 **QUASAR XR**  
**QUASAR SR**



## User Manual



Includes Quasar Engine and Engine RPS

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# Notices and Cautions

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## Notices

The installation and service instructions in this manual are for use by qualified personnel only. To avoid electric shock, do not perform any servicing other than that contained in the operating instructions unless you are qualified to do so. Refer all servicing to qualified personnel.

This instrument has an autoranging line voltage input. Ensure the power voltage is within the specified range of 100-240v. The ~ symbol, if used, indicates an alternating current supply.



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



This symbol, wherever it appears, alerts you to important operating and maintenance instructions. Read the manual.

## Caution: Double Pole/Neutral Fusing

The instrument power supply incorporates an internal fuse. Hazardous voltages may still be present on some of the primary parts even when the fuse has blown. If fuse replacement is required, replace fuse only with same type and value for continued protection against fire.

## Warning

The product's power cord is the primary disconnect device. The socket outlet should be located near the device and easily accessible. The unit should not be located such that access to the power cord is impaired. If the unit is incorporated into an equipment rack, an easily accessible safety disconnect device should be included in the rack design.

To reduce the risk of electrical shock, do not expose this product to rain or moisture. This unit is for indoor use only.

This equipment requires the free flow of air for adequate cooling. Do not block the ventilation openings in the top and sides of the unit. Failure to allow proper ventilation could damage the unit or create a fire hazard. Do not place the units on a carpet, bedding, or other materials that could interfere with any panel ventilation openings.

If the equipment is used in a manner not specified by the manufacturer, the protection provided by the equipment may be impaired.

## USA Class A Computing Device Information to User. Warning

This equipment generates, uses, and can radiate radio-frequency energy. If it is not installed and used as directed by this manual, it may cause interference to radio communication. This equipment complies with the limits for a class a computing device, as specified by FCC rules, part 15, subpart J, which are designed to provide reasonable protection against such interference when this type of equipment is operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference. If it does, the user will be required to eliminate the interference at the user's expense. Note: objectionable interference to tv or radio reception can occur if other devices are connected to this device without the use of shielded interconnect cables. FCC rules require the use of shielded cables.

## Canada Warning

"This digital apparatus does not exceed the class A limits for radio noise emissions set out in the radio interference regulations of the Canadian department of communications."

"Le présent appareil numérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils numériques (de class A) prescrites dans le règlement sur le brouillage radioélectrique édicté par le ministère des communications du Canada."

## CE Conformity Information

This device complies with the requirements of the EEC council directives:

- 93/68/EEC (CE MARKING)
- 73/23/EEC (SAFETY - LOW VOLTAGE DIRECTIVE)
- 89/336/EEC (ELECTROMAGNETIC COMPATIBILITY)

Conformity is declared to these standards: EN50081-1, EN50082-1.

A copy of the CE Certificate of Conformity for this product is available in the last page of this User Manual.

# Quasar User Manual

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## Updates

The operation of Quasar is determined largely by software. We routinely release new versions to add features and fix bugs. Check the Axia Audio web site for the latest. We encourage you to sign-up for the email notification service offered on the site.

## Feedback

We welcome feedback on any aspect of Quasar, or this manual. In the past, many good ideas from users have made their way into software revisions or new products. Please contact us with your comments. If you find anything in this manual that you feel needs clarification or correction, please let us know by sending an e-mail to [quasar@telosalliance.com](mailto:quasar@telosalliance.com)

## Serial Numbers

All units produced by Axia are given a serial number and are booked into a central record system at the time of manufacture. These records are updated whenever a piece of hardware is dispatched to or received from a customer. When contacting Axia Customer Support with a hardware inquiry it is important to provide the correct Quasar serial number.

This is printed on a label on the Rear I/O Module, located at the back of the console frame.

## Service

You must contact Axia before returning any equipment for factory service. We will need your unit's serial number, located on the back of the unit. Axia will issue a return authorization number, which must be written on the exterior of your shipping container. Please do not include cables or accessories unless specifically requested by the Technical Support Engineer. Be sure to adequately insure your shipment for its replacement value. Packages without proper authorization may be refused. US customers, please contact Axia Technical Support at +1-216-622-0247. For customers outside the US, please shipping to our European Repair Centre via the distributor, in order to avoid having to deal with export paperwork. If there is a need to send direct to Axia, contact Support beforehand to log the incoming repair and for assistance with export documents.

## After Sales Modifications

Please be aware that any modifications other than those made or approved by Axia Audio, may invalidate the console's warranty. This includes changes to cabling provided by Axia and variations to the recommended installation as detailed in Axia documentation.

Modifications to this equipment by any party other than Axia Audio may invalidate EMC and safety features designed into the equipment. Axia Audio can not be liable for any legal proceedings or problems that may arise relating to such modifications.

## Credits...

Kudos to the entire Quasar Surface and Engine Teams, for their tireless work, incredible perseverance and commitment to this project. Many, many thanks to: Serhiy Borisov (you'll be always in our hearts!), Sergey Eremeenkov, Dmytriy Gritsay, Steve Kiffmeier, Andris Kalejs, Oleg Krylov, Gunta Lazdina, Gints Linis, Traian Mohan, Andrew Rogovenko, Normunds Veselis. There wouldn't be any Quasar without all of you guys, you have my respect.

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Many Thanks to Dan Bays for adding all that cool stuff into Pathfinder that makes Quasar so versatile, and to Maciej Slazpka for providing us the hooks to control the best IP intercom system on this planet.

And last but not least... Ojigi to "Sir" Derek Pilkington for his continued trust in our team, and a very large Grazie goes to our guru, and Padrino, Frank Foti, and the whole of Telos, for the opportunity they gave us!

# We Support You

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## By Phone / Fax

- You may reach our 24/7 Support team anytime around the clock by calling +1-216-622-0247.
- For billing questions or other non-emergency technical questions, call +1-216-241-7225 between 9:30 am to 6:00 PM, USA Eastern time, Monday through Friday.
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## By Email

- Technical support is available at **support@telosalliance.com**.
- All other questions, please email **inquiry@telosalliance.com**.

## Via World Wide Web:

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## Register Your Product

Please take a moment to activate your coverage online at <http://telosalliance.com/product-registration/>.

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# Creating the Most Exciting and Engaging Audio Experiences Imaginable

## **Congratulations on your new Telos Alliance product!**

The gang here at Telos is committed to shaping the future of audio by delivering innovative, intuitive solutions that inspire our customers to create the most exciting and engaging audio experiences imaginable.

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**The Telos Alliance**

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# Introduction

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## About This User Manual

With the design of the Quasar console, our goal was to provide you with the most ergonomic and efficient human-machine interface possible, for addressing any type of On Air workflow: from fast-paced self-operated radio programs to complex broadcasts where board operators must multi-task without error. As such, Quasar brings a whole new level of control and sophistication to the broadcast studio, while providing very intuitive operation and fast access to controls.

It is not an easy mission to write a fully descriptive User Manual that can be as easy to read for the novice operator as it is informative and detailed for the experienced console engineer. In fact, the purpose of this Manual is to be a reference for operators of any level, as well as for those who are responsible for the advanced configuration of the system.

This document is not intended for the experienced Technician, or System Integrator, who is looking for all the technical information required to plan for a Quasar installation.

The chapters in this manual are grouped in four main categories:

- Startup
- Operation
- Configuration
- Appendixes

It begins with the basic setup information, the Quick Start Guide (the same we're shipping with the console as a printed document), then continues with chapters dedicated to the Console operation and configuration, then to the Engine configuration. The last pages are dedicated to Appendixes and Technical specifications.

We have clearly separated the operational concepts from the configuration instructions and avoided mixing the two to make reading easier for operators, and searching for configuration parameters faster for technicians. But we encourage operators not only to dive into the first few chapters, dedicated to the console Operation, but also to take a look at the Configuration section, to get a broader view of the console workflow and a deeper understanding of the whole system, its user interface and control logic.

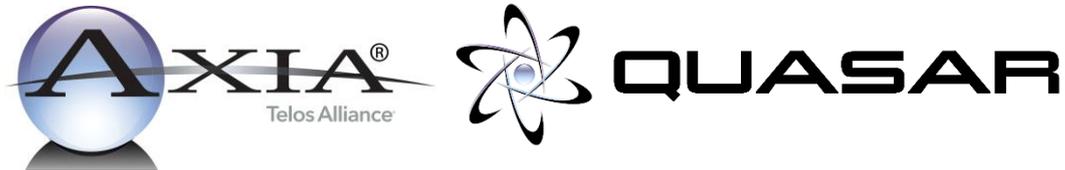
Thank you for purchasing the Axia Quasar. We hope you will enjoy using it, as much as we enjoyed designing it for you!

**Luca La Rosa**  
Senior Project Manager  
Axia Quasar Lead Designer

# Startup

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## Quick Start Guide



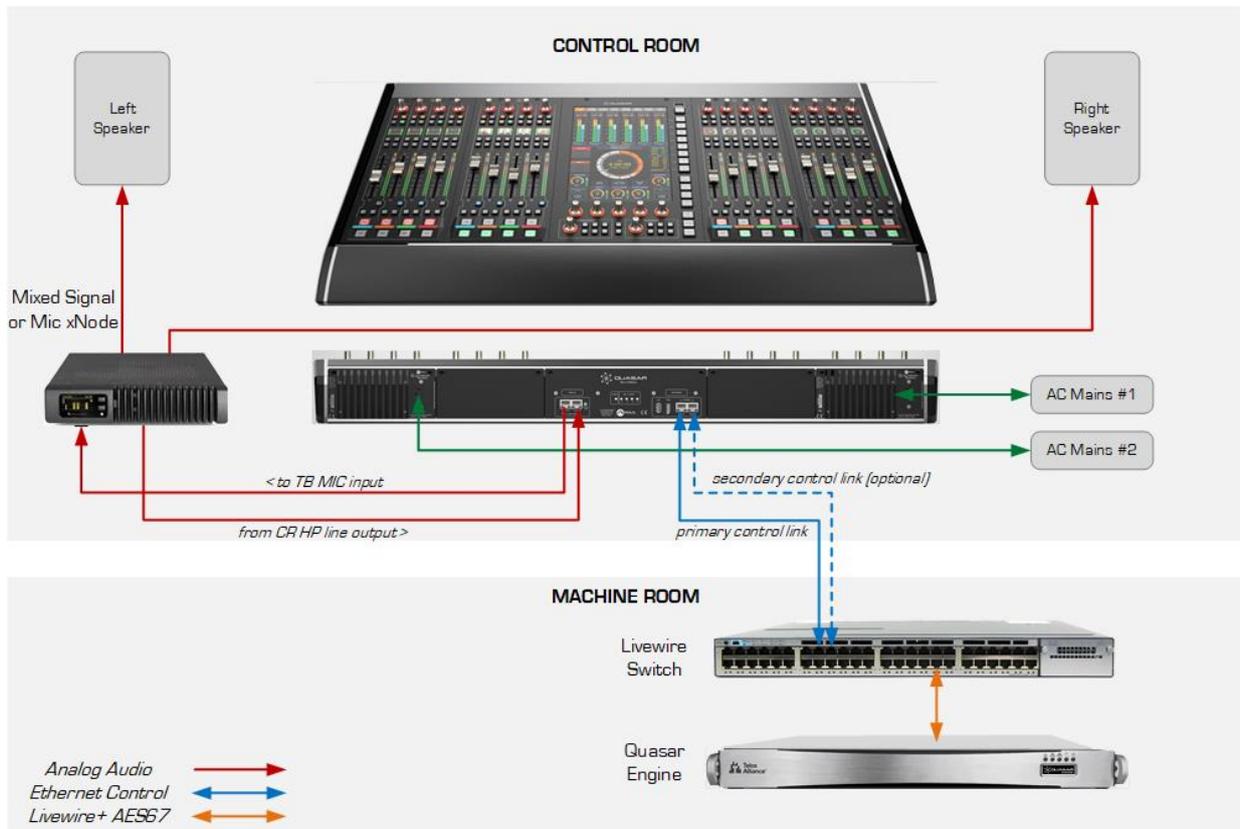
## Welcome to Quasar XR and Quasar SR

Setting up a networked audio mixing console is challenging, but the following pages are intended to make it as easy as possible and help the busy engineer get up and running quickly.

### **This Quick Start Guide Assumes a Few Things:**

- Basic networking skills and a familiarity with networking terminology
- Familiarity with Audio over IP and Axia Livewire products
- An approved network switch properly configured for use on a Livewire network

## Quasar System Anatomy



Example of simple Studio configuration. Connection of the optional TBP-IO module is included in this drawing.

## Required Equipment, Cables, & Accessories:

The following items are necessary to complete the installation and configuration.

### Required items that are included:

- Quasar Control Surface
- Power (mains) cords
- CAT5 Ethernet cable

### Required items that are NOT included:

- Quasar Engine platform
- Gigabit Switch configured for use on an Axia network
- PC with access to the Axia network

**Note** - Only the Quasar Engine can be connected to the Quasar Surface. Other mixing engines such as the fanless Axia Studio Engine, or Axia Powerstation, cannot be used with Quasar.

## Steps to complete in order to get audio from the console:

1. Network Switch Configuration
2. Quasar Surface Connections
3. Quasar Surface Network Configuration
4. Quasar XR Modules Discovery and Configuration
5. Quasar SR Modules Configuration
6. Quasar Engine Installation and Network Configuration
7. Checking Connection to the Quasar Engine



1. Quasar Engine Audio Outputs Configuration
2. Quasar XR Surface Layer Configuration
3. Creating and Configuring Input Sources
4. Assigning Sources to Input Channels
5. Program Assignment and Monitoring



## Network Switch Configuration

Audio over IP requires the use of an approved and properly configured switch. Not all switches are capable of handling the traffic generated by AoIP, and an improper configuration can lead to a flood of traffic on the network.

Details of switch configuration are beyond the scope of this Quick Start Guide, and it is assumed the person responsible for installation has the necessary configuration skills or the resources of an IT engineer.

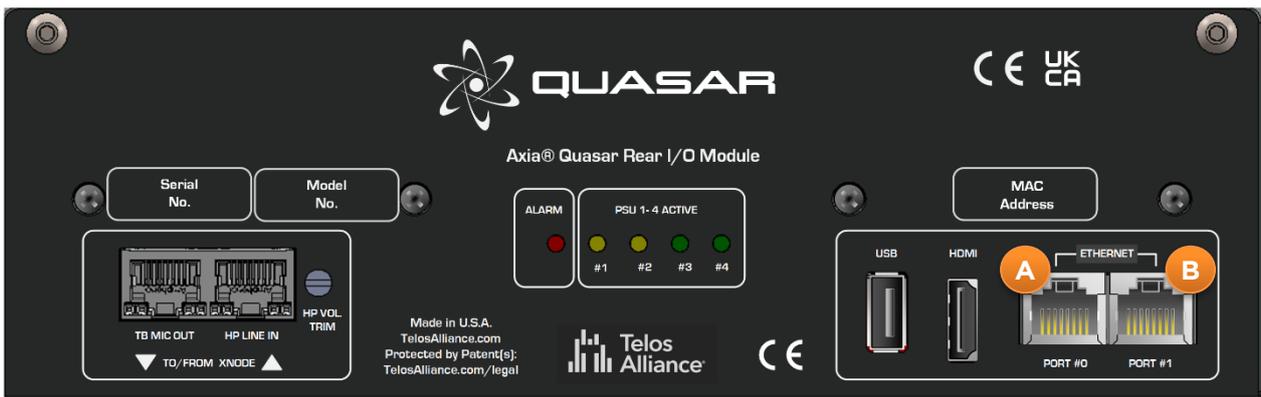
A list of Telos Alliance-approved Ethernet switches is available on our website, at the following url: <https://www.telosalliance.com/Axia/What-Ethernet-Switches-has-Axia-Approved> or you may submit an online support request at the following url: <https://www.telosalliance.com/support-request>.

# Quasar Surface Connections

## Network Connections

Connect an Ethernet cable to the Primary network port (A), labeled “PORT #0” on the console’s rear I/O board. Connect the other end of the cable to a Gigabit port of your network switch.

The Secondary network port (B), labeled “PORT #1”, can be used to create an optional redundant network connection. Doing so will require a special configuration of the switch ports as outlined in the Quasar Installation Guide.



## Mains Power Connections

Use the provided AC power cable(s) to connect the surface’s PSU module (C) to a grounded power receptacle. If you have more than one power supply in your system, we suggest connecting them to different power sources for redundancy and highly recommend the use of a UPS (uninterruptable power supply) for each source.



**Important** - The Quasar Surface is grounded through the power cable and therefore relies upon proper grounding of the circuit providing power. There is no separate chassis ground.

## Quasar Surface Network Configuration

Quasar Surface and Quasar Engine ship with the following default IP addresses:

- Surface (Master Module) - 192.168.2.10
- Surface (XR Modules) - 192.168.2.11, 192.168.2.12, etc.
- Surface (SR Modules) - no IP address required
- Engine - 192.168.2.100

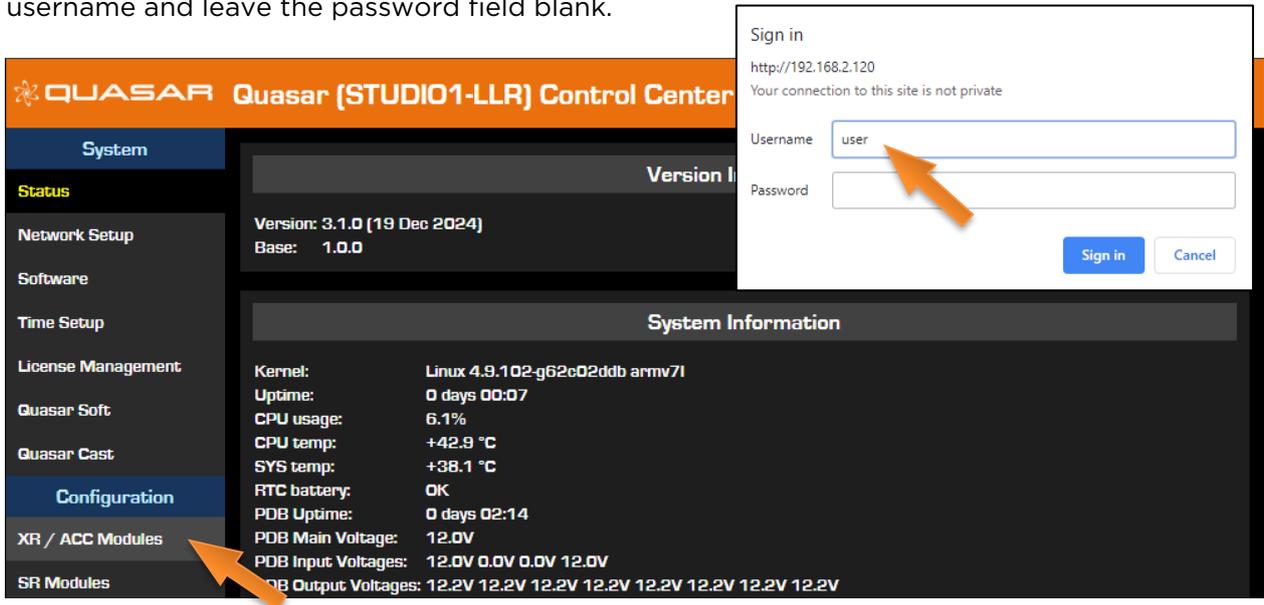
Should you need to change the default settings, or if the warning message: “No Connection to Engine” should appear on the surface’s home screen (just above the clock) once the system boot is completed, then you will need to enter new network settings manually.

- Access the System Setup page on the Master Touchscreen Module, by pushing and holding the “Monitor Options” key (labeled “MON OPT” and located between the two Control Room volume pots) for 5 seconds.
- Enter the desired IP address and Netmask for the Surface and Engine using the touchscreen or the corresponding rotary encoders beneath the display. A Gateway is optional but typically not required when the console is part of a closed network.
- Press the **Save & Reboot** button. A confirmation dialog will appear.

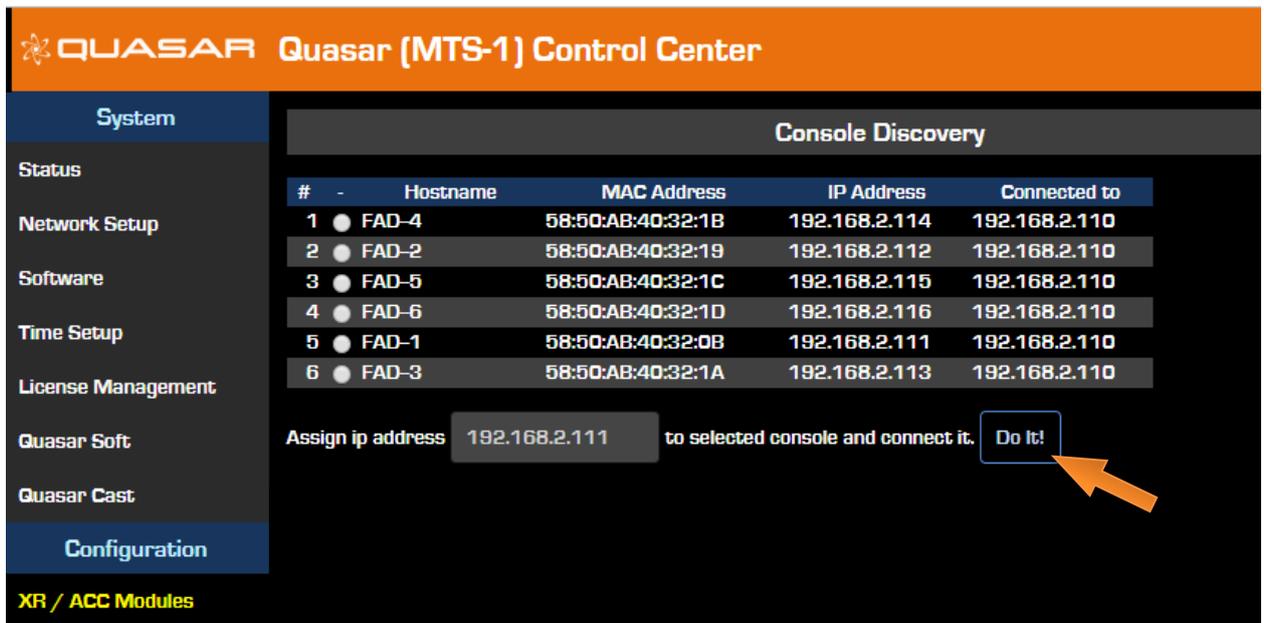


# Quasar XR Module Discovery & Configuration

Each fader module on the Quasar XR Surface has a unique IP address to permit configuration via web GUI. Begin by launching a web browser and entering the IP address of the MTS, then click on the **XR / ACC Modules** menu. When prompted for authentication, enter “user” for the username and leave the password field blank.



All connected fader modules will be listed along with their MAC Address and IP Address. If the pre-assigned IP addresses don't fit into your network scheme, you can change them here by selecting a module's radio button, entering the new IP address, and clicking the “Do It!” button.



Once all modules have been set, check the access to each module by entering its IP address in your browser, or by clicking directly on each of the addresses listed in the Console Discovery menu.

## Quasar XR Surface Layer Configuration

You can skip this paragraph if you're setting up a Quasar SR console.

### Understanding Layers

Layers are useful when the number of DSP input channels you need to access exceeds the number of faders or physical channel strips on your Quasar surface. Since any fader on any module can access any input channel on the Quasar engine, you will need to assign four input channels to each of the four available layers on every Fader Module.

### Configuring Layers

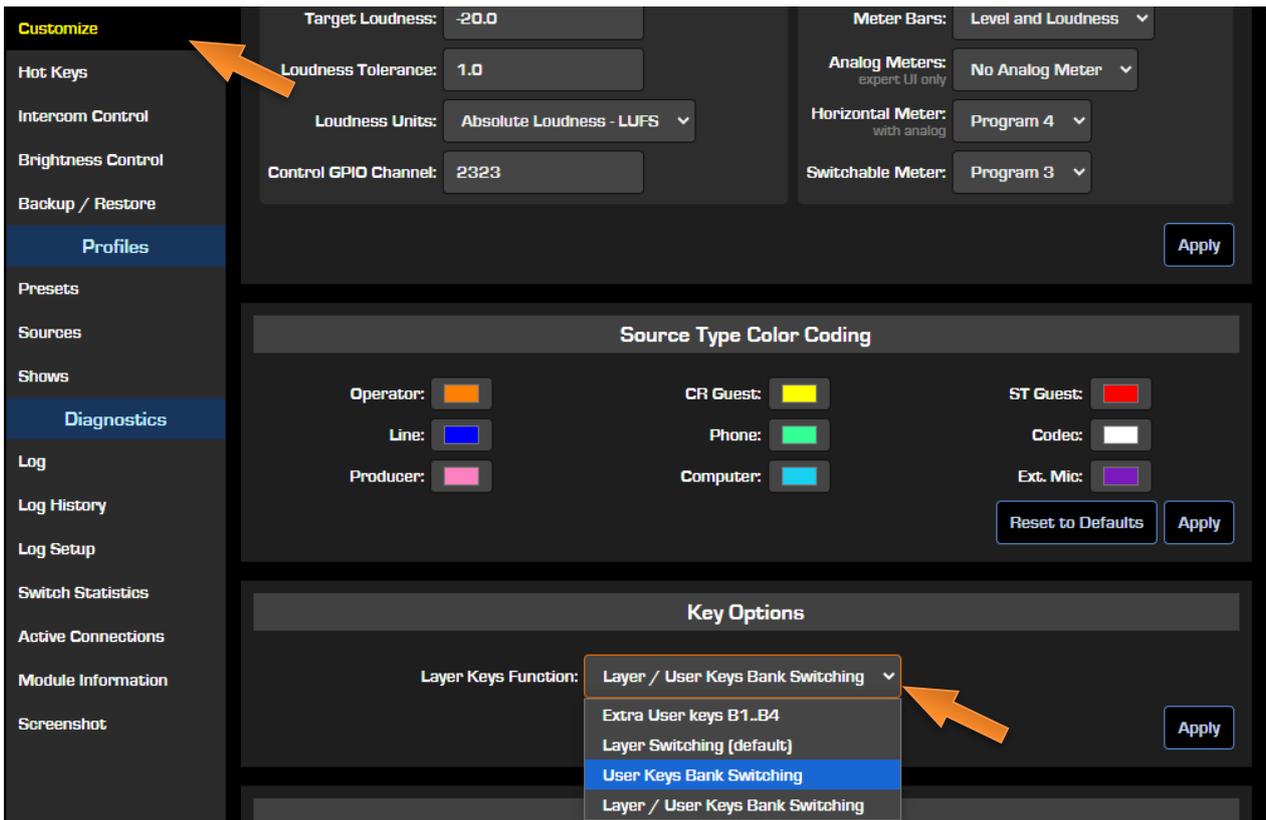
- Using a PC connected to your studio network, launch a web browser and enter the IP address assigned to the first (left-most) fader module. When prompted for authentication, enter the username "user" and leave the password field blank.
- In the Configuration menu, click on Layers Setup.
- Enter the desired input channel number for each layer of each physical fader; the **+4** and **-4** buttons can be used to quickly increment or decrement channels in groups of four.

	Fader 1	Fader 2	Fader 3	Fader 4		
Layer 1:	1	2	3	4	+4	-4
Layer 2:	13	14	15	16	+4	-4
Layer 3:	25	26	27	28	+4	-4
Layer 4:	PGM1	PGM2	PGM3	PGM4	+4	-4

The table shown in the picture above shows which of the 64 input channels available in the Quasar Engine, will be loaded on the four channel strips each time the LAYER 1, 2, 3, 4 keys are pushed on the Master Module.

### Disabling Layers

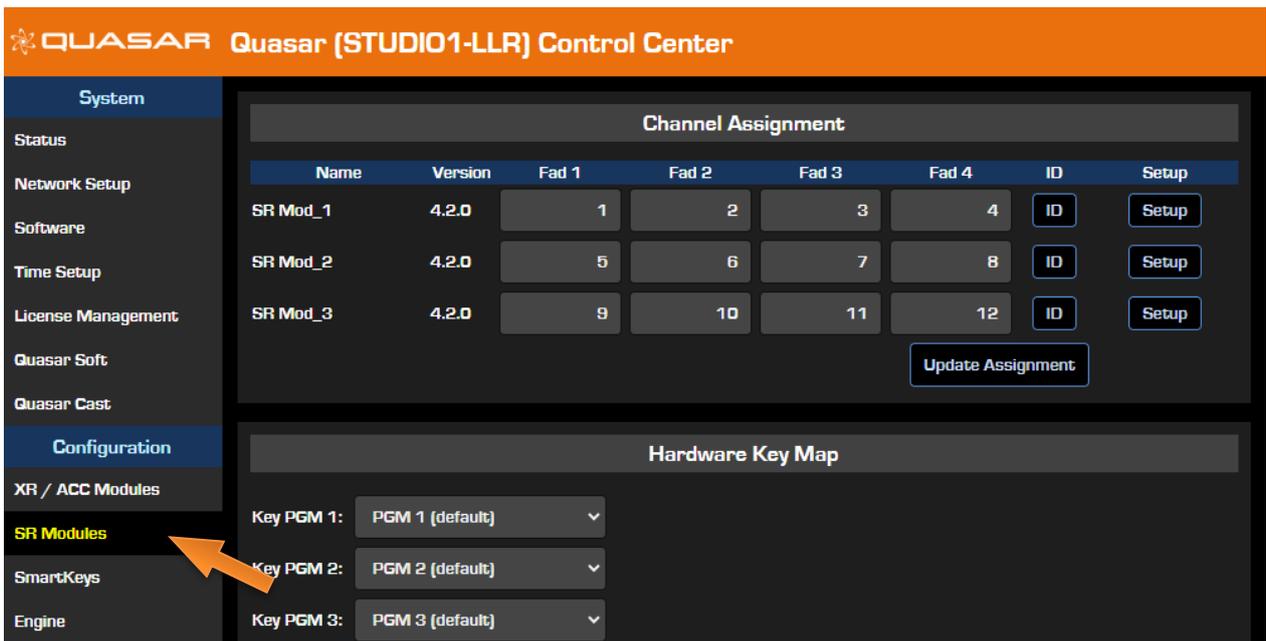
If you have a Quasar SR console or simply no need for Layers, you can disable them by navigating to the Master Module web UI and selecting the **Customize** menu. In the Key Options section, select the "User Keys Bank Switching" or "Extra User Keys b1 ..B4" options from the **Layers Keys Function** dropdown.



This will switch off the Layer buttons on the Master module.

## Quasar SR Module Configuration

Quasar SR Fader modules have no individual IP address and therefore do not need to be discovered. They are always connected to the Master and automatically linked to it. Launch a browser and enter the IP address of the MTS module, then click on the **SR Modules** menu.



Here you can assign Input channels of your Quasar Engine to the SR modules' faders. Simply type the desired Input Channel number in each of the four Fader fields for each of the SR Modules listed in the Channel Assignment section. Then press the "Update Assignment button".

# Quasar Engine Installation & Network Configuration



## Physical Installation

The Quasar Engine is designed to be installed in a climate-controlled environment where any noise generated from its fans will not be an issue, such as a machine room. Please refer to the Quasar Installation Guide for complete installation details.

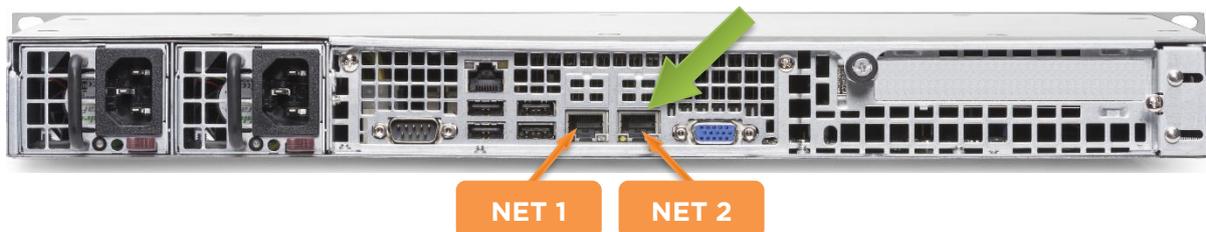
## Mains Power Connections

Use the provided AC power cables to connect the Engine to grounded power receptacles. We suggest connecting each supply to different power sources for redundancy and highly recommend the use of a UPS (uninterruptable power supply) for each source.

**Note** - The Quasar Engine is grounded through the power cable and therefore, relies upon proper grounding of the circuit providing power. There is no separate chassis ground.

## Network Connections

Connect a network cable to the **NET 2** port (the right-most port when looking at the rear panel). Connect the other end of the cable to a Gigabit port of your network switch.



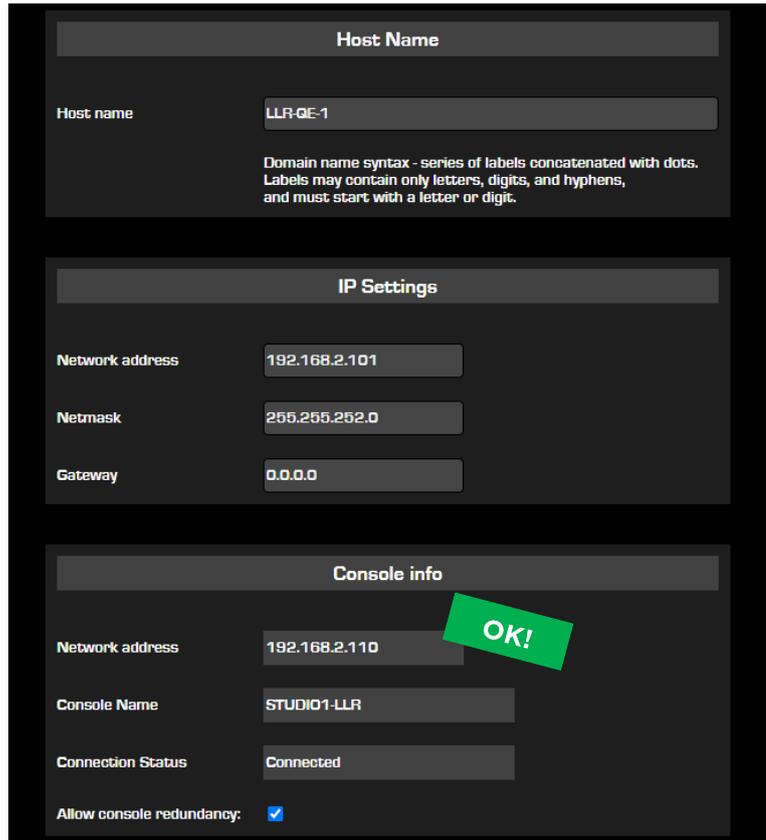
Quasar Engine ships with a default IP address of 192.168.2.100 and a netmask of 255.255.255.0. This address can be changed using the web GUI once it is connected to the network. If the IP address must be changed before connecting the Engine to the network, follow the steps below:

- Connect a VGA monitor and a USB keyboard to the appropriate rear panel ports; the Basic Setup screen will be displayed.
- Select “1-LAN Settings” with the numeric keypad or up/down arrow keys, then press Enter.
- Select “2-Configure Network Settings”, then press Enter.
- Type the new network settings with the numeric keypad, then press Enter.
- The Engine will prompt you to reboot; select “OK” to reboot with the new settings.

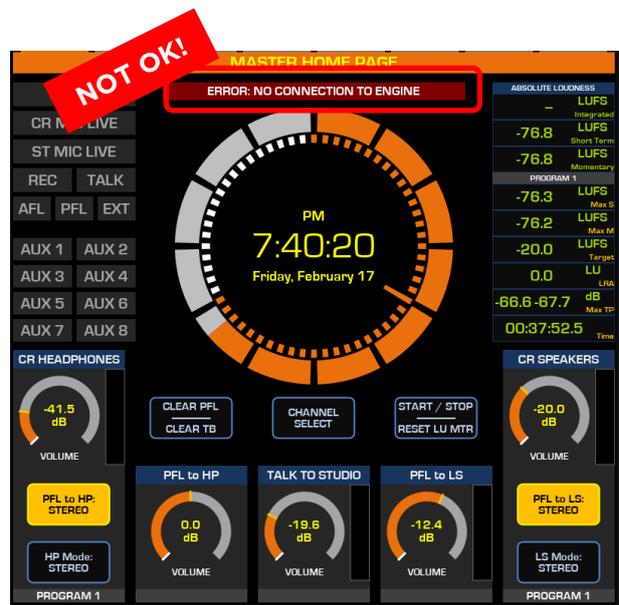
## Checking the Connection

Using a PC connected to your studio network, launch a web browser and enter the IP address of the Quasar Engine. When prompted for authentication, enter “user” in the username field, and leave the password field blank.

Navigate to the Network Setup menu, then to the “Console Info section”. Verify that the IP address of the Quasar Master Touchscreen (MTS) module appears and that the connection status shows as “Connected.”



Next, verify that there is no error message above the clock on the MTS home page.

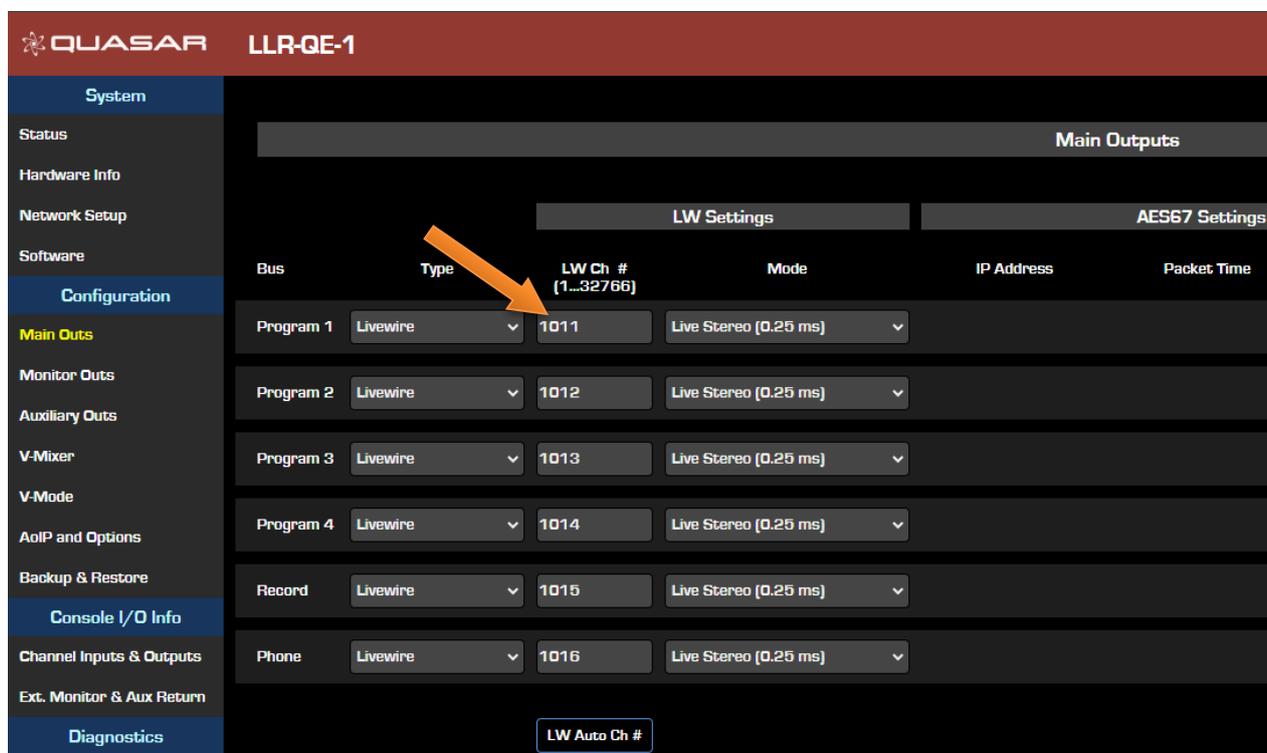


# Quasar Engine Audio Outputs Configuration

Quasar Engine audio setup begins with configuring the audio outputs.

## Program Outputs

- Using a PC connected to your studio network, enter the IP address of the Quasar Engine to display the web GUI.
- From the Configuration menu, select Main Outs.
- Enter the channel numbers you planned for the Quasar Engine (Livewire Sources) outputs.
- Enable the streams you need to be active on the Engine
- Click the **Apply** button.



Note - We recommend adopting a numbering scheme for Channel ID's that uses the first two or three digits of the last octet of the Quasar Engine IP address. In the example above, the Engine has an IP address of 192.168.2.101, and the Channel Numbers are 1011, 1012, 1013, etc. This will make it easier to identify the channels when browsing a large network.

## Monitor Outputs

In order to hear audio through your speakers and headphones, assign the CR Monitor and CR Headphone streams to the Destinations (outputs) of the xNode connected to the speakers and headphone amplifier.

Bus	Channel # (1...32766)	Stream Type	Status
CR Monitor Direct	1040	Live Stereo	OK
CR Monitor	1041	Live Stereo	OK
CR Headphones	1042	Live Stereo	OK
PFL/AFL	1043	Live Stereo	OK
Talk to CR	1044	Live Stereo	OK
Guest Headphones	1045	Live Stereo	OK
Studio Monitor	1046	Live Stereo	OK
Talent Headphones	1047	Live Stereo	OK
Talk to External	1048	Live Stereo	OK

Auto Ch #

Apply

## Creating and Configuring Input Sources

In order for sources to appear as Channel inputs on the Quasar Surface, they must first be created and a source type specified.

- Using a PC connected to your studio network, enter the IP address of the Quasar Console to display the web GUI.
- Navigate to the Profiles menu, then to Sources.
- Click **Create New Source Profile**.
- Select the Source Type (A). Enter the Source Name (B).
- Click the browse button to the right of the Primary Source field (C), and select the desired source from the pop-up list. Click **Ok**.
- Repeat these steps for additional sources which will then appear in the Channel Input > Sources menu of your Quasar Surface.

The screenshot displays the 'Source Profile "LUCA"' configuration interface. The left sidebar shows the navigation menu with 'Sources' highlighted. The main content area is divided into three main sections: 'Source Settings', 'Stream I/O Settings', and 'Source Logic'. The 'Source Settings' section includes fields for 'Source name' (LUCA), 'Source name override' (Show sourcename), 'Source type' (Operator Microphone), 'Source color', 'Input signal phase' (Normal), 'Signal mode for Record bus' (Stereo), 'Input trim' (+0.0 dB), 'Pan setting' (0 %), 'Audio delay' (0 ms), and 'Automix parameters' (AM Weight 0, Automix ON). The 'Stream I/O Settings' section includes 'Source Inputs' and 'Source Outputs' tables. The 'Source Inputs' table has columns for Input, Type, Livewire Channel / IP Address, Format, and Channel Map. The 'Source Outputs' table has columns for Output, Type, Livewire Channel / IP Address, Packet Time, Stream Type, and Channel Map. The 'Source Logic' section includes 'Backfeed mode' (Disabled) and 'Fader mode' (Normal). An orange arrow points to the 'Sources' menu item in the left sidebar.

For more information on how to create and configure all your Network sources, please refer to Configuration Chapter 2 of this manual: [Source Profiles](#).

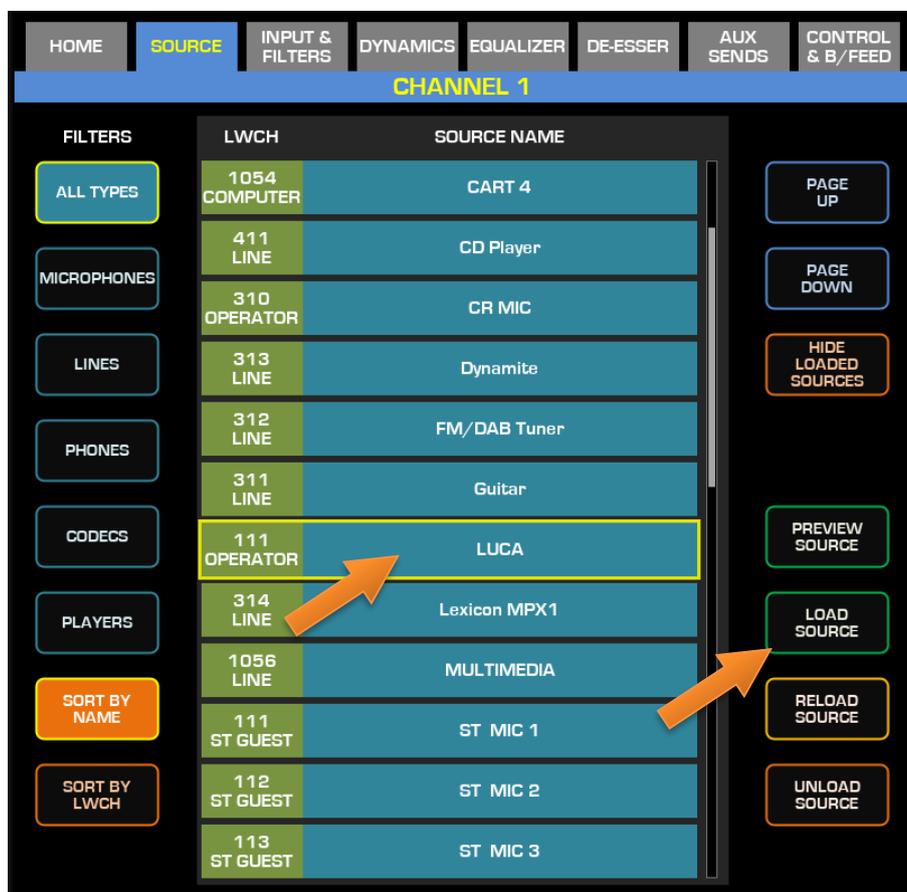
## Assigning Sources to Input Channels

Once Sources have been created, they must be assigned to input channels on the Quasar Engine and then saved to a Show Profile. Show Profiles can be created in two ways:

- Sources can be assigned directly from the console and the resulting configuration captured as a Show.
- A Show Profile can be created from the Web UI and sources can be assigned to channels

### Creating a Show Profile from The Console

- Push the top encoder of the channel strip to which you want to assign a source, then select the **SOURCE** tab in the Master Touchscreen module.
- All configured and available sources on the network will appear; it may be necessary to use the touchscreen to scroll down to show more sources.
- Push the **Load Source** button.



**Note** - Inactive sources (that is, those generated by devices that are disconnected from the network or turned off) will not be detected by Quasar and will therefore not appear in the list.

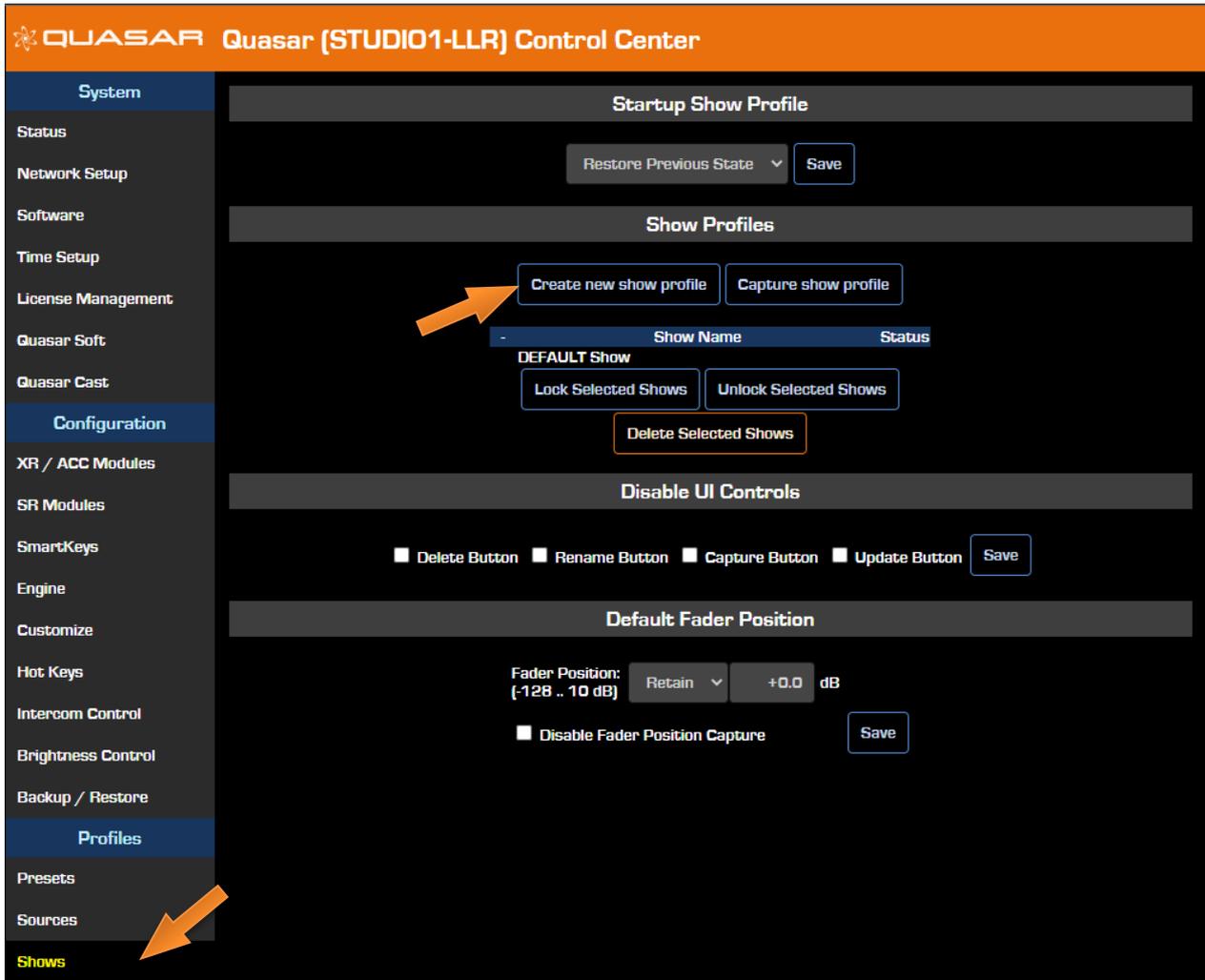
Once you have assigned all sources to the channels, press the **PROFILES** key on the top left corner of the Master module, and select **Capture Show** from the Touchscreen UI.

You will be prompted to enter a name for your new Show Profile. Type a name and press **Ok**.

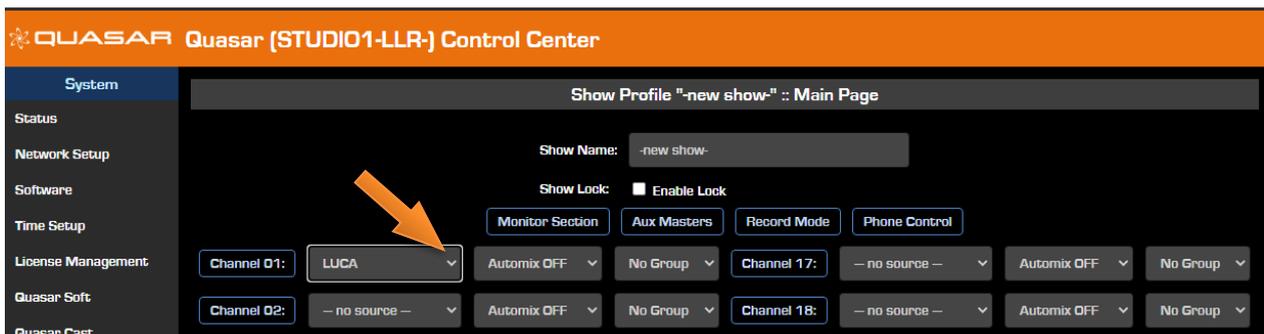
## Creating a Show Profile from The Web UI

Using a PC connected to your studio network, enter the IP address of the Quasar Surface in a browser to display the web GUI.

- Navigate to the Profiles menu, then to **Shows**.
- Click on the **Create new show profile** button.
- Enter a name for the new Profile and confirm.



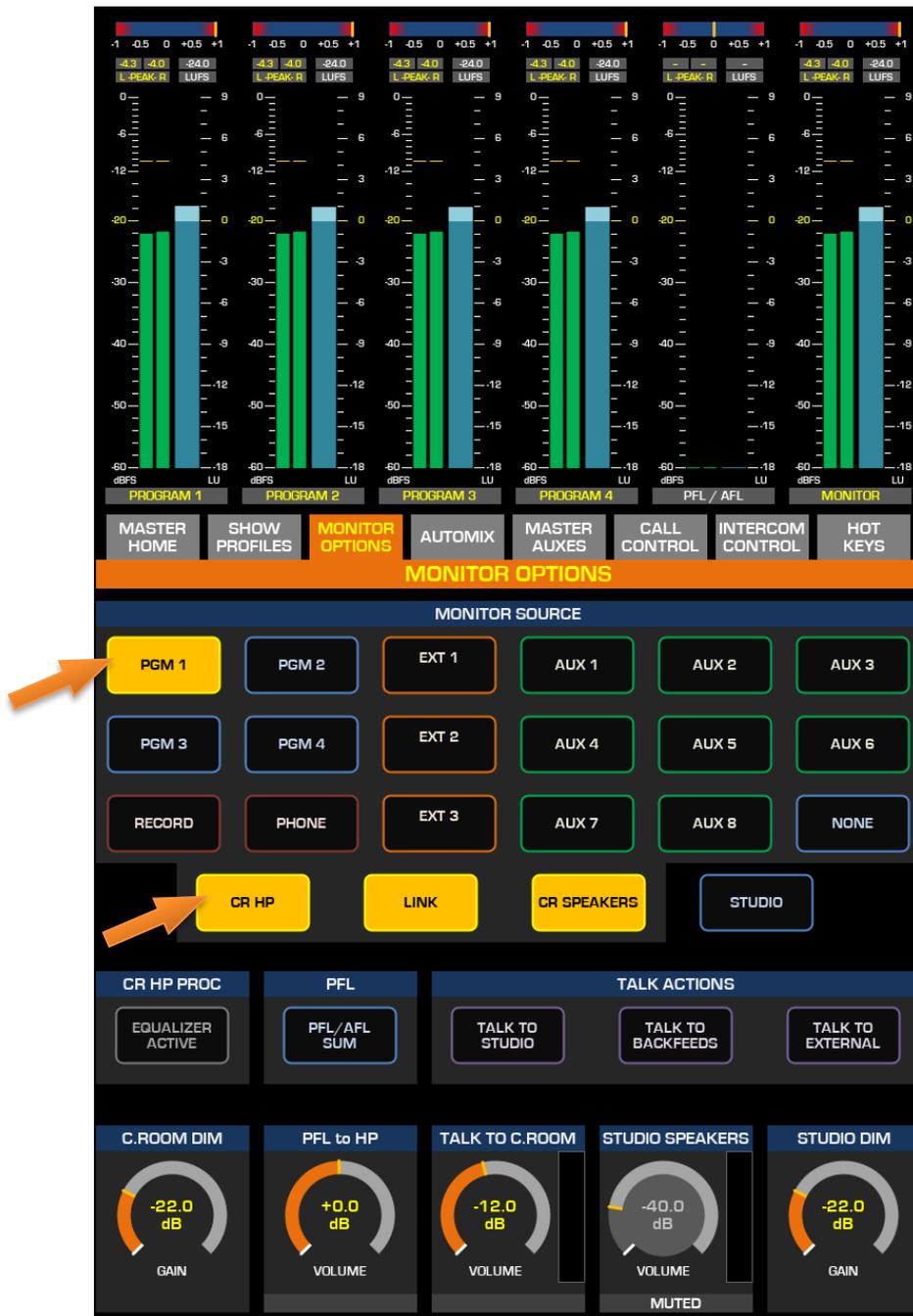
A new Show Profile configuration page will appear where Sources can be assigned to input channels from a drop-down menu.



## Program Assignment & Monitoring

The output of each fader strip must be assigned to a Program output.

- Press one of the **PGM** keys on the fader strip; it will illuminate to indicate that it has been assigned.
- Press the **ON** key at the bottom of the fader strip and move the fader up.
- Your meters should be active as shown below.



To hear the audio, make sure you have selected the same Program assigned above as source to your CR Headphones or Speakers, using the controls on your Monitor module, and that the volume is set at an appropriate level.

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# Operation 1 – System Overview

## Control Surface Overview

Quasar was designed with control in mind. The needs of each facility are unique, and the possibility to customize the Station's workflow is important. Since different Stations have different workflows, we've made it possible to access the most important functions in multiple ways. This way, the board operators can work the way they want – not the way somebody else *thinks* they should work!

This chapter will start with an overview of Quasar's control surface (a Quasar XR, in this example). Then, we'll have a detailed look at each of its control modules.



1. MTS-MON Module: Main Master and Monitor Control Module.
2. XR-4FAD Modules: Expert Motorized Fader Modules.
3. Rear PSU Modules: Power Supply Units. Can be up to four, redundant.
4. Rear I/O Module: Where Network, USB, HDMI and Audio connections are located.



## Terminology Used in This Manual

This manual adopts the following object definitions and associated text formatting when describing the different controls and objects available in the control surface and Web UI:

- **KEY** : Mechanical hardware pushbutton on the control surface.
- **Button** : Graphical pushbutton, displayed in the Touchscreen GUI, or in the Web UI.
- **Encoder**: Rotary knob (endless turn), with integrated mechanical pushbutton.
- **TAB** : Touchscreen view. Selectable as menu page, with a “tab-style” button.
- **Page**: Web UI page, link to pages or screens
- “Label” : Static text window displayed in the Touchscreen, or drop-down menu option available in the Web UI.

## The MTS-MON Master Module

The MTS-MON Module (or “Master Monitor Module”) is the main control module in the system. This module offers a combination of fixed-function and soft hardware keys (that change function based on the user operation), and touch display controls.

1. **Touchscreen:** This is a 12.1” Industrial-grade display with a robust protective glass. The graphics on this display change according to the operations being triggered. The next section of this chapter will cover this item in more detail.
2. **Touch sense encoders:** These five high quality optical rotary encoders are soft driven: their functionality changes based on the UI tabs selected by the operator.

The display will dynamically change graphics to indicate the function of the encoder. In some cases, the touch sense function will trigger a *pop-up*, providing a control and feedback when the touchscreen is not displaying the current functionality of the rotary encoder.

In some screens, pushing these encoders will reset the corresponding parameter to its default value.

3. **CR Headphones section:** This section is dedicated to the Control Room (operator) headphone.

The 6 keys are used to select the more common sources that need to be monitored. A rotation of the encoder will adjust the Control Room Headphones level.



When the rotary encoder is touched, and a UI tab other than the MASTER HOME tab is selected in the UI, a small pop-up window will appear on the display (1), showing the Headphone Volume information.

4. **CR Speakers section:** The same as above (CR HP) where six keys and a touch sense rotary encoder are dedicated to the main CR monitor Loudspeakers.
5. **MON OPT:** This key gives the operator access to more control options over the monitoring. Pressing this key, changes the display view (1) and provides additional sources to be monitored, control of the dim values, dedicated talkback functions and activation of CR Headphone 4-Band EQ.

**LINK:** Located between the two CR Monitor sections, this key links their source selection: selecting a mix in the CR Speaker section will also select that same mix in the Headphone section.

6. **Fixed Function Keys:** These keys provide direct access to the most useful functions required to operate the console and the navigate the touchscreen UI:

The **HOME/CH.SEL** key returns the display to the MASTER HOME tab (starting point of the navigation) when pressed momentarily. A press & hold of this key will make the Channel & Bus Select screen appear. Channel modification options are covered later.

The **RECORD** key will enter into (or exit) the Record Mode. The Record Mode is a customizable “Macro”, needed for recording workflows. Its operation is described later on.

Currently the **TALK** key will trigger the Talk to Studio function.

7. **Layer / User Bank keys:** Quasar offers 4 surface Layers, and 4 banks of User Keys. These keys enable the operator to quickly select each of the layers, and/or each of the User Banks, depending on the options selected in the console Web UI.

Layers can be configured by accessing the Web UI built in each fader module, while the four Banks A, B, C and D of eight user-programmable keys must be configured via Pathfinder: up to 32 console User Keys can be assigned to any function programmed in PF, to let users address the customization needs of the facility.

Layers are not available in Quasar SR consoles. The Layer keys can be completely disabled in the MTS Module Web UI, by browsing the Web UI *Customize* page.

8. **USER 1-8 keys:** Eight user-programmable keys with capacitive touch sense are available for adding custom functionality. With the use of Pathfinder, these keys can be programmed to execute virtually any action within the facility. The touch sense function triggers the pop-up of “dynamic labels” on the display: the simple touch of each key will reveal its label before the key is actually pressed. Pathfinder can change the text in the pop-up and change the color of the key (RGB color illumination). Please see Chapter 3 of this manual for a more detailed description on how to use these keys.

**Warning -** The USER 1-8 Keys use special key caps that cannot be disassembled by users to insert custom labels. Attempting to do so would damage the Touch Sensor built inside the switch. Please be aware that this would generate an out-of-warranty support case.

9. **Profile key:** A fixed-function key which shows a list of the defined Show Profiles onto the display for user selection. Press & hold this key for 5 secs to Reload the current profile back to its *Defaults*, (originally saved settings).

10. **USB DATA port:** For future use.

**Warning -** Do not use this port for charging phones! A damaged cable or phone could seriously damage the console!

## The Touchscreen User Interface – Expert UI Mode

The Quasar MTS-MON module includes a Touchscreen with an interactive User Interface. Its workflow is based on the following principles:

- When none of the encoders at the top of each channel strip are selected (the LED rings are all off), the UI is displaying the MASTER control menu. This menu is characterized by having orange-colored tabs
- When any one of the encoders at the top of each channel strip is selected (the LED ring illuminates), the UI is displaying the CHANNEL control menu. This menu is characterized by having blue colored tabs.

Quasar comes with its MTS-MON module set to display the **Expert Mode** UI. This is the default UI mode.

The display is divided in three areas:

**1. Meters Area:** The top area is the meter area. Within the Master Menu (orange tabs) this area always shows the Main Meters view. If a channel is selected, and one of the processing tabs (blue) is accessed by the operator, this area is switched to display the graphs associated to these pages.

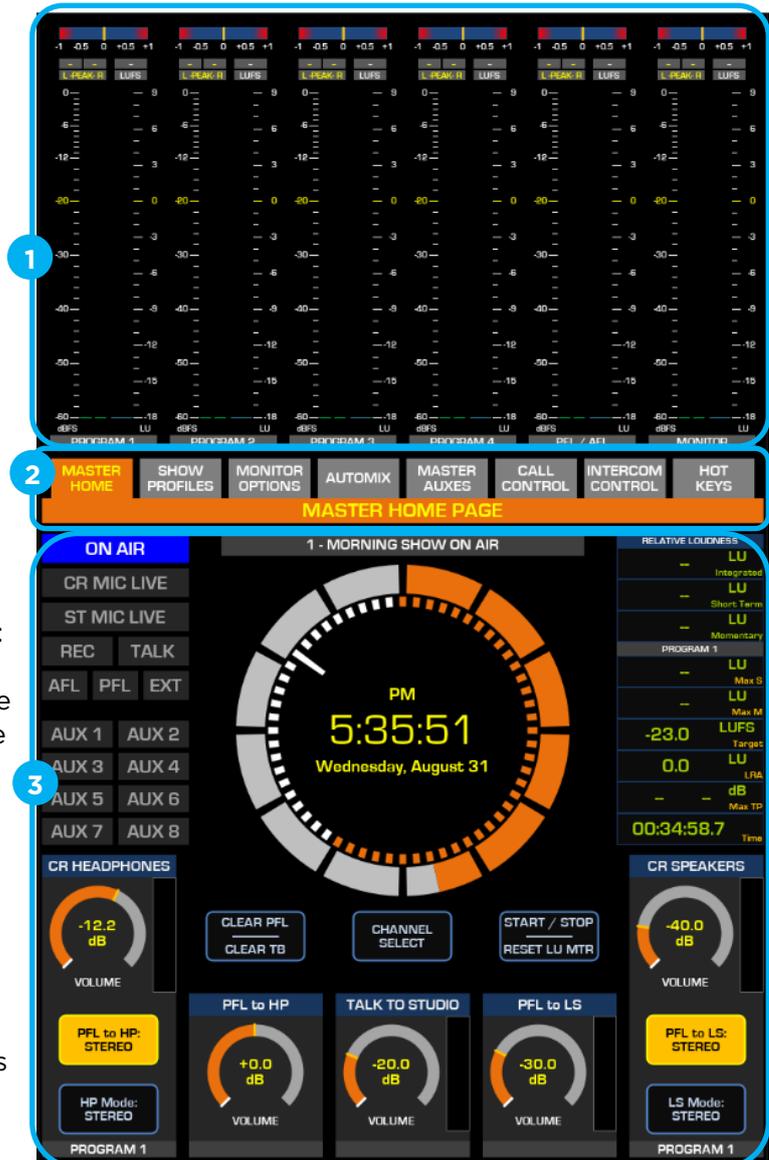
**2. Menu tabs:** The tabs bar is for navigating the various UI pages. The tabs have two colors to help users identify the type of controls: when a tab is selected, that tab will be illuminated in its color while others within the bar are grey. The bottom area will display the items related to selected tab.

**Master tabs** (orange): all tabs in this menu provide access to controls typically found in any console’s Master Module. These controls define functions that belong to the entire desk.

**Channel tabs** (blue): all tabs in this menu provide access to controls typically found in any console’s Channel Strip.

These controls define functions belonging to the single input channel.

**3. Function area:** The bottom area shows the properties associated with the selected tab. The area will be populated with feedback information or control items (buttons, dials, etc.). The remainder of this section will cover various functions available. This area is graphically divided in vertical strips, each aligned with an optical encoder positioned at the bottom of the touchscreen. Touching one of these graphical rotary objects (we call them “Gauges”) associates the Gauge to the Encoder.

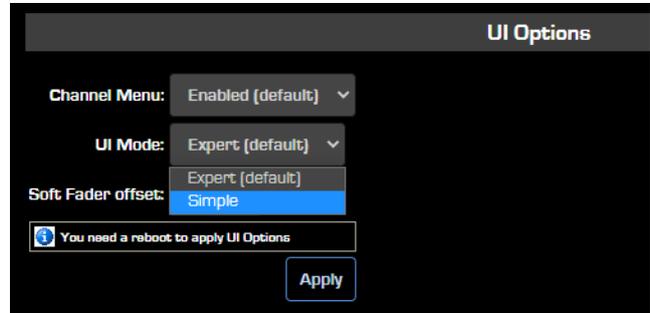


## The Touchscreen User Interface – Simple UI Mode

As an option, you can select a less sophisticated user interface. Particularly useful for those fast-paced self-operated workflows, this view provides access to only the most important parameters, allowing users to find their way around the user interface more quickly.

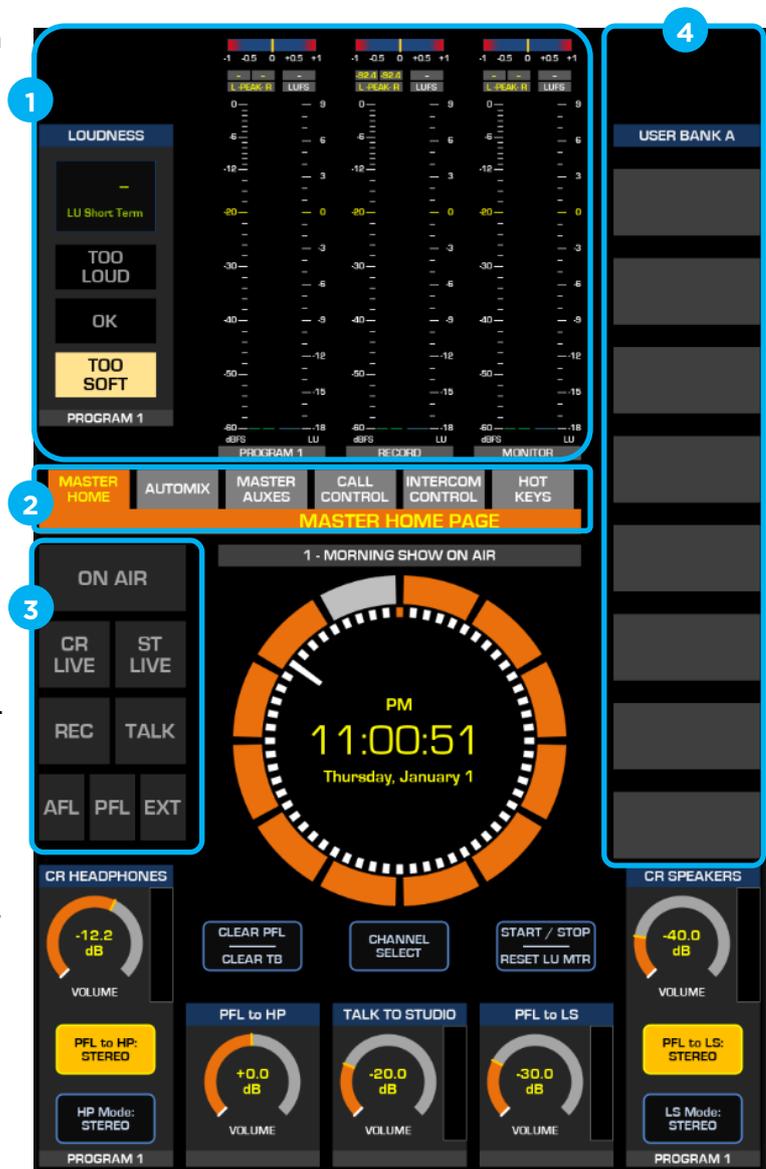
To select this option, navigate to the **Customize** menu of the Master module’s Web UI, and in the **UI Options** section at the bottom of the page, select **UI Mode: Simple**.

**Note - You will need to reboot the MTS module to view the new UI. You can do so from the **Software** menu page.**



The basic principles explained in the previous page apply to the Simple UI as well. The graphics have the following differences:

- 1. Meters Area:** The Main Meter area displays only the two most important meters (PGM1 and RECORD) plus an automatically switched MONITOR/PFL meter. A super-intuitive “Traffic Light” Loudness Meter offers real-time information about the Loudness level of your program, based on the Target Loudness and Tolerance parameters entered in the Web UI.
- 2. Menu tabs:** The tabs bar offers a reduced view, hiding those tabs that can be accessed via the hardware pushbuttons on the module.
- 3. Signals:** The ON AIR, RECORD, TALK and PFL/AFL activation signals are larger and more visible.
- 4. User Keys Labels:** Static labels for the four User Key Banks A,B,C,D, are displayed in this area, instead of the Loudness Counters. In the Expert UI Mode view, these are normally hidden and show up only when the surface of the key is touched by the operator. In this view they are permanently displayed for faster access to the custom functions.



The bottom area and the associated Gauges are the same as those found in the Expert UI Mode.

## The Channel Select Screen

There are three possible ways to select a channel strip in Quasar. You can:

1. Press the **CHANNEL SELECT** button below the clock, in the HOME Tab
2. Press & hold the **HOME --- CH.SEL** key for 1 second
3. Press the encoder at the top of the desired channel strip



The first two will make the Channel Select pop up on the console Touchscreen. From this screen you can select any of the input channels that are available on your Quasar Engine:

- When a Channel is turned ON, the corresponding button will have a Yellow outline
- When a Source is loaded to a channel the corresponding selection button will display the Source Name below the Channel Number.
- If your Engine has a Base License, only the first 16 buttons will be active (blue outline, black fill), allowing you to select channels 1 to 16. The remaining buttons will be disabled, and greyed out in the UI.
- If your Engine has one or more 16-Channel Expansion Licenses, then more buttons will be active, and only those which correspond to unlicensed channels will be left disabled.

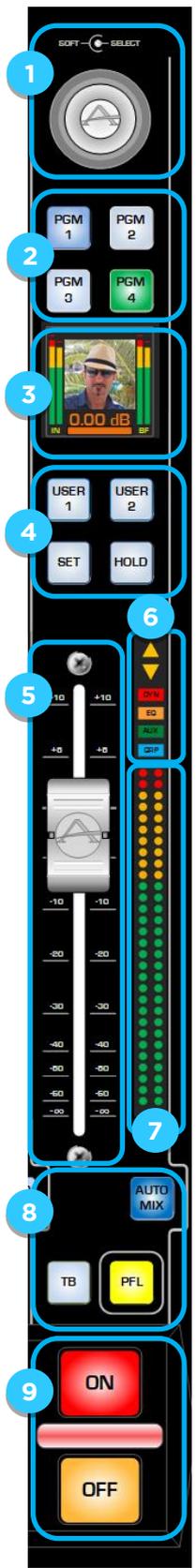
Press the **CANCEL** button in the bottom right corner of the screen, if you want to return to the same UI page you were in, before the Channel Select screen was displayed.

Or you can simply press the **HOME** key to return to the MTS HOME Tab.

## The XR-4FAD Fader Module

The XR 4-Fader module is a fully-featured control module that connects to the Master through an internal Ethernet network. The XR 4-Fader module is composed of four channel strips, each with the following control elements:

1. **Channel Options encoder:** Can respond to touch, press, and rotate. A short press can trigger the Channel select which drives the master section into Channel options (blue tabs). The rotation function will depend on the configuration of the source profile and can be Fader-trim control, Pre-Amp gain control, Automix control, or Source select.
2. **PGM 1-4 keys:** Main program mix assignments are the default configuration for these 4 keys which assign the channel to feed the designated mix bus.
3. **TFT Color display:** Provides feedback to the user on the state of the channel, which profile is loaded, gain values, Input and Backfeed confidence metering. Users can choose to display an image here by configuring this option in the source profile.
4. **USER keys:** Two of these four keys are generally configured as User keys, while the other two are generally configured for integrated Phone workflows and assigned by default to the following functions:
  - **SET** -selects the channel or seizes the line
  - **HOLD** - puts a call on hold.
5. **Channel Fader:** This high-quality spill-proof motorized fader controls the gain of the input channel. The fader is Touch-Sensitive.
6. **LED feedback:** The arrow indicators show the motorized fader position in relation to the logical gain level in the Engine DSP. If the fader's physical position doesn't match its logical level (i.e., the fader is at 0 dB position, but the display indicates -10.0dB) one of the two arrows (in this example, the one pointing down) will light up to indicate that the fader needs to be moved (down) by the operator, in order to regain control of the level. When the fader is touched by a finger, both arrows light up to indicate that the touch sensor is active. After movement, this double indication will turn off after 2 seconds. The four rectangular LEDs provide status indication for the channel's Dynamics processing (DYN), Equalization (EQ), AUX send assignment (AUX), and Group assignment (GRP).
7. **Channel Bargraph:** This LED bar normally works as stereo level meter for the channel input (pre-fader, pre-trim). When the Automixer is active on the channel, the mono sum is displayed onto the left bar while the right bar shows the AM gain reduction.
8. **Dedicated keys for:** Talkback, PFL, and Automix functions.
  - Talkback (**TB**) lets the operator communicate through a backfeed to the source if a backfeed is enabled. A quick tap will latch the Talkback ON, while a press & hold will have a momentary action.
  - Pre-Fader Listen (**PFL**) is also known as "cue" or "preview" in some circles. Pressing this will route the source to a monitoring mix for off line listening. In this case too, a quick tap will latch the **PFL** ON where a press & hold will be momentary for the period of the press.
  - The **AUTO MIX** key is used to add the channel to the Automixer group.
9. **ON/OFF keys:** A red ON key and a yellow OFF key for the channel's state.

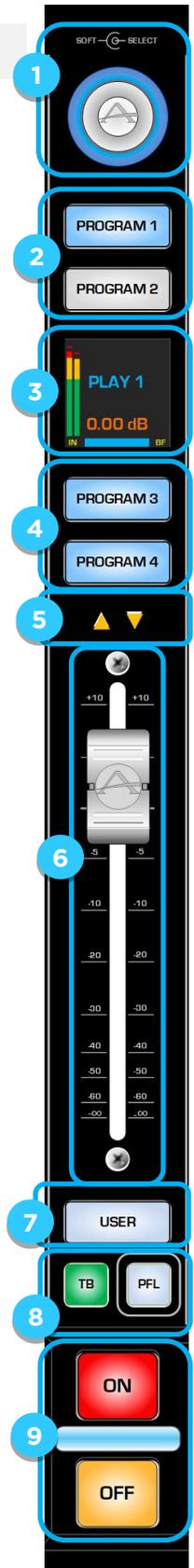


- ON/OFF keys:** Enable the channel audio (post-fader) to the selected busses. A color-coding LED bar indicator is integrated between the two.

## The SR-4FAD Fader Module

The SR 4-Fader module is a fully-featured control module that connects to the Master through an internal USB network. The SR 4-Fader module is composed of four channel strips , each with the following control elements:

- Channel Options encoder:** Can respond to touch, press, and rotate. A short press can trigger the Channel select which drives the master section into Channel options (blue tabs). The rotation function will depend on the configuration of the source profile and can be Fader-trim control, Pre-Amp gain control, Automix control, or Source select.
- PGM 1-2 keys:** Main program mix assignments are the default configuration for these 2 keys which assign the channel to feed the designated mix bus.
- TFT Color display:** Provides feedback to the user on the state of the channel, which profile is loaded, gain values, Input and Backfeed confidence metering. Users can choose to display an image here by configuring this option in the source profile.
- PGM 3-4 keys:** Main program mix assignments are the default configuration for these 2 keys which assign the channel to feed the designated mix bus.
- LED feedback:** : The arrow indicators show the fader position in relation to the logical level in the Engine DSP. If the fader’s physical position doesn’t match its logical level (i.e., the fader is at 0 dB position, but the display indicates -10.0dB) one of the two arrows (in this example, the one pointing down) will light up to indicate that the fader needs to be moved (down) by the operator, in order to regain control of the level.
- Channel Fader:** This high-quality spill-proof fader controls the gain of the input channel. The fader is non-motorized and Touch-Sensitive: When the fader is touched by a finger, both arrows light up to indicate that the touch sensor is active. After movement, this double indication will turn off after 2 seconds.
- USER key:** Normally set to AUTOMIX function or SET/HOLD, depending on the source type.
- Dedicated keys for:** Talkback and PFL functions.
  - Talkback (**TB**) lets the operator communicate through a backfeed to the source if a backfeed is enabled. A quick tap will latch the Talkback ON, while a press & hold will have a momentary action.
  - Pre-Fader Listen (**PFL**) is also known as “cue” or “preview” in some circles. Pressing this will route the source to a monitoring mix for off line listening. In this case too, a quick tap will latch the **PFL** ON where a press & hold will have a momentary action.
- ON/OFF keys:** Enable the channel audio (post-fader) to the selected busses. A color-coding LED bar indicator is integrated between the two.



## The MPC-ACC and MF1-ACC Modules

The Accessory Modules are fully-featured control modules that connect to the Quasar console via PoE, over the Livewire network. They are intended to let an external operator (not the console operator) control some functions associated to a specific Source. A typical example is that of a Talent that wants to control his own Studio Microphone.

Two types of Accessory Modules are available:

- The MF1-ACC : Motorized Fader Accessory Module
- The MPC-ACC : Mic & Headphone Control Accessory Module

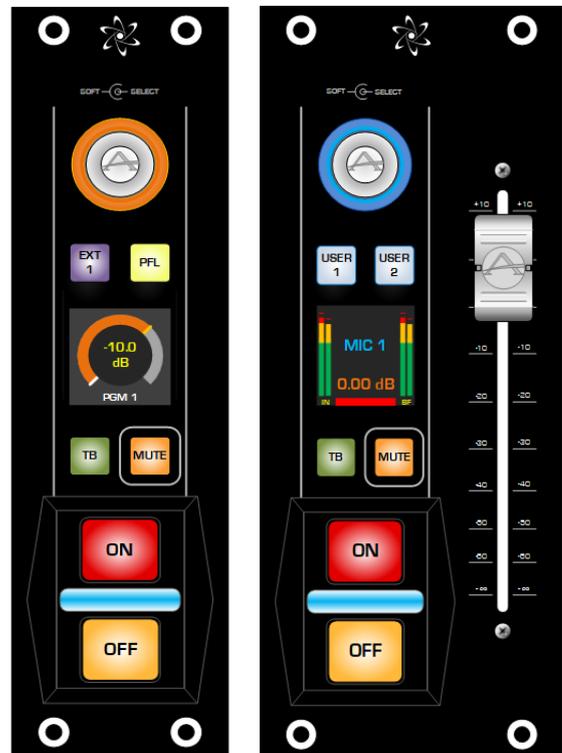
Both units are designed for quiet, fanless operation, and for installation in a worksurface.

From a functionality standpoint, the MF1-ACC is equivalent to the MPC-ACC with the addition of a motorized fader.

From a form factor standpoint, they are different: the MF1-ACC has a wider footprint.

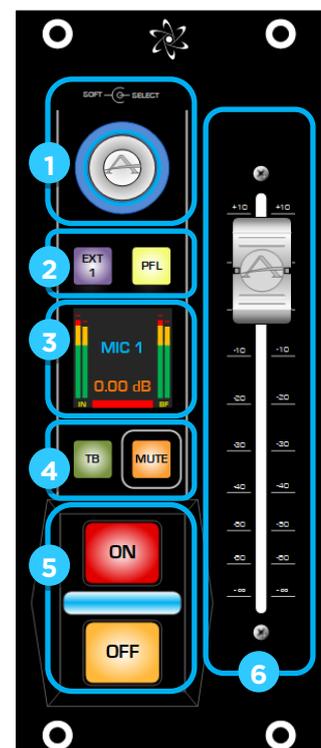
Both have the same features and the same control elements (except the fader): these may change their function according to the unit's *Control Mode*, and internal Key Mapping.

The following paragraphs describe the functions of each Hardware control, for each of the available Control Modes.



### Control Mode: Studio Guest Position

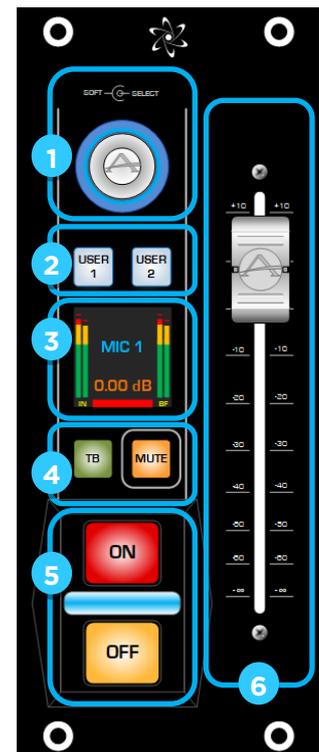
1. **Options encoder:** Can respond to touch, press, and rotate. A short press can trigger the Monitor select which drives the master section into Channel options (blue tabs). The rotation function will depend on the selected Operating Mode.
2. **External Monitor** and **PFL** keys (fixed function):
  - External Monitor (**EXT 1**) displays a list of External Monitor sources. Scroll the list and push the encoder knob to select the desired source and return to the volume control screen
  - Preview (**PFL**): Pressing this key will route the source directly to the headphones for off line listening. This key has (always) a momentary action.
3. **TFT Color display:** Provides different types of feedback to the user, depending on the selected operating mode
4. **Talk To Control Room** and **MUTE** keys:  
(Functionality available in Studio Guest Position Mode only)



- Talk to CR (**TB**) lets the Studio Guest communicate with the console Operator. A quick tap will latch the Talkback ON, while a press & hold will have a momentary action.
  - **MUTE** will momentarily turn the channel OFF (cough).
5. **ON/OFF keys:** Enable the channel audio (post-fader) to the selected busses. A color-coding LED bar indicator is integrated between the two.
  6. **Fader:** This high-quality spill-proof motorized fader controls the gain of the input channel. The fader is Touch-Sensitive.

### Control Mode: Channel Strip Control

1. **Options encoder:** Can respond to touch, press, and rotate. A short press drives the master section into Channel options (blue tabs). The rotation function will depend on the configuration of the source profile and can be Fader-trim control, Pre-Amp gain control, Automix control, or Source select.
2. **USER 1** and **USER 2** keys (programmable function): A small kit with two keycaps labeled “USER 1” and “USER 2” is provided for replacing the stock Keycaps, in case the unit is set to operate in Channel Strip Control mode. In this mode the keys become user-programmable, and their function can be defined in the Hardware Keymap menu of the Web UI, following the same rules of the XR Fader Modules’ User Keys.
3. **TFT Color display:** Provides different types of feedback to the user, depending on the selected operating mode
3. **Talkback** and **MUTE** keys (programmable function): In this mode the keys become user-programmable, and their function can be defined in the Hardware Keymap menu of the Web UI, following the same rules of the XR Fader Modules’ User Keys.



**Note -** The Accessory Modules are configured in Studio Guest Position Mode at the time of shipping, and the corresponding keycaps are installed. The keycaps for using the module in Channel Strip Mode are provided as spares in the package in case the user prefers to replace them.

5. **ON/OFF keys:** Enable the channel audio (post-fader) to the selected busses. A color-coding LED bar indicator is integrated between the two.
6. **Fader:** This high-quality spill-proof motorized fader can be configured to control the gain of any console input channel. The fader is Touch-Sensitive.

## Control Mode: VMIX Control

New in v3.0 is the VMIX Control Mode, which allows the MF1-ACC and the MPC-ACC to control any of the five input channels of a VMIX instance in the Quasar Engine.

1. **Options encoder:** A short press can trigger the Input Selector which is the equivalent of the Input top drop-down menu in the Engine-s VMIX Settings page . The rotation function will control the Gain level for the specific VMIX input channel.
2. **External Monitor** and **PFL** keys: External LW input selector by Name (EXT1) and by LW Channel # (PFL).
3. **TFT Color display:** The Module's display will show the selected source and its level by default, and will switch to , or a list of inputs for selection, depending what control is
4. **Talk To Control Room** and **MUTE** keys: no function assigned
5. **ON/OFF keys:** The Module's ON-OFF keys will control the Input channel ON-OFF function.
6. **Fader:** In case you are using an Accessory Module with Motorized Fader (MF1-ACC Module) the Fader will automatically control the VMIX Sub OUT.



A full description of how to configure Accessory Modules for different Operating Modes is available in Configuration Chapter 1: [Web UI Reference: ACC Modules](#).

A full description of the Accessory Module's Operating Modes, and of the available functionalities, is available in Operation Chapter 4 of this Manual: [Working with the Accessory Modules](#).

## The SK-BTN SmartKey Modules

Although the Quasar MTS module comes equipped with 4 Banks of 8 User-Programmable touch-sense keys, for some Customers these are not enough. In some cases, their fast workflows require a way to directly access multiple groups of User Keys without having to page between banks.

For this reason, we created the Quasar SmartKey Modules. This is a new range of Hardware Modules for the Quasar control surface, designed for adding User-Programmable LCD Buttons to the console.

The SmartKey Module is a fully-featured control module that connects to the Master through an internal USB network.

The module can be programmed only via Pathfinder, and comes in 4 versions, with 6, 12, 18, or 24 User Buttons. Modules with 6, 12 and 18 buttons can be daisy chained (via USB) in various combinations, and up to 8 GROUPS of User Buttons can be present in total, on the surface.

Modules with 24 buttons can not be daisy chained.

The SmartKey Module require Quasar SW v2.2 or greater, and Pathfinder Core Pro v1.9.12.60 or greater, in order to operate.

Please make sure you download the latest Firmware available from the Telos' website, or ask Support for the latest update.

Shown here:  
SK-24BTN module



# Operation 2 – The GUI

## The Display GUIs

The main console Touchscreen GUI (also called "TS GUI") provides the user with a graphic and tactile interface to interact with the control surface, and ultimately with the entire Console System. It augments the level of access provided by the individual hardware controls available on the surface and enriches the User Experience with interactive touchscreen controls, real-time visual feedback and contextual menus. Particular attention has been given to the ergonomics of the application, with the main concept of providing the visual grouping of control elements, visual styles, behavioral standards of the various objects (menus, buttons, labels, etc.) with a minimal effort required in terms of user actions.

The Channel Displays GUI is dedicated to the channel strips available on the XR-4FAD, SR-4FAD and ACC Modules. It provides basic indication of Source Names, Input and Backfeed audio levels, fader position, color coding and Auxiliary Text information messages.

### GUI Elements : Widgets & Windows

The Surface GUI consists of a series of web pages, populated with Widgets. these are a set of graphic elements like menu tabs, buttons, gauges, meters, faders, indicators, text labels, frames, etc. Also, there are various types of embedded windows showing more complex graphical objects like the EQ , the Dynamics, and the De-Esser graphs.

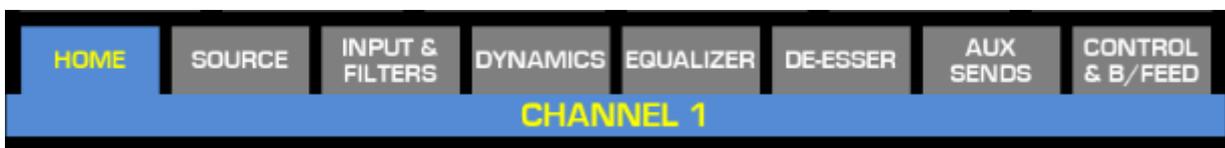
## GUI Widgets' General Rules

### Menu Tabs

The 12.1" touchscreen can be navigated via a tabbed menu available in each page, or by interacting with the hardware controls present on the surface modules. In case of the tabbed menus, only one tab can be selected by the user at any one time. Where appropriate, each tab's contents are further divided into sections, according to a particular logical and functional grouping.



These are located always in the middle of each page, (for easy reach and to split the Main Meters area from the Controls area) and can be operated as buttons (although they don't look like a button widget) to access each of the GUI pages. The tab names and their order are fixed, not user-defined.



## GUI Buttons

GUI Buttons are always rectangular shape, with rounded corners to differentiate them from text labels. Buttons can have various functions:

- ON/OFF switch,
- Option Select (i.e. slope type selection for a filter band)
- Channel Select with ON status indication (no control, only visual feedback)
- Open/Close Menu (i.e. open EQ preset menu)

Different graphical styles (colors) are associated with each type of function, so the user can intuitively understand what type of button he is operating by just looking at it.

**Note** - The MENU TABS are the only exception to this rule, because they are buttons with sharp corners.

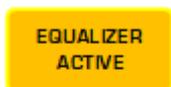
Buttons can have different logic (momentary or latching) and behaviors according to the above listed functions. These are described below.

### ACTIVATION buttons

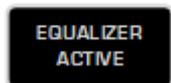
**Action:** Latching (typical) , momentary (only in some specific cases), press & hold.

**States:** This type of button has two states: ENABLED & ON (pressed), ENABLED & OFF (de-pressed)

**Visual Aspect:**



Button Active (ON state)



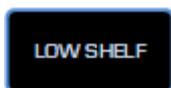
Button Inactive (OFF state)

### OPTION SELECT buttons

**Action:** Momentary (typical), Latching , press & hold.

**States:** This type of button has three states: ENABLED & OFF (de-pressed), ENABLED & ON (pressed), DISABLED or LOCKED (greyed out) .

**Visual Aspect:**



Option button Enabled, not pushed (de-pressed)



Option button Enabled, and pushed down (pressed).  
(This applies to both momentary and latching style buttons)



Option button Disabled, or Locked. It is not possible to press it.

In some cases, a short press (or tap) can cycle through the different states (or options). A long press (>1s) will open a popup menu on its side, showing the extra buttons corresponding to all available options.

## MENU SELECT buttons

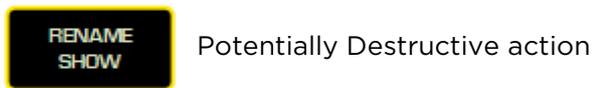
**Action:** Momentary only

**States:** Same logic as that of the Option Select buttons.

**Visual Aspect:**



In some cases, the button fillet (frame line) color is used to indicate the type of action associated with that button: green indicates non-destructive, yellow indicates potentially dangerous, red indicates a potentially destructive action



**Note** – After pressing a Destructive type button, the UI will always present the user with a YES/NO confirmation dialog.

## Labels

Text Labels (i.e., “**COMPRESSOR RATIO**”) are simple text fields. They are always rectangular, with no rounding on corners in order to differentiate them from buttons, dark (blue or grey) color and white text.

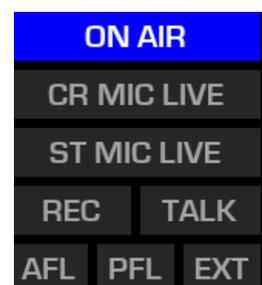
There are two different types of labels, system-wide:

- Fixed Text Labels, which typically have text which is defined either by the user or the system.
- Dynamic Text Labels, which display an information which is contextually changed by the UI, according to the function selected by the user.



## Indicators

Indicators are a particular type of dynamic text label: which can change its text as well as its color, following certain user operation (like when PFL is activated). In general, indicators are dark grey or black when inactive and light up with a particular color when active.



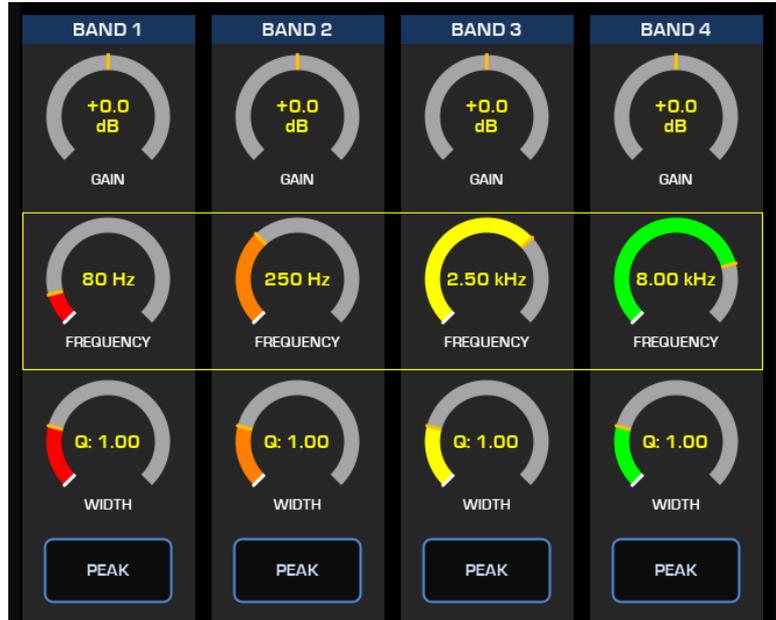
## Frames

Frames are empty rectangular boxes with a yellow outline, used to group a set of widgets that belong to the same class, like for example, the three gauges of a single EQ and, or all the subgroups bus buttons of a certain channel. (See next page picture)

Frames are used in the UI to indicate which set of Gauges (rotary touch controls) is associated to the physical encoders in the Master Module.

## Gauges

All encoders on the surface are “endless turn”, with touch sensor circuit. In some cases (on the Fader modules) these encoders will have mechanical detents, while in some other (in the Master/Monitor module) they won't. In order to show the logical position of an encoder to users, the GUI uses rotary touch controls that we call “Gauges”. Gauges are dynamic indicators, with a circular field in the center containing text information about the parameter value and unit.



For example: a gauge associated to a trim pot will display “Gain +/-xx.x dB” in its center field. Each Gauge has a light grey circular background and a colored circular indicator, which varies according to the associated DSP parameter . When selected (via a frame) the associated hardware encoder will illuminate of the same color.

A gauge becomes “greyed out” when it is either in Disabled, Muted, or Locked state. An indication will appear below the gauge describing the actual status.



A white start detent can be located either at the beginning (left edge) of the circular background, or in the middle, according to the nature of the associated parameter (example: *pan* will have a center detent, while *gain* will have a left detent). In case the function, or parameter, associated with a Gauge is turned off, the Gauge will reflect this by changing the color of the indicator, and the text.



In addition to being operated from physical encoders, upon selection with a frame, Gauges can also be operated directly from the console touchscreen, by touching their area and swiping horizontally, (right to increase, left to decrease value) with your fingertips.



## Faders

The fader widget is used in the CONTROL & B/FEED page of the Input Channel GUI. Although this might seem redundant in a control surface which has physical faders, it is very useful in a split frame console, or when the GUI is accessed from remote. The fader widget is designed to be exactly the same size of a physical fader: 100mm long.

# TS GUI: Master Menu (orange tabs)

This is the starting page of the Quasar Touchscreen User Interface.

A quick access to this view is also provided by the large **HOME** key located in the lower right area of the Master Module.

## MASTER HOME

The MASTER HOME tab displays the normal operational view which provides high-resolution level and loudness meters, some console status signalization (“ON AIR”, “RECORD”, “TALK”, “PFL” and AUX contribution), and the title of the currently loaded show profile.

There is a large visual aid clock indicating the time and visual clues for the hour location. At the bottom of this view are some touch screen buttons for operation assistance.

- A short press on the **CLEAR PFL – CLEAR TB** button will clear any PFL or AFL selection, on all channels. A long press will clear any TB selection.
- Push **CHANNEL SELECT** to display an option screen from where you can select any available input channel. It’s an alternative to pushing the encoder located at the top of each fader strip. The same can be done with a long press of the HOME button. Channels availability will depend on the type of Engine License (16, 32, 48, or 64 channels).
- The **LOUDNESS START/STOP /RESET** button controls the Loudness metering counters.



Below these buttons, an indicator (new in v3.0) will appear whenever a Quasar Soft session has been opened to control the console from remote, informing the operator he’s not “alone”.

Next to these buttons are the two rotary indicators for the CR HEADPHONES and CR SPEAKERS volume controls. Below, there are some controls used for configuring the Monitors:

- **PFL to HP** : routes the PFL (Pre-Fader Listen) bus to the Control Room Headphones out.
- **HP Mode** : Selects the listen mode for the CR Headphones (Stereo, Left, Right, Mono Sum).
- **PFL to LS** : routes the PFL (Pre-Fader Listen) bus to the Control Room Loudspeakers out.
- **LS Mode** : selects the listen mode for the CR Loudspeakers (Stereo, Left, Right, Mono Sum).

For more information about how to configure PFL options, please refer to Configuration Chapter 5 of this manual: [Setting up the Monitors & PFL/AFL](#).

## Console Status Indicators

### ON AIR

The ON AIR signal in the Quasar Master Home screen can be driven by custom Pathfinder logic or dedicated external GPI pin, to signalize when the console, or the studio, is live on the air.

For more details on how to trigger the ON AIR signal, please refer to Configuration Chapter 5 of this manual: [Triggering an ON AIR Light](#).



### CR MIC LIVE and ST MIC LIVE

Quasar console logic offers two independent statuses for signaling when a Control Room type Microphone (Operator Mic, Producer Mic, or CR Guest Mic) or a Studio Type microphone are turned ON, regardless of their assignment to the PGM busses. Each MIC LIVE signal becomes active when the corresponding microphone is activated or when an external MUTE GPI (pin 11) is closed on its Control Room Monitor or Studio Monitor GPIO port. When the MUTE GPI pin is triggered, the corresponding monitors will be muted. At the same time, a MIC LIVE GPO on the same port will be closed, allowing the mounting of external signaling devices. The desired GPO channel # can be defined in the Show Profile Monitor Section, using the Logic Port field.

For more detailed information about configuring the Quasar Monitor GPIOs, please refer to Configuration Chapter 7 of this manual: [GPIO Configuration](#).

### RECORD

Quasar offers two Record Mode options: Standard and Flexible. The indicator will display the “**REC**” signal when the Standard Record Mode is active, and the “**FLEX**” signal when the Flexible Record Mode is active.

### TALK/LISTEN

This indicator will light up every time the operator microphone audio is routed to the Talkback bus, instead of the PGM bus. If the CR Microphone is used both as Talk Mic and On Air Mic, this will be temporarily removed from the program bus (muted to the Air) so that the conversation to the talk bus is not sent to the mix. If the CR Microphone is used as Talk Mic only, then this indicator will not light up when a TALK or TB key is pushed by the operator. When a TB key is pressed on one of the Accessory Modules, and Talk to CR activated, the indication will change to “LISTEN”.

### AFL and PFL

These indicators are active every time the AFL or PFL function is selected on any of the input channels.

### EXT

This indicator is active every time the EXTERNAL PFL function is triggered via a GPIO port. In case Quasar is configured to control an Infinity Intercom Panel, this indicator will be active each time a Listen Key is enabled on the Intercom.

### AUX 1-8

These indicators are active every time one of the AUX Send is turned ON in any of the input channels. They are intended for monitoring the contribution to the bus, rather than the ON-OFF status of the Master Auxes.

### Quasar Soft Active

This indicator (added in SW v3.0) will warn the operator that a Quasar Soft session is active on his console... so that he know he’s not alone using the desk!

## SHOW PROFILES

This is the page where you can store and recall all the different Show Profiles in your Quasar. A quick access to this page is also provided by the large **PROFILES** key located in the top right area of the Master Module.



A list of show profiles that have been configured in the Quasar Web UI will be presented here. Select the desired show with the touch screen so that it is highlighted, and press the **LOAD SHOW** button to the right. The soft buttons to the right of the show profile list let the user perform the following operations:

- **PAGE UP** : pages up the list
- **PAGE DOWN** : pages down the list
- **LOAD SHOW** : Loads the currently selected show.
- **CAPTURE** : takes a snapshot of the current state of the surface, and stores the state in a new show. Once the button is pressed, a dialog box pops up to enter the name for the new show.
- **RENAME** : changes the name of the selected show.
- **UPDATE** : takes a snapshot of the current state of the surface, and stores the state in the currently active show. Once the button is pressed, a dialog box pops up to confirm the overwriting action.
- **DELETE** : removes the profile from the library of show profiles.

MODIFIED	SHOW NAME
00:05 01.01.06	0 - EMPTY Show
00:01 01.01.70	1 - MORNING SHOW IN UK!
00:56 01.01.70	1 - MORNING SHOW ON AIR
00:05 01.01.06	2 - AFTERNOON TALK!
00:05 01.01.06	3 - NEWS PROGRAM
00:05 01.01.06	4 - EVENING SHOW
00:05 01.01.06	5 - NIGHT MUSIC
00:18 01.01.70	-new show- TEST1
00:02 01.01.70	-new show- TEST3
00:00 01.01.70	-new show-TEST2
00:05 01.01.06	-new TEST show-
00:05 01.01.06	-prova manuale-

**Tip** - If you think that the **DELETE SHOW** button is too dangerous, and fellow operators might inadvertently delete important show profiles, you can always disable it from the Web UI and make the button completely disappear from the touchscreen. The functionality to delete shows will still remain available on the Web UI.

## Managing Show Profiles from The Console

The following basic concepts have to be always kept in mind when Capturing, Updating, and editing Show Profiles directly from the console's Touchscreen UI:

- Show Profile settings are applied only at the time of loading the Show.
- If, after loading a Show, a new Source is manually loaded on a channel (or the same Source is reloaded), the new Source settings will be always applied on top of the settings that were saved in the Show, for that Source.

- At this point, if the Show is updated (using the **UPDATE SHOW** button on the Touchscreen) then the settings that came from the Source will be saved back into the Show Channel, and the Web GUI will reflect this update.
- But, if after loading a new Source, the Show is not updated, and the user will open the Show in the Web GUI for editing, then he will notice a mismatch between the settings in the console and the settings stored in the Show.

## MONITOR OPTIONS

As detailed earlier, there are some dedicated hardware keys for monitor control which provide the most common sources. If the operator needs to monitor other mixes, adjust the dimming levels, or adjust the TALK gains, this is the location for those controls. The gain indicators at the bottom are tied to the rotary encoders below. It is also possible to adjust the level directly from the touchscreen, by touching the desired gauge and sliding the finger to its left or right.

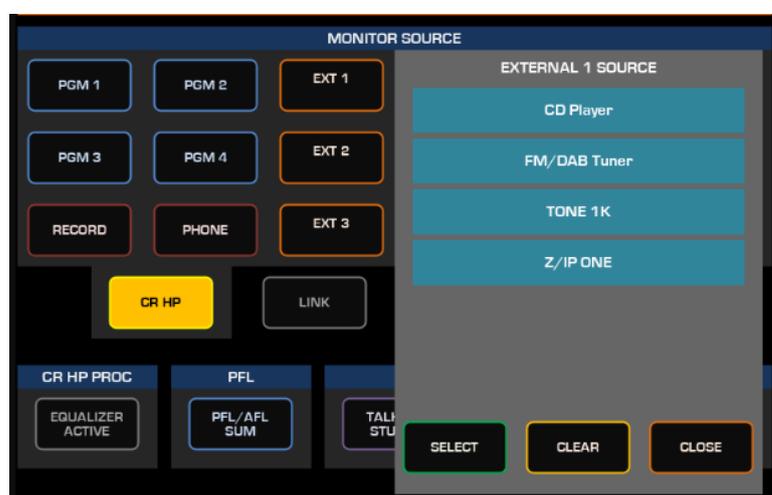
The MONITOR SOURCE section provides source selection for the three Monitor Sections:

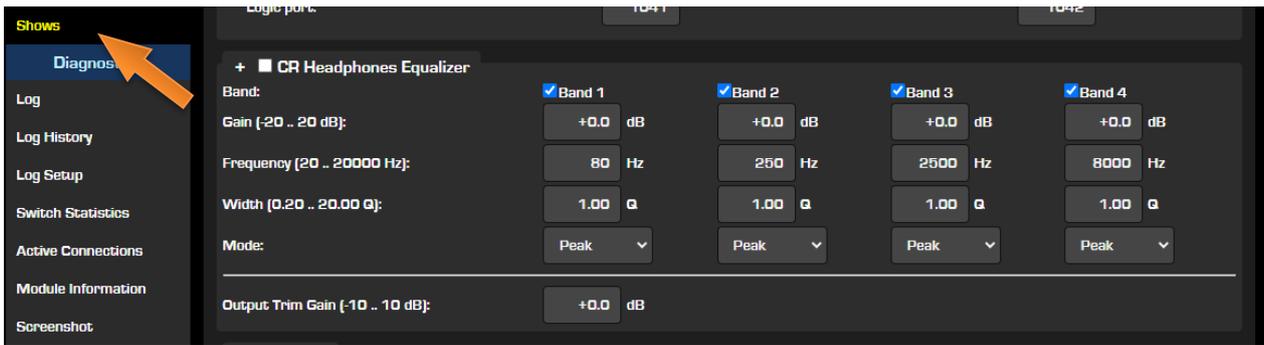
- C. Room Headphones (**CR HP**)
- C. Room Speakers (**CR SPK**)
- Studio Monitors (**STUDIO**)

For each one of these sections, selectable with the corresponding buttons, it is possible to select different sources to listen, such as Program 1 to 4 bus, Record and Phone bus, Auxiliary 1 to 8 bus, and three EXTERNAL monitor inputs available on the Engine. Or you can select **NONE** to clear the assignment.

To assign a source to the External inputs, press & hold any of the **EXT** buttons, and a popup list will appear, presenting all the sources which, in their profile, have been configured to show up as externals.

The CR HP PROC section provides the control to engage the 4-Band Fully parametric EQ integrated in the CR HP bus. The settings for this EQ are *only* available in the **Monitor Section** of the Show Profile Web UI.





The PFL section provides the control to change the switching mode for the PFL/AFL bus:

- **PFL/AFL SUM** mode lets the operator activate the PFL or AFL function on multiple channel strips at the same time. This will result in a sum of all the channels Pre-fader (or After-Fader) outputs, being sent to the bus.
- **PFL/AFL INTERCANCEL** mode lets the operator toggle the PFL or AFL function on a single channel strip at a time. This will send to the bus only the Pre-fader (or Post-Fader) output of the last selected channel.

**Note** - By default, the Channel AFL function is accessed only from the touchscreen UI in the Channel Control & Backfeed menu. However, this function can be assigned to any hardware key on your fader modules. See Chapter 3 - Remapping the XR-4FAD Module Keys

The TALK ACTIONS section provides the controls to activate different types of Talk functions within the console:

- **TALK TO STUDIO** will insert the Operator Microphone signal into the Studio Monitor, Talent Headphones, and Studio Guest Headphones. This function is also triggered by the **TALK** key, located in the bottom right corner of the MTS module.
- **TALK TO BACKFEEDS** will insert the Operator Microphone signal into the Studio Guest Headphones, and any other Backfeed configured for Telephone or Codec systems.
- **TALK TO EXTERNAL** will insert the Operator Microphone signal into the dedicated “Talk to External” bus of the Quasar Engine. This is normally used in conjunction with the “External PFL” function only in the Monitor Section of the Show Profile Web UI, to set up inter-studio communication.

The TALK TO C.ROOM section provides the control to adjust the gain of the talkback which is activated from the studio, to communicate back to the Control Room.

The STUDIO SPEAKERS section provides the control to adjust the volume of the Studio Speakers output.

The STUDIO DIM section provides the control to adjust the level of attenuation applied to the Studio Speakers output, when the Talk to Studio function is engaged.

For more detailed information about configuring the Talkback section, please refer to Configuration Chapter 5 of this manual: [Setting up the console Talkback](#).

# AUTOMIX

The view is divided in 3 sections:

The CHANNEL SELECTOR section at the top, allows to toggle selection of the channels in the system (momentary action) and to see which ones are enabled into the Automixer. Channels with AUTOMIX ON will have the selector button highlighted by a yellow fillet.

The CHANNEL PARAMETERS section at the bottom left, allows to change the ON-OFF state or adjust the WEIGHT (priority) of the channel within the Automixer.

The GLOBAL PARAMETERS section at the bottom right, allows you to adjust the global settings of the entire Automixer, such as Attack and Release times and the Pre/Post fader position of the Automix gain control stage.



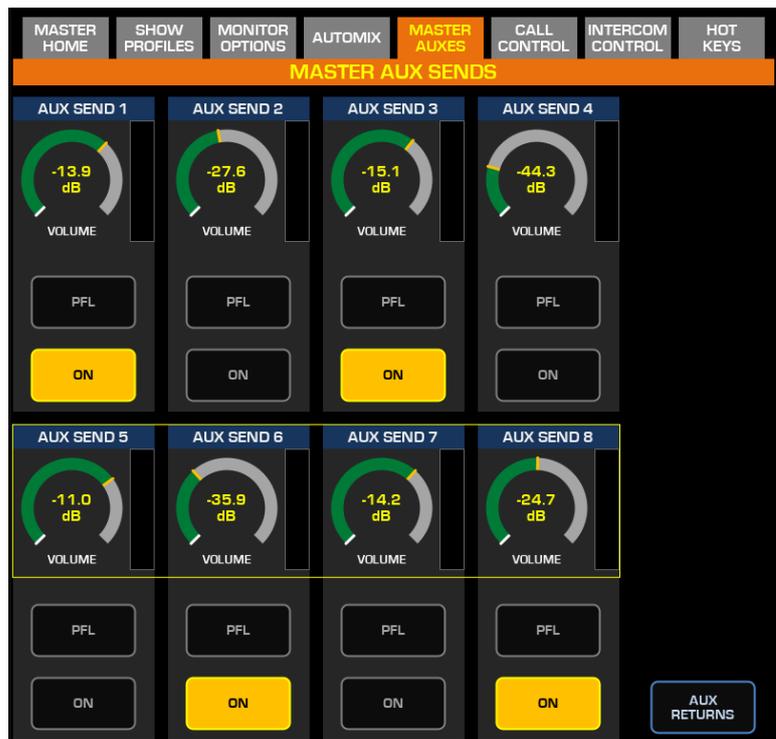
# MASTER AUXES

The system has 8 AUX sends and each can have a gain adjusted, enabled, and can be monitored into the PFL bus.

The view defaults to AUX 1-4 controls. If you want to control AUX 5-8 touch the lower area of the screen to highlight these controls with the yellow box.

Pushing the encoder at the bottom of the display will immediately bring the gain up to 0.0dB level for the associated gauge.

The **PFL** button has a momentary action, so you won't risk leaving it active in the background, and once out of this screen, wasting your time looking for a hidden **PFL** button.



## AUX RETURNS

Pushing the **AUX RETURNS** button brings you to the AUX RETURN screen. These two AUX return inputs are basically two “short” input channels, (without DSP processing) typically used for bringing any external FX processor return back into the mix.

This view provides gain control, pan control, and assignment to any of the four main program mixes.



Additionally, the AUX returns can be monitored as part of PFL via the dedicated soft buttons that are provided in this view. The action of these buttons is always momentary.

In order to exit this view and return to the main MASTER AUXES screen, select the MASTER AUX Tab.

## CALL CONTROL

For use in multiline workflows, such as those available with the Telos VX system, the CALL CONTROL tab will provide access to a soft “Call Controller” panel. This can be configured from the web UI to manage 8 or 12 phone lines letting you assign any two of those lines, to two different channel strips.

The 8-line selector is organized with two columns of 8 selector buttons, for assigning lines to two different channels. In conjunction with the **SET** and **HOLD** keys on the channel, the operator can take callers and make calls through this interface.

A **DROP** button for each of the two line columns will let the board operator drop the selected line instantly, while the **NEXT** button will let him switch to the following call.

The **BLOCK ALL** and **NEXT** buttons will respectively block all incoming calls and transfer one call to

A Numeric Keypad is permanently shown, for quickly dialing numbers and making calls.

