User Manual

Quasar User Manual v1.4.0.18 - by L. La Rosa
Console SW v1.4.0.24 – Engine SW v1.2.0.8
June 2021 – p/n 1490-00224-001

Telos Alliance®

Intentionally Left Blank

Notices and Cautions

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

Disclaimer

All versions, claims of compatibility, trademarks, etc. of hardware and software products not made by Axia Audio which are mentioned in this manual or accompanying material are informational only. Axia makes no endorsement of any particular product for any purpose, nor claims any responsibility for operation or accuracy. We reserve the right to make improvements or changes in the products described in this manual which may affect the product specifications, or to revise the manual without notice.

Trademarks

Axia Audio and the Axia and Quasar logos are trademarks of TLS Corp. All other trademarks are the property of their respective holders.

Warranty

This product is covered by a two-year limited warranty. For more details, please visit https://www.telosalliance.com/warranty-information

After-Sales Information

Should you require any technical information or assistance with your Axia product please contact your local Axia distributor. Customers within the United States could contact Axia directly. For a complete list of worldwide distributors by region, go to www.telosalliance.com or contact us for more information.Our Support Team works closely with our global distributor network to provide the highest level of after sales support. Your distributor should be your first point of contact and will often be able to provide an instant solution, be it technical advice, spares or a site visit by an engineer.

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

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 prented 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 Team, for their tireless work, incredible perseverance and commitment to this project. Many, many thanks to: Serhiy Borisov, 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.

Much Gratitude goes to our CTO Greg Shay. and to fellow managers John Granchi and Milos Nemcik for their precious consulting work, and for being always there when I needed their support.

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

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.
- Our Fax number is +1-216-241-4103.

By Email

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

Via World Wide Web:

The Axia Audio web site has a variety of information which may be useful for product selection and support. The url is **telosalliance.com.**

Register Your Product

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

AXIA AUDIO

1241 Superior Avenue E.
Cleveland, OH., 44114 USA

+1-216-241-7225 (phone)
+1-216-241-4103 (fax)
+1-216-622-0247 (24/7 technical support)

support@telosalliance.com inquiry@telosalliance.com

Intentionally Left Blank

Contents

Intentionally Left Blank

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.

We're grateful that you have chosen audio tools from Telos® Systems, Omnia® Audio, Axia® Audio, Linear Acoustic®, 25-Seven Systems®, and Minnetonka Audio®. We're here to help you make your work truly shine. We hope that you enjoy your Telos Alliance product for many years to come and won't hesitate to let us know if we can help in any way.

The Telos Alliance

Intentionally Left Blank

Introduction

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:

- Operation
- Configuration
- Engine
- 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 encourage the operators to not only dive in the first chapters, dedicated to the Console operation, but also take a look at the configuration section, to get a broader view of the console workflow, and a deeper understanding of the entire 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 Team

Startup

Quick Start Guide

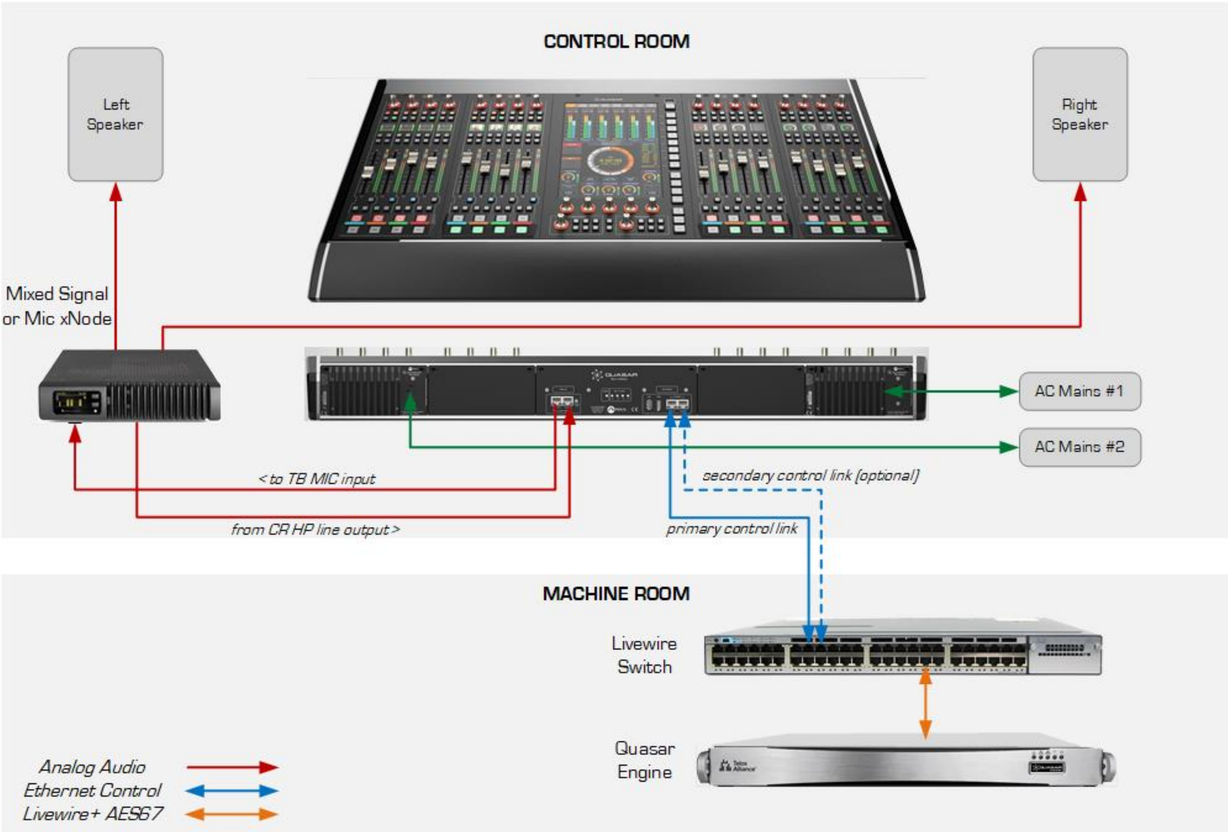
Welcome To Quasar

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 Quickstart 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:

Initial Setup

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

Console Configuration

1. Quasar Engine Audio Outputs Configuration
2. Quasar 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

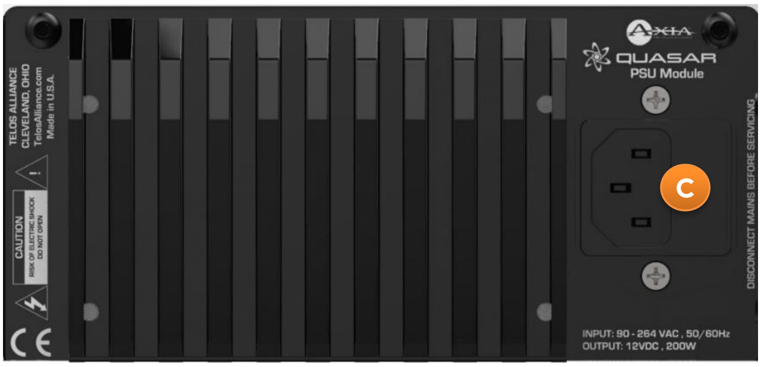
Network Connections

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:

