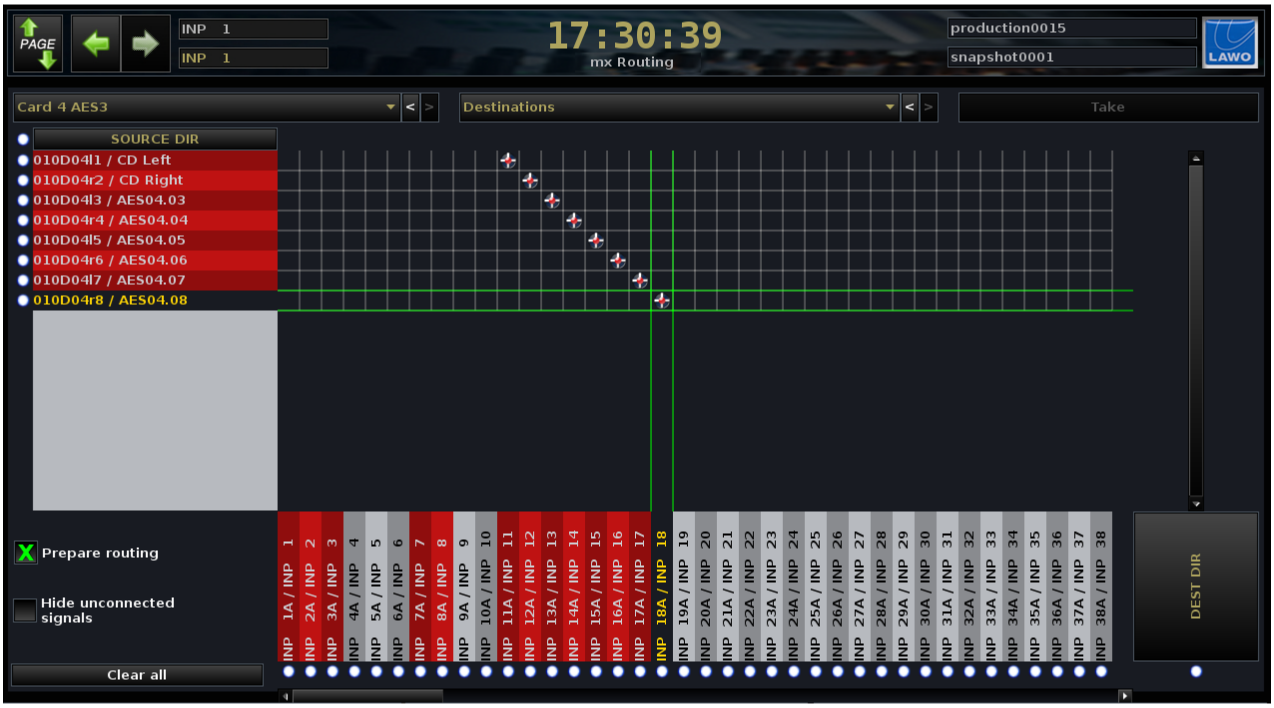
   This puts the display into 'prepare' mode.

2. Now make (or unmake) the connections – in our example, AES returns to INP channels 11 to 18.

   At this stage, the connections have only been prepared and are not yet active; therefore they are displayed with a different icon:

3. When you have completed the prepared routes, select the **Take** button at the top right of the display.

All prepared connections (and disconnections) are actioned, and the icons change state to reflect the routes made:



4. You can now prepare another set of connections and action them from the **Take** button.

5. When you are finished, remember to deselect the **Prepare Routing** checkbox to return thedisplay to its normal mode of operation.

Partial Snapshots

A partial snapshot is designed to store selected routing crosspoints. For example, you could use a partial snapshot to route tone to all transmission feeds for a line check without affecting other aspects of the mix.

A partial snapshot also stores and recalls signal parameters such as mic pre-amp gain and SRC on/off for the selected sources and destinations.

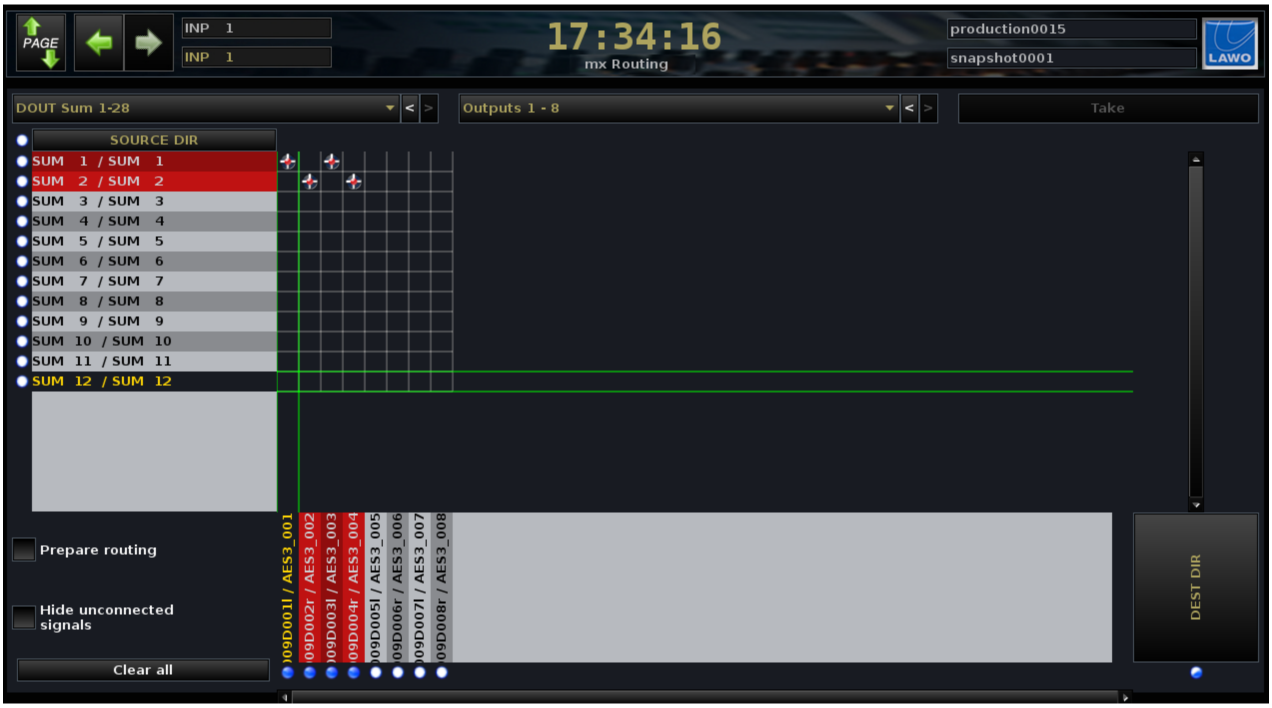
Partial snapshots are prepared from the **mx Routing** display, and then saved and loaded from the **Snapshots** display.

1. Open the mx Routing display.
2. Use the circles beside each source and destination to select which will be stored within the partial snapshot.

When a source or destination is selected, its circle turns blue.

- If you select a destination, the partial snapshot stores the route made to the destination and the destination's I/O parameters.
- If you select a source, the partial snapshot stores only the source I/O parameters.

Therefore, to store crosspoints in a partial snapshot, *always* select the destinations. In our example, we have selected four AES destinations:



Note that the half blue circle beside **DEST DIR** indicates that some signals within the current directory are selected. To select all sources or all destinations within a directory, click on this circle to urn it fully blue.

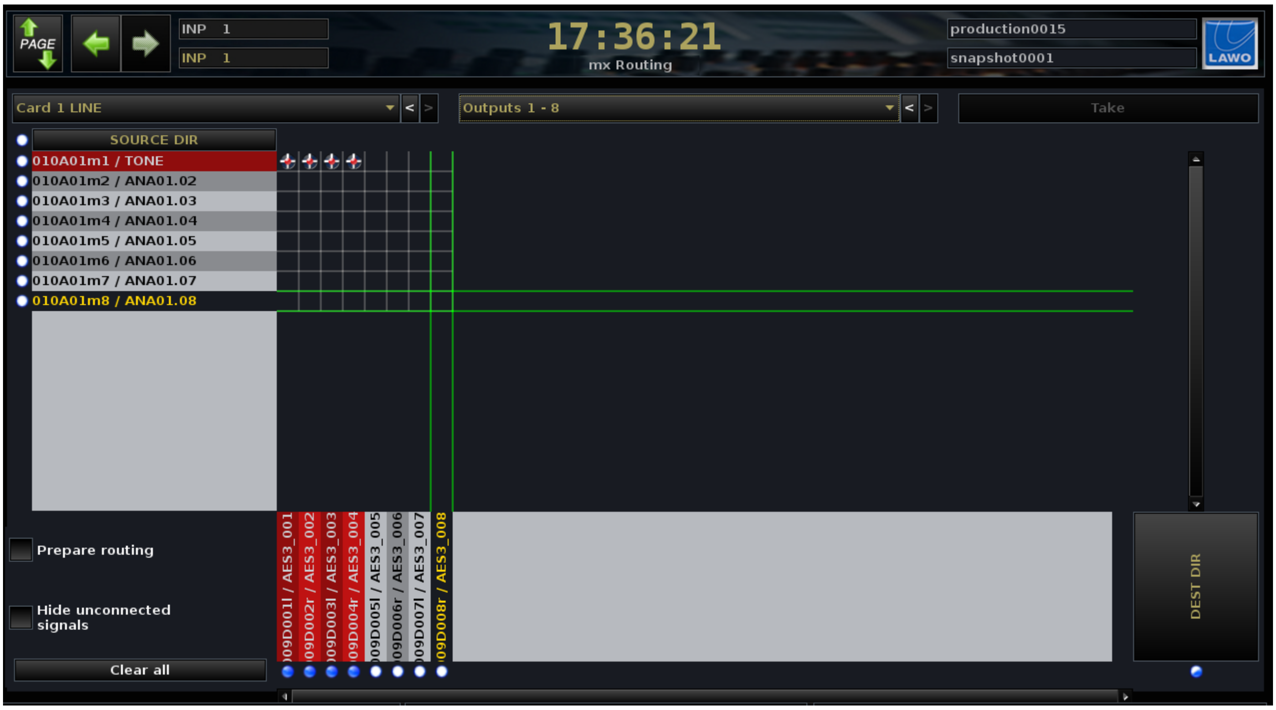
Alternatively, select **Clear All** to clear all partial snapshot selections made throughout the entire routing matrix. Use this when you wish to clear down any active selections in preparation for a new partial snapshot.

3. Now open the Snapshots display and select **Save Partial** at the bottom of the display.

The system saves the routes made to the selected destinations in a new partial snapshot:

Note that the type of snapshot is marked in the **Type** column to distinguish **partial** snapshots from **full** snapshots.

4. Return to the mx Routing display and make the new routes to your selected destinations – in our example, Tone to the transmission feeds:



5. And save another partial snapshot from the [Snapshots](#) display:

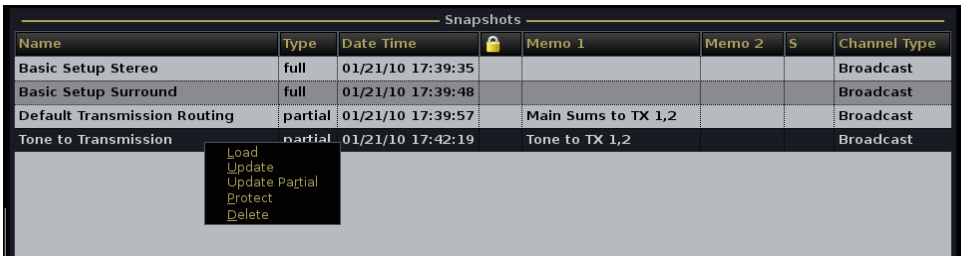
6. At any time you can now load the partial snapshots to recall routes made only to the transmission feed destinations.

Note that it is the blue circle selections when the partial snapshot is saved which defines which routes and I/O settings are stored. This allows you to save partial snapshots for different subsets of signals.

Note that you can use signal **Isolate** or **SNAP ISO** to [isolate](#) a source or destination from the partial snapshot recall.

Partial snapshots are treated in exactly the same way as full snapshots, so you can load, update, protect or delete them from the [Snapshots](#) display.

7. To update an existing partial snapshot, be sure to select **Update Partial**:

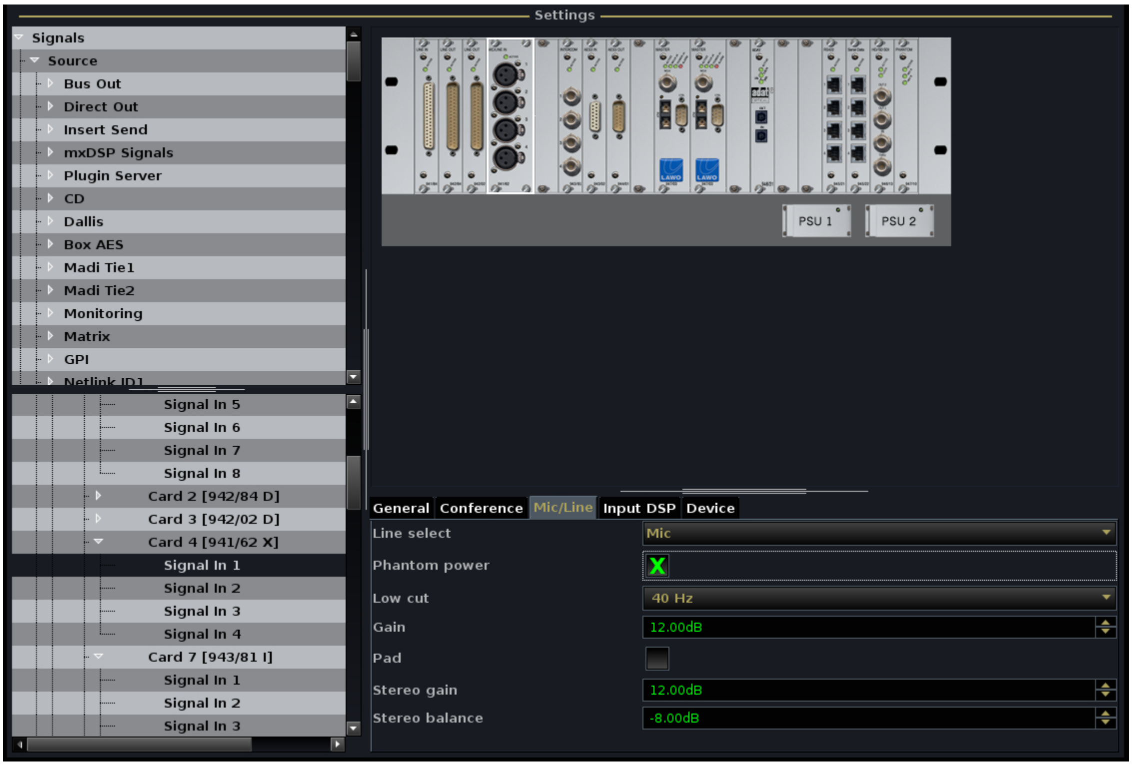


8. Remember to [save](#) or [update](#) the production in order to save snapshots permanently to the user data flashcard.

The Signal Settings Display

The **Signal Settings** display has two functions: to monitor the status of system hardware, and to set parameters for individual input and output signals.

1. Press the **SIGNAL** button, located on the SCREEN CONTROL panel, to view the **Signal Settings** display:



The two "trees" on the left of the display show the location of a signal within the **Signal List** (top) and its physical location in the **System** (bottom). Whenever a signal is selected at the top, the **System** tree follows, and vice versa.

You can open or close branches of the **Signal List** or **System** tree by clicking on the arrows or double-clicking on a directory/component name.

You can resize the different areas by clicking and dragging the grey separator bars - for example, during normal operation you might hide the **System** tree until it is needed. If information within an area is hidden, then left/right or up/down scroll bars will automatically appear.

As you select signals, a graphical representation appears in the middle of the display – in our example, we can see the DALLIS where our mic signal is connected.

If all is well with the system hardware, then the components are coloured grey. However, if there is a problem, the component will be highlighted in red, and you will see a red/white cross next to the component name in the system tree.

When running mxGUI offline, all components appear as if they are in error.

When you select an individual signal, a number of parameter tabs appear at the bottom of the display – in our example, **General**, **Conference**, **Mic/Line**, **Input DSP** and **Device**:

Note that the parameter tabs depend on the type of signal selected.

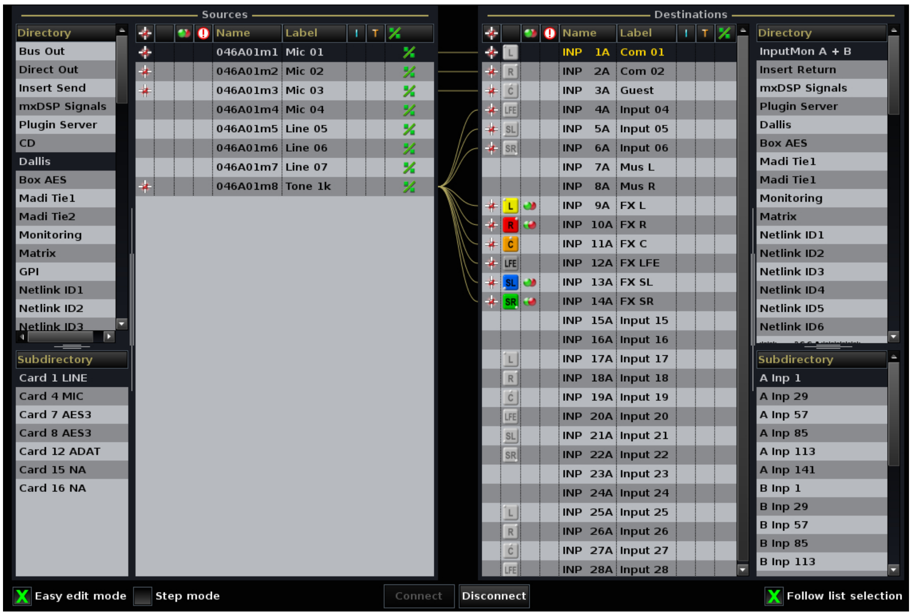
2. Select a tab to access I/O parameters for the selected signal.

3. Press the **COLLAPSE** soft key to collapse **System** tree in order to get a quick overview of system components.

Follow list selection

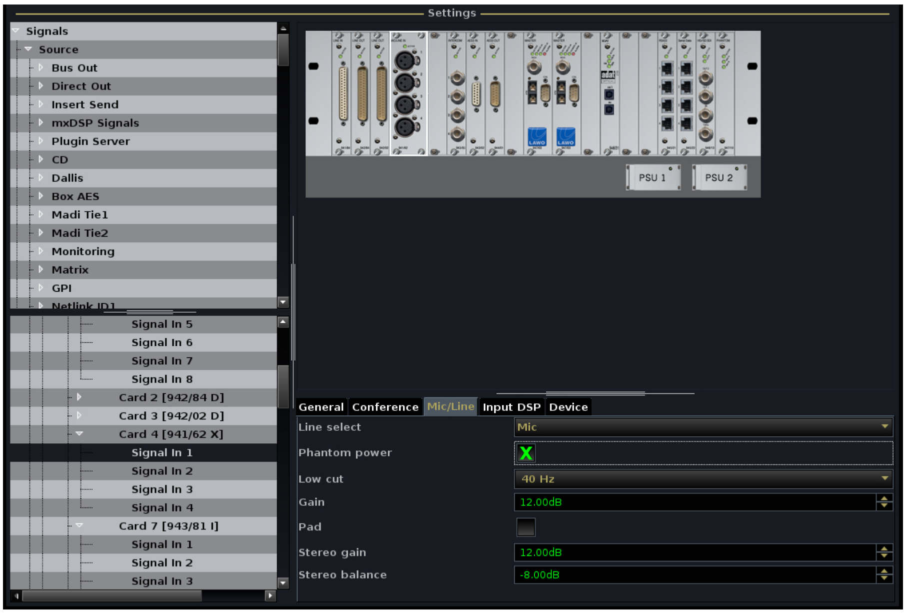
You can link the Signal List and Signal Settings displays so that when you select a signal from the **Signal List** display, and switch to **System Settings**, the selected signal follows. To do this:

1. Open the **Signal List** display and select a source – e.g. **Mic 01**:



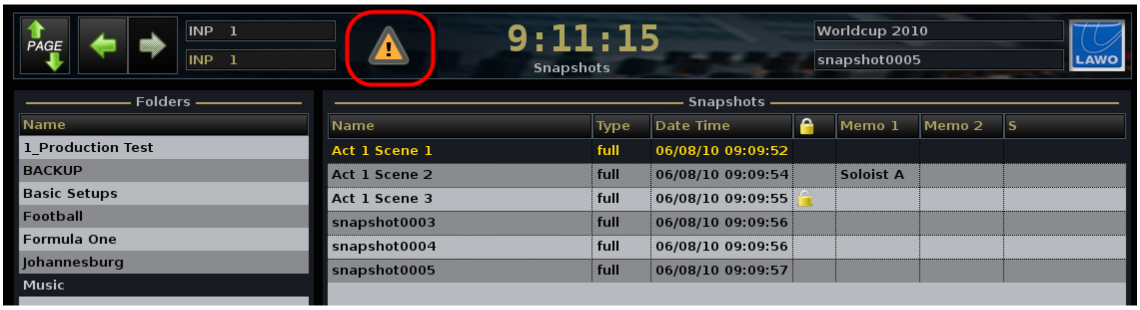
2. Make sure that the **Follow list selection** option is checked at the bottom of the display.
3. Then switch back to the **Signal Settings** display.

*The **System** tree should have automatically opened to reveal your selected source:*



Diagnosing System Errors

In the event of a component failure, a hazard warning flag appears in the title bar of the Central GUI. Note that this flag will appear at the top of any display, so you don't need to be viewing the **Signal Settings** display to monitor your system hardware:

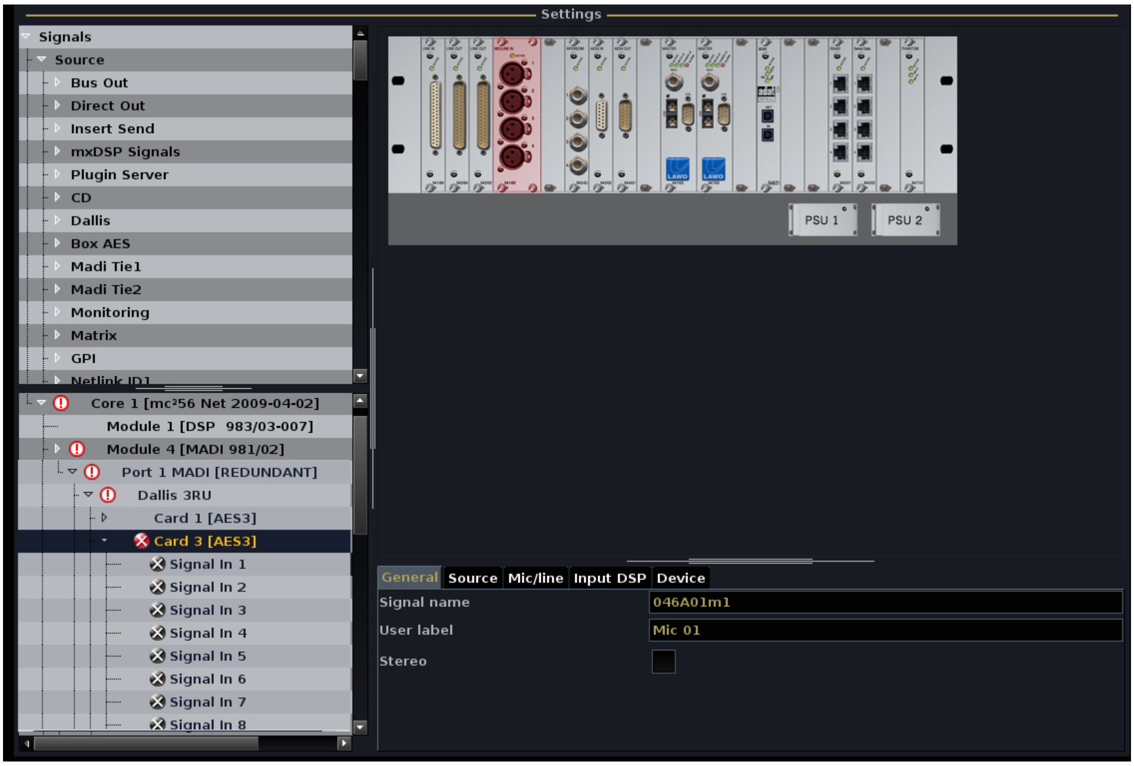


1. Press the **SIGNAL** button, located on the SCREEN CONTROL panel, to view the **Signal Settings** display.

*A red/white cross in the **System** tree, and a red highlighted card, show the location the problem.*

2. If the fault is hidden within the **System** tree, follow the red warning flags and open each branch of the tree to find the problem – in our example, a DALLIS card.

3. Open the DALLIS card further, and you will see grey/white crosses beside **Signal In 1**, **Signal In 2**, etc. These show that the AES signals are no longer available:



3. Check and replace the card if necessary.

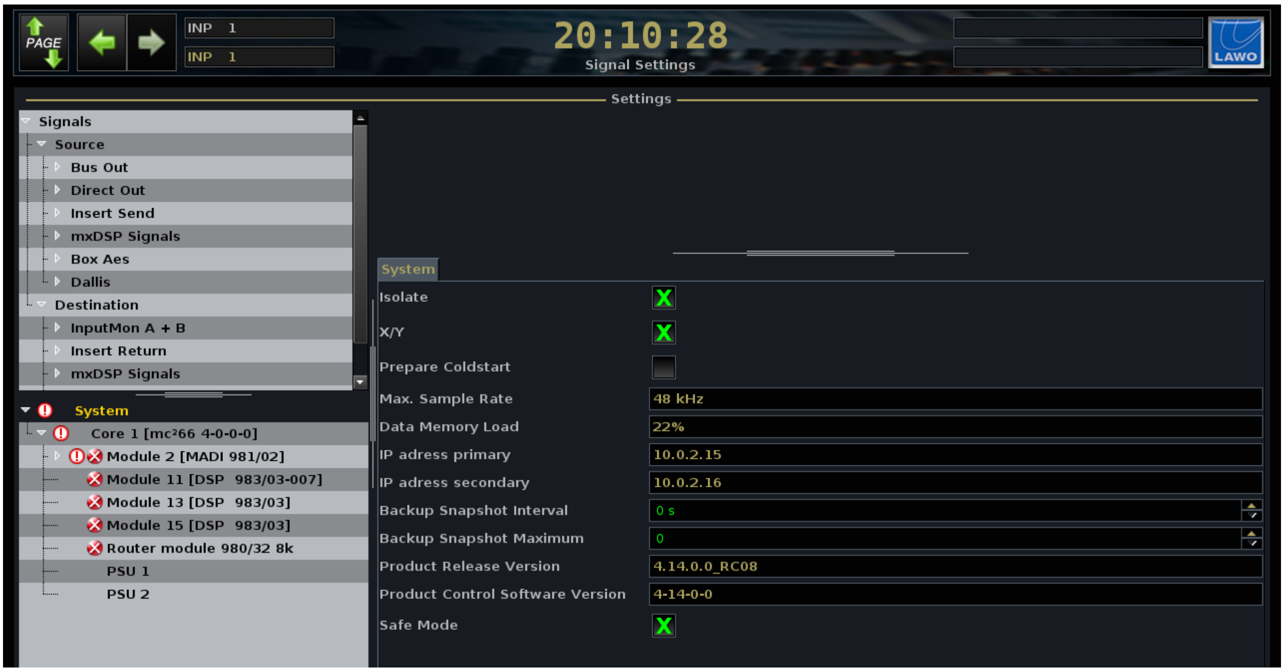
*Once all components are connected and working correctly, the red/white crosses disappear from the **System Settings** display and the hazard warning flag in the title bar is cleared.*

System Tree Structure

The **System** tree is structured as follows:

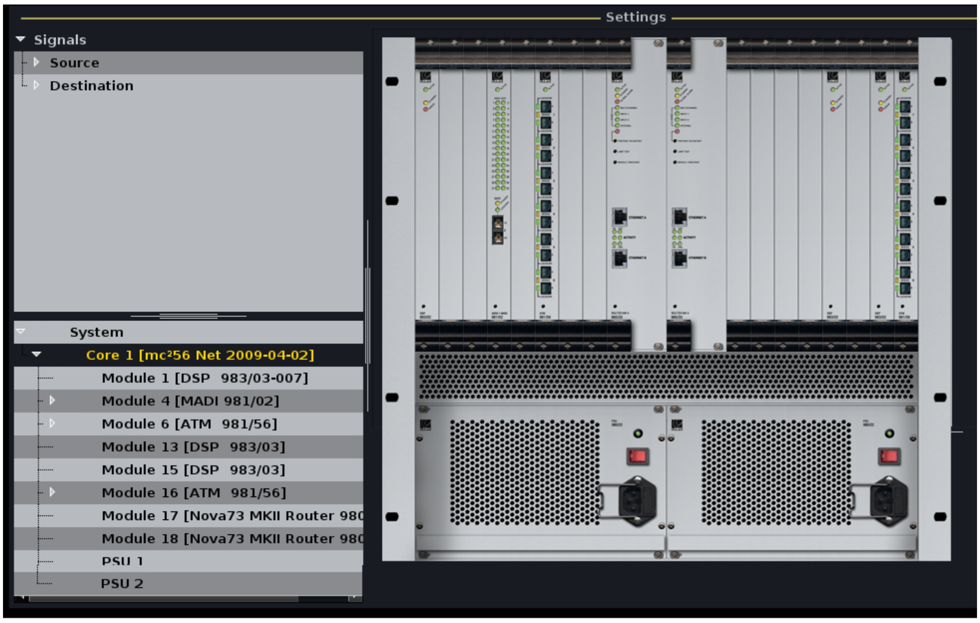
1. At the top level - **System** - you can view general information about the system. Many of these fields are duplicates of System Settings options.

> The **IP address primary** field displays the IP address of the main control system.
>
> The **IP address secondary** field displays the IP address of the redundant control system (if fitted).



2. Open the **System** to see all the **Cores** contained within your system network – e.g. **Core 1**.

3. Open **Core 1** to see all the **Modules** fitted to the core, and its power supplies – **PSU 1** and **PSU 2**:



4. And open a **Module** to view its ports and then any DALLIS units connected to those ports:

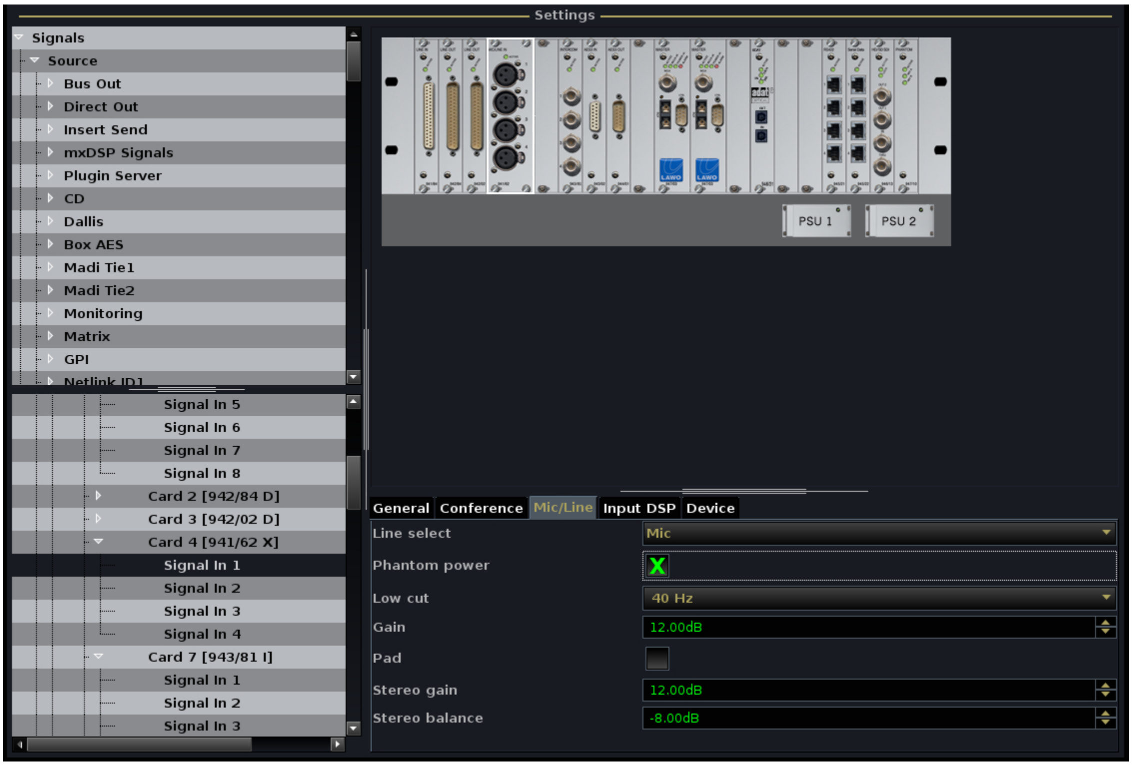
Signal Settings: I/O Parameters

Each time you select an individual signal within the **Signal Settings** display, you can adjust its I/O parameters from the bottom of the display.

1. Open up the system tree until you find the signal you wish to adjust – in our example, **Mic 01**.

A number of parameter tabs appear at the bottom of the display.

2. Select a tab to access the I/O parameters for the selected signal:



The parameters vary depending on the type of signal and whether you have selected an input or output.

You can find details for all parameters by referring to the I/O card data sheets.

Here we will cover the most common parameters.

General Parameters

These parameters appear for most signals:

- **Signal name** - the system name defined by AdminHD. This is identical to the Name field in the **Signal List** display.
- **User label** - the user label defined by the Label field in the **Signal List** display.
- **Signal Stereo** - links odd/even signals as stereo. This is identical to the Stereo option in the **Signal List** display.
- **Signal Isolate** - isolates a signal from snapshot recall. This is identical to the Isolate option in the **Signal List** display.
- **Inherited Label** (output signals only) - if the selected output is routed from a source, then this field displays the inherited user label, as defined by the Label field in the **Signal List** display.

Conference Parameters (Input Signals Only)

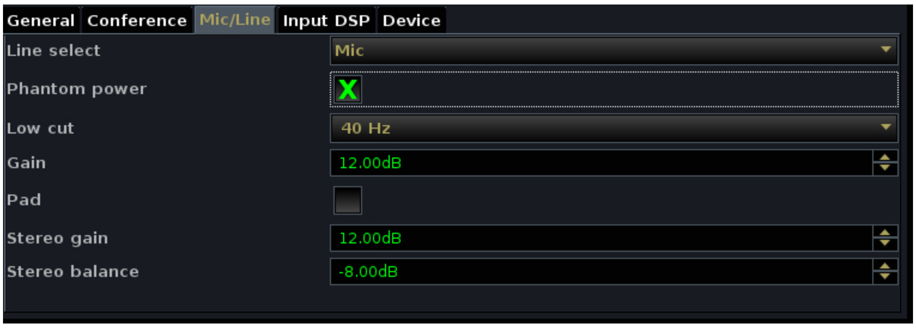
These parameters appear when an input signal is selected:

They define options for the source's mix minus (N-1):

- **Mix minus Self Monitoring** - as a default, this parameter is disabled (unchecked). Enable this parameter if you wish to add the selected signal back onto the mix minus feed. For example, if the Talent wants to hear their own microphone.
- **Mix Minus Bus** - use this field to assign an auxiliary send or track bus as the mix minus send for the selected input signal. This result is the same way as assigning a mix minus bus from the **Channel** display touch-screen. However, from this field you may assign any aux (1 to 32) or any track bus (1 to 96).

Mic/Line Parameters (Mic/Line Signals Only)

These parameters appear when an input signal from a mic/line card is selected:



They duplicate the mic/line parameters available from the INPUT Control section:

- **Line select** - selects mic or line level.
- **Phantom power** - enables 48V phantom power.
- **Low cut** - enables the high pass filter.
- **Gain** - adjusts the mic/line input gain.
- **Pad** - enables the 20dB pad.
- **Stereo gain** & **Stereo balance** - if a signal is designated as a stereo source, then you may use these fields to adjust the gain and balance of both left and right signals.

To enter a gain value (in dB), either click on the existing entry and type in a value from the keyboard, or click on the up/down arrows beside the field to increment or decrement the value in 1dB steps.

AES/EBU Inputs (AES/EBU Signals Only)

These parameters appear when a digital input signal is selected:

- **Sample Rate Converter On** - enables sample rate conversion.

Note that not all digital inputs support sample rate conversion so this option may not be available for all signals.

To make a digital path suitable for Dolby E operation, you should turn off the I/O DSP for both the input and output, and disable any sample rate conversion.

AES/EBU Outputs (AES/EBU Signals Only)

These parameters appear when a digital output signal is selected:

For an AES/EBU output signal you may adjust the Sample Rate and Wordlength.

Note that both options affect the status of the **Sample Rate Converter**, and therefore this option is for display purposes only.

To disable sample rate conversion, to make the output path suitable for Dolby E operation, set the **Sample Rate** and **Wordlength** according to the Digital Output Settings Appendix.

Sample Rate & Use System Sample Rate

The default state is that digital outputs are referenced to the console's system clock – in other words, the **Use System Sample Rate** option is checked, and the **Sample Rate** field is set accordingly:

The system's internal sample rate is set by the Sample Rate option in the **System Settings** display.

On digital outputs with sample rate conversion (SRC), you may alter the clock selection. For example, to send a 44.1kHz feed to a CDR. Note that not all digital outputs support sample rate conversion so this option may not be available for all signals.

To change the sample rate of the selected output:

1. Select **Sample Rate** and choose a drop-down menu option:

- **follow** – sets the output sample rate to follow the input sample rate from which it is routed.
- **44.1** – 44.1kHz.
- **48** – 48kHz.

On systems running at higher sample rates, you can also select:

- **88.2** – 88.2kHz.
- **96** – 96kHz.

*Selecting a different sample rate automatically unchecks the **Use System Sample Rate** option, and checks the **SRC** status flag:*

2. To reset the digital output, so that it is referenced to system clock, reselect **Use System Sample Rate**.

Word Length

The word length for each digital output defaults to 24-bit unless you select otherwise:

Note that dither is automatically applied to signals reduced to 20- or 16-bits. In addition, your wordlength selection may change the status of output sample rate conversion. See the Digital Output Settings Appendix for details.

To change the wordlength of the selected output:

1. Select **Wordlength** and choose a drop-down menu option:

- **24 bit**
- **20 bit**
- **16 bit**

I/O DSP

These parameters are available for all types of input and output signal, and control a small amount of DSP which exists on the I/O card:

To make a digital path suitable for Dolby E operation, you should turn off the I/O DSP for both the input and output, and disable any sample rate conversion.

- **I/O DSP** - enables or disables the I/O DSP. This option must be turned on (checked) for **Volume** and **Phase** to be active. Note that, for fixed gain analogue and digital inputs, I/O DSP is enabled/disabled from the **LINE/ON** button on the INPUT Control section.

- **Volume** - this field allows you to set an offset level for the selected input or output signal. It is particularly useful if you are routing a bus to multiple destinations that require slightly different line up levels, as you may use the **Volume** to adjust each individual output level.

Click to enter a value from the keyboard, or click on the up/down arrows to increment or decrement the level in 0.5dB steps.

*The **Volume** may be adjusted from -128dB to +15dB.*

Note that, for fixed gain analogue and digital inputs, **Volume** is adjusted from the **GAIN** control on the INPUT Control section.

- **Phase** - check this option to reverse the phase of the signal. Note that, for fixed gain analogue and digital inputs, **Phase** is adjusted from the **Ø** button on the INPUT Control section.

Device Parameters

- **HLSD** - this field displays the Lawo system address which is used to identify the signal within the system. It is a unique address which cannot be modified by the user.

You may need to copy and paste the HLSD when programming a Custom Function involving signals, see Entering a HLSD Address.

Tone Generator Control (Internal Tone only)

All mc²/Nova73 systems fitted with a Router Module (MKII) support four internal generator sources: two sine wave, one pink noise and one white noise.

This feature is supported from V4.12 software onwards. If you have updated your software to V4.12, then you must update the **gui_config.tcl** file using AdminHD and cold start the system to add the generator sources to a directory within the **Signal List** display. You may then make routes from each of the four generator sources in the usual manner.

From V4.24 software onwards, the first internal generator source (sine 1) may be switched to any Input or Monitor channel using the INPUT panel's TONE button.

When an internal generator signal is selected, the **Signal Generator** tab appears in the i/o parameter area of the **Signal Settings** display:

- **Frequency** - for the two sine wave generator sources, you may adjust the frequency. Click on the up/down arrows to step through the following pre-defined options:

20, **49.9**, **100**, **200**, **400**, **440**, **1000**, **2000**, **2998**, **3999**, **4987**, **6997** Hz and **10.0**, **15.0**, **20.0** kHz

Alternatively, you can type in any frequency within the parameter area.

- **Level** - this field adjusts the level of the generator signal.

You can either click on the existing entry and type in a value from the keyboard, or click on the up/ down arrows beside the field to increment or decrement the value in 1dB steps.

The level may be adjusted from 0dB to -128 dB.

SDI Parameters

SDI parameters vary depending on the card type (3G or non 3G). Please see the following Appendices for further details:

- SDI Parameters (3G SDI Card)
- SDI Parameters (non 3G SDI Cards)

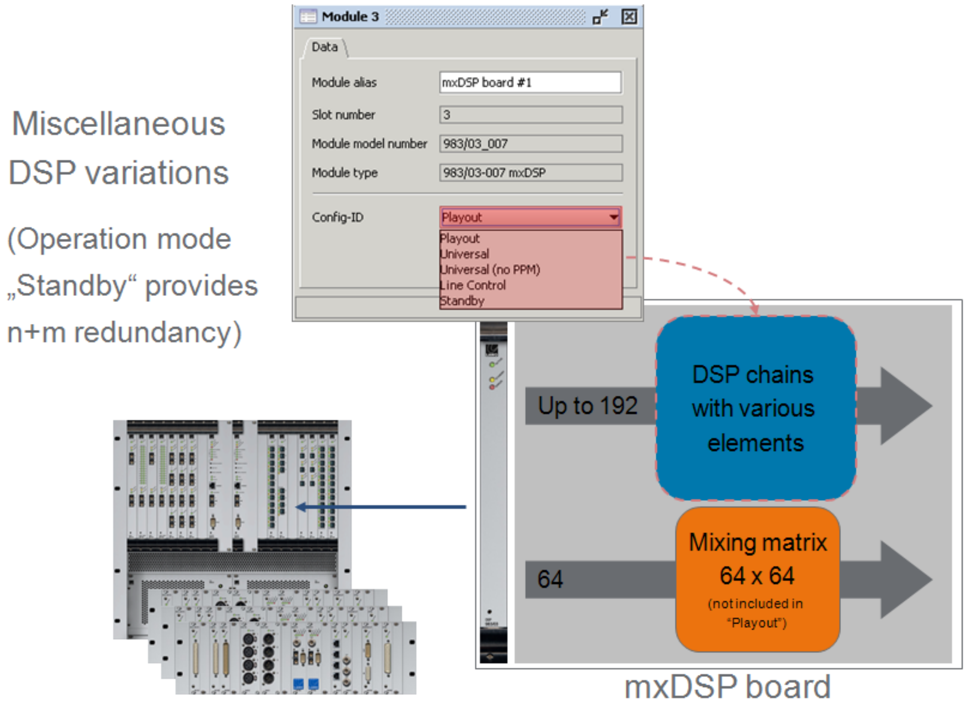
The mxDSP Settings Display

This display can be used to control the DSP settings of any mxDSP modules fitted to your system.

An mxDSP module provides a pool of DSP resource which may be applied to signal paths within the routing matrix. For example, to apply fixed DSP settings to line arrays.

Physically, each mxDSP module is identical to a normal channel DSP board and occupies one slot within the Nova73. However, rather than DSP channels, which can be assigned to the console surface, the mxDSP provides DSP "chains" which can be viewed and controlled from the **mxDSP Settings** display.

Several configuration options are supported, providing up to 192 DSP chains plus a 64 x 64 mixing matrix per module. The DSP chains are configured from various elements including level, mute, delay, EQ, etc. The number of DSP chains, and their signal flow, is determined by the AdminHD configuration:



At least one 983/03-007 mxDSP module must be configured, using AdminHD, and new software loaded to the board, before the mxDSP features become available. If not, then the **mxDSP Settings** display will appear empty. For details, please refer to the "mc²56 Technical Manual".

Note that the **Config-ID**, which determines the mxDSP mode, can be changed using AdminHD while running online.

Routing Signals to/from the mxDSP Module

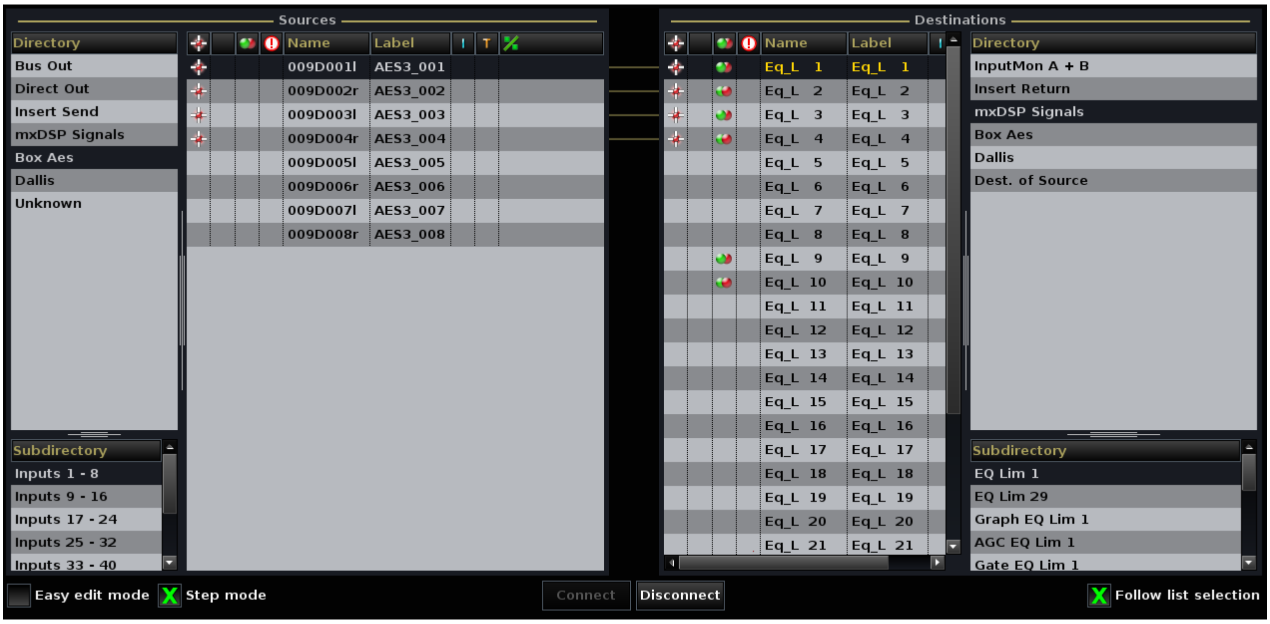
Each DSP Chain or Summing Matrix in/out may be routed from any source, and to one or more destinations using either the Signal List or mx Routing displays.

The **mxDSP Signals** appear within their own Directory. So, to route a source to an mxDSP destination (using the **Signal List** display):

1. Select your source in the usual manner.
2. Then select the destination:

- Select **mxDSP Signals** from the **Directory** list.
- Select the DSP Chain type from the Subdirectories – e.g. **EQ Lim 1**.
- Select the DSP Chain from the Destinations list – e.g. **EQ_L 1.**

3. Press **CONNECT** to make the route:



4. Then select mxDSP Settings display, and you will see the **Label** of the assigned source beside the DSP Chain.

Controlling DSP Parameters

1. Press the **MATRIX** button, located on the SCREEN CONTROL panel, to view the **mxDSP Settings** display:

On the left of the display you will see the:

- **DSP Chain Type** – this lists all the DSP chain types offered by the card. The types are pre-defined by the card configuration (defined by AdminHD). Types in grey are not supported by the current configuration.
- **DSP Chain** – this lists the individual DSP chains. Here you can name and label each chain and view its source and mono/stereo configuration.

Note that you cannot change the **Stereo** configuration of a DSP Chain from the **mxDSP Settings** display. This operation must be performed from the Signal List display.

To control routing to and from the DSP chains, use either the **Signal List** or **mx Routing** displays, see Routing mxDSP Signals.

2. Select a DSP Chain from the list to view its signal flow.
3. Then click on one of the signal flow blocks – e.g. **Par-EQ** – to display the current parameters:

4. Adjust parameter values using either the trackball (click on the up/down arrows) or keyboard.

You may adjust parameter values for any DSP block within any DSP chain.

The 64x64 Summing Matrix

Depending on the AdminHD configuration, each mxDSP module may support a 64 x 64 summing matrix.

1. Scroll through the entries in the **DSP Chain Type** list and select **Sum Matrix**.

*The display updates to show settings for the selected summing matrix – in our example, **Matrix 1**:*

For each of the 64 summing matrix inputs and outputs, you may adjust the following settings:

- **Input level**, **phase** and **mute**.
- **Output level**, **phase** and **mute**.
- **Crosspoint level** and **on/off** status.

Note that the summing matrix defaults to all levels at 0dB, all phase, mutes and crosspoints off.

On the left of the display, the **Sum Matrices** area lists all matrices configured within the system. For example, if you have several mxDSP modules, configured with a summing matrix, then you will see Matrix 1, Matrix 2, etc.

The Views list can be used to filter the number of signals in view.

Controlling the Matrix Settings

The main area of the display shows the crosspoint on/off status and levels for the signals in view:

In our example, inputs 1 to 64 run down the left hand side, and outputs 1 to 64 across the bottom.

1. Use the scroll bars to access all 64 signals.
2. Select **Inherit Label** (bottom left) to view the source and destination labels, from the Signal List, rather than the default labels shown above.

In the main grid, each box shows the matrix crosspoint level in dB. If a crosspoint is active, then its box has a heavy green outline.

The yellow outlines provide a reference to show which input, output and crosspoint will be affected by the DSP buttons on the left of the display (**Input Phase**, **Input Mute**, etc.)

The circles beside each input and output signal are used to create views.

To Adjust a Matrix Crosspoint

1. Click on the crosspoint you wish to adjust.

The yellow outline updates.

2. Turn the rotary scroller on the SCREEN CONTROL panel to adjust the level. (Or you can click on the up and down arrows or type in a new level.)

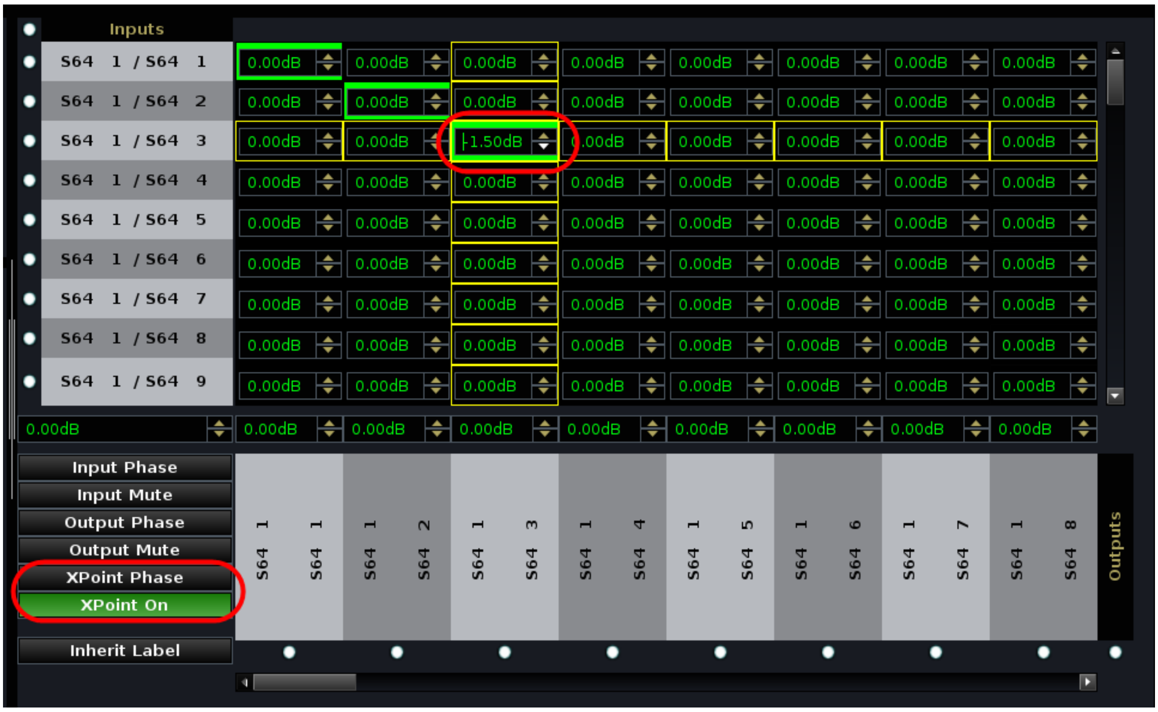
The crosspoint level may be adjusted from -128dB to +15dB.

3. Select **XPoint On** to turn the crosspoint on or off.

When active, the crosspoint box has a heavy green outline.

4. Select **XPoint Phase** to reverse the phase of the crosspoint.

When active, the button turns blue.



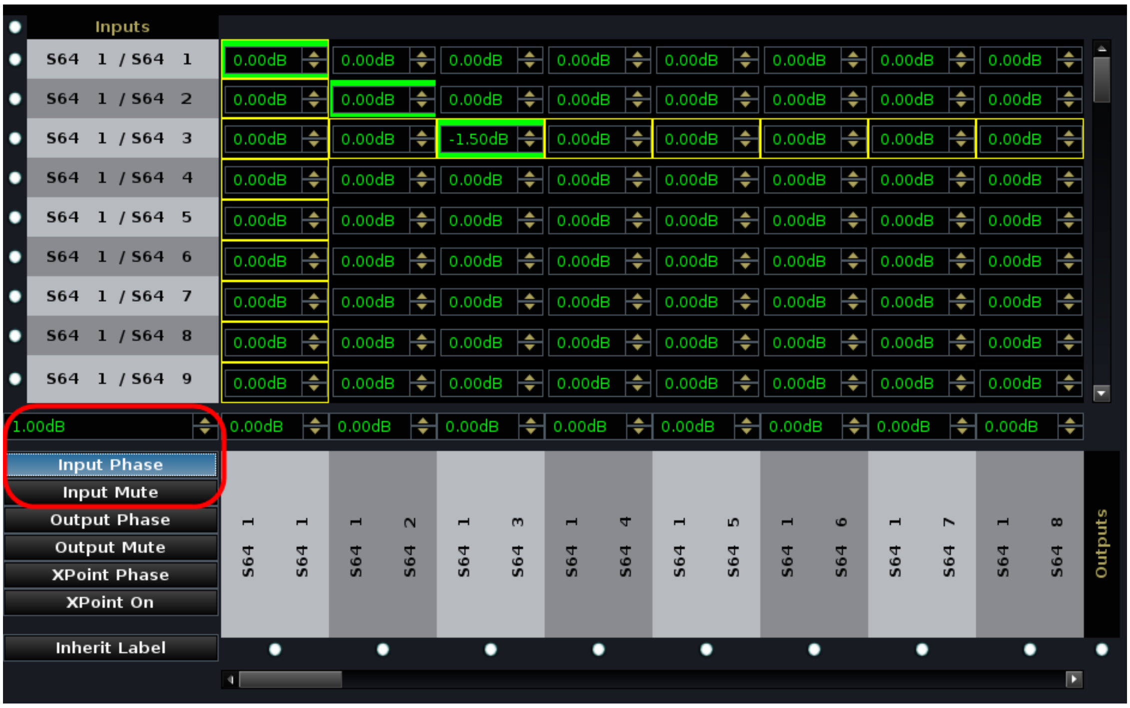
To Adjust a Matrix Input

1. Click on any crosspoint within the input row you wish to adjust – for example, input 3.
2. Use the level box below the **Inputs** list to adjust the input level.

The input level may be adjusted from -128dB to +15dB.

3. Select **Input Phase** to reverse the phase of the summing matrix input.
4. Select **Input Mute** to mute the input.

The input level box turns red if the input is muted.



To Adjust a Matrix Output

1. Click on any crosspoint within the output column you wish to adjust – for example, output 5.
2. Use the level box at the bottom of the column to adjust the output level.

Output level may be adjusted from -128dB to +15dB.

3. Select **Output Phase** to reverse the phase of the summing matrix output.
4. Select **Output Mute** to mute the output.

The output level box turns red if the output is muted.

Views

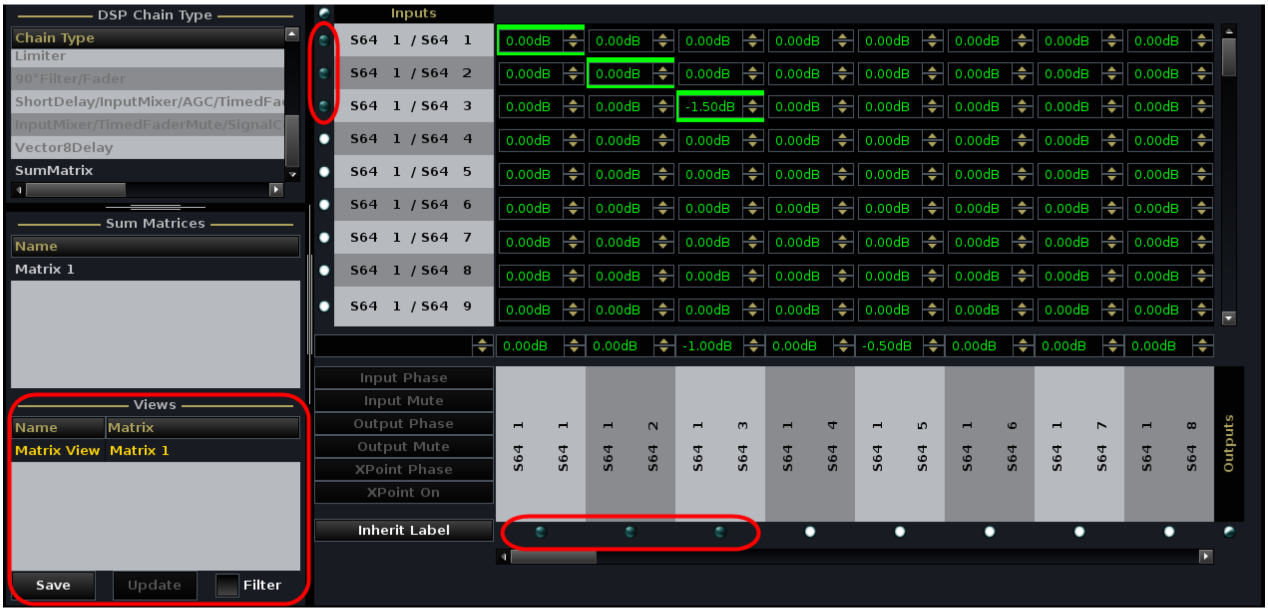
To reduce the number of signals in view to a more manageable number you can use **Views**.

1. Select the circles beside each input and output signal you wish to include within the matrix View.

   When a signal is selected, its circle turns blue.

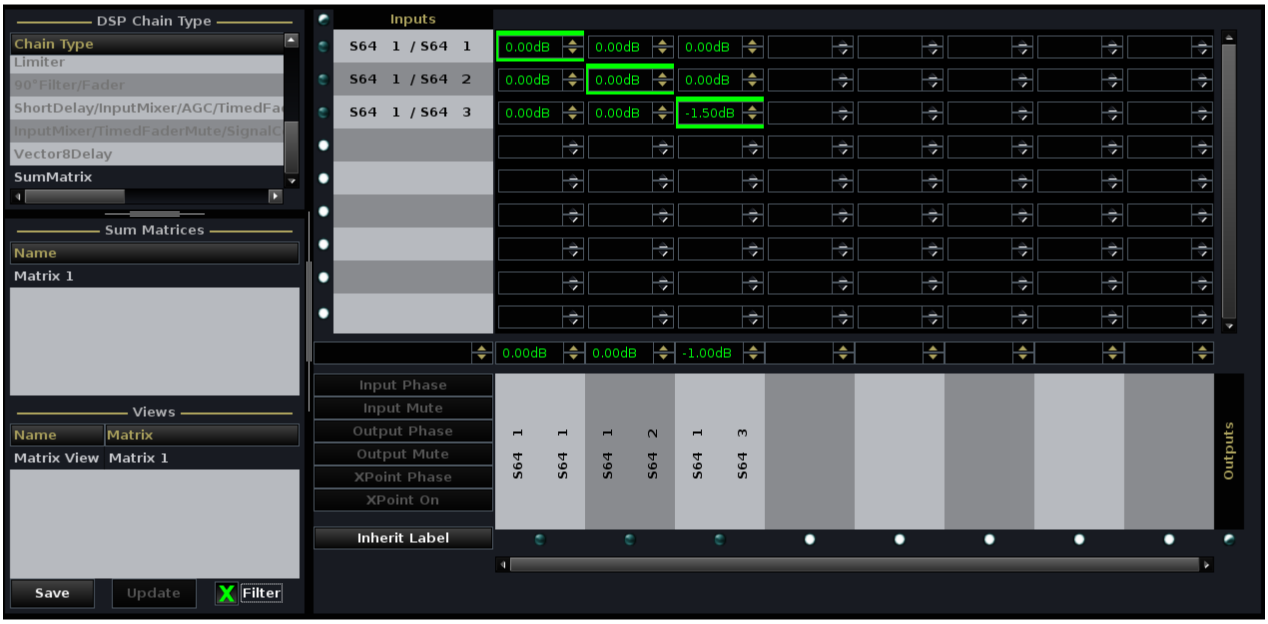
2. Now select **Save** at the bottom of the **Views** area.

   *The **Views** list updates accordingly:*



3. To apply the View, select the checkbox beside **Filter**.

   *When the **Filter** checkbox is active, the crosspoint grid only shows signals stored within the selected **View**:*



4. To return to all signals, deselect the **Filter** checkbox.

You can store as many Views as you wish, and perform the following operations by right-clicking on a **View**:

- **Update** – select a different set of signals and click on Update to update an existing View.
- **Delete** – deletes the selected View.
- **Rename** – renames the selected View.
- **Reload** – reloads the selected View.

Note that the half blue circle beside **Inputs** and **Outputs** indicates that some but not all signals are selected:

- To select all signals, click on this circle to make it fully blue.
- To deselect all signals, click it again to make it fully white.

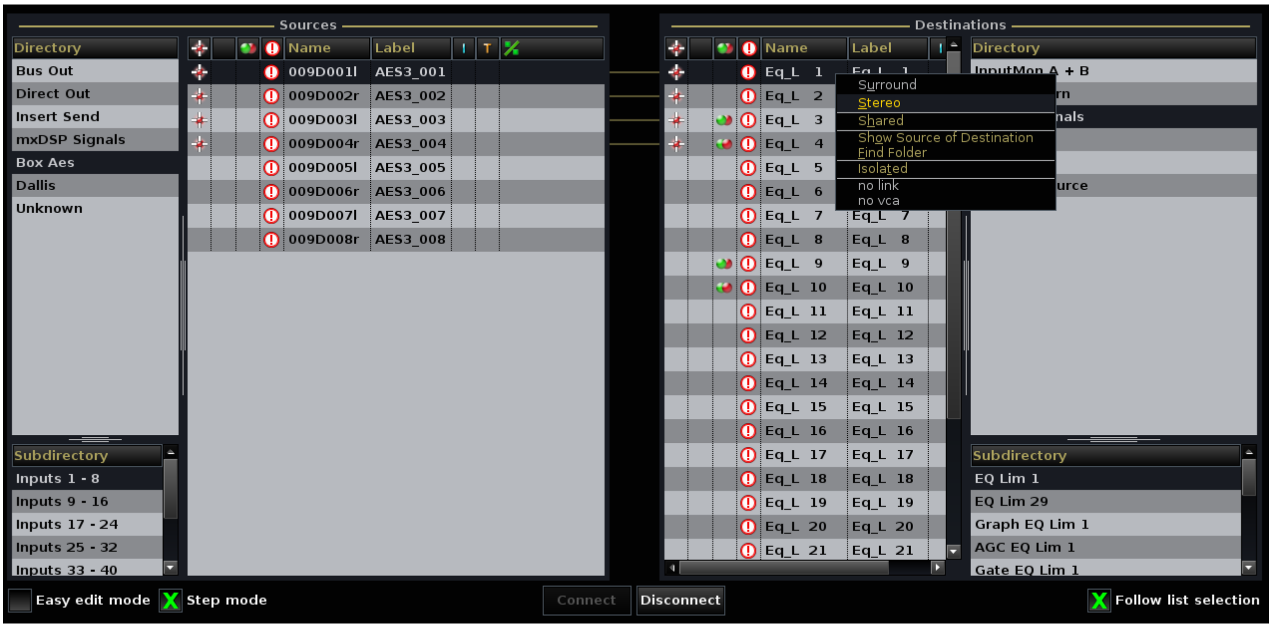
Stereo Configuration

An odd/even pair of DSP Chains can be configured for stereo operation.