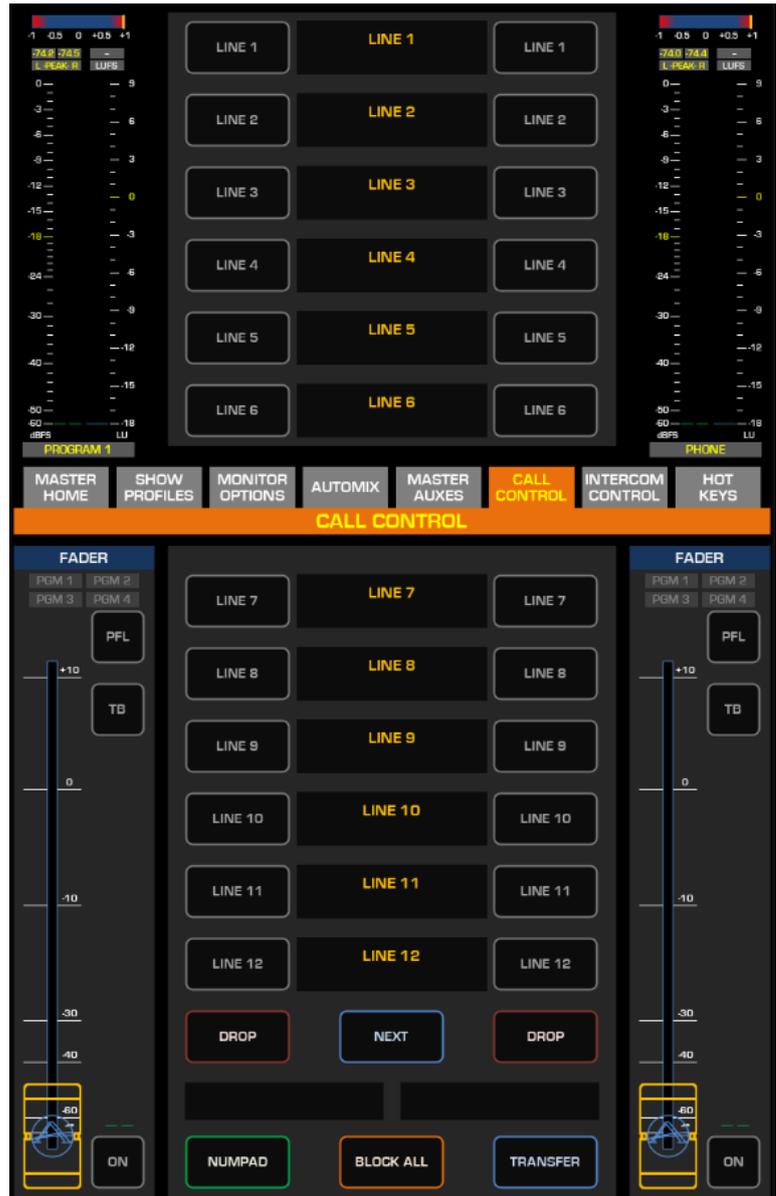


Above: 8-Lines Call Controller screen

The 12-line selector is organized with two columns of 12 selector buttons, for assigning lines to two different channels. In conjunction with the **SET** and **HOLD** keys on the channel, the operator can take callers and make calls through this interface.

Two convenient virtual faders are displayed on each side of the Line selector matrix, and will replicate the physical controls of the two fader strips assigned to the phone hybrids.

Note that these virtual faders will be active even if the hybrid phone channel strips are not present on the surface (not assigned to physical faders) effectively adding two extra faders to your surface. For example, on a 12-Fader console with 16 or more engine input channels, the phone hybrids channel strips could be accessed from the **Channel Select** menu and full control of the channel could be achieved from the CONTROL & BACKFEED tab, while the basic controls (Faders, ON/OFF, TB and PFL) could be accessed directly from the 12-Lines Call Controller screen.



Above: 12-Lines Call Controller screen

When the Call Control tab is selected, one of the hardware encoders becomes active (lit in orange), indicating that the PFL to LS (LoudSpeakers) control will pop up when the encoder is touched, so as to let you adjust the caller's level while listening to him through the PFL bus.



In this configuration, the numeric Keypad is activated by pressing the **NUMPAD** button, and it will automatically disappear once a number is dialed.

For more information about how to use the Call Controller, please refer to Operation Chapter 3 of this manual: [Working with the Call Controller](#)



# INTERCOM CONTROL

The INTERCOM CONTROL tab offers a complete remote GUI to control any Infinity Intercom panel, both hardware and virtual.

The PANEL section at the top, replicates a standard 16-keys Infinity panel: can be an MP-16 or a VIP panel.

The OPTIONS section at the bottom offers some of the controls found on the Infinity panels:

- The **PAGE** button replicates the front panel's Shift Key and scrolls through the different Shift Pages (layers) of contacts, only accessing pages that have been assigned a contact until finally returning to the first page at the end of the cycle.
- The **MIC ON** button activates the panel's microphone input. This could either be the CR Operator Microphone, or a network source or the gooseneck connected to the front panel input, depending on the unit's configuration.
- The **MAIN VOLUME** control, replicates all the functions of the top rotary encoder found on the Infinity front panel ("Vol" knob):
  - It provides the volume control function for the main output (either the internal speaker, headset or the rear Panel output when configured to output the incoming talkback).
  - It mutes/unmutes the listening function, when the corresponding encoder is pushed in
  - It lets you adjust the listening level of each individual source when this is in listen mode.
- The **PFL VOLUME** control is the console PFL level control, useful to control the monitor level when incoming audio is mixed into the EXTERNAL PFL input of the Quasar's Monitoring section.
- The **LW OUT GAIN** control lets you adjust the level of the Infinity "Spkr/HS LW Output Channel Gain".



For more information about how to configure the Intercom Controller, please refer to Configuration Chapter 5 of this manual: [Setting up the Intercom Controller](#).

# HOT KEYS

Hot Keys can turn the Quasar touchscreen display into a remote controller for your Playback System's Hot Keys.

A new feature of v3.0 enables the Hotkeys to operate in two different modes:

1. **Direct TCP/IP control** - Any external device (such as a playback system or jingle machine) can be controlled via TCP / IP. This operating mode helps create the shortest and fastest possible control loop, without the need to use Pathfinder. This control mode is unidirectional.
2. **LWCP control** - Any LWCP-capable device can be controlled via Axia LWCP control protocol. This operating mode basically makes the HotKey simulate hardware user buttons found in all Axia consoles



30 user-programmable on-screen buttons can send a control string to any device connected over a TCP link, providing that the IP address and Port # of the receiving ends are entered into the console Web UI.

**Note** - At the time of writing this manual, the Hot Keys' control is unidirectional. We are planning to add bidirectional communication.

For detailed information about configuring the Hot Keys, please refer to Configuration Chapter 1 of this Manual: [Hot Keys](#).

## SYSTEM SETUP

This screen is meant for accessing Setup options which are not required during normal operation; therefore, it is hidden. In order to access the System Setup page on the Master Touchscreen Module, you need to push and hold the “Monitor Options” key (labeled “MON OPT” and located between the two Control Room volume pots) for 5 seconds.



The IP ADDRESS, NETMASK, GATEWAY and ENGINE sections are for configuring the network parameters of the Console and Engine, without using their Web UIs.

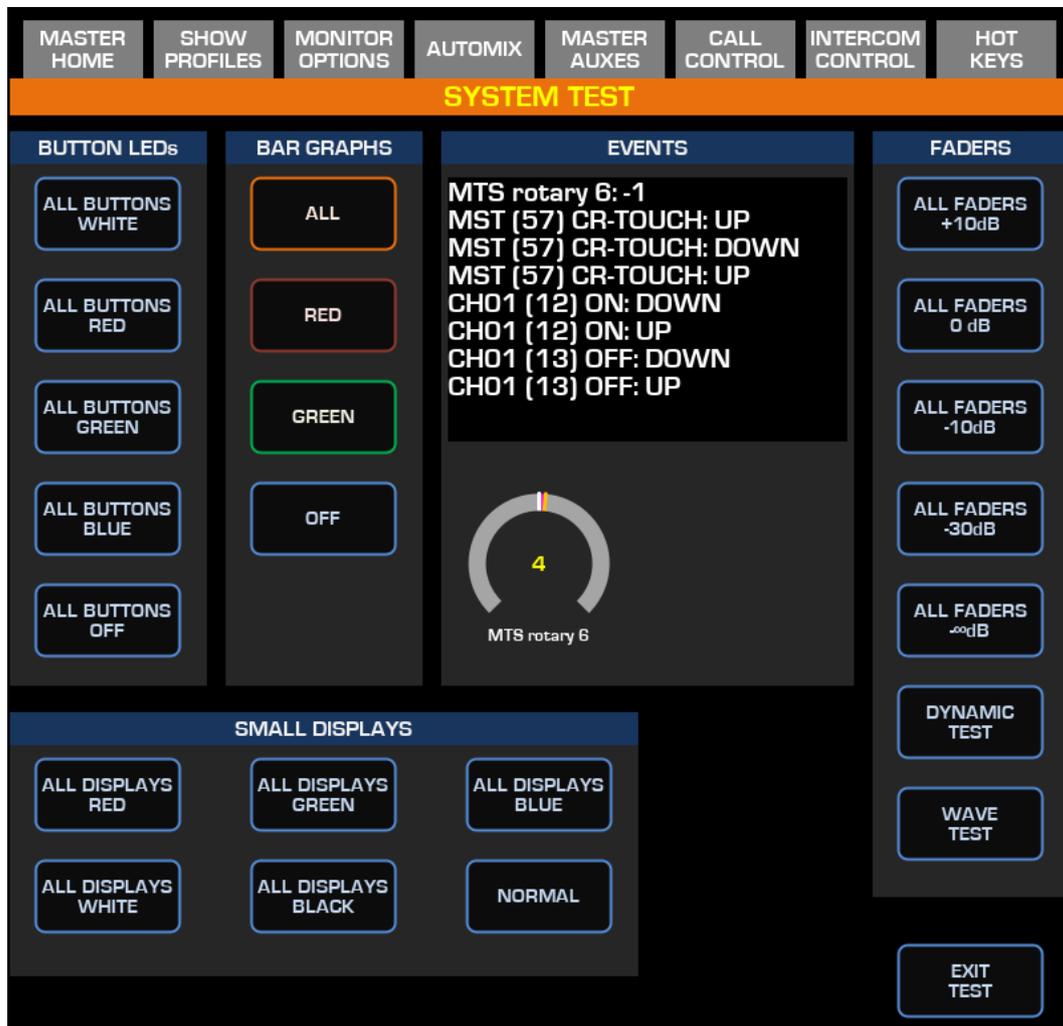
The **SAVE & REBOOT** button is for applying the new network settings, while the **UNDO CHANGES** button is for reverting to the saved settings if an unwanted change was made.

The VERSION section is useful for checking which software version is running in the active Software Bank, without having to access the web UI.

The **LOCK TOUCH SCREEN** button allows to lock the touch sensor on the display, for cleaning it while the console is on the air. After locking the touchscreen, a semi-transparent overlay is applied to the GUI, with a message indicating how to remove the lock.

The lock can be removed by pressing the leftmost and rightmost encoders below the screen at the same time.

The **SYSTEM TEST** button will enter a special maintenance mode, required for testing every single hardware control on the console: Switches, LEDs, Bargraphs Encoders, Displays, Faders and Smartkeys (if present in the surface).



**Note** - For a complete description of the System test, and how to use it, please refer to the Quasar Install Guide document.

**Warning** - We recommend not enter SYSTEM TEST while the console is on the air, unless required to do so by a Telos Support Engineer during planned troubleshooting activity. Upon exiting the System Test, the MTS module is not fully reset to its previous operating state and will need a soft reboot to return to a fully working state.

# TS GUI: Quasar Channel Menu (blue tabs)

The channel control tabs, or blue tabs, are made available when the option knob is pressed on one of the channel strips, or one of the channels is selected from the Master module's Touchscreen UI.

## CHANNEL HOME

This is the default screen visible anytime the channel control mode is active. The screen provides a complete status overview of the input channel, such as bus assignments, EQ curve, Dynamics, Panpot, and audio meters. There are buttons to enable or disable the various audio processing stages, the Automixer, and for configuring the Channel Insert Send and Return.



The Channel meters provide monitoring at-a-glance of the Direct Out (insert Send) and Insert Return levels. The EQUALIZER, DYNAMICS, and AUX SENDS graphs can be touched to go directly to the corresponding processing tab.

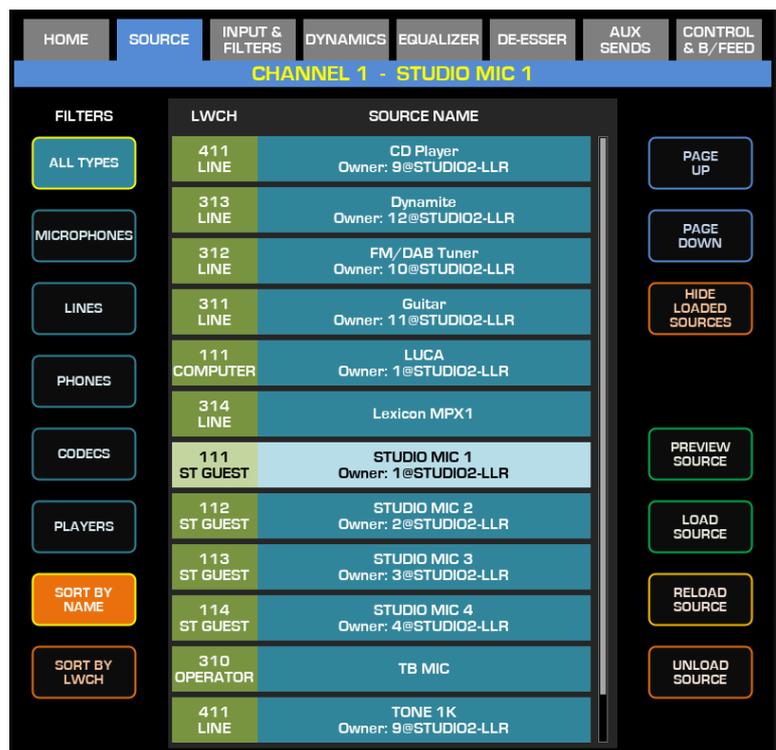
**Note -** The position of the FILTERS, DYNAMICS, EQUALIZER and DE-ESSER tabs in the channel UI, follows the order of the DSP processing blocks within the channel.

## SOURCE

This view provides a list of Source Profiles that have been defined to be accessible to the selected channel. If the list is long, there are filter options to help organize the view.

The left-hand column of buttons allows for organizing the view by source type: MICROPHONES, LINES, PHONE, CODECS, and PLAYERS. The list can also be sorted by name or by Livewire channel (LWCH).

The right-hand column offers additional buttons to navigate the list, preview the selected source before it is loaded onto a channel, and Load, Reload, or Unload the selected source.



## INPUT & FILTERS

This view provides various controls to the input and useful metering information. The INPUT section provides:

- **Input meter:** monitors the input phase and source level (after the gain stages in the node, before the fader and channel processing). The small confidence meter in each of the channel displays copies this meter.
- **Input Trim:** controls the digital gain of the channel input stage. It can be used to offset the fader level: if you like the 0.0dB mark to be at the top of the fader throw, and not at 75%, you can set it to -10.0dB
- **Input Mode:** determines which leg of the input signal feeds the channel strip. Available options change according to the type of source which is selected.



The NODE CONTROL section provides:

- **Line Gain:** controls the digital line gain of the xNode. This is the gain applied to the Node's matrix output that feeds the sources.
- **Mic Gain:** controls the analog Mic Preamp gain of the Microphone xNode. It is applied before its internal I/O matrix
- **Phantom power:** turns on +48VDC supply in the Microphone xNode

The FILTERS section has full-band adjustable HP and LP filters, useful to remove unwanted signal noise before it is processed. It provides:

- **High Pass Freq and Low Pass Freq:** controls to adjust the center frequency of each filter
- **Slope buttons:** can be set to FLAT, 6, 12, 24, 36 and 48dB/Octave
- **Clear, Copy, Paste:** clear or copy settings from the selected channel into a different one
- **Presets:** displays a list of User Presets. The list is editable from the Web UI **Presets** menu.

The INPUT DELAY section provides control over the **Audio Delay** of the channel's input. Setting any value different from 0ms will engage a delay block positioned right before the Input Trim. Can be set from 1 to 400ms in steps of 1ms.

The OUTPUT section provides:

- **Output meter:** monitors phase and level of the signal sent to the mix bus, post-fader
- **Pan:** adjusts the panorama position of the post-fader signal
- **Phase:** inverts the signal phase. Depending on the selected Input Mode, different settings for inverting one leg only, or both legs of the signal, will be made available.

## CHANNEL DYNAMICS

New and improved DSP algorithms in the Quasar Engine offer more accurate control over the audio giving it smooth and musical sound. The COMPRESSOR / LIMITER section provides:

- **Threshold:** sets the level at which the Compressor or Limiter is engaged.
- **Ratio:** sets how aggressive the processing would be.
- **Knee:** finely adjusts how smoothly the transition across the threshold occurs.
- **Auto Makeup:** a special type of Makeup Gain that adjusts the compressor's output, letting you manually match the level of the compressed output signal to the input signal level. Its exclusive design lets you tweak the Threshold and Ratio even while the channel is on the air, because the output level is always kept constant.
- **Attack Time:** controls how quickly the processing is applied, after the Threshold is passed.
- **Release Time:** sets how slowly the processing is released, after the Threshold is passed.
- **Auto Attack/ Release:** provides optimized Attack and Release time settings, for either music or vocal programs. The Speech mode is optimized to react quickly to voice transients, while the Music mode will provide softer compression times for music programs.



The EXPANDER / NOISE GATE section provides:

- **Threshold:** sets the level at which the Expander or Gate is engaged.
- **Ratio:** sets how aggressive the processing would be.
- **Variable Knee:** finely adjusts how smooth transition across the threshold needs to be.
- **Depth:** sets the limit for the maximum amount of signal attenuation.
- **Attack Time:** controls how quickly the processing is applied, after the Threshold is passed.
- **Release Time:** sets how slowly the processing is released, after the Threshold is passed.

The OPTIONS section provides the ability to copy your settings and to paste those same settings to other channels. To do so, create your settings, press **COPY**, then select a different channel (by pressing its top encoder, or using the MTS Channel Select menu) and press **PASTE**. The **CLEAR** button will clear all parameters and disengage the Dynamics. The **PRESETS** button displays a list of User Presets, editable from the Web UI **Presets** menu.

**Trim Gain:** Offsets the overall output level of the entire Dynamics processor by +/- 10dB

For detailed information about configuring the Dynamics, please refer to the next Chapter.

# EQUALIZER

The Quasar EQ was modeled after many hours of listening to the best analog EQs available on the market, including some really cool vintage gear. The 4 bands are completely overlapping, and each band provides:

- **Gain:** adjusts the Gain of the selected band.
- **Frequency:** adjusts the center frequency of the selected band
- **Width:** adjusts the size (Q factor) of the selected band.
- **Peak/Low/High Shelf:** determines the type of filter applied to that band
- **Band Active:** engages each individual band.

The OPTIONS section provides the ability to copy your settings and to paste those same settings to other channels. To do so, create your settings, press **COPY**, then select a different channel on the surface (by pressing its top encoder, or using the MTS Channel Select menu) and press **PASTE**.

The **CLEAR** button will clear all parameters and disengage the EQ. The **PRESETS** button displays a list of User Presets, editable from the Web UI **Presets** menu.

### Layout Vertical/Horizontal:

rearranges the EQ controls to a different layout. The controls are tied to the encoders below the screen and this button allows for controlling the frequency of all 4 bands at the same time or changing the parameters of a single band at the same time.

**Equalizer Active:** is the enable/disable button for the entire EQ processing chain. A **Trim Gain** is provided, in order to allow compensation of the signal level post EQ processing.



New in v3.0 is the possibility to adjust the EQ directly from the touchscreen graph points.

## DE-ESSER & LIMITER

A powerful De-Esser algorithm in the Quasar introduces more control over the challenges audio engineers face with sibilance. The controls provided are:

**Threshold:** sets the level at which the De-Esser is engaged.

- **Ratio:** sets how aggressive the processing would be.
- **Depth:** sets the amount of attenuation the processor will apply, after it is engaged.
- **Attack Time:** controls how quickly the processing is applied.
- **Release Time:** sets how slowly the processing is released.
- **Sidechain Filters:** define the bandwidth of the signal that triggers the processor.
- **PFL Output Selector:** allows monitoring of the signal along each step of the processing chain.
- **Limiters:** New in v3.0, is a low-latency limiter (without look-ahead delay), designed to control the fastest peaks of your audio signal right before it is sent to the output busses.



For detailed information about configuring the De-Esser, please refer to Operation Chapter 3

## AUX SENDS

Eight AUX sends are available per channel. The controls are:

- **Gain:** sets the send level. 0dB means no additional gain or attenuation.
- **ON/OFF:** enables the send
- **Post Fader:** makes the send level dependent from the fader level.
- **Post ON:** makes the send activation dependent from the channel's ON state.

The OPTIONS section provides the same functionalities described in the previous screens.



## CONTROL & B/FEED

Backfeed is our definition for return streams. Some source profiles require a backfeed to be generated and this backfeed can be either a mix-minus bus or an individual headphone feed. In either case, the backfeed will be a copy of one of various mixes of the console. In the case of mix-minus for example, the backfeed would be the final mix, minus the source (i.e., Program 1 minus the Codec).

In case a Source is owned by the console, the following three types of control screens are available for a backfeed, depending on its type (set in the Source Profile):

The **DEFAULT M-1 BACKFEED selector** is displayed only in case the Source type is a Codec or a Phone and the Backfeed is set as Default type. It lets you select all the available busses, and it offers an *AUTO* option which is a state-sensitive setting: when the channel is OFF, the PHONE bus is selected. and when the channel is ON, the PGM 1 bus is automatically selected. The Phone bus is a special mix bus created Pre-fader and Pre-ON.



The **CUSTOM BACKFEED selector** is displayed only in case the Source type is a Microphone, a Codec or a Phone and the Backfeed is set as Custom type. It provides:

- Individual Level control for the Left and Right channels of the stereo Feed
- Dedicated source selection buttons for each channel, using the **SET LEFT SOURCE** and the **SET RIGHT SOURCE** buttons.



The **INDIVIDUAL HEADPHONES selector** is displayed only in case the Source type is a Studio Microphone, and the Backfeed is set as Default type. It provides:

- Level control for the HP Feed to the Guest Microphone (Mic Backfeed)
- Dedicated source selection buttons for the internally generated Monitor feeds
- An **EXTERNAL** HP source selector
- An **EXTERNAL MODE** switch to set listening in Stereo, Left only, Right only, or Mono SUM.

This section replicates the controls available on the Quasar Accessory Modules, and allows remote control of those modules, which are typically installed in the Studio.



The other controls provided in this tab are:

- **Backfeed Meter:** Higher resolution alternative to the small one visible on each channel display. Useful to monitor the signal returning to the codec or the guest
- **Backfeed Selector:** appears only in case the Source type is a Codec or a Phone and the Backfeed is set as Default type. Lets you select all the mixes available. There is an *AUTO* option which is a state-sensitive setting: when the channel is OFF, the backfeed is the PHONE mix and when the channel is ON, the backfeed is the PGM 1 mix. The Phone mix is a Pre-fader and Pre-ON mix.
- **Channel Controls:** such as Fader, Automixer, main PGM assignments, Talkback, PFL, channel ON/OFF, and AFL. They mirror the hardware controls found on the channel strip.
- **Dim gain:** determines by how much the main audio will be dimmed when the Talkback is engaged.
- **Preview Feed:** momentarily adds the Backfeed audio to the PFL bus, allowing the operator to check its content.

### Backfeed source ownership

The console will let the operator control a Source Backfeed only in case it owns that Source. In case the Source loaded is owned by another console (already loaded by that console), the Source will appear in “Listen Only” mode: no backfeed controls will be available and instead, a button for requesting Source Unload to the owner will be available.

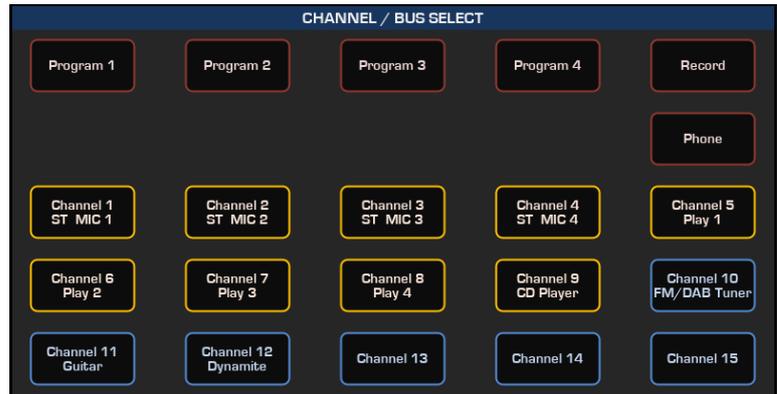
If the button is pressed, a confirmation request will pop up on the screen, and in if the **OK** button is pressed, the Source will be taken from the original owner, but at a condition: the source must be off the air, and the channel where it’s loaded must be OFF.



# TS GUI: Quasar Bus Menu (red tabs)

A new feature in v3.0, the BUS CONTROL Tabs are an additional group of red tabs available when one of the main busses is selected from the Master module's Touchscreen UI.

Busses can now be selected using the Channel & Bus Select screen, activated by pressing and holding the HOME key, or by pressing the **CHANNEL SELECT** button in the clock screen.



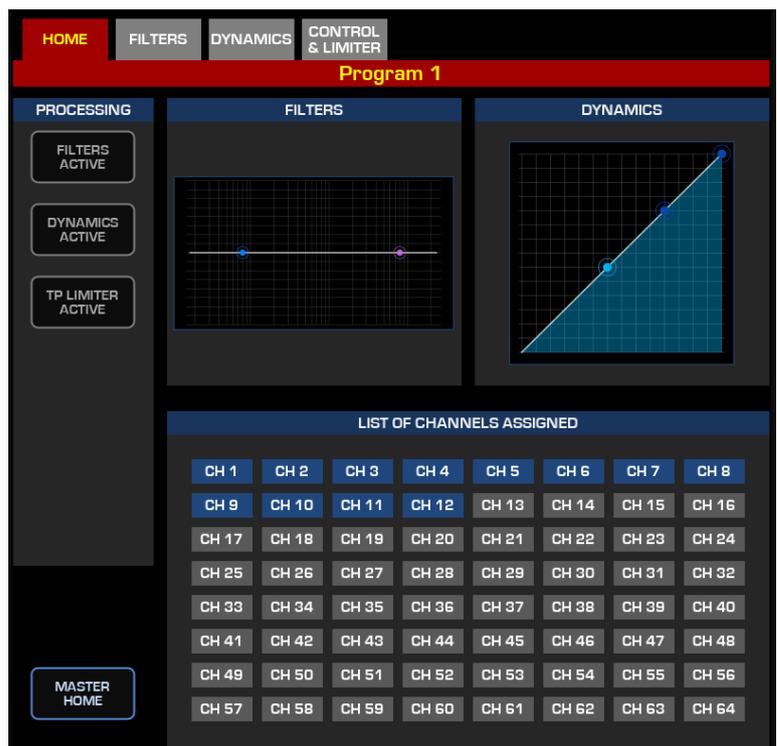
## BUS HOME

This is the default screen visible anytime the bus control mode is active.

There are buttons to enable or disable the various audio processing stages on the selected bus: Filters, Dynamics, and True Peak Limiter

The FILTERS and DYNAMICS graphs provide monitoring at-a-glance of the processing applied to the output signal.

A complete status overview is available to monitor which channel is contributing to the selected bus (also known as "Reverse Interrogation system").



**Note** – The position of the FILTERS, DYNAMICS, TP LIMITER and MASTER FADER tabs in the Bus UI follows the order of the DSP processing blocks within the bus.

## BUS FILTERS

This view provides various controls to the Bus Filters and useful metering information:

- **Input meter:** monitors the signal phase and audio level at the Bus input, before processing is applied.
- **Output meter:** monitors the signal phase and audio level after processing is applied by the two Filters.

The FILTERS section has full-band adjustable HP and LP filters, useful to remove unwanted signal noise from the bus output. It provides:



- **High Pass Freq and Low Pass Freq:** controls to adjust the center frequency of each filter
- **Slope buttons:** can be set to FLAT, 6, 12, 24, 36 and 48dB/Octave
- **Clear, Copy, Paste:** clear or copy/paste settings from the selected channel into a different one
- **Presets:** displays a list of User Presets. The list is editable from the Web UI **Presets** menu.

## BUS DYNAMICS

New and improved DSP algorithms in the Quasar Engine offer more accurate control over the audio giving it smooth and musical sound. The COMPRESSOR / LIMITER section provides:

- **Threshold:** sets the level at which the Compressor or Limiter is engaged.
- **Ratio:** sets how aggressive the processing would be.
- **Limiter:** is engaged when the Ratio is set to more than 50:1
- **Variable Knee:** finely adjusts how smooth the transition across the threshold occurs.
- **Auto Makeup:** a special type of Makeup Gain that adjusts the output gain of the compressor, and manually match the level of the compressed signal to that of the original input signal. This is designed to let you tweak the Threshold and Ratio while on the air, because the output level is always kept constant.
- **Attack Time:** controls how quickly the processing is applied, after the Threshold is passed.
- **Release Time:** sets how slowly the processing is released, after the Threshold is passed.
- **Auto Attack/ Release:** provides optimized Attack and Release time settings, for either music or vocal programs. The Speech mode is optimized to react quickly to voice transients, while the Music mode will provide softer compression times for music programs.



The EXPANDER / NOISE GATE section provides:

- **Threshold:** sets the level at which the Expander or Gate is engaged.
- **Ratio:** sets how aggressive the processing would be.
- **Variable Knee:** finely adjusts how smooth transition across the threshold needs to be.
- **Depth:** sets the limit for the maximum amount of signal attenuation.
- **Attack Time:** controls how quickly the processing is applied, after the Threshold is passed.
- **Release Time:** sets how slowly the processing is released, after the Threshold is passed.

**Trim Gain:** Offsets the overall output level of the entire Dynamics processor by +/- 10dB.

The OPTIONS section provides the ability to copy your settings and to paste those same settings to other channels. To do so, create your settings, press **COPY**, then select a different channel (by pressing its top encoder, or using the MTS Channel Select menu) and press **PASTE**. The **CLEAR** button will clear all parameters and disengage the Dynamics. The **PRESETS** button displays a list of User Presets, editable from the Web UI **Presets** menu.

For detailed information about configuring the Dynamics, please refer to the next Chapter.

# BUS CONTROL & LIMITER

The CONTROL & LIMITER screen allows you to add a final level adjustment to the entire bus. The Bus output level can be manually adjusted in three ways:

- From the MTS touchscreen, via the Soft Fader or MASTER GAIN gauge
- From the hardware encoder that will be highlighted below the GAIN gauge
- From a physical fader of an XR-4FAD module (configured accordingly)

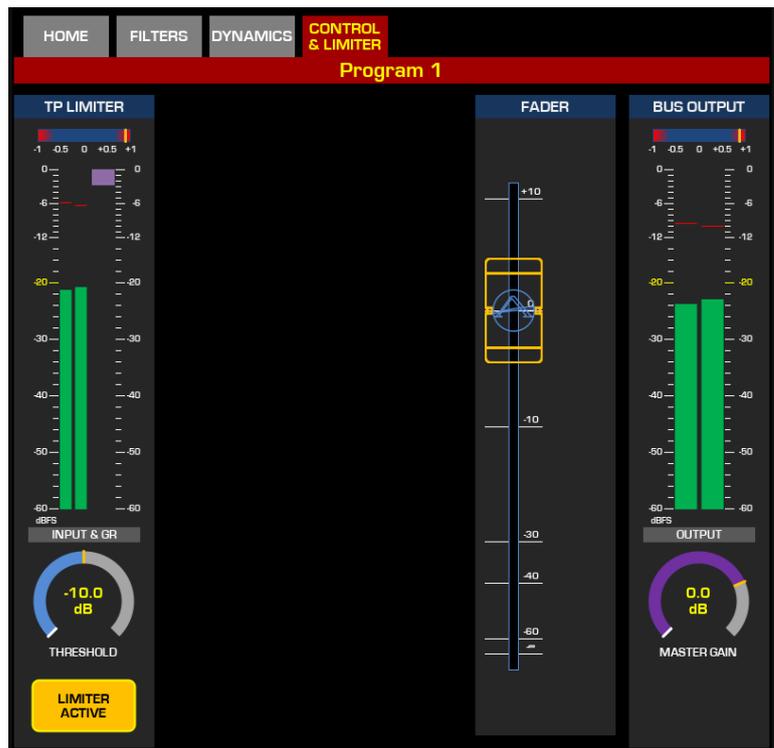
Coupled with the new BUS Assignment functionality that we added to the Layers menu of the XR-4FAD Modules, the Master Gain function allows one XR Fader Module to control Master BUS levels.

The new TRUE PEAK LIMITER is designed to control even the fastest peaks, with minimal overshoot and distortion.

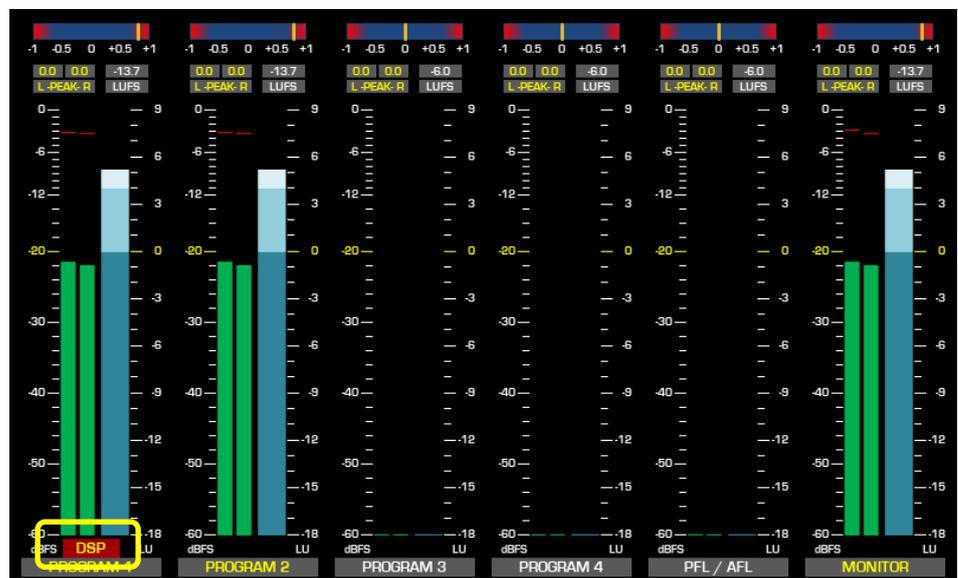
Being a True Peak type, this limiter includes an oversampling function and a Look-Ahead Delay.

## Monitoring when DSP is active on a Bus

Note that the entire Monitoring section picks up the audio right after Bus Filters and Dynamics and before the Limiter and Master Fader, so that the Limiter's delay will not add latency to the audio being monitored and its meters.



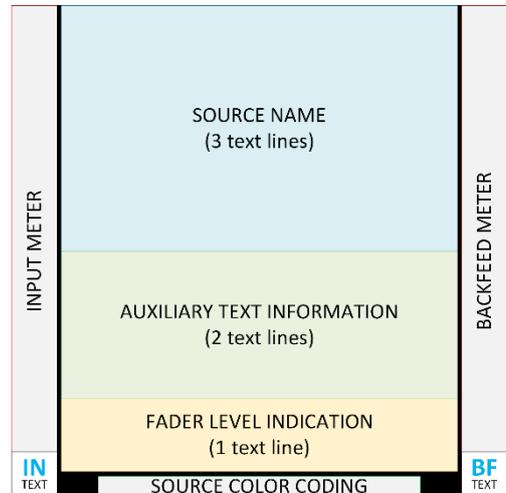
A **DSP** signal will appear below each of the Main Meters, for indicating when any of the available DSP processing is active on the corresponding Bus.



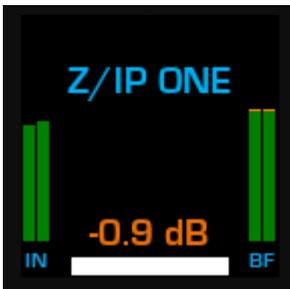
# Channel Strip Display GUI

The Channel Strip Displays are primarily designed with the goal of providing the operator with as much feedback as possible about the status of an input channel. Although small, they have a resolution of 240x240 pixels and can display:

- Source Picture
- Source Name
- Auxiliary Text Information
- Fader Level Indication
- Source Color Coding Bar
- Input Meter
- Backfeed Meter



## Source Name and Picture



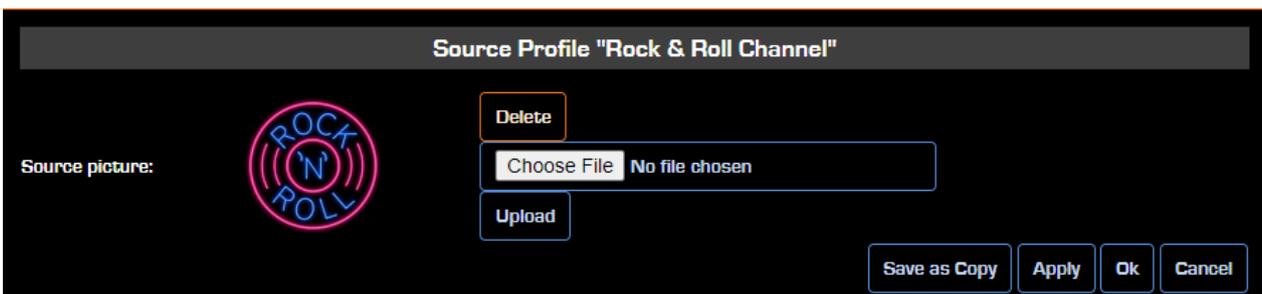
The Source Name is always shown by default, and can appear on up to 3 lines of text. A longer name will be truncated.

Alternatively, if a Picture is associated with the Source Profile at the time of creating a Source, this will appear on top of both the Source Name and Auxiliary Text Information areas.



In order to apply a picture, once a New Source Profile is created, and the basic Source Settings are applied, the page will refresh and a dialog will appear at the top of the Source Profile page from where you can upload a picture.

1. Select a picture file by clicking on the **Choose File** button,
2. Wait until the picture file name appears next to the button.
3. Click on the **Upload** button, and wait for it to appear in the **Source picture:** field.



The picture must be .PNG or .JPG type, with a maximum size of 192x192pixels, and max file size of 8kB, in order to be uploaded correctly to the Profile. You will need to Reload the Source onto the surface, for the picture to appear on the displays.

The touch sensor built in the Channel top knob will automatically hide the picture to reveal the Source name and any messages or indications related to the parameter assigned to that knob.

**Note** - Only one 2-lines text message can be displayed at a time in the Auxiliary Text Area. It is not possible to display two messages. The Gain Indicator has priority over the "Listen Only" message, and in case of a Source in Listen Only mode, with a picture loaded, when the knob is touched and the picture removed, only the gain will be displayed. The fact that in this case it will not be possible to change gain suggests that the Source is in Listen Only mode because this status removes any control over the source.

## Auxiliary Text Information

The Auxiliary Text information can condense a variety of information such as:

1. Knob Function
2. Phone Messages
3. Source Status
4. Change Waiting
5. Source Label

There are five classes of such messages, each with a different priority. The priority determines the order in which they can appear on the display. The higher priority messages will hide the lower priority messages.

### Priority 1 - Knob Function:

- Fader Trim Gain
- Pan level
- Source (Line) Gain
- Mic Preamp Gain
- Automix Weight

### Priority 2 - Phone Messages:

- Caller ID
- Line Ringing
- Transfer

### Priority 3 - Source Status:

- Turn off, clear
- Turn off, load
- Source inactive
- Source not found
- No backfeed
- Listen only

- General error
- Operation error

**Priority 4 - Change Waiting:**

- Change Waiting

**Priority 5 - Source Label:**

- *User text label*

**Note** - Please refer to Configuration Chapter 2 of this manual: [Source Name Override](#) for more information on how to display User Text Labels.

### Description of Priority 3 Messages

**Turn off, clear:** You must turn OFF this channel, in order for the source to be cleared

**Turn off, load:** You must turn OFF this channel, in order for the source to be loaded

**Source inactive:** The Source Profile uses a Source which is not advertised on the network. The corresponding endpoint might be turned off.

**Source not found:** The console is trying to load a Source Profile which is no longer available (Source Profile ID does not exist because it was deleted) or it was de-selected in the Profile's Availability Mask.

**No backfeed:** The Source has a backfeed associated, but failed to create it. Engine might not have enough resources to execute the action.

**Listen only:** The Source is controlled by another console. No GPIO control, Backfeed, and Microphone Control is available. Only its audio feed. Use the REQUEST UNLOAD function if you need to gain control of this Source.

**General error:** Generic console-related error associated to data corruption or mismatch in the Source Profile. It could appear in case the Source Profile Database becomes corrupted. The Surface Log will contain useful information to analyze the nature of this error.

**Operation error:** Generic engine-related error which indicates that the Engine is not acknowledging commands sent by the surface. It could appear in case the console is trying to set some parameters, or value, that the Engine does not recognize as valid, or considers out of range. The Engine Trace will contain useful information to analyze the nature of this error.

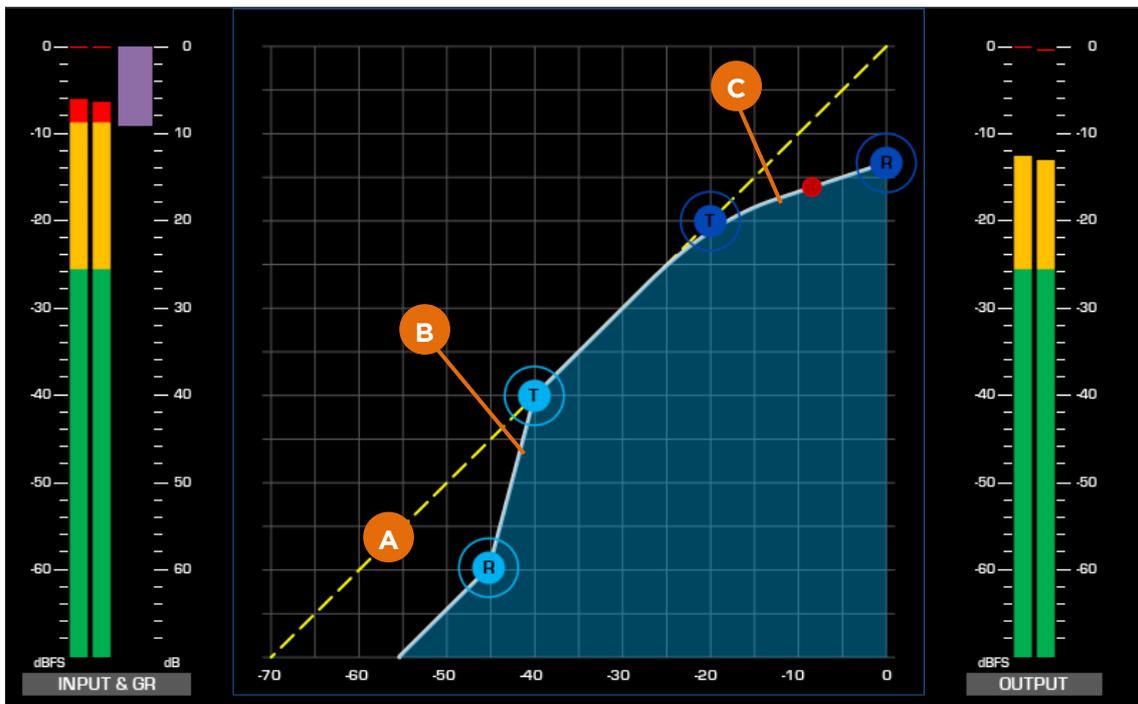
# Operation 3 – DSP & Device Control

## Working with the Dynamics

The Channel Dynamics processor in the Quasar Engine is a very powerful tool: musical, sophisticated, yet very easy and intuitive to operate. Its wide range of settings lets the experienced user not only creatively shape the sound of vocal channels, but also finely adjust the dynamic range of any other audio signal like codecs, phones, or even a playout system.

### Dynamics UI

The UI provides two level meters: input & gain reduction level (left side) and output level (right side). These meters can be used to easily compare pre- and post-processing levels, when setting the compressor/expander.



The yellow dashed line in the graph (A) shows the UNITY GAIN LINE. It represents the signal gain curve when no compression or expansion is applied.

The white line in the graph (B) shows the TARGET GAIN REDUCTION CURVE applied to the signal, based on the Threshold, Ratio, Knee, and Make-Up parameters. The dark blue circled dots represent the compressor THRESHOLD and RATIO settings, while the light blue circled dots do the same for the expander.

**Note** - In SW v3.0 we added the possibility to tweak the curve directly from the touchscreen, by dragging the blue dots with your finger.

Dynamics Processing is a bi-dimensional process: it is not just about controlling the signal output level; it is about *controlling dynamic range over time*. Therefore, we included a special

graphic solution to provide augmented feedback to the engineer, and help him in *real-time* when he's tweaking the parameters: the red SIGNAL DOT.

The SIGNAL DOT (C) continuously shows the audio signal gain reduction vs. its input level, over time, during dynamics processing, thus providing a very accurate feedback about the effective behavior of the signal level, after the Attack time and Release time constants are applied to the gain reduction.

- When the DOT sticks to the white line, it indicates that very fast attacks and release time have been set, and the Dynamics is performing a very tight control of the signal.
- When the DOT tries to follow the white line, but keeps moving below or above it, it indicates that very slow attacks and release time have been set, and the Dynamics is performing a very loose control of the signal.

**Tip** - The SIGNAL DOT is very useful when setting the THRESHOLD parameter: it will show you immediately if the signal is below or above the threshold point.

## Compressor and Limiter

In addition to the standard THRESHOLD, RATIO and KNEE parameters we mentioned, (and described in the previous chapter of this manual) the compressor has the following special functions:

### Auto Make-up Gain

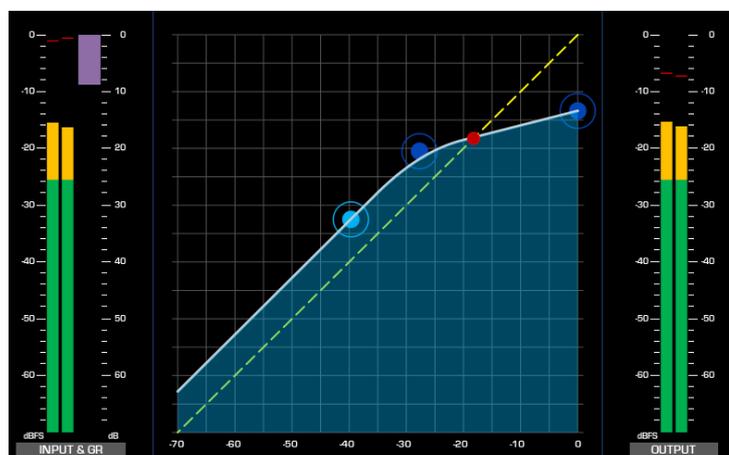
This function, that we designed specifically for broadcast, will let you adjust the output gain compensation *while you are on the air*, without altering the overall level of the audio sent to the program bus. Check the output meter to the right, while adjusting this Make-up Gain, and you will notice it does not change the level after the threshold point, unlike all other compressors, but only before this point.

### Auto Attack/Release

Two broadcast-specific modes for automatically setting Attack & Release times: **Speech** and **Music**. Both include an automatic time control function that reduces the Attack and Release time during fast changes of the signal. This lets the Quasar dynamically adapt the attack and release times to the voice transients or music program.

We recommend you to enable one of these two settings for quickly getting good sound out of the compressor. Doing so will bypass the manual time controls.

**Tip** - Setting very fast compression attack times will shape the signal in such a way that its low-frequency content is reduced. This is perfectly normal. So, if you are setting compression on a microphone, and want to preserve that nice "proximity effect" that makes the voice deep and present, we recommend using Attack times greater than 20ms.



**Note** - In v3.0 the Limiter was updated and moved out of the DYNAMICS page. Now its settings are found in the new DE-ESSER & Limiter tab.

## Expander/Gate

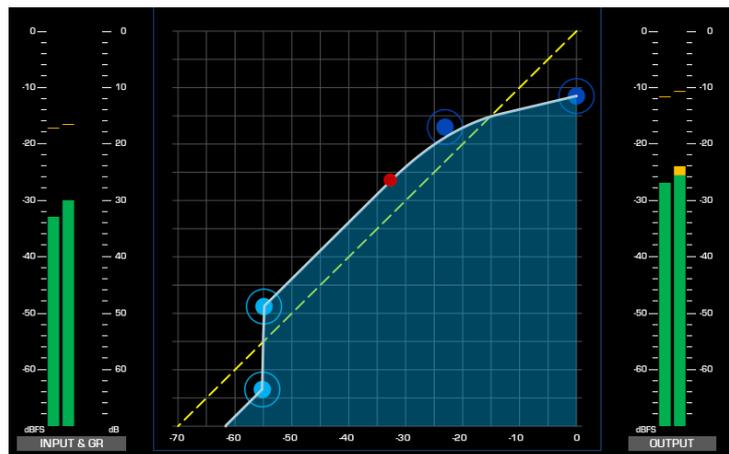
In addition to the standard THRESHOLD, RATIO and KNEE (described in the previous chapter of this manual) the Expander/Gate has the following functions:

### Gain Depth

This parameter sets the limit for the maximum attenuation applied to the signal after the Threshold is passed. Used in conjunction with the Expander RATIO, it will let you seamlessly morph the expander into a noise gate.

### Auto Attack/Release

Like the compressor, the expander offers two Automatic Attack & Release modes for Speech and Music. When expansion is applied to a microphone channel, we recommend using the Attack & Release Speech mode, to avoid lag in silencing the microphone when the talent is not talking to it.

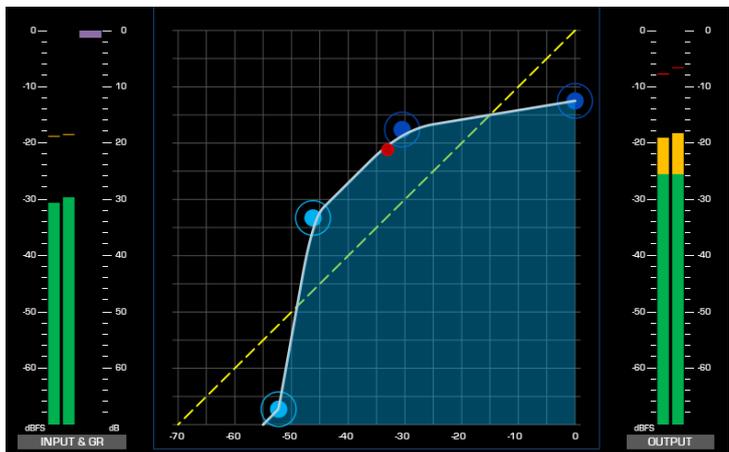


## Some Configuration Examples

### Compressor + Expander

Also known as “Compander”, this is the most common way to process a broadcast microphone.

Applying compression to the Talent’s microphone will help the voice sound closer and more detailed, letting you play with the proximity effect of the microphone to generate deep and warm voice tone, while controlling peaks, and make it stand out from your mix.



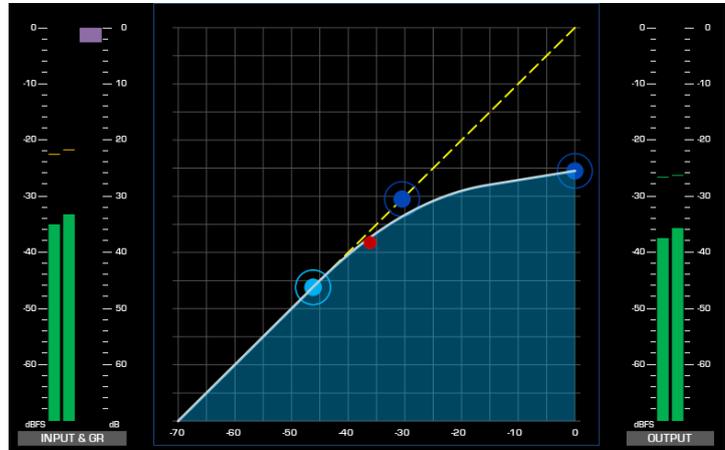
But at the same time, compression has the unwanted effect of bringing up the lower end of the signal, increasing the background noise, the leakage from nearby talents speaking to other open microphones in the room, and amplifying any ambient reverberation. You will need to add an expander, to prevent this from happening.

With Quasar, you can *combine* the compressor with the expander, and control both of them from the same view, in order to treat both ends of the signal range – the one you need to enhance, and the one you may need to attenuate – at the same time.

The complex part is adjusting the Expander THRESHOLD: that's where our Signal Dot comes in to help. You want to make sure the threshold point is not set too high, or the expander will kick in too late, attenuate only portions of the signal, across transition between silence and speech revealing its effect.

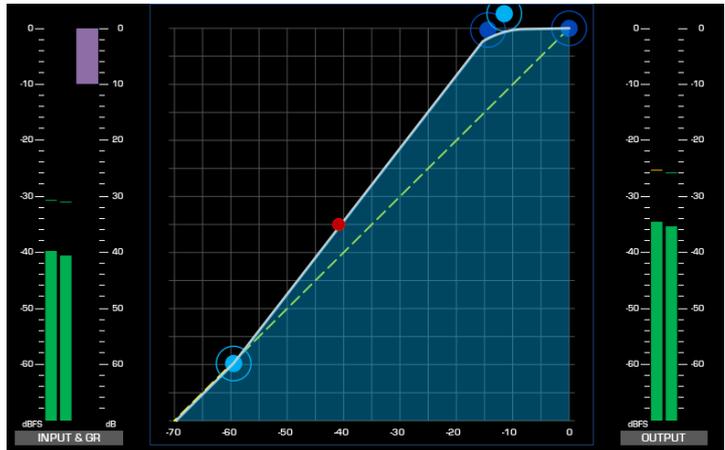
### Variable Ratio Compressor

Would you like to have a compressor which has very gentle gain reduction effect when the signal has just passed the Threshold, but very strong effect in case the signal goes too far beyond? You can set the compressor KNEE to its maximum level and play with the RATIO until you bend the compression curve.



### Upward Expander

There are some cases when the audio signal sounds “squashed” because it has a very limited dynamic range, (for example, if it’s overly compressed at the source, or comes from an FM tuner, or an old analog tape). In these cases, it would be useful to decompress the signal, and increase its dynamic range to make it more consistent with the other channels in the mix.



You can combine the action of the Compressor and expander in such a way to obtain a so-called “Upward Expander”:

Set the compressor THRESHOLD so that the ramp starts from the unity gain line, set AUTO-MAKEUP to the maximum, then adjust both compressor and expander THRESHOLD to about the same value, making sure this is well below the signal lowest level. Then carefully adjust the expander Ratio to values between 1 and 2.

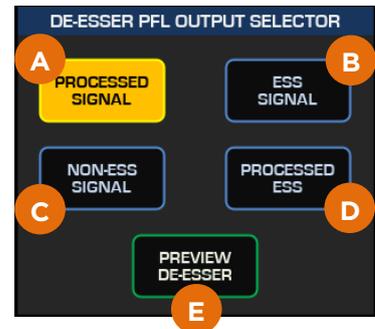
The result is that the signal is never attenuated, and it is actually “pumped up” a bit, as most DJs do when they push the fader up following the music beat. This will actually increase the signal’s dynamic range.

## De-Esser

The Quasar De-Esser was designed to precisely target the voice's sibilants or plosives, and remove them without altering the spectral balance of the remaining part of the signal. While most of the de-Essers in the market compress the entire audio signal when the sibilants are detected, this algorithm will extract the unwanted frequencies, process only that portion, and then mix it with the rest of the signal, producing a much cleaner sound.

### De-Esser Sidechain

The key to successfully remove sibilants from a signal, is to precisely set the sidechain filters so that the detector can work more accurately, and trigger the compression only when the sibilant occurs. For this reason, we provided a complete set of sidechain listen points, to let you monitor the signal along the entire processing path, through the PFL bus. The sidechain listen points are:



- **Processed Signal:** For listening to the entire signal through the PFL bus, after De-Esser processing is applied. (A = C + D)
- **Ess Signal:** For listening to the raw sidechain signal only, after the filters, before processing is applied. (B = Sidechain Signal)
- **Non-Ess Signal:** For listening to the portion of the input signal which is left unprocessed by the sidechain filters. (C = Input Signal - Sidechain)
- **Processed Ess:** For listening to the Ess Signal after processing is applied (D = Processed Signal - Non-Ess signal)
- **Preview De-Esser:** for quickly listening to the De-Esser output through the PFL bus.

### Setting The De-Esser

In order to set the De-Esser:

- Make sure the **PFL to LS** or **PFL to HP** function is enabled in the Monitor section
- Select the desired input channel
- Select the DE-ESSER tab and press the **DE-ESSER ACTIVE** button.
- In the DE-ESSER PFL OUTPUT SELECTOR section, press the **ESS SIGNAL** button then press and hold the **PREVIEW DE-ESSER** button to listen to the ESS signals
- In the SIDECHAIN FILTER section, adjust the filters searching for the ESS portion, by narrowing down the selected range of frequencies until you hear only the unwanted Ess frequencies. This is your "Ess Signal".
- In the DE-ESSER PFL OUTPUT SELECTOR section, press the **PROCESSED ESS** button, then press and hold the **PREVIEW DE-ESSER** button to listen
- Adjust the THRESHOLD by rotating the corresponding encoder backwards, from the 0.0dB position, until the gain reduction meter located in the top left area of the screen starts working.
- Set the **Ratio** to the desired value (higher values apply more aggressive compression)
- Set the **Attack** and **Release** times (lower values will provide faster response).
- Set the **Depth** to fix a limit to the gain attenuation applied to the Ess Signal.

## Channel and Bus Limiters

With SW v3.0 we de-coupled the Limiter from the Compressor, and we added it as a separate processing block at the end of the channel's signal chain. This will let you enable the Compressor and the Limiter on a channel at the same time, for shaping different parts of the signal with both these tools. We also added True-Peak Limiter to the final stage of the console Program 1-4, Record, and Phone Busses

### Channel Low-Latency Limiter

The Channel Limiter has a brand algorithm which is optimized for the console inputs, and features a very low-latency so as to not delay the audio signal before it is sent to the output bus and the monitor section.

In order to achieve this goal, we chose to not include a look-ahead delay, therefore, we suggest setting the threshold of this limiter in such a way that it will only work on the loudest peaks:

If the input gain (and all the other Channel DSP processing) is set correctly, and the signal is loud enough after the de-esser, the Limiter's threshold should never be set below -10.0dB, to avoid "squashing" the signal. The gain reduction meter in this case, should never go beyond the -10dB mark.



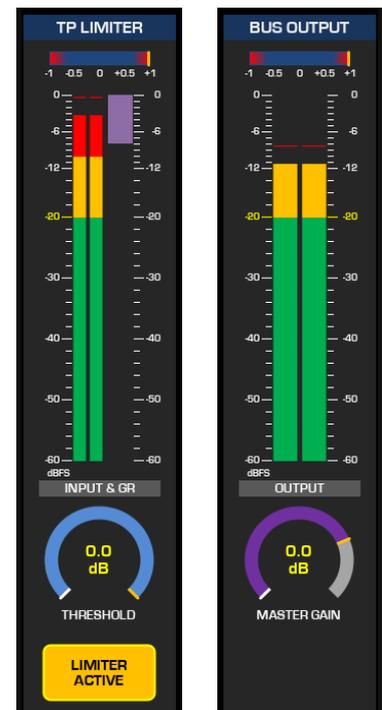
If the input level of the signal is such that very low thresholds are required, in order to see some gain reduction activity on the meter, we suggest adjusting the input gain or the previous channel processing stages to "beef up" the signal, maintaining the proper operating levels throughout the entire input channel.

### Bus True-Peak Limiter

In the Bus Limiter we privileged sound quality and signal density: this True Peak Limiter adds a small amount of Look ahead delay and applies gain reduction in a softer way and proportionally apply limiting so that peaks do not arrive to a point where they can pass the threshold. This is what we do, in order to reduce distortion, below audible levels (but when the waveform is modified, technically it is distorted, it's inevitable).

**Note** - For a limiter, the Threshold sets the point after which there should be no peaks in the signal. It is not the point where the limiter starts working. This limiter might start working (reducing peaks) before the signal reaches the set Threshold if the signal has a high dynamic range which exceeds a certain Peak/RMS ratio

**Tip** - The Bus TP Limiter could be left Active, and with the Threshold set to 0.0dB just to make sure no transients will cause distortion to the signal. In this case it will process audio without being audible, and will sound extremely transparent.



# Working with the Automixer

The Automixer function can be used to automatically control the levels of a selection of channels (mono or stereo) , keeping the overall level of the mix constant.

In order to achieve this goal, the Automixer *has the effect* of boosting channels with higher signal levels relative to other paths in the group, and attenuating those with lower signal levels. But in reality, the Automixer only ever applies attenuation, and signals' gain is never actually increased.

An exclusive gain sharing algorithm compares each input level with the sum of all inputs and adjust the gain of each mic so that the potential acoustic gain of the sum off all mics is constantly equal to that of one mic. The attenuation is applied by an additional gain stage that is placed after the Fader level control, while the detector which drives the AM can be placed pre- or post-fader. The default setting is Post-Fader. In this case, the amount of attenuation remains proportional to the fader balance set by the operator.



In broadcast, typical examples of usage scenarios for this feature, are:

- Providing automatic voice-over control (where a DJ speaks over the music). This feature is also known as “auto ducking”.
- Controlling the attenuation of the Guest microphones while the Host (or Presenter) is speaking in a talkshow where all the microphone channels are kept open, it can be used to , in order to keep the level of the overall program constant
- Improve signal to noise ratio of a group of open microphones (noise is the room ambience).

In all of the above cases, the Automixer will produce a mix in which the total ambient (or background) noise level remains fairly constant.

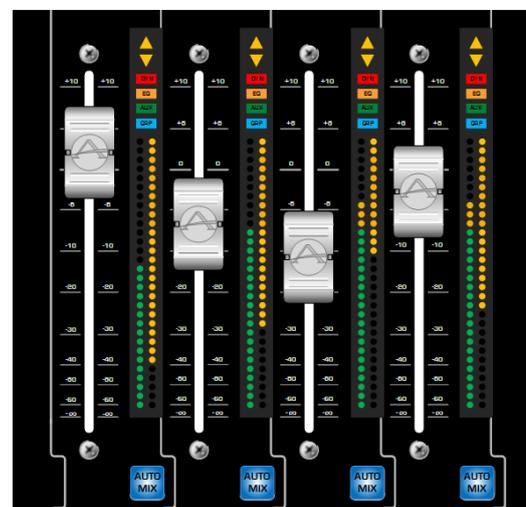
A “Weight” factor can be applied individually to each channel: the higher the weight, the more priority is given to that channel in the ratio calculation, giving it more prominence within the mix.



For example, in the ‘talkshow’ use case described above, if all contributors talk at the same time you may wish to give a higher weight to the Presenter’s microphone so that his voice is always heard during the discussion.

A gain reduction meter is provided right next to each fader (XR-4FAD modules) so you can quickly see how much gain reduction is currently being applied to that channel.

Quasar offers two Automixer groups, one for the Program outputs and one for the Record mix. The settings of these two groups are always linked, and not independent. The Record Automixer is automatically enabled if the fader channel is assigned to the PGM4/REC bus.



# Working with the Intercom Controller

Operating the Quasar Intercom controller requires no special knowledge, other than that required to operate the Infinity Intercom panels. The operational workflow is identical to that of the Infinity MP-16 panel (hardware) or VIP panel (virtual): the same operating controls are simply duplicated on the Quasar's Touchscreen.

The Intercom Controller is a pure remote controller:

- There is no dedicated DSP processing happening in the Quasar Engine in order to make the console interact with the Infinity system.
- The Intercom Controller and the physical (or virtual) Infinity panels it connects to, can be operated in parallel.

**Note** - The Intercom Controller in Quasar does not support the legacy Axia Intercom system. It only supports Telos Infinity Intercoms.

Please refer to Operation Chapter 2 of this manual: [INTERCOM CONTROL](#), for a detailed description of the operating controls. You can also refer to Configuration Chapter 5: [Setting Up the Intercom Controller](#), for comprehensive setup instructions.

# Working with the Record Modes

Record Modes consist in a special “macro function” that help the operator quickly prepare to record Jingles, phone bits, interviews or other program segments for later airing. Any source assigned to the Program 4 bus automatically feeds the Record and Phone busses as well.

Quasar offers two different types of Record Mode: Standard and Flexible. These options are selected in the **Record Mode** menu of the Show Profile main page.

The above settings are explained in more detail in the Configuration Chapter 2 of this manual: [Record Mode Configuration page](#).

## The Standard Record Mode

The **Standard Record Mode** can be configured and enabled in the Show Profiles, on a global, per-Show basis. When configured, in order to enter the Standard Record Mode:



- Load the Show you have configured for this mode
- Press the **RECORD** key in the bottom right corner of the Master Monitor module.

When this mode is turned ON:

- The hardware **RECORD** key will turn Red
- The “REC” indicator will turn on in the MTS Home page
- The **PGM4** keys on each channel will change to DARK RED color, to indicate that they have been activated by the record mode.
- Record Bus is selected in all Monitors: CR SPK, CR HP and STUDIO SPK.
- In the Control Room Headphone section, **HP Mode** is automatically set to SUM
- In the Control Room Headphone section, **PFL to HP** is turned OFF
- The corresponding GPIO triggers will be executed.



The different color of the **PGM4** keys will also remind the operator that the Record Mode is active, and that PGM4 key is actually controlling the special RECORD bus, and therefore it will mute the CR Monitors if turned on.

When this mode is turned OFF:

- All parameters that were configured to change state (on entering), will revert back to their original states. This includes monitor settings.
- The RECORD key will turn OFF
- Corresponding GPIO triggers will be executed.

**Note** - When Standard Record Mode is turned OFF the console will not return to the entire state it had before Record Mode was activated. Only those settings changed by the Record Mode configuration will be reverted, but none of the changes made by the operator.

For more detailed information about configuring GPIO for the Record Mode, please refer to Configuration Chapter 7 of this Manual: [GPIO Record Mode Logic](#).

## The Flexible Record Mode

When the options available in the Standard Record Mode settings do not suit your workflow, you can use the Flexible Record Mode. The **Flexible Record Mode** options are defined in two places:

1. The Monitoring options are set in the Show Profile Global settings,
2. The Channel options for that Show must be configured in each of the Source Profiles

When this mode is turned ON:

- The hardware **RECORD** key will turn ORANGE.
- The “FLEX” indicator will turn on in the MTS Home page.
- MTS Home page will turn the “FLEX RECORD” indication on.
- All parameters that were configured to change state, will follow the options set in the “On Entering” section.
- The **PGM4** keys on each channel will not change color, to indicate that they have been activated by the record mode.
- Corresponding GPIO PIO triggers will be executed.



When this mode is turned OFF:

- All parameters that were configured to change state, will follow the options set in the “On Exiting” section.
- The RECORD key will turn OFF.
- Corresponding GPIO triggers will be executed.

Please see Configuration Chapter 2 - [Show Profiles Record Mode Configuration Page](#), for configuring Monitor Options in the Flexible Record Mode.

## Configuring Sources for Flexible Record Mode

The Flexible Record Mode lets you configure, per source, the state of all the main settings (Ch ON/OFF, PGM, AUX), upon entering and exiting the Record Mode.

In addition, you can use Custom Backfeeds to route different audio signals to the Left and Right channels. This could be done also when using the Standard mode, although it's less common.

## Configuring Enter and Exit States

Navigate to the **Flexible Record Mode Options:** section of the Source Profile.

The options available here will let you independently Retain, Recall, or Assign a pre-defined state to each of the listed channel controls.

	On Entering	On Exiting
Source:	Retain	Retain
On switch:	Retain	Retain
Program 1:	Retain	Retain
Program 2:	Retain	Retain
Program 3:	Retain	Retain
Program 4:	Retain	Retain
AUX Send 1:	Retain	Retain
AUX Send 2:	Retain	Retain
AUX Send 3:	Retain	Retain
AUX Send 4:	Retain	Retain
AUX Send 5:	Retain	Retain
AUX Send 6:	Retain	Retain
AUX Send 7:	Retain	Retain
AUX Send 8:	Retain	Retain

Buttons: Save as Copy, Apply, Ok, Cancel

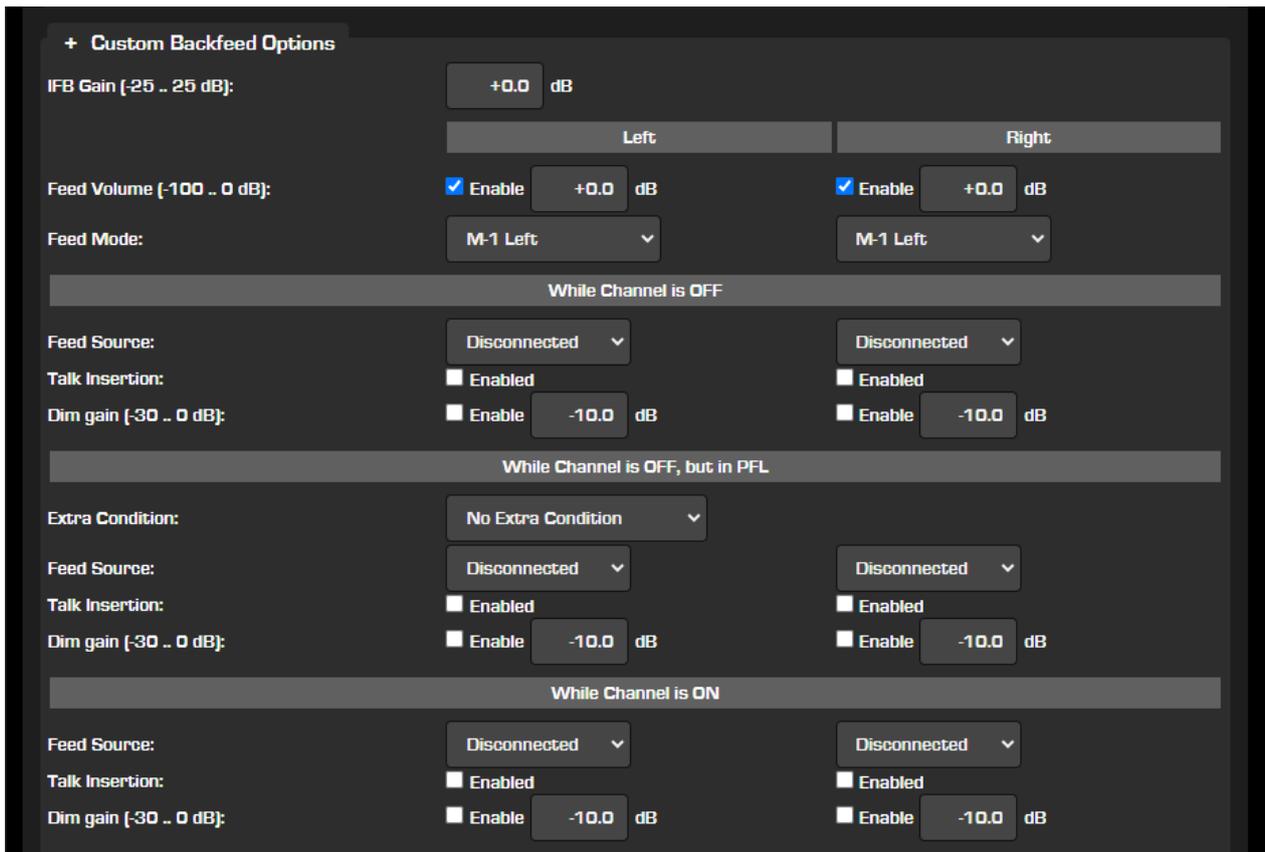
- **Retain:** Will leave the current state of the selected control unaltered.
- **Turn OFF:** Will turn OFF the selected control when the Record mode is activated (“On Entering” column) or de-activated (“On Exiting” column).
- **Turn ON:** Will turn ON the selected control when the Record mode is activated (“On Entering” column) or de-activated (“On Exiting” column).
- **Recall:** Keeps in memory the state of the parameter while Record Mode is Active and Inactive, and will recall it each time the Rec Mode is activated or deactivated.

## Routing Backfeed Audio

Navigate to the **Feed to Source:** section of the Source Profile.

The following options are available only to Sources with a Custom Backfeed configured. They will let you independently assign any Backfeed source to the Left and Right channel when the Record Mode is engaged, based on the following Channel statuses:

- While Channel is OFF
- While Channel is OFF, but in PFL
- While Channel is ON



**Note** - The **Backfeed mode**, in the Source Logic section of the Source Profile must be set to “Custom” for these options to appear.

For more detailed information about configuring the Flexible Record Mode, and the Custom Backfeed options, please refer to Configuration Chapter 2 of this manual: [Feed to Source](#)

## Standard Vs. Flexible Record Modes

In case Standard Record Mode is selected, the logic is associated to the Show Profiles:

- the options that define the channel behavior must be configured in each Channel of the Show Profile.
- The Standard Record Mode allows you to define a state change (for a Source parameter) only when the mode is entered (activated). On exit, the state will be reverted.

In case Flexible Record Mode is selected, the logic is associated to the Source Profiles:

- you will have to configure the behavior in each Source, via the **Feed to Source** options section.
- The Flexible Record Mode allows you to define different state changes for both enter and exit. This is because Flexible Record Mode has the ability to recall (switch between) different Sources within a certain input channel.

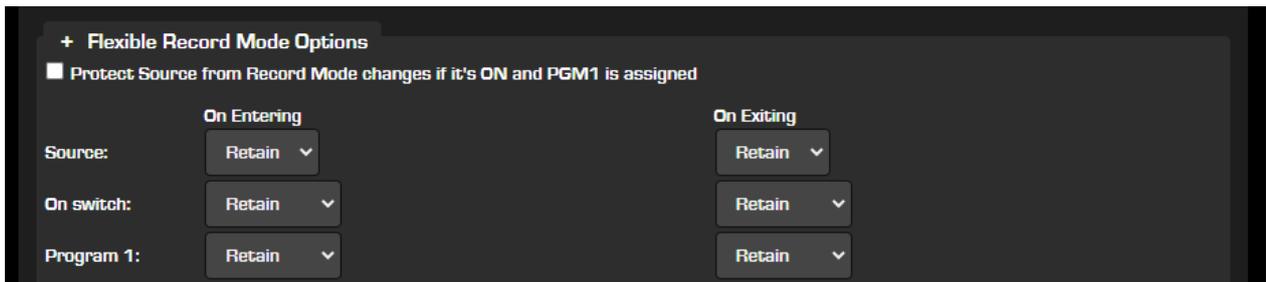
## Source Load Priorities

The Flexible Record Mode Options saved in a Source's Profile, are only loaded in one of the following two cases:

- When the Source is loaded by a Show Profile
- When the Source is loaded manually and Record mode is OFF

**Note** - When a Show is loaded, Record Mode is *a/ways* OFF: a Show cannot be loaded while RECORD is ON.

Basically, if a new Source is loaded by the operator while Record Mode is ON , the Flexible Record Mode Options saved in the new Source will not be loaded, or applied.



## Flexible Record Mode Source Protection

A new option was added in SW v3.2, to protect a Source from any changes made by the activation of the Flexible Record Mode to the Source's bus assignments, while the Source is on the air.

Selecting the **Protect Source from Record Mode changes if it's ON and PGM1 is assigned** checkbox, will prevent the Flexible Record mode to execute any of the routing options selected configured, in case the Source is ON and PGM1 is assigned. If only one of the two conditions is not verified, this protection will not be applied, and the Flexible Record mode will execute the changes programmed in the **On Entering** and **On Exiting** drop-down menus.

# Operation 4 – Accessory Modules

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## Working with the Accessory Modules

The Accessory Modules are multipurpose devices that can be used in a variety of applications. Being connected via Ethernet, they can be installed in the Studio, in a Producer's desk in the Control Room, or in any other remote location.

The main concept behind their operation is what we call "Control Modes". These are different configurations of the module's internal software, used to tailor the hardware module to a specific use case.

The way the Accessory Modules interact with the user depends on a combination of two settings:

- 1) The **Control Mode**, selected in the Module's Web UI
- 2) The **Source Backfeed Mode**, selected in the Source Profiles

Both the MPC-ACC Module, and the MF1-ACC Module have the following Operation Modes:

- Studio Guest Position Control Mode
- Channel Strip Control Mode
- V-Mixer Control Mode

The basic difference between these three modes, from a logical standpoint, is:

- In Studio Guest Position Mode, the Accessory Module is linked directly to a Source and will follow the Source when this is loaded by different consoles (in case of Switching between different Studios)
- In Channel Strip Control Mode, the Accessory Module is linked to a console channel and will remain attached to the console (linked to its MTS module) even in case the source is unloaded and loaded by a different console
- In V-Mixer Control Mode, the Accessory Module is linked to the console Engine.

**Note** - Operation Modes are set and configured through the Web UI, and not during normal operation. More Control Modes are in the works. Stay tuned with us!

### Operation in Multi-Room Environments

In a multi-Room environment, where multiple Control Rooms (so multiple consoles) share the same Studio, and therefore the same ACC modules, there can be two cases:

1. The ACC Module is set to operate in Studio Guest Position mode. In this case the Master Address does not need to be changed, when the Studio is "taken by", or switched to, a different Control Room. It can be left to any of the other consoles, (as long as they are turned on) because the control will follow the Microphone Source.
2. The ACC Module is set to operate in Channel Strip Control mode. In this case the control will follow the MTS-MON and it will be necessary to switch the IP address to connect the ACC module to the active Console.

## Working in Studio Guest Position Mode

The Studio Guest Position mode is the default Control Mode. The primary scope of this Control Mode is to let the Studio Talent control his own Headphone feed with Source selection and Volume adjustment, as well his own microphone ON-OFF state.

**Note** - With this operation mode, it is not possible to Remap the function of each key, as it can be done with the Channel Strip Control Mode.

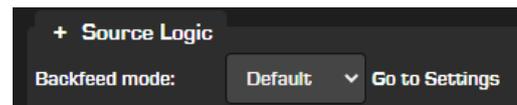
When this mode is selected, the module works like an Ethernet-Connected GPIO panel, and it uses an extension of the LWGPIO protocol to communicate directly with the Quasar Engine.

In this case the module is linked directly to a Source, allowing multiple consoles to connect to a single panel, depending on which console owns the source. So, recalling different Show profiles in different consoles will associate the panel to different consoles.

**Note** - In this mode it is not possible to freely remap buttons, like it is in Channel Strip Control Mode. No user-defined functionality is possible.

## Display UI in Studio Guest Position Mode

Depending on the **Source Backfeed Mode** selected in the Source Profile, the module will display two different User Interface graphics:

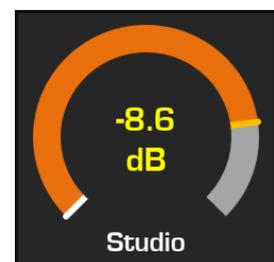
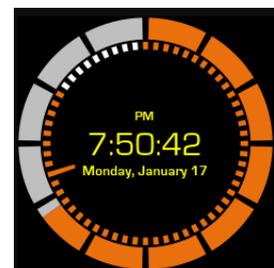


### Default Backfeed UI

The Default Backfeed is the most common setting, and covers the majority of applications. It offers a simple and intuitive screen for visualizing and adjusting your HP Volume.

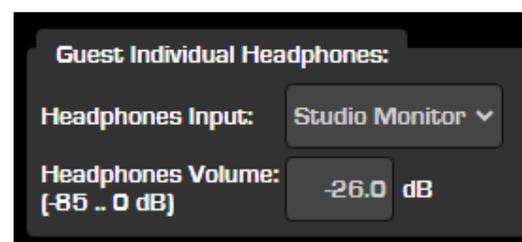
A clock can be displayed, and will automatically disappear each time the top encoder knob is touched for operation, or when a Source list is displayed. This option is enabled by selecting the **Clock Display** checkbox.

Upon touching the knob, the Volume gauge for the selected feed will be displayed, along with its name. The level adjustment is stereo linked.



Push the encoder to display the Headphone Input source list. Scroll the list and push the encoder knob to select the desired source and return to the volume control screen.

These sources correspond to those available in the **Headphone Input** drop-down menu of the Guest Individual Headphones section, in the Channel's Show Profile.



Before v2.1 was released, the Accessory Modules have been shipping with two keys labeled “USER 1” and “USER 2”, located between the encoder and the display. These keys were only active when the Channel Strip Control Mode is selected, and were inactive when the Studio Guest Position Mode was selected.

In software v2.1 we introduced a new functionality for these keys, when operating the modules in Studio Guest Position Mode. The new Accessory Modules are now shipping with new caps labeled “EXT 1” and “PFL” installed by default, and with the “USER 1” and “USER 2” supplied as spares (for those who need to run the ACC module in Channel Strip mode). Existing users can check with their preferred Axia Dealer for the availability of a conversion kit.

Press **EXT 1** key (was USER 1) to display a list of External sources. Scroll the list and push the encoder knob to select the desired source and return to the volume control screen. You can define what Source will appear on the External Source list of the ACC Modules by flagging the “Guest HP” checkbox in each of the Source Profiles.

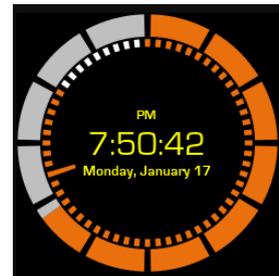


Press **PFL** key (was USER 2) to listen to, or preview, any of the sources, while scrolling the Internal, or External Source lists. The action of this key is always momentary.

**Note -** The **EXT 1** and **PFL** keys are active only when the Default Backfeed type is selected, in the corresponding Source Profile.

### Custom Backfeed UI

You must set your Source to enable Custom Backfeed when you need to select different feeds for your HP Left and Right channels.



A clock can be displayed in this case too, by selecting the **Clock Display** checkbox.



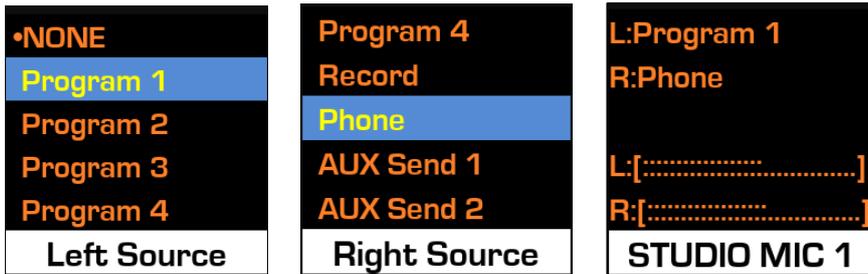
Upon touching the knob, two volume level bars for the selected feed will be displayed. The Volume level, and Source selection, can be adjusted independently for the Left and Right channels.

The bottom label in this case displays the primary source name, while the feed name for each channel is displayed at the top.

In the case of the MF1-ACC module (with Fader), the bottom label will turn yellow every time the fader is touched.

Push the encoder once to select a source for both channels. Push multiple times to select the Left Channel only, or the Right only, then rotate to select the desired Source for that channel. Rotate the encoder to scroll the list and push again the knob to confirm your selection.





Custom Backfeeds offer the possibility to set different source assignments (routing) for each of the following Channel states:

- 1) Channel is OFF
- 2) Channel is OFF, but in PFL
- 3) Channel is ON
- 4) Channel is OFF with Record Mode Active
- 5) Channel is OFF, but in PFL with Record Mode Active
- 6) Channel is ON with Record Mode Active

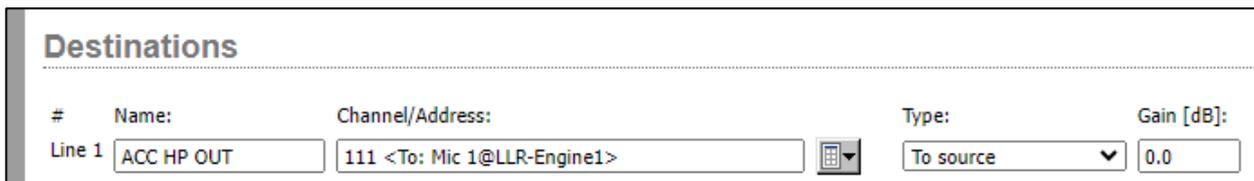
Every time the status of the channel is changed, the backfeed will automatically change according to what is configured in Show Profile for those Custom Backfeed Options, and consequently, the HP feed will be switched.

For more details about setting different Backfeed Modes, please refer to Configuration Chapter 5 of this Manual: [Setting Up Mix-Minus](#).

**Note** - In this mode it is not possible to freely remap the module's keys, as it is in the Channel Strip control mode.

### Routing the Backfeed Audio

In order to listen to the backfeed audio selected with the Accessory Module, you will have to route the Source Backfeed to the desired physical output of the xNode which feeds your HP amplifier. This is done in Destination menu of the xNode by selecting the desired Source in the Channel field and choosing the "To Source" from the drop-down menu in the **Type** section.



### Studio Guest Position Talkback

The Talkback function is activated by pressing the **TB** key. This key has different functions, depending on the selected operating mode and on the type of Source.

- In Studio Guest Position Mode, the **TB** key activates the Talk to CR function
- In Channel Strip Control Mode, the **TB** key activates the Channel Talkback function

The **Talk to CR** function, active when Studio Guest Position Mode is selected, is only available from CR Guest Mic, Studio Guest Mic, External Mic, and Codecs Source types. It is not available from the Operator and CR Producer Microphones Source types.

- 1) The “Default” Backfeed mode is not available to the Operator Microphone, CR Producer, Computer Player and Line source types.
- 2) The “Custom” Backfeed mode is available to all types of sources.

Detailed Talkback operation is described in the Configuration Chapter 5 of this Manual: [Setting Up The Console Talkback](#).

**Note** - When this operation mode is selected, if the Accessory Module is rebooted (either by removing and reconnecting its cable, or by performing a soft reboot from the Web UI) its HW controls will work immediately but the module’s LCD display will show a black screen with a white bar at the bottom, instead of the normal level control. This is normal, and you just need to push the encore twice, or select EXT, in order to display the standard controls.

## Working in Channel Strip Control Mode

In channel strip mode, the Accessory Module replicates the main controls available on a 4-Fader Module. It simply works as an additional fader strip to the surface, and can be assigned to control any of the Quasar Engine input channels.

An Accessory module can be set to work in Channel Strip Control Mode in order to:

- Control one of the Engine Input channels which are already loaded onto a surface (to give someone else, like a Producer, parallel control of that channel strip). In this case the controls will be mirrored.
- Add one more channel strip to an existing surface and control an Engine Input channel which is not loaded onto that surface

In both cases, operating the Accessory Modules when Channel Strip Control Mode is selected, is exactly like operating one of the Console Channel Strips, but with a reduced hardware functionality: some function keys found on the XR or SR channel strips are simply missing from the ACC modules, and only the essential controls are present.

**Note** - The individual HP Monitoring functions are not available in this operating mode.

### Channel Strip Mode Control UI

In this Operation Mode, the front display UI looks and works exactly like that found on one of the Quasar XR-4FAD module’s channel strips. Operation of all controls is the same as that of a single channel strip found on an XR-4FAD module.



### Channel Strip Control Keys

In this Operation Mode, the Hardware Keys operation is the same as that of any other Channel Strip on the console, and depends on the function defined in the Hardware Keymap Menu of the module’s web UI.

**Note** - When this operation mode is selected, the Accessory Module Key Mapping, and the keys located between the top Encoder and the Display will assume the USER 1 and USER 2 function. These keys will work with the same functionality they have in the corresponding console channel strip (if assigned from Pathfinder).

## Working in VMIX Control Mode

This operation mode is introduced with the Quasar v3.0 update. When this mode selected, the ACC Module is capable of controlling any of the five input channels of a VMIX instance, by communicating directly with the Engine using LWCP over TCP/IP.

### V-MIX Control Mode UI

In V-Mix Control Mode, the display UI switches to show the Input Gain level. When the encoder is pressed, it is possible to control the Input Selector of the selected VMIX input channel, and choose one of the internal sources generated by the engine from the list.

Record  
Phone  
**AUX Send 1**  
AUX Send 2  
AUX Send 3

Rotate the encoder to select an internal bus, and press to apply the selection.

+7.0 dB  
AUX Send 1

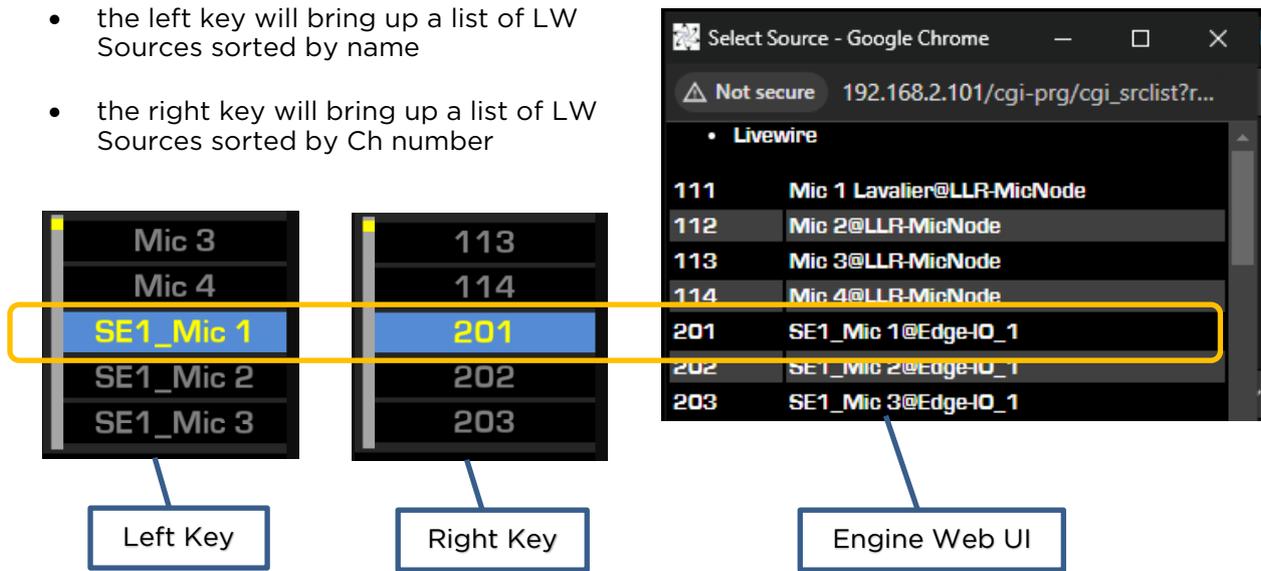
Once the selection is applied, the UI returns to Input Gain adjustment

ON  
OFF

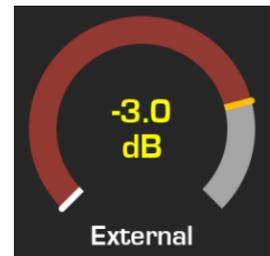
Input 1  
Input Selector: AUX SEND 1  
External Input Name  
External Input LW Ch#/IP Address  
External Input LW Mode / AES67 Stream Type  
Input Channel Map (Left from: / Right from:): 1st ch 2nd ch  
Phase: Normal  
Gain (-128.0...10.0dB): +7.0 dB  
Pan (-100%...0...+100%): 0% CENTER  
Fade In Time (0.0 to 60.0 sec): .5 ON  
Fade Out Time (0.0 to 60.0 sec): .5 OFF  
Direct Out Type: Livewire  
Direct Out Name: VMIX 16 fader 1  
Direct Out LW Ch#/IP Address: 10661  
Direct Out LW Mode / AES67 Stream Type: - Disabled -  
Direct Out Channel Map top: 1-3-5-7 bot: 2-4-6-8  
Export SDP

If instead, you wish to select a LW Source from the network, you can press one of the two Keys below the encoder knob (above the display):

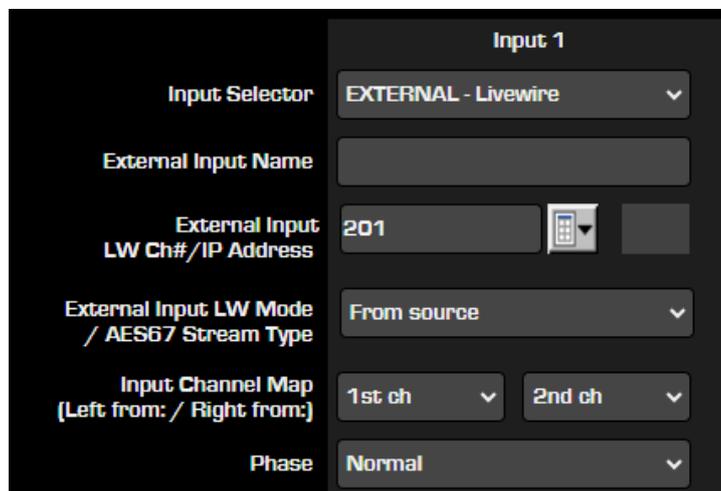
- the left key will bring up a list of LW Sources sorted by name
- the right key will bring up a list of LW Sources sorted by Ch number



In this case, once the External Input selection is applied, the display will show the “External” indication below the level gauge



Navigating to the VMIX Web UI, this selection will appear as shown in this picture:



Remember: in case you are using an Accessory Module with Motorized Fader (MF1-ACC Module) the Fader will automatically control the VMIX Sub OUT.

**Tip** - You could create a very basic 5-Channel Mixer using 4x MPC-ACC Modules and 1x MF1-ACC module.

# Operation 5 – Quasar Soft

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## Working with Quasar Soft & Cast

**Quasar Soft** is a powerful remote control tool that is capable of representing the entire surface on a browser page, providing remote access to every function of the desk. It builds up on the previous HTML5 UI, introducing simultaneous control of all channels and adding tremendous power and flexibility.

**Quasar Cast** is a remote monitoring tool that is capable of streaming any LW channel generated in the studio from the console to any browser, providing remote listening directly to the control computer workstation.

Together, they form the most powerful solution for driving your studio from a remote location.

The setup of Quasar Soft and Quasar Cast is fully described in Configuration Chapter 1 of this manual: [Quasar Soft](#) and [Quasar Cast](#). For more details about installing a Quasar Soft License upgrade, please refer to Configuration chapter 1 of this Manual: [License Management](#).

### Main Features

At the time of writing this Manual, **Quasar Soft offers:**

- Control from any browser-enabled device, without the need to install a dedicated application
- Control of every input channel, simplified monitoring section, and full master section of the console.
- Up to a total of eight instances can be started from one or multiple devices accessing the same console
- Each page can be resized and arranged freely on the display thanks to the built-in scaling engine
- Possibility to freely add, remove, and arrange its control widgets within channel strips
- Possibility to switch between fader banks similar to Layers, by using the top Tabs.
- The Call Controller, Intercom Controller, and Hot Keys Tab functionality
- Remote control of the programmable User Keys found on the Control Surface
- The ON AIR, TALK and LISTEN indicators functionality

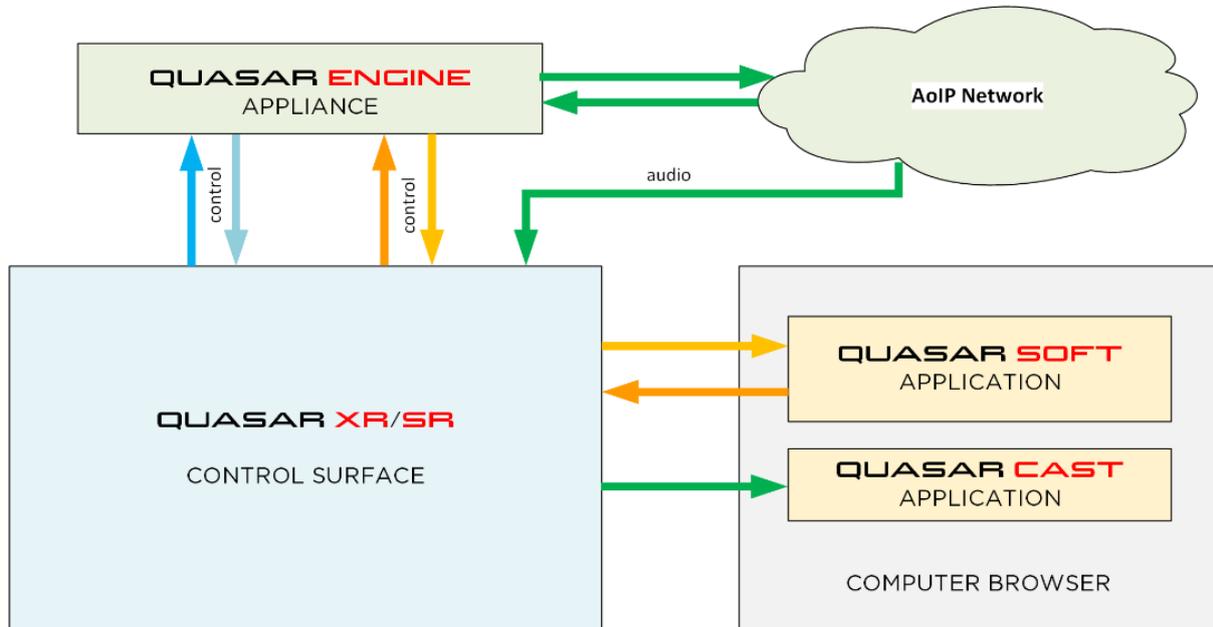
At the time of writing this Manual, **Quasar Soft does not offer:**

- Control of external device such as xNodes
- Remote control of the SmartKey Modules
- Dynamic text push to Channel labels via LWCP

We are continuously working to improve Quasar Soft and add new features. In the meantime, any additional external control functionality can be compensated by opening separate browser windows with the corresponding Web GUIs of the external devices you may want to control, (xNodes, Telos VX, Pathfinder and Infinity panels) and arrange them next to the Quasar Soft window.

## General Operating Principles

The soft controls in the Quasar Soft UI do not mirror, or copy the hardware controls of the Quasar Surface. They act independently, leaving the local (Desk) operator freedom to access other controls like, for example, the EQ tab, while the remote operator (Browser) is accessing the AUTOMIX tab.



The operation of Quasar Soft and Quasar Cast completely depends on the browser, and on the connection it establishes with the web server located inside the surface. Therefore:

- If any changes are made to the console Configuration Menu in the web UI, the new settings will be applied only after Quasar Soft's Browser page is reloaded.
- If Quasar Soft is used on a portable device, such as a Tablet, and the device enters sleep mode, on wake up the browser page will have to be reloaded.
- Up to 8 instances of Quasar Soft can be started, from a given console.
- Up to 8 instances of Quasar Cast can be started, from a given console.

**Note** - Running multiple instances of Quasar Cast at high bitrates will impact the Console's CPU load.

## Quasar Soft Applications

### Remote Control

Quasar Soft can be set up to control a console from remote, in case this is left unattended or to assist an operator which is unfamiliar with the console.

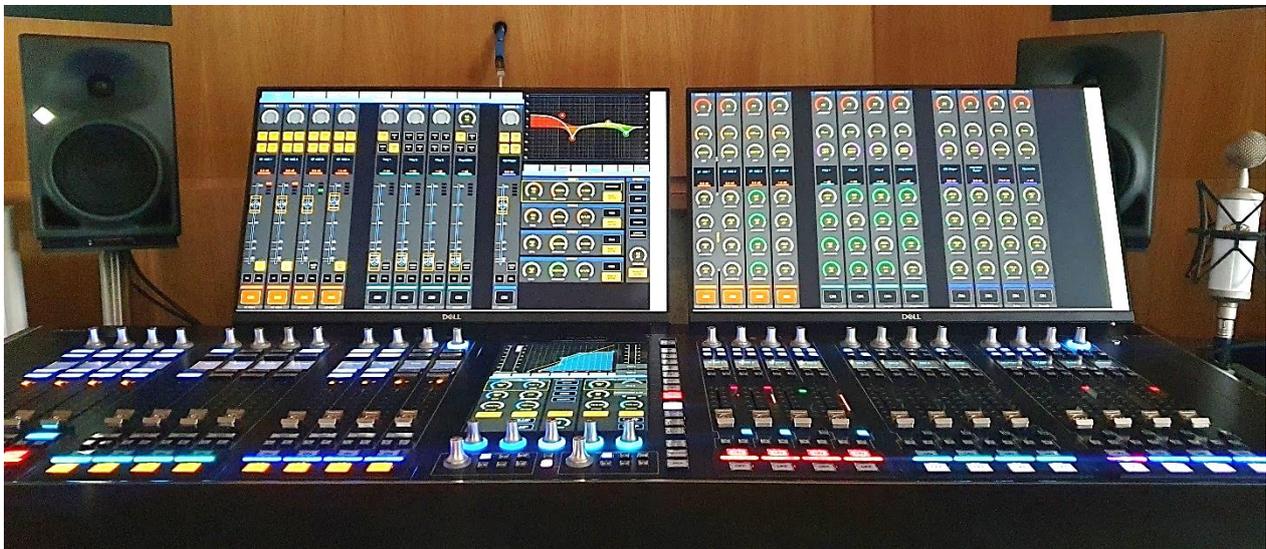
### Multi-Operator workflows

With Quasar Soft, it is possible to use a Touchscreen PC, or a Tablet, to let a second operator or a Producer, *interact* with the main operator in case of particularly complex workflows.

## Extending a Quasar Surface with Quasar Soft

Quasar Soft can be used to *extend* the physical controls of your surface by installing touchscreens positioned directly above the console. Those channel controls that are normally only accessible through the Master Touchscreen Monitor (MTS) module, such as: Aux Sends, Pan, Automix Weight, Channel Delay, can be laid out above the surface for direct access. And soon more controls will be added.

In case a standard 24" Full HD display (1920x1080 pixel resolution) is used, the soft channel strips will have the same width of the physical fader strips on the console. Furthermore, it is possible to add blank strips every four channel, in order to match the physical alignment of the console fader strips and modules.



## Quasar Soft Configuration & Startup

For details about how to configure and start a Quasar Soft instance, please refer to the Web UI Reference found in Configuration Chapter 1 of this Manual: [Quasar Soft](#)

# Quasar Soft Operation

When Quasar Soft is launched, all the controls accessible from the surface can be replicated on the browser's window, allowing the System Administrator, or the Operator, to access and operate the console entirely from remote.

Typically, a Quasar Soft Screen can be divided into two sections:

1. The Channel Strip Section
2. The Master and Monitor Section



**Note** - This view can be arranged from the Web UI to display the master section to the Left, or to the Right.

## The Channel Strip Section

### Channel Tabs

Channel Tabs let you page between blocks of fader strips. Faders numbers always correspond to input channels of the DSP Engine. Tabs are automatically generated if enabled from the Web UI.

### Channel Selection

In order to select a channel, or fader strip in Quasar Soft, you must click on the Channel Display. Since Quasar Soft is supposed to be controlled from touchscreens, the Channel display offers a large enough touch area for this purpose, eliminating the need for a dedicated button.

In case the Large Monitoring section is displayed, selecting a fader strip will toggle the monitor section to the channel menu, reproducing the behavior of the physical surface when a channel encoder is pressed.

All controls follow the same logic and operation rules of the hardware controls in the Console: All logic is set in the Source and Show Profiles.

### Widgets

Each graphical object in Quasar Soft, is called a widget. The Quasar Soft Configuration page offers a default arrangement for the Channel Strip Widgets.

The Widgets' type and position within all Channel Strips can be chosen at the time of starting a new Quasar Soft instance: simply select a different widget from each of the drop-downs. The final widget position, from top to bottom of the channel strip, will follow the selection of the drop-downs.

### Rotary Gauges

In order to control the rotary Gauges from the controlling device touchscreen (or mouse), you will need to click inside the Gauge area and then drag your finger (or cursor) horizontally.



These gauges work exactly like those found in the Quasar Touchscreen GUI.



### Faders

The Fader widget includes the Bar Meters, the channel status indicators (DYN, EQ, AUX, GROUP), and the AUTOMIX button. These are all part of a single block. You must touch or click the yellow fader cap, in order to slide the fader up and down. Touching outside the cap area will not make the fader move in any direction.

### Sliders

The Slider widget includes the Bar Meters, the channel status indicators (DYN, EQ, AUX, GROUP), and the AUTOMIX button. These are all part of a single block. You can touch or click the slider anywhere on the entire slot (moving bar + background slot), and drag it in order to slide the fader up and down. Touching outside the slot area will not make the slider move in any direction.



## The Master/Monitor Section

Starting from SW v3.2 , the Master section in Quasar Soft support the new Soft USER 1-8 Keys (User Buttons located on the MTS module), as well as the new Soft SmartKey modules.

### Full Master Section

This is a full representation of the Master Section found in the Quasar Surface. The Widgets copy its physical controls, and the Touchscreen area offers the same functionality, of the MTS Touchscreen module.

## Compact Monitor Section

This is a streamlined version of the Master Section found in the Quasar Surface, with only the essential monitor controls, and no TS GUI. No control of the input channels is possible, when the Small Monitor Section is selected.



## Storing your Favorite Quasar Soft Layouts

Once you have created some customized Layout views in Quasar Soft, you can simply store them in your Browser's Favorites, as you would normally do for a website link.

In fact, all Quasar Soft layouts information, is stored in the HTTP link which is created once the [Start Quasar Soft Instance](#) button is pressed.

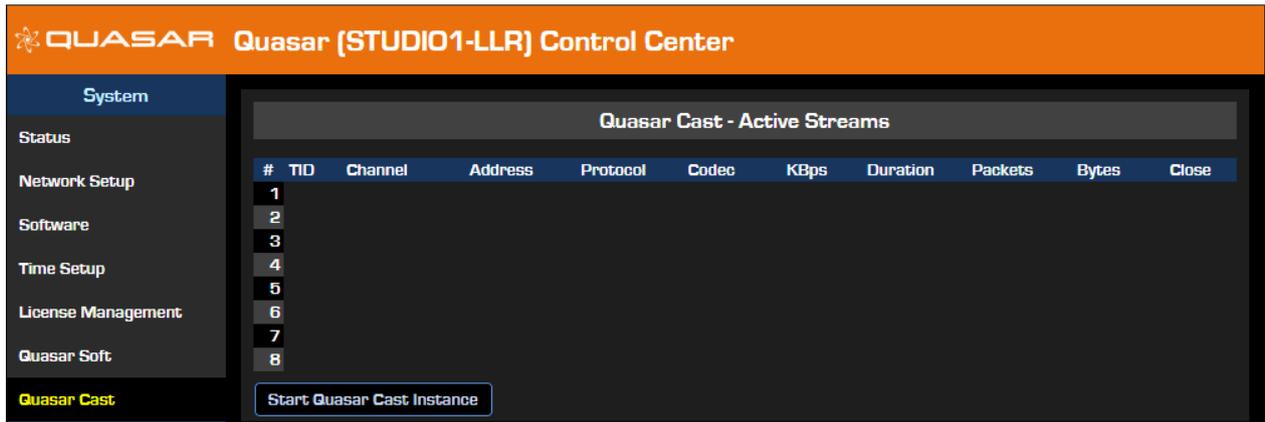
**Note** - If you would like to open the same customized Layouts from a different computer, or device, such as a Tablet, you will need to export those favorites and import them into the other device. Or simply save them as text files into a network location accessible from all your devices.

## Quasar Cast Operation

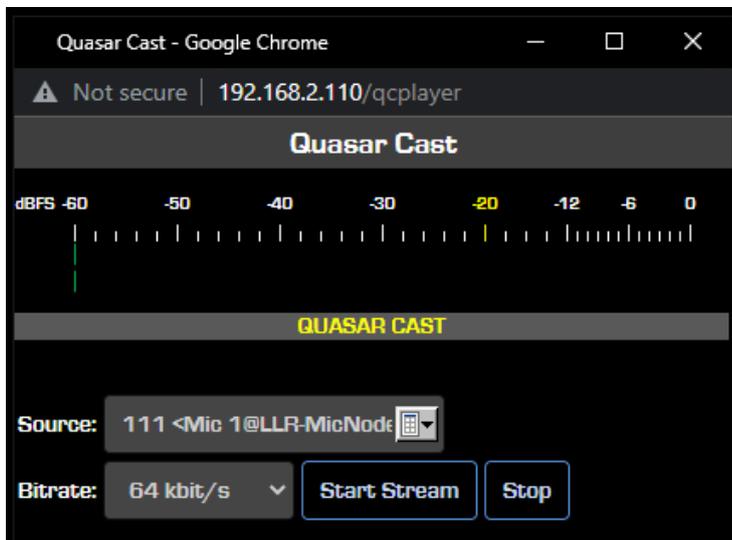
Quasar Cast is a streaming encoder based on the license-free OPUS encoding algorithm. Quasar Cast shares the same Telos license with Quasar Soft. You need a Quasar Soft License in order to use Quasar Cast.

Quasar Cast operation is as easy as Starting an instance, choosing an AoIP stream from the network, and selecting the desired Bitrate. In order to do so:

Navigate to the **Quasar Cast** web UI page



Click the **Start Quasar Cast Instance** button. A popup window will appear.



In the **Source** field, click on the drop-down and select a network Stream from the popup menu. The stream will be added into the Source field.



In the Bitrate field, select the desired bitrate for encoding your stream. Possible Bitrates range from 32 to 320 kbit/s

Click the **Start Stream** button. If the selected stream is active, but no audio is present, the first segments of the audio meter bar will be displayed (see picture above).

You can create up to 8 instances of Quasar Cast. These will appear in the Active Streams list after the page is refreshed. When a stream is stopped, it's not possible to restart it from within the same instance it was created. You will need to close the instance and start a new one.

## PC Configuration

### Display Settings

#### Multi-Touch Support

Quasar Soft supports multi-touch displays. Its fader strips can be operated simultaneously exactly like the physical channel strips of the console. The performance (how many touch points, sensitivity and response latency) depend solely on the computer hardware.

#### Optimal Display Size

Normally, a perfect 1:1 copy of the console's 12" touchscreen can be displayed on a 24" touchscreen monitor, by setting the browser's zoom at 75% magnification and Full Screen view. In this case, a standard Full HD monitor will be able to fully display:

- Up to 16 Channel Strips with no Monitor section
- Up to 14 Channel Strips + Small Monitor section
- Up to 11 Channel Strips + Full Master section

#### Picture Scaling

New in SW v3.2 is the possibility to scale a Quasar Soft page by simply resizing the browser window. You can now scale the image to fit the browser's window maintaining QSoft's graphics original aspect ration (Best Fit), scale to Fit height only or width only, or stretch to fill the browser window without maintaining the aspect ratio. This is configurable from the Web UI.

You no longer need to use the browser bult-in zoom feature, but if for some reason you need to do so, please keep in mind that in Chrome, this zoom will apply to all browser tabs that point to the same domain (the console's IP address), including other console Web UI pages you may have open. To work around this problem, you can launch a different browser application (like MS Edge, for example) and dedicate this to Quasar Soft, while you operate the console UI from Chrome.

### PC Settings

#### Energy Saving

Energy Saving Setting in some computers may interrupt the connection when the browser tab is in the background. This depends on both the browser and the host computer settings. In this case you will need to bring the browser tab back to the foreground and refresh the page.

#### Anti-Virus

Anti Virus programs can create noticeable delays to the execution of Quasar Soft, when it is started. For controlling a Quasar console within the same facility, we suggest running Quasar Soft from a computer which is sitting on the same network where the console connects to, without connection to the internet and no Antivirus installed.

For controlling Quasar over a WAN, trough the public internet, we suggest installing a firewall with properly configured DMZ and NAT between the Internet router and the AoIP network.

### **Some notes about Internet Browsers**

Quasar's Remote GUI was fully tested with Firefox, Chrome and MS Edge, and it was only lightly tested with Safari. Although we support all these browsers, we recommend using Chrome, MS Edge (now fully equivalent to Chrome), or Firefox.

### **Browser-to-Console Connection Link**

Note that in case the Ethernet Link between the Quasar Soft PC and the Quasar Console has not enough bandwidth, the performance of the physical control surface might be affected and its responsiveness reduced. For example, the console might perform certain operations (like loading a Show Profile) slowly, because it will take longer for the MTS to update Quasar Soft pushing its state to the browser over a slow ethernet link.

The recommended link bandwidth is 2Mbps, with a minimum of 1Mbps.

# Operation 6 – Engine RPS

## The Quasar Engine RPS

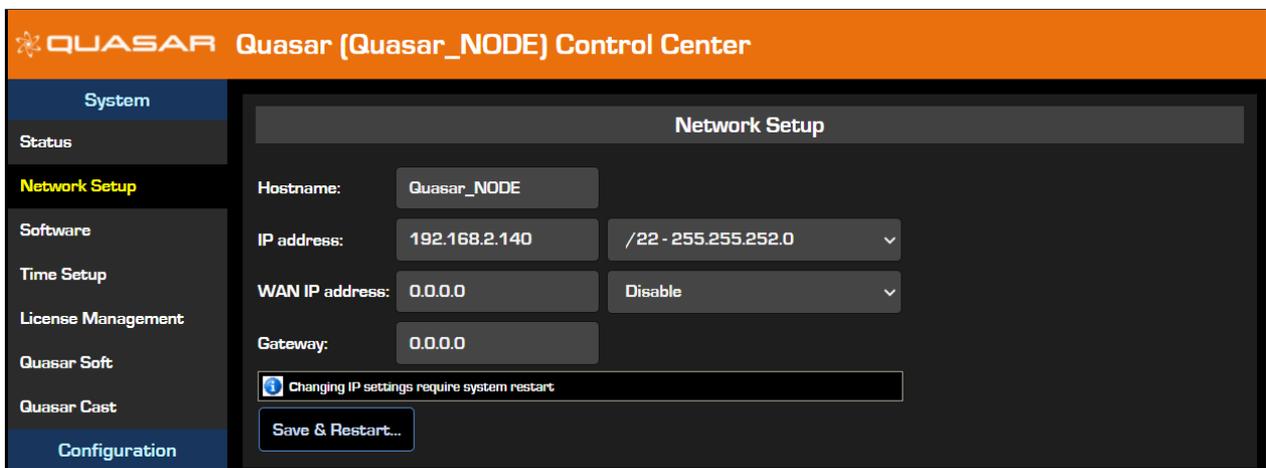
Quasar Engine RPS is an “augmented version” of a Quasar Engine, where all the functionalities of the MTS module normally found in every Quasar console, are added directly into the engine hardware platform thanks to an additional CPU board, similar to that found inside the Quasar MTS-MON Module.