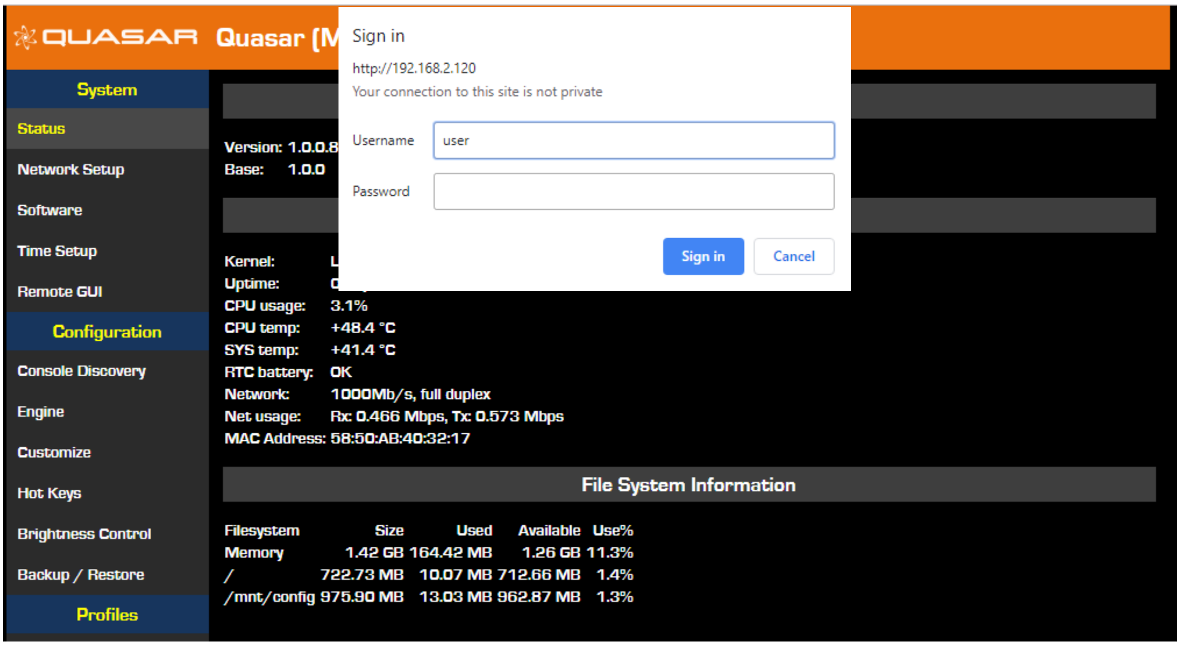
- Surface (Master) - 192.168.2.10
- Surface (Modules) - 192.168.2.11, 192.168.2.12, etc.
- Engine - 192.168.2.100

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.

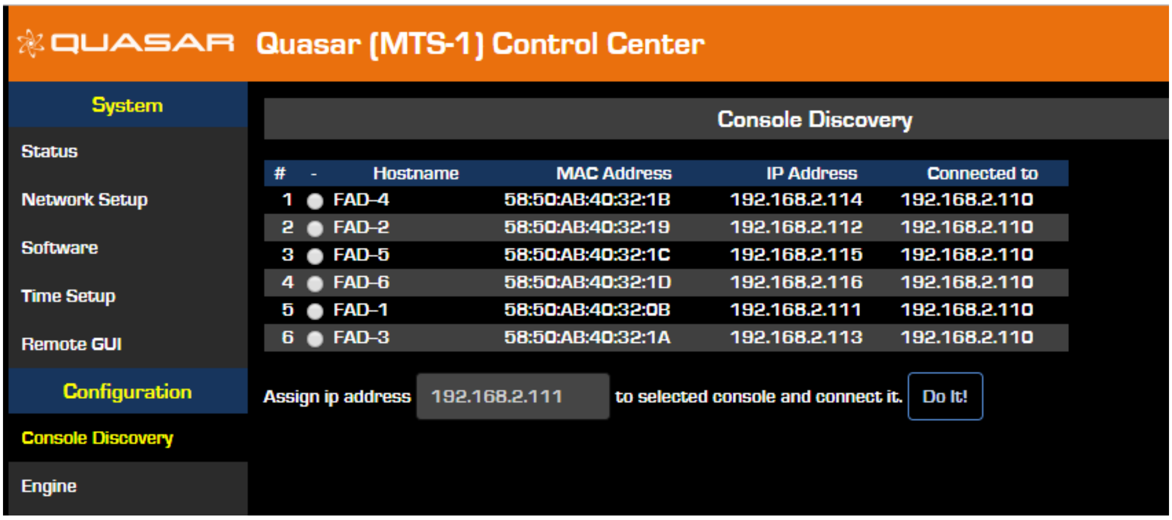
- Access the System Setup page on the Master Touchscreen Module, by pushing and holding the "Monitor Options" key (labeled "MON OPT" and located between the two Control Room volume pots) for 5 seconds.
- Enter the desired IP address and Netmask for the Surface and Engine using the touchscreen or the 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.

Quasar Console Discovery & Configuration

Each module on the Quasar Surface has a unique IP address to allow configuration via web GUI. Begin by launching a web browser and entering the IP address of the MTS Module, then click on the **Console Discovery** menu. When prompted for authentication, enter "user" for the user name 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 by 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 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.

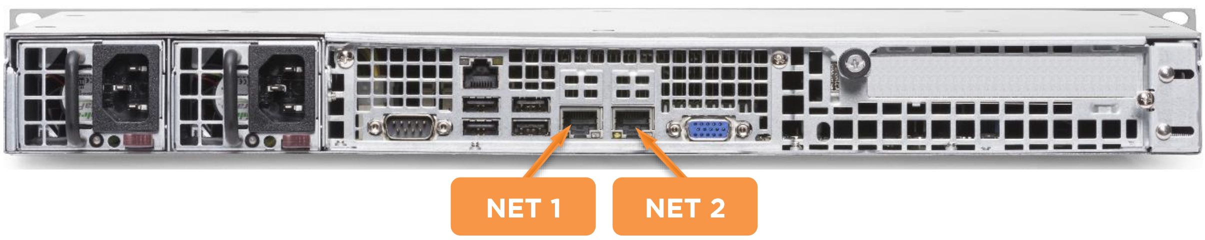
Mains Power Connections

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.



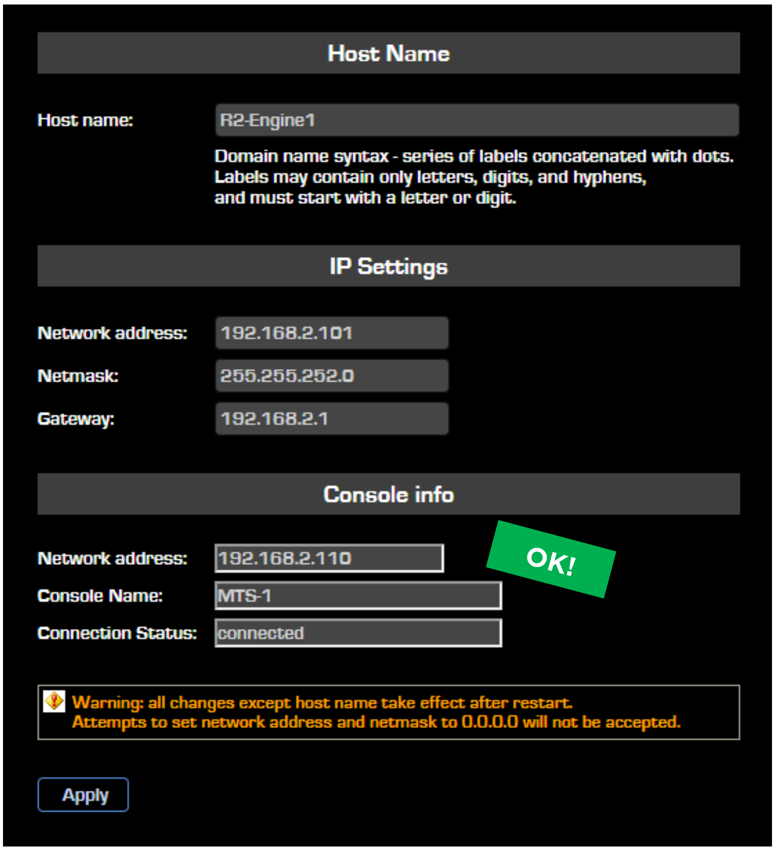
Quasar Engine ships with a default IP address of 192.168.2.100 and a netmask of 255.255.255.0. This address can be changed using the web GUI once it is connected to the network. If the IP address must be changed before connecting the Engine to the network, follow the steps below:

- 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.