Note that surround configuration is not supported.

This operation is performed from the Signal List display:

1. Select the DSP chain you wish to make stereo and right-click:



2. Select the **Stereo** option.

Green/red circles appear beside the DSP chains to indicate that they are now linked for stereo.

3. Return to the **mxDSP Settings** display and you will see the stereo status indicated beside the DSP Chain and within the main **Settings** area:

Saving and Loading mxDSP Settings

The settings for each mxDSP module are stored within snapshots and productions, so remember to save or update a production to save any changes.

You can isolate all mxDSP signals so that they will not be affected by a snapshot load using the **mxDSP** Global Snapshot Isolate option.

The Downmix Display

The **Downmix** display provides on-screen control of any downmix matrices supported by your system. For example, if you have a 5.1 surround to stereo downmix, then you may adjust how much level from the front LR, Centre, LFE and rear LR channels feed the stereo output.

1. Press the **MATRIX** button, located on the SCREEN CONTROL panel, to view the **Downmix** display.

In the top half of the display you will see a list of all available downmixes for your system. In our example, we have one downmix named ***5.1 Mains****:*

Note that to support downmix matrices, the required DSP resources must be fitted to your system's hardware and configured using AdminHD. If not, then the **Available Downmixes** list will be empty. For details, please refer to the "mc²56 Technical Manual".

The downmix matrix **Name** is also defined by AdminHD.

2. You may edit the Downmix **Label** field to apply a user name to the matrix.

User labels are inherited into the Signal List and mx Routing displays.

Controlling Downmix Parameters

1. Select a downmix from the **Available Downmixes** list to view its parameters.

Our example shows an 8 x 8 matrix which is configured to produce 4 stereo outputs (Downmix 1 to 4) from a 5.1 input:

2. Using the trackball or console keyboard, you may adjust the following parameters for Downmix 1 to 4:

- **Front** level – from inputs 1 (Left) and 2 (Right).
- **Center** level – from input 3 (Centre), unless **Alt Center** is active, see below.
- **LFE** level – from input 4 (LFE).
- **Surround** level – from inputs 5 (Surround Left) and 6 (Surround Right).
- **Alt Center** – use this option to replace the Center input with an alternate centre channel:
  - **Off** = no alternate centre is used. Input 3 feeds the Centre channel.
  - **1** = input 7 replaces input 3.
  - **2** – input 8 replaces input 3.

You can use this option to generate a clean feed or alternate language downmix. For example, Downmix 1 might be your main programme, Downmix 2 the clean feed, and Downmix 3 an alternate language version.

- **Output** level – adjusts the output level for the stereo downmix.

3. Select **Reset levels** to reset the downmix to its default parameters, and **Yes** to confirm.

All parameters are reset to the default values stored in the AdminHD configuration.

Saving and Loading Downmix Settings

The settings for each Downmix matrix are stored within snapshots and productions, so remember to save or update a production to save any changes.

By default each matrix is isolated so that it will not be affected by a snapshot load. You can adjust this by selecting the **Isolate** box beside the matrix name in the **Downmix** display:

Chapter 9: System Configuration

Introduction

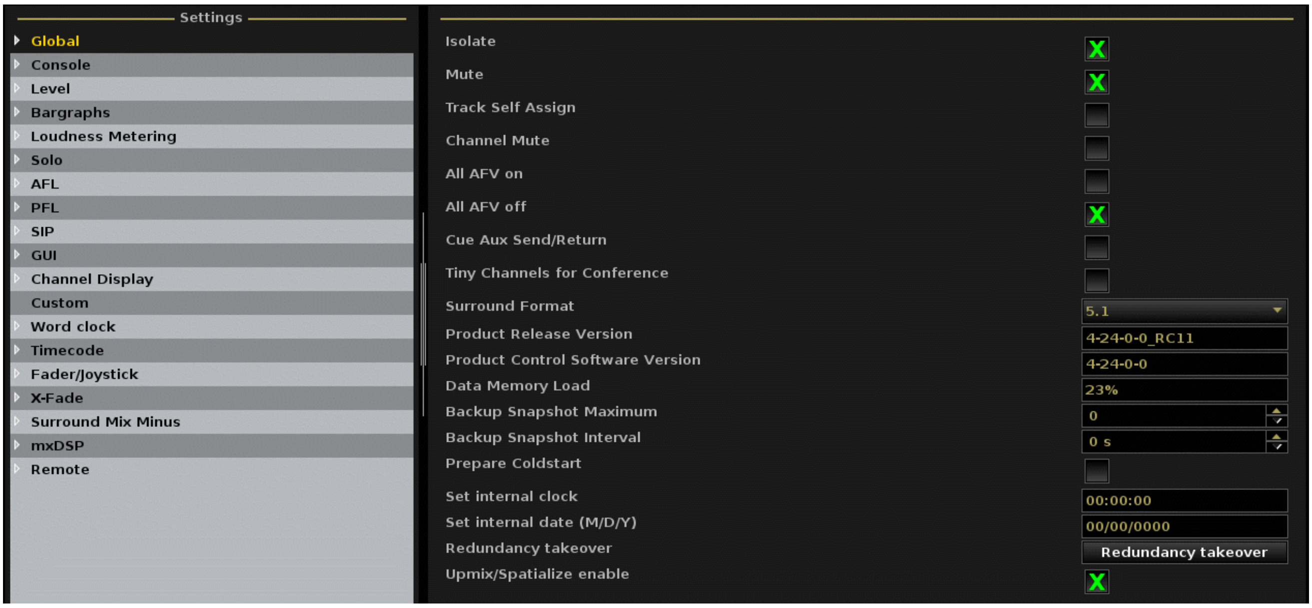
This chapter deals with the **System Settings** and **Custom Functions** displays. It also covers the system hardware, redundancy, sample rate and synchronisation, and procedures for system shutdown and restart.

Topics covered are:

The System Settings Display

The **System Settings** display configures all the system options which may be modified by the user. These options are stored and recalled with productions, but not snapshots.

1. Press the **SYSTEM DSP** button, located on the SCREEN CONTROL panel, to view the **System Settings** display:



On the left you will see a list of topics.

2. Using the trackball or navigation controls select a topic – for example, **Global**.

*The right hand side of the display updates to show a list of options within the selected topic – for example, **Isolate**, **Mute**, **Track self assign**, etc.*

3. Depending on the option it can be modified as follows:

- **Checkbox on/off** (e.g. **Isolate**) - use the trackball to select the checkbox beside the option.

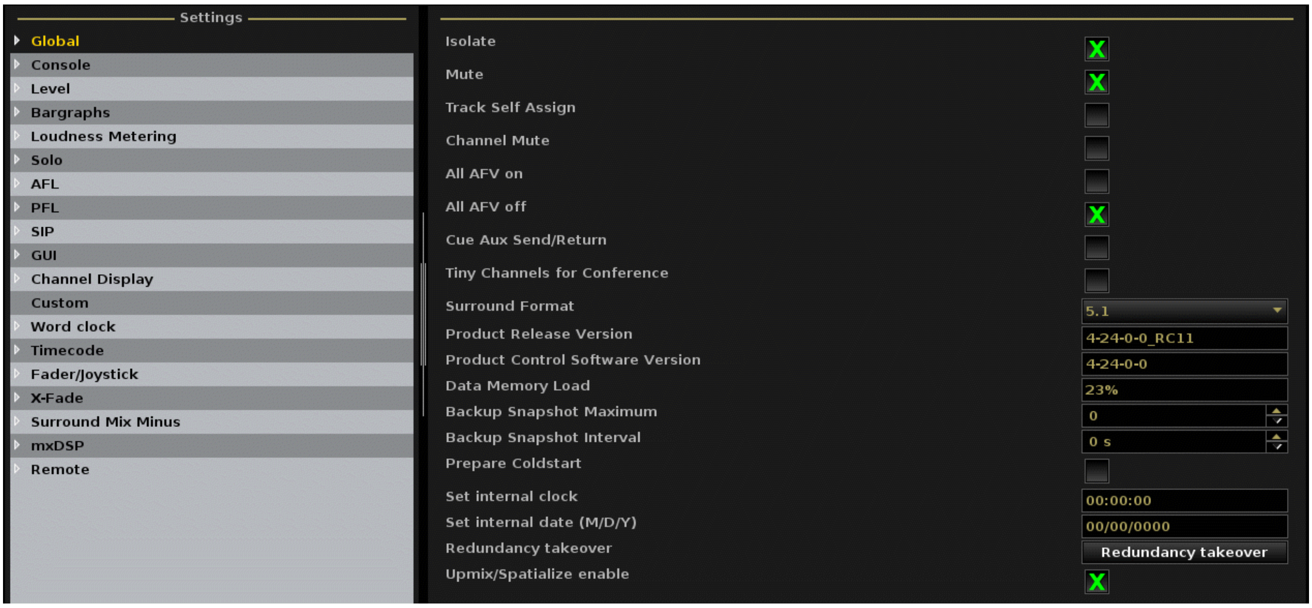
*A green cross appears when the option is enabled – for example, **Isolate** is **ON**.*

- **Drop-down selections** (e.g. **Surround format**) – using the trackball select an option from the drop-down list.
- **Numeric Entries** (e.g. **Backup Snapshot Maximum**) – some options require a number to be entered. You can click on the existing entry and type in a value from the keyboard; or click on the up/down arrows beside the number to increment or decrement its value; or select the option, press the **SET** soft key and then use the rotary scroller to increment or decrement the value.

If you hover the cursor above each option name, you will see a 'Tool Tip'. This is a helpful description which acts as a brief reminder of the option's function.

Global Options

Select the **Global** topic to set the following global options:



Isolate

This option enables or disables the SNAP ISO (Snapshot Isolate) buttons across the console:

- **Isolate (on)** – you can select **SNAP ISO** (Snapshot Isolate) buttons to isolate channels from a snapshot or automation load.
- **Isolate (off)** – prohibits selection of **SNAP ISO** buttons across the console; any existing **SNAP ISO** selections will be cleared. Use this mode to ensure that all stored parameters are recalled to all channels from any snapshot or automation load.

Mute

This option enables or disables the fader strip MUTE buttons across the console:

- **Mute** (on) – you can select **MUTE** buttons to mute/cut a channel.
- **Mute** (off) – prohibits selection of **MUTE** buttons across the console; any existing **MUTE** selections will be cleared. Use this mode to prevent accidental muting of channels during a live production.

Track Self Assign

This option determines whether a monitor channel can be assigned to its associated track bus. For example, whether monitor channel 8 can be assigned to track bus 8:

- **Track Self Assign** (on) – allows monitor channel x to be assigned to track bus x. This mode is designed for non-multitrack applications where you wish to route to track busses from any channel.
- **Track Self Assign** (off) – prohibits the assignment of monitor channel x to track bus x; any existing assignments to associated track busses will be cleared. Use this mode when working with a multitrack machine to prevent monitor channel x routing to track bus x and generating feedback.

Channel Mute

This option determines where in the signal flow a channel is muted when the MUTE button is selected:

- **Channel Mute** (on) - the **MUTE** button mutes the channel after the input mixer. In this mode all channel outputs including pre-fader sends are muted. Note that, from Version 4.24 software onwards, PFL is NOT muted to enable pre-fader listen.
- **Channel Mute** (off) - the **MUTE** button mutes the channel after the fader. In this mode only post fader outputs are muted, pre fader sends remain active.

All AFV On/Off

This option sets AFV (Audio Follow Video) to either on or off across all channels:

- **All AFV on** (on) – AFV is switched on across all channels.
- **All AFV off** (on) – AFV is switched off across all channels.

Cue Aux Send/Return

This option determines the behaviour of auxiliary sends 17 to 32 when assigned from monitor channels.

- **Cue Aux Send/Return** (on) – aux sends 17 to 32 can be switched between send and return. This mode is ideal for cue feeds when overdubbing.
- **Cue Aux Send/Return** (off) – aux sends 17 to 32 return to normal aux send operation and can be switched post fader, pre fader or pre EQ.

Tiny Channels for Conference

From V4.24 software onwards, this option determines whether tiny input channels can feed onto mix minus (N-1) sends:

- **Tiny Channels for Conference** (on) – tiny input channels can feed mix minus sends.
- **Tiny Channels for Conference** (off) – tiny input channels cannot feed mix minus sends.

When using a Recording Channel DSP configuration, and creating a mix minus from tiny channels, you *MUST* use auxiliary busses for mix minus sends (as track bus conference facilities are not supported from tiny DSP channels).

Surround Format

This option defines the global surround format used for surround channels, pan laws and monitoring.

Use the drop-down menu to select an option:

- **4.0 –** L, R, C, S for Dolby ProLogic.
- **5.1 –** L, R, C, LFE, Ls, Rs for Dolby Digital and DTS.
- **6.1** – L, R, C, LFE, Ls, Rs, Cs for Dolby Digital EX and DTS ES.
- **SDDS** – L, R, Lc, Rc, C, LFE, Ls, Rs for 7.1 SDDS.
- **7.1 –** L, R, C, LFE, Lm, Rm, Ls, Rs for DTS-HD.

Product Release and Control Software Versions

These fields are for display purposes only, and tell you the software versions running on your system. Note that there are two different releases, both important when reporting software versions to a service engineer:

- **Product Release Version** – this is the release version of your product software.
- **Product Control Software Version** – this is the release version of the control system software.

Data Memory Load

This field is for display purposes only, and indicates the amount of used data storage space (%).

Backup Snapshot Maximum & Interval

These options define the system's backup snapshots

- **Backup Snapshot Maximum** – sets the number of backup snapshots which will be automatically stored before the first backup snapshot is overwritten. The number may be adjusted from 0 to 1000. Enter 0 to turn off the backup snapshots function.
- **Backup Snapshot Interval** – sets the time interval between backup snapshots, and may be adjusted from 60 seconds to 24 hours (86400s).

Prepare Cold Start

This option sets whether the system will cold or warm start on the next power-on:

- **Prepare Coldstart** (on) - the system will cold start. This means that no user data is loaded. Use this option if you wish to clear all user settings from the system.
- **Prepare Coldstart** (off) - the system will warm start. This means that the console is restored with same settings as before the power off.

Note that following a restart this option is always reset to off. This ensures that by default, warm start data is loaded at the end of every power-on or restart.

Set Internal Clock

Using this option you can set the internal clock.

1. Type in the time you wish to set and then press Enter.

   A confirmation pop-up appears.

2. Select **OK** to confirm.

   The new time is set.

The time may be displayed on the Central GUI by adjusting the Time Display option.

Set Internal Date

Using this option you can set the internal date.

1. Type in the date in the format: Month/Day/Year (e.g. 25/01/2010) and then press Enter.

A confirmation pop-up appears.

2. Select **OK** to confirm and the date is set.

The date stamp is used when saving user data such as Productions and Snapshots.

Redundancy Takeover

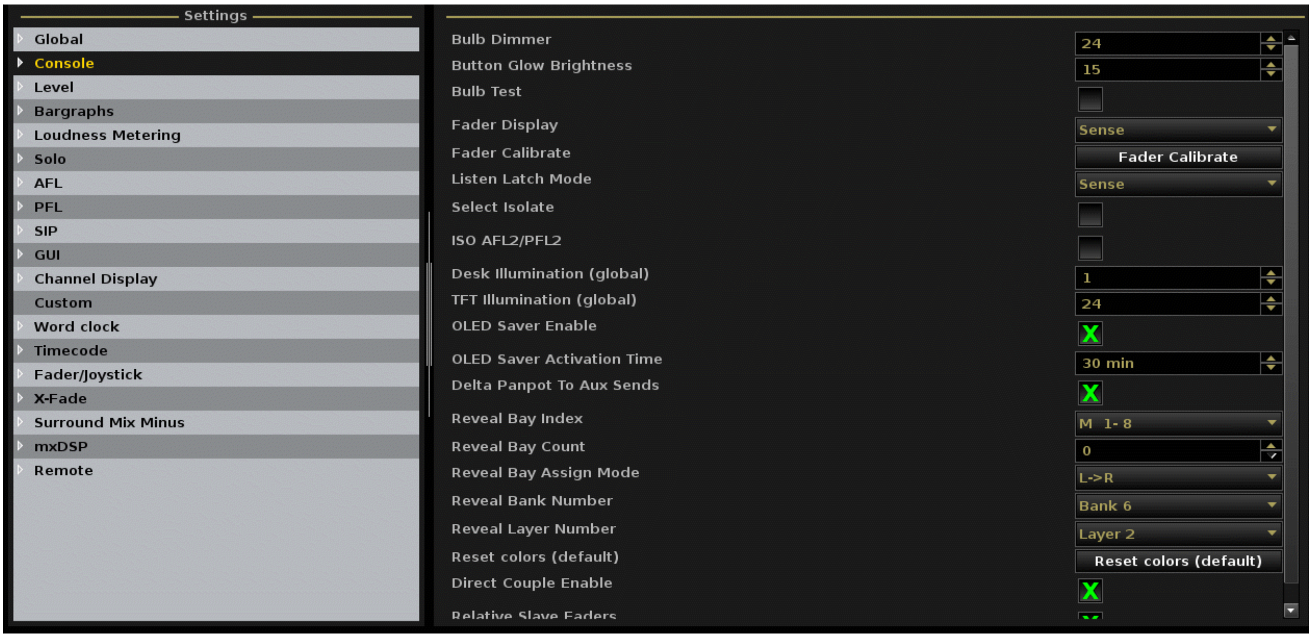
Use this option to force a manual takeover from the redundant control system (if fitted). See Redundant Router Module and Control System.

Upmix/Spatialize Enable

Use this option to enable or disable the AMBIT upmix and spatialize module for 5.1 surround channels.

Console Options

Select the **Console** topic to set the following console surface options:



Bulb Dimmer

This option sets the brightness of all LEDs, bulbs and text displays across the console surface.

The brightness may be set from 0 = low to 31 = high.

Note that there is a fixed relationship between the **Bulb Dimmer** value and Button Glow Brightness.

Button Glow Brightness

When this option is enabled, some fader strip buttons (in their off state) are dimly lit according to the channel colour code. This makes it easy to identify which channels are assigned to fader strips, especially useful in low-light conditions. The fader strip buttons affected are A/B input switching, Free Control on/off buttons, the four channel user buttons, AFL and PFL.

To enable button-glow, set the **Button Glow Brightness** to any value > **0**. We recommend **20** as a good starting number.

Note that there is a fixed relationship between the **Button Glow Brightness** and **Bulb Dimmer** value. Therefore, if you adjust the Bulb Dimmer setting it may affect **Button Glow Brightness**.

To disable button-glow, set the **Button Glow Brightness** to **0**.

Note that this function is not supported by classic mc²56 (the option can be adjusted, but performs no action).

Bulb Test

This option lights all LEDs, bulbs and text displays across the console surface in order to check for defects:

- **Bulb Test** (on) - enters the test mode. All LEDs, bulbs and displays will illuminate across the console. Note that all dual coloured bulbs should be orange. If not, then this indicates that either the red or green bulb is faulty.
- **Bulb Test** (off) - exits the test mode.

Fader Display

This option determines whether the fader label displays show channel levels when faders are touched:

- Select **Sense** to enable the fader sense mode. The displays show the channel name or label until a fader is touched; while the fader is touched the display shows fader level in dB.
- Select **Name** to disable the fader sense mode. Use this mode if you *always* want to view the channel name or label, even while faders are touched.

Fader Calibrate

This option is used to calibrate the faders on the console.

1. Select **Fader Calibrate** to calibrate all faders.

Each fader across the console opens and closes to calibrate.

Listen Latch Mode

This option defines whether AFL monitoring actioned from the LISTEN buttons is momentary (sensing) or latching:

- Select **Sense** for momentary AFL. The output of the LISTEN module feeds the AFL bus as long as you touch the control. Once the control is released, AFL is cancelled.
- Select **Latch** for latching AFL. AFL latches on and remains on even if you release the control. AFL is cancelled when you touch a control within another module or deselect the LISTEN button.

Select Isolate

This option determines whether fader select (**SEL**) buttons within isolated fader bays update the channel in access:

- **Select Isolate** (on) – the **SEL** buttons within isolated bays do NOT update the channel in access. Use this mode when you want isolated bays to work independently from the rest of the console. For example, when one engineer is working on an isolated fader bay and another with the rest of the console.
- **Select Isolate** (off) – the **SEL** buttons within isolated fader bays do update the channel in access. This mode is ideal for single operator use where you wish the channel in access to follow selections within isolated fader bays.

ISO AFL2/PFL2

This option is used to route AFL and PFL selections made within isolated fader bays onto a second AFL and PFL bus:

- **ISO AFL2/PFL2** (on) – enables the second AFL/PFL output; any AFL and PFL selections made from an isolated fader bay route to the AFL2 and PFL2 outputs. AFL and PFL selections from the rest of the console route to AFL1 and PFL1.
- **ISO AFL2/PFL2** (off) – disables the second AFL/PFL output; all AFL and PFL selections, including those within isolated fader bays, route to AFL1 and PFL1.

You can find the **AFL2** and **PFL2** bus outputs in the Signal List display, under **Bus Out**:

Depending on your monitoring configuration, you may have options to switch **AFL2** and **PFL2** to the control room monitors or headphones.

Desk Illumination

This option sets the brightness for the console desk light.

The brightness may be set from 0 = off to 15 = high.

TFT Illumination

This option sets the brightness for the TFTs.

The brightness may be set from 0 to 32 = high.

OLED Saver Enable and Activation Time

These options enable the OLED (text displays) screensaver and set the time in minutes before the screensaver is activated. The time may be set from 5 to 60 minutes. The screensaver will deactivate as soon as you touch any fader, rotary control or press a button.

Use this mode to prolong the lifetime of the OLED (text displays).

Delta Panpot to Aux Sends (Aux Panning Link)

This option determines whether aux send panning follows channel fader panning, across the console, for stereo aux sends:

- **Delta Panpot to Aux Sends** (on) – aux panning follows channel fader panning.
- **Delta Panpot to Aux Sends** (off) – aux panning may be set independently from channel fader panning.

Reveal

These five options determine where the VCA slave faders appear when the REVEAL function is active.

The first three options determine the location and number of fader strips to be used, and whether faders are revealed from left to right or right to left:

- **Reveal bay index** – enter the first channel or main fader bay you wish to use.
- **Reveal bay count** – select the total number of fader bays you wish to use. For example, **1** will allocate 8 fader strips, **2** will allocate 16 fader strips, etc. If you enter **0**, then no fader bays are allocated, and the **REVEAL** button will perform no function.

Using more than 8 faders can be very useful if you are using **REVEAL** with normal VCA grouping. For example, by setting the Reveal bay count to **2** you will be able to reveal 16 slaves.

- **Reveal bay assign mode** – this option determines whether slaves appear from left to right (L->R) or right to left (R->L) across the defined fader bay(s).

The last two options determine which fader strip bank(s) and layer are used to implement the reveal function:

- **Reveal bank number** – selects the fader strip bank used to store revealed slaves.
- **Reveal layer number** - selects the fader strip layer used to store revealed slaves.

Whenever you put a surround VCA or normal VCA master into access, its slaves are automatically assigned to the designated "Reveal bank and layer"; the **REVEAL** button then simply flips these fader strips to the current surface. Therefore, select a bank and layer of fader strips which you do not need for your normal operation.

Reset Colours (default)

This option resets all DSP channels to their default colour codes, see Channel Colour Coding.

Direct Couple Enable

When this option is enabled (default), a couple can be created by pressing and holding the fader **SEL** buttons.

If the option is disabled, then you must use the **COUPLE** button.

Relative Slave Faders

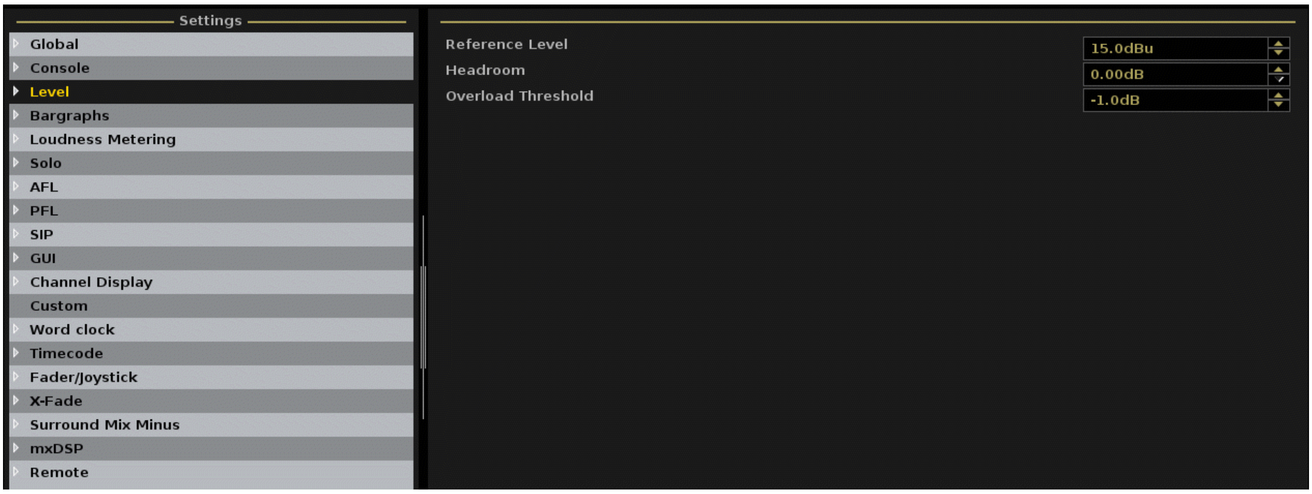
When this option is enabled (default), the slave faders of a VCA group are non-moving, as in an analogue VCA. This allows you to see and update slave positions even if the VCA master is closed.

Uncheck the option to turn on the fader motors for VCA slaves; move a VCA master and the slaves will follow.

Note that this option *only* affects VCA grouping. Surround VCAs, Links and the Couple group always use moving faders.

Level Options

Select the **Level** topic to make changes to the system's reference level, headroom or overload threshold:



These levels affect the maximum analogue level from your system according to the following equation:

- Maximum Analogue Level = **Reference Level** + **Headroom**

The system supports a maximum analogue level = +24dB, and a minimum analogue level = +12dBu.

Warning

Changing the **Reference Level** or **Headroom** options move the internal 0dB operating point for the system and therefore will change the behaviour of any level dependent settings such as dynamics processing and metering. Therefore, it is not advisable to alter these levels once dynamics processing has been set.

For systems fitted with fixed level analogue I/O cards:

- The **Headroom** and **Reference Level** cannot be altered independently. For example, with a +15dBu fixed analogue I/O card and +9dB **Headroom**, the **Reference Level** *must* be +6dBu.
- The Maximum Analogue Level of the whole system is defined by the DALLIS card with the lowest GDA (General Device Address) - this is the card with the lowest address fitted to the DALLIS frame connected to the lowest port number of the first Nova73. (If a different fixed level analogue card is fitted elsewhere within the system, then a warning appears in the log file; however, the card with the lowest GDA still wins.)

Reference Level

Sets the reference level of your analogue interfaces in dBu.

Reference level may be set from 0dBu to +24dBu, depending on the **Headroom**.

Headroom

Sets the operating headroom to the external world; this is the difference between the analogue reference level and digital full scale (0dBFS).

Headroom may be set from 0dB to +20dB depending on the **Reference level**.

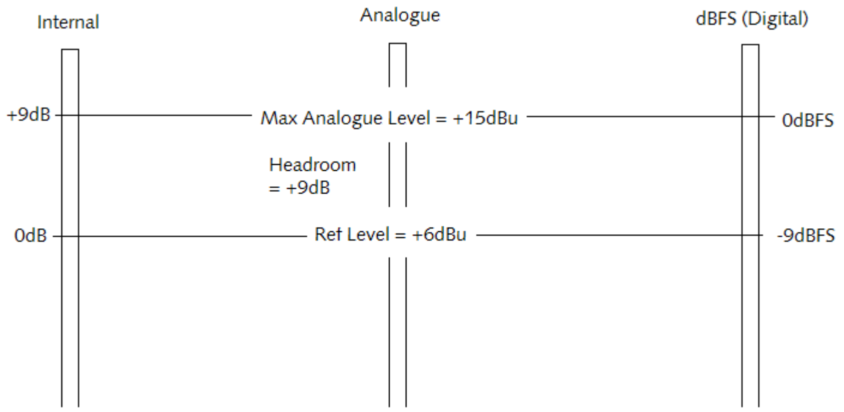
Note that the internal Headroom is more then 380dB which means, if you route from input to group to group to sum, you can overdrive the level more then 380dB before clipping!

Overload Threshold

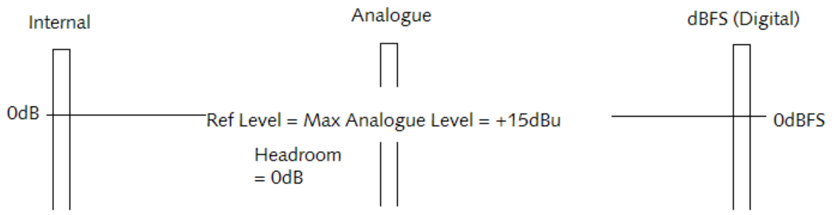
Sets the overload threshold of your system relative to digital full scale. It can be set from -6dBFS to -0.5dBFS or switched off.

Note that **OVR** is only indicated if you meter signals input to or output from the routing matrix. Internally, the system headroom exceeds 380dB!

The diagram below shows the normal operating levels for DIN scale operation in Germany:



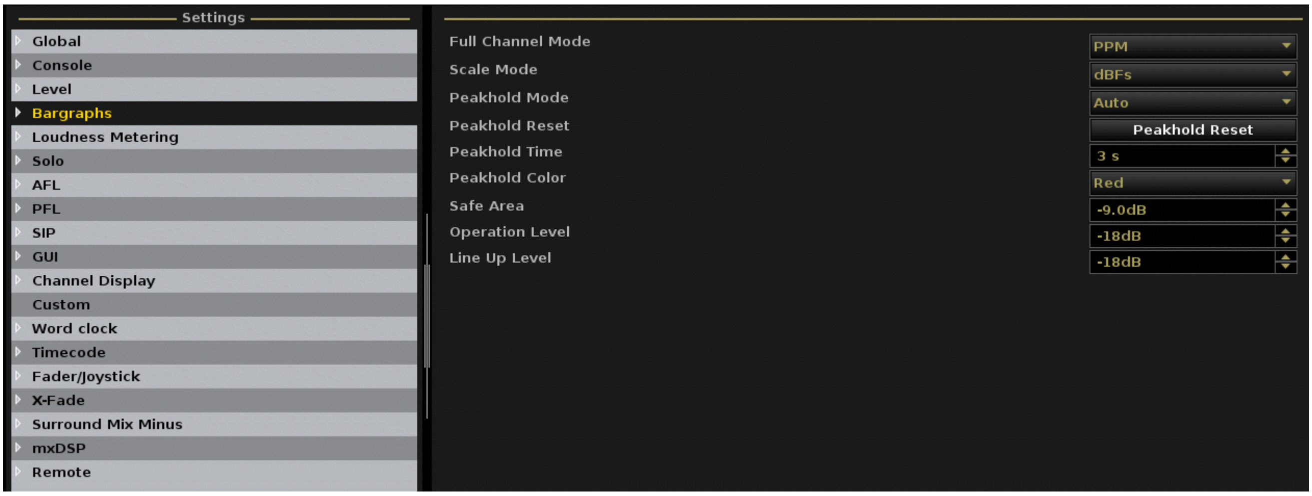
However, if you intend to work with the **dBFS** digital meter scale option, or an external AES meter, then you should set the **Reference Level** equal to your maximum analogue level (e.g. +15dBu) and **Headroom** to 0dB as shown below:



This ensures that the dBFS metering on the **Channel** display matches any external AES metering you may have. But be aware that the internal 0dB operating level now equals 0dBFS. This means that you are responsible for your own headroom. For example, if you still want a headroom of +9dB, then you will need to set your limiter threshold points to -9dB, etc.

Bargraph Options

Select the **Bargraphs** topic to set the peak metering options:



Full Channel mode

This option defines the peak meter characteristics across the console. Choose from the following options:

- **PPM** – Peak metering; 10ms attack time and 1.5s release.
- **True Peak** – True peak metering with 2 x oversampling, 0ms attack time and 1.5s release.
- **fast** – fast response peak metering; 1ms attack time and 1.5s release.
- **VU** – RMS metering; 300ms attack and 300ms release.

For ITU compliant operation, you should choose **True Peak**.

Scale mode

This option defines the peak meter scale across the console. Choose from the following options:

- **DIN PPM** – conforming to IEC 268-10.
- **UK PPM** – conforming to IEC 268-10 IIA.
- **Nordic** – conforming to IEC 268-10 I.
- **dBFS** – dB Full Scale digital meter scale (shown opposite).

When using the **dBFS** meter scale, it is recommended that you return to the Level options and set the **Reference Level** equal to your maximum Analogue Level and the **Headroom** to 0dB. This ensures that the dBFS metering across the console matches any external AES metering you may have.

Peakhold Options

Peakhold Mode

This option defines the behaviour of the peak hold indicator, which monitors and marks the peak level reached on each meter across the console. Choose from:

- **Auto** – peak hold automatically clears after the **Peakhold Time** value (see below).
- **Manual** – peak hold remains set until you select **CLEAR**.
- **Off** – the peak hold indicator is disabled.

Peakhold Reset

This option clears the peak hold indicators and reset peak level monitoring.

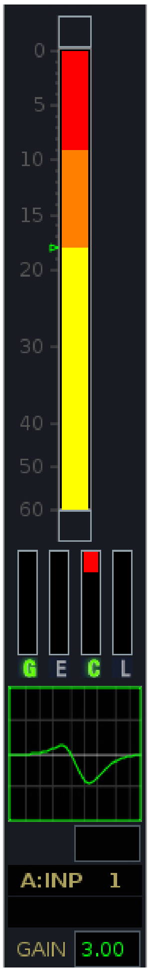
Peakhold Time

This option sets the peak hold time used in the **Auto** peak hold mode (see above). Set the value in seconds.

Peakhold Colour

This option sets the colour of the peak hold indicator.

Safe Area, Operation Level, Line Up Level



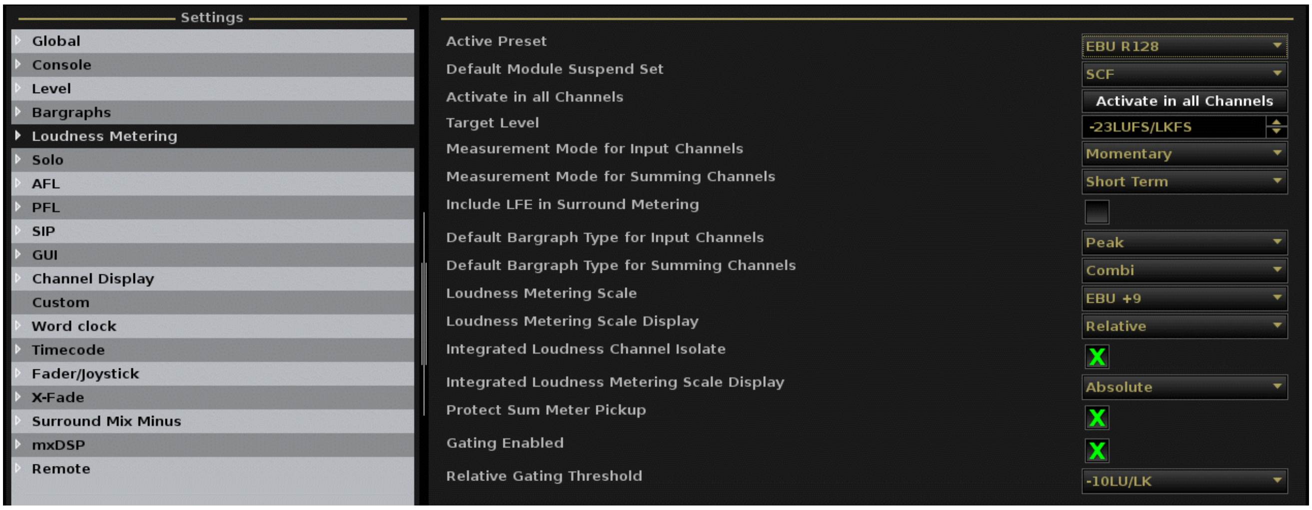
These options colour code the peak meter bargraphs, and can be used to help manage your headroom:

- **Safe Area** - this option sets the point where the meters change from red to orange. For example, you could set this to -6dB to mark 6dB's of headroom.
- **Operation Level** - this option sets the point where the meters change from orange to yellow in the middle of the meter scale. For example, you might set this to -12.0dB so that when signals peak within the orange area (-12dB to -6dB) you know that they are at a good operating level for the type of programme.
- **Line Up Level** - this option sets the position of the green 'Line up level' mark.

In each case, the levels are adjusted relative to the 0dB meter point.

Loudness Metering Options

Select the **Loudness Metering** topic to set the loudness metering options:



Active Preset

The **Active Preset** automatically recalls the correct loudness metering settings to comply with either the **EBU R128** or **ATSC A/85** & **ARIB** implementation standards.

The preset determines how loudness is displayed - in **LU**/**LUFS** for **EBU R128**, or **LK**/**LKFS** for **ATSC/ARIB** compliance. Note that LUFS is identical to LKFS defined by the ITU standard (BS 1770).

The preset also resets the following options:

- Target Level
- Loudness Metering Scale
- Gating Enabled
- Relative Gating Threshold

Note that if you change any of these options, then you are deviating from the EBU or ATSC/ARIB recommendations; to indicate this, the **Active Preset** changes to **Custom**.

Note that you may change the **Loudness Metering Scale** to the extended scale (e.g. from **EBU +9** to **EBU +18**) without affecting compliance.

Default Module Suspend Set

This option defines the default DSP module, or modules, which will be disabled (suspended) when loudness metering is activated.

Use the drop-down menu to make a selection.

The default selection can be modified on a channel by channel basis from the Channel Config display.

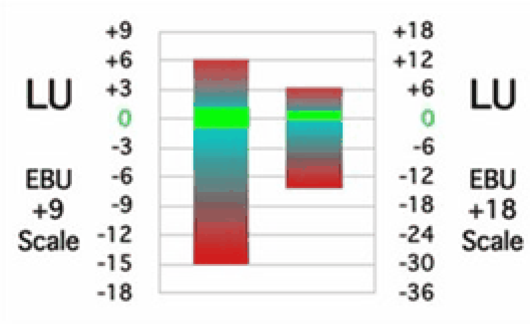
Activate in all Channels

Select this option to turn on loudness metering for all channels which support it. See Activating Loudness Metering for details.

Target Level

This option adjusts the target level for programme loudness, and is recalled by the Active Preset.

The EBU R128 recommends a target level of **-23 LUFS** +/- 1 LU. The target level is equivalent to 0 LU on the EBU loudness metering scale:



The ATSC A/85 & ARIB standards recommend a target level of **-24 LKFS** +/- 2 LK.

Note that LUFS is identical to LKFS defined by the ITU standard (BS 1770).

Note that the **Target Level** may be adjusted from -31 to -14 LUFS/LKFS. However, any changes will deviate from the EBU or ATSC/ARIB recommendations.

Measurement Mode for Input/Summing Channels

The next two options adjust the measurement mode for the loudness meter bargraphs. You may select:

- **Momentary (M)** – integration time operates over a 400ms sliding window.
- **Short Term (S)** – integration time operates over a 3 second sliding window.

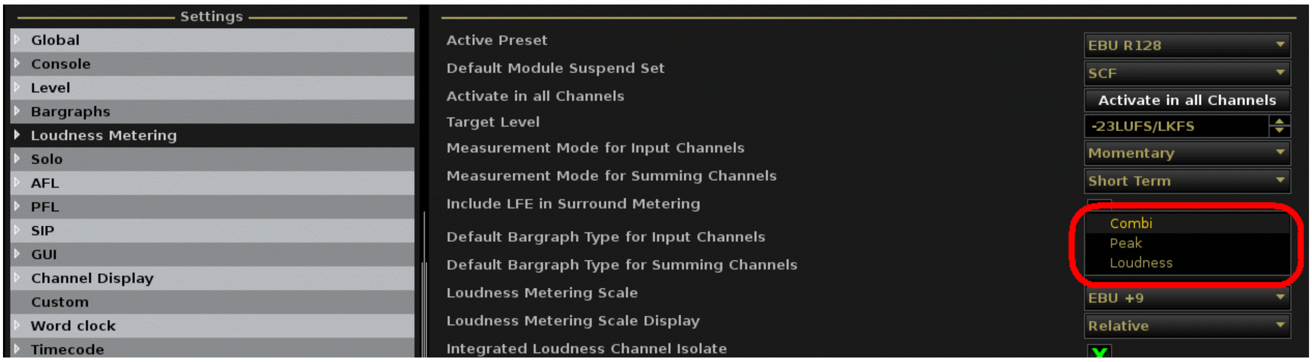
The loudness bargraphs include either an **M** or **S** representing the integration time. You may adjust this option separately for input channels and summing channels.

Include LFE in Surround Metering

Check this option if you wish to include the LFE channel in surround channel loudness measurements.

Default Bargraph Type for Inp/Summing Channels

The next two options adjust the default bargraph type for input and summing channels:



- **Default Bargraph Type for Input Channels** sets the option for all input and monitor channels.
- **Default Bargraph Type for Summing Channels** sets the option for all summing channels (groups, sums and auxes).

In each case, you may choose from:

- o **Combi** – peak and loudness metering side by side.
- o **Peak** – peak metering only.
- o **Loudness** – loudness metering only.

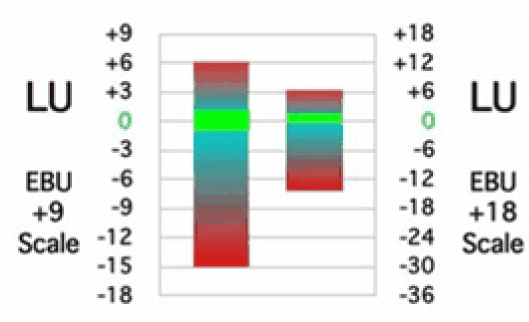
Loudness Metering Scale

This option define the scale for the loudness meter bargraphs.

Note that a default scale is recalled by the Active Preset.

The scale options comply either with the EBU R128 or ATSC A/85 / ARIB recommendations:

- **EBU +9** – the EBU standard scale.
- **EBU +18** – the EBU extended scale (covering twice the dynamic range).



- **ITU-R BS.1771** – the ITU standard scale (-21 LU to +9 LU).
- **Extended ITU-R BS.1771** – the extended ITU standard scale (-42 LU to +18 LU).

Loudness Metering Scale Display

This option determines how the loudness meter bargraph scale values are displayed. You may select:

- **Absolute** – scale values are displayed as absolute values in LUFS/LKFS.
- **Relative** – scale values are displayed relative to the Target Level.

Integrated Loudness Channel Isolate

This option affects channels using the integrated loudness measurement.

Check the option to automatically isolate a channel once an integrated measurement is started. The option turns on **SNAP ISO** on the channel's fader strip so that any snapshot recalls will *not* affect the channel.

If this option is *not* checked, then a snapshot saved when the **LOUD** DSP module was turned off, will reset the channel's signal flow and therefore destroy any active integrated loudness measurement.

Integrated Loudness Metering Scale Display

This option determines how the integrated loudness value is displayed. You may select:

- **Absolute –** value is displayed as an absolute value in LUFS/LKFS.
- **Relative –** value is displayed relative to the Target Level.

Protect Sum Meter Pickup

When this option is checked, you cannot alter the position of the loudness meter pickup point (the **LOUD** DSP module) for summing channels (Groups, Auxes or Sums).

Gating Enabled

When this option is checked, two-step gating is enabled for integrated loudness measurements.

Note that this option is recalled by the Active Preset. Any change will deviate from the EBU or ATSC recommendations.

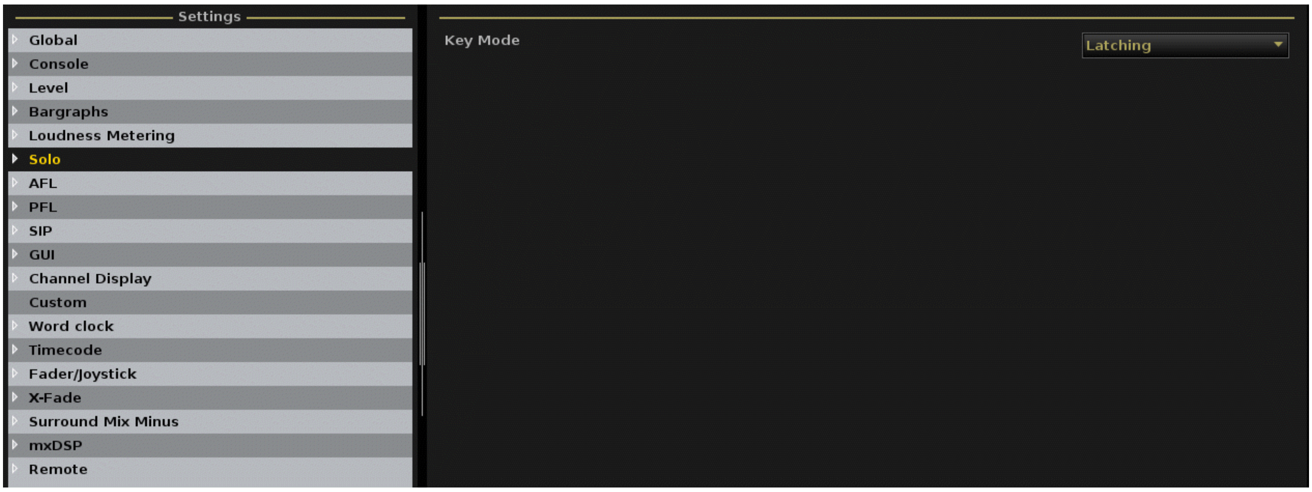
Relative Gating Threshold

This option sets the relative gating threshold to either -8 or -10 LU/LK. Gating is only applied if the **Gating Enabled** option above is checked.

Note that this option is recalled by the Active Preset. Any change will deviate from the EBU or ATSC/ARIB recommendations.

Solo Button Options

Select the **Solo** topic to set the behaviour of the console's solo buttons:



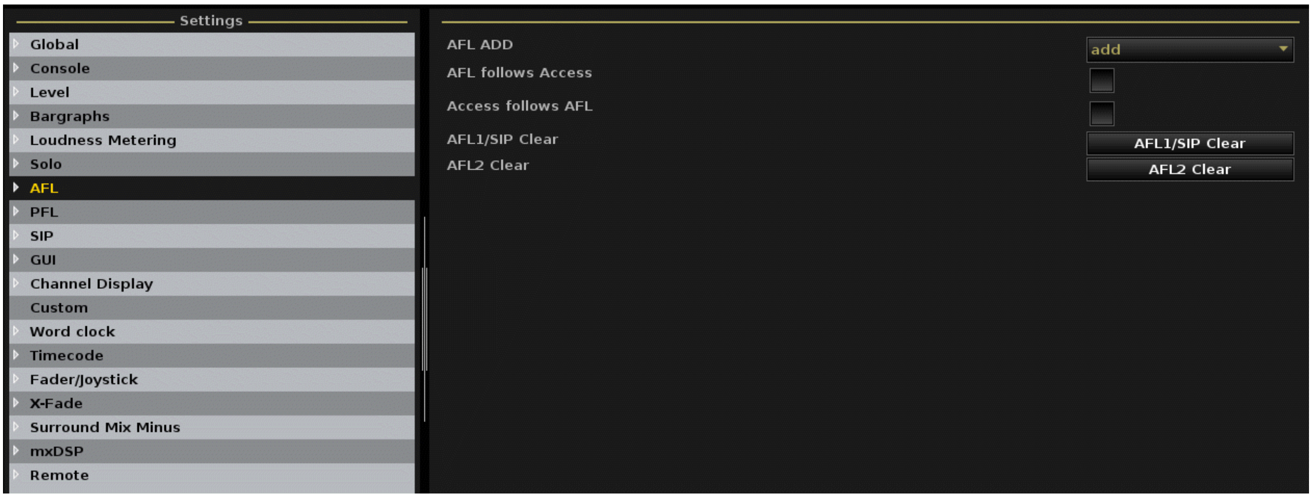
Key Mode

This option defines whether **AFL** (or Solo-in-Place) and **PFL** buttons are latching or momentary:

- **Key Mode** (**Latching**) – AFL and PFL buttons latch on or off when pressed.
- **Key Mode** (**Momentary**) – AFL and PFL buttons are only active while pressed.

AFL Options

Select the **AFL** topic to set the AFL options:



AFL Add

This option defines whether **AFL** selections are additive or exclusive:

- **EXCL** – only one AFL may be active at any time; selecting a new AFL cancels the previous selection.
- **ADD** – allows multiple AFL buttons to be combined, thereby enabling a group of channels to be monitored in context.

AFL follows Access

This option controls the behaviour of AFL when you update the channel in access. It works best with exclusive AFL.

- **AFL follows access** (on) – AFL selections follow the channel in access.
- **AFL follows access** (off) – updating the channel in access does not automatically select AFL.

Access Follows AFL

This option determines whether the channel in access automatically follows AFL selections:

- **Access follows AFL** (on) – selecting a channel AFL automatically updates the channel in access.
- **Access follows AFL** (off) – the channel in access is not updated by AFL selections.

AFL/SIP Clear

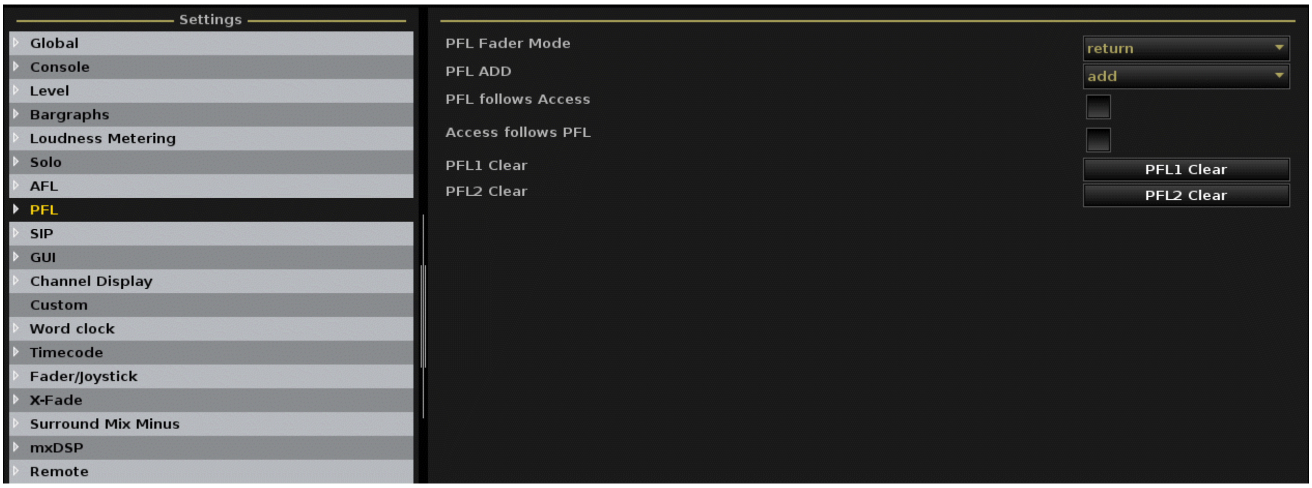
Select these buttons to clear all active AFL1 or AFL2 selections.

Note that Clear A/PFL may also be available from the console's monitoring section.

Note AFL buttons may operate as Solo-in-Place if SIP is active.

PFL Options

Select the **PFL** topic to set the PFL options:



PFL Fader Mode

This option controls how PFL responds to fader open and fader closed:

- **off** – the fader position does not affect PFL selections.
- **on** – choose this option to cancel PFL when a fader opens.
- **return** – choose this option to activate PFL when a fader closes.

PFL can also be actioned from the fader backstop, see the Fader Backstop options.

PFL Add

This option defines whether PFL selections are additive or exclusive:

- **EXCL** – only one PFL may be active at any time; selecting a new PFL cancels the previous selection.
- **ADD** – allows multiple PFL buttons to be combined, thereby enabling a group of channels to be monitored in context.

PFL follows Access

This option controls the behaviour of PFL when you update the channel in access. It works best with exclusive PFL:

- **PFL follows access** (on) – PFL selections follow the channel in access.
- **PFL follows access** (off) – updating the channel in access does not automatically select PFL.

Access follows PFL

This option determines whether the channel in access automatically follows PFL selections:

- **Access follows PFL** (on) – selecting a channel PFL automatically updates the channel 'in access'.
- **Access follows PFL** (off) – the channel in access is not updated by PFL selections.

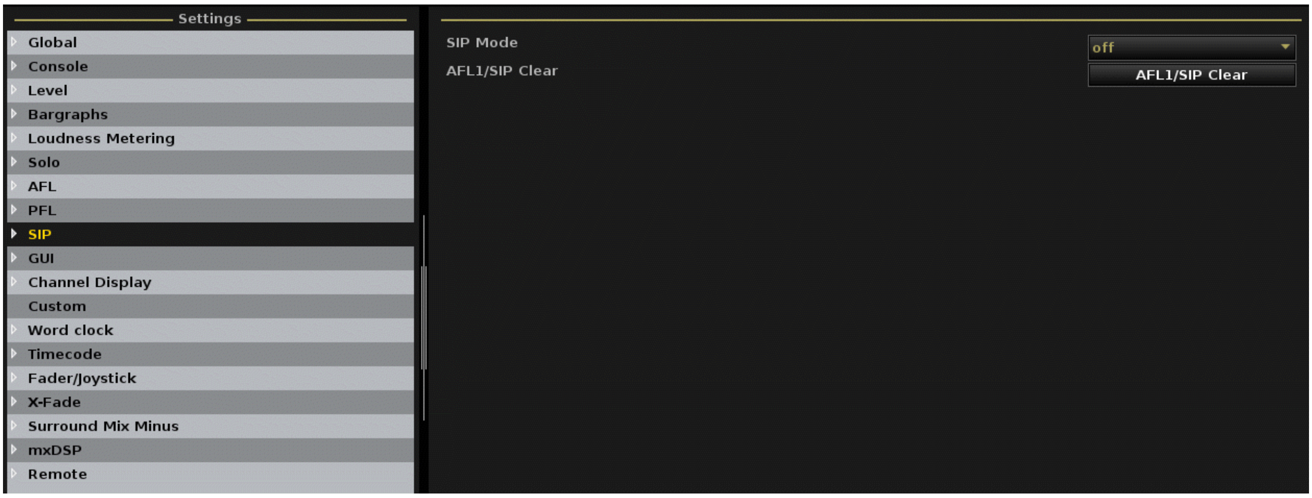
PFL Clear

Select these buttons to clear all active PFL1 or PFL2 selections.

Note that Clear A/PFL may also be available from the console's monitoring section.

Solo-in-Place Options

Select the **SIP** topic to set the Solo-in-Place options:



SIP Mode

This option enables or disables destructive Solo-in-Place:

- **OFF** – all AFLs act as non-destructive AFL.
- **MON** – all AFLs act as non-destructive AFL except on monitor channels where the AFL button provides Solo-in-Place for multitrack returns.
- **INP+MON** – all AFLs act as destructive Solo-in-Place.

Warning

Solo-in-Place works by muting any channels not in Solo, so that they no longer feed the bus outputs. This is very useful for post production. However, you should *NOT* use Solo-in-Place when working on a live broadcast, to avoid accidentally interrupting the main programme mix.

AFL1/SIP Clear

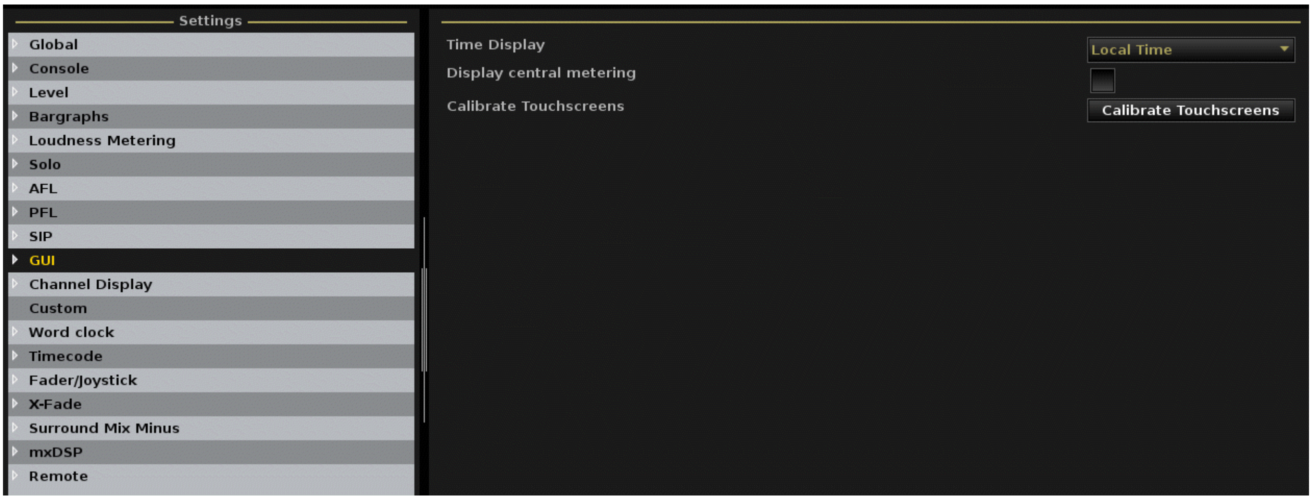
Select this button to clear all active Solo-in-Place (or AFL) selections.

Note that Clear A/PFL may also be available from the console's monitoring section.

Use the Solo Safe option, in the **Channel Configuration** display, to prevent individual channels being muted when a Solo is active.

GUI Options

Select the **GUI** topic to set the Graphical User Interface options:



Time Display

At the top of the Central GUI, a time is displayed. This may be the local time, in 24 hour clock, or timecode:

- **Local** – displays the local system time in 24 hour clock.
- **Timecode** - displays SMPTE timecode from your selected timecode reference.
- **Offset TC** - displays SMPTE timecode + the Midnight offset.

This option applies only to the Central GUI (GUI_0).

You can also set this option by clicking in the Headline:

The time display may be replaced by the integrated loudness measurement for a summing channel, see the Title Bar.

Display Central Metering

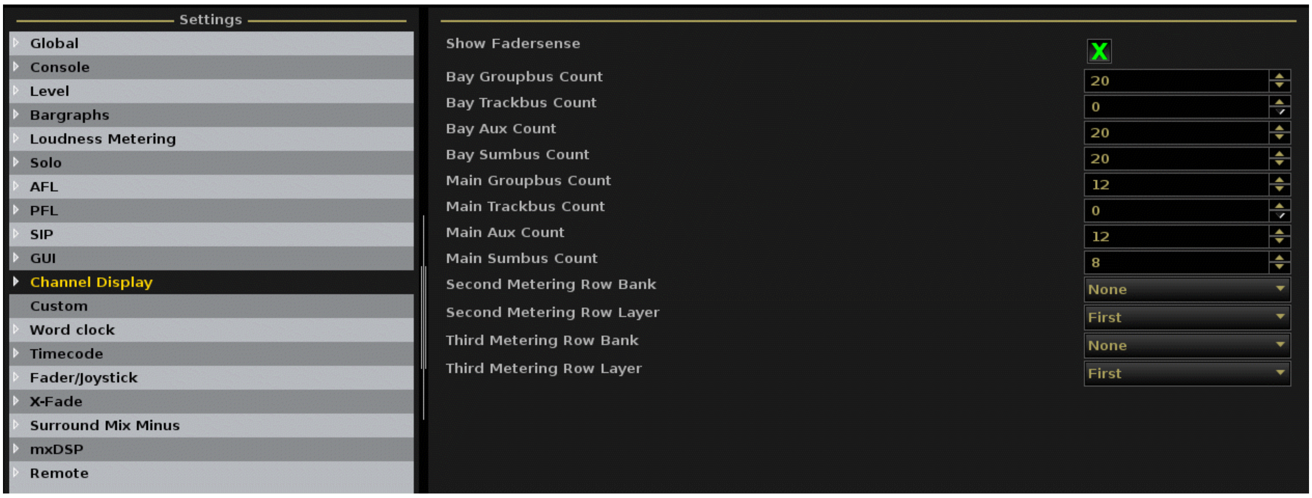
This option activates the mini main fader metering display for the 16 main fader strips:

Screen Calibration

Starts the touch-screen calibration. Select the option and follow the on-screen instructions.

Channel Display Options

Select this topic to set the following **Channel Display** options:



Show Fadersense

This option enables or disables the show Fader/encoder sense mode on the Channel display:

- **Show Fadersense** (on) – touch a fader or encoder and the channel highlights on the **Channel** display. The colour of the outline matches the colour coding for the channel type, selected from the Channel Config display.
- **Show Fadersense** (off) – nothing changes on the **Channel** display when you touch a fader or encoder.

Bay & Main Bus Count

The next eight options enable you to change the number of busses shown on:

- **Bay Bus Count** - the Channel display.
- **Main Bus Count** - the Main Fader Metering display.

For example, you may wish to display different numbers of **Group**, **Track**, **Aux** and **Sum** Busses depending on your production and choice of DSP configuration. For each entry, you can enter the number of busses you wish to display; the display resizes accordingly.

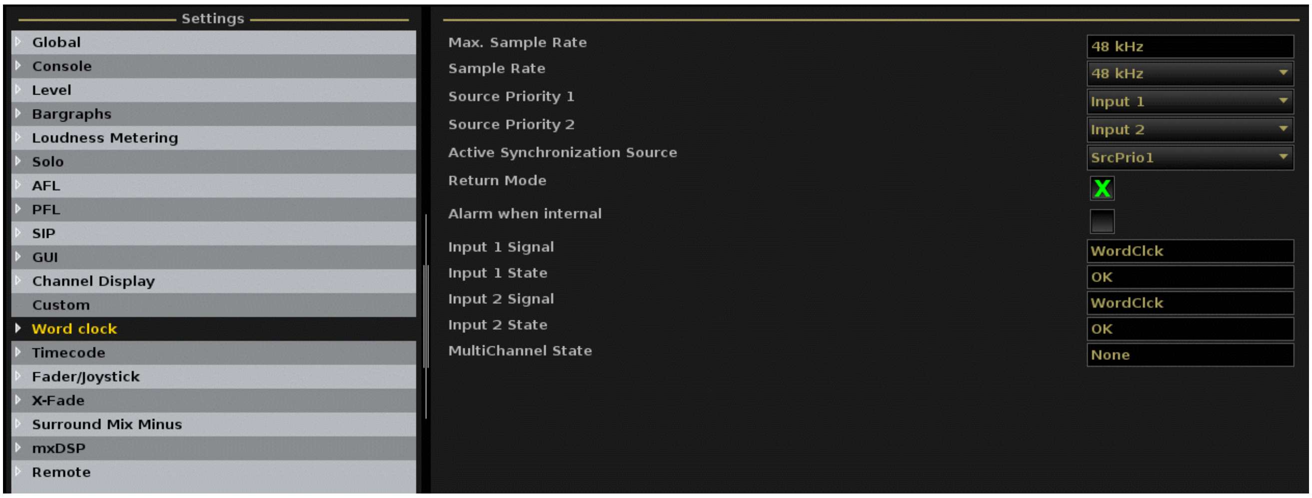
Note that if multiple meter rows are enabled, then the maximum number of busses is limited by the physical size of the display.

Metering Row Options

The next set of options configure the second and third metering rows on the Channel display. See Multi-row Metering Configuration for details.

Wordclock Options

The **Wordclock** topic covers a range of options for selecting the internal sample rate of the system, and defining source priorities for system sync.



Max Sample Rate & Sample Rate

The option to run at higher (96kHz or 88.2kHz) or lower (48kHz or 44.1kHz) sample rates is made within the AdminHD configuration and cannot be modified from the console.

> Higher sample rates use twice as much DSP resource as lower sample rates; this is reflected in the DSP Configurations display.
>
> Higher sample rates also affect the crosspoint capacity of the routing matrix ($8k^2$ at 48kHz, or $4k^2$ at 96kHz).

Max Sample Rate

The maximum rate is displayed in the **Max Sample Rate** field. This field is for display purposes only, and determines the available **Sample Rate** options below.

Sample Rate

This option selects the internal sample rate of the system.

If the system is configured to run at lower sample rates (by AdminHD), then you may select either **48kHz** or **44.1kHz** operation.

If the system is configured to run at higher sample rates, then you may select **96kHz**, **88.2kHz**, **48kHz** or **44.1kHz** operation.

> It is recommended that you mute your loudspeakers when changing the system sample rate.
>
> When running the system referenced to Wordclock, the frequency of the sync source *MUST* match the internal operating sample rate of the system.

System Clock Options

The remaining **Wordclock** options define the system's sync reference.

The Nova73 offers a fully redundant clock source structure with two independent clock inputs, an internal sync generator and the ability to lock to sync from an incoming multi-channel signal. This allows the console to be clocked from a variety of sync sources and recover from loss of external sync.

Source Priority 1 & 2

These two options define the main and redundant clock source.

If sync is lost or a signal of an incorrect frequency appears on **Source Priority 1**, the system automatically switches to **Source Priority 2**. Similarly, if sync is lost on **Source Priority 2**, the system automatically switches to internal sync.