This makes it possible to operate a mixing engine without a physical console attached to it.

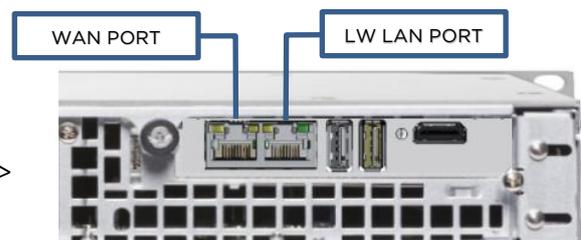


## Engine RPS Web UI

The Engine RPS Web UI is the same as the standard MTS Web UI, with only one exception: The Network Setup menu has an additional Network port labeled “WAN” , that can be configured to connect to a different network.



**Note** - The WAN port is disabled by default.



*The RPS Board installed in the Quasar Engine RPS >*

Since this an Operation Manual, we have included all information about how to connect the Engine RPS network ports into the Quasar Quick Start Guide, and we added information on how to configure these ports in the Quasar Installation Guide. Please ask your Dealer for this document if you're planning a Quasar Console, or Quasar Engine RPS installation.

The Engine RPS built-in Surface web UI is the same as that found in a normal Quasar Console: for a complete description of the RPS Web UI, please refer to Configuration Chapter 1 of this manual - [MTS-MON Module Web UI](#).

The Engine RPS built-in Engine web UI is the same as that found in a normal Quasar Engine: for a complete description of the RPS Engine Web UI, please refer to Configuration Chapter 9 of this manual - [The Quasar Engine Web UI](#).

## Engine RPS Operation

Quasar Engine RPS can be operated in the following ways:

1. With a Touchscreen directly connected to the unit
2. With a browser, by launching the Quasar Soft HTML5 application from the RPS Web UI

### Direct Touchscreen Operation

In this case a standard PC Touchscreen monitor is connected via HDMI and USB to the RPS board at the back of the Engine. At the time of writing this chapter, the monitor will display the same GUI available from a Quasar MTS module, but rearranged in landscape format. All control functions available on a Quasar MTS module will be available, and it will be possible to access and control the Input Channels only one at a time.

### Browser-based Operation

In this case Quasar Engine RPS is operated via Quasar Soft like a normal Quasar console. All the Quasar Soft instructions found in this manual equally apply to the Quasar Soft found in a Quasar Engine RPS.

For more details about using Quasar Soft, please refer to Operation Chapter 5 of this Manual - [Quasar Soft](#).

## Engine RPS Applications

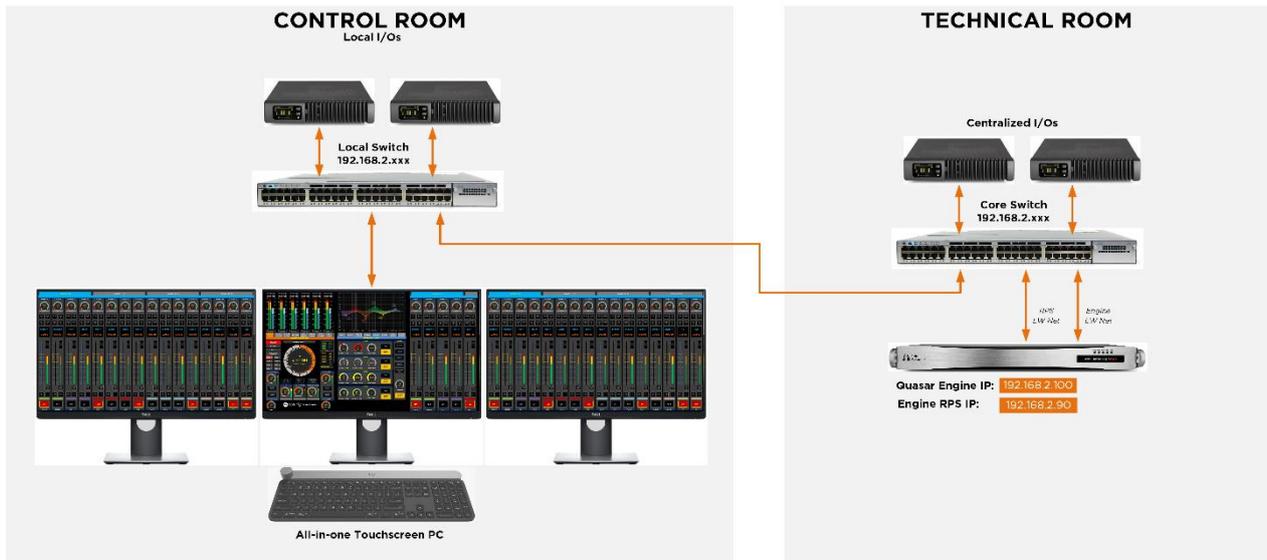
Quasar Engine RPS is ideal for situations where a physical console is either unnecessary or impractical, such as small studios or backup facilities. Each Quasar Engine RPS includes Quasar Soft (for mixing), Quasar Cast (for low-latency monitoring), and like the standard Quasar Engine, starts with 16 input channels. Additional channel expansion licenses are optionally available.

The Engine RPS can also be used in situations where a physical Quasar XR or SR console is already installed, providing a convenient backup solution to a main hardware console or acting as a "proxy console" for Remote Production workflows. In this case the user can choose whether to control the Engine mixing functions from the hardware Control Surface, from the Quasar Soft built into that surface, or from the Quasar Soft built into Quasar Engine RPS.

Let's have a more detailed overview of the most common usage scenarios:

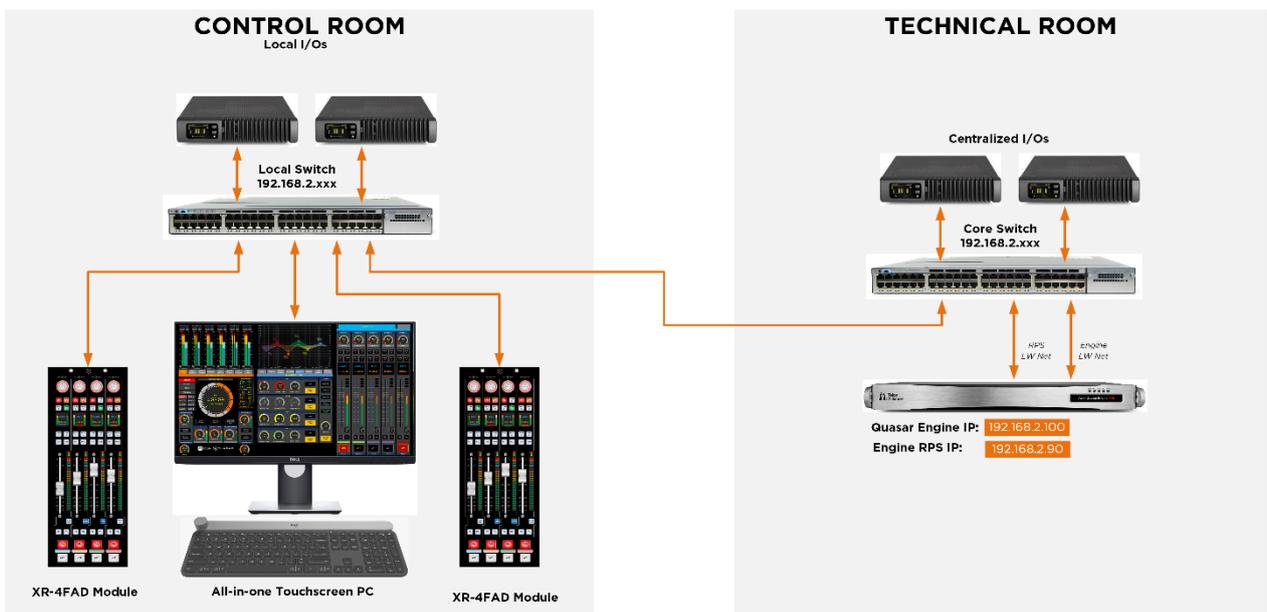
## Using Engine RPS as a full soft console

When a full-blown hardware control surface is not required, or when a Touchscreen-based surface is preferred, the Quasar Engine RPS is the ideal solution: the built-in Quasar Web UI with Quasar Soft can be used to exploit the full power of the Quasar Mixing Engine. In such case, the Quasar Engine RPS becomes an all-in-one solution for “glass mixing” production. Just add a PC with a web browser and one or multiple Touchscreen monitors.



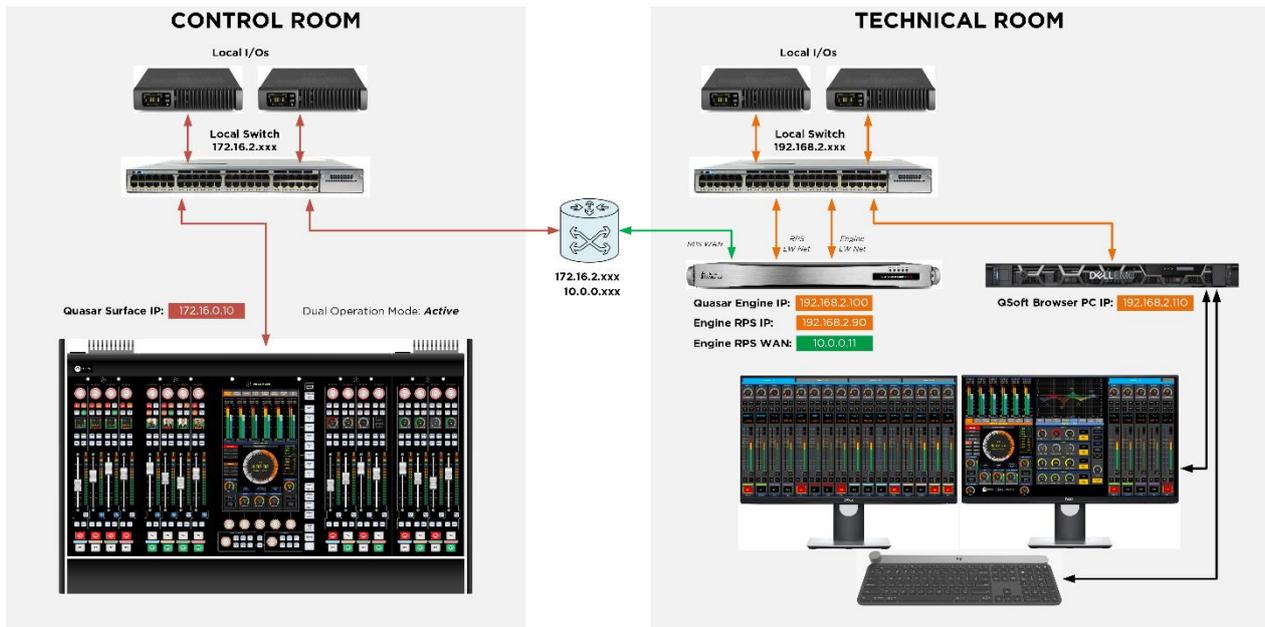
## Using Engine RPS as a hybrid console

The RPS board in the Quasar Engine RPS can be externally connected to Quasar XR-4FAD fader modules via Ethernet, or to Quasar SR-4FAD fader modules via USB, allowing the creation of a hybrid control surface based on a soft Master module and hardware Faders:



## Using Engine RPS as a Remote Production System

The RPS board in the Quasar Engine RPS can be used as a “proxy console”, providing a remote production solution in cases where the main hardware control surface, is located far from the engine and its I/Os, and being connected on a different network it could not connect directly to its engine.



In this case, the Engine RPS WAN interface will allow connection to the router and will pass network traffic internally, controlling the Engine which is sitting on the local LAN.

The Dual Operator Mode can be used to address this particular use case, and make the physical (main) console control the soft (remote proxy) control surface, allowing for Remote Production over WANs.

For more details about how to configure Dual Operation mode please refer to [Configuration Chapter 8](#) of this manual.

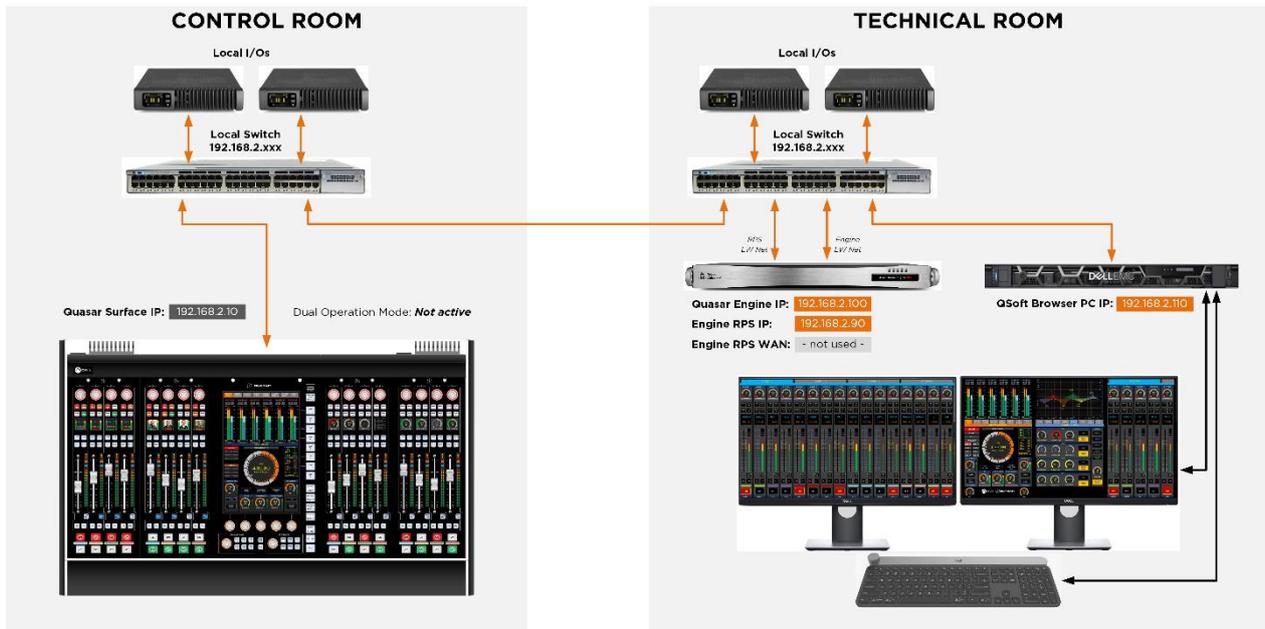
## Using Engine RPS as an “Air Chain Processor”

Some large installations require centralized content manipulation (like audio insertion, processing, routing and/or mixing) to be added to the studio Program feeds, before these reach the On Air processor. This is especially the case of Broadcast Syndications for example, where licensed content is inserted onto the local programs and audio feeds may require dedicated processing and time-based routing.

An RPS Engine can be installed in the Technical Operation Center and its input channels (up to 64) could take Program feeds from all rooms and thanks to its onboard channel DSP processing, VMIXes and VMODEs, these feeds could be manipulated and routed by Pathfinder with custom Logic Flows, and could be monitored using a custom dashboard.

## Using Engine RPS as a soft backup console

The Quasar Engine RPS can be used as mixing engine for a hardware Quasar surface. The RPS board included with it can be used as a “soft surface”, providing a backup solution for those cases where the main hardware control surface needs to be turned off for programmed maintenance or other reasons, while control of the mixing engine needs to be maintained.



In this case, both the Quasar surface, and the Engine RPS can all be part of the same network. The Alternative Console Operation Mode, was specifically designed to address this particular use case, and make the changeover between the physical (main) and soft (backup) control surfaces automatic.

For more details about how to configure automatic backup switching please refer to [Configuration Chapter 8](#) of this manual.

**Note** - At the time of writing this manual the CALL CONTROL, INTERCOM CONTROL, and HOT KEYS tabs are not active in Quasar Soft, therefore control of the Telos VX, Infinity Intercom, and of External HotKey machines is not possible yet.

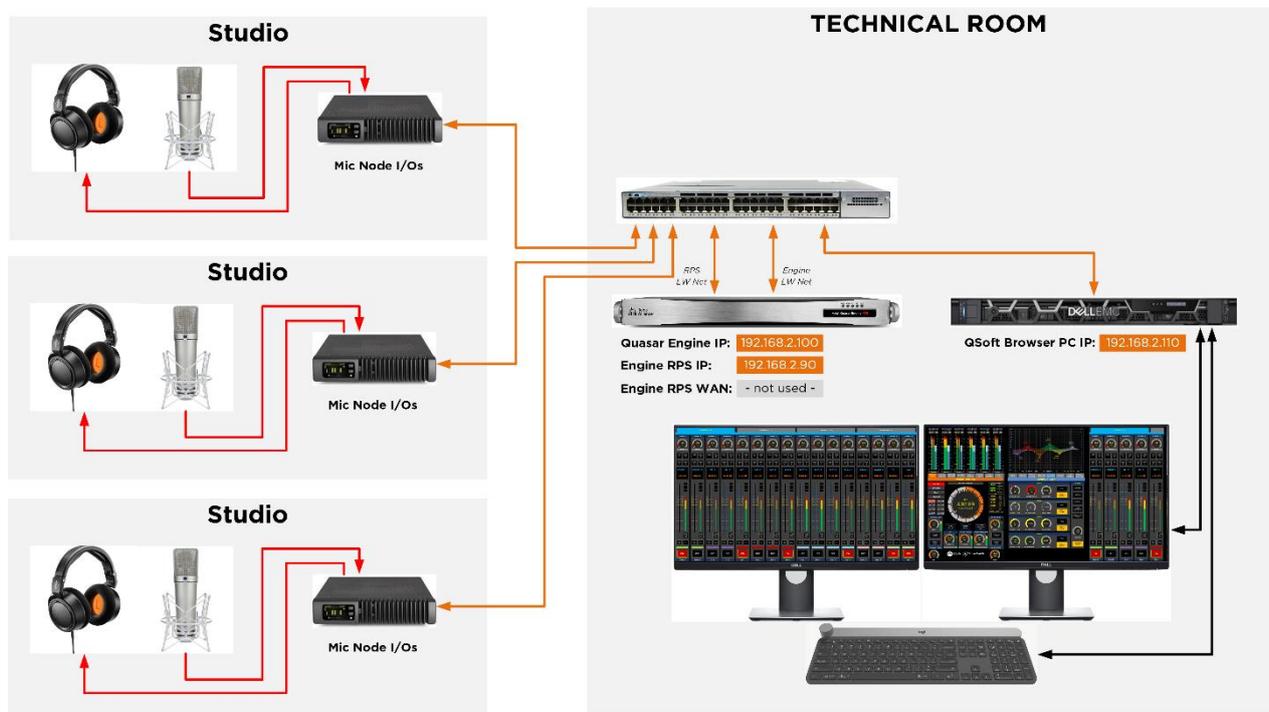
Please refer to the [Quasar Console Installation Guide](#) for more details about how to connect and wire Quasar Engine RPS in the above mentioned use cases.

## Using Engine RPS as a 64-Channel Microphone processor

The Quasar Engine RPS can be used as centralized Microphone processor in large installations where not every studio is equipped with a Quasar console, yet high-quality microphone processing is required for feeding newsrooms, edit rooms, and production areas.

This can be a very convenient solution for multi-room environments where inexpensive AoIP consoles - or even no consoles - are present.

The Quasar Engine RPS can provide the renowned Quasar processing to up to 64 Microphone sources: just feed each of the available channel inputs with a microphone and use the corresponding Channel Direct Out to feed the desired studio. The Direct Out will have to be set to Post-Fader position, in order to apply full channel processing including Filters, Dynamics, EQ, De-Esser and Limiter.



The RPS Engine's Program and Aux Busses could be used for creating Mike groups, while its Monitoring section could be used for listening off the air to each microphone feed.

Adding Pathfinder Pro control turns the Engine RPS into a highly customizable and automated solution for Centralized Microphone processing.

The next part of this User Manual focuses on the console Configuration: it provides a broader view of the console workflow and allows for a deeper understanding of the entire system, its user interface and control logic.

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# Configuration 1 – Web UI Reference

## MTS-MON Module Web UI

What we refer to as “Quasar Web UI” or “Console Web UI” in this manual, is the MTS-MON (Master Monitor) Module Web UI. This module is the “brain” of the system; therefore, all the most important Studio configuration is stored within this module, and access to its UI is required for the initial setup as well as for all the daily configuration management activities, including configuration backup.

There is no need for the board operator to access the console Web UI during normal console operation. This is designed to let the Station engineer or system Administrator, entirely configure the console, without the need for a specialized Telos engineer. That’s the beauty of our system!

Connect a PC to the console network, and open up a browser. Enter the MTS-MON Module’s IP address. You will be presented with the console’s **System Status** Page:

**QUASAR Quasar (STUDIO1-LLR) Control Center**

**System**

**Status**

**Version Information**

Version: 3.1.0 (19 Dec 2024)  
Base: 1.0.0

**System Information**

Kernel: Linux 4.9.102-g62c02ddb armv7l  
Uptime: 0 days 03:17  
CPU usage: 3.8%  
CPU temp: +44.5 °C  
SYS temp: +40.2 °C  
RTC battery: OK  
PDB Uptime: 0 days 03:18  
PDB Main Voltage: 12.0V  
PDB Input Voltages: 12.0V 0.0V 0.0V 12.0V  
PDB Output Voltages: 12.2V 12.2V 12.2V 12.2V 12.2V 12.2V 12.2V 12.2V  
Network: 1000 Mbps Full-duplex  
Net usage: Rx: 0.373 Mbps, Tx: 0.174 Mbps  
MAC Address: 58:50:AB:40:32:17

**File System Information**

Filesystem	Size	Used	Available	Use%
Memory	1.42 GB	179.40 MB	1.24 GB	12.4%
/	722.45 MB	10.95 MB	711.50 MB	1.5%
/mnt/config	975.90 MB	8.48 MB	967.42 MB	0.9%

**Software Components**

quasar: 117ab4625882eb91fedeb2846fd6b0c1838b48fc  
acl: be83ff8fb50a4f19183391602cb477846b053516

[Quasar User Manual](#)

**Configuration**

XR / ACC Modules

SR Modules

SmartKeys

Engine

Customize

Hot Keys

Intercom Control

Brightness Control

Backup / Restore

**Profiles**

Presets

Sources

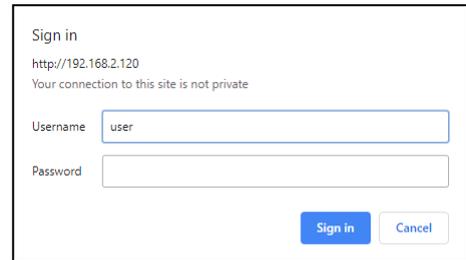
Shows

**Diagnostics**

Log

The first time the Web UI is accessed from a browser, a login prompt will appear, where a user password, if configured, needs to be entered. If no password has been configured, you can enter the default credentials: *user* and *-no password-*.

**Note** - The XR-4FAD (Fader) Modules also have a Web UI. These modules come normally pre-configured from the factory. However, users might need to tweak some parameters like Hardware Controls, Brightness or Layers, after they gain some practice with the console.



Sign in  
http://192.168.2.120  
Your connection to this site is not private

Username:

Password:

The left column menu is divided into four menu sections: **System, Configuration, Profiles and Diagnostics.**

## System Menu

Access to the System Menu is mostly required for initial setup of the system:

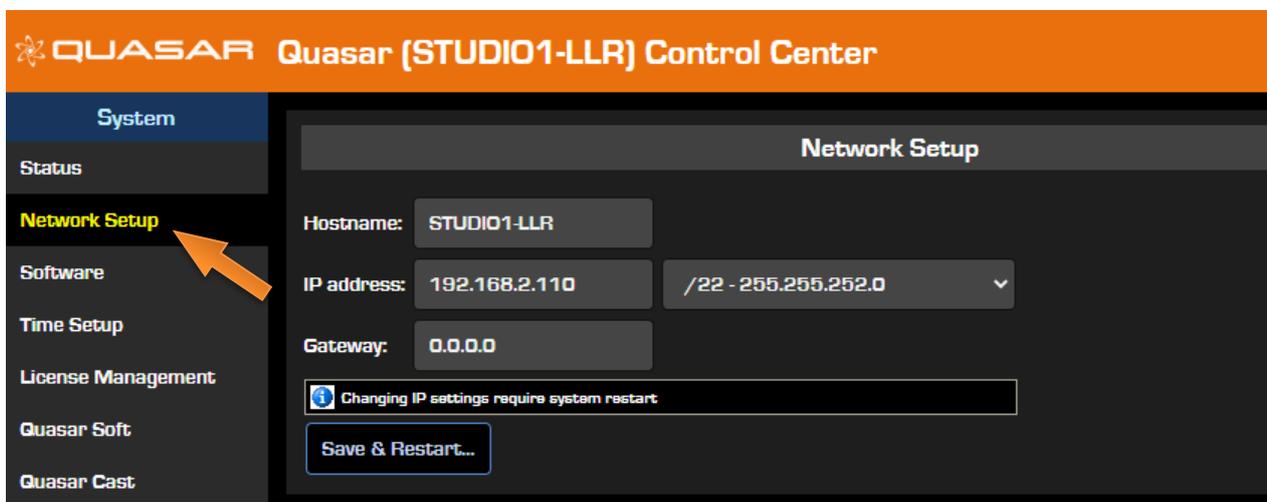
### Status

This landing page displays all the most important information about the installed Software Version, the System’s internal hardware components, and its memory usage. It is also possible to view and download this User Manual by pushing the [Quasar User Manual](#) button.

### Network Setup

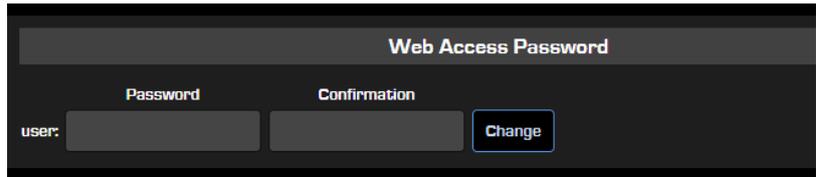
From this page you can set up or change your Quasar network configuration. Here you can enter the Console Name (Hostname), the IP Address and Subnet information.

Entering a Gateway is not required, unless you need the console to access a public NTP Server. However, we recommend that you leave this field empty, and use an internal NTP server that sits on your network, for updating the Time on your Quasar.



The Default user Password can be changed in the lower part of this page:

The Username is always “user” and cannot be changed.



The Quasar Soft Access Whitelist section allows you to enter one IP address of a privileged computer which you want to allow access to Quasar Soft mixer GUI without being asked for the login credentials.

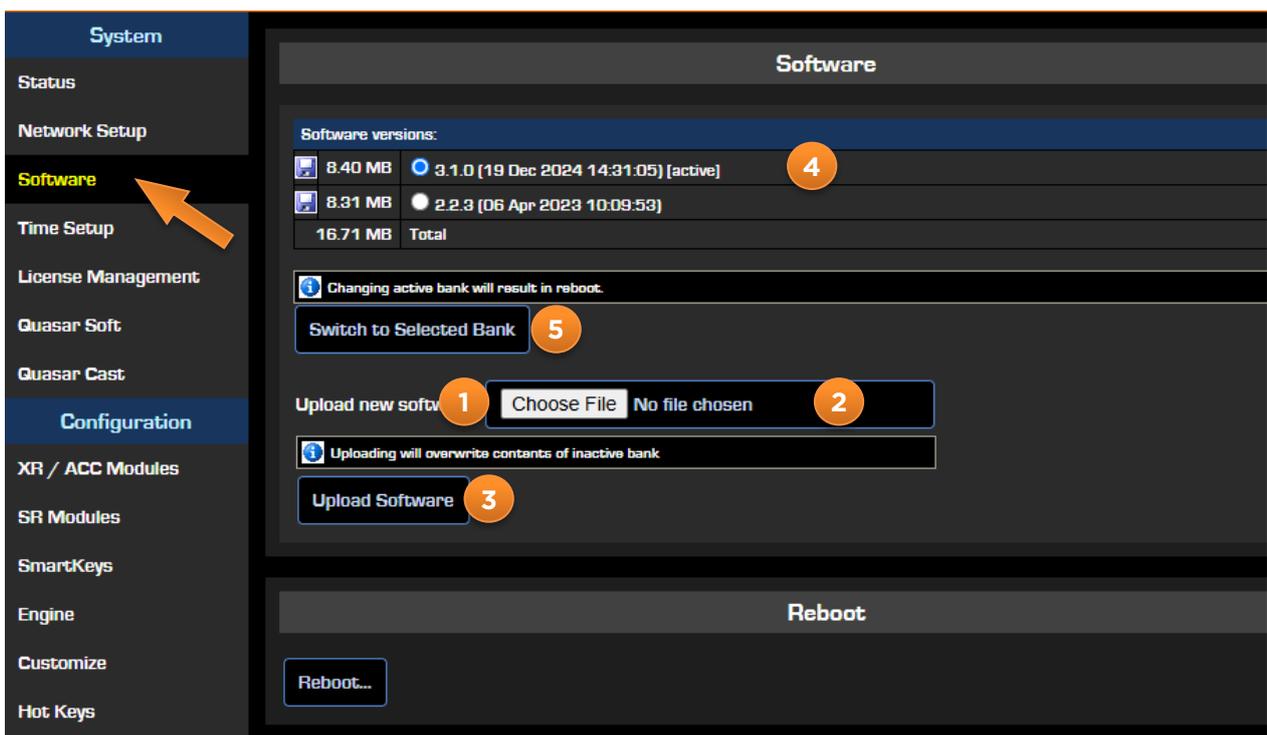


## Software

What we refer to as “Software” is the Top-Level application which runs on the main CPU and includes the embedded Linux Operating System, Web Server, Console Logic and GUI. This software is not what many engineers call “Firmware”, which instead is a Low-level configuration file used to determine the behavior of all the hardware controls such as buttons, faders, encoders and LEDs, normally found on a control surface module. When talking to our Support, please keep in mind that each surface module has a Software and a Firmware file.

From the **Software** page it is possible to update the Quasar System software. In order to install a new software package, you must perform the following steps:

1. Select a new “web update file” by clicking on the **Choose File** button,
2. Wait until the software package appears in the field next to the button.
3. Click on the **Upload Software** button, and wait for it to appear in the currently inactive software bank (the bank which is not selected).
4. Select the new software bank.
5. Click the **Switch to Selected Bank** button.

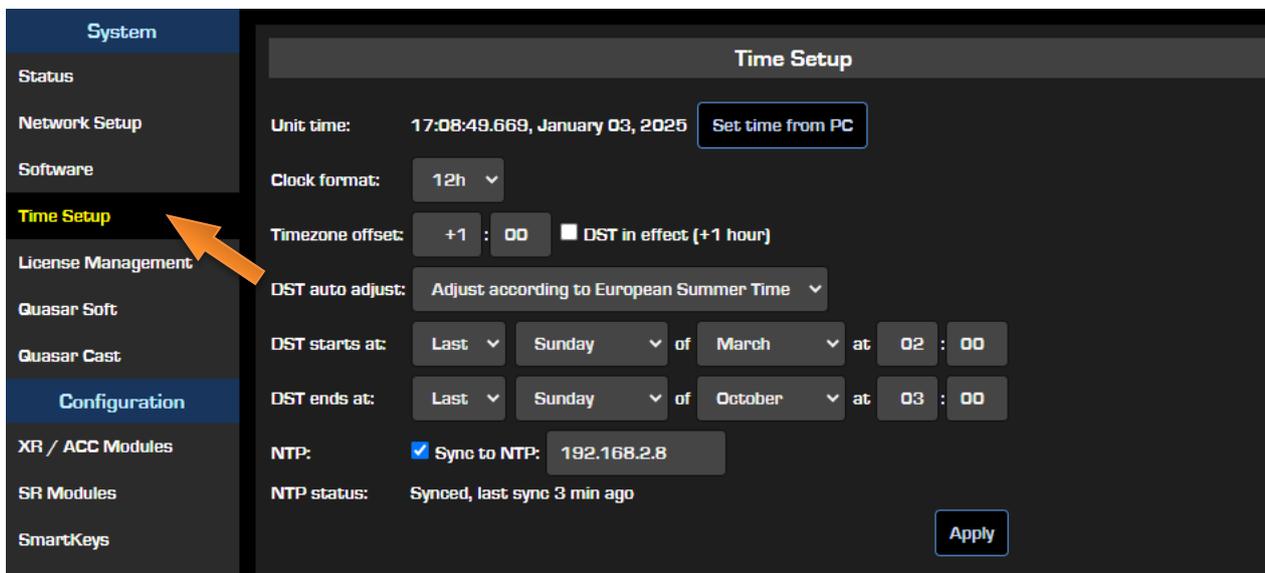


**Warning** - Quasar will automatically reboot after selecting a new bank, connections to all modules and Engine will be reset, and audio will be interrupted for a few secs.

It is also possible to manually **Reboot** the system, without the need to cycle power to the surface. To do so, click the **Reboot...** button. A confirmation dialog will appear.

## Time Setup

The Quasar MASTER HOME tab always displays a large circular clock in its center. The MTS-MON module requires time to be set for this clock, and for its logging activity. From this page you can set the system time from you PC, simply by clicking on the **Set Time from PC** button. The clock can be set to display 24-hour time (0:00 to 23:59) or 12-hour time (AM/ PM). Time zone offset and Day Light Saving (DST) options are also available. It is also possible to enter the IP address of an **NTP server**, to automatically update the clock. The update rate is 5 mins. Press the **Apply** button before you exit the page.



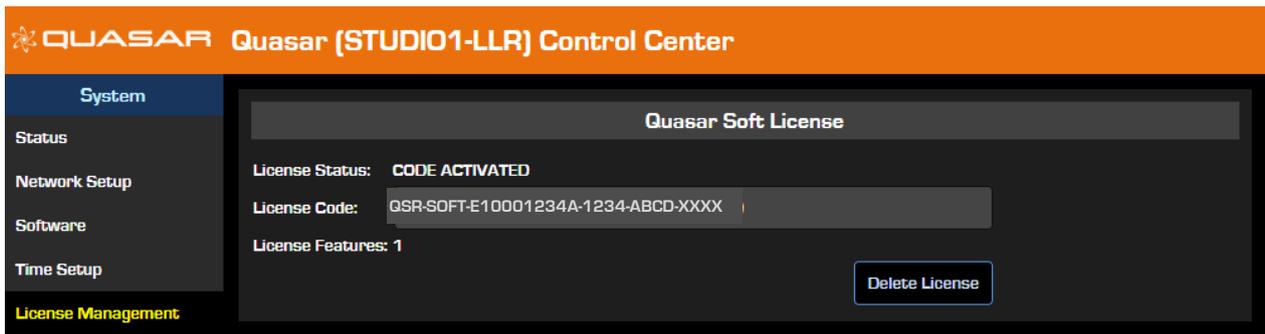
Whether it is set manually or via NTP, the MTS Module clock will be used for:

- the Console Touchscreen (Home) Clock,
- the MTS module's internal Logs,
- providing time info to XR Fader Modules when they connect to the MTS, in case they have NTP disabled
- providing time info to Accessory Modules when they connect to the MTS, in case they have NTP disabled

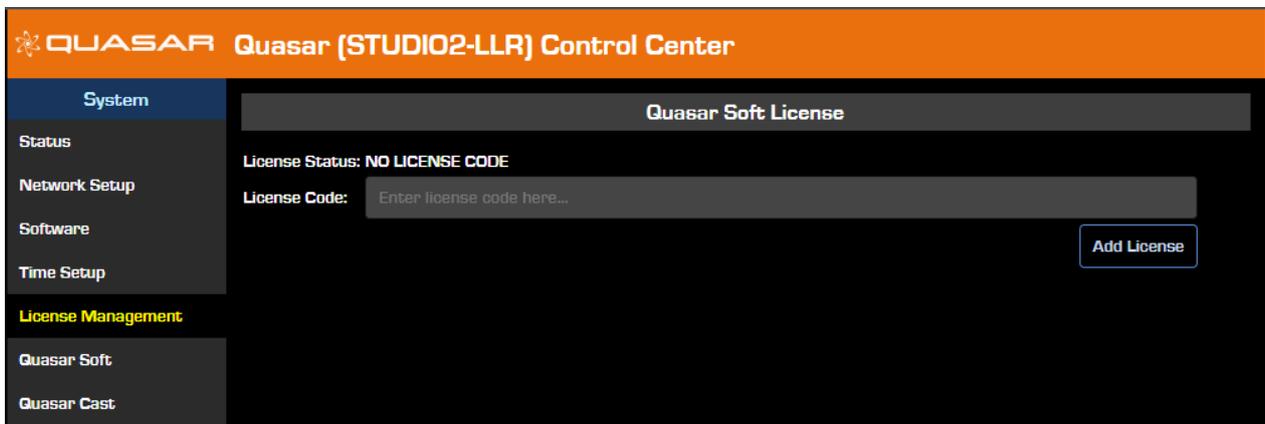
## License Management

This page is for entering the Quasar Soft/Cast License. You can also remove a License, for use on another console, and then reinstall it at later time.

If you see a screen like this:



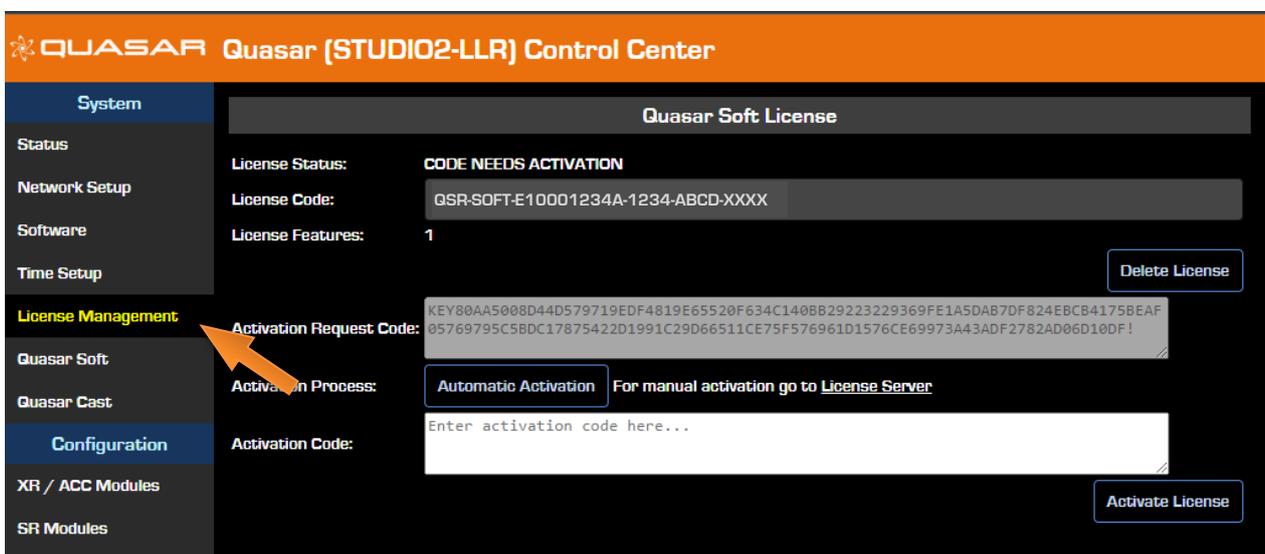
Then you have a valid Quasar Soft License already installed in your system, and you can skip this paragraph. If instead, you see a screen like this:



you do not have a license installed, and you may want to read the next pages in case you purchased a Quasar Soft license and you would like to install it.

### License Installation

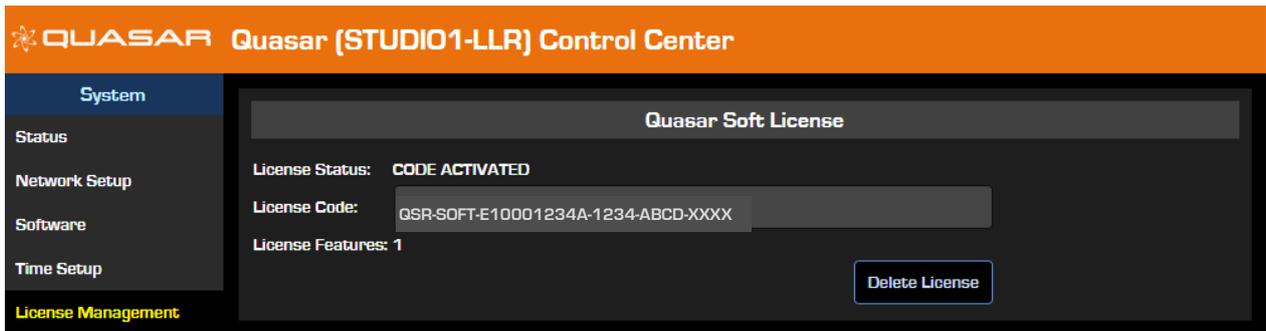
- Navigate to the **License Management** menu. Your PC browser must have an internet connection.
- Obtain a valid License from your preferred Dealer and paste it into the **License Code** field.
- Click the **Add License** button. The following screen will appear:



and an Activation Request Code will be generated.

- At this point, if your browser is connected to the Internet, you can click the **Automatic Activation** button, or the **Delete License** button to delete the license in case you pasted the wrong code.
- If your code is valid, you will receive a positive response and the License will be installed.

If you see this screen:



you successfully installed your license and you can skip the next points.

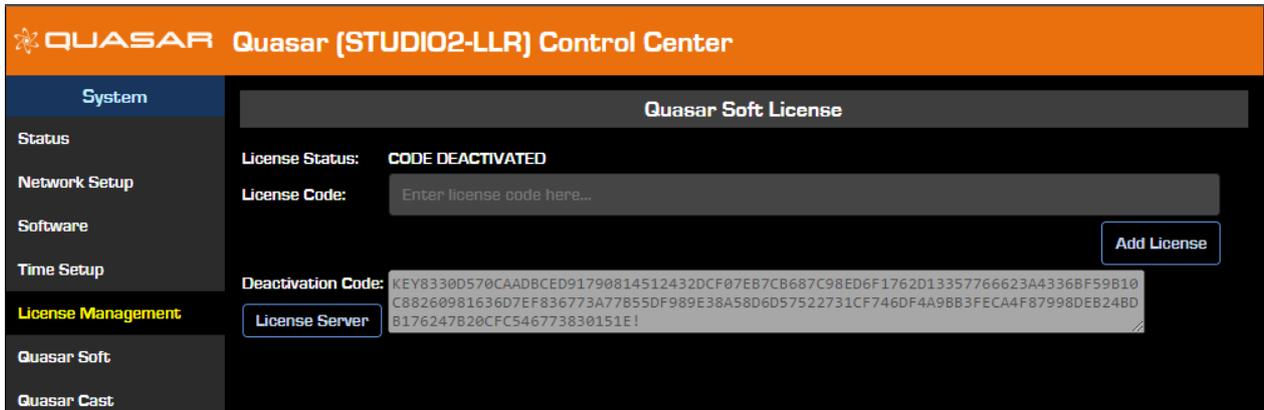
- In case the code is already used on another machine, you will get a “Failed” message as response from the server.
- If your computer has no Internet connection (you might be installing the license from a PC in your Studio, with no external connection) you can proceed to the manual activation, by copying the Request Code to another device which has an internet connection, (like your mobile phone) navigate from that device to the License Server, and paste your Request code there:



Then click the Submit button to obtain an Activation Code that you will need to paste back into the Quasar Soft License page.

## License Removal and Migration

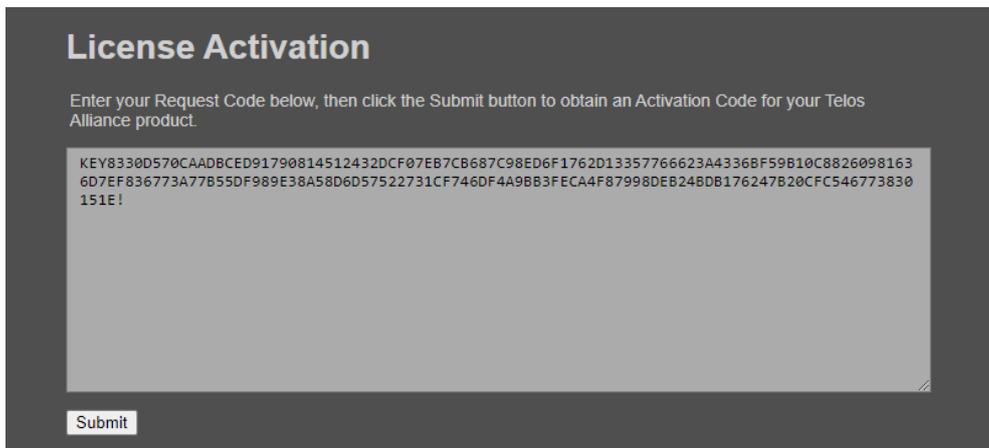
You can always remove an installed license, if you need to migrate it to a different console. In order to do so, click the **Delete License** button. The following screen will appear:



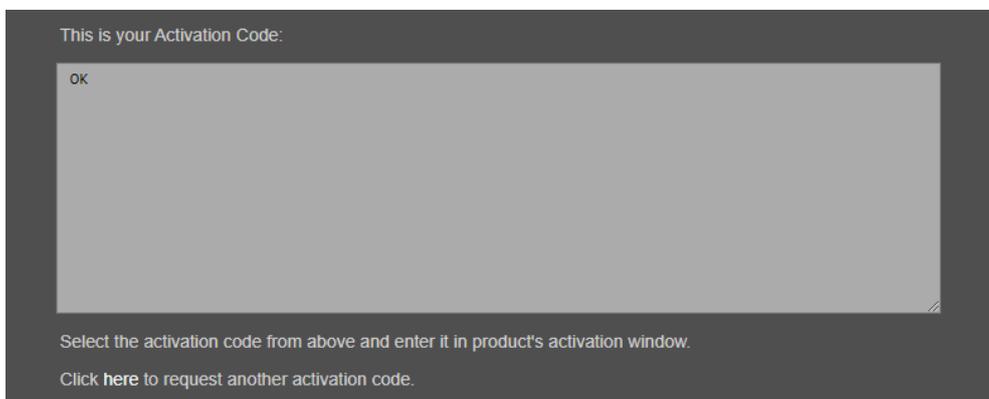
- Copy the Deactivation Code in the grey field and click the **License Server** button. This will open a new tab in your browser, taking you to the Telos License Server web page.

**Note** - Even if the page says “License Activation” , you are actually deactivating your license, and asking the Server to make it available for another request (from a different console).

- Paste the Code into the grey field



- Click the **Submit** button. You will get an “OK” confirmation response like this:



Now you can re-use your license elsewhere.

## Quasar Soft

Selecting this link will take you to the Quasar Soft configuration page. In order to see this screen, you must have a valid Quasar Soft License. Please see the previous paragraph if you have to install a Quasar Soft License.

The **Quasar Soft Surface** section is divided into three configuration areas: the Channel Section, the Monitoring Section, and the Channel Strip Layout.

- The Channel Section lets you determine which input channels will be accessed by Quasar Soft, and how they are arranged in the UI.
- The Monitoring Section lets you choose type and position of the Monitor Section
- The Channel Strip Layout lets you arrange widgets within all channel strips.

**System**

**Quasar (STUDIO1-LLR) Control Center**

**Quasar Soft Surface**

**Channel Section**

Number of Visible Faders: [0..64]

First Channel: [1..64]

Last Channel: [1..64]

Channel Gaps:

Channel Faders:

Channel Tabs:

Channel Select Linking:

Channel List [comma separated list, only if no tabs present]:

**Monitoring Section**

Type:

Position:

User:

SmartKeys:

Scaling:

**Channel Strip Layout**

Top:

Bottom:

**Channel User Button Function**

User 1:

User 2:

User 3:

User 4:

**Quasar Soft Accessory Module**

Fader Number:

### Channel Section

**Number of Visible Faders:** This field lets you determine how many Faders you want to be visible on the Quasar Soft GUI

**First Channel:** In case you want to display and access a whole range of channel strips, enter here the number of the first channel you want to access, and see in the leftmost part of the Channel Strip area. You can enter a value between 1 and 64.

**Last Channel:** In case you want to display and access a whole range of channel strips, enter here the number of the last channel you want to access, and see in the rightmost part of the Channel Strip area. You can enter a value between 1 and 64.

The **Channel Gaps** drop-down offers the following options:

- **No Gaps** will lay out all the channel strips next to each other, for maximum viewing density
- **Gap Each 4 Channel** will add a blank space between blocks of 4 channel strips, reproducing the layout on the physical surface, where faders are grouped in blocks of four.

The **Channel Fader Type** drop-down, (added in SW v3.2) lets you select between:

- Standard **Faders**, adjustable by touching the yellow fader cap
- The Altus-style **Sliders**, adjustable by touching anywhere in the slider's vertical strip.

The **Channel Tabs** drop-down offers the following options:

- **No Tabs** will hide the Input Channel Tabs on top of the Channel Strips. This is useful to prevent unwanted access to all channels of the console.
- **Channel Tabs Shown** will show the Input Channel Tabs on top of the Channel Strips.

**Note** - The number of tabs is not user-definable: it is automatically calculated on the basis of the Number of Visible Faders per each Layout, and the total number of Channels.

**Channel Select Linking:** This new option, introduced with SW v3.0 allows you to link the Channel Select function across multiple Quasar Soft windows.

- **No Link** will behave like SW v2.0: channel selection (by clicking or touching the channel strip display) is disabled in a Quasar Soft screen where no Master is displayed
- **MTS** will link the channel selection function to the physical MTS-MON module of the hardware console, and will switch it to display the blue Input Channel Tabs when the Channel Strips are selected from Quasar Soft
- **Group 1 - 8** will link the channel selection function between different Quasar Soft instances (browser windows).

For example, if you have two Quasar Soft windows open on two different PC monitors, and one of the two shows the Full Master Section, assigning both instances to the same Group ( can be any of the 8 groups available) will make the Master Section switch to the channel menu when any channel is selected, on both instances.

**Channel List:** In case you want to display an arbitrary sequence of channel strips (like: 8, 4, 9, 3, 2, 17 for example), enter here the channel numbers separated by commas and without spaces. The Channel List will be shown only if the **No Tabs** option is selected, otherwise it will be ignored.

## Monitoring Section

The **Type:** drop-down offers the following options:

- **No Monitor Section** will simply display only channel strips and no monitor controls
- **Small Monitor Section** will display a “reduced” version of the monitor section, with only monitor controls and without reproducing the Touchscreen GUI found in the console’s MTS module.
- **Full Master Section** will display a full reproduction of the Touchscreen GUI found in the console’s MTS module.
- **Full Master Section with Volume and Buttons** will display a full reproduction of the console’s MTS module, including the Touchscreen GUI, the Volume controls, and the HOME, TALK; and RECORD hardware keys.

The **Position:** drop-down offers the following options:

- **Left Side** will draw the monitor section to the left
- **Right Side** will draw the monitor section to the right
- **Center** will draw the Monitor Section in the center of your surface. This option is available only in case **No Tabs** is selected in the Channel Tabs drop-down.

The **User:** drop-down offers the following options:

- **No User Buttons** will not draw any button panel next to the monitor section
- **8 Buttons** will draw a Panel with 8 User buttons and 4 Bank switching buttons next to the monitor section
- **12 User Buttons** will draw a Panel with 12 User buttons next to the monitor section
- **32 User Buttons** will draw a Panel with 32 User buttons next to the monitor section

The **SmartKeys:** drop-down offers the possibility to add two “Soft SmartKey Modules” to the Quasar Soft screen. Each Module can have up to 4 Key Groups (6-Button Keypads), like the hardware SmartKey modules of the Quasar console. In order to function properly, the Soft SmartKeys must be configured in the **SmartKeys** menu of the Web UI, even if no hardware module is installed on your console. Please see [this paragraph](#) for configuring SmartKeys.

## Scaling

The **Scaling:** function, added in SW v3.2 applies to the entire Quasar Soft GUI (not just the Monitoring section). The drop-down offers the following options:

- **Best Fit** will scale a Quasar Soft GUI maintaining the graphics original aspect ratio, and according to either the width or the height of the browser window
- **Fit height** will scale a Quasar Soft GUI according to the height of the browser window, and cropping the GUI when the width is reduced
- **Fit width** will scale a Quasar Soft GUI according to the width of the browser window and cropping the GUI when the height is reduced
- **Fill** will stretch a Quasar Soft GUI without maintaining the graphics original aspect ratio, and according to either the width or the height of the browser window
- **No Scaling** will apply no changes to the Quasar Soft GUI, leaving its aspect fixed.

## Channel Strip Layout

This section lets you arrange the Widgets’ position within all Channel Strips, before starting a new Quasar Soft instance: just select a different widget for each slot using the drop-downs.

The available widgets are:

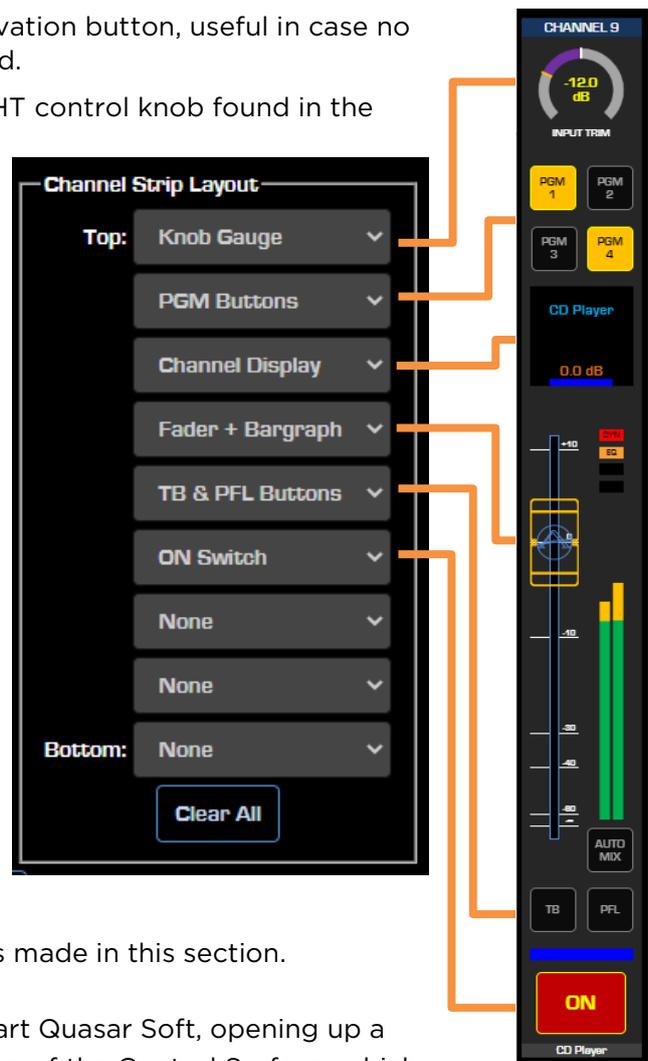
- **None:** No widget is loaded in the selected slot.
- **Knob Gauge:** replicates the knob located on top of a fader module's channel strip.
- **PGM Buttons:** replicates the 4 PGM keys of an XR-4FAD module.
- **SR PGM1+2:** replicates the PGM1 and PGM2 keys of an SR-4FAD module.
- **SR PGM3+4:** replicates the PGM3 and PGM4 keys of an SR-4FAD module.
- **Channel Display:** replicates the channel display of an XR or SR module. The display also acts as channel strips selection button when a Full Master Section is displayed.
- **Fader + Bargraph:** replicates the fader and input meter of an XR-4FAD module. It includes a small size AUTOMIX on/off button.
- **Rotary Fader:** shows a channel fader in the form of a rotary encoder. Ideal for creating compact views.
- **TB & PFL Buttons:** replicates the TB and PFL keys of an XR or SR module.
- **ON Switch:** shows a single ON/OFF switch with latching action.
- **ON & OFF:** replicates the ON and OFF switches found on XR and SR modules.
- **Pan:** replicates the PAN control knob found in the MTS Channel Input Tab.
- **Automix:** shows a large AUTOMIX activation button, useful in case no fader, or only a rotary fader is displayed.
- **Automix Weight:** replicates the WEIGHT control knob found in the MTS Automix Tab.
- **Audio Delay:** replicates the INPUT DELAY control knob found in the MTS Channel Input Tab.
- **AUX Send 1-8:** replicates the AUX SEND 1-8 control knobs found in the MTS Channel Aux Sends Tab.

The final widget position, from top to bottom of the channel strip, will follow the selection made with the drop-downs.

**Note** - The Knob Gauge function is automatically disabled, and the knob will be greyed out, in case the selected knob function (in the Source Profile) is either **Mic Gain** or **Line Gain**. This is because as we explained previously, Quasar Soft cannot control external devices such as xNodes.

The **Clear All** button will clear all assignments made in this section.

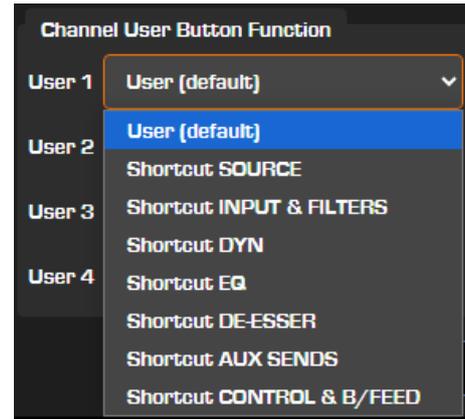
The **Start Quasar Soft Instance** button will start Quasar Soft, opening up a new browser tab with an HTML5 representation of the Control Surface, which includes all the Faders and Master Touchscreen modules. Up to 8 instances can be started, from a Quasar console.



### Channel User Button Function

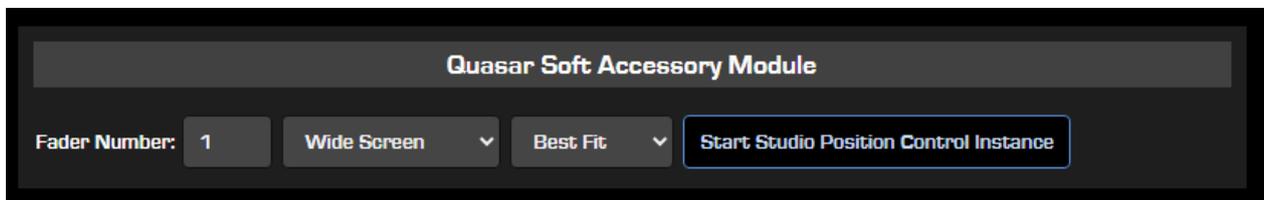
This section offers the possibility to assign Shortcuts to the Channel User Buttons when these widgets are enabled. Select **User 1&2 Buttons** and/or **User 3&4 Buttons** in one of the drop-downs of the Channel Strip Layout section, then assign a shortcut to each of those User Buttons from these 4 drop-downs.

The selected shortcut will display the corresponding Channel Tab on the Full Master section, without the need for selecting the channel first.



### Quasar Soft Accessory Module

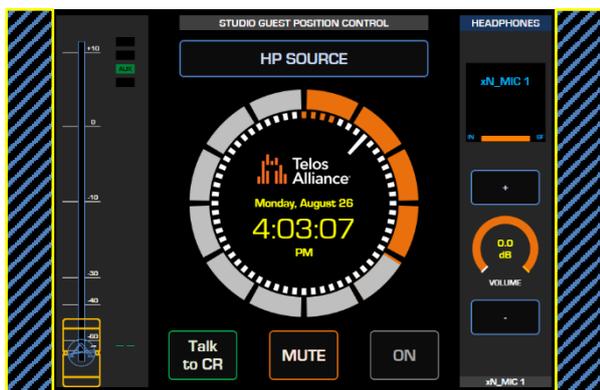
This section located at the bottom of the Quasar Soft configuration page, offers the possibility to create a convenient control panel which can be considered like a soft version of the Quasar MF1-ACC Accessory Module.



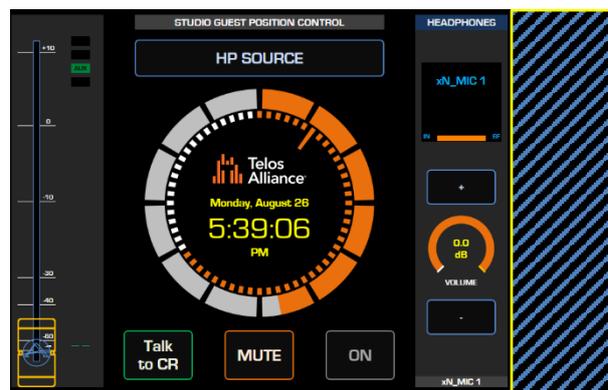
**Fader Number:** is where you enter the number of the console fader you want to control.

In the drop-down menu, the **Wide Screen** option allows to position the panel in the center of the display (with at least a 1024x768 pixel resolution) when viewed in full screen mode on your browser.

The **Compact Screen** option allows to position the panel to the left of the display (with at least a 1024x768 pixel resolution) when viewed in full screen mode on your browser. This is useful for displaying some other html5 window (like a small PF Button panel for example) in the empty space.



Wide Screen view



Compact Screen view

The **Start Studio Position Control Instance** button will start the Soft Accessory Panel, opening up a new browser tab with an HTML5 screen which includes the all controls found on the Quasar MF1-ACC Accessory Module.

Once started, the Panel offers the following controls:

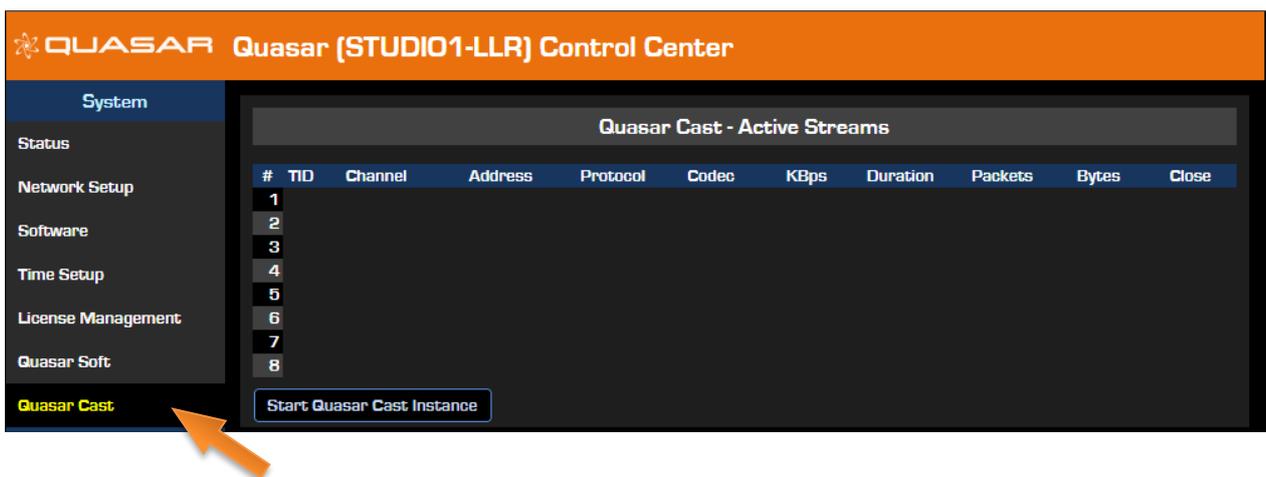


The large **+** and **-** buttons allow for fine adjustment of the headphone level in steps of 1dB while the Volume gauge allows for quick adjustment by swiping the control left or right.

The Monitor Feed Selection (normally accessed by pushing the top encoder, on the MF1-ACC module) is accessed by pressing the **HP SOURCE** button above the clock.

## Quasar Cast

Selecting this link will take you to the Quasar Cast status page. In order to see this screen, you must have a valid Quasar Soft License. Please see the previous paragraph if you have to install a Quasar Soft License.



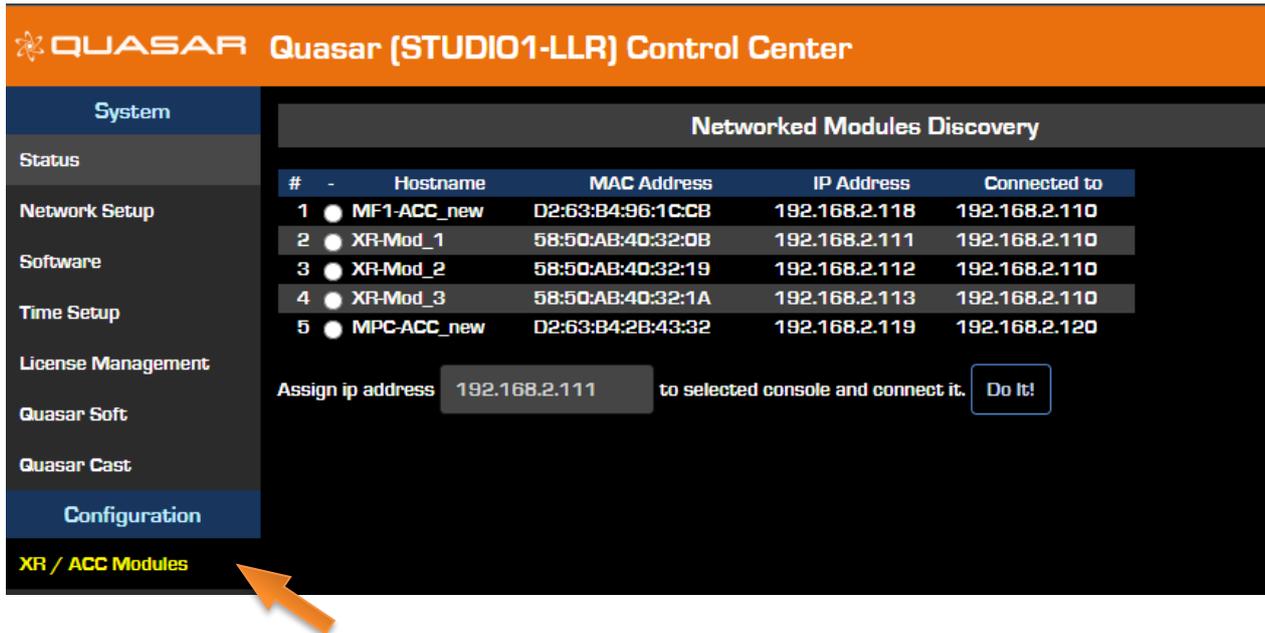
The operation of Quasar Soft and Quasar Cast is fully described in Operation Chapter 5 of this manual: [Working with Quasar Soft & Cast](#).

## Configuration Menu

Access to the Configuration Menu is mostly required during the initial system setup.

### XR/ACC Modules (Console Discovery)

Select this link to discover and list all the networked Modules (XR Fader and Accessory Modules) which are active on the network.



**QUASAR Quasar (STUDIO1-LLR) Control Center**

**System**

**Networked Modules Discovery**

#	Hostname	MAC Address	IP Address	Connected to
1	MF1-ACC_new	D2:63:B4:96:1C:CB	192.168.2.118	192.168.2.110
2	XR-Mod_1	58:50:AB:40:32:0B	192.168.2.111	192.168.2.110
3	XR-Mod_2	58:50:AB:40:32:19	192.168.2.112	192.168.2.110
4	XR-Mod_3	58:50:AB:40:32:1A	192.168.2.113	192.168.2.110
5	MPC-ACC_new	D2:63:B4:2B:43:32	192.168.2.119	192.168.2.120

Assign ip address  to selected console and connect it.

**XR / ACC Modules**

This page is normally used during factory configuration, but the user, or system administrator, might want to access it in case he needs to connect a new (additional or replacement) module to the console, or simply find out about the IP addresses of the modules which are active on the network.

Selecting the **XR/ACC Modules** link, activates the discovery process, during which the Master module will scan the network for available modules:

The active modules will be listed according their MAC addresses, even if their IP address is not configured. In order to change, or assign a new IP address to one of the modules in the list, perform the following steps:

1. Select the module you want to assign, using the radio button in the left column
2. Enter the desired IP address in the field below
3. Click on the **Do It!** button. This will trigger a reboot on the selected module
4. Wait for a moment until the module restarts, then refresh the browser page to check that the module has connected to the Master with the new IP address.

**Warning** - In case multiple Quasar consoles are connected to the same network, the Discovery process will find and list ALL the modules active on the network, and connected to all consoles. Carefully check which MTS address each module is connected to, before changing its connection settings.

**Tip** – You can quickly access each of the modules’ web UIs by right-clicking directly onto their IP address, and selecting “Open link in a new tab” .

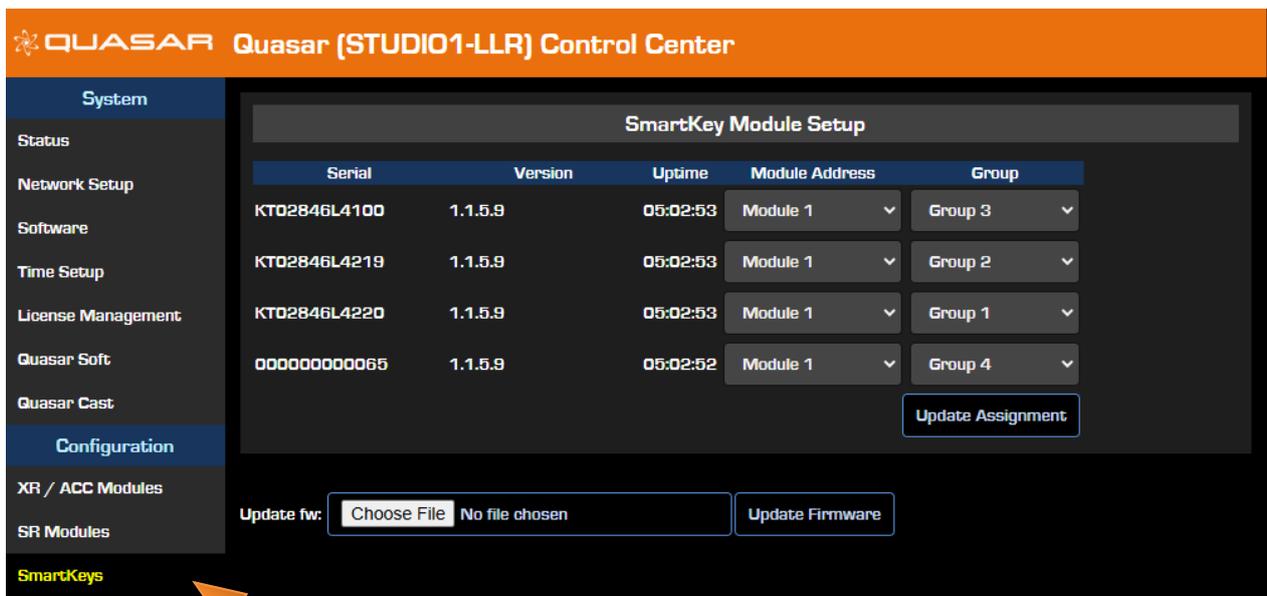


## SR Modules

The Quasar SR-4FAD modules do not have their own Web UI because they are not Ethernet-connected, but USB-connected, and therefore they are managed from the MTS Module Web UI. But for more clarity, we are describing SR Module’s UI later on in this chapter, in the [Web UI Reference: SR-4FAD Modules](#).

## Smart Keys

Select the **SmartKeys** link to discover and list all the Quasar SmartKey button groups which are active and connected to the Master.



SmartKey Modules may contain up to 4 Keypads, or Key “Groups”. The system will not detect SmartKey Modules as a whole device. The system will only detect individual Keypads as devices. These are listed in the **Serial** column, which will show the serial numbers (USB ID#) of each of the Keypads connected.

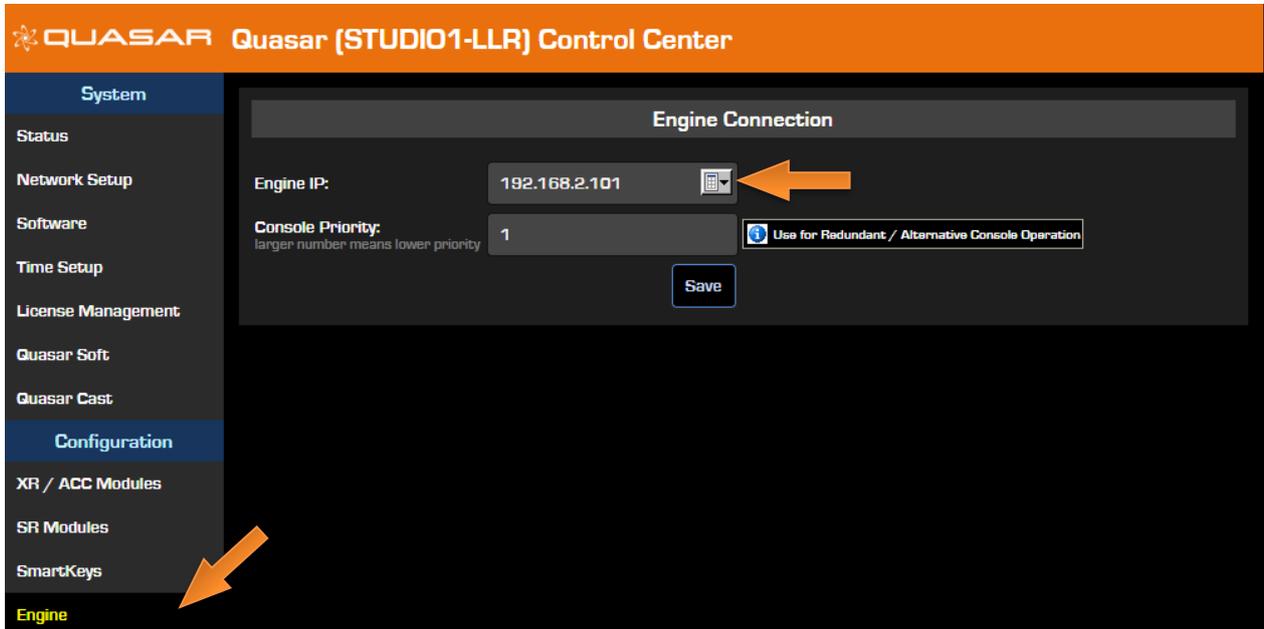
- Use the **Module Address** drop-down to assign each Keypad to a physical SmartKey Module, or to disable it.
- Use the **Group** drop-down to assign each Keypad a position within that physical Module, or to disable it.

Press the **Update Assignment** button to assign Keypads to Modules.

When a new Firmware is released for the SmartKey modules, use the “Choose File” button to upload a new FW file, and press the **Update Firmware** button to update ALL the Keypads.

## Engine

The **Engine** menu lets you assign and connect a Quasar Engine to your surface. This is normally used during factory configuration, but the user, or system administrator, might need to access it when connecting a new (or different) Engine to the console, or for checking which Console is assigned to which Engine in the entire network.



To configure Engine connection, just enter the IP Address of the Engine you want to connect to, in the **Engine IP** field, and press the **Save** button.

Alternatively, you can click on the drop-down menu and a list of available engines will appear, showing their IP Address, Software version, Licensed Channels, Operating Status and Console Connection information.

The screenshot shows a browser window titled 'Select Engine - Google Chrome' with the URL '192.168.2.110/englist'. The page displays a table with the following data:

IP Address	Version	Channels	Status	Assigned Console
192.168.2.101	3.1.2-3	64	Working	192.168.2.110 [1]

In the **Console Priority** field, you can enter a number ranging from 0 to 255 to set the connection priority that the Engine will assign to this surface. Default Value is 0 (zero).

Please Refer to Configuration Chapter 8 of this manual - [Redundant/Alternative Console Operation Mode](#), for more details about this option.

## Customize

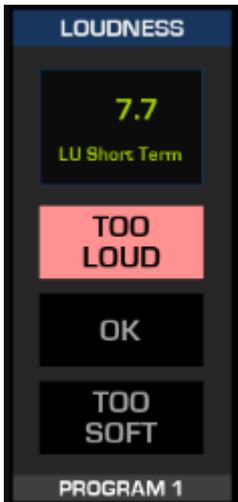
The **Customize** menu has undergone major improvements in SW v3.0, with a new graphical layout and new features added. This page offers the following configuration areas:

- User Logo
- Meter Options
- Source Type Color Coding

- Key Options
- UI Options

### Loudness Meters Options

These options apply only to the Loudness Meter Box which is visible in the Master Home screen, just above the CR SPEAKERS control section. They do not apply to the vertical bar-style loudness meters located next to each Main PPM meter.



The **Meter Type** drop-down lets you choose between the **Loudness Meter Box**, and the **Loudness Traffic Light** options. Pressing the **Apply** button will show the desired metering option on the MTS Home UI page. There is an option for not showing any box.

Two text fields are provided for entering your desired **Target Loudness** value and the **Loudness Tolerance** (acceptable +/- deviation from the target). These fields are used for driving the Traffic Light and the GPIO control. Depending on region, these values can vary: for example, EBU R128 calls for -23 LUFS, while the American Calm Act recommends -24 LKFS (LUFS = LKFS).



The **Loudness Units** drop-down lets you choose between Short Term LU and LUFS (Loudness Unit Full Scale). This applies to both the Loudness Box and the Traffic Light.

The **Control GPIO Channel** field is for entering a GPIO Channel # in order to activate a GPO alarm when the Target Loudness is exceeded.

**Note** – in SW v3.0 a new feature was introduced: GPO indication of Loudness state. One GPO pin is triggered for each state: too soft, within range, too loud. There will be a state change every time the level deviates by the amount (in dB) specified in the Loudness Tolerance field, from the Target Loudness.

The **Source** drop-down (added in SW v3.2) allows you to assign a specific bus to the Loudness Meter (both Traffic Light and Meter Box styles), with the additional option of automatically switching between the PGM1 and Record bus when the Record Mode is enabled.

### Meter Options

This section lets you select the **Meter Ballistics** options for the six Main Meters in the Master Home screen, and for the Channel I/O meters available in the bottom part of the Channel UI (Blue Tabs). Ballistics is not selectable for the Dynamics and De-Esser meters.

The **Meter Bars** drop-down lets you choose between two styles of vertical bar-style meters: narrow Stereo bars with Loudness bar, or full width Stereo Bar without Loudness bar.

The **Analog Meters** drop-down lets you select between standard vertical bar-style meters (by selecting *No Analog Meter*), or an **Analog VU** meter with needles. Select **Analog VU +3dB** for calibrating the Analog VU meter to Peak signals. For our British friends, an **Analog PPM** meter is also available.

In case the Analog VU or Analog PPM meters are selected, a **Horizontal Meter** is displayed below the needles, which can monitor the PGM1-4, Record, Phone and Monitor busses and

switch automatically to the PFL bus when this function is active on any channel.

The **Switchable Meter** option (new in SW v3.0), lets you assign:

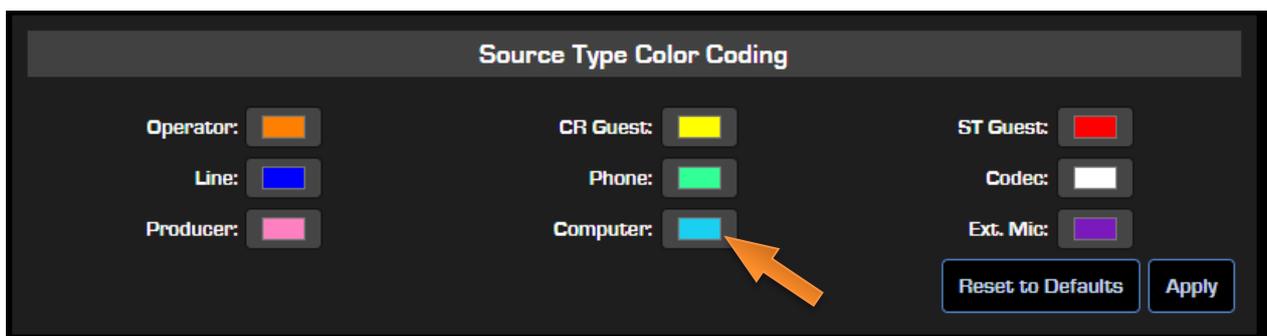
- The 4<sup>th</sup> Main Meter of the Expert UI (normally assigned to PGM4) to the Record or Phone busses.
- The 2<sup>th</sup> Main Meter of the Simple UI (normally assigned to PHONE) to PGM3, PGM4, Record, and Phone busses.

All these options, and the meters' technical specifications are covered in detail later on, in the Configuration Chapter 6 of this manual - [Audio Metering Options](#).

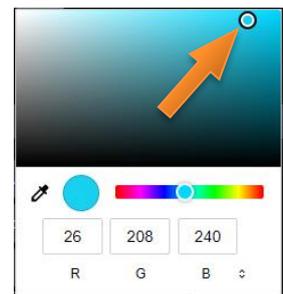
### Source Type Color Coding

This menu lets the user define colors for each type of Source Profiles, so that when a Source is loaded onto a channel, and appears on the fader strip, the color-coding indicators will let the operator easily spot all the sources that belong the same type, like Microphones, Codecs, Phones, and so on. Quasar offers two color-coding indicator bars on its fader strips:

1. The first appears at the bottom of the channel displays
2. The second one is located between the ON and OFF keys.



Selecting one of the color boxes, will display a pop-up window that lets you pick a custom RGB color and assign it to the selected Source Type. You can directly enter an RGB code, or pick a color by dragging the small circled dot.



### Key Options

The **Layer Keys Function** drop-down menu offers the following three options:

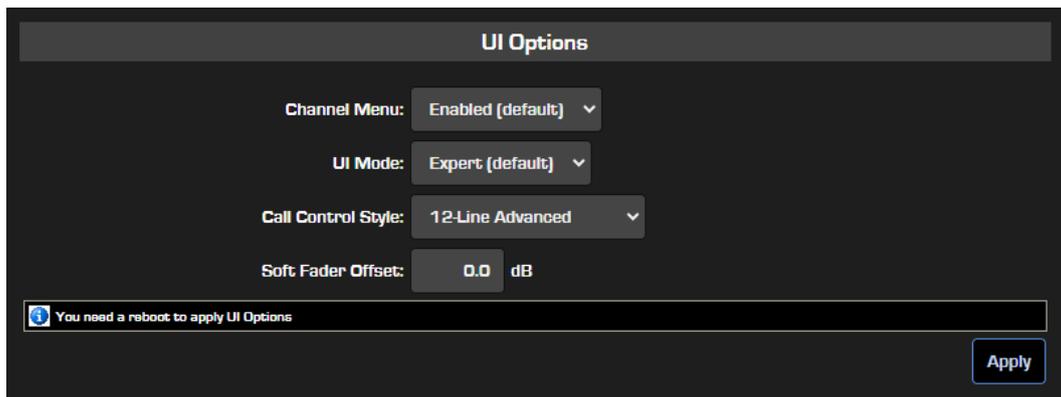
- **Extra User keys B1...B4:** will enable the four hardware Keys to work as extra User Buttons, with no Bank switching. They will be equivalent to the first 4 buttons of bank B, giving you 12 User Buttons in total. A dedicated Key Caps kit is available for purchase.
- **Layer Switching (default):** Enables Layer selection from the LAYER-BANK keys on the surface. Switching between Layers happens instantly, at the press of one of the four keys. User Banks B,C,D selection is disabled. Bank A remains active.
- **User Button Bank Switching:** Enables User Bank A-B-C-D selection from the LAYER-BANK keys on the surface. Switching between Banks happens instantly, at the press of one of the four keys. Layers are disabled. The surface will work with Layer 1 only.
- **Layer / User Button Bank Switching:** Enables selection of BOTH Layers and User Banks, from the LAYER-BANK keys on the surface: Switching between Layers happens

instantly, at the press of one of the four keys. Switching between User Banks happens when one of the four keys is pressed and held for one second.

Layers and User Keys operation is described in Configuration Chapter 4 of this Manual - [Layers & User Keys](#).

## UI Options

The **Channel Menu** drop-down menu offers two options:



- **Enabled (default):** normal channel selection functionality is enabled. It is possible to access an Input Channel through the following controls:
  - The **CHANNEL SELECT** button below the round clock in the MTS HOME screen
  - The **HOME-CH.SEL** hardware key on the MTS Module (press & hold).
  - The Option knobs located on top of each fader strip.
- **Locked:** Completely disables the access to any of the input channels, from anywhere on the control surface. The following changes are applied:
  - The **CHANNEL SELECT** button below the round clock in the MTS HOME screen is removed
  - The functionality that allows to select a channel by holding the **HOME-CH.SEL** key on the surface, is disabled.
  - The functionality that allows to select a channel by pressing one of the channel knobs, is disabled.

This lock applies to both Expert and Simple UI modes. After the Channel Menu lock is enabled, you will need to reboot the MTS Module to apply the lock.

The **UI Mode** drop-down menu enables the selection of a less sophisticated user interface. Two options are available:

- **Expert (default):** The standard Touchscreen UI on the MTS-MON Module is displayed.
- **Simple:** When selected, the Touchscreen UI changes to display only the most relevant widgets, and hides those UI tabs that are duplicated by hardware keys on the MTS module.

The Expert and Simple UI modes are described in detail in Operation Chapter 1 of this Manual. After the Simple UI Mode is enabled, you will need to reboot the MTS Module to apply the new UI. You can do so from the **Software** page.

The **Call Control Style** drop-down menu lets you choose between the standard 8-Lines Call Controller, and a new 12-lines version, which includes two soft faders for full control of your VX system.

For more details about the Call Controller options, please refer to Operation Chapter 2 of this Manual - [CALL CONTROL](#).

The **Soft Fader Offset** option lets you introduce an offset to the position of an UI-generated fader. This option applies to the Soft Fader found in the CONTROL & B/FEED Tab of the Channel Menu (in the Touchscreen UI), and to all Faders created by Quasar Soft instances. It does not apply to XR and SR Modules' physical faders.

**Note** - You will need to reboot the MTS Module, in order to apply any of the options set in the **UI Options** menu. You can do so from the **Software** page.

## Hot Keys

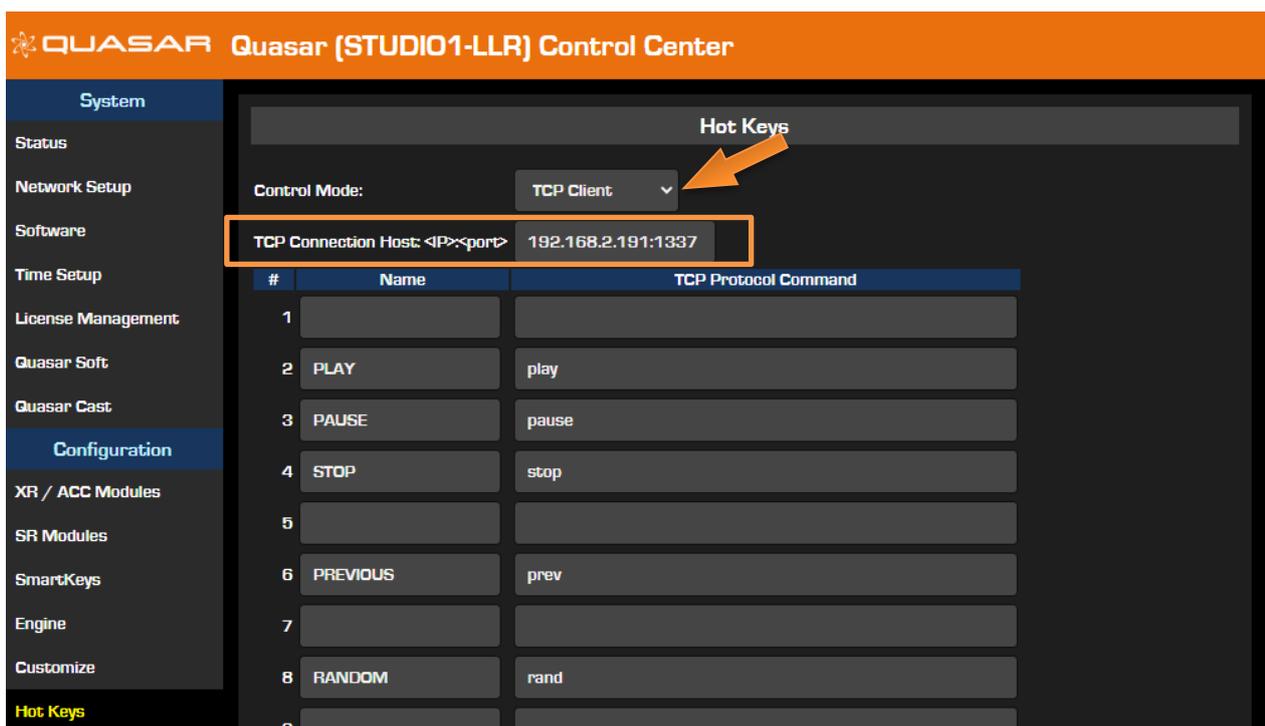
This section of the Customize menu lets you configure the Hot Keys functionality available from the Touchscreen's HOT KEY tab. When this tab is selected, the Touchscreen turns into a touch controller for an external device, such as a Jingle Machine, or a Playout System.

New in SW v3.0 is the possibility to choose between two different Control Modes: **TCP Client**, and **LWCP Control**.

### TCP Client mode

This is the same mode which was available in previous SW versions. When this mode is selected, the Hot Keys control is unidirectional: commands can only be sent from the surface, with no possibility of receiving feedback (or "tallies") from the device.

In order to control your device, the communication from the Quasar must be configured according to the device port and protocol specifications.



The screenshot shows the 'Hot Keys' configuration page in the Quasar (STUDIO1-LLR) Control Center. The 'Control Mode' is set to 'TCP Client' (indicated by an orange arrow). The 'TCP Connection Host' is set to '192.168.2.191:1337' (indicated by an orange box). Below this, a table lists 8 hot keys with their names and corresponding TCP Protocol Commands.

#	Name	TCP Protocol Command
1		
2	PLAY	play
3	PAUSE	pause
4	STOP	stop
5		
6	PREVIOUS	prev
7		
8	RANDOM	rand
9		

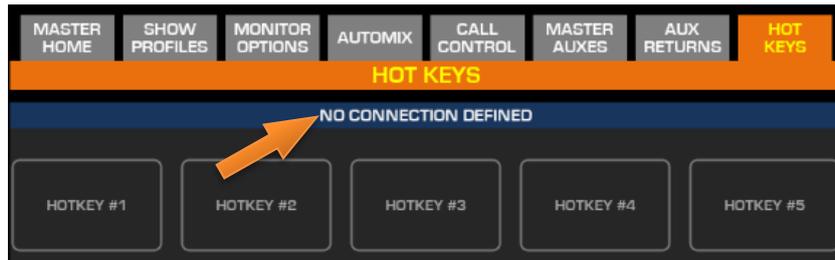
In the **TCP Connection Host** field, enter the IP address of the device you want to control (like a Cart Player or a Jingle Machine) , followed by “:” and its TCP port number.

In the **Name** field, type the name you want to be displayed by each of the Hot Keys.

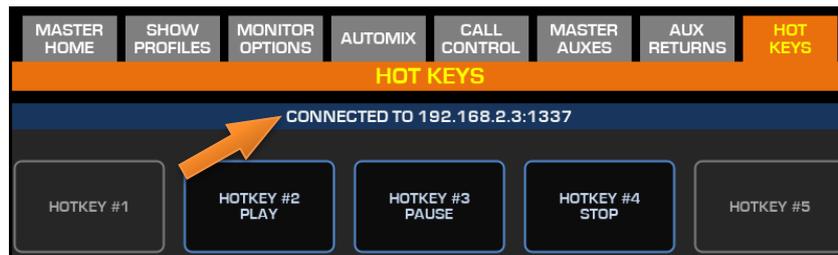
In the **Protocol Command** field, type the command string your device is expecting to receive, with the required syntax. Please refer to the device’s User Manual to obtain this information.

Click **Apply** .

Before the Hot Keys are configured properly, the console Touchscreen UI will look like this:



When a successful connection to the device is established, this will be reported in the top blue banner above the hot keys with the message: "CONNECTED TO xxx.yyy.zzz"



**Note** - Only single line commands are supported in TCP Client mode. Carriage Return or special mid-string characters are not supported.

**Note** - When a new configuration is applied, changes are pushed in real time BUT in order to view them you will need to refresh the screen by switching to a different tab and then back to the HOTKEY tab.

### LWCP Control Mode

The LWCP mode turns the SmartKey module into an LWCP button panel: for the buttons we added a new object server called 'mon.HotKey' (#1 to #30) to our internal LWCP server. In this mode, communication between the console and the device is bi-directional.

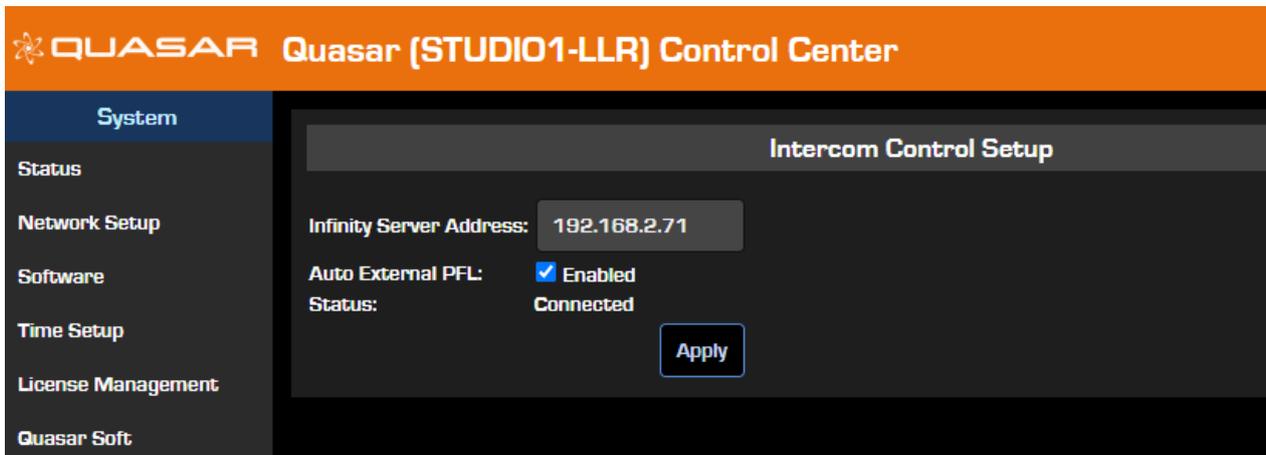
This follows the same rules of the MOD\_USER.BUT object we had in Fusion for the LCD Buttons: just program the SmartKey as you were doing for the LCD buttons using the **mon.HotKey** object, and it will work.

We didn't add the new object to the LWCP protocol yet, so Pathfinder is not yet aware of this.

**Note** - In LWCP mode, state changes and labels are pushed to the buttons in real time and the information displayed will be dynamically updated without the need refresh the screen by switching to a different tab.

## Intercom Control

This page is used to set up the connection to a Telos Infinity Intercom panel.



To configure Intercom connection, just enter the IP Address of the Telos Infinity panel you want to connect to, in the **Infinity Server Address** field, and press the **Apply** button.

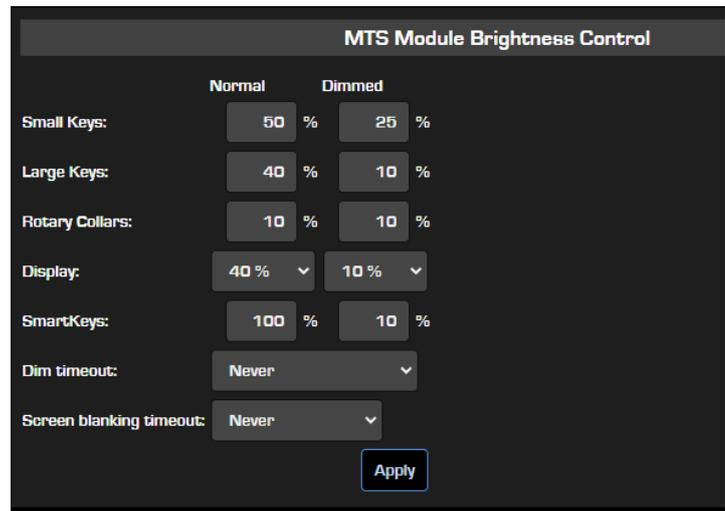
Enable the **Auto External PFL** option in order to automatically activate the Quasar External PFL input (found in Shows Monitoring Section) each time a connected Infinity panel activates its Talkback function.

Please see the dedicated Practical Configuration Guide in Configuration Chapter 5 of this Manual, to learn about [Setting Up the Intercom Controller](#).

## Brightness Control

This page is for setting the brightness of each group of hardware controls found in the Master Module.

Two settings are available for controlling the **Normal** and **Dimmed** states of each group.



Enter the desired value, directly in each of the field and click **Apply** .

**Tip** - The Brightness scale is not linear, so we suggest you to try different settings, like 30%, 50%, and 70% to work out what's the best brightness level for your environment.

The **Small Keys**, **Large Keys**, and **Rotary Collars** settings apply to the LED built in the small and large pushbuttons, and in the encoder's illuminated rings.

The **Display** control field allows setting of the Touchscreen brightness.

The default value is 40%. If the room ambient lighting condition permit, we recommend leaving this parameter set at the default value. This will help extending the lifetime of the display's backlight unit.

The **SmartKeys** settings apply to the User Button Keypads in the SmartKey Modules.

The **Dim Timeout** is the time it takes to transition from the Normal to the Dimmed state.

In order to select a different value, pick one of the preset levels from the drop-down menu and press **Apply**.

The **Screen Blanking Timeout** will let you turn the Touchscreen display completely off after a pre-set period of time. This is very useful in order to avoid the image retention effect which is normally affecting LCD displays. Set this parameter so that when the console is left unattended (during the night for example) the screen will be switched off.

The screen will turn on again as soon as any of the physical controls on the Master section are touched, including the Touchscreen itself.

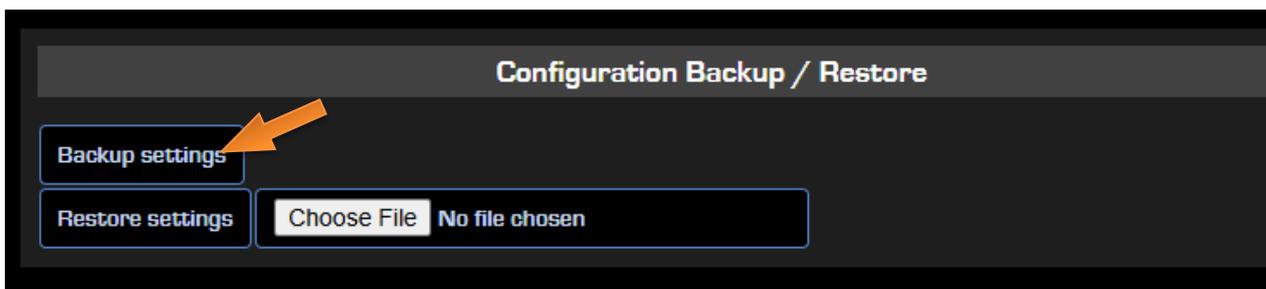
**Note -** A Similar page is available in the Fader Module (XR-4FAD) Web UI. To adjust the fader modules' brightness, just navigate to their IP Addresses with your browser and repeat the above steps for each module.

## Backup / Restore

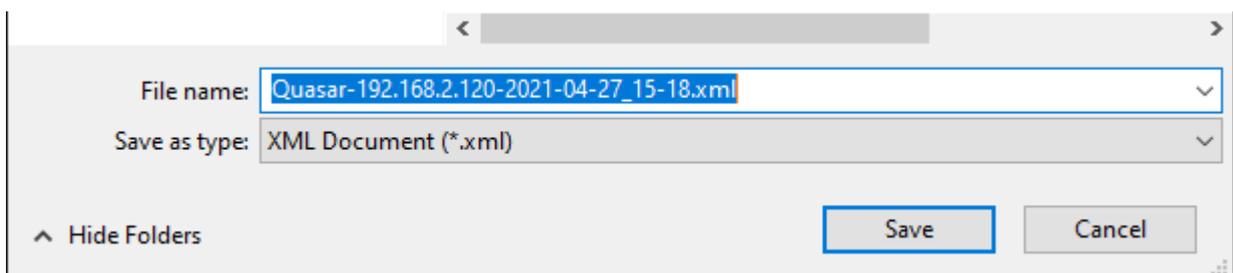
From this page, it is possible to download and save to your PC a full backup of the console configuration, in the form of an .xml file.

### Backup your Quasar Configuration

Select **Backup Settings** if you want to create and download a Backup File:



A dialog will appear, to let you save the file, in XML format. The file name will be automatically generated, based on the MTS module's IP address, Date and Time of the day.



## Restore a Backup onto your Quasar

Select **Choose File**, choose an existing backup file, then press **Restore settings**, if you want to upload a previously created backup from your archive.

The configuration backup includes:

- Connected Engine information
- NTP Settings
- Brightness Settings
- Presets
- Source Profiles (including description of each source)
- Shows Profiles (including settings for each channel and Startup Show information)
- Accessory modules configuration
- Phone Connection Settings (to console)
- UI Options
- Metering Options
- HotKey Setup

The configuration backup does not include:

- Console's IP Address, Subnet, and Gateway information
- Fader Modules configuration (you will need to back it up from each module's Web UI)
- Web Access Password

**Note** - At the moment, it is not possible to export Profiles and Presets selectively, from one console to another.

## Exporting a backup from a console and restoring it to a different one

Quasar is fundamentally different from previous Axia consoles, in the fact that its size is not fader-driven, but channel-driven. So, it depends on the Engine size, and not the surface size. A Quasar 16-fader surface in fact, with its 4 Layers, behaves like a 64-fader surface. There is no difference between a Quasar surface with 16, 32, or 64 faders. So, importing backups between different size surfaces will always work, no matter if it's from a smaller to a bigger surface, or vice versa. That's because the backup file always contains data for all 64 faders, plus Monitoring, Rec Mode, etc...

### **Example: Backup a 16-Fader console configuration and Restore it into a 12-Fader console:**

The 16-fader console could have Show Profiles which use any number of channels, from 1 to 64 (not necessarily only 16) . If you are interested in copying a show which uses 44 channels, and import a backup taken from that 16-fader console into a 12-fader console, you will be able to open the 44-ch show and load all channels using 4 layers of your 12-fader surface, if you want.

### **Example: Backup a 12-Fader console configuration and Restore it into a 24-Fader console:**

The 12-fader console could have Show Profiles which use any number of channels, from 1 to 64 (not necessarily only 16) . If you are interested in copying a show which uses only 8 channels, and import a backup taken from that 12-fader console into a 24-fader console, you will be able to open the 8-ch show and load all channels on your 12-fader surface. The unused channels on

the larger surface, will simply load default values (those generated when a new Profile is created).

**Note** - The Backup does not include the surface IP Address (Master and Fader modules) therefore, restoring a backup from a different console will not replace the IP Address of the console it is loaded to, with that of the source console.

**Warning** - Restoring a Backup will take the console off the air! Do not attempt to restore a Backup while the console is On the Air!

A Restore will :

- Remove all sources from the console database, by deleting them
- Load the new database and restore the new source, (no check or comparison between the two database is possible in this case)
- Load the Engine IP settings saved in the Backup file, and connect to this Engine (attention - they might be different from the current engine)
- Load the Startup Show Profile, following the settings saved in the backup file.

## Profiles Menu

Access to the Profiles Menu is required when the system is configured for the first time and for managing all configuration during the entire life of the system.

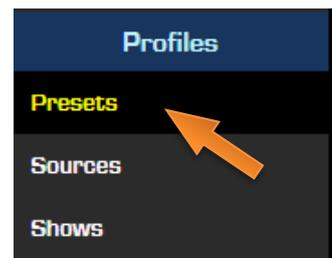
This section offers separate menus for:

- Presets
- Source Profiles
- Show Profiles

It is very important to master this part of the Quasar configuration, so we dedicated [the next Configuration Chapter](#) of this manual entirely to Source and Show Profiles, providing a detailed explanation of how these are created and managed.

### Presets

Presets are memory slots where individual settings for Filters, Equalizer, Dynamics, and De-Esser can be stored. They allow you to create a library of your favorite processing settings, and easily import them into a Show, by loading a preset to any of its channels. Presets are not part of a Show Profile, or a Source Profile: they are independent.

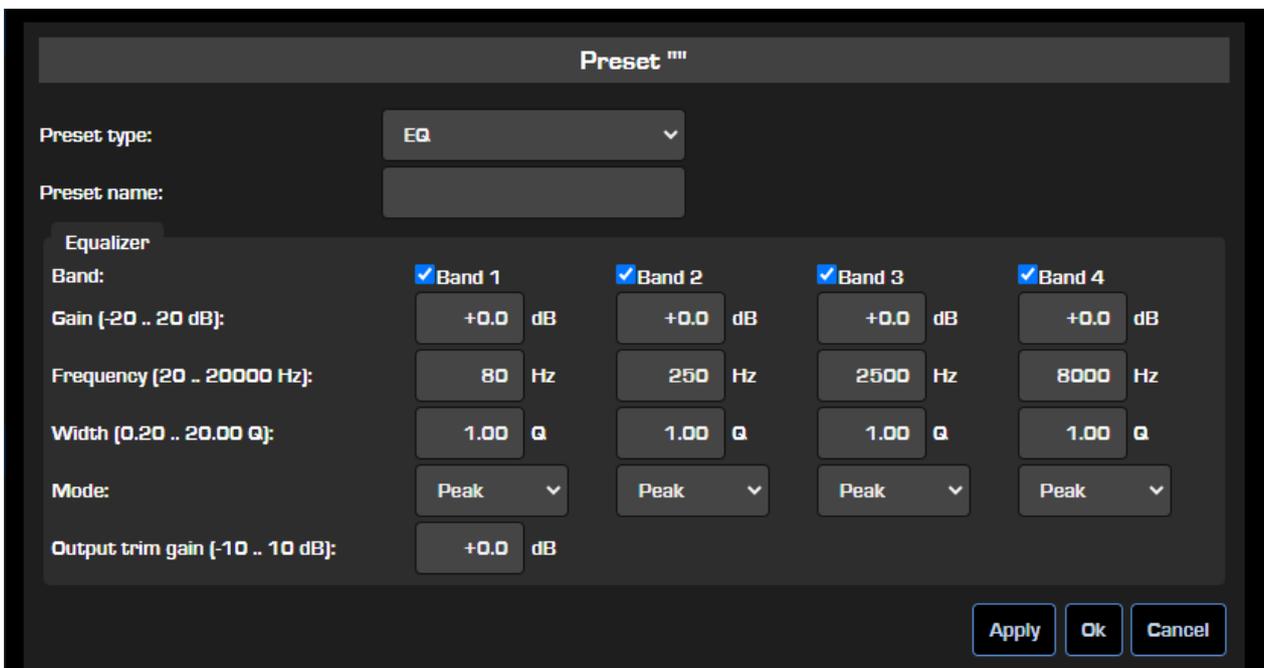
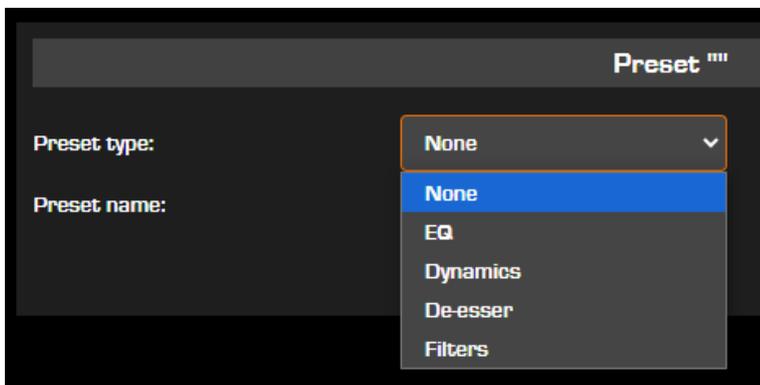
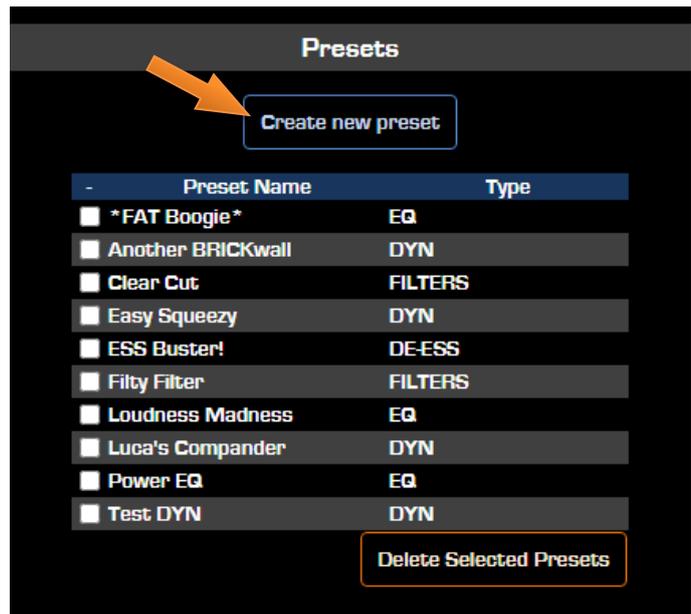


Presets also let you organize and move settings between Shows without having to manually copy each setting from the Show, or from a Source.

## Creating Presets from the Web UI

Press the **Create new preset** Button:

Then select a **Preset Type** from the drop-down menu, and the corresponding parameter settings window will appear below.



Enter a name in the **Preset Name** field. Press **Apply**, to remain in this page and proceed with entering values in the four EQ Bands. Press **Ok** when done.

## Creating Presets from the console

You can follow these steps to create Presets directly from the console UI:

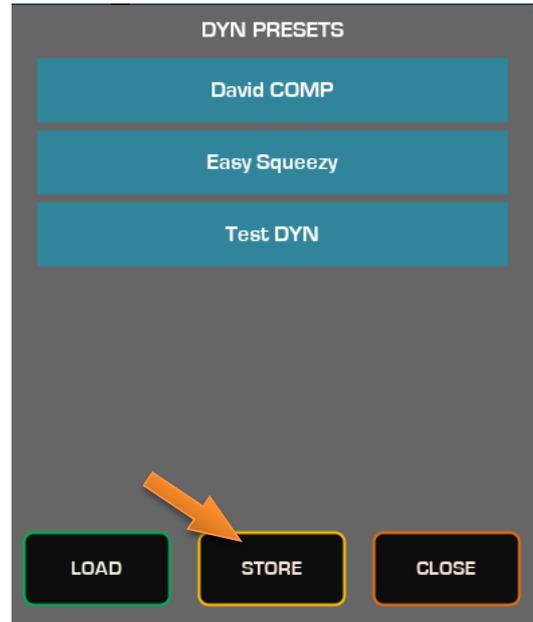
- Select any channel and create your favorite processing (EQ, DYN, etc.) settings

--- OR ---

- Load the Show with the known good settings for a given channel, that you'd like to export, and select the desired channel and processing tab

--- THEN ---

- Press the **PRESET** button on the Touchscreen UI: a "PRESET" list window will pop up.
- Press the **STORE** button: a "STORE NEW PRESET" window will appear, from which you'll be able to enter the name for your new Preset.
- Press the (green) **OK** button. The new Preset will be added to the list.
- Press the **CLOSE** button to close the list.



## Copying processing settings between Shows

If you would like to copy your favorite EQ settings (for example) from one Show Profile to another, you can do so in two different ways:

1. Load the Show with the EQ settings you want to copy, save them as a Preset, then Load the target Show, load the desired Preset so that its parameters are transferred into the Show's EQ settings, Then, update the Show in order to permanently store those settings in the Show configuration.
2. Or, you can copy the EQ settings "on the fly" (without saving them to a Preset), by simply pressing the **COPY** button, then load the target Show Profile, select the target channel, and press the **PASTE** button in its EQUALIZER tab. Then, update the Show.

**Note** - Once a Preset is created, it is possible to associate it to a Source when this is loaded. You can do so from the Source Profiles web UI.

## Sources

Please refer to [the next Configuration Chapter](#) of this manual for a detailed description of Source Profiles.

## Shows

Please refer to [the next Configuration Chapter](#) of this manual for a detailed description of Show Profiles.

## Diagnostics Menu

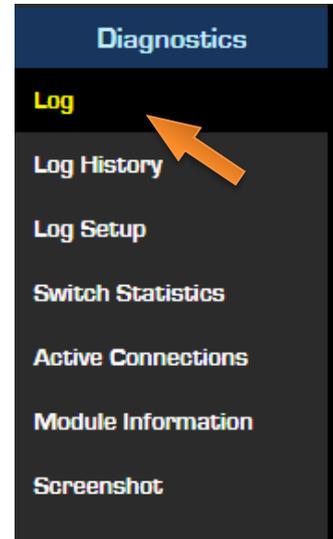
Access to Diagnostics is never required during Console Operation, or System Configuration. Generally, this menu is used to retrieve specific information about the System when asked by Telos Support, such as: Logs, Network Ports activity, Connected Modules and FW version.

### Log

The Quasar has a circular buffer which keeps the last 256 log entries in its memory. This page shows those entries, and clears after each reboot, or it can be cleared manually by pressing the

**Clear Log** button located at the bottom of the list.

The entries are sorted top to bottom, from newest to oldest.



### Log History

At each reboot a new Log file is created and saved. The files are sorted top to bottom, from newest to oldest. Each file has an upper size limit of 1MB. Click on the floppy icon to download individual files, or delete each file by selecting it with the checkbox and pressing the **Delete Selected Files** button.

### Log Setup

**Syslog Server:** This field lets you configure the IP Address for your Syslog Server. If the server is not on the same subnet as the console, you'll have to configure a Gateway in the Network Setup menu.

**Syslog Severity Filter:** Filtering depends on the type of events you want to detect, and on the type of server. Normally Error Level should be selected (intermediate).

### Switch Statistics

This page is for monitoring the activity of the Ethernet switch built-in the Master Module.

#### Switch Port Status

This section displays the status of the three ports of the switch built inside the MTS Module.