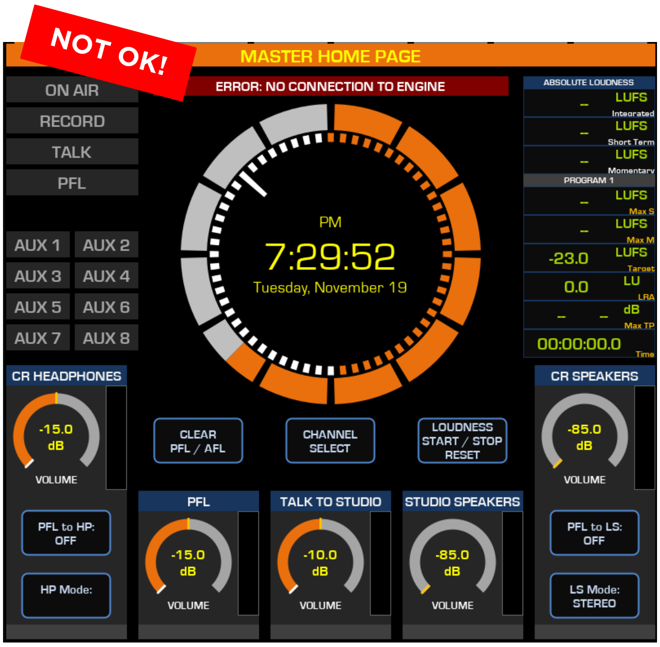
Checking the Connection

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 the user name “user” and leave the password field blank.

Navigate to the **Network** 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.

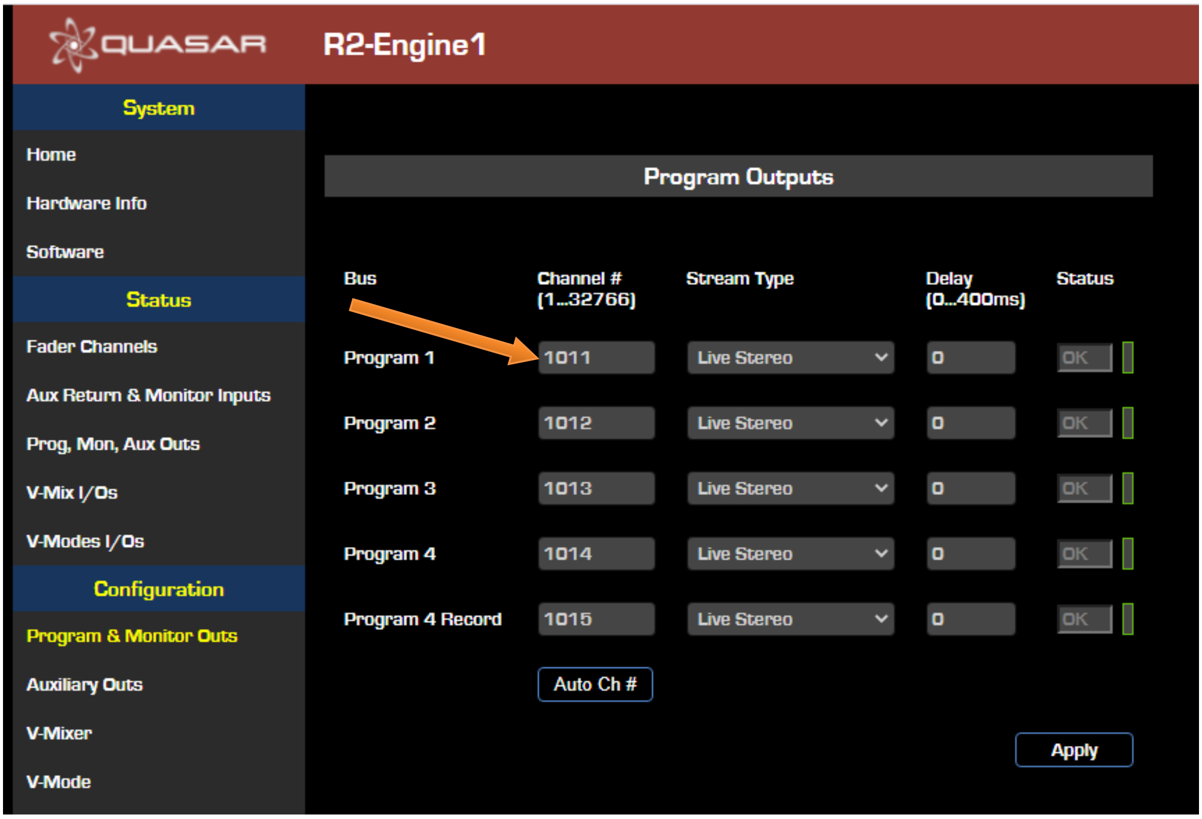


Quasar Engine Audio Outputs Configuration

Quasar Engine audio setup begins with configuring the audio outputs.

Program Outputs

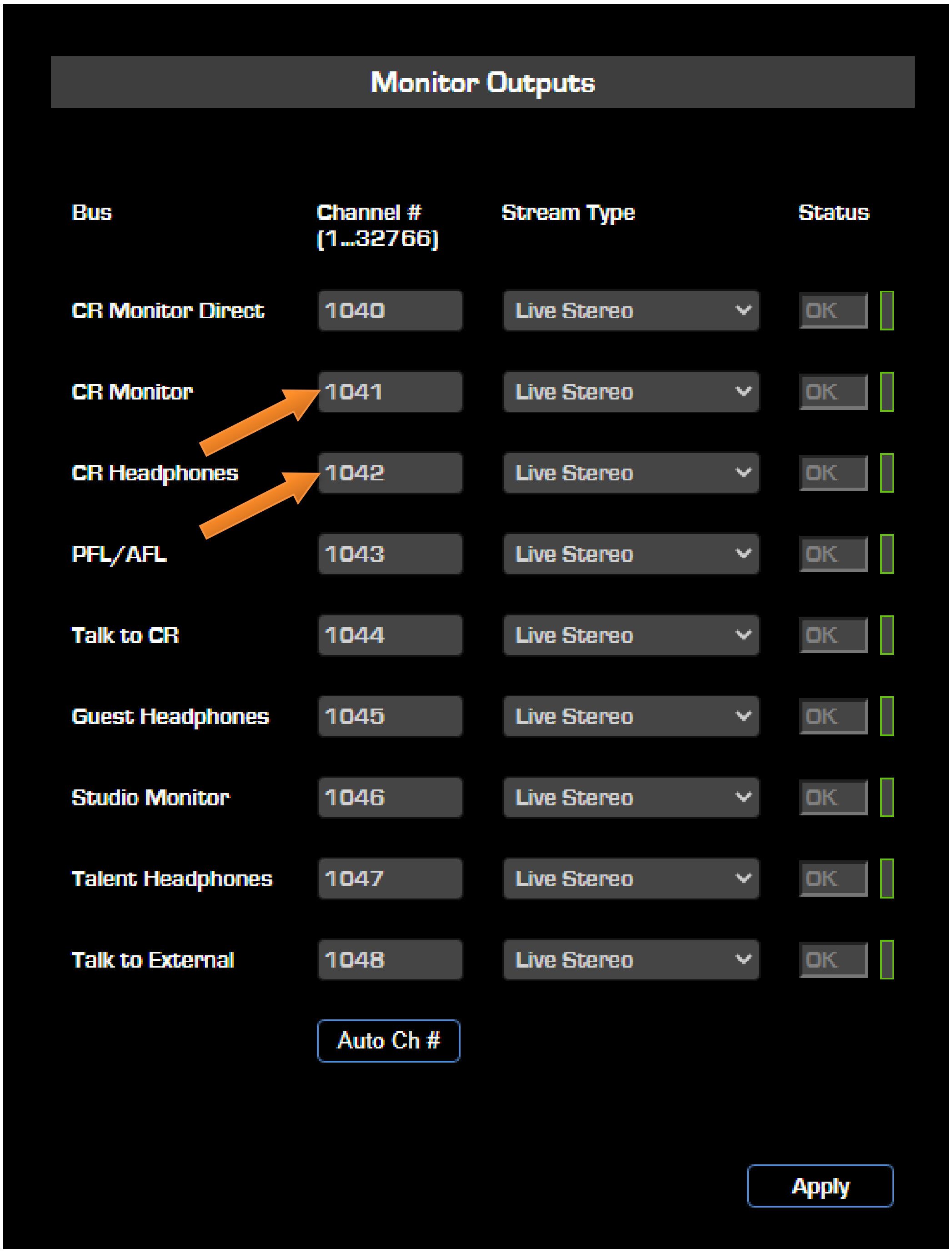
- Using a PC connected to your studio network, enter the IP address of the Quasar
- Engine to display the web GUI.
- From the Configuration menu, select **Program & Monitor Outs**.
- Enter the channel numbers you planned for the Quasar Engine (Livewire Sources)
- outputs.
- Enable the streams you need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