You can set each of the options to:

- **Input 1** – connected to the Nova73 rear panel (Wordclock, Video Black Burst or AES3-id).
- **Input 2** – connected to the Nova73 rear panel (Wordclock, Video Black Burst or AES3-id).
- **MultiCh** - Multichannel Sync (MADI or ATM, as defined by AdminHD).

Active Synchronization Source

This option displays and sets the active sync source for the system:

- **Src Prio 1** – the input selected as **Source Priority 1**.
- **Src Prio 2** – the input selected as **Source Priority 2**.
- **Internal**.

Return Mode

This option activates a return mode so that the system will switch back to **Source Priority 1** (or **2**) when it returns. The system even checks whether the return sync is valid and will not switch until the sync source matches the chosen operating frequency of the console.

- **Return Mode** (On) - activates the return mode.
- **Return Mode** (Off) - deactivates the return mode.

To force the system to run on internal sync, deactivate the return mode and set the **Active Source** to **Internal**.

Alarm when Internal

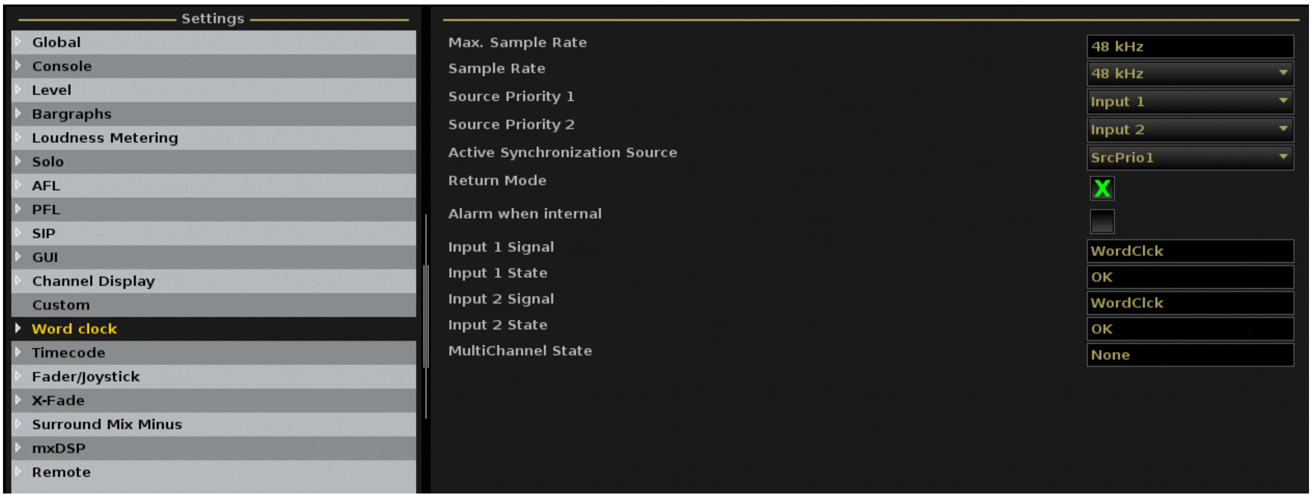
This option activates an alarm when the system is running on internal sync:

- **Alarm when internal** (On) - activates the alarm.
- **Alarm when internal** (Off) - deactivates the alarm.

The alarm triggers on-screen Warning flag and illuminates the red LED on the front panel of the Nova73 Router Card.

External Sync Input Status

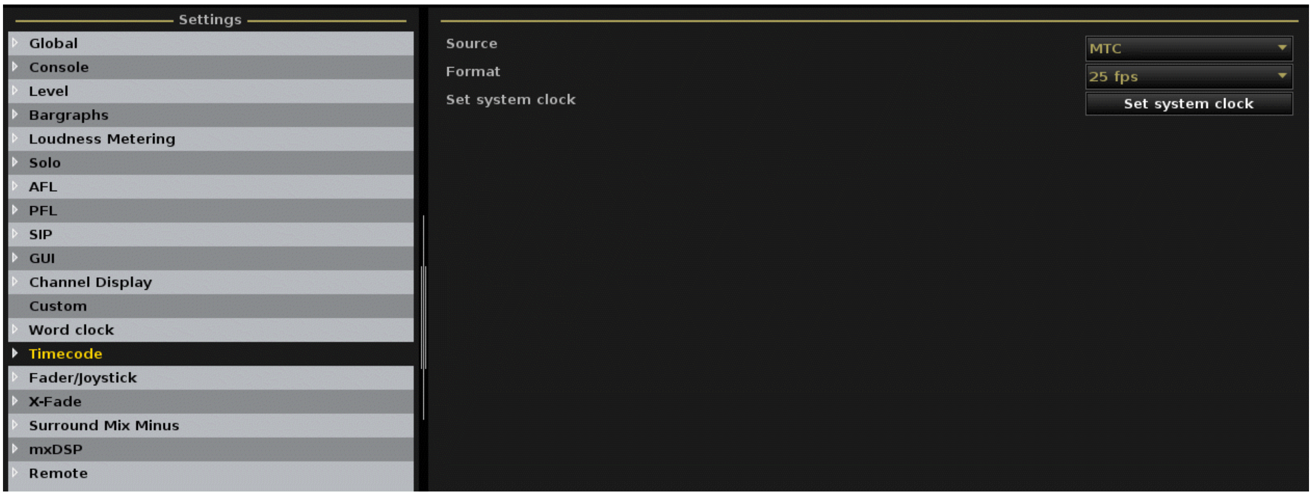
The next five options are for display purposes only and show the status of the external and multi-channel sync signals.



The example above shows that a valid Wordclock signal is connected to external inputs 1 and 2, and the **Active Synchronisation Source** is **Src Prio 1** = **Input 1**.

Timecode/Frame Rate Options

The **Timecode** topic selects the timecode source and frame rate when running timecode automation.



Source

Sets the timecode source. You may choose from:

- **MTC** – the automation system will slave to the external Midi timecode (MTC) input.
- **Internal** – the automation system will slave to internal timecode.
- **LTC** – the automation system will slave to the external Linear timecode (LTC) input.
- **Machine** – the automation system will slave to the active Sony 9-pin machine (selected from the optional Machine Control panel).

Format

When running on internal timecode, this option sets the frame rate.

Note that the frame rate also sets the delay time for channel delay when adjusting delay in frames. For example, if you select 25 fps, then delay time for 1 frame will be 40ms (1/25s).

- **24 fps** - 24 frames per second - Film.
- **25 fps** - 25 frames per second - EBU (PAL or SECAM)
- **30d fps** - 30 drop frame timecode - NTSC colour TV.
- **30 fps** - 30 frames per second – monochrome TV.

If you have selected an external timecode source, this field displays the incoming frame rate.

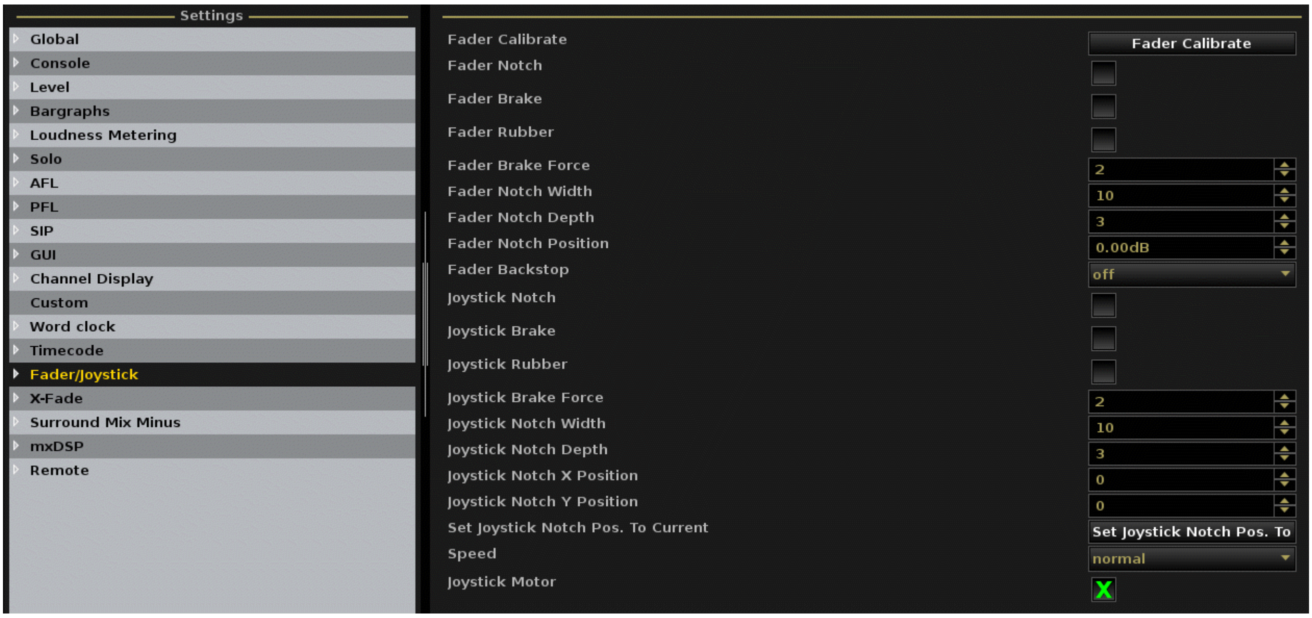
Set System Clock

Select this button to set the timecode to the local system time.

Fader/Joystick Options

The **Fader/Joystick** topic provides options to customise the feel of your faders and joystick. For example, you may wish to work with a 0dB level fader notch, increase or decrease the brake resistance of the faders and/or work with a PFL Overpress. You can also create notches and brake resistance for the surround joystick to allow you to feel specific room positions as you pan.

Note that these options apply globally to all console faders and/or joysticks.



Fader Calibrate

This option is used to calibrate the faders on the console.

1. Select **Fader Calibrate** to calibrate all faders.

Each fader across the console opens and closes to calibrate.

Fader Notch and Brake Resistance

The next seven parameters set notches and brake resistance for the console's faders:

- **Fader Notch** (on) - activates a fader notch at a certain position (e.g. 0dB).
- **Fader Brake** (on) - activates fader brake resistance.
- **Fader Rubber** (on) - activates the fader brake force if the fader moves away from the notch position.
- **Fader Brake force** – sets the amount of resistance which will be applied when Fader brake is active. 1 = smooth; 3 = stiff.
- **Fader Notch Width** – sets the width of the notch when Fader notch is active. 1 = narrow; 20 = wide.
- **Fader Notch Depth** – sets the depth of the notch when Fader notch is active. 1 = flat; 5 = deep.
- **Fader Notch Position** – sets the position of the notch when Fader notch is active. The position may be set from -128dB (fader closed) to +15dB (fader open).

You may select multiple options, for example, to activate a fader notch and brake resistance.

Fader Backstop

This option activates the fader backstop. The fader backstop switch can be used to select PFL monitoring when a fader is pulled back against its endstop. Or, to trigger an external event such as a fader start:

- **Off** - disables the backstop switch.
- **On** - enables the backstop switch. Use this option if you wish to trigger a fader start, or other external event, by pulling back on a fader.
- **On + PFL** - enables PFL monitoring from the backstop, otherwise known as backstop PFL monitoring.

Joystick Notch and Brake Resistance

The next seven parameters set notches and brake resistance parameters for the console's joystick:

- **Joystick Notch** (on) - activates a joystick notch at a certain position (e.g. Front Centre).
- **Joystick Brake** (on) - activates joystick brake resistance.
- **Joystick Rubber** (on) - activates the joystick brake force if the joystick moves away from the notch position.
- **Joystick Brake Force** – sets the amount of resistance which will be applied when Joystick brake is active. 1 = smooth; 3 = stiff.
- **Joystick Notch Width** – sets the width of the notch when Joystick notch is active. 1 = narrow; 20 = wide.
- **Joystick Notch Depth** – sets the depth of the notch when Joystick notch is active. 1 = flat; 5 = deep.
- **Joystick Notch X Position** – sets the x-axis position of the notch when Joystick notch is active. The position may be set from -20 (Left) to +20 (Right).
- **Joystick Notch Y Position** – sets the y-axis position of the notch when Joystick notch is active. The position may be set from -20 (Rear) to +20 (Front).
- **Set Joystick Notch Pos. To Current** – this option allows you to set the joystick notch position from the current position of the control. Place the joystick control at the desired position then select this button.

Speed

This option adjusts the speed of all console faders when they respond to automated control, for example snapshot reset or timecode automation. You can select:

- **fast** - fast fader speed.
- **normal** - normal fader speed.
- **slow** - slow fader speed.

Joystick Motor

This option can be used to enable or disable the motors on the console's joystick.

- **Joystick Motor** (on) - enables the joystick motor.
- **Joystick Motor** (off) - disables the joystick motor.

Note that on US systems, the joystick motor may not be enabled.

X-Fade Options

The **X-Fade** topic sets the default cross fade parameters which are saved with a snapshot when it is saved or updated. These parameters are applied when snapshots are played out from a Sequence, see Snapshot Cross Fades.

Default Switch Mode

This option sets whether switched functions, such as mutes, will change state at the start or at the end of the cross fade. Choose from:

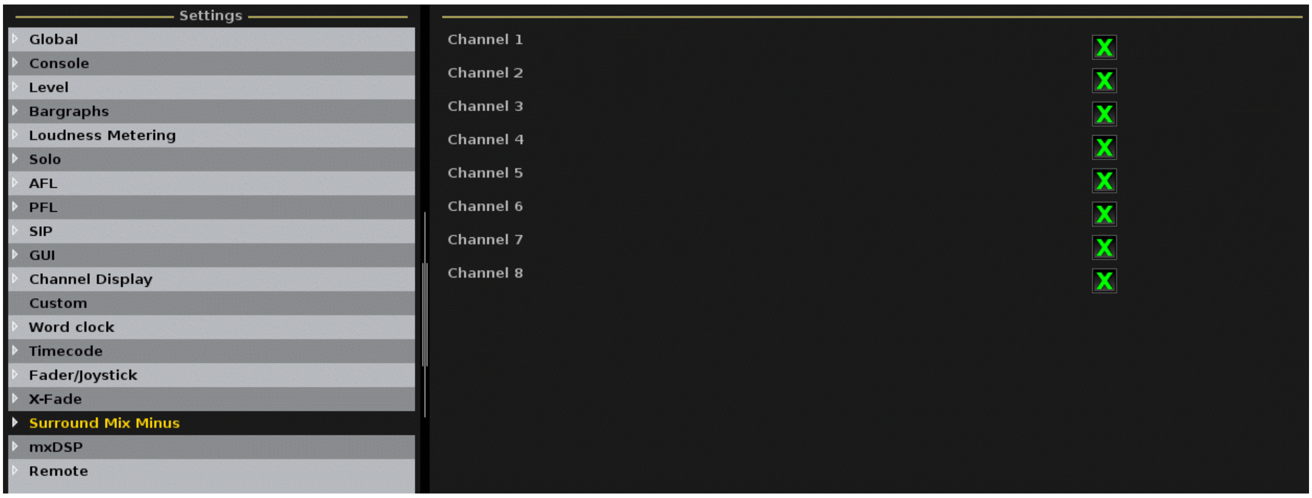
- **at start** – switched functions change state at the start of the cross fade.
- **at end** – switched functions change state at the end of the cross fade.

Default X-Fade Time

This option sets the default cross fade time. It is adjusted in ms steps.

Surround Mix Minus Options

The **Surround Mix Minus** topic determines which components of a surround channel feed the mix minus bus when configured:

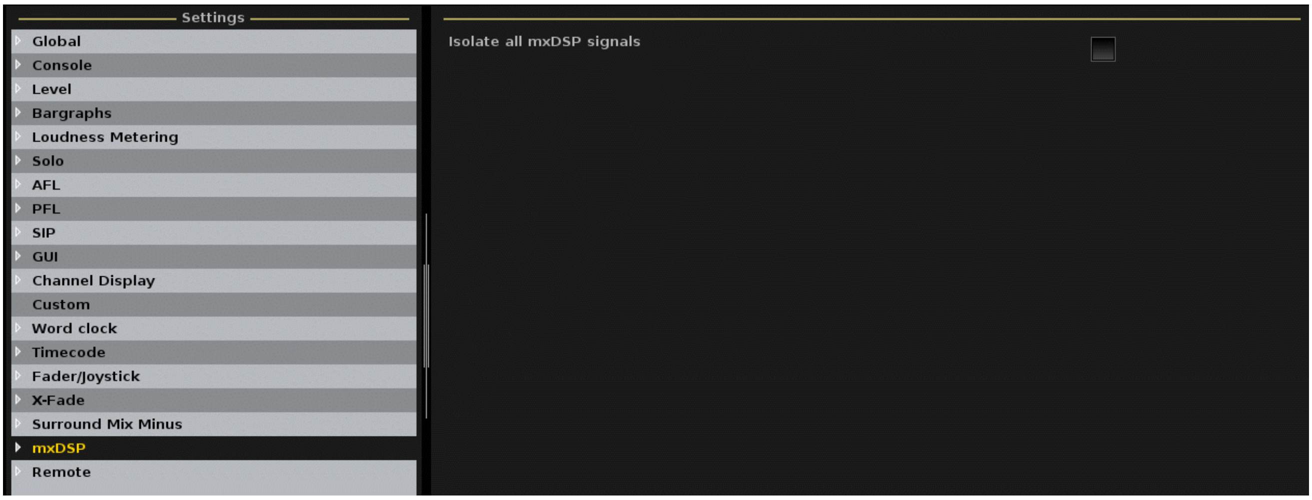


Select the channels if you wish them to feed the mix minus bus. For example, you might select only Channel 1 (left) and Channel 3 (centre).

The default configuration is all flags selected (as above).

mxDSP Options

The **mxDSP** topic provides options for the optional mxDSP module:



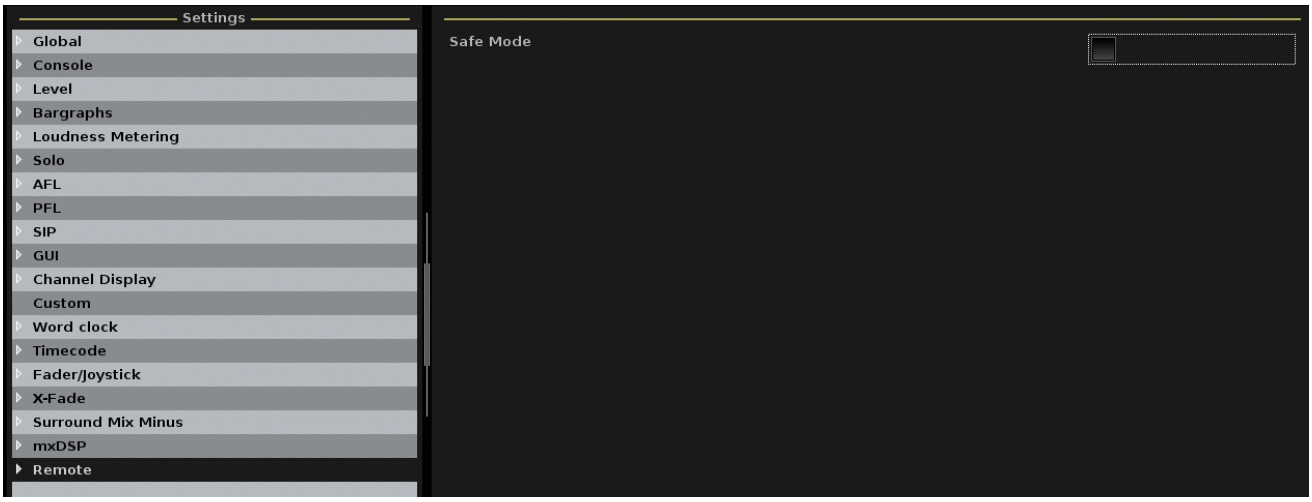
The option determines whether mxDSP signals are affected by snapshot loads:

- **Isolate all mxDSP signals** (on) – isolates all mxDSP signals so that they are not affected by a snapshot load. Use this option to protect the current mxDSP settings.
- **Isolate all mxDSP signals** (off) – settings will be reset by a snapshot load. Use this option if you wish to recall mxDSP settings from snapshots.

Note that the same option can be selected using the Global Snapshot ISO **MXDSP** button.

Remote Options

The **Remote** topic provides options for the Lawo Remote App.



The option determines whether the console may be controlled from a remote device running the Lawo Remote App:

- **Safe Mode** (on) – access from remote devices is denied. Use this mode to prevent unauthorised control of the console.
- **Safe Mode** (off) – the console may be controlled by a remote device running the Lawo Remote App.

The Custom Functions Display

This display provides access to factory-configured custom functions, such as the mapping of user buttons, so that users can reconfigure the console without assistance from Lawo.

The functions configured from this display are stored as part of the system configuration, which means that any changes will affect all users. In addition, there are many powerful features. It is recommended that users have a good understanding of the system, are familiar with the programming of user buttons, and understand how to connect to the console via ftp or telnet. For information on these procedures, please refer to the "mc²56 Technical Manual".

Note that the **Custom Functions** display may be hidden from the console GUI to protect the current configuration. If this is the case, you may still access the display from an mxGUI computer. Please contact Lawo service for advice on how to show or hide the **Custom Functions** display on your console.

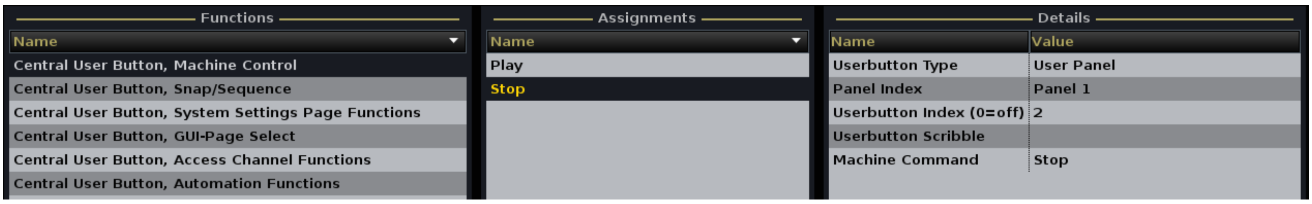
1. Press the **SYSTEM DSP** button, located on the SCREEN CONTROL panel, to view the **Custom Functions** display.

   *The **Functions** column on the left lists the different types of function which can be configured. A brief description appears when you hover over each title.*

2. Select a function to interrogate any existing assignments.

   *Each time you select a different function, the **Assignments** column updates – in our example, we have two Machine Control user button assignments named **Play** and **Stop**.*

3. Select an **Assignment** to interrogate its **Details**:



Operating Principles

Triggering a Custom Function

Most functions listed on the **Custom Functions** display are actioned from a user button. However, some functions are designed for other purposes. For example, Snap Iso List configures a list of sources or destinations which you do not wish to be reset by snapshots.

When looking at the **Functions** list, use the naming as a guide:

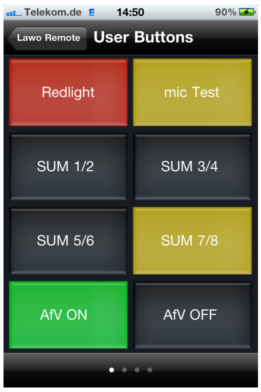
- "**Central User Button, xxx**" - master functions such as machine control Play, Stop.
- "**Fader User Button, xxx**" - channel-related functions such as Snapshot Isolate, Aux on/off.
- Anything else is a special case!

User Button Panel Types

Fader User Buttons

User Button Screen Control Panel

Lawo Remote App



When assigning **Fader User Button** functions, they can be mapped to any available user button on the fader strip. The function then becomes available globally across the console (e.g. Fader User Button 3 = **SNAP ISO**).

When assigning **Central User Button** functions, you have a choice of several panel types. These refer to different user button locations:

- **Monitoring Panel** = the Touch-screen Monitoring Buttons 1 to 24 on the Central GUI.
- **User Button Monitoring Panel mc²90** = not supported by the mc²56. It is used on the mc²90 to programme the hardware user buttons on the Monitor Panel.
- **User Panel 40 Button** = an optional user panel which may be fitted to the Overbridge.
- **User Button Screen Control Panel** = the Central User Buttons 1 to 9 on the SCREEN CONTROL panel.
- **Talkback Panel** = the Talkback User Buttons 1 to 4 beside the monitor level controls.
- **Lawo Remote App** = the user buttons on an iPhone, iPod or iPad running the Lawo Remote App.

Please refer to the User Button Numbering Appendix for further details.

Programming Custom Functions

All custom functions are programmed in a similar manner, so this section deals with how to create, edit and delete an assignment. For a complete list of functions see The Custom Functions list.

Note that the MKII mc²56 ships from the factory with some default custom functions.

Warning

Before changing the function of a user button, make sure that there is nothing assigned to it. Otherwise, the button *will* perform multiple functions!

In particular, take care with the Monitoring touch-screen panel. The pages cannot be accessed from a custom function, therefore the button location *MUST* be free across *ALL* pages.

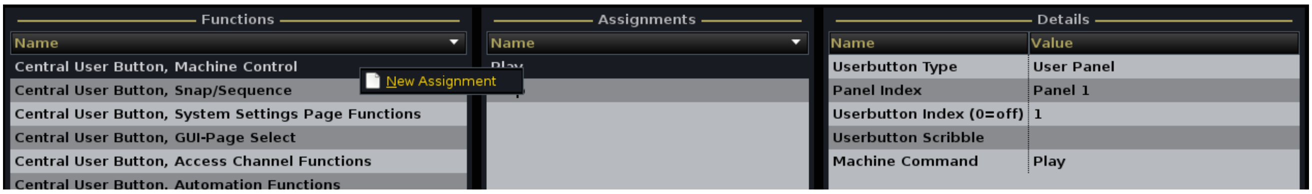
Also be aware that factory-configured user functions do *NOT* appear in the **Custom Functions** display. If you wish to reprogramme these, then you should contact Lawo to remove the factory configuration first. Otherwise, you may have buttons performing more than one function.

Note that as soon as you create or edit a custom function assignment, a Custom Template file (for the assignment) is stored in the system's configuration data. Custom functions are stored as part of the configuration, and not in productions, and therefore any changes will affect *all* users of the system.

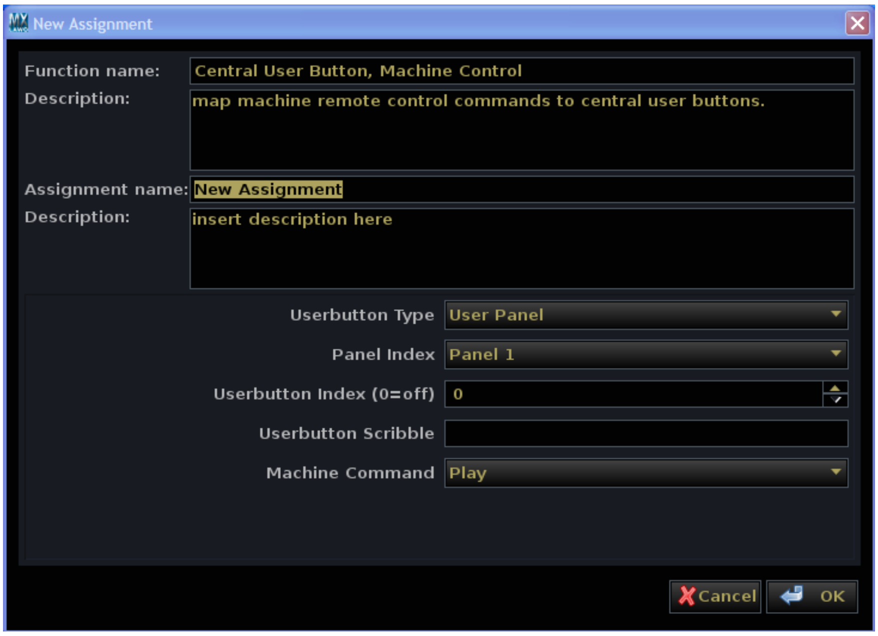
Creating a New Assignment

Let’s take the example of mapping a Central User Button to a machine control command such as Record:

1. Right-click on the function from the **Functions** list and select **New Assignment:**



A pop-up window appears listing the assignment details:



2. Edit each field as follows:

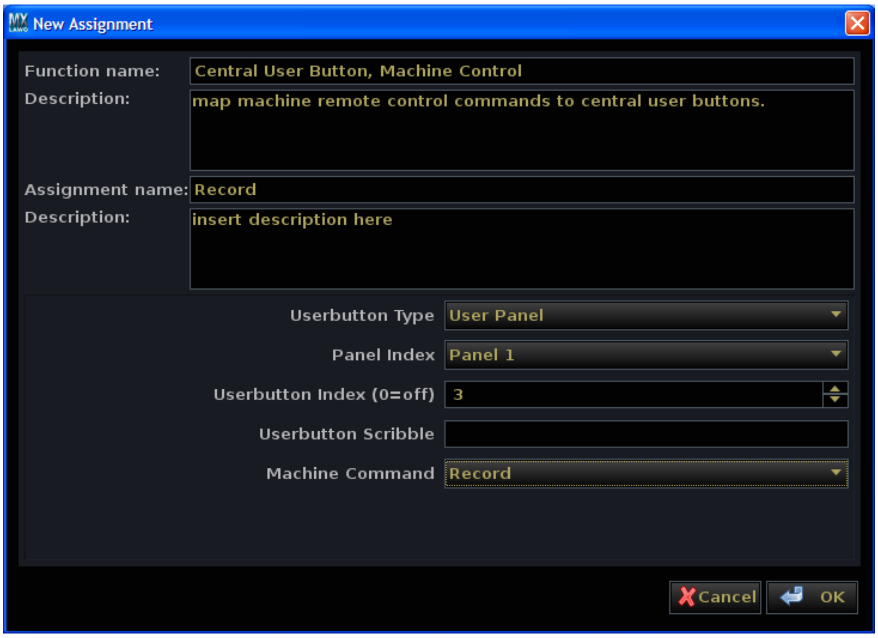
- **Function Name** and **Description**: these fields are for information only and cannot be edited. They describe what the function does.
- **Assignment Name**: enter a name for the assignment, for example - **Record**.

> You *must* enter a unique name for each custom function you create.

- **Description**: enter a user description for your assignment (optional).

- **Userbutton Type**: select the panel location for the button assignment - e.g. **User Button Screen Control Panel**.
- **Panel Index**: select the panel number for the assignment, see User Button Numbering.
- **Userbutton Index**: select the button number for the assignment, see User Button Numbering.

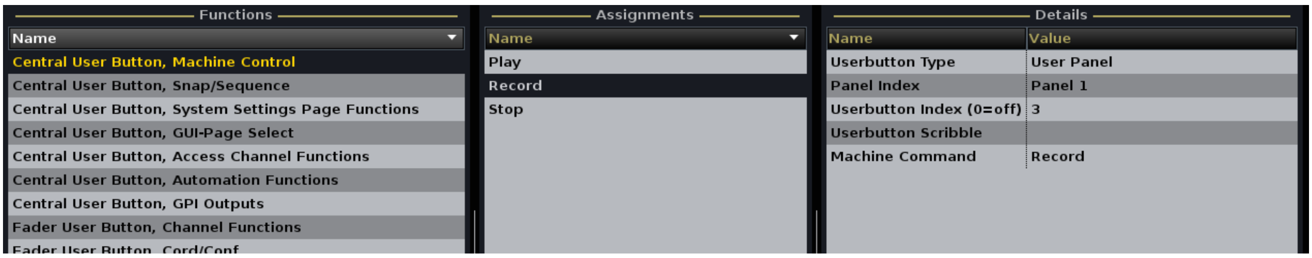
In our example, we have selected button 3 on User Panel 1:



- **Userbutton Scribble**: if the selected user button has an accompanying scribble strip display, then you can enter the text to be displayed in this field. Up to 8 characters. Text is only displayed *IF* the button has a scribble strip - for example, on the Lawo Remote App.
- **Machine Command**: select the function to be assigned, for example - **Record**.

3. Once you are happy with everything select **OK**.

*The assignment is made and you will see its name appear in the **Assignments** list:*



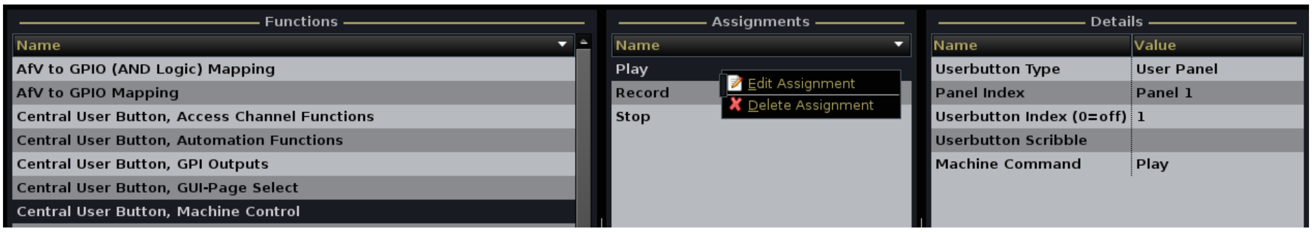
4. Repeat these steps to configure other custom functions.

Editing & Deleting Assignments

Note that as soon as you edit or delete an assignment, the changes update the Custom Template file (for the assignment). Custom functions are stored as part of the configuration, and not in productions, and therefore any changes will affect *all* users of the system.

➢ **To edit an existing assignment:**

1. Select the **Function** and **Assignment** you wish to edit,
2. Right-click and select **Edit Assignment**:



*The **Edit Assignment** pop-up window appears showing the current details of the assignment.*

3. Edit the fields as before and select **OK** to confirm the changes.

➢ **To delete an existing assignment:**

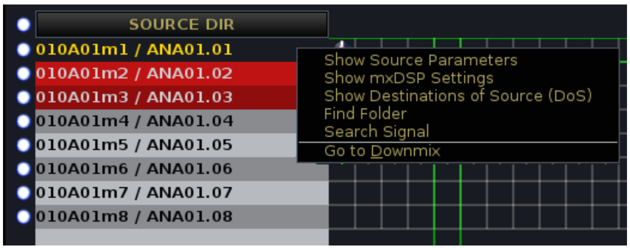
1. Select the **Function** and **Assignment** you wish to delete.
2. Right-click and select **Delete Assignment**.
3. Confirm by selecting **OK**.

The assignment is deleted.

Entering a HLSD Address

Some functions require you to enter the Lawo system address (HLSD address) for a signal. You can copy and paste this address from the Signal Settings display, or from the **mx Routing** display as follows:

1. Open the mx Routing display and locate the signal.
2. Right-click and select **Show Source Parameters** (or **Show Destination Parameters**):

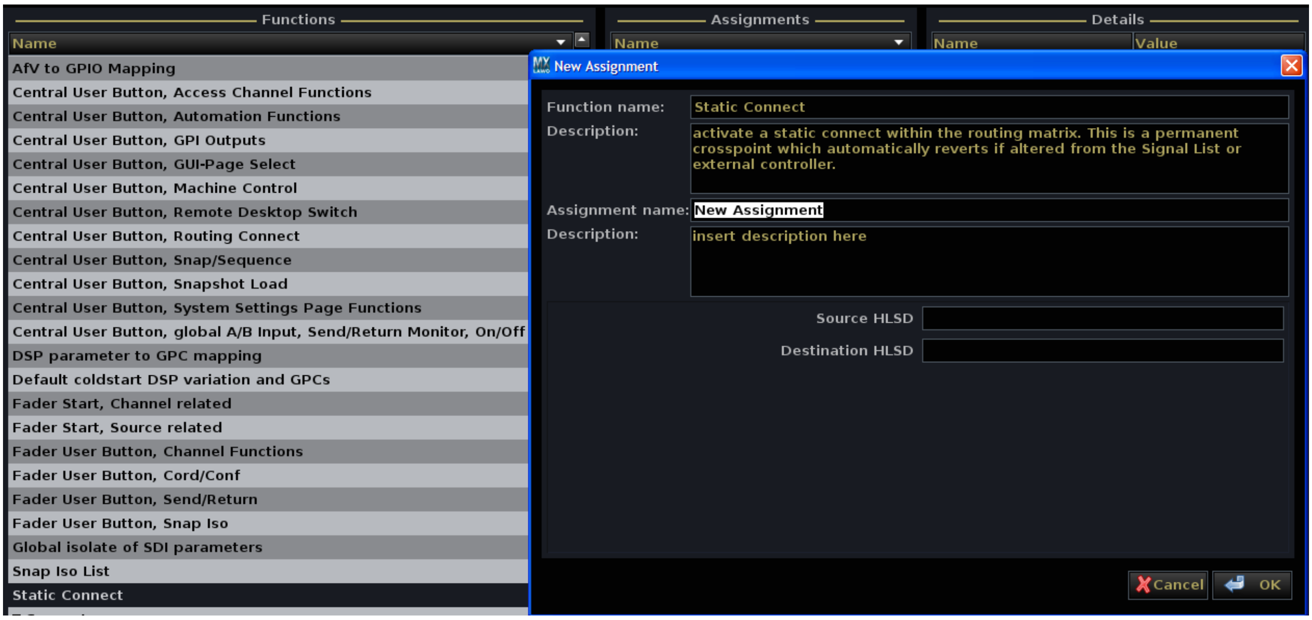


The Signal Parameters pop-up window appears:

3. Select the **Device** tab.
4. Select the **HLSD** address field, right-click and select **Copy** to copy the address:

5. Now return to the **Custom Functions** display.

6. Create a new function assignment, or edit an existing assignment – for example, a **Static Connect**:



7. Right-click on the **HLSD** field and select **Paste** to paste the copied address.

Default Custom Functions

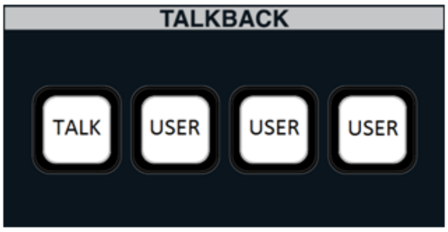
The following **Custom Functions** are pre-configured at the factory. You may edit these assignments at any time, or replace them with alternate functions.

Fader Strip User Buttons

The first four fader strip user buttons (User 1 to User 4) are programmed as follows:

- **Channel Userbutton 1 = CORD**, see Fader User Button, Cord/Conf.
- **Channel Userbutton 2 = CONF**, see Fader User Button, Cord/Conf.
- **Channel Userbutton 3** = **SNAP ISO**, see Fader User Button, Snap ISO.
- **Channel Userbutton 4** = **TALK**, see Fader User Button, Talkback to Channel. This button routes the console's talkback source to the channel's N-1 bus.

TALKBACK User Buttons



The first user button on the TALKBACK panel is pre-configured as follows:

- **Talkback Userbutton 1** = **TALK**, see Fader User Button, Talkback to Channel. This button routes the console's talkback source to the access channel's N-1 bus.

Importing and Exporting Custom Functions

When you create a custom function, the assignment is stored as a Custom Template file within the console's configuration data.

These files cannot be accessed from the console GUI. However, they can be transferred to an mxGUI computer via the File Transfer display. This provides a way of importing and exporting functions between systems, or creating a backup for your installation.

The Custom Functions List

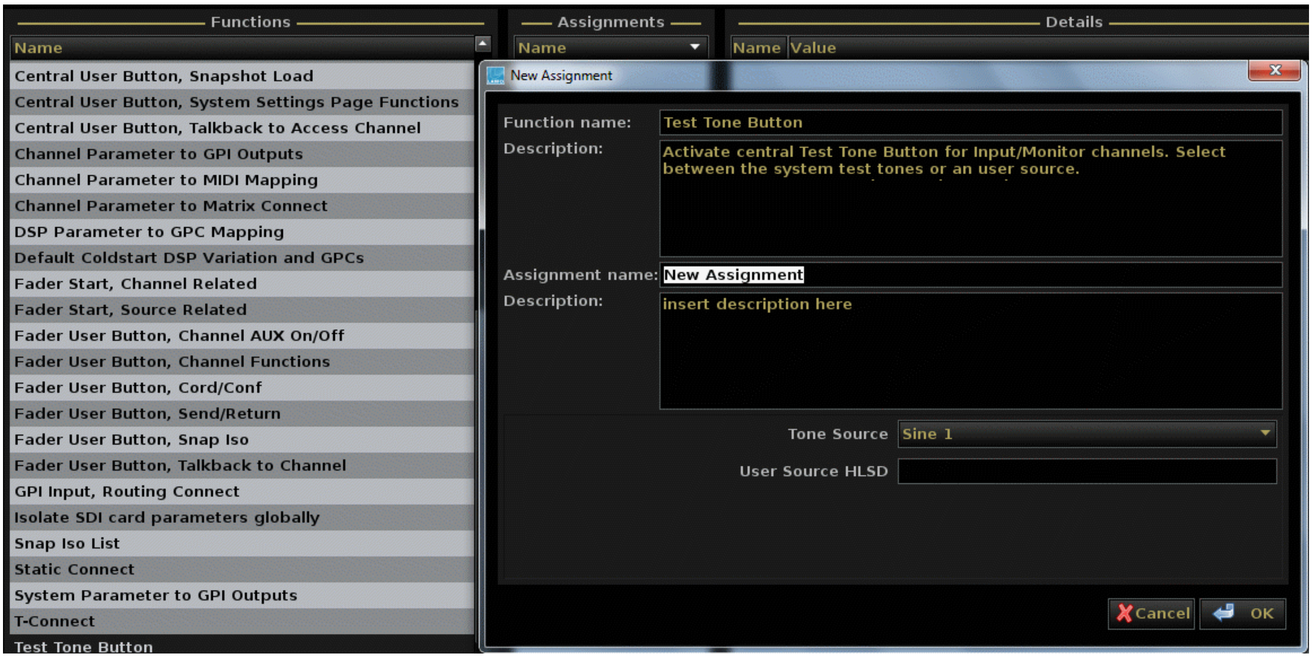
This section describes each of the custom functions, in alphabetical order:

Click on **Name** at the top of the **Functions** list to sort by name order.

- Activate Test Tone Button
- AFV to GPIO (and Logic) Mapping
- AFV to GPIO Mapping
- Bird Beater Aux
- Central User Button, Access Channel Functions
- Central User Button, Automation Functions
- Central User Button, GPI Outputs
- Central User Button, GUI-Page Select
- Central User Button, Global A/B Input, Send/Return Monitor, On/Off AFV
- Central User Button, Loudness Metering
- Central User Button, Loudness Start/Pause/Reset
- Central User Button, MIDI Command
- Central User Button, Machine Control
- Central User Button, Machine Locators
- Central User Button, Multi Row Metering
- Central User Button, Remote Desktop Switch
- Central User Button, Routing Connect
- Central User Button, Routing Toggle Connect
- Central User Button, Snap/Sequence
- Central User Button, Snapshot Load
- Central User Button, System Settings Page Functions
- Central User Button, Talkback to Access Channel
- Channel Parameter to GPI Outputs
- Channel Parameter to MIDI Mapping
- Channel Parameter to Matrix Connect
- DSP Parameter to GPC Mapping
- Default coldstart DSP variation and GPCs

- Fader Start, Channel related
- Fader Start, Source related
- Fader User Button, Channel Aux On/Off
- Fader User Button, Channel Functions
- Fader User Button, Cord/Conf
- Fader User Button, Send/Return
- Fader User Button, Snap Iso
- Fader User Button, Talkback to Channel
- GPI Input, Routing Connect
- Isolate SDI card parameters globally
- Snap Iso List
- Static Connect
- System Parameter to GPI Outputs
- T-Connect

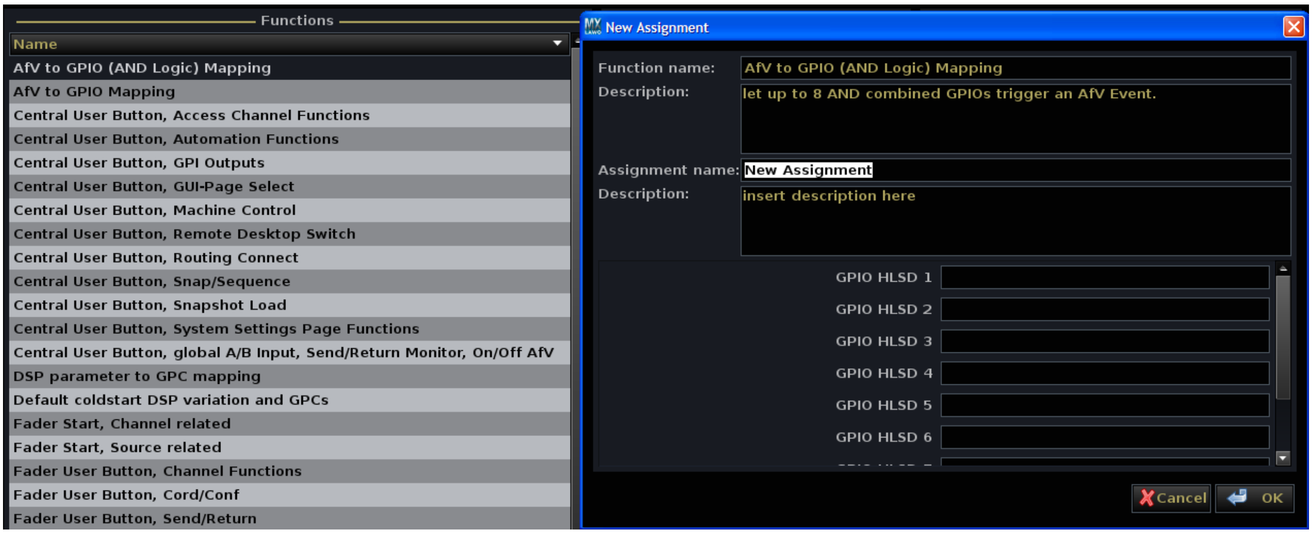
Activate Test Tone Button



From V4.24 software onwards, this function activates and specifies the source used for tone to channel switching. Once programmed, test tone is always active.

- **Tone Source** - select one of the drop-down options to specify either an internal tone generator source (**Sine 1**, **Sine 2**, **White Noise**, **Pink Noise**), or **User Input**.
- **User Source HLSD** - if **User Input** is selected, enter the HLSD address for the source here.

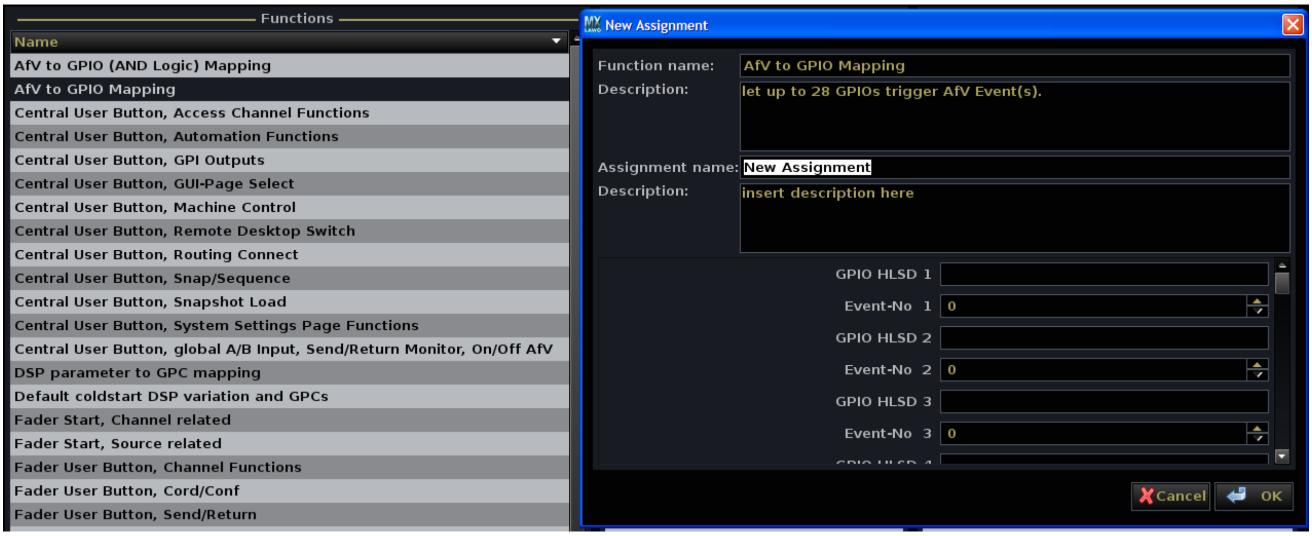
AFV to GPIO (and Logic) Mapping



This function triggers a single Audio Follow Video event from up to 8 AND combined GPIOs. The GPIO events can be In, Out, Relays or Optocoupler. Create multiple assignments if you wish to trigger several AFV events with AND combined GPIO logic.

You may create multiple instances of this template.

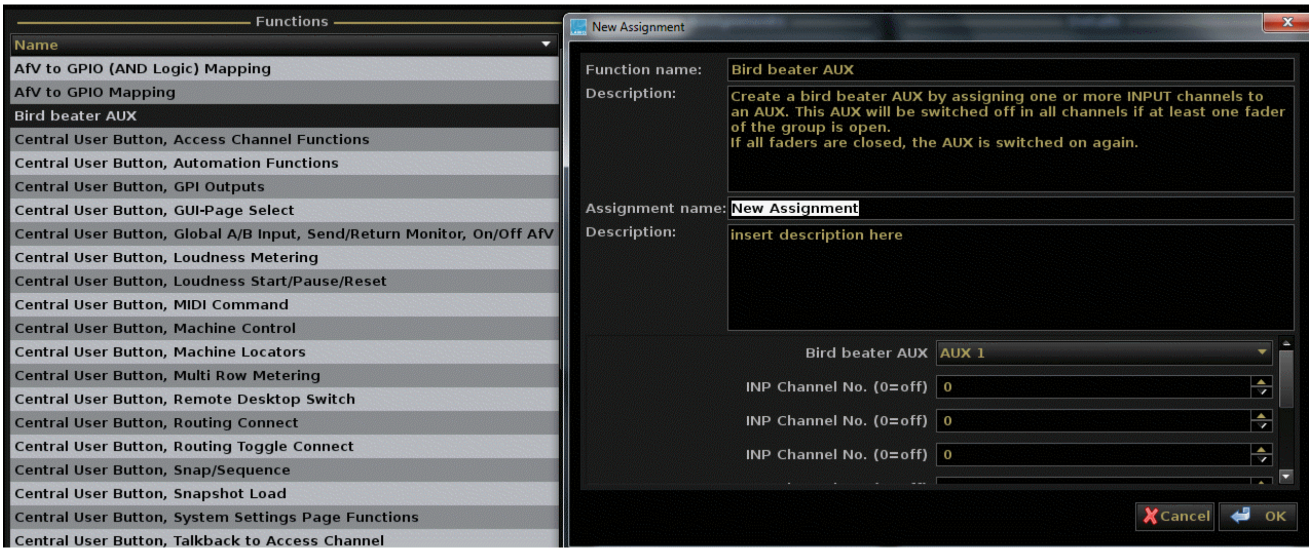
AFV to GPIO Mapping



This function triggers Audio Follow Video events from up to 28 GPIOs. You can create an OR combined GPIO by entering the same AFV event for all 28 GPIOs. Alternatively, you can mix OR combines GPIOs with a direct AFV Event assignment.

You may create multiple instances of this template.

Bird Beater Aux



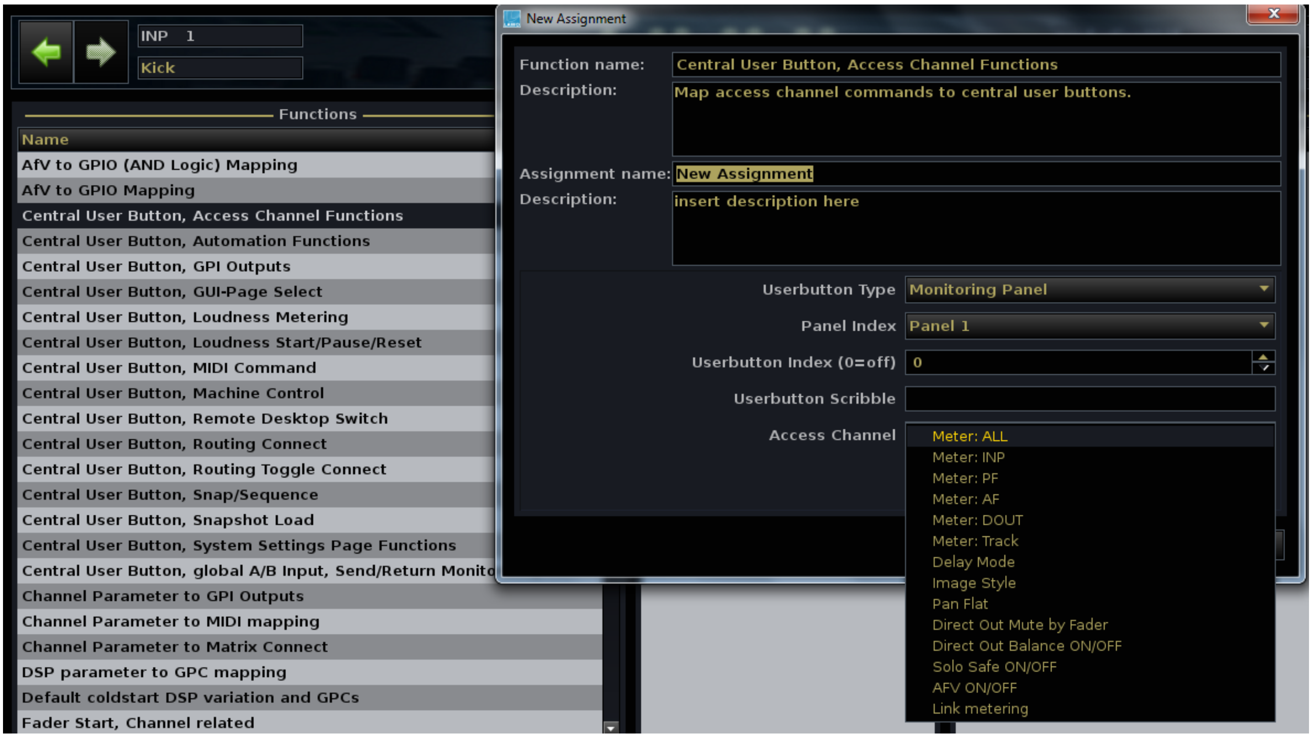
From V4.24 software onwards, this function allows you to create a "bird beater" aux - an aux send which automatically mutes when the channel fader opens and is "on air". This feature is similar to the Direct-Out Mute by Fader option, but affects an aux instead of the channel's direct output.

Note that you may OR the function so that the aux send mutes if at least one fader within the OR'd group opens. The group may contain up to 8 channels.

1. Use the **Bird beater AUX** field to select the aux bus you wish to use (from 1 to 32).
2. Use the **INP Channel No.** fields to define the input channel(s) which will trigger the "bird beater" aux - you should enter the mono DSP channel number. So, for example, if channels are stereo, enter the left channel number (1, 3, 5, 7, etc.).

You may create multiple instances of this template.

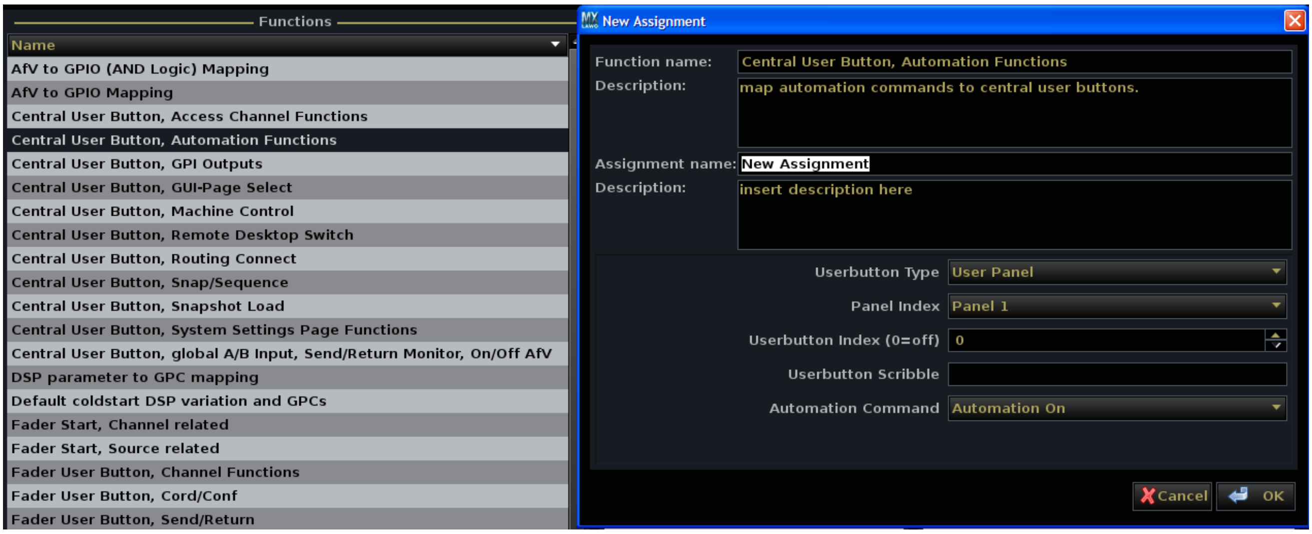
Central User Button, Access Channel Functions



Maps a central user button to functions which will act on the channel in access. Functions include:

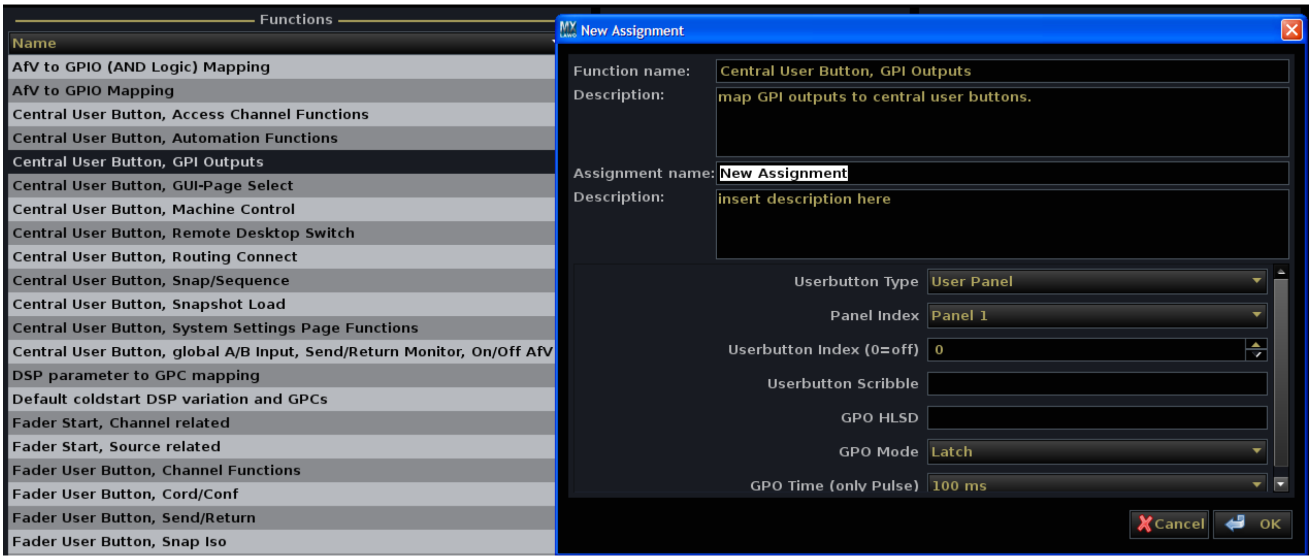
- Meter pickup point selection
- Delay mode, Image style and Pan flat
- Direct Out options
- Solo Safe on/off
- AFV (Audio Follow Video) on/off
- Link metering on/off

Central User Button, Automation Functions



Map a central user button to timecode automation functions.

Central User Button, GPI Outputs



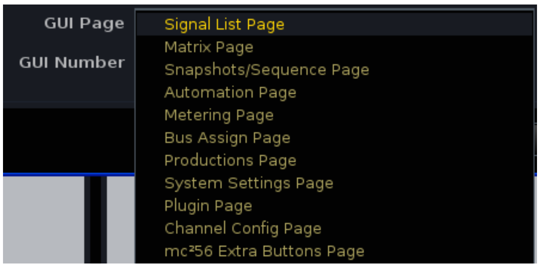
Maps a central user button to external relays (GPI Outputs). For each user button define the:

- **GPO HLSD** – this is the Lawo system address of the GPO which will be triggered.
- **GPO Mode** – latching, momentary, pulse, etc.
- **GPO Time –** for a pulsed relay.

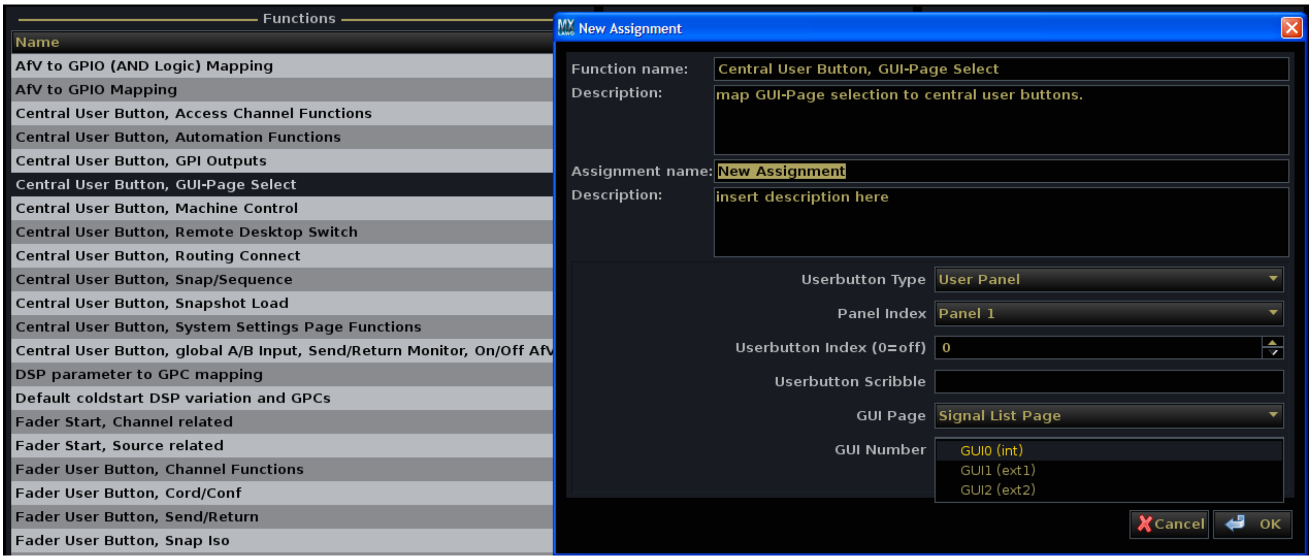
Central User Button, GUI-Page Select

Maps a central user button to GUI page switching. For example, you could assign user buttons to switch different displays to an external screen. For each user button define the:

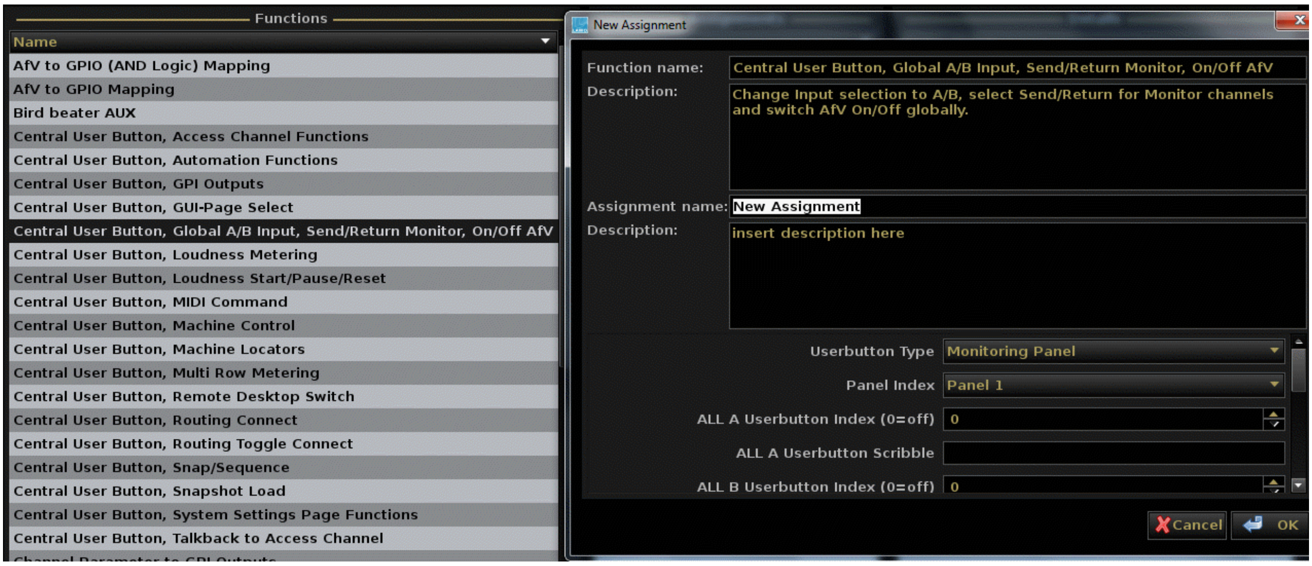
- **GUI Page** which the button will select – **Signal List**, **Matrix**, **Snapshots/Sequence**, etc.