**Eth1** and **Eth2** are the two Ethernet ports at the back of the module, while **SoM** is the internal port which communicates with the CPU.

#### Switch Statistics

This section gives details about the data traffic on each of the switch ports:

- RxByte and TxByte Counters are a good indication of the amount of traffic that is handled by the switch in the module.
- Octets Counters are a good indication of the type (packet size) of traffic that is handled by the switch in the module.



# XR-4FAD Module Web UI

The XR-4FAD (Fader) Modules also have a Web UI. These modules come normally pre-configured from the factory. However, users might need to tweak some parameters like Hardware Controls, Brightness or Layers, after they gain some practice with the console.

Connect a PC to the console network, and open up a browser. Enter the IP address of one of the XR-4FAD Modules.

The first time the Web UI is accessed from a browser, a login prompt will appear, where a user password, if previously configured, needs to be entered.

If no password has been configured, you can enter the default credentials: *user* and *-no password-*.

You will be presented with the Fader Module's **System Status** Page:

**System**

**Status**

**Version Information**

Version: 3.1.0-2 (20 Jun 2024)  
Base: 1.0.0

**System Information**

Kernel: Linux 4.9.102-g62c02ddb armv7l  
Uptime: 0 days 01:56  
CPU usage: 1.4%  
CPU temp: +41.0 °C  
SYS temp: +34.3 °C  
Net Link: 1000 Mbps Full-duplex  
Net Usage: Rx: 0.097 Mbps, Tx: 0.011 Mbps  
MAC Address: 58:50:AB:40:32:0B

**File System Information**

Filesystem	Size	Used	Available	Use%
Memory	501.97 MB	63.15 MB	438.82 MB	12.6%
/	248.66 MB	8.43 MB	240.22 MB	3.4%
/mnt/config	975.90 MB	2.53 MB	973.37 MB	0.3%

**Software Components**

qsconn: dea3bbda8ea63b4cb613966960fab269029f7a8a

**Navigation Menu:** System, Status, Network Setup, Software, Time Setup, Configuration, Connection, Layers Setup, Hardware Key Map, Fader Options, Brightness Control, Backup / Restore, Diagnostics, Log, Log Setup, Switch Statistics, Module Information, Fader Calibration, Screenshot.

The left column menu is divided into three menu sections: **System**, **Configuration**, and **Diagnostics**. For simplicity, since the System and Diagnostics menus are identical to those found in the MTS-MON web UI, (we described these in the previous paragraph), we will offer a condensed description of these two menus, and a full description of the Configuration menu.

## System Menu

Access to the System Menu is mostly required for initial setup of the system.

### Status

This landing page displays all the most important information about the installed Software Version, the System's internal hardware components, and its memory usage.

### Network Setup

From this page you can set up or change your Quasar network configuration. Here you can enter the Module's Name (Hostname), the IP Address and Subnet information. Entering a Gateway is not required, unless you need to access the module from outside the network.

### Software

From this page it is possible to update the XR Module's software. In order to install a new software package, you must perform the following steps:

1. Select a new "web update file" by clicking on the **Choose File** button,
2. Wait until the software package appears in the field next to the button.
3. Click on the **Upload Software** button, and wait for it to appear in the currently inactive software bank (the bank which is not selected).
4. Select the new software bank
5. Click the **Switch to Selected Bank** button.

### Time Setup

The Quasar XR-4FAD Module requires time to be set for its logging activity. From this page you can set the System time from you PC, simply by clicking on the **Set Time from PC** button. The clock can be set to display 24-hour time (0:00 to 23:59) or 12-hour time (AM/ PM). Timezone and Day Light Saving (DST) options are also available.

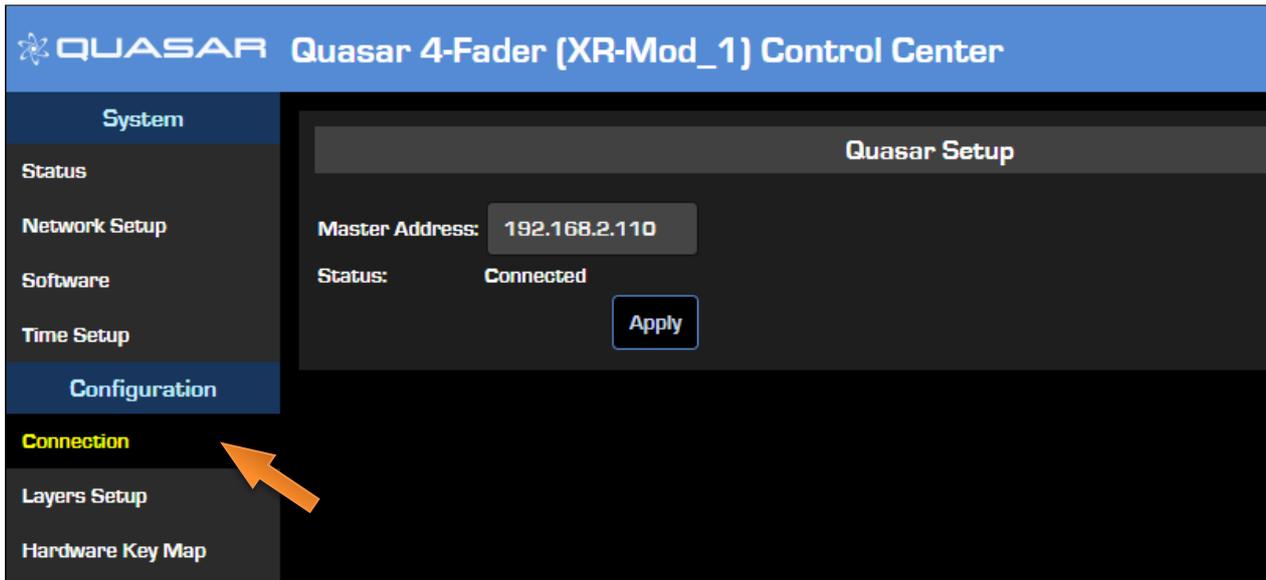
It is also possible to enter the IP address of an NTP server, to automatically update the clock every 5 minutes. Incase the **Sync to NTP** option is not selected, the module will get time from the MTS module at every connection or reboot, and will use it for its internal Logs.

## Configuration Menu

Access to the Configuration Menu is mostly required during the initial system setup.

### Connection

This page displays the IP address of the Master module which the Fader Module is connected to, and its connection Status.



It is possible to change the connected Master either from the Fader module's Web UI, using this menu, or from the desired Master module, navigating to the **XR/ACC Modules** menu found in the MTS-MON web UI.

**Note** - It is also possible to change this parameter using an external LWRP command.

### Layers Setup

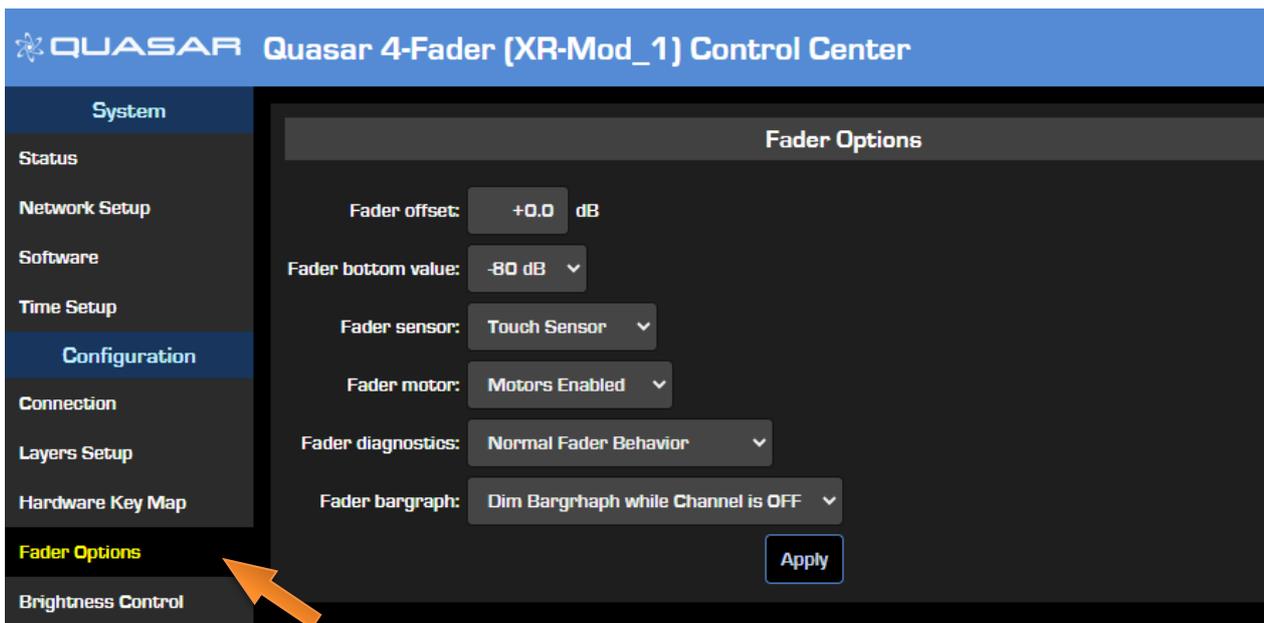
The Layers configuration is explained in Configuration Chapter 4 of this manual: [Configuring Layers](#). Please refer to this chapter for detailed instructions about setting up Layers in Quasar.

### Hardware Key Map

The Hardware Key Map configuration is explained in Configuration Chapter 4 of this manual: [Remapping the XR-4FAD Module Keys](#). Please refer to this chapter for detailed instructions about remapping buttons in Quasar XR.

### Fader Options

Some users may want to change the behavior of their faders. In this menu there are some global options which apply equally to all 4 faders:



### Fader offset

Shifts the faders' scale and level setting. For example: a setting of -10 will shift the 0dB mark (also known as "Unity Gain" point) up to the top of the fader's travel. This is useful for those operators who like working with faders all the way up.

A setting of +6 will increase the "sensitivity" of your faders by 6 dB. The faders in this example would have to be run 6 dB lower in order for the PGM audio output to be the same as the -0dB default setting.

**Note** - Meters will display the actual audio level, regardless of the offset.

### Fader Bottom Value

This function lets you set the value in dB to the bottom of the fader's scale. 5 settings are available: -80dB, -75dB, -70dB, -65dB, -60dB . This value represents the transition point between the maximum fader attenuation and the -infinite. For example, a setting of -60dB will reduce the fader gain range from +10/-80dB to +10/-60 and increase the -infinite range.

**Note** - In case the Fader Start function is used, changing the Fader Bottom Value will also move the fader-start point, since this function is triggered when the transition point is reached.

### Fader Sensor

This menu lets you choose between two different ways of activating a fader, using the built-in Touch Sensors or the Motion Sensor.

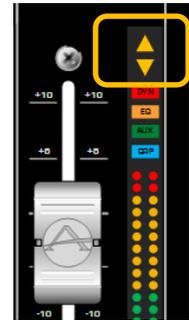
- **Touch Sensor** is a more advanced control technology, which offers protection against unwanted movement of the fader: if a non-human object (like a newspaper, or a mobile phone) ends up hitting and moving a fader which is on the air, the fader will not change gain and its motor will try to return it to its original position. The fader will be enabled only in case it is touched by a finger, allowing the operator to move it to the desired position.

**Warning** - In case the operator has long fingernails and pushes the fader with the tip of his/her nails, the fader will fight back thinking it is being moved accidentally. This is because nails are non-conductive objects similar to plastic.

- **Motion Sensor** is a traditional control technology, which offers no protection against unwanted movement of the fader. In this case the fader will always work, no matter what is moving it, and it will behave like the fader in an analog console.

### Fader Motor

This menu allows you to disable the motor of all 4 Faders, in the module. In this case, if the fader is controlled by another fader (like a Quasar Soft fader, or a different fader controlling the same input channel) the two triangular LEDs above the meter bar will light up to indicate that the fader's logical position has changed, and is no longer matching the fader's physical position.



### Fader Diagnostics

For normal operation you must leave this option set to its default value: **Normal Fader Behavior**. You should set it to **Fader Movement Simulation** only in case you are diagnosing some problems with the fader's motor, and Telos Support asked you to enable this option.

### Fader Bargraph

New in SW v3.0, the **Dim Bargraph while Channel is OFF** option allows you to reduce the brightness of the LED Bargraph meters when the channel is OFF, and return to full brightness when the channel is ON. Select **Normal Bargraph Behavior** to disable this function.

**Note** - FW v4.3 or later must be installed on the XR Module to enable this function.

## Brightness Control

This page is for setting the brightness of each group of hardware controls found in the Fader Module. Two settings are available for controlling the **Normal** and **Dimmed** state of each group. Enter the desired value, directly in each of the field and click **Apply**.

	Normal	Dimmed
Small Keys:	50 %	25 %
ON-OFF Keys:	40 %	20 %
Key Guards:	15 %	5 %
Rotary Collars:	20 %	10 %
Bargraphs:	50 %	20 %
Display:	100 %	52 %
Dim timeout:	1 Minute	
Screen blanking timeout:	Follow MTS (default)	

The **Small Keys** drop-down sets the brightness for the PGM 1-4, USER 1-4, SET and HOLD (if installed in place of USER 3-4), AUTOMIX, TB and PFL keys.

The **ON-OFF Keys** drop-down sets the brightness for the fader ON and OFF keys.

The **Key Guards** drop-down sets the brightness for the LED strip between the ON and OFF keys.

The **Rotary Collars** drop-down sets the brightness for the Channel Knob LED collar.

The **Bargraphs** drop-down sets the brightness for the fader LED Meters.

The **Display** drop-down sets the brightness of the four Channel Displays. In order to select a different value, pick one of the preset levels from the drop-down menu and press **Apply**.

The **Dim Timeout** sets the time it takes to transition from the Normal to the Dimmed state. Select **Follow MTS** to make the XR module dim when the MTS Module enters the dimming state, and return to normal state when the MTS module is touched.

**Tip** - The Brightness scale is not linear: you can try different settings, like 30%, 50%, and 70% to work out what's the best brightness level for your environment.

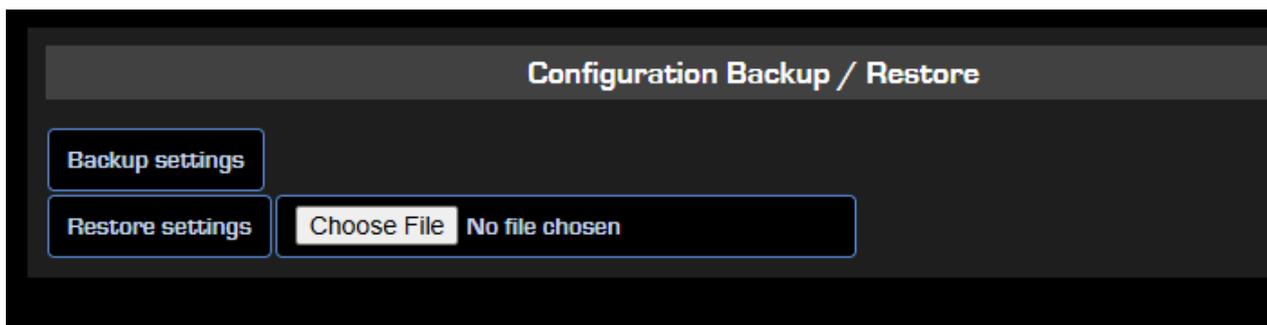
## Backup / Restore

From this page, it is possible to download and save to your PC a backup of the selected Fader Module configuration, in the form of an .xml file.

### Saving a Backup

Select **Backup Settings** if you want to create and download a Backup File:

A dialog will appear, to let you save the file, in XML format. The file name will be automatically generated, based on the MTS module's IP address, Date and Time of the day.



### Restoring a Backup

Select **Choose File**, choose an existing backup file, then press **Restore settings**, if you want to upload a previously created backup from your archive.

## Diagnostics Menu

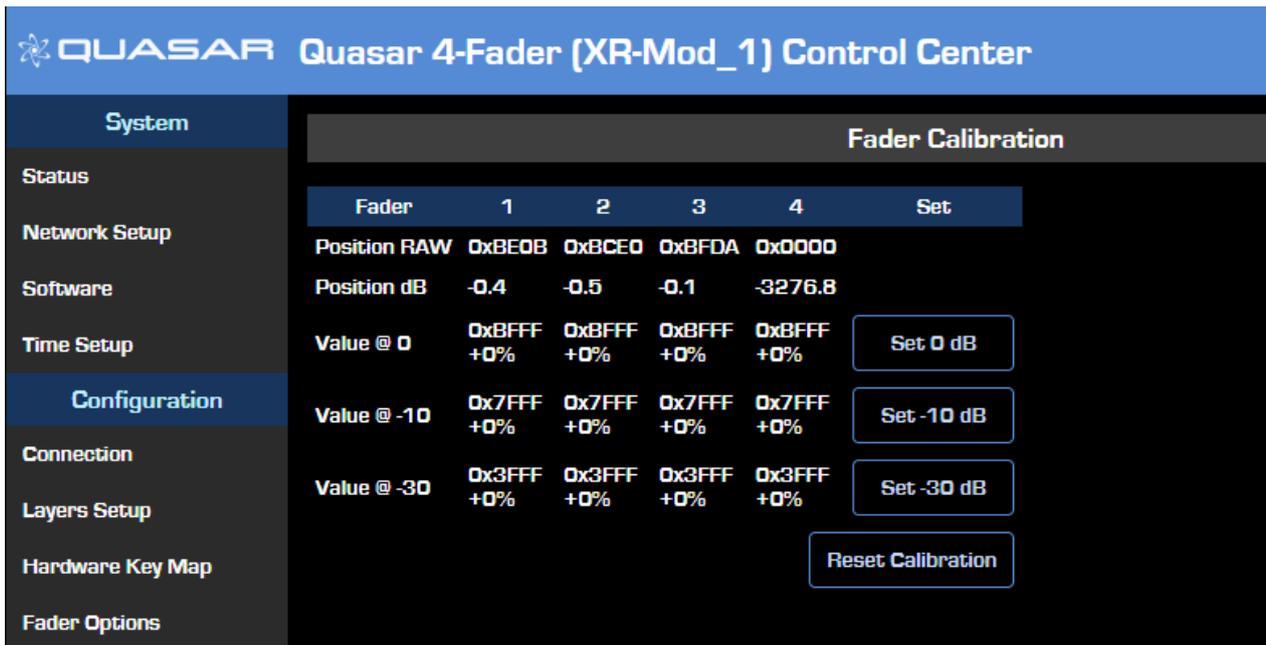
Access to the Fader Diagnostics Menu is never required during Console Operation, or System Configuration. Generally, this section is used to retrieve specific information about the System when asked by Telos Support, such as Logs, Network Ports activity, Connected Modules and Active Firmware version.

The description of the **Log**, **Log Setup**, **Switch Statistics**, and **Module Information** pages found in this menu is equivalent to that of the [MTS-MON Module Diagnostics](#), given in the previous paragraph.

## Fader Calibration

**Fader Calibration** is used to finely adjust the position of each fader with reference to the dB scale printed on the Fader Module surface.

The fader calibration procedure should be performed by a qualified technician. Normally this process is performed at the factory before shipping. Please do not attempt to calibrate faders, and do not reset the calibration, unless you have experience in this field, or you have been specifically instructed to do so by one of our Support Engineers.



The screenshot displays the 'Quasar 4-Fader (XR-Mod\_1) Control Center' interface. On the left is a navigation menu with options: System, Status, Network Setup, Software, Time Setup, Configuration (highlighted), Connection, Layers Setup, Hardware Key Map, and Fader Options. The main area shows the 'Fader Calibration' settings for four faders. The table below summarizes the data shown in the interface.

Fader	1	2	3	4	Set
Position RAW	0xBE0B	0xBCE0	0xBFDA	0x0000	
Position dB	-0.4	-0.5	-0.1	-3276.8	
Value @ 0	0x8FFF +0%	0x8FFF +0%	0x8FFF +0%	0x8FFF +0%	Set 0 dB
Value @ -10	0x7FFF +0%	0x7FFF +0%	0x7FFF +0%	0x7FFF +0%	Set -10 dB
Value @ -30	0x3FFF +0%	0x3FFF +0%	0x3FFF +0%	0x3FFF +0%	Set -30 dB
Reset Calibration					

In case it might be necessary to re-align the faders after the console has been shipped to its final destination, and has been used for a while, please contact Telos Support to obtain instructions and help on how to perform Fader Calibration.

## Screenshot

This useful feature lets you retrieve a snapshot of the graphics displayed by the four Channel Displays available in each Fader Module. It is very useful in case a remote operator needs to quickly monitor the module's state, as if he was in front of the console.

# SR-4FAD Module Web UI

The Quasar SR-4FAD Modules do not have their own IP address because they are not Ethernet-connected. Their UI is included in the MTS-MON module Web UI and therefore they are managed from the MTS Module. Nevertheless, we will devote an entire paragraph to these modules.

Select the **SR Modules** link. The following page will appear, showing a list of connected SR-4Fad Modules:

The screenshot displays the Quasar (STUDIO1-LLR) Control Center web interface. The left sidebar contains a navigation menu with categories: System, Configuration, Profiles, and Diagnostics. Under Configuration, the 'SR Modules' link is highlighted with a red arrow. The main content area is divided into three sections: Channel Assignment, Hardware Key Map, and Fader Options.

**Channel Assignment**

Name	Version	Fad 1	Fad 2	Fad 3	Fad 4	ID	Setup
SR Mod_1	4.2.0	1	2	3	4	ID	Setup
SR Mod_2	4.2.0	5	6	7	8	ID	Setup
SR Mod_3	4.2.0	9	10	11	12	ID	Setup

**Hardware Key Map**

Key PGM 1: PGM 1 (default) ▾  
Key PGM 2: PGM 2 (default) ▾  
Key PGM 3: PGM 3 (default) ▾  
Key PGM 4: PGM 4 (default) ▾  
Key USER 1: AUTO PHONE ▾  
Key TB: TB (default) ▾  
Key PFL: PFL (default) ▾  
Key ON: ON (default) ▾  
Key OFF: OFF (default) ▾

Buttons: Reset to Default, Apply

**Fader Options**

Physical Fader offset: 0.0 dB  
Physical Fader bottom value: -60 dB ▾  
Fader sensor: Motion Sensor ▾ Apply

Unlike the XR-4FAD modules, which have a Web UI dedicated to each module, and all settings can be different for each single module, Quasar SR-4FAD modules configuration includes some settings which are global, and some others which are individual and applied per-module.

The **SR Modules** landing page contains the global configuration parameters. In order to setup each of the SR modules, you must press the **Setup** button next to the desired module, in the **Channel Assignment** section at the top of this page.

## Channel Assignment

The options in this section allow you to assign an Input Channel to each of the 4 Faders (labeled “Fad1” – “Fad4”) in each of the SR modules listed.

The **ID** button lets you identify the physical module in your surface: when pressed, its channel displays will start flashing. This is useful to make sure you are configuring the correct module in your surface.

The **Setup** button enters the individual settings page, for each of the SR modules.

Once done, press the **Update Assignment** button in order to apply the settings to all modules.

To summarize, the Global Configuration settings applied to all SR modules are:

- Hardware Key Map
- Fader Options

And the individual settings applied to each SR module are:

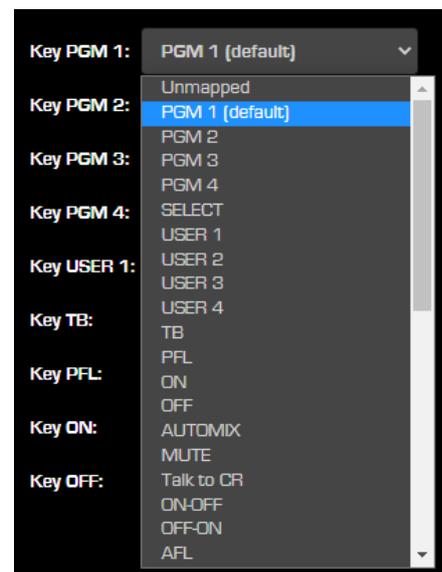
- Module Information (Name)
- Brightness Control
- Fader Calibration
- Firmware Update

## SR Modules – Global Configuration

### Hardware Key Map

The options in this section are the same as those found in the XR Module’s Hardware Keymap. But in case of the SR Modules, it is not possible to remap each module with different functions.

The Hardware Key Map configuration is explained in Configuration Chapter 4 of this manual: [Remapping the SR-4FAD Module Keys](#) Please refer to this chapter for detailed instructions about remapping buttons in Quasar SR.



## Fader Options

The options in this section are the same as those found in the XR Module's Fader Options. But in case of the SR Modules, it is not possible to configure each module with different fader options.

### Physical Fader offset

Shifts the faders' scale and level setting.

Example1: a setting of -10 will shift the 0dB mark (also known as "Unity Gain" point) up to the top of the fader's travel. This is useful for those operators who like working with faders all the way up.

Example 2: a setting of +6 will increase the "sensitivity" of your faders by 6 dB. The faders in this example would have to be run 6 dB lower in order for the PGM audio output to be the same as the -0dB default setting.

**Note** - Your meters will always display the actual audio levels, no matter what the offset is.

### Physical Fader Bottom Value

This function lets you set the value in dB to the bottom of the fader's scale. 5 settings are available: -80dB, -75dB, -70dB, -65dB, -60dB . This value represents the transition point between the maximum fader attenuation and the -infinite.

For example, a setting of -60dB will reduce the fader gain range from +10/-80dB to +10/-60 and increase the -infinite range.

**Note** - In case the Fader Start function is used, changing the Fader Bottom to a higher value (from -80 to -75) will also move the fader-start point up, since this function is triggered when the transition point is reached.

### Fader Sensor

This menu lets you choose between two different ways of activating a fader, using the built-in Touch Sensors or the Motion Sensor.

- **Touch Sensor** is a more advanced control technology, which offers protection against unwanted movement of the fader: if a non-human object (like a newspaper, or a mobile phone) ends up hitting and moving a fader which is on the air, the fader will not change gain and its motor will try to return it to its original position. The fader will be enabled only in case it is touched by a finger, allowing the operator to move it to the desired position.

**Warning** - In case the operator has long fingernails and pushes the fader with the tip of his/her nails, the fader will fight back thinking it is being moved accidentally. This is because nails are non-conductive objects similar to plastic.

- **Motion Sensor** is a traditional control technology, which offers no protection against unwanted movement of the fader. In this case the fader will always work, no matter what is moving it, and it will behave like the fader in an analog console.

# SR Modules – Individual Setup

Press the **Setup** button to enter each of the modules' Setup page:

**QUASAR Quasar (STUDIO1-LLR) Control Center**

**System**

- Status
- Network Setup
- Software
- Time Setup
- License Management
- Quasar Soft
- Quasar Cast

**Configuration**

- XR / ACC Modules
- SR Modules** (indicated by an orange arrow)
- SmartKeys
- Engine
- Customize
- Hot Keys
- Intercom Control
- Brightness Control
- Backup / Restore

**Profiles**

- Presets
- Sources
- Shows

**Diagnostics**

- Log
- Log History
- Log Setup
- Switch Statistics
- Active Connections
- Module Information
- Screenshot

---

**SR Module Information**

Module Name:

CH01	CH02	CH03	CH04
xN_MIC 1 Operator	xN_MIC 2 Producer	xN_MIC 3 CR Guest	xN_MIC 4 ST Guest
+0.6 dB	+0.2 dB	+1.4 dB	+1.1 dB
126880	126880	126880	126880

Module uptime is 04:09:06  
Time since last event 04:09:06

---

**Brightness Control**

	Normal	Dimmed
Small Keys:	50 %	25 %
ON-OFF Keys:	30 %	10 %
Key Guards:	15 %	5 %
Rotary Collars:	20 %	10 %
Fader Arrows:	40 %	20 %
Wide Keys:	30 %	15 %
Displays:	100 %	40 %

---

**Fader Calibration**

Fader	1	2	3	4	Set
Position RAW	0x0000	0x0000	0x0000	0x0000	
Position dB	-3276.8	-3276.8	-3276.8	-3276.8	
Value @ 0	0xBFFF +0%	0xBFFF +0%	0xBFFF +0%	0xBFFF +0%	<input type="button" value="Set 0 dB"/>
Value @ -10	0x7FFF +0%	0x7FFF +0%	0x7FFF +0%	0x7FFF +0%	<input type="button" value="Set -10 dB"/>
Value @ -30	0x3FFF +0%	0x3FFF +0%	0x3FFF +0%	0x3FFF +0%	<input type="button" value="Set -30 dB"/>

---

**Firmware Update**

Running Version 4.2.0 "4.2.0" <axia,qs-mcu-mk24> 05cac3e2e1fd8f68d13b6b90be35cd4c504d3855

Firmware Image:  No file chosen

## SR Module Information

This section is for entering the name of each Fader Module, and monitor its state, uptime and last event update. Press **Apply** when done. A screenshot of the four Channel Displays is provided, to let you monitor the status of each of the channel strips.

## Brightness Control

Controls in this section are equivalent to those found in the MTS-MON and XR-4FAD modules.

**Note** - You will notice that there are no Timeout settings in SR. That's because SR Modules' Brightness always follows the Timeout set in the MTS-MON module.

## Fader Calibration

Fader Calibration is used to finely adjust the position of each fader with reference to the dB scale printed on the Fader Module surface.

The fader Calibration procedure should be performed by a qualified technician. Normally this process is performed at the factory before shipping. Please do not attempt to calibrate faders, and do not Reset the Calibration, unless you have experience in this field, or you have been specifically instructed to do so by one of our Support Engineers.

Fader Calibration					
Fader	1	2	3	4	Set
Position RAW	0x0000	0x0000	0x0000	0x0000	
Position dB	-3276.8	-3276.8	-3276.8	-3276.8	
Value @ 0	0xBFFF +0%	0xBFFF +0%	0xBFFF +0%	0xBFFF +0%	Set 0 dB
Value @ -10	0x7FFF +0%	0x7FFF +0%	0x7FFF +0%	0x7FFF +0%	Set -10 dB
Value @ -30	0x3FFF +0%	0x3FFF +0%	0x3FFF +0%	0x3FFF +0%	Set -30 dB
Reset Calibration					

In case it might be necessary to re-align the faders after the console has been shipped to its final destination, and has been used for a while, please contact Telos Support to obtain instructions and help on how to perform Fader Calibration.

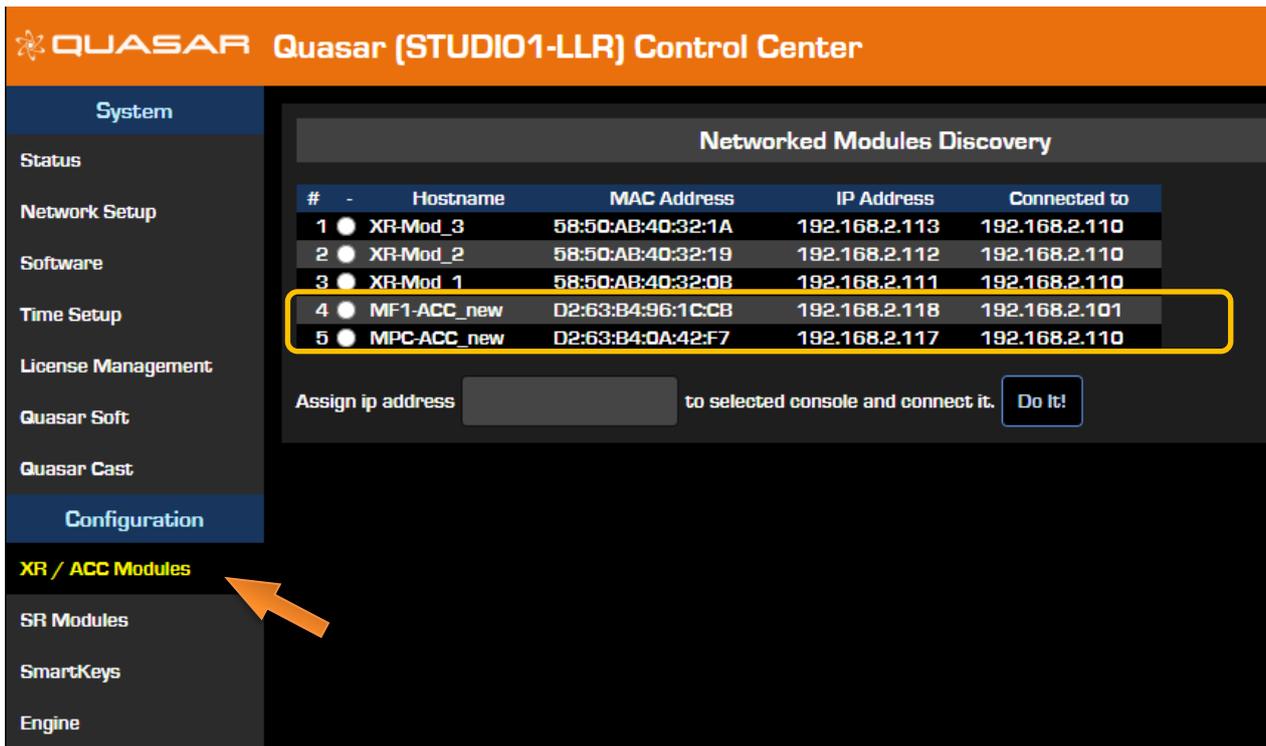
## Firmware Update

1. Select the update file (has an .itb extension) by clicking on the **Choose File** button,
2. Wait until the software package appears in the field next to the button.
3. Click on the **Upload Firmware** button. Click the **SR Modules** link to check that the new Firmware version has been uploaded, by looking at the Channel Assignment section.

# ACC Modules Web UI

Accessory Modules are discovered and assigned to a Quasar surface by navigating to the MTS-MON Module Web UI, and selecting the **XR/ACC Module** link of the MTS Web UI, from the Configuration menu.

From this page, you can select the ACC Module with the radio button, assign it an IP address, and connect the Accessory Module to the console Master Module.



**Note** - ACC Modules are supported in Quasar SW v.2.0.1 and later versions.

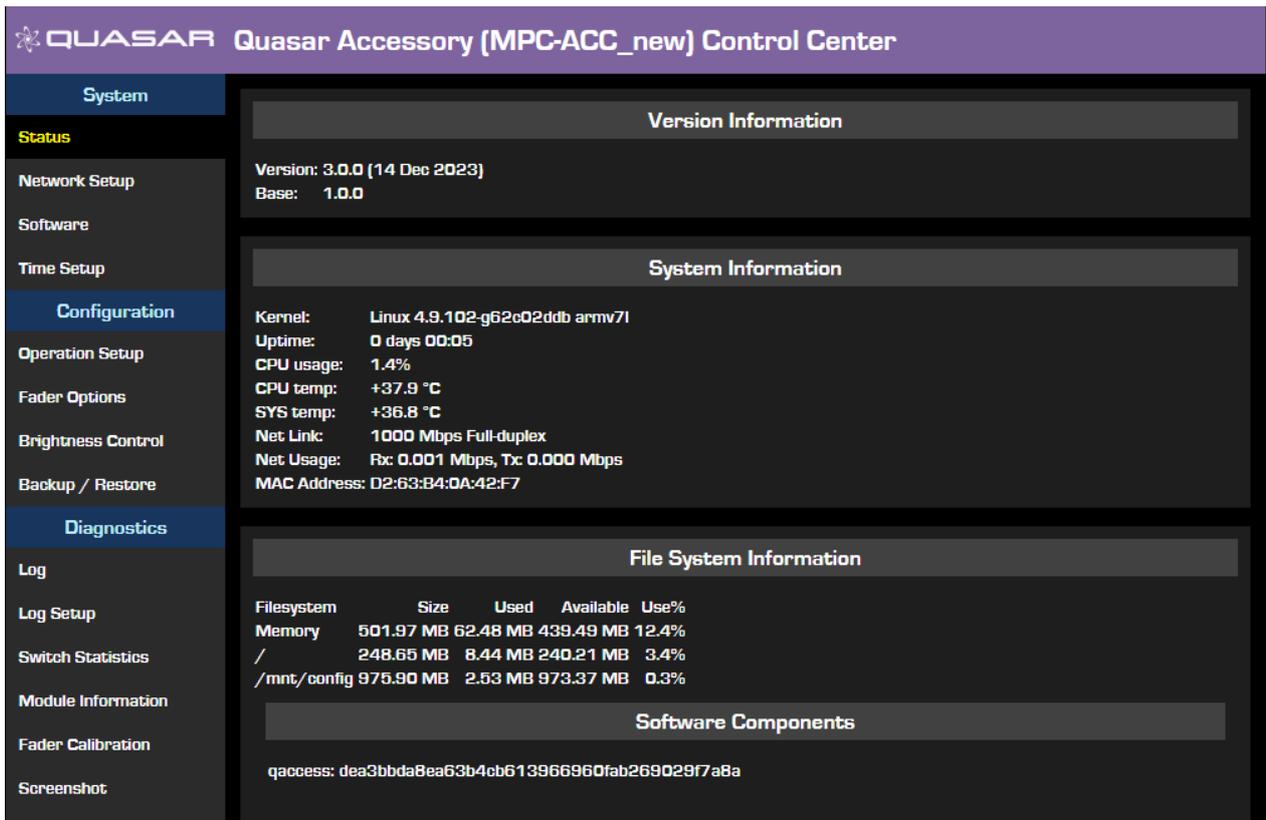
Once the IP address is assigned, it is possible to navigate the Accessory Module Web UI, by clicking its IP in the Networked Modules Discovery page (shown above) or by simply entering its IP in your browser.

The first time the Web UI is accessed from a browser, a login prompt will appear, where a user password, if previously configured, needs to be entered.

On a brand new unit, the default access credentials are: *user - no password*

The screenshot shows a 'Sign in' dialog box. It displays the URL 'http://192.168.2.120' and a warning: 'Your connection to this site is not private'. There are two input fields: 'Username' with the value 'user' and an empty 'Password' field. At the bottom right, there are 'Sign in' and 'Cancel' buttons.

The following **Status** page will appear:



The left column menu is divided into three menu sections: **System**, **Configuration**, and **Diagnostics**. For simplicity, since the System and Diagnostics menus are identical to those found in the MTS-MON web UI, (described in the previous paragraph), we will offer a condensed description of these two menus, and a full description of the Configuration menu.

## System Menu

The **Status**, **Network Setup** and **Software** pages in the System Menu are the same as those found on the Master and XR Fader Modules Web UIs. They will let you monitor the status of the module, enter the complete network settings for it, and install software updates.

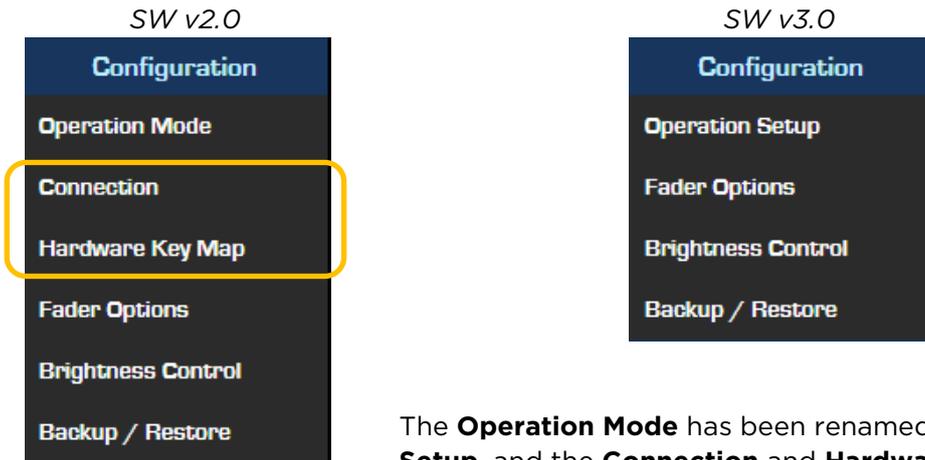
The **Time Setup** page is also the same as that found on the Master and Fader Modules, but it is important to remember that the Accessory Modules internal clock is not battery-backed up: when the module is connected to a console, it will get time information from the MTS-MON Module. Alternatively, an NTP server can be configured.

For a full description, please refer to the previous paragraphs in this chapter: [MTS-MON Module System Menu](#) or [XR-4FAD Module System Menu](#).

# Configuration Menu

Access to the Configuration Menu is mostly required during the initial system setup.

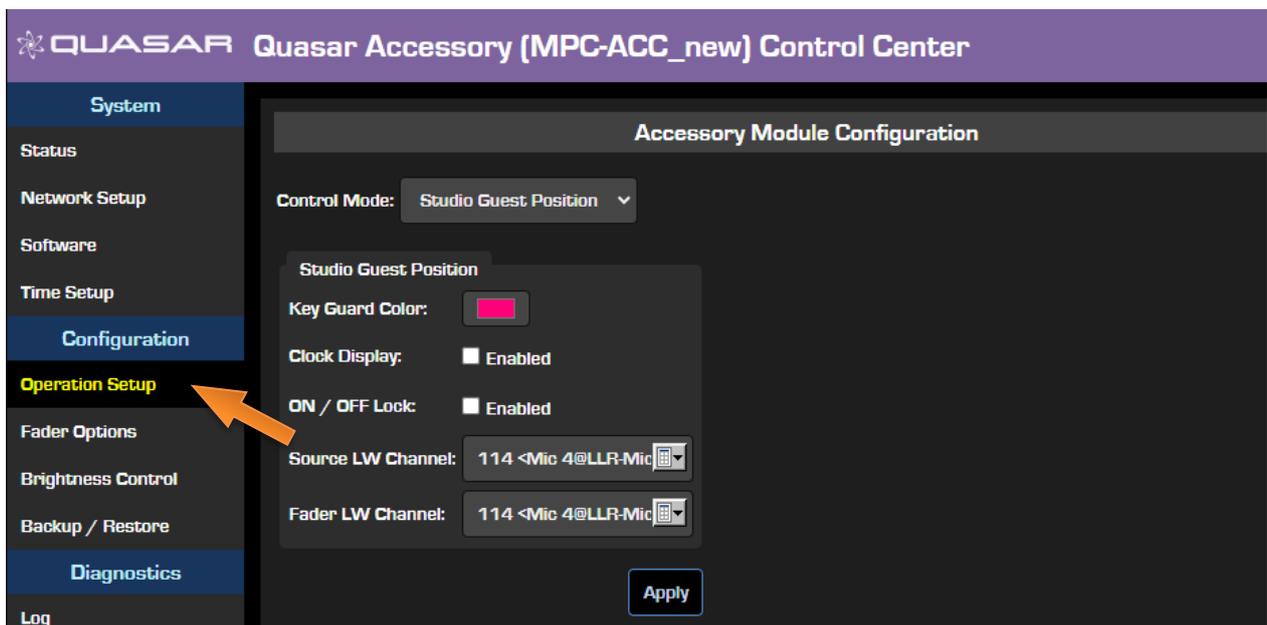
The Configuration menu has been greatly simplified in SW v3.0. In previous versions, some settings related only to the Channel Strip Control mode, were exposed in the main menu, leading to potential confusion in case other control modes were selected. In v3.0 the UI has been rationalized and updated to display only the relevant settings for the selected Control Mode. Please take a look at the differences:



The **Operation Mode** has been renamed **Operation Setup**, and the **Connection** and **Hardware KeyMap** settings, active only when the Channel Strip Control mode is selected, have now been moved to the Operation Setup menu and appear only when this particular mode is selected.

## Operation Setup

From the **Configuration** menu of the module's Web UI, select **Operation Setup**: The Accessory Module Configuration page will be displayed:



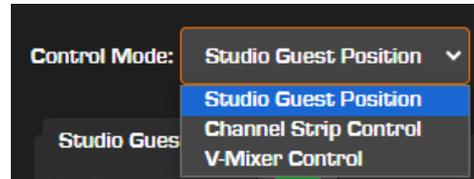
From the **Control Mode** drop-down, you can select the desired Control Mode for the module. This defines the entire user workflow for the accessory module. Three options are available:

1. Studio Guest Position (default)
2. Channel Strip Control
3. V-Mixer Control

### Studio Guest Position Mode

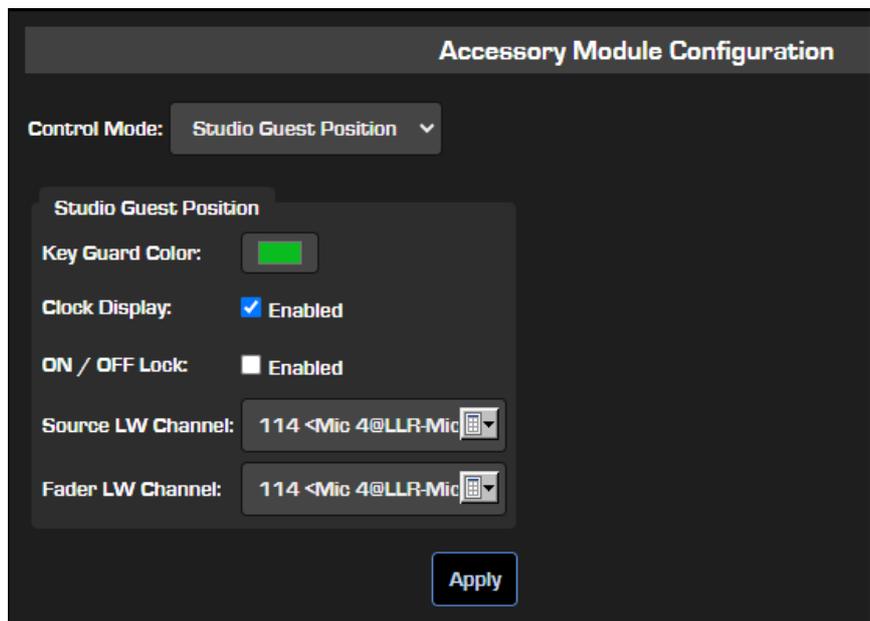
Every Accessory Module comes with this Control Mode selected by default.

For this Mode, the Web UI offers the following configuration options:



**Key Guard Color:** lets you associate a user-defined color to the LED bar located between the ON and OFF keys. Select the rectangular colored button to display a color picker.

**Clock Display:** When the “Enabled” checkbox is flagged, a miniature clock will appear on the display. The clock will disappear every time the knob is touched, or a source selection menu is activated.



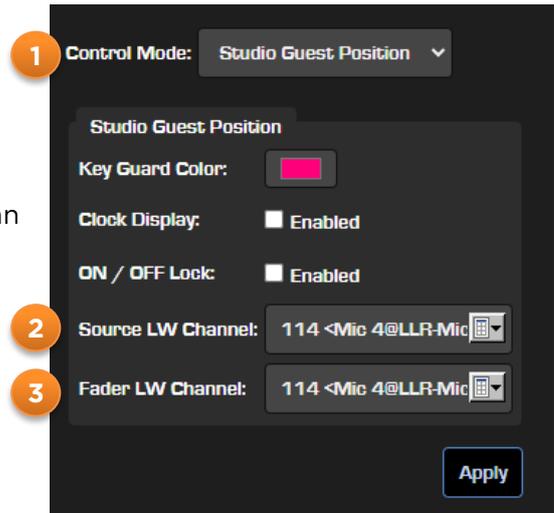
**ON/OFF Lock:** is useful for locking the ON and OFF buttons, in case the console operator wants to have exclusive control of this function, and let the Studio Talent only control his HP feed source and volume. This lock is global, and not related to any Show or Source Profiles.

**Source LW Channel:** Enter the LW Channel number of the source you want the module to control. This setting is applied to all controls except the Fader (if available on the module).

**Fader LW Channel** This field is only relevant for the MF1-ACC module (with Fader). Enter the LW Channel number you want the Fader to control. Normally it is the same as the Source LW Channel, unless you want to control two different sources from one module (which is possible). Entering a value for this parameter in the MPC-ACC module (without fader) will be accepted, but simply ignored.

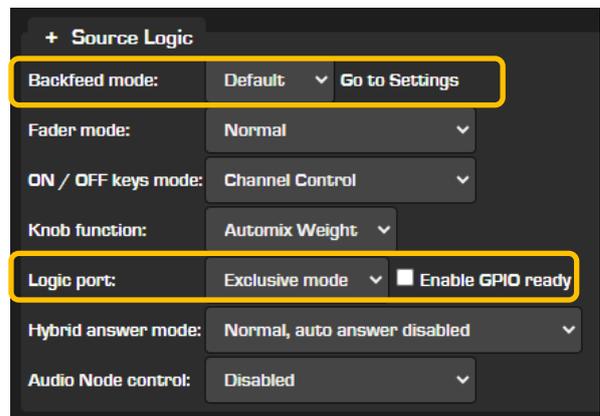
Follow these steps for configuring the Module for Studio Headphone Monitoring:

- 1- Select **Studio Guest Position** mode from the Control Mode drop-down menu.
- 2- Associate the Module to a Source by selecting its LW channel.
- 3- In case of a module with fader (MF1-ACC) you can also select a fader to control. Select the same Livewire Channel if you want complete control of your Mic channel, or a different one in case you want to control the ON-OFF, TB and MUTE functions of your mic, and the level of a different source (like an effect, or background music).



- 4- In the **Source Logic** section of the Mic Source Profile, enable the Backfeed by choosing between Default or Custom Mode in the **Feed to Source** option menu. This will enable the HP Source selector (display and top knob).

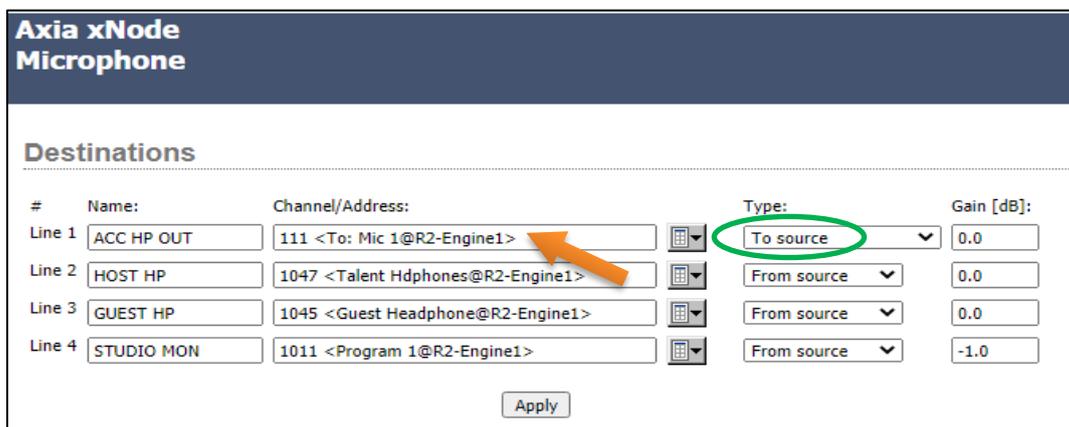
**Note** - The “Default” Backfeed mode is not available to the Operator and CR Producer Mics, Computer Player and Line source types.  
The “Custom” Backfeed mode is available to all type of sources.



- 5- Then, down below enable the **Logic Port** (usually set to Exclusive Mode) in order for the module to work in this particular Control Mode.

**Note** - If the Logic port is disabled, no controls will be available when the module operates in Studio Guest Position mode.

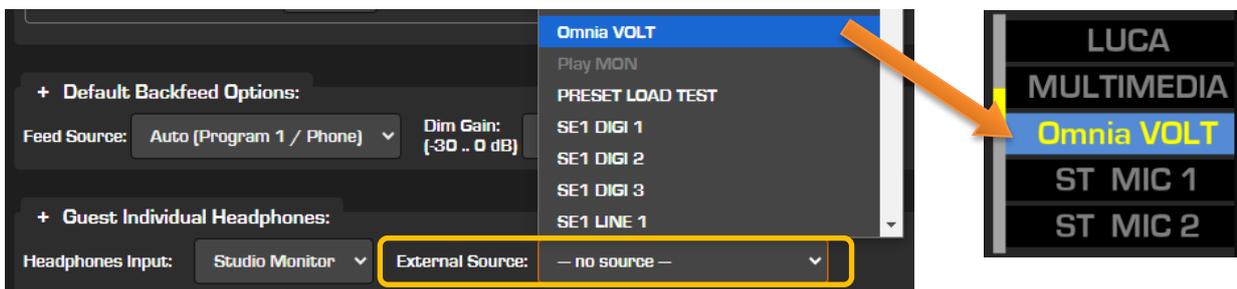
- 6- Select the desired Microphone backfeed in the Node output which will be used to monitor the HP feed. Make sure that “To Source” is selected in the Source Type menu. In case the Source is a Microphone type, you can only setup the backfeed for that Microphone in the Show Profiles **Guest Individual Headphones** section.



In this case, the Accessory module, if associated to this source, will let you control the Headphones Input and Headphones Volume parameters.



The **External Source** drop-down menu contains all those sources that were configured as “External” by selecting the corresponding flag in the Source Availability section. They correspond to those available on the module when the **EXT** key is pressed.



In case the Source is a Codec or Phone type you have the choice of disabling, enabling, or customizing the return feed (backfeed) associated with it. You can do so from the Source Profile **Feed to Source** option menu.

For more details and usage examples about this particular Control Mode, please refer to Operation Chapter 4 of this manual - [Working in Studio Guest Position Mode](#).

## Channel Strip Control Mode

From the **Control Mode** drop-down menu, choose the **Channel Strip Control** option.

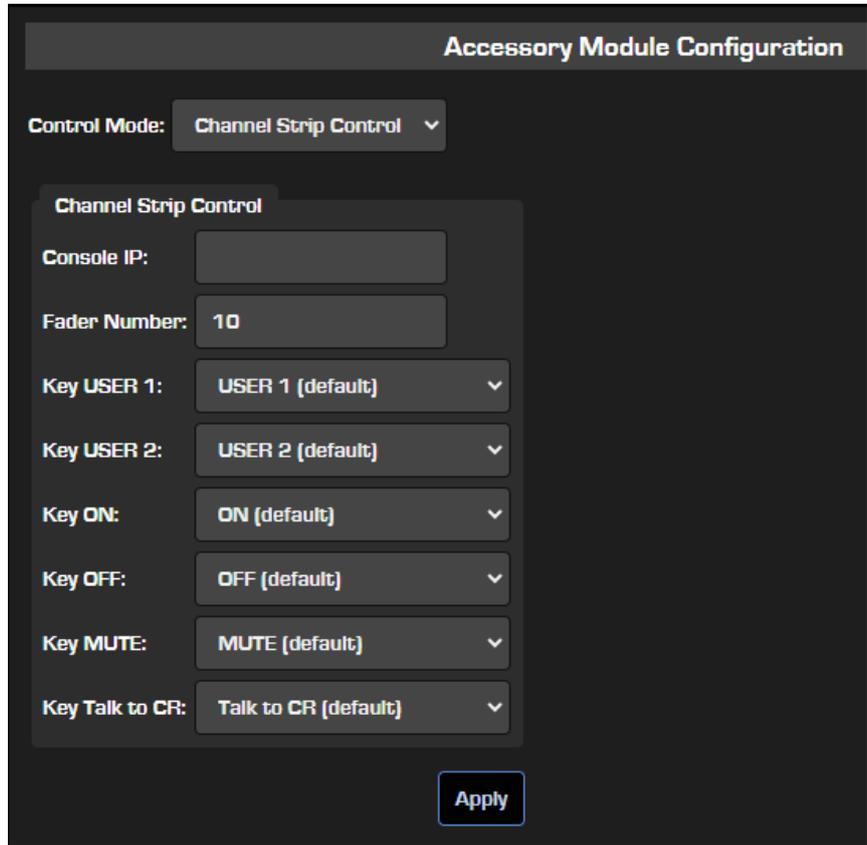
For this Mode, the Web UI offers the following configuration options:



**Console IP:** Enter in this field the IP Address of the console MTS-MON module, that you wish to connect to. It is also possible to change the connected Master from the desired MTS-MON module web UI, navigating to the **XR/ACC Modules** menu.

**Note** - It is also possible to change this setting using an external LWRP command.

**Fader Number:** Enter in this field the number of the Fader (channel) that you want to control. It could be any of the available Engine input channels, and it doesn't matter if it is currently loaded on the surface, or not.

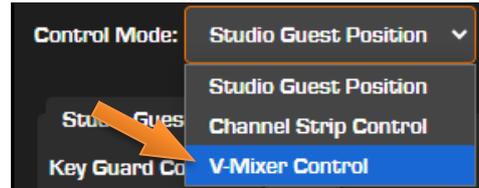


Following, are a series of **KeyMap** options, which can be configured only in this particular operation mode, and let you associate a different function to each key, from the default. The options available are the same as those found in the Fader Modules' KeyMap menu. These settings apply to both MPC-ACC and MF1-ACC accessory modules, but only when they are operating in Channel Strip Control Mode.

For more details and usage examples about this particular Control Mode, please refer to Operation Chapter 4 of this manual - [Working in Channel Strip Control Mode](#).

## V-Mixer Control Mode

from the Control Mode drop-down, select the **V-Mixer Control** option.

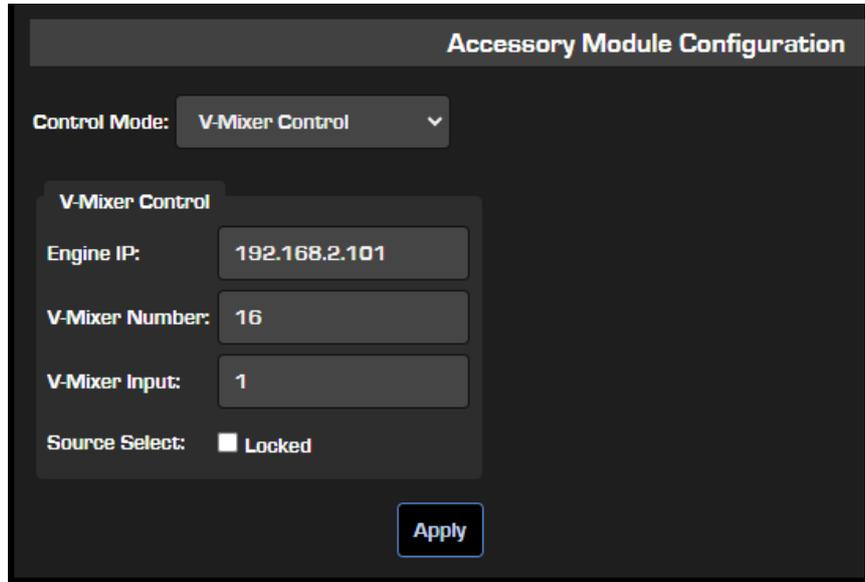


The UI will now display the relevant settings for this mode:

In the **Engine IP** field, you must configure the IP address of the Engine which hosts the VMIX you would like to control.

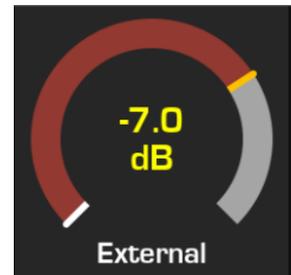
In the **V-Mixer Number** field, you must type the number of the VMIX instance you wish to control

In the **V-Mixer Input** field, you must type the Input Channel number of the VMIX instance you wish to control.



- The input level for this channel will be controlled by the Accessory Module's Top Encoder. The display will show the selected source and its level.
- The Module's ON-OFF keys will control the Input channel ON-OFF function.

Flag the **Source Select** option if you want to lock the Source selector, normally activated by pressing the top encoder, and avoid users changing the Input Source.

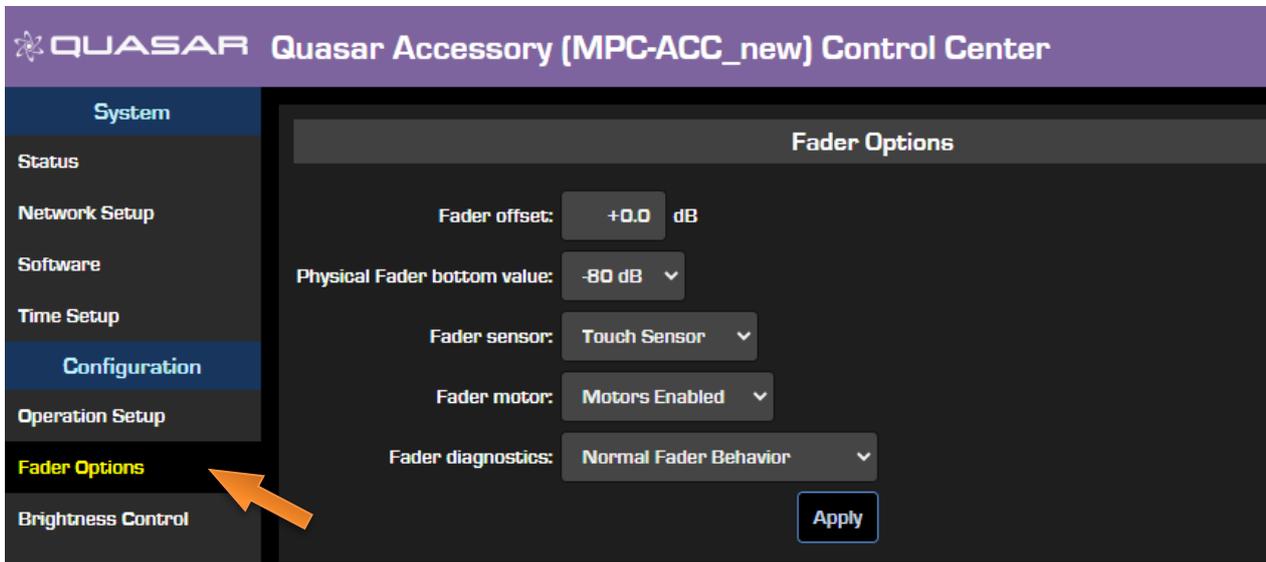


In case you are using an Accessory Module with Motorized Fader (MF1-ACC Module) the Fader will automatically control the VMIX Sub OUT.

For more details and usage examples about this particular Control Mode, please refer to Operation Chapter 4 of this manual - [Working in VMIX Control Mode](#).

## Fader Options

The settings found in this Menu apply only to the Accessory Module with Motorized Fader (MF1-ACC) when it is operating in any of its three Control Modes:



### Fader offset

Shifts the faders' scale and level setting.

Example 1: a setting of -10 will shift the 0dB mark (also known as "Unity Gain" point) up to the top of the fader's travel. This is useful for those operators who like working with faders all the way up.

Example 2: a setting of +6 will increase the "sensitivity" of your faders by 6 dB. The faders in this example would have to be run 6 dB lower in order for the PGM audio output to be the same as the -0dB default setting.

**Note** - Your meters will always display the actual audio levels, no matter what the offset is.

### Physical Fader bottom Value

This function lets you set the value in dB to the bottom of the fader's scale. 5 settings are available: -80dB, -75dB, -70dB, -65dB, -60dB . This value represents the transition point between the maximum fader attenuation and the -infinite.

For example, a setting of -60dB will reduce the fader gain range from +10/-80dB to +10/-60 and increase the -infinite range.

**Note** - In case the Fader Start function is used, changing the Fader Bottom Value will also move the fader-start point, since this function is triggered when the transition point is reached.

### Fader Sensor

This menu lets you choose between two different ways of activating a fader, using the built-in Touch Sensors or the Motion Sensor:

- **Touch Sensor** is a more advanced control technology, which offers protection against unwanted movement of the fader: if a non-human object (like a newspaper, or a mobile

phone) ends up hitting and moving a fader which is on the air, the fader will not change gain and its motor will try to return it to its original position. The fader will be enabled only in case it is touched by a finger, allowing the operator to move it to the desired position.

**Warning** - In case the operator has long fingernails and pushes the fader with the tip of his/her nails, the fader will fight back thinking it is being moved accidentally. This is because nails are non-conductive objects similar to plastic.

- **Motion Sensor** is a traditional control technology, which offers no protection against unwanted movement of the fader. In this case the fader will always work, no matter what is moving it, and it will behave like the fader in an analog console.

### Fader Motor

This menu allows you to disable the Fader’s motor. In this case, if the fader is controlled by another fader (like a Quasar Soft fader, or a different fader controlling the same input channel) the only way to tell if the fader’s logical position has changed and is no longer matching the fader’s physical position, will be to check the fader position against the dB level shown in the small display

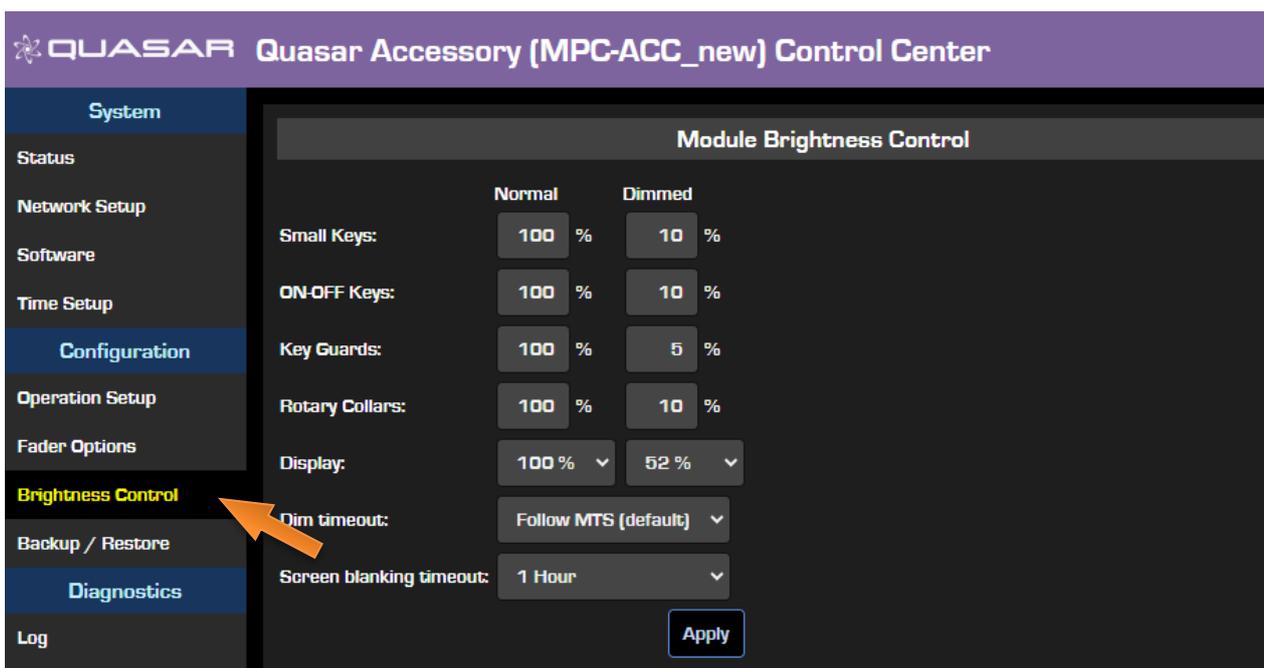
### Fader Diagnostics

For normal operation you must leave this option set to its default value:

**Normal Fader Behavior.** You should set it to **Fader Movement Simulation** only in case you are diagnosing some problems with the fader’s motor, and Telos Support asked you to enable this option.

## Brightness Control

This page is for setting the brightness of each group of hardware controls found in the Accessory Module. Controls in this section are equivalent to those found in the MTS-MON and XR-4FAD modules.



Two settings are available for controlling the **Normal** and **Dimmed** state of each group. Enter the desired value, directly in each of the field and click **Apply** .

The **Display** drop-down menu allows setting of the four Channel Displays' brightness. In order to select a different value, pick one of the preset levels from the drop-down menu and press **Apply** .

The **Dim Timeout** is the time it takes to transition from the Normal to the Dimmed state.

The Dimming Timeout in the fader XR Module can be set to **Follow the MTS Module** timeout, so that all XR modules will dim when the MTS Module enters the dimming state, and will return to normal state when the MTS module is touched.

**Tip** - The Brightness scale is not linear, so we suggest you to try different settings, like 30%, 50%, and 70% to work out what's the best brightness level for your environment.

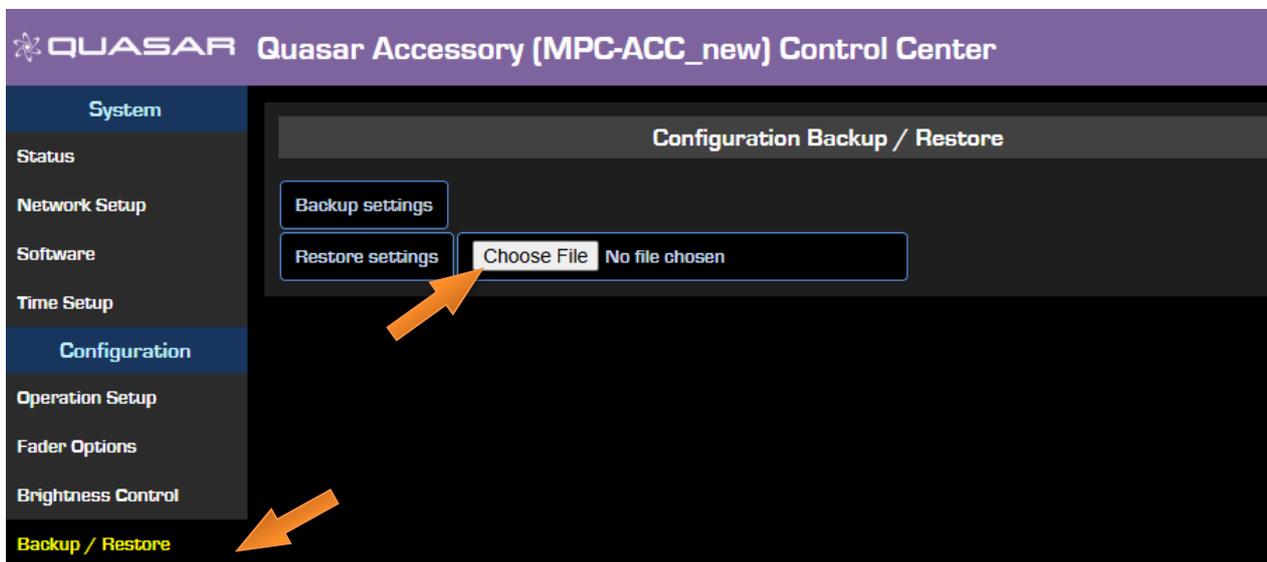
## Backup / Restore

From this page, it is possible to download and save to your PC a backup of the selected Fader Module configuration, in the form of an .xml file.

### Saving a Backup

Select **Backup Settings** if you want to create and download a Backup File:

A dialog will appear, to let you save the file, in XML format. The file name will be automatically generated, based on the MTS module's IP address, Date and Time of the day.



### Restoring a Backup

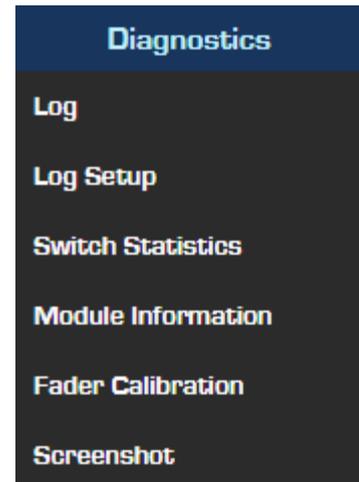
Select **Choose File** , choose an existing backup file, then press **Restore settings** , if you want to upload a previously created backup from your archive.

## Diagnostics Menu

Access to the Accessory Module Diagnostics Menu is never required during Console Operation, or System Configuration. Generally, this section is used to retrieve specific information about the System when asked by Telos Support, such as Logs, Network Ports activity, Connected Modules and Active Firmware version.

The description of the **Log**, **Log Setup**, **Switch Statistics**, and **Module Information** pages found in this menu is equivalent to that of the MTS-MON Module, given in the previous paragraph.

The description of the **Fader Calibration** and **Screenshot** page found in this menu is equivalent to that given in the [XR-4FAD Module Diagnostics Menu](#).



# Configuration 2 – Profiles

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## Show and Source Profiles

The Axia terminology for the console preset configurations is “Profiles”. There are two types of Profiles:

1. **Source Profiles**, which define the different audio sources and how they function within the console system. These are essentially preset configurations for an Input Source.
2. **Show Profiles**, which define what sources are placed on which console input channels, which parameters are stored within each channel strip, and how the console’s Monitor section is configured. These are “scene presets” or “snapshots”, of the entire console.

This chapter begins with a basic description of how a Source Profile is created, then reviews some specific Profiles configuration examples, and then moves onto Show Profiles, offering all the essential the knowledge needed to configure the Quasar surface.

## Source Profiles

A very important aspect of the console configuration, is defining the sources that will be used on the console.

The fundamental difference between an Axia console (not just Quasar, but all our models) and its competitors, is that we created a *set of logic attributes* that can be associated to any audio source, which describes how that source behaves when it is loaded onto the console. This “custom logic layer” can be manipulated directly by the end user via the console built-in Web UI (no configuration software required), and once associated to the source, it is carried on with it across the entire network.

Quasar eliminates the complexity associated to this process by automatically merging audio, logic and program data into a single, routable information stream. Since audio in an AoIP network is transported as packetized data, it’s easy for us to associate other data along with audio.

This is accomplished through the use of Source Profiles. Source Profiles allow you to specify which network audio sources are to be used on each console in your facility. They also include Input processing, Panpot, EQ and Dynamics settings. Even mix-minus source selections, can all be set in the Source Profile and are then automatically loaded whenever that source is assigned to a console fader strip.

Along with declaring the audio source (giving it a user-friendly name), we also must define how loading the source to a fader will modify the operation of the console, and how the source in turn is modified by user interaction. Once the console has its sources configured, we can go one additional step and define channel layout and monitor settings.

To get started, connect a computer to the Quasar and access its configuration Web UI by entering the MTS Module IP address into your Web browser. If this is the first time you access the MTS UI, you will be prompted for entering user access credentials. Default Username is: *user*. No password is needed. Leave this field empty and press Enter.

Sign in

http://192.168.2.120

Your connection to this site is not private

Username

Password

Select the **Sources** link, as shown here:

**QUASAR Quasar (STUDIO1-LLR) Control Center**

**System**

- Status
- Network Setup
- Software
- Time Setup
- License Management
- Quasar Soft
- Quasar Cast

**Configuration**

- XR / ACC Modules
- SR Modules
- SmartKeys
- Engine
- Customize
- Hot Keys
- Intercom Control
- Brightness Control
- Backup / Restore

**Profiles**

- Presets
- Sources** (highlighted with orange arrow)
- Shows

**Source Profiles**

Profile Name	Type	Source
<input type="checkbox"/> AES Analyzer	Line Input	215 <Digital 3@Edge-IO_1>
<input type="checkbox"/> AES-67 Source	Line Input	239.192.0.110
<input type="checkbox"/> Blu-Ray	Line Input	214 <BDP AES@Edge-IO_1>
<input type="checkbox"/> CART 1	Computer Player	1051 <VMIX 1 Sub@LLR-QE-1>
<input type="checkbox"/> CART 2	Computer Player	1052 <VMIX 2 Sub@LLR-QE-1>
<input type="checkbox"/> CART 3	Computer Player	1053 <VMIX 3 Sub@LLR-QE-1>
<input type="checkbox"/> CART 4	Computer Player	1054 <VMIX 4 Sub@LLR-QE-1>
<input type="checkbox"/> CD Player	Line Input	213 <Sony CD@Edge-IO_1>
<input type="checkbox"/> CR MIC	Operator Microphone	310 <DESK TB MIC@LLR-MixNode>
<input type="checkbox"/> Dynamite	Line Input	313 <DYNAMITE Ret@LLR-MixNode>
<input type="checkbox"/> EXT MIC	External Microphone	3101
<input type="checkbox"/> FM/DAB Tuner	Line Input	209 <DAB/FM Tuner@Edge-IO_1>
<input type="checkbox"/> Guitar	Line Input	311 <JMP-1 Master@LLR-MixNode>
<input type="checkbox"/> Lexicon MPX1	Line Input	314 <MPX1 Ret@LLR-MixNode>
<input type="checkbox"/> LUCA	Operator Microphone	3050
<input type="checkbox"/> MP-16 MIC	External Microphone	711
<input type="checkbox"/> MP-16 OUT	Line Input	712
<input type="checkbox"/> MULTIMEDIA	Line Input	1056 <VMIX 6 Sub@LLR-QE-1>
<input type="checkbox"/> Omnia VOLT	Line Input	811 <VOLT Out@Omnia-VOLT>
<input type="checkbox"/> PC_Play 1	Computer Player	611
<input type="checkbox"/> PC_Play 2	Computer Player	612
<input type="checkbox"/> PC_Play 3	Computer Player	613
<input type="checkbox"/> PC_Play 4	Computer Player	614
<input type="checkbox"/> PGM 1!	Line Input	1011 <Program 1@LLR-QE-1>
<input type="checkbox"/> PGM 2!	Line Input	1012 <Program 2@LLR-QE-1>
<input type="checkbox"/> Play MON	Computer Player	615
<input type="checkbox"/> PRESET LOAD TEST	Line Input	311 <JMP-1 Master@LLR-MixNode>
<input type="checkbox"/> SE1 DIGI 1	Line Input	213 <Sony CD@Edge-IO_1>
<input type="checkbox"/> SE1 DIGI 2	Line Input	214 <BDP AES@Edge-IO_1>
<input type="checkbox"/> SE1 DIGI 3	Line Input	215 <Digital 3@Edge-IO_1>
<input type="checkbox"/> SE1 LINE 1	Line Input	205 <BD FL+FR@Edge-IO_1>
<input type="checkbox"/> SE1 LINE 2	Line Input	206 <BD SL+SR@Edge-IO_1>
<input type="checkbox"/> SE1 LINE 3	Line Input	207 <BD C+SUB@Edge-IO_1>
<input type="checkbox"/> SE1 LINE 4	Line Input	208 <BD SBL+SBR@Edge-IO_1>
<input type="checkbox"/> SE1 LINE 5	Line Input	209 <DAB/FM Tuner@Edge-IO_1>
<input type="checkbox"/> SE1 LINE 6	Line Input	210 <SE1_Line 6@Edge-IO_1>
<input type="checkbox"/> SE1 LINE 7	Line Input	211 <SE1_Line 7@Edge-IO_1>

In this screen, you'll be able to see any sources that have already been configured, as well as to create a new Source by clicking on the **Create new source profile** button at the top of the page.

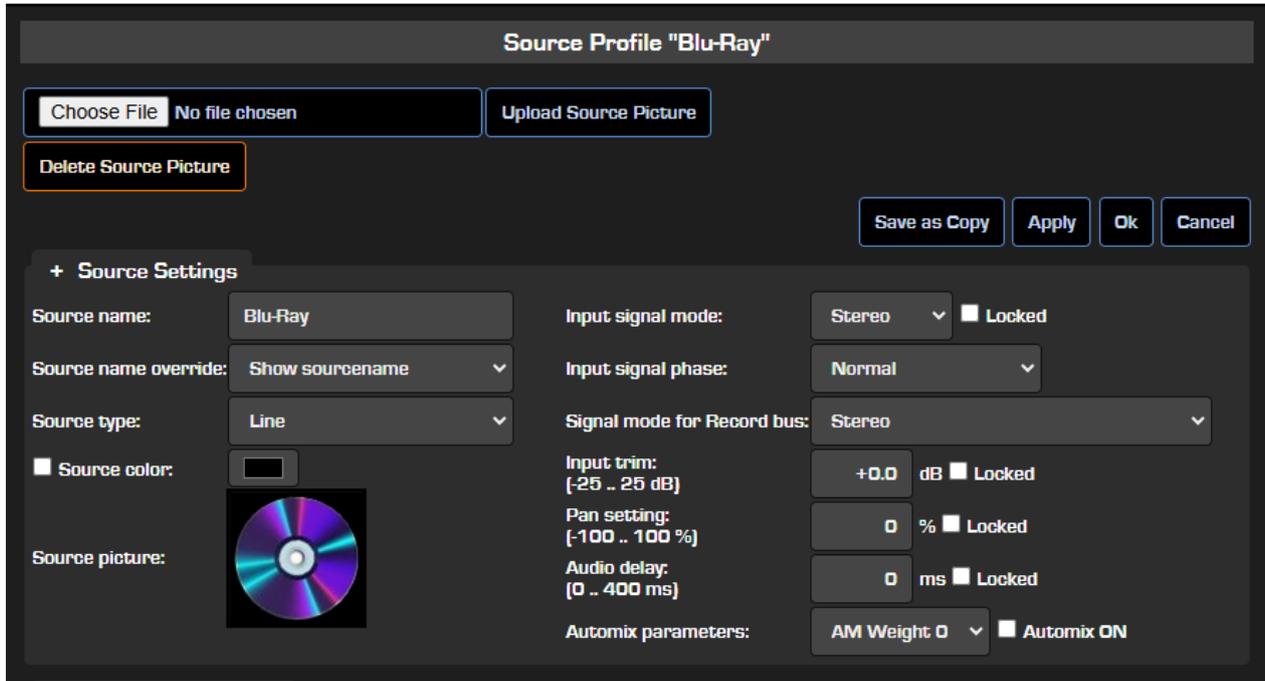
In order to delete Source Profiles, select any of the checkboxes to the left, and press the **Delete Selected Sources** button located at the bottom of the page.

Selecting **Create new source profile**, or opening an existing source from the list, displays a configuration page.

## Source Settings

In v3.0 we added Direct Outs (Insert Sends) and Insert Returns to every input channel in the console, as well as full support for AES67 stream configuration in source profiles.

This required a complete redesign of the source profile page: a more compact layout was adopted to minimize scrolling when navigating, and settings were grouped into different sections that can now be collapsed or expanded for easier navigation.



**Note** - While the options visible in the left part of this section don't change with the Source Type, the options visible in the right side will change according to the type of Source which is selected, with more or less drop-down menus becoming available.

### Source Picture

It is possible to associate a picture to the Source configuration.

If a new Source has just been created, click **Apply** : the page will refresh and display a new field where you can **Choose a File** , select a picture from your own library and press the **Upload Source Picture** button to upload it. The picture will show up on the console channel strip displays when the source is loaded. The picture must be .PNG or .JPG type, 192x192 pixels max. resolution, and 8kB max. file size.

If, after choosing the file and pressing the **Upload** button, a preview of the picture does not appear in the UI, then your picture is not the correct format.

### Source Name

Here you can type a 8-character descriptive name for your source. This is the description that will appear on the Console's Channel Display , above the fader. Names that should not fit in one line of the channel display, will be correctly reported on the Touchscreen UI's SOURCES tab, but will be truncated, followed by the "... " text.

## Source Name Override

Every Livewire Source can have two descriptors: A primary Source Name and a Label. If the source you're defining can supply Program Associated Data (for instance, a computer playout system that sends song/artist information), you can choose to have that text dynamically replaced in the console fader strip displays.

From the xNode Web UI, in the SOURCES tab, you can only assign the Primary Source Name. The Label instead, can only be assigned from Pathfinder, or from a software that supports dynamic source labelling over a serial connection. So, a playout system could change the Source labels, if required.



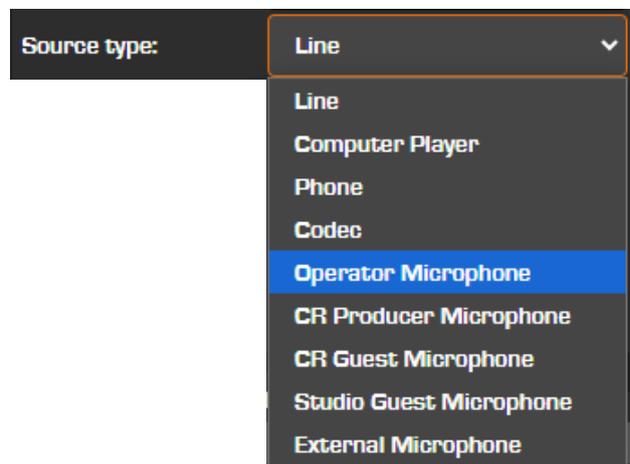
We use our LWRP protocol for communicating with the console, in this case.

- Selecting **Show sourcename** will display the name specified in the **Source name:** field onto the input channels' displays when the Source is loaded.
- Selecting **Allow application to override** will override the Source name and display the LABEL information sent by the external system in the Source name field.
- Selecting **Show both** will display both the Source name and the Label on two lines of the input channels' displays.

**Note** – Please refer to Operation Chapter 2 of this manual: [The Channel Strip Display GUI](#) for more information on the priority assigned to the various messages that can appear on the displays.

## Source Type

From this drop-down menu you can select the type of source, choosing between nine different options:



- Line
- Computer Player
- Phone
- Codec
- Operator Microphone
- CR Producer Microphone
- CR Guest Microphone
- Studio Guest Microphone
- External Microphone

The table in the next page shows which options are available to each of the Source Types:

Source Type	Input Mode	Input Phase	Rec Mode	Input Trim	Pan	Delay	Logic Port	GPIO Ready	Knob Funct.	Hybrid Answer
Line	•	•	•	•	•	•	•	•	•	
Computer Player	•	•	•	•	•	•	•	•	•	
Phone	•	•	•	•	•	•	•	•	•	•
Codec	•	•	•	•	•	•	•		•	
Operator Microphone		•	•	•	•	•	•		•	
CR Producer Microphone		•	•	•	•	•	•		•	
CR Guest Microphone		•	•	•	•	•	•		•	
Studio Guest Microphone		•	•	•	•	•	•		•	
External Microphone		•	•	•	•	•	•		•	

### Source Color

This option lets you assign a color directly to the Source: this will override the Source Type Color Coding configured in the Customize page of the Configuration Menu.

### Input Signal Mode

The Input Mode drop-down menu is not available when a Microphone source type is selected. It defines the mono/stereo setting for the Source. Four options are available:

- **Stereo:** feeds incoming L/R signal to left and right channels of the assigned bus.
- **Left:** feeds incoming left channel to both channels of the assigned bus.
- **Right:** feeds incoming right channel to both channels of the assigned bus.
- **Sum:** creates L+R mono mix of incoming stereo source and feeds it to both channels of the assigned bus.

Select the **Locked** checkbox to prevent the user from changing the source's signal mode (as set above) from the Console surface. This lock applies to the Console UI (the corresponding control is grayed out) and to external LWCP controllers (like PF).

### Input Signal Phase

Defines the Phase setting for the Source. Four options are available:

- **Normal:** No change is applied to the input signal phase when the Source is loaded.
- **Invert Left:** Reverses the phase of the left leg only, of the input channel.
- **Invert Right:** Reverses the phase of the right leg only, of the input channel.
- **Invert Left and Right:** Reverses the phase of both input channel's legs.

**Note** – The “Invert Right” and “Invert Left and Right” options will apply only if a Stereo Source is loaded to the channel. Remember that Microphone sources are considered mono and therefore, only the “Normal” and “Invert Left” options will be applied. The other two will be ignored, and not applied.

### Signal mode for Record bus

This option allows you to define how this source is sent to the **PGM-4/Record** bus. **Record** is a special pre-on/off, post-fader bus with a special output for a dedicated recording device.

- **Stereo:** sends both sides of the source to both sides of the recording device, as normal.
- **If Phone or Codec Left, otherwise Right:** sends the source to the recorder’s left channel, ONLY if the Source Type is defined as **Phone** or **Codec**. Any other Source Type will be fed to the recorder’s right channel.
- **Summed-Mono to Left:** sums the left and right sides of the audio source and sends the summed signal to the recorder’s left channel.
- **Summed-Mono to Right:** Same as above, but sends summed audio to the recorder’s right channel.

### Input Trim

This field lets you specify a fixed trim level that is applied to the signal after the channel Input, and just before it goes to the fader. You may specify any amount from -25.0 to +25.0 dB.

Select the **Locked** checkbox to prevent operators from making changes to the above trim gain setting, from the Console surface. This lock applies to the Console UI (the corresponding control is grayed out) and to external LWCP controllers (like PF).

### Pan setting

Lets you predefine Panpot settings for this Source. Pan is variable in % steps, with center being 0%. You may specify any amount from 100% Left to 100% Right.

Select the **Locked** checkbox to prevent the user from changing the source’s Panpot from the Console surface. This lock applies to the Console UI (the corresponding control is grayed out) and to external LWCP controllers (like PF).

### Audio Delay

Lets you add a delay to the source, at the moment it is loaded onto the input channel. The delay stage is in the Quasar Engine. You may specify any amount from 0 to 400ms. Only integer values are accepted.

Select the **Locked** checkbox to prevent the user from changing the source’s Delay from the Console surface. This lock applies to the Console UI (the corresponding control is grayed out) and to external LWCP controllers (like PF).

### Automix Parameters

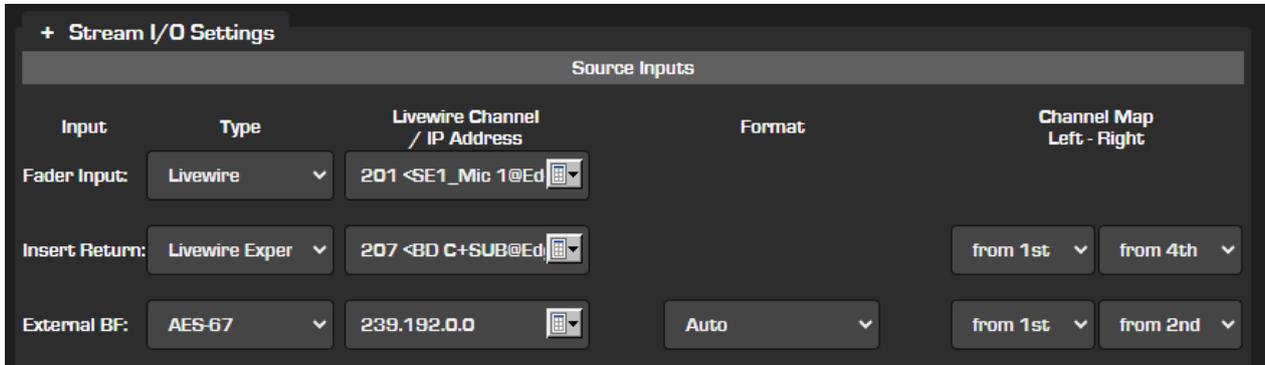
In v3.0 we moved the main settings for channel Automix to Source Profiles, and added the possibility to retain or override those settings in Shows.

This option lets you configure a Source with Automix ON/OFF and Weight settings: these can be retained or overridden by the settings saved per-Channel, in the [Show Profile Main Page](#).

## Stream I/O Settings

This new section, added in v3.0, will let you define each Source Input and Output as **Livewire**, **Livewire Expert**, or **AES67**, and enter the appropriate settings for the corresponding type.

The available options for the **Source Inputs** are:



The screenshot shows the 'Stream I/O Settings' window with a 'Source Inputs' section. It contains a table with columns: Input, Type, Livewire Channel / IP Address, Format, and Channel Map Left - Right. The table has three rows: 'Fader Input' with Type 'Livewire' and Channel '201 <SE1\_Mic 1@Ed'; 'Insert Return' with Type 'Livewire Exper' and Channel '207 <BD C+SUB@Ed'; and 'External BF' with Type 'AES-67' and IP '239.192.0.0'. The 'Format' column has a dropdown set to 'Auto'. The 'Channel Map' column has dropdowns for 'from 1st' and 'from 4th' for the first two rows, and 'from 1st' and 'from 2nd' for the third row.

### Fader Input

Here you can define the Primary Source input. Select **Type** from the drop-down menu, then, in the Livewire **Channel / IP Address** field:

- Enter the desired **LW Channel** by typing the source channel ID or its multicast address
- OR pick the desired Livewire channel from your Livewire Network using the drop-down Source Selector.
- OR enter an AES67 Multicast address by typing its IP in the box.

You can type “0” (zero) if you want to clear the stream for this Source.

### Insert Return

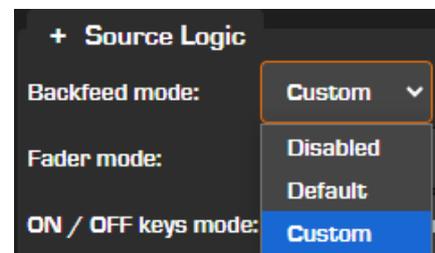
Select the appropriate Source **Type** from the drop-down menu.

Then, in the **Livewire Channel / IP Address** field, enter settings as described above.

This option lets you define only the AoIP settings for the Insert Return. Move to the **Direct Out/ Insert Return** section of the Source Profile for configuring Enable/Disable, Position and Locking options.

### External BF

The external Backfeed input is enabled only in case Custom Backfeed mode is selected in the Backfeed Mode drop-down, described in the next section: [Source Logic](#). In this case, its settings are the same as above.

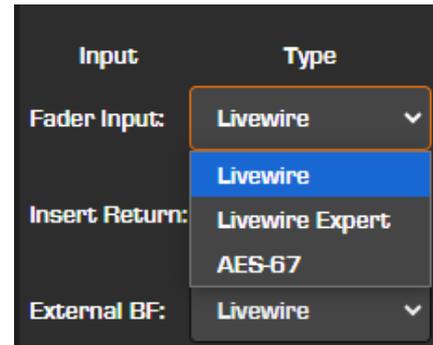


The screenshot shows the 'Source Logic' settings. It has three rows: 'Backfeed mode:' with a dropdown set to 'Custom'; 'Fader mode:' with a dropdown set to 'Disabled'; and 'ON / OFF keys mode:' with a dropdown set to 'Custom'.

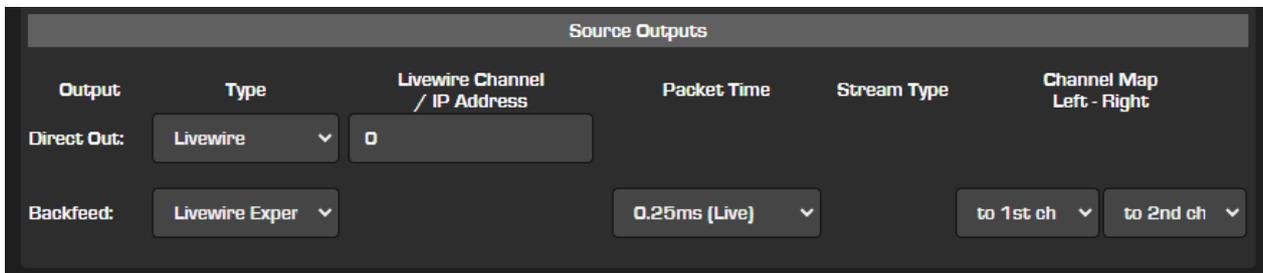
### Source Input Types

For each of the three Source Inputs available per channel - Fader Input, Insert Return, and External Backfeed - the following Input Types are available:

- **Livewire:** is the standard setting: in case a Livewire Surround stream (5.1+Stereo bundle) is selected, the Stereo portion of the stream (Ch 7 and 8) will be routed to the console channel Left and Right inputs
- **Livewire Expert:** adds the possibility to select which audio signal, from an incoming Livewire Surround stream, should be mapped to the console channel Left and Right inputs. If no changes are made to the default **Channel Map** settings, the UI will switch back to standard Livewire Type upon pressing the **Apply** button.
- **AES67:** lets you configure the input stream with AES67 parameters.



The available options for the **Source Outputs** are:



### Direct Out

Select the appropriate Source **Type** from the drop-down menu. Then, in the Livewire **Channel / IP Address field**:

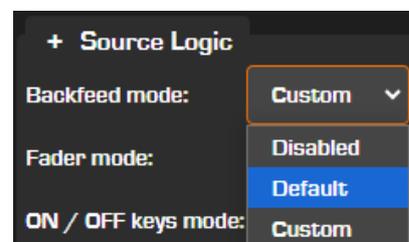
- Enter the desired **LW Channel** by typing the source channel ID or its multicast address. A new Source will be generated by the console engine
- OR enter an AES67 Multicast address by typing its IP in the box.

You can type “0” (zero) if you want to clear the stream for this Source.

This option lets you define only the AoIP settings for the Direct out. Move to the **Direct Out/ Insert Return** section of the Source Profile for configuring Source mode and Locking options.

### Backfeed

The Backfeed output is enabled only in case Default or Custom Backfeed modes are selected in the Backfeed Mode drop-down, described in the next section: [Source Logic](#). In this case, its settings are the same as above, with the exception that if Livewire Expert type is selected, the IP address will be automatically generated and no field will appear to enter this setting.



### Source Output Types

For each of the two Source Outputs available per channel - Direct Out and Backfeed - the following Output Types are available:

- **Livewire:** is the standard setting: enter the desired Ch# in this field. This will generate a new LW Source
- **Livewire Expert:** adds the possibility to select a different Packet Time (from that of the input Source), Stream Type (*From Source* or *To Source*) and multichannel output mapping. If no changes are made to the default settings, the UI will switch back to Livewire Type upon pressing the **Apply** button.
- **AES67:** lets you configure the same parameters as Livewire Expert (packet Time, Stream Type and Channel Map), and expects an AES67 multicast IP address in the Livewire Channel/IP Address field. Note that the Stream Type field in case the output is configured as AES67, will allow the selection of the payload type.

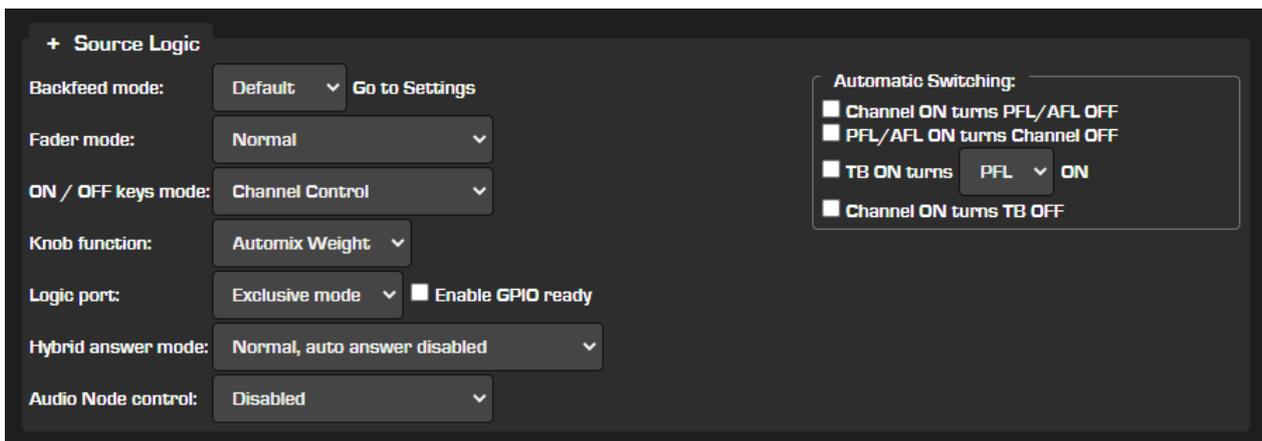


**IMPORTANT:** the following automatic logic is applied to the Backfeed Type selector depending on the combination of Input and Output Types selected:

1. If LW is selected for the Fader Input Type, it will not be possible to configure the corresponding Backfeed as AES67. The UI will reject this setting combination and will configure the backfeed as Livewire Expert.
2. If AES67 is selected for the Fader Input Type, it will not be possible to configure the corresponding Backfeed as LW. The UI will reject this setting combination and will disable the backfeed because no IP address for it could be automatically derived from the input address. You will need to manually enter both the Input and Output IPs in case you want to configure a channel with AES67 input and backfeed.

## Source Logic

In v3.0 the **Source Logic** section of the Source Profile UI was also updated.



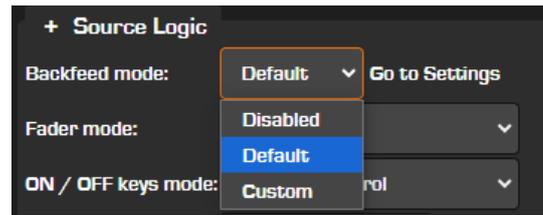
The options available are:

### Backfeed Mode

This option was renamed from v2.0, where it was called “Feed to Source”. It was also moved to from the original Backfeed options section up to the Source Logic section to minimize vertical scrolling when checking a Source Profile configuration.

- **Disabled:** removes completely the backfeed from the channel

- **Default:** is only available to Phone or Codec type Sources.
- **Custom:** will activate the Custom Backfeed options for this source, described later on in this chapter. A convenient *Go to Settings* link was added next to the drop-down menu, to let you skip quickly all the way down to the Backfeed Options section.



Please see the [Default and Custom Backfeed Options](#) paragraph for more details on configuring Backfeeds.

### Fader Mode

Defines fader start actions and states. Three types of Fader Operation mode are available:

- **Normal:** is the standard fader operation, which does not affect the ON-OFF keys logic. The operator must manually turn the channel on and off.
- **Fader Start:** turns the Channel ON as soon as the fader is moved up from the  $-\infty$  position. A GPIO closure can be triggered, if associated to the source.
- **Fader Start with Arm:** works just like the Normal setting when the fader is not set to minimum position. When the fader is set to the minimum position, it can be in an Armed state. The armed state is entered by pressing the **ON** key when the fader is down. This state means that the channel will go to the ON state when the fader is moved up. When in Arm state, the channel OFF lamp will be illuminated. The channel ON lamp will be flashing in “wink” mode - a long on-time and short off-time.

For all fader modes, pressing the **OFF** key at any time will set the channel to the OFF state. For complete instructions on how to associate a GPIO contact to Fader Start, please refer to Configuration Chapter 7 of this manual: [GPIO Computer Playback Device Logic](#).

### ON / OFF keys mode

Determines if the channel's ON/OFF keys will control the input channel ON/OFF function, or if they control an external device.

- Select **Channel Control** to control the channel ON/OFF function (normal operation).
- Select **External Control** to disconnect the keys from the channel, and use them to control an external device, such as playout machine. The device will be able to control the ON/OFF keys' lamps. This external control works bidirectionally via LWCP only (not via GPIO).

**Tip -** When External Control is selected, you can still turn the Channel ON or OFF manually from the console, using the ON button in the CONTROL & B/FEED TAB.

For more information about using External Control mode, please see Configuration Chapter 5 of this manual: [Setting Up the ON/OFF Keys as Remote Control](#)

### Knob Function

This drop-down menu assigns a function to the Option Knob located at the top of each channel strip. It allows you to completely disable it, or to associate some of the parameters which are normally available from the Touchscreen UI, directly to the knob, for immediate access to that function. At the time of writing this manual, the following options are available:

- None
- Input Trim
- Mic Gain
- Line Gain
- Automix Weight
- Channel Pan

### Logic Port

- **Disabled:** Allows you disable GPIO machine logic to this audio source device or enable it in two modes.
- **Exclusive mode:** permits GPIO control associated with only a single fader.
- **Shared mode:** allows more than one user to send on/off/start signals to a source via GPIO.

### Enable GPIO ready

Enabling this checkbox allows external devices such as CD players or tape machines to signal their READY state to the operator through the OFF key LED. If a device is cueing, its OFF lamp will not be lit. Tying the device’s OFF lamp to the Ready command means that a machine must be cued and ready before the OFF lamp illuminates. If enabled, the OFF lamp illuminates only when the fader is OFF and the Ready command is active on the GPIO. If disabled, the OFF lamp illuminates normally whenever the fader is turned OFF.

**Note** – This functionality is not available to Microphone type Sources.

### Hybrid Answer Mode

This setting allows you to tailor Quasar’s Hybrid Answer mode to suit your facility’s operating style. It is intended for hybrids that use external GPIO for device control:

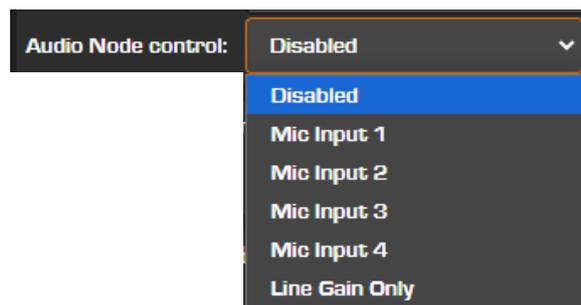
- **Normal, Auto Answer Disabled:** Quasar provides no auto-answer logic when a fader with a hybrid assigned is turned ON.
- **Channel ON Answers Hybrid:** When a hybrid has an incoming call, turning its assigned fader ON will answer the call.
- **Channel ON or Preview ON Answers Hybrid:** When a hybrid has an incoming call, turning its assigned fader ON or pressing that channel’s **PFL** key will answer the call.

**Note** – This functionality is only available to Phone type Sources. The corresponding drop-down menu will not appear on other source types.

### Audio Node Control

We’ve added the possibility to control your xNodes parameters, such as:

- Line Gain
- Microphone Preamp Gain
- +48V Phantom Power



To configure xNode control, you must select the physical node input (**MIC Input 1 to 4**) where the Source is originating from. You have the options to enable **Line Gain Only**, or to completely **Disable** this control.

**Warning** - In case the selected Source is an IP Driver, the Line Gain control will adjust the Driver's output Digital Gain. Adjusting IP Driver gain while listening, or even worse, while the Source is on the air, will produce audible artifacts. You can always lock out this function, if you want to ensure trouble-free operation.

### Automatic Switching

Determines what will happen to a source's off-air PFL assignment when that source is taken to air, by turning its channel ON:

- All options un-selected: The **PFL/AFL** bus assignment will be retained even after the fader is turned ON.
- **CHANNEL ON turns PFL/AFL OFF:** If the PFL/AFL function is active on this Source, turning its channel ON will clear its assignment.
- **PFL/AFL ON turns Channel OFF:** Does just what it says - useful for Phone channels.
- **TB ON turns PFL [or AFL] ON:** It links the TB activation only, to the PFL [or AFL] activation. After the TB is disengaged, the PFL/AFL will remain on and will need to be manually turned off.
- **Channel ON turns TB OFF:** Does just what it says - useful for Mic channels.

### Direct Out / Insert Return

This section is for configuring the activation and behavior of the Direct Out and Insert Return:

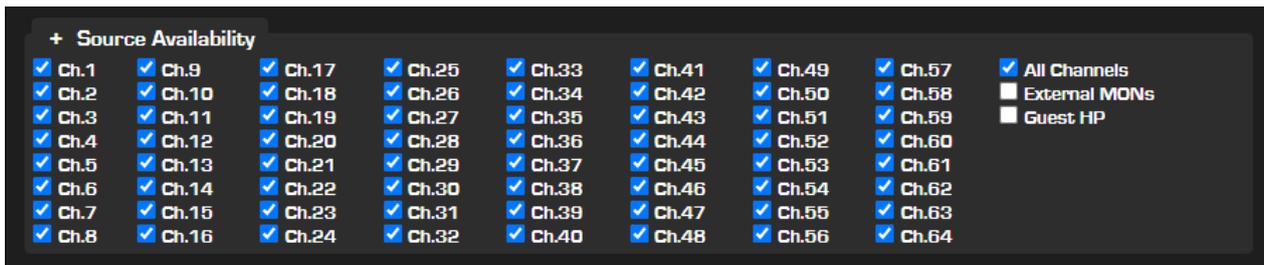
#### Direct Out Source

Determines the point in the Channel DSP processing chain, where the signal is picked up for sending it to the Direct Output:

- **Disabled:** turns the Direct Out OFF disabling its AoIP stream.
- **Pre-Processing:** turns the Direct Out ON and takes the signal right after the Channel Input & Filters, and before the Dynamics section.
- **Pre-Fader:** turns the Direct Out ON and takes the signal right after the Channel De-Esser & Limiter section, and before the Channel Mute, Fader and Pan section.
- **Post-Fader:** turns the Direct Out ON and takes the signal right after the Channel Mute, Fader and Pan section, and before the channel ON-OFF and Bus assignment section
- **Post-On:** turns the Direct Out ON and takes the signal right after the Channel ON-OFF switch.

### Source Availability

This section has checkboxes which determine which Input Channel (not fader strips) , External Monitor input, or Guest Headphone, will be allowed to load this Source. To prevent the selected Source from appearing on these inputs' selection lists, uncheck the desired box.



- **Ch.1 to 64 checkboxes:** can be selected or deselected to allow/prevent the selected source from being loaded by the desired channel
- **All Channels:** is flagged by default when a new source is created. This checkbox makes it possible to load the selected Source on all channels
- **External MONs:** this checkbox must be flagged for allowing the selected Source to be loaded onto the following external inputs:
  - External 1, 2, 3 Monitors
  - External PFL
  - External Talk to CR
  - Studio Talkback
 And appear in the Show Profiles > Monitor Section > Externals drop-down menus.
- **Guest HP:** this checkbox must be flagged for making the selected Source available to an Accessory Module, running in Studio Guest Position Mode.

For example:

1. If your source is a Control Room microphone, and you only want it to appear on channel 5, you would check only the “Ch. 5” box.
2. You might want to make a Phone Hybrid available on any fader, so you would check the box marked as “All Channels”.
3. Some sources – especially air monitors – you might wish to prevent from ever being assigned to a fader, and make them visible only to the Monitor section, so you would uncheck the boxes for all channels and just check the “External MON” box.

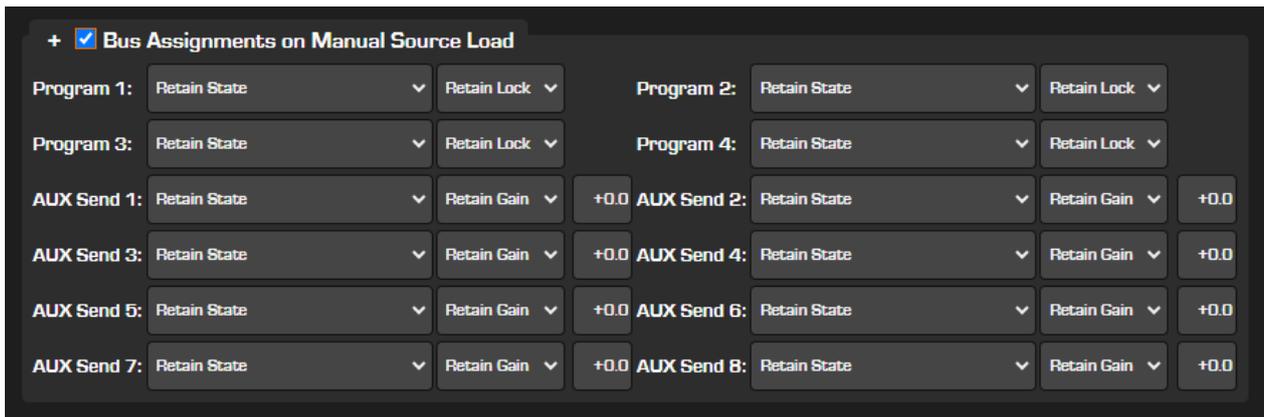
**Tip -** A quick way of removing checkmarks from all boxes is to double-click the “All Channels” checkbox.

**Tip -** Monitor Sources such as off-air receivers, or on-air processors, should have the “External MONs” checkbox flagged.

## Bus Assignment on Manual Source Load

This section covers a very important aspect: it defines the default assignment status for the four Program and eight Aux Busses, when the Source is loaded manually, to a channel. In fact, the behavior of the Source when this is loaded through a Show profile, is defined in the Show Profile settings page.

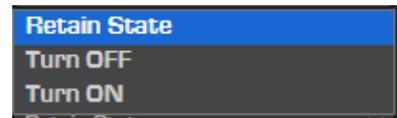
Select the checkbox if you want the assignments (and level, where applicable) to be different from defaults “Retain” settings.



### Program 1 - 4

The options available for the Program busses are:

**Retain State , Turn OFF , Turn ON.**



- The **Turn ON** and **Turn OFF** options can be used to pre-define the ON/OFF state.
- The **Retain State** option will leave the current bus status unchanged.

**Retain Lock , Unlock , Lock.**

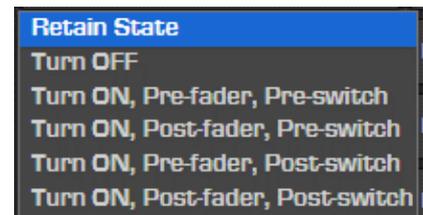
- The **Unlock** and **Lock** options can be used to pre-defined the lock state.
- The **Retain Lock** option will leave the current lock state unchanged.



### Auxiliary 1 - 8

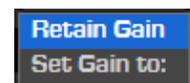
The options available for the Auxiliary busses are:

- Retain State
- Turn OFF
- Turn ON, Pre-fader, Pre-switch
- Turn ON, Post-fader, Pre-switch
- Turn ON, Pre-fader, Post-switch
- Turn ON, Post-fader, Post-switch



These options simply reflect the option available from the Channel settings UI, and allow you to pre-define any combination of Pre/Post fader, and Pre/Post ON Switch upon Source manual load.

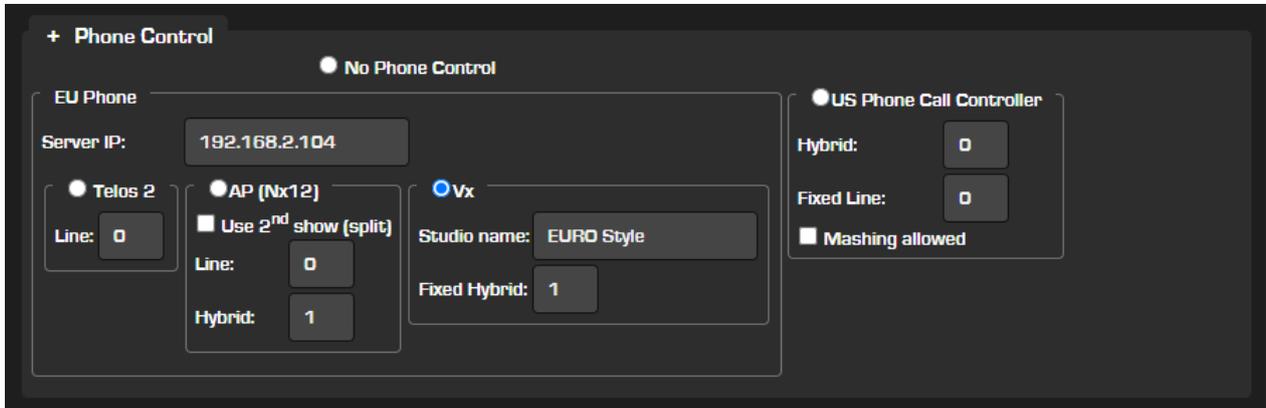
Also, you can select **Set Gain to:** from the drop-down menu to set the bus level by a fixed dB amount. Enter the desired level in the next box. You may specify any level between -60.0dB and +10.0dB. Or choose **Retain Gain** to leave the Aux Level unchanged.