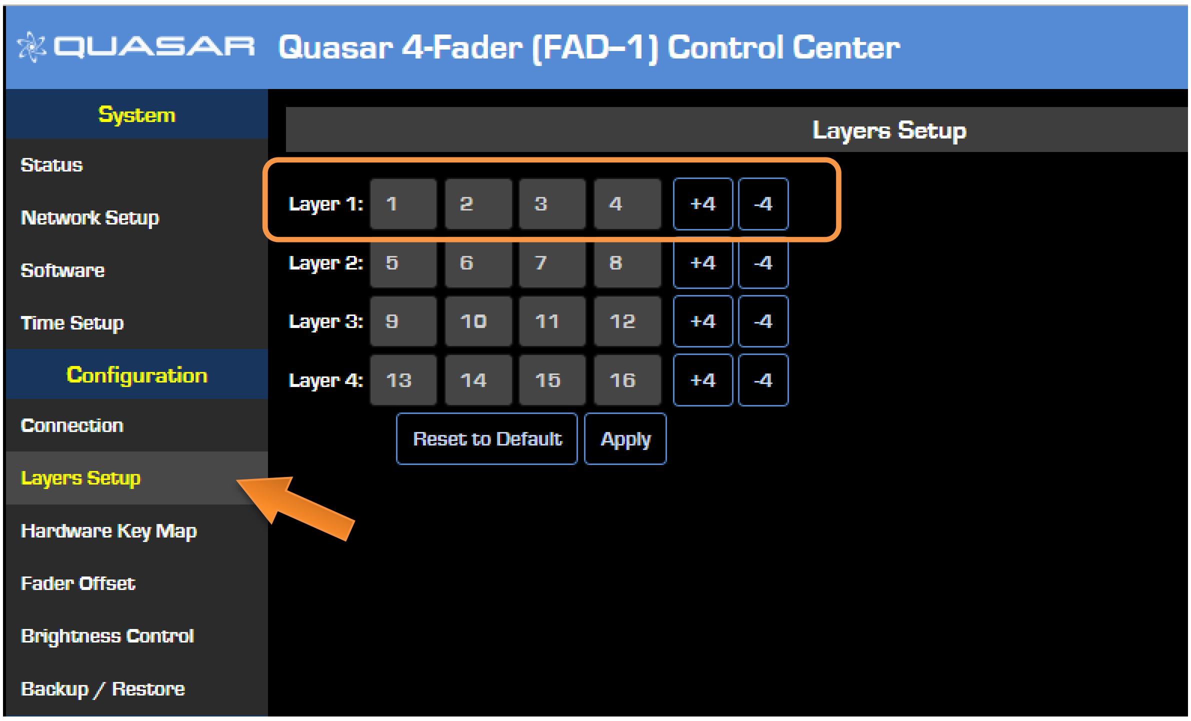
Quasar Surface Layer Configuration

Understanding Layers

Layers are useful when you need to access more input channels from the Engine than you have available faders (physical channel strips) on your Surface. Since any fader on any module could access any input channel on the Quasar Engine, you will need to assign four input channels to each of the 4 layers available, on every Fader Module.

Configuring Layers

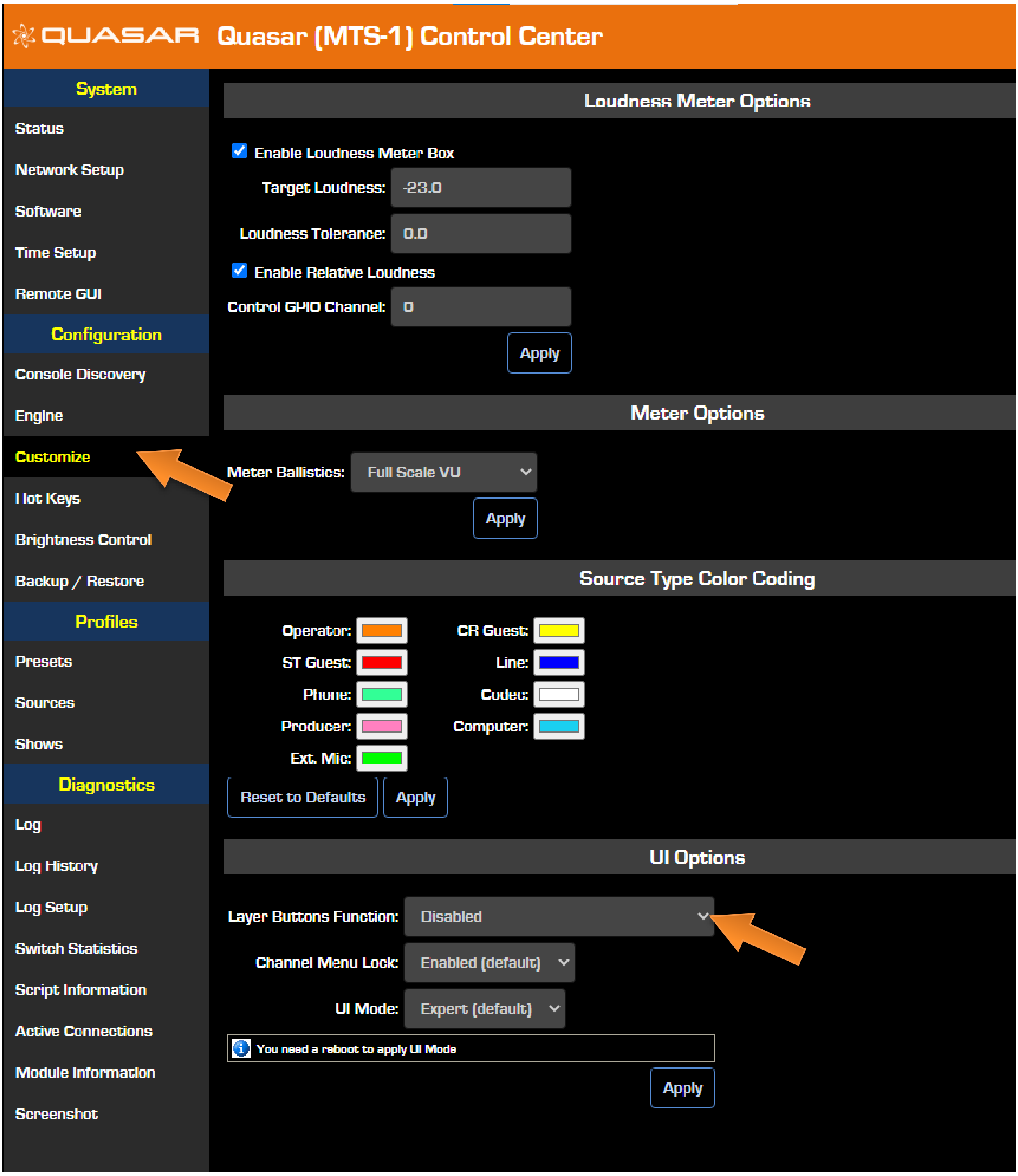
- Using a PC connected to your studio network, launch a web browser and enter the IP address assigned to the first (left-most) fader module. When prompted for authentication, enter the user name "user" and leave the password field blank.
- In the Configuration menu, click on **Layers Setup**.
  Enter 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.