- **GUI Number** (internal, external 1 or external 2) which will be switched:



Central User Button, Global A/B Input, Send/Return Monitor, On/Off AFV



This function allows you to switch a number of operations globally from a central user button. You may assign:

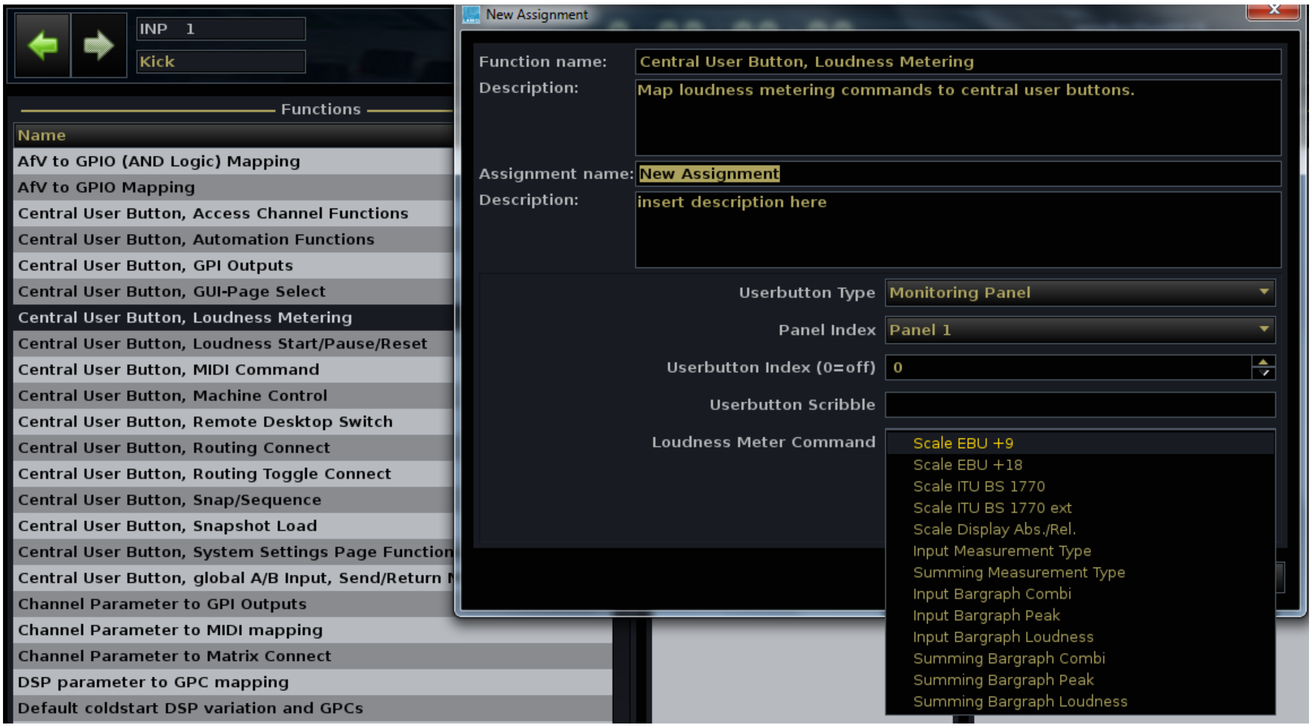
- Global A/B Input Switching – use this function to switch all input channels between A and B sources (if assigned).

Note that if B Inputs are not assigned, then the status LED of the "All B" button will not light. The button will still switch to Input B on Inputs where a source is assigned.

- Global Send/Return Switching on monitor channels – handy for multitrack recording sessions.
- Global AFV On/Off Switching – handy if cameras are rehearsing (you can switch AFV off).

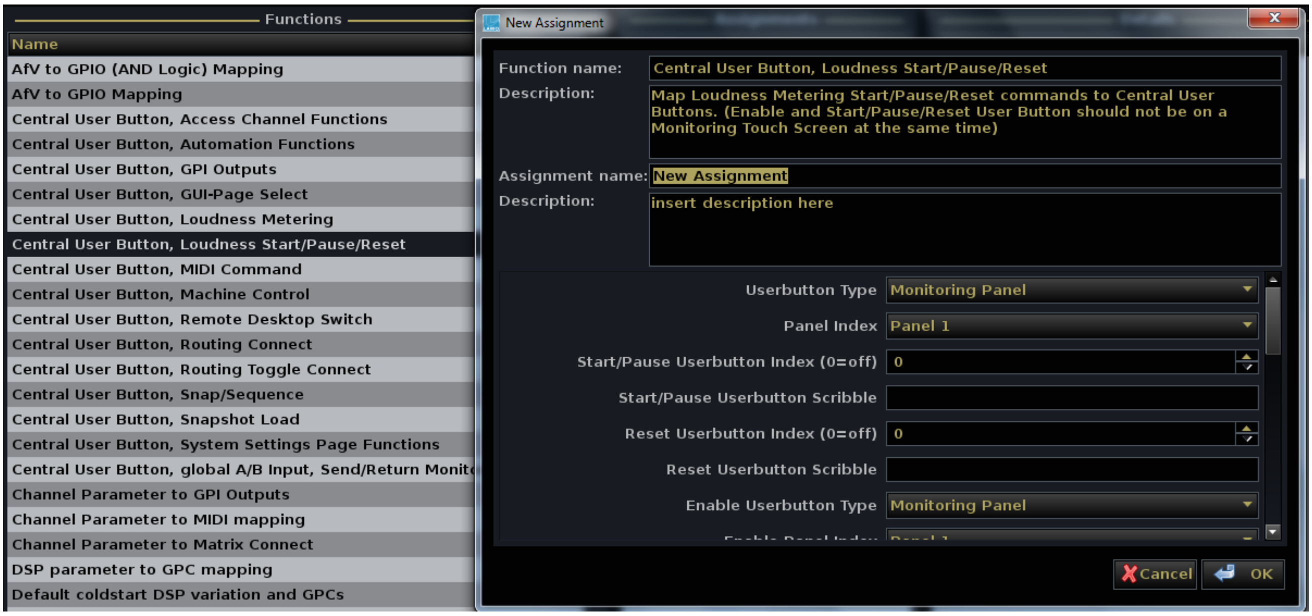
You may create multiple instances of this template if you wish to have functions on different User Button Panels.

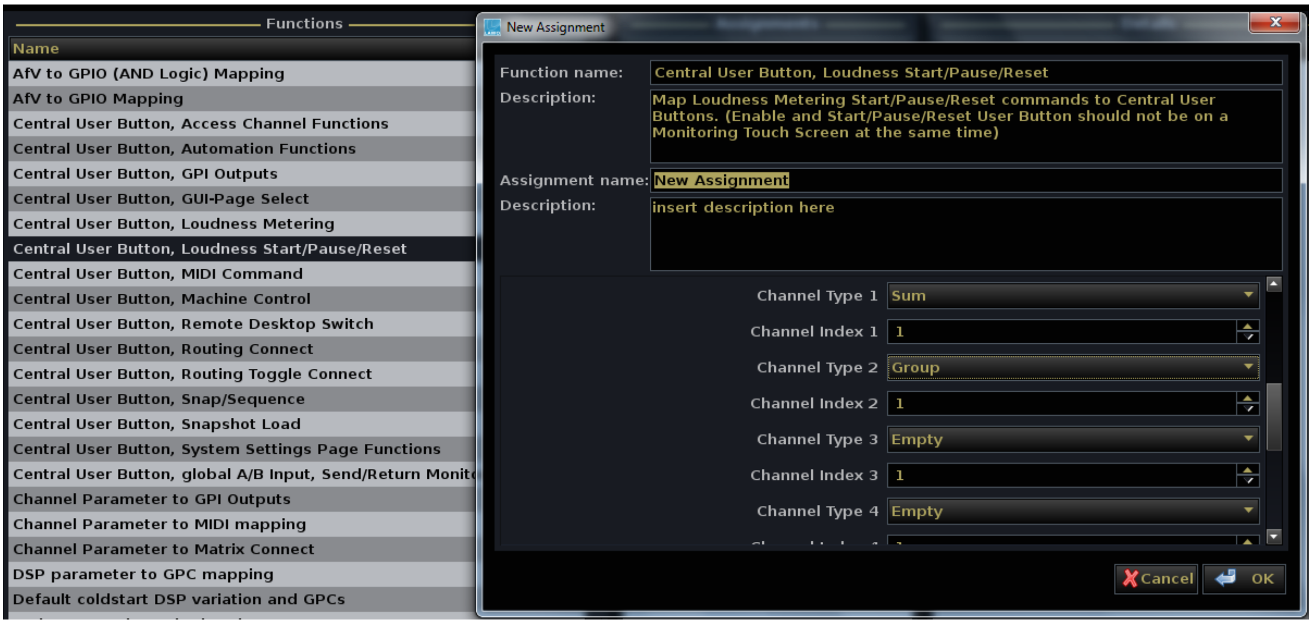
Central User Button, Loudness Metering



Maps a central user button to the Loudness Metering options available from the **System Settings** display.

Central User Button, Loudness Start/Pause/Reset

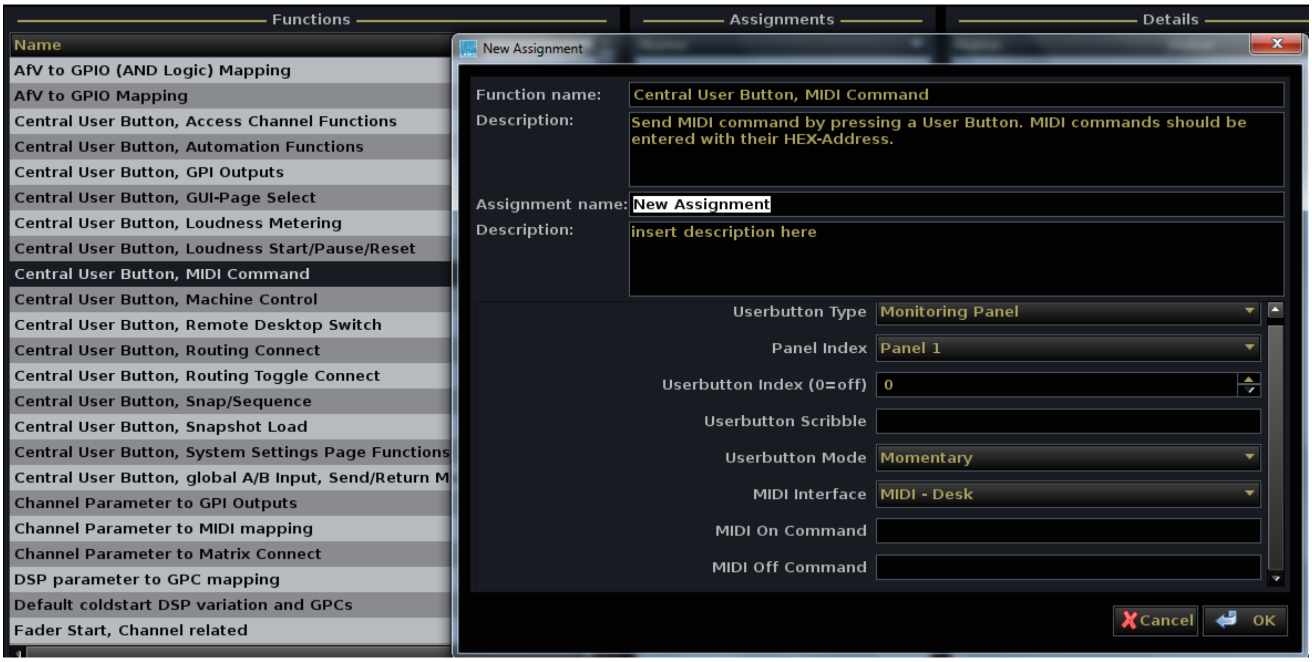




Maps a central user button to start, or reset, the integrated loudness measurement on up to 8 specific summing channels, see Integrated Loudness Measurement.

Note that you these functions can also be mapped to fader strip user buttons, using the Fader User Button, Channel Functions template.

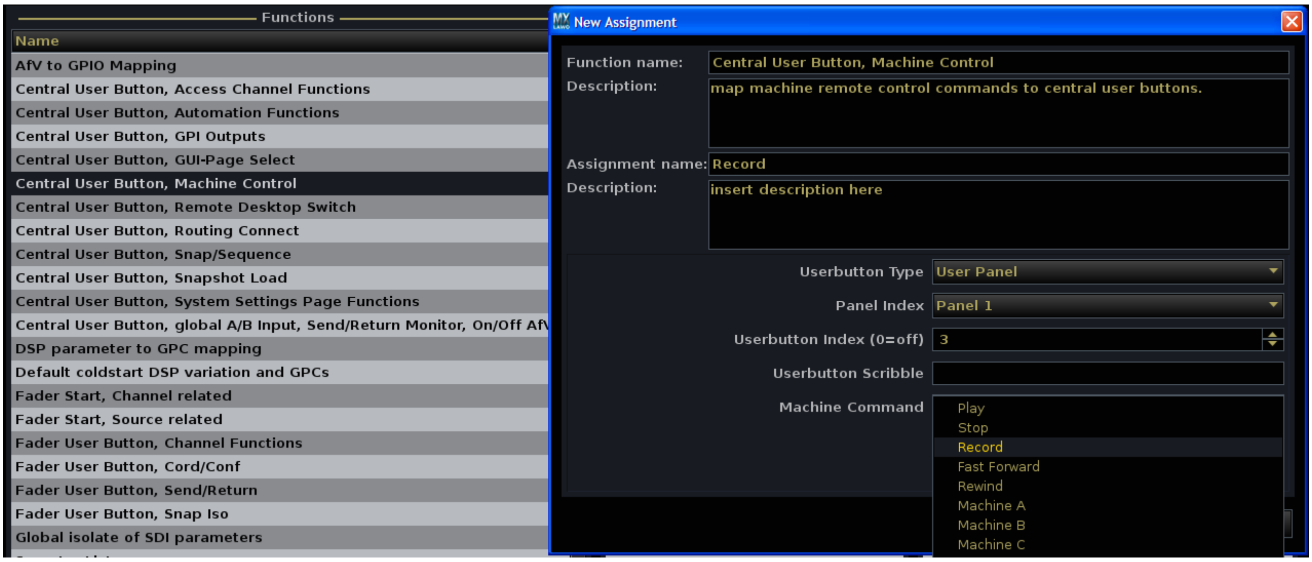
Central User Button, MIDI Command



Maps a central user button to MIDI commands. For each user button define the:

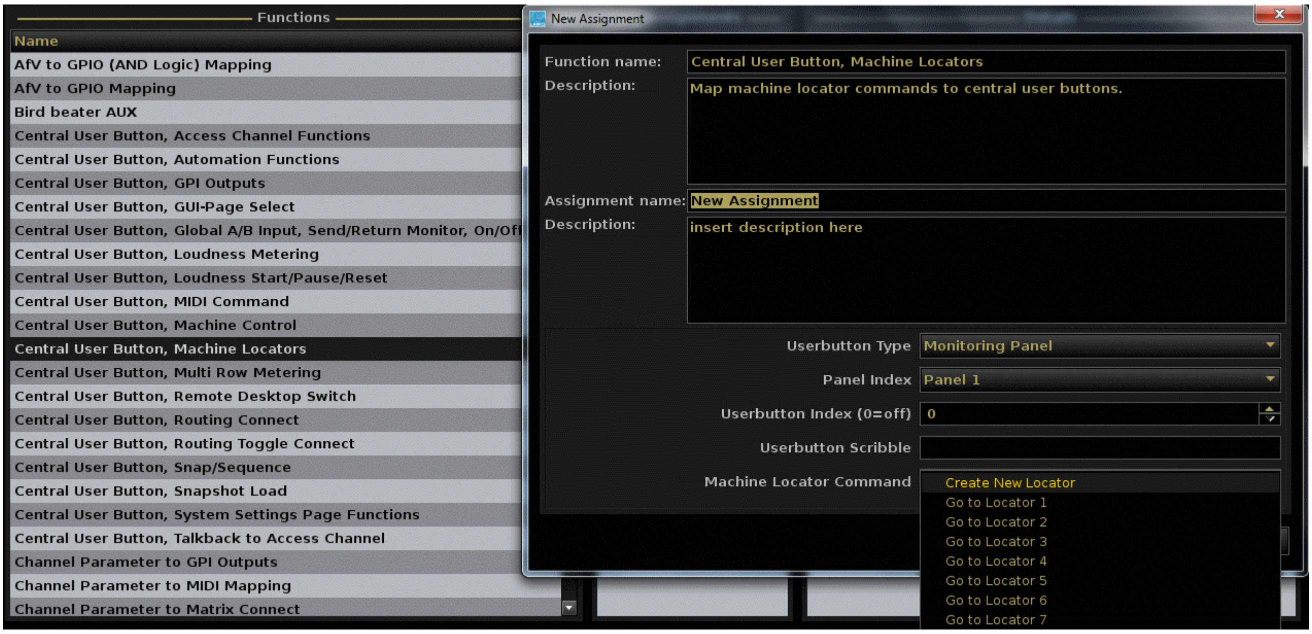
- **MIDI Interface**:
  - o **DESK** – MIDI is connected to the MIDI IN/OUT sockets on the rear of the console.
  - o **LAN 1 to 16** – MIDI is transmitted via the Lawo network; select the network client from 1 to 16.
- **MIDI On/Off Commands** – enter the hexadecimal address for the MIDI Command. For example:
  - o **0xc0 0x07** = Program Change to MIDI ch 1; Patch Number 8.
  - o **0xc2 0x03** = Program Change to MIDI ch 3; Patch Number 4.

Central User Button, Machine Control



Maps a central user button to machine control commands.

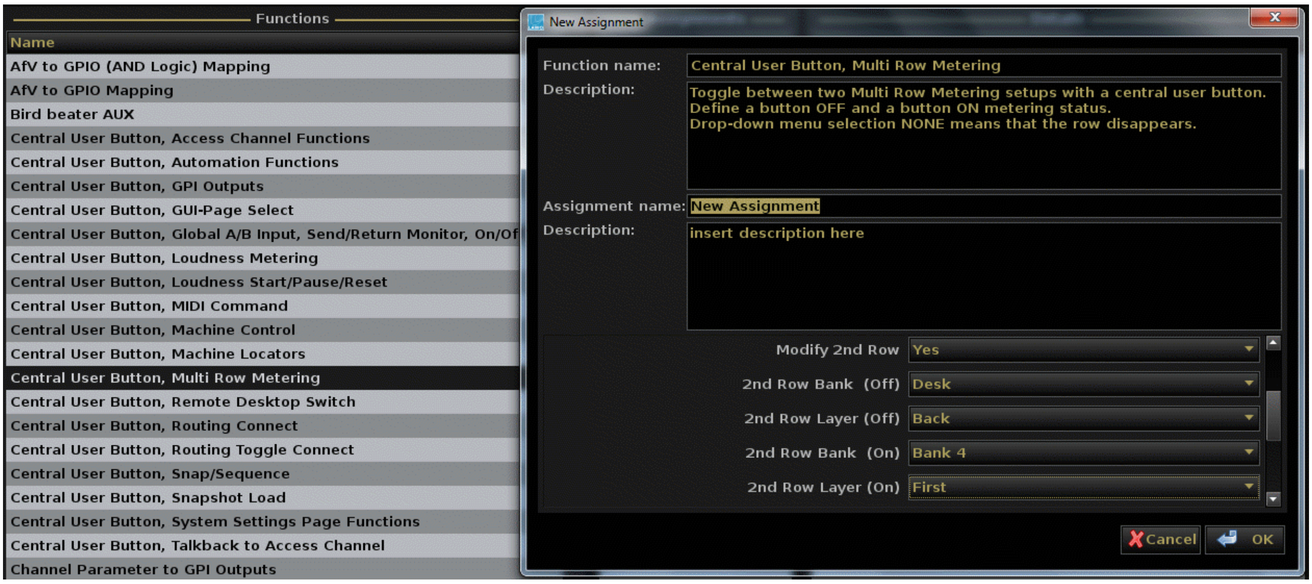
Central User Button, Machine Locators



Maps a central user button to machine locator commands (cue points). The drop-down options include:

- **Create New Locator** - stores the current timecode position into a new locator (identical to **New** on the **Machine Locators** display).
- **Go to Locator 1** to **10** - recalls a stored locator from ID 1 to 10.

Central User Button, Multi Row Metering



From V4.24 software onwards, this function maps a central user button to switch the second and third **Metering Row** options available from the **System Settings** display, see Multi-row Metering. This allows you to toggle between two metering setups.

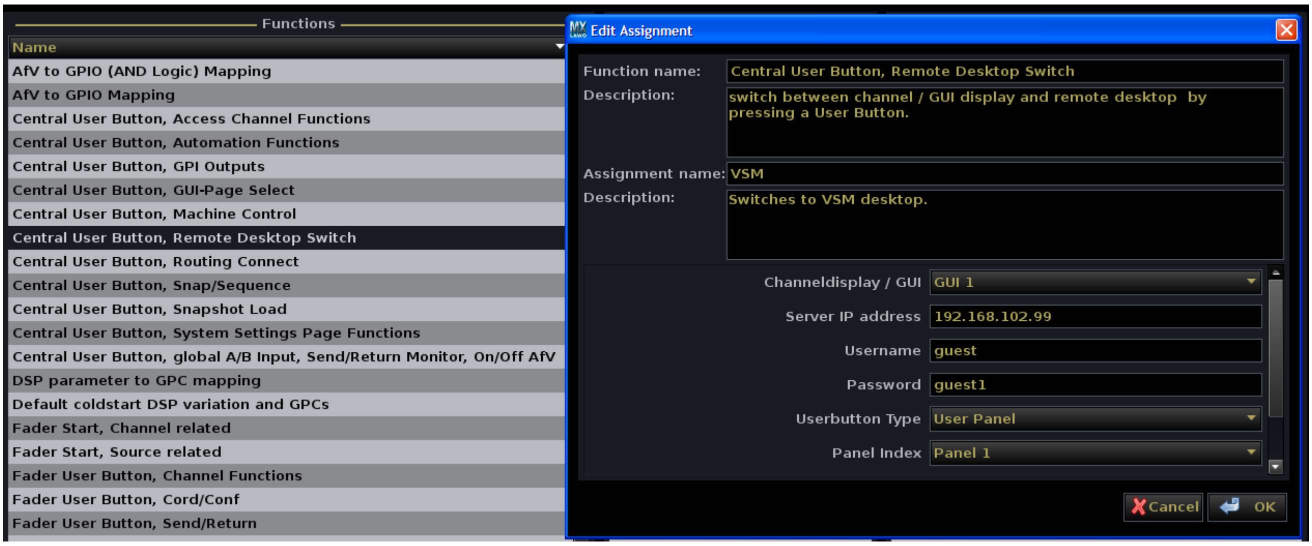
To enable user button switching, set the **Modify 2nd Row** field to **Yes**. (If this field is set to **No**, then the metering row is always assigned to the options defined in the System Settings display.)

Each user button has an "On" and "Off" state - so, in our example, when the user button is "Off", the 2nd row *always* meter channels assigned to the alternate Layer of the active Bank (**Desk** + **Back**); when the user button is "On", the 2nd row switches to meter channels assigned to the Bank 4, Layer 1 (**Bank 4** + **First**).

Select **None** from the **2nd Row Bank** options to disable the second metering row.

On the MKII mc²56, you can map user buttons to switch the third metering row in a similar manner.

Central User Button, Remote Desktop Switch



This function allows any of the console's TFT displays to be switched to a remote desktop in order to view and control other applications – for example, a playback system or DAW. You may use any central user button to action the function.

On the Remote Server:

Connect the remote desktop server to the Lawo system network, via ETHERNET B on the Router Module (MKII).

The server should have a fixed IP address (fixed IP from DHCP server, or static IP address) within the same range as that of the control system, for example **192.168.102.xxx**. You can check the IP address of your control system from the **Signal Settings** display, see System Tree Structure.

Create a new user and password for the remote desktop login. We advise creating a new user, as the password is displayed and stored in clear text on the mc² mixing console. The user must be a member of the "Remote Desktop Users" Group.

On the Console:

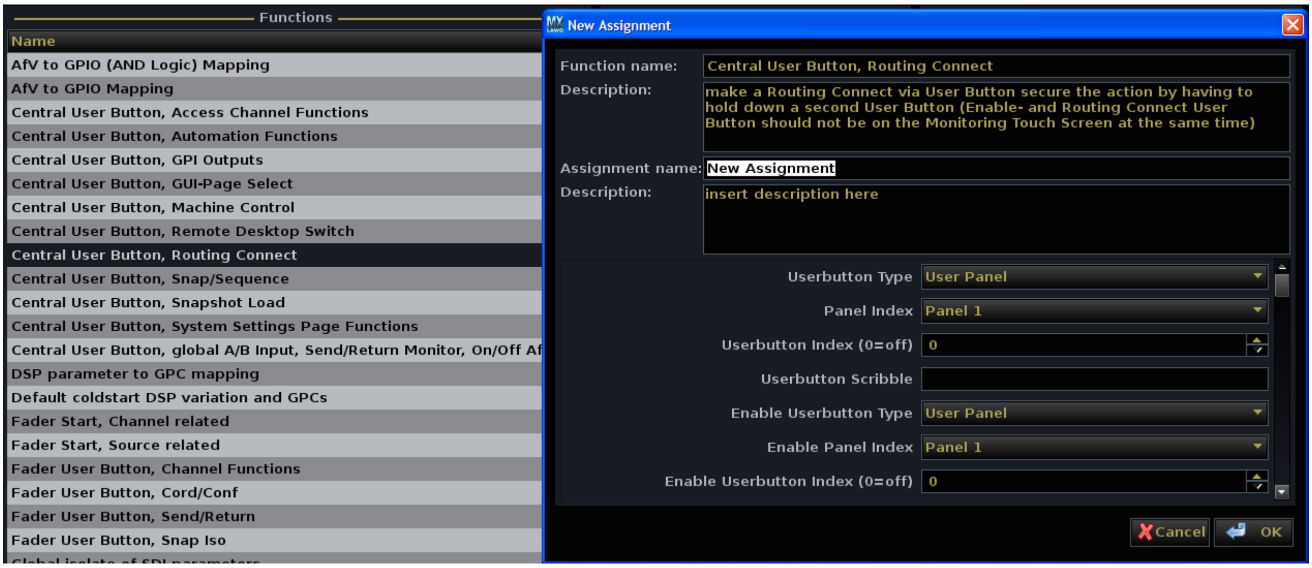
For each remote desktop you wish to connect to, define the:

- **Channeldisplay / GUI** – select the console display which will display the remote desktop. You may choose any Channel display or Central GUI.
- **Server IP address** – enter the IP address of the server (as configured above).

Note that the control system must have an IP route to the remote server. If the server is not in the same subnet range, then an appropriate gateway must be configured. Please consult your network administrator for assistance. The default Subnet Mask is **255.255.255.0**.

- **Username** – enter the remote server's Username configured above..
- **Password** – enter the remote server's Password configured above.
- **Userbutton Type**, **Panel Index**, etc. – assign the user button which will switch to and from the remote desktop.

Central User Button, Routing Connect



This function allows you to perform signal routing from a central user button. Up to 28 connects/ disconnects can be assigned to one button. Routes may be made to/from any source or destination including DSP channels.

You may create multiple instances of this template.

Copy the source and destination **HLSD** from the **mx Routing** display, see Entering a HLSD Address.

To create a disconnect, type **DISCONNECT** into the field for the Source HLSD.

To secure the operation, you can define an **Enable Userbutton**. Once defined, you will need to hold down **Enable** while pressing the **Connect** user button in order to action the connects/disconnects.

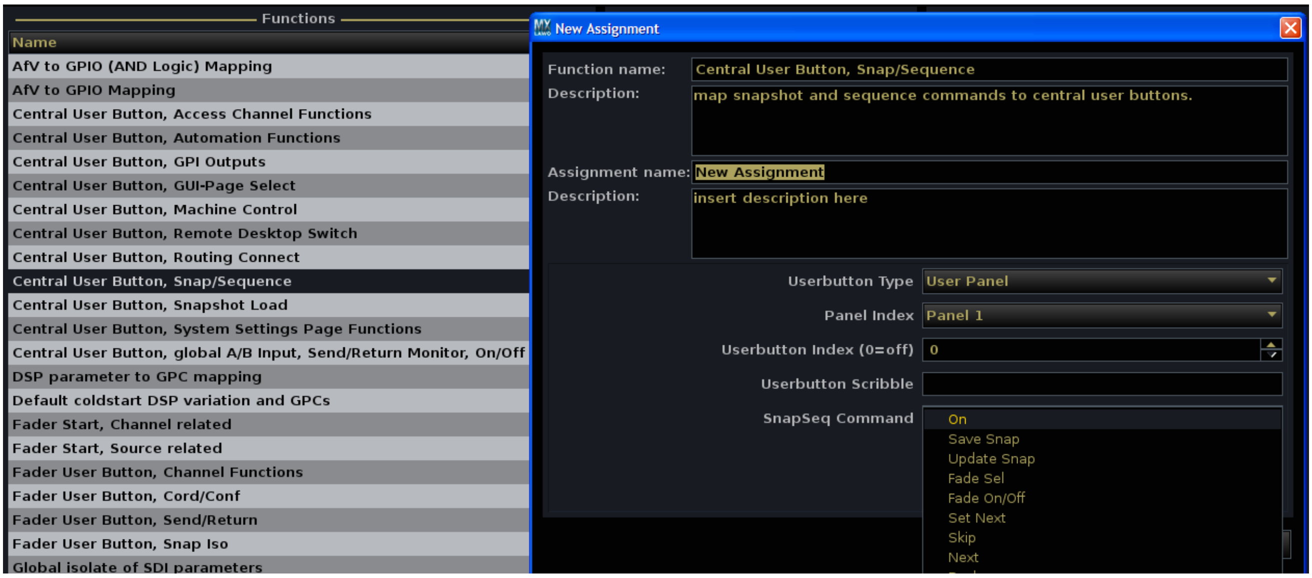
Note that the **Enable** and **Routing Connect** user buttons are not multi-touch capable, therefore do not assign them to a touch-screen.

Central User Button, Routing Toggle Connect

This function is similar to the Central User Button, Routing Connect function. But it provides source on and source off states so that routes may toggle. Up to 16 connects/disconnects can be assigned to one user button.

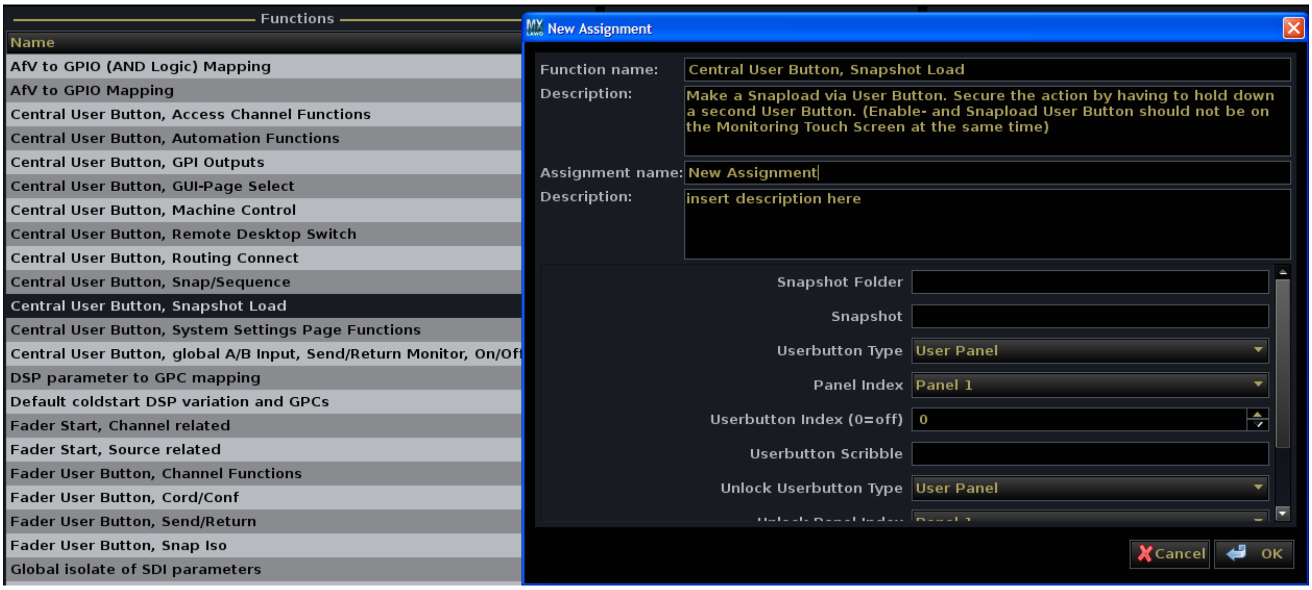
You may create multiple instances of this template.

Central User Button, Snap/Sequence



Maps a central user button to snapshot and sequence commands.

Central User Button, Snapshot Load



This function allows you to load a specific snapshot from a single user button press. The snapshot may come from any folder within the active production.

You can make the operation more secure by defining an **Unlock** user button. This means that the operator must press and hold the **Unlock** button while pressing the **Snapshot Load** in order to recall the snapshot. You may use any central user button to action the functions.

> Note that the **Unlock** and **Snapshot Load** user buttons are not multi-touch capable, therefore do not assign them to a touch-screen.

For each function, define the:

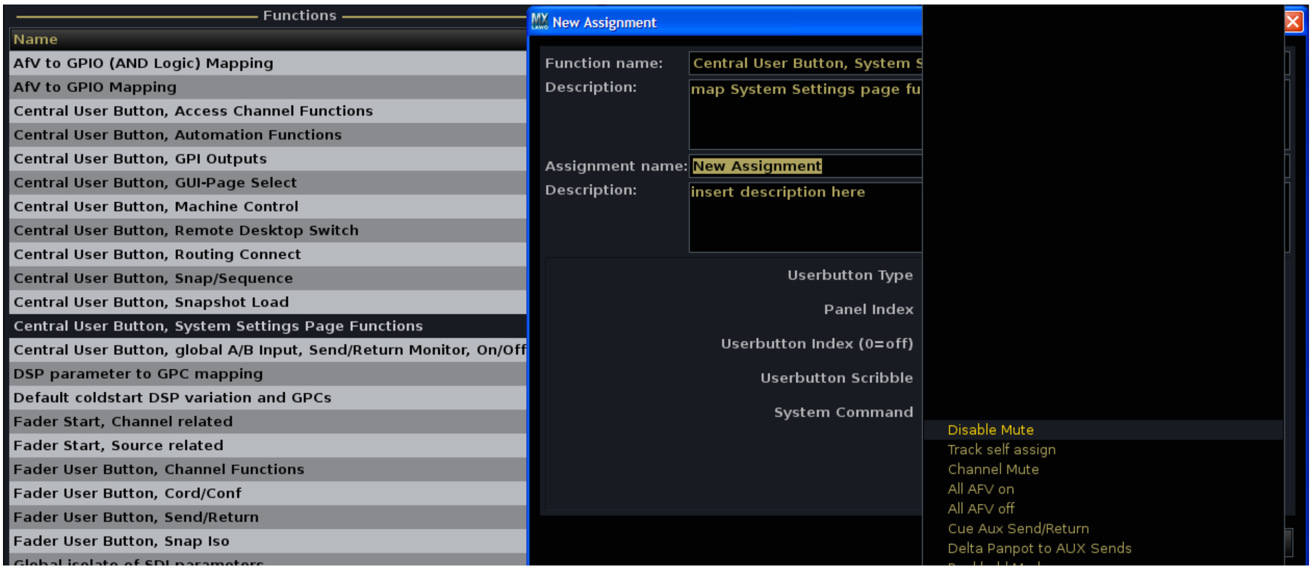
- **Snapshot Folder** – the name of the Folder where the snapshot is stored.
- **Snapshot** – the name of the Snapshot you wish to load.
- **Userbutton Type**, **Panel Index**, etc. – the user button which will action the Snapshot Load.
- **Unlock Userbutton Type**, **Panel Index**, etc. – the user button which will action the Unlock function.

If the **Unlock** user button is empty, then the **Snapshot Load** will action on a single press of the first user button.

Central User Button, System Settings Page Functions

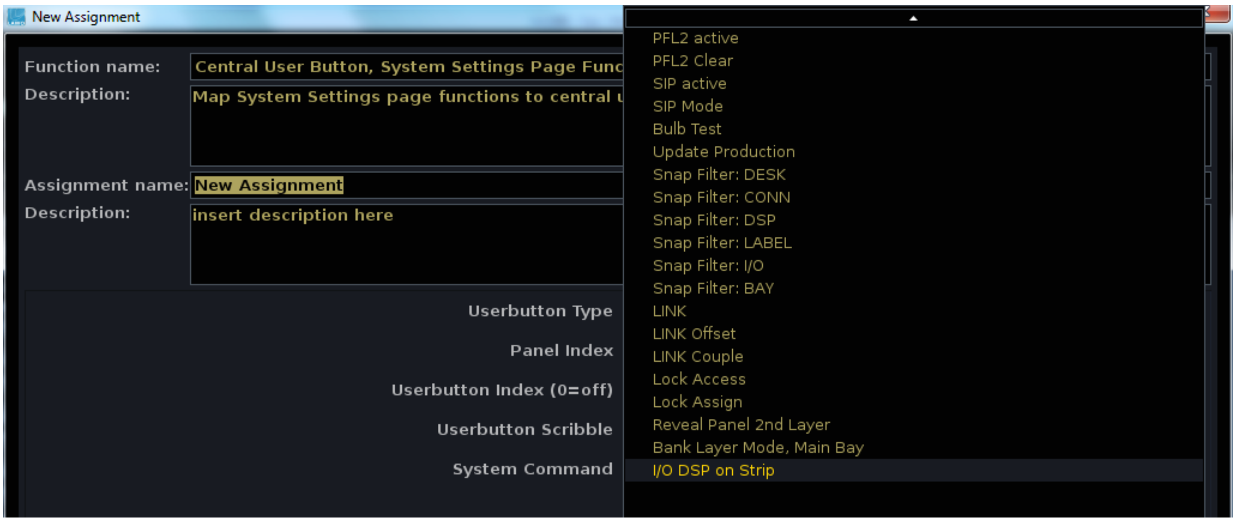
Maps a central user button to system options, allowing them to be changed 'on the fly'.

Most of the options in this list come from the System Settings display:

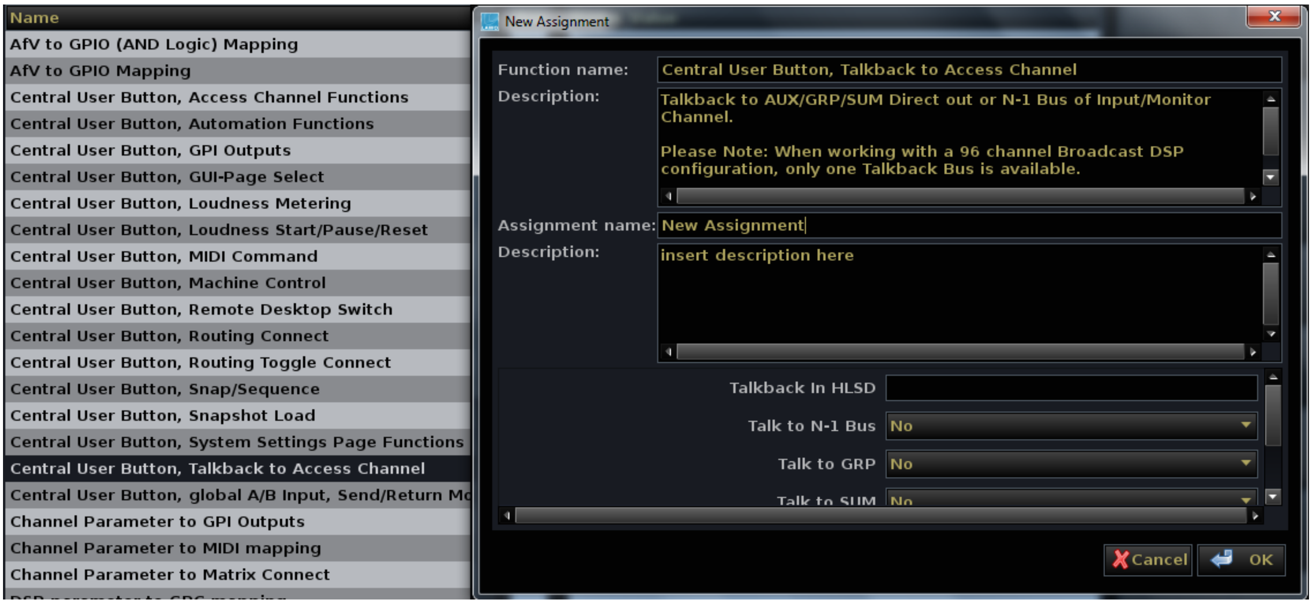


Those which do not are:

- **Snap Filter** – mimic the Global Snapshot ISO buttons.
- **LINK**, **Lock** and **Bank Layer Mode: Main Bay** – mimic the front panel buttons of the same name.
- **I/O DSP on Strip** – changes the INPUT MIXER mode from SOURCE to INMIX.



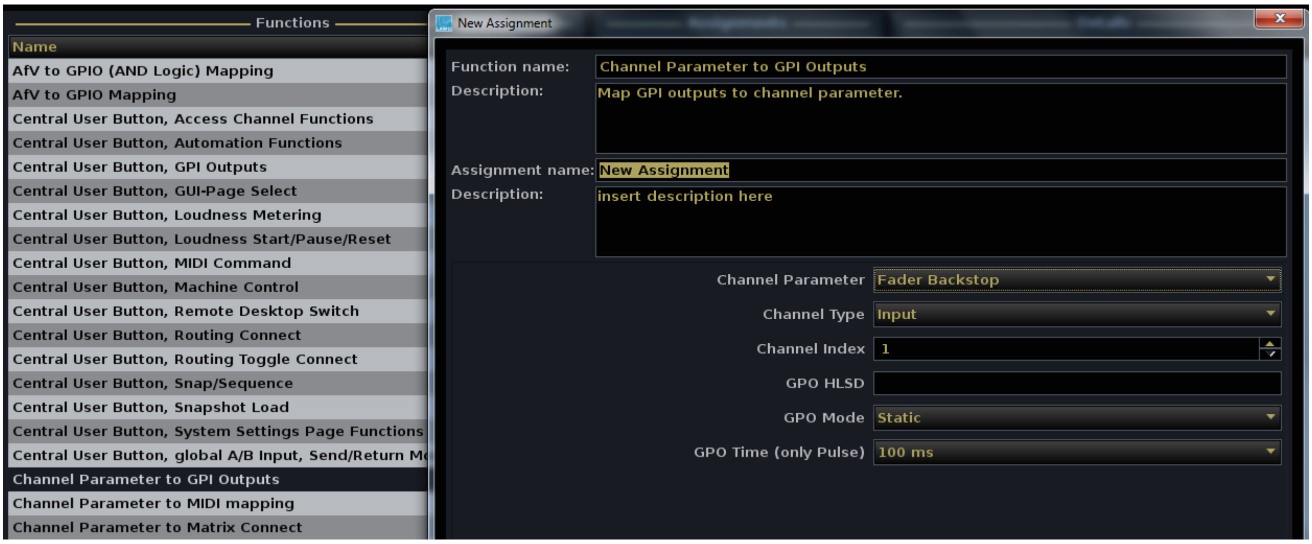
Central User Button, Talkback to Access Channel



Maps a central user button to talkback switching.

This function is programmed in an identical manner to the fader user button talkback function, see Fader User Button, Talkback to Channel. The only difference being that talkback is applied to the channel in access.

Channel Parameter to GPI Outputs



Maps a channel parameter to a GPI output.

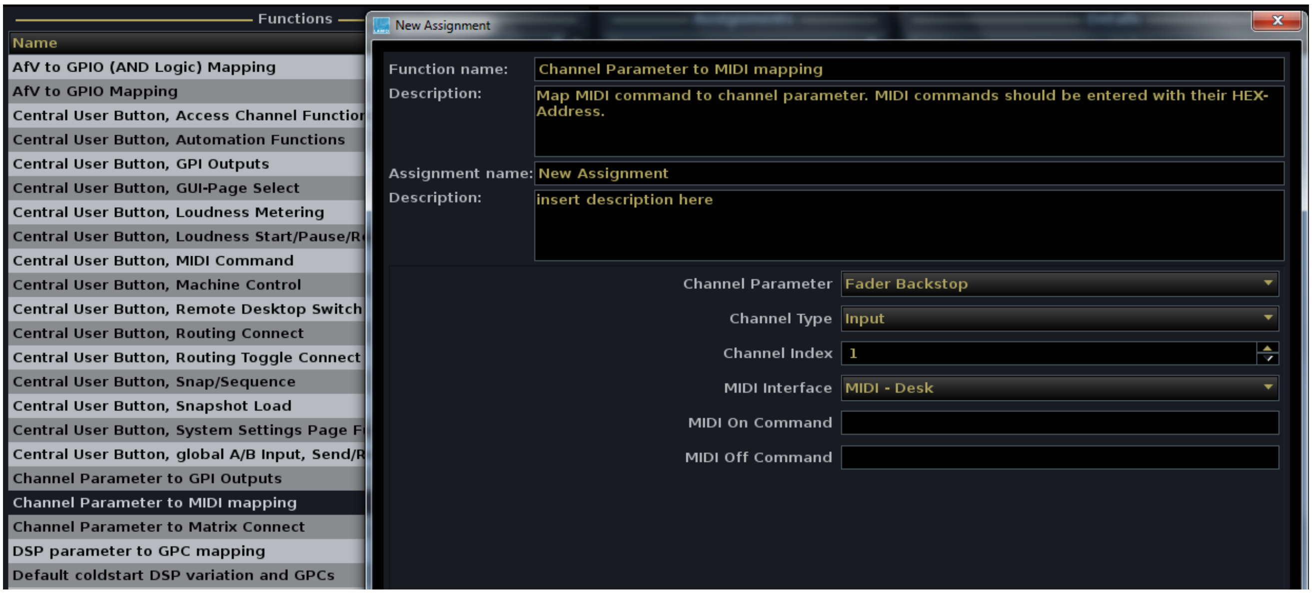
The custom function defines the channel parameter, type and number, and then the HLSD, Mode and Time for the GPI. Note that the GPI is triggered from the channel, and therefore will follow if the channel is assigned to a different fader strip.

Channel parameters include:

- **Fader Backstop** – active when you pull back on the fader. Note that Fader Backstop must be turned **On** in the **System Settings** display.
- **Fader start** – active whenever the fader is opened.
- **Userbuttons 1** to **n** – active when the fader strip user buttons are turned on.

Note that fader start GPIs may also be programmed using the Fader Start, Source related or Fader Start, Channel related custom functions (allowing multiple channels or sources to be assigned to each relay).

Channel Parameter to MIDI Mapping

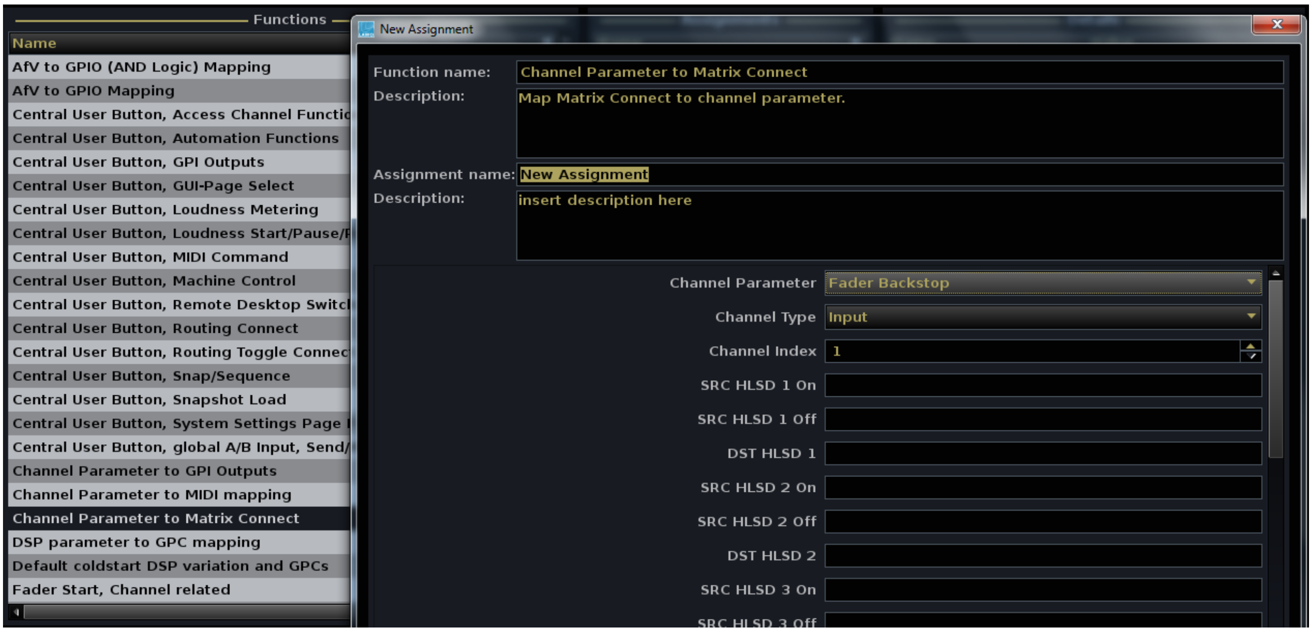


Maps a channel parameter to a MIDI Command. The channel parameters are identical to those available for Channel Parameter to GPI outputs function.

For the MIDI command, define the:

- **MIDI Interface**:
  - o **DESK** – MIDI is connected to the MIDI IN/OUT sockets on the rear of the console.
  - o **LAN 1 to 16** – MIDI is transmitted via the Lawo network; select the network client from 1 to 16.
- **MIDI On/Off Commands** – enter the hexadecimal address for the MIDI Command. For example:
  - o **0xc0 0x07** = Program Change to MIDI ch 1; Patch Number 8.
  - o **0xc2 0x03** = Program Change to MIDI ch 3; Patch Number 4.

Channel Parameter to Matrix Connect



Maps a channel parameter to signal routing. For example, you could choose to disconnect certain routes, such as a studio loudspeaker, when a channel fader is opened. Routes may be made to/from any source or destination including DSP channels.

Channel parameters include:

- **Fader Backstop** – active when you pull back on the fader. Note that Fader Backstop must be turned **On** in the **System Settings** display.
- **Fader start** – active whenever the fader is opened.
- **Userbuttons 1** to **n** – active when the fader strip user buttons are turned on.
- **Aux 29 to 32 On/Off** – active when the channel Aux on/off button is turned on.

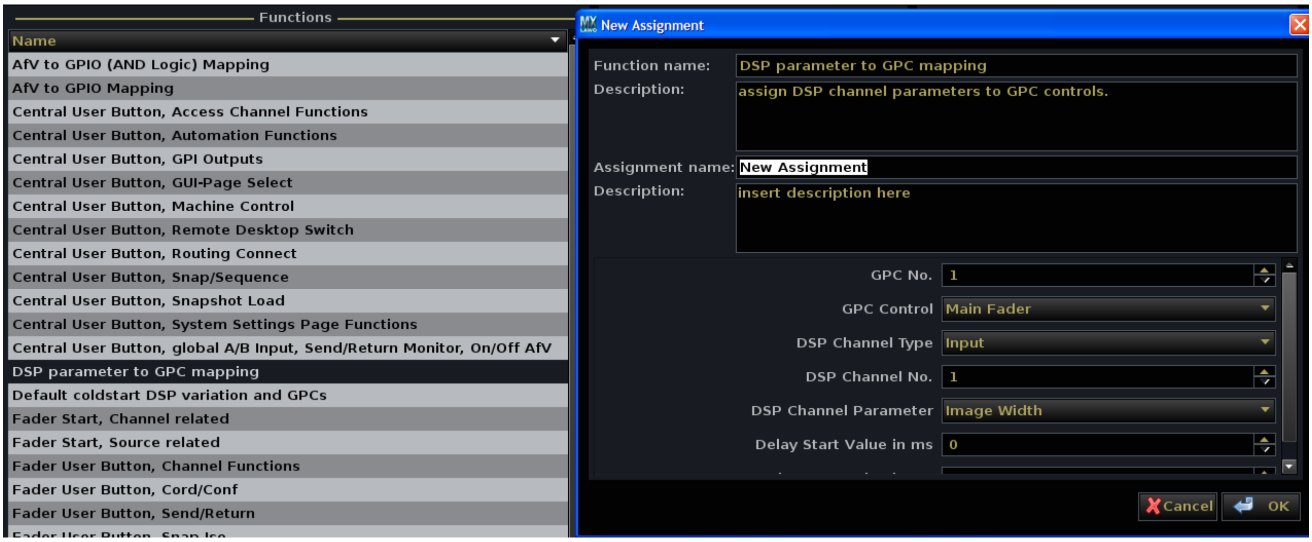
Copy the source and destination **HLSD** from the **mx Routing** display, see Entering a HLSD Address.

To create a disconnect, type **DISCONNECT** into the field for the Source HLSD.

Up to 8 connects/disconnects can be assigned to one custom function.

You may create multiple instances of this template.

DSP Parameter to GPC Mapping



This function offers a new way to control and automate DSP parameters by assigning a DSP channel parameter to a General Purpose Channel (GPC) control.

The **GPC Control** can be the Main fader or one of the GPC Auxes.

The **DSP Channel Type** can be any Input, Monitor, Group, Sum, Aux or Surround VCA channel.

The **DSP Channel Parameter** can be: Image Width or Position; Panning Left-Right or Front-Back; Slope; Hyperpanning: Turn, Front and Back Width, Depth; EQ Gain for Bands 01, 02, 03, 04; Digiamp; Insert Send; Direct Out or Delay.

Each GPC control is assigned to a single DSP parameter, so if you wish to control more than one parameter at a time, then do this by linking the GPCs. 256 GPCs are available.

By combining this function with the AFV to GPIO template, you can change DSP parameters from an Audio Follow Video event.

For example, to adjust delay for wireless cameras automatically:

1. Create an AFV to GPIO custom function where all GPIOs from the wireless cameras OR combine to trigger one AFV event.

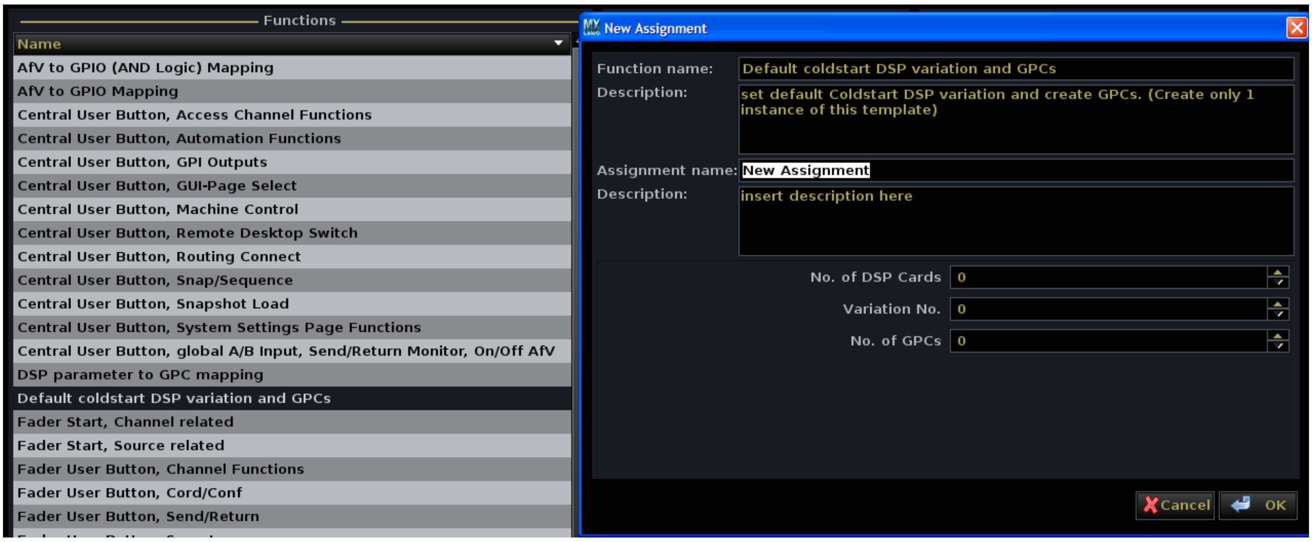
2. Then use a **DSP Parameter to GPC Mapping** custom function to assign the Delay parameter of the audio Group (mixing the wireless cameras) to a GPC.

3. Assign the AFV event to the General Purpose Channel.

Warning

Do *NOT* link GPCs which control the same DSP parameter in a contradictory manner, as the system may react badly.

Default coldstart DSP variation and GPCs



This function sets the DSP Configuration which will be loaded after a cold start.

Enter the number of DSP boards fitted, and the **Variation No.** which you wish to load.

Note that the **Variation No.** is *NOT* the **Index** number displayed on the DSP Configuration display.

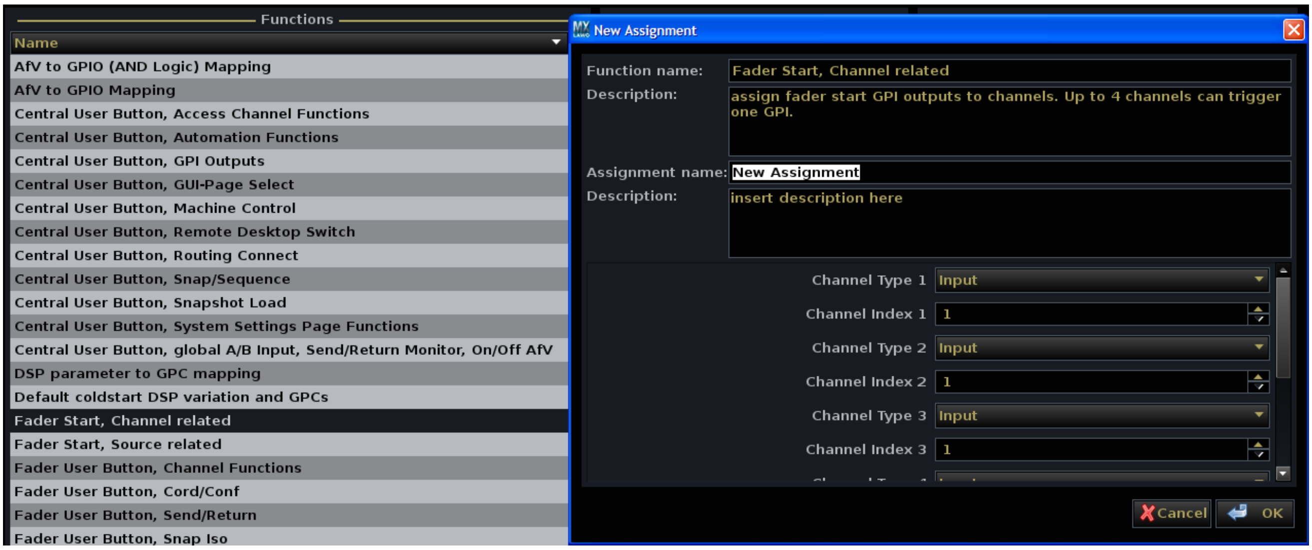
To calculate the **Variation No.**, open the **DSP Configuration** display, and sort the **Configuration Presets** list by the number of Inputs, in descending order. Now count down from the top of the list to find the Variation number.

Also note that there is no feedback from the system if the chosen variation is not available. If the DSP configuration preset cannot be loaded, then you see that there is no active DSP configuration preset once the system restarts.

The **No of GPCs** field has no function in the current release of software, and is reserved for future implementation.

Although you can create multiple instances of this template, it is not recommended - the last one initialised wins.

Fader Start, Channel related



Maps DSP channels to an external relay (GPI output) in order to create a fader start. Once the DSP channel is assigned to a physical fader, the fader triggers the start. Up to 4 DSP channels can be assigned to each relay.

For each of the 4 channels, define the:

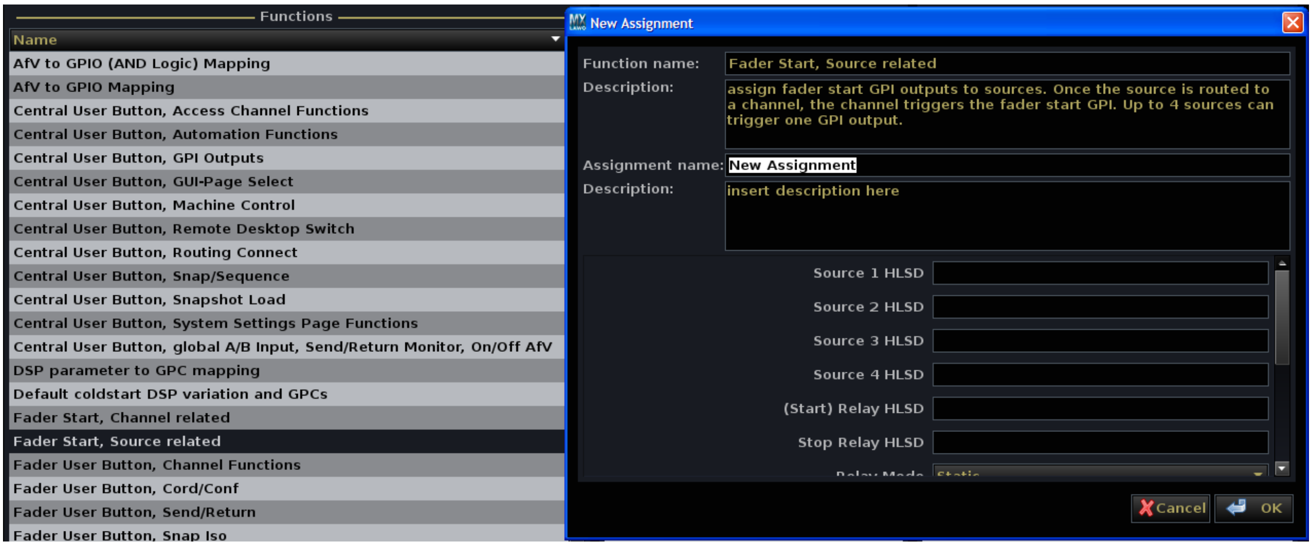
- **Channel Type** – input, monitor, group, sum, aux, surround VCA or GPC (General Purpose Channel).
- **Channel Index** – the channel number.

Scroll down the **New Assignment** window to define options for the relay output:

- **Relay HLSD** – this is the Lawo system address of the relay which will be triggered.
- **Relay mode** – static, pulse, etc.
- **Relay Time** – can be set for a pulsed relay.
- **Consider Cut** – has two states:
  - **Yes** – if the channel mute is active, then the fader start will not trigger when the fader is opened.
  - **No** – the fader start always triggers when the fader opens regardless of the channel mute status.

Note that the fader start is assigned to the DSP channel and not a physical fader. Therefore, if INP 1 is reassigned to a different fader strip, the fader start follows.

Fader Start, Source related



Maps source signals to an external relay (GPI output) in order to create a fader start. Once the source is routed to a DSP channel, and the channel assigned to a physical fader, the fader triggers the start.

Up to 4 sources can be assigned to each relay:

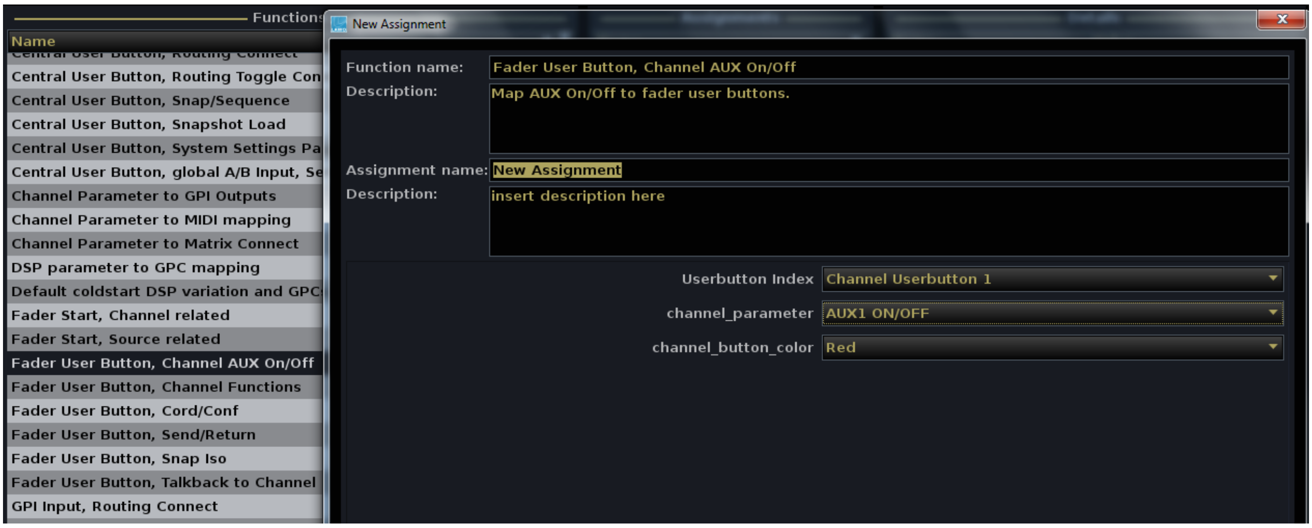
- **Source HLSD** – the Lawo system address of each source.
- **(Start) Relay HLSD** – the Lawo system address of the start relay which will be triggered.
- **Stop Relay HLSD** – the Lawo system address of the stop relay which will be triggered.
- **Relay mode** – static, pulse, etc.
- **Relay Time** – can be set for a pulsed relay.
- **Consider Cut** – has two states:
    - o **Yes** – if the channel mute is active, then the fader start will not trigger when the fader is opened.
    - o **No** – the fader start always triggers when the fader opens regardless of the channel mute status.