Press the **Apply** button to apply the settings and continue with configuring the same Source, or **Ok** to go back to the main Source Profiles menu. Remember to select the top checkbox before you apply the new settings: leaving the checkbox empty and pressing **Apply** or **Ok** will reset all settings to the default “Retain State” and “Retain Gain” status.

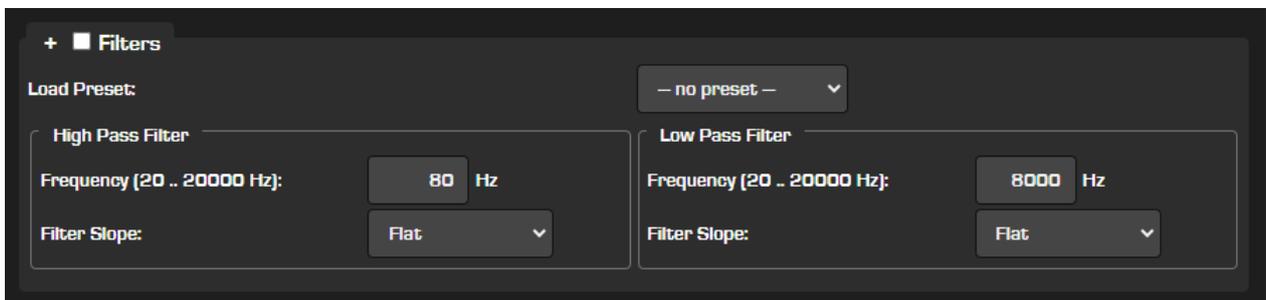
## Phone Control

This section of the Sources UI is only available to Phone type Sources. Setting up Phone Sources and their control workflow can be challenging, so we dedicated a full chapter to this topic in this manual. Please see Configuration Chapter 3 – Setting up Phones.



## Filters

This section lets you predefine the default Filters settings for a Source, when this is loaded to a channel, either manually, or through a Show Profile, and the show channel has these parameters set to “Load from Source Profile”.



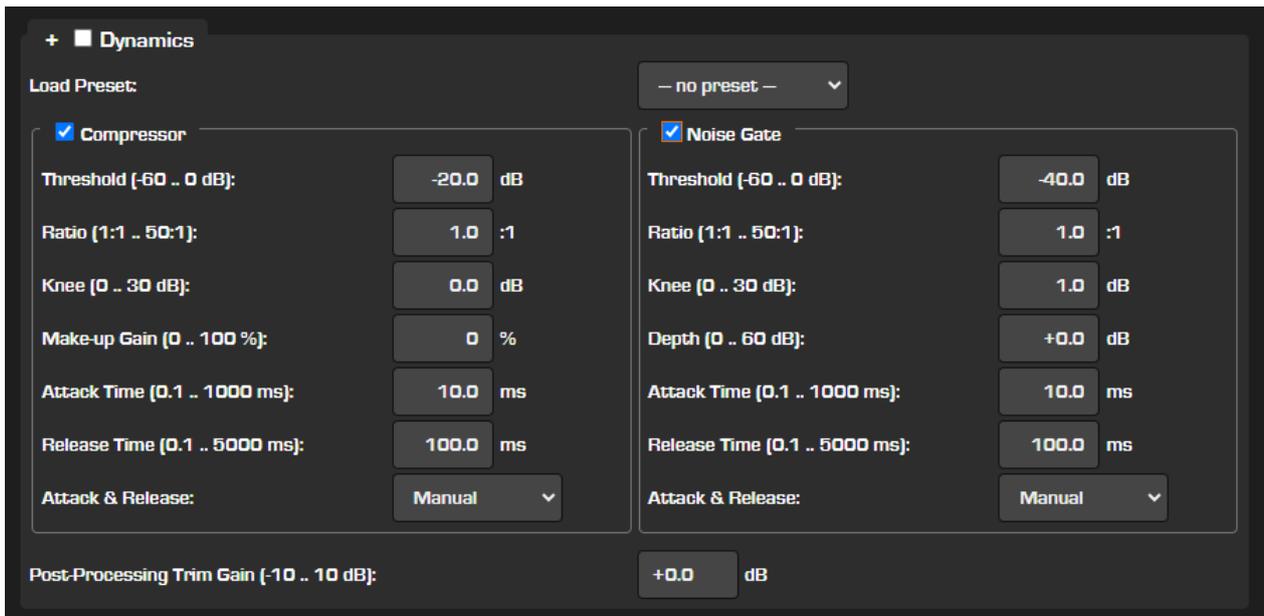
### Load Preset

This option will let you associate one of the FILTERS presets (if any were created) to the Source at the time of loading. When - no preset - is selected in the drop-down, the settings saved with the source (shown below) will be loaded.

**Note** - This function does not actually *copy the preset values into the Source*, and that's why if a Preset is chosen from the drop-down menu, the parameter fields below remain unchanged.

## Dynamics

This section lets you predefine the default Dynamics settings for a Source, when this is loaded to a channel, either manually, or through a Show Profile, and the show channel has these parameters set to “Load from Source Profile”.



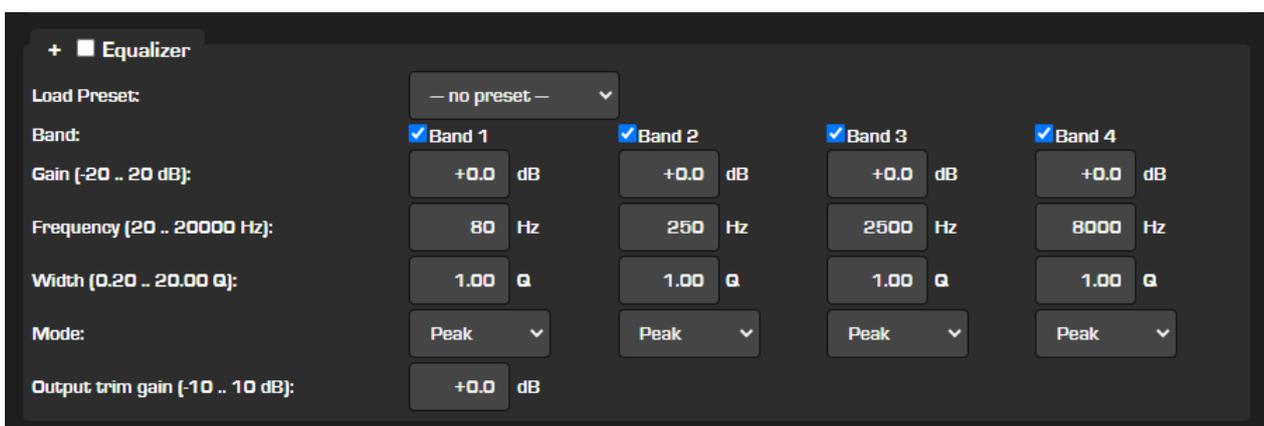
## Load Preset

As above, this option will let you associate one of the DYN presets (if any were created) to the Source at the time of loading. When “- no preset -” is selected in the drop-down, the settings saved with the source (shown below) will be loaded.

**Note** – Dynamics settings can be pre-defined in the Source Profile, for ALL type source types. You can also store these settings in the Show profile Channels, if you desire.

## Equalizer

This section lets you predefine the default EQ settings for a Source, when this is loaded to a channel, either manually, or through a Show Profile, and the show channel has these parameters set to “Load from Source Profile”.

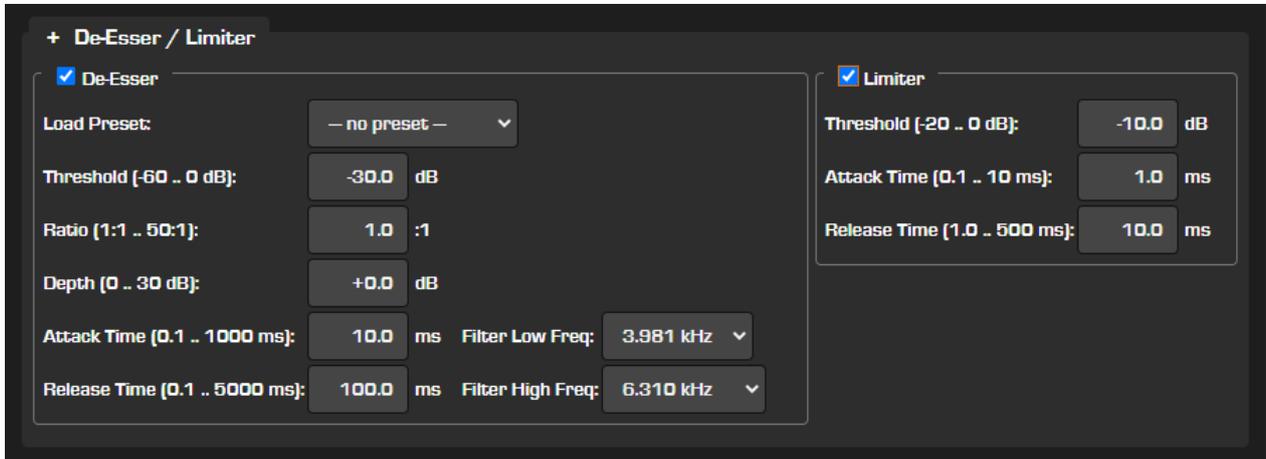


## Load Preset

Again, this option will let you associate one of the EQ presets (if any were created) to the Source at the time of loading. When “- no preset -” is selected in the drop-down, the settings saved with the source (shown below) will be loaded.

## De-Esser / Limiter

This section lets you predefine the default De-Esser and Limiter settings for a Source, when this is loaded to a channel, either manually, or through a Show Profile, and the show channel has these parameters set to “Load from Source Profile”.



### Load Preset

This option will let you associate one of the De-Esser presets (if any were created) to the Source at the time of loading. When “- no preset -” is selected in the drop-down, the settings saved with the source (shown below) will be loaded.

**Note** - De-Esser settings can be pre-defined in the Source Profile, for ALL type source types. You can also store these settings in the Show profile Channels, if you desire.

## Default and Custom Backfeed Options

This section is where complex backfeed configuration can be created for the Source. Three **Backfeed Modes** are available from the drop-down menu located at the top of the [Source Logic](#) section: “Disabled” , “Default”, and “Custom”.

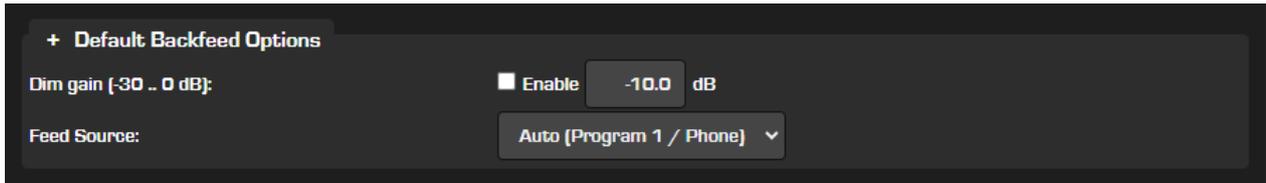
### Backfeed Mode: Disabled

The **Disabled** option is available for ALL Source types and will completely disable the return feed to the Source. In this case, the backfeed section disappears from the page, and the next visible UI section becomes the **Flexible Record Mode Options**.

**Note** - When the backfeed is disabled, the Source will lose all audio return functionality including Individual Guest Headphones.

### Backfeed Mode: Default

The **Default** option is available to Phone, Codecs, Guest and External Microphones. When selected, the Default Backfeed Options section appears. Here, a Gain control for when the backfeed is in Dimmed state (i.e., when TB is injected) is available.



In case of Phone and Codecs Source types only, the **Feed Source** menu is added to the section, adding simple Routing options for creating the Mix-Minus: When Auto (Program 1 / Phone) is selected, the Mix-Minus will be created:

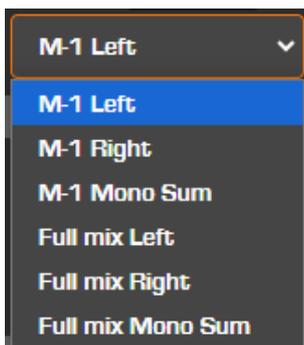
- from the PGM1 bus when the channel is in ON state,
- from the PHONE bus when the channel is in OFF state.

### Backfeed Mode: Custom

The **Custom** option is available to ALL Source Types. When selected, the Custom Backfeed Options section appears. At the top of this section the following options are available, which define the basic settings of the backfeed when the corresponding Input Channel on the console is in ON state:



- **External Feed Src:** lets you pick any audio stream from the network, and assign it to your source's backfeed.
- **IFB Gain:** it's a Gain control for the channel's IFB input only. It does not apply to the External Feed Source. It's normally used to adjust the level of TB injected from an intercom system directly into the channel backfeed. It can cut or boost the audio up to 25 dB.
- **Feed Volume:** it's a volume control which applies to the entire Backfeed. It allows you to reduce the volume by up to 100 dB for either or both sides of the stereo return channel, respectively.
- **Feed Mode:** enables you to pick from a variety of Backfeed styles:



- **M-1 Left/Right/Mono Sum:** sends a mix of the stereo Program out, minus the source, as either a summed Mono signal, or as the Left or Right channel of the stereo Program output, minus the source.

**Note** - No Mix-Minus will be created in case "Studio Monitor" or "External" is selected in the **Feed Source** menu.

- **Full Mix Left/Right/Mono Sum:** disables the Mix-Minus and supplies the complete mix to the Backfeed. You may pick the right or left channel, or choose a summed Mono signal.

**Note** - This option is active for both sides of the stereo Backfeed channel, so that you can completely customize the style of the audio flowing back to your remote user.

Next come a series of **Feed Source** options that can be applied to any or all of 6 different console fader channel states to completely customize backfeed based upon the logical state of the fader strip itself. These logical states are:

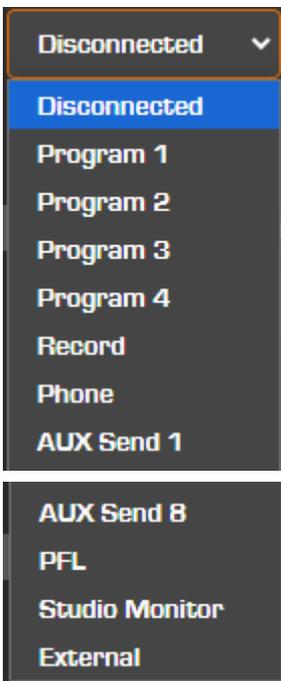
In normal operation Mode

- While Channel is OFF
- While Channel is OFF, But in PFL
- While Channel is ON

In Record Mode

- Record Mode: While Channel is OFF
- Record Mode: While Channel is OFF, But in PFL
- Record Mode: While Channel is ON

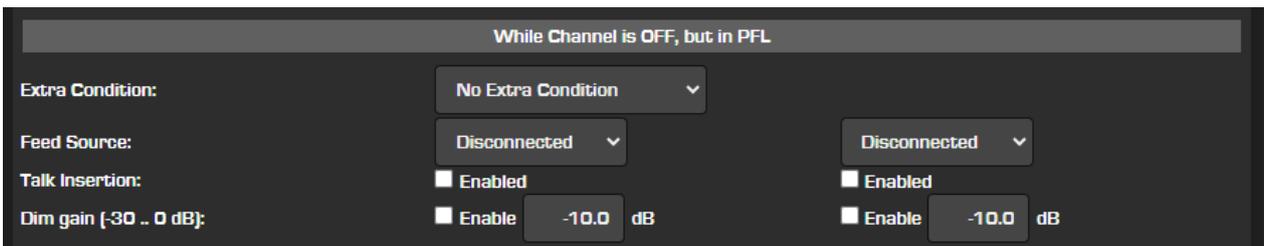
Using the **Feed Source** drop-downs, you can pick the Backfeed source that will be fed to the Codec's IFB channel in each of the six possible channel states noted above. Your choices are:



- **Disconnected:** Use this when you want to disable the Backfeed (send no audio) for a particular channel state.
- **Program 1 – Program 4:** Sends the audio from the selected Program bus.
- **Record:** Sends audio from the console's Record bus. The Record bus is a special variant of the PGM4 bus: The bus mix is post-fader and pre-On/Off, to provide offline recording with volume control.
- **Phone:** Quasar has an off-line Phone bus that is actually a special variant of PGM4. The Phone bus is mono-sum, pre-fader and pre-ON/OFF to allow speaker-phone style operation thru the Operator's mic. Selecting Phone will feed the Phone bus, minus the source, so that the listener can hear other Phone callers who are waiting in the air queue.
- **Aux Send 1 – Aux Send 8:** Sends the audio from the selected Auxiliary mixing bus.
- **PFL:** Sends the audio from any sources assigned to the PFL bus.
- **Studio Monitor:** This is the source typically sent to Guest Studio monitors & headphones by the Control Room board op. It is assigned using the "Studio Monitor" controls on the Quasar Monitor Module.
- **External:** Sends the audio from the channel you specified in the **External Feed Src.** field at the top of the section.

**Note** - A detailed description of how to use the record Mode function can be found in the Configuration Chapter - 5 of this Manual: [Working with the Record Modes](#).

**Extra Condition:** it's an additional logic condition that applies only to Channels which are in OFF state and have PFL active (both in normal operating mode and Record mode).



If the Extra Condition is selected, it will be checked after checking if the channel is being assigned to PFL. The options which follow will be applied only in case the Extra Condition is met. If it's not met, then the channel is considered not being assigned to PFL, therefore it will follow the settings for the *Channel OFF* (only) state.

**Talk Insertion:** it's a switch that enables injection of the Console Talkback bus and/or External IFB source directly into the backfeed. It also enables the Backfeed to be dimmed, while Talk is inserted. De-selecting this option will disable both Talkback and Dim to the backfeed.

**Note** - there is no input source menu on the Web UI for the IFB. This is not exposed to the user. The desired stream will be routed by an Infinity Intercom directly to the Source's LW channel using LWCP.

**Dim Gain:** it's a Gain control for the Dim function which is active when Talk is inserted into the channel's IFB input.

## Flexible Record Mode Options

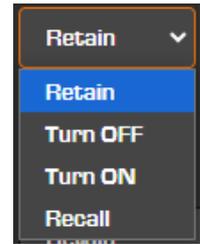
When "Flexible" is chosen as the Record Mode activation option (in the Show Profile's **Record Mode** menu) these options become active, and are applied to the Source.

These menus let you determine the behavior of the channel strip when the console *enters* Flexible Record Mode leaving its normal state, and when it *returns to its normal state* after exiting the Flexible Record mode.

+ Flexible Record Mode Options	
On Entering	On Exiting
Source: Retain	Retain
On switch: Turn OFF	Turn ON
Program 1: Recall	Recall
Program 2: Retain	Recall
Program 3: Retain	Retain
Program 4: Retain	Retain
AUX Send 1: Retain	Retain
AUX Send 2: Retain	Retain
AUX Send 3: Retain	Retain
AUX Send 4: Retain	Retain
AUX Send 5: Retain	Retain
AUX Send 6: Retain	Retain
AUX Send 7: Retain	Retain
AUX Send 8: Retain	Retain

### Source, ON Switch, Program 1-4

- **Retain:** will keep the current parameter's state when entering and/or exiting the Record Mode.
- **Turn OFF:** will turn that parameter to OFF state
- **Turn ON:** will turn that parameter to ON state
- **Recall:** The console maintains two separate states in its internal memory (Standard and Flexible Record modes) for the Source, ON-OFF, Program and Auxiliary Sends parameters. By selecting this option, each time one of the two states is entered, the corresponding parameters will be recalled.

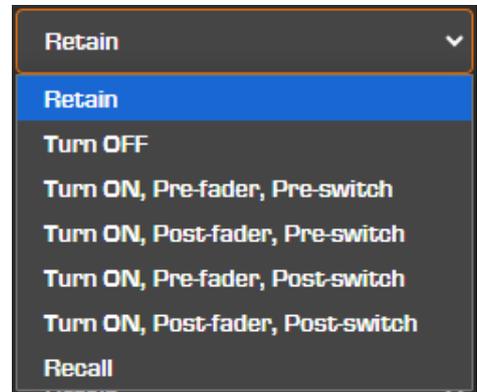


**Note** – The Recall functionality is not available in the Standard Record Mode.

### AUX Send 1-8

The options available for the Auxiliary busses are:

- Retain State
- Turn OFF
- Turn ON, Pre-fader, Pre-switch
- Turn ON, Post-fader, Pre-switch
- Turn ON, Pre-fader, Post-switch
- Turn ON, Post-fader, Post-switch



These options simply reflect the option available from the Channel settings UI, and allow you to pre-define any combination of Pre/Post fader, and Pre/Post ON Switch when the console *enters* Flexible Record Mode.

In order to better understand the operating principles behind the Standard and Flexible Record Modes, and catch all the differences between the two, we invite you to read Operation Chapter 3 of this manual: [Working with The Record Modes](#).

That's it! We covered all the settings available to a Source Profile. Remember to Save Changes at the top or bottom of the page. You have three options for doing that:

- **Save as Copy** creates a copy of the current source, applies the new settings and saves the Source copy in the Console source List.
- **Apply** will apply the new settings, leaving you in the Source Profile Settings page
- **Ok** will apply the new settings, and return you to the main Source Profiles Page.
- **Cancel** will clear the changes and return you to the main Source Profiles Page.

**Note** – Every time you update the Source settings through the Web UI, you will need to reload the Source on the console, for the new settings to take effect.

## Source Ownership

One of the great advantages of the Livewire+ protocol, is that it can associate control data to Sources. When a Source, like a Microphone for example, is loaded into a different Studio, the Preamp Gain control for that Source will be moved with it. But what happens when a source is shared? Who will take control of that Source?

When a Source is shared, it could be loaded on multiple channels of the same console, or loaded by different consoles. In this case, a simple safety mechanism is used to prevent control conflicts: the control of that Source is given only to the channel which loads it for the first time, and not given to the others. These other channels will only be able to receive audio from the shared Source, but not send control data to it. Their display will show the “**listen only**” message.



In case Full Control of a Source which is currently loaded onto a different console is desired, the source must be unloaded so that its control is released by its current owner.

It is possible to check the “owner” and request unload, from the CONTROL & B/FEED Tab of the console Touchscreen GUI.



Pushing the **REQUEST UNLOAD** button will make a confirmation message appear on the screen. If **OK** is selected, and if the channel which currently owns it, is turned OFF, then the Source will be unloaded and control released to the requesting channel.

This mechanism is useful in many cases, for example when console A is left unattended in a studio, which is locked for the night, and an operator in a different studio loads a Profile on console B, which uses the same sources which are left loaded onto the unattended console A. If the console A has those sources off the air, it will be possible (and safe) for the operator, to load them on his console (B). If they are on the air with console A, it will not be possible to load them on console B and take them off the air by mistake.

**Note** - Source ownership will be dropped in case the Engine is disconnected from the console, or in case the number of licensed channels is changed.

## Source Configuration Examples

Following are five examples which represent the basic types of sources found in most studios. A good way to understand Source Profile configuration options is to jump in and build a few common sources that almost any studio would need, such as:

**Operator Microphone** (Operator source type) – the Board Op’s mic

**Guest Microphone** (CR Guest source type) – an additional mic in the same studio

**CD Player** (Line source type) – any basic audio source

**Caller** (Phone source type) – a source that would require a mono mix-minus return

**Codec** (Codec source type) – a source that would need a custom stereo return feed

## Create The Operator’s Mic Source Profile

The Operator’s mic is intended for the operator of the Quasar console, so let’s create this first. Click [Create new source profile](#). The top section will look like this:

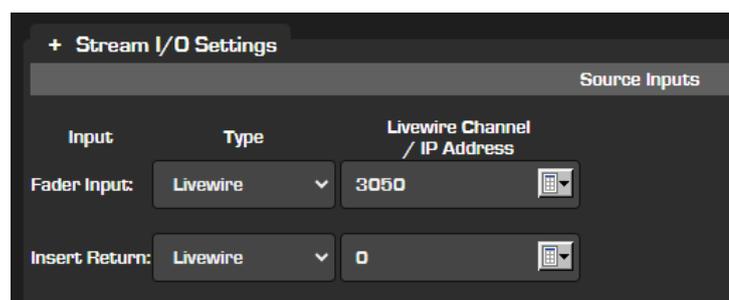


In the **Source Settings** section, select **Operator Microphone** from the **Source type** drop-down list. In the **Source name** field, type a useful name, like “Host” or “Board Op”, that the DJ can easily identify. This is the name that will be shown on the display of fader strip the source is loaded to.

Move down to the **Stream I/O Settings** section, and in the **Fader Input** field, enter the board mic’s Channel Number.

Each audio source in the network has its own unique Channel ID number:

If you know the number, just type it in; if not, use the browse button to the right of the field to select the source from your network.



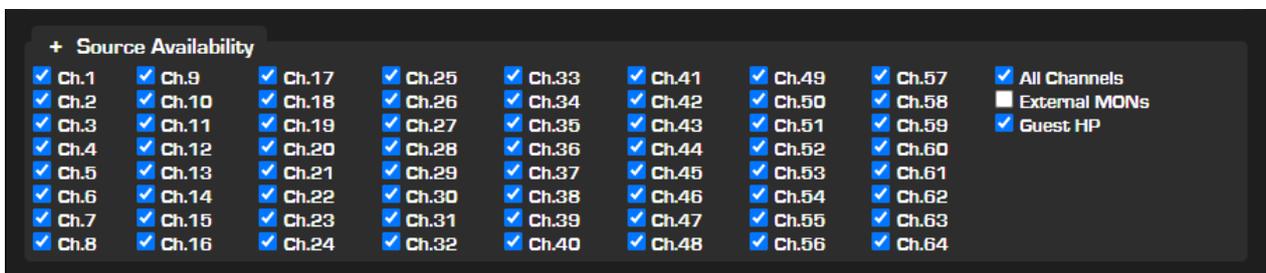
With just these three basic options, you have enough for a working Source Profile!

You could easily leave the remainder of the options at their default values, but a couple of additional adjustments will help eliminate Control Room errors.

As previously explained, in case you want to be able to control the Mic Preamp directly from the console (these are settings normally belonging to the xNodes' Web UI) you can do so by selecting the physical input of the Node to which your microphone is connected to, in the **Audio Node Control** drop-down menu.

**Note** - On all UI pages, the **Apply** button will apply the settings you just entered and leave you in the current page, while the **Ok** button will apply the settings and return you to the upper level of the UI.

In the Source Availability box (down below), you have the option to control what faders and Monitor channels the source may be assigned to. Uncheck the **External MONs** option — it's not wise to assign the Operator's mic directly to the monitors!



Press the **Ok** button at the bottom of the page and your new Mic source is ready to use - you've just created the Operator Mic Source Profile.

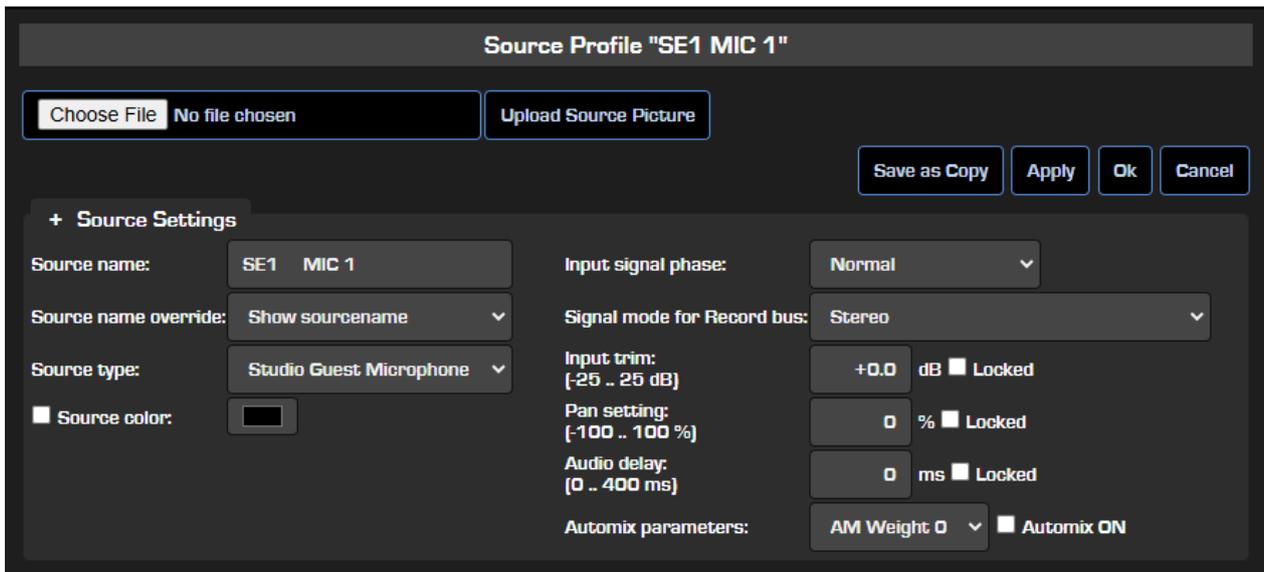
**Note** - There should only be a single Operator source type loaded to the console's faders at any one time. This is because the Operator source type contains some preset logic functions specific to the type, such as muting of the Control Room (CR) monitors when the mic is open, as well as being the default source for any Talkback commands that are engaged. Therefore, any additional Microphone source should be one of the other microphone source types.

**IMPORTANT:** The Operator Microphone is also used for the Talkback function. In order for the TALK function to work, this has to be assigned either to one of the console's channels (any of the 64, no matter which Layer it belongs to) or to the **Studio TB** input found in the Monitor Section of the Show Profiles. Please see Chapter 2 of this manual, for [Setting up the Console Talkback](#).

If you want to add a picture to your Microphone Source, please see the [Source Settings](#) paragraph in this chapter.

## Create a Guest Mic Source Profile

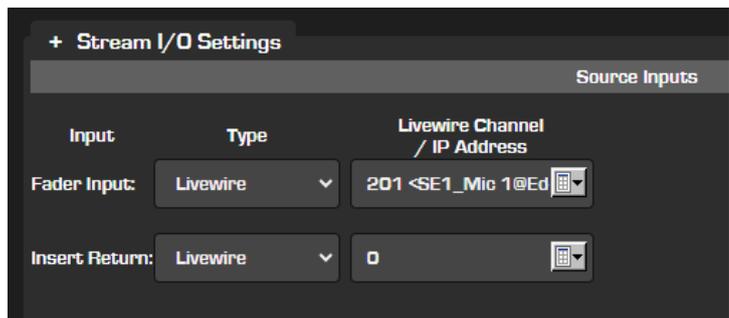
The "CR guest" source type is intended for microphones located at guest positions within the same control room as the Quasar surface. The built-in logic functions will mute the CR monitors when the source is turned on, and provide GPIO logic for an optional Guest Control Panel that uses GPIO to remotely control Channel ON/OFF/MUTE and Talkback functions.



The steps for setup are similar to the ones outlined in the previous paragraph:

From the **Source type** drop-down, select **CR Guest Microphone** and type in a friendly name of up to 10 characters in **Source name**.

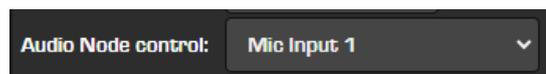
Move down to the **Stream I/O Settings** section, and in the **Fader Input** field, enter the mic's Channel Number, either by typing it in or using the browse button to the right of the field to select the source from your network's source list.



If you would like to control the mic preamp directly from the knob at the top of the channel strip, you can select **Mic Gain** in the **Knob function** drop down.



For additional control, you have the option to define the **Mic Input** which allows for input gain and Phantom Power control directly from the surface.



You can leave the rest of the options at their default settings. Press the **Ok** button at the bottom of the page and you've just created a Guest Mic Source Profile.

## Other Helpful Options: The Knob Function

Pressing the **Channel Options** encoder at the top of any fader strip on your Quasar surface opens the Channel Option screen on the Master module. However, the action taken when the board operator rotates the Options knob *without* pressing it can be tailored to your studio's preferred operating style via the setting in the Source Profile's **Knob Function** drop-down menu. The options are:

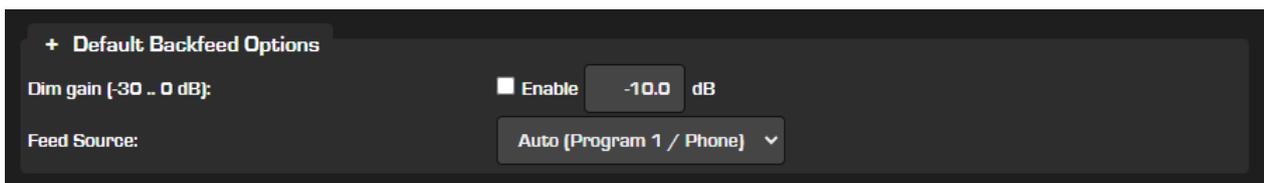
- **None:** will assign no functionality to the knob in case it is rotated without pushing it.
- **Input Trim:** adjusts the digital gain of the channel input stage. This is useful when one

source within a group of similar audio sources has higher or lower level, and the operator wants to maintain a similar physical fader position. This control does not affect the Input gain of the source at the node.

- **Mic Gain:** allows the board operator to use the knob to quickly boost or cut the level of the Analog Mic Preamp stage in the xNode.
- **Line Gain:** allows the board operator to use the knob to quickly boost or cut the level of the Line Input stage in the xNode.
- **Automix Weight:** is for adjusting the Automix weight priority for the selected channel strip. The Automix can then be Enabled or Disabled using the dedicated key below the fader.
- **Channel Pan:** is for adjusting the channel panorama position.

## Other Helpful Options: Default Backfeed

The other Source Profile option that's important to microphone sources is found in the **Default Backfeed Options** box.



In Axia terminology, “Backfeed” refers to any *audio return signal* that is sent back to an audio source (such as a microphone, codec or phone caller) from the console. When pressing the **TB** key on a fader strip, the board-op’s mic is routed, pre-fader, to the “backfeed” of that channel — an IFB function, basically. This may be a Mix-Minus for phone or codec sources, or a private headphone feed for microphone positions.

The **Dim gain** setting defines the amount of cut, in dB, by which the Backfeeds normal audio is adjusted, so that the board-op can be better heard in the mic-user’s headphones.

## Other Mic Profile Types

In addition to the **Operator** and **CR Guest** mic profiles, several other types are provided: CR producer, Studio guest, and External microphone are also available.

- **CR producer** is intended for a Show Producer’s mic located in the same studio as the mixing console, so its GPIO logic functions mute the CR monitors, and specialized GPIO functionality permits the producer to talk (using **Backfeeds**, Axia’s name for internal foldback, or IFB) to any source with a Backfeed that is assigned to the PFL (Pre-Fader Listen) bus.
- **Studio guest** is used for mics located in an adjacent studio — for instance, a talk studio for stations hosting Talk formats, or a music format with a morning show crew. The

**Studio guest** source type logic mutes the Studio monitor mix when the mic is turned on, and provides GPIO logic that permits optional control panels to control ON/OFF/MUTE states, and make use of the Talkback channel to the Control Room board operator.

- The **External** microphone source type is used for any microphone that will benefit from a headphone feed, and located in a space that does not require monitor speaker muting.

## Create a CD Player Source (Line Source Type)

The **Line Source** type is the basic source for inputs other than microphones. It will not mute the monitor speakers when ON, and doesn't require a Backfeed (mix-minus or IFB).

**Line Source** is perfect for creating Source Profiles for devices such as CD players, DAT decks, Satellite receivers, PCs, etc.

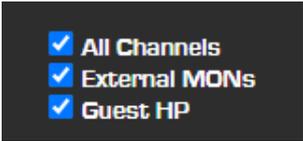
From the **Source Type** drop-down, select **Line**, then enter a name for the device.

Move down to the **Stream I/O Settings** section, and in the **Fader Input** field, enter the input's Channel Number, either by typing it in or using the browse button to the right of the field to select the source from your network's source list.

Input	Type	Livewire Channel / IP Address	Format	Channel Map Left - Right
Fader Input:	Livewire	213 <Sony CD@Edge>		
Insert Return:	Livewire	0		

Now click the **Ok** button at the bottom of the screen; your CD player is configured and ready to be assigned to a fader.

You can select the **External MONs** checkbox in the **Source Availability** section. Doing this allows the board op to assign the CD player to the monitors for direct auditioning.

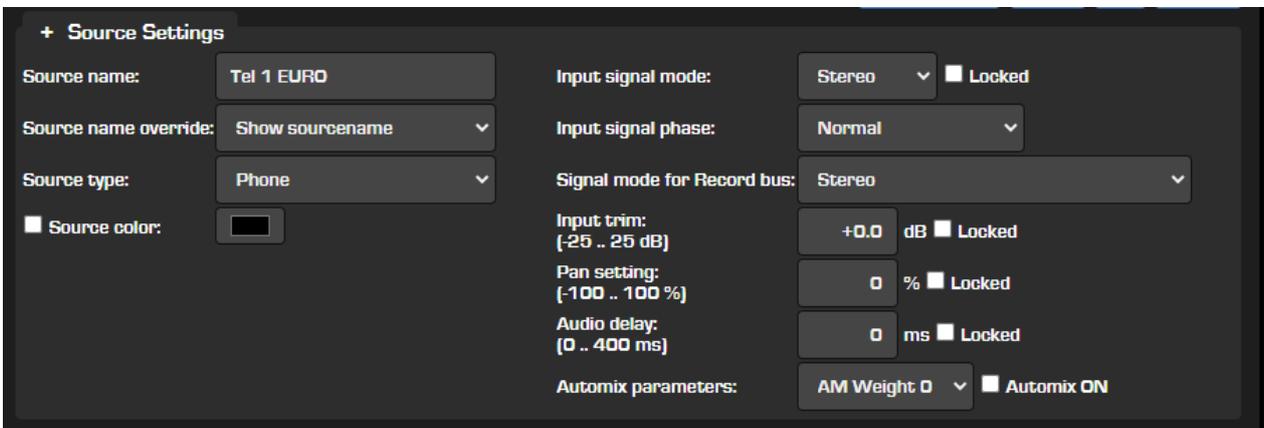


A more practical application for this option is when creating a Source Profile for an Air Monitor, so that talent can directly monitor the over-the-air broadcast signal. To do this, you'd create a **Line** source type for the air monitor receiver, and *uncheck* all the Channel availability boxes *except* the External. This way, the over-the-air signal can feed the monitors — but *not* be assigned to a fader and sent back to the transmitter! An easy method to do this is to select the **All Channels** checkbox twice which will unselect all 64 channels with the second click.

Another checkbox that's useful for Line source types is the one marked **GPIO ready enabled**. This controls the OFF lamp on the fader strip that the source is assigned to. Some operation practices require an indication of source readiness; when this box is checked, the fader's OFF lamp will only illuminate when the device provides a "ready" logic state. Many professional CD players provide GPIO closures for such states, as well as most modern Automation systems, but make certain that you understand this feature prior to enabling and, that your device really does support a "Ready" indication. If it doesn't, your operators may think the OFF lamp is broken because it never illuminates!

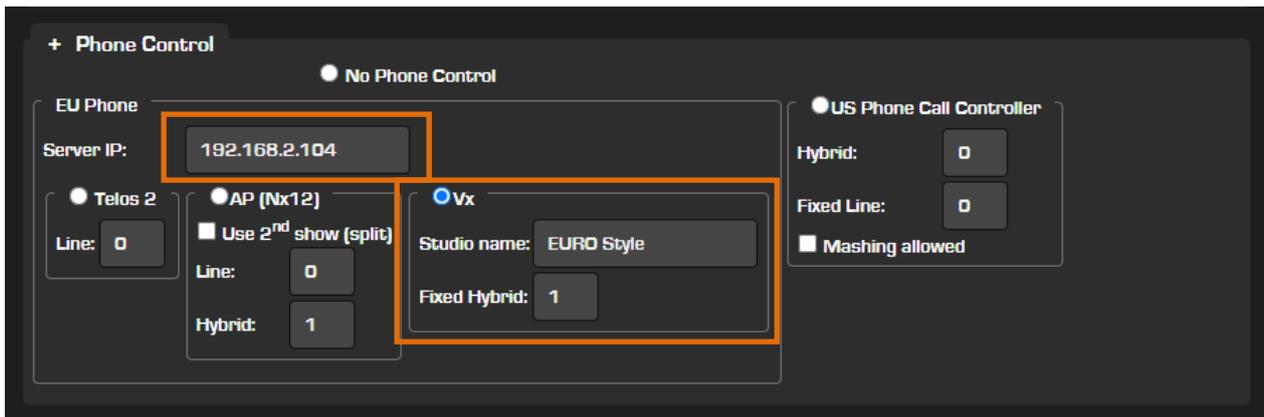
## Create a Telephone Source (Phone Source Type)

Putting phones on-air is one of the basic operations of the modern studio. Quasar's **Phone** Source Type helps ease the task of handling outboard phone hybrids.



First, select "Phone" from the **Source Type** drop-down, and some phone-specific options will appear. In the **Source Name** field, type the name you'd like to be displayed on the console. Then select one of the following options that best suits your studio's phone hybrid:

- **No Phone Control:** This is used for Telos Hx products, or for non-network-controlled hybrids from other manufacturers. You'll still be able to control the hybrid via GPIO: move to the **Hybrid Answer Mode** drop-down menu and select either "Channel ON answer hybrid" or "Channel ON or PFL ON answers hybrid":  
In the first case, when the Phone source is assigned to a console fader strip, turning that channel ON answers the phone. In the second case, the operator may either turn the channel ON *or* use the channel's PFL key to answer the phone.
- **EU Phone:** allows you to map specific lines/hybrids from a Telos Multi-Line phone system (such as TWOx12, Nx12, Nx6 or VX) to a single fader.



In the example shown above, we're mapping a hybrid from a VX Broadcast VoIP system, using the sections highlighted in orange:

- Type in the IP address of the VX Server into the **Server IP** field
- Select the **VX** radio button.
- Enter the **Studio name** configured in your VX system into the **Studio Name** field.
- Enter the number of the VX Channel (hybrid) from the specified VX Studio into the **Fixed Hybrid** field.

**Tip** - If the Server IP has authentication requirements, the syntax to use in the Server IP field is: *username:password@ipaddress*.

For example, *user:test@192.168.100.200* attempts to log into the specified IP address with username *user* and password *test*.

- **US Phone Call Controller** ties the Source Profile to the Call Controller available in the Master module.

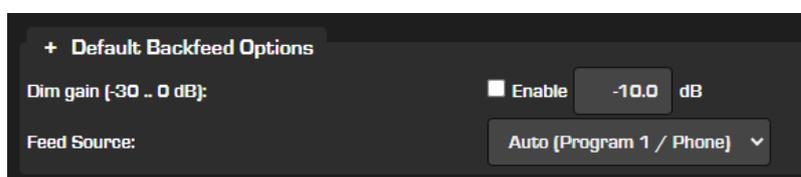
To access the Call Controller, select the CALL CONTROL tab in the Master module's touchscreen. A detailed explanation of how to configure and use the Quasar Call Controller is given in the next Chapter of this manual, which is specifically dedicated to Phone workflows.

- In the **Hybrid** field, enter the hybrid number you want this Source Profile to be associated with.
- The **Fixed Line** field is another method to do what the EU Phone method accomplishes; i.e., assigning a dedicated line to a dedicated fader. Most users with Call Controllers will not use this option.
- The **Mashing Allowed** checkbox permits a single hybrid to handle multiple callers at once by allowing the board operator to select multiple line buttons. Check (or un-check) this box as you desire. This works like a "Conference" mode.



The final step is to define how you want to handle mix-minus. Quasar (and all Axia consoles) automatically generates mix-minus (N-1, or "clean feed") for each phone caller taken to air.

Scroll down to the **Default Backfeed Options** box and select the desired audio mix from the **Feed Source** drop-down.

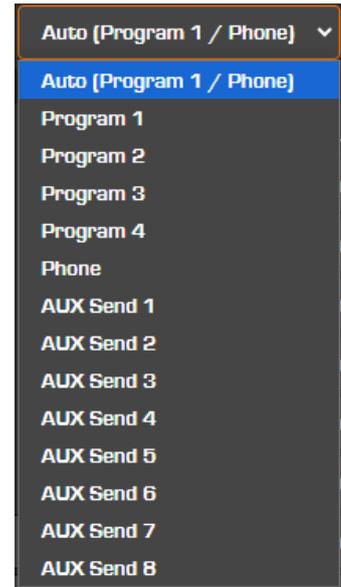


Thirteen different Manual Backfeed mixes plus an Auto smart mode are available:

The default option is **Auto**. Choosing this option eliminates manual mix-minus building by switching the source of the mix-minus based on the ON state of the fader the Phone source is loaded to. When the fader is in the OFF state, the caller hears the off-line PHONE mix. The moment the channel is turned ON, the audio feeding the caller switches to the Program 1 bus, minus the caller's own audio. Here are the manual options:

- **PGM-1** through **PGM-4** feed the caller the output of the selected Program bus, minus their own voice.
- **AUX Send 1** through **AUX Send 8** feed the caller the output of the selected Aux Send bus, minus their own voice.
- **PHONE** is a mix designed to harmonize with typical, traditional radio operations.

A channel is assigned to the PHONE mix by selecting the fader strip's **PGM 4** key; the PHONE mix is then created pre-fader and pre-ON/OFF, so that pressing the **PGM 4** key on any channel will send the audio of that channel to the phone mix at unity gain.



We suggest leaving the selection at its default, **Auto**, unless special circumstances dictate otherwise. No matter what you choose here, the board operator can quickly override it and select a different function, using the controls made available at the master modules touch display.

When your setup is complete, click the **Ok** button at the bottom of the screen.

For more detailed information on how to set up Phone Sources, and how to use the Call Controller, please read the [Configuration Chapter 3: "Setting Up Phones"](#).

## Create a Codec Source with Custom Backfeed

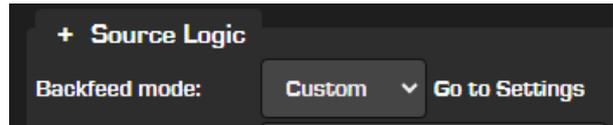
If you have created a Phone Source Profile as described in the previous section, creating a Profile for a codec will appear similar (minus the phone hybrid controls).

Since the codec is a bi-directional device, your codec Source Profile will need a Backfeed configuration as well.

The Default Backfeed behavior of the Codec provides a mono sum to both the Left and Right channels on the return audio signal. However, when the Talkback key (TB) on the channel strip, the board operator's mic audio is routed (pre-fader) to the left channel only; the right channel remains unchanged.

When you create a Source Profile from scratch, all Profile types have the **Backfeed Mode** option configured as "Default". But your workflow might require a more specialized Backfeed than the Default behavior.

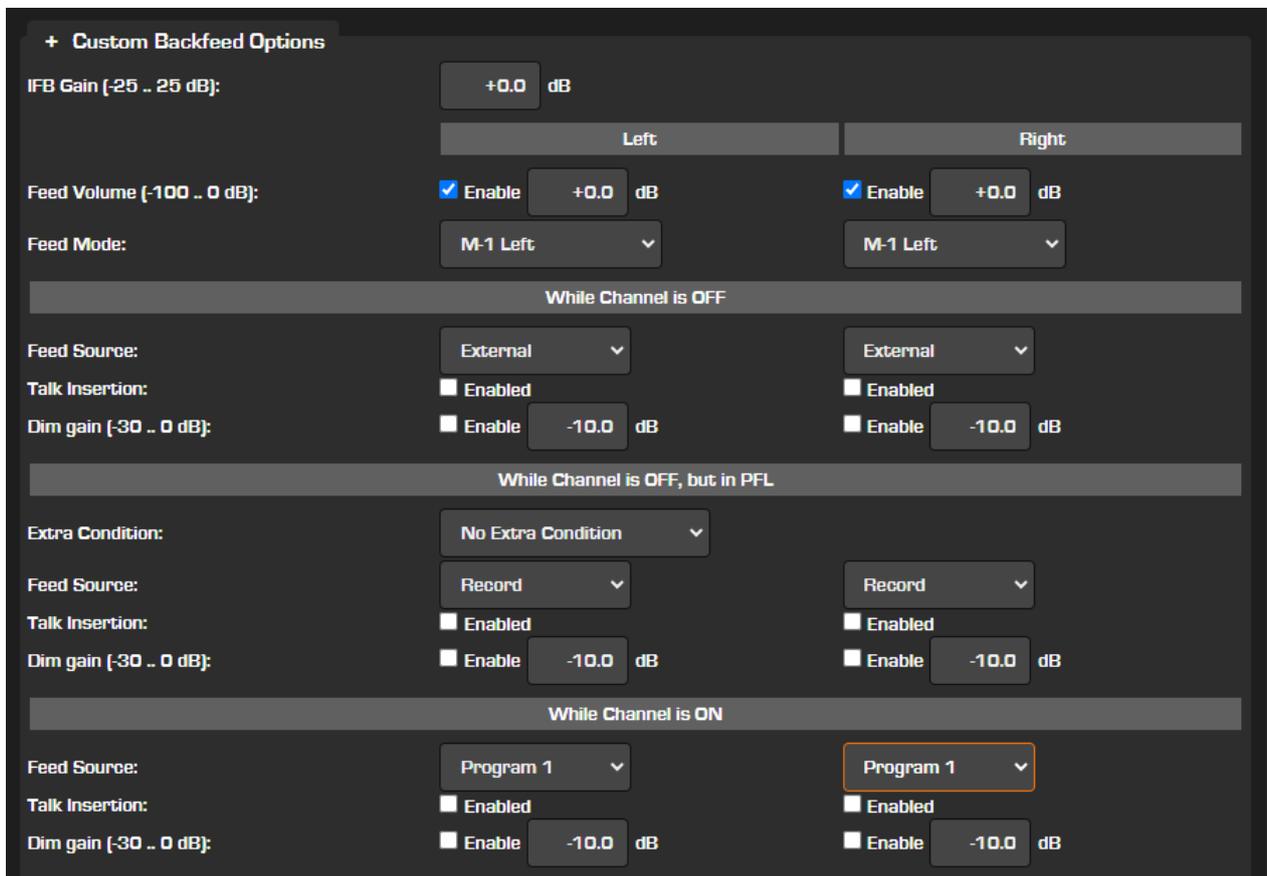
That's what the **Custom** option is for: it allows high-level control of mix-minus behavior based on channel state logic.



As you can see from the screenshot below, the **Custom** option provides significantly enhanced backfeed routing options, including independent control of the L and R sides and a completely different IFB signal insertion point. In some cases, an Extra Condition can be added to switch the automatic backfeed switch only when this additional condition is met.

**Note** - The grey outlined headers (like “While Channel is OFF”, for example) always refer to the settings below the header and not to the ones above. This is a standard convention adopted in all Quasar UI pages.

For example, let's say you want to set up a custom Conditional Backfeed for your codec, that sends the CR Headphone when the channel is OFF, the contents of the Record bus when the channel is OFF but assigned to PFL, and a PGM 1 mix-minus when the channel is ON. You would have to set it up as follows:



Naturally, this level of complexity is not necessary in every radio station, but we made it available for specialized situations where highly-tailored Backfeeds are required.

# Show Profiles

Now that you know how Source Profiles work, let's talk about Show Profiles:

In the same way that Source Profiles allow you to pre-determine how an individual channel is configured when a source is loaded, Show Profiles let you build configuration files that can be loaded to determine how the entire Quasar behaves, which sources are loaded, and what console channels strips they appear on.

Show Profiles are, essentially, a collection of Source Profiles. You can also consider a Show like a snapshot of an entire console. A Show Profile keeps track of :

- Input channels Source assignment and configuration
- All channel-related settings like: Ch Control & Locks, Bus Assignments, Automixer settings, Filters, EQ, Dynamics, De-Esser, Backfeed options, and Record Mode parameters
- All the Monitor section settings
- All the Automixer settings
- All the Master Auxiliary (sends & returns) section settings
- The Record Mode parameters and monitor settings
- Your Telos Phone system control parameters and settings

**Note** - With the words “console channels” we always indicate the *Quasar Engine's DSP input channels*, and not the physical channel strips on your mixing surface.

You can also use Show Profiles to define different types of broadcasts – one for the morning show, one for talk segments, one for musical guest interviews, one for unattended operation – that instantly recall your saved configuration when loaded. Up to 9999 Show Profiles can be saved in a console.

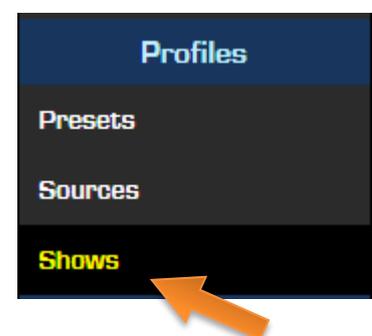
**Note** - Loading a Show Profile will never take an active audio source off the air: any changes to a fader's source assignment are queued until after the OFF key for that fader is pressed by the operator onto the desk.

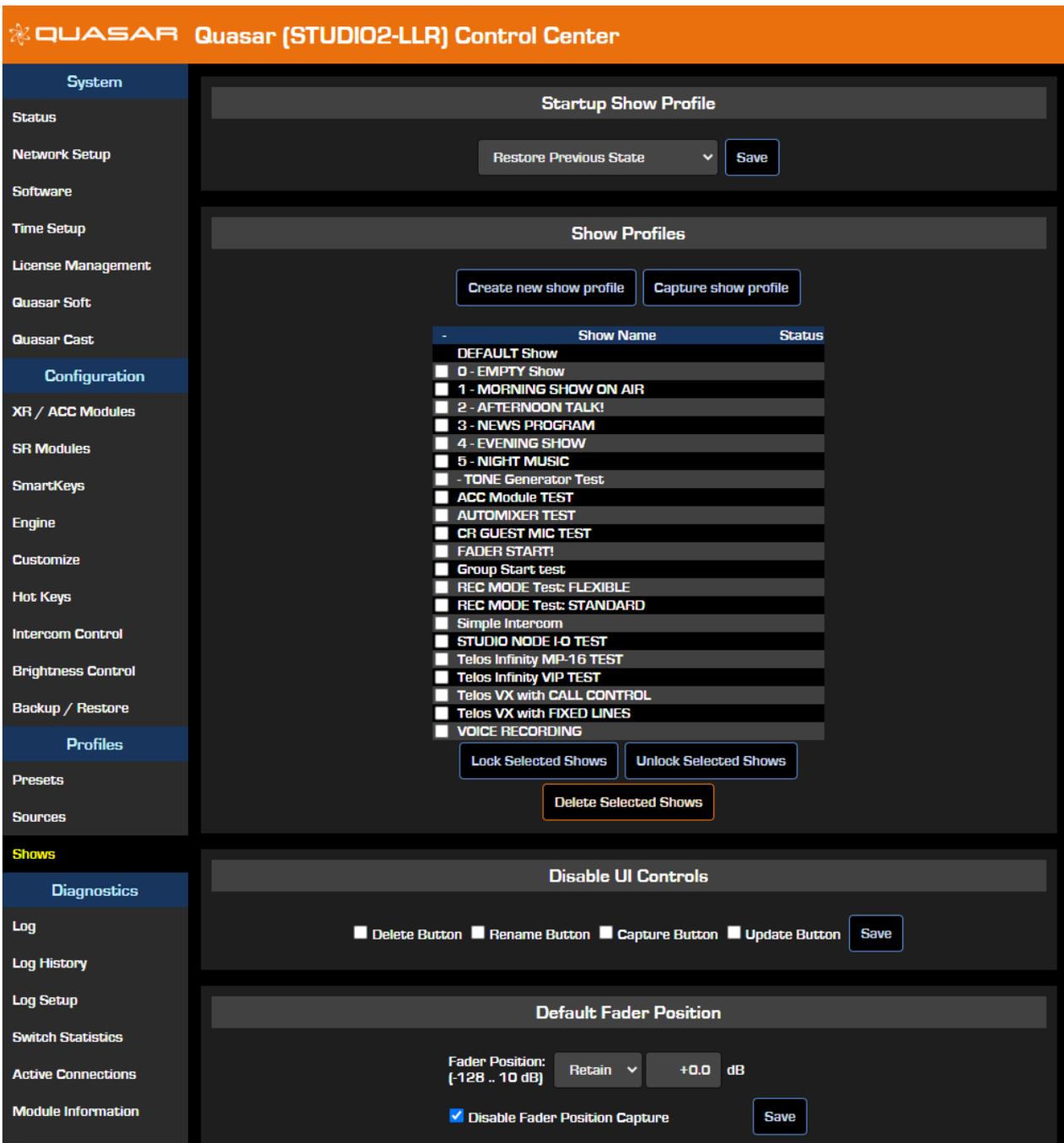
## The Show Profile Web UI

To get started, connect a computer to the Quasar and access its configuration Web UI by entering the MTS Module IP address into your Web browser. Select the **Shows** link, as shown here:

The landing page is divided into the following four sections:

- Startup Show Profile
- Show Profiles List
- Disable UI Controls
- Default Fader Position





The settings in these sections are applied globally, and will let you customize how the Shows UI will behave and appear to the console operator:

### Startup Show Profile

This section lets you decide which Show will be automatically loaded by Quasar, when the console is started, or rebooted.

Choose **Restore Previous State** from the drop-down menu, if you want the console to load the last active state it was in, before it was shut down, or rebooted.

Choose **Default Show** from the drop-down menu, if you want the console to load a specific Show that you define as Default. The Default Show cannot be deleted.

Choose any Shows from the drop-down menu, if you want the console to load that Show at startup. The Default Show cannot be deleted. In case the Show selected as Startup Show will be deleted (from the list), the system will default to the **Restore Previous State** setting.

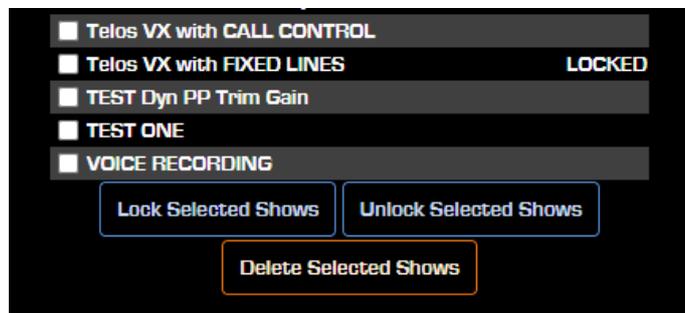
Press the **Save** button to apply the desired setting.

### Show Profiles

The section offers a list of all the Shows that have been created. Here you can create new Show Profiles from scratch, or “capture” them directly from the console. We’ll see how to do that in the next paragraph.

You can also Lock, Unlock or Delete Shows (except the Default), by selecting the checkbox to the left of each entry and clicking on the **Lock Selected Shows**, **Unlock Selected Shows**, or **Delete Selected Shows** buttons available at the bottom of the list.

In case one or multiple Shows are locked, the “LOCKED” status will be shown to the right of each entry.



### Disable UI Controls

In this Section you can individually enable or disable the Delete, Rename, Capture and Update buttons that appear in the SHOW PROFILE TAB of the Touchscreen GUI.

Disabled buttons will disappear from the console touchscreen, once tabs are switched and the screen is refreshed. This is useful to prevent unexperienced operators perform destructive operations from the console.

Press the **Save** button to apply the desired setting.

### Default Fader Position

This section lets you define a default value for the Fader’s Position, so that when a new Show is created, all channels will have their faders preset to this value. Available settings are:

- **Retain:** The current Fader position is kept unchanged, when the Show is loaded.
- Select **-∞** from the drop-down menu to set the fader level to the minus infinite position.
- Select **Use:** from the drop-down menu to specify the level at which the fader should be set when the Show Profile is loaded. Enter the desired dB level in the next field. You may specify any level between -128.0dB and +10.0dB.

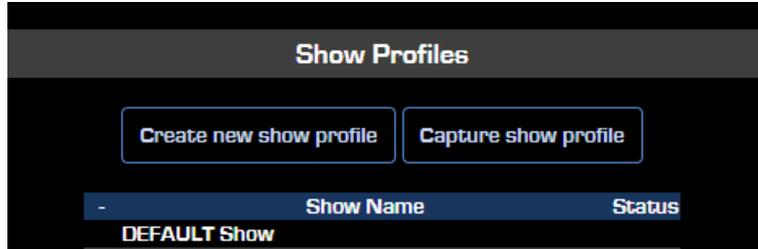
This is helpful to avoid having to configure fader position on all 64 channels after a new show has been created, and it is essentially the same set of parameters you will find in the [Channel Configuration Page](#).

The **Disable Fader Position Capture** option makes the console ignore, and not store the fader position when a Show is Captured. This option was specifically introduced for **Quasar SR** consoles which have non-motorized faders. In this case, the faders position is saved as **Retain**.

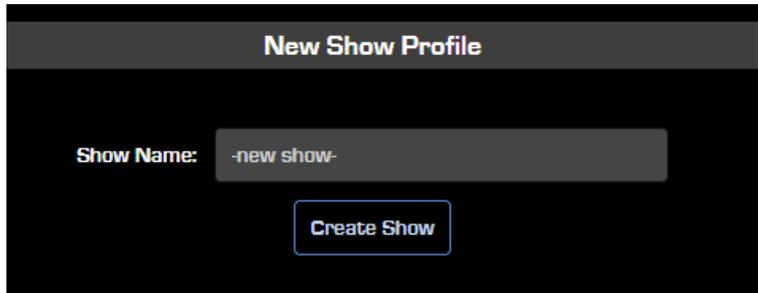
# How To Create a Show Profile

## Create a Show from the Web UI

The **Create new show profile** button allows you to create an entire Show Profile completely from scratch and configure it using your computer's on-screen controls.

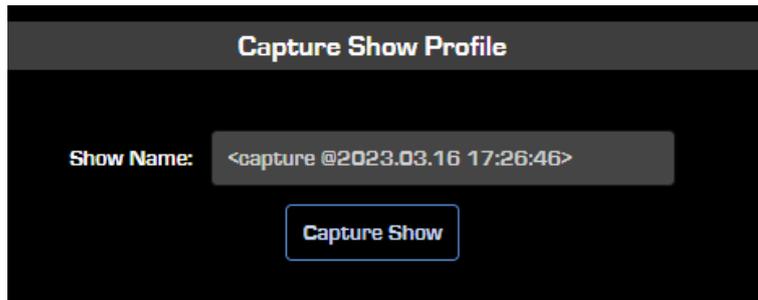


This is the “expert” way of making a show profile. You can save up multiple Show Profiles to make console reconfiguration fast and easy. Click the **Create Show** button when done.



Alternatively, simply click on the **Capture show profile** button, and you'll be prompted to name your new Show Profile.

Type in a name, overwrite the default capture name with your own, press **Capture Show**.



Now you've got a new Show Profile that can be loaded by pressing the **PROFILES** key at the console's Master Monitor module.

## Create a Show from the Console

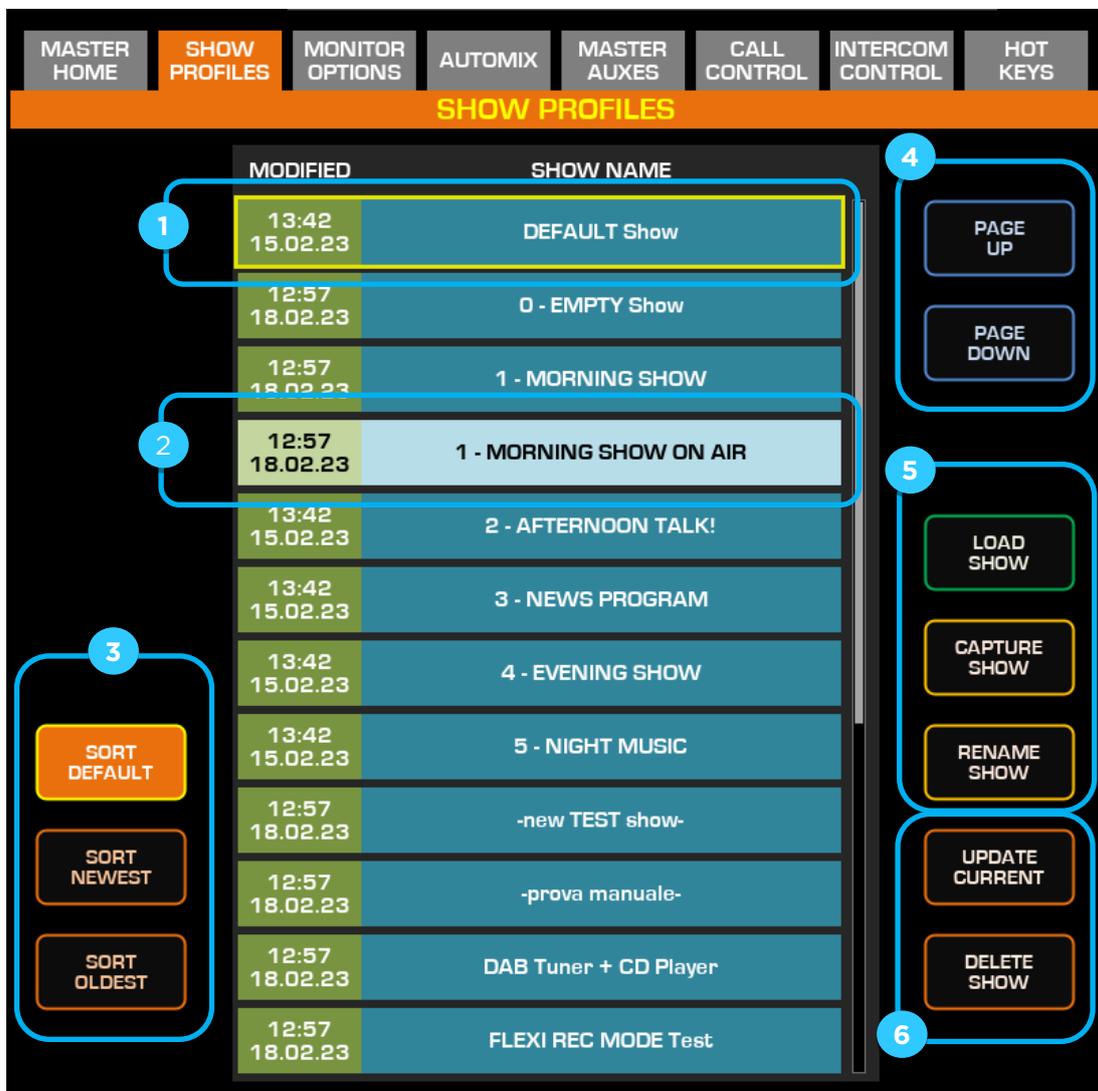
The easiest way to create a Show Profile is to set up the console the way you like it using the controls on the board itself – then use the **CAPTURE SHOW** button found in the SHOW PROFILES tab of your Quasar Touchscreen. This will take a snapshot of the entire console – fader assignments, EQ, Dynamics, Monitor assignments, even headphone EQ settings – and save them to a new Show Profile that you can use as-is, or edit later. Alternatively, you can do the same from the Web GUI.

- Assign a source to each fader strip using the Options knob at the top of each strip, and selecting a source from the Current Source selection box.
- Once that's done, assign each source to the Program bus you want it to feed, using the PGM keys at the top of each fader strip.
- Finally, select the desired audio feed for your Monitors and Headphones, using the keys at the bottom of the Master Monitor Module.

Now that your console is set up the way you want, press the **PROFILES** key on the top right of your Master Module (or select the SHOW PROFILES tab on the Touchscreen) and push the **CAPTURE SHOW** button: an on-screen keyboard will appear: type a name for your new Profile, and press **OK**.

The picture below shows the Show Profile page on the Touchscreen UI. From a logical standpoint this is divided into the following areas:

1. **Selected Show:** relates to **LOAD SHOW**, **RENAME SHOW**, **DELETE SHOW**, buttons
2. **Currently Loaded Show:** relates to **UPDATE CURRENT** button
3. **Sort buttons:** reorganize the Shows List according to alphabetical order (Default) or saving date/time. (Newest and Oldest)
4. **Page buttons:** useful alternative to touch scroll, when lots of Show Profiles are listed
5. **Non-Destructive buttons:** used for every-day Show Profile management
6. **Destructive buttons:** used for expert Show Profile management.



**Note** - Delete, Rename, Capture and Update buttons can be disabled from the Web UI.

## Show Profile Main Page

Once a Show Profile is created, select one from the list to access its Main Configuration page.

- The **Show Name** field can be used to update the profile's name.
- The **Show Lock** feature is an option that will lock the show, preventing the user from renaming or deleting the show from the console Touchscreen GUI.

The **Monitor Section** , **Aux Masters** , **Record Mode** and **Phone Control** menu buttons will let you access the configuration pages for those sections. These are explained in detail later on in this chapter.

Remember to save the Show after you've made changes in any of these menus.

**Note** - There are also some options, such as Record Mode setup, that cannot be set from the console surface, and must be set up by editing the Show Profile itself.

## Quick Channel Options

Some quick options are available next to each channel select button. This is to easily compare the most important settings of all channels at a glance, and in case edit, without the need for entering the configuration pages for each channel. For each Input Channel, it is possible to:

- pick a **Source** from the drop-down list (A)
- set the **Automixer** activation and Weight parameter (B)
- set the **Group** activation and Master/Slave status (C)

The screenshot displays the 'Main Page' for 'Show Profile "1 - MORNING SHOW ON AIR"'. At the top, the 'Show Name' is '1 - MORNING SHOW ON AIR' and 'Show Lock' is 'Enable Lock'. Below are navigation buttons: 'Monitor Section', 'Bus Processing', 'Aux Masters', 'Record Mode', 'Phone Control', and 'Misc Options'. The main area is a grid of 24 channels. Each channel has three settings: Source (e.g., 'xN\_MIC 1 Operat'), Automixer (e.g., 'Automix OFF'), and Group (e.g., 'No Group'). Three blue circles labeled A, B, and C are overlaid on the grid, pointing to the Source, Automixer, and Group settings for Channel 04.

### Source

This drop-down lets you quickly assign, or change, a Source to the selected channel, from a list of configured Source Profiles. It duplicates the function of the **Source** menu found inside each of the Channel Options pages.

## Automixer

Quasar offers an entirely new **Automixer**. Here you can set and compare the Weight for each input channel, or disengage a channel from the Automixer. The value ranges from 0 to 9.

**Note** - This is the only place in the Quasar Web UI where you can configure the Automixer Weight. There is no option to access this parameter from inside the Channel Options pages.



## Groups

Quasar offers a **Group** feature that enables the user to turn several faders ON by pressing the **ON** key of the Master fader. The options for this function can only be controlled in this screen. Unlike Source Selection, you won't find these settings duplicated in each of the Channel Options pages.

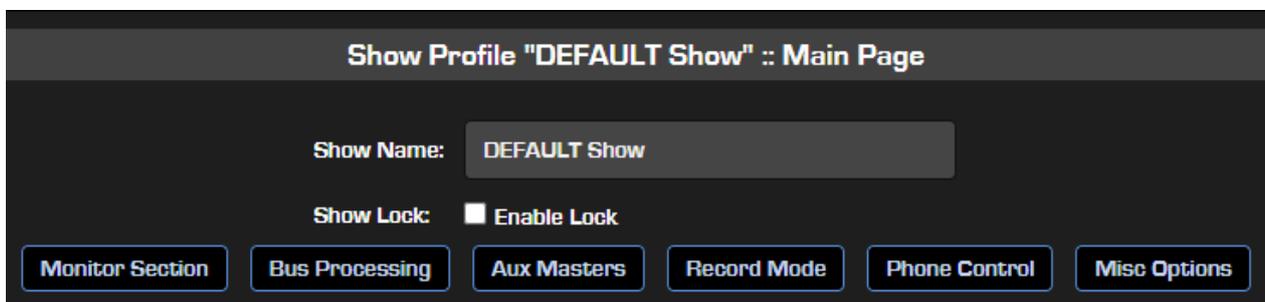
- **Master:** Designates this fader channel as a Group Start Master. Pressing its **ON** or **OFF** keys will turn Slave faders on and off as well.
- **Slave:** Designates this fader channel as a Slave. It will mirror the ON/OFF state of the Group Start Master fader.
- **No Group:** Normal ON/OFF operation.



## Show Profile Top Menu

Next, is a complete description of the Channel Configuration page: this can be accessed by selecting any of the 64 Input Channels available in this screen, grouped in two blocks of 32.

Following is a description of each of the five pages that can be accessed via the top menu in the Show Profile Main screen: **Monitor Section**, **Bus Processing**, **Aux Masters**, **Record Mode**, **Phone Control**, and **Miscellaneous Options**.



**Note** - The Bus Processing section was added with SW v3.0.

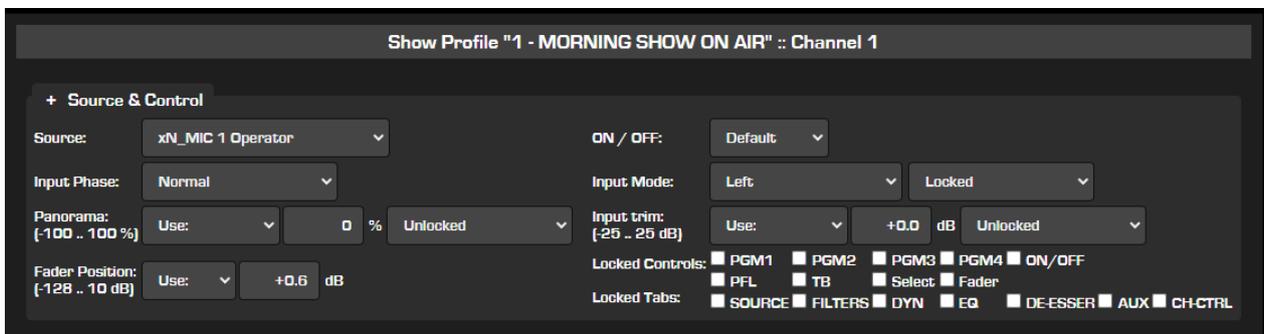
## Channel Configuration Page

In each of the Show Profiles' Main Pages, there is a list of the available Input Channels: selecting any of the channels will take you to a configuration page where all parameters related to that particular Channel can be edited and stored within the Profile. After clicking on one of the channels, a full set of options is displayed.

**Note** - choosing the “Retain Source Setting” option for any item leaves that item unchanged when loading the Show Profile

Next is a full reference listing of the various options found on each Channel Configuration page.

### Source & Control



#### Source

This drop-down menu allows you to select a Source to assign to the fader, from a list of saved Source Profiles

#### Input Phase

- **Phase is from Source:** The Phase setting saved with the source is kept unchanged, when the Show is loaded.
- **Normal:** No change is applied to the phase of input signal when the Show is loaded.
- **Invert Left:** Reverses the phase of the left leg only, of the input channel.
- **Invert Right:** Reverses the phase of the right leg only, of the input channel.
- **Invert Left and Right:** Reverses the phase of both input channel's legs.

**Note** - The “Invert Right” and “Invert Left and Right” options will apply only if a Stereo Source is loaded to the channel. Remember that Microphone sources are considered mono and therefore, only the “Normal” and “Invert Left” options will be applied. The other two will be ignored.

#### Panorama

- **From Source:** The Pan setting saved with the source is kept unchanged, when the Show is loaded.
- Select **Use:** from the drop-down menu to set the Pan of the input signal left or right by a fixed % amount. Enter the desired amount in the next field.

#### Fader Position

- **Retain:** The current Fader position is kept unchanged, when the Show is loaded.
- Select **-∞** from the drop-down menu to set the fader level to the minus infinite position.

- Select **Use:** from the drop-down menu to specify the level at which the fader should be set when the Show Profile is loaded. Enter the desired dB level in the next field. You may specify any level between -128.0dB and +10.0dB.

This setting operates in conjunction with the **Channel On/Off Status** described below. Using these two settings, you can set a channel to turn ON, and its fader to assume a preset value, when the Show Profile is loaded. This is useful when creating a Show Profile for use with automation systems; combine it with the **Locked Controls** to set up a Show Profile for unattended, automated operation with controls that cannot be inadvertently changed by careless operators.

## ON / OFF

- **DEFAULT:**

IF a new source is being loaded to the channel:

1. The console waits until the channel status is OFF. The source, and its channel parameters are not loaded until the channel is turned OFF.
2. When the channel is OFF, the source and channel parameters are loaded.
3. The channel is left OFF.

OR if the same source is to be "reloaded" to the channel:

4. The current channel ON/OFF state is ignored (if ON, it is left ON)
5. The channel parameters are just reloaded, without reloading the source.

- **Safe ON:**

IF a new source is being loaded to the channel:

1. The console waits until the channel status is OFF. The source, and its channel parameters are not loaded until the channel is turned OFF.
2. When the channel is OFF, the source and channel parameters are loaded.
3. The channel is turned ON.

OR if Same source is to be "reloaded" to the channel:

4. The current channel ON/OFF state is ignored (if ON, it is left ON)
5. The channel parameters are just reloaded, without reloading the source.
6. The channel is turned ON if it's currently OFF.

- **Force OFF:**

1. The console forces the channel OFF regardless of its ON/OFF state.
2. Loads the source and channel parameters.
3. Leaves the channel OFF.
4. IF the same source is to be "reloaded", the audio will not be interrupted and only its parameters will be reloaded.

- **Force ON:**

1. The console unloads the current source regardless of the channel ON/OFF state
2. Loads the source and channel parameters
3. THEN turns the channel ON (if not ON already)
4. IF the same source is to be "reloaded", the audio will not be interrupted and only its parameters will be reloaded.

## Input Mode

- **Mode is from Source:** The Mode setting saved with the source is kept unchanged,

when the Show is loaded.

- **Stereo:** Sets the source signal mode to Stereo.
- **Left / Right:** Takes the chosen side of the input signal and sends it to both sides of the stereo channel.
- **Sum:** Creates a sum of both input channels and sends the sum to both sides of the stereo channel.

**Note** - if the Mode is set to “Lock is from Source” and the Source type is a Microphone, the Show will be loaded with the Source locked to Left channel mode. This is because Mic Sources are considered as mono, and always saved with a “hidden lock to Left” mode. Only Stereo line sources have the possibility to unlock the Mode in the Source profiles.

### Input Trim

- **From Source:** The Input Trim level saved with the source is kept unchanged, when the Show is loaded. Select **Use:** from the drop-down menu to set the input Trim level by a fixed dB amount. Enter the desired level in the next field. You may specify any level between -25.0dB and +25.0dB.

Also, it is possible to select the following lock options for both the **Input Mode** and **Input Trim** parameters, from a drop-down menu:

- **Lock is from Source:** will retain lock settings from the source
- **Unlocked:** allows you to change the Signal Mode from the console’s UI.
- **Locked:** prevents the console operator from making changes.

### Locked Controls and Locked Tabs

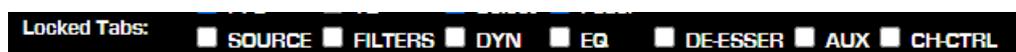
From the **Locked Controls** section, it is possible to lock any of the following controls: PGM 1-2-3-4 , ON/OFF, PFL, Talkback, Select (Option Knob pushbutton) and Fader, by selecting any of the checkboxes.



The above Locks will prevent the operator from changing the status of the selected parameters, from the console’s Channel Strips. These locks apply only to the Fader Module hardware controls (pushbuttons, faders, knobs), and not the corresponding TS GUI controls.

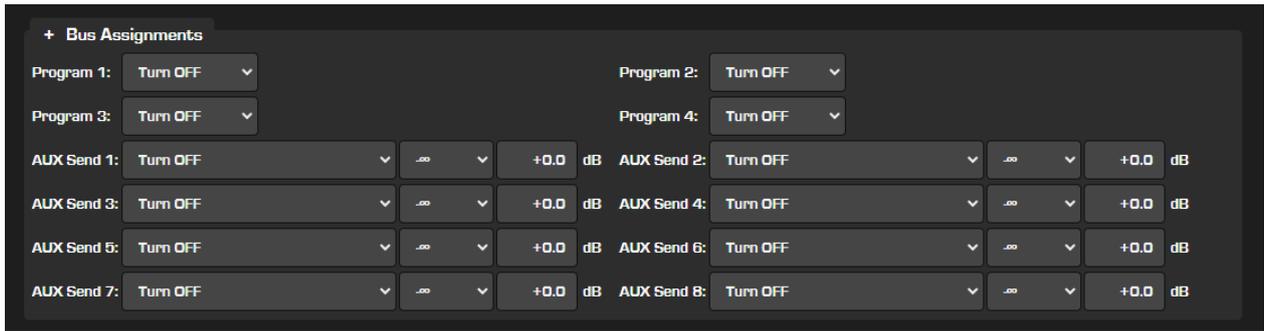
**Note** - It will still be possible to control those functions from the Master Module’s Touchscreen UI. For example, it will still be possible to control PGM bus activation from the CONTROL & BACKFEED tab, or to select a channel by holding the HOME key and pushing one of the Channel Select buttons.

From the **Locked Tabs** section, it is possible to lock selection of any of the channel menu tabs in the Touchscreen UI



The above Locks will prevent the operator from accessing the selected channel tabs from the console’s Touchscreen UI.

## Bus Assignments



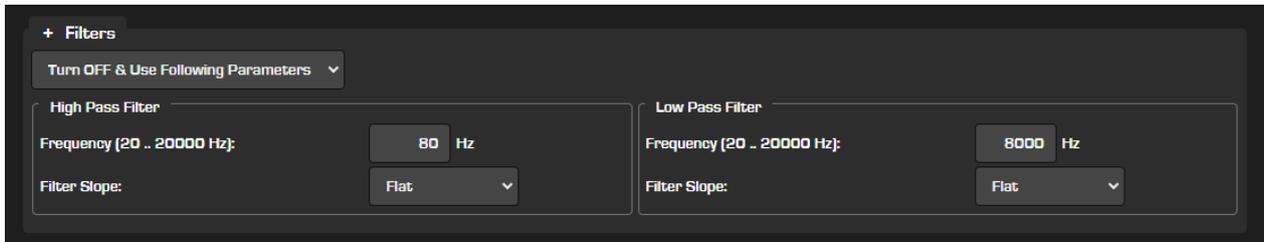
In this section, it is possible to pre-define the activation status of each of the four Program Busses, and each of the eight AUX busses, upon Show Profile load.

- The **Turn ON** and **Turn OFF** options can be used to pre-defined the ON/OFF state.
- The **Retain State** option will leave the current bus status unchanged.

For the eight AUX busses it is also possible to:

- Pre-define any combination of Pre/Post fader, and Pre/Post ON Switch
- Select **-∞** from the drop-down menu to set the bus level to the minus infinite position.
- Select **Use:** from the drop-down menu to set the bus level by a fixed dB amount. Enter the desired level in the next field. You may specify any level between -60.0dB and +10.0dB.

## Filters



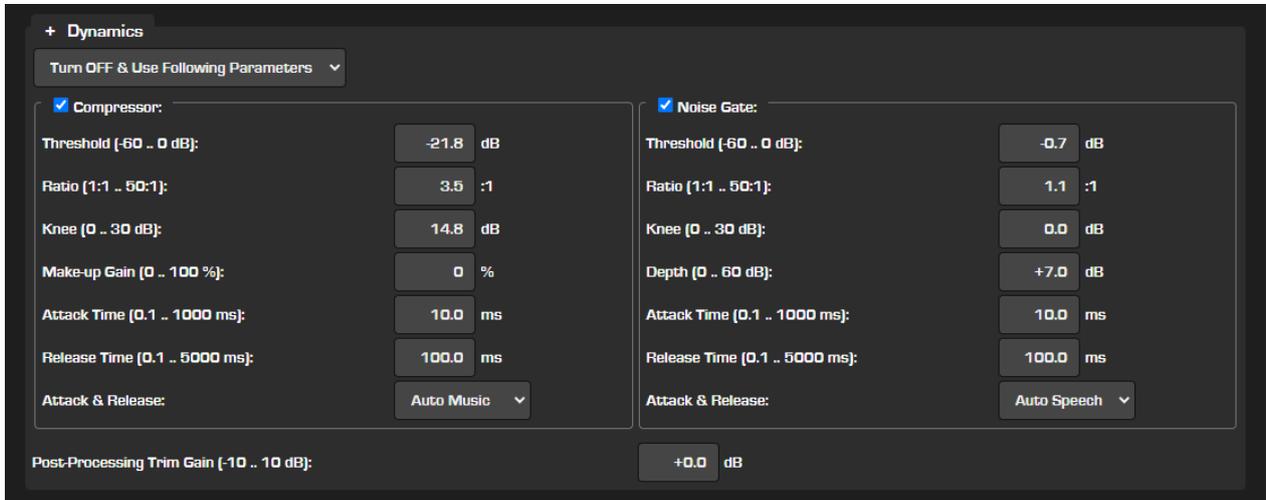
In this section, it is possible to pre-define the activation status and parameter settings of the channel's High-Pass and Low-Pass filters, upon Show Profile load. Three options are available:

- Select **Load from Source Profile** from the drop-down menu to load filter settings from the Source Profile.
- Select **Turn OFF & Use Following Parameters** from the drop-down menu to load the Show Profile with filters in OFF state, and some pre-defined parameters already loaded.
- Select **Turn ON & Use Following Parameters** from the drop-down menu to load the Show Profile with filters in ON state, and some pre-defined parameters already loaded.

## Dynamics

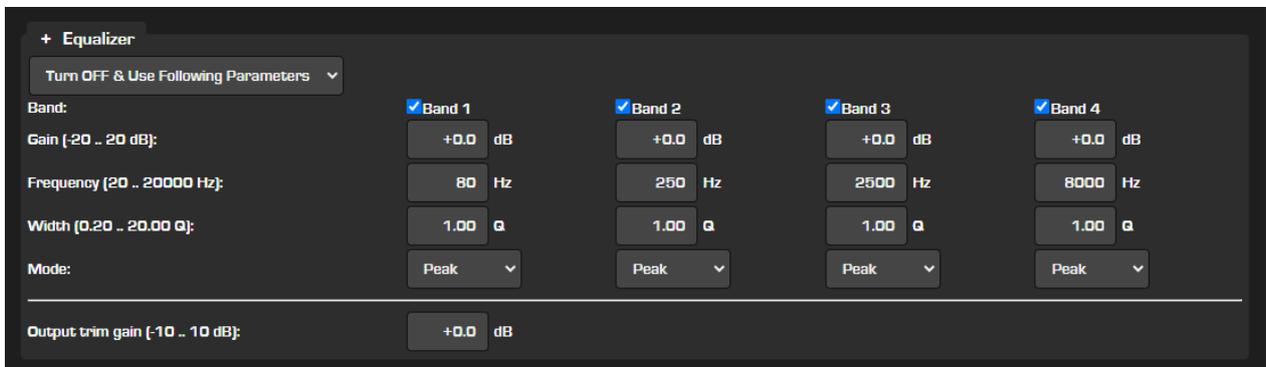
In this section, it is possible to pre-define the activation status and parameter settings of the channel's Compressor/Limiter and Expander/Noise Gate, upon Show Profile load.

The same loading options available in the previous section are also available for this section.



It is also possible to Pre-set the Post-Processing Trim Gain, by entering a fixed dB level. You may specify any level between -20.0dB and +20.0dB.

## Equalizer

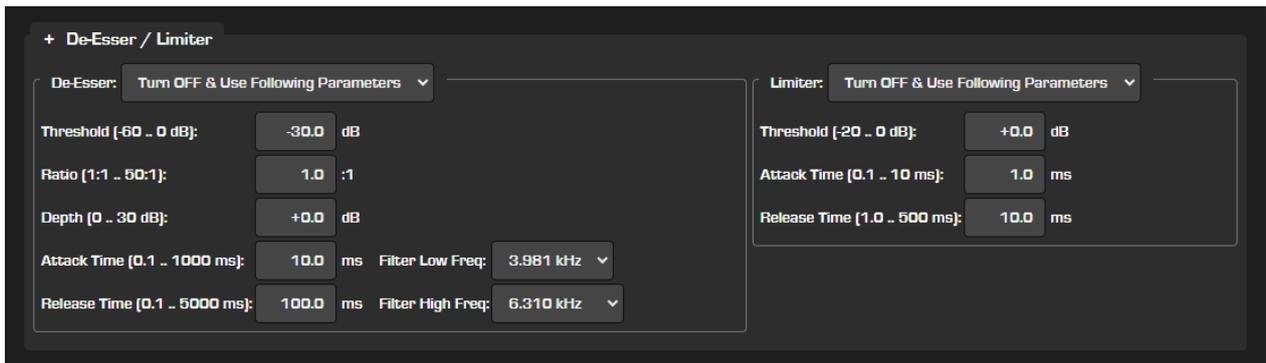


In this section, it is possible to:

- pre-define the activation status and parameter settings of the channel's Equalizer, upon Show Profile load.
- enable or disable each individual band, using the checkbox at the top.
- pre-select each of the bands' mode using the **Mode** drop-down menu at the bottom.
- Pre-set the EQ Output Trim Gain, by entering a fixed dB level. You may specify any level between -10.0dB and +10.0dB.

The same loading options available in the previous sections are also available for this section.

## De-Esser / Limiter

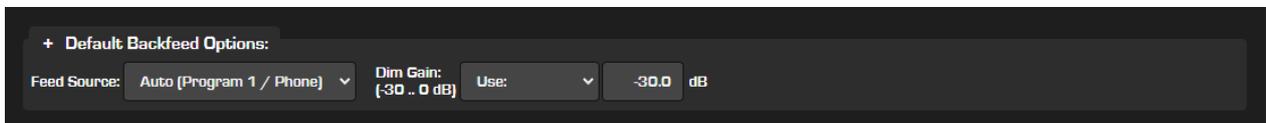


In this section, it is possible to pre-define the activation status and parameter settings of the channel's De-Esser and the Channel Limiter (added in SW v3.0), upon Show Profile load.

The same loading options available in the previous sections are also available for this section.

## Default Backfeed Options

In this section, it is possible to set the default options for the channel Backfeed, upon Show Profile load.



### Feed Source

If the loaded source has a mix-minus, you can choose the source to create the mix-minus from this drop-down menu. Mix-minus is also known as N-1 or N-2 (depending on whether it's stereo or mono)

- **Auto (Program 1 / Phone):** lets the console choose a mix-minus source automatically.
- **Phone, PROGRAM 1-4** and **AUX SEND 1-8** options let you manually choose the bus to use as the basis for the mix-minus.

### Dim Gain

Allows to pre-define the amount of level attenuation when the Talkback is injected into the backfeed.

- Select **OFF** from the drop-down menu to set no pre-defined attenuation
- Select **Use:** from the drop-down menu to set the Dim level by a fixed dB amount. Enter the desired attenuation level in the next field. You may specify any level between -30.0dB and 0.0dB.

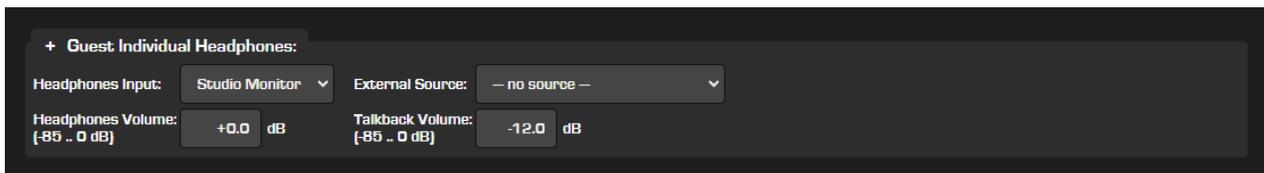
## Guest Individual Headphones

In this section, it is possible to set the default options for the Guest Headphones, upon Show Profile load, providing that the Source is a Guest Microphone type.

Any Quasar input defined as a Microphone source can have a dedicated headphone feed, to facilitate individual Talkback (IFB) to and from the board operator or other talent. The options below affect this dedicated headphone feed, if enabled.

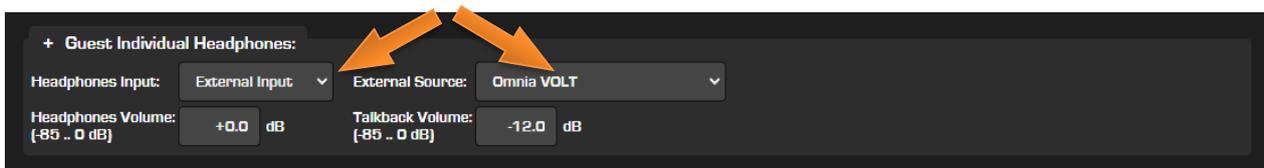
### Headphone Input

From this drop-down menu it is possible to select the Headphone Feed to the Guest Mic position. Any of the internal Quasar busses, or an external Input, can be selected.



### External Source

From this menu you can choose any of the Sources configured in the console, and feed it to the Guest Headphones.



**Note** - The External Input selection will be active only if “External Input” is selected as source for the Headphone Input in the left-side drop-down.

### Headphones Volume

It is possible to specify the level at which the Guest Individual Headphone channel should be fed. You may specify any level between -85.0dB and 0.0dB.

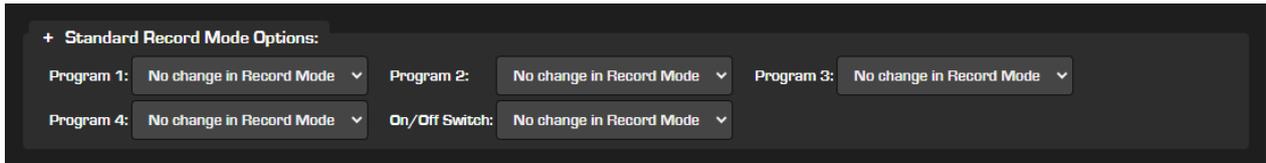
### Talkback Volume (Low Limit)

Here you can specify the initial level at which Talkback will be fed to the Guest Individual Headphone channel.

**Note** - If a Quasar Accessory Module is used to control the same Guest position, the settings from this section will be applied to the Accessory Module when the Show is loaded. Any changes made on the Accessory Module will be saved in this section, when the Show is saved.

## Standard Record Mode Options

Quasar’s Record Mode is a “macro” that allows complex pre-defined operations to take place with a single button press. The options below define bus assignments for the channel that occur when the Standard Record Mode is entered and exited.



**Note** - The options for defining the Flexible Record mode are stored in the Source Profiles

### Program 1 - 4

- **No Change in Record Mode:** This channel's bus assignments do not change when Record Mode is active.
- **Assign in Record Mode:** Assigns this channel to the selected Program bus when Record Mode is active.
- **Remove in Record Mode:** Removes this channel from the selected Program bus until Record Mode is exited.

### On/Off Switch

- **No Change in Record Mode:** Channel ON and OFF keys function normally during Record Mode.
- **Disable in Record Mode:** Locks the buttons, preventing the operator from changing the channel ON/OFF state when Record Mode is active. It also prevents any external controller to change the ON/OFF state.

When done editing the Channel options, be sure to click the **Save Show** button. You will be returned to the Show Profile options screen.

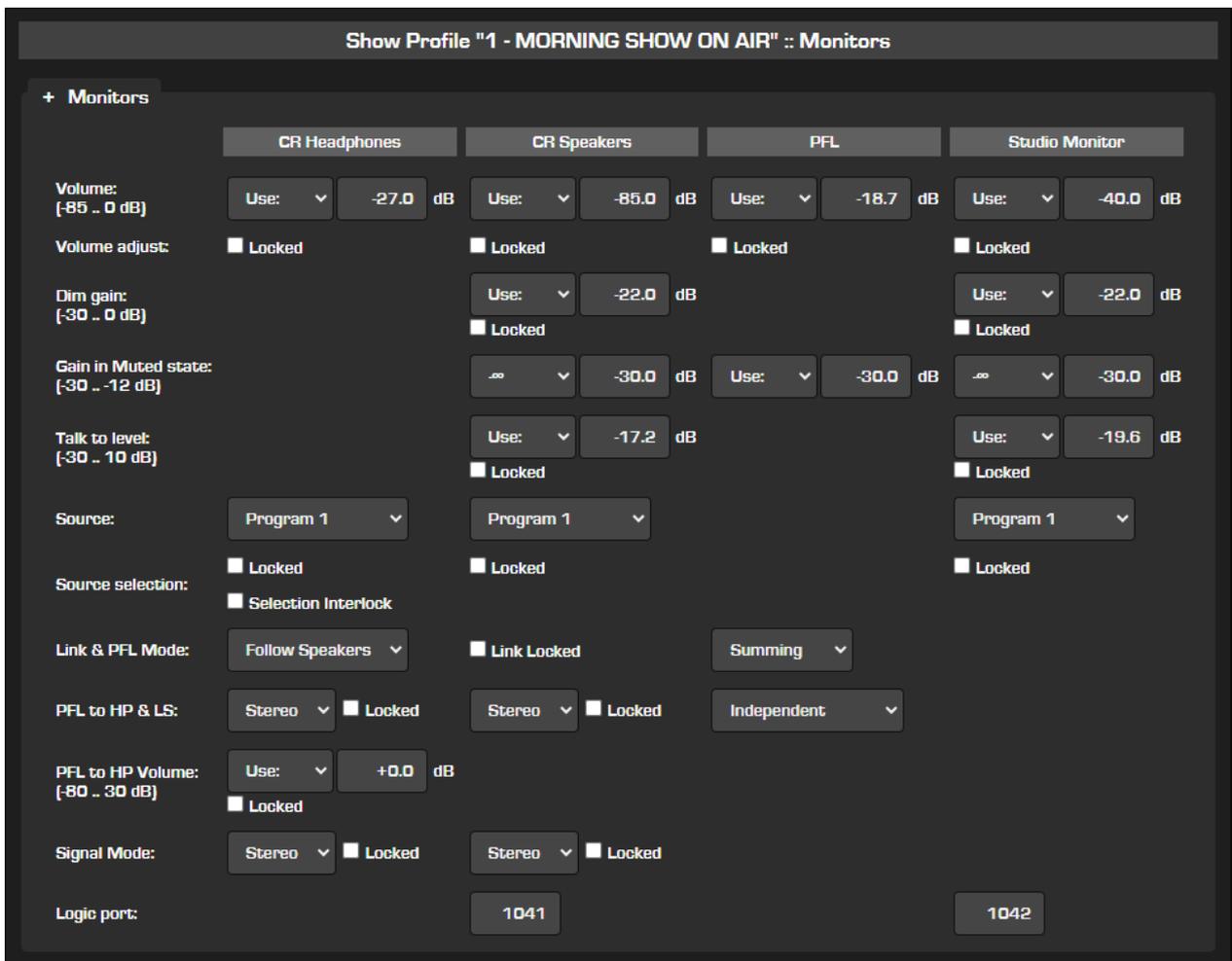
# Monitor Section Configuration Page

The options found in this section let you define the following settings, for each Show Profile:

- the Control Room Speakers monitor settings,
- the Control Room Headphones monitor settings,
- the PFL section settings,
- the Studio Monitors settings,
- the Control Room Headphone Equalizer parameters,
- the Automixer Global settings,
- the External 1-3 Inputs, Talk to CR, and Studio TB Sources.

**Note** - choosing the “Retain Source Setting” option for any item leaves that item unchanged when loading the Show Profile

## Control Room and Studio Monitors



### Volume

- Select **Retain**: to keep the current Volume level unchanged, when the Show is loaded.
- Select **Use**: from the drop-down menu to set the Volume level to a fixed dB value. Enter the desired value in the next field. You may specify any level between -85.0dB and 0.0dB.

## Volume Adjust

You can essentially prevent any Volume Level on the board from being changed if you want. If desired, these options can be flagged to Lock the console's Volume pots, preventing the operator from changing the Show Profile's pre-selected Headphones and Monitor levels.

## Dim Gain

- Select **Retain:** to keep the current Dim Gain level unchanged, when the Show is loaded.
- Select **-∞** to enter a muted state when Dim is active.
- Select **Use:** from the drop-down menu to set the Dim level to a fixed dB amount. Enter the desired attenuation level in the next field. You may specify any level between -30.0dB and 0.0dB.

## Gain in Muted State

- Select **Retain:** to keep the current Volume level unchanged, when the selected Monitor is in a Muted State.
- Select **-∞** from the drop-down menu to completely mute the selected monitor when this state is entered.
- Select **Use:** from the drop-down menu to set the Volume level to a fixed dB value for the muted state. Enter the desired value in the next field. You may specify any level between -85.0dB and 0.0dB.

**Note** - Normally, turning a Control Room microphone channel ON mutes the PFL speakers entirely. This setting allows you to let the PFL bus be heard in the Control Room at a reduced level, even while CR Mics are active.

## Talk to Level

- Select **Retain:** to keep the current Talk to CR, or Talk to Studio levels unchanged, when the Show is loaded.
- Select **Use:** from the drop-down menus to set the level of the Talk to CR, or Talk to Studio channel to a fixed dB value. You may specify any level of cut or boost between -30.0dB and +10.0dB.

Dedicated locking options are available for each of these controls.

## Source

This drop-down allows you to select one of the console busses, or External Inputs, and assign it to the corresponding Monitor section. This will let you pre-define what Source the operator will be monitoring, when the Show is loaded.

## Source Selection (Control Lock Map)

You can essentially prevent any Monitor Source on the board from being changed by the operator if you want.

If desired, the three **Locked** checkboxes can be flagged to lock each of the monitor Source selectors, preventing the operator from changing the Show Profile's pre-selected options.

Select the **Selection Interlock** checkbox to prevent clearing the monitor selection when the corresponding button (in the monitor section of the console) is pressed twice by the operator.

### Link and PFL Mode

- Select **Retain** from the drop-down menu to keep the current Monitor Link status, when the Show is loaded.
- Select **Follow Speakers** to link the CR HP to the CR Speakers selection
- Select **Independent** to make the CR HP selection independent from the CR Speakers selection.

A dedicated locking option is available for the monitor Link control.

### PFL to HP & LS (PFL to Headphones and Loudspeakers)

- Select **Retain** from the CR Headphones or CR Speakers drop-down menu to keep the current PFL monitoring status, when the Show is loaded.
- Select **Off** to disable the PFL to Loudspeakers and PFL to Headphones selection.
- Select **Split** to retain the current selection on the left channel, and send a mono sum of the PFL bus to the right channel of the selected monitor section.
- Select **Independent** from the PFL drop-down menu, to leave the PFL level independent from the Loudspeaker Level.
- Select **Link Volume to LS** to have the PFL always linked to the Loudspeakers.

### PFL to HP Volume:

- Select **Retain:** to keep the current Volume level unchanged, when the Show is loaded.
- Select **Use:** from the drop-down menu to set the Volume level to a fixed dB value. Enter the desired value in the next field. You may specify any level between -80.0dB and 30.0dB.

### Signal Mode

- Select **Retain** from the drop-down menu to keep the current Monitor Mode status, when the Show is loaded
- **Stereo** sets the Monitor channel to Stereo.
- **Left / Right** takes the chosen side of the Monitor channel and sends it to both speakers.
- **Sum** Creates a sum of both sides of the Monitor channel and sends it to both speakers.

### Logic Port

This is a bi-directional GPIO port that can be used to receive and/or trigger commands: Being a port, it has 5 Inputs (for receiving commands) and 5 Outputs (for indicating events).

#### Console Receive side :

- Mute CR Speaker (pin #11)
- Dim CR Speaker (pin #12)
- Trigger EXT PFL (pin #13)
- Trigger Talk to EXT (pin #14)
- Trigger ON AIR signal on the console TS GUI (pin #15)

#### Console Send side :

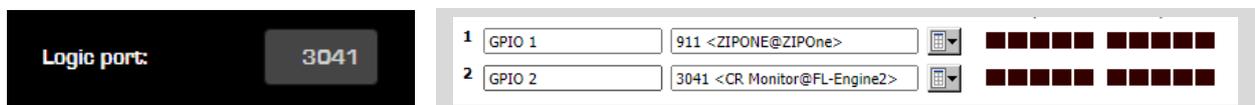
- CR Mic Live (pin #1)
- CR SPK DIM Active State indication (pin #2)

- PFL/AFL/EXT Active State indication (pin #3)
- Talk To EXT Active State indication (pin #4)
- Talk to CR Active State indication (pin #5)

You could use any arbitrary number here, providing that the same number is used on the other end, to identify the correct GPIO port of your node. It is good practice to use the CR monitor LW channel number, so that the IDs can be easily identified, and possible conflicts are avoided.

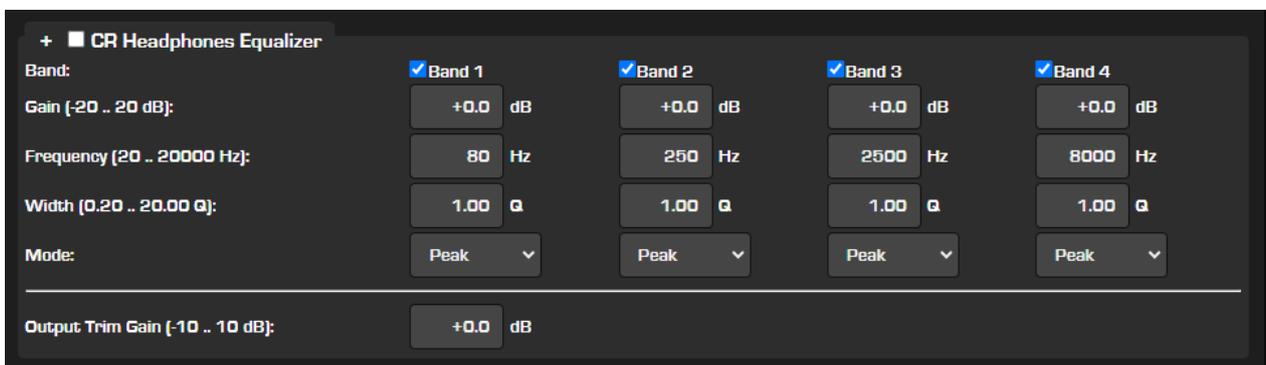
If you associate this Logic port to a HW port on any Axia device, the physical wiring on that HW port will determine which GPIO pin # is triggered, and therefore which command is activated in the console, and which indication is received from it.

The example below shows how to link the Console's Logic port to a GPIO port on an xNode.



**Tip** - The Logic port can be used in conjunction with the EXT PFL Source selection to trigger the injection of an external signal into the PFL bus, to interface the console with an intercom system. This option should typically be programmed with the same logic channel number for all Show Profiles.

## CR Headphones Equalizer



Check this option box to apply a 4-Band Fully Parametric EQ to the CR Headphones feed using the following settings:

### Band Active

Check this option box to engage each individual band.

### Gain

Use this field to enter a Gain cut or boost value between -20.0 and +20.0 dB to the selected frequency band.

## Frequency

Use this field to enter the center frequency of the selected EQ Band. You may specify any level between 20Hz and 20.0kHz.

## Width

Use this field to adjust the size, or width (Q factor ) of the selected band. You may specify any Q factor between 0.20 and 20.0.

## Mode

**Peak/Low/High Shelf:** determines the type of filter applied to that band

## Output Trim Gain

Enter a fixed dB level to offset the output of the Equalizer. You may specify any level between -10.0dB and +10.0dB.

## Externals



### External 1-3

This drop-down menu allows to select any Source with the **External** flag active (in the Source availability menu) and assign it to one of the three External Inputs of the Monitor Section.

### Ext PFL

This drop-down menu allows to select any Source with the **External** flag active (in the Source availability menu) and inject it (by summing it, not switching it) into the PFL channel.

**Note** - this Input is enabled by GPI pin #13 on the CR Monitor GPIO. Please see Configuration Chapter 7, for obtaining the wiring diagram pinout.

Example #1: you have an intercom system that you wish to feed into the console's PFL bus. Use the drop-down to select the intercom's audio source, then take the intercom's GPO and use it to gate open the External PFL input, which would be fed by the intercom audio. The GPO channel # will have to be entered in the **Logic Port** field of the CR Monitor section.

Example #2: you have a playout system you wish to feed into the console's PFL bus. Use the drop-down to select the Playout Computer's PFL output channel, then take the PC's GPO and use it to gate open the External PFL input, which would be fed by the PC PFL audio. The GPO channel # will have to be entered in the **Logic Port** field of the CR Monitor section.

## Talk to CR

This drop-down menu allows to select any Source with the **External MONs** flag active (in the Source availability menu) and permanently mix it to the Talk to CR bus. Summing of the external Source is done without dimming, and without activation switch. The audio will be directly mixed with the bus if a Source is assigned from the drop-down. This function is useful when the user wants to make sure he will always hear an intercom feed, no matter what his monitor levels are, for example.

## Studio TB

This drop-down menu allows any Operator Mic type Source to be used as Talkback Microphone in the CR Room, to talk directly to the Studio Guests when the console's **TALK** key is pressed. You do not need to assign the Operator's Microphone to one of the console channels. When done editing the Monitor Section options, be sure to click the **Save Show** button. You'll be returned to the Show Profile options screen.

# Aux Masters Configuration Page

Aux Masters are the main Auxiliary Output Busses of your Quasar. There are eight Aux Send busses and 2 Aux Returns: Aux Returns are two Stereo Input channels (they are not busses) that come handy when you don't want to consume faders on your surface for mixing the FX returns back into the Program. Imagine them as if they were "short" input channels, without any DSP processing feature.

## Aux Master Sends 1-8



### On/Off Status

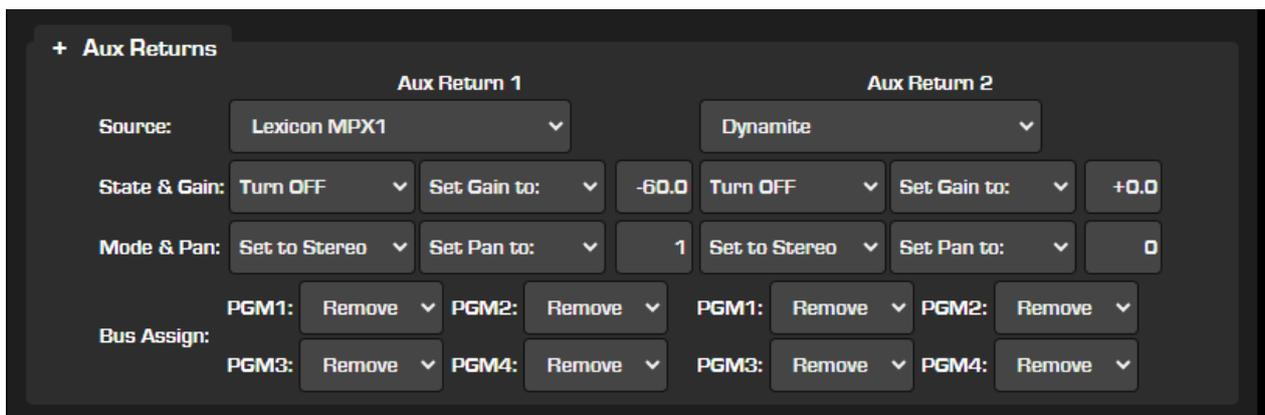
- Select **Retain State**: if you want the On/Off state to remain unchanged when the Show profile is loaded
- Select **Turn OFF**: to switch off the selected Aux Send bus upon Show Profile load
- Select **Turn ON**: to activate the selected Aux Send bus upon Show Profile load.

### Aux Send 1-8 Master Gain

- Select **Set Gain to:** from the drop-down menu to specify the master gain level at which the selected Master Aux bus will be set. This is the gain applied to the bus after any sources have been assigned to it. You may specify any level between -60.0dB and +10.0dB.
- Select **Retain Gain** if you want the Aux Master send Gain value to remain unchanged when the Show profile is loaded.

**Note** - "Retain" always refers to keeping the current console state, and NOT to keeping settings from the previous Source Profile.

## Aux Master Returns 1-2



### Source

This drop-down lets you quickly assign a Source to the selected Return channel, from a list of configured Source Profiles.

### State & Gain

- Select **Retain State:** if you want the On/Off state to remain unchanged when the Show profile is loaded
- Select **Turn OFF:** to switch off the selected Aux Return channel upon Show Profile load
- Select **Turn ON:** to activate the selected Aux Return channel upon Show Profile load.
- Select **Retain Gain:** if you want the Gain level to remain unchanged when the Show profile is loaded
- Select **Set Gain to:** from the drop-down menu to specify the gain level at which the selected Master Aux Return channel will be set upon Show load. You may specify any level between -60.0dB and +10.0dB.

### Mode & Pan

- Select **Retain Mode:** if you want the signal mode to remain unchanged when the Show profile is loaded.
- Select **Set to Stereo:** to set the bus signal mode to Stereo.
- Select **Set to Left / Right:** to take the chosen side of the bus' stereo signal and sends it to both sides of the stereo bus output channel.
- Select **Set to Sum:** to create a sum of both stereo channels and send it to both sides

of the stereo bus output channel

- Select **Retain Pan:** if you want the channel Pan to remain unchanged when the Show profile is loaded.
- Select **Set Pan to:** from the drop-down menu to Pan the output signal of the Aux Return channel to the left or right.

### Bus Assign

- Select **Retain:** if you want the Assignment state to remain unchanged when the Show profile is loaded
- Select **Assign:** to assign the output of the specified Aux Return channel to the selected Program bus(es) upon Show Profile load
- Select **Remove:** to remove the output of the specified Aux Return channel from the selected Program bus(es) upon Show Profile load.

**Tip** - When done editing the Aux Send & Return options, be sure to click the **Save Show** button. You'll be returned to the Show Profile options screen.