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 will be pushed on the Master Module.

Disabling Layers

To disable layers, log into the Master Module, navigate its Web UI to the Configuration menu, then to the **Customize** menu, and turn them off in the "Layer Button Function" drop-down menu.



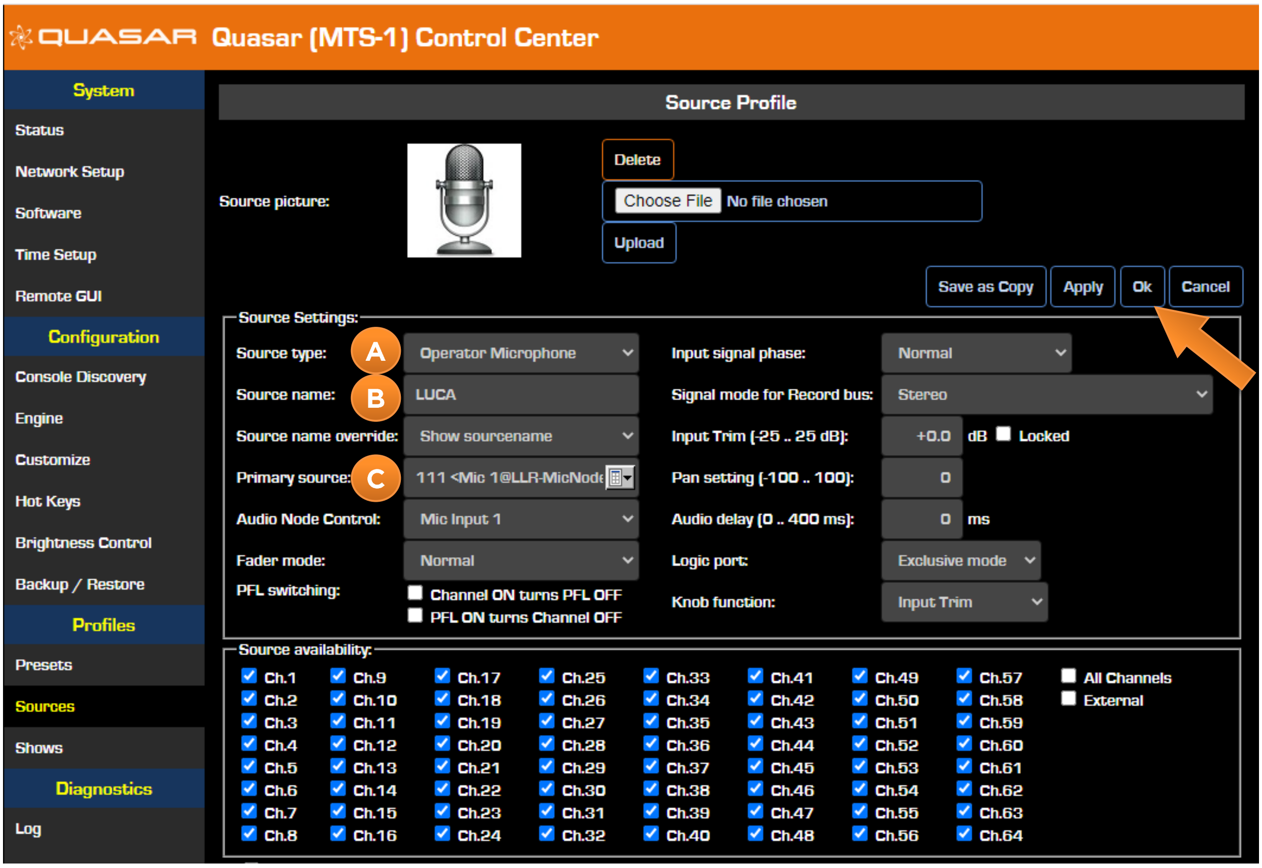
Creating and Configuring Input Sources

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.



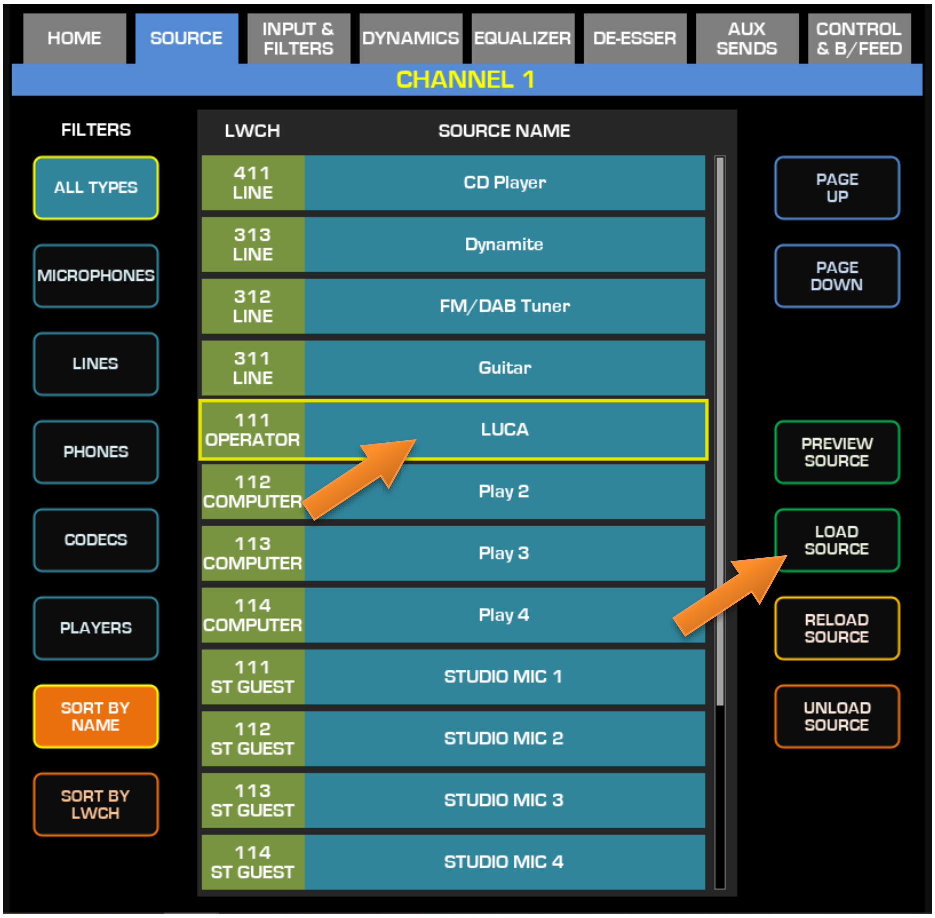
Assigning Sources to Input Channels

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.