The last three options assign a user button which can be used to trigger the relays:

- **Enable Userbutton Type** – the panel type.
- **Enable Userbutton Index** – the panel number.
- **Enable Userbutton Scribble** – the text displayed if the user button has an accompanying scribble strip display.

Note that the fader start is assigned to the source. Therefore, if the source is reassigned to a different DSP channel, the fader start follows.

Fader User Button, Channel Aux On/Off

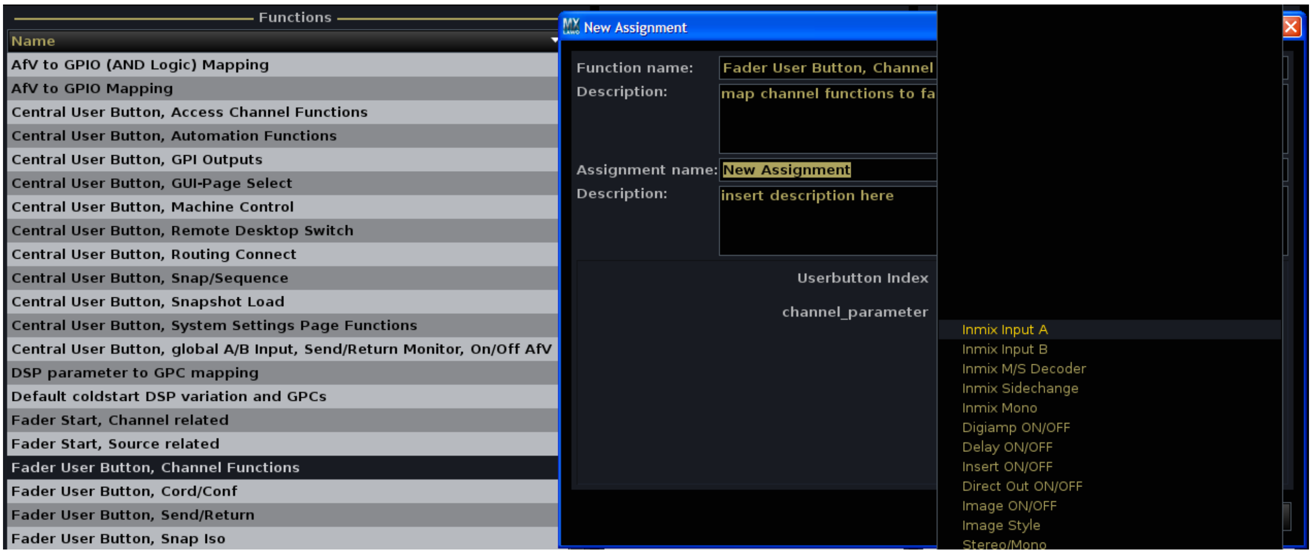


Maps a fader strip user button to the channel's aux send on/off.

For each function, define the:

- **Userbutton Index** – the user button number (1 to 12).
- **Channel parameter** – e.g. Aux 1 ON/OFF.
- **Channel Button Color** – select the colour for the user button on state: red, yellow or green.

Fader User Button, Channel Functions



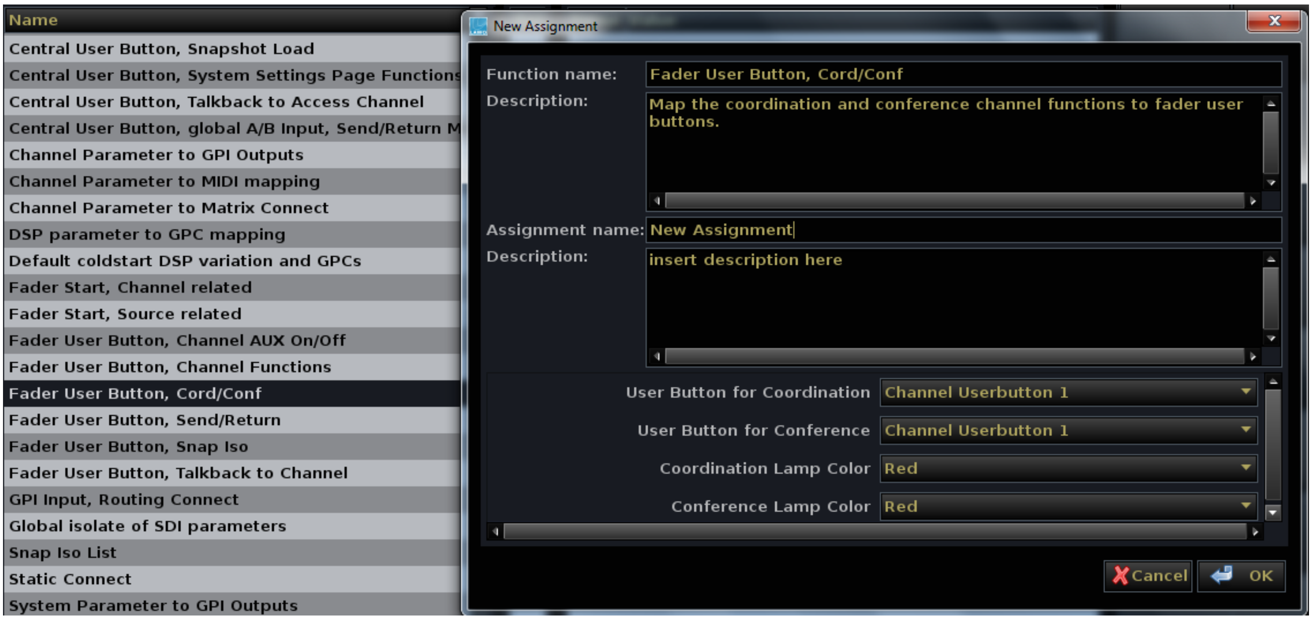
Maps a fader strip user button to a channel function such as:

- A/B input switching
- MS Decode
- Delay on/off
- EQ on/off
- Tone to Channel (from V4.24 software onwards).
- Fader R/W (from V4.24 software onwards; particularly useful for the mc²56 which has no dedicated **R/W** button.)
- Trim/Absolute
- etc.

For each function, define the:

- **Userbutton Index** – the user button number (1 to 12).
- **Channel parameter** – e.g. Delay ON/OFF.
- **Channel Button Color** – select the colour for the user button on state: red, yellow or green.

Fader User Button, Cord/Conf

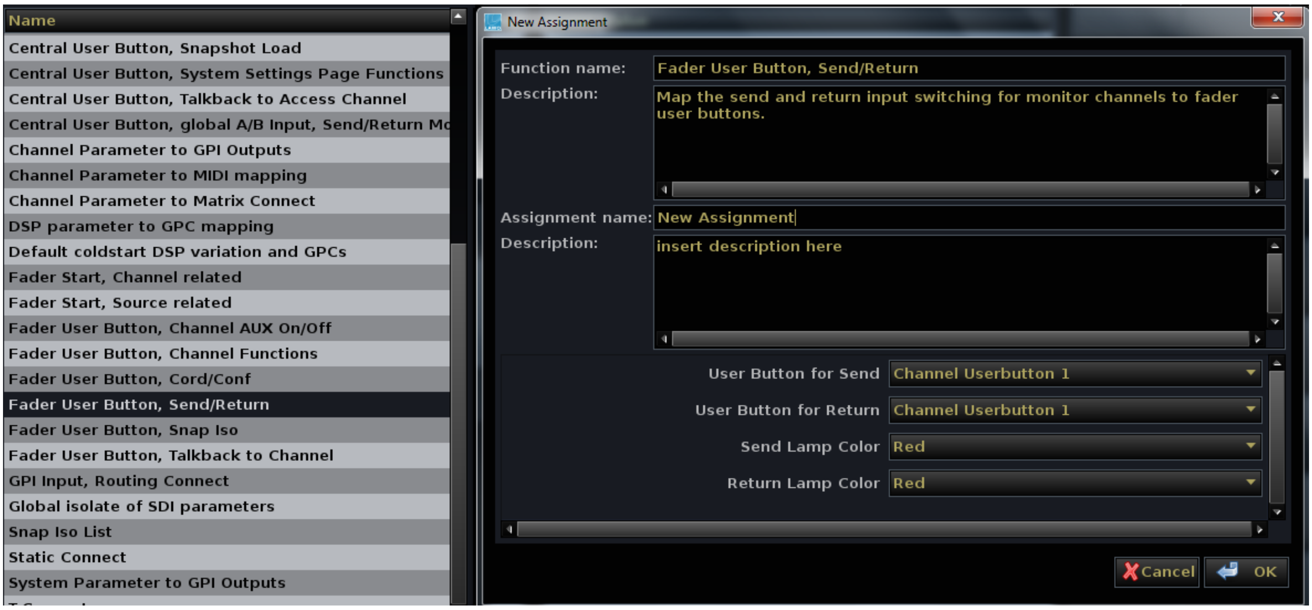


Maps fader strip user buttons to the **CONF** and **CORD** mix minus controls.

For each function, define the:

- **Userbutton Number** – 1 to 12.
- **Lamp Color** – select the colour for the user button on state: red, yellow or green.

Fader User Button, Send/Return

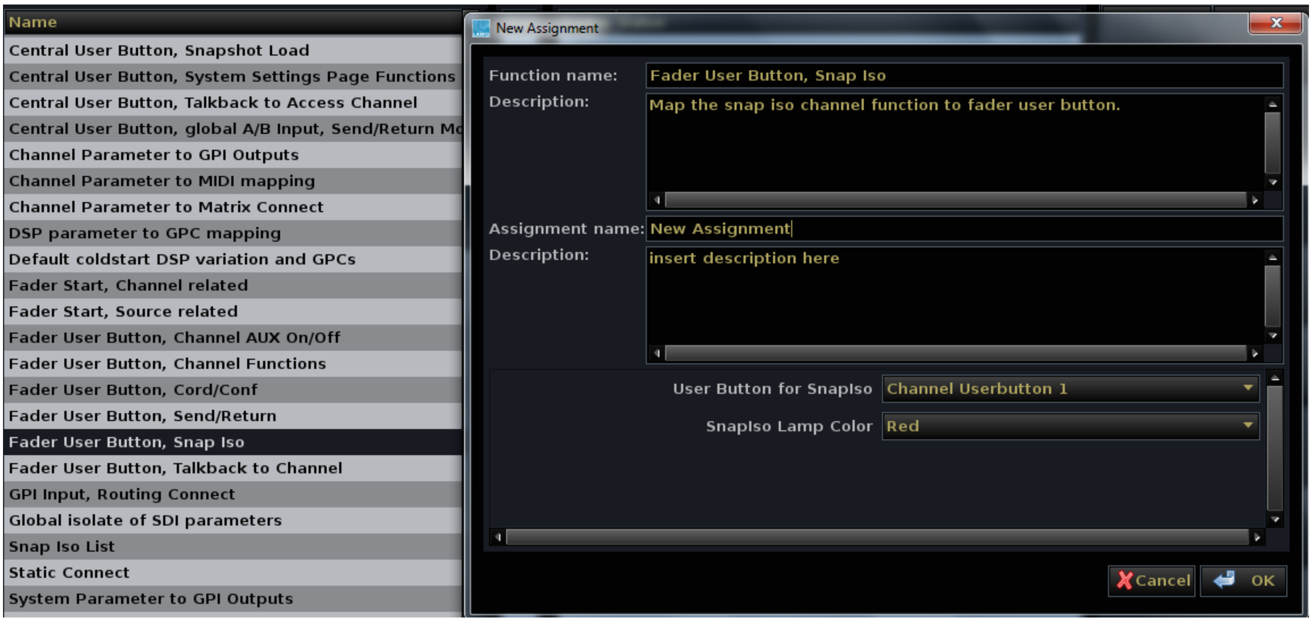


Maps fader strip user buttons to the multitrack **SEND** and **RETURN** switching for monitor channels.

For each function, define the:

- **Userbutton Number** – 1 to 12.
- **Lamp Color** – select the colour for the user button on state: red, yellow or green.

Fader User Button, Snap Iso

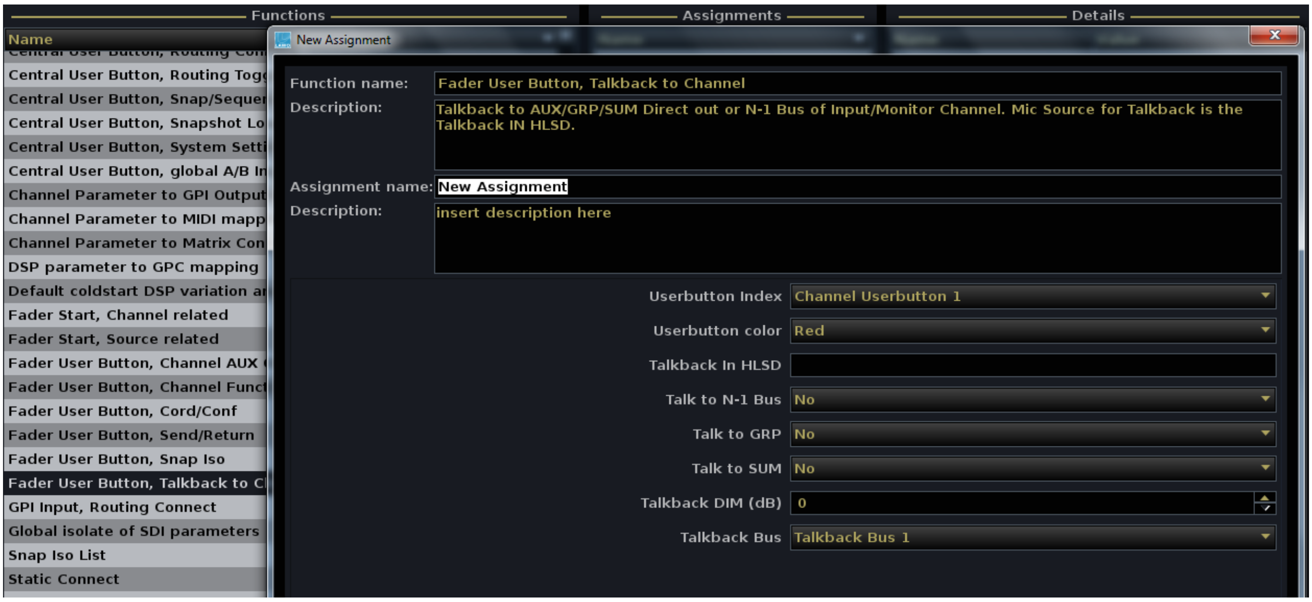


Maps a fader strip user button to the snapshot isolate function (**SNAP ISO**):

Define the:

- **Userbutton Number** – 1 to 12.
- **Lamp Color** – select the colour for the user button on state: red, yellow or green.

Fader User Button, Talkback to Channel



Maps a fader strip user button to talkback switching.

When active, talkback is routed from the talkback source onto one of 8 talkback busses. (These busses appear in the Signal List display under the **Input/Mon A + B** -> **Command Bus** Source Directory. Note that when using a 96 channel broadcast channel DSP configuraton, only one talkback bus is available).

The talkback bus may then feed the channel's N-1 bus (on input or monitor channels), the Group direct output (on group channels) or the Sum direct output (on sum channels).

You can also decide whether to dim the console monitoring when the talkback user button is active.

1. Define the **Userbutton Index** and **Userbutton Color** in the usual manner.

2. Enter the HLSD address for the talkback source into the **Talkback In HLSD** field. The easiest way is to copy and paste the signal HLSD from the **mx Routing** display, see Entering a HLSD Address.

3. Define whether the user button will activate talkback on input/monitor channels, group channels and/or sums:

   - **Talk to N-1** = **Yes** – on an input or monitor channel, the user button routes talkback to the mix minus bus assigned to the channel's source. This can be any aux or track bus as defined by the mix minus configuration.

   - **Talk to GRP** = **Yes** – on a group channel, the user button routes the talkback bus to the group's direct out.

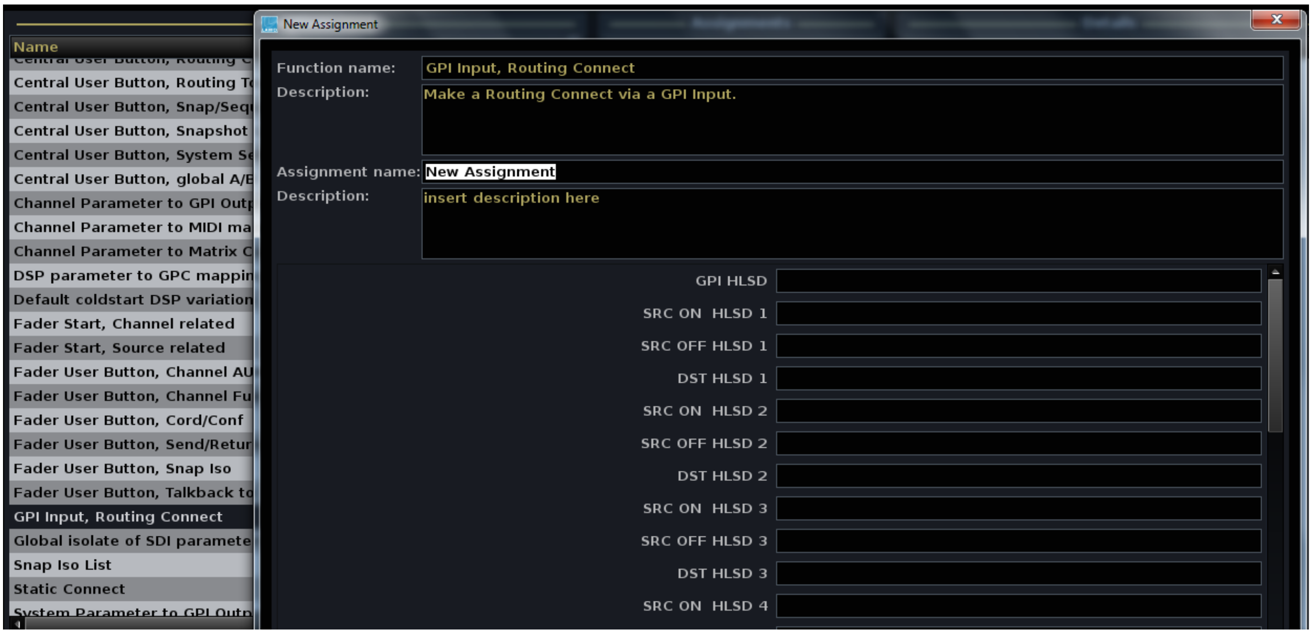
   - **Talk to SUM** = **Yes** – on a sum channel, the user button routes the talkback bus to the sum's direct out.

   For example, if you set **Talk to N-1** = **Yes**, and **Talk to GRP/Talk to SUM** = **No**, then the user button will *only* activate talkback on fader strips assigned to input or monitor channels, and route talkback to the channel's N-1 bus.

4. Use the **Talkback DIM (dB)** field to enter the amount of dim applied to the console's monitoring when the user button is active.

5. Use **Talkback Bus** to select one of the 8 available talkback busses. This option is useful when programming multiple **TALK** buttons, as you can have each user button working with a different talkback bus.

GPI Input, Routing Connect



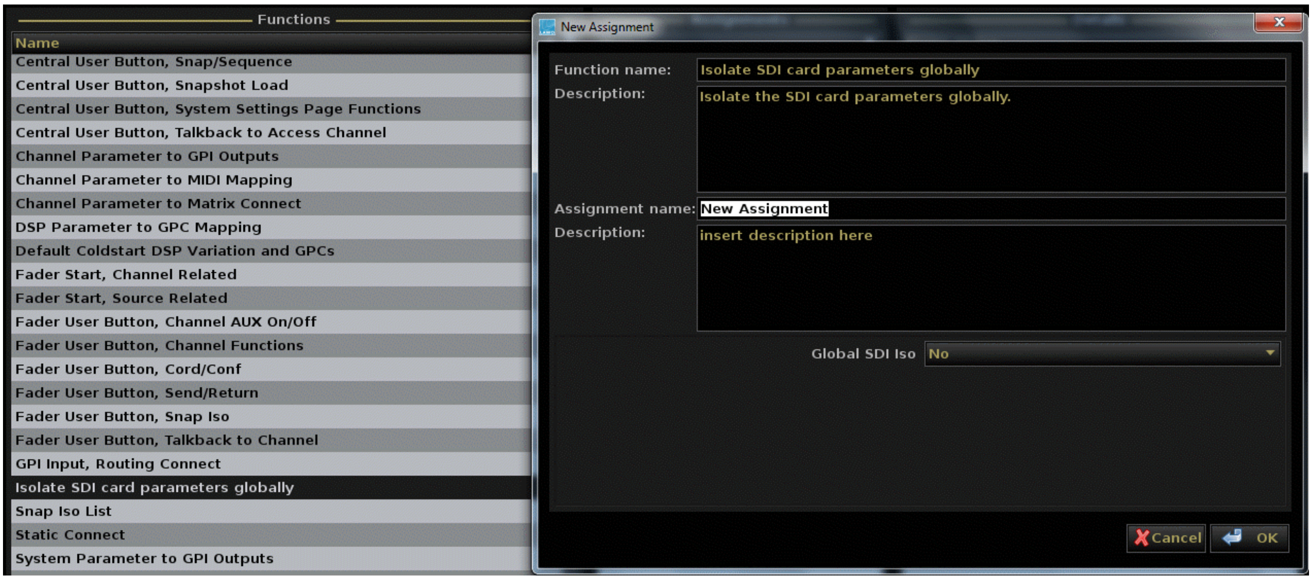
This function allows you to perform signal routing from a GPI Input. Up to 16 connects/disconnects can be assigned to one input.

You may create multiple instances of this template.

Copy the source and destination **HLSD** from the **mx Routing** display, see Entering a HLSD Address.

To create a disconnect, type **DISCONNECT** into the field for the Source HLSD.

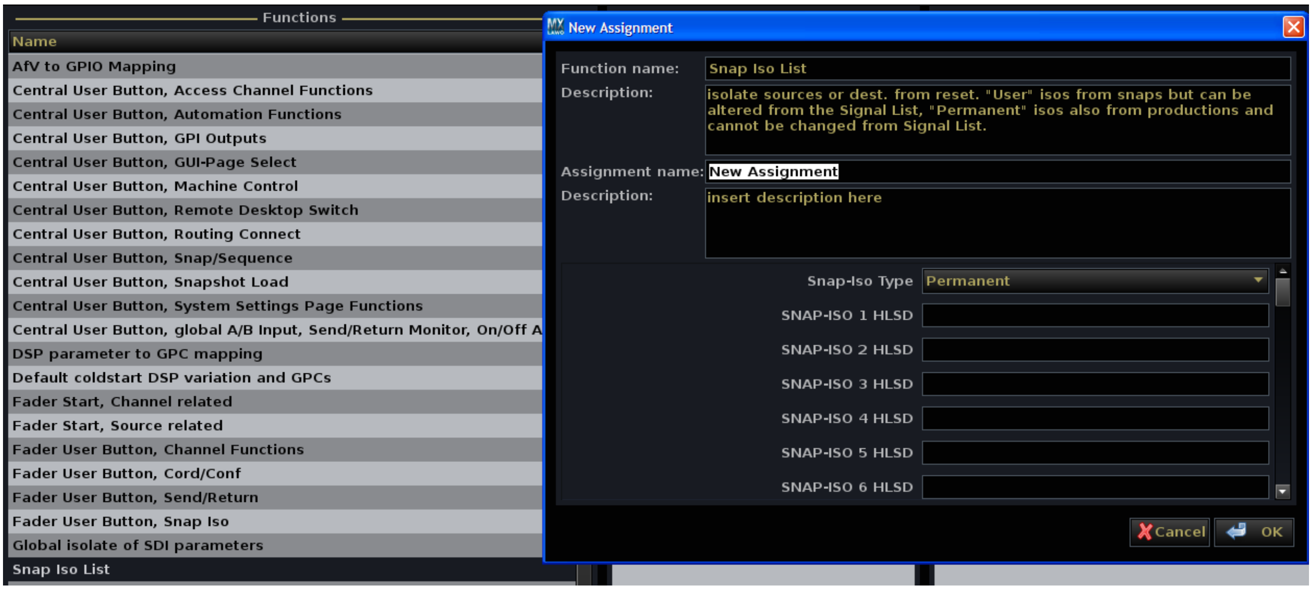
Isolate SDI card parameters globally



SDI parameters are never stored by snapshots. From Version 4.8.0.2 onwards, they are stored and recalled by productions. This function can be used to isolate all SDI parameters so that settings are not affected by a production load.

This template should only be created once. If created several times, the last initialised one wins.

Snap Iso List



This function allows you to isolate sources or destinations to prevent them being reset by snapshots, and/or from productions or the Signal List display.

Up to 48 signals may be defined within each **Snap Iso List** assignment; you can create multiple assignments to isolate lots of signals.

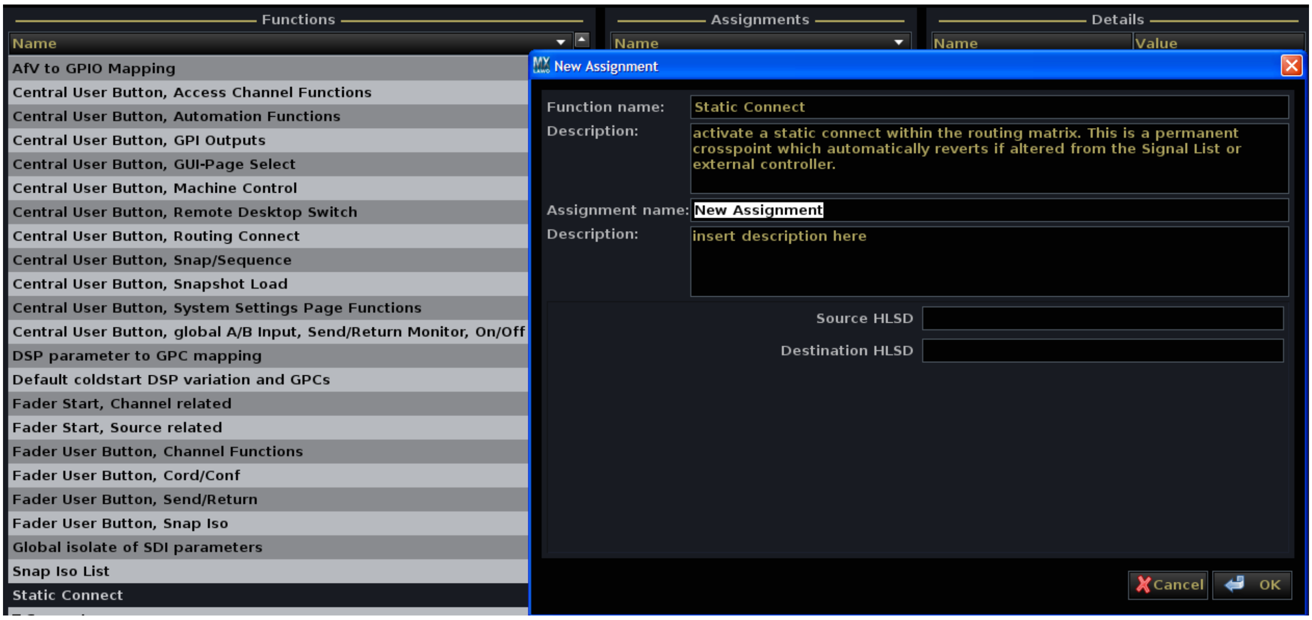
Within each assignment, the **Snap-iso Type** can be:

- **Permanent** – signals are not reset by snapshots or productions, and cannot be adjusted from the **Signal List** display.
- **User** – signals are not reset by snapshots, but will be reset by productions and can be adjusted manually from the **Signal List** display.

For each **Snap Iso List** assignment, enter the **HLSD** (Lawo system address) of the signals you wish to isolate.

> Copy the source and destination **HLSD** from the **mx Routing** display, see Entering a HLSD Address.

Static Connect



This function allows you to define a Static Connect by entering the **HLSD** (Lawo system address) for a Source and a Destination.

A Static Connect is a routing crosspoint which will *always* be active. If it is disconnected by any means, for example by the console operator or by an external controller, the crosspoint is automatically remade. You might use this function to prevent vital crosspoints from being accidentally reset.

Note that having defined a Static Connect, the only way to change or disconnect the crosspoint is to delete the Static Connect from the **Custom Functions** display.

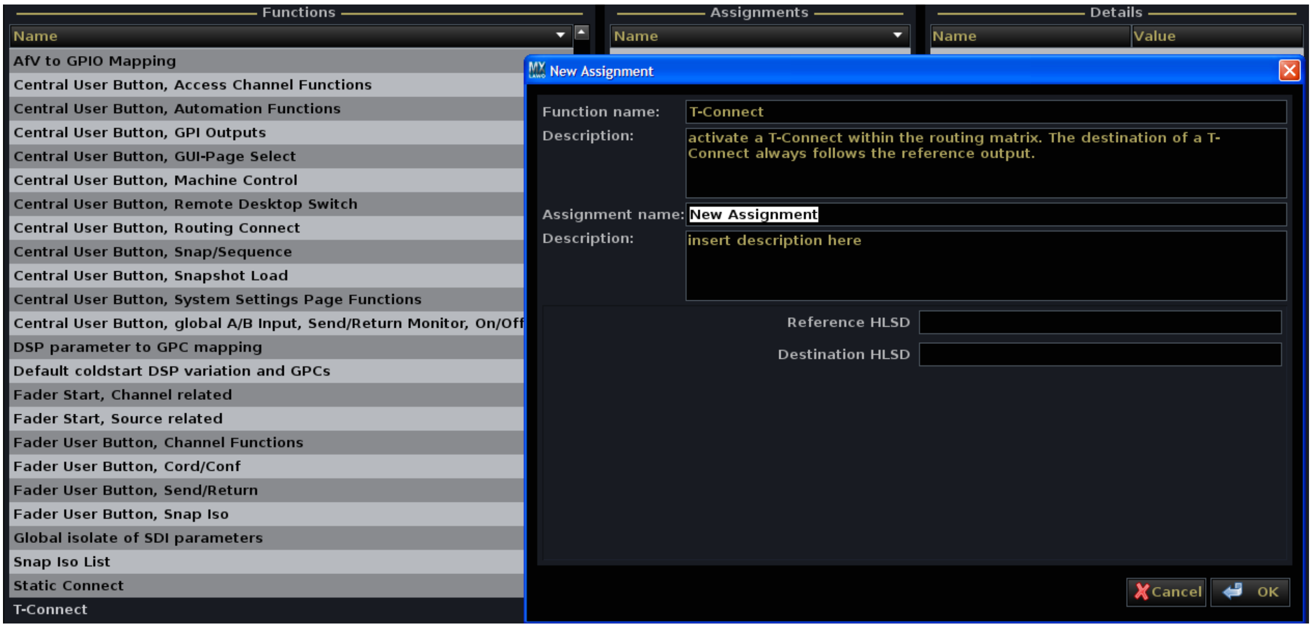
Copy the source and destination **HLSD** from the **mx Routing** display, see Entering a HLSD Address.

System Parameter to GPI Outputs

Maps system parameters to GPI Outputs. Select the parameter – for example, **PFL active** – and then enter the **HLSD**, **Mode** and **Time** for the GPO.

Copy the source and destination **HLSD** from the **mx Routing** display, see Entering a HLSD Address.

T-Connect



This function allows you to define a T-Connect by entering the **HLSD** (Lawo system address) for a Reference output and a Destination output.

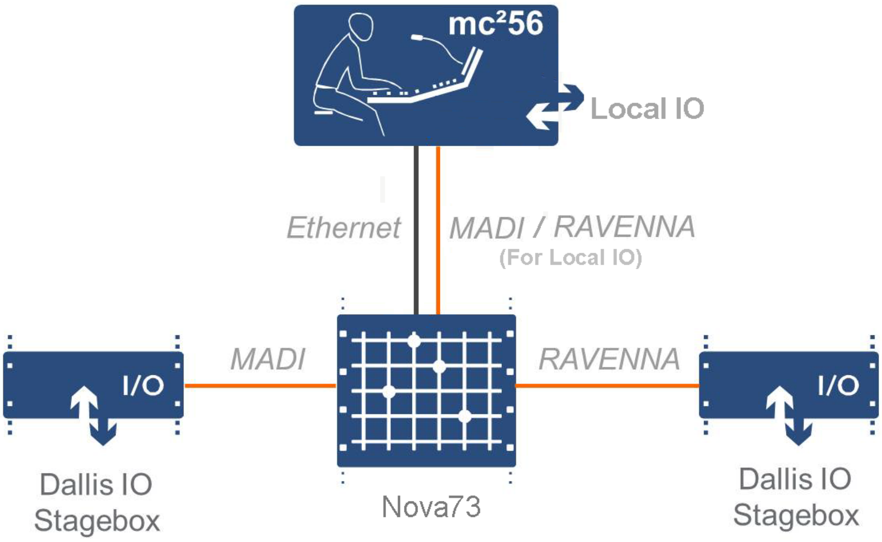
The Destination output always follows the Reference output. So, for example, if the source to the reference output is Sum 3, the destination output source is also Sum 3. You might use this function if you have several transmission feeds all requiring identical routing changes.

Copy the source and destination **HLSD** from the **mx Routing** display, see Entering a HLSD Address.

System Components

The **mc²56** consists of three principal components:

- **Console control surface** - with integrated power supplies and local I/O connections.
- **Nova73** – with Router Modules, DSP boards and AES, MADI, ATM or RAVENNA I/O. Available in two sizes: **Nova73 HD** (10RU) or **Nova73 Compact** (7RU).
- **DALLIS I/O** – offering further I/O breakout options; connected to the Nova73 via MADI, ATM or RAVENNA I/O.

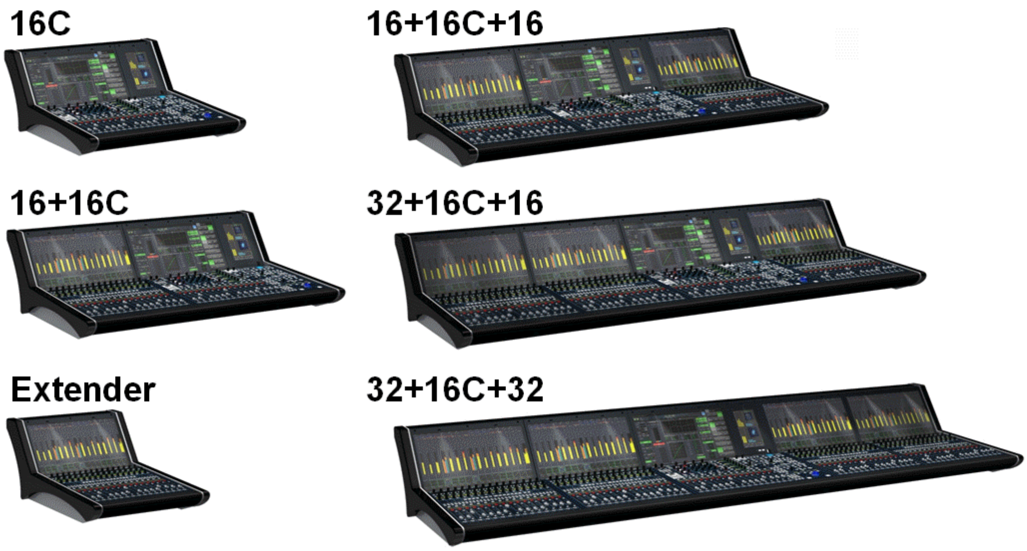


The exact hardware specification defines how many analogue and digital connections are available for external equipment, and how much DSP processing is available for input channels, monitor return channels, groups, sums and auxiliary sends. For a summary of the system capabilities, please see Technical Data.

> In the classic mc²56 there are no integrated local I/O cards, and so the only connection between the control surface and Nova73 is Ethernet.

Console Control Surface

The **mc²56** control surface is constructed in 16-fader sections, with frame sizes scaling from 16 faders up to 80 faders. You may add 16-fader extenders to expand the number of fader strips.



A range of console options offer wide (studio) or narrow (OB) side panels, table-top or stand mounting, overbridge metering, etc.

Control surface power is provided by internal power supplies, with n+1 redundancy and two mains connections for phase redundancy. PSU status may be monitored from the console GUI.

All application software and user data is handled by the control system, located on the Router Module (MKII) within the Nova73. The surface connects to the Nova73 via TCP/IP Ethernet; if a redundant Router Module is fitted, then main and backup connections can be installed.

The control surface also houses a local I/O board, for monitoring, metering, talkback and headphones. This is available in two versions, connecting to the Nova73 via either MADI or RAVENNA.

> As the control system is integrated within the Nova73, the control surface may be powered off without loss of user data or audio!

Nova73

The **Nova73** forms the "heart" of the system, and is available in two sizes - **Nova73 HD** (10RU) or **Nova73 Compact** (7RU):

Nova73 HD (10RU)

Nova73 Compact (7RU)

In each case, the front of the frame houses the:

- **Router Modules MKII** - two central slots are available for a main and redundant Router Module. The Router Module MKII (980/33) contains the summing matrix *AND* control system. The summing matrix offers a $8k^2$ capacity* router at 48kHz (or $4k^2$ capacity at 96kHz). The control system runs on an embedded Linux operating system, and stores both the application software and user data. Connections are made via the two TCP/IP Ethernet ports:
  - ETHERNET A – connects to the control surface.
  - ETHERNET B – connects to the Lawo system network (to other Lawo devices; third-party controllers; computers running configuration, maintenance or remote control software).
- **DSP and I/O Modules** - 16* slots are available for plug-in DSP or I/O modules. Up to 8* DSP boards may be fitted supporting a range of DSP configurations; I/O options include AES/EBU, MADI, ATM and RAVENNA. All modules are hot-pluggable enabling them to be be replaced without affecting other aspects of the system. Further breakout formats are realised by connecting to DALLIS I/O.
- **Power Supply Units** - two slots are available for main and redundant power supplies.

The rear of the frame houses the:

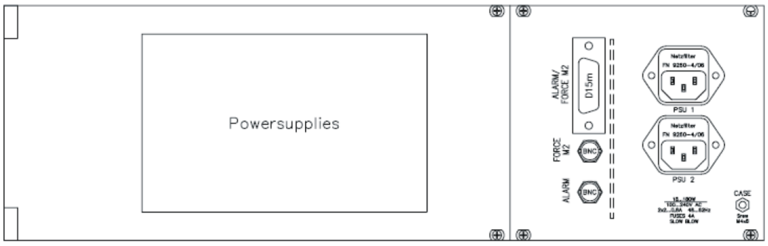
- Sync ports - accepting Wordclock, AES/EBU (AES3-id) or Video Black Burst (PAL or NTSC).
- Alarm and control contacts - including a global alarm; prepare cold start; force redundant Router Module takeover.
- AES connector panels - for front-mounted AES3 I/O modules.
- 5* Cooling Fans - hot-pluggable and easily accessible.

* The figures above are for the **Nova73 HD**. For more details on the Compact core, see the "mc²56 Technical Manual".

DALLIS I/O

Front View

Rear View



The I/O capabilities of the system are expanded by adding **DALLIS** unit(s). Each may be either 3RU (shown above) or 6RU in height, and may be remote from the rest of the system.

The front of the frame houses the:

- **DALLIS Master Boards** - two central slots are provided for a main and redundant master board. A choice of board types provide connection to/from the Nova73 via MADI, ATM or RAVENNA.

> The type of DALLIS master board, and hence the connection, determines the maximum number of audio channels to/from the Nova73: up to 60 (MADI), 80 (ATM) or 128 (RAVENNA).

- **DALLIS I/O cards** - 18 slots are available for a range of I/O breakout options (Mic/Line, Line, AES, SDI, GPIO, etc.).

All cards are hot plug-able, with the exception of Phantom Power.

The rear of the frame provides access to:

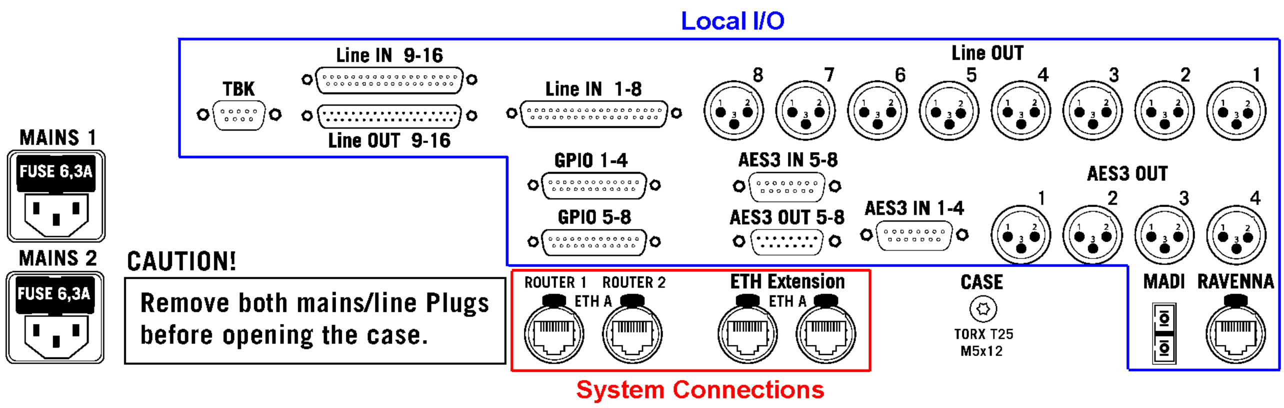
- Main and redundant power supplies
- Alarm and control contacts - including a local DALLIS alarm.

Local I/O

The MKII mc²56 control surface includes an integrated local I/O board. This provides dedicated connections for local devices such as monitoring, metering, talkback and headphones.

(Note that on the classic mc²56, similar functionality may be provided via an external DALLIS.)

All local I/O connections are accessed from the control surface rear panel:



The local I/O provides:

- **16 Line In** - wired to 2 x DSub (female).

Note that **Line In 16** may be fed from the integrated talkback mic preamp, according to the jumper switch positions set for the Local I/O.

- **16 Line Out**:
    - o Line Out 1-8 - wired to 8 x XLR (male). By default, these outputs are routed from the CRM 1 monitor output.
    - o Line Out 9-16 - wired to 1 x DSub (male).
- **8 AES3 In** - wired to 2 x DSub (female).
- **8 AES3 Out:**
    - o AES3 Out 1-4 - wired to 4 x XLR (male).
    - o AES3 Out 5-8 - wired to 1 x DSub (male).

Note that **AES IN 5-8** and **AES OUT 5-8** connect to the RTW meter, if either of the TM 7 or TM 9 Overbridge options are fitted.

- **8 GPIO** - wired to 2 x DSub.
- **2 Stereo Headphones** - wired to the headphone 1 & 2 connectors on the console's front buffer.
- **1 MADI or 1 RAVENNA** - the local I/O board is available in two versions, connecting to the Nova73 via either MADI or RAVENNA. You will need to reserve one MADI, or one RAVENNA, port within the Nova73 for this connection.

Please see Local I/O Wiring for more details on wiring, pin-outs and jumper switch options.

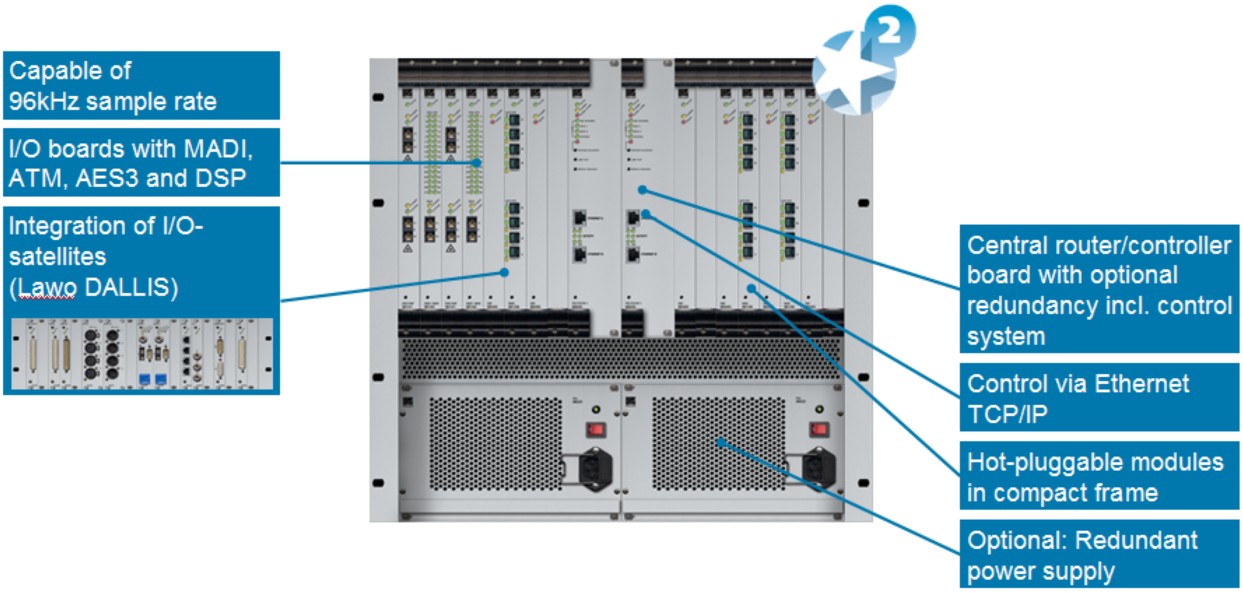
Redundancy

One of the strengths of the **mc²56** is its ability to withstand component failures, and every component is designed with fault tolerance in mind:

- Star[2] Technology
- Link & Port Redundancy
- Nova73 & DALLIS Power
- Redundant DSP
- Control System
- Redundant Router Module and Control System
- Control Surface Power
- Control Surface Internal Wiring

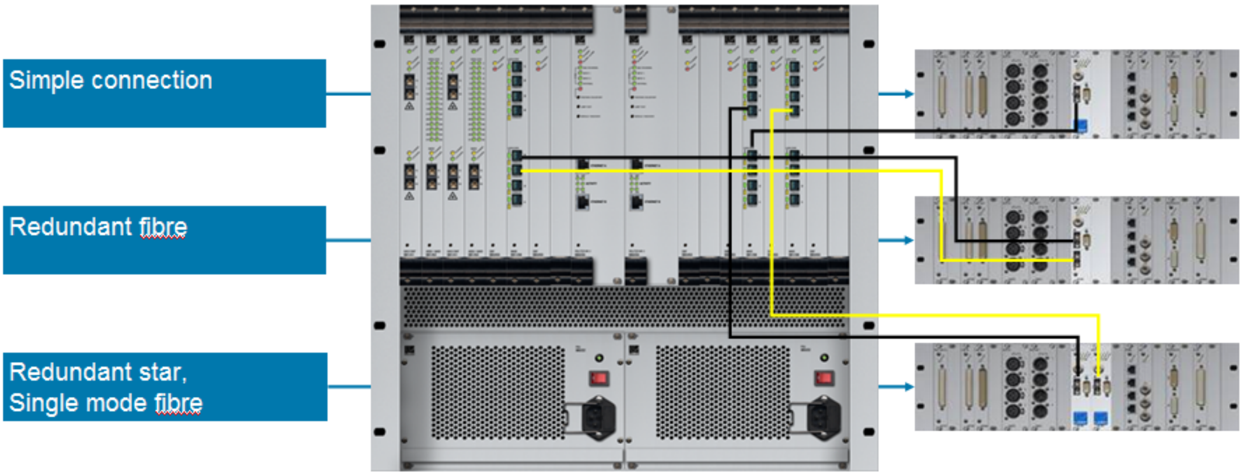
Star2 Technology

All components within the system utilise Lawo's Star² technology:



- **Point-to-point connections** – with point-to-point connections, a fault only affects that part of the system, unlike a TDM bus architecture where a fault may disrupt everything connected to the bus!
- **Dual star topology** – with redundant Router Modules fitted to the Nova73, and redundant Master Boards in every DALLIS, then components connect in a dual 'star' mode. This protects signal paths from any single point-of-failure. See Link & Port Redundancy.
- **Hot-swappable Modules/Cards** – every plug-in module or card can be hot-swapped without affecting the rest of the system enabling online maintenance of the system.
- **Redundant Power Supply Units** – both Nova73 and DALLIS units can be fitted with dual redundant power supplies, which can be isolated and exchanged from the front or rear. See Nova73 & DALLIS power.
- **Passive backplanes** – the frame backplanes are entirely passive. With no active components, this increases reliability.

Link & Port Redundancy

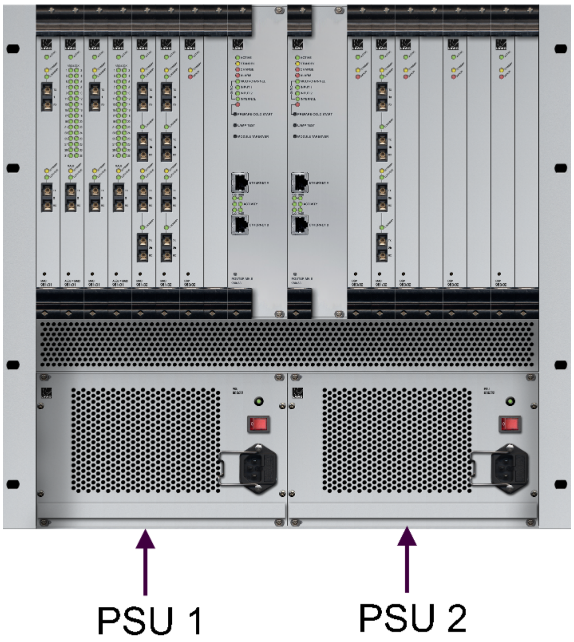


For crucial interconnections between DALLIS and Nova73 units, you can specify either link, or link and port, redundancy:

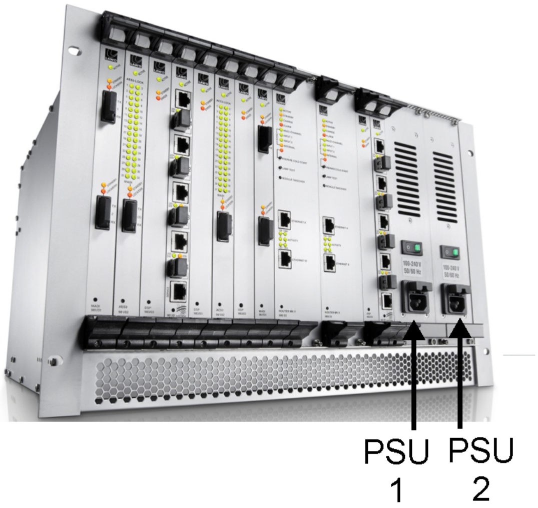
- **Link Redundancy** – two physical connections (MADI, ATM or RAVENNA) are made from the DALLIS master board to the Nova73. If the active link fails, then the redundant link ensures an automatic recovery.
- **Link & Port Redundancy** - two master boards are fitted to each DALLIS, and connect to different Nova73 ports (preferably on a different module). Port redundancy provides automatic recovery from a:
  - o Failure of the active physical link (MADI, ATM or RAVENNA).
  - o Malfunction of the active DALLIS master board.
  - o Malfunction of the Nova73 module.

Nova73 & DALLIS Power

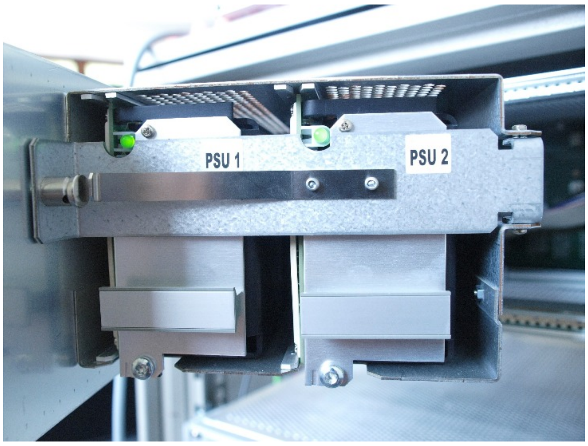
Nova73 HD front view



Nova73 Compact front view



DALLIS internal view of PSUs



The Nova73 HD, Nova73 Compact and DALLIS provide two slots for dual redundant power supplies. Their status may be monitored from the console GUI using the Signal Settings display.

Redundant DSP

Within the Nova73 a DSP board may be reserved to provide redundant processing (indicated by the **STANDBY** LED).

In the unlikely event of a failure, the system automatically switches all DSP resources and settings from the faulty board to the spare; the faulty board may then be safely removed and replaced.

This option is enabled from the Central GUI using the DSP Configurations display, and is saved within the production.

Control System

The control system resides on the Router Module MKII (980/33) within the Nova73.

It runs on an embedded Linux operating system for speed and increased reliability, and stores both the application software and user data.

The Router Module MKII (980/33) contains a backup power unit which provides up to 3 seconds of backup power to deal with short interruptions to mains (AC) power.

Warm Start & Cold Start

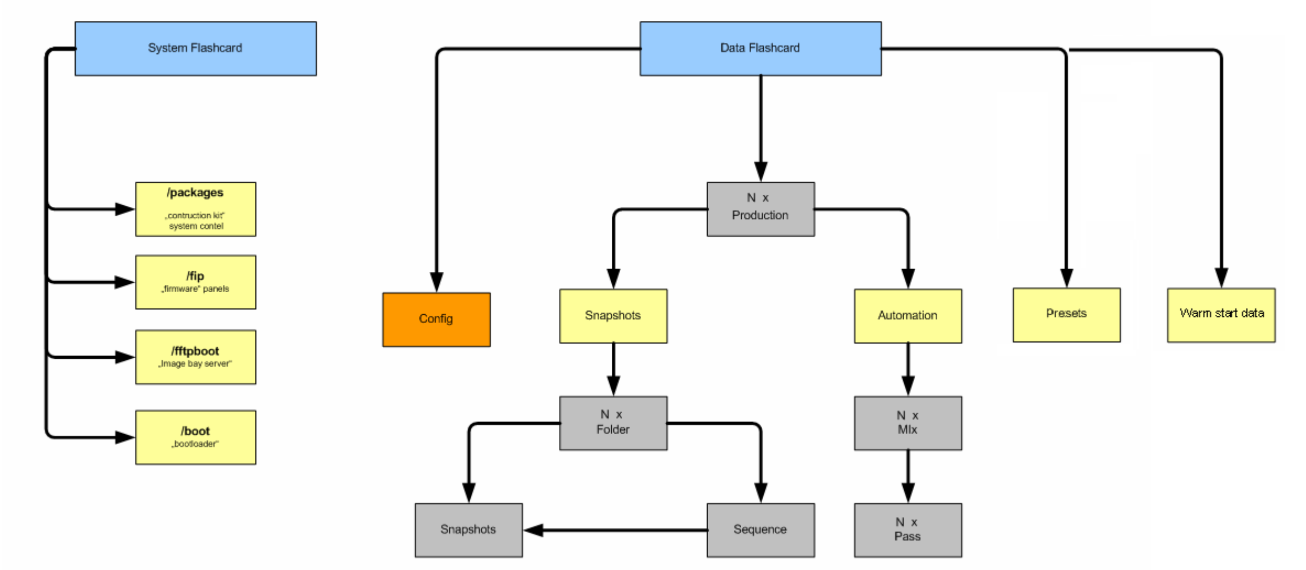
Following switch-off, power is provided to the control system for a further 18 seconds. During this time, all current settings are saved to flash memory; this is known as the system's warm start data.

By default, the warm start data is loaded at the end of boot-up. This means that the console comes back exactly as it was when you last shut down, ensuring fast recovery of all previous settings following a loss of power.

Alternatively, you can perform a cold start if you suspect a problem with the warm start data.

Data Recovery

Two flash cards are used to store the application software (**System Flashcard**) and user data (**Data Flashcard**) separately. You may create a backup copy of the flashcards so that they may be replaced if necessary ,see the "mc²56 Technical Manual".

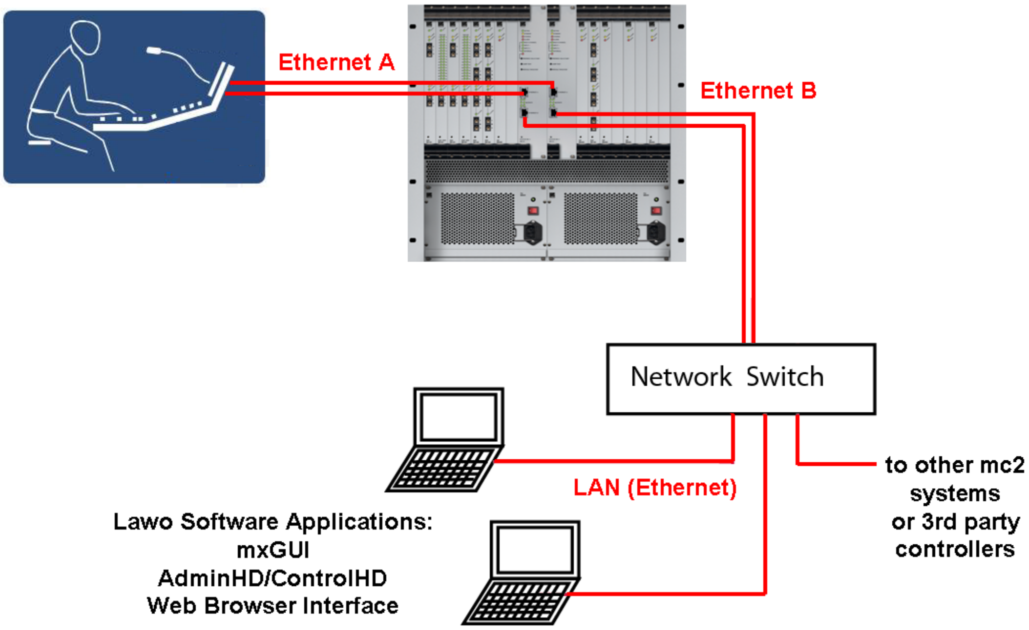


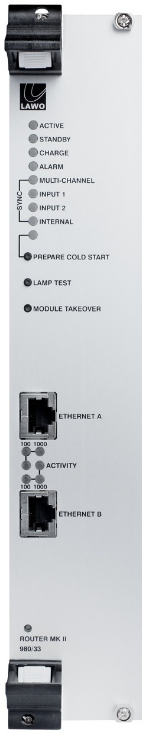
During operation, any errors generated by the control system are stored in the **message** logfile. This can be copied to USB via the File display, or monitored remotely via the Web Browser Interface.

Redundant Router Module and Control System

By fitting a second Router Module to the Nova73, the system can provide redundancy for the routing matrix and control system.

In order to provide redundancy, your Nova73 must be fitted with two Router Modules MKII; Ethernet A and Ethernet B connections from both the main and redundant modules are required:





Automatic Takeover

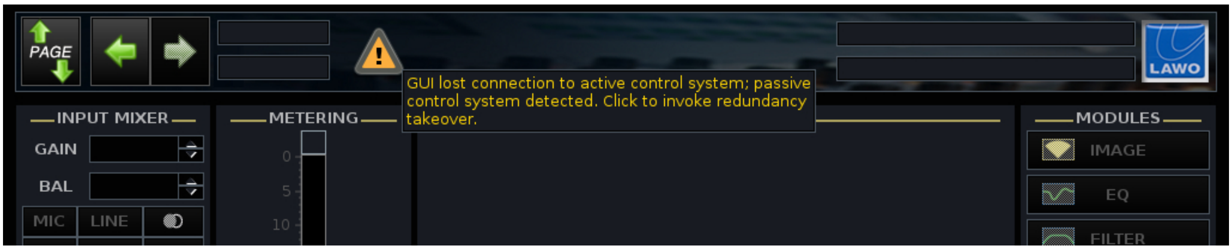
If the main Router Module fails, then the redundant module automatically takes over. This ensures a seamless recovery without any interruption to operation.

> Note that a brief interruption to audio will occur while routes are reconfigured.

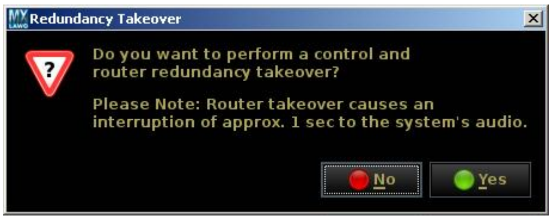
The redundant control system is automatically activated if, internally, a loss of connection is noticed by the redundant system. This could be due to a software failure, hardware error or reboot of the main control system.

If the Ethernet connection between the control surface and Router Module fails, then an automatic takeover does *not* occur, as the failure may be deliberate (for example, if you disconnect the cable).

Instead the operator is presented with an error message:



1. Click on the message and a confirmation pop-up appears:



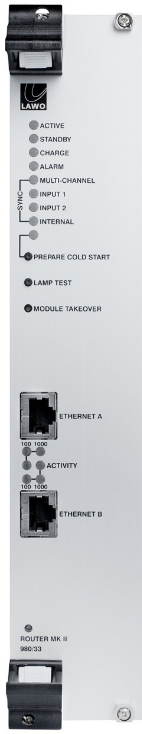
2. Select **Yes** to switch to the redundant control system or **No** to cancel.

Selecting **Yes** causes an interruption to the audio.

If you select **No**, then you *MUST* fix the problem with the connection before you can regain control of the audio.

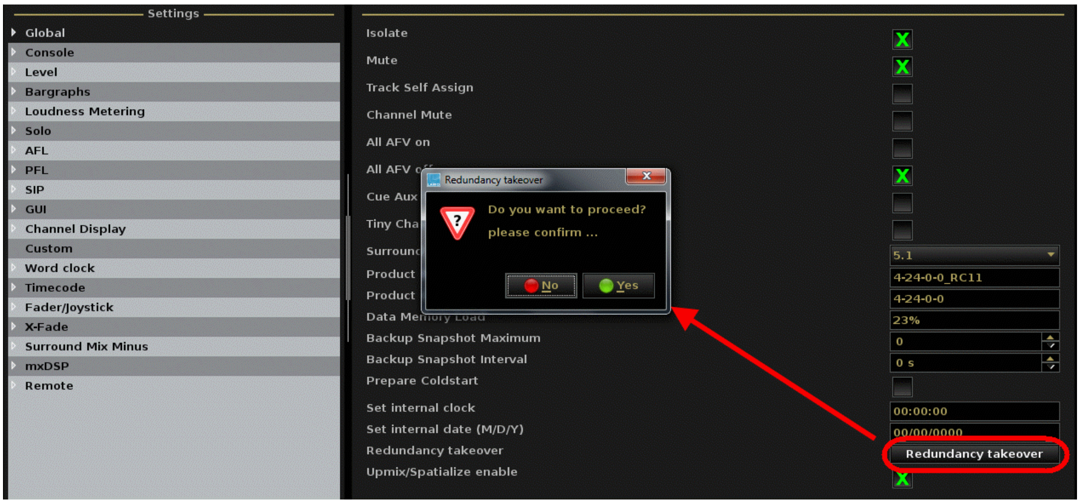
Manual Takeover

You can force a manual takeover at any time, using the Redundancy takeover option in the **System Settings** display:



1. Select the **Global** topic followed by the **Redundancy takeover** option.

   A confirmation dialogue box appears:



2. Select **Yes** to confirm or **No** to cancel the operation.

   *Selecting **Yes** switches to the redundant control system.*

Alternatively, press the **Module Takeover** button on the front of the redundant Router Module.

A manual takeover may also be forced using the **ROUTER TAKEOVER** contact, connected to **GPI 1** on the Nova73 rear panel.

Control Surface Power

Control surface power is provided by internal power supplies. Depending on the frame size, either one or two PSU blocks are fitted to each frame. Each block is equipped with two power supplies running in parallel. Both share the current load; if one fails, then the second is powerful enough to handle the required load alone.

Power Supply Desk Alarm

From Version 4.8 software onwards, the Central GUI offers status monitoring for all PSU blocks fitted to the control surface.

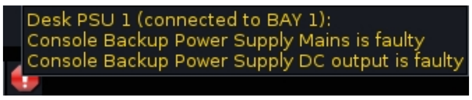
The status of each PSU block is represented by a symbol which appears at the bottom right of every console display:

The number of symbols relates to the number of PSU blocks within the control surface and whether any extender bays with their own PSU are fitted. PSU blocks are represented from left to right, and each block consists of two supplies running in parallel.

The symbols indicate:

- **Green Circle** – the PSU block is working fine.
- **Yellow triangle with an exclamation mark** – the PSU block is working fine, but there was a fault in the past which has now been cleared. Click on the icon to reset it.
- **Red circle with an exclamation mark** – there is a fault.