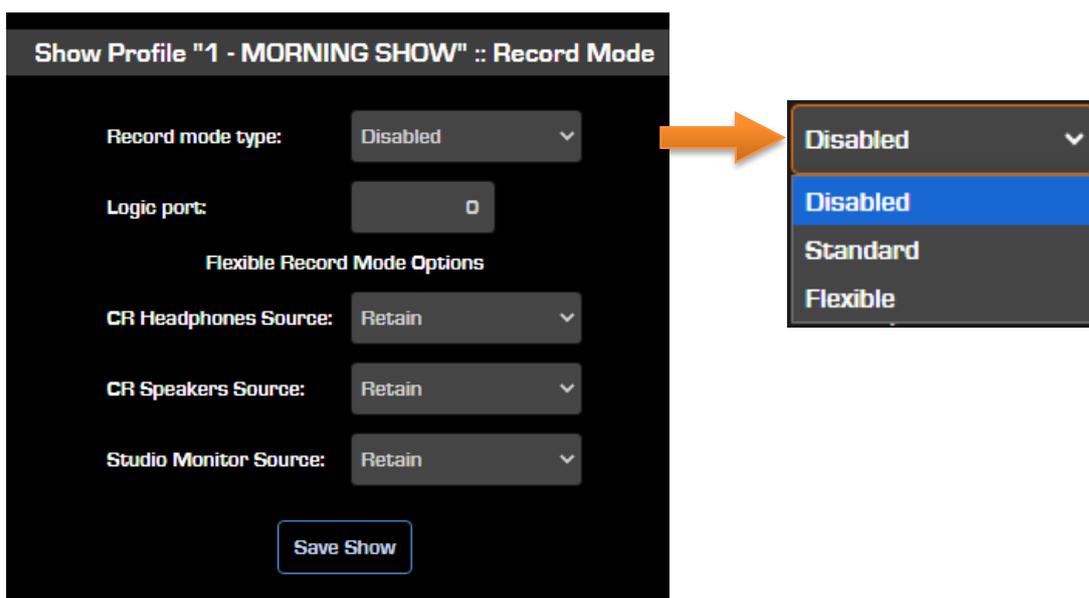
## Record Mode Configuration Page

From this section of the Show Profiles Web UI it is possible to set the desired type of Record Mode. Some options allow you to customize the Monitor and Headphone settings that are automatically selected when Flexible Record Mode is entered. The initial (original) Show Profile settings will be loaded upon exiting the Record Mode.

They primarily affect the behavior of Monitors when the Record Mode is activated, while Sources assigned to Program-4/Record will follow the Record Mode options in their own Profile (see previous section of this chapter - Source Profiles).

### Record Mode type

- **Disabled:** Disables Record Mode entirely for this Show Profile.
- **Standard:** Allows activation of Standard Record Mode for this Show Profile. CR Monitor and CR Headphone assignments automatically switch to the Program 4 bus, and the bus assignment keys for channels assigned to Program 4 flash.
- **Flexible:** Allows activation of Flexible Record Mode, with custom Monitor, Headphone and Meter options set in the “Flexible Record Mode Options” section that follows.



### Logic Port

In this field you can enter the GPIO channel number used to trigger your dedicated recording device. This option should typically be programmed with the same logic channel number for all Show Profiles. Once a number will be entered here, the same number will have to be entered on the “other end” of your GPIO chain, in the desired port of the GPIO node that is connected to the recorder.

**Note** - The Logic Port is only used when a HW GPIO needs to be sent to a device for triggering recording. Start, Stop pulses and ON/OFF indication lamps are supported.

For more detailed information about configuring GPIO for the Record Mode, please refer to Configuration Chapter 7 of this Manual: [GPIO Record Mode Logic](#).

The following options apply only to the Flexible Record Mode:

### CR Headphones Source

- Select **Retain**: to keep the CR HP Monitor feed unchanged, when the Record Mode is engaged. This option keeps the CR HP selection the board operator was using before entering the Record Mode.
- Select **Program 1 - 4, Record, Phone, Auxiliary 1 - 8, External 1 - 3**: to change the CR HP Monitor feed to the selected bus or Monitor channel when Record Mode is engaged.
- Select **Recall**: to re-load the Monitor selection that was manually selected by the board operator *the last time* Record Mode was active.

### CR Speakers Source

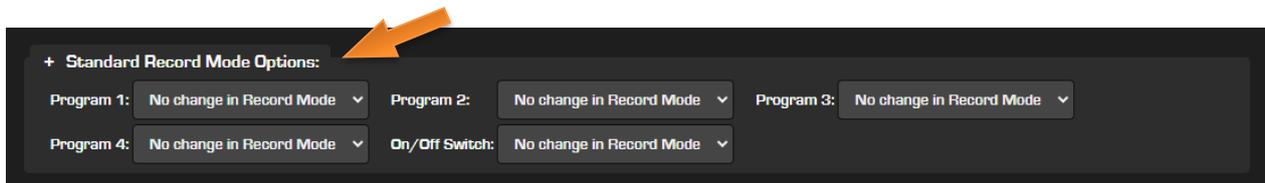
The same options as described in the previous menu are available for the Control Room Speakers section.

### Studio Monitor Source

The same options as described in the CR HP menu are available for the Studio Monitor section.

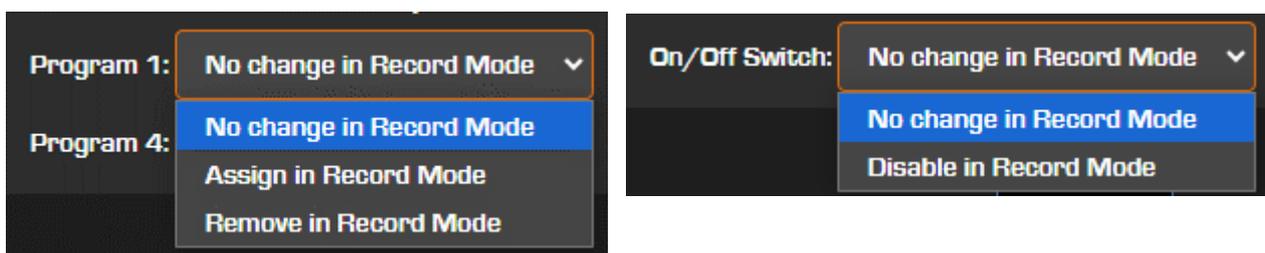
## Configuring Show Channels for Standard Record Mode

Navigate to the desired Show Profile, and select one of the Channels. Then scroll down the bottom of the page, to the **Standard Record Mode Options**: section.



Each of the Show Channels can be configured to :

- Activate different PGM bus assignments
- Enable or Disable control the Channel ON/OFF switches



For more detailed information about configuring the Channel Record Mode options, please refer to the Configuration Chapter 2 of this manual: [Standard Record Mode Options](#).

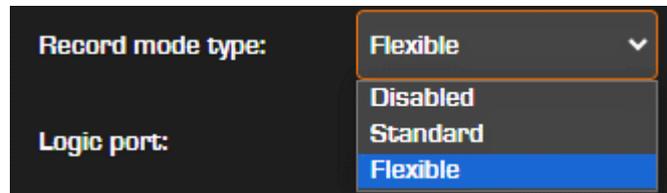
## Controlling the Record Mode via LWCP

Two parameters associated to the Record Mode can be read via LWCP:

- Activation
- Enable

Only one of these parameters can be written, or set via LWCP: Activation.

The Enable parameter is internally saved in the show, and can be either “Standard”, or “Flexible”. So, it can only be read.



### Controlling Record Mode Activation

Activation is a function that can be executed in 2 ways:

1. By pushing the RECORD MODE key on the console
2. By sending a command through LWCP

property name: **recmode\_stat**

examples:

SET AppControl recmode_stat=ON	(turns Record Mode ON)
SET AppControl recmode_stat=OFF	(turns Record Mode OFF)
GET AppControl recmode_stat	(queries the state)
INDI AppControl recmode_stat=OFF	(console's reply to the GET query)

### Reading the Record Mode Enable state

Enable is a read-only parameter saved in Show profile, in the "Record mode type" field

property name: **recmode\_enbl**

examples:

GET AppControl recmode_enbl	(queries the state)
INDI AppControl recmode_enbl=OFF	(console's reply to the GET query)

**Note** - LWCP is case sensitive - use the exact same syntax shown above.

## Phone Control Configuration Page

The Phone Control is used when setting up a Telos talkshow system for use with your console. Here you can find the global, per-show configuration of your Phone system. Two different menus are available, for selecting and configuring the desired Phone System type: Telos VX (left side) and AP-type, Nx12 compatible systems (right side).

**Show Profile "1 - MORNING SHOW ON AIR" :: Phone Control**

No Phone Control

Telos Vx

AP (Nx12)

**Phone Server IP:**  
format: [user[:pass]@]host

**Phone Server IP:**  
format: [user[:pass]@]host

**Studio Name:**

**Show Name:**

**Show Name:**  
leave blank to use current

**Show Host Name:**

**Show Password:**

**Hybrid mode:** Normal

This is due to the different communication protocols used by these two classes of systems.

## Telos VX

### Phone Server IP

Click on the drop-down menu, to view a list of available connected Telos VX Engines and select the desired VX unit. Alternatively, you can enter here the IP address of your Telos VX call server. In case user and password are configured for this device, use the syntax:  
*username:password@ipaddress*

### Studio Name

This field is used to identify the Studio configuration when using a Telos VX system. These names are defined in your VX system's setup. Click on the drop-down menu, to view a list of configured Studios, within the selected VX Engine, and select the desired Studio Name. Alternatively, you can manually enter the configuration name you wish to log into.

### Show Name

This optional field is for the VX Show names you wish to load, while using your Quasar. Click on the drop-down menu, to view a list of available Shows, within the selected VX Engine, and select the desired Show Name.

**Tip** - with the VX system, this field can be left blank, permitting the changing of the show assigned with a VSet phone, or the VX interface.

## Telos AP

Prior to the VX system, the AP protocol was used. The AP option is to support legacy Telos products still in use, otherwise you would likely select the Telos VX option.

### Phone Server IP

You can enter here the IP address of your legacy Telos Phone System. In case user and password are configured for this device, use the syntax: *username:password@ipaddress*

### Show Host Name

This field is used to log into the appropriate configuration:

- “Hybrid1”, “Hybrid2”, “Hybrid1&2” (used in **TWO-x-12**)
- “Hybrid 1&2”, “Hybrid 3&4”, “Hybrid 1-4” (used in **Nx6 & Nx12**)

### Show Password

This field is used to permit access to the Call system that has password protection at show levels. If you have a password for the phone system Show you want to use, enter it here.

## Hybrid Settings

### Hybrid Mode

This option applies to both Phone System types and swaps the banks of the Call Controller. Normally, the left column controls Hybrid 1, and the right column controls Hybrid 2; selecting the reverse option will reverse the order.

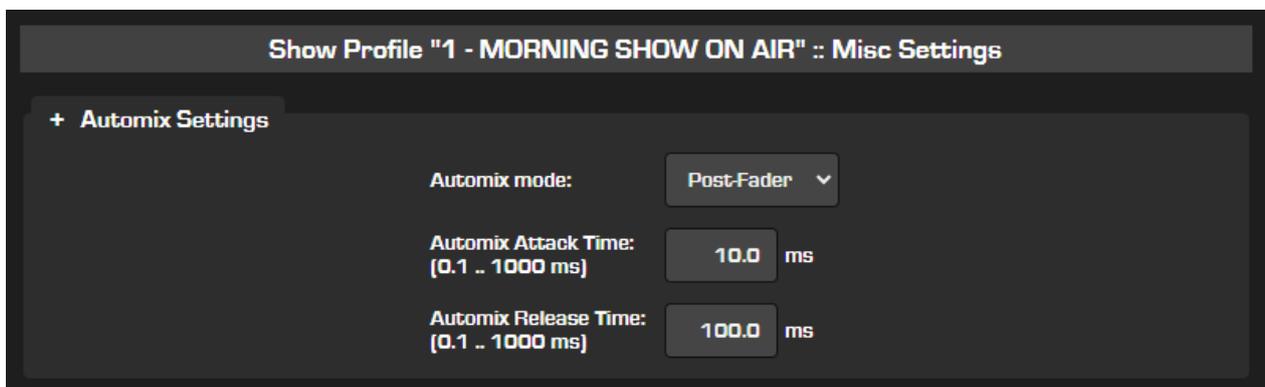
Please refer to Configuration Chapter 3 - [Setting up Phones](#), for more details.

## Miscellaneous Options Configuration Page

From this section of the Show Profiles Web UI, it is possible to set some global parameters related to the Automixer, Touchscreen GUI, Program Bus activations, and Backfeed gain.

### Automix Settings

This section allows to predetermine the Global Parameters for the entire Automixer:



### Automix Mode

This drop-down menu allows to pre-set the Pre/Post fader position of the Automix gain control stage.

### Automix Attack Time

Enter a fixed ms value to set the Global Automix Attack Time. You may specify any value between 0.1ms and 1000.0ms.

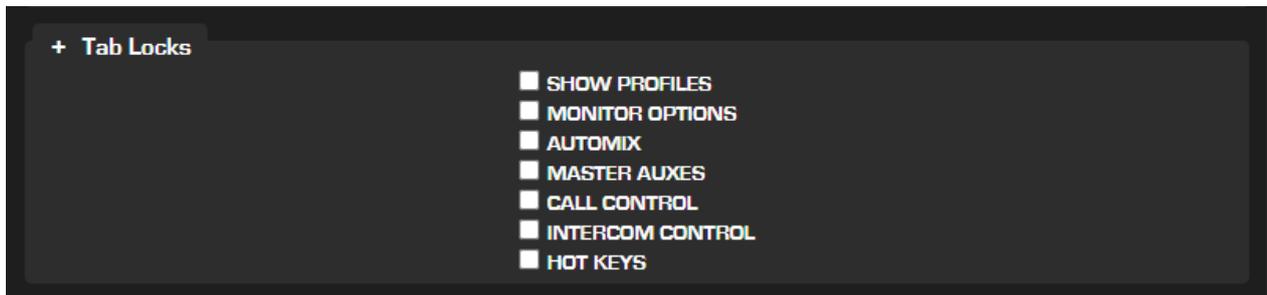
### Automix Release Time

Enter a fixed ms value to set the Global Automix Release Time. You may specify any value between 0.1ms and 1000.0ms.

The individual Channel Parameters, (such as Activation and Weight) can be pre-set in the Show Profiles Main Page.

## Tab Locks

This section allows to lock each of the Master Menu Tabs, except the Home Tab. Once locked, they will appear greyed out in the Touchscreen GUI, and the user will not be able to select them. This is useful in case access to some critical parts of the console configuration wants to be protected by the System Administrator.



**Warning** - If a Show saved with the "Show Profiles" Tab locked is loaded, the operator will be left without the ability to load any other Show. But he will still be able to reload the same show. So the only way to regain the possibility of loading Shows, is to remove the Tab lock from within the web UI, save the Profile, then press and hold the hardware **PROFILES** key on the console until the show is reloaded, and the Show Profiles tab will be unlocked.

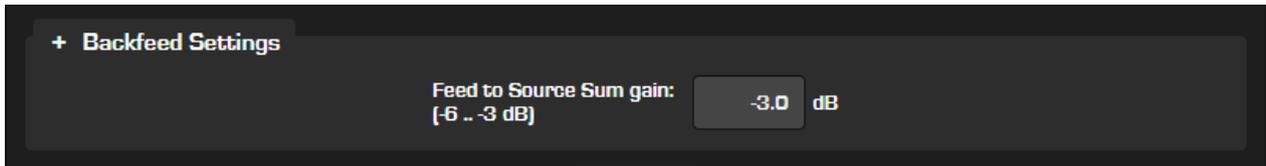
## Prevent Program Deselection when Channel is ON



## PGM 1 - 4

This section allows to protect each of the PGM busses while the console channels are active. It lets you prevent the bus from being de-selected (turned OFF) when the corresponding channel is ON. This safety feature is applied to all channels in the Show Profile.

## Backfeed Settings



### Feed To Source Sum Gain

This field allows to enter a gain compensation for mono-sum of Talkback audio sources. You may specify between -6 dB and -3 dB of attenuation. This functionality helps keeping your levels consistent when you're summing content to mono, over a backfeed. In case of dynamic Talkshow-type speech, it can be useful to set a -6dB compensation. For all other use cases, better to use -3.0dB.

# Configuration 3 – Phone Control

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## Setting up Phones

One of the advantages of a Livewire+ studio is the smooth integration of Telos telephone interface equipment with the mixing console. Console telephone control modules let operators work phone without taking their eyes or hands off the console; mix-minus is handled automatically, and each fader has its own mix-minus capability, so you will never run out of busses. Since advanced Telos telephone interfaces have Ethernet connections, they integrate easily with Axia networks, exchanging control signaling between console and phone system without the need for the usual parallel connections.

This Chapter describes how to configure Sources in Quasar, for use with Telos talkshow systems, and how to control these directly from your Quasar using the integrated soft Call Controller module. In case you don't own a Telos phone system, GPIO control of telephone equipment without a Livewire interface is also possible; refer to the GPIO chapter for details.

## Phone Control Types

There are three methods of setting up phone control with the Quasar console:

- **EU Phone** (networked)
- **US Phone** (networked)
- **No Phone Control** (GPIO enabled control)

**EU Phone** is the method most used in European countries, wherein a single line is assigned to a single hybrid. This method requires configuration of the Source Profiles only. You do not need to configure the Phone Control options in the Show Profile Menu. This is also known as “hybrid per fader” mode; that is, there is no switching between multiple lines on a single hybrid, but a 1:1 direct mapping, and each line is presented on its own dedicated console fader. So, for example, if 4 phone lines are needed, each line's hybrid is presented on a separate fader. This mode does not make use of the full capabilities of the CALL CONTROL tab. Only the Numeric Keypad will be active, while the hybrids' lines can be controlled with the SET and HOLD functions available on each channel strip.

**Note** – The SET and HOLD keys must have their function programmed in the web UI of the desired Fader Module.

**US Phone** is the method most common in North America, where the operator has the ability to choose between multiple incoming lines to feed the telephone hybrid. Commonly, two hybrids are presented on separate faders, and the user dynamically switches between incoming lines. The Quasar CALL CONTROL tab is used for this mode of operation, allowing for full 8-Line selection and switching functionalities. This method requires configuration to be entered in the Show Profiles and in the Source Profiles. Most Telos multi-line call systems support both of these two methods.

**No Phone Control** is used for controlling Telos Hx1 and Hx2 products, or third-party hybrids which do not support Telos control protocols. In this second case, basic GPIO closures are used to “take” and “drop” lines.

## EU Phone Operation Setup

In this case you only need to create a Source Profile for each of your fixed line hybrids. In legacy Telos Phone systems, you may have 1, 2, or even 4 hybrids; with VX systems, the number may be many more.

### Source Profile Settings for EU Phone Operation

1. Create a new Source Profile, and select the **Phone Source** type.
2. In the Stream I/O Settings section, select the appropriate **Fader Input** Source using the adjacent drop-down Source Selector box to pick the desired Livewire channel from your Telos phone system.
3. Below in the Source Profile, look for the **EU Phone** section in the **Phone Control** area.

The screenshot shows the 'Phone Control' settings panel. At the top, there is a '+ Phone Control' header and a radio button for 'No Phone Control'. Below this, there are two main sections: 'EU Phone' and 'US Phone Call Controller'. The 'EU Phone' section is active and contains a 'Server IP' field with the value '192.168.2.104'. Underneath, there are three radio buttons: 'Telos 2', 'AP (Nx12)', and 'Vx'. The 'Vx' radio button is selected. The 'Telos 2' section has a 'Line' field with '0'. The 'AP (Nx12)' section has a 'Use 2<sup>nd</sup> show (split)' checkbox, a 'Line' field with '0', and a 'Hybrid' field with '1'. The 'Vx' section has a 'Studio name' field with 'EURO Style' and a 'Fixed Hybrid' field with '1'. The 'US Phone Call Controller' section has a 'Hybrid' field with '0', a 'Fixed Line' field with '0', and a 'Mashing allowed' checkbox.

For all the three EU Phone options, you must enter the IP Address of the Telos phone system in the **Server IP** field.

4. Then select the Telos phone system you're interfacing with. Choose from **Telos 2**, **Nx-Series** (shown as *AP (Nx12)*) or **VX** and enter the appropriate settings for the device:
  - If you're using a Telos TWO, enter the number of the line to be used in the **Line** field.
  - If using an Nx system, enter the Line and Hybrid numbers in their respective fields. Select the **Use 2<sup>nd</sup> Show** checkbox if split shows are enabled on your Nx system. In the **Hybrid** field, enter the hybrid number you want this Source Profile to be associated with.
5. If using a VX system, enter the **Studio Name** as configured in your VX Engine, and the number of the Fixed Channel (this is the VX terminology) in the **Fixed Hybrid** field.

This corresponds to the Channel ID number found in the Studio Information page of the VX Web UI.

6. When finished, remember to save your Source Profile. You do not need to configure your Shows for this workflow.

## Telos VX settings for EU Phone operation

In case you're setting up EU Phone Sources with a Telos VX System, and you would like to obtain a "Hybrid per Fader" setup, the VX Studio configuration will drive the entire workflow:

In VX, you will need to:

1. Create a **Studio** with the desired number of *Fixed Channels*. Typically, you would create a Channel for each "hybrid" you want to have on the console, and assign their Livewire channels to each EU Phone Source on your Quasar. These channels, in practice, will work as your fixed hybrids.
2. Create a **Show** with the desired number of SIP Lines you want to associate to each hybrid on your console.

Let's consider an example of how a Studio can be configured in VX to work with a EU-style workflow which uses 4 fixed lines on 4 faders.

The Web UI of the VX Show associated to this Studio could look like this:

**TelosVX Control Center**

### EURO Style Configuration

Studio is using show "4-Lines Show"

Current Show: 4-Lines Show

Studio Name	Fixed channels	Selectable channels	Auto Answer Fixed Lines	Lockless Conferencing
<input type="text" value="EURO Style"/>	<input type="text" value="4"/>	<input type="text" value="0"/>	<input type="checkbox"/>	<input type="checkbox"/>

**Fixed Channels:**

Enable	Name	Channel	Advanced Receive
<input checked="" type="checkbox"/>	<input type="text" value="Studio2: Fix 1"/>	<input type="text" value="10403"/> <input type="text" value="Live Stereo"/> <input type="button" value="v"/>	<input type="checkbox"/> <input type="text" value="auto"/> <input type="text" value="To Source"/> <input type="button" value="v"/>
<input checked="" type="checkbox"/>	<input type="text" value="Studio2: Fix 2"/>	<input type="text" value="10404"/> <input type="text" value="Live Stereo"/> <input type="button" value="v"/>	<input type="checkbox"/> <input type="text" value="auto"/> <input type="text" value="To Source"/> <input type="button" value="v"/>
<input checked="" type="checkbox"/>	<input type="text" value="Studio2: Fix 3"/>	<input type="text" value="10405"/> <input type="text" value="Live Stereo"/> <input type="button" value="v"/>	<input type="checkbox"/> <input type="text" value="auto"/> <input type="text" value="To Source"/> <input type="button" value="v"/>
<input checked="" type="checkbox"/>	<input type="text" value="Studio2: Fix 4"/>	<input type="text" value="10406"/> <input type="text" value="Live Stereo"/> <input type="button" value="v"/>	<input type="checkbox"/> <input type="text" value="auto"/> <input type="text" value="To Source"/> <input type="button" value="v"/>

**Selectable Channels: None**

## 4-Lines Show Configuration

Show is in use by studio(s) "EURO Style"

Show Name

Lines

4-Lines Show

4

### Lines:

Position	Name	Extension	Server	Config	Fader	Busy All?	Ringer
1	VX Line 1	1	192.168.2.104 ▼	»	Fixed #1 ▼	<input checked="" type="checkbox"/>	Default ▼
2	VX Line 2	2	192.168.2.104 ▼	»	Fixed #2 ▼	<input checked="" type="checkbox"/>	Default ▼
3	VX Line 3	3	192.168.2.104 ▼	»	Fixed #3 ▼	<input checked="" type="checkbox"/>	Default ▼
4	VX Line 4	4	192.168.2.104 ▼	»	Fixed #4 ▼	<input checked="" type="checkbox"/>	Default ▼

Apply

**Note** - These instructions presume that the VX system has been configured properly, has extensions registered with a SIP server, Shows and Studios have been configured, is present on the Axia Ethernet network, and is running up-to-date software.

## US Phone Operation Setup

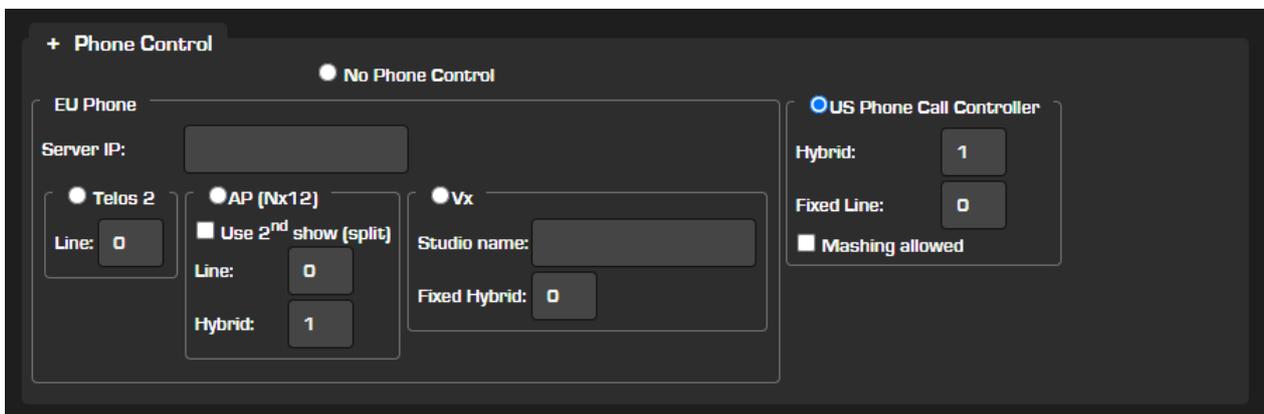
Using the US Phone method of operation requires the use of the Quasar Call Controller: this soft module lets you control up to 8 incoming lines from 2 faders, in a highly flexible way. With this method, you can use Quasar’s Show Profiles feature to instantly recall show setups that choose between different phone systems, or even different phone system configurations. For example, one Show Profile could call up the configuration needed for a Talk format, while another might recall the configuration needed for music request lines, allowing a single studio to be the Control Room for any station in a cluster.

**Note** - Using this mode requires two steps: defining **Source Profiles**, then defining the **Show Profile**. Once these are configured, please check [Operation Chapter 3: Working with the Call Controller](#).

### Source Profile Settings for US Phone Operation – VX & Nx Systems

You will need to create two Source Profiles for your Selectable lines hybrid (as described in the section in Chapter 2 of this manual entitled “Working With Profiles”).

1. Create a new Source Profile, and select the **Phone Source** type.
2. In the Stream I/O Settings section, select the appropriate **Fader Input** Source using the adjacent drop-down Source Selector box to pick the desired Livewire channel from your Telos phone system.
3. Below in the Source Profile, select the **US Phone Call Controller** section in the **Phone Control** area (you can skip the EU Phone section).



4. The **Hybrid** field is for entering the hybrid number that you want to associate to this Source Profile.
5. The **Fixed Line** field lets you configure a workflow similar to the EU Phone operation: assigning a dedicated line to a dedicated fader. So, leave this set to 0 if you intend to use a Call Controller with selectable lines.

**Note** - If your Phone Source uses a Fixed Channel from the VX, you must set the **Fixed Line** parameter to any number different from 0 (it doesn’t matter which number). But in this case, you will not be able to control this line from a Call

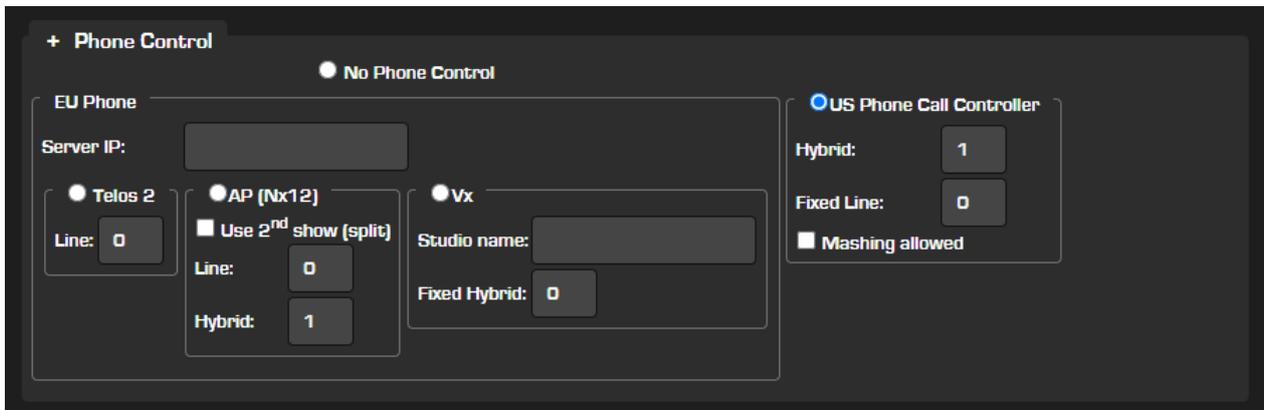
Controller, but directly from the console fader strip. You will still be able to use the Call Controller keypad to dial a number for making an outgoing call.

6. Select the **Mashing Allowed** checkbox if you want a single hybrid to handle multiple callers at once by allowing the board operator to select multiple line buttons. If used with a Telos VX, the VX System will mix audio from all lines into one source, and will create a conference bus so that all callers will hear each other.
7. Proceed to the **Phone Control** menu of your show profile, to configure console access to the Telos phone system.

## Source Profile Settings for US Phone Operation – Hx6 & iQ6 Systems

As with the other systems discussed here, you'll begin by creating a Source Profile for each of your Hx6 or iQ6 hybrids:

1. Create a new Source Profile, and select the **Phone Source** type.
2. In the Stream I/O Settings section, select the appropriate **Fader Input** Source using the adjacent drop-down Source Selector box to pick the desired Livewire channel from your Telos phone system.
3. Below in the Source Profile, select the **US Phone Call Controller** section in the **Phone Control** area (you can skip the EU Phone section).



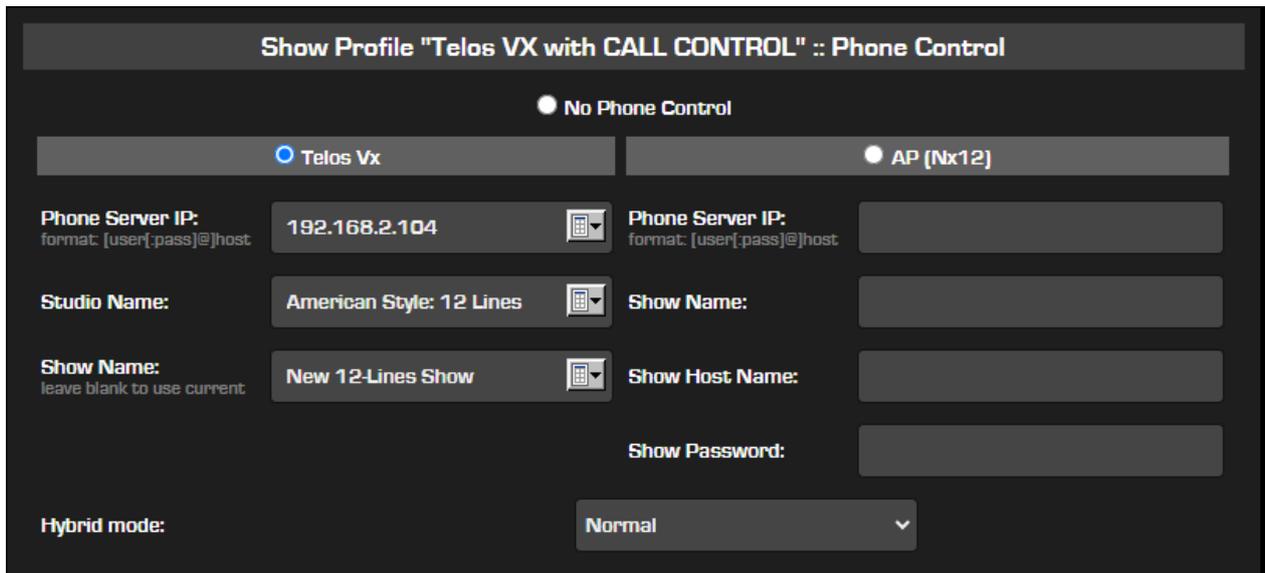
4. Enter the hybrid number in the **Hybrid** field. The iQ6 and Hx6 have two hybrids, so the option will be either 1 or 2.
5. Leave the **Fixed Line** entry at 0.
6. Repeat these steps to create a Source Profile for your phone system's second hybrid.
7. Proceed to the **Phone Control** menu of your show profile, to configure console access to the Telos phone systems (AP).

## Show Profile Settings for US Phone Operation

In case US-style phone control is required, the Show Profile itself also needs configuration. This will let Quasar log into the Telos Phone system as a client, to enable the use of the Quasar CALL CONTROL Tab.

In order to do this, select the desired Show Profile in your Quasar's Web UI, (or create a new

one following the instructions found in Chapter 2) and click on the **Phone Control** button to display the following screen:



These options let you select the protocol used to communicate with the phone server: Prior to the VX system, the AP protocol was used. The AP option is to support legacy Telos products still in use.

New Telos products (VX Protocol)	Legacy products (AP Protocol)
Telos VX	Telos Nx12 with v1.1 software
Telos Hx6 and iQ6	Telos TWO
Telos Nx12 with VX compatibility protocol update	Telos 2101

### Telos VX settings

The **Phone Server IP** is where you identify the IP address of the Telos product that is managing the calls.

- In case the standard (default) access credentials are used (user: *user* , password: *no password* ), just enter the VX IP address here.
- In cases where the non-standard credentials are used, you would add those here using the following syntax: *username:password@ipaddress*
- Hx6 and iQ6 systems use Telos VX control protocol, so for those you can use the *vx:username:password@ipaddress* . The **vx** prefix in this example is correct.

The **Studio Name** field is required by the Quasar Call Controller to interface with the appropriate configuration as defined in the VX. An incorrect name instructs the Quasar to connect to a non-existing configuration, which is indicated by the text “Studio: NULL” displayed in the INFORMATION field of the Call Controller.

The **Show Name** field is used to identify a phone system configuration when using a Telos Series 2101 or VX system. These names are defined in your phone system’s setup. Enter the configuration name you wish your Call Controller to log into.

With the VX system, this field can be left blank, allowing users to change the assigned show with a VSet phone, with the VX Web UI, or through a remote control connection. If you set the

show name here, the Quasar will select that show in the VX upon connection. Otherwise, the VX will keep the current show configuration regardless of the show profile loaded on the console.

Tip - You can select the VX Phone Server, Studio and Show Names directly from the selector box. A list will popup, with the parameters of the available VX Servers.

### Legacy Telos systems (AP) settings

The **Phone Server IP** is where you identify the IP address of the Telos product that is managing the calls.

- In case the standard access credentials are used (user: *user* , password: *no password* ), just enter your phone system IP address here.
- In cases where the non-standard credentials are used, you would add those here using the following syntax: *username:password@ipaddress*

The **Show Name** field is used to identify a phone system configuration when using a Telos Series 2101 or VX system. These names are defined in your phone system’s setup. Enter the configuration name you wish your Call Controller to log into.

The screenshot shows a configuration window titled "Show Profile 'Telos NX12 Test Show' :: Phone Control". At the top, there are three radio buttons: "No Phone Control" (selected), "Telos Vx", and "AP (Nx12)". Below this, the interface is split into two columns. The "AP (Nx12)" column contains the following fields: "Phone Server IP" (format: [user[.pass]@]host) with the value "nxuser@192.168.2.106"; "Studio Name"; "Show Name"; "Show Host Name" with the value "Hybrid1+2"; and "Show Password". The "Telos Vx" column contains: "Phone Server IP" (format: [user[.pass]@]host); "Studio Name"; and "Show Name" (format: [user[.pass]@]host). At the bottom, there is a "Hybrid mode" dropdown menu currently set to "Normal".

The **Show Host Name** field is used to log into the appropriate configuration:

- “Hybrid1”, “Hybrid2”, “Hybrid1&2” (used in **TWO-x-12**)
- “Hybrid 1&2”, “Hybrid 3&4”, “Hybrid 1-4” (used in **Nx6 & Nx12**)

The **Show Password** field is used to permit access between your Quasar console and any Telos Call system that has password protection at show levels. If you have a password for the phone system Show you want to use, enter it here.

### Settings that apply to both Phone System types

The **Hybrid Mode** drop-down lets you swap the banks of the Call Controller. By default, the left column controls Hybrid 1, and the right column controls Hybrid 2; selecting this option reverses these.

**Save** your changes and load the Show Profile to your console. The Call Controller will show a dot in the first 6 lines selections if the lines are present. If the first line selector is showing a spinning square, there is difficulty logging into the Phone system. Check your settings and verify the Telos phone system is online.

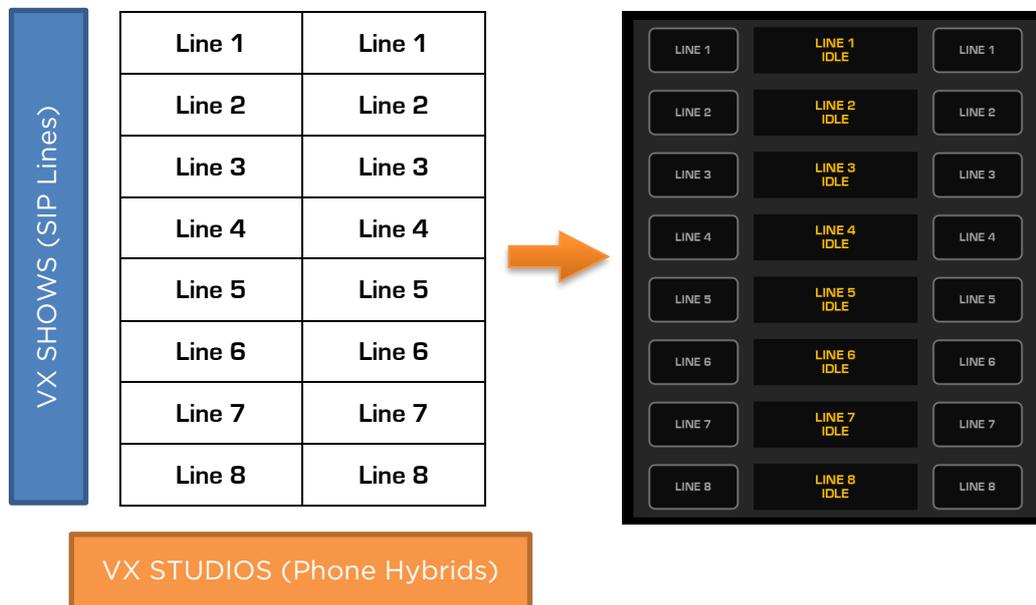
**Tip** - When selecting a line directly using the Call Controller, the default behavior is for the Left column of buttons to assign the line to Hybrid 1, and the Right column to assign the line to Hybrid 2. But when using phone systems with more than two hybrids, you may assign lines to hybrids other than the default by using the appropriate Source Profile options.

## Telos VX settings for US Phone operation

In case you're setting up US Phone Sources with a Telos VX System, and you would like to use the Call Controller, the VX Studio configuration will drive the entire workflow.

In VX, you will need to:

1. Create a **Studio** with the desired number of *Selectable Channels*. Typically, you would create two Channels, one for each "hybrid controller" in the Call Controller, and assign their Livewire channels to each US Phone Source on your Quasar. These channels, in practice, will work as your selectable line hybrids.
2. Create a **Show** with the desired number of SIP Lines you want to associate to each hybrid on your console.



The picture above shows how Studios and Shows within the Telos VX Phone System can be associated to Phone Hybrids and Phone Lines, for use with the Quasar Call Controller.

Below is an example of how a Studio can be configured in VX to work with a US-style workflow which uses 8 selectable lines on 2 faders:

## TelosVX Control Center

### 8-Lines Show Configuration

Show is in use by studio(s) "U.S. Style"

**Show Name**

**Lines**

**Lines:**

Position	Name	Extension	Server	Config	Fader	Busy All?	Ringer
1	<input type="text" value="VX Line 1"/>	<input type="text" value="1"/>	<input type="text" value="192.168.2.104"/> ▼	»	<input type="text" value="Selectable"/> ▼	<input checked="" type="checkbox"/>	<input type="text" value="Default"/> ▼
2	<input type="text" value="VX Line 2"/>	<input type="text" value="2"/>	<input type="text" value="192.168.2.104"/> ▼	»	<input type="text" value="Selectable"/> ▼	<input checked="" type="checkbox"/>	<input type="text" value="Default"/> ▼
3	<input type="text" value="VX Line 3"/>	<input type="text" value="3"/>	<input type="text" value="192.168.2.104"/> ▼	»	<input type="text" value="Selectable"/> ▼	<input checked="" type="checkbox"/>	<input type="text" value="Default"/> ▼
4	<input type="text" value="VX Line 4"/>	<input type="text" value="4"/>	<input type="text" value="192.168.2.104"/> ▼	»	<input type="text" value="Selectable"/> ▼	<input checked="" type="checkbox"/>	<input type="text" value="Default"/> ▼
5	<input type="text" value="VX Line 5"/>	<input type="text" value="5"/>	<input type="text" value="192.168.2.104"/> ▼	»	<input type="text" value="Selectable"/> ▼	<input checked="" type="checkbox"/>	<input type="text" value="Default"/> ▼
6	<input type="text" value="VX Line 6"/>	<input type="text" value="6"/>	<input type="text" value="192.168.2.104"/> ▼	»	<input type="text" value="Selectable"/> ▼	<input checked="" type="checkbox"/>	<input type="text" value="Default"/> ▼
7	<input type="text" value="VX Line 7"/>	<input type="text" value="7"/>	<input type="text" value="192.168.2.104"/> ▼	»	<input type="text" value="Selectable"/> ▼	<input checked="" type="checkbox"/>	<input type="text" value="Default"/> ▼
8	<input type="text" value="VX Line 8"/>	<input type="text" value="8"/>	<input type="text" value="192.168.2.104"/> ▼	»	<input type="text" value="Selectable"/> ▼	<input checked="" type="checkbox"/>	<input type="text" value="Default"/> ▼

Following, is how the VX Show associated to this Studio could look like:

## TelosVX Control Center

### U.S. Style Configuration

Studio is using show "8-Lines Show"

**Current Show**  ▼

**Studio Name**

**Fixed channels**

**Selectable channels**

**Auto Answer Fixed Lines**

**Lockless Conferencing**

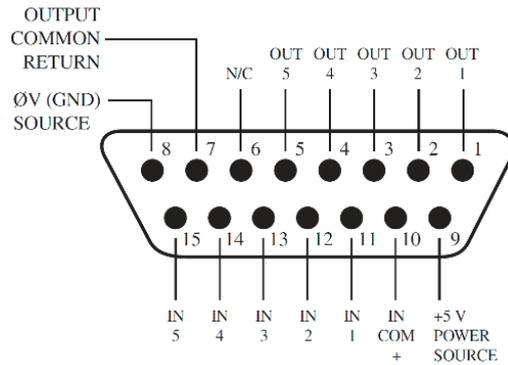
**Fixed Channels: None**

**Selectable Channels:**

Enable	Name	Channel	Advanced Receive
<input checked="" type="checkbox"/>	<input type="text" value="Studio1: Sel 1"/>	<input type="text" value="10401"/> <input type="text" value="Live Stereo"/> ▼	<input type="checkbox"/> <input type="text" value="auto"/> <input type="text" value="To Source"/> ▼
<input checked="" type="checkbox"/>	<input type="text" value="Studio1: Sel 2"/>	<input type="text" value="10402"/> <input type="text" value="Live Stereo"/> ▼	<input type="checkbox"/> <input type="text" value="auto"/> <input type="text" value="To Source"/> ▼

## GPIO Control Operation Style

The **No Phone Control** option is the default selection when you create a new Phone Source Profile. Leave **No Phone Control** selected in case you need to connect Quasar to a telephone Hybrid system which is controlled over GPIO only, and has no control over IP. This type is intended for single line hybrids like the Telos Hx1 and Hx2, but can also be used with older single-line Telos phone systems as well as those made by other phone system vendors.

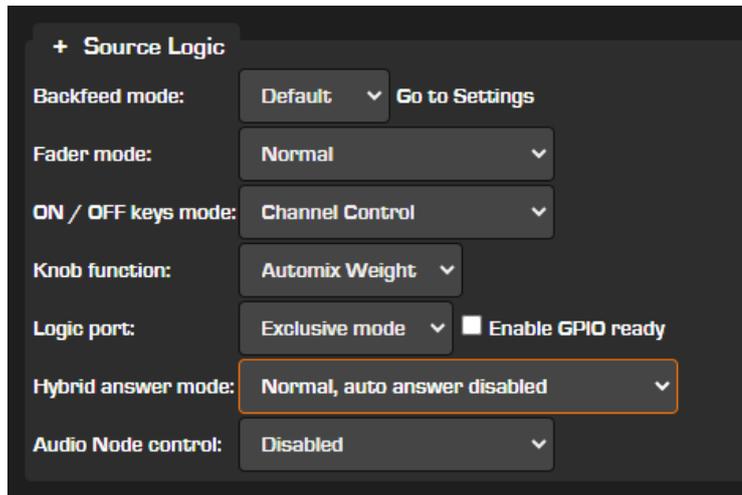


**Note** - Please refer to the [GPIO Telephone Hybrid Logic chart](#) found in Appendix B of this manual for the appropriate pinouts.

When configuring a Source Profile for this operational mode, the important field is the **Hybrid Answer Mode** drop-down menu: This type is intended for single line hybrids like the Telos Hx1 and Hx2, but can also be used with older single-line Telos phone systems as well as those made by other phone system vendors.

The default setting for Hybrid Answer mode is **Normal, auto answer disabled**. If your desire is to achieve pulses on pins 4 and 5 of the GPIO logic port associated with your phone hybrid, then you must choose one of the two other options. These other options will create pulses when the channel changes ON state or PFL state. For the **Hybrid answer mode**, your choices are:

- » **Normal, auto answer disabled.** This is the default; turning on the fader that the hybrid is assigned to does not pick up the selected line.
- » **Channel ON answers hybrid.** Turning the fader on immediately picks up the selected line.
- » **Channel ON or PFL ON answers hybrid.** As above. Additionally, placing the fader channel in **PFL** answers the call; the board operator can then talk to the caller through the Operator Mic.



Once the source profile is configured, you will need to configure the GPIO port on the xNode (or similar device) such that the GPIO port is assigned the same Primary source number as this profile was configured to.

Wiring from a GPIO port that has been configured with this channel to the hybrid is the next step. Again, refer to GPIO Telephone Hybrid Logic chart for the appropriate pinout charts, but here's a quick reference chart to interface a Telos HX1 to an Axia GPIO port:

Telos Hx1 DE-9 Pin	Axia GPIO port DA-15
Pin-1	Pin-7
Pin-2	Pin-4
Pin-3	Pin-5

## GPIO Telephone Hybrid Logic

The following table applies to a GPIO port on an xNode or similar device.

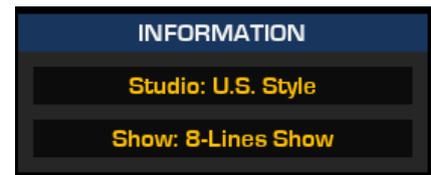
<b>Name</b>	<b>Pin</b>	<b>Type</b>	<b>Notes</b>
<b>INPUTS</b>			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF
PFL Command	13	Active Low Input	Turns PFL ON
RESET Command	14	Active Low Input	Turns channel off while not sending a STOP pulse
READY Command	15	Active Low Input	Illuminates OFF lamp to indicate source's readiness
<b>OUTPUTS</b>			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
PFL Lamp	3	Open Collector to Logic Common Return	Illuminates when PFL is ON
START Pulse	4	Open Collector to Logic Common Return	A 100ms PULSE is sent when channel is first turned ON or when PVW is first selected
STOP Pulse	5	Open Collector to Logic Common Return	A 100ms PULSE sent when channel is turned OFF.
<b>POWER &amp; COMMAND</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		

# Working with the Call Controller

With a Call Controller you can assign lines (calls) to faders that have a phone type Source loaded. The following user guidelines assume that both the Show Profile and Source profiles have been correctly configured to work with the Call Controller, according to the [US Phone operation style](#). Detailed instructions on how to configure this phone workflow are given in Configuration Chapter 3 of this Manual: [US Phone Operation setup](#).

## U.S. Phone Operation Style

Select the CALL CONTROL tab on the MTS UI. If the Show Profile's Phone Control menu has been configured correctly, the INFORMATION section in the upper right corner of the screen should report the VX Studio and Show names.



The control interface shows a 2x8 matrix: the two columns represent 2 faders, while the eight rows correspond to the 8 selectable lines. Each horizontal row has a display with two associated **LINE** buttons and a Line Status display in between. Pressing the left **LINE** button puts that line on Hybrid 1. Pressing the right **LINE** button puts that line on Hybrid 2. If a call is already on the air, pressing the button a second time locks the line, or if already locked, unlocks it. Lines that are locked cannot be dropped or put on hold. A system is built in to prevent calls from being accidentally dropped. A locked or unlocked call can be placed on the opposite Hybrid at any time by pressing the **LINE** button from the corresponding column.

Now, how do you map faders to calls? With the 1st phone fader, it is easy: it is always on the left key column (1).

The right key column (3) works as follows: it looks for the first phone fader available on the console (excluding the first) that is 2nd, 3rd and so on. From this list it looks for the first with either no locked calls, or with mashing mode (conferencing) allowed.

So, if you wish to take a call on fader 3, first you need to take a call on fader 2, then lock that line (press **LINE** button again). Then the next line would be answered on fader 3, since this is the first free fader.



8-Lines Call Controller GUI shown above.  
A 12-Line version is also available

## Talent and Producer Modes

Because the Quasar Call Controller Module is meant to be the board operator's interface, the controller is permanently set to operate in Talent Mode. This means that all lines selected from the line columns will be sent to the corresponding hybrid and that there is no means for screening calls in Producer mode like it is possible on the VSet6 or VSet12 phone-sets.

**Tip** - If the caller cannot hear the operator, the most common cause is having the phone channel turned ON. In fact, when the phone channel is ON, and the Source Backfeed is set to the Default mode (Auto PGM1/Phone), the backfeed to the caller is a mix-minus of PGM-1. Only when the phone channel is OFF the caller will hear what is assigned to PGM-4 (Phone).

## LINE 1-8

The **LINE** button is used to "Seize" a line: select it from your pool to make a call. This is the equivalent of "picking up the handset" in the old analog phone days. Once the line is Seized, you can dial a number from the Numeric Keypad and press the GO button to initiate the call. In this case the call the Line will be put on the same hybrid it was seized on.

When a Line is in On Air state, you can press again the currently active **LINE** button to LOCK and UNLOCK that Line. When locked, it will not be possible to DROP that line.

When a Line is in On Air state, you can press the inactive **LINE** button (opposite to the active one) to immediately switch the call on the other fader, or hybrid.

## Line Status Display

Each phone Line has a status display that shows the following info on two text lines:

- LINE # / Line Name / Caller ID on the first text line. The Line Name will be displayed if configured in Telos VX, and the Caller ID will always override the Line # and Name.
- Line Status on the second text line

			The Line is ready for incoming or outgoing calls
			The Line is on Fader 1
			The Line is on Fader 2
			The Line is ringing (Caller ID shown)
			The Line is On Air
			The Line is Locked on Fader 1



The Line is Locked on Fader 2



The Line is On Hold



The Line is Blocked and in Busy state



The Line is On Hold Ready and will be put On Air when NEXT is pressed (\*)



The call is being answered through a handset (\*)



The Line is seized elsewhere, perhaps in another studio. (\*\*)



The Line is used elsewhere, perhaps in another studio. (\*\*)

(\*) This status can only be generated by an external controller operating in Producer Mode.

(\*\*) This status can be generated by any external controller including other Quasar Call Controllers.

## DROP

This button drops all calls in a column that are active and unlocked. There are separate drop buttons for the left and right line columns.

When lines are Seized, they still do not belong to any fader (or hybrid) therefore, ANY Drop button will be able to drop that line. In all other cases, when the lines are On Air, You cannot drop a Line when it is:

- On Hold
- Locked
- Ringing

## NEXT

This button puts the “next” call on Hybrid 1. The NEXT priority is as follows:

1. Longest waiting screened hold
2. Longest waiting hold
3. Longest ringing-in

When the Next button is used to take a new call, if the current call is not locked, it will be dropped when the new call is placed on the hybrid. If the line is locked, the new call will be added to the hybrid by the Next button.

## BLOCK ALL

Pressing this button will cause all inactive and ringing lines to be dropped and blocked from handling any calls. Calls on-air, on the handset, on hold and the fixed lines will not be affected. The usual application for this function is to let you prevent early callers from getting in on contests until after you've made the announcement and released the lines to accept calls.

When Block All mode is selected, as lines become available, they will automatically "go busy." Pressing Block All again will release the lines and allow incoming and outgoing calls.

**Note** - Block All will also affect any lines that are shared between studios. So, if you are seeing a blocked line on a VSet that is not in Block All mode, it is possible that there is a studio in the building that has Block All active.

There is no way to block individual lines from the Quasar Call Controller. This functionality is normally available from a VSET running in Producer mode.

## TRANSFER

You can only transfer lines when they're active - which means when they are On Air. In order to transfer a call after you picked it up from one of your lines, you need to push the **TRANSFER** button, then you need to dial the phone number you want to transfer your call to, then you have the following two choices:

- Push the **GO** button and it will transfer the first active line from your pull
- In case you have multiple lines active, select a specific line and that line, if it is on air, will be transferred. If you have only one line active, the two options will coincide

## INFORMATION

This section (4) reports the configuration settings entered in the Phone Control section of the Show Profile configuration menu.

## NUMERIC KEYPAD

This section (5) is used for dialing out a call.

**Tip** - The DTMF "Touch Tones" used by telephones can be easily decoded. When dialing on the air, do not enter private numbers or passwords without first turning off the hybrid's audio on your console or this private information will be broadcast.

## Fader Modules' Hardware Keys

On top of the soft buttons available in the Call Controller, two hardware keys could be available on each channel strip, for use with both US and EU Phone workflows.

### SET key

The **SET** key is used to:

- Show which Hybrid is currently selected, when operating in US Mode
- Drop a Line (when active) when operating in US Mode
- Select a fader for seizing a Line in Euro Mode, and dialing a call through the Call Controller Numeric Keypad.

## HOLD key

In order to put a Line on Hold, push the **HOLD** key on the console channel strip. When a line is put on hold, it is temporarily removed from the fader and it goes back into the pool of lines available to be selected on a fader.

The HOLD key on a fader strip always works on a whole Phone Hybrid:

- In EURO Style operation mode, Hybrids and Lines have 1:1 correspondence (one hybrid has a single line only) therefore the Hold button can be used to put a Line on Hold.
- In U.S. Style operation mode, Hybrids can have up to 12 Lines therefore pressing HOLD will put all active, non-locked lines on hold, removing them from that particular hybrid. Pressing again the HOLD button will not restore the original active Lines. They will have to be reassigned manually.

**Note** - The SET and HOLD keys must have their function programmed in the Hardware Keymap menu of the corresponding Fader Module, to be enabled. Since it is possible to remap these keys to different functions, in cases where they do not appear to be working, we suggest checking the Web UI of the Fader Module to make sure they are enabled.

The SET and HOLD keys are currently not installed in the XR-4FAD modules. The XR modules are currently shipping with the USER 3 and USER 4 button caps installed.

## AUTO PHONE key

For Quasar SR users, who have a single USER Key on the channel strip, we developed a useful feature that lets you obtain two functions from a single key: AUTO PHONE.

It is possible to map a Hardware Key on a fader strip, to the AUTO PHONE function. In this case:

1. If the channel is not a Phone, it works as USER key, and it is mappable with Pathfinder.
2. If the channel is a Phone, then a short Press will trigger the SET function.
3. If the channel is a Phone, then a long Press will trigger the HOLD function.

# Configuration 4 – Layers & User Keys

## Working with Layers

Quasar XR is the first Axia console to introduce the concept of *Layers*. Layers offer the possibility to map any Input DSP channel, to any fader strip on the Surface, providing up to Four user-definable combinations.

**Note** - Layers are available only to Quasar XR consoles, and are not available to Quasar SR consoles. This is because Quasar SR fader modules are not motorized.

## The Layer Concept Explained

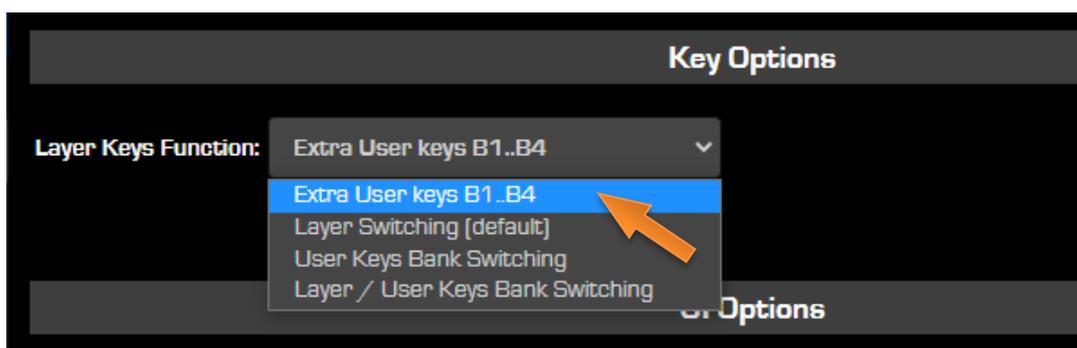
The Quasar Surface can be ordered with a number of faders strips, which might be different from the number of DSP Input channels in the Quasar Engine. In fact, a Quasar Engine comes with a standard Base License of 16 DSP channels, and can be expanded up to 64 channels.

Since there is no correspondence between physical fader strips and DSP input channels, a way to map faders to channels is required, and this is what Layers are for. Layers let you create a specific view of the Engine input channels onto each of the surface modules in any order, and multiple views of the same channels, if desired. Up to 4 Layers can be configured on a Quasar XR surface. In practice, the difference between a Show Profile and a Layer is the following:

- **Shows** are used to map **input sources** from your network to the **input channels** of your Engine.
- **Layers** are used to map the **input channels** of the Engine to the physical **fader strips** of the Surface.

Layers are enabled by default. If you use Layers, proceed to the next section **Configuring Layers**. If you have a Quasar SR console or simply no need for Layers, make sure they are disabled by following these steps:

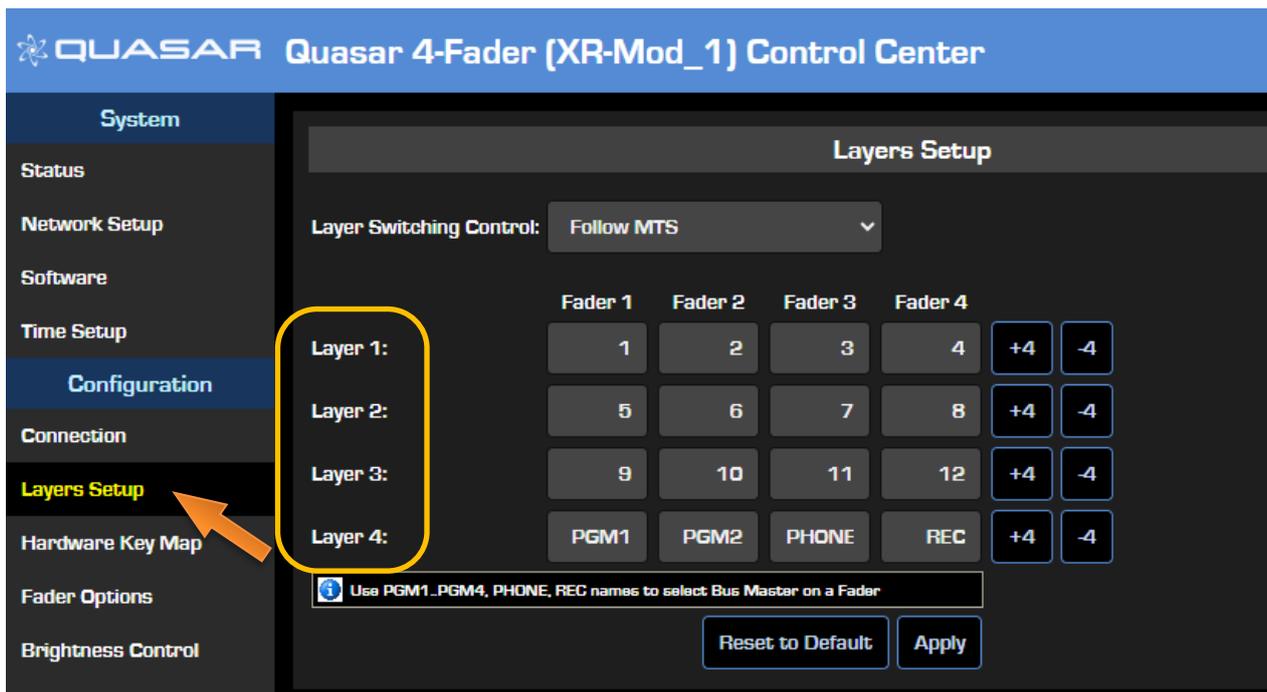
Connect a PC to your network, launch a web browser, and enter the IP address of your MTS module. Navigate to the **Configuration** section of the Master Module web UI and select the **Customize** menu. In the Key Options section, select the “User Keys Bank Switching” or “Extra User Keys B1 ..B4” option from the **Layers Keys Function** drop-down list, then click **Apply**.



## Configuring Layers on Quasar XR

Quasar XR consoles ship with all fader strips assigned, in incremental order, to an equal number of Engine input channels, starting from channel 1. So, for example, a 16-Fader surface will come with Layer 1 assigned to Input channels 1 to 16 of your Engine. Also Layers 2, 3 and 4 will be configured to access the same 16 channels.

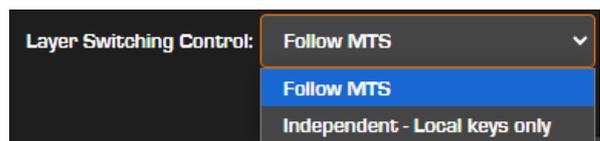
If you want to modify this configuration and create your own Layers to access different input channels of the Engine at the touch of a button, you can do so by connecting a computer to your XR Fader Modules, typing their IP addresses into your Web browser, and selecting the **Layers Setup** link, as shown here:



Enter the number of the Engine input channel you want to assign to each physical fader for each of the four Layers. The table shown in the picture above shows which input channel, of the 64 available in the Quasar Engine, will be loaded on each of the four channel strips each time you press the **LAYER 1, 2, 3, or 4** keys on the Master Touchscreen module.

**Note** - SW v3.0 adds the possibility to assign PGM1, PGM2, PGM3, PGM4, PHONE or REC to a Layer and use the fader module as a bus master controller. Just type the bus name in the box as shown in the picture above.

From the **Layer Switching Control** drop-down, select **Follow MTS** if you want the XR Module to switch layers following the Main Layer Keys in the MTS Module. Select **Independent** if you want to switch layers on a particular module only from its user keys (you will need to configure this function in its KeyMap menu).



With Layers, you could also mirror two or more fader modules by setting both modules to control the same input channel, or channels, providing that they belong to the same logical surface (connected to the same Master). And it doesn't matter where they are physically installed - they can be both in the same surface frame or be distant from each other, as long

as they are on the same LAN.

Here is an example of how to configure Layers on a 16-Fader console, in order to access all 64 input channels available in the Engine:

Fader Module # 1

Layer 1:	1	2	3	4
Layer 2:	17	18	19	20
Layer 3:	33	34	35	36
Layer 4:	49	50	51	52

Fader Module # 2

Layer 1:	5	6	7	8
Layer 2:	21	22	23	24
Layer 3:	37	38	39	40
Layer 4:	53	54	55	56

Fader Module # 3

Layer 1:	9	10	11	12
Layer 2:	25	26	27	28
Layer 3:	41	42	43	44
Layer 4:	57	58	59	60

Fader Module # 4

Layer 1:	13	14	15	16
Layer 2:	29	30	31	32
Layer 3:	45	46	47	48
Layer 4:	61	62	63	64

Layer 1:	1	2	3	4	+4	-4
Layer 2:	17	18	19	20	+4	-4
Layer 3:	33	34	35	36	+4	-4
Layer 4:	49	50	51	52	+4	-4

Reset to Default    Apply

**Tip** - This work can be done manually, by typing the Input Channel number in each field, or by using the **+4** or **-4** buttons available to the right of each layer line. These will increment or decrement the channel assignments by four at a time.

Do not forget to press the **Apply** button when done, with setting Layers on each module. You can also go back to the default settings by hitting the **Reset to Defaults** button.

**Note** - Layers' configuration is saved inside each XR-4FAD fader module, and therefore it is a global setting not stored in the MTS-MON module. Layers' configuration is NOT stored in Show profiles .

# User Keys/Buttons

A “User Key” or “User Button” is a hardware key, or a soft button, which function can be defined by the user. There are two classes of functions: Pre-defined and Pathfinder-defined.

**Note** - Pathfinder refers to them as “buttons” (being soft), so we will refer to hardware keys as “buttons”, only in this chapter.

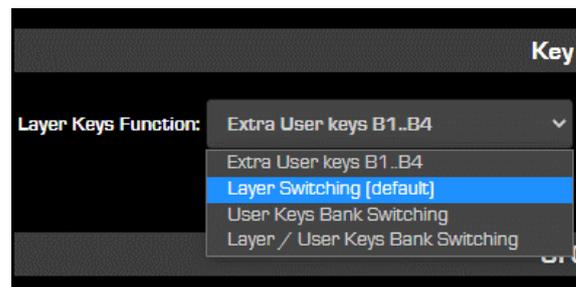
1. A Pre-defined function is a shortcut to a function which already exists in the console, but is simply not available directly from its hardware keys. The keys can be configured from each module’s Web UI, and their function selected from a drop-down list.
2. A Custom PF function is a function defined with Pathfinder. When a User button is defined as Custom PF in the web UI, it becomes a PF button.

Quasar offers 4 Banks of eight User-Programmable keys in the MTS module, 4 dedicated User Keys per channel strip in the XR-4FAD module, and 1 dedicated User Key in the SR-4FAD module. Users who need a high level of customization can even remap any of the Fader module keys.

## MTS-MON Module User 1-8 Buttons

The User buttons found in the Quasar MTS module belong to the second class, and are only defined with Pathfinder.

By default, Quasar comes with the first bank (Bank A) of User 1-8 buttons enabled, as well as the Layer Switching function enabled.



In order to activate the other three Banks of User 1-8 buttons (Bank B, C and D) navigate to the MTS module Web UI with your browser. Under the **Configuration** header, click on **Customize**. In the **Key Options** section, you can choose to enable the following options from the **Layer Keys Function** drop-down list:

- **Extra User Keys B1 ..B4** will disable Layers and enable the four hardware Keys to work as extra User Buttons, with no Bank switching. In PF, they will be equivalent to the first 4 buttons of bank B, giving you 12 User Buttons in total.
- **User Button Bank Switching** will enable all the 4 user Banks, and disable Layer 2, 3, 4 selections. In this case the console will only operate on a single Layer.
- **Layer / User Button Bank Switching** will enable bot the Layers and User Banks selection

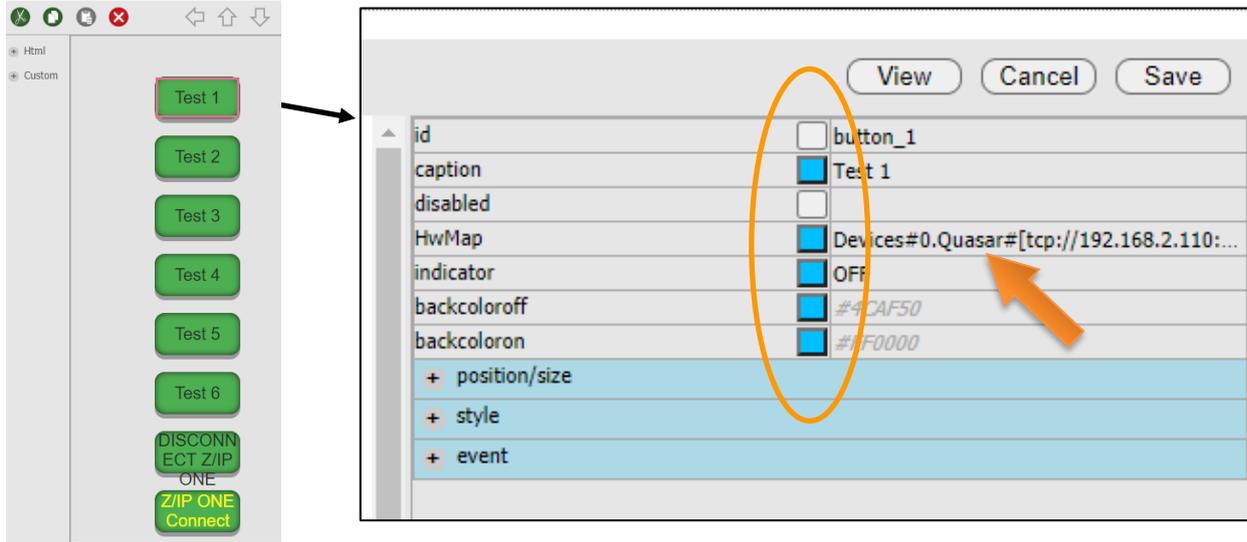
Don't forget to click the **Apply** button.

## Programming MTS User Buttons with Pathfinder

User 1-8 Buttons can be programmed via Pathfinder User Panels or Logic Flows. It is beyond the scope of this manual to provide full instructions on how to program Pathfinder.

Quasar Modules follow the same discovery rules and have their own set of object properties, like other Axia Devices do. The following example, based on using Pathfinder User Panels assumes that the user has a basic knowledge of Pathfinder:

1. Make sure that PF has discovered all the surface modules, by starting the Livewire Endpoint Discovery service. You can find this in Pathfinder's System Config menu.
2. Create a simple User Panel with 8 buttons in PF. These can be mapped to the MTS User Buttons by selecting each button and clicking on the **HwMap** property.



3. Pathfinder will detect 32 buttons for each Master Module detected, and display the following list of properties:

Select Hardware Button

Show 100 entries

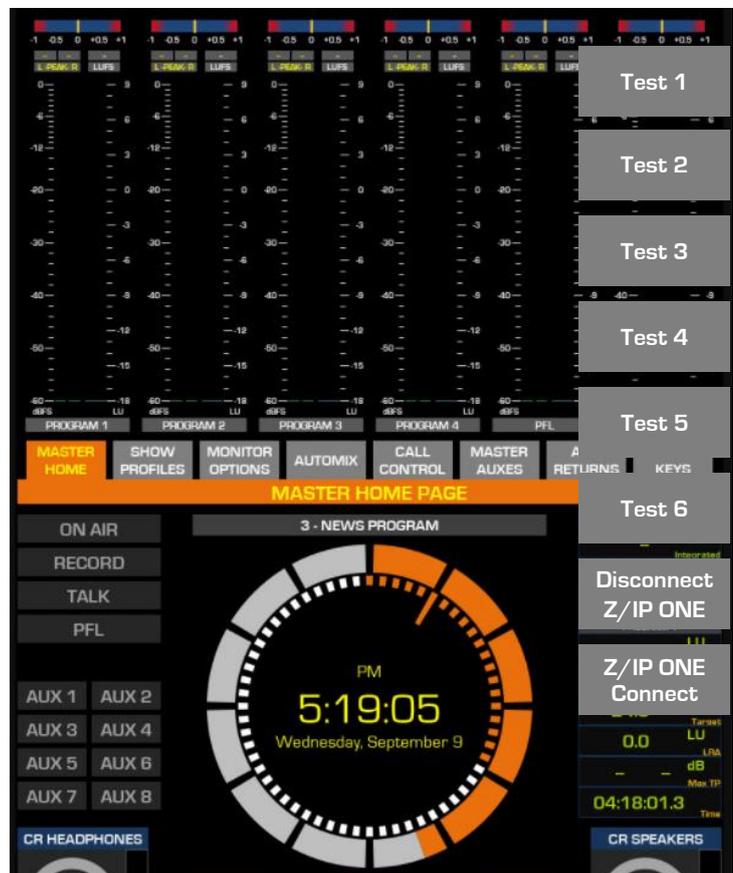
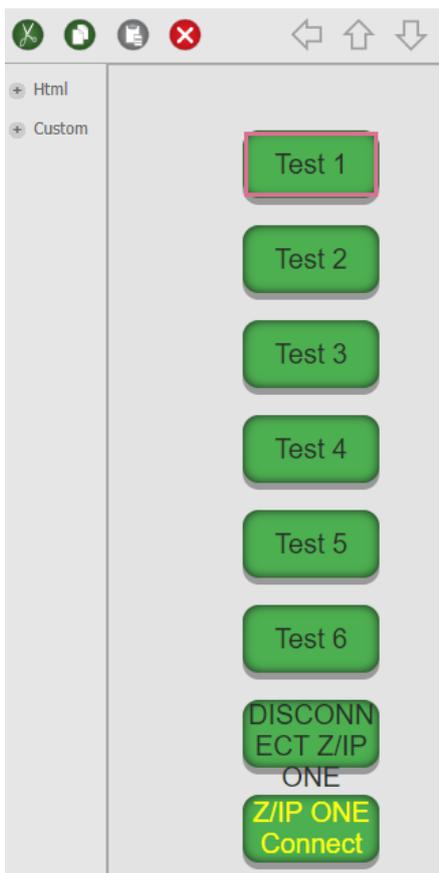
Device Name	IP Address	Module	Button
None		0	0
Quasar	192.168.2.110	mon#0	1
Quasar	192.168.2.110	mon#0	2
Quasar	192.168.2.110	mon#0	3
Quasar	192.168.2.110	mon#0	4
Quasar	192.168.2.110	mon#0	5
Quasar	192.168.2.110	mon#0	6
Quasar	192.168.2.110	mon#0	7
Quasar	192.168.2.110	mon#0	8
Quasar	192.168.2.110	mon#0	9
Quasar	192.168.2.110	mon#0	10
Quasar	192.168.2.110	mon#0	11
Quasar	192.168.2.110	mon#0	12
Quasar	192.168.2.110	mon#0	13
Quasar	192.168.2.110	mon#0	14
Quasar	192.168.2.110	mon#0	15
Quasar	192.168.2.110	mon#0	16
Quasar	192.168.2.110	mon#0	17
Quasar	192.168.2.110	mon#0	18
Quasar	192.168.2.110	mon#0	19
Quasar	192.168.2.110	mon#0	20
Quasar	192.168.2.110	mon#0	21
Quasar	192.168.2.110	mon#0	22
Quasar	192.168.2.110	mon#0	23
Quasar	192.168.2.110	mon#0	24
Quasar	192.168.2.110	mon#0	25
Quasar	192.168.2.110	mon#0	26
Quasar	192.168.2.110	mon#0	27
Quasar	192.168.2.110	mon#0	28
Quasar	192.168.2.110	mon#0	29
Quasar	192.168.2.110	mon#0	30
Quasar	192.168.2.110	mon#0	31
Quasar	192.168.2.110	mon#0	32

Showing 1 to 33 of 33 entries 1 row selected

Select Cancel

4. The column indicated by the yellow arrow shows if the function is related to the Fader or Master module. Lines with “mon #0” indicate Master (User 1-8 keys) and Lines with “FaCH xx” indicate Fader (User 1-4 keys).
5. The column indicated by the blue arrow shows the key identifier. The Master module has four Banks of 8 buttons therefore:
  - Button IDs #1 to 8 correspond to User 1-8 Buttons of Bank A
  - Button IDs #9 to 16 correspond to User 1-8 Buttons of Bank B
  - Button IDs #17 to 24 correspond to User 1-8 Buttons of Bank C
  - Button IDs #25 to 32 correspond to User 1-8 Buttons of Bank D

When a label is assigned to a button, within the Pathfinder User Panel, this will be displayed as soon as the corresponding button on the MTS module is touched, and before this is actually pushed.



For more information on how to create a User Panel in Pathfinder, please refer to the Pathfinder Core Pro User Manual.

# Fader Modules' User Buttons

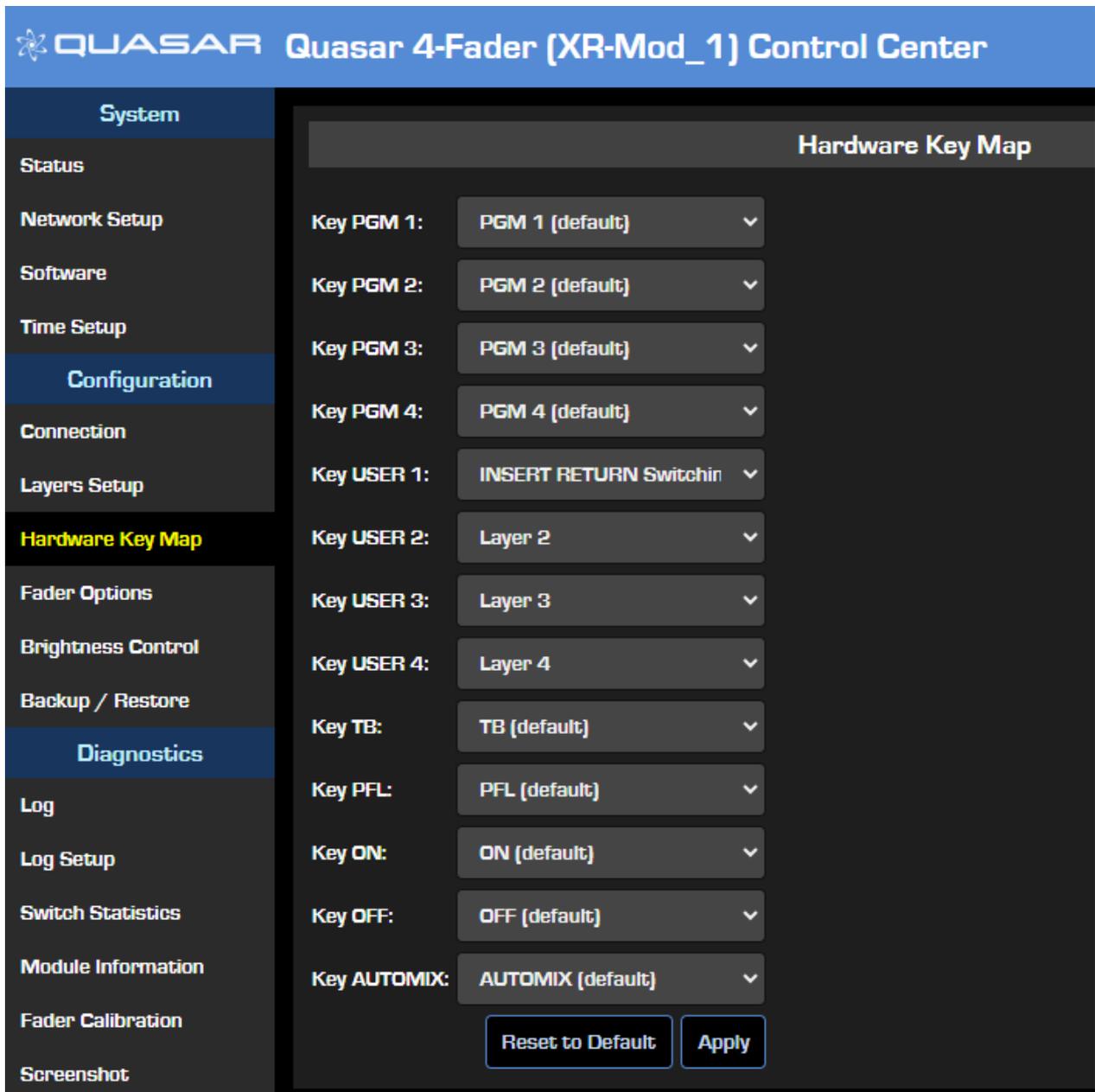
## Remapping the XR-4FAD Module Keys

User Buttons 1-4 are available in the XR-4FAD Module, and located on each channel strip right below the channel display.

On top of defining a custom function for the User 1-4 buttons, it is also possible to map a different function to any of the hardware keys on the XR-4Fader Module.

In fact, every XR Fader module ships with a default set of functions, that corresponds to the labels found on the key caps. But these defaults can be changed. We call this “Key Mapping”

In order to remap the default function to any of the XR Fader Module keys, open up a browser on your PC, navigate to the module’s Web, and select **Hardware Key Map** from the Configuration menu.



**Select** the new function you want to associate to each hardware key, from the drop-down list, and click **Apply** . USER1-4 functions must be programmed with Pathfinder, while all the others are pre-determined. Below is the list of all possible functions available for each key:

PGM 1	Activates the Program 1 bus
PGM 2	Activates the Program 2 bus
PGM 3	Activates the Program 3 bus
PGM 4	Activates the Program4 bus
SELECT	Selects the Channel Strip (duplicates the Encoder Knob pushbutton)
USER 1	Activates a custom function programmed with Pathfinder
USER 2	Activates a custom function programmed with Pathfinder
USER 3	Activates a custom function programmed with Pathfinder
USER 4	Activates a custom function programmed with Pathfinder
TB	Activates the Talkback function
PFL	Activates the Pre-Fader Listen function
ON	Turns the Channel ON, only
OFF	Turns the Channel OFF, only
AUTOMIX	Activates the Automix function
MUTE	Temporarily mutes the channel audio
TALK TO CR	Talks to the Control Room Monitors activating the LISTEN signal
EQ Active	Activates the Equalizer on the selected channel
DYN Active	Activates the Compressor and Expander on the selected channel
DE-ESSER Active	Activates the Equalizer on the selected channel
FILTERS Active	Activates the Filters on the selected channel
LIMITER Active	Activates the Limiter on the selected channel
INSERT RETURN Switching	Toggles Insert Return positions: OFF, PRE-PROCESS, PRE-FADER, POST-FADER
ON-OFF	Ch defaults to OFF state - push to turn channel ON
OFF-ON	Ch defaults to ON state - push to turn channel OFF (CUT function)
AFL	Activates the After-Fader Listen function
SET	Selects the Phone channel for controlling a Hybrid/VoIP System
HOLD	Puts the Hybrid/VoIP System call on hold
AUTO PHONE	If Phone Control is configured for the Source, Short press = SET and long press = HOLD. If not configured, it works as USER 1
AUX 1	Activates the Auxiliary 1 bus
AUX 2	Activates the Auxiliary 2 bus
AUX 3	Activates the Auxiliary 3 bus
AUX 4	Activates the Auxiliary 4 bus
AUX 5	Activates the Auxiliary 5 bus
AUX 6	Activates the Auxiliary 6 bus
AUX 7	Activates the Auxiliary 7 bus
AUX 8	Activates the Auxiliary 8 bus
Shortcut SOURCE	Selects the SOURCE Tab from the Touchscreen Channel Menu
Shortcut INPUT & FILTERS	Selects the INPUT & FILTERS Tab from the Touchscreen Channel Menu
Shortcut DYN	Selects the DYNAMICS Tab from the Touchscreen Channel Menu
Shortcut EQ	Selects the EQUALIZER Tab from the Touchscreen Channel Menu
Shortcut DE-ESSER	Selects the DE-ESSER Tab from the Touchscreen Channel Menu
Shortcut AUX SENDS	Selects the AUX SENDS Tab from the Touchscreen Channel Menu

Shortcut to CTRL & B/FEED	Selects the CONTROL & B/FEED Tab from the Touchscreen Channel Menu
LAYER 1	Recalls Layer 1 on the selected channel strip
LAYER 2	Recalls Layer 2 on the selected channel strip
LAYER 3	Recalls Layer 3 on the selected channel strip
LAYER 4	Recalls Layer 4 on the selected channel strip

**Note** - These functions are available per module, not per channel strip. So, choosing one custom function for a key only applies to all 4 channel strips within that XR module.

In case custom key caps are required by the user, Telos can offer a custom labeling service. Please contact your Axia Dealer for more details.

## Remapping the SR-4FAD Module Keys

One User Button is available in the SR-4FAD Module, and located on each channel strip right below the fader. On top of defining a custom function for the User button, it is also possible to map a different function to any of the hardware keys on the SR-4Fader Module.

In fact, every SR Fader module ship with a default set of functions, that corresponds to the labels found on the key caps. But these defaults can be changed.

The same set of functions available to the XR keys, is also available to the SR keys. Please see the table above for a full list of functions.

**Note** - In SR, these functions are available only globally, per console, not per channel strip. So, choosing one custom function for a key applies to all channel strips within that SR console.

In case custom key caps are required by the user, Telos can offer a custom labeling service. Please contact your Axia Dealer for more details.

## Programming Fader User Buttons with Pathfinder

In order to assign a function to these buttons from PF, you need to look for the “FaCH#xx” entries, (where xx is a number ranging from 0 to 64, which corresponds to the Engine Input Ch number):

MTS-1	192.168.2.120	FaCH#1	1
MTS-1	192.168.2.120	FaCH#1	2
MTS-1	192.168.2.120	FaCH#1	3
MTS-1	192.168.2.120	FaCH#1	4

Pathfinder will associate the function of the User 1-4 buttons to the Engine Input channels, not to the physical fader strips. This means that the function will follow the input channel, if this is moved on a different fader strip. This would be the case when different layers are recalled.

## Programming SmartKey Buttons with Pathfinder

In order to assign a function to these buttons from PF, first we need to create a User. Please refer to the Pathfinder Core Pro User manual, for creating User Panels.

Once the panel is created, with an equivalent number of buttons to those present in your SmartKey Module, you can edit the panel and select each of the buttons to assign the required properties.

In the **HwMap** field, you will be able to assign the selected button to one of the modules' hardware Keys.

In this case, you must look for the "SK#x.GROUPy" entries, in the Console's Master Module (MTS):

Select Hardware Button			
Device Name	IP Address	Module	Button
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#1	1
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#1	2
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#1	3
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#1	4
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#1	5
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#1	6
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#2	1
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#2	2
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#2	3
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#2	4
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#2	5
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#2	6
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#3	1
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#3	2
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#3	3
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#3	4
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#3	5
STUDIO1-LLR	192.168.2.110	SK#1.GROUP#3	6

You can find these more quickly by entering "SK#" in the Search box in the top-right corner of your PF web UI screen:

Search Filenames:

**x** and **y** are numbers ranging from 1 to 4, which correspond to the SmartKey Module Address and Group number, which are configured in the Console's Web UI, in the **SmartKeys** Setup page.

SmartKey Module Setup		
Uptime	Module Address	Group
08:44:14	Module 1	Group 3
08:44:14	Module 1	Group 2
08:44:14	Module 1	Group 1
08:44:13	Module 1	Group 4

Please see Configuration Chapter 1 - Configuration Menu - [SmartKeys](#), for more details on how to configure SmartKeys Modules in Quasar.

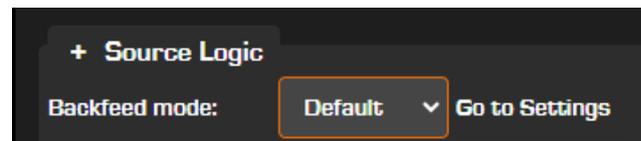
# Configuration 5 – Practical Guides

This chapter contains some Practical Help Guides for the most common setups. If we forgot one, please let us know!

## Setting Up the Console Talkback

The first step is to configure a source for use it with the Talkback circuit. The Source Type must be set as Operator Microphone”. Please check Chapter 2 of this manual, for [Creating the Operator’s Mic Source Profile](#).

The studio microphones that will need to receive this TB feed (when the **TALK** key is pressed) must have their Source Backfeed Enabled as **Default**. In order to do so, go to the [Source Profile](#) menu and set the **Backfeed mode** drop-down as **Default**, on each of the studio microphone sources.



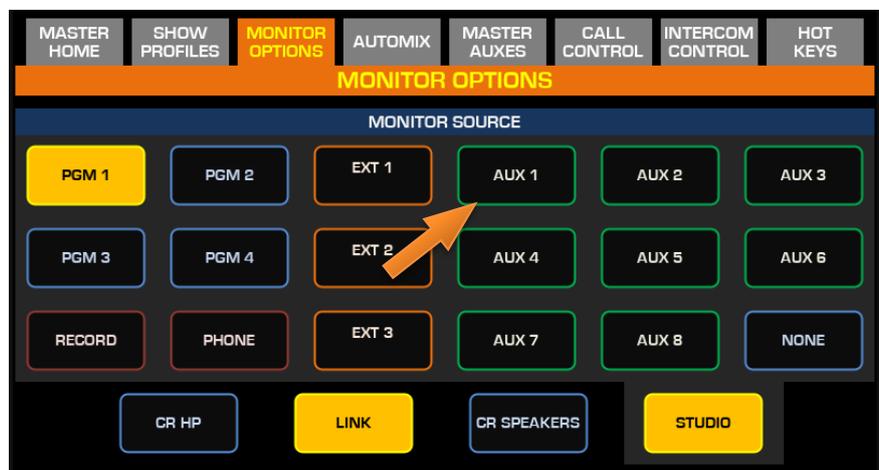
The second step is to choose from one of the following two workflows:

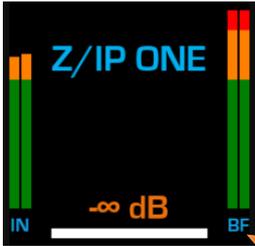
1. *Self-Operated*: The console operator is the program host, and he also uses the CR Microphone to talk to other microphones in the Studio, to phones, and codecs.  
In this case, it is sufficient to create an Operator Mic source, and load it on one of the console channels, to operate.
2. *Engineer + Studio Guest*: The Studio Guest is the program host, and uses the Studio Guest Microphone to talk to the Control Room. The console operator only uses the CR Microphone for the talkback, and never goes on air from it.

In this case, having the CR Microphone loaded on one of the console channels is not required.

## When the Operator is the Host

To set the Program feed to the Studio Microphones, push the **MON OPT** key on your Master Module, select the STUDIO section, and select the desired program bus.





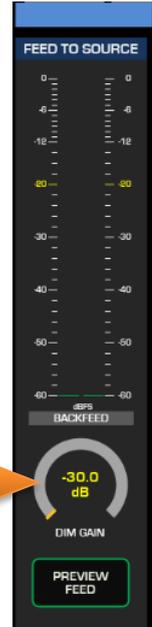
When the desired Program bus is selected, you will see the audio level in the studio microphone’s channel strip **BF** (backfeed) meters.

The amount of attenuation of the monitor audio when the TB is active can be set individually for each of the studio microphones.

This can be done with the Backfeed Dim Gain control, found in each of the channels’ Control & Backfeed tab. The channel BF meter will display the level drop when the TB is engaged.

### When the Studio Guest is the Host

Unlike the Fusion, Quasar does not require the TB Mic Source to be always assigned to a physical or virtual fader of the console, in order to operate the Talkback. In cases where the CR Microphone is not needed as a channel on the console, and only used as TB Mic, the Source designated as TB microphone can be simply assigned to the **Studio TB** input of the Show Profile’s Monitor Section. Pushing the **TALK** key will inject the TB Mic Source directly into the Talk Bus.



Only when the Talkback is properly configured in one of the two ways we just described, the TB circuit will be active. To talk to the Studio, push the hardware **TALK** key located in the bottom right corner of the Master Monitor module. To talk to one particular channel only, push the small **TB** key onto the fader strip.

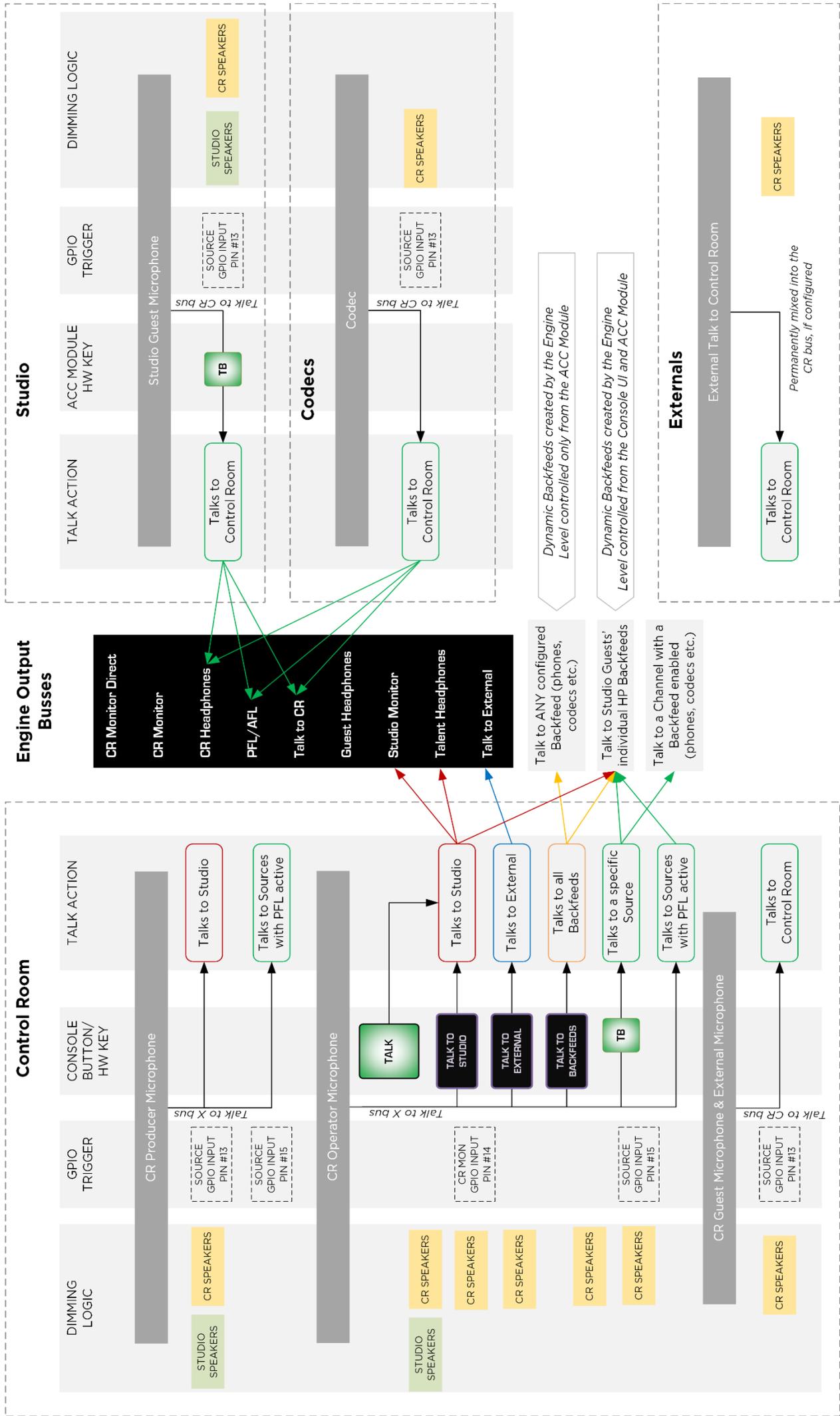


Both these actions will enable the Talkback function, turning the corresponding signal on in the MTS Home screen.

In case the Studio Host will press the small **TB** key on an accessory module, or activate the Talk To CR function via a GPI button, this indicator will change to “LISTEN”, meaning that the TB circuit is now working in the opposite direction, from the Studio to the Control Room.

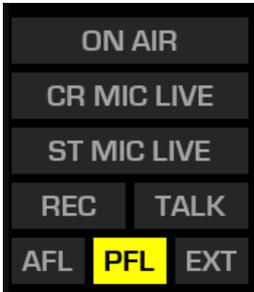
**Note** - The green TALK indicator will light up every time the ops’ microphone audio is routed to the Talk bus, instead of the PGM bus. If the CR Mic is used both as Talk Mic and On Air Mic, this will be temporarily removed from the PGM bus (muted to the Air) so that the conversation to the Talk bus is not sent to the mix. If the CR Mic is used as Talk Mic only, then this indicator will not light up when a TALK or TB key is pushed.

For details on how to drive external lights using the TALK and LISTEN indicators, please Refer to the paragraph: [Detecting the TALK and LISTEN Lights](#).



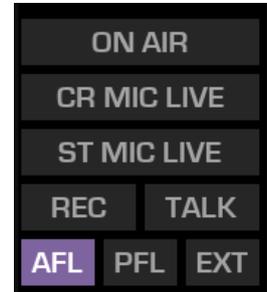
## Setting Up the Monitors & PFL/AFL

Quasar’s Monitoring section offers independent PFL/AFL operation modes for the Control Room Headphones and Loudspeakers. While PFL stands for “Pre-Fader Listen”, AFL stands for “After-Fader Listen”.



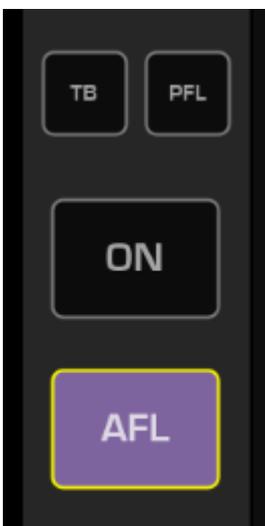
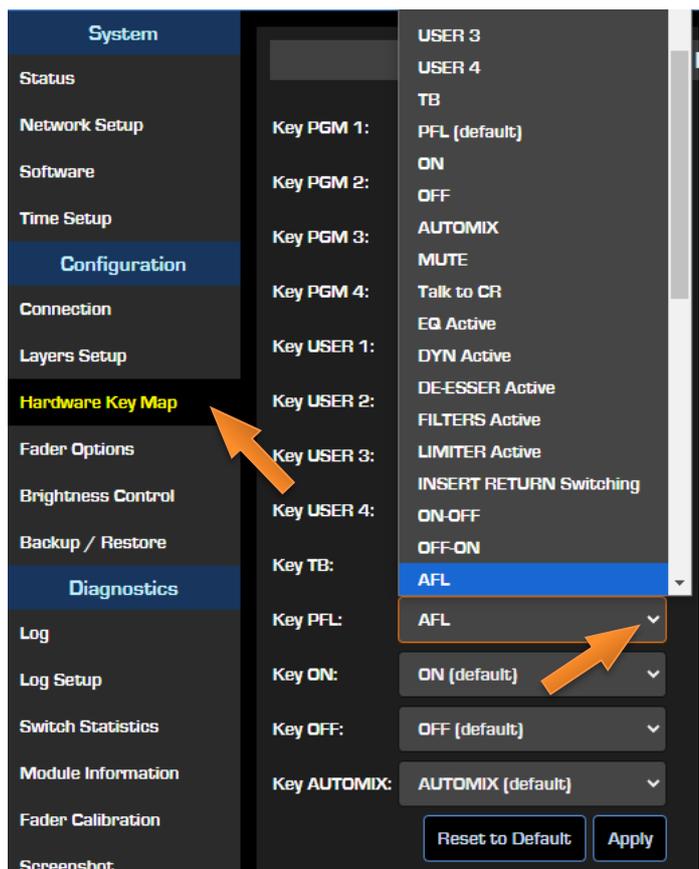
In general, when the **PFL** key is pressed on a channel, the console will indicate that Pre-Fader Listening is active by turning on the “PFL” signal in the Master Home screen.

Similarly, if the After-Fader Listening will be engaged (either from the UI soft channel, or by a custom-mapped key on the surface) an “AFL” signal will be turned on in the UI.



**Note** - In Quasar, we use industry-standard terminology such as “PFL” and “AFL” because using a generic term like “Preview” or “Cue” would not let the operator distinguish between Pre-Fader and Post-Fader listening modes. We also use the word “Preview” in different cases, such as when we are *previewing* an Input or a Monitor Source directly from the network.

There is a dedicated **PFL** key on each of the Quasar surface’s channel strip. However, this key (or any other key in the Fader Module) can be reprogrammed to work as an **AFL** key, by navigating each Fader Module’s web UI and selecting the **Hardware Key Map** menu.

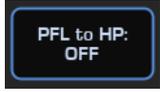


It is also possible to access the AFL function directly, on every channel strip, by selecting the CONTROL & B/FEED tab from the Touchscreen UI, and pressing the large **AFL** key located in the bottom right corner of the screen.

## Monitoring PFL in the CR Headphones

### PFL to HP Workflow

When the operator is monitoring a program through his Headphones, and the **PFL** key is pressed onto a channel:



1. **PFL to HP : OFF.** The HP monitoring does not change. The PFL bus is routed only to the dedicated PFL speaker output. Its output level is controlled by the PFL to HP control.



2. **PFL to HP : STEREO.** The PFL bus is routed to the L & R HP outputs before the CR HEADPHONES volume. The PFL bus level is fixed, and the CR HEADPHONES volume controls the CR HP output level.

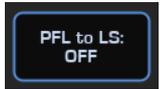


3. **PFL to HP : SPLIT.** The PFL bus is routed to the Right HP output only, before the CR HEADPHONES volume. The level of this bus is controlled only by the PFL to HP volume. The previously selected monitor bus remains routed to the Left HP output only. The CR HEADPHONES volume controls the level of both Left & Right outputs.

## Monitoring PFL in the CR Loudspeakers

### PFL to LS Workflow

When the operator is monitoring a program through the CR Loudspeakers, and the **PFL** key is pressed onto a channel:



1. **PFL to LS : OFF.** The LS monitoring does not change. The PFL bus is routed only to the dedicated PFL speaker output. In this case the level is controlled by the PFL to LS volume only.



2. **PFL to LS : STEREO.** The PFL bus is routed to the Left & Right LS outputs after the CR SPEAKERS volume, therefore the CR SPEAKERS volume control has no effect on the output level and is bypassed. In this case, the CR Speakers knob changes its ring color from orange to yellow, to indicate its function is changed, and it is now controlling the PFL to LS volume.



3. **PFL to LS : SPLIT.** The PFL bus is routed to the Right LS output only, after the CR SPEAKERS volume. The PFL to LS volume controls the Right Speaker output only. The previously selected monitor bus is still routed to the Left Speaker output only, and the CR SPEAKERS volume controls the level of this output.

## Monitoring PFL from a dedicated Loudspeaker

In this case, the **PFL to LS** button must be set to **OFF**. The PFL level must be adjusted from the **PFL to LS** control, while the CR Speakers knob will keep adjusting only the Control Room monitor speakers, and not the PFL speaker.

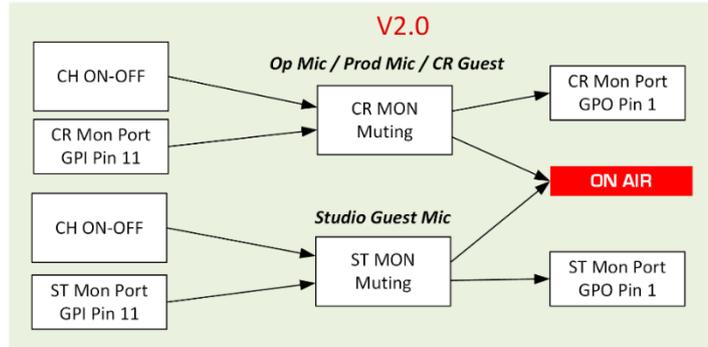


In SW v3.1 and earlier versions, the **PFL to LS** control was only available from the MTS HOME and CALL CONTROLLER Tabs, while in SW v3.2 this is available in all Master Tabs except the MONITOR OPTIONS tab, from the rightmost soft encoder, permanently illuminated with orange color. The PFL to LS control has been added in the form of a dynamic widget which will pop up as soon as the encoder knob is touched.

## Triggering an ON-AIR Light

In v2.0 and previous software versions, the console GUI was showing a combined indication of the two MON MUTE states, and was translating it into an "ON AIR" signal. So, in reality, when any mic was on (either CR or Studio), and the corresponding monitor was muted, the ON AIR light would be turned on.

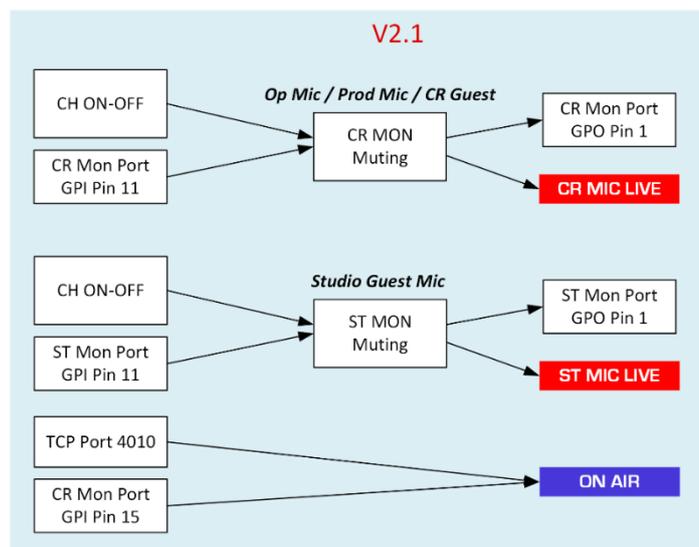
This was not an optimal approach, since this indication could be false if the program bus is not assigned, or if it's assigned but not routed to the transmitter.



In v2.1 we improved the current signalization by splitting the combined signal in two dedicated signals: CR MIC LIVE and ST MIC LIVE, and create a new, generic ON AIR indicator, that can be triggered only by an external TCP command, or GPI pin.

This approach takes into consideration modern multi-studio workflows where only the On Air routing matrix is aware of which studio is actually on the air (routed to the transmitter), and which one is not.

**Note** - Starting from SW v3.2, the ON AIR indicator has been enabled in Quasar Soft, and its state synchronized with the ON AIR lamp of the MTS-MON GUI.



### Triggering the ON AIR signal

The new ON AIR indicator is a pure signal, (like a "user lamp" built into your console) that can be activated with either a GPI or Pathfinder logic under certain user-defined conditions, in order to display the real ON AIR status of the console. This implies that it is an input-only indicator which does not generate any event, since it is not aware of the console on air status.

A new Property has been added to an existing LWCP 'mon' object. Its name is: 'on\_air'. Possible values are ON and OFF. The Property can be written with SET and read with GET. State changes are not followed by an event notification. Since only an LWCP or a GPI can turn this light on, there is no need to generate an event notification.

Example: **SET mon on\_air=ON** will activate the ON AIR signal in MTS HOME tab.

**Note** - ON AIR indicator logic: The TCP Port and GPI Port have true multiplexing logic - both external ports are monitored and their states are

combined into a single state for the indicator. No one overrides the other.

## Detecting the TALK and LISTEN Lights

### Driving an external device, or light, from the TALK signal

The TALK indicator is an “intelligent lamp” driven by the internal console logic that can be detected by Pathfinder or any other LWCP-capable device, for driving external devices (such as a light signal, or a camera) when ANY Talk function is activated.



The TALK indicator will light up when any of the following talkback functionalities are active:

- Talk to all Backfeeds (Phones, Codecs, Studio Guests)
- Talk to Studio
- Talk to External
- Talk to Source (channel TB key)
- Talk to all Sources with PFL active

This implies that it is an output-only indicator which generates only events.

A new Property has been added to the existing LWCP 'mon' object. Its name is: 'talk'. Possible values are ON and OFF. The Property can not be written or read, but every state change is followed by an event notification.

Example: **EVENT mon talk=ON** will indicate the TALK signal in MTS HOME tab is active.

```
192.168.2.110 - PuTTY
SUB mon
ACK mon OP=SUB,Status=OK

EVENT mon cr_dim=DIMMED
EVENT mon st_dim=DIMMED
EVENT mon talk=ON
EVENT mon tt_st=ON
EVENT mon talk_stud=ON
EVENT mon st_dim=NORMAL
EVENT mon st_vol=-19.5
EVENT mon cr_dim=NORMAL
EVENT mon cr_vol=-49
EVENT mon talk=OFF
EVENT mon tt_st=OFF
EVENT mon talk_stud=OFF
```

### Driving an external device, or light, from the LISTEN signal

The LISTEN indicator is an “intelligent lamp” driven by the internal console logic that can be detected by Pathfinder or any other LWCP-capable device, for driving external devices (such as a light signal, or a camera) when the Talk to Control Room function is activated.



The LISTEN indicator will light up only when any the following talkback functionality is active:

- Talk to CR

This implies that it is an output-only indicator which generates only events.

A new Property has been added to an existing LWCP 'mon' object. Its name is: 'tt\_cr'.

Possible values are ON and OFF. The Property can not be written or read, but every state change is followed by an event notification.

Example: **EVENT mon tt\_cr=ON** will indicate the LISTEN signal in MTS HOME tab is active.

```
192.168.2.110 - PuTTY
SUB mon
ACK mon OP=SUB,Status=OK

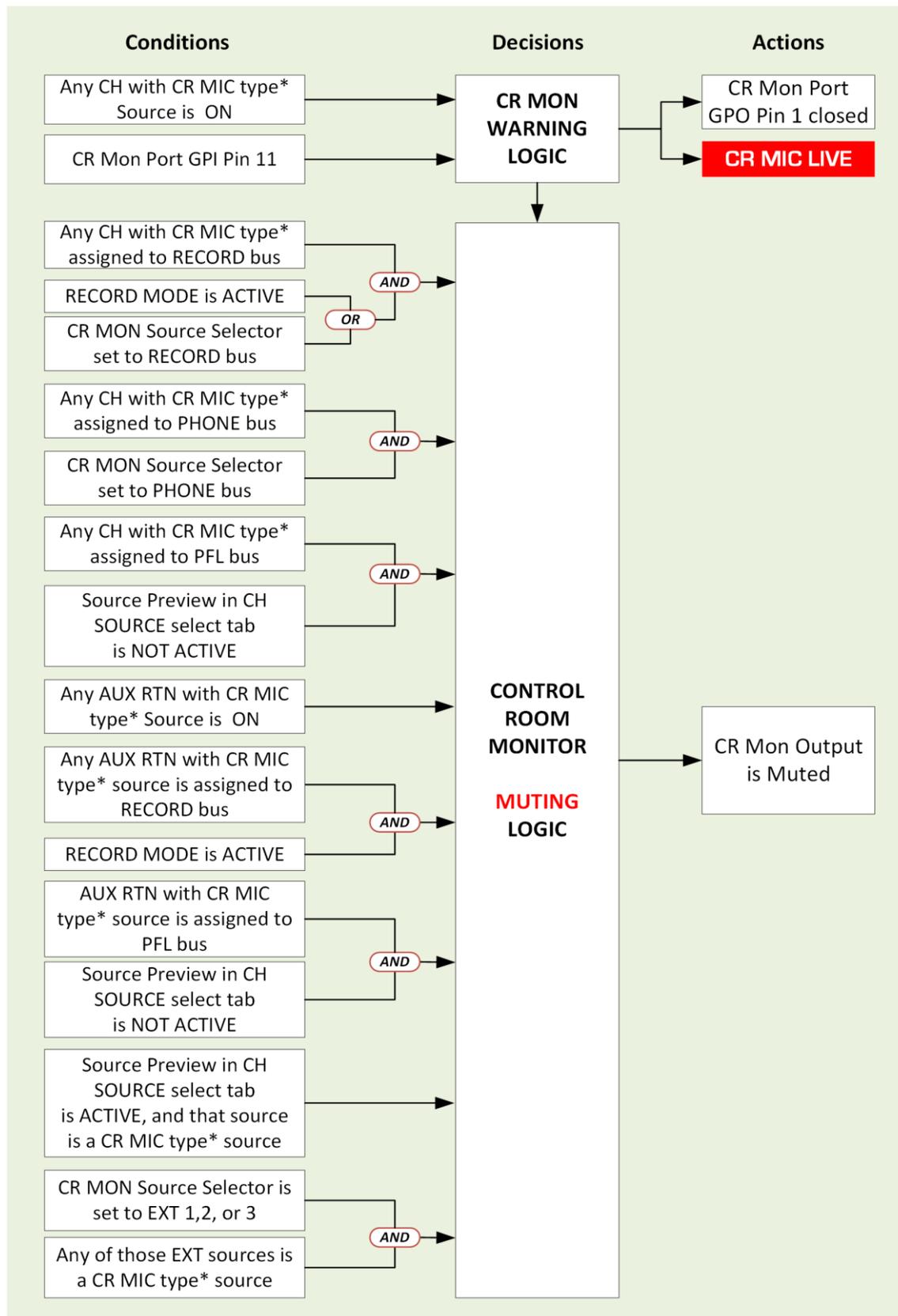
EVENT mon cr_dim=DIMMED
EVENT mon tt_cr=ON
EVENT FaCH#3 tt_cr=ON
EVENT LwCH#113 tt_cr=ON
EVENT mon tt_cr=OFF
EVENT mon cr_dim=NORMAL
EVENT mon cr_vol=-49
EVENT FaCH#3 tt_cr=OFF
EVENT LwCH#113 tt_cr=OFF
```

**Note** - A subscription to the Monitor is required in order to read such events.

The LWCP command to initiate the subscription is: **SUB mon**

## Triggering the Monitor Logic

The internal logic will **MUTE** the Control Room Speakers when:



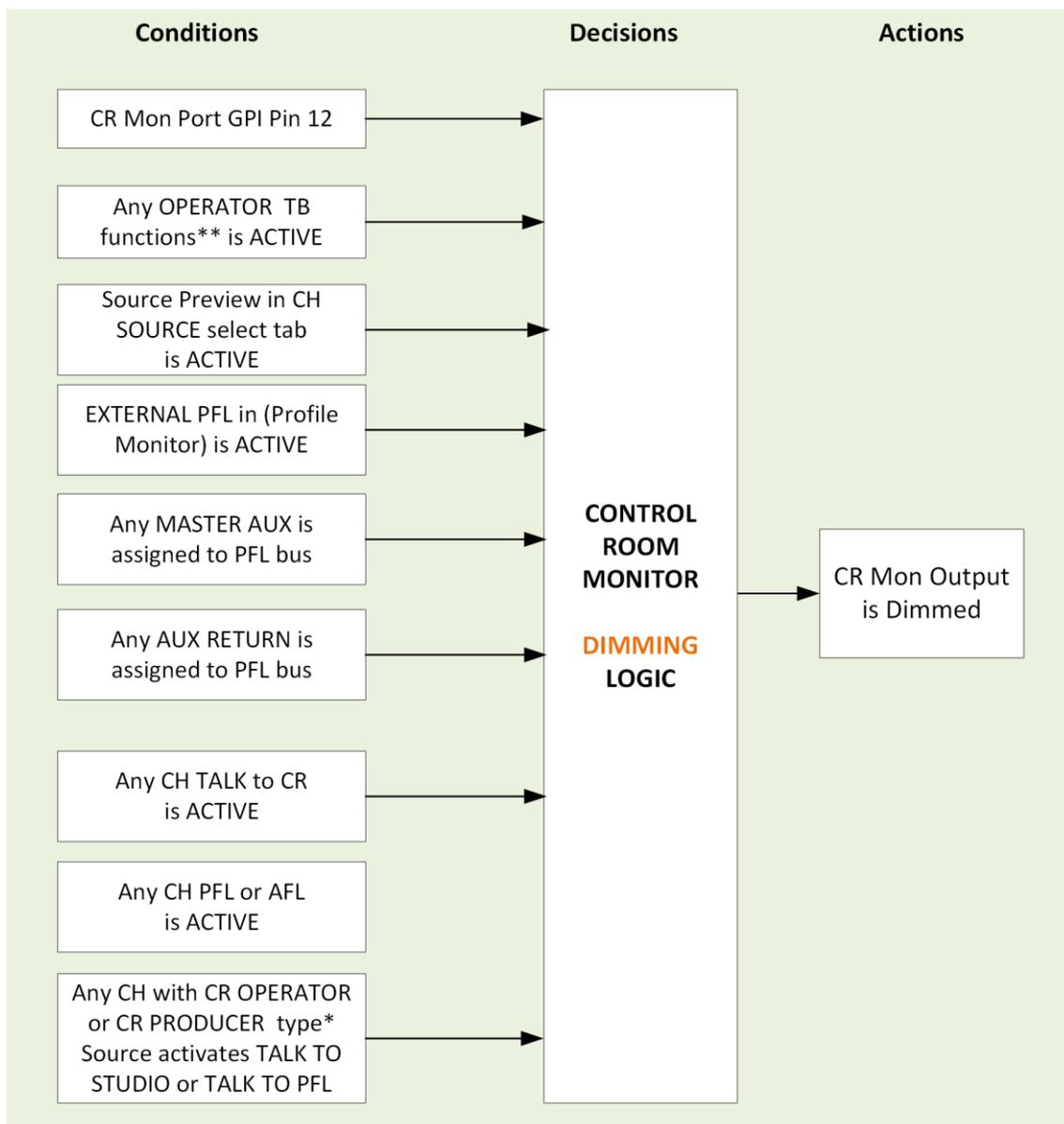
\*Operator Mic / CR Guest Mic / CR Producer Mic

### Muting the CR Speakers while recording

The PGM4, Record and Phone busses are tightly coupled (since the Element times): this means that when you push the PGM4 button, you are also pushing two more "invisible" buttons, that activate the Rec and Phone busses. But once a CR MIC channel is assigned to PGM4, the console cannot know if the receiving end is recording from the Record or PGM4 bus so it will mute the CR monitors because the Rec bus has been activated (as well) and it is pre-on, so the speakers must be muted to avoid feedbacks.