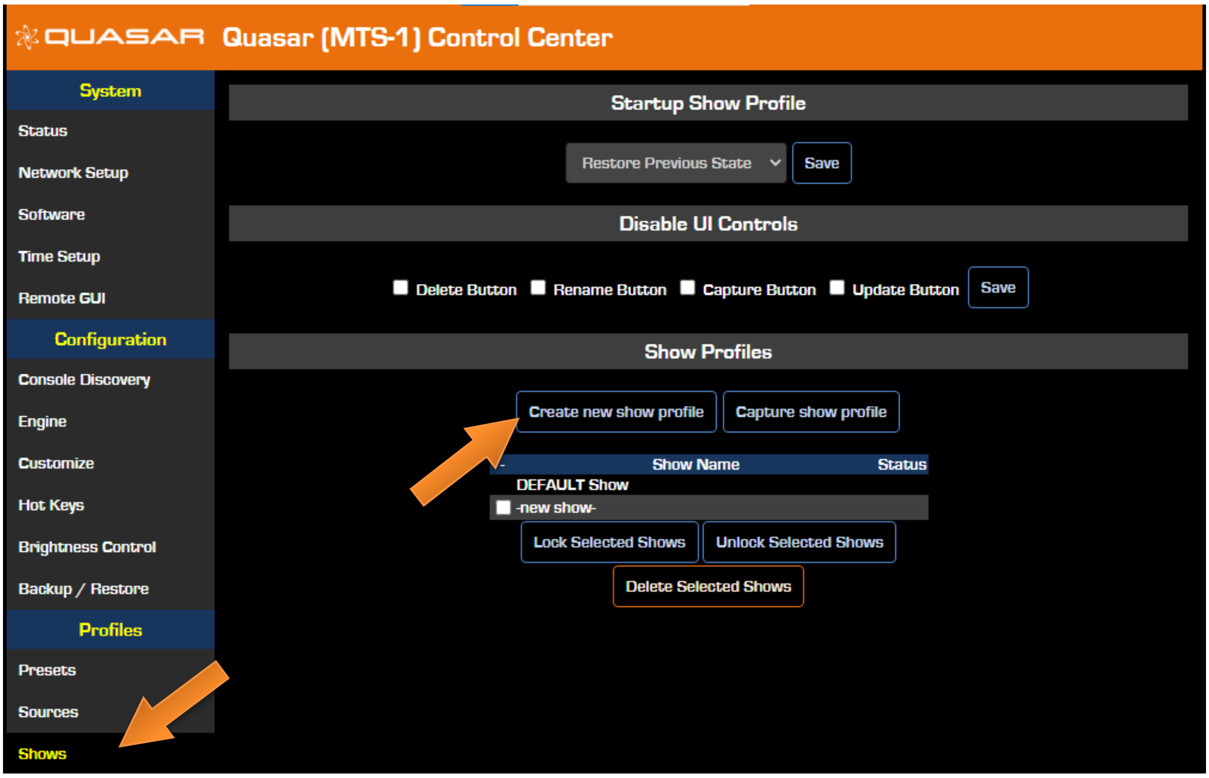
Note – Inactive sources (that is, those generated by devices that are disconnected from the network or turned off) will not be detected by Quasar and will therefore not appear in the list.

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**.

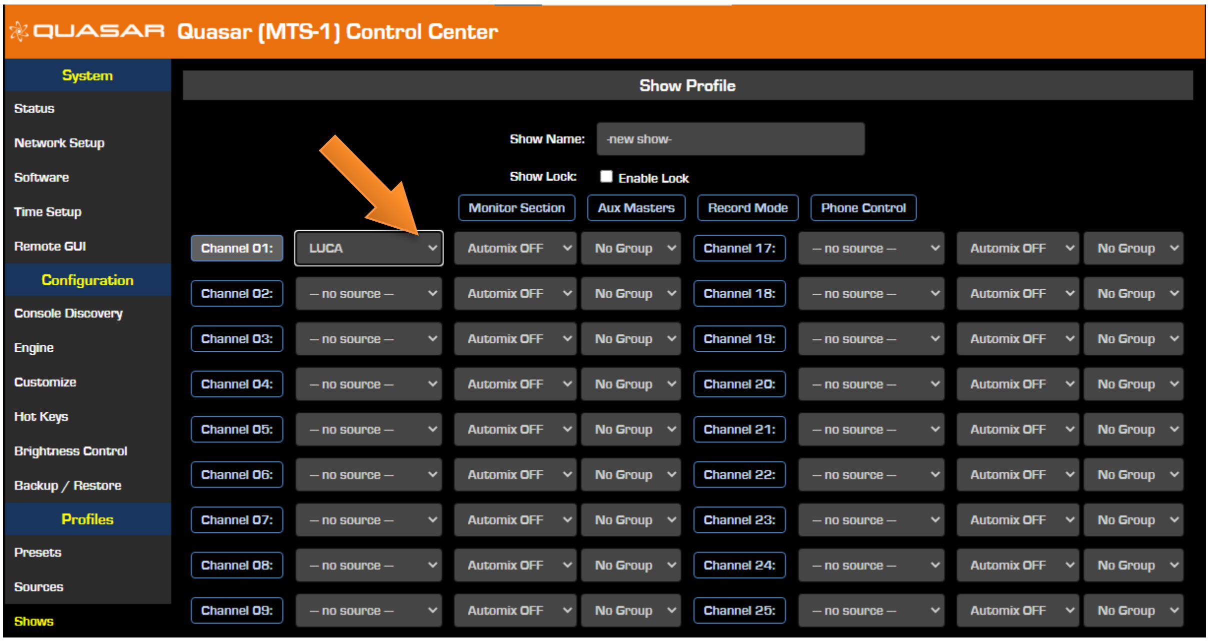
Creating a Show Profile From The Web UI

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 the **Shows** menu.
- 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.

Intentionally Left Blank

Operation - 1

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. 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 pushbuttons, 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 soft graphical button.
- **Page**: Web UI page, link to pages or screens
- "Label" : Static text window or displayed in the Touchscreen, or 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.

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 (1) view 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 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.

   Alternatively, if Layers and Banks are not needed, the Layer keys can be completely disabled in the MTS Module Web UI, (by browsing the Web UI *Customize* page).

8. **Touch sense User 1-8 keys:** Eight user-programmable keys with capacitive touch sense are available for 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.

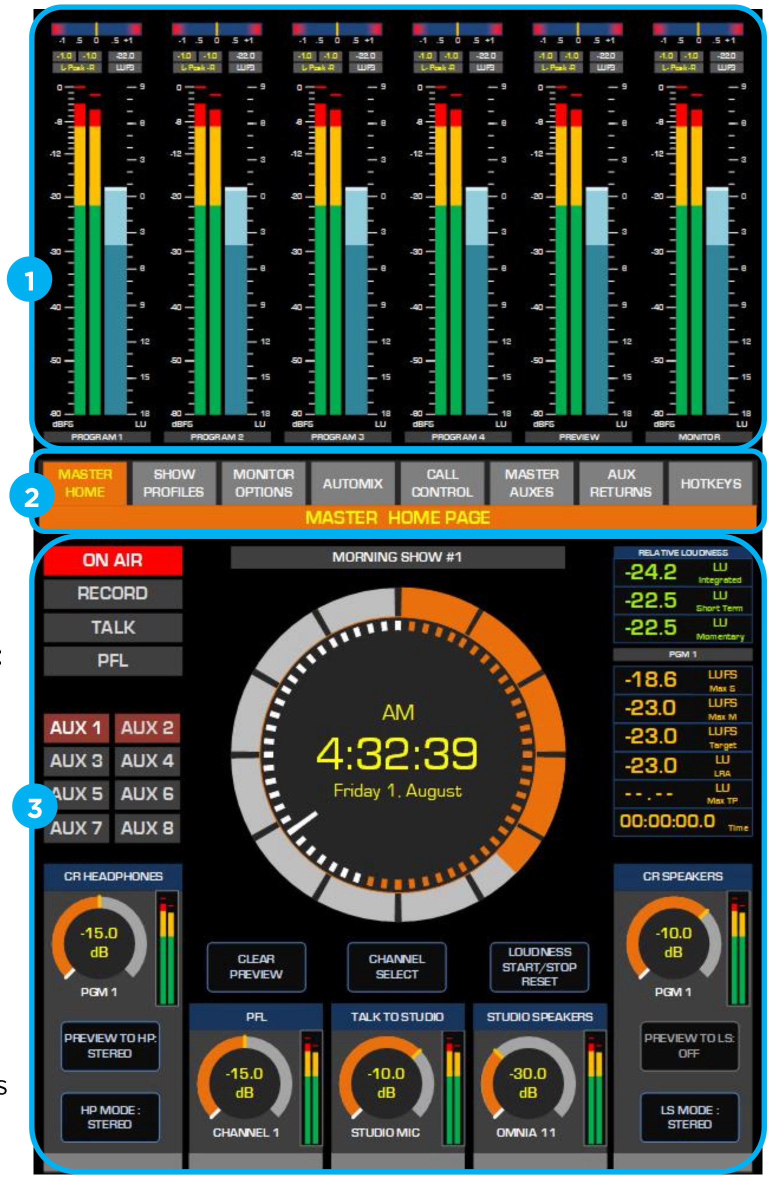
9. **Profile key:** A fixed-function key which shows the defined show profiles onto the display for user selection. Currently, a press & hold of the key will load the *last loaded profile* (back to its default settings).