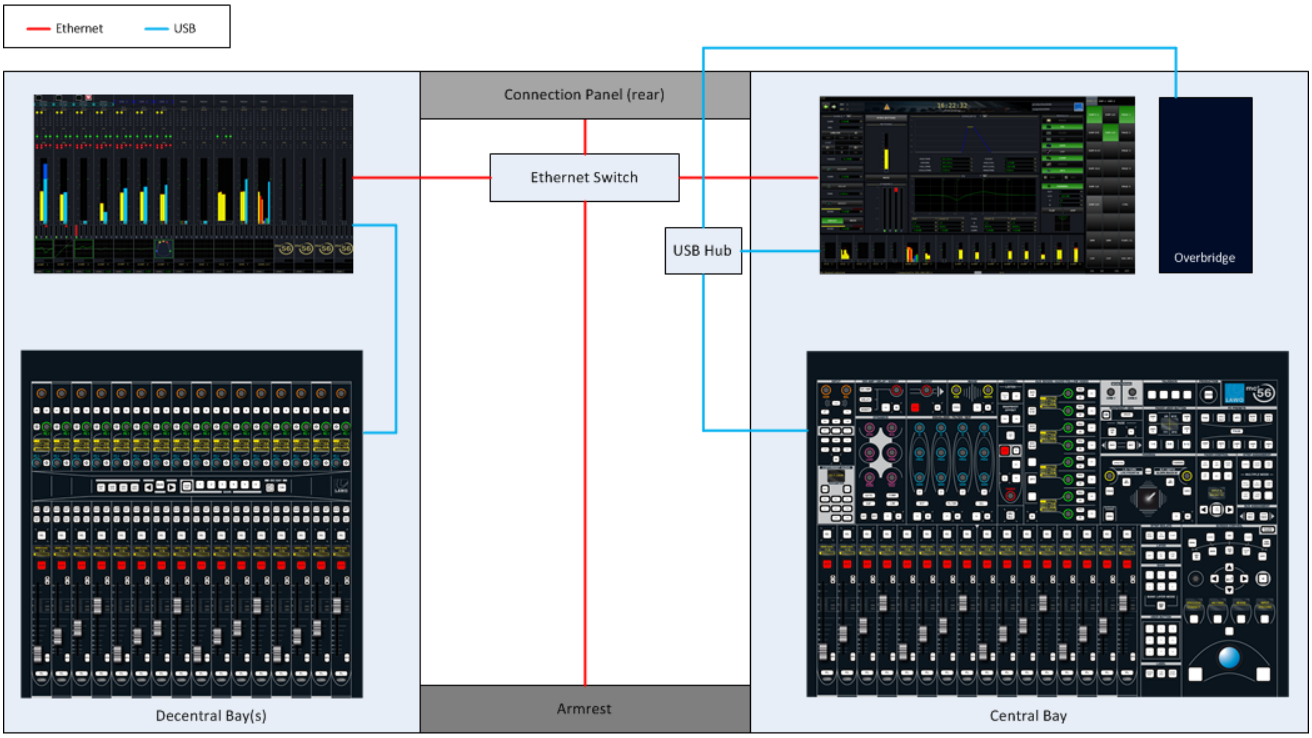
Hover over the symbol to reveal more information:



Our example shows the status for Bay 1 - the first bay on the left of the console.

> Note that within the text on the GUI, bays are counted from BAY 1 upwards. However, internally bays are addressed from 0. This means that BAY 1 on the GUI relates to an internal Bay Server address of 0, BAY 2 to an internal address of 1, and so on.

Control Surface Internal Wiring



Within each channel and central bay, individual panels and displays connect to an Ethernet Bay Server. (Control surface panels via USB; displays via LVDS for high resolution graphics and USB for touch control).

Each Bay Server then connects to an Ethernet switch, mounted inside the control surface frame. The network connection from the switch appears on the rear panel as ETHERNET A.

By fitting a redundant Router Module to the Nova73, a second Ethernet A connection can be installed for automatic redundancy.

Internally, point-to-point connections provide fault tolerance, and allow any bay or panel to be isolated from the rest of the console.

Control surface panels and displays are hot-pluggable making them easy to service.

Sample Rate & System Clock

Internal Sample Rate

The system may operate at a choice of internal sampling rates including 96kHz, 88.2kHz, 48kHz and 44.1kHz.

The maximum sample rate (96kHz or 48kHz) is set by the AdminHD configuration and cannot be modified from the Central GUI.

Having configured the maximum rate, you may use the Sample rate option, in the **System Settings** display, to change from 48kHz to 44.1kHz, or from 96kHz to 88.2kHz, 48kHz or 44.1kHz.

System Clock (Sync Reference)

The Nova73 offers a fully redundant clock source structure with two independent clock inputs, an internal sync generator and the ability to lock to sync from an incoming multi-channel signal. This allows the console to be clocked from a variety of sync sources and recover from loss of external sync.

External sync connections are located on the Nova73 rear panel.

The sync signal priorities are defined using the Wordclock options in the **System Settings** display.

System Shutdown and Restart

Shutdown

The console should be shut down by powering off the control surface (mains connections at rear) and Nova73 (mains connections at front).

Note that the control system is located on the Router Module (MKII) within the Nova73. Therefore, it is here where your user data is stored.

Following switch-off, power is provided to the control system for a further 18 seconds. During this time, all current settings are saved to flash memory; this is known as the warm start data. You will hear several tones signalling that the shut down operation has been successfully completed. The system is shut down when the blue LED of the trackball is off.

You may switch off the power to other system components (e.g. DALLIS units) at any time.

Starting the System (Warm Start)

To start the system, turn on the power to the control surface (mains connections at rear) and Nova73 (mains connections at front). The components may be powered in any order, but note that the control system resides within the Nova73. Therefore, the system boots when you turn on power to the Nova73.

You may switch on the power to other system components (e.g. DALLIS units) at any time.

The control system boots in a few seconds; during this time the Central GUI reports back on the boot-up progress.

By default, the warm start data is loaded at the end of boot-up. This means that the system comes back exactly as it was when you last shut down, ensuring fast recovery of all previous settings following a loss of power.

Depending on who was last using the console, you may be sat in front of a fully configured control surface with DSP settings or a series of blank fader strips! In either case, the fastest way to reset the console is to load a production.

The control surface and Nova73 may be booted before DALLIS units. This enables you to prepare settings, including signal routing, before remote DALLIS stageboxes are connected or have received power.

Starting the System (Cold Start)

Alternatively, the system may be set to cold start, following the next reboot, using the Prepare Coldstart option in the **System Settings** display. Or, the **Prepare Coldstart** button on the front of the Router Module (MKII).

Select the **Prepare Coldstart** option, and then force a restart by powering off, and then on, the Nova73.

A cold start boots without loading any warm start data. You should perform a cold start *only* if there is a problem with the warm start data, or if you wish to clear all warm start data from the system.

The best way to reset the console for a new job or show is to load a production. (A cold start resets the system back to its cold start data and factory default settings.)

Warm Start & Cold Start Data

Warm Start Data

The following settings are stored in the warm start data, and are recalled following a warm start:

- Matrix crosspoints.
- The DSP configuration.
- The console's complete settings (control surface layout, etc.)
- All DSP parameters (EQ, Dynamics, etc.).
- All I/O parameters (Mic preamp gain, SRC on/off, etc.)
- Any Core configuration settings changed by an online AdminHD computer.

Cold Start Data

Following a cold start:

- All matrix connections are cleared, unless protected by a factory configuration (.tcl) file.
- The default DSP configuration is loaded. This can be defined from the Custom Functions display.
- The control surface will appear blank (no fader strip assignments).
- All DSP parameters are set to factory default values.
- All I/O parameters are set to factory default values
- All configuration files return to their cold start defaults (config.tcl, gui_config.tcl, etc.)

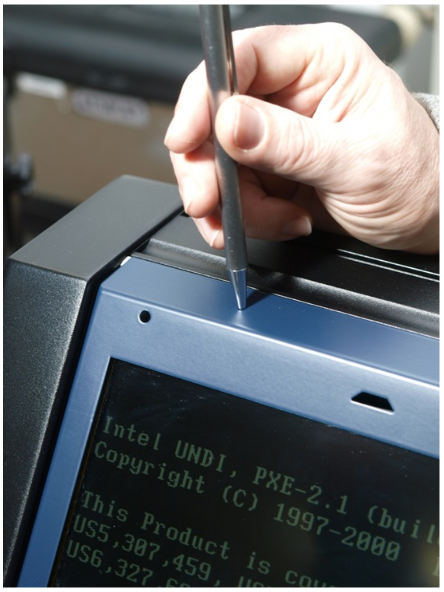
Restarting a Bay Server

Each TFT display on the mc²56 has its own Ethernet Bay Server which can be restarted from the front panel. You should perform this procedure, rather than a system restart, if:

- the graphics on an individual display freeze or look odd.
- the controls on a panel are not responding; indicators not updating.

These symptoms can sometimes occur if a Bay Server looses its Ethernet connection to the control system.

1. Using a pointed object, press the recessed button on the top of the display:



The Bay Server restarts.

Once the restart is complete, communication with the control system is re-established, and the selected display reinstated.

System Software Versions

Compatibility

From version 4.0.2.2 onwards, all Lawo products have adopted a consistent software release numbering system to indicate compatibility. This affects system networking, mxGUI and AdminHD. In each case, the first three digits of the software version must match.

Checking the Software Version

You can check the software version of your mc² system from the Global Options in the **System Settings** display.

Upgrading Software

Please register at www.lawo.de (click on **Login**) and go to the **Download-Center** to download software and documentation for your product.

Information about each software release can be found in the "Release_Notes_X.xx".

Instructions on how to perform each upgrade are included with the first software release candidate ("Product_systemupdate_VX-xx-x-x"). To perform a system update, you will need a computer connected to the Lawo system network, and installed with suitable FTP and Telnet clients. For details, see the "mc²56 Technical Manual".

Chapter 10: mxGUI

Introduction

This chapter covers **mxGUI**, the Lawo software programme which runs on an external computer to provide offline setup or remote operation of any mc^2 or Nova73 system.

Topics covered are:

- Overview
- Compatibility
- Computer System Requirements
- Software Installation
- Starting mxGUI
- Online/Offline Status
- Operating Principles
- Closing mxGUI
- Online Operation
- Offline Setup
- The File Transfer Display
- The Shared Folder
- The Strip Assign Display
- The Access/Assign Window

Overview

mxGUI (Matrix GUI) is a software programme which runs on an external computer to provide offline setup or remote operation of any mc² system:

- **Offline Setup** - productions, snapshots, sequences, mixes and presets can be prepared and stored on the mxGUI computer, and then transferred to the system at a later date; thus saving valuable setup time before a show.
- **Remote Operation** - mxGUI can run online by connecting the mxGUI computer to the mc²56 Control System (via Ethernet). This provides additional screen displays or remote operation for a second engineer.

mxGUI runs an emulation of the mc² control system, providing identical displays to those found on the mc²56, 66 and 90 Central GUI. This enables the creation of a complete production offline, including signal routing, labels, fader strip assignments, processing settings, snapshots, sequences, etc.

Compatibility

mxGUI may connect to any mc² system or Nova73 running Version 4.6 software or later.

From version 4.0.2.2 onwards, all Lawo products have adopted a consistent software release numbering system to indicate compatibility. In each case, the first three digits of the software version *must* match.

So, to connect to a mc²56 running **4.20.2.0**, you will need a computer running mxGUI **4.20.2.x**. You can check the mc²56 software version from the Global Options in the **System Settings** display, and the mxGUI version from the ? main menu.

Computer System Requirements

To install and run the mxGUI software, your computer *MUST* meet or exceed the following system requirements:

Windows PC:

- **Hardware**: 1.5 GHz (required for VirtualBox).
- **Operating System**: Windows XP, Windows Vista (32-bit), Windows 7 (32-bit and 64-bit)
- **RAM**: 1.5GB RAM (Windows XP), 2GB RAM (Windows Vista/7).
- **Hard Disc**: minimum 200 MB free space.
- **Operation**: Keyboard and mouse.
- **Interface**: Ethernet 10/100Mbit.

MAC:

- **Hardware**: 1.5 GHz (required for VirtualBox).
- **Operating System**: MAC OS X 10.6 (Snow Leopard), 10.7 (Lion), 10.8 (Mountain Lion)
- **Utilities**: X11 must be installed, see additional notes for MAC OS X.
- **RAM**: 2GB RAM
- **Hard Disc**: minimum 200 MB free space.
- **Operation**: Keyboard and mouse.
- **Interface**: Ethernet 10/100Mbit.

Software Installation

Lawo's mxGUI software runs on a "virtual Linux machine" inside your computer. This provides the same operating platform as on a real mc² or Nova73 system. To achieve this, three separate programmes are installed by the mxGUI installer:

- **mxGUI** – Lawo's application software.
- **Oracle VM VirtualBox** – this programme creates the "virtual machine" which runs the Linux operating system.
- **Xming X Server** – this programme deals with the management of TCP/IP ports within the mxGUI computer.

Warning

Having completed the installation process you should not need to open or modify the Oracle VirtualBox or Xming programmes, as all settings are automatically dealt with by the mxGUI installer (except in MAC OS X 10.8).

Licensing

From Version 4.14 onwards, Lawo's mxGUI application is free of charge and does not require a software licence.

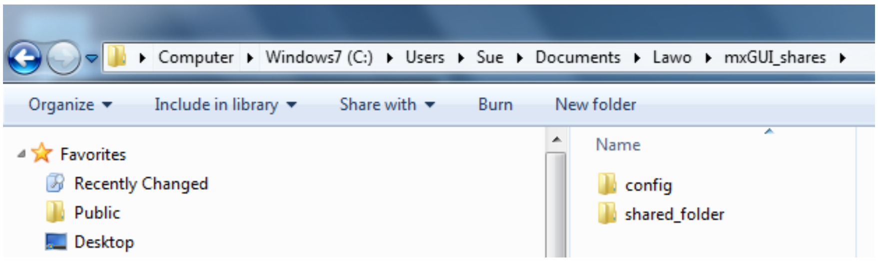
Installation Procedure

Please refer to the separate "mxGUI Installation Manual" for step-by-step instructions on how to install the software, taking note of the User Defined Folder location, and additional notes for MAC OS X.

If mxGUI is already installed on your computer, and you wish to install a different version, then it is recommended that you uninstall the current version first, see Uninstall & Update.

User Defined Folder (mxGUI_shares)

During the installation, please note that the location of the user defined folder (**mxGUI_shares**) is where the **config** and **shared_folder** will be stored:



The Shared Folder provides access to mxGUI files from your host operating system. You will need access to this folder to copy mc²56 user data onto your computer (e.g. to/from USB, email, etc.)

The **config** folder stores the complete “Local Control System” for the mxGUI computer. You should not need to access this folder. However, make sure you don’t edit or delete the **config** folder contents, otherwise you may edit or delete the mxGUI control system!

The default location of the **mxGUI_shares** folder is inside the user's Home Directory:

- **Windows XP/Vista/Windows7**: <HOMEPATH>\My Documents\Lawo\mxGUI_shares
- **MAC OSX**: <HOMEPATH>\Library\Lawo\mxGUI_shares

Depending on your computer’s configuration, this location may be hidden to normal users.

On **Windows**, use the "Show hidden files, folders and drives" option within the Control Panel to reveal hidden files and folders.

On **MAC OSX**, you can unhide the "mxGUI_shares" folder as follows:

Select **Go -> Go to Folder** from the Apple menu bar and type in:

~/Library

To permanently unhide the folder, use the Terminal application (included in your Utilities folder) and type:

Chflags nohidden ~/Library/

With the default path, each user has their own mxGUI configuration. If you wish all users to share the same mxGUI configuration, then change the default "User Defined Folder" path to outside the Home Directory during the mxGUI installation.

Additional Notes for MAC OS X Installation

➢ ORACLE Virtual Box for OS X 10.8 (Mountain Lion):

If you install mxGUI on a MAC OS X 10.8.x.x (Mountain Lion) computer, then you *MUST* update your **Oracle VirtualBox** to a Mountain Lion compatible version such as **ORACLE Virtual BOX 4.2.xx**. (The Oracle Virtual BOX 4.16, installed by the mxGUI installer, will *NOT* work on the Mountain Lion operating system.)

You can download this software from http://www.virtualbox.org

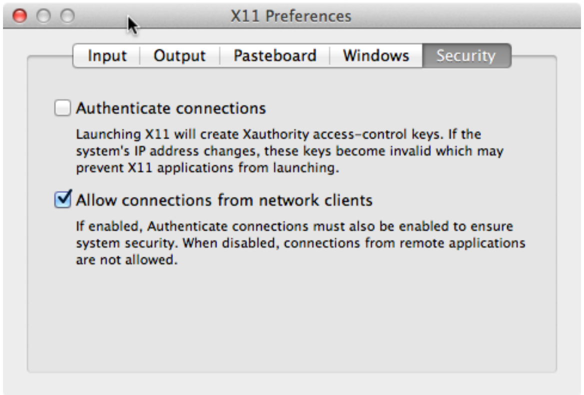
➢ X11 Installation (All OS X Versions):

The **X11** Window System utility *MUST* be installed in order to run mxGUI. Look in the "Applications -> Utilities" folder - **X11** must be present. If not:

- **10.6 (Snow Leopard); 10.7 (Lion)** - **X11** can be installed from the OS X installation CD (look under "Optional Installs").
- **10.8 (Mountain Lion)** - **X11** can be downloaded from http://xquartz.macosforge.org/trac

After installation, it is important to set the **X11** Security Preferences as follows:

1. Open **X11** (select "Applications -> Utilities -> X11").
2. Open the **Preferences** dialogue box and select the **Security** tab.
3. Enable the "**Allow connections from network clients**" checkbox:



Uninstall & Update

Uninstalling mxGUI

To uninstall mxGUI completely, remove all three programmes from your computer:

- mxGUI
- Oracle VM VirtualBox
- Xming X Server

On Windows 7, you can use the “Uninstall a program” option within the Control Panel to do this.

On MAC OS X, remove the programmes from your "Applications" folder.

Updating mxGUI

To install a different version of mxGUI, then you should uninstall the current version first.

So, in Windows 7:

1. Use the “Uninstall a program” option within the Control Panel to remove mxGUI.

Note that it is not necessary to remove the Oracle VirtualBox or Xming X Server programmes.

2. Run the new mxGUI installer to re-install mxGUI.

At the end of the install, the Oracle VirtualBox installer automatically opens - cancel the VirtualBox installer as it is not necessary to re-install this programme.

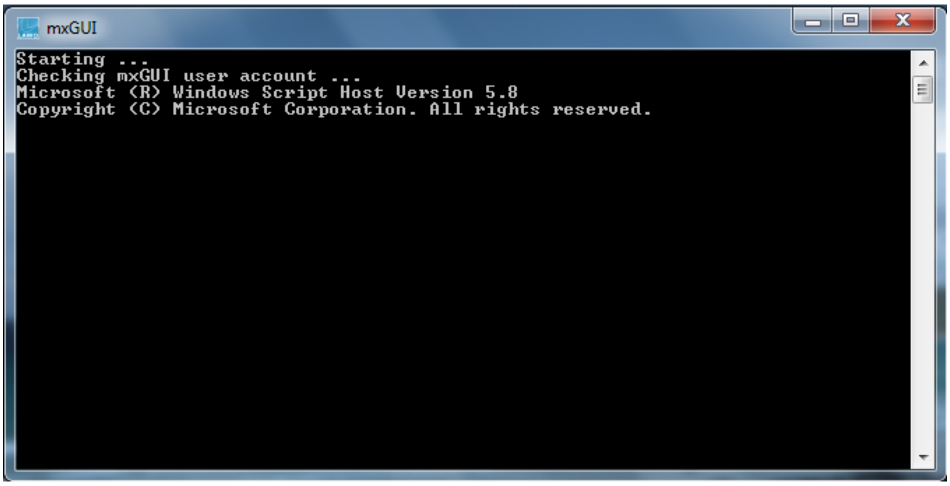
3. Following the installation or re-installation of mxGUI, a restart of the computer is advised.

Starting mxGUI

1. Start the programme, by selecting **mxGUI** from the START menu (Windows) or Applications folder (MAC). Alternatively, click on on the desktop icon:

The programme automatically launches the Xming X server and the Oracle VirtualBox to provide the “virtual Linux machine” which will be the platform for the mxGUI application.

The following window appears while these programmes start up; this may take a while:

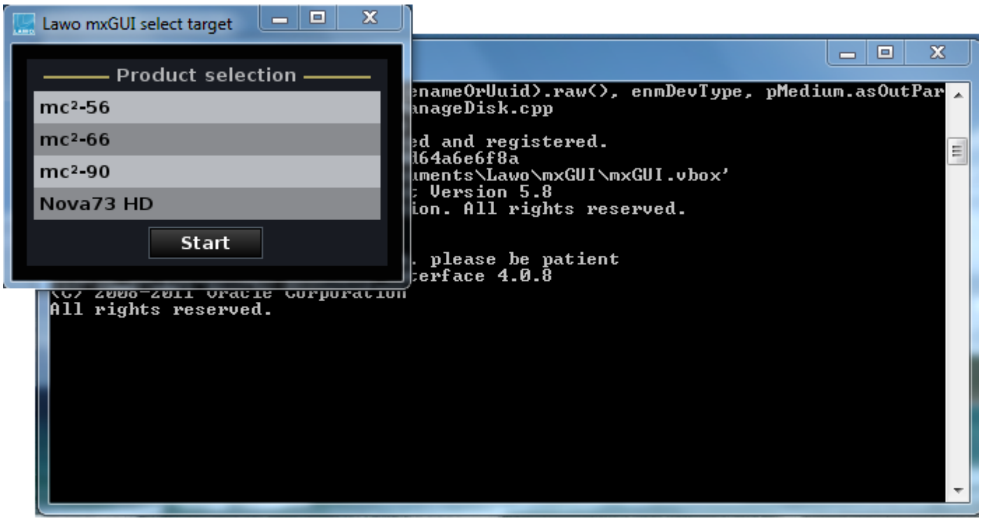


If you have a firewall installed on your computer, you will need to unblock the firewall access for the Xming X Server programme. Once you have authorised the firewall access, you shouldn’t need to deal with this security alert again.

If you are running Windows 7, then you may also be prompted to allow changes to your User Account.

2. Select **Yes** on any pop-up windows to authorise these changes and continue.

Once the VirtualBox and Xming have booted, you will see the mxGUI launch window:



From here you can choose which system you wish to emulate – mc²56, mc²66, mc²90 or Nova73. This ensures that only the features relevant to your product are available from the mxGUI displays.

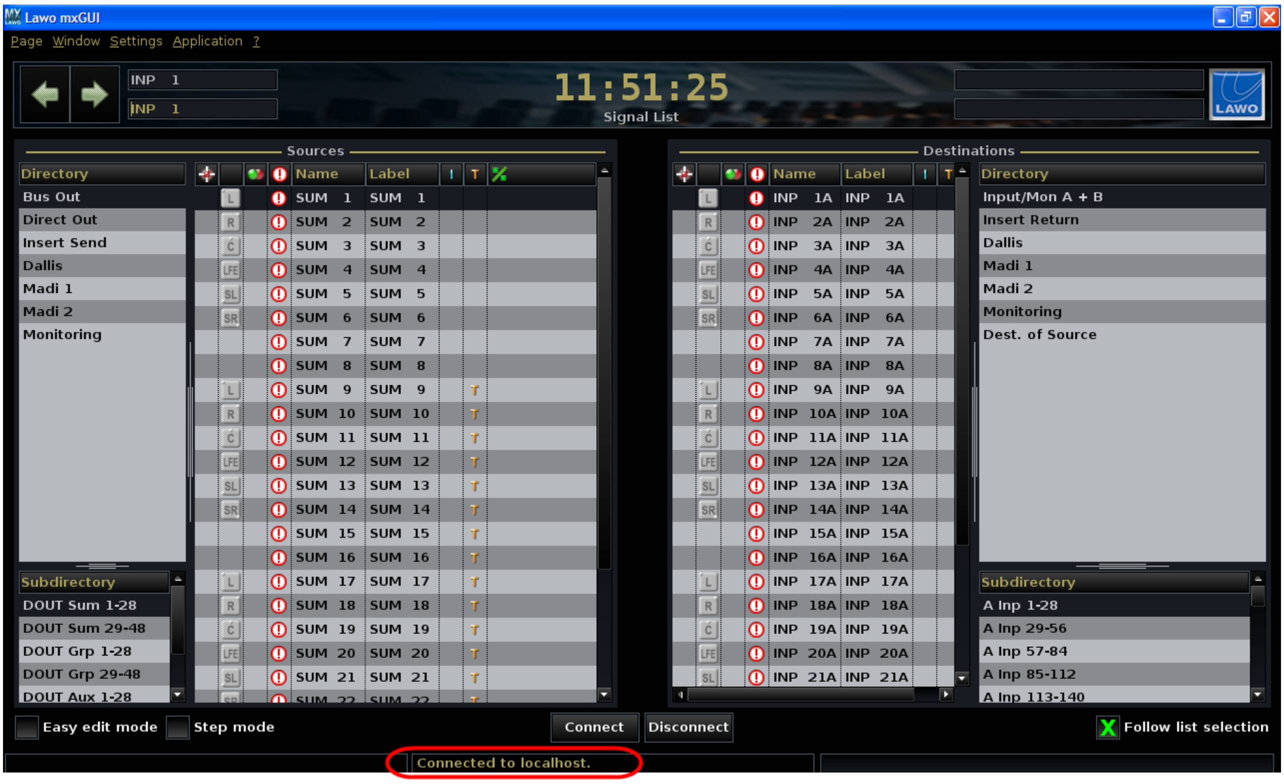
Note that only the systems selected during the software installation appear. In our example, all four systems are available.

3. Select an option (e.g. mc²56) and click **Start** to launch mxGUI.

mxGUI boots up and loads its warm start data (the settings saved when mxGUI was last shutdown). The programme is ready to use once you see the **Lawo mxGUI** operating window:

Online/Offline Status

The first time you start mxGUI, it opens in offline mode. (Once configured, you can use the Reconnect option to automatically start in online mode.) The online/offline status is shown in the status bar at the bottom of the display:



Offline

When offline, mxGUI is connected to the "local host". This means that data is being saved and loaded to/from the "Local Control System", i.e. on your computer.

If this is the first time you have started mxGUI, then your **Signal List**, **Productions**, **Snapshots**, etc. will bear no resemblance to the displays on your system. This is because your "Local Control System" is running from a default configuration.

The best solution is to transfer the complete configuration (and some productions) from your mc²56 to the mxgui computer. See Synchronising the Configuration.

If you don't have access to a system, and want to play around with the mxGUI interface, you can build a simple production using the default configuration installed by the mxGUI installer.

> The default mxgui configuration is very basic, and provides limited tools for offline setup (as elements such as signals will not match those of your system).

Online

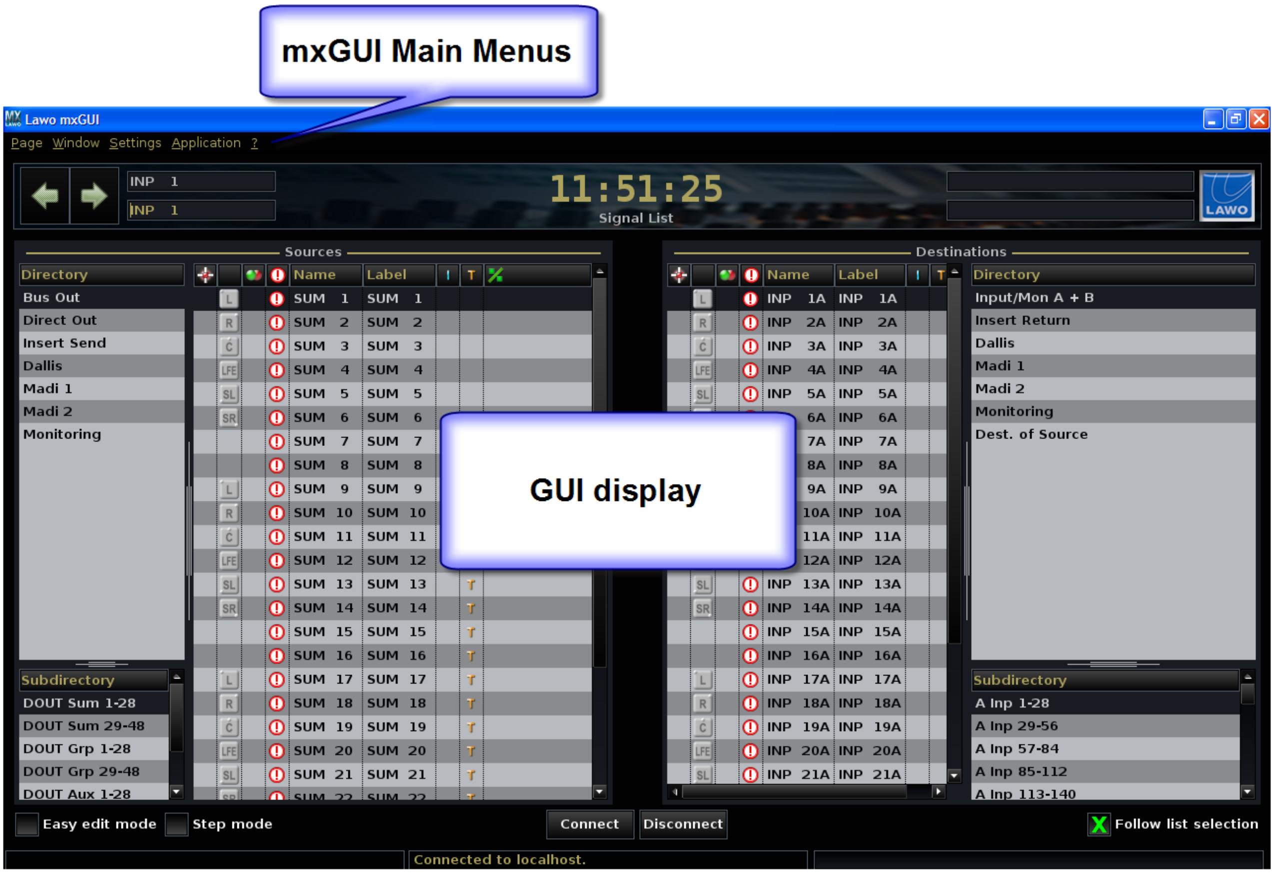
When mxGUI operates online, the status bar shows the IP address of the connected host:

Any changes you make immediately affect the online system, and all data is saved and loaded to/ from the host mc²56 control system.

For more details, see Online Operation.

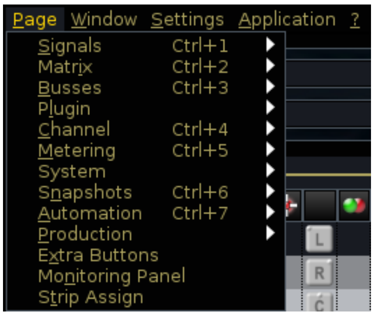
Operating Principles

The mxGUI operating window is virtually identical to the Central GUI on an mc² console:



Changing Display

1. Click on the drop-down **Page** menu to access the same Screen Control displays as on your mc² system: **Signal List**, **Signal Settings**, etc:



You can use the next/previous page buttons or keyboard "hot keys" for faster access to displays.

You will find two displays which appear only within mxGUI (and not on the console):

- **Strip Assign** – provides an overview of channel and main fader strip assignments. It may be used with the Access/Assign pop-up window to change fader strip assignments, change bus routing, copy audio parameters, etc. In addition, you can control fader levels and enter user labels from this display.
- **Production -> File Transfer** - replaces the console's File display, and allows you to transfer files between your computer and any mc² system.

Other Main Menus

- **Window -> Access/Assign** - opens the Access/Assign pop-up window. This mimics the console's ACCESS/ASSIGN control panel.
- **Settings -> Connection** - opens the Connection pop-up window. This is used to connect to a real system in order to work online.
- **Application -> Quit** – closes the mxGUI application.
- **? -> Info** – shows the release version of the mxGUI software and Lawo service contact details:

Adjusting Settings

mxGUI adjusts settings using the mouse and keyboard on your computer. These operations are also available on your mc² console via the trackball and console keyboard. Therefore, please see the links below for details. Depending on the function, you may:

- Click on a dedicated on-screen button.
- Right-click on a selection to reveal the context menu options.
- Enter names using the keyboard.
- Click on the up/down arrows beside a parameter, or use the keyboard to adjust parameter values.

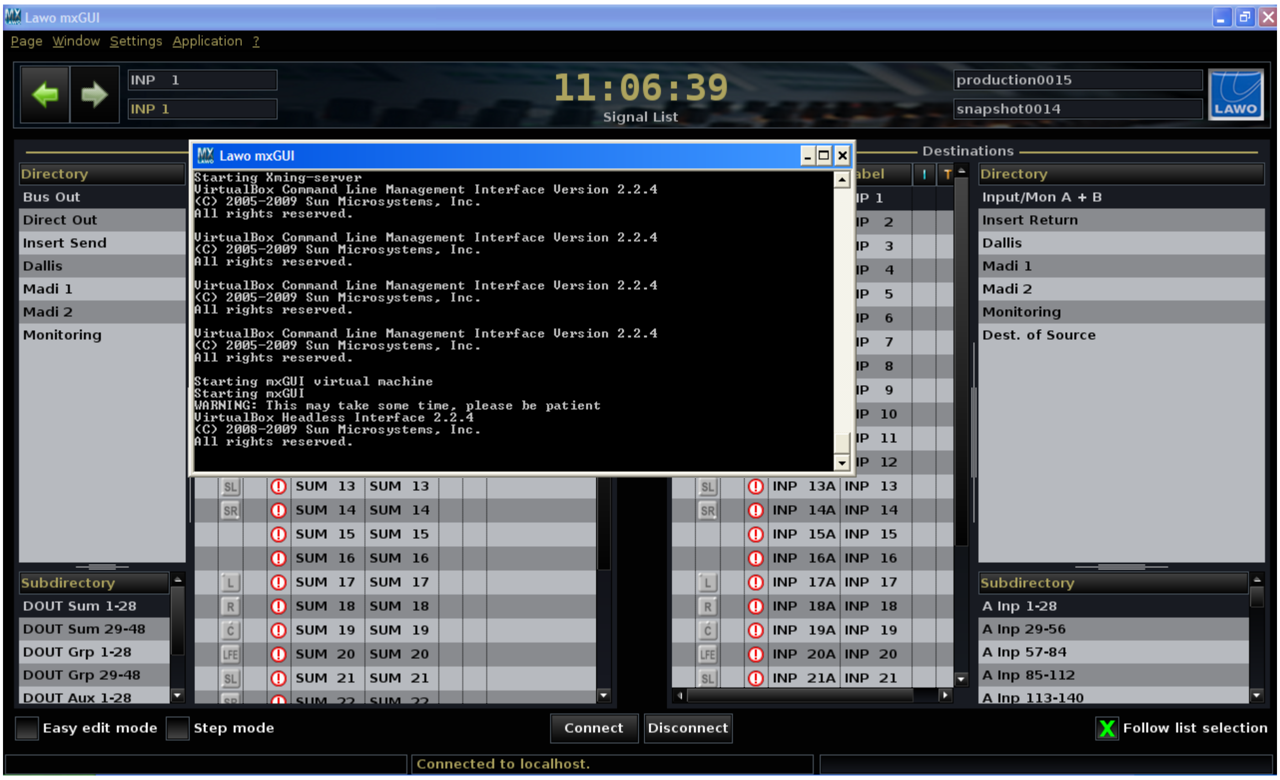
> You can use the mouse wheel (if you have one) to adjust parameter values or scroll up/ down lists in focus.

Keyboard Shortcuts

See Keyboard Shortcuts for a list of useful "Hot Key" functions.

Closing mxGUI

mxGUI runs on a virtual Linux machine inside your computer. Therefore, when running the software, you will notice that two windows are open: the mxGUI operating window and the virtual machine:



You can maximise or minimise these windows in the usual manner. So, for normal operation, maximise the mxGUI operating window to hide the virtual machine.

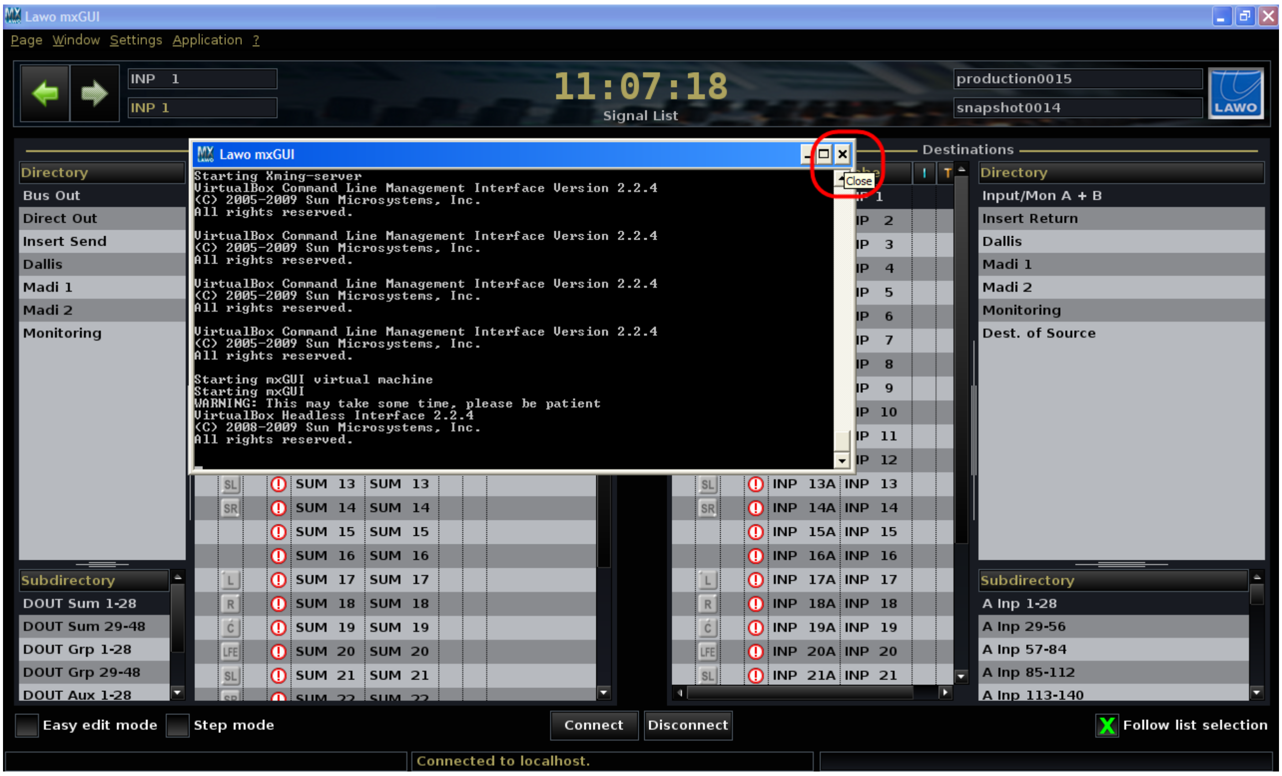
➢ **To close the mxGUI programme:**

1. Select **Application** -> **Quit** from the main menu bar, or click on the close icon at the top right of the display:

Either operation quits mxGUI and exits back to the launch options.

> When running offline, the latest settings are saved to the local control system's warm start data.

2. Alternatively, click on the VirtualBox close icon to quit the virtual machine:



Note that if you close the Virtual Box window *BEFORE* closing the mxGUI operating window, then mxGUI shuts down without storing any warm start data to the local control system.

Online Operation

When operating online, the mxGUI computer talks to a real mc²56 control system via its control network (Ethernet).

In this mode, mxGUI is simply acting as a remote control. You will see the same **Signal List**, **Productions** list, etc. as on the mc² system, and all data (productions, presets, configuration, etc.) is being saved and loaded to/from the host control system.

Network Connection

The mxGUI computer must be connected to the Lawo system network port of the mc²56 control system.

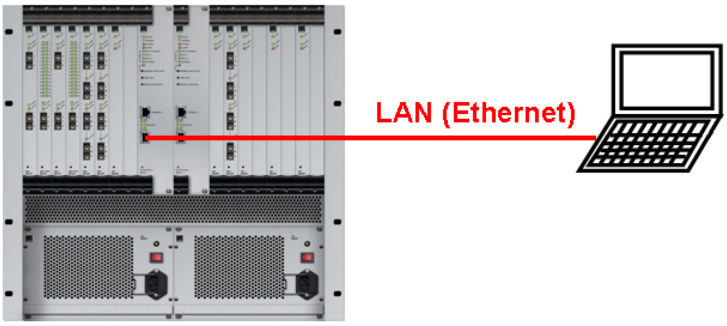
> Note that the location of the control system varies depending on the Lawo product, see Control System Locations.

On the mc²56, the **ETHERNET B** port on the Nova73 Router Module (MKII) should be used.

For a Direct Connection

Use a *crossed* network cable (STP-CAT 5 with RJ45 connectors):

1. Connect the device to the **ETHERNET B** port on the active Router Module MKII:



> If a redundant Router Module is fitted, and a control system takeover is actioned, you will lose your network connection. Therefore, a network switch is recommended.

For a Connection via a Network Switch

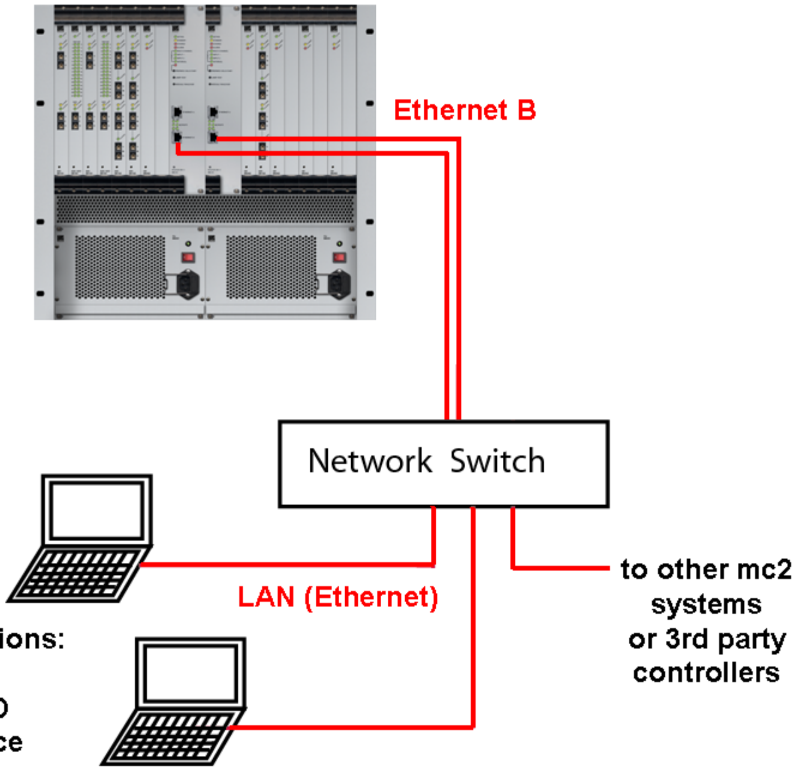
Use a *straight* (1:1) network cable (STP-CAT 5 with RJ45 connectors):

1. Connect the device to the network switch.

And, if not already installed:

2. Connect the network switch to the **ETHERNET B** port on the Router Module MKII.

3. If a redundant Router Module is fitted, then run a second network connection. This ensures continued operation should a control system takeover occur:



Warning

You must use a network switch and *NOT* a hub.

Keep the Lawo network separate from other network traffic within the installation.

For more information on installing a suitable network switch, please contact your local Lawo representative or email service@lawo.de.

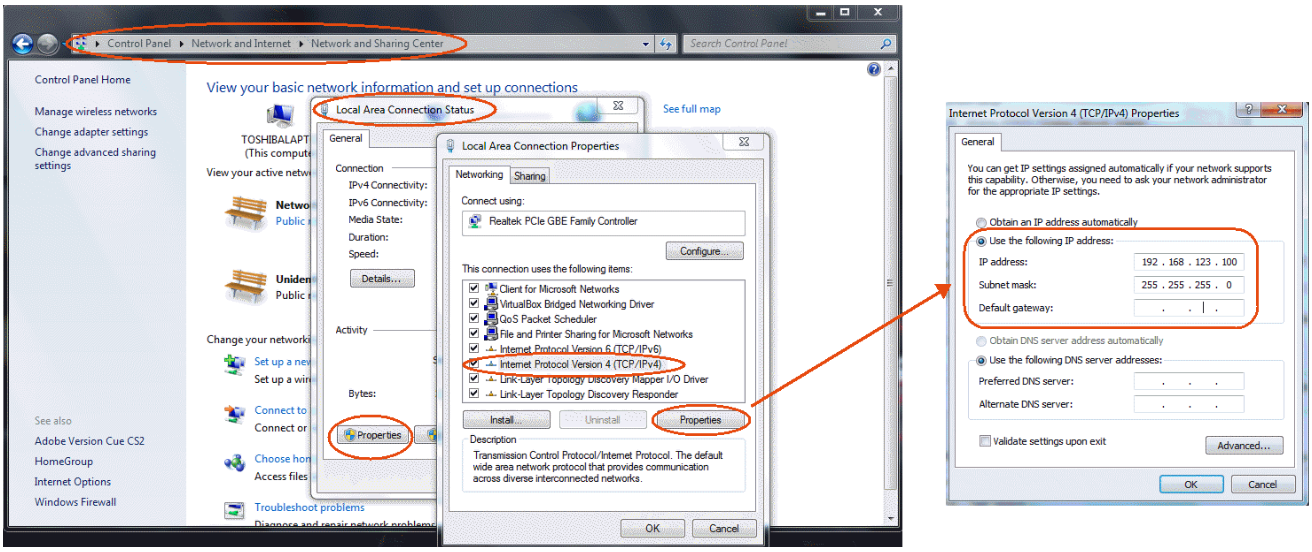
Depending on the number of network connections, one mc²56 system is able to support up to 16 clients simultaneously

TCP/IP Configuration

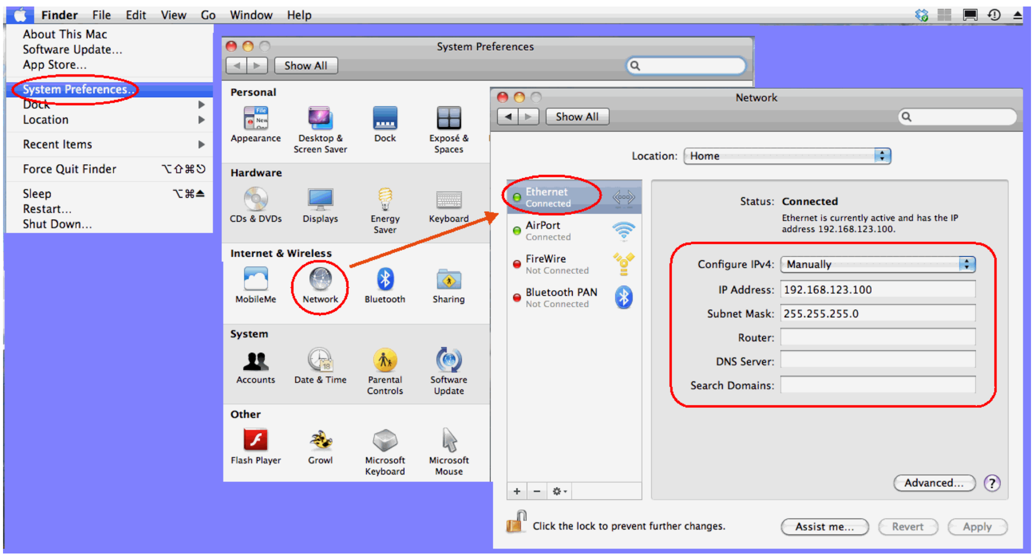
To establish communication with the control system, you will need to configure the TCP/IP settings for your device's network interface card.

The following screenshots demonstrate how to do so on a computer running Windows 7 and Mac OS X:

Windows 7:



Mac OS X:



You can find further information from www.microsoft.com or www.apple.com.

IP Address

The IP address of your device's network interface card must be unique, and set within the same range as that of the mc²56 control system.

You can check the IP address of your control system from the console GUI (using the Signal Settings display). See TCP/IP Addresses for a list of the default IP addresses for different Lawo products.

For example, to connect to a mc²66 with a default IP address = 192.168.102.65, set your device's IP address to **192.168.102.101**.

In a networked installation, it is likely that you will be connecting via an Ethernet switch, so please consult your network administrator for further details.

> Take care when setting the IP address of your device. If there is an IP conflict within the network, then the console may not operate correctly.

Subnet Mask

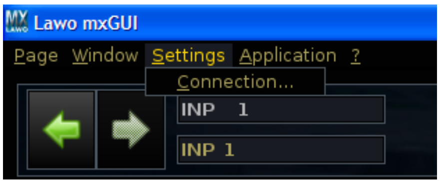
The Subnet Mask of your device's network interface card should be identical to that of the system. For all products, the default Subnet Mask is **255.255.255.0**.

Checking Network Communication

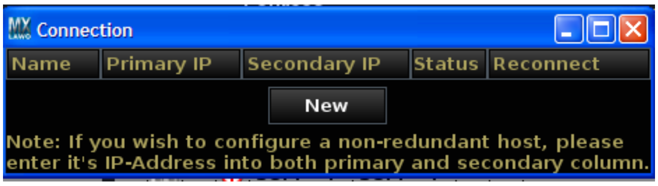
You can use AdminHD, mxGUI or the Web Browser Interface to check the network communication.

Getting Online

1. Start mxGUI, making sure that you open the correct emulation (e.g. mc²56).
2. Select **Settings** -> **Connection** from the mxGUI main menus:



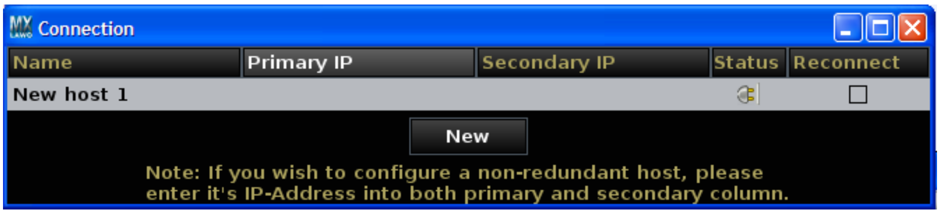
The 'Connection' pop-up window appears:



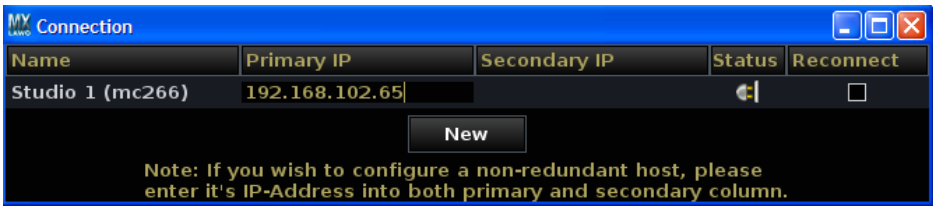
If this is the first time you have used the 'Connection' window, then it will be blank. (This window lists all the systems which mxGUI can connect to, each with a **Name**, **Primary IP** address (main control system), **Secondary IP** address (redundant control system) and connection **Status**).

3. Click on **New** to create a new connection.

A generic host control system is added to the connections list:



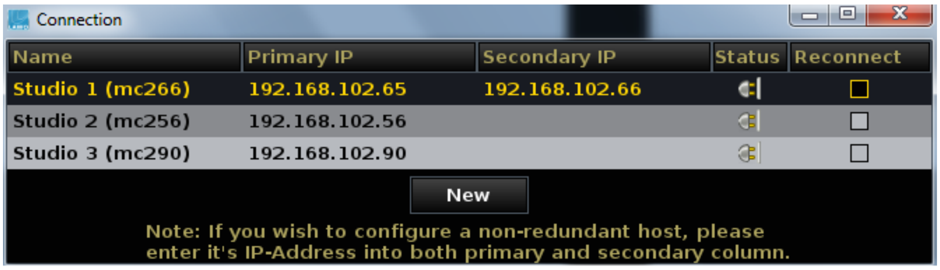
4. Click on **New host 1** to enter a name for this system – in our example, we have chosen **Studio 1 (mc266)**.
5. Then enter the **Primary IP** address of the main control system, for example:



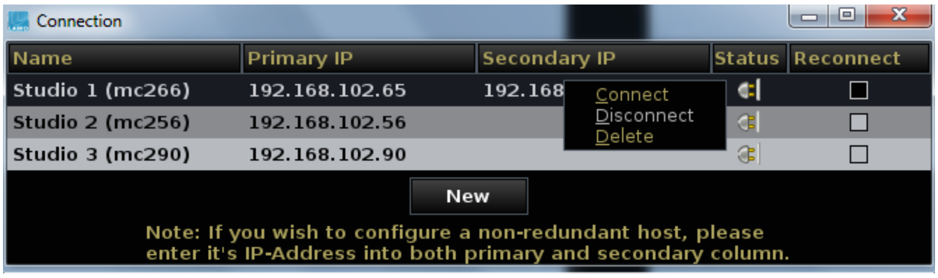
The connection is now prepared and you are ready to go online.

> If the system does *NOT* have a redundant control system, then you only need enter the **Primary IP** address.
>
> If there is a redundant control system, then the **Secondary IP** address must also be entered. This is *always* 1 above the **Primary IP**. So, for example, if the **Primary IP** address is **192.168.102.65**, enter **102.168.102.66** for the **Secondary IP**.

You can prepare several connections for systems which you may wish to connect to at a later date. Our example below shows three different mc² connections, all currently offline:



6. To connect to one of the systems in the list, right-click on its entry and select **Connect**:



The mxGUI computer will now attempt to connect to the selected system:

- If the connection is successful, then the **Status** column updates to show the "plugged in" icon.
- If the connection fails, then the **Status** remains as "unplugged":
  - o Check the network connection and TCP/IP settings of your computer's network interface card.
  - o Check that mxGUI is compatible with the mc² or Nova73 system (the first three digits of the software versions *must* match.)
  - o See also the trouble-shooting tips to resolve the problem.

7. If you wish mxGUI to automatically reconnect if the system loses its online status, then tick **Reconnect**. mxGUI will also start up in online mode if a valid network connection is present.

8. Once you have a valid connection, you can minimise the **Connection** window.

Notice that the status bar at the bottom of all mxGUI displays shows the IP address of the connected host:

You can now use the mxGUI displays to view or change settings on the online system.

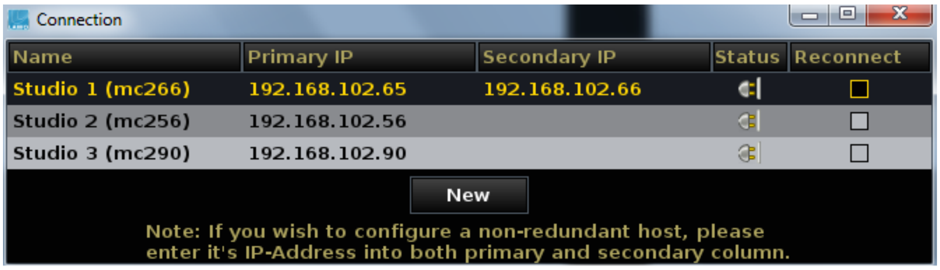
Any changes you make are actioned immediately, and all data is saved and loaded to/from the host control system. So, make sure any other operators are aware that you are online!

Disconnecting mxGUI

To disconnect mxGUI from the mc²56 system:

1. Maximise or open the **Settings** -> **Connection** window.
2. Select the online system, right-click and choose **Disconnect**.

mxGUI disconnects and the **Status** of the Connection returns to is "unplugged" state:



mxGUI may connect to one system at a time. Therefore, if you connect to a different mc² or Nova73 system, any existing connection is automatically cancelled.

Offline Setup

When running offline, mxGUI runs an emulation of the mc²56 control system. All data is saved and loaded to/from your computer (known as the “Local Control System”).

Settings are prepared offline by saving productions from the **Productions** display, presets from the **Main** display, etc. These files are then transferred back to the mc²56 either by copying to USB (via the Shared Folder), or going online and using the File Transfer display.

Files & Compatibility

What can be Prepared Offline?

Anything which can be saved on a real system can be saved offline onto the mxGUI local control system:

- **Productions**, **Snapshots** and **Mixes** - are saved from the Productions, Snapshots and Mixes displays.
- **Presets** – module or channel presets are saved from the Main display.
- **Custom Function Assignments** – the mapping of user buttons and other custom function assignments can be edited from the Custom Functions display.

> When mxGUI is started in Nova73 mode, presets and mixes are not accessible (as these features are not supported by a stand-alone Nova73 matrix).

Compatibility

- **Productions** can be loaded on any mc² or Nova73 system. However, only transferable elements will load. For example, you can recall a snapshot created on one console to another providing you are using the same channel type. However, inputs and outputs are specific to the system, so signal routing will not load unless supported by an identical Core and Signal List Configuration.

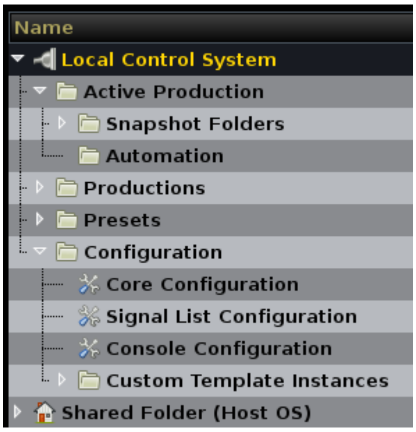
> When preparing a production, it is important that the local control system configuration, on your mxGUI computer, matches that of your actual system. This ensures that any productions you create will load in full when they are transferred back to the mc²56. See Synchronising the Configuration.

- **Presets** can be loaded to any mc² console regardless of the configuration or mc² mode.
- **Custom Function Assignments** – these files are specific to the function. This means that you can transfer a custom assignment file created on mxGUI to any console or Nova73; if the receiving system does not support the same User Panel or HLSD, the custom assignment may be edited from the Custom Functions display.

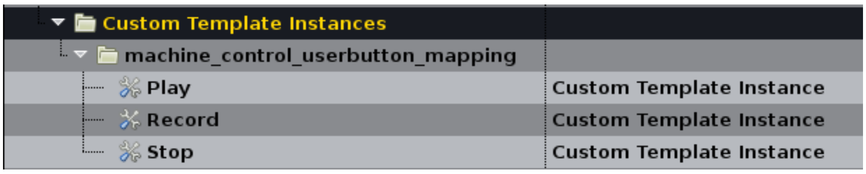
Please see Transferring User Data for more information on exchanging data within a production (snapshots, mixes).

Where are the Files Stored?

The File Transfer display provides access to all user data stored on the **Local Control System**:



- **Active production** – the active production can be opened to access individual snapshot folders, snapshots and automation mixes.
- **Productions** – contains all zipped productions; these can be transferred as a complete file.
- **Presets** – contains all module and channel presets.
- **Configuration** – contains the configuration data (see below). You may open the **Custom Template Instances** folder to access assignments made from the Custom Functions display:



Configuration Data

The complete configuration set contains four individual component files:

- **Core Configuration** – defines the Nova73/DALLIS System and its signal parameters (config.tcl).
- **Signal List Configuration** – defines the Directories, Subdirectories, Signal Names and Labels of the Signal List display (gui_config.tcl).
- **Console Configuration** – defines the console surface (console_config.tcl).
- **Custom Template Instances** – store the custom function assignments edited from the Custom Functions display.

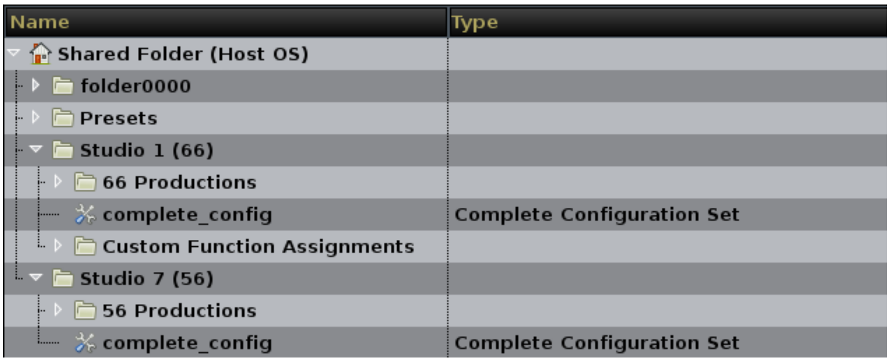
The first three files cannot be edited by mxGUI and are included for service/AdminHD access.

For simplicity, the complete configuration can be zipped and transferred as a single file - called the **complete_config**.

Organising Your Files

If you are going to prepare settings for a range of different systems, then it is a good idea to organise your files carefully before you start working with mxGUI.

1. Use the File Transfer display to create a separate folder, inside the **Shared Folder**, for each mc² or Nova73 system.
2. Then within each studio's folder, create a sub folder for productions, custom functions, etc:



This will allow you to keep all the relevant files together for each studio's configuration: **Productions**, **Custom Function Assignments** and configuration (**complete_config**).

We have also created a **Presets** folder to store module or channel presets. Because presets can be recalled on any mc² console, this is a top level folder and is not system specific.

Synchronising the Configuration

In order to prepare a production offline, it is important that the local control system configuration, on your mxGUI computer, matches that of your mc²56. If not, some parts of the production, such as signal routing, may not load correctly, see File Compatibility.

The best solution is to transfer the complete configuration from your mc²56 to the mxGUI computer.

> You only need to perform this operation once (providing there are no changes to your system configuration).

For a fail safe approach, it is best to perform this operation in two stages using the mxGUI Shared Folder:

- Step 1 - Go online, and transfer the **complete_config** from the mc²56 to the **Shared Folder**. This places a copy of the system configuration on your computer. (If you do not have online access, then request a copy of the "complete_config" file from your system administrator, and copy this into the **Shared Folder** using your host operating system, see Shared Folder.)
- Step 2 - Disconnect (go offline) and transfer the copied **complete_config** from the **Shared Folder** to the local control system. Then cold start mxGUI, and it will boot-up using the new configuration data.

Warning

While it is possible to transfer the **complete_config** from the Remote to the local control system in one step, this is *NOT* recommended. *IF* you transfer in the wrong direction, then you may overwrite the configuration data on your mc²56!

Step 1. Transfer the complete_config to the Shared Folder

1. Connect your mxGUI computer to the remote system, and configure its TCP/IP settings.
2. Open an online connection.
3. Open the File Transfer display.
4. Select a location within the **Shared Folder** (on the left) as your destination - in our example, the sub folder **Studio 1 (66)**.
5. Then right-click on the console's **Configuration** directory (on the right) and choose **Transfer**:

All the configuration files are zipped and transferred to the mxGUI **Shared Folder** as a single file - **complete_config**.

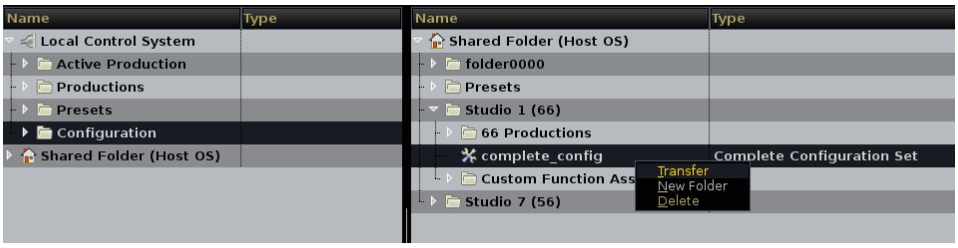
While online, it is a good idea to transfer some productions from the online system (on the right) to the mxGUI computer (on the left). You can then load a production later, rather than having to build your offline setup from scratch.

Step 2. Change the mxGUI Configuration

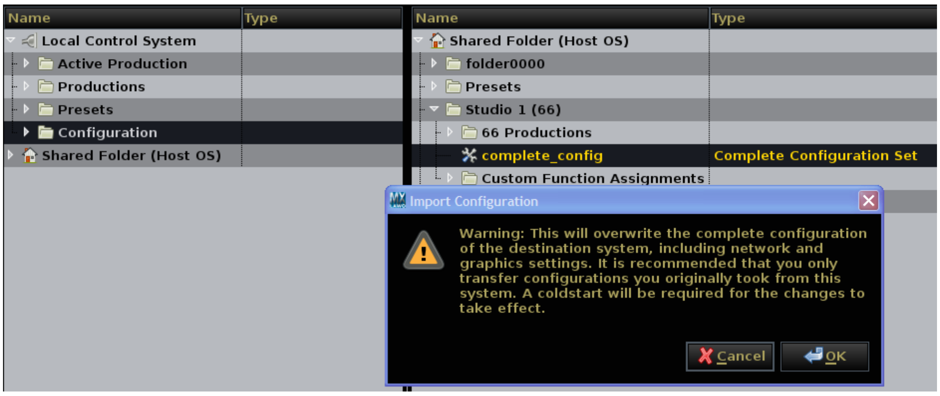
1. Disconnect mxGUI from the online system.