**Note** - CR MIC LIVE indicators logic: The Internal logic state (Mic ON/OFF) and GPI Port have true multiplexing logic - both states are monitored and combined into a single state for the corresponding indicator. No one overrides the other.

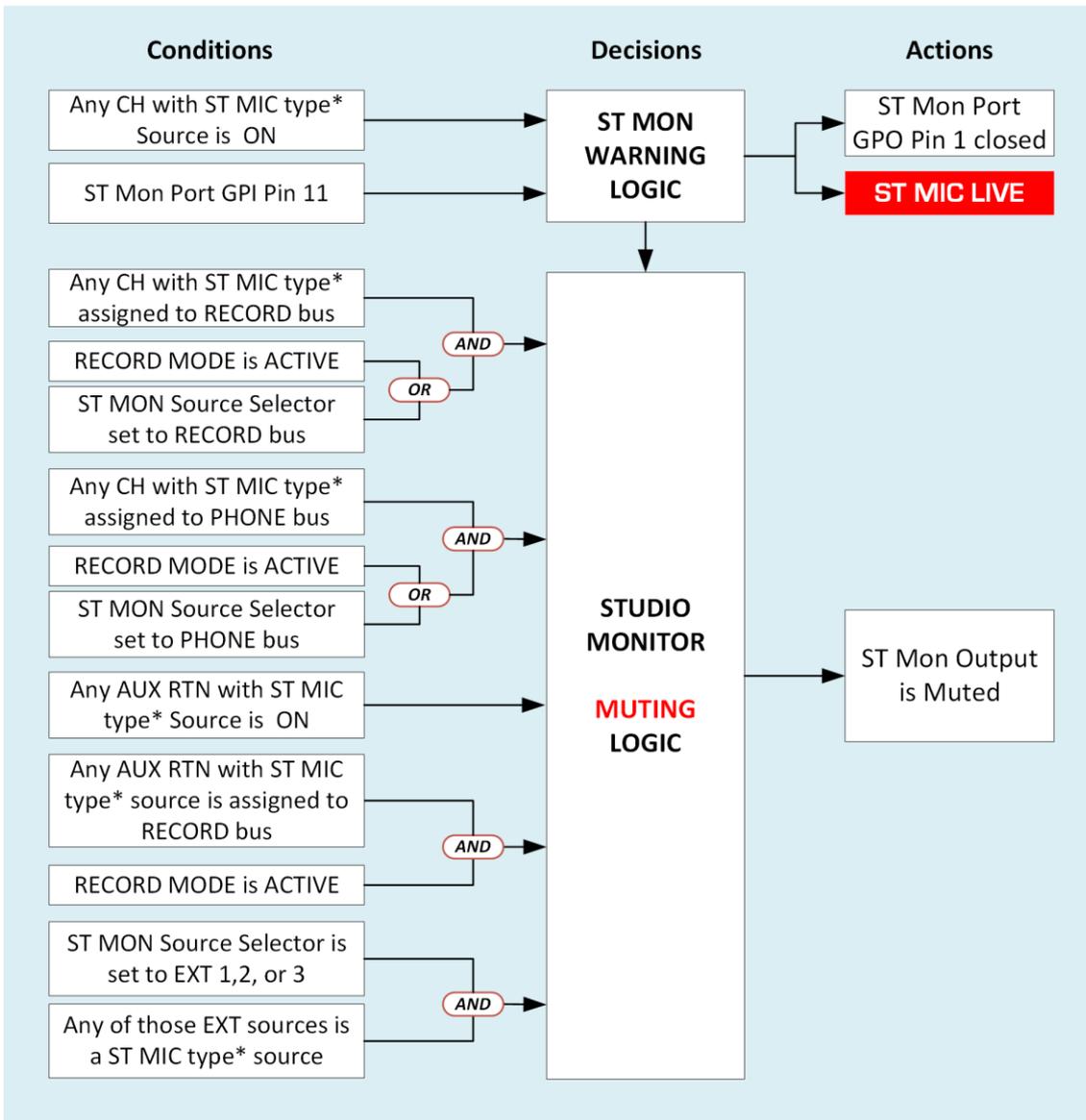
The internal logic will **DIM** the Control Room Speakers when:



\*Operator Mic / CR Guest Mic / CR Producer Mic

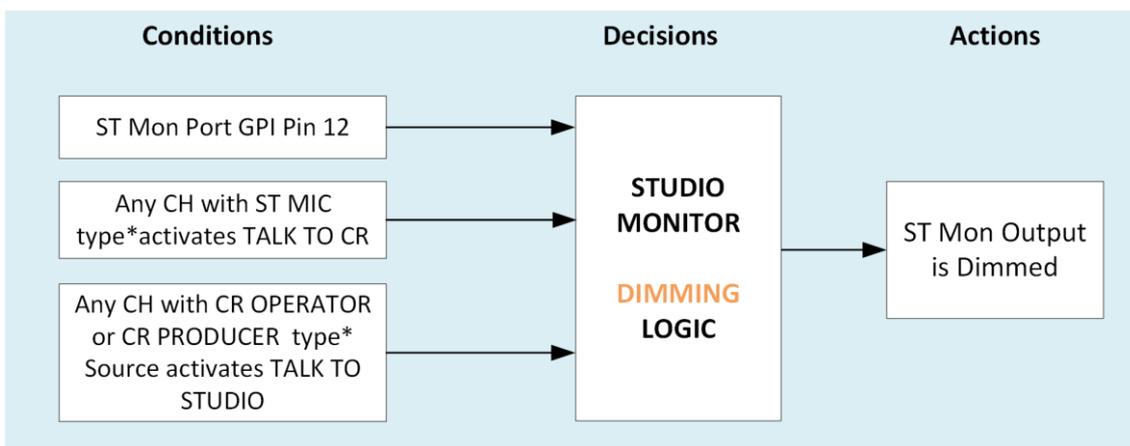
\*\*Talk to EXT, Talk to STUDIO, Talk to Backfeeds, any CH TB

The internal logic will **MUTE** the Studio Speakers when:



*\*STUDIO Guest Mic*

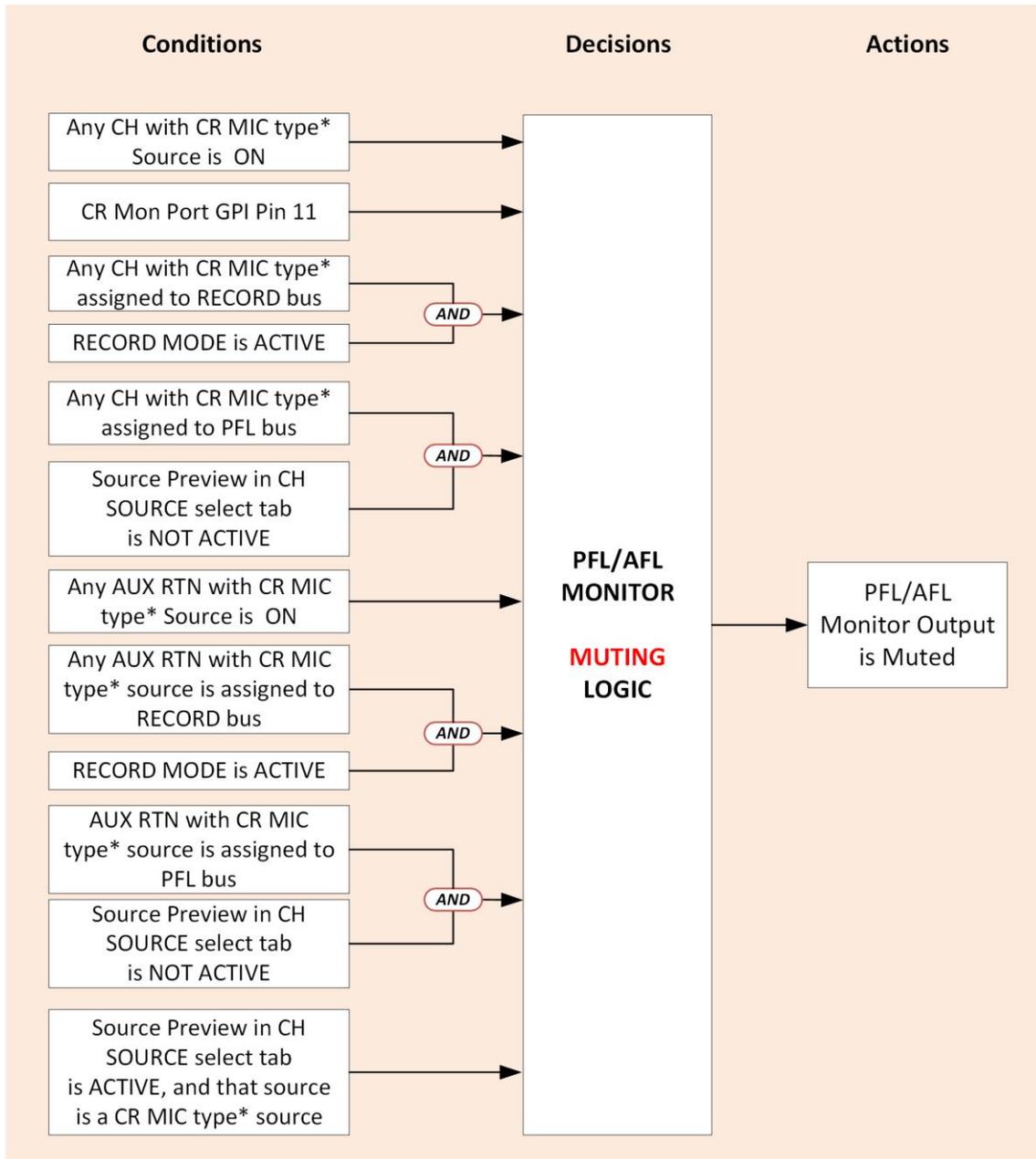
The internal logic will **DIM** the Studio Speakers when:



*\*STUDIO Guest Mic*

**Note** - ST MIC LIVE indicators logic: The Internal logic state (Mic ON/OFF) and GPI Port have true multiplexing logic - both states are monitored and combined into a single state for the corresponding indicator. No one overrides the other.

The internal logic will **MUTE** the PFL Speakers when:



\*Operator Mic / CR Guest Mic / CR Producer Mic

**Note** - There is no Dimming Logic for the PFL Speakers

## CR MIC LIVE and ST MIC LIVE signals

According to the diagrams shown in the previous pages, the condition for triggering the "CR MIC LIVE" lamps is:

- Any Channel with a Control Room Mic type source must be turned ON
- OR**
- Corresponding GPI pin is closed

And the condition for triggering the "ST MIC ON" lamps is

- Any Channel with a Studio Mic type source must be turned ON
- OR**
- Corresponding GPI pin is closed

In case of the MIC LIVE signals (not in case of MON MUTE) the internal logic checks only the Microphones' ON state, without checking the muting logic or the Record Mode state. Also, the GPO pins copy the state of the Mic ON lamps, and no longer copy the muting state.

## MIC LIVE Indicators in Pathfinder

Nothing changed from the LWCP/Pathfinder point of view, for the CR and ST MIC LIVE indicators: the existing implementation is retained.

## MIC LIVE Indicators in Quasar Soft

Quasar Soft offers the two CR MIC LIVE and ST MIC LIVE indicators, but it does not offer the ON AIR indicator. This can be only controlled via an external LWCP command because Quasar Soft does not tunnel LWCP (yet...!).

**Note** - Quasar Soft for now only passes CMSG protocol in the browser-to-console communication. This is the same reason why the Intercom Controller, the Call Controller, and the Hotkeys don't work in Quasar Soft.

## GPIO Control Room Monitor Logic

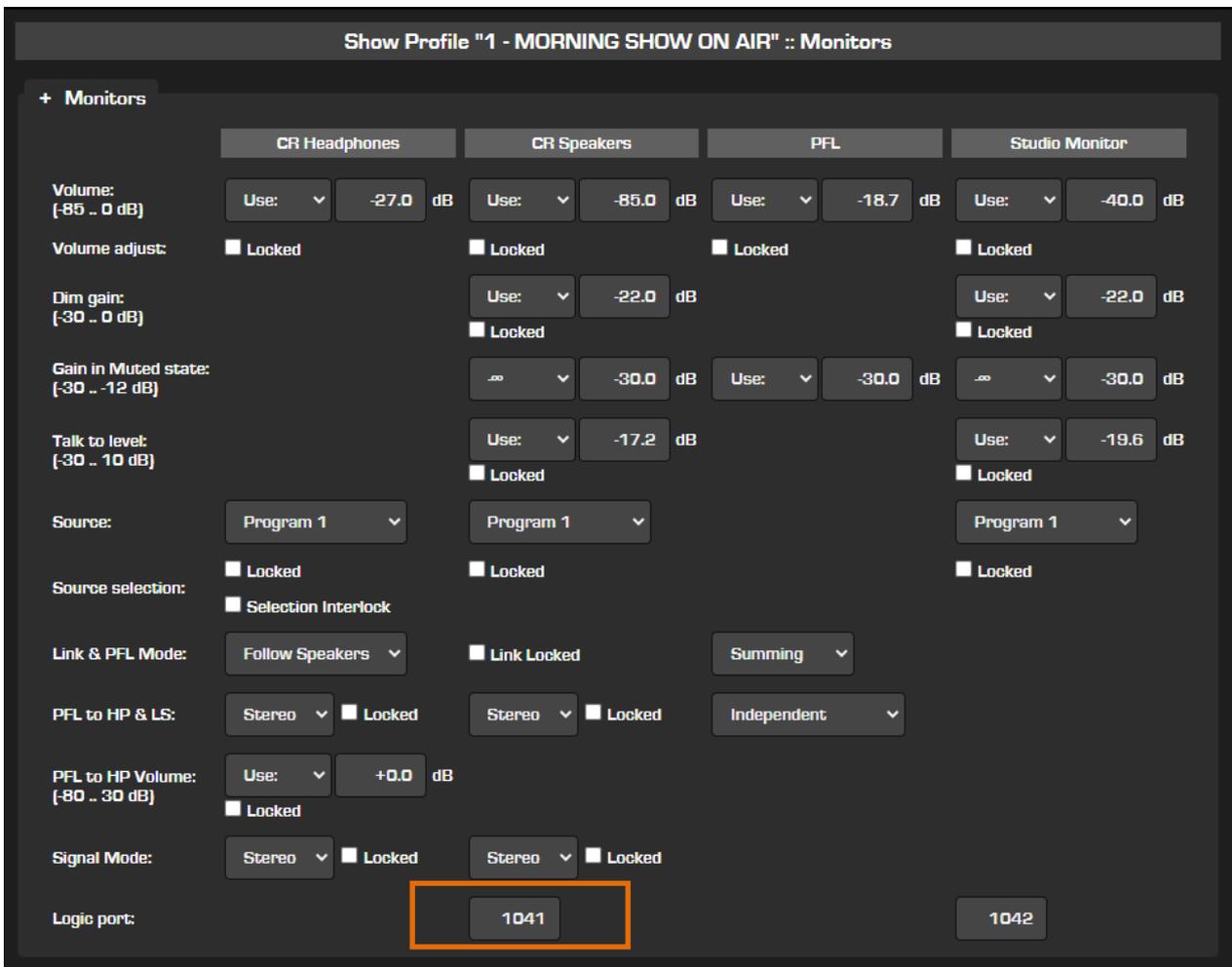
The Control Room (CR) speakers have GPIO logic built-in. Additionally, the console has logic built-in which will mute the monitor speakers if a localized microphone is ON.

Name	Pin	Type	Notes
<b>INPUTS</b>			
MUTE CR MON Command	11	Active Low Input	Mutes CR monitors and PFL speakers
DIM CR Command	12	Active Low Input	Allows external dimming of CR monitor speakers.
Enable EXT PFL Command	13	Active Low Input	Feeds External Audio Input to PFL/AFL bus
TALK TO EXT Command	14	Active Low Input	Turns on Talk to External Audio.
ON AIR Signal*	15	Active Low Input	Turns on the ON AIR indicator in MTS GUI (Home Tab)
<b>OUTPUTS</b>			

CR ON AIR Lamp	1	Open Collector to Logic Common Return	Illuminates whenever CR monitors are muted
DIM CR Lamp	2	Open Collector to Logic Common Return	Illuminates whenever control room monitors are DIMMED
PFL Lamp	3	Open Collector to Logic Common Return	Illuminates when PFL/AFL is active.
TALK TO EXT Lamp	4	Open Collector to Logic Common Return	Illuminates when Talk to External is active.
TALK (to CR) Active Lamp	5	Open Collector to Logic Common Return	Active whenever a source has activated its TALK (to CR) function

In order to use this logic to trigger a light circuit you must follow these steps:

1. Navigate to the Monitor Section of your Show Profile.
2. Locate the **Logic port** field for the CR Speakers: Enter a unique Livewire channel number. A common practice is to enter the LW ch# used for the CR Speaker audio.



3. Save the show profile and load the profile to the surface. When a microphone is turned on, the CR monitor speakers will mute and the logic state will be reported into the network.

The next step is to assign a GPIO port in the network to respond to this message.

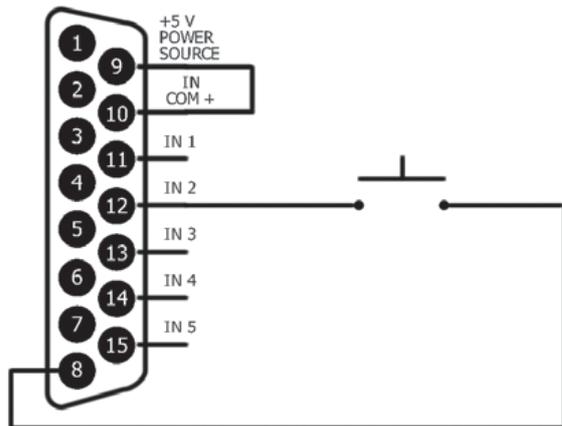
4. Log into an xNode that has GPIO functionality.

5. Assign one of the ports to the same channel that was assigned earlier. In the example here, 1041 was used.

### GPIO

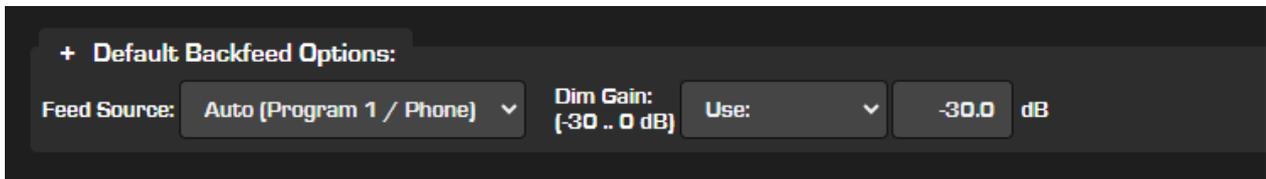
	# Name:	Channel:		Inputs	Outputs
1	<input type="text" value="CR MON DIM/MIC LIVE"/>	<input type="text" value="1041 &lt;CR_SPKR@LLR-Engine1&gt;"/>	<input type="button" value="⌵"/>	■ ■ ■ ■ ■ ■ ■ ■	■ ■ ■ ■ ■ ■ ■ ■
2	<input type="text" value="INTERCOM"/>	<input type="text" value="712 &lt;Intercom Out@MP16_Panel&gt;"/>	<input type="button" value="⌵"/>	■ ■ ■ ■ ■ ■ ■ ■	■ ■ ■ ■ ■ ■ ■ ■

6. In the case of electrically isolated circuit, you would wire the GPIO port as shown in the pinout diagram below:

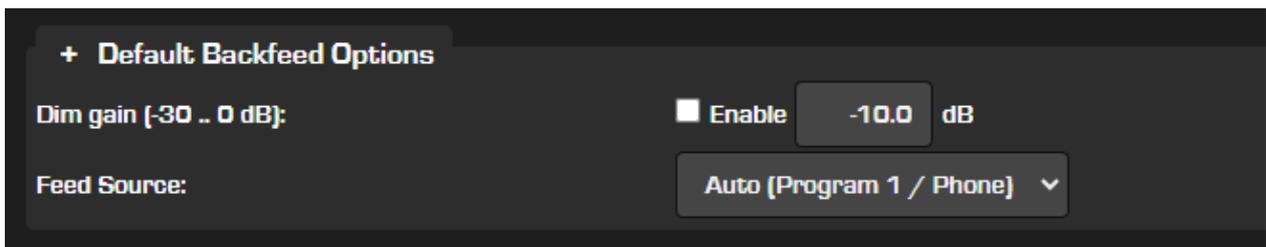


## Setting Up Mix-Minus

In Show profiles, this is where you set the Mix-Minus for Codec and Phone type sources:



There is an equivalent section in the Source Profile:



Again, this will appear only in case of Codec and Phone type sources.

In case the Source is a Microphone type, you can setup the backfeed for that Microphone in the Show Profiles menu, using the **Headphone Input** drop-down menu, and choosing amongst one of the internal console Engine busses, or a predefined external source.



**Note** - In this case, if there is an Accessory module associated to this source, this will be able to control the **Headphones Input** and **Headphones Volume** parameters. This is the place where the Accessory module settings are stored, within a Show.

If the left drop-down menu is set to “External Input”, it is possible to load a Predefined Source to this input by using the right drop-down.

The source selected in the above menu, will also appear on the Accessory Module display, when this is set to “Accessory Panel Mode”. The Show Profile Channel will save any selection entered from the Accessory module in this section.

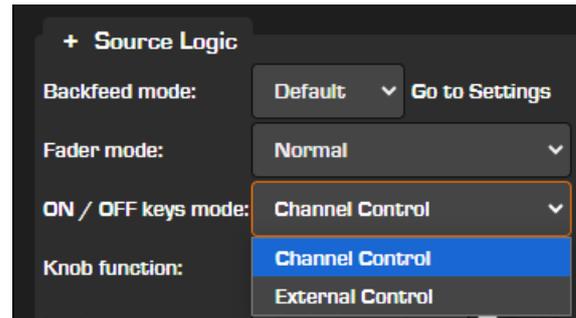
## Setting Up the ON/OFF Keys as Remote Controls

In Software v2.1.0 we have added *an option* to make the Channel ON / OFF keys work as remote control objects in the Source Profile: We can now decouple the ON and OFF keys (the hardware pushbuttons, not the logical or soft buttons) and the built-in Lamps, from the engine channel ON/OFF function, making these keys and lamps controllable directly by LWCP.

This special logic is associated to the Source, so that when the Source is moved across the console fader strips, it will follow the Source. Two modes are available for the ON/OFF Keys logic:

1. **Channel Control** (default)
2. **External Control** (new feature)

**Channel Control** is the standard operation mode, where the ON/OFF keys control the ON/OFF function of the Input channel where the Source is loaded.



When **External Control** is selected and the source is loaded to a fader, the ON/OFF keys will work as follows:

- The ON and OFF keys will no longer control the channel ON/OFF state.
- The ON and OFF lamps will not indicate the channel's ON and OFF states
- The keys and lamps are controlled via the LWCP (Livewire Control Protocol) over TCP.

**Note** - The difference between using this setting and mapping ON/OFF keys to USER keys (using the Keymap function), is that this is Source Type driven, while keys mapped as USER are applied *globally*, to a whole fader module, regardless of the source loaded.

### LWCP Syntax

- Objects are LwCH and FaCH just as all other channel controls.

Pressing the ON and OFF keys while in this mode, generates a press EVENT (as in v2.0.0) where the LwCH# / FaCH# values **ON\_But** and **OFF\_But** follow the key state and where the values **ON\_State** and **OFF\_State** are left unchanged (since they are no longer controlled by the key)

- A remote device can still control the ON state of the channel by sending ON\_State=ON (or OFF) command.
- The remote device can also control ON and OFF key lamps by sending ON\_Lamp=ON (or OFF) and OFF\_Lamp=ON (or OFF)

## Setting Up the Intercom Controller

With the Intercom Controller you can control a Telos Infinity Intercom Panel directly from your surface.

**Note** - Quasar does not support the legacy Axia Intercom system.

Quasar's Intercom Controller is a "soft panel" that simply replicates all the objects found on an Infinity MP-16 or a 16-keys VIP panel. It does not add or remove any feature, and it is a 1:1 copy of the Infinity panel. It is not a substitute for an Infinity panel, and will work only if connected to a panel. No license is required to operate this remote controller.

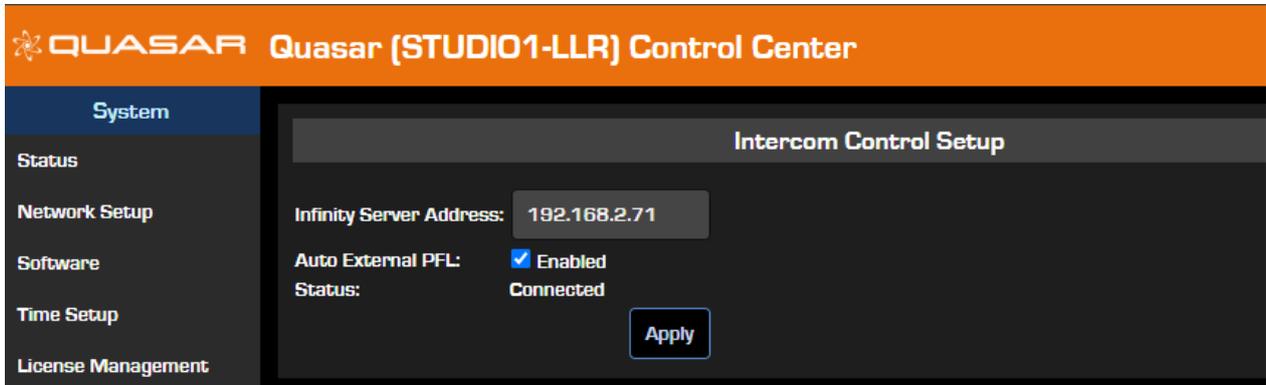


Setup is very easy and consists of just two steps:

- Connection to the panel
- Configuring Monitor integration

## Setting up the connection to the Infinity Panel

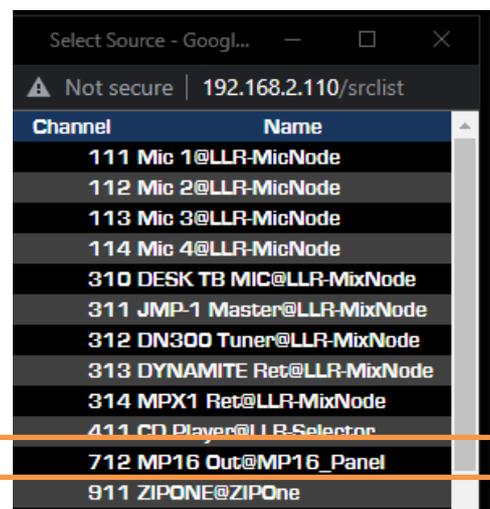
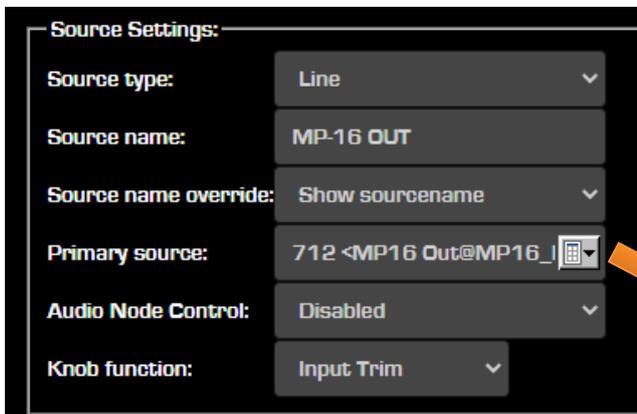
From the Quasar Web UI, select the **Intercom Control** link. In the Infinity Server Address field, type the IP address of the VIP or MP-16 panel you want to control. Press **Apply**



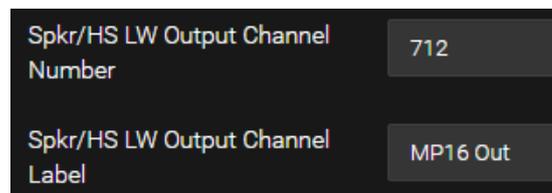
## Setting up the Quasar Monitoring section

The Quasar-Infinity Monitoring integration consists in adding the Infinity panel output to the console's EXT Monitor Input, controlling the External PFL function from the Infinity, and controlling the Infinity Speaker muting function from the console. You can set this up by performing the following steps:

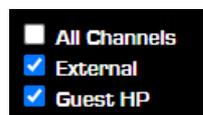
- Create a new Source Profile, and assign the output of the Infinity Panel as source to the profile



The LW channel and Name for this can be found in the **Livewire+/AES67** section of the Infinity **Setup** page. You must look for the "Spkr/HS LW Output Channel Number" field.



**Note** - You must flag the **External** box in the source Availability section, or it will not appear in the Ext PFL drop-down menu.



- Then, select the desired Show Profile, and navigate to its **Monitor Section**: Scroll Down

and go to the **Externals** section of this page. Here you will select the Source you just created, in the **Ext PFL** drop-down. This will let Quasar inject the intercom audio into its CR Monitors, and DIM the CR speakers every time the EXT PFL input is triggered by the Infinity Panel.

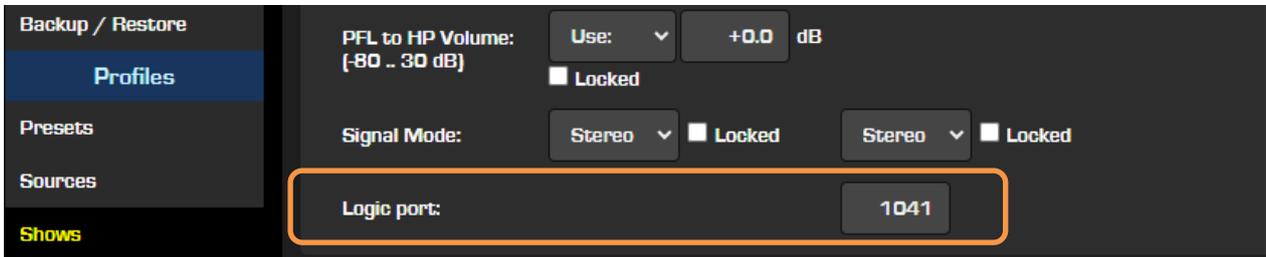


### Automatic Activation of Quasar’s External PFL

In case your Infinity Intercom is running SW v3.1 or later, you can activate the Quasar External PFL function by simply checking the **Auto External PFL** option: the new software will make the Infinity signalize that a call is active without the need to use non-routable Multicast GPIOs.

In case your Infinity Intercom is running an earlier version of software, a standard Multicast GPIO must be used to trigger the activation of the External PFL function in Quasar:

- Scroll the page up and enter the desired LWGPIO channel# in the **Logic Port** field of the CR Speakers section



- Access the Infinity Panel’s Web UI, and navigate to the GPIO section of the **Setup** page. In the field labeled “Axia console Control Monitor GPIO port LW Channel”, enter the same LWGPIO Channel # you entered above, in the Quasar Logic Port field.



**Note** – The first method is the preferred one, so we encourage you to update your infinity panels to the latest version. This will also free up the Logic port field in the monitor section of your console, making it available to other applications.

## Connecting to a VIP Panel

One of the differences between the hardware and virtual panels is that VIPs support audio on the /remote WEB page. This feature is not needed in case of remote control from Quasar, and blocks the connection to the console.

In order to connect Quasar to a VIP panel, the WebAudio functionality must be disabled.

To disable that feature, add a parameter to the VIP container: WEBAUDIO=disabled

This is how it looks in Portainer:

name	WEBAUDIO
value	disabled

[- Remove value](#)

**Note** - In this case, Audio on the VIP panel will be disabled.

# Configuration 6 – Metering

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## Audio Metering Options

Both the Master and Fader Modules in Quasar offer extensive Metering capabilities. The options provided will let you adapt the console to your facility workflow, and monitoring requirements.

### Main Meters

Located in the top section of the MTS Module Touchscreen UI, this group of large hi-resolution meters is provided to accurately monitor the following signals:

- Program 1 bus
- Program 2 bus
- Program 3 bus
- Switchable Meter: Program 4 bus is Default – can be assigned from the Web UI
- PFL / AFL bus
- CR Monitor bus

The ballistics of these meters is programmable from the Web UI, and does not require a reboot of the console, in order to be applied.

### Monitor Output Meters

In the lower part of the MTS Module Touchscreen UI, a number of confidence-class meters is also available for the following Monitor outputs:

- CR Headphones
- CR Speakers
- Talk to Studio
- Talk to Control Room
- Studio Loudspeakers
- PFL / AFL output

These meters show the post-volume knob output level.

### Channel Meters

From the Master Module Touchscreen UI, the following meters are available for each channel:

- Channel Direct Out/Insert Send (follows the user-selectable position)
- Channel Insert Return (follows the user-selectable position)
- Channel Input Meter (right after source input, before Delay and Trim Gain)
- Channel Output Meter (after fader, before Automix Gain stage)
- Backfeed Output Meter

The ballistics of these meters is programmable from the Web UI, but it requires a reboot of the console, in order to be applied.

- Dynamics Input, Gain Reduction, and Output meters
- Ess Signal Meter (monitors the De-Esser sidechain)

- Auxiliary 1-8 Sends Meters (confidence class)

The ballistics of these meters is not programmable. They are all high resolution full-scale VU with Quasi-PPM reading (small peak segment) except for the Aux Send Meters, which are confidence class.

From the Fader Modules, the following meters are available, for each channel:

- Channel Input Meter (confidence class) - XR and SR modules
- Backfeed Output Meter (confidence class) - XR and SR modules
- Fader Bargraph Meter (confidence class) - XR modules only
- Automixer Gain Reduction meter (confidence class) - XR modules only

### Auxiliary Meters

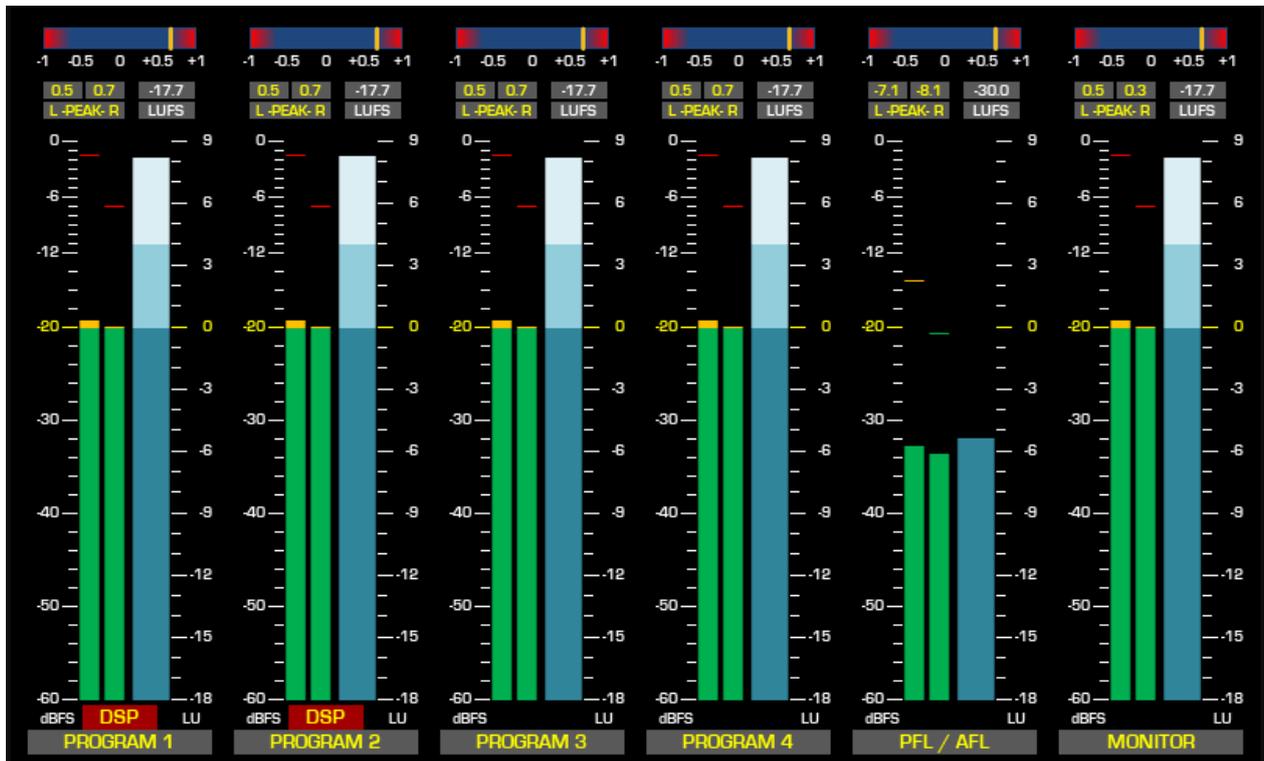
Eight confidence-class meters are available for the Master Auxiliary Sends. These always display the Aux Bus level, post-Gain.

Two confidence-class meters are available for the Master Auxiliary Returns. These always display the Return Channel Input level, Pre-Gain.

Eight confidence-class meters are available for the Channel Auxiliary sends. These will display the Aux Send level, following the Pre/Post-Fader and Pre/Post ON settings of each Send.

## The Main Meters

Starting from the leftmost meter, the first four are permanently assigned to each of the four Program busses, the fifth meter is assigned to the PFL/AFL bus, and the sixth one is assigned to the CR Monitor bus.



**Program 1-4:** The level is displayed for the PGM1-4 bus before Master Gain and Bus Processing

**PFL/AFL:** The level is displayed for the PFL/AFL bus before PFL Level

**Monitor:** The level is displayed for the CR Monitor bus after Master Gain and Bus Processing, and before CR Speakers Level

**Note** – The meters indicate the level of the audio signal before DSP processing is applied in order to be visually synchronized with the audio monitored through the CR Headphones or Speakers. The CR monitor outputs are picked up after Filters and Dynamics processing, but before the TP Limiter, in order to avoid the latency associated with True Peak Limiting.

The Label below each of the meters, indicating the bus name can be of two different colors depending on the assignment state of each of the channels:



White color indicates that no input channels is assigned to that particular bus regardless if it is turned ON, and audio is present.



Yellow color indicates that at least one of the input channels is assigned to that particular bus, regardless if it is turned ON, and audio is present.

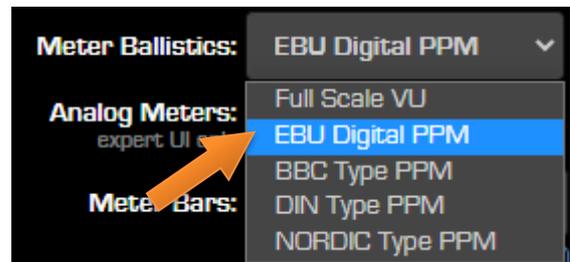
The presence of the red “DSP” Label above the bus name (new in v3.0) indicates that DSP processing (either Filters, Dynamics, or True Peak Limiter) has been activated on that particular bus.



## Meter Ballistics

The **Meter Ballistics** option, available in the Customize section of Web UI Configuration menu, lets the user choose between six possible Meter Types, each with its own Ballistics, for the Main Meters:

- Full Scale VU
- EBU Digital PPM
- BBC Type PPM
- DIN Type PPM
- NORDIC Type PPM



## Quasar Meter Types Specifications

### Full Scale VU

Scale: -60 to 0dBFS  
Reference level mark: -20  
Color transition: -20dBFS  
Attack time: 300ms for 20dB  
Time constant: 65ms to reach 63% of target  
Release time: same as attack time  
Calibration: -20dBFS = +4dBu

### Analog VU

Scale: A VU scale, extending from -20 to +3, with 0 marked at about 71% max scale reading  
Reference level mark: 0

Color transition: 0dB/-20dBFS  
 Attack time: 300ms for 20dB  
 Release time: same as attack time  
 Calibration: 0VU = +4dBu

### EBU Digital PPM

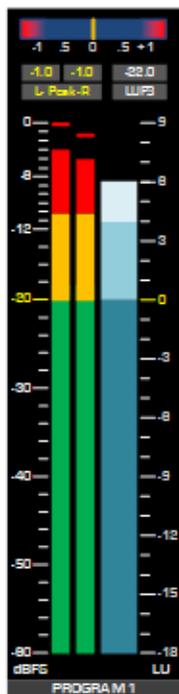
Scale: -60 to 0dBFS  
 0dB mark: 0dBfs  
 Reference level mark: -18dBFS  
 Color transition: -9dBFS  
 Attack time: 80% in 5ms  
 Release time: 20dB in 1.7s (12dB/s)  
 Calibration: -18dBFS = 0dBu

### BBC Type PPM

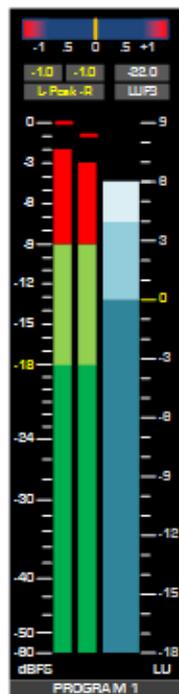
Reference level mark: none (4 usually corresponds to -18dBFS and to 0dBu analog)  
 Color transition: 6.2 (-9dBFS)  
 Attack time: 80% in 10ms  
 Release time: 24dB in 2.8s (8.6dB/s)  
 Calibration: -18dBFS = 0dBu  
 Normally, “4” corresponds to -18dBFS. But some installations prefer to have -12dBFS at this mark.

### DIN Type PPM

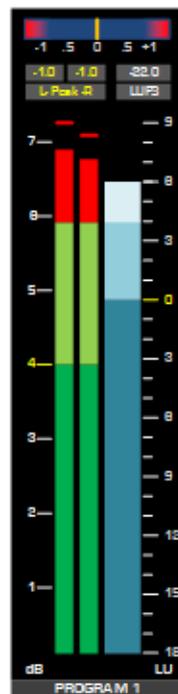
Scale: -60 to +5dB  
 0dB mark: -9dBfs  
 Reference level mark: -9dB/-18dBFS  
 Color transition: 0dB/-10dBFS  
 Attack time: 80% in 5ms  
 Release time: 20dB in 1.7s (12dB/s)  
 Node: -18dBFS = 0dBu



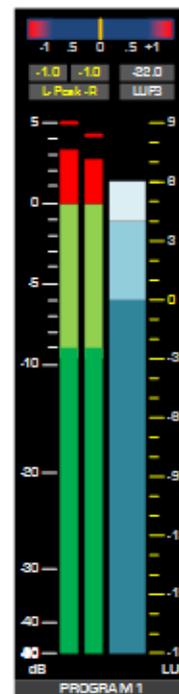
FULL SCALE VU



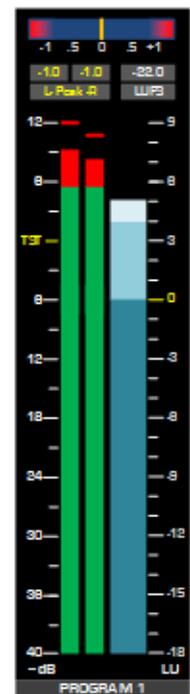
EBU DIGITAL



BBC-TYPE PPM



DIN-TYPE PPM



NORDIC N9

## Nordic Type PPM

Reference level mark: "TEST" at 0dBr/-18dBFS

Color transition: +6dBr/-9dBFS

Attack time: 80% in 5ms

Release time: 20dB in 1.5s

Calibration: -18dBFS = 0dBU

Programme Meter Type	Recommendation	PML <sup>a</sup> 100%	Limit Level	Scale	Attack time (integration)	Decay time (fall-back)	Invisible peaks
VU Meter	ANSI C 16.5 IEC 268-17	0 VU +0 dBU		-20 to +3 [dB]	300 ms / 90%	300 ms / 10%	+13 ... +16 dB
DIN PPM (QPPM)	DIN 45406 IEC 268-10/1 ARD Pfl.H.3/6	0 dBr +9 dBU	+16 dBr +25 dBU	-50 to +5 [dB]	10 ms / 90% 5 ms / 80%	20 dB / 1.5 s = 13 dB/s	+3 ... +4 dB
BBC PPM (QPPM)	IEC 268-10 / IIa	'6' +8 dBU		1 to 7 [ ]	10 ms / 80%	24 dB / 2.8 s = 8.6 dB/s	+4 ... +6 dB
EBU PPM Std (QPPM)	EBU 3205 E IEC 268-10 / IIb	+9 dB +9 dBU		-12 to +12 [dB]	10 ms / 80%	24 dB / 2.8 s = 8.6 dB/s	+4 ... +6 dB
EBU Digi PPM (QPPM)	EBU IEC 268-18	-9 dBFS	0 dBFS	-40 to +0 [dB]	≤ 5 ms / 80%	20 dB / 1.7 s = 12 dB/s	+3 ... +4 dB
IRT Digi PPM (QPPM)	IRT proposal	0 dBr 100%	≤ +10 dBr	-50 to +10 [dB]	5 ... 10 ms / to 80%	20 dB / 1.7 s = 12 dB/s	+3 ... +4 dB

## Peak Meter with Hold

All meter types have a True Peak meter that displays as a single segment. This has instant attack time and a non-linear release time that holds a new peak value for three seconds with no change, then releases at a 20dB/1.5sec rate.

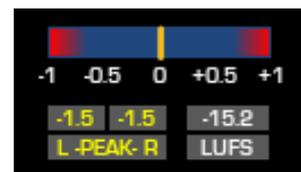
## Max True Peak

Above each main meter, there is a Max True Peak Meter, and Max LU Meter. The L-PEAK-R Indicator shows max true peak level, as described in ITU-R BS.1771 and uses oversampling to calculate the maximum peak levels. LUFS shows integrated loudness value since reset in LUFS units.

Peaks are measured by the single segments and also by the text indicators with the label "Peak". You must push the **START/STOP/RESET LU MTR** button to reset the peak readings. Playing a -20dFS Test Tone calibrated at the peaks, you will see the solid green bar reading -23dBFS (which corresponds to -3VU), and you will notice the peak segments measuring -20dBFS.

## Phase Meter

A phase meter is included at the top of each main meter. A +1 reading indicates the Left and Right channels are in phase together. A -1 reading indicates the Left and Right channels are out of phase with one another. A presentation of a single yellow center indicator states complete stereo separation between the left and right channels.



## The Loudness Meters

Each of the six Main Meters has a Loudness bargraph Meter associated with it. These particular meters implement the Standard “EBU Mode” and are based on ITU.BS-1770. Some of the options provided are specific to EBU R128, and compliant with the EBU Tech 3341 specifications.

The ABSOLUTE LOUDNESS and the RELATIVE LOUDNESS boxes will display numerical values for the following types of loudness measurements:



- Momentary (defined as ungated loudness with an LPF time constant of 400ms),
- Short (integration over a time interval of 3000ms),
- Integrated (integration performed over the whole time period between two Resets).

You can choose between Absolute and Relative modes, by selecting an option in the Web UI.

Below the three Relative Loudness readings, the Maximum value recorded for Short and Momentary term within the time segment are also shown. This segment is shown in the last field, and is defined by operating the **START/STOP/RESET** button on the Touchscreen.



Below there are two Max True Peak values recorded for the Left and Right channels. Below the true peak meter is the time counter.

At the time of writing this manual, the Loudness Counters apply only to the PGM 1 bus.

### Relative Loudness

Enabling the relative loudness option changes the numerical presentation so that the number is in reference to the target. A value of 0 LU indicates the target value is achieved. Lower than target value will be negative (-) LU while positive (+) LU indicates higher than target.

### Loudness Range

Loudness range (LRA) is a supplemental measurement for loudness. It is a statistical distribution of measured loudness that quantifies the variations. The measurement is presented in LU. A typical promo would have an LRA of 5 +/-1 LU. A typical theatrical drama would have an LRA of 15 +/-1 LU.

### Control

Other than enabling the loudness meter for your facility’s needs, the program segment length is needed to be defined by the operator or other control point. When loudness metering is enabled, the **LOUDNESS START/STOP/RESET** button on the Touchscreen UI is used for control of the integration timer. In addition, a GPIO channel number may be defined to

provide control of the integration time from another location other than the operator.

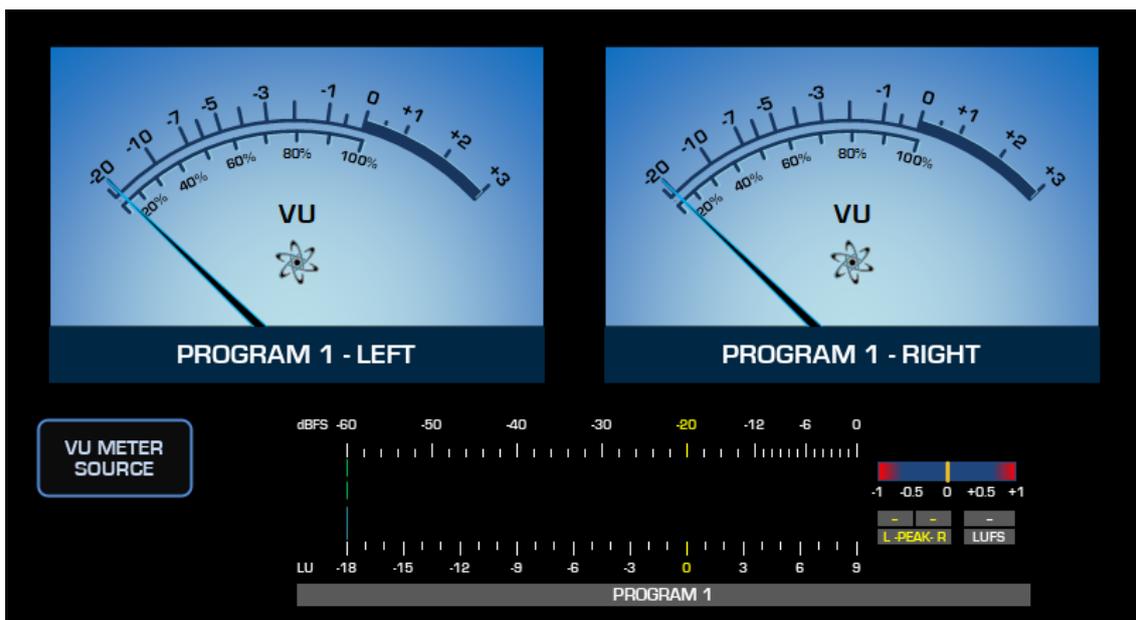
The pinout is:

Input Pin	Connector Pin	Function
1	11	START
2	12	STOP
3	13	RESET

Please refer to the next Chapter of this manual: [GPIO Loudness Metering Logic](#), for more details.

## The Analog VU Meters

Volume units (VU) were originally developed in 1939 by Bell Labs and broadcasters CBS and NBC for measuring and standardizing the levels of telephone lines. The instrument used to measure VU was called the Volume Indicator (VI) meter.



Everyone ignores this, of course, and calls it a *VU meter*. The behavior of VU meters is an official standard, originally defined in ANSI C16.5-1942 and later in the international standard IEC 60268-17. These specify that the meter should take 300ms to rise 20 dB to the 0 dB mark when a constant sine wave of amplitude 0 VU is first applied. It should also take 300ms to fall back to 20 dB when the tone is removed. This integration time is quite long relative to audio wavelengths, so the meter effectively incorporates a filter that removes peaks in order to show a long-term average value.

The input signal to the VU Meter can be changed between PGM1 to PGM4 and RECORD busses by using the **METER SOURCE** button below it.

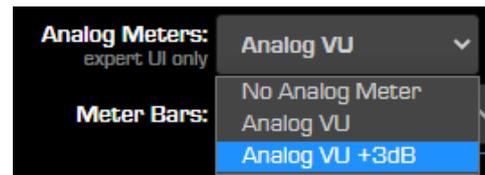
## Reference Level Alignment

For the VU meter, the 0-VU point is used for both the setting of sine-wave reference tones, and as the usual operator target for maximum audio signal level. On the other hand, the PPM's greater sensitivity to peaks means that the lower crest factor of sine waves will not deflect the meter as much as typical audio signals. So, sine-wave alignment reference tones are set at a lower level than the typical audio gain-riding target on PPMs, typically 14-18 dB below the meter's maximum level. (This level is sometimes labeled "TEST" on PPMs.) So, the VU meter retains the singular advantage of having its "nominal level" and "reference level" at the same value (0 VU), marked with a visible change in the scale, from a thin line to a thick arch.

Some important factors need to be taken into account, when calibrating your system:

- Quasar's Analog VU Meters are fully compliant with IEC Standard 60268-17 and therefore they are measuring RMS signal levels.
- when tested with a tone generator whose output has been calibrated to the Peaks of the sine wave, the Analog VU will measure the RMS values from that peak-to-peak signal, and therefore it will read a level which is 3 dB lower. That's what happens when an RMS meter is used to measure a Peak-to-Peak signal!

But we thought about this case, and we added the possibility to calibrate VU meter scale to *Peak signals*, by means of selecting the **Analog VU +3dB** option.



- when tested with a tone generator whose output has been calibrated to the RMS value of the sine wave, the Analog VU will read the correct input amplitude, so 0VU.

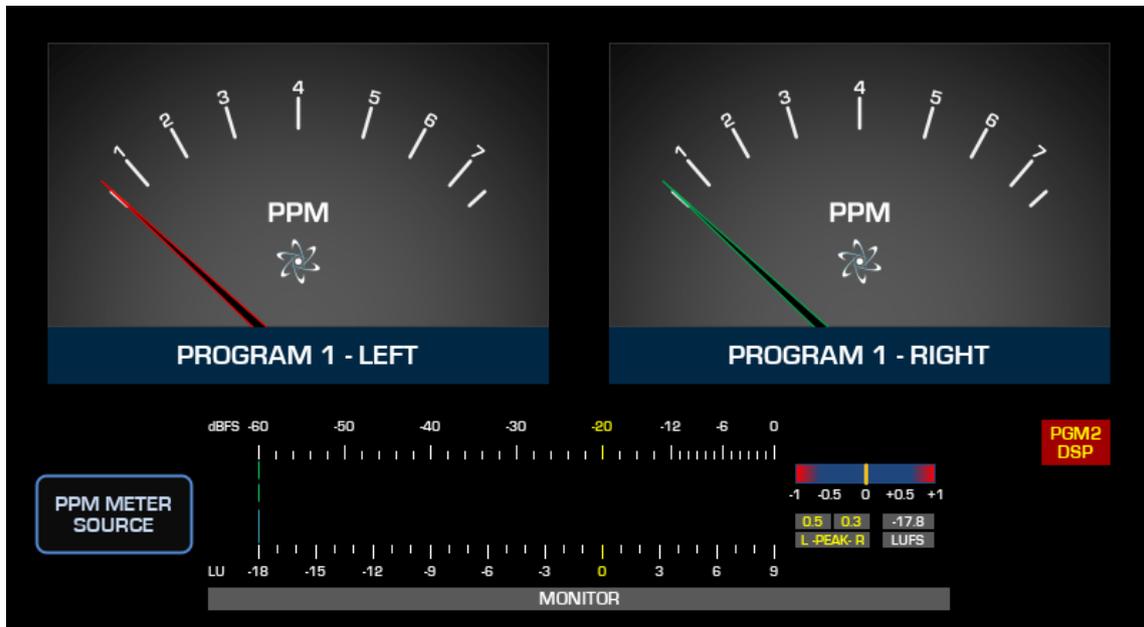
That's because in this case, an RMS meter is being used to measure an RMS signal.

**Note** - IEC 60268-17 Standard defines that the reading of the Volume Indicator shall be 0 VU when it is connected to an AC voltage equal to 1.228 Volts RMS across a 600ohm load, equal to +4dBu. This corresponds to a power of 2.5 milliwatts. The level specification is meant at 1000 Hz.

Below the Analog VU we have added a digital bar-type Full Scale VU with True Peak indicators. Being a Full Scale VU, this horizontal bargraph meter is still measuring RMS levels, and it is not a Peak Program Meter (PPM).

## The Analog PPM Meters

The principal standard for the Quasar PPMs is IEC 60268-10 Type II, equivalent to the British standard BS 6840-10:1991. It describes a quasi-PPM design that has a 4 dB spacing between the scale marks, and only show the true level of the peak if it exceeds a certain duration, typically a few milliseconds. On peaks of shorter duration, it indicates less than the true peak level.



Quasi-Peak Program meters (QPPMs) have their roots in meters originally developed in the 1930s for the AM radio broadcasting networks of the United Kingdom.

Many engineers prefer the PPM to the much slower VU meter used in the US, but it does require some experience to correctly read its indicators. Though it gives a useful warning about a potential level overload, it does not represent either true peak level or subjective loudness.

Regardless of the kind of program, there is usually a nominal Permitted Maximum Level, as indicated on a PPM. Operators are expected to keep levels below it, within reason. Practices vary between countries and organizations. In the UK, the Permitted Maximum Level is 8 dB above Alignment Level, corresponding to '6' on the British PPM scale.

Out of curiosity... unlike many would assume, the colors of the needles do not follow the audio standard (where an input or an output marked red usually indicates the right channel). Instead, the red needle is used for the left channel and the green needle for the right channel, following a color convention borrowed from navigation: port and starboard lights are in fact red for the left side and green for the right side of the boat.

## The Channel I/O Meters

The Input and Output Meters in the Channel Input tab, have a bargraph which shows the RMS Level, plus a Peak Segment, unlike the Main Meters in the MTS HOME page, which have a bargraph which shows the Peak Level.

At first glance it may appear that the PFL meters don't match the Input/Output meters in the Source Menu, but in fact they do match: only the presentation is different  
The picture >>

Shows that the Peak segment is exactly at the same height in both the PFL and Input meters. The bar in the Input meter is lower because it shows RMS values.

# Configuration 7 – GPIOs

## GPIO Configuration

The Axia IP-Audio system is capable of transporting routable machine logic along with each audio signal. Unlike conventional logic connections which require each command circuit to be wired individually, Axia sends machine controls over the same Ethernet port upon which your audio travels. Quasar’s GPIO capabilities provide control of external audio devices, logic commands for routine studio/control room operations such as tally lights, monitor muting, On-Air lights and more, and even “virtual” GPIO for routing system commands, using Axia Pathfinder routing controls tools.

This chapter provides a fast overview of these GPIO functions. Please refer to the [Axia xNode User’s Manual](#) for more in-depth information on configuring GPIO.

### GPIO Port Definitions

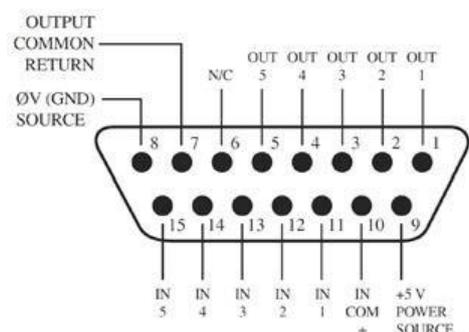
Axia Quasar consoles must be paired with xNode GPIO Nodes, in order to control, or be controlled via GPIO. Each GPIO port can be associated with a device in your studio, and provides five opto- isolated inputs and five opto- isolated outputs per device.

GPIO ports are pre-programmed to support several different types of devices; when you construct Source Profiles, the GPIO type best suited for the Source Type you choose is associated with that Profile. When the source is assigned to a console fader, this Source Profile selection tells the GPIO port what sort of command to send to the attached device. If the Source Profile defines the attached device as a microphone, the GPIO port sends logic for On, Off, Remote Mute and Remote Talk commands on the appropriate pins. If the Source Profile is configured for a line input, the GPIO port sends Start, Stop and Reset commands, plus closures for Ready lights, etc.

Axia GPIO ports can deliver unique command sets for the following types of devices:

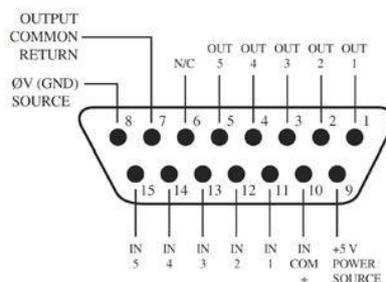
1. Microphone (Operator, Guest or Producer)
2. Line Input
3. Codec
4. Telephone Hybrid
5. Computer Playback Device
6. Control Room Monitor
7. Studio Monitor
8. Accessory Button Panel Device

The next few pages contain tables that explain what function the GPIO port pins provide in each different device mode.



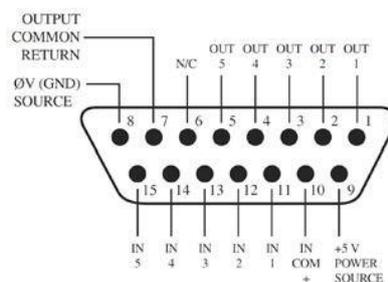
## GPIO Control Room Operator's Microphone Logic

Name	Pin	Type	Notes
<b>INPUTS</b>			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to STUDIO) Command	13	Active Low Input	Activates the TALK TO STUDIO function and routes mic audio to the Talkback bus.
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to SOURCE with PFL active) Command	15	Active Low Input	Activates the TALK key on every source currently in PFL and routes mic audio to the Talkback bus.
<b>OUTPUTS</b>			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
TALK (to STUDIO) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK TO STUDIO is active
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to SOURCE with PFL active) Lamp	5	Open Collector to Logic Common Return	Illuminates when TALK to PFL SOURCE is active.
<b>POWER &amp; COMMON</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic +5 Volt Supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Input Common	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



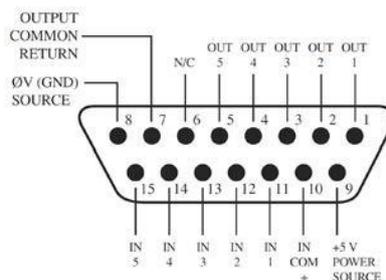
## GPIO Control Room Guest Microphone Logic

Name	Pin	Type	Notes
<b>INPUTS</b>			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to CR) Command	13	Active Low Input	Mutes channel outputs and routes source audio to PFL speakers
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to SOURCE) Command	15	Active Low Input	Allows an external key to activate channel TALK TO SOURCE function.
<b>OUTPUTS</b>			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
TALK (to CR) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK is active
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to SOURCE) Lamp	5		
<b>POWER &amp; COMMON</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



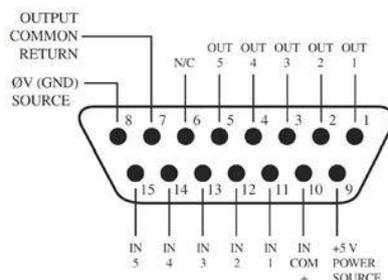
## GPIO Control Room Producer's Microphone Logic

Name	Pin	Type	Notes
<b>INPUTS</b>			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to STUDIO) Command	13	Active Low Input	Activates the TALK to STUDIO function and routes mic audio to the Talkback bus.
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to SOURCE with PFL active) Command	15	Active Low Input	Activates the TALK key on every source currently in PFL and routes mic audio to the Talkback bus.
<b>OUTPUTS</b>			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
TALK (to STUDIO) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK to STUDIO is active.
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to SOURCE with PFL active) Lamp	5	Open Collector to Logic Common Return	Illuminates when TALK to PFL SOURCE is active.
<b>POWER &amp; COMMON</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



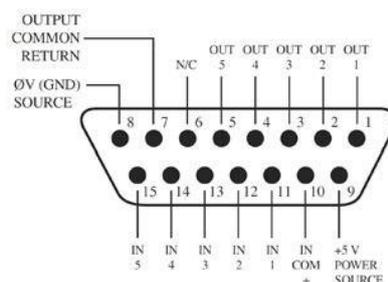
## GPIO Studio Guest Microphone Logic

Name	Pin	Type	Notes
<b>INPUTS</b>			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to CR) Command	13	Active Low Input	Mutes channel outputs and routes source audio to PFL speakers
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to SOURCE) Command	15	Active Low Input	Allows an external key to activate channel TALK TO SOURCE function.
<b>OUTPUTS</b>			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
TALK (to CR) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK is active.
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when the channel TALK TO SOURCE function is active.
<b>POWER &amp; COMMON</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



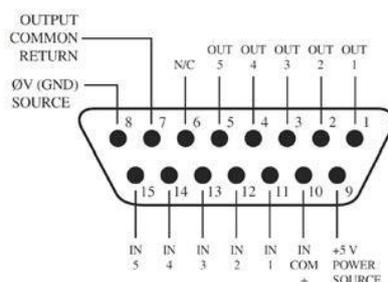
## GPIO External Microphone Logic

Name	Pin	Type	Notes
<b>INPUTS</b>			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to CR) Command	13	Active Low Input	Mutes channel outputs and routes source audio to PFL speakers
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to SOURCE) Command	15	Active Low Input	Allows an external key to activate channel TALK TO SOURCE function.
<b>OUTPUTS</b>			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
TALK (to CR) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK is active.
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when the channel TALK TO SOURCE function is active.
<b>POWER &amp; COMMON</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



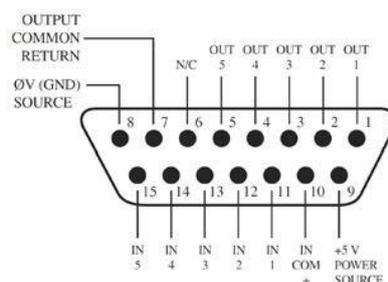
## GPIO Line Input Logic

Name	Pin	Type	Notes
<b>INPUTS</b>			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF & sends 100 msec STOP pulse
PFL Command	13	Active Low Input	Turns PFL/AFL ON
RESET Command	14	Active Low Input	Turns channel OFF, while not sending a STOP pulse
READY Command	15	Active Low Input	Illuminates OFF lamp to indicate source's readiness
<b>OUTPUTS</b>			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF and READY is active
PFL/AFL Lamp	3	Open Collector to Logic Common Return	Illuminates when PFL is ON
START Pulse	4	Open Collector to Logic Common Return	A 100 msec pulse when the channel status changes from OFF to ON
STOP Pulse	5	Open Collector to Logic Common Return	A 100 msec pulse when the channel status changes from ON to OFF
<b>POWER &amp; COMMON</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



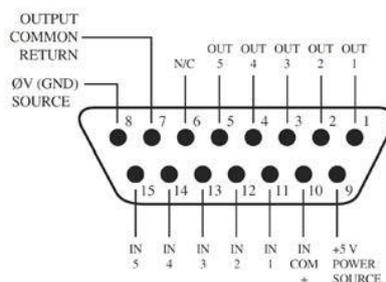
## GPIO Codec Logic

Name	Pin	Type	Notes
<b>INPUTS</b>			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF
TALK (to CR) Command	13	Active Low Input	Mutes channel outputs and routes source audio to PFL speakers
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to SOURCE) Command	15	Active Low Input	Allows an external key to activate channel TALK TO SOURCE function.
<b>OUTPUTS</b>			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE are active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF.
TALK (to CR) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK is active
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when the channel TALK TO SOURCE function is active.
<b>POWER &amp; COMMON</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



## GPIO Telephone Hybrid Logic

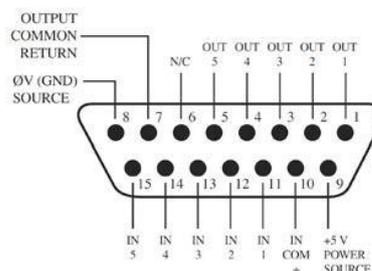
Name	Pin	Type	Notes
<b>INPUTS</b>			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF
PFL Command	13	Active Low Input	Turns PFL/AFL ON
RESET Command	14	Active Low Input	Turns channel off while not sending a STOP pulse
READY Command	15	Active Low Input	Illuminates OFF lamp to indicate source's readiness
<b>OUTPUTS</b>			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
PFL Lamp	3	Open Collector to Logic Common Return	Illuminates when PFL/AFL is ON
START Pulse	4	Open Collector to Logic Common Return	A 100 ms PULSE is sent when channel is first turned ON or when PVW is first selected
STOP Pulse	5	Open Collector to Logic Common Return	A 100 ms PULSE sent when channel is turned OFF.
<b>POWER &amp; COMMAND</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



## GPIO Control Room Monitor Logic

Name	Pin	Type	Notes
<b>INPUTS</b>			
MUTE CR MON Command	11	Active Low Input	Mutes CR monitors and PFL speakers
DIM CR MON Command	12	Active Low Input	Allows external dimming of CR monitor speakers.
Enable EXT PFL Command	13	Active Low Input	Feeds External Audio Input to PFL bus
TALK TO EXT Command	14	Active Low Input	Turns on Talk to External Audio.
ON AIR Signal*	15	Active Low Input	Turns on the ON AIR indicator in MTS GUI (Home Tab)
<b>OUTPUTS</b>			
CR MIC LIVE Lamp *	1	Open Collector to Logic Common Return	Illuminates whenever a MIC is LIVE in the Control Room
DIM CR Lamp	2	Open Collector to Logic Common Return	Illuminates whenever control room monitors are DIMMED
PFL Lamp	3	Open Collector to Logic Common Return	Illuminates when PFL/AFL is active.
TALK TO EXT Lamp	4	Open Collector to Logic Common Return	Illuminates when Talk to External is active.
TALK (to CR) Active Lamp	5	Open Collector to Logic Common Return	Active whenever a source has activated its TALK (to CR) function
<b>POWER &amp; COMMON</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		

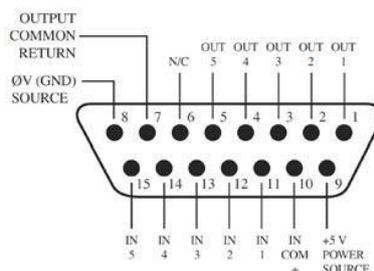
\*Feature added with software v2.1



## GPIO Studio Monitor Logic

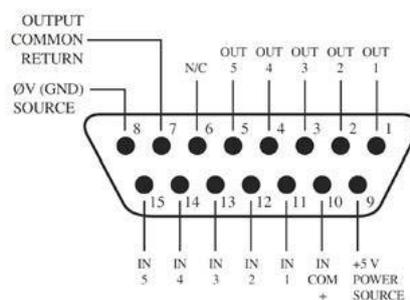
Name	Pin	Type	Notes
<b>INPUTS</b>			
MUTE STUDIO MON Command	11	Active Low Input	Mutes Studio monitor speakers.
DIM STUDIO MON Command	12	Active Low Input	Allows external dimming of Studio monitor speakers.
Not used.	13	Active Low Input	
Not used.	14	Active Low Input	
Not used.	15	Active Low Input	
<b>OUTPUTS</b>			
ST MIC LIVE Lamp *	1	Open Collector to Logic Common Return	Illuminates whenever a MIC is LIVE in the Studio
DIM ST Lamp	2	Open Collector to Logic Common Return	Illuminates whenever Studio monitors are DIMMED
Not used.	3	Open Collector to Logic Common Return	
Not used.	4	Open Collector to Logic Common Return	
TALK (to STUDIO) Active Lamp	5	Open Collector to Logic Common Return	Active whenever a source has activated its TALK (to STUDIO) function
<b>POWER &amp; COMMON</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		

\*Feature added with software v2.1



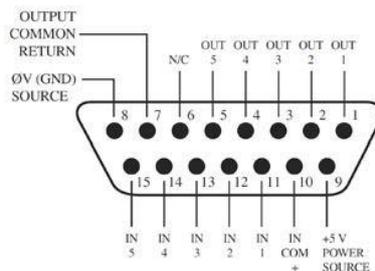
## GPIO Computer Playback Device Logic

Name	Pin	Type	Notes
<b>INPUTS</b>			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF & sends 100 msec STOP pulse
PFL Command	13	Active Low Input	Turns PFL/AFL ON
Not Used	14	Active Low Input	
READY Command	15	Active Low Input	Illuminates OFF lamp to indicate source's readiness
<b>OUTPUTS</b>			
NEXT Pulse	1	Open Collector to Logic Common Return	A 100 mS PULSE sent when ON key is depressed, except when initially turned ON.
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF and READY is active
PFL Lamp	3	Open Collector to Logic Common Return	Illuminates when PFL/AFL is ON
START Pulse	4	Open Collector to Logic Common Return	A 100 ms PULSE sent when channel is first turned ON.
STOP Pulse	5	Open Collector to Logic Common Return	A 100 ms PULSE sent when channel is turned OFF.
<b>POWER &amp; COMMON</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



## GPIO Record Mode Logic

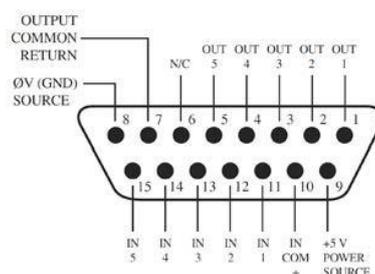
Name	Pin	Type	Notes
<b>INPUTS</b>			
Not used.	11	Active Low Input	
Not used.	12	Active Low Input	
Not used.	13	Active Low Input	
Not used.	14	Active Low Input	
Not used.	15	Active Low Input	
<b>OUTPUTS</b>			
RECORD mode ON	1	Open Collector to Logic Common Return	Illuminates whenever Record Mode is active
RECORD mode OFF	2	Open Collector to Logic Common Return	Illuminates whenever Record Mode is inactive
Not used.	3	Open Collector to Logic Common Return	
Recorder START Pulse	4	Open Collector to Logic Common Return	A 100 ms PULSE sent when Record Mode is turned ON.
Recorder STOP Pulse	5	Open Collector to Logic Common Return	A 100 ms PULSE sent when Record Mode is turned OFF.
<b>POWER &amp; COMMON</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



## GPIO Loudness Metering Logic

Name	Pin	Type	Notes
<b>INPUTS</b>			
START	11	Active Low Input	
STOP	12	Active Low Input	
RESET	13	Active Low Input	
Not used.	14	Active Low Input	
Not used.	15	Active Low Input	
<b>OUTPUTS</b>			
TOO LOUD*	1	Open Collector to Logic Common Return	
OK*	2	Open Collector to Logic Common Return	
TOO SOFT*	3	Open Collector to Logic Common Return	
	4	Open Collector to Logic Common Return	
	5	Open Collector to Logic Common Return	
<b>POWER &amp; COMMON</b>			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		

\*Feature added with software v3.0



## Comparative Table of GPIO Commands

Source Type	GPI / GPO				
	11 / 1	12 / 2	13 / 3	14 / 4	15 / 5
<b>Line</b>	ON	OFF	PFL/AFL	Reset / Start*	Ready / Stop*
<b>Computer</b>	ON / NEXT*	OFF	PFL/AFL	- / Start*	Ready / Stop*
<b>Phone</b>	ON	OFF	PFL/AFL	Reset / Start*	Ready / Stop*
<b>Codec</b>	ON	OFF	Talk to CR	MUTE	Talk to Source^
<b>Operator</b>	ON	OFF	Talk to Studio	MUTE	Talk to Preview
<b>Producer</b>	ON	OFF	Talk to Studio	MUTE	Talk to Preview
<b>CR Guest</b>	ON	OFF	Talk to CR	MUTE	Talk to Source^
<b>Studio Guest</b>	ON	OFF	Talk to CR	MUTE	Talk to Source^
<b>External Mic</b>	ON	OFF	Talk to CR	MUTE	Talk to Source^
<b>CR Monitor</b>	Mute CR / CR MIC LIVE	DIM CR	EXT PFL / PFL	Talk to External	ON AIR+ / Talk to CR
<b>ST Monitor</b>	Mute ST / ST MIC LIVE	DIM ST			- / Talk to ST
<b>Record</b>	- / ON	- / OFF	-	- / Start*	- / Stop*

\* Sends 100ms pulse.

+ From *software* v2.1 Quasar is set to ON AIR state only from this pin, or from LWCP command.

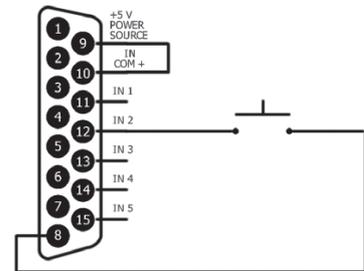
^ Requires active Backfeed.

## Wiring GPIO Connections

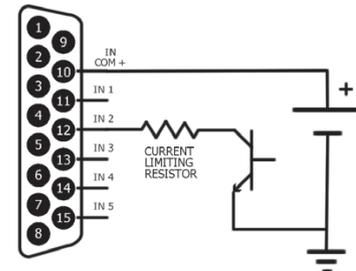
The maximum voltage allowed for an external power supply for logic control is 48 Volts DC. The use of a current limiting resistor is required for some voltages:

I NPUT	
VDC	External Series Resistor
5	0
6	0
12	680 @ 1/4 watt
24	1.8k @ 1/2 watt
48	3.9k @ 1 watt

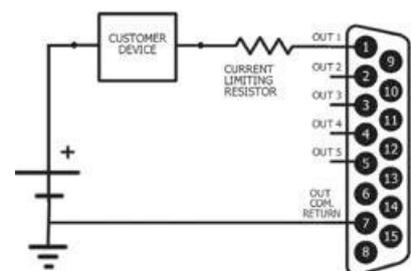
If the equipment being controlled is electrically isolated, than the use of the GPIO port's own power supply is acceptable as shown here. In this case the 5vDC internal supply can be used, maintaining a limit of 100mA on the current drawn.

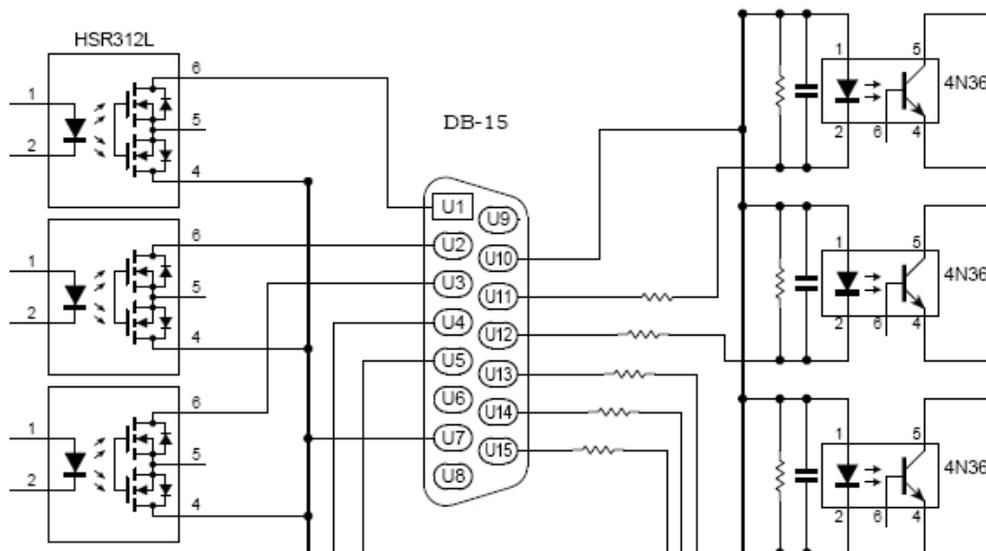


Take note to use current limiting resistors per the above chart if the voltage supplied is above 6vdc. The intention is to limit the current to 20mA for each GPI pin.



The GPO portion of the GPIO ports are solid state relays. Current should be limited to a combined 100mA through all the pins of a port. Maximum allowed voltage is 24 volts. The following diagram shows the recommended connections for outputs with the use of an external power supply.





The Axia accessory modules use the 5vDC supply to illuminate LED based keys. So, a one-to-one pin connection is all that is needed between any accessory modules and a GPIO port.

**Note** - All of the inputs and outputs on a specific GPIO port are “grouped together”. The 5 “Outputs” are on 5 separate output pins; however, they share the same “Common Return” connection on Pin #7. Similarly, the 5 “Inputs” pins would be pulled to ground to activate them, and they share a common pin for a high-side rail, on Pin #10. If more than one remotely-controlled device is to be connected to a single 15-pin I/O port, you must make sure that the two units in question have the same ground potential or ground loops will occur. Therefore, it is recommended that only one remote device be connected to each I/O port connector to assure complete electrical isolation.

# Configuration 8 – Console Modes

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## Dual and Redundant Operation

Two new functions were introduced with quasar SW v3.0: **Dual Console Operation Mode** and **Redundant/Alternative Console Operation Mode**:

- **Dual Console Operation** is configured in the **Network Setup** menu
- **Redundant/Alternative Console Operation** is configured in the **Engine Connection** menu

You must not confuse "Dual Console Operation Mode" with "Redundant/Alternative Console Operation": they are two different features designed for two different purposes. Let's see what they are about:

### Dual Console Operation Mode

Dual Operation Mode is typically used when two operators are sitting at the same console (with either single or split surface frame) and they may want to control different parameters of the console at the same time.

Dual Operation Mode can also be used to control a Quasar Engine RPS from a remote location where physical console is installed. In this case the remote console is connected to a different Network or Subnet, from that where the Engine RPS is connected. By establishing a direct communication link between the two, the remote console will behave as if it was connected to the network where the Engine RPS connects to, and will detect all sources and devices on the RPS' network.

In both cases, the remote console controls the Engine RPS like a Quasar Soft instance would, if launched from the RPS Web UI.

### Dual Operator Workflow

Let's take for example the case where the main operator must work on the microphones and music faders, monitoring the PGM levels in the Home Tab, while the second operator needs to work on the phone channels, control their EQs, and make calls using the built-in Call Controller screen.

With this mode, 2x MTS modules (either separate, or mounted into the same console chassis, or in a split XR frame) are used *at the same time* by two different engineers to control the (same) console Engine.

In this case, the two MTS Modules are part of the same logical console: one unit controls the other, and they work in a master/slave mode controlling one engine. The engine is not aware that the secondary MTS unit is present, since this talks only to the primary MTS.

For this reason, the secondary MTS control unit will not allow access to any of the Profiles menus. Selecting Presets, Sources or Shows from the Web UI of the slave unit will generate an error message on your browser. This is perfectly normal, and the Profiles menus are available only on the primary control unit.

**Note** - At the time of writing this chapter, there are no dedicated Headphone Monitor and PFL busses for the second operator, but please stay tuned with us: they are on our roadmap!

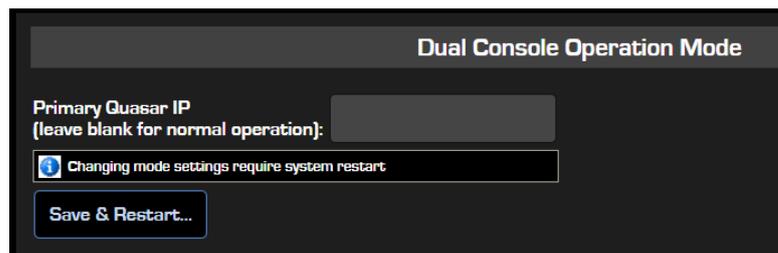
## Setting Up Dual Console Operation

You will need to access both units' Web UI:

1. Open up the Web UI of the primary console which is going to be used by the first Operator, and navigate to the **Network Setup** menu.

In the Dual Console Operation Mode section, make sure that the "Primary Quasar IP" address field, is left blank.

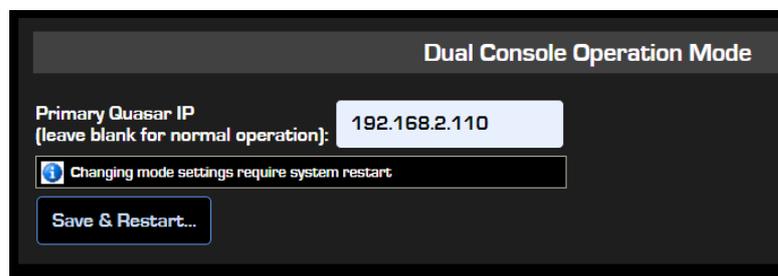
*Primary Console (192.168.2.110)  
settings >*



2. Open up the Web UI of the console which is going to be used by the second Operator, and navigate to the **Network Setup** menu.

In the Dual Console Operation Mode section, enter the IP address of the Primary console in the "Primary Quasar IP" address field.

*Secondary Console (192.168.2.120)  
settings >*



## Redundant/Alternative Console Operation Mode

Redundant Console Operation can be set up in cases where:

1. A spare console, or just a spare MTS module, is used as emergency backup for the main console
2. The RPS control unit of a Quasar Engine RPS is used as emergency backup for the main console

Alternative Console Operation can be set up in cases where:

1. Two studios (two consoles, or a console and an Engine RPS) need to share one engine: control of this engine needs to be taken by the Primary console for some time, and then by the secondary console. This could be the case when for example, a local and a remote console need to alternate with each other on the air.
2. The main console is expected to be taken off the air for some planned maintenance, and a spare console needs to go seamlessly on the air connecting to the main studio engine, so that no audio switching is required.

With this mode, 2x MTS modules, or 1x MTS and 1x RPS are used in alternative to each other, for controlling the same Quasar Engine.

Only one control unit (MTS, or RPS) will actively control the Engine, and the other unit will work in hot-standby mode. A priority mechanism built into the Engine will determine which one, of the two consoles, will be allowed to connect to the Engine.

### Redundant/Alternative Console Operation

Let's consider an example where an Engine RPS is used as emergency backup for a Quasar console. This also applies to existing Quasar systems, where the standard Quasar Engine has been upgraded to a Quasar Engine RPS.

With this mode, we assign to each console a priority number. You will need to access both units' Web UI, and navigate to the **Engine Connection** menu and:

- Set the MTS module in the Quasar console with priority 0
- Set the RPS with have priority 1 or a bigger number

When the MTS with priority 0 (the physical console) is disconnected from the engine (via Web UI) or from the network (by unplugging the Ethernet cable) , the Engine will automatically connect to the RPS.

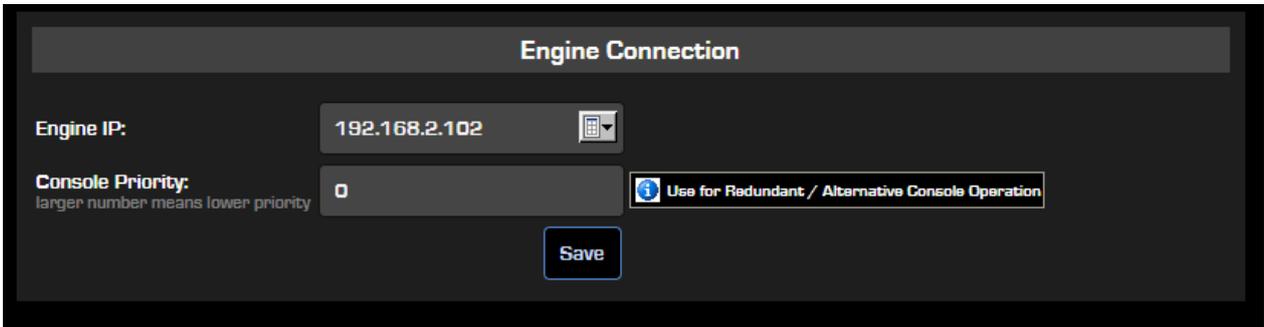
When the MTS with priority 1 is re-connected to the network, the Engine will automatically disconnect from the RPS and connect to the MTS because it has a higher priority.

In this case, the customer should regularly backup the console MTS and upload (restore) this backup onto the RPS, so that all shows are synchronized.

**Note** - In this case, both the MTS and the RPS will need to have the Dual Console Operation Mode field, left blank.

## Setting Up Redundant/Alternative Console Operation

In the Web UI, navigate to the Engine menu:



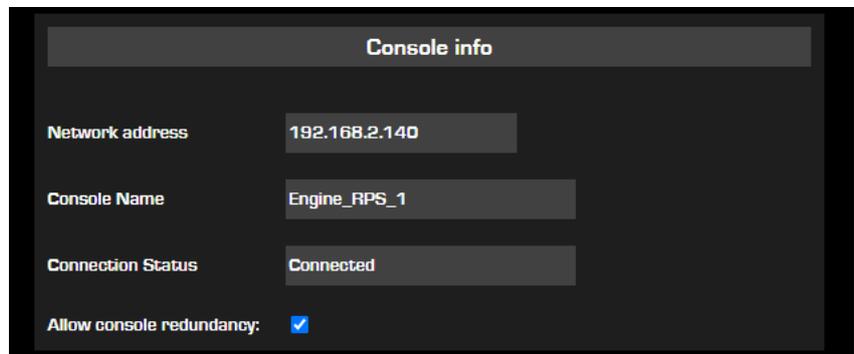
In the **Engine IP** field, enter the IP address of the Engine that you want the console to connect to.

In the **Console Priority** field, enter the priority you want to assign to this console. Numbers range from 0 to 255:

- Number 0 is the highest priority, and it is assigned by default
- Number 255 is the lowest priority

Open up the [Engine Web UI](#), and navigate to the **Network Setup** menu.

In the **Console Info** section, check the “Allow console redundancy” box to enable this option.



### Changeover Mechanism

The Engine will activate a priority control mechanism to decide which console can connect to it:

1. The console with higher priority will be allowed to connect to, and take control of the Engine.
2. In case this console disconnects from the Engine (or from the Network), the Engine will become available for connection, and will let the next console in order of priority take control.
3. In case a console with higher priority should appear on the network, the engine will drop the connection from the current console and will become available for connection to the new console with higher priority.

## Changeover Time

There is a difference in the reaction times when the changeover is triggered by the user or when it is triggered automatically. Let's analyze these two cases:

1. Manual Changeover can be performed by the user by changing the connection priorities on the Web UIs. For example, this is the case when the user wants to switch between two surfaces in alternative, or when he needs to perform some planned maintenance.

**Tip** - It is good practice to set the main console (that with the highest priority) always to Priority 1 so that if you desire to force the backup console to take control of the engine, priority on this one can be changed from 2 to 0, and leaving the main console's Priority unchanged.

2. Automatic Changeover is triggered when the console with higher priority disconnects from the Engine. This is the case when the main console or its MTS Module is turned off, or when the Ethernet connection between the main surface and its Network Switch is interrupted because a cable is damaged, or the switch itself is faulty, or turned off.

In the first case, the Manual Changeover from the Primary Console to the Secondary Console is almost instant, and the changeover in the opposite direction is instant.

In the second case, the Automatic Changeover from the Primary Console to the Secondary Console will take approx. 30 seconds and the changeover in the opposite direction will take approx. 4-6 seconds. That's because we added some timeouts in order to avoid unwanted console switching:

- a) The timeout that a console uses to detect an engine disappear from the network is set to 10 seconds.
- b) The engine uses two timeouts to detect a console disappear from the network: a short one and a long one. The first is set to 9 seconds, the second one to 30 seconds. After a short timeout (9s) the engine stops sending meter data to the console, but keeps the engine assigned to the console. After another 21 seconds (30s in total) the engine disconnects from the console and becomes available to other consoles with lower priority. The second timeout is necessary to make sure that no changeover is triggered automatically in case the main console is rebooted (since reboot takes 15 seconds).

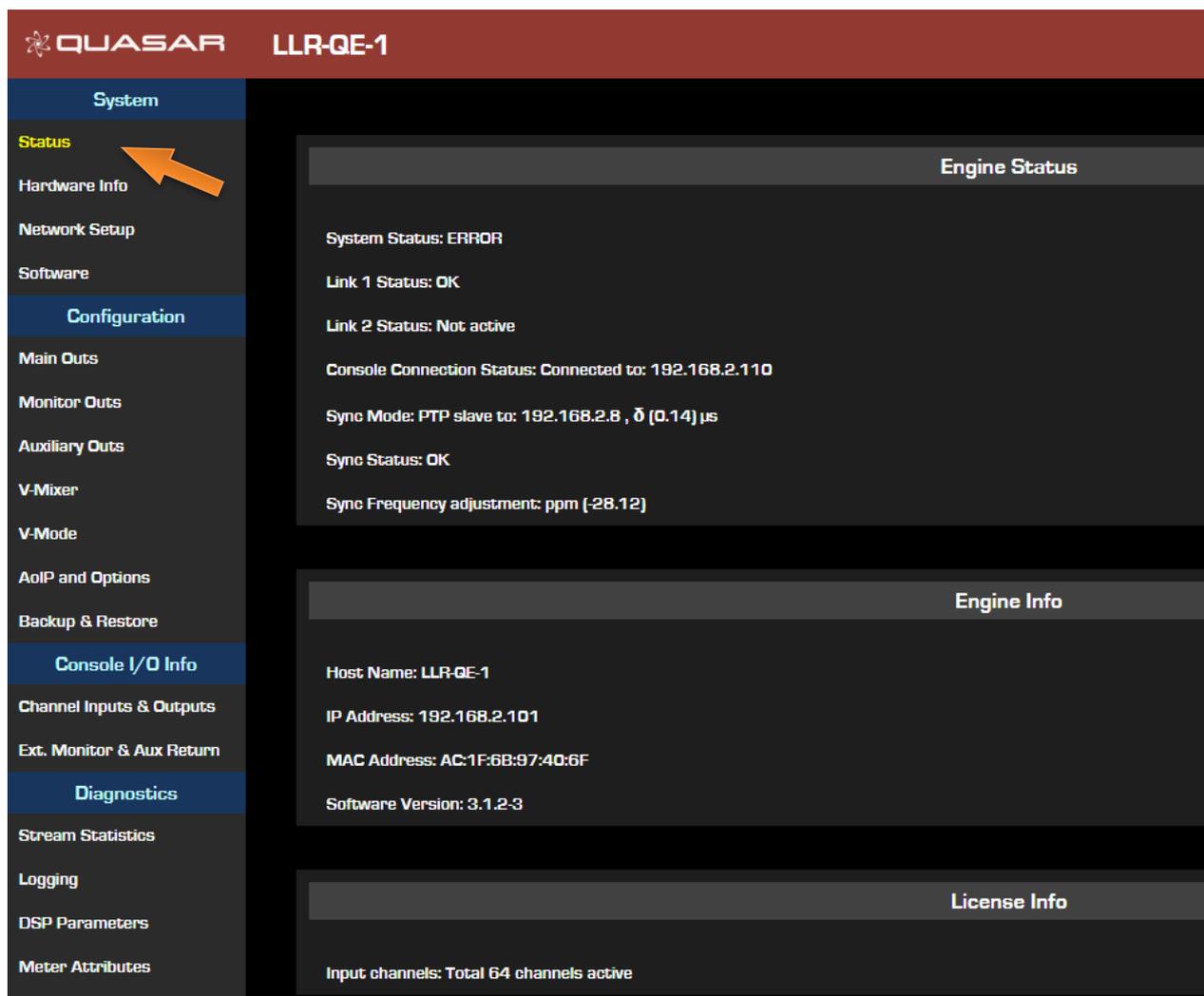
**Note** - The Engine never stops audio processing, unless instructed to do so by a connected console. There is a mechanism in Quasar to avoid audio loss in case of console connection problems.

# Configuration 9 – The Mix Engine

## The Quasar Engine Web UI

The Quasar Engine is a powerful DSP platform that can serve not only as Mixing Engine for the Quasar Console, but also for stand-alone mixing and processing applications.

In order to configure all the Engine's inputs, outputs, and options, you will need to access its web UI. Connect a PC to the console network, and open up a browser. Enter the Engine's IP address. You will be presented with the Engine's **Home** page:



The first time the Web UI is accessed from a browser, a login prompt will appear, where a user password, if configured, needs to be entered. If no password has been configured, you can enter the default credentials: *user* and *-no password-*.

The “Host Name” field at the top of the page shows the name you’ve defined for this Engine. The left column menu is divided into four menu sections: **System**, **Configuration**, **Console I/O Info** and **Diagnostics**.

## System Menu

This menu offers a general overview of system, as well as some basic information about the hardware platform and the software.

### Status

This landing page displays all the most important information about:

Engine Status: System Status, Console Connection, Ethernet Link activity, Audio Sync Mode and Status

Engine Info: Network address, MAC Address, Running Software Version

Engine Licensing: total number of Active Licensed channels

### Engine Status

- **System Status:** The System Status must be “OK” when both the PSU are connected to the AC mains. If an ERROR message appears here, you need to check the **Hardware Info** page to see what’s generating this error.
- **Network Link 1/2 Status:** The Link 1 Status must be “OK”. This corresponds to the rightmost port, by looking at the back of the Engine. At the time of writing this manual, the other network port is disabled, so the Link 2 will display a “Not Active” message. This is normal.
- **Console Connection Status:** This line shows the IP address of the connected Quasar Surface. This is equivalent to the IP Address of the MTS-MON (Master Monitor) module.
- **Sync Mode / Status:** This line displays the IP address of the device in the AoIP network, which is generating the Master clock reference. The **Sync Mode** must be “Slave to:...” and the **Sync Status** must be “OK” for the Engine to operate properly.

**Note -** xNodes, xNodes2 and legacy 8x8 Nodes (silver, 1RU model) are supported. GPIO nodes do not supply audio clock.

**Important -** The Quasar Engine has NO internal Audio Clock Generator, and requires an external Livewire or PTP Clock source to operate: Typically, the LW Clock source will be an xNode in your network which is set as MASTER, while the PTP Clock source will be a GPS-enabled PTP Grandmaster.

### Engine Info

- **Host Name:** is the user-defined Name for the Engine. This can be set in the Network Setup page.
- **IP Address:** is the user-defined Address for the Engine. Default is 192.168.2.101. This can be changed in the Network Setup page.
- **Software version:** Version of software saved in the active bank. This can be changed in the Software page.

## License Info

The Engine license is pre-installed at the factory onto new engine platforms. In case an old platform is updated to v2.0 it will be automatically installed during field update, if the Engine detects valid hardware. You do not need to obtain a license from Telos, if you are updating your Engine with a v3.0 web update downloaded from the Telos website.

New Engines come with a “Base License” of 16 Input Channels. Additional 16-Channel expansion licenses can be purchased separately.

## Hardware Info

The following Info Blocks in this page are displayed for viewing only:

The “ERROR” status refers to one of the two rear PSUs being disconnected. In this example it’s PSU #2

The N/A status refers to the optional FAN being not installed

System Temperature must be < 40°C.  
CPU Temperature must be <50°C  
DSP Core usage is avg/peak percent

This info refers to NET2 port. NET1 port is not active.

System Time Info is useful to determine if/when the Engine was restarted.

System Status	
<b>System status:</b>	ERROR
<b>PSU1:</b>	OK
<b>PSU2:</b>	OFF
<b>12VDC Power:</b>	12.38
<b>5VDC Power:</b>	5.03
<b>3.3VDC Power:</b>	3.42
<b>RTC Battery:</b>	3.00
<b>FAN 1:</b>	3500
<b>FAN 2:</b>	3400
<b>FAN 3:</b>	3600
<b>FAN 4:</b>	N/A
<b>FAN A:</b>	3600
<b>System Temperature:</b>	25°C
<b>CPU Temperature:</b>	38°C
<b>CPU usage (sys avg/DSP peak(%))</b>	
<b>Core 0:</b>	16 / 22
<b>Core 1:</b>	1 / 99
<b>Core 2:</b>	8 / 12
<b>Core 3:</b>	1 / 1
<b>Link status:</b>	OK
<b>Link speed:</b>	1 Gbs
<b>Network usage In:</b>	4.0%
<b>Network usage Out:</b>	6.5%
<b>System Time:</b>	Mon May 3 13:16:20 2021
<b>Uptime:</b>	0 days, 0 hr, 17 min, 13 sec
<b>MAC Address:</b>	ac:1f:6b:97:40:6f
<b>Kernel info:</b>	Linux version 2.6.38.8-rtai+

## Network Setup

From this page you can set up or change your Quasar network configuration, as well as check which Console is connected.

### Host Name

Here you can enter the Engine Name (Hostname),  
A description of the correct syntax to use is provided below the corresponding text field.

**Note** - A new Hostname is applied immediately, when the **Apply** button is pressed.

### IP Settings

Here you can enter the IP Address and Subnet information. A Gateway is not required. Attempts to set network address and netmask to 0.0.0.0 will not be accepted.

**Note** - Changes to these settings take effect only after restart.

### Console Info

The first three fields in this section are for viewing purposes only, and will show information about the connected console. In order to change the connection parameters of the console connected to this Engine, you need to access the console's MTS Web UI, and select the **Engine** menu.

Flag the **Allow console redundancy** checkbox, if you want the Engine to automatically connect to a secondary console, configured as redundant.

Please see Configuration Chapter 8 of this manual - [Setting Up Redundant/Alternative Console Operation](#) for more details about this option.

## Software

From this page it is possible to update the Quasar System software. In order to install a new software package, you must perform the following steps:

1. In case both software banks are used, select the current bank and click **Commit**
2. Select a new "web update file" by clicking on the **Choose File** button,
3. Wait until the software package appears in the field next to the button.

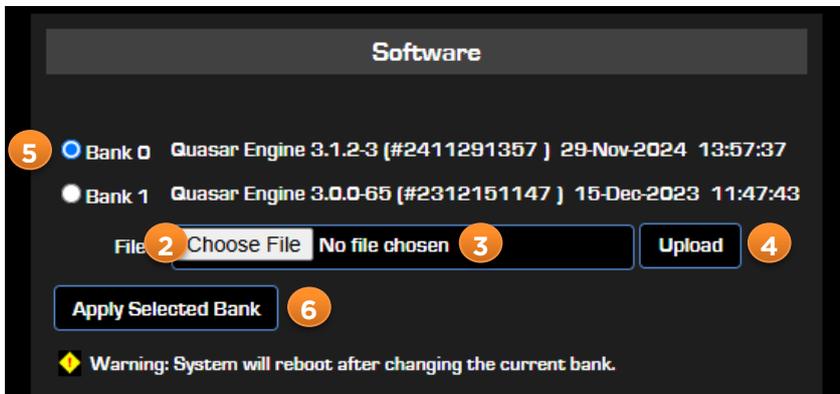
The screenshot displays a dark-themed configuration interface with three main sections:

- Host Name:** A text input field contains "LLR-QE-1". Below it, a note states: "Domain name syntax - series of labels concatenated with dots. Labels may contain only letters, digits, and hyphens, and must start with a letter or digit."
- IP Settings:** Three text input fields are shown: "Network address" with "192.168.2.101", "Netmask" with "255.255.252.0", and "Gateway" with "0.0.0.0".
- Console info:** Four fields are present: "Network address" (192.168.2.110), "Console Name" (STUDIO1-LLR), "Connection Status" (Connected), and a checked "Allow console redundancy" checkbox.

A close-up of the "Allow console redundancy" checkbox, which is checked with a blue checkmark.

A close-up of the "Commit" button, which is highlighted with a red circle and the number 1.

- Click on the **Upload** button. A message will appear to inform you that the file is being uploaded. Once complete, you will receive a confirmation message. Refresh the page
- Select the new software bank by clicking on the radio button.



- Click the **Apply Selected Bank** button. A confirmation dialog will appear, and after confirmation, the Engine will be restarted.

## Restart

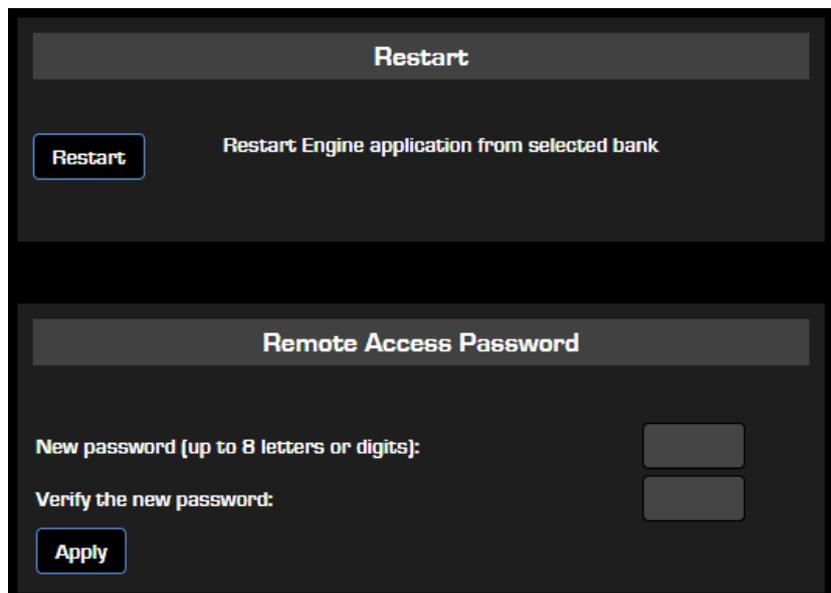
Use the **Restart** button to reboot the system directly from the web UI.

**Warning -** When the Engine reboots after selecting a new software bank, or after the user Restarts the unit, connections to the surface will be reset, and audio will be interrupted for more than one minute.

## Remote Access Password

The Default user Password can be changed in the lower part of this page.

The Username is always “user” and cannot be changed.



# Configuration Menu

Engine SW v3.0 introduces full AES67 compatibility, with some new features and a completely new graphical layout for the Web UI.

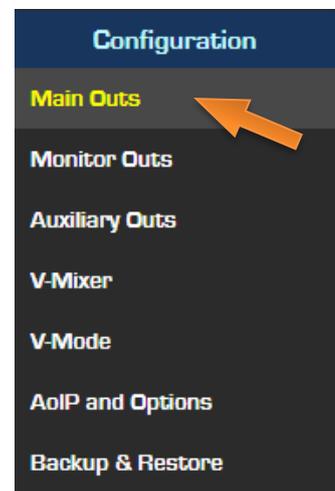
The all-new Configuration Menu has been moved up, right below the System Menu. Here you can configure the Engine's Program, Monitor and Auxiliary output busses, as well as the Monitor Inputs, the V-Mixers and the V-MODEs.

In this paragraph only the Outputs Configuration basics is described. V-Mixer and V-Mode are described in the next two paragraphs of this Chapter.

## Main Outs

### Bus Outputs

- **Program 1 - 4:** Your console's main Program bus outputs. Each bus has a dedicated activation button on the console's channel strips.
- **Record:** A special bus for recording devices which feeds the contains the Program 4 bus post-fader and pre-On/Off switch. It is activated by pressing the PGM4 button on the channel strip.
- **Phone:** The main Phone bus output, without the individual mix-minus which is automatically created for each channel. Useful to create special Backfeeds to external devices. It is activated by pressing the PGM4 button on the channel strip.



For each Bus Output, a selector is now available for defining the Output Type: **Livewire** or **AES67**. The UI will show a different set of parameters for each selection.

**Note -** Disabling an output or switching between Livewire and AES67 type will not delete the settings entered in both modes.



### LW Settings

#### LW Ch # (1...32766):

This is where you assign a unique Livewire channel number for the selected output.

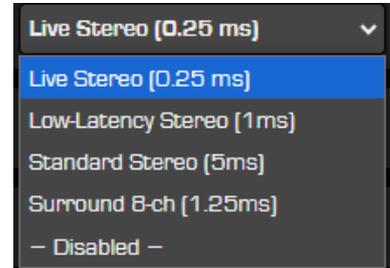


The Quasar Engine is capable of generating up to 264 different AoIP streams (if all VMIX and VMODE outputs were to be used), so we suggest following a precise scheme when assigning a Channel Number to each output.

Different methods for making up channel numbers can be used: a combination of the Engine's IP address last octet, with an ID number, is one of the possible solutions.

Enter the desired **LW Channel** by typing the source channel ID or its multicast address. In case you want to assign continuous numbering to the outputs, you can type the desired initial channel # in the first output field, then press the **LW Auto Ch #** button to automatically generate the following numbers. Remember to press **Apply** after you're done, otherwise you will lose all your settings after selecting a different page.

**Mode:** This drop-down lets you select the type of AoIP stream generated by the Engine: different Stream types use different packet structures optimized for either high efficiency or low delay, therefore this choice has an impact on the AoIP Network traffic, as well as on the audio latency.



The possible settings are:

- **Live Stereo:** these are the fastest available streams, specialized for low delay, so we pack only a few audio samples into each packet. Because they are smaller, less buffering is needed, and that means the latency is lower. These are usually chosen for anything that is sent to the air or to the live DJ microphone-to-headphone paths, so we suggest selecting this stream type for all Program and Monitoring outputs. Packet time is 250us.
- **Standard Streams:** these use large packets to be efficient with both computer resources and network bandwidth. They are usually chosen when PCs are the audio devices, so we do not suggest selecting this stream type for your Program, or Monitoring outputs, but rather to use it for IP Drivers setup, or specialized VMIX use. Packet time is 5ms.
- **Surround Streams:** Livewire inherently carries multiple audio streams and surround mixing is a built-in feature of the Axia Element console and Engine, so it is ready for radio and TV surround. Surround streams accommodate eight channels, carrying the 5.1 multichannel and a stereo mix version simultaneously. Surround streams carry these eight channels in the following order: front left, front right, center, low-frequency enhancement (LFE), surround left, surround right, stereo left, and stereo right.
- **Disabled:** Turns the stream off, making it invisible to the entire AoIP network.

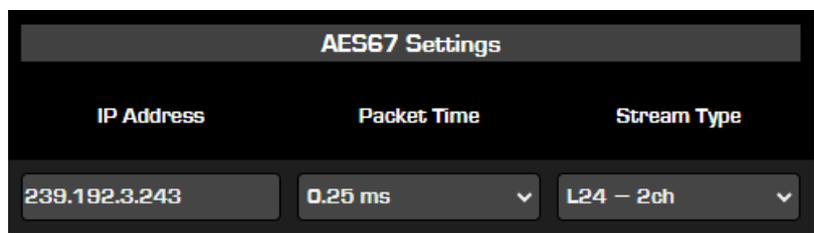
**Note -** Typically, all the output Streams are all set to “Live Stereo”, unless you have limited bandwidth budget on your AoIP switch, and require keeping the traffic as low as possible.

### AES67 Settings

**IP Address:** enter an AES67 Multicast address by typing its IP in the box.

**Packet Time:** choose the desired packet time from the drop-down menu. The standard setting for AES67 is 1ms.

**Stream Type:** choose the desired Bit Length and Channel#. The default setting is: **L24 - 2ch** (24-Bit, Stereo). You can also disable the AES67 stream and leave its settings in place for later activation, by selecting the - **Disabled** - option.

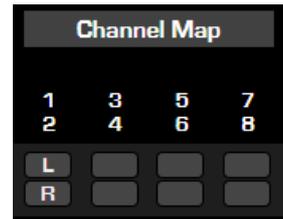


### Channel Map

This section and the following ones, apply to both the Livewire and AES67 Output types.

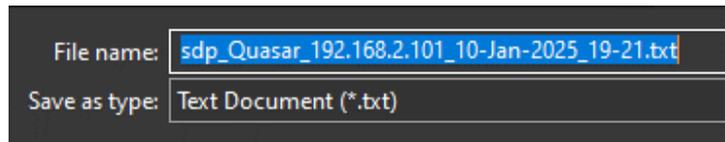
Use this section to Map Left and Right channels of the selected bus to output channels 1-8 of the LW Surround Stream, or in an AES67 multichannel stream. Type "L" or "R" in the desired Channel Field. You can populate all 8 fields, if required. For standard Stereo Stream operation type "L" in field #1 and "R" in field #2 (below).