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

The Touchscreen User Interface – Expert 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.

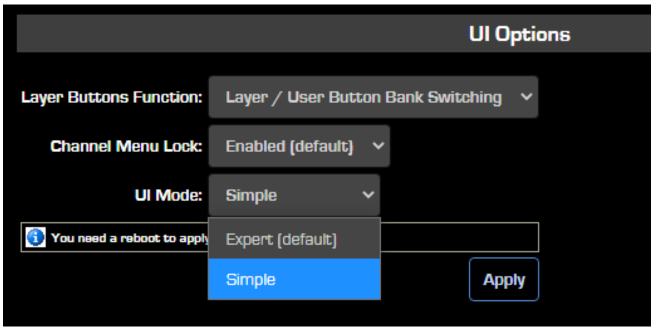
   Master tabs (orange): all tabs in this menu provide access to controls typically found in any console's Master Module. These controls define functions that belong to the entire desk.

   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 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.

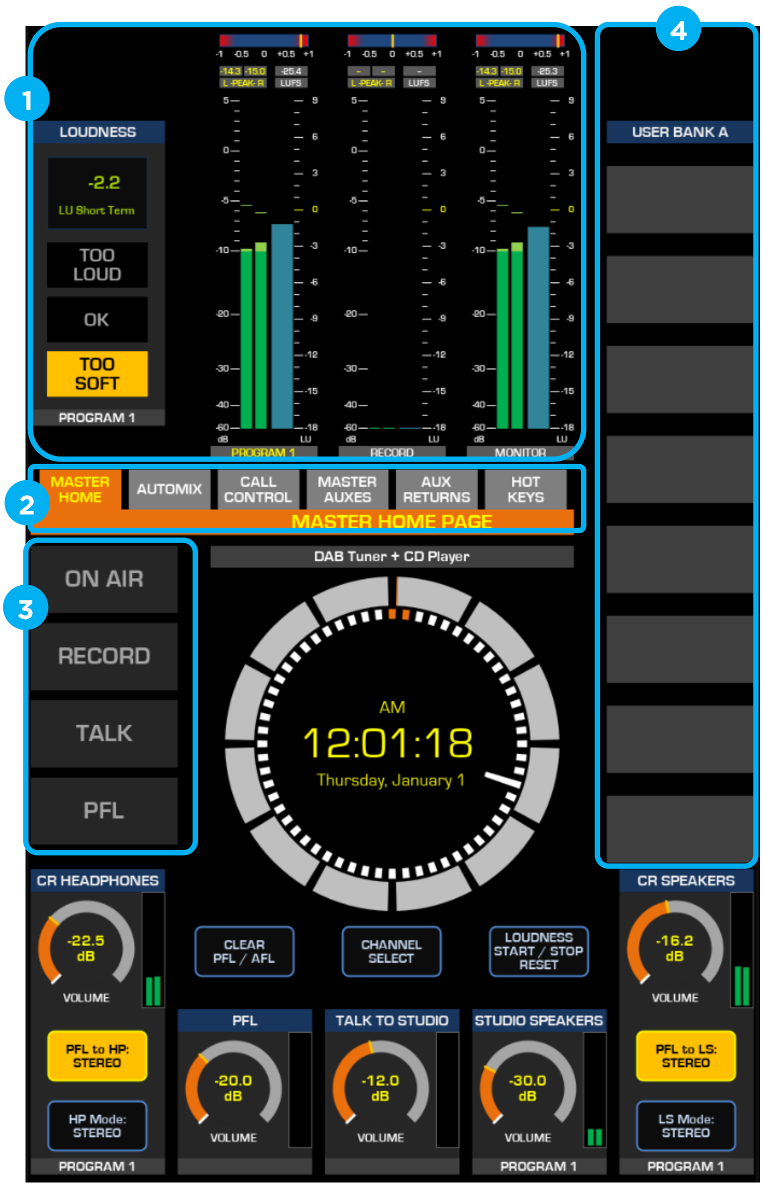
To select this option, navigate to the Master module's Web UI, and in the Customize > UI Options menu, select **UI Mode:** Simple.



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

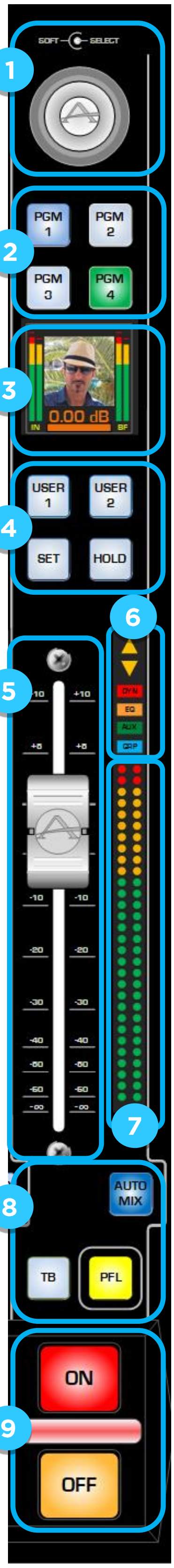
The basic principles explained in the previous page apply to the Simple UI as well. The graphics have the following differences:

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 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.

   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.
2. **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.
3. **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.
4. **Channel Fader**: This high-quality spill-proof motorized fader controls the gain of the input channel.

   LED feedback: The arrow indicators show the fader position in relation to the logical level in the Engine DSP. 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).
5. **Channel Bargraph**: This LED bar normally works as stereo level meter for the channel input (prefader). When Automixer is active on the channel, the mono sum is displayed onto the left bar while the right bar shows the AM gain reduction.
6. **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 where a press & hold will be momentary for the period of the press.
   - 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.
7. **ON/OFF keys:** Enable the channel audio (post-fader) to the selected busses.

   A color-coding LED bar indicator is integrated between the two.

Operation - 2

The Touchscreen UI

The Console Touchscreen UI provides the user with a graphical and tactile interface, in order 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 present on the surface, and enriches the User Experience with interactive touchscreen controls, real-time visual feedback, and context-sensitive menus.

Special consideration was given to the application's ergonomics, with the main concept being to provide visual grouping of control elements, visual styles, behavioral standards of the various objects (menus, pushbuttons, labels, etc.) with minimal effort required in terms of user actions.

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.

General GUI Widgets' 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 easier reach and to divide 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 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)

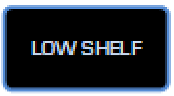
Button Inactive (OFF state)

OPTION SELECT buttons

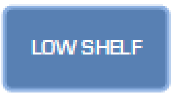
Action: Momentary (typical), Latching , press & hold.