2. From the File Transfer display, select the local control system's **Configuration** folder (on the left).

3. Locate the **complete_config** file you transferred earlier (on the right), right-click and choose **Transfer**:



A pop-up appears warning you that **OK** will overwrite the configuration of the mxGUI local control system:

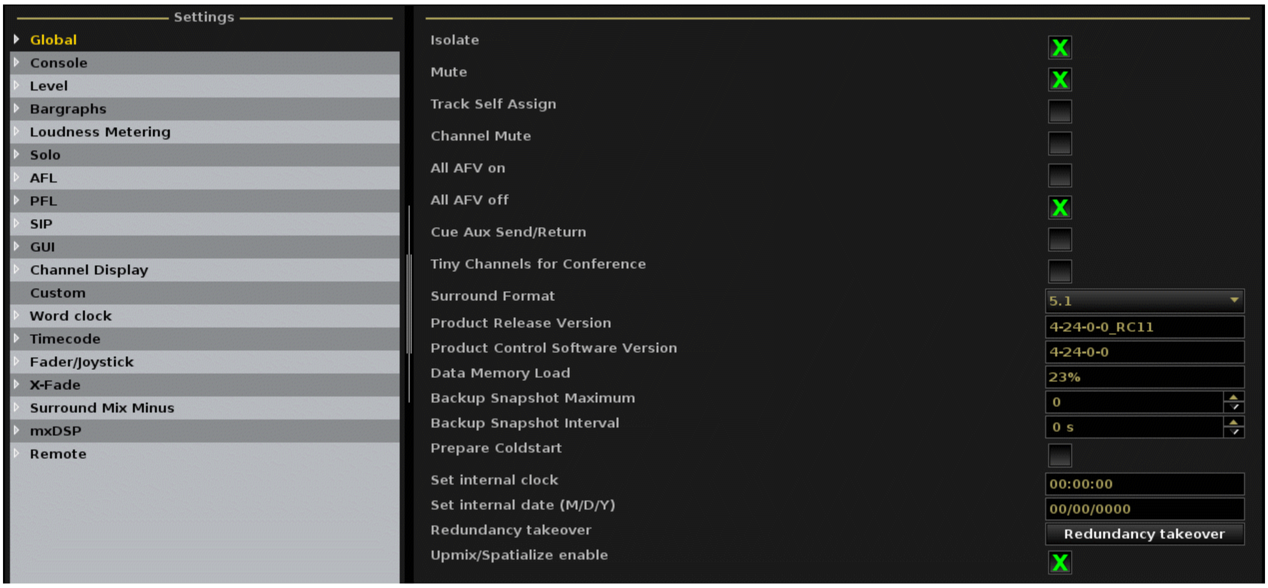


4. Select **OK** to continue.

The configuration is transferred.

You now need to cold start mxGUI before the new configuration data takes effect. To do this:

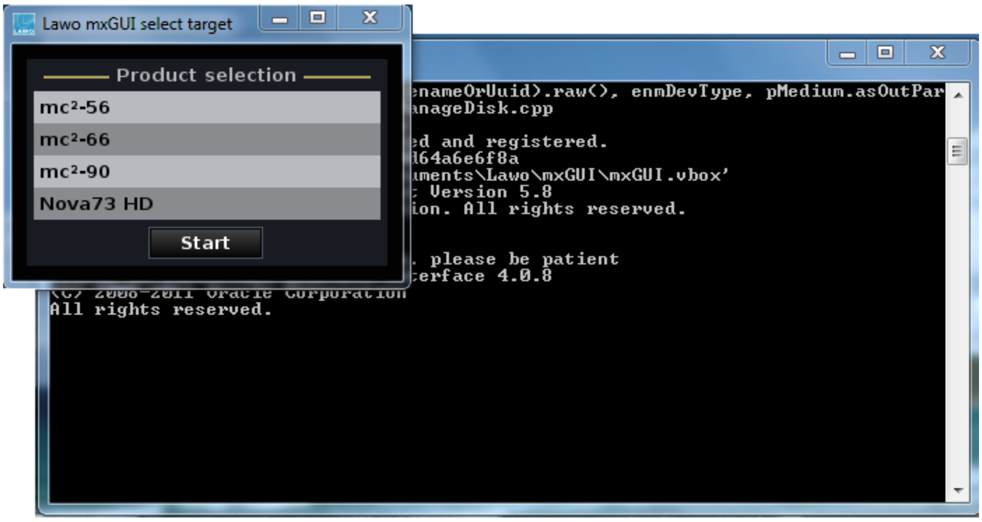
1. Select the **System** -> **System Settings** display.
2. And select the **Global** topic:



3. Enable the **Prepare Coldstart** option.

This prepares mxGUI so that on the next restart it will perform a cold start rather than warm starting from the current configuration.

4. Close mxGUI, by selecting **Application** -> **Quit**.
5. Then restart the programme from the launch options:



After the restart, you will be running the new configuration.

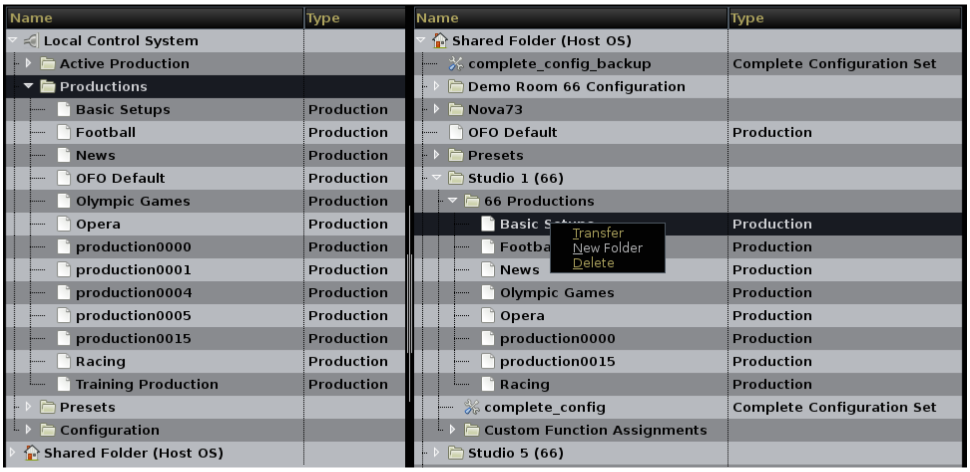
You can check this by looking at the Directories and Subdirectories within the Signal List and/or the Nova73 configuration in the Signal Settings display.

Preparing & Saving Settings

You may now begin your offline setup:

Productions

1. If you transferred some productions from the online system, then copy these into the **Productions** folder of the local control system using the File Transfer display:



2. Load a production from the Productions display. This provides a great starting point for your offline setup.

3. Now make changes, and save your settings either by saving or updating the production.

You can use the Strip Assign display, and Access/Assign pop-up window to perform tasks which normally use the console front panel - for example, changing the channel in access, assigning channels to fader strips, etc.

Presets

Module or channel presets may be saved in the usual manner from the Main display.

To modify an existing preset, copy it into the **Presets** folder of the local control system.

Custom Functions

Custom Functions may be programmed in the usual manner from the Custom Functions display.

To modify an existing function, copy it into the **Custom Template Instances** folder of the local control system.

Other Data

You may also save snapshots and mixes from mxGUI, and access these individual files from the **Active Production** folder of the local control system.

Remember to save or update the production to save the snapshot or mix permanently onto the local control system. If not, your snapshots and mixes are only held in temporary memory.

Transferring Data Back to the System

Having prepared a file, it can be transferred back to the mc²56 either by copying to USB (via the Shared Folder), or going online and using the File Transfer display.

For example, to transfer a production via the **File Transfer** display:

1. Connect your mxGUI computer to the remote system, and configure its TCP/IP settings.
2. Open an online connection.
3. Open the File Transfer display.
4. Select the online system's **Productions** folder as the destination (on the right).
5. Then on the left, right-click on the production you wish to import and choose **Transfer**.

The production is copied to the online control system.

6. Now go to the console and load the production.

Your setup is recalled!

You can transfer any type of file: productions, snapshots, automation, presets, and custom function assignments to the online control system.

A production created offline will only load completely if:

- The configuration running on mxGUI matches that of the online system.
- mxGUI is running the correct mc²/Nova73 emulation.

Good Housekeeping

Having completed a transfer, it is a good idea to keep a copy of the production in the mxGUI **Shared Folder**. This ensures that you keep a backup of everything needed for this offline setup: the **complete_config**, **production**, etc:

Name	Type
Shared Folder (Host OS)	
folder0000	
Presets	
Studio 1 (66)	
66 Productions	
complete_config	Complete Configuration Set
Custom Function Assignments	
Studio 7 (56)	
56 Productions	
complete_config	Complete Configuration Set

Note that when you change the mxGUI configuration, all other folders – **Active production**, **Productions** and **Presets** – remain intact. This means that you may end up with a mixture of productions from different systems on the same mxGUI local control system.

We recommend keeping a backup of all files within the Shared Folder. Create a sub folder for each mc² and Nova73 system so that you can store all configuration data and productions together. This way you will know which productions match which configuration in a few weeks time! See Organising Your Files.

The File Transfer Display

The **File Transfer** display allows you to transfer Productions, Presets and Configuration files between the local control system (your mxGUI computer) and an online control system (mc^2 console or Nova73). You might use this display to:

- Transfer configuration data to mxGUI.
- Transfer productions, snapshots, presets, custom function files to/from an mc^2/Nova73 system.
- Transfer files to/from the **Shared Folder** so that files can be accessed by your host operating system.

1. Select **Page** -> **Production** -> **File Transfer** to open the display:

The display is divided into two halves:

- **Local Control System** - on the left you are *always* viewing files or directories on the mxGUI computer.
- **Online Control System** - on the right you can view files or directories on any online system plus the shared folder (host operating system shared folder).

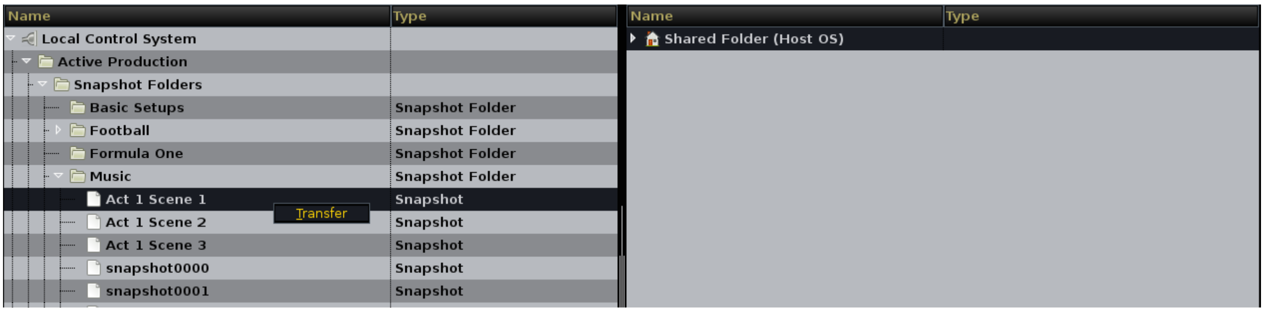
Note that the **Shared Folder (Host OS)** is represented on both sides of the display so that it can accept files from the local control system (your offline mxGUI) or an online system.

In the example above we are connected to a mc^{2}66 control system (online). Note that if mxGUI is offline, then the only folder on the right of the display is the **Shared Folder**.

The method of operation is very similar to the File display on an mc² console:

1. Open or close directories by double-clicking on the directory name (or click on the arrow beside the directory name).

2. Having selected a source and a valid destination, right-click on the source file to select **Transfer**:



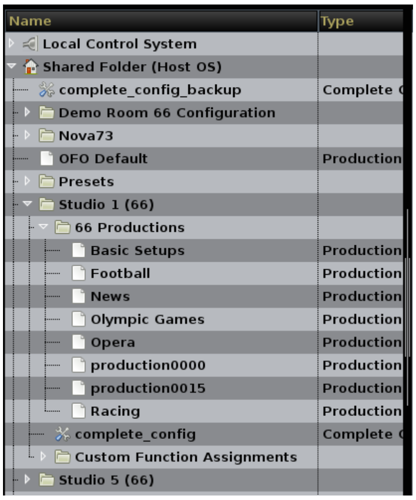
Note that:

- Files can be transferred from left to right or right to left.
- Each file or folder is clearly marked with its **Type** – e.g. production, snapshot, channel preset, EQ preset, etc. This is important as files can only be transferred to a valid destination. For example, you cannot transfer a snapshot into the Automation directory!
- For safety reasons you cannot delete productions, snapshots, configuration files, etc. from the **File Transfer** display.

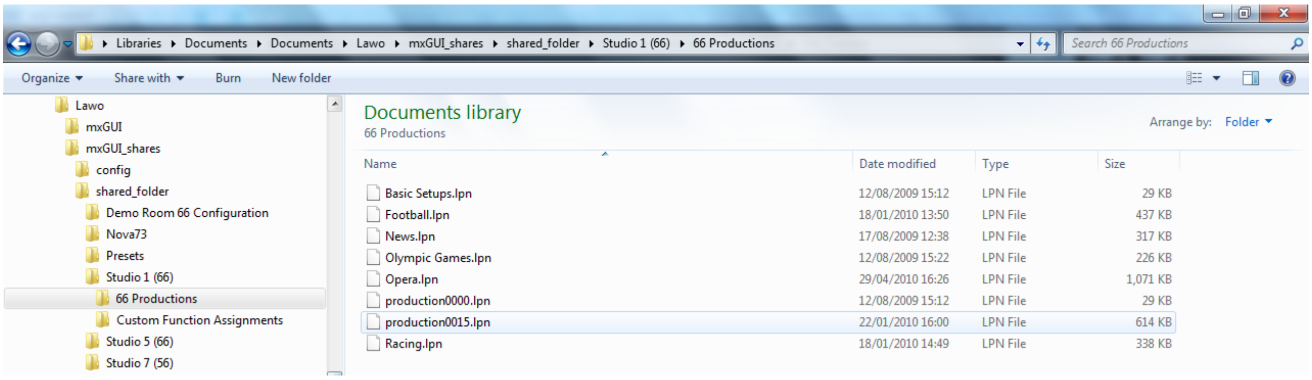
The Shared Folder

The contents of the **Shared Folder** can be accessed from the File Transfer display, and outside mxGUI by your host operating system. You may use the **Shared Folder** to organise files or transfer files externally (e.g. to USB or email):

The Shared Folder (in mxGUI):



The Shared Folder (in Windows Explorer):

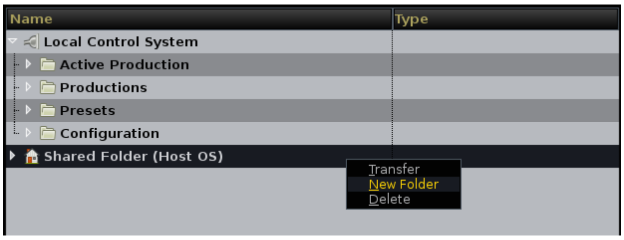


mxGUI Operations

Within mxGUI, you may use the File Transfer display to perform some basic file management tasks:

Creating a Sub Folder

1. Right-click on the **Shared Folder** and select **New Folder**:



A new folder is added with a generic name.

2. Type to rename the folder.

You can create folders within folders simply by right-clicking on the sub folder name.

Deleting Files or Folders

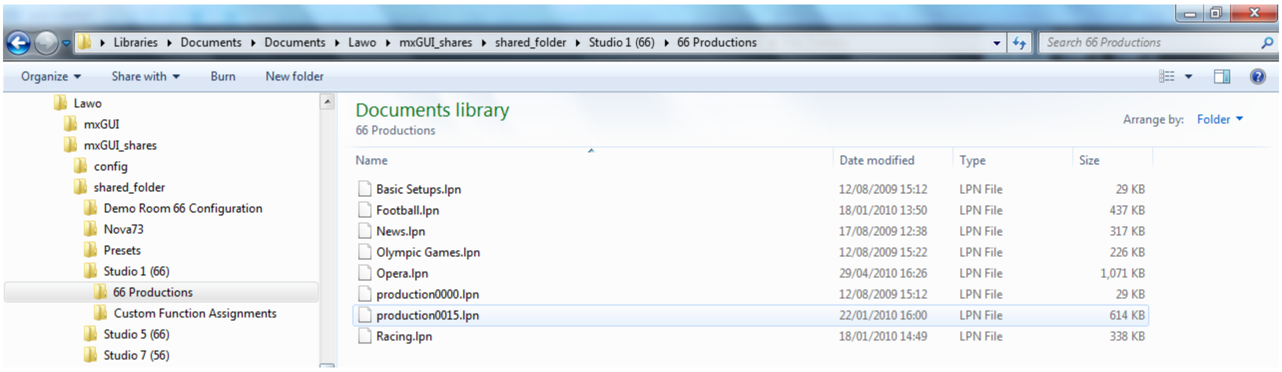
1. Right-click on the file or sub folder and select **Delete**.
2. Select **OK** to confirm.

The file or folder is deleted from the Shared Folder.

Host OS Operations

You may use your host operating software to perform any of the usual file management tasks: create folder, move or delete files, etc.

For more details on locating the **Shared_Folder**, see Software Installation: User Defined Folder. The default location for a Windows 7 installation is shown below:



Note that each file has a specific file extension which should not be modified, otherwise the file cannot be loaded by the Lawo system. For example, **.lpn** is the file extension for all zipped productions. See File Types & Extensions.

The Strip Assign Display

This display provides an overview of channel and main fader strip assignments. It may be used with the [Access/Assign](#) pop-up window to change fader strip assignments, change bus routing, copy audio parameters, etc. In addition, you can control fader levels and enter user labels from this display.

1. Select **Page** -> **Strip Assign** to open the **Strip Assign** display.

This display represents the physical fader strips on the surface of the console:

2. To view the whole of the surface use the left and right scroll bars at the bottom of the display.

> The size of the surface is defined by the console configuration. When working online, this data is read directly from the online control system. However, if you are operating offline, the surface size is dependent on the [console configuration](#) stored on the local control system (the console_config.tcl file).

On each 'fader strip' you will see:

- **STRIP N** – identifies the fader strip. Note that main fader strips are marked as **MAIN N**.
- **Layer 2 Controls**:
  - o **SEL LAY 2** – the select (**SEL**) button.
  - o **Channel Name** – always displayed.
  - o **Channel Name/User Label/Source Label** – this label is switched by the **LABEL** buttons on the Access/Assign pop-up window.
  - o **Main Level (dB)** – the fader level.
- **Layer 1 Controls** – as above but for layer 1.

Note that the empty grey area which separates layer 1 and layer 2 controls is deliberate; this may be used for new features in a future release of software!

3. Press a **SEL** button to select a fader strip.

The channel in access updates.

4. Click on the main level (dB) to adjust a fader level.

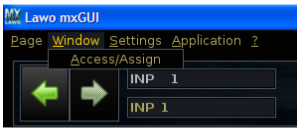
You can use the mouse wheel (if you have one) to adjust fader level or type in a value or use the up/down arrows.

On its own, these are the only operations which can be performed from the **Strip Assign** display. However, if you open the Access/Assign pop-up window, then you can use the **SEL** buttons to perform assignment operations such as fader strip assignments, bus assignments, copy/ reset audio parameters, etc.

The Access/Assign Window

The Access/Assign window is a pop-up window that replicates *all* the access and assignment functions found on the front panel of a mc^2 console.

1. Select **Window** -> **Access/Assign**:



The window opens, and may be moved to any position above another display, or minimised until needed:

2. Use the on-screen controls in exactly the same way as the console front panel:

- **ACCESS** – selects the channel in access, see ACCESS CHANNEL/ASSIGN.
- **COPY AUDIO** – used to copy or reset audio parameters, see Copy & Reset.
- **LABEL** – switches the fader strip Labels.
- **LINK** – used to create link groups or couple groups.
- **Bus Assign - Fader** – makes bus/VCA assignments, see Bus Assign.
- **BANK** and **LAYER** - switches Banks and Layers.
- **STRIP ASSIGNMENT** - assigns channels to fader strips, see Fader Strip Assignment.
- **SEL** – replicates the SEL buttons for each audio module (EQ, Gate, Compressor, etc.) within the Central Control Section, see Selecting Channel Parameters.

Chapter 11: Lawo Remote App

Introduction

This chapter covers the **Lawo Remote App**, a free App which allows you to operate any fader of a mc² console, recall snapshots and control user-defined functions remotely from an iPhone, iPod or iPad.

Topics covered in this chapter are:

- Installation & Configuration
- Starting the Lawo Remote App
- Configuring a New Connection
- Connecting to the System
- Controlling Parameters
- Disconnecting from the System

Installation & Configuration

Installing the Lawo Remote App

The Lawo Remote App can be downloaded, for free, from the App store, and installed on an iPhone, iPod or iPad. Download and install the App on your device in the usual manner.

Configuring the Network

The remote device communicates with the mc² or Nova73 control system via WLAN (Wireless Local Area Network).

To use the Lawo Remote App you must have a properly configured wireless network access point and know the IP address of your control system. There are several configuration options depending on your network infrastructure, so please consult your network administrator or refer to the technical document "TD_AccessPoint.iApp" for details.

You can check the IP address of the mc²56 control system from the **Signal Settings** display, see System Tree Structure.

Enabling App Control

To prevent unauthorised control, remote access must be enabled using the Safe Mode option in the **System Settings** display.

1. Select the **Remote** Topic and make sure that the **Safe Mode** option is unchecked.

The console may now be controlled from a Lawo Remote App device.

> There is no limit on the number of clients. However, if more than one device sets a parameter, the last change wins!

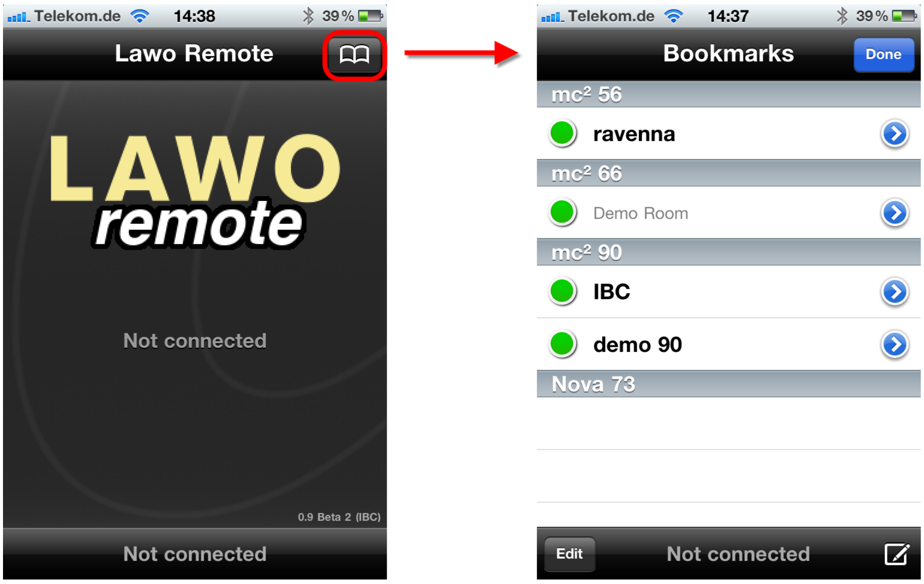
Starting the Lawo Remote App

1. On your device, open the Lawo Remote App.

The Lawo Remote welcome page appears showing the status of the existing connection – in our example, “Not connected”.

2. Touch the Bookmark icon at the top right of the display.

*The **Bookmarks** page opens listing all configured connections:*



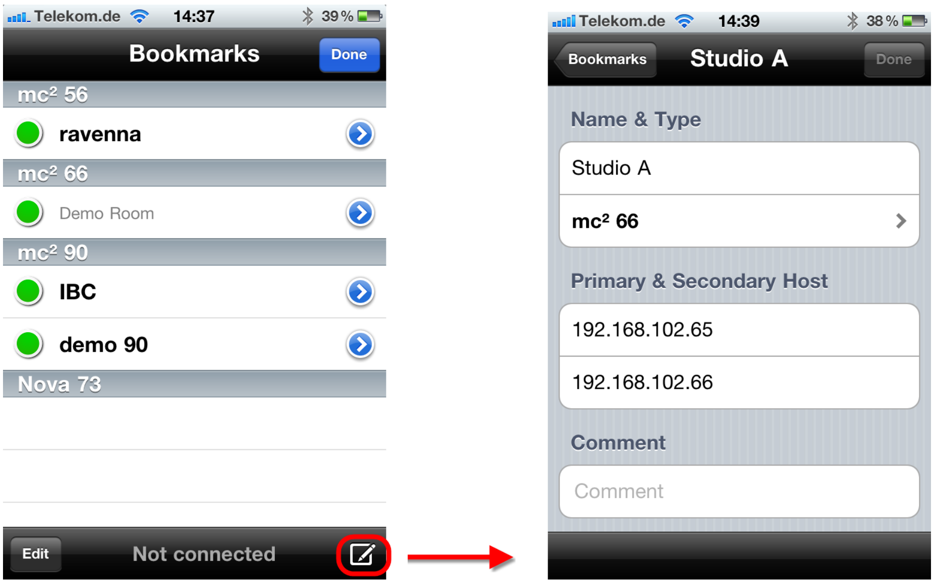
The list will be empty if no connections have been configured.

Configuring a New Connection

To configure a new connection:

1. Select the New Bookmark icon at the lower right of the display.

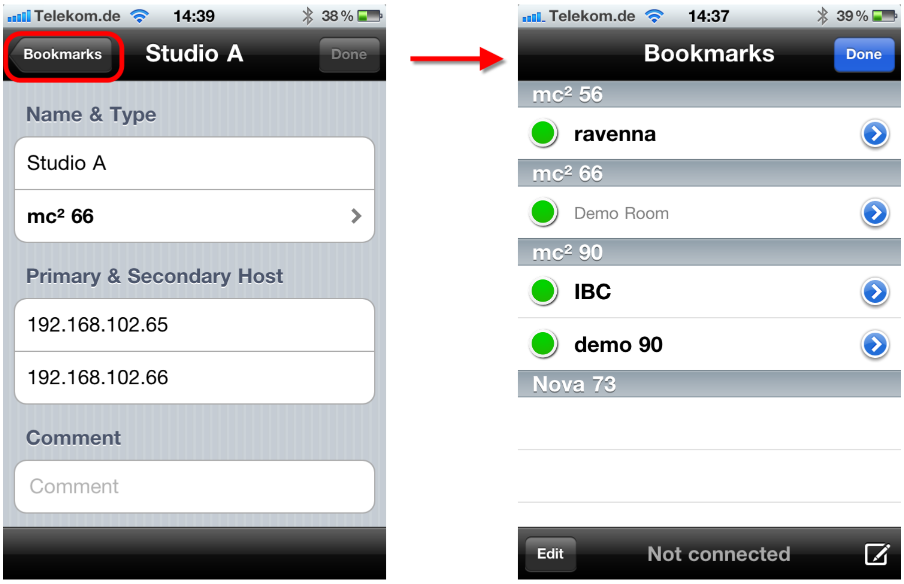
The 'New Bookmark' page opens, and a new connection is automatically configured:



2. Touch each entry to edit the:

- **System Name** – e.g. **Studio A**. This name is used for reference within the Lawo Remote App.
- **System Type** – e.g. **mc²66**. Choose from the list of supported systems. The system type must match that of the system you wish to connect to.
- **Primary & Secondary Host** – enter the IP address of the control system you wish to connect to. If you have a redundant control system, then you will need to enter the primary and secondary IP addresses. (You can check the IP address of the mc² control system from the **Signal Settings** display, see System Tree Structure.)
- **Comment** - enter a Comment if you wish.

3. When you have completed each field, touch **Bookmarks** to return to the Bookmarks page.



The name of your new connection appears in the list.

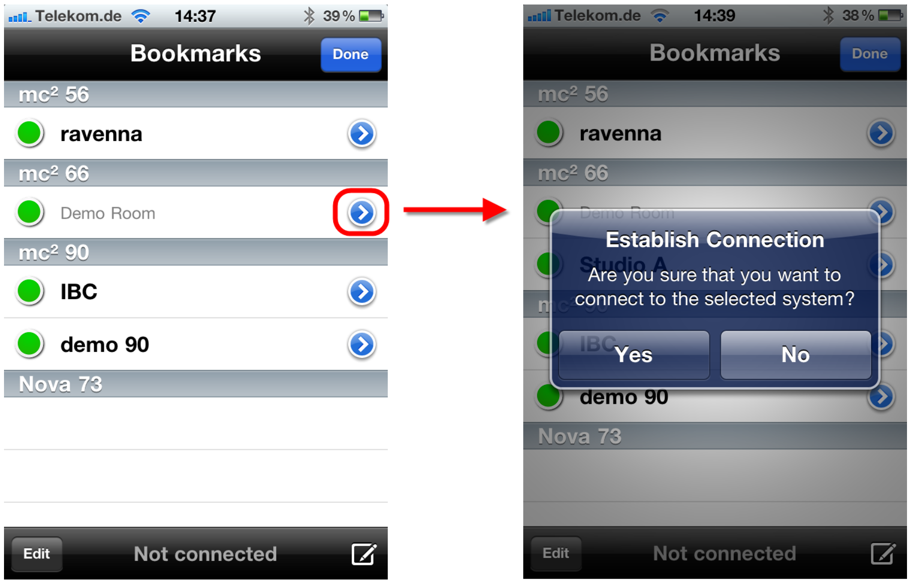
4. At any time you can edit an existing connection, by touching the **Edit** button at the lower left of the **Bookmarks** page.

Connecting to the System

You can connect to any system configured within the **Bookmarks** page. You may only connect to one system at a time.

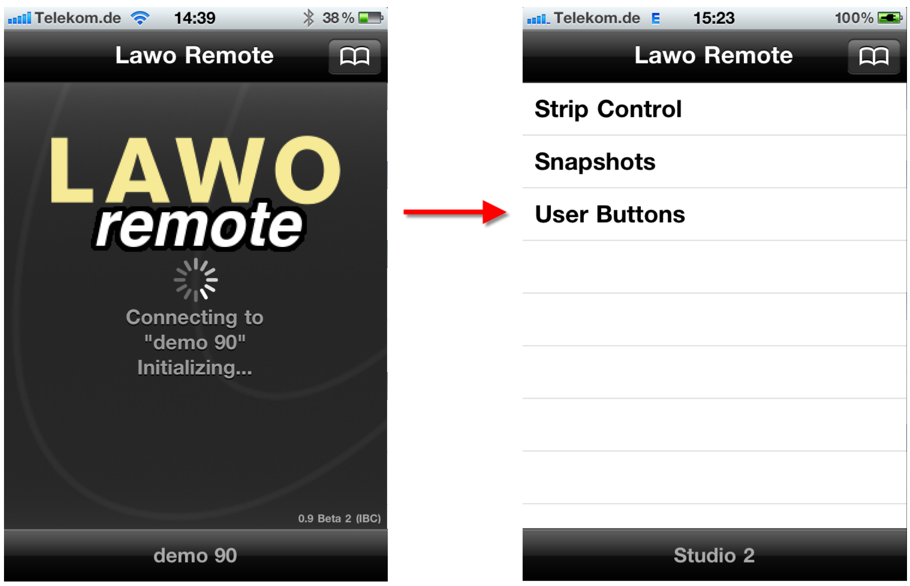
1. Open the **Bookmarks** page, and touch the arrow beside the name of the system you wish to connect to.

The following confirmation dialogue appears:



2. Touch **Yes** to continue.

The device attempts to connect. If successful, the operational menus appear and the connection status, at the bottom of the display, updates to show the system name:



If the connection fails, then an error will appear. Check the system type and IP settings from the **Bookmarks** page. Check that the iPhone, iPod or iPad is connected to the correct WLAN. If the connection still fails, then there is a problem with your network or its configuration. Please contact your network administrator for assistance.

Controlling Parameters

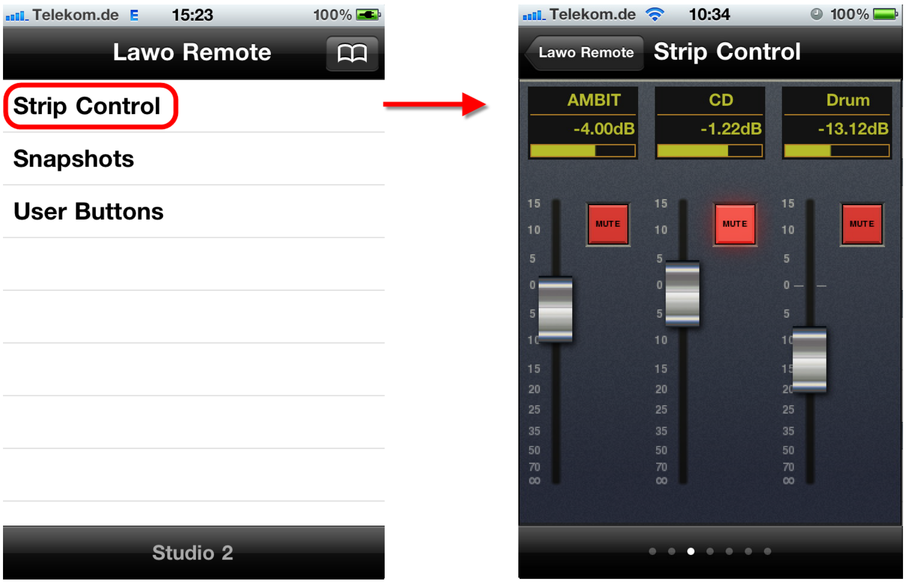
The main operational menus appear once you have an active connection to the mc²56 system.

From the Lawo Remote App you have access to:

- Strip Control – fader level, mute and metering for any fader assigned to the active Bank and Layer.
- Snapshots – load any Snapshot from any folder within the active production.
- User Buttons – a special page of buttons allow you to control user defined functions such as monitoring, GPI control, etc. The button assignments are made from the Custom Functions display and stored as part of the configuration.

Strip Control

1. Select **Strip Control** to control the fader level and mute for any fader assigned to the active Bank and Layer:



You will see the label and level in dB for three fader strips at a time.

2. Touch and drag up or down on a fader to adjust the fader level.
3. Touch the **MUTE** button to mute or unmute the channel.

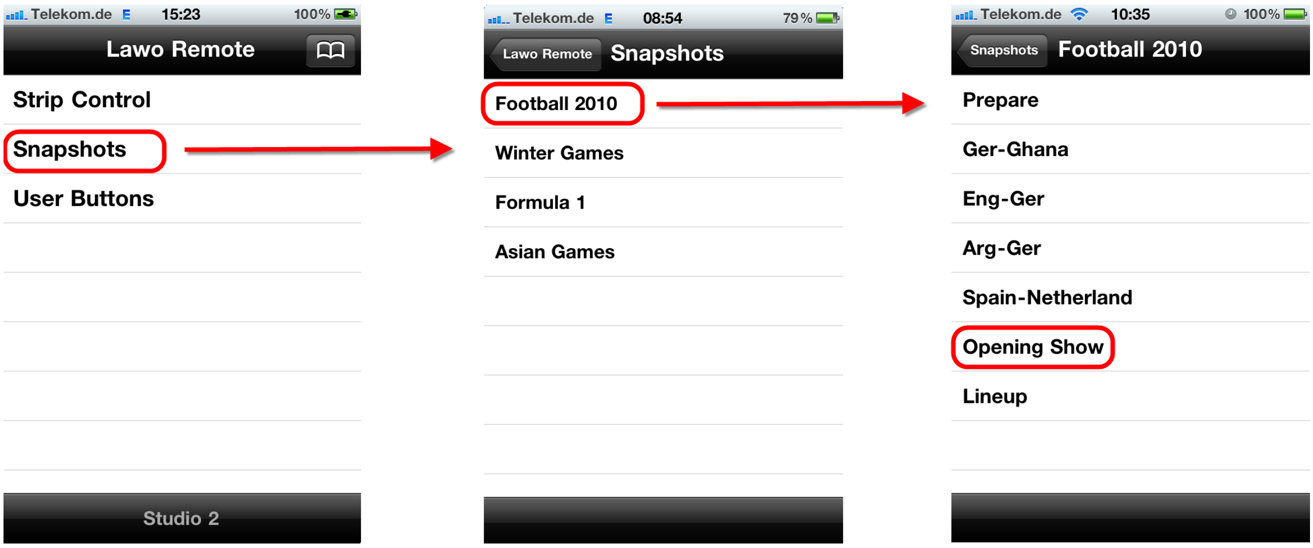
Any changes are reflected on the console control surface.

4. Touch and drag to the left or right to scroll across the fader bay.
5. Touch one of the dots at the bottom of the page to access a different fader bay. Each dot represents a fader bay (of 8 strips). The number of dots depends on the size of the control surface.
6. To return to the main menus, select the **Lawo Remote** button (top left).

Note that you cannot change Banks or Layers from the Lawo Remote App.

Snapshots

1. Select **Snapshots** to load a Snapshot from any folder within the active production:



2. Select the folder:
3. Then select the snapshot followed by **Load**.

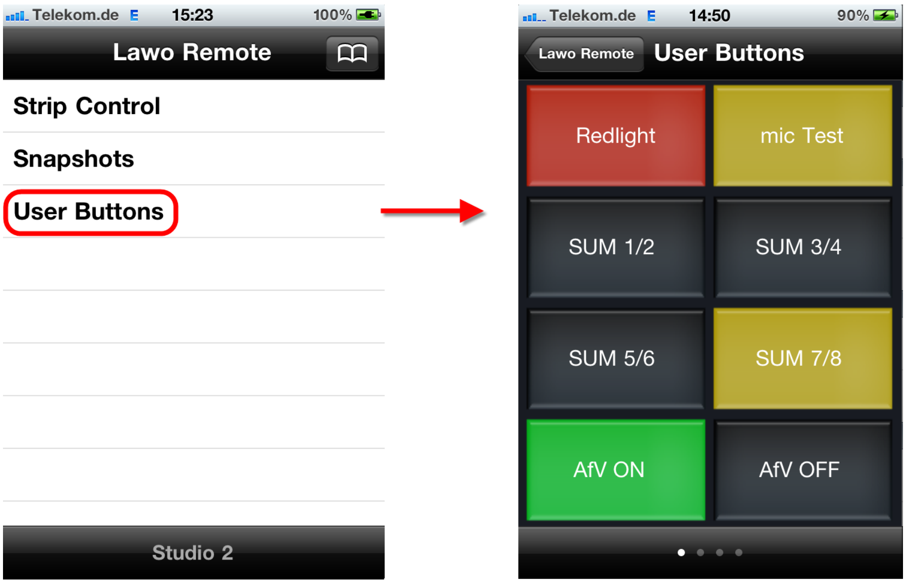
The snapshot is loaded to the console. If any [snapshot isolates] are active, then these are applied.

4. To return to the main menus, select **Snapshots** followed by the **Lawo Remote** button (top left).

Note that you cannot save or update snapshots from the Lawo Remote App, or change production.

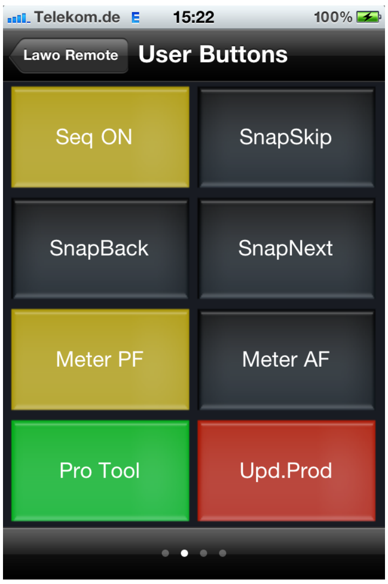
User Buttons

1. Select **User Buttons** to access a special page of buttons designed for monitoring functions such as source selection and monitor level control:



2. Touch a button to action its function.

3. Touch one of the dots at the bottom of the page to access a different page of functions:



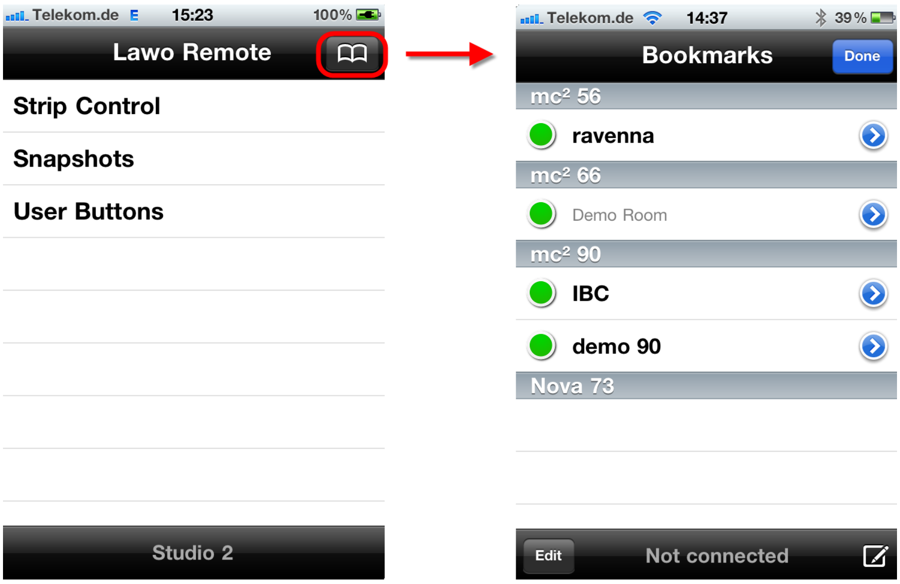
4. To return to the main menus, select the **Lawo Remote** button (top left).

Note that the Lawo Remote user buttons are assigned from the Custom Functions display, and therefore may vary when you connect to a different console.

Disconnecting from the System

1. To disconnect from the system, select the Bookmarks icon at the top right of the display.

*The **Bookmarks** page re-opens and the connection status, at the bottom of the display, updates to “Not connected”:*



Trouble-shooting

Introduction

This section includes a series of example problems and tips to help you fault find the mc²56 system.

For further assistance, please contact your local Lawo representative or the service department at service@lawo.de.

- The system will not boot or does not boot correctly
- The system boots up but I have no audio
- The complete control surface is not responding
- One of the control surface panels is not working
- The graphics on the TFT displays freeze temporarily
- The graphics on one of the displays freezes or looks odd
- The network connection between my computer and the control system is not working
- Running a PING test (to check network communcation)

The system will not boot or does not boot correctly

1. Power off the Nova73 and wait for the system to shutdown.

The system has completed its shutdown when the blue LED of the trackball is off.

2. Power on to try a warm start.

3. If this is unsuccessful perform a cold start.

If the system now boots correctly, then your warm start user data is corrupt. Check your production data by loading a production. If this is the cause of the problem, perform another cold start and try a different production. If there is a problem with all production data, then you may need to replace the Data Flashcard.

4. If this is still unsuccessful, then you should try replacing the System Flashcard with a backup copy.

The system boots up but I have no audio

1. Check the System Settings display to see if there any reported errors.

If a Nova73 module or DALLIS I/O card is shown in red, then there is a problem with the connection or module/card.

2. Check the connections between the Nova73 I/O module and any DALLIS units.

Are the fibres reversed?

3. Check that all the **ACTIVE** LEDs on modules within the Nova73, and cards within the DALLIS, are green and flashing synchronously.

The **ACTIVE** LED on each Nova73 module, or DALLIS card, should blink in time with all other **ACTIVE** LEDs (at approximately 100Hz). This shows that the card is synchronous to the rest of the system. If an LED is out of sync, then check that the card is fitted correctly, and if the symptom persists, replace the card.

4. If everything still looks ok, then try reloading the DSP configuration from the DSP Configurations display.

The complete control surface is not responding

1. Check the Ethernet A connections between the control surface and Nova73 Router Module.

2. If main and redundant Router Modules are fitted to the Nova73, try forcing a manual takeover to the redundant control system.

3. If not, power off the Nova73 and wait for the system to shutdown. And power on to try a warm start.

One of the control surface panels is not working

1. Try restarting the Ethernet Bay Server.
2. Carefully remove the panel, and check the connections.
3. Try disconnecting and reconnecting the USB and power connectors to the panel.

Try this a few times to see if the panel will boot. If not, then the panel may be faulty so please contact your local service representative.

The graphics on the TFT displays freeze temporarily

This may occur if the load on the CPU exceeds 95% - for example, during a production load. Audio processing is unaffected, and therefore the behaviour should be ignored. Once the production has loaded, and the CPU returns to normal levels of operation, all graphics should update correctly.

The graphics on one of the displays freezes or looks odd

This may occur if a Bay Server looses its Ethernet connection to the Control System.

1. Try restarting the Ethernet Bay Server.

If the problem persists, then the display or Bay Server may be faulty so please contact your local service representative.

The network connection to the control system is not working

If you cannot establish network communication between your computer and the control system:

1. Check the network connection and TCP/IP settings of your computer's network interface card.

2. If applicable, check that the software you are running is compatible with the mc^2 system. When connecting from mxGUI or AdminHD, the first three digits of the software versions *must* match.

3. Try a PING command to test whether you have a valid network connection:

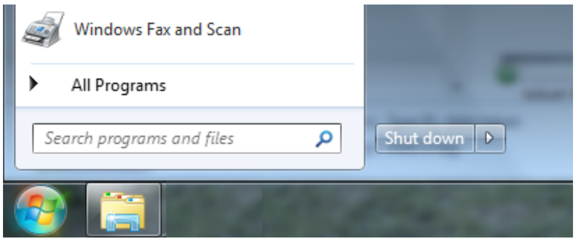
- If the ping test fails, then there is something wrong with your network configuration.
- If the ping test is successful, then this confirms that the network communication is working. If you still cannot connect to the mc^{2}56 control system, then something on your computer is blocking the network connection. Try disabling any firewall and/or antivirus software.

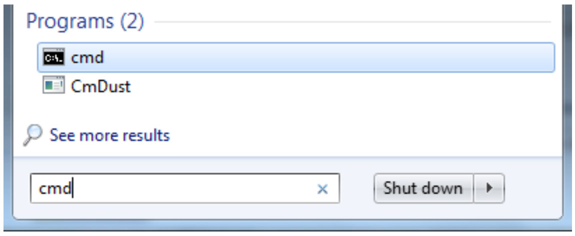
Running a PING test

The PING command is a built-in Windows and Mac function, that allows you to test whether you have a valid network connection to and from any networked device.

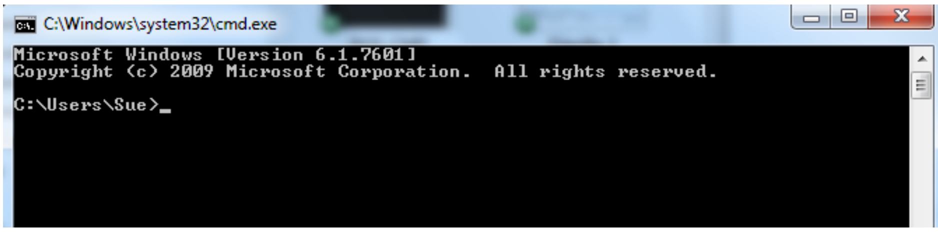
1. Make sure that your computer is connected to the Lawo system network, and that you have configured the TCP/IP settings of your computer's network interface card.

2. On a Windows 7 PC, type **cmd** into the "*Search programs and files*" field under the **Start** menu and press Enter.





This opens the DOS command prompt window:



Alternatively, on a Mac, open the **Terminal** program (found in the **Applications** -> **Utilities** folder).

3. On both platforms, perform the ping test as follows:

Type **ping xxx.xxx.xxx.xx** (where **xx..** is the IP address of the device you are trying to connect to) and press Enter.

You can check the IP address of your control system from the console GUI (using the Signal Settings display). See also TCP/IP Addresses for a list of the default IP addresses for different Lawo products.

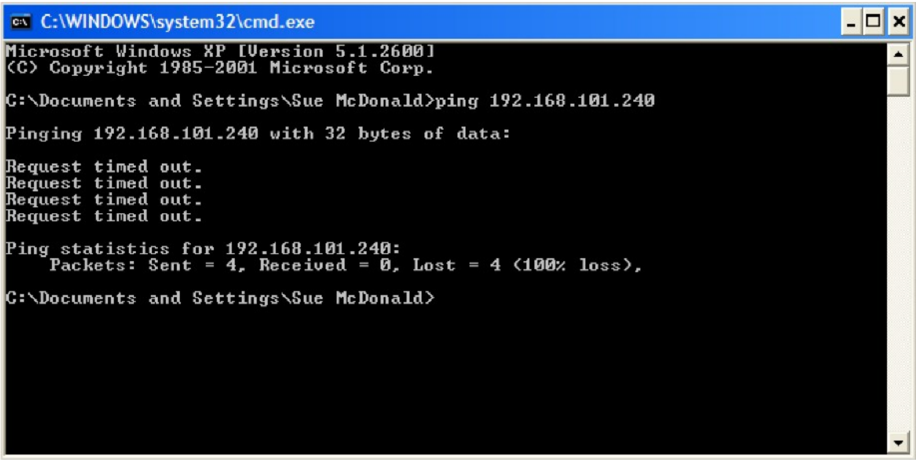
For example, to test the connection to a mc²56 with a default IP address, you would type:

ping 192.168.102.56

Your computer will now try to establish communication...

➢ Ping Test Fail

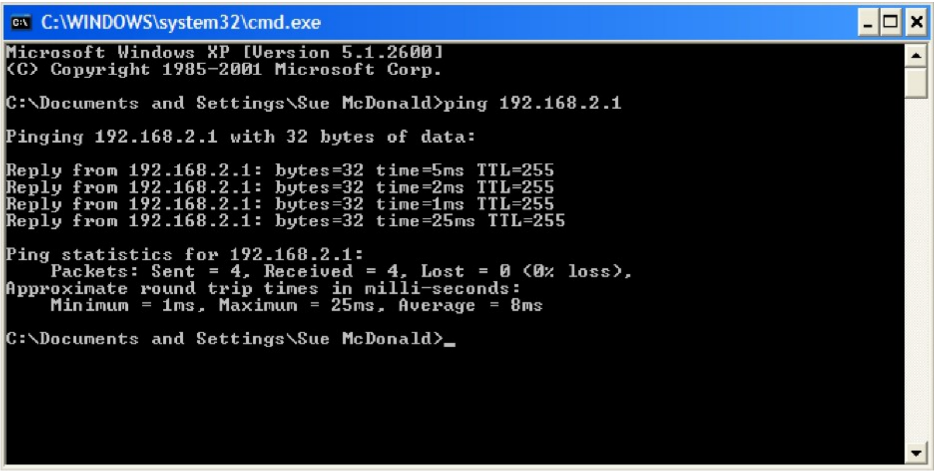
If the ping test fails, then the request will time out, and you will not receive any successful packets:



There is something wrong with your network configuration, so check the network connections, and TCP/IP settings again. Or contact your network administrator.

➢ Ping Test Success

If the ping test is successful, then the result will show that the Sent packets have been successfully Received:



This confirms that the network communication is working. If you still cannot connect to the mc²56 control system, then something on your computer is blocking the network connection. Try disabling any firewall and/or antivirus software.

Technical Data

Control Surface

- Frame widths from 16 to 80 faders.
- Remote stand-alone extender frames from 16 faders.
- 6 banks each with 2 layers.
- 100mm fader + two freely adjustable rotary controls (Free Controls) + Input-Gain rotary control + Channel GUI display for each fader.
- TFT display: mono, stereo or up to 7.1 metering + bus assignment + gain reduction for dynamics + AfV status + VCA assignment + Mix Minus bus assign.
- External display of GUI pages (via mxGUI), e.g. metering.
- Fader colour coding, reset via snapshots.
- Fader notch and PFL overpress (Backstop PFL).
- 12 channel user buttons.
- 9 central user buttons.
- Optional: one integrated user panel (automation, 40 user buttons, reveal fader surround or intercom), RTW goniometer integration, script tray.

Signal Processing

- 888 channels + 144 summing busses, 40bit floating point.*
- Up to 760 inputs with A/B input, up to 64 sub-groups, 32 aux sends, up to 96 track busses, up to 48 main sums.*
- Rapid switching of channel and bus to mono/stereo/surround.
- Up to 96 surround masters + 128 VCA groups with metering + 256 GPCs (General Purpose Channels).
- Surround formats: DTS/Dolby ® Digital 5.1, Dolby ® Prologic 4.0, DTS ES/Dolby ® EX 6.1, SDDS 7.1 or DTS-HD 7.1. Diverse panning characteristics + surround aux bus.
- 2 AFL: 1 surround 8-channel + 1 stereo.
- 2 PFL, both stereo.
- Audio-follow-Video with 128 events, controlled via RemoteMNOPL, GPI or matrix connection. Envelope adjustable up to 10s fade time.
- Solo in place (enabled/disabled from System Settings display).
- Permanent input meter beside fader + adjustable INPUT, PF, AF, DIROUT meter point in Channel display.
- Loudness Metering according to EBU R128 and ATSC A/85, momentary or short term in every channel, integrated measurement on sum channels with display of integrated LUFS value in the Central GUI headline.

- Processing Modules**: INMIX with MS decoder, digital amp (DAMP), 2-band fully parametric FILTER, 4-band fully parametric EQ, 2-band fully parametric side chain filter (SCF), INSERT, DELAY up to 1800ms (switchable to metres, milliseconds or frames), 4 independent dynamic modules: EXPANDER, GATE, COMPRESSOR, LIMITER, stereo IMAGE, METER point, and DIRECT OUT.
- AMBIT Upmix, available on every 5.1 channel, fully Downmix compatible.
- Inline configuration, with per channel or global send/return switching.
- Fully-equipped surround channel with coupling of all channel parameters and hyperpanning.

* Figures are for a Nova73 HD fitted with 8 DSP boards running at 48kHz. At higher sample rates, the number of channels and summing busses is halved.

** The processing modules listed are for a Recording channel. Broadcast channels offer less processing modules in return for twice as many channels.

See Channel Types for more on the differences between Recording and Broadcast channels.

Routing Matrix

- Up to 8192 cross points, non-blocking.
- Up to 96kHz, 24-bit. (Higher sample rate operation is defined by AdminHD).
- Fully redundant signal path.
- Level adjustment for all inputs and outputs.
- Downmix from surround (up to 7.1) to stereo.
- Integrated monitoring devices for remote locations, e.g. director's room.
- Full networking of up to 16 systems, share and import sources and destinations, studio intervention.
- Full snapshot and production portability independent of matrix and DSP size.
- Level control for every input and output.

Plug-in Server

- Full VST plug-in integration with storage of plug-in parameters in snapshots and production data.

Static and Dynamic Automation

- Snapshot automation.
- Sequence automation with trim and cross fade.
- Dynamic timecode-based automation for all parameters including bus assign. Modes include touch, glide, join, punch in/out and absolute/trim (including trim "on the fly").
- Offline timecode automation editing (copy, paste, delete, cut, insert, etc.)

Interfaces

- Mic/Line, Line Out, AES, 3G SDI, MADI, ATM, GPIO, Serial, MIDI. (For details, see DALLIS product information.)
- Stereo and surround monitoring systems.
- Local I/O within console frame: 16 Line I/O, 8 AES I/O, 8 GPIO, 2 Headphones.

Synchronisation

- 2 redundant sync inputs with automatic Blackburst, Wordclock, AES 3, MADI detection.

Redundancy

- Redundant PSUs (standard).
- Redundant DSP board (defined in the **DSP Configurations** display).
- Optional Redundant Router card provides redundant routing matrix and control system, exchangeable during run time, with full data redundancy.
- Optional Redundant DALLIS cards offer fully redundant signal paths.

Control Features

- Bay Isolate (ISO BAY) with separate bank and layer switching + second PFL/AFL bus.
- Global A/B input switching.
- Enhanced mix-minus control with independent off-air conference.
- Direct out mute by fader.
- Fader control of all level parameters.
- Diverse tally and fader start modes.
- Program switch.
- Machine control.
- Audio-follow-Video, up to 128 camera tallies, Ethernet or GPI controlled.
- Extensive talkback system integration.
- Camera mic remote via GPI or voltage control.
- Remote desktop access from TFT to external computer.
- mxGUI: remote control of GUI pages via external computer.
- iPhone App: remote control of fader level, monitoring and snapshots.

Remote Maintenance

- Connection via Internet remote software.
- Software updates, error diagnostics, remote assistance.

External Control Systems

- Remote control of all routing parameters via network.
- Remote control of monitoring units in remote locations.
- Remote control of integrated matrix monitoring units.
- Remote control of signal parameters such as SDI, silence detects, mic gain, etc. (for details, see the Remote MNOPL documentation).
- Online configuration with AdminHD, graphical configuration of Nova73 and DALLIS components.
- External matrix controllers: VSM, Evertz, Quartz, BFE, Pharos, and others.
- EmBER+ control protocol (available 2013).

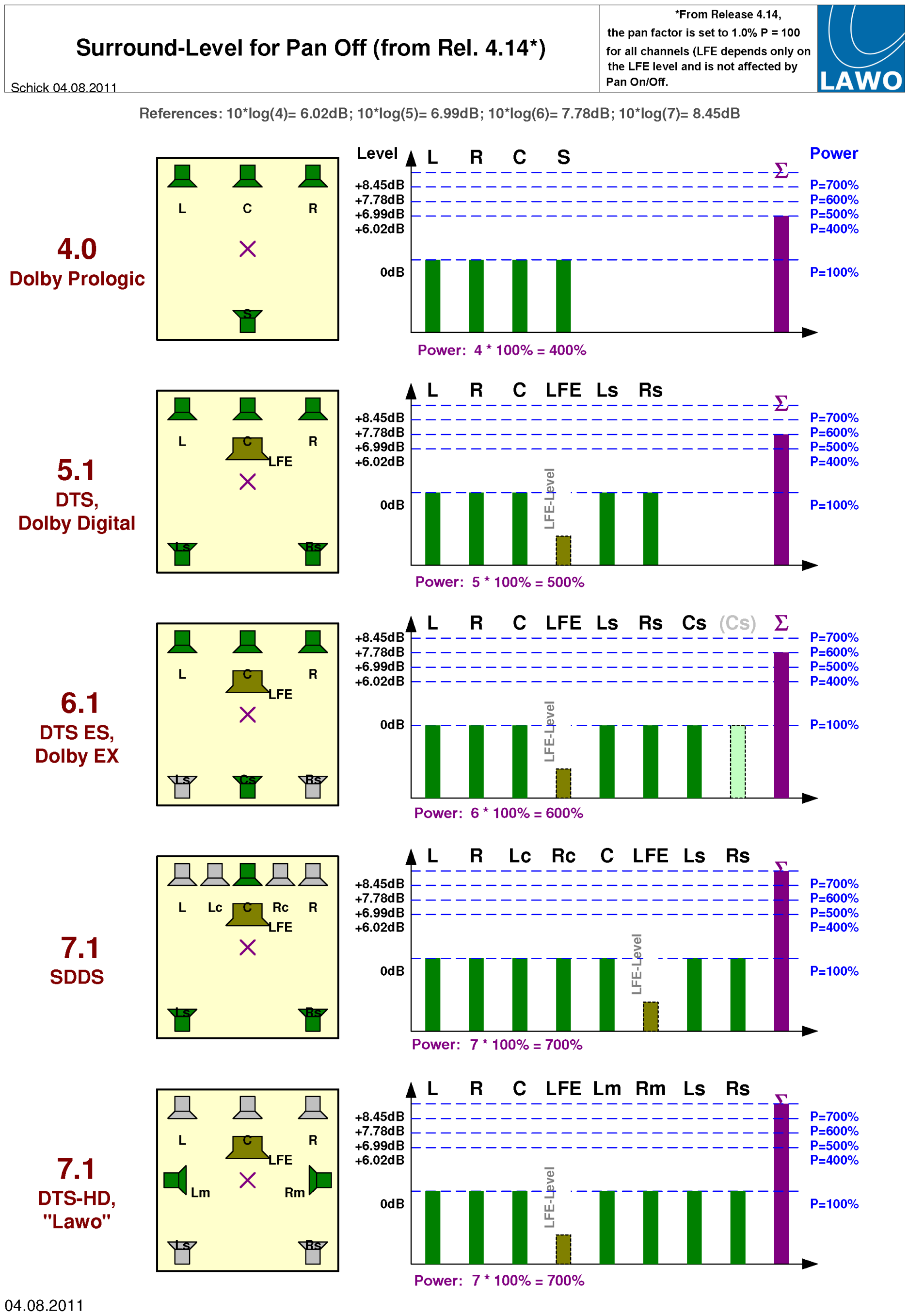
Appendices

This section includes the following appendices:

- Surround Levels
- Pan Slope
- Digital Output Settings
- VCA, Surround, Link and Couple Masters
- User Button Numbering
- Local I/O Wiring
- DSP Configurations
- SDI Parameters
- Control System Locations
- TCP/IP Addresses

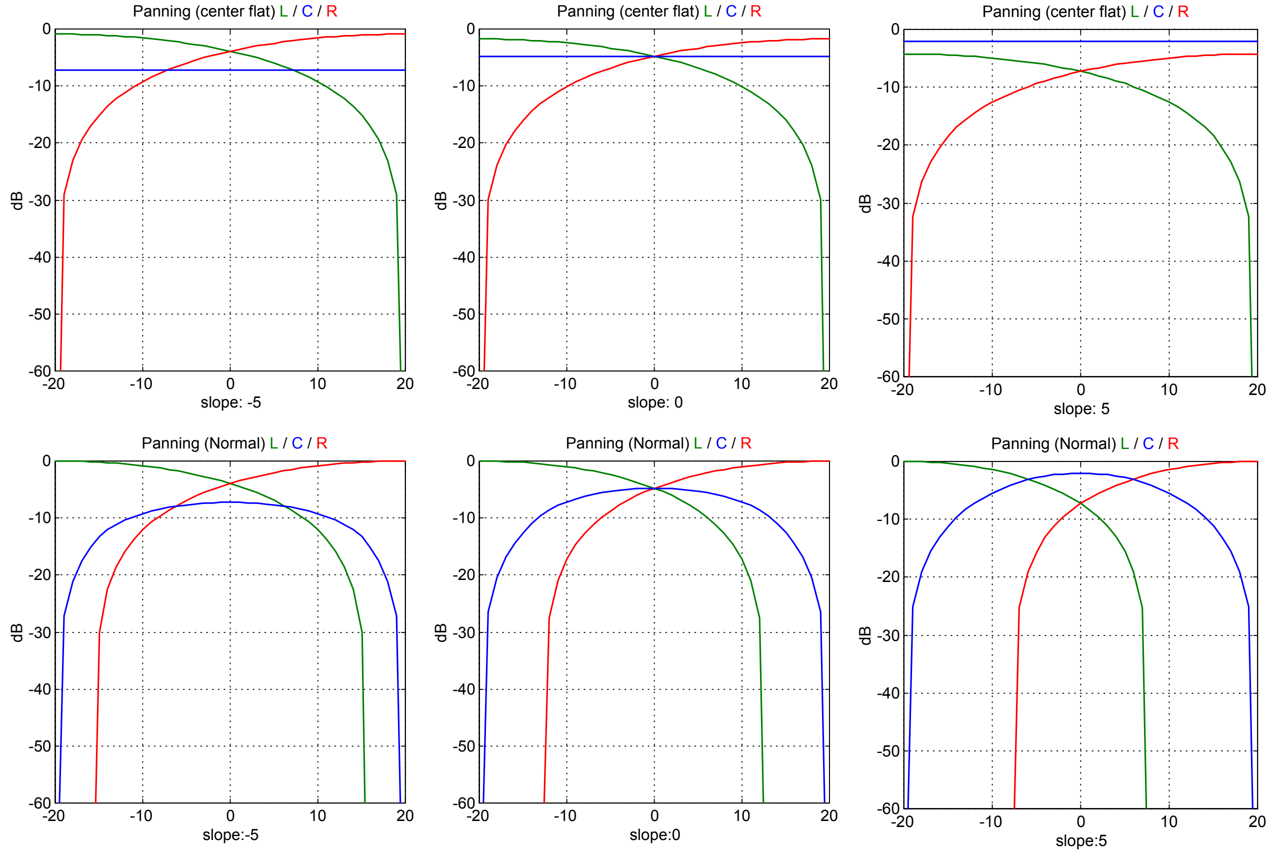
Surround Levels

The following diagrams show the power output to each channel, when panning is off, for each of the console's surround formats. Note that the power factor changed in Version 4.14 software, so that 100% of the source feeds all channels, except the LFE, to make it easier to measure your loudspeaker and metering systems.



Pan Slope

The following diagrams demonstrate how the slope control affects Left-Centre-Right panning:



Digital Output Settings

For each digital output, sample rate conversion and dither are applied automatically depending on your choice of **sample rate** and **word length** from the Signal Settings display.

The following table explains the results of each clock selection and word length combination:

Clock Selection	Word Length Selection	SRC on/off	Dither Status
System	24-bit	SRC off	Off (Truncate)
System	20-bit	SRC on	Dither on
System	16-bit	SRC on	Dither on
44.1kHz	24-bit	SRC on	Off (Truncate)
44.1kHz	20-bit	SRC on	Dither on
44.1kHz	16-bit	SRC on	Dither on
48kHz	24-bit	SRC on	Off (Truncate)
48kHz	20-bit	SRC on	Dither on
48kHz	16-bit	SRC on	Dither on
Follow Input	24-bit	SRC on	Off (Truncate)
Follow Input	20-bit	SRC on	Dither on
Follow Input	16-bit	SRC on	Dither on

VCA, Surround, Link and Couple Masters

The master/slave behaviour of VCA, Surround VCA, Link and Couple masters vary depending on the parameter and type. Firstly, each parameter behaves according to a mode:

- **Relative Control** – these parameters are controlled relatively, allowing you to offset slave positions.
- **Absolute Control** - these parameters are set by the master; any change is inherited by all slaves.
- **On Master** – for some switches the parameter may be switched ON from a master but not OFF. For example, you can use a VCA master MUTE button to mute all slaves, and then individually unmute slaves.
- **Off Master** – for other switches the parameter may be switched OFF by a master but not ON.