Click the **Default Output Map** button to revert to standard stereo Stream operation on all outputs.



### SDP Export

Press this button to export the SDP configuration file for the selected bus.



### Delay

This drop-down lets you add up to 400ms delay to each Program output, for aligning the Engine's feeds to other devices within the network, or to different broadcast chains.

### Stream Status

These indicators are present on all outputs of the Engine.



The **Status** window will normally display:

- An "OK" message, when the stream is successfully loaded
- An "ERR" message, when the audio stream is not loaded successfully
- A solid green bar indicates that the stream is active, and audio is present, on particular output.
- An empty bar with green outline indicates that the stream is active but no audio is present, on that output.
- No bar indicates the stream is not active, or not enabled.

## Monitor Outs

**CR Monitor Direct:** This is the output of the Control Room Monitor selector before the operator's volume controls and mutes; useful for feeding a Producer's position, or any other monitoring station with independent headphone volume controls.

**CR Monitor:** The source selected to feed the Control Room Loudspeakers is routed to this output

**CR Headphones:** The source selected to feed the Control Room Headphones is routed to this output

**PFL/AFL:** The PFL or AFL bus is routed to this output. Useful to feed a dedicated PFL speaker, in case you prefer not to use the PFL to Loudspeaker function on the console

**Talk to CR:** This channel feeds audio to the Control Room talkback channel whenever the Talk key is pressed on any Accessory Module

**Guest Headphones:** The source selected to feed the Guest Headphones is routed to this out

**Studio Monitor:** The source selected to feed the Studio Monitors is routed to this out

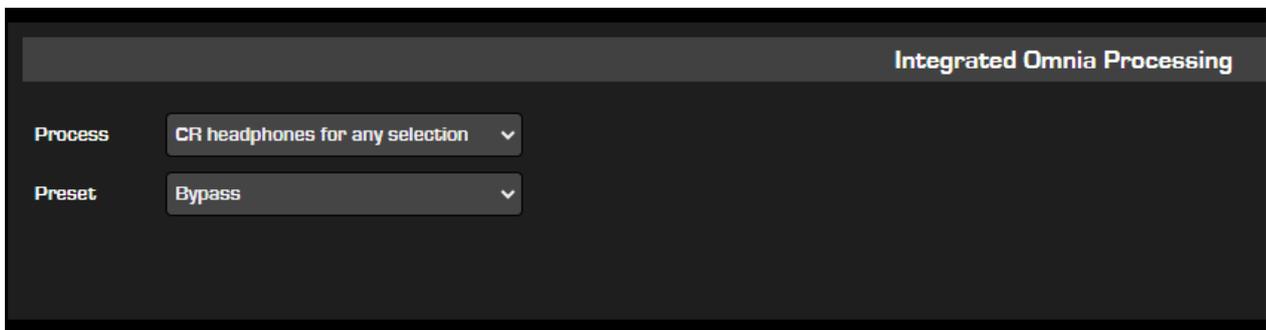
**Talent Headphones:** The source selected to feed the Talent Headphones is routed to this out

**Talk to External:** When any console Talk key is pressed, CR Mic audio is routed, pre-fader, to this channel.

The same output settings found in the Main Outs (LW, AES67, Channel Map, SDP, Delay and Status) apply to these outputs. Please refer to the previous paragraph for a complete description of these settings.

### Integrated Omnia Processing

Monitoring the **Program** bus leaves most talents unhappy with the sound coming off their headphones. To address this problem, most engineers route the Station's On Air Processor output to the Studio Headphone feeds, but this normally has an audible delay that can make monitoring very tiring, if not impossible.



Quasar offers customizable headphone EQ and built-in headphone dynamics processing by Omnia. When this feature is used, Talents can monitor **Program** audio with no delay and still hear the comfortable “air” sound they’re used to without the hassle and expense of outboard processors dedicated to the headphone channel.

The Quasar Engine contains special audio processing presets by Omnia that you can apply to the either:

- Control Room Headphone channel (fixed output, any source monitored).
- Program 1 in all monitoring paths (fixed source, any monitor output)

This will let you simulate the “air sound” provided by an on-air processor, so that your jocks can have real-time Program monitoring that sounds closer to your tuner, without having to set up a separate processor chain.

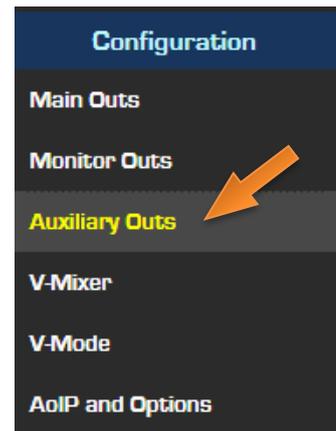
Two presets are provided: both are based on a 3-band Omnia processor, with Wideband AGC and single bands dynamic processing. Choose one from the drop-down and click **Apply** .

**Note** - You can also apply a 4-band EQ on the CR Headphones, and activate it from the console. Settings for this EQ are stored in the Show Profiles. Please see Operation Chapter 2 of this Manual - [MONITOR OPTIONS tab](#).

## Auxiliary Outs

This is where you enable and configure all of your console's Auxiliary outputs. If any of your Quasar's Aux outputs are disabled in this menu, that Aux bus will have no audio output.

The same output settings found in the Main Outs (LW, AES67, Channel Map, SDP, Delay and Status) apply to these outputs. Please refer to the previous paragraph for a complete description of these settings.



Auxiliary	Type	LW Settings		AES67 Settings		
		LW Ch # (1...32766)	Mode	IP Address	Packet Time	Stream Type
Auxiliary 1	Livewire	1017	Live Stereo (0.25ms)			
Auxiliary 2	Livewire	1018	Live Stereo (0.25ms)			
Auxiliary 3	Livewire	1019	Live Stereo (0.25ms)			
Auxiliary 4	Livewire	1020	Live Stereo (0.25ms)			
Auxiliary 5	AES 67			239.192.3.253	0.25 ms	L24 - 2ch
Auxiliary 6	AES 67			239.192.3.254	0.25 ms	L24 - 2ch
Auxiliary 7	AES 67			239.192.3.255	0.25 ms	L24 - 2ch
Auxiliary 8	AES 67			239.192.4.0	0.25 ms	L24 - 2ch

## V-Mixer and V-Mode

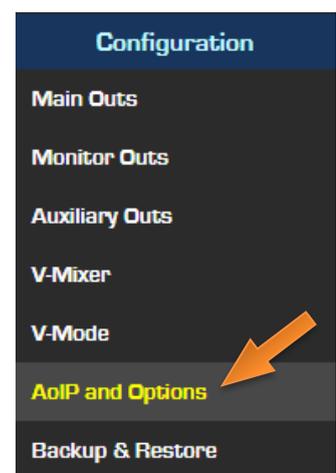
The V-Mixer and V-Mode menus deserve their own Paragraph, so we will describe these in detail in the [next paragraphs](#) of this chapter.

## AoIP and Options

This Menu is for setting the Audio Sync mode and the Global Options for the Livewire and AES67 streams.

### Synchronization

Two options are available:

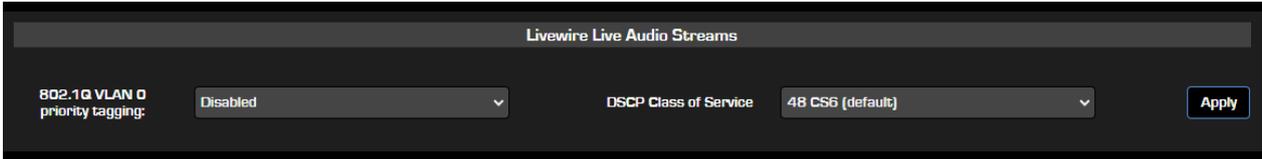


Synchronization	
Synchronization mode:	Livewire slave only
PTP domain number:	0 default
<input type="button" value="Apply"/>	

- **Synchronization Mode:** lets you configure the Engine as Livewire Slave only, or PTP slave only. The engine cannot be configured to be the clock Master in a system.
- **PTP domain number:** lets you configure the PTP domain in case the engine is set to be a PTP Slave

### Livewire Live Audio Streams

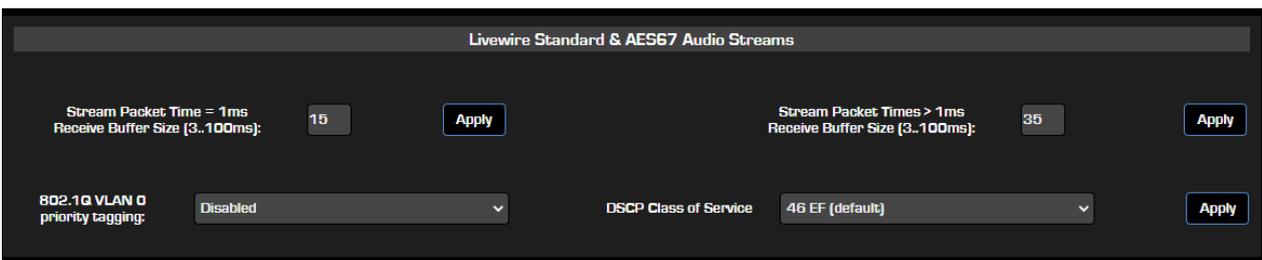
Two options are available:



- **802.1Q VLAN 0 priority tagging:** The 802.1Q standard defines a system of VLAN tagging for Ethernet frames and also contains a provision for a quality of service (QoS) prioritization scheme known as 802.1P, which indicates the priority level of the frame. The priority level values range from zero (best effort) to seven (highest). These values can be used to prioritize different classes of traffic such as AoIP. The VLAN ID tag specifies the VLAN to which the frame belongs. The priority bits define the priority with which the frames are processed.
- **DSCP Class of Service:** The Differentiated Services Code Point (DSCP) is a 6-bit field in an IP header for the classification of packets. Differentiated Services is a technique used to classify and prioritize network traffic (for example, requesting high priority or best effort delivery for traffic), and it helps to provide QoS for modern Internet networks. We suggest leaving this parameter to its default setting: 48 CS6 .

### Livewire Standard & AES67 Audio Streams

Two options are available for Streams with 1ms packet time (like the AES67 streams), and two for the streams with a larger packet time (like the Livewire Standard streams):



- **Stream Packet Time=1ms:** The Quasar Engine offers the possibility to tune the streams receive buffer size, except for the "Live Stereo" type of streams, in order to ensure optimal latency performance on every specific network, and maintain the lowest possible jitter. This setting will let you define the Receive Buffer Size for AES67 streams. The values range from 3ms to 100ms. The default value is 15ms.

Users could experiment with the receive buffer setting to find the optimum values - to some degree it is always specific per individual network.

Small size stream receive buffers reduce the Audio Stream transmission latency, but require fast, QoS-controlled networks, with enough bandwidth *consistently available*, in

order to support the desired audio transmission bitrate. Short receive buffers cannot ride out bandwidth variations. Large size receive buffers are more forgiving in case the network is not able to convey audio streams at the required rate *consistently*, but increase packet transmission latency.

Jitter is another important factor because it determines the minimum receive buffer length. The buffer has to be long enough that is it able to catch the latest arriving packet. Any packet that turns up past the buffered time is lost. Jitter is mostly caused by queuing delays. If an outgoing link is busy, packets have to wait around in a buffer until their turn arrives.

Bear in mind that if there is no QoS at all, tuning the buffers within this range is unlikely to change much. This tuning is rather useful to adapt to source properties, not to work around a lack of QoS. If you have only Axia nodes on the network, you will choose a lower setting (default is good), compared to a network of Windows machines.

**Note** - A wrong setting might generate buffer overruns or underruns. Please refer to the [Diagnostics](#) paragraph for more details about Stream statistics.

- **802.1Q VLAN 0 priority tagging:** As above, the priority level values range from zero (best effort) to seven (highest). These values can be used to prioritize different classes of traffic such as AoIP. The VLAN ID tag specifies the VLAN to which the frame belongs. The priority bits define the priority with which the frames are processed.
- **Stream Packet Time>1ms:** The Quasar Engine offers the possibility to tune the streams receive buffer size, except for the "Live Stereo" type of streams, in order to ensure optimal latency performance on every specific network, and maintain the lowest possible jitter. This setting will let you define the Receive Buffer Size for Livewire Standard streams, or AES67 streams with packet time greater than 1ms. The values range from 3ms to 100ms. The default value is 35ms.
- **DSCP Class of Service:** The Differentiated Services Code Point (DSCP) is a 6-bit field in an IP header for the classification of packets. Differentiated Services is a technique used to classify and prioritize network traffic (for example, requesting high priority or best effort delivery for traffic), and it helps to provide QoS for modern Internet networks. We suggest leaving this parameter to its default setting: 46 EF.

**Note** - This is a global setting that will be applied to all received audio streams, except Livewire "Live Streams". To ensure stable reception of all streams, the buffer size must be at least three times the longest packet time to be received by this device.

**Warning** - For active streams, any changes will take effect and be applied after the stream is restarted.

### Audio Mixing Mode

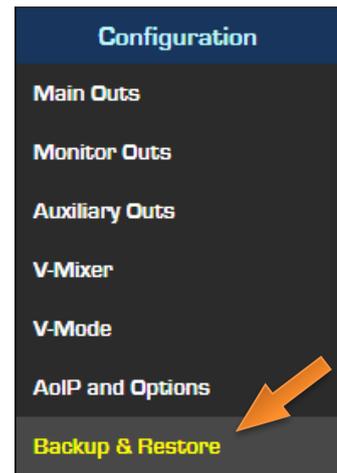
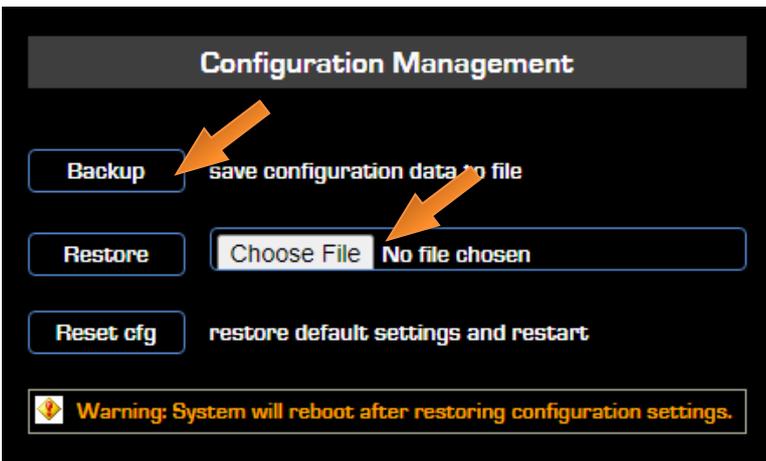
This section is for viewing only. It simply shows the Engine's mixing mode.



## Backup & Restore

From this page, it is possible to download and save to your PC a full backup of the console configuration, in the form of an .xml file.

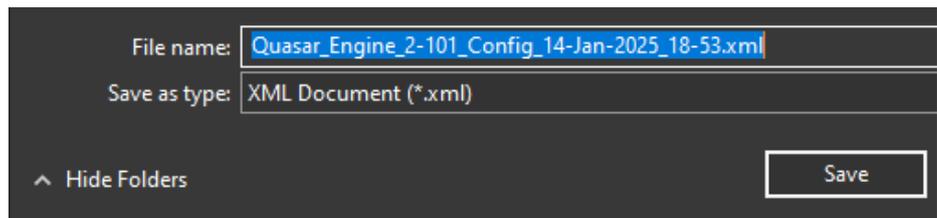
Select **Backup** if you want to create and download a Backup File. A dialog will appear, to let you save the file, in XML format.



Select **Choose File**, if you want to upload a previously created backup file from your archive. then press **Restore**.

The configuration backup includes all parameters, except:

- License codes
- IP settings, conditionally by dialog option
- UI password



**Warning** - The System will reboot after restoring the configuration settings.

## Console I/O Info Menu

The original Status menu was renamed **Console I/O Info** in SW v3.0, and was moved down below the Configuration menu. This menu was also streamlined and pages that were essentially a view-only duplicate of the Configuration pages were removed.

It provides feedback about the status of all Engine Channels' Inputs and Outputs, as well as External Monitor and Aux Return Inputs.

**Note** - This entire section, with its two pages, is for viewing only. No parameters can be configured in this section.

## Channel Inputs & Outputs

The status of all Fader Input Channels is reported here. The view is now organized with four tabs, each showing a block of 16 Input channels.



Channel Inputs Status										Channel Outputs Status			
Ch #	FADER IN		HP & BF EXT IN		IFB / INTERCOM		INS RETURN		BACKFEED		DIR OUT / INS SEND		
	Name Ch#/Address	Map L-R Status	Name Ch#/Address	Map L-R Status	Name Ch#/Address	Map L-R Status	Name Ch#/Address	Map L-R Status	Name Ch#/Address	Map L-R Status	Name Ch#/Address	Map L-R Status	
1	Mic 1 Lavalier 111	1 2 OK							To: Mic 1 Laval 239.193.0.111	1 2 OK	Dir: xN_MIC 1 D 239.192.39.8	1 2 OK	
2	Mic 2 112	1 2 OK					DAB/FM Tuner 209	1 2 OK	To: Mic 2 239.193.0.112	1 2 OK			
3	Mic 3 113	1 2 OK		209 OK					To: Mic 3 239.193.0.113	1 2 OK	Dir: xN_MIC 3 C 239.192.39.9	1 2 OK	
4	Digital 3 215	1 2 OK											

Each input channel is displayed as a horizontal row, with four inputs and two outputs:

- FADER IN
- HP & BACKFEED IN
- IFB / INTERCOM
- INSERT RETURN
- BACKFEED
- DIRECT OUT (INSERT SEND)

For each I/O an indication of the assigned Source Name, AoIP Channel (Livewire Ch# or AES67 IP Address), Stream and Mapping Status is available, along with audio presence indicators.

## External Monitor & Aux Returns

In this page, an indication of the Source loaded with the Show Profile, its AoIP Channel (Livewire Ch# or AES67 IP Address), Stream Type and Packet Time is available, along with Channel Mapping and audio presence indicators, is provided for each of the Monitor Input and Auxiliary Return Input.



It is a useful diagnostics tool when there are problems with listening to some source through the console monitor section.

External Monitor Inputs Status							
Bus	Source Loaded	LW Ch #/Address	Stream Type	P.Time	Map L-R	Status	
EXT 1 Monitor		209	From source (Stereo)	0.25ms	1 2	OK	
EXT 2 Monitor							
EXT 3 Monitor							
Source Preview							
EXT PFL		114	From source (Stereo)	0.25ms	1 2	OK	
EXT Talk To CR							
STUDIO Talkback		310	From source (Stereo)	0.25ms	1 2	OK	
Auxiliary Return Inputs Status							
AUX Return 1		314	From source (Stereo)	0.25ms	1 2	OK	
AUX Return 2		314	From source (Stereo)	0.25ms	1 2	OK	

## Diagnostics Menu

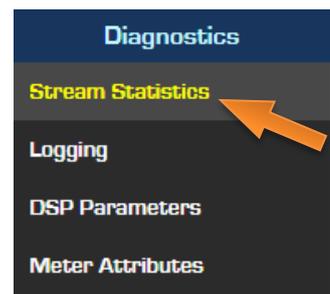
The following menu includes some information which is not relevant for everyday use, and might sound a little obscure to some users, so if it is too deep for you, you can always skip this part. But in case some issues are observed with the Engine, this turns out to be a precious source of information for Telos' Support engineers.

We will provide some very basic concepts and explanations of what the parameters in these diagnostics pages relate to, just in case someone who likes to know what's going on "under the hood" is interested.

### Stream Statistics

This page reports in detail, the Stream Receiver status of each of the Engine's Inputs.

Since up to 361 inputs are available, the UI offers the possibility to view statistics only for the **Connected Inputs** or for **All Inputs**. The Default view is Active Inputs only.



Show all inputs   Clear counters

Fader Inputs													
	Source Name:	Channel or Address:	Receive Port:	Source IP Address:	Session Count:	Packets Received (%):	Packets Lost:	Packets too soon:	Packets too late:	MAX Jitter ms:	Timestamp Correction:	Samples Lost:	Double Stream IP:
Input 1	111	111/0	5004	192.168.2.11	0	14634269 (100.00)	0	0	804	0	13	653799	0.0.0.0
Input 2	112	112/0	5004	192.168.2.11	0	14634262 (100.00)	0	0	804	0	13	641606	0.0.0.0
Input 3	113	113/0	5004	192.168.2.11	0	14634260 (100.00)	0	0	805	0	13	705113	0.0.0.0
Input 4	215	215/0	5004	192.168.2.20	0	14631039 (100.00)	0	0	804	0	13	682204	0.0.0.0
Input 5	310	310/0	5004	192.168.2.31	0	14635252 (100.00)	0	0	804	0	-42791	727685	0.0.0.0
Input 6	911	911/0	5004	192.168.2.91	2	731708 (100.00)	0	2	645	5	214	150187	0.0.0.0
Input 7	213	213/0	5004	192.168.2.20	0	14634048 (100.00)	0	0	804	0	13	683137	0.0.0.0
Input 8	214	214/0	5004	192.168.2.20	0	14634044 (100.00)	0	0	805	0	13	682086	0.0.0.0
Input 9	611	611/0	5004	0.0.0.0	0	0 (0.00)	0	0	0	0	0	175602184	0.0.0.0
Input 10	612	612/0	5004	0.0.0.0	0	0 (0.00)	0	0	0	0	0	175601812	0.0.0.0
Input 11	613	613/0	5004	0.0.0.0	0	0 (0.00)	0	0	0	0	0	175601452	0.0.0.0
Input 12	614	614/0	5004	0.0.0.0	0	0 (0.00)	0	0	0	0	0	175601044	0.0.0.0

In this view, all Engine's Inputs are divided into the following groups:

- Fader Inputs
- Individual Headphone Inputs
- Interruptible Feedback Inputs (IFBs)
- Insert Return Inputs
- Aux Return and Monitor Inputs
- VMIXER Inputs
- VMODE External Inputs

For each Group of inputs, the following fields are displayed:

- Source Name
- Channel or Address
- Receive Port
- Source IP Address
- Session Count
- Packets Received (%)
- Packets Lost
- Packets too soon
- Packets too late
- MAX Jitter (ms)
- Timestamp Correction
- Samples Lost
- Double Stream IP

### Source Name

This field shows the name of the Source which is connected to the input

### **Channel or Address**

This field shows the Livewire Channel, or AES67 IP address, of the connected source

### **Receive Port**

This field shows the Engine TCP port used to receive the connected source

### **Source IP Address**

This field shows the Multicast IP address of the connected source

### **Session Count**

The Session Counter is increased by one unit if, after it was receiving packets, these are lost or no longer sent for more than 100ms. On this event the receiver clears its internal states and waits for the stream packets.

### **Packets Received**

This field shows the total number and percentage of audio packets received by the input, the last time the web page was refreshed. The packets counted here arrived to the stream receiver on time and in the correct order.

### **Packets Lost**

This field shows the total number of audio packets lost by the input, the last time the web page was refreshed. The algorithm checks the packet sequence numbers and increases the counter by the number of lost packets. The algorithm is capable of handling swapped packets. For example, sequence numbers 1,2,5,3,4,6. Ideally, the value in this field should be 0.

### **Packets too soon**

The time-aligned receiver accepts packets originated in a certain time window from the moment audio is synchronized. The counter is increased if the packet timestamp shows time later than time of synchronization. This is an abnormal situation caused by a synchronization error in the stream sender or in the Engine. This will cause resynchronization of the Stream receiver with some sample loss. Ideally, the value in this field should be 0.

### **Packets too late**

The time-aligned receiver accepts packets originated in a certain time window from the moment audio is synchronized. The counter is increased if the packet timestamp shows time earlier than time of synchronization, minus the buffer size. As a result, this packet will be dropped and samples will be lost. This is caused by the buffer being too short for this particular packet time, including the jitter in the sender and the delay and jitter of the ETH network. Ideally, the value in this field should be 0.

### **MAX Jitter**

Measured maximum packet delay from time of synchronization and timestamp in packet. This measurement helps the user to select the proper buffer size for the receiver. The buffer size must always be set larger than MAX jitter.

### Timestamp Correction

This value is "0" if time-aligned receiving is possible. This means that the time of synchronization in the sender and in the Engine is correct, the sender sets correct packet timestamp values and the jitter is in an acceptable range. If all of the above conditions are not verified, the receiver calculates a correction value to provide stable receiving without precise time alignment. The correction value shows the number of samples which are not time-aligned.

### Samples Lost

This counter is increased if a sample is not sent for DSP processing. This could be caused by a Stream Sender which is not active, or has wrong settings for the channel count and coding, or could be caused by packet loss, packets being too late or too soon, or by a synchronization instability.

### Double Stream IP

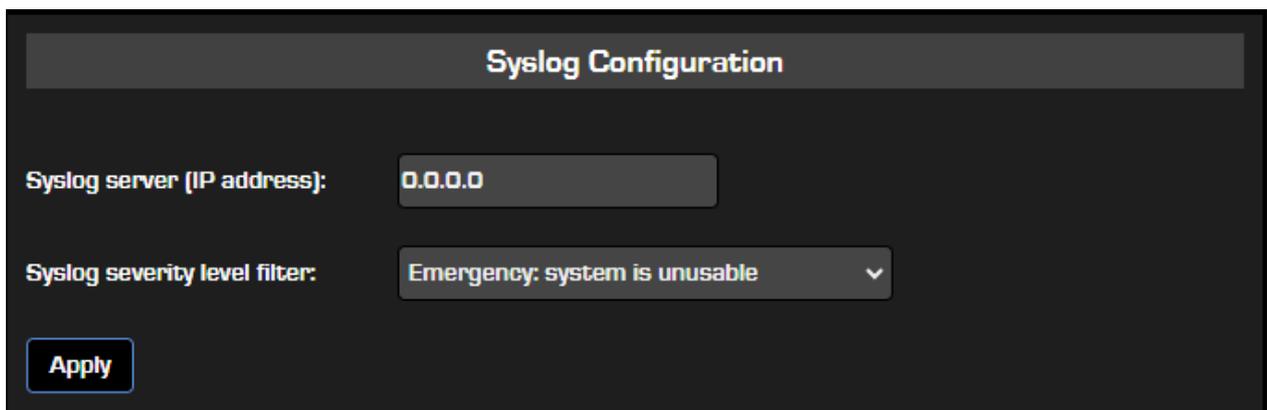
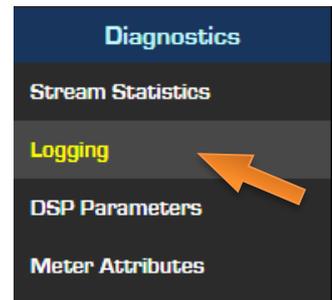
An IP address different from 0.0.0.0 appearing in this field indicates a configuration error: cases where two or more senders with the same multicast address or Livewire channel are active on the network, or cases where a unicast sender tries to send packets to the same destination address and port. This field shows the address of the second sender, which is not being received. These audio packets are dropped.

## Logging

### Syslog Configuration

**Syslog Server:** if it's not on the same subnet, you'll have to set a Gateway in the network setup menu.

**Syslog Severity Filter:** it depends on the type of server. Normally Error Level should be selected (intermediate).



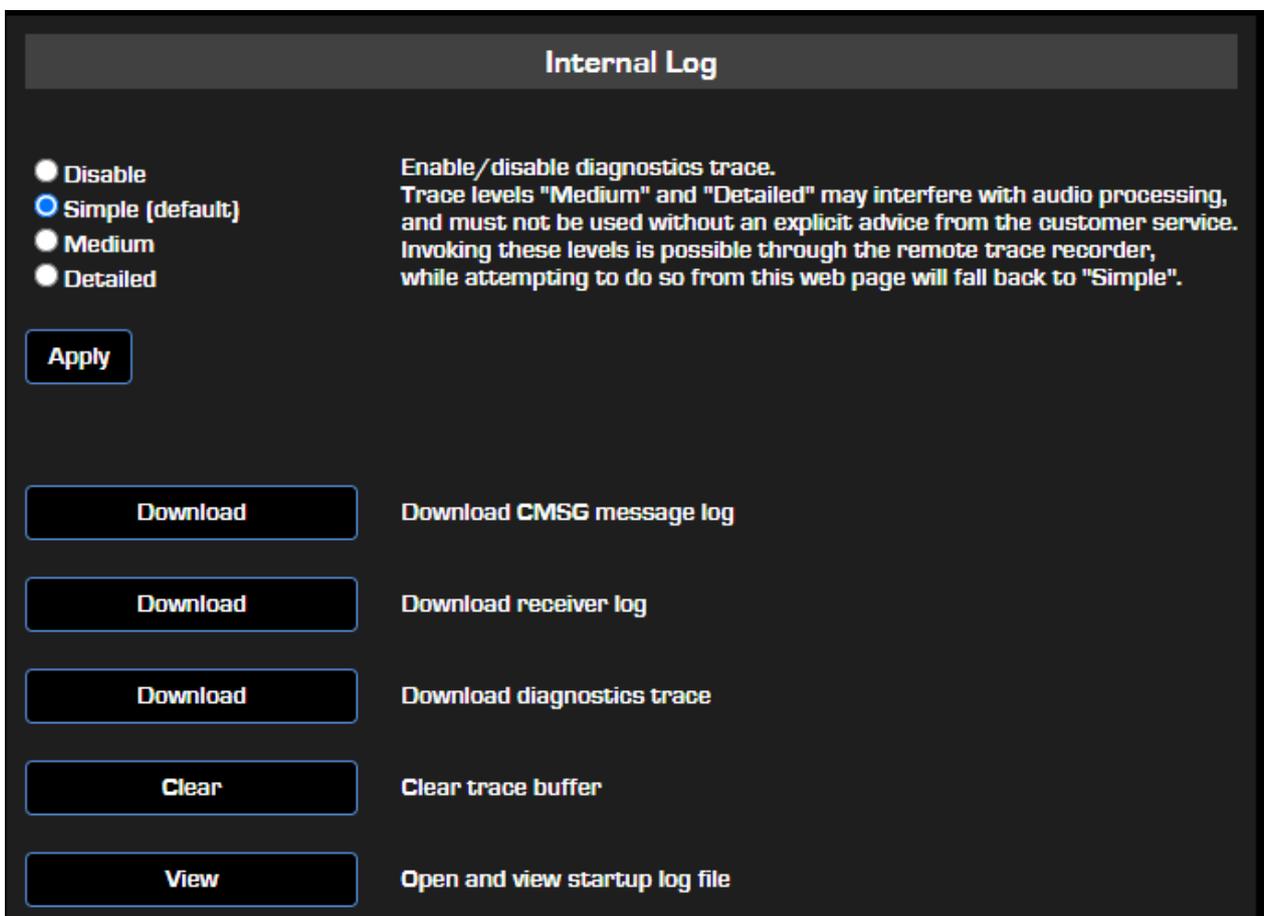
### Internal Log

You know what syslog is: when something happens that is considered "important enough" to be reported, a record is made on the server. Typically, this information is meaningful for the end user, including studio engineers, IT department, etc.

If enabled, the Engine can record an internal “Diagnostics Trace” that once downloaded, and sent to the Telos Support, can reveal details about an operation failure, of misbehavior.

Think of the Engine trace like an airplane’s black box, where you can record detailed internal states and parameter values for later analysis - as long as you know how to decode them and what they mean. The information in this trace is precisely synchronized to Engine’s internal processing by indicating the sequence number of the current 0.25 ms processing period since the startup. This information can be very dense (a short fragment can be stored inside the Engine in RAM, but that is limited to 10,000 records only, and it is volatile) and can easily grow to many Gigabytes of data. So, If you want to log at a high resolution for days and months, and need the log to survive Engine restarts, you have to use an external Syslog server.

This trace is not supposed to be of any use to the customer, except it helps developers to help the user. It is also extensively used in the development process.

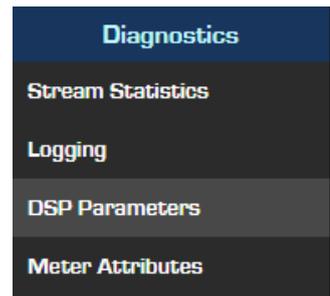


The **Download Trace** button will open a dialog to save the Trace information in the form of a Text (.txt) file. Pressing the **Clear Trace** button, the internal buffer is emptied.

The **View Startup Log** button will display a view of the entire log generated since the last time the Engine was started, up to present. The entire text can be selected, copied and pasted into a text file, or it can be quickly exported by right-clicking on the web page and printing it as a .pdf file.

## DSP Parameters

This page displays the status of every single DSP Logic attribute contained in the Engine: This is a very useful tool, capable of revealing if a command sent from the control surface, is correctly acknowledged and executed by the Engine.



Return to Menu      Meter Attributes...

	Chan 1:	Chan 2:	Chan 3:	Chan 4:	Chan 5:	Chan 6:	Chan 7:	Chan 8:	Chan 9:	Chan 10:	Chan 11:	Chan 12:	Chan 13:	Chan 14:	Chan 15:	Chan 16:
SRCT	10	10	10	4	1	10	4	4	8	8	8	8	0	0	0	0
FATG	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
FAxx	6	2	14	0	7	3	4	-20	-17	-32768	-32768	-32768	70	0	0	0
MUTE	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
ONSW	0	0	0	0	0	0	0	1	1	0	0	0	0	0	0	0
PGM1	0	1	1	1	1	1	1	1	1	0	0	0	0	0	0	0
PGM2	0	1	1	1	1	1	1	0	0	0	0	0	0	0	0	0
PGM3	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
PGM4	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
PREV	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0

**Note** – Every time a parameter is changed on the console, this page must be refreshed to display the change

Two different buttons are available for viewing the corresponding attributes:

[Mixing Attributes](#) and [Meter Attributes](#).

When you’ve done checking the DSP parameters, use the [Back to Main Page](#) button to go back to the main menu. Using the browser “Back” button will result in slower performance.

## V-Mixer and V-Mode

In addition to the 64-input mixing capacity of your Quasar Engine, there are 16 Virtual Mixers, or “VMix”, and 16 routing processors called “V-Modes” that you can control from the Engine’s Web UI, or from Pathfinder. VMix and V-Modes work independently of the Quasar surface.

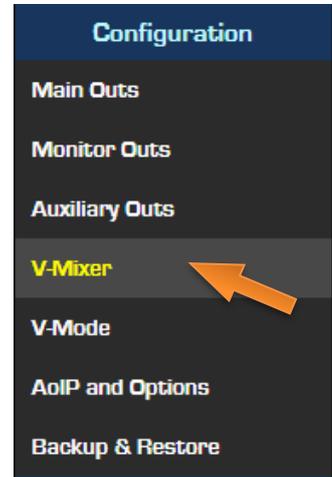
### V-Mixer

V-Mixers have also benefited from the massive update to version 3.0: full AES67 compatibility has been added to all VMIX I/Os, making it no longer necessary to use VMODE to convert LW streams generated with V-Mixers to AES67 streams.

Each of the 16 VMix is a 5-Input “virtual mixer”, which can be used as a subgroup mixer to premix up to 5 audio sources and generate a new Submix source. The outputs of the 16 V-Mixers are directly available as new Sources, and also summed into the Main VMix Output bus, with its own Main Output Source. This gives you the equivalent of an 80-input mixer.

In addition to static control of VMix through its web pages, Axia’s Pathfinder routing control tools can also be used to dynamically control VMix and create mixing functions based on a variety of system-wide parameters. Also, a VMix can be controlled in real time from a Quasar ACC Module running VMIX Control mode.

Finally, a VMix source can be applied to a console fader, assigned to an audio node destination, or monitored by Pathfinder.



## V-Mixer Main page

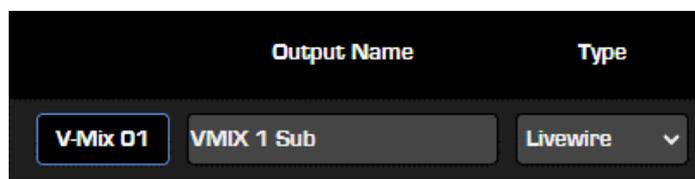
This page offers an overview of all the 16 V-Mixers Outputs available, and allows for quick configuration and an easier comparison of these outputs.

V-MIX - Home page														
LW Settings					AES67 Settings			Channel Map				SDP	Audio	Stream
Output Name	Type	LW Ch. # [1...32766]	Mode	IP Address	Packet Time	Stream Type	1	3	5	7	Gain (dB)	Status		
							2	4	6	8				
V-Mix 01	VMIX 1 Sub	Livewire	1051	Live Stereo (0.25ms)			L				Export SDP	0.0	OK	
V-Mix 02	VMIX 2 Sub	Livewire	1052	Live Stereo (0.25ms)			R				Export SDP	0.0	OK	
V-Mix 03	VMIX 3 Sub	Livewire	1053	Live Stereo (0.25ms)			L				Export SDP	0.0	OK	
V-Mix 04	VMIX 4 Sub	Livewire	1054	Live Stereo (0.25ms)			R				Export SDP	0.0	OK	
V-Mix 05	VMIX 5 Sub	Livewire	1055	Live Stereo (0.25ms)			L				Export SDP	0.0	OK	
V-Mix 06	VMIX 6 Sub	Livewire	1056	Live Stereo (0.25ms)			R				Export SDP	0.0	OK	
V-Mix 07	VMIX 7 Sub	Livewire	1057	Live Stereo (0.25ms)			L				Export SDP	0.0	OK	
V-Mix 08	VMIX 8 Sub	Livewire	1058	Live Stereo (0.25ms)			R				Export SDP	0.0	OK	

For each V-Mixer Output, the following basic parameters are displayed:

### Output Name

Here you can type in the name you want to assign to each V-Mixer



### Type

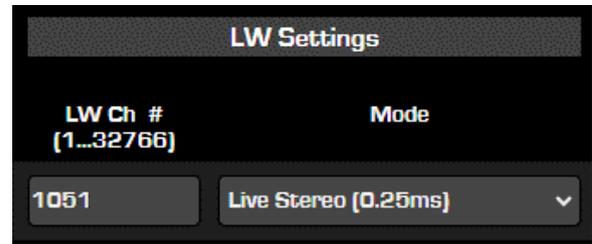
For each VMIX Sub Output, a selector is now available for defining the Output Type: **Livewire** or **AES67**. The UI will show a different set of parameters for each selection.

### LW Settings

**LW Ch # (1...32766):** Here you can type in the Livewire Channel number that you want to assign to the output of each V-Mixer. This will be the V-Mix Source ID. In case you want to assign continuous numbering to the outputs, you can type the desired initial channel # in the

first output field, then press the **LW Auto Ch #** button to automatically generate the following numbers. Remember to press **Apply** after you're done, otherwise you will lose all your settings after selecting a different page.

**Mode:** This drop-down lets you select the type of AoIP stream generated by the Engine: different Stream types use different packet structures optimized for either high efficiency or low delay, therefore this choice has an impact on the AoIP Network traffic, as well as on the audio latency.



The possible settings are:

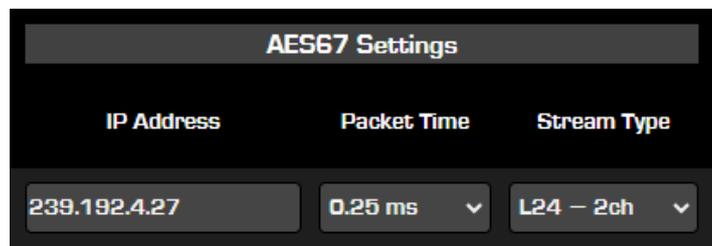
- **Live Stereo:** these are the fastest available streams, specialized for low delay, so we pack only a few audio samples into each packet. Because they are smaller, less buffering is needed, and that means the latency is lower. These are usually chosen for anything that is sent to the air or to the live DJ microphone-to-headphone paths, so we suggest selecting this stream type for all Program and Monitoring outputs. Packet time is 250us.
- **Standard Streams:** these use large packets to be efficient with both computer resources and network bandwidth. They are usually chosen when PCs are the audio devices, so we do not suggest selecting this stream type for your Program, or Monitoring outputs, but rather to use it for IP Drivers setup, or specialized VMIX use. Packet time is 5ms.
- **Disabled:** Turns the stream off, making it invisible to the entire AoIP network.

### AES67 Settings

**IP Address:** enter an AES67 Multicast address by typing its IP in the box.

**Packet Time:** choose the desired packet time from the drop-down menu. The standard setting for AES67 is 1ms.

**Stream Type:** choose the desired Bit Length and Channel#.

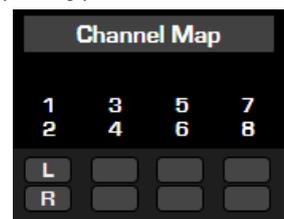


The default setting is: **L24 - 2ch** (24-Bit, Stereo). You can also disable the AES67 stream and leave its settings in place for later activation, by selecting the - **Disabled** - option.

### Channel Map

This section and the following, apply to both the Livewire and AES67 Output types.

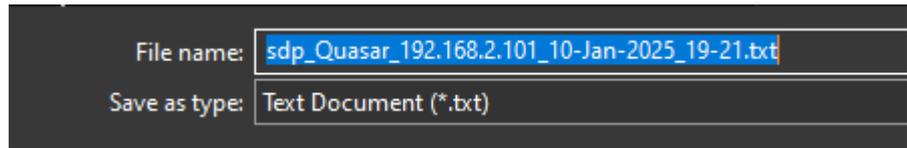
Use this section to Map Left and Right channels of the selected bus to output channels 1-8 of the LW Surround Stream, or in an AES67 multichannel stream. Type "L" or "R" in the desired Channel Field. You can populate all 8 fields, if required. For standard Stereo Stream operation type "L" in field #1 and "R" in field #2 (below).



Click the **Default Output Map** button to revert to standard stereo Stream operation on all outputs.

### SDP Export

Press this button to export the SDP configuration file for the selected bus.



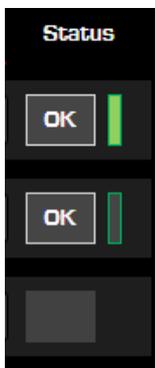
### Gain

Here you can set the “fader gain level” for each of the VMIX outputs, by entering any value between -80.0dB and +10.0dB.

### Stream Status

The same audio presence and stream status indicators as in other sections of the Engine are available for each VMIX output.

The **Status** window will normally display:



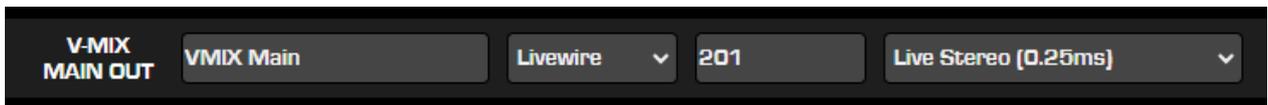
- An “OK” message, when the stream is successfully loaded
- An “ERR” message, when the audio stream is not loaded successfully
- A solid green bar indicates that the stream is active, and audio is present, on particular output.
- An empty bar with green outline indicates that the stream is active but no audio is present, on that output.
- No bar indicates the stream is not active, or not enabled.

**Note** - These settings are the same found in the [Main Outs Configuration](#).

- The **Status** window will normally display “OK” when the stream is enabled.
- The **Audio Indicator** will have a green outline in case the stream is enabled, but no audio is present, and will be solid green when audio is present.

### VMIX MAIN Out

The outputs of the 16 V-Mixers are summed into a Main V-MIX bus. In the bottom part of the V-MIX Main page you can set the name, Livewire channel number, and Stream type for the output of the Main V-MIX bus.



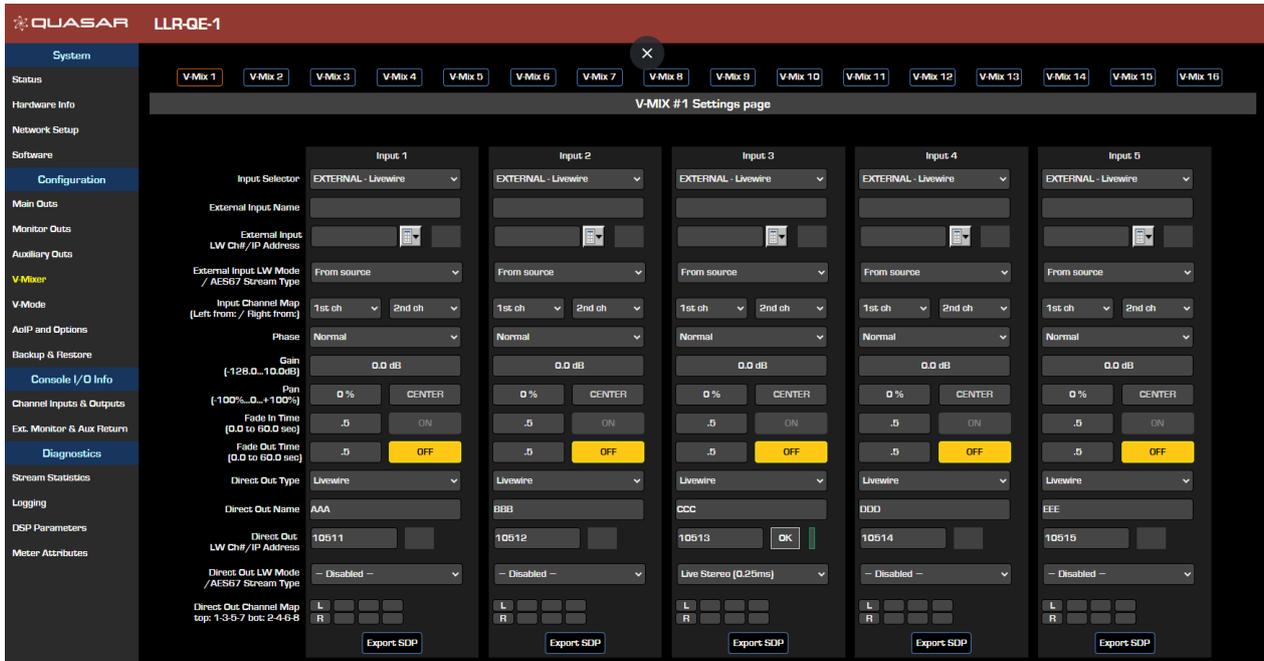
Unless you need a single output that combines the audio from of all the Submixes, you may leave this disabled – it doesn’t need to be enabled for each VMIX to work.



## V-Mix Submixer Settings

From the Main V-Mixer page, you can enter a setup page for each of the 16 V-Mixes, by pressing the buttons to the left of each V-Mix line. This will open the VMix #1-16 Settings page.

Each V-Mixer consists in a 5-Channel Mixer. The workflow for configuring each channel follows a top-to-bottom logic, and has been improved to always show only the relevant parameter fields, for each selected option.



The controls available for each of the channels are:

### Input Selector

Use this drop-down to choose between one of the “Internal” busses, generated by the Engine itself, or any of the streams active on your AoIP Network. These are considered to be “External” sources, and can be of two *Types*: **Livewire** or **AES67**.

If one of the two **EXTERNAL** types is selected, then the following three fields will appear below.

### External Input Name

In this field, enter the name of the source you’ll be assigning to the input.

### External Input LW Ch#/IP Address

In this field, enter the unique Livewire channel number of your audio source. Alternatively, you can pick a channel from the list of active sources by pressing the button next to the field. If “EXTERNAL – AES67” is selected, the field will let you enter the Multicast IP address of the desired AES67 stream.

Once a *valid* channel is selected, a small status/audio presence indicator will appear next to the button.

## External Input LW Mode/AES67 Stream Type

- **EXTERNAL - Livewire selected:** This will normally be set to **From Source**, meaning that the source itself (a mic, CD player, etc.) will be providing the audio stream. However, you can select **To Source** to use the source's automatically generated Backfeed (mix-minus audio) as an input.  
*Example:* Selecting **From Source** when a phone hybrid is assigned as a VMix Sub input would use the caller audio; selecting **To Source** would instead use the Mix Minus sent to the hybrid.
- **EXTERNAL - AES67 selected:** choose the desired Bit Length and Channel#. The default setting is: **L24 - 2ch** (24-Bit, Stereo). You can also choose **Auto**: this option will automatically detect the bit length and channel count of the stream.

## Phase

Here you can set the phase of the input signal: you have the option to invert phase on the Left Channel Only, Right Channel only, or both channels.

## Gain

Here you can set the “fader gain level” for each of the Submixer input channels, by entering any value between -128.0dB and +10.0dB.

## Pan

Here you can set the Panorama, or “Panpot” of the input signal by entering any value between -100 (100% Left), and 100 (100% Right). Typing 0 will display “CENTER” in the next field, once the **Apply** button is pressed at the bottom of the page.

## Fade In Time

This field controls the OFF to ON **Fade In time**, This function won't be used in normal operation, but can be used to create automatic cross fades between sources when Pathfinder is dynamically making changes to the VMix.

*Example:* If 1.0 is entered in the first field, the Submixer channel's audio will rise from  $-\infty$  to the value set in the **Gain** field in 1.0 seconds. If the field is set to 0, the audio will simply turn on at the gain value specified, immediately.

## Fade Out Time

This field controls the ON to OFF **Fade Out time**, and works in the same way.

## ON/OFF

The channel **ON** and **OFF** buttons are displayed next to the Fade In and Fade Out fields. The logic is intercanceling.

## Direct Out Type

Select **Livewire** or **AES67** to define the Direct Out type

## Direct Out Name

The name you give the post-on/off Submix channel to send back to the Axia network.

### Direct Out LW Ch#/IP Address

In case you need to create a Direct Output (post-on/off) for this input channel, you can enter the unique Livewire channel number you want to assign to it in this field. This will generate a new Source that will appear onto the AoIP network, if enabled.

### Direct Out LW Mode/AES67 Stream Type

Here you can choose between **Live Stereo [0.25ms]** , **Low-Latency Stereo [1ms]** , and **Standard Stereo [5ms]** to enable the direct output stream for this VMix channel. Choose **-Disabled-** if no stream needs to be generated.

### Direct Out Channel Map

Use this section to Map Left and Right channels of the selected bus to output channels 1-8 of the LW Surround Stream, or in an AES67 multichannel stream. Type "L" or "R" in the desired Channel Field. You can populate all 8 fields, if required. For standard Stereo Stream operation type "L" in field #1 and "R" in field #2 (below).

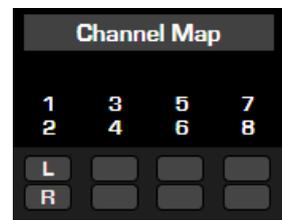
This is all you need to do to configure inputs for a VMix Submixer. You can configure up to 5 input streams per VMix Submixer.

### V-Mixer Submixer Out Settings

In the bottom part of the V-MIX Settings page below the 5 input channels, you can set the name, Livewire channel number, and Stream type for the output of the V-MIX Submixer. These duplicate for your convenience the settings available in the Main V-Mixer Page.



At the bottom of each of the Submix sections is an **Apply** button. The adjustments made to a V-Mixer take effect as soon as you apply them, so changes saved “on the fly” will affect your output streams immediately.



Be sure to save the changes for each Submixer as they are configured.

**Tip** - In most cases, the VMix Subs are the only channels you will need to enable, since each Sub has its own direct output. Only enable the Submixes that you intend to actively use; active Submixes without any activity clutter up your network with empty streams.

## Some V-Mix Examples

Now that you know how to enable and set up VMix Submixers, what might you do with it? Here are some examples.

### Input Groups

A typical application for the V-Mixers is to create audio Subgroups for up to 5 inputs and present them onto your console as a single Source.

### Pre-Mixing your Playout System

A widely used application for the V-Mixes, is that of mixing clean (non-processed) audio program from the console with Split Ads played by a dedicated Split Player, before it is sent to the Split processor.

Some facilities may have a Multi-channel Playout system connected to a small console, with not so many faders. Only some very advanced (and expensive) Playout Systems have their own software audio Engine, capable to internally mix and route the individual output channels, before they're sent to the PC audio driver. In this case, having the possibility to premix these sources into one, to save faders on the console, can be very useful.

### Pre-Mixing the PFL bus

Another application is to premix the PFL channel of the Playout system directly with the Engine's PFL/AFL output. The VMix Sub Channel would then be the audio source that you would route to the PFL speaker.

## Combining VMix with Pathfinder Core PRO Routing Control

Axia Pathfinder Core PRO routing tools can be used to “control” VMix in several different ways.

First, as a background controller, Pathfinder can monitor Livewire system parameters or receive commands from external devices like satellites, button panels, or automation systems and react to them by changing the state of VMIX ON/OFF, Gain, Time Up, and/ or Time Down fields.

This provides many different possibilities for facility automation, Intercom functions, or whatever else you might imagine. For example, the combo of Pathfinder Core PRO and VMix could duplicate the function provided by other products that are controlling audio switching in many radio facilities. (Refer to documentation on [Pathfinder Core PRO](#) for further information.)



Second, Pathfinder Core PRO has a built-in VMIX controller web page specifically to provide a quick mixer user interface to the V-Mixers in your network. You can use parameters to specify which engines and V-Mixers a user may access.

Third, you can access V-Mixers through the user panel designer to create your own custom user interface. This tool can be used to drag faders, labels, buttons, gauges and other controls onto a panel and then configure them to interact with Engine V-Mixers in whatever way you wish.

There are several other ways that Pathfinder Core PRO can be used for background control of VMix. VMix functions can be used both as qualifiers and actions in routing salvos, scene changes, and logic flows. This means that a designer can select GPIO triggers, time-based events on a calendar, user button presses, serial port commands, and other options and combinations of options to decide when to make changes to any Virtual Fader in a VMix. The user can make a gain change based on these events, turn a channel off or on, and or adjust the fade times, giving complete control over the VMix based on any of the logic flows.

Finally, Pathfinder Core PRO’s API and Generic emulators include commands to control any VMix fader that’s active, so any machine that can send user defined serial or TCP commands can also control and read VMix functions through Pathfinder Core PRO.

Using these techniques VMIX can be used as a fully automated virtual mixer in the background of each console.



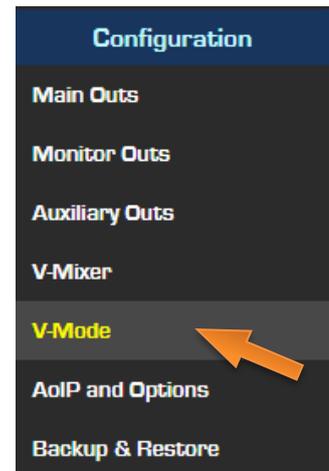
## GPIO control of your VMIX with Pathfinder

Imagine that you have a night DJ that *should* monitor all four radio stations in your cluster. To help make sure this actually happens, you could send all four off-air signals as sources into a VMix Submixer, and take the output of that Submix to a monitor. A Fusion accessory panel or external button wired to a GPIO port could then provide a “press & hold” function to allow the DJ to monitor the sources momentarily. (This example is only possible with Pathfinder control of VMix.)

## V-Mode

V-Mode (short for Virtual Mode converter) can be used to convert channel count, channel order, and/or channel content between audio streams at its input and output.

For example, passing some channels while cutting others, summing stereo to mono, upmixing or downmixing between stereo and multi-channel, combining channels from two inputs into one output. Select audio as input, define the “Mode” (conversion type, or in other words, the routing within an internal matrix), and define the output as a network source into the AoIP network.



There are 16 V-Mode instances in a Quasar Engine:

Each instance supports audio streams with up to 8 channels at the input and up to 8 channels at the output. Between the input and output, there is a matrix, where the conversion is performed, according to the selected audio routing option (“Mode”). Livewire 8-channel streams carry 5.1 content in channels 1-6, and stereo content in channels 7 and 8, and the matrix is partitioned the same way, to fit them natively:

- Channels 1 to 6 of the matrix, form a multi-channel part
- Channels 7 and 8 of the matrix, form a stereo part

Routing option (“Mode”) names refer to the parts of the matrix and conversions performed between them. They do not refer to channel maps of the input/output streams.

For receiving and transmitting, two stream classes are distinguished:

- Class “mono/stereo”: streams with 1 or 2 channels
- Class “multi-channel”: streams with 3 to 8 channels

Input streams put their content into, and output streams take their content from the stereo or multi-channel parts of the matrix, according to their mono/stereo or multi-channel classes. Multi-channel streams with 7 and 8 channels span across the both parts of the matrix, starting from channel 1, in the natural order. Livewire 8-channel “Surround” streams fit the matrix mapping without reordering.

## V-Mode Main Page

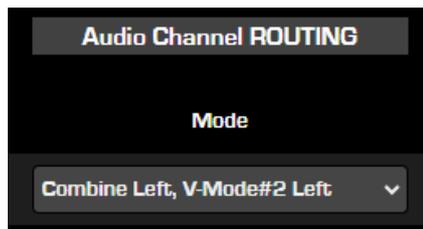
The V-Mode Main page offers an overview of all the 16 V-Modes available in the Quasar Engine. For each of the 16 V-Mode instances, the following basic parameters are displayed:

- Selected Input
- Input Name
- Input Address
- Input Channel Mapping
- Input Status

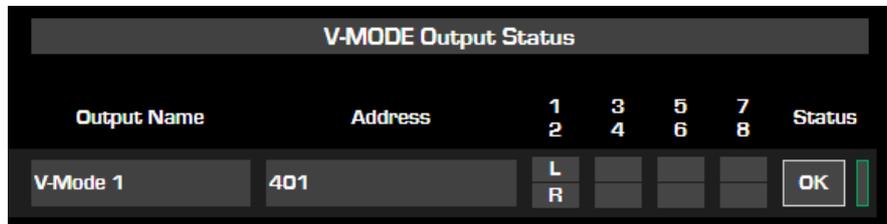
A screenshot of a table titled "V-MODE Input Status". The table has columns for "Selected Input", "Input Name", "Address", "FL FR", "FC LFE", "SL SR", "Left Right", and "Status". The first row shows "V-Mode 01" as the selected input, "LIVEWIRE" as the input name, "1011" as the address, and "1" and "2" for the left and right channels respectively. There is an "OK" button in the status column.

Selected Input	Input Name	Address	FL FR	FC LFE	SL SR	Left Right	Status
V-Mode 01	LIVEWIRE	1011				1 2	OK

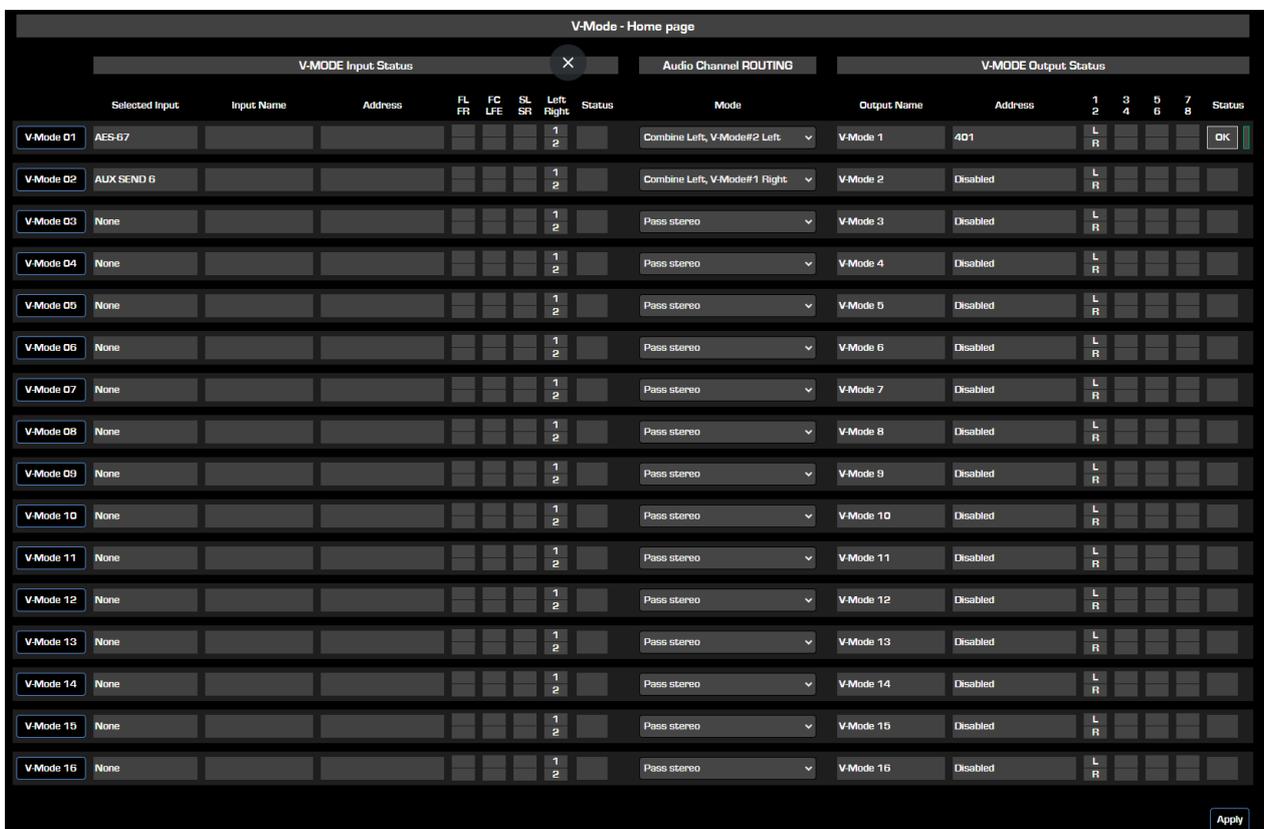
- Mode



- Output Name
- Output Address
- Output Channel Mapping
- Output Status



Only the **Audio Channel Routing** settings could be edited from this page, using the **Mode** drop-downs: we duplicated this drop-down menu here for the user's convenience.

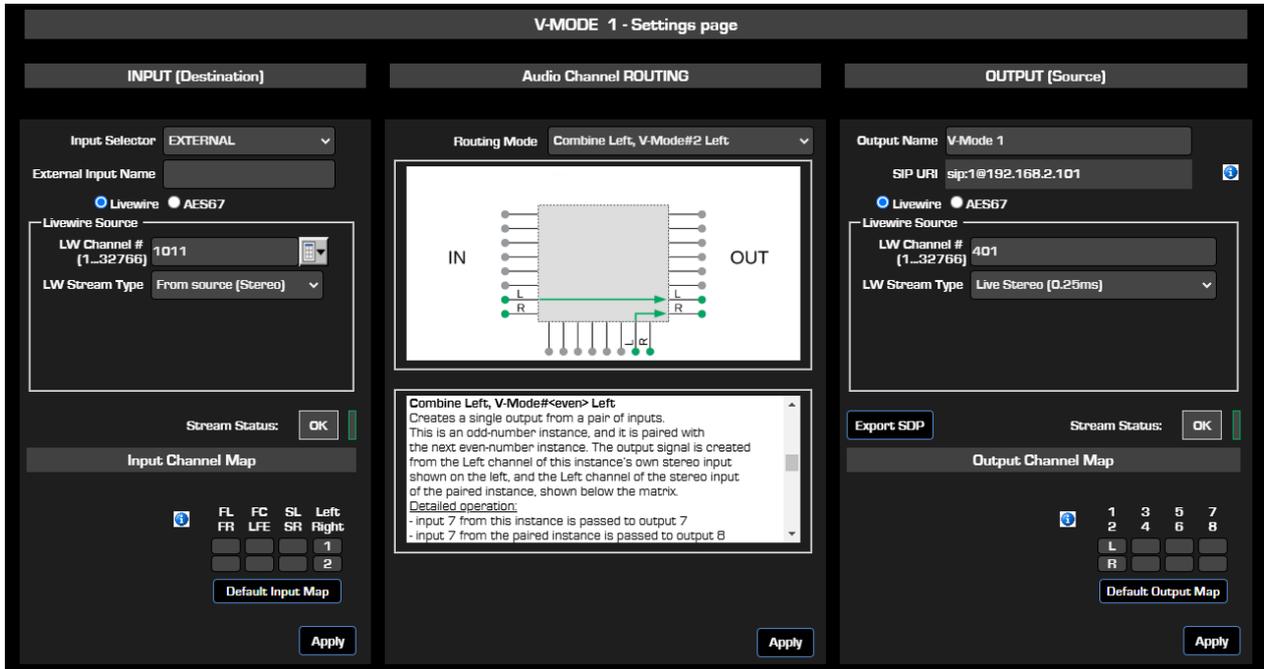


The other parameters fields in this page, which was designed with the goal of providing a view of all 16 V-Modes at a glance, are for view only.

## V-Mode Settings page

In order to access and configure each V-Mode, press one of the selection buttons to the left of the V-Mode Home Page: The **V-Mode Settings page** will be opened, where it is possible to configure each parameter of the selected V-Mode.

Each V-Mode consists in a signal processor with 3-Blocks: **Input** (left block), **Routing** (center block), **Output** (right block).



You would normally configure the Input block first, then choose the desired Routing Mode, and then configure the Output block. The workflow for configuring each block follows a top-to-bottom logic, and has been improved to always show only the relevant parameter fields, for each selected option.

Some configuration fields may dynamically appear or disappear, depending on the options chosen, in order to hide the fields that do not require configuration and to make the overall configuration process, trouble-free.

The controls available for each block are:

### Input (Destination)

The **Input Selector** drop-down is used to select any source from two possible categories:

- INTERNAL – A source generated by the Engine. In this case, no more parameter are displayed for the input block.
- EXTERNAL - A source from the network. In this case the following additional parameters will appear:

The **External Input name** is a text field used to label the input.

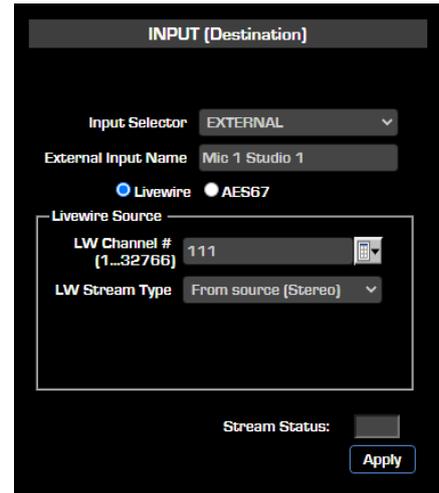
The **External Input Type** radio buttons let you choose between Livewire and AES67:

Selecting **Livewire** will display the **Livewire Source** configuration field, where a **LW Channel** can be entered directly as a number, or selected from the pop-up menu, by clicking on the button next to the field.

From the **LW Stream Type** drop-down, choose between:

- From Source (Stereo)
- From Source (Surround)
- To Source

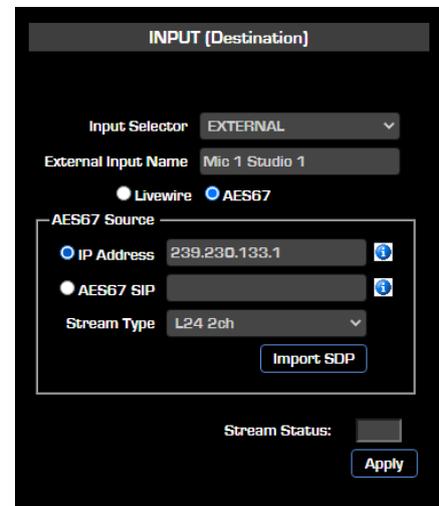
If you want to select the Source's main audio feed, in either Stereo or Surround format, or the return feed (backfeed) from that Source.



Selecting **AES67** will display the **AES67 Source** configuration field, where an **IP Address** can be entered directly using the following syntax options:

- Multicast IP: Address of AES67 multicast stream. Standard RTP port number 5004 will be used.
- [0.0.0.0]:<port> Unicast stream receive port number. A stream from any sender will be accepted.
- <Unicast IP>:port Unicast stream sender, and the receive port number. Only a stream from the indicated IP address will be accepted.

Or an **AES67 SIP** connection string can be entered, in order to initiate a SIP Unicast connection, in AES67 format. The maximum length of the string is 299 characters.

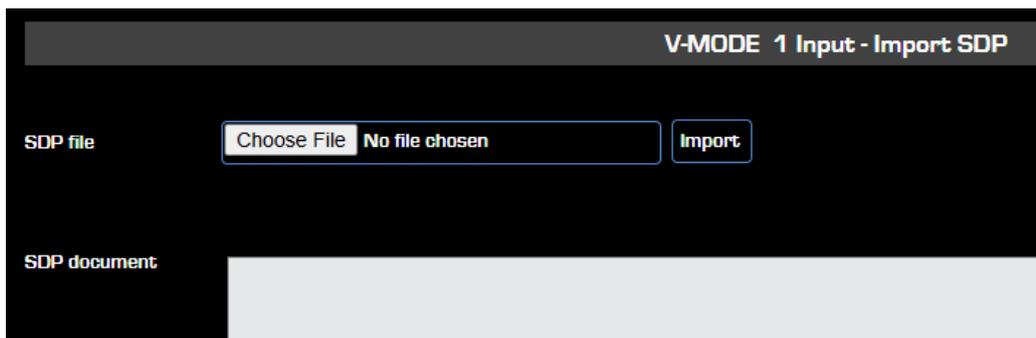


**Note** - Different devices may have different requirements for SIP unicast connections. Please check with the device's manufacturer what convention is used for establishing a SIP connection.

**Stream type** is used to define the source. With AES67, the stream could be 24-bit linear, or 16-bit linear, up to 8 channels. Make sure you select the correct type.

Alternatively, you can import an SDP (Session Description Protocol) string and apply it directly to the AES67 Source configuration, by clicking on the **Import SDP** button.

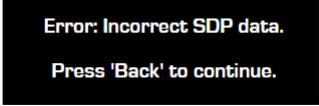
The following page will appear:



Where you can Choose a File, **Import** it, and visualize it from the **SDP Document** field to verify it is correct.

You can also enter the Document text directly into the field, if you are copy and pasting it from some other application. Then click the **Apply to AES67 Settings** button at the bottom of the page, to transfer it back into the Input configuration.

The Engine will automatically check the SDP data: In case the syntax should not be valid, or incorrect, an error message will be displayed.

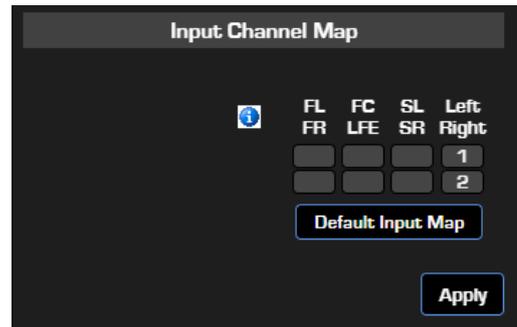


### Input Channel Map

Use this section to change the default routing of the channels bundled with a LW Surround Stream, or with an AES67 multichannel stream, BEFORE they are sent to the 8x8 Matrix.

You can Map the 8 channels of the Input Stream to inputs 1-8 of the Matrix. You can type the following characters in the desired Channel Field:

- "1" for routing the 1st Stream channel
- "2" for routing the 2nd Stream channel
- "3" for routing the 3rd Stream channel
- "4" for routing the 4th Stream channel
- "5" for routing the 5th Stream channel
- "6" for routing the 6th Stream channel
- "7" for routing the 7th Stream channel
- "8" for routing the 8th Stream channel



You can populate all 8 fields, if required. For standard Stereo Stream operation type "1" in field "L" and "2" in field "R" (below).

Click "Default Input Map" to revert to standard stereo Stream operation on all inputs if you are not using multichannel streams.

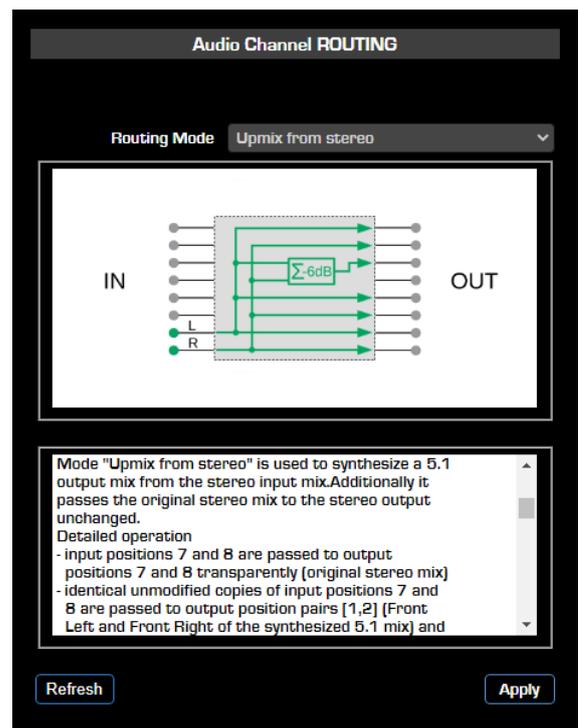
### Audio Channel Routing (Mode)

This block consists essentially in an I/O Matrix. Modes are a collection of routing presets that we made available to manipulate audio channels within a stream, when this enters the matrix.

19 different Modes are available from the drop-down menu. Press the **Preview** button to display a block diagram of the selected routing mode. In case you change the mode, the button will turn into a **Refresh** button, in order to let you display the last mode you selected. Press **Apply** to apply the routing.

Two stream classes are utilized within the different mode designations:

- Class "stereo": streams with 1 or 2 channels ("Left" or "Right" may also be used in this class)
- Class "multi-channel": streams with 3 to 8 channels



Two “Pass” modes are provided, to pass input signals unaltered in both their gain level and routing, to the outputs of the matrix. These are normally chosen when only a stream format conversion needs to be applied (example: from Livewire to AES67 or vice versa).

**Pass stereo** passes only the stereo part of the matrix transparently, and blocks the multichannel part.

**Pass multi-channel** passes all 8 matrix inputs to the corresponding outputs transparently:

- Unity gain in all channels
- No up-/down-mixing
- Preserving the channel order regardless of the actual number of channels in the input and output streams
- No signal in output channels, where corresponding channels do not exist at input

Furthermore:

11 Modes are provided, that alter the internal channel routing and gain level within the matrix,

and 12 Modes are provided, that route signals between two adjacent V-Modes and alter the signal routing inside each matrix, to recombine the signals before they are sent to the outputs of the two V-Modes.

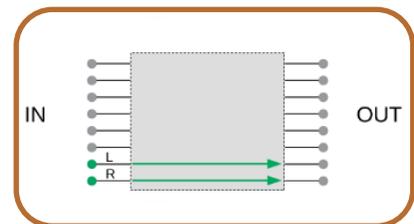
The next diagrams offer a detailed description of each Mode option:

**Pass stereo**

transparently passes a mono or stereo input to the output.

Detailed operation:

- input 7 and 8 are passed to output 7 and 8 transparently
- silence is fed to output 1 to 6

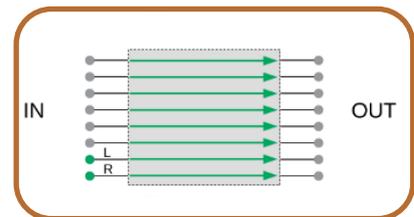


**Pass multi-channel**

transparently passes a multi-channel input to the output.

Detailed operation:

- input 1 to 8 are passed transparently through the matrix

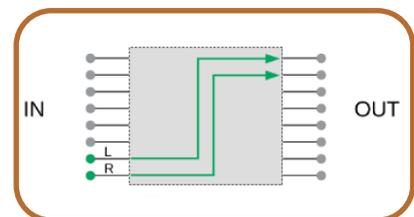


**Transpose stereo to multi - channel [1, 2]**

passes a mono or stereo input to the lowest channel numbers of a multi-channel output.

Detailed operation:

- input 7 and 8 are passed to output 1 and 2
- silence is fed to output 3 to 8

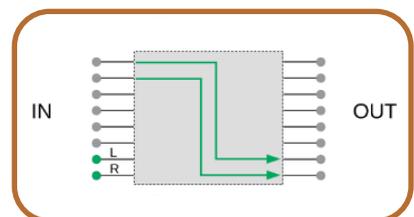


**Transpose multi - channel [1, 2] to stereo**

passes the lowest channel numbers of a multi-channel input to a mono or stereo output.

Detailed operation:

- input 1 and 2 are passed to output 7 and 8
- silence is fed to output 1 to 6

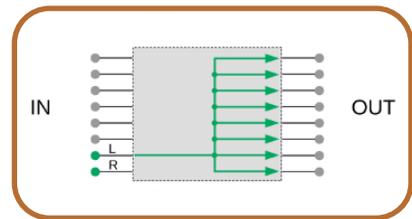


### Upmix from Left

replicates a mono input or the Left channel of a stereo input to all 8 output channels.

Detailed operation:

- identical unmodified copies of input 7 are passed to output 1 to 8

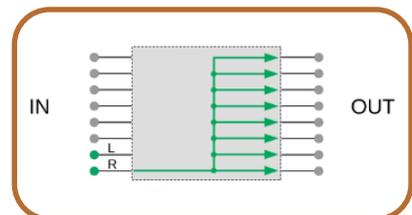


### Upmix from Right

replicates the Right channel of a stereo input to all 8 output channels.

Detailed operation:

- identical unmodified copies of input 8 are passed to output 1 to 8

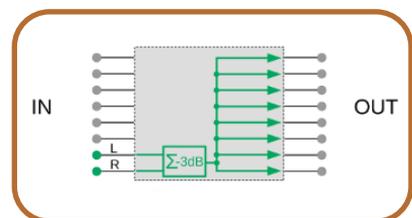


### Upmix from Left + Right

creates a mono sum of a stereo input and replicate it to all 8 output channels.

Detailed operation:

- identical copies of the sum of input 7 and 8, attenuated by 3 dB, are passed to output 1 to 8

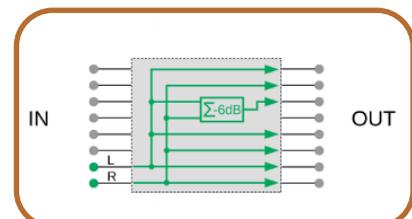


### Upmix from stereo

synthesizes a 5.1 output mix from the stereo input mix. Additionally it passes the original stereo mix to the stereo output unchanged.

Detailed operation:

- input 7 and 8 are passed transparently to output 7 and 8 (original stereo mix)
- identical unmodified copies of input 7 and 8 are passed to output pairs [1,2] (Front Left and Front Right of the synthesized 5.1 mix) and [5,6] (Surround Left and Surround Right of the synthesized 5.1 mix)
- the sum of input 7 and 8, attenuated by 6 dB, is passed to output 3 (Front Center of the synthesized 5.1 mix)
- silence is fed to output position 4 (LFE of the synthesized 5.1 mix)

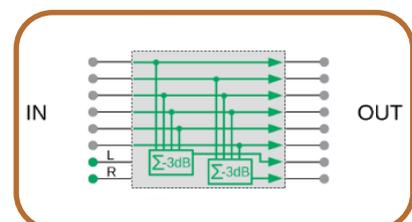


### Downmix from 5.1

synthesizes a stereo output mix from the 5.1 input mix. Additionally it passes the original 5.1 mix to the 5.1 output unchanged.

Detailed operation:

- input 1 to 6 are passed transparently to output 1 to 6 (original 5.1 mix)
- the sum of input 1, 3, 4, and 5 (Front Left, Front Center, LFE, and Surround Left of the original 5.1 mix), attenuated by 3 dB, is passed to output position 7 (Left of the synthesized stereo mix)
- the sum of input 2, 3, 4, and 6 (Front Right, Front center, LFE, and Surround Right of the original 5.1 mix), attenuated by 3 dB, is passed to output position 8 (Right of the synthesized stereo mix)

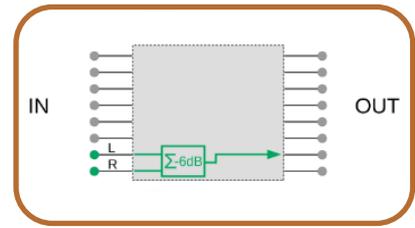


### Split Left - 6dB

Sends a mono sum of the input stereo mix to the Left channel of the stereo output.

Detailed operation:

- the sum of input 7 and 8, attenuated by 6 dB, is passed to output 7
- silence is fed to output 1 to 6, and 8

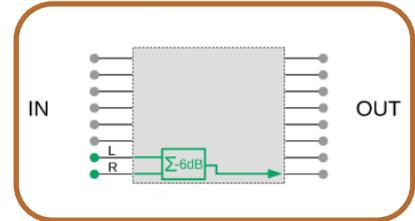


### Split Right - 6dB

Sends a mono sum of the input stereo mix to the Right channel of the stereo output.

Detailed operation:

- the sum of input 7 and 8, attenuated by 6 dB, is passed to output 8
- silence is fed to output 1 to 7

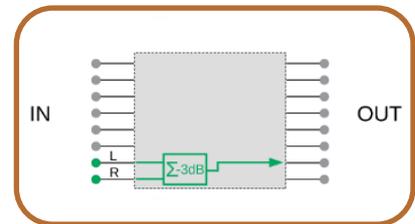


### Split Left - 3dB

Sends a mono sum of the input stereo mix to the Left channel of the stereo output.

Detailed operation:

- the sum of input 7 and 8, attenuated by 3 dB, is passed to output 7
- silence is fed to output 1 to 6, and 8

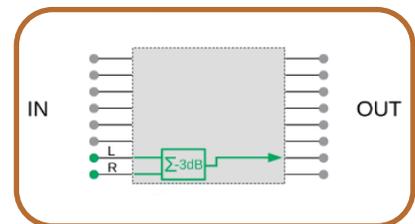


### Split Right - 3dB

Sends a mono sum of the input stereo mix to the Right channel of the stereo output.

Detailed operation:

- the sum of input 7 and 8, attenuated by 3 dB, is passed to output 8
- silence is fed to output 1 to 7



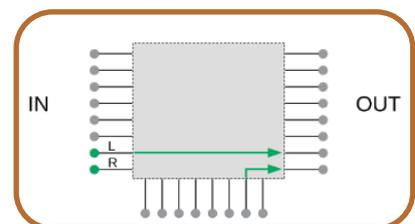
### Combine Left, V-Mode#2 Left

Creates a single output from a pair of inputs.

This is an odd-number instance, and it is paired with the next even-number instance. The output signal is created from the Left channel of this instance's own stereo input shown on the left, and the Left channel of the stereo input of the paired instance, shown below the matrix.

Detailed operation:

- input 7 from this instance is passed to output 7
- input 7 from the paired instance is passed to output 8
- silence is fed to output 1 to 6



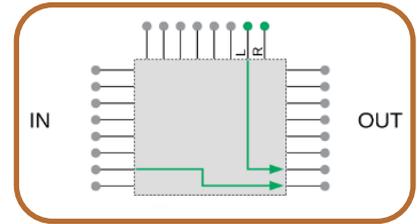
### Combine Left, V-Mode#1 Left

Creates a single output from a pair of inputs.

This is an even-number instance, and it is paired with the previous odd-number instance. The output signal is created from the Left channel of this instance's own stereo input shown on the left, and the Left channel of the stereo input of the paired instance, shown above the matrix.

Detailed operation:

- input 7 from the paired instance is passed to output 7
- input 7 from this instance is passed to output 8
- silence is fed to output 1 to 6



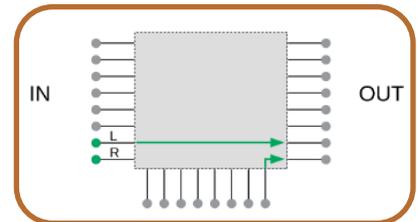
### Combine Left, V-Mode#2 Right

Creates a single output from a pair of inputs.

This is an odd-number instance, and it is paired with the next even-number instance. The output signal is created from the Left channel of this instance's own stereo input shown on the left, and the Left channel of the stereo input of the paired instance, shown below the matrix.

Detailed operation:

- input 7 from this instance is passed to output 7
- input 8 from the paired instance is passed to output 8
- silence is fed to output 1 to 6



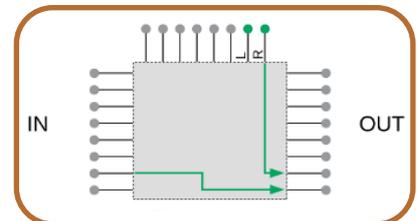
### Combine Left, V-Mode#1 Right

Creates a single output from a pair of inputs.

This is an even-number instance, and it is paired with the previous odd-number instance. The output signal is created from the Left channel of this instance's own stereo input shown on the left, and the Right channel of the stereo input of the paired instance, shown above the matrix.

Detailed operation:

- input 8 from the paired instance is passed to output 7
- input 7 from this instance is passed to output 8
- silence is fed to output 1 to 6



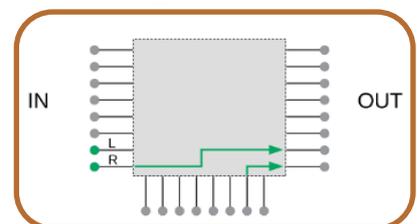
### Combine Right, V-Mode#2 Left

Creates a single output from a pair of inputs.

This is an odd-number instance, and it is paired with the next even-number instance. The output signal is created from the Right channel of this instance's own stereo input shown on the left, and the Left channel of the stereo input of the paired instance, shown below the matrix.

Detailed operation:

- input 8 from this instance is passed to output 7
- input 7 from the paired instance is passed to output 8
- silence is fed to output 1 to 6



### Combine Right, V-Mode#1 Left

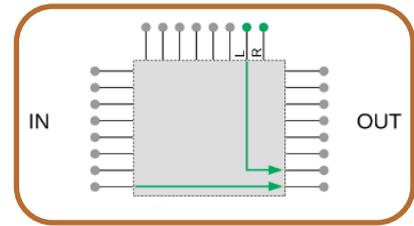
Creates a single output from a pair of inputs.

This is an even-number instance, and it is paired with the previous odd-number instance.

The output signal is created from the Right channel of this instance's own stereo input shown on the left, and the Left channel of the stereo input of the paired instance, shown above the matrix.

Detailed operation:

- input 7 from the paired instance is passed to output 7
- input 8 from this instance is passed to output 8
- silence is fed to output 1 to 6



### Combine Right, V-Mode#2 Right

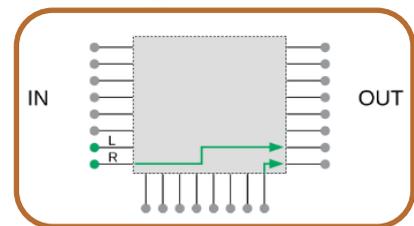
Creates a single output from a pair of inputs.

This is an odd-number instance, and it is paired with the next even-number instance.

The output signal is created from the Right channel of this instance's own stereo input shown on the left, and the Right channel of the stereo input of the paired instance, shown below the matrix.

Detailed operation:

- input 8 from this instance is passed to output 7
- input from the paired instance is passed to output 8
- silence is fed to output 1 to 6



### Combine Right, V-Mode#1 Right

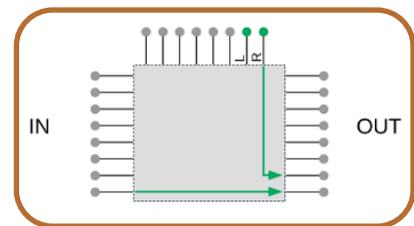
Creates a single output from a pair of inputs.

This is an even-number instance, and it is paired with the previous odd-number instance.

The output signal is created from the Right channel of this instance's own stereo input shown on the left, and the Right channel of the stereo input of the paired instance, shown above the matrix.

Detailed operation:

- input 8 from the paired instance is passed to output 7
- input 8 from this instance is passed to output 8
- silence is fed to output 1 to 6



### Combine stereo, V-Mode#2 5.1

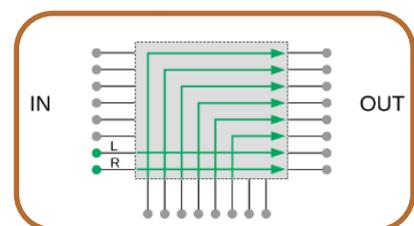
Create a single output from a pair of inputs.

This is an odd-number instance, and it is paired with the next even-number instance.

The output signal is created from this instance's own stereo input shown on the left, and the 5.1 input of the paired instance, shown below the matrix.

Detailed operation:

- input 7 and 8 from this instance are passed to output 7 and 8
- input 1 to 6 from the paired instance are passed to output 1 to 6



### Combine stereo, V-Mode#1 5.1

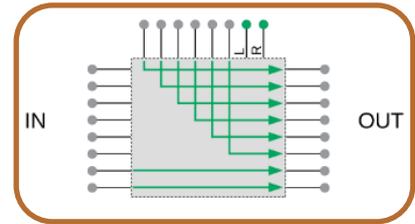
Creates a single output from a pair of inputs.

This is an even-number instance, and it is paired with the previous odd-number instance.

The output signal is created from this instance's own stereo input shown on the left, and the 5.1 input of the paired instance, shown above the matrix.

Detailed operation:

- input 7 and 8 from this instance are passed to output 7 and 8
- input to 6 from the paired instance are passed to output 1 to 6



### Combine 5.1, V-Mode#2 stereo

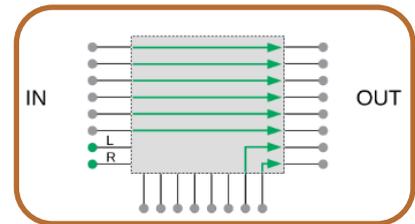
Creates a single output from a pair of inputs.

This is an odd-number instance, and it is paired with the next even-number instance.

The output signal is created from this instance's own 5.1 input shown on the left, and the stereo input of the paired instance, shown below the matrix.

Detailed operation:

- input 1 to 6 from this instance are passed to output 1 to 6
- input 7 and 8 from the paired instance are passed to output 7 and 8



### Combine 5.1, V-Mode#1 stereo

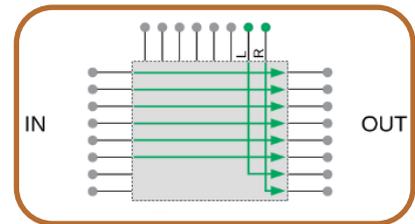
Creates a single output from a pair of inputs.

This is an even-number instance, and it is paired with the previous odd-number instance.

The output signal is created from this instance's own 5.1 input shown on the left, and the stereo input of the paired instance, shown above the matrix.

Detailed operation:

- input 1 to 6 from this instance are passed to output 1 to 6
- input 7 and 8 from the paired instance are passed to output 7 and 8



## Output (Source)

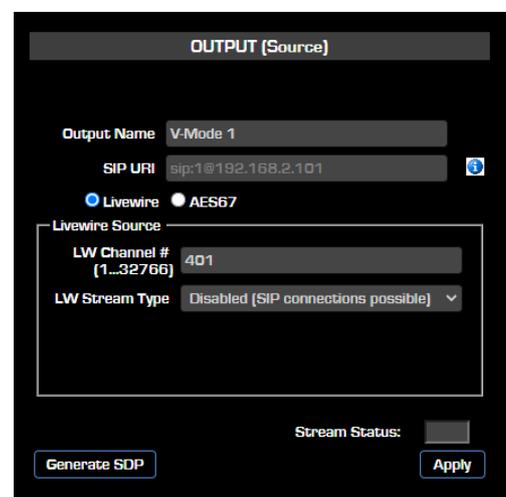
The **Output name** is a text field used to label the output.

The **Output Type** radio buttons let you choose between Livewire and AES67.

Selecting **Livewire** will display the **Livewire Source** configuration field, where a **LW Channel** can be entered directly as a number, or selected from the pop-up menu, by clicking on the button next to the field.

From the **LW Stream Type** drop-down, choose between:

- Live Stereo
- Standard Stereo
- Surround
- Disabled



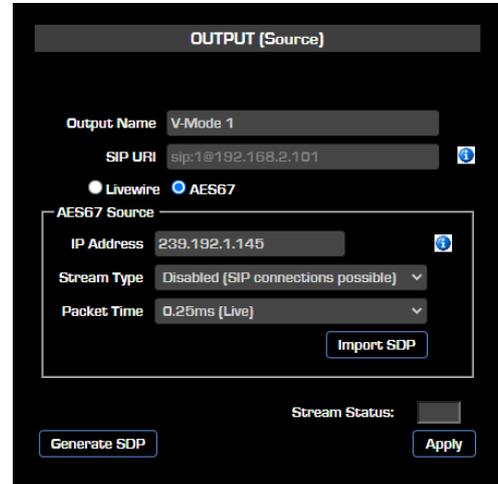
to select the Source stream type, in either Live Stereo, Standard Stereo, or Surround formats.

Choose Disabled if you want to allow SIP connections to this output.

Selecting **AES67** will display the **AES67 Source** configuration field, where an **IP Address** can be entered directly using the following syntax options:

- Multicast IP: Address of AES67 multicast stream. Standard RTP port number 5004 will be used.
- [0.0.0.0]:<port> Unicast stream receive port number. A stream from any sender will be accepted.
- <Unicast IP>:port Unicast stream sender, and the receive port number. Only a stream from the indicated IP address will be accepted.

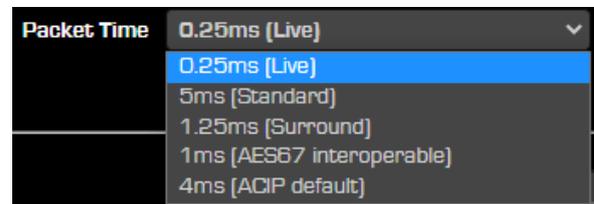
Or an **AES67 SIP** connection string can be entered, in order to initiate a SIP Unicast connection, in AES67 format. The maximum length of the string is 299 characters.



**Note** - Different devices may have different requirements for SIP unicast connections. Please check with the device's manufacturer what convention is used for establishing a SIP connection.

**Stream type** is used to define the source. With AES67, the stream could be 24-bit linear, or 16-bit linear, up to 8 channels. Make sure you select the correct type.

**Packet Time** - This field is for entering the required AES67 Stream Packet Time.



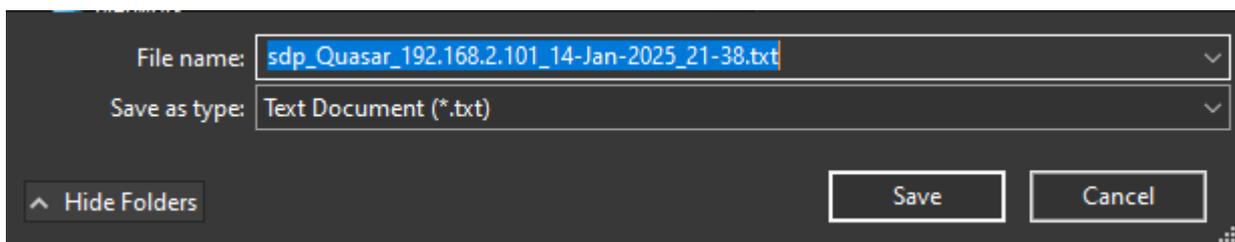
Choose between 0.25ms, 5ms, and 1.25 for creating an output stream compatible with LW Live, Standard, or Surround formats.

Choose between 1ms and 4ms to create an AES67 interoperable stream or an ACIP-compliant stream.

**Note** - Different Packet Times will provide different stream latencies.

**SIP URI** - This field is for entering an address to make a SIP connection to this Output.

Alternatively, you can export an SDP (Session Description Protocol) string and apply it directly to the AES67 Source configuration, by clicking on the **Export SDP** button. The following page will appear:



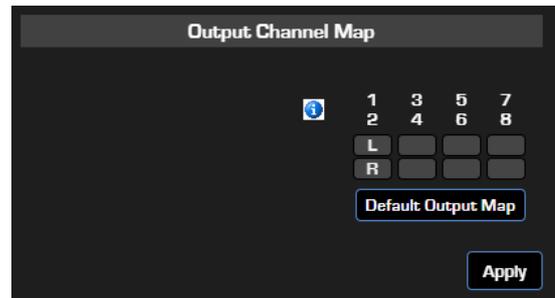
Where you can Choose a Filename, **Export** it, and save it for Import in a different system.

### Output Channel Map

Use this section to change the default routing of the channels coming out of the 8x8 Matrix BEFORE they are bundled in a LW Surround Stream, or in an AES67 multichannel stream.

You can Map the 8 Outputs of the Matrix to channels 1-8 of the Output Stream. You can type the following characters in the desired Channel Field:

- "FL" for routing Front Left signal
- "FR" for routing Front Right signal
- "FC" for routing Front Center signal
- "LF" for routing LFE signal
- "SL" for routing Surround Left signal
- "SR" for routing Surround Right signal
- "L" for routing Stereo Left signal
- "R" for routing Stereo Right signal



You can populate all 8 fields, if required. For standard Stereo Stream operation type "L" in field #1 and "R" in field #2 (below).

Click "Default Output Map" to revert to standard stereo Stream operation on all outputs if you are not using multichannel streams.

Intentionally Left Blank

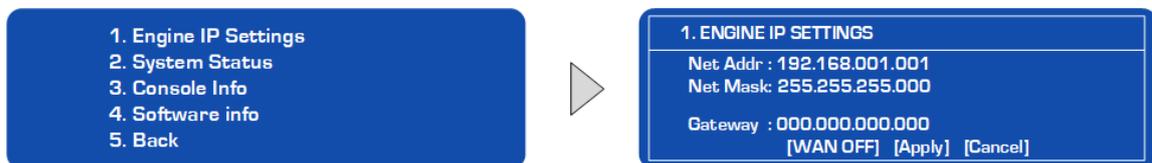
# Appendix A

## Previous Quasar Engine Platforms

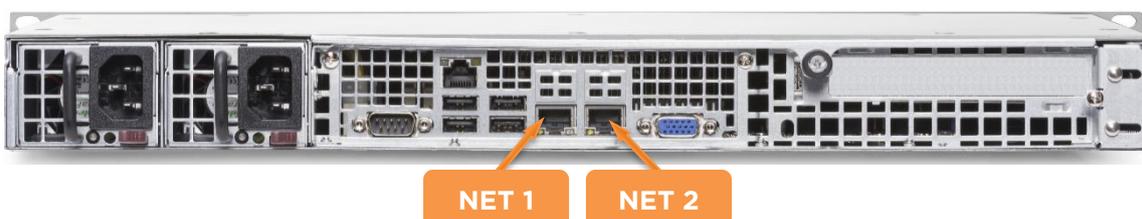
During its first year of life, in 2020, the Quasar Engine used to ship with a different hardware platform. The only difference with the new platform is in the front panel which, in the previous version, had an integrated LCD display.



- Install your Quasar Engine platform in a suitable environment, such as an air conditioned machine room. Please leave one empty RU space above and below the Engine for proper airflow through the sides of the chassis.
- Connect the mains cables to your Quasar Engine. Push the ON/OFF (D) button on the front panel. At the end of the boot process, the four LEDs to the left of the front display could indicate errors due to missing sync, console not yet connected, or connection of a single Power Supply only. This is normal.
- Push the **V** button (E) on the front panel to access the main menu. Select *Engine IP settings* using the arrow buttons, then push again the **V** button to select.



- Push the UP/DOWN arrows buttons to select *Net Address* field, and Push the **V** button to enter.
- Use the LEFT/RIGHT arrow buttons to select each digit, the UP/DOWN arrows buttons to change the value, then Push the **V** button to enter.
- Move the cursor to the right and select the **↵** symbol. Push the **V** button to confirm the setting.
- Repeat the above steps to set the Netmask.
- Once on the Engine IP settings are entered, move the cursor down to *[Apply]* and Push the **V** button to select. The Engine will prompt you with a request to reboot.
- Select OK and Push the **V** button. Confirm your choice in order to reboot the Engine with the new settings.
- Connect the network cable to the NET2 port (rightmost port, if looking at the rear panel).



# Appendix B

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## Quasar Surface Specifications

Power Supply
Universal AC Input Range (90 - 264VAC)
Operating Mains Frequency 47-63Hz
-40 to +85 degrees Celsius, no condensation
IEC receptacle, with locking clip. Internal fuse.
Mains Outlet rating: 200W minimum (for each PSU module)
Standard Operating Power consumption: 50W (each PSU module)

Surface Operating Temperatures
-10 degrees C to +40 degrees C, <70% humidity, no condensation
Mechanical Data
Width sizes: 430mm (2.5U), 585mm (3.5U), 740mm (4.5U), 895mm (5.5U), 1050mm (6.5U), 1205mm (7.5U), 1360mm (8.5U) Depth: 580mm Front Height (measured after armrest) : 50mm Rear Height (measured at highest point) :110mm
From 4 to 28 faders in a single frame. Up to 60 with two frames (split or bolted together)
Unit Weight: N/A kgs
Shipping Weight: N/A kgs

# Quasar Engine Specifications

## Input Filters

Filter 1 & 2 Frequency : 20Hz to 20.0kHz

Filter 1 & 2 Slope: 6-12-24-36-48 dB/Octave, selectable

## 4-Band Equalizer

Bands 1-2-3-4 Frequency : 20Hz to 20.0kHz

Bands 1-2-3-4 Gain: -25.0dB to +20.0dB

Bands 1-2-3-4 Q: 0.2 to 20.0

Bands 1-2-3-4 Type: Peak, Low Shelf, High Shelf selectable

EQ Output Trim Gain: -10.0dB to +10.0dB

## Compressor

Threshold: -60.0dB to 0.0dB

Ratio: 1.0 : 1 to 50.0 : 1

Knee: 0.0dB to +30.0dB

Auto Gain Make-up: Adjustable from 0% to 100% (+30dB)

Attack Time: 0.1ms to 1.00s

Release Time: 0.1ms to 5.00s

Automatic Attack & Release Time selectable

## Expander/Noise Gate

Threshold: -60.0dB to 0.0dB

Ratio: 1.0 : 1 to 50.0 : 1

Knee: 0.0dB to +30.0dB

Depth: 0.0dB to +60.0dB

Attack Time: 0.1ms to 1.00s

Release Time: 0.1ms to 5.00s

Low Frequency Filter: 1.99kHz to 6.31kHz

High Frequency Filter: 3.98kHz to 12.5kHz

## De-Esser

Threshold: -60.0dB to 0.0dB

Ratio: 1.0 : 1 to 50.0 : 1
Depth: 0.0dB to +30.0dB
Attack Time: 0.1ms to 1.00s
Release Time: 0.1ms to 5.00s
Automatic Attack & Release Time selectable
<b>Mechanical Data</b>
1 RU chassis, Industrial-Grade hardware platform
Width: 482mm - 19"
Depth: 457mm - 18"
Height: 1 Rack Unit
Unit Weight: - depends on hw configuration -
Shipping Weight: - depends on hw configuration -
<b>Power Supply</b>
Dual Redundant, hot-swap capable Power Supply modules
Auto-sensing, auto-ranging power supplies. 90 - 132 / 187 - 264 VAC, 50Hz/60Hz.
IEC receptacle, internal fuse.
Power consumption: 150 Watts
<b>Operating Temperatures</b>
-10 degrees C to +40 degrees C, <90% humidity, no condensation
<b>CE CONFORMANCE INFORMATION:</b>
This device complies with the requirements of the EEC council directives:
• 93/68/EEC (CE MARKING)
• 73/23/EEC (SAFETY - LOW VOLTAGE DIRECTIVE)
• 89/336/EEC (ELECTROMAGNETIC COMPATIBILITY)
Conformity is declared to those standards: EN50081-1, EN50082-1.

# Quasar I/O Specifications (Axia xNodes)

## Microphone Preamplifiers

Source Impedance: 150 Ohms

Input Impedance: 4 k Ohms minimum, balanced

Nominal Level Range: Adjustable, -75dBu to -20dBu

Input Headroom: >20 dB above nominal input

Output Level: +4dBu, nominal

## Analog Line Inputs

Input Impedance: >40 k Ohms, balanced

Nominal Level Range: Selectable, +4dBu or -10dBv

Input Headroom: 20 dB above nominal input

## Analog Line Outputs

Output Source Impedance: <50 Ohms balanced

Output Load Impedance: 600 Ohms, minimum

Nominal Output Level: +4dBu

Maximum Output Level: +24dBu

## Digital Audio Inputs and Outputs

Reference Level: +4 dBu (-20 dB FSD)

Impedance: 110 Ohms, balanced (XLR)

Signal Format: AES-3 (AES/EBU)

AES-3 Input Compliance: 24-bit with selectable sample rate conversion,

32 kHz to 96kHz input sample rate capable.

AES-3 Output Compliance: 24-bit

Digital Reference: Internal (network timebase) or external reference 48 kHz, +/-2ppm

Internal Sampling Rate: 48 kHz

Output Sample Rate: 44.1 kHz or 48 kHz

A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling

D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling

Latency <3 ms, mic in to monitor out, including network and processor loop

## Frequency Response

Any input to any output: +0.5 / -0.5 dB, 20 Hz to 20 kHz

## Dynamic Range

Analog Input to Analog Output: 102 dB referenced to 0dBFS,

105 dB "A" weighted to 0dBFS

Analog Input to Digital Output: 105 dB referenced to 0dBFS

Digital Input to Analog Output: 103 dB referenced to 0dBFS, 106 dB "A" weighted

Digital Input to Digital Output: 138 dB

## Equivalent Input Noise

Microphone Preamp: -128dBu, 150 ohm source, reference -50dBu input level

Total Harmonic Distortion + Noise

Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38dBu input, +18dBu output

Analog Input to Analog Output: <0.008%, 1 kHz, +18dBu input, +18dBu output

Digital Input to Digital Output: <0.0003%, 1 kHz, -20dBFS

Digital Input to Analog Output: <0.005%, 1 kHz, -6dBFS input, +18dBu output

## Crosstalk Isolation, Stereo Separation and CMRR

Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kHz

Microphone channel to channel isolation: 80 dB isolation minimum, 20 Hz to 20 kHz

Analog Line Stereo separation: 85 dB isolation minimum, 20Hz to 20 kHz

Analog Line Input CMRR: >60 dB, 20 Hz to 20 kHz

Microphone Input CMRR: >55 dB, 20 Hz to 20 kHz

# Appendix C

## CE Declaration of Conformity – EU



### EU DECLARATION OF CONFORMITY

**Declaration:** The listed product is in conformity with following Union harmonization legislation:

**Directive 2014/30/EU** of the European Parliament and of the Council of 26 February 2014 on the harmonization of the laws of the Member States relating to electromagnetic compatibility (recast).

The Technical Documentation demonstrates the fulfilment of the essential requirements as set out in Annex I of Directive 2014/30/EU

**Directive 2011/65/EU + (EU) 2015/863** of the European Parliament and of the Council of 8 June 2011 on the restriction of the use of certain hazardous substances in electrical and electronic equipment.

**Directive 2014/35/EU** of the European Parliament and of the Council of 26 February 2014 on the harmonization of the laws of the Member States relating to the making available on the market of electrical equipment designed for use within certain voltage limits (recast).

**Manufacturer:** Telos Alliance  
1241 Superior Avenue  
Cleveland, OH 44114  
USA

This declaration of conformity is issued under the sole responsibility of the manufacturer.

<b>Product Name :</b>	Quasar MTS-MON Module	2001-00568 : Standalone 2001-00568-999
	Quasar SR-4FAD Module, Non Motorized	2001-00571 : Standalone 2001-00571-999
	Quasar XR-4FAD Module, Motorized	2001-00569 : Standalone 2001-00569-999
	Quasar Mic in/HP out Module	2001-00579
	Quasar table-top frame 2.5U thru 8.5U	2001-00550 thru 2001-00556
	Quasar flush-mount frame 2.5U thru 8.5U	2001-00557 thru 2001-00563
	QUASAR SK-6BTN MODULE	2001-00612
	QUASAR SK-12BTN MODULE	2001-00613
	QUASAR SK-18BTN MODULE	2001-00614
	QUASAR SK-24BTN MODULE	2001-00615

Referenced harmonized standards to which conformity to 2014/30/EU is declared:

EN 55032:2015 Electromagnetic compatibility of multimedia equipment - Emission requirements (CISPR 32:2012(EQV)).

EN 55035:2017 Electromagnetic compatibility of multimedia equipment – Immunity requirements

Referenced harmonized standard to which conformity to 2011/65/EU + (EU) 2015/863 is declared:

EN 63000:2018 Technical documentation for the assessment of electrical and electronic products with respect to the restriction of hazardous substances.

Referenced harmonized standard to which conformity to 2014/35/EU is declared:

EN 62368-1:2020 Audio/video , information and communication technology equipment – Part 1: Safety requirements.

Signature of Manufacturer: \_\_\_\_\_

Date: 2/23/23

## EU DECLARATION OF CONFORMITY

**Declaration:** The listed product is in conformity with following Union harmonization legislation:

**Directive 2014/30/EU** of the European Parliament and of the Council of 26 February 2014 on the harmonization of the laws of the Member States relating to electromagnetic compatibility (recast).

The Technical Documentation demonstrates the fulfilment of the essential requirements as set out in Annex I of Directive 2014/30/EU

**Directive 2011/65/EU + (EU) 2015/863** of the European Parliament and of the Council of 8 June 2011 on the restriction of the use of certain hazardous substances in electrical and electronic equipment.

**Directive 2014/35/EU** of the European Parliament and of the Council of 26 February 2014 on the harmonization of the laws of the Member States relating to the making available on the market of electrical equipment designed for use within certain voltage limits (recast).

**Manufacturer:** Telos Alliance  
1241 Superior Avenue  
Cleveland, OH 44114  
USA

This declaration of conformity is issued under the sole responsibility of the manufacturer.

**Product Name:** 2001-00606 Infinity VIP Server Appliance  
2001-00602 Pathfinder Core Pro AE-2000  
2001-00605 IPORT High Density  
2001-00510 VX Prime +  
2001-00599 IPORT High Density + Content Delay  
2001-00583 Quasar Engine  
2001-00581 Quasar Engine RPS Base (16 channel)  
2011-00835 Quasar Engine RPS Upgrade Kit  
2001-00515 Pathfinder Core Pro MK2  
2001-00503 VX Enterprise  
2001-00437 Z/IPStream R/2

Referenced harmonized standards to which conformity to 2014/30/EU is declared:

EN 55032:2015 Electromagnetic compatibility of multimedia equipment - Emission requirements (CISPR 32:2012(EQV)).

EN 55035:2017 Electromagnetic compatibility of multimedia equipment – Immunity requirements

Referenced harmonized standard to which conformity to 2011/65/EU + (EU) 2015/863 is declared:

EN 63000:2018 Technical documentation for the assessment of electrical and electronic products with respect to the restriction of hazardous substances.

Referenced harmonized standard to which conformity to 2014/35/EU is declared:

EN 62368-1:2020 Audio/video , information and communication technology equipment – Part 1: Safety requirements.

Signed for and on behalf of Telos Alliance, 1241 Superior Avenue, Cleveland, Ohio 44114 USA:

Signature of Manufacturer:



Date: FEB 6, 2024

### DECLARATION OF CONFORMITY

**MANUFACTURER:** Telos Alliance  
1241 Superior Avenue  
Cleveland, Ohio 44114 USA

**PRODUCT :**

Quasar MTS-MON Module	2001-00568
Quasar MTS-MON Module, Standalone	2001-00568-999
Quasar SR-4FAD Module, Non Motorized	2001-00571
Quasar SR-4FAD Module, Standalone	2001-00571-999
Quasar XR-4FAD Module, Motorized	2001-00569
Quasar XR-4FAD Module, Standalone	2001-00569-999
Quasar Mic in/HP out Module	2001-0000579
Quasar table-top frame 2.5U thru 8.5U	2001-00550 thru 2001-00556
Quasar flush-mount frame 2.5U thru 8.5U	2001-00557 thru 2001-00563
QUASAR SK-6BTN MODULE	2001-00612
QUASAR SK-12BTN MODULE	2001-00613
QUASAR SK-18BTN MODULE	2001-00614
QUASAR SK-24BTN MODULE	2001-00615

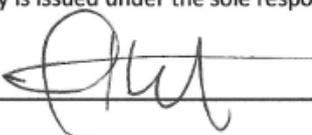
**REPORTS:** F2P23529A-01E, F2P23529A-01S, F2P26704-0E, F2P26704-01S

Conforms to the following standards:

**UK LEGISLATION:** Electromagnetic Compatibility Regulations 2016  
Electrical Equipment (Safety) Regulations 2016  
The Restriction of the Use of Certain Hazardous Substances in Electrical and Electronic Equipment Regulations 2012

**STANDARDS:** BS 55032:2015  
BS 55035:2017  
BS 62368-1:2020  
BS 63000:2018

This declaration of conformity is issued under the sole responsibility of the manufacturer.

Signature of manufacturer:  DATE 2/23/23

## DECLARATION OF CONFORMITY

**MANUFACTURER:** Telos Alliance  
1241 Superior Avenue  
Cleveland, Ohio 44114 USA

<b>PRODUCT / MODEL:</b>	Infinity VIP Server Appliance	2001-00606
	Pathfinder Core Pro AE-2000	2001-00602
	Pathfinder Core Pro MK2	2001-00515
	IPOINT High Density	2001-00605
	Quasar Engine	2001-00583
	Quasar Engine RPS Base (16 Channel)	2001-00581
	Quasar Engine RPS Upgrade Kit	2011-00835
	VX Prime+	2001-00510
	VX Enterprise	2001-00503
	Z/IPStream R/2	2001-00437
	IPOINT High Density+Content Delay	2001-00599

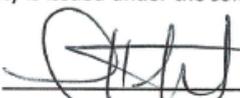
**REPORTS:** HA201216-GGC-012-E01, HA201216-GGC-012-S01

Conforms to the following standards:

**UK LEGISLATION:** Electromagnetic Compatibility Regulations 2016  
Electrical Equipment (Safety) Regulations 2016  
The Restriction of the Use of Certain Hazardous Substances in Electrical and Electronic Equipment Regulations 2012

**STANDARDS:** BS 55032:2015  
BS 55035:2017  
BS 62368-1:2020  
BS 63000:2018

This declaration of conformity is issued under the sole responsibility of the manufacturer.

**Signature of manufacturer:**  \_\_\_\_\_ **DATE** FEB 6, 2024