The following conditions affect what happens when channels are linked or unlinked:

- **Apply on Assign** – the parameter value of the master is added to the slave channel when a link is created.
- **Restore** – the parameter value of the master is subtracted from the slave channel when a link is removed.

Some special conditions apply to faders and AFV:

- **Relative Faders** – this condition sets whether slave faders are moving (Relative Fader OFF) or non-moving (Relative Fader ON).
- **Slave Controls Master** – this condition determines whether a change on a slave fader updates the master.
- **Invert** - for the Audio Follow Video enable parameter (ON function), the Invert condition inverts settings between the master and slave.

These conditions apply to Link groups:

- **Ignore Module Link** – this condition means that a parameter is linked as soon as a Link group is created, whether any modules are selected for linking or not.
- **Ignore Suspend** - this condition means that the parameter cannot be suspended from the link or couple.

Each mode and condition may differ between a VCA, a Surround VCA, a Link or the Couple master so please use the following tables to check the behaviour for specific parameters.

Remember that when using VCA grouping, slave faders can be moving or non-moving, defined by the Relative Slave faders option in the **System Settings** display. For all other group types (Surround VCAs, Links and the Couple group), slave faders always move.

VCA Link Table

VCA Link Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Isolate	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
DigiAmp level	RELATIVE	On	Off	Off	On	Off	Off	Off	Off
DigiAmp on	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Input gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Input balance	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Input phase revert left	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Input phase revert right	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Input stereo swap	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Input left to both	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Input right to both	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Input M/S matrix	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Input mono	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Insert send level	RELATIVE	On	Off	Off	On	Off	Off	Off	Off
Insert on	OFF_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Insert level bypass	OFF	On	Off	Off	Off	Off	Off	Off	Off
Insert soft clip	OFF	On	Off	Off	Off	Off	Off	Off	Off
Insert meter source	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 frequency	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 Q	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 on	OFF_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 slope	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 type	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 2 & 3 notch	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer on	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Filter /SCF gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF frequency	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off

VCA Link Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Filter/SCF Q	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF 1 & 2 on	OFF_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF slope	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF type	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF on	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Delay time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Delay on	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Image width	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Image position	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Image on	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Image style	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Direct out level	RELATIVE	On	On	On	On	Off	Off	Off	Off
Direct on	OFF_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Direct out level bypass	OFF	On	Off	Off	Off	Off	Off	Off	Off
Direct out soft clip	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Aux send level	RELATIVE	On	On	On	On	Off	Off	Off	Off
Aux send pan/ balance	RELATIVE	On	On	On	Off	Off	Off	Off	Off
Aux send on	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Aux send mix cue	OFF	On	Off	Off	Off	Off	Off	Off	Off
Aux send independent	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Metering position	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Direct out position	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Track path position	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Track switch	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Aux send position	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Channel module order	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Fader Level	RELATIVE	On	On	On	On	Off	Off	Off	Off

VCA Link Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Mute	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Left-right panning	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Front-back panning	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Pan slope	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
LFE level	RELATIVE	On	Off	Off	On	Off	Off	Off	Off
Hyperpan front width	ABSOLUTE	On	Off	Off	Off	On	Off	Off	Off
Hyperpan back width	ABSOLUTE	On	Off	Off	Off	On	Off	Off	Off
Hyperpan depth	ABSOLUTE	On	Off	Off	Off	On	Off	Off	Off
Hyperpan turn	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Pan on	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Pan mode center-flat	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan on	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan turn pre pan	ABSOLUTE	On	Off	Off	Off	On	Off	Off	Off
Direct out balance	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Busses	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
PFL	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AFL	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
DirOut mute by fader	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Coordination	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
Conference	ON_MASTER	On	Off	Off	Off	Off	Off	Off	Off
AfV on level	RELATIVE	On	Off	Off	On	Off	Off	Off	Off
AfV off level	RELATIVE	On	Off	Off	On	Off	Off	Off	Off
AfV attack time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV hold time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV release time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV event number	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AfV state	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AfV enable	OFF_MASTER	On	Off	Off	Off	Off	On	Off	Off
AfV event state	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AfV hold time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off

VCA Link Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
AfV max event time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Module Link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Channel source selection	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off

Surround Master Table

Note that analogue input gain is not linked by the Surround Master; only digital input gain for AES or MADI sources is linked when using a Surround Master.

Surround Master Table

Parameter	Mode	Slave Operation	Apply On Assign	Restore	Relative Fader	Slave Controls Master	Invert	Ignore Module Link	Ignore Suspend
Isolate	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
DigiAmp level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
DigiAmp on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Input gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Input balance	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Input phase revert left	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Input phase revert right	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Input stereo swap	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Input left to both	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Input right to both	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Input M/S matrix	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Input mono	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Insert send level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Insert on	ON_MASTER	On	On	On	Off	Off	Off	Off	Off
Insert level bypass	OFF	On	Off	Off	Off	Off	Off	Off	Off
Insert soft clip	OFF	On	Off	Off	Off	Off	Off	Off	Off
Insert meter source	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 frequency	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Equalizer 1 to 4 Q	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Equalizer 1 to 4 on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Equalizer 1 to 4 slope	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Equalizer 1 to 4 type	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Equalizer 2 & 3 notch	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Equalizer on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off

Surround Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF frequency	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Filter/SCF Q	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Filter/SCF 1 & 2 on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Filter/SCF slope	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Filter/SCF type	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Filter on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Expander threshold	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Expander gain	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Expander ratio	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Expander attack	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Expander hold	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Expander release	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Expander delay	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Expander on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Gate threshold	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Gate floor	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Gate hysteresis	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Gate attack	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Gate hold	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Gate release	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Gate delay	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Gate on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Gate SCF on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Gate external key on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Gate external key	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Compressor threshold	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Compressor gain	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Compressor ratio	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Compressor attack	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Compressor hold	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off

Surround Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Compressor release	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Compressor delay	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Compressor on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Compressor SCF on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Compressor external key on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Compressor external key	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Compressor soft knee	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Limiter threshold	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Limiter gain	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Limiter hysteresis	OFF	On	Off	Off	Off	Off	Off	Off	Off
Limiter attack	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Limiter hold	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Limiter release	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Limiter delay	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Limiter on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Limiter soft knee	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Delay time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Delay on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Image width	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Image position	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Image on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Image style	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Direct out level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Direct on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Direct out level bypass	OFF	On	Off	Off	Off	Off	Off	Off	Off
Direct out soft clip	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Aux send level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off

Surround Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Aux send pan/ balance	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Aux send on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Aux send mix cue	OFF	On	Off	Off	Off	Off	Off	Off	Off
Aux send independent	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Metering position	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Direct out position	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Track path position	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Track switch	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Aux send position	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Channel module order	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Fader Level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Mute	ON_MASTER	On	On	On	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Left-right panning	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Front-back panning	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Pan slope	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
LFE level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan front width	ABSOLUTE	On	On	Off	Off	On	Off	Off	Off
Hyperpan back width	ABSOLUTE	On	On	Off	Off	On	Off	Off	Off
Hyperpan depth	ABSOLUTE	On	On	Off	Off	On	Off	Off	Off
Hyperpan turn	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Pan on	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Pan mode center-flat	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Hyperpan on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan turn pre pan	ABSOLUTE	On	On	On	Off	On	Off	Off	Off
Direct out balance	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Busses	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off

Surround Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
PFL	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AFL	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
DirOut mute by fader	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Coordination	ON_MASTER	On	On	On	Off	Off	Off	Off	Off
Conference	ON_MASTER	On	On	On	Off	Off	Off	Off	Off
AfV on level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV off level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV attack time	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
AfV hold time	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
AfV release time	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
AfV event number	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
AfV state	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
AfV enable	OFF_MASTER	On	Off	On	Off	Off	On	Off	Off
AfV event state	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
AfV hold time	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
AfV max event time	ABSOLUTE	On	On	On	Off	Off	Off	Off	Off
Module Link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Channel source selection	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off

Link Master Table

Link Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Isolate	ABSOLUTE	On	Off	Off	Off	Off	Off	On	Off
DigiAmp level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
DigiAmp on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Input gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Input balance	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Input phase revert left	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Input phase revert right	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Input stereo swap	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Input left to both	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Input right to both	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Input M/S matrix	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Input mono	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Insert send level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Insert on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Insert level bypass	OFF	On	Off	Off	Off	Off	Off	Off	Off
Insert soft clip	OFF	On	Off	Off	Off	Off	Off	Off	Off
Insert meter source	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 frequency	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 Q	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 slope	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 type	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 2 & 3 notch	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF frequency	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off

Link Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Filter/SCF Q	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF 1 & 2 on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF slope	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF type	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Filter on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Expander threshold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander ratio	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander attack	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander hold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander release	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander delay	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Gate threshold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate floor	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate hysteresis	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate attack	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate hold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate release	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate delay	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Gate SCF on	OFF	On	Off	Off	Off	Off	Off	Off	Off
Gate external key on	OFF	On	Off	Off	Off	Off	Off	Off	Off
Gate external key	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Compressor threshold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor ratio	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor attack	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor hold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor release	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor delay	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off

Link Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Compressor SCF on	OFF	On	Off	Off	Off	Off	Off	Off	Off
Compressor external key on	OFF	On	Off	Off	Off	Off	Off	Off	Off
Compressor external key	OFF	On	Off	Off	Off	Off	Off	Off	Off
Compressor soft knee	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Limiter threshold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter hysteresis	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter attack	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter hold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter release	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter delay	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Limiter soft knee	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Delay time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Delay on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Image width	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Image position	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Image on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Image style	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Direct out level	RELATIVE	On	Off	Off	Off	On	Off	Off	Off
Direct on	ABSOLUTE	On	Off	Off	Off	On	Off	Off	Off
Direct out level bypass	ABSOLUTE	On	Off	Off	Off	On	Off	Off	Off
Direct out soft clip	ABSOLUTE	On	Off	Off	Off	On	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Aux send level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Aux send pan/ balance	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Aux send on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Aux send mix cue	OFF	On	Off	Off	Off	Off	Off	Off	Off

Link Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Aux send independent	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Metering position	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Direct out position	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Track path position	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Track switch	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Aux send position	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Channel module order	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Fader Level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	Off	Off	Off	Off	Off	Off	Off	Off
Mute	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Left-right panning	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Front-back panning	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Pan slope	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
LFE level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan front width	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan back width	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan depth	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan turn	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Pan on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Pan mode center-flat	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan turn pre pan	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Direct out balance	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Busses	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
PFL	OFF	On	Off	Off	Off	Off	Off	Off	Off
AFL	OFF	On	Off	Off	Off	Off	Off	Off	Off
DirOut mute by fader	OFF	On	Off	Off	Off	Off	Off	Off	Off
Coordination	OFF	On	Off	Off	Off	Off	Off	Off	Off

Link Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Conference	OFF	On	Off	Off	Off	Off	Off	Off	Off
AfV on level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV off level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV attack time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV hold time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV release time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV event number	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AfV state	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AfV enable	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AfV event state	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AfV hold time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV max event time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Module Link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Channel source selection	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off

Couple Group Master Table

Couple Group Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Isolate	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
DigiAmp level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
DigiAmp on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Input gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Input balance	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Input phase revert left	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Input phase revert right	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Input stereo swap	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Input left to both	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Input right to both	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Input M/S matrix	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Input mono	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Insert send level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Insert on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Insert level bypass	OFF	On	Off	Off	Off	Off	Off	Off	Off
Insert soft clip	OFF	On	Off	Off	Off	Off	Off	Off	Off
Insert meter source	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 frequency	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 Q	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 slope	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 1 to 4 type	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer 2 & 3 notch	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Equalizer on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF frequency	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off

Couple Group Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Filter/SCF Q	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF 1 & 2 on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF slope	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Filter/SCF type	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Filter on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Expander threshold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander ratio	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander attack	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander hold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander release	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander delay	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Expander on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Gate threshold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate floor	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate hysteresis	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate attack	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate hold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate release	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate delay	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Gate on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Gate SCF on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Gate external key on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Gate external key	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Compressor threshold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor ratio	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor attack	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor hold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor release	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor delay	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Compressor on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off

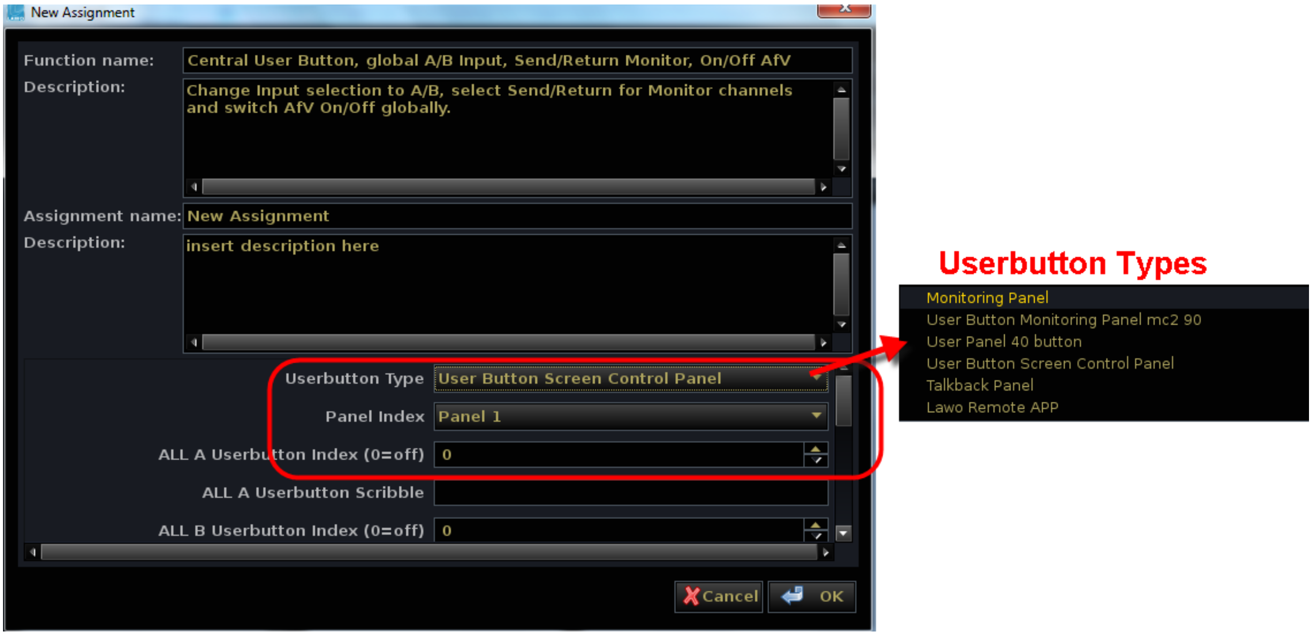
Couple Group Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Compressor SCF on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Compressor external key on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Compressor external key	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Compressor soft knee	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Limiter threshold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter gain	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter hysteresis	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter attack	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter hold	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter release	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter delay	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Limiter on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Limiter soft knee	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Delay time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Delay on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Image width	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Image position	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Image on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Image style	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Direct out level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Direct on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Direct out level bypass	OFF	On	Off	Off	Off	Off	Off	Off	Off
Direct out soft clip	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Aux send level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Aux send pan/ balance	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Aux send on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Aux send mix cue	OFF	On	Off	Off	Off	Off	Off	Off	Off

Couple Group Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Aux send independent	OFF	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Metering position	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Direct out position	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Track path position	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Track switch	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Aux send position	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Channel module order	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Fader Level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Mute	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Left-right panning	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Front-back panning	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Pan slope	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
LFE level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan front width	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan back width	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan depth	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan turn	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Pan on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Pan mode center-flat	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan on	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Hyperpan turn pre pan	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Direct out balance	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Busses	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Module link	OFF	On	Off	Off	Off	Off	Off	Off	Off
PFL	OFF	On	Off	Off	Off	Off	Off	Off	Off
AFL	OFF	On	Off	Off	Off	Off	Off	Off	Off
DirOut mute by fader	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
Coordination	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off

Couple Group Master Table									
Parameter	**Mode**	**Slave Operation**	**Apply On Assign**	**Restore**	**Relative Fader**	**Slave Controls Master**	**Invert**	**Ignore Module Link**	**Ignore Suspend**
Conference	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AfV on level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV off level	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV attack time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV hold time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV release time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV event number	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AfV state	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AfV enable	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AfV event state	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off
AfV hold time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
AfV max event time	RELATIVE	On	Off	Off	Off	Off	Off	Off	Off
Module Link	OFF	On	Off	Off	Off	Off	Off	Off	Off
Channel source selection	ABSOLUTE	On	Off	Off	Off	Off	Off	Off	Off

User Button Numbering

The following information provides the panel type, index and button numbering for each user button panel on the mc²56. You will need this information to address user buttons from the Custom Functions display:



Monitoring Panel

This panel type addresses the Touch-screen Monitoring Buttons 1 to 24 on the Central GUI.

The panel index and button numbering is as follows:

mc²56 monitor panel (touch / GUI)

User Button Monitoring Panel mc290

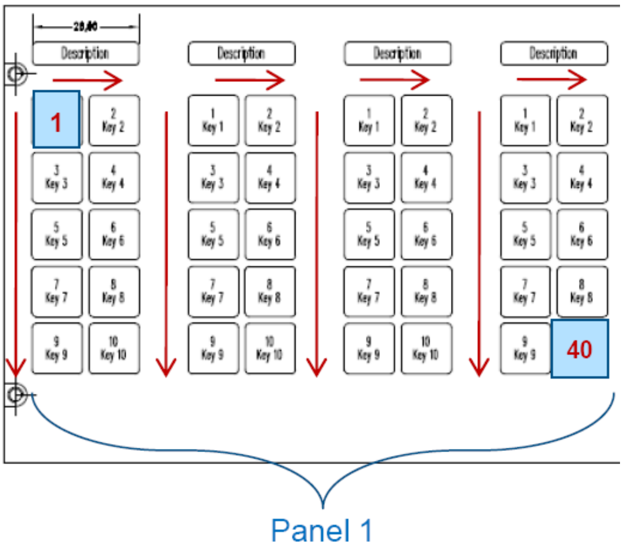
This panel type is not supported by the mc²56. It is used on the mc²90 to programme the hardware user buttons on the Monitor Panel.

User Panel 40 Button

This is an optional user panel which may be fitted to the Overbridge.

The panel index and button numbering are shown below:

User Panel 40

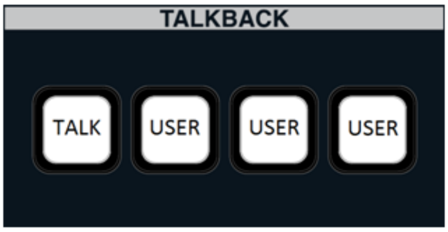


User Button Screen Control Panel

This panel type addresses the Central User Buttons in the centre section:

Talkback Panel

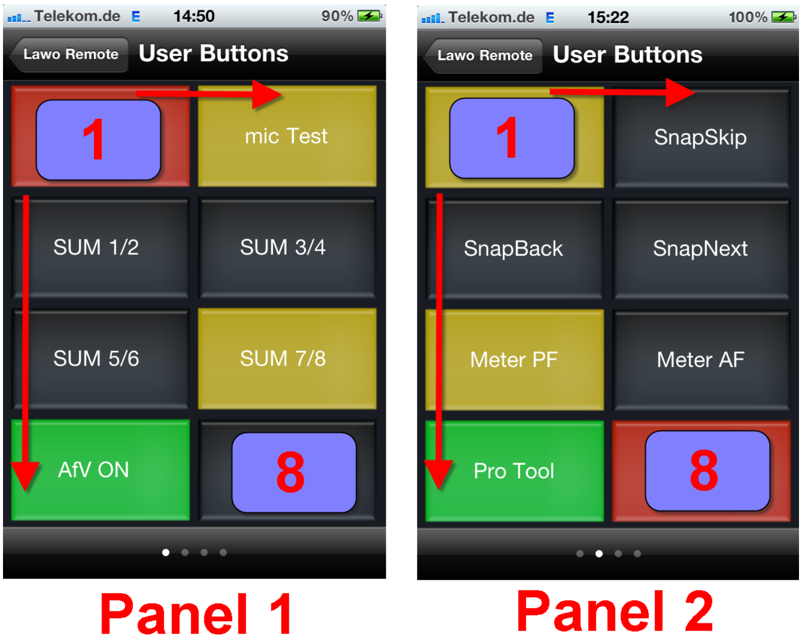
This panel type addresses the Talkback User Buttons beside the monitor level controls; buttons are number 1 to 4 from left to right:



Lawo Remote APP

This panel type refers the user buttons available from a device running the Lawo Remote App (iPhone, iPod or iPad).

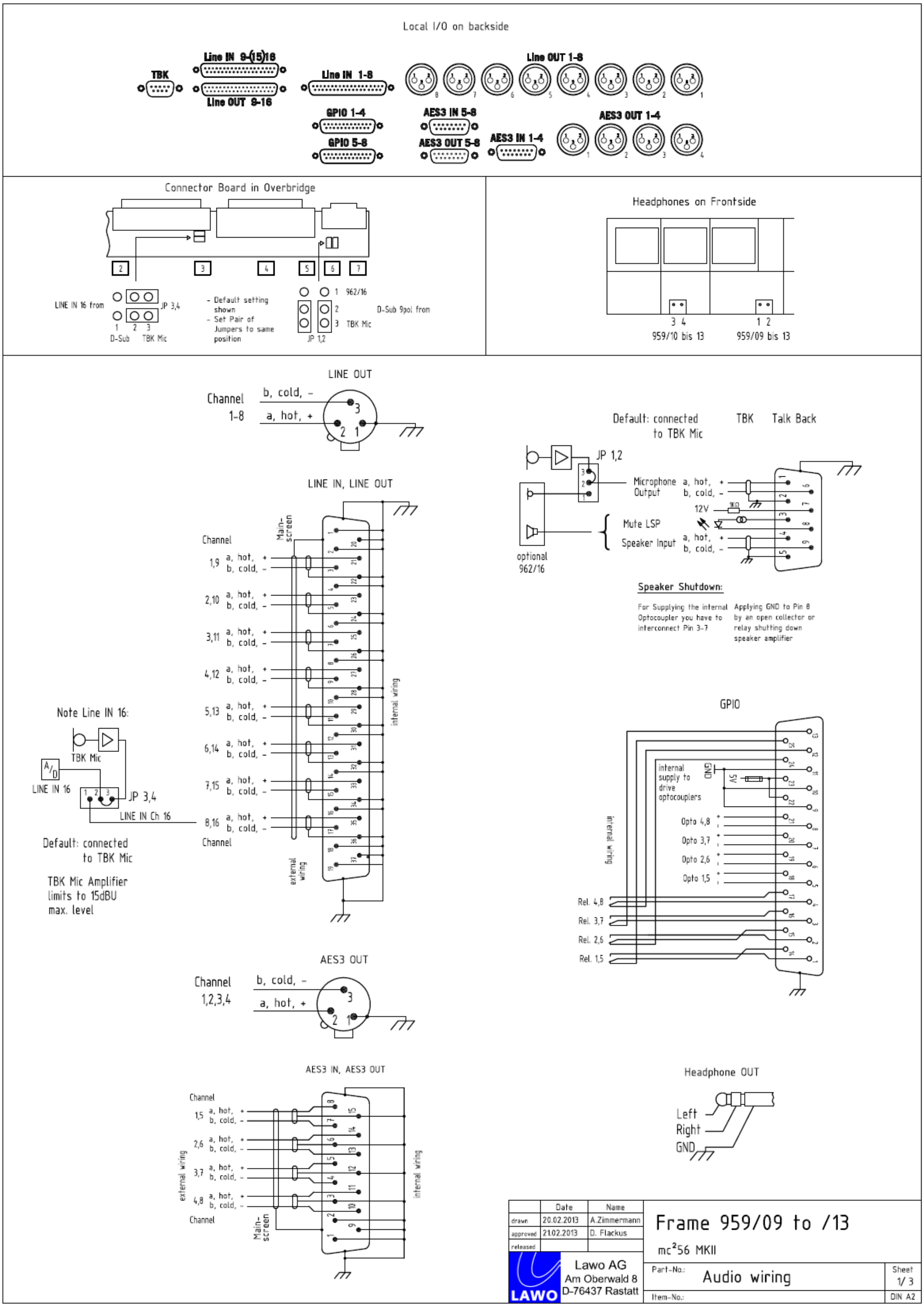
The panel index and button numbering for the first two panels is shown below. Up to four User Button panels may be configured.

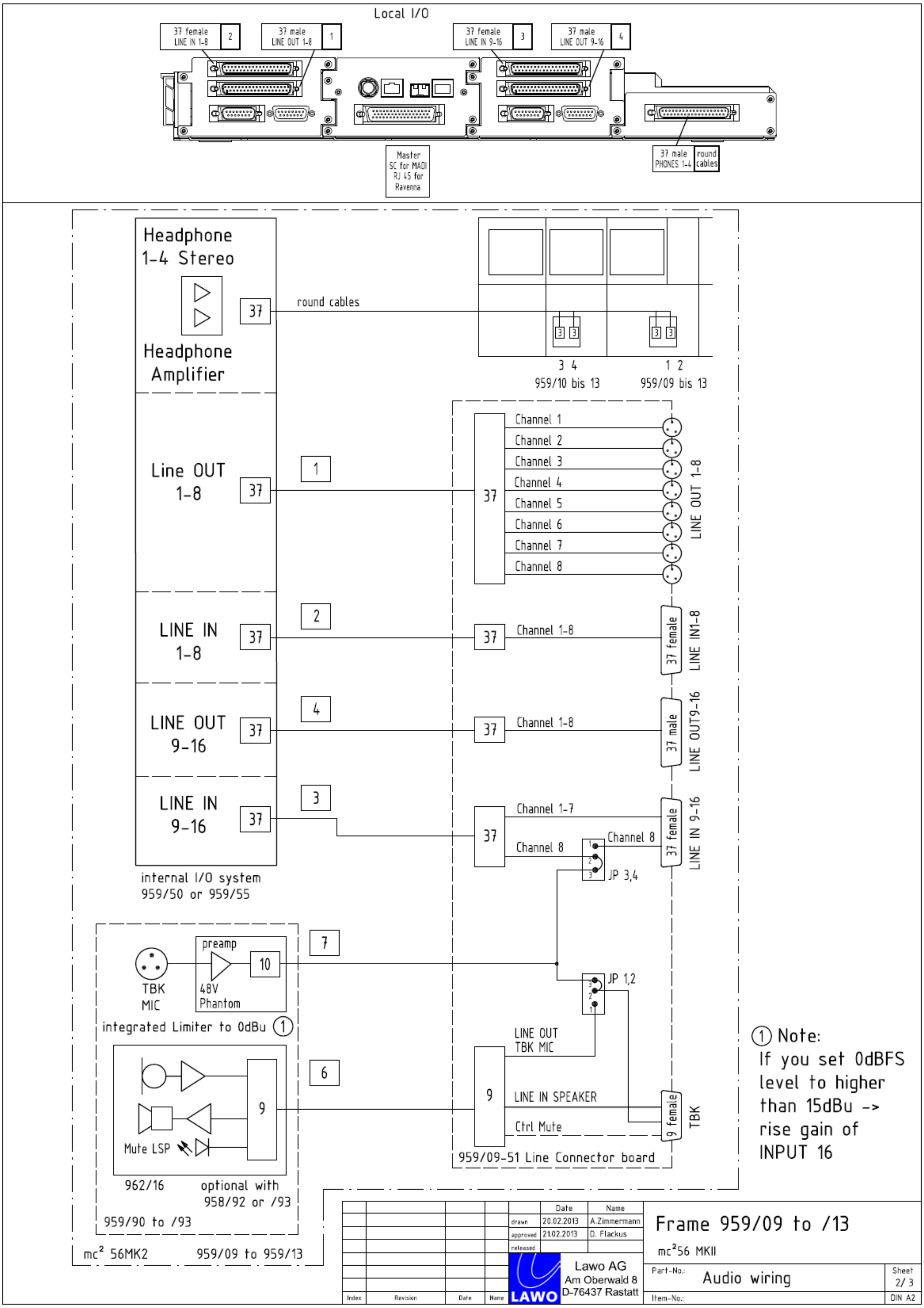


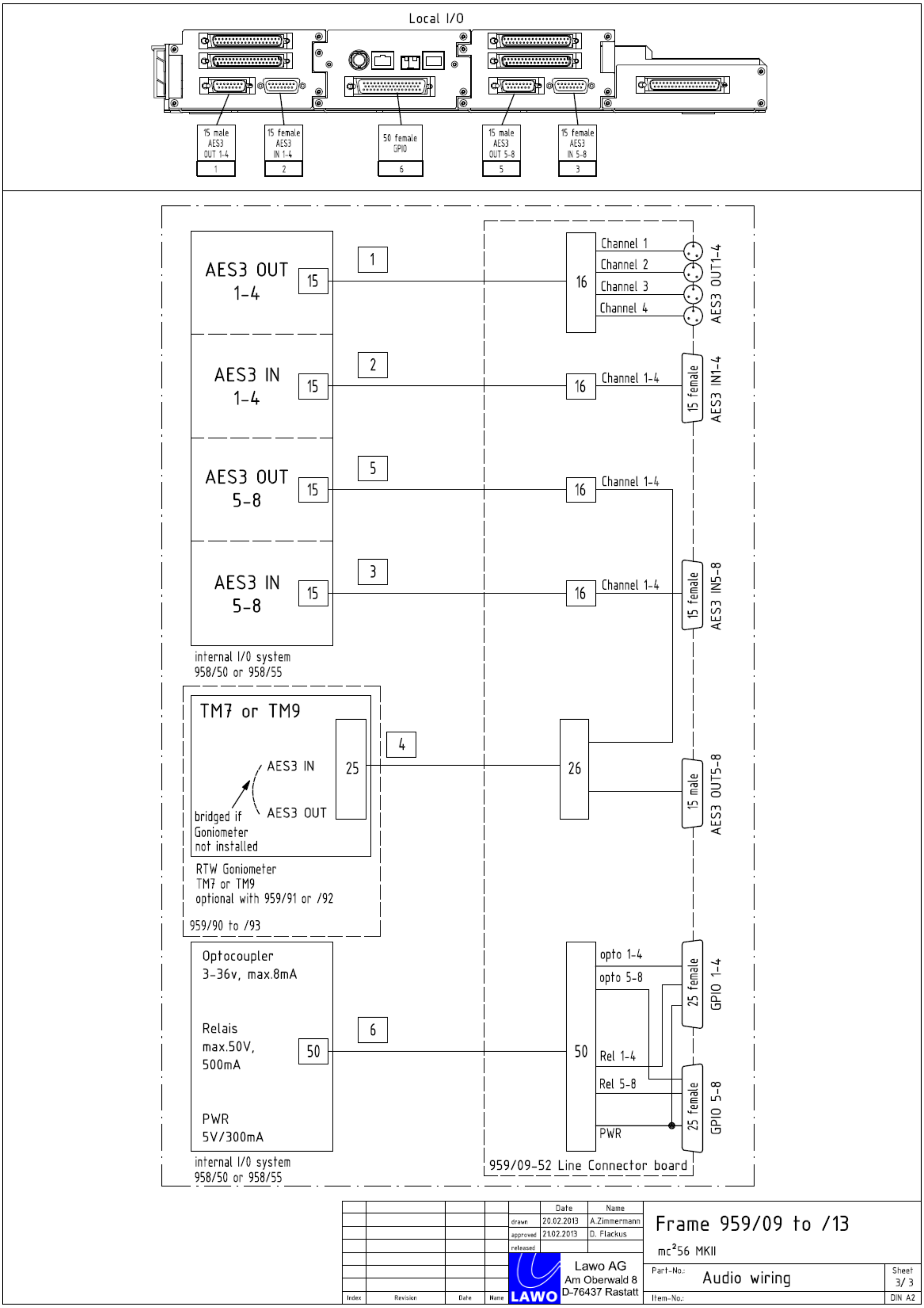
Local I/O Wiring

The following diagrams illustrate the wiring, pin-outs and default jumper switch positions for the local I/O.

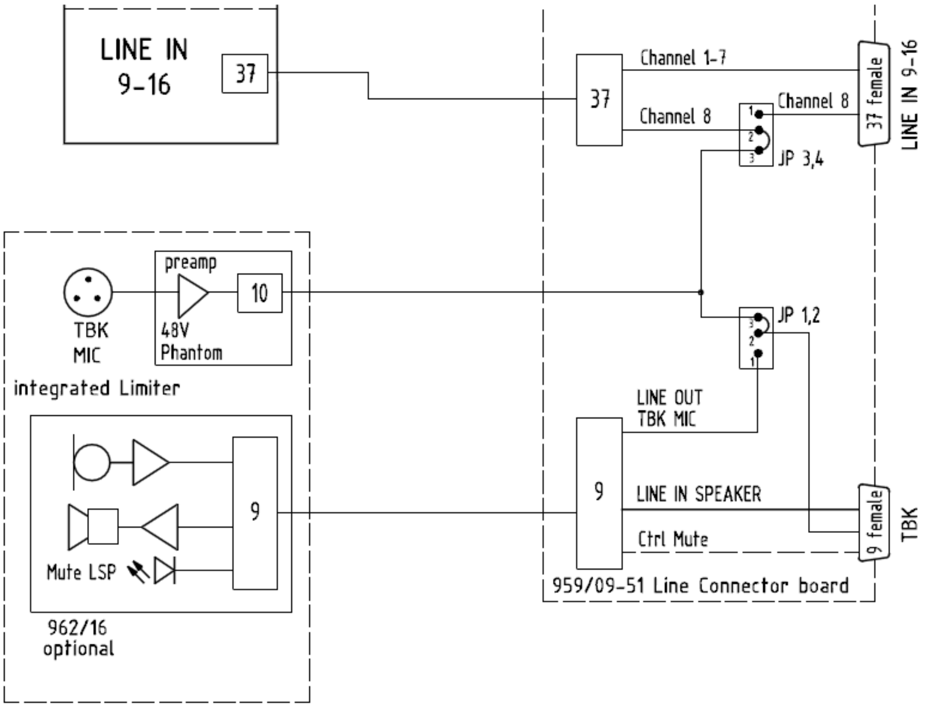
Note that **AES IN 5-8** and **AES OUT 5-8** connect to the RTW meter if either of the TM 7 or TM 9 Overbridge options are fitted.


Local I/O on backside
TBK
Line IN 9-(15)16
Line OUT 9-16
Line IN 1-8
Line OUT 1-8
GPIO 1-4
GPIO 5-8
AES3 IN 5-8
AES3 OUT 5-8
AES3 IN 1-4
AES3 OUT 1-4
Connector Board in Overbridge
LINE IN 16 from
JP 3,4
D-Sub
TBK Mic
- Default setting shown
- Set Pair of Jumpers to same position
962/16
D-Sub 9pol from
TBK Mic
JP 1,2
Headphones on Frontside
959/10 bis 13
959/09 bis 13
LINE OUT
Channel 1-8
b, cold, -
a, hot, +
LINE IN, LINE OUT
Main-screen
Channel
1,9
2,10
3,11
4,12
5,13
6,14
7,15
8,16
a, hot, +
b, cold, -
internal wiring
external wiring
Note Line IN 16:
TBK Mic
A/D
LINE IN 16
JP 3,4
LINE IN Ch 16
Default: connected to TBK Mic
TBK Mic Amplifier limits to 15dBU max. level
Default: connected to TBK Mic
TBK Talk Back
JP 1,2
Microphone Output
12V
Mute LSP
Speaker Input
optional 962/16
Speaker Shutdown:
For Supplying the internal Optocoupler you have to interconnect Pin 3-7
Applying GND to Pin 8 by an open collector or relay shutting down speaker amplifier
GPIO
internal supply to drive optocouplers
GND
5V
Opto 4,8
Opto 3,7
Opto 2,6
Opto 1,5
Rel. 4,8
Rel. 3,7
Rel. 2,6
Rel. 1,5
AES3 OUT
Channel 1,2,3,4
AES3 IN, AES3 OUT
Headphone OUT
Left
Right
GND
Date
Name
drawn
20.02.2013
A.Zimmermann
approved
21.02.2013
D. Flackus
released
Frame 959/09 to /13
mc²56 MKII
Lawo AG
Am Oberwald 8
D-76437 Rastatt
Part-No.:
Audio wiring
Sheet 1/3
Item-No.:
DIN A2


Local I/O
37 female LINE IN 1-8
2
37 male LINE OUT 1-8
1
37 female LINE IN 9-16
3
37 male LINE OUT 9-16
4
Master SC for MADI RJ 45 for Ravenna
37 male PHONES 1-4
round cables
Headphone 1-4 Stereo
Headphone Amplifier
37
round cables
3 4
959/10 bis 13
1 2
959/09 bis 13
Line OUT 1-8
Channel 1
Channel 2
Channel 3
Channel 4
Channel 5
Channel 6
Channel 7
Channel 8
LINE OUT 1-8
LINE IN 1-8
Channel 1-8
37 female
LINE IN1-8
LINE OUT 9-16
37 male
LINE OUT9-16
LINE IN 9-16
Channel 1-7
Channel 8
JP 3,4
37 female
LINE IN 9-16
internal I/O system
959/50 or 959/55
preamp
10
7
TBK MIC
48V Phantom
integrated Limiter to 0dBu
JP 1,2
LINE OUT TBK MIC
LINE IN SPEAKER
Ctrl Mute
9 female
TBK
6
9
Mute LSP
962/16
optional with 958/92 or /93
959/90 to /93
959/09-51 Line Connector board
mc² 56MK2
959/09 to 959/13
Note:
If you set 0dBFS level to higher than 15dBu -> rise gain of INPUT 16
Date
Name
drawn
20.02.2013
A.Zimmermann
approved
21.02.2013
D. Flackus
released
Lawo AG
Am Oberwald 8
D-76437 Rastatt
Index
Revision
Date
Name
Frame 959/09 to /13
mc²56 MKII
Part-No.:
Audio wiring
Sheet 2/3
Item-No.:
DIN A2


Local I/O
15 male AES3 OUT 1-4
1
15 female AES3 IN 1-4
2
50 female GPIO
6
15 male AES3 OUT 5-8
5
15 female AES3 IN 5-8
3
AES3 OUT 1-4
15
1
16
Channel 1
Channel 2
Channel 3
Channel 4
AES3 OUT1-4
AES3 IN 1-4
15
2
16
Channel 1-4
15 female
AES3 IN1-4
AES3 OUT 5-8
15
5
16
Channel 1-4
AES3 IN 5-8
15
3
16
Channel 1-4
15 female
AES3 IN5-8
internal I/O system
958/50 or 958/55
TM7 or TM9
AES3 IN
AES3 OUT
25
4
26
bridged if
Goniometer
not installed
15 male
AES3 OUT5-8
RTW Goniometer
TM7 or TM9
optional with 959/91 or /92
959/90 to /93
Optocoupler
3-36v, max.8mA
Relais
max.50V,
500mA
PWR
5V/300mA
50
6
50
opto 1-4
opto 5-8
Rel 1-4
Rel 5-8
PWR
25 female
GPIO 1-4
25 female
GPIO 5-8
internal I/O system
958/50 or 958/55
959/09-52 Line Connector board
Date
Name
drawn
20.02.2013
A.Zimmermann
approved
21.02.2013
D. Flackus
released
Frame 959/09 to /13
mc²56 MKII
Lawo AG
Am Oberwald 8
D-76437 Rastatt
Part-No.:
Audio wiring
Sheet
3/ 3
Index
Revision
Date
Name
Item-No.:
DIN A2

Local I/O Jumper Switch Positions



There are four jumper switches on the local I/O connector board, which control two settings:

- **JP 3,4** - set the connection to the **Line input 16** A-D converter. This can be taken from:
  - o an "internal talkback mic preamp" (set by **JP 1,2**).
  - o the **LINE IN 16** connection from the rear panel.
- **JP 1,2** - set the "internal talkback mic preamp" to:
  - o the integrated talkback mic preamp (fitted as standard).
  - o the talkback mic preamp fitted to the optional 962/16 INTERCOM user panel.

Note that the **JP 1,2** switch positions affect both the connection to the **Line input 16** A-D converter, and the line level talkback output available via the **TBK** connector.

The factory default positions, shown above, support talkback via the integrated talkback mic preamp.

You may need to adjust the jumper switches if:

- the 962/16 INTERCOM user panel is fitted. Move **JP 1,2** to connect talkback from the INTERCOM panel's talkback mic preamp.
- you are using an external talkback source, and wish to "free up" **Line input 16** for another application. Move **JP 3,4** to connect Line input 16 from the **LINE IN 9-16** connector.

Please see the "mc²56 Technical Manual" for details on adjusting the jumper switch positions.

For more details on the 962/16 INTERCOM user panel, please refer to the relevant data sheet.

DSP Configurations

DSP resources are allocated using DSP configurations.

Please note that the variation with **1 DSP** board includes: 1 x stereo PFL; 1 x stereo AFL.

All other variations include: 2 x stereo PFL; 1 x stereo AFL; 1 x surround AFL (7.1).

Higher sample rates use twice as much DSP resource as lower sample rates.

More channels, from the same DSP resource, become available if you use Broadcast channels.

For further details on the variations available, we recommend installing mxGUI and viewing the **DSP Configurations** display.

SDI Parameters

AdminHD can define a number of parameters for the SDI Card, SDI Signal In and SDI Signal Out. The parameters are what the system resets to after a cold start.

The parameters are similar to those on the mc² GUI's **Signal settings** display, and vary depending on the type card:

- SDI Parameters (3G SDI Card)
- SDI Parameters (non 3G SDI Cards)

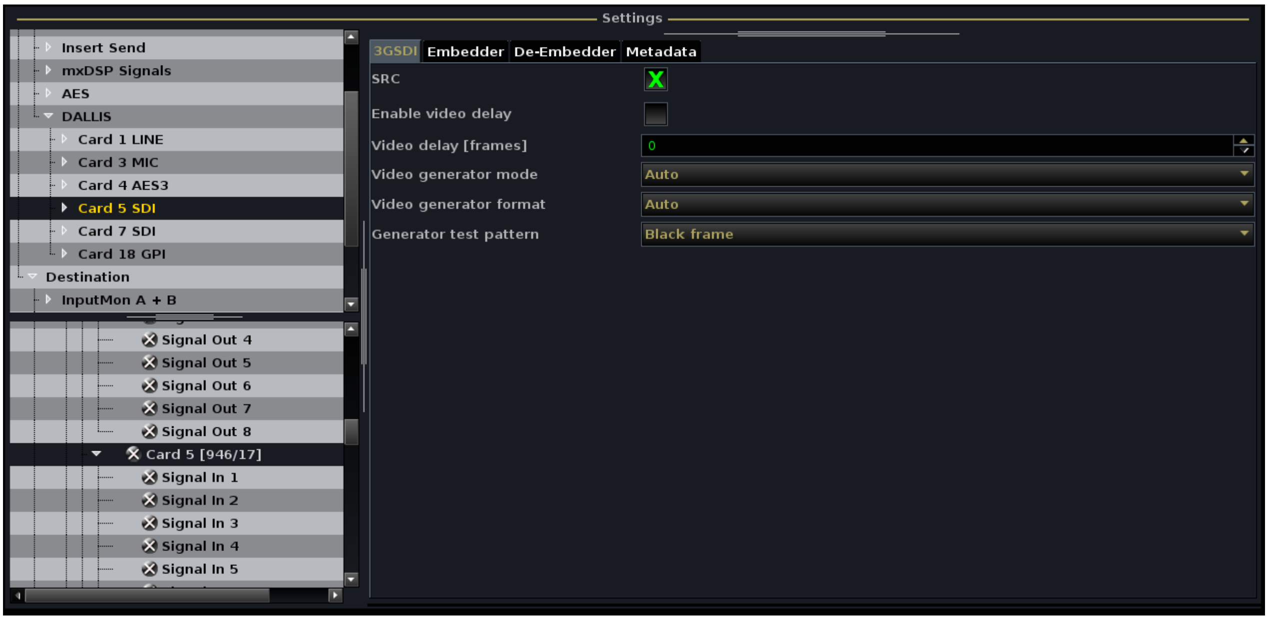
SDI Parameters (3G SDI Card)

The DALLIS 3G/HD/SD SDI card (946/17) is a multi-rate SDI card with BNC input, thru and two outputs. It contains an audio embedder and de-embedder for up to 16 audio channels, and a VANC embedder and de-embedder for two independent Dolby E Metadata streams. There is onboard video and audio delay, and an integrated sample rate converter. It occupies two DALLIS card slots and may be configured to run in a number of different modes using AdminHD.

Further information can be found in the data sheet, available in the "mc2_Nova73_documentation" guide.

Note that SDI signals have parameters for both the signal and the card. The SDI parameters are adjusted by selecting the card:

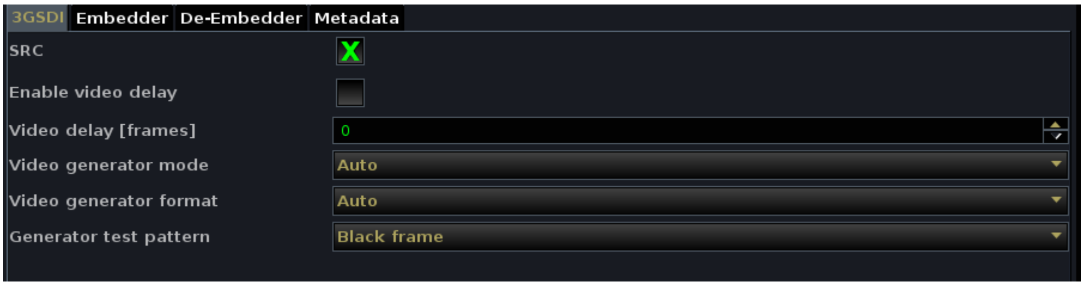
1. Select the **946/17** card from the **System** tree.
2. Then select one of the four parameter tabs:



Note that SDI card parameters may be adjusted whether the card is local to the system, or fitted to a remote network partner.

Note that SDI parameters are never stored by snapshots. From Version 4.8.0.2 onwards, they are stored and recalled by productions. You may use the Global isolate of SDI parameters custom function to isolate SDI parameters so that settings are not affected by a production load.

3G SDI



- **SRC** – check this option to enable sample rate conversion. Note that SRC is applied to all channels on the card. Normally, SRC should be enabled. If SRC is off (unchecked), then the system must be clocked to the same reference as the sending device.
- **Enable video delay** & **Video delay** (frames) – this option applies a delay to the SDI data from the de-embedder to embedder. Video and audio containted in the stream are delayed by the same amount. Set the amount of Video delay in steps of 1 video frame.
- **Video generator mode**, **format** & **test pattern** – the SDI card is equipped with a free-running video test pattern generator. Set the mode to either:
  - **Auto** – if the input is locked to an incoming video signal, then the output will automatically track the format of the input. If the input fails, then the video test pattern generator transmits the last received video format. When the SDI module is part of a SDI chain, this option is recommended.
  - **Force On** – in this mode it is assumed that the card is used as a video master and that no SDI input signal is applied. The test pattern generator is forced on all the time. Use the **Video generator format** and **Generator test pattern** options to define the video signal. In this mode the embedder sample rate is derived from the generator, and the SDI receiver is switched off. Note that the de-embedder cannot be used.

Embedder

- **Clean** – check this option to set the embedder mode to "Clean". In this mode the incoming audio stream is deleted and a new data structure generated according to your embedder settings. Note that if you select this mode any existing audio data will be lost.
- **Embedder Group Enable** – audio is embedded in groups of four channels into SDI. There is a total of four groups per SDI, resulting in 16 audio channels. For each group, this checkbox determines whether the incoming SDI stream is replaced:
  - o Enable the checkbox to replace the audio group content.
  - o Disable the checkbox to leave the audio group untouched.

If there is no audio at the SDI input, then a new audio group will be generated.

Note that in AdminHD modes 16/0 and 8/0, all embedder group enables are turned off as the whole embedder section is bypassed.

- **Embedder source 1 to 16** – use these options to define the source for each embedder.

De-Embedder

- **DALLIS signal in source 1 to 16 –** use these options to define the source for each de-embedder.

Metadata

The SDI module offers 2 metadata ports according to SMPTE RDD-2008. This allows embedding, de-embedding and transport of two independent Dolby metadata streams alongside with the video. The streams can be accessed via two D-Sub connectors at the front panel.

- **Metadata de-emb. & emb. to port 1, 2** - use these options to define the streams for the Metadata ports.
- **Metadata embedder mode & line** – set the mode to **Auto** to track the input, or select **Pre-selected line** and define a **Metdata embedder line**.

SDI Parameters (non 3G SDI Cards)

The DALLIS HD or SD SDI cards (946/13, 09, 05, 01) provide the ability to route a maximum of 8 channels to/from the SDI stream. Sample rate conversion may be applied to the whole card (all 8 channels), and delay may be applied to either the embedded or de-embedded signals.

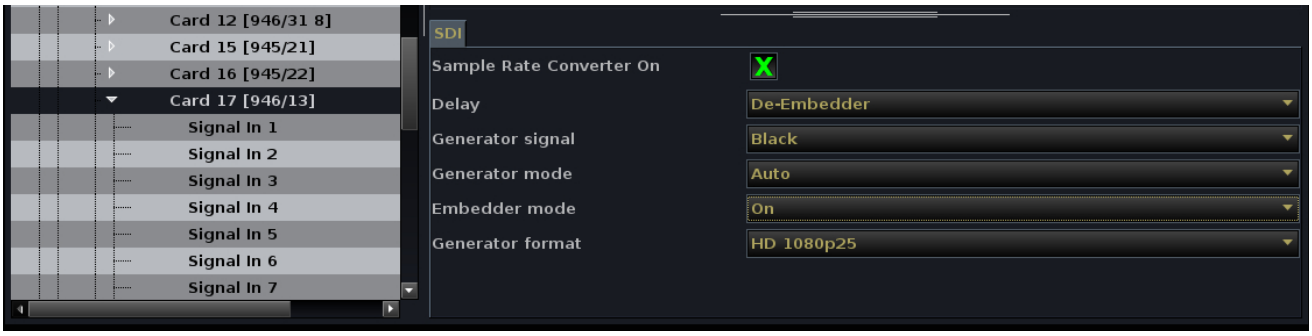
Further information can be found in the data sheet, available in the "mc2_Nova73_documentation" guide.

SDI parameters can be adjusted for the card and for individual input and output signals.

Note that SDI parameters are never stored by snapshots. From Version 4.8.0.2 onwards, they are stored and recalled by productions. You may use the Global isolate of SDI parameters custom function to isolate SDI parameters so that settings are not affected by a production load.

SDI Card

Select an SDI card from the **System** tree, and click on **SDI** to adjust the following card parameters:



- **SRC** – check this option to enable sample rate conversion. Note that SRC is applied to all 8 channels on the card. Normally, SRC should be enabled. If **SRC** is off (unchecked), then the system must be clocked to the same reference as the sending device.
- **Delay** – select whether delay is enabled for the **Embedded** (SDI output) or **De-embedded** (SDI input) signals; delay cannot be applied to both.
- **Generator signal**, **mode** and **format** – defines the output generator signal for the SDI stream.
- **Embedder mode** – select from:
  - o **On** – audio channels will be replaced within the existing SDI data structure according to your SDI output group selections.
  - o **Off** – no audio replacement; the SDI stream remains unaltered.
  - o **Clean** – deletes the incoming audio stream and generates a new data structure according to your embedder settings. Note that if you select this mode any existing audio data will be lost.

SDI Inputs

Select an SDI input signal from the **System** tree, and click on **SDI** to adjust the following signal parameters:

- **Group select** – this field defines which pair of SDI channels will map to the selected SDI card input. In our example, **Group 2 Channels 3&4** from the SDI stream will be de-embedded to **SDI Signal In 1** and **2**.
- **Delay time** & **Delay** – check the Delay option to enable delay for the stereo input, and set the delay time in ms. Delay time can be adjusted from 0 to 240ms.

Delay is only applied to SDI inputs if the SDI card **Delay** parameter is set to **De-embedder**.

SDI Outputs

Select an SDI output signal from the **System** tree, and click on **SDI** to adjust the following parameters:

- **Group select** – this field defines which pair of SDI channels will map to the selected SDI card output. In our example, **Group 1 Channels 1&2** from the SDI stream will be embedded to **SDI Signal Out 1** and **2**.

The assignment is only active if the SDI card **Embedder mode** is set to **On** or **Clean**.

- **Delay time** & **Delay** – check the **Delay** option to enable delay for the stereo output, and set the delay time in ms.

Delay is only applied to SDI outputs if the SDI card **Delay** parameter is set to **Embedder**.

- **Wordlength** – choose from the available drop-down menu options.

When 16 or 20-bit are selected, dither is automatically applied.

Control System Locations

The table below shows the location of the control system for different mc^2 and Nova73 products.

Note that:

- The Router Module MKII (980/33) control system provides two network ports: **ETHERNET A** connects to the mc^2 control surface; **ETHERNET B** connects to the Lawo system network.
- For control systems located inside the control surface, you may use any console **ETHERNET** port to connect to the Lawo system network.

System	Router Module	Control System	Location	System Network Port
mc² 56 classic	980/33	Intel	Nova73	ETHERNET B
mc² 56 MKII	980/33	Intel	Nova73	ETHERNET B
mc² 66 classic	980/31, 32	Intel	control surface	ETHERNET (any)
mc² 66 top1	980/31, 32	Intel	control surface	ETHERNET (any)
mc² 66 MKII	980/33	Intel	Nova73	ETHERNET B
mc² 90	980/31, 32	Intel	control surface	ETHERNET (any)
mc² 90	980/33	Intel	Nova73	ETHERNET B
mc² 90 star²	980/33	Intel	Nova73	ETHERNET B
Nova73 Standalone MKI	980/31, 32	Motorola	Nova73	ETHERNET
Nova73 Ripper MKI	980/31, 32	Intel	1HE Ripper	see Technical Documentation
Nova73 DSHS MKI	980/32	Intel	1HE Ripper	see Technical Documentation
Nova73 HD MKII	980/33	Intel	Nova73	ETHERNET B
Nova73 Compact MKII	980/33	Intel	Nova73	ETHERNET B
Nova73 HD DSHS MKII	980/33	Intel	Nova73	ETHERNET B

TCP/IP Addresses

Default IP Addresses

The default IP addresses, for different Lawo product control systems, are:

- **mc²56** = 192.168.102.56
- **mc²66** = 192.168.102.65
- **mc²90** = 192.168.102.90
- **Nova73** (HD & Compact) = 192.168.102.143
- **Nova73 HD DSHS**: primary 'star' = 192.168.102.32; secondary 'star' = 192.168.102.160
- **mxGUI** (local control system) = 192.168.56.101

You can check the IP address of your control system from the GUI (using the Signal Settings display).

Subnet Mask

For all products, the default Subnet Mask is **255.255.255.0**.

Other IP Addresses

The table below lists the other IP addresses used within a mc²56 installation:

Device	Port	IP Address	Notes
Router Module Slot A	ETHERNET A	192.168.105.1	Fixed address.
Router Module Slot A	ETHERNET B	192.168.102.xxx	Default address of the control system (as listed above). This address can be modified by AdminHD.
Router Module Slot B (optional)	ETHERNET A	192.168.106.1	Fixed address.
Router Module Slot B (optional)	ETHERNET B	192.168.102.xxx	This address is *always* one digit higher than that of the main control system. It is set automatically by AdminHD.
Ethernet Switch (optional)	-	192.168.102.250	Default address.
ISDN Dialup Router (optional)	-	192.168.102.200	Default address.

We recommend keeping the default IP addresses, where possible, as this will simplify remote maintenance.

Glossary

48kHz or 44.1kHz	See Sample Rate.
Access	On mc² consoles, much of the channel parameter operation is performed by assigning a fader strip to the Central Control Section. This is otherwise known as putting a source 'in access'.
AdminHD	Lawo's configuration and control software for Nova73 systems.
ATM	Asynchronous Transfer Mode (Packing of signals in small portions; commonly used and highly standardised network protocol).
Attack Time	In the context of dynamics processing (compressor, limiter, gate or expander), the attack time defines the duration over which an input signal is measured. The longer the attack time, the slower the processor will react. For example, when using a gate, a fast attack time causes the gate to open quickly when signal exceeds the gate threshold.
Aux	Auxiliary An Aux is a general purpose mono, stereo or multi-channel summing bus which can be used for a variety of applications such as sending to outboard effects devices.
Aux Send	Auxiliary Send Source channels feed onto each aux via their Aux Send. The aux send from each channel can be either pre or post fader and has variable level control.
Aux Master	Auxiliary Master The Aux Master is a master source channel used to control the level and processing of the Aux output. The direct output of the Aux Master is the signal routed to the outboard effects send.
Aux Return	Auxiliary Return The Aux Return is the name given to the return channel from the outboard effects device. This channel controls the level and processing of the effect as it is summed into the rest of the mix.
Band Pass Filter	See Filters.
Balance	Balance is applied to the input of a stereo channel and is the ratio between the left and right input levels. When Balance is set to its default value, the level of left and right inputs are equally weighted.
Bargraph	An optical display instrument in the shape of a LED bar for displaying signal level.
Clean Feed	See Mix Minus.

Compressor

A dynamics processor used to smooth out uneven signal levels. For example, when a presenter shouts and then whispers, they are producing sound which has a wide dynamic range; one moment it is very loud and the next very quiet. This can mean that if we listened to this signal on our radio without compression, we would forever be turning the level up and down! A compressor smoothes the signal such very loud audio is reduced in level and very quiet audio is increased in level. This results in smaller dynamic range ideal for radio transmission.

Configuration

The system configuration is a file created by the AdminHD software. The file may be exported and uploaded to the system's cold start data where it will load following a cold start. Or, the file may be uploaded to the system's warm start data where it is then loaded every time the system reboots or powers on. The configuration defines key elements of the system such as the hardware components, and default signal parameters.

ControlHD

Lawo's control software for Nova73 systems.

DALLIS

Lawo's modular I/O interfacing system based on 19" frames using plug-in cards for different interfaces.

dB

deciBel

A unit of transmission giving the ratio of two powers.

The number of bels is the logarithm to the base 10 of the ratio of the two powers. One decibel equals one tenth of a bel.

dBU

dBU is used to describe levels within the analogue domain, and is a measure of absolute voltage level based on 0dBU = 0.775 Volts (RMS). dBU is often used to indicate nominal broadcast operating levels.

dBFS

dB Full Scale

dBFS is used to describe levels within the digital domain. 0dBFS describes the system's internal clipping point; this is the maximum level which may be handled by the system without signal distortion.

Delay

The signal output from a delay module is x ms behind the signal input to the module. Delay is often applied to audio sources whose video has undergone digital video processing; delay is required such the audio remains in sync with the video.

Direct Out

Direct Output

The direct output of a channel is the output of the individual source. Direct Outputs are often used to provide a record or 'snoop' feed of a single source, and may be taken from various points within signal flow: pre fader, post fader, etc.

Drop-out

Interruption of the audio signal caused by an error in the signal transfer or recording.

DSP	Digital Signal Processing Digital signal processing (DSP) is the study of signals in a digital representation and the processing methods of these signals. Within mc^2 consoles and the Nova73, DSP is also used as the collective name given to the processing cards, within the Nova73, which provide audio signal processing such as equalization, dynamics and delay.
Dynamics	Dynamics is the collective terms given to audio processing which responds to changes in signal level. For example, a Compressor, Limiter, Gate or Expander.
EQ	Equaliser. An equaliser is a processor which changes the frequency characteristics of a signal, for example to increase the amount of treble or bass components in the signal.
Expander	A dynamics processor used to magnify changes in the dynamic range of the input signal. For example, to reduce noise in speech pauses. See also Compressor.
Fader	A potentiometer used to adjust the gain of a signal.
Filters	Filters are equaliser sections which are used to cut out or reduce specific frequency bands within the signal. For example, a Low Pass Filter cuts out high frequencies so will result in less treble to the sound. A High Pass Filter cuts out low frequencies, for example you may use this to remove unwanted low frequencies like hum or rumble. A Band Pass Filter cuts out both high and low frequencies allowing frequency components within the band to pass through the signal; for example, you may use this type of filter to create a telephone effect on a normal voice.
Gain	Adjusting the gain of a signal results in a change in the perceived level or amplitude. An increase in gain (positive values) results in amplification and a reduction in gain (negative values) in attenuation.
Gate	A dynamics processor used to remove unwanted signals below a certain threshold level. For example, if a gate is applied to a presenter's microphone source, then when they speak signal level exceeds the gate threshold and the gate opens, while if they make a low level sound, like shuffling in their seat, the gate remains closed. The result is that only the signal we want to hear is output from the source channel!
GPI	General Purpose Interface (IEEE488) is a standardised platform independent short-range digital interface, to allow switching connections between broadcast equipment from different manufacturers.
Headroom	The amount of operating level which is in reserve between normal operating level and 0dBFS.
High Pass Filter	See Filters.

Insert Point	A connection point within the source channel which interrupts the signal flow and routes out to a piece of external equipment and returns back to the source channel. Insert send = route out from the source channel to the external device. Insert return = input to the source channel from the external device.
Limiter	A dynamics processor used to stop signals exceeding a certain threshold level. For example, you may place a limiter across the main output of the programme to prevent a sudden increase in level exceeding the clipping point of your transmission feed and causing signal distortion.
Low Pass Filter	See Filters.
MADI	Multi-channel Audio Interface; digital interface for combining audio signals of 56 or 64 channels.
Mix Minus	Mix Minus, Clean Feed and N-1 are all terms used to describe a feed which is created from a number of channels minus a particular channel or channels. For example, to provide telephone hybrids with a feed of the programme minus the incoming phone call.
Monitor	Term used to describe the outputs and functionality of feeds to loudspeakers or headphones for the purpose of listening to a mix.
ms	milliseconds Unit of time measurement.
M-S	Middle and Side Stereo Used to describe an arrangement of two coincident microphones, one pointing to the front (Middle) and the other (bidirectional) at right angles providing a Side signal. The mc² consoles provide M-S to X-Y decoding to turn the Middle and Side signal into normal Left and Right stereo.
mxGUI	Lawo's control software for mc² and Nova73 systems. The software runs GUI displays from an external PC and can be used either online or offline.
Nova73	The heart of the mc² system (includes the routing matrix, control system, I/O modules and DSP). Can exist as a stand-alone routing matrix with networking capabilities.
N-1	See Mix Minus.
On-Air	Term used to indicate that a radio or TV programme is being broadcast.
Overload	Occurs when the signal level is too large for the system, resulting in signal distortion.
Panning	Used to control the left/right position of a mono source when routed to a stereo or multi-channel output. For example, if a source is panned left, then you will all signal from the source is routed to the left side of the summing bus. If a source is panned centre, equal levels are applied to the left and right sides of the summing bus, etc.

PFL	Pre Fade Listen Used to listen to signals before the application of fader level. Provides a way of listening to a source when the fader is closed to check its signal before the fader is opened to route it onto the programme output.
Phantom Power	This is the power supply required when working with condenser microphones. The console supplies 48V to the microphone via the audio connector.
Programme	The main output of a live broadcast console. This is the mix which feeds the transmission chain.
RAS	Radio Automation System control protocol is Lawo's universal protocol for communication between a mixing console (MIXER) and a radio automation system (RAS).
Ratio	In the context of a compressor or expander, the ratio defines how much compression or expansion is applied. For example, the higher the compressor ratio, the more signal levels above the compressor threshold will be compressed.
Release Time	In the context of dynamics processing (compressor, limiter, gate or expander), the release time defines the time taken for the action of the processor to subside. For example, when using a gate, a short release time will cause the gate to close quickly after signal falls below the gate threshold.
Remote MNOPL	The remote control protocol RemoteMNOPL is a LAN based client-server network byte order protocol to enable third party systems to control Lawo's digital mixing consoles or standalone routers.
Roll-off Frequency	See Shelving EQ.
Routing	Signal Routing Term used to describe the connection made between an input and output.
RS422	Type of serial interface used to communicate with external devices.
RU	Rack Units ⇒ 44,45 mm respectively 1,75 inch
Sample Rate	The speed at which the internal processing of the system takes samples respective to values from a continuous, analogue audio signal to make a discrete, digital one. For example, when running at 48kHz, incoming analogue audio is sampled at a rate of 48000 values per second.
Shelving EQ	A shelving equaliser band is used to increase or decrease high or low frequency components of a signal. The slope of the shelf defines how steeply the gain increase/decrease is applied. The roll-off frequency defines the frequency at which signal level is reduced by 3dB.
Slope	See Shelving EQ.
SMPTE	Abbreviation for Society of Motion Picture and Television Engineers Standardised protocol for the synchronisation of audio and video technology - timecode.

SRC	Sample Rate Converter.
Sum	Summing Bus The result of several audio signals mixed together within the console. Within mc^2 consoles, the name given to the main output busses (programme busses).
Telephone Hybrid	Device which deals with bi-directional signals to/from a 2-wire phone line. One line provides an incoming feed from the phone line (e.g. the guests voice), and the other sends signal back to the receiver (e.g. the mix minus feed).
Threshold	In the context of dynamics processing (compressor, limiter, gate or expander), the threshold defines the signal level at which the processor starts to act. For example, the gate threshold sets the level at which the gate will open and then close.

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