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:

User Menu

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

Non-Destructive action

Potentially Destructive action

Destructive action

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

Labels

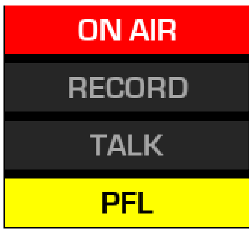
Text Labels (i.e."**COMPRESSOR RATIO**") are simple text fields. They are always rectangular, with no rounding on corners in order to differentiate them from buttons, dark (blue or grey) color and white text.
There are two different types of labels, system-wide:

- 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.

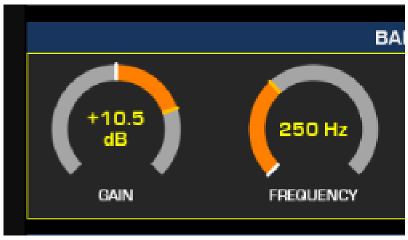
Gauges

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 user the logical position of an encoder, 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 trimpot 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 dscribing 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 when the GUI is accessed from remote. The fader widget is designed to be exactly the same size of a physical fader: 100mm long.

Quasar Master Menu (orange tabs)

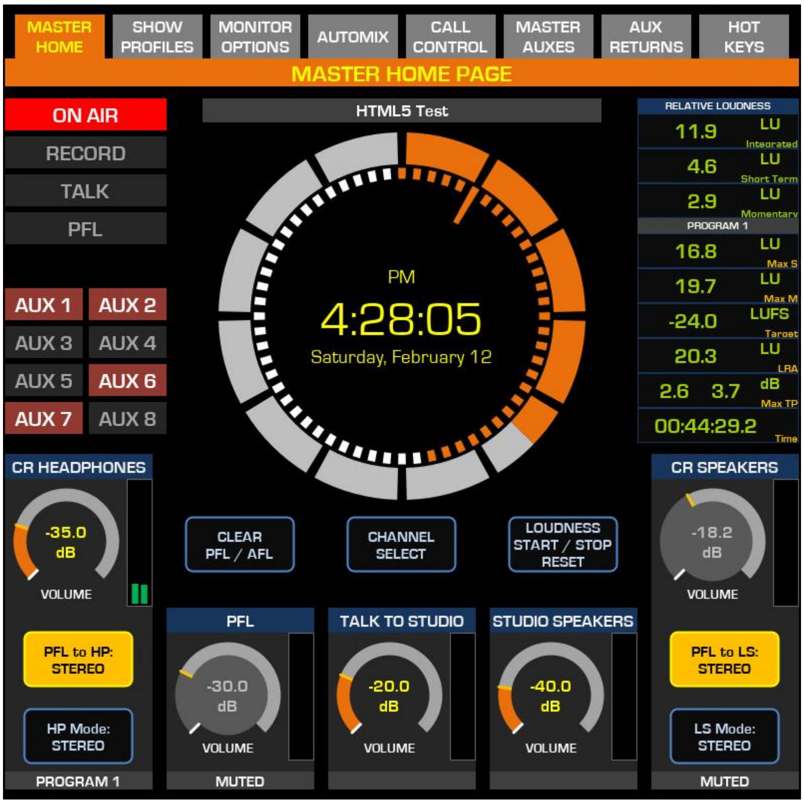
MASTER HOME

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.

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.

- **CLEAR PFL:** Lights up when any channel is assigned to the PFL bus. When pressed it will clear the selection on all channels assigned to the bus. A quick fix de-select button.
- **CHAN SELECT:** Displays an option screen to select any channel from the touchscreen, as an alternative to pushing the encoder located at the top of each fader strip. This function can also be obtained by pressing and holding the HOME button for 1 second.
- **LOUDNESS START/STOP /RESET:** Controls the Loudness metering counters.

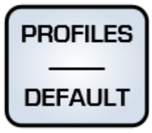


Next to these buttons are the two rotary indicators for the CR HEADPHONES and CR SPEAKERS volume controls. Below, there are some controls used for configuring the Monitoring section:

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

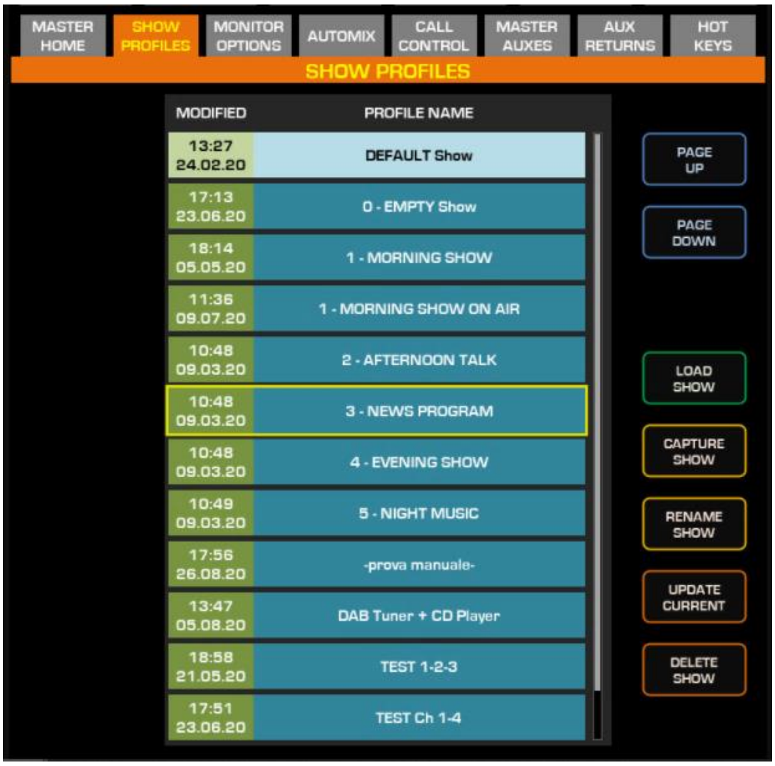
SHOW PROFILES

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.

> **Tip -** If you think that the DELETE SHOW button is too dangerous, and fellow operators migh 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 <u>at the time of loading</u> the Show.
- If, after loading a Show, a <u>new</u> 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

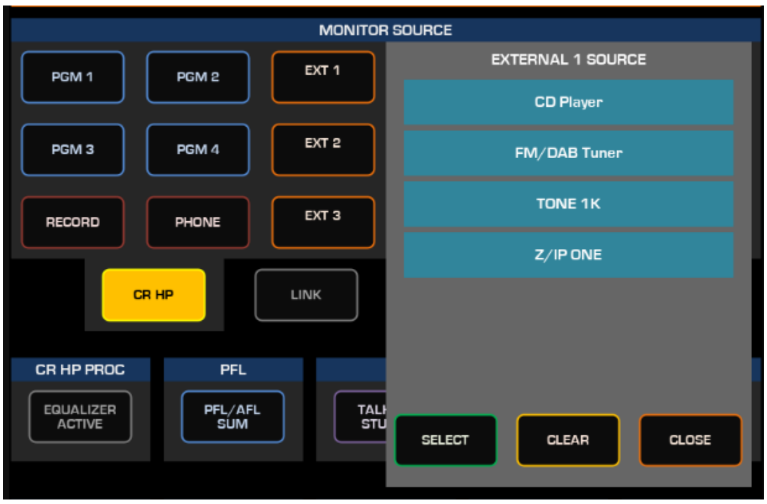
As detailed earlier, there are some dedicated hardware keys for monitor control which provide the most common sources. If the operator needs to monitor other mixes, adjust the dimming levels, or adjust the TALK gains, this is the location for those controls. The gain indicators at the bottom are tied 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 section provides source selection for the three Monitor Sections:

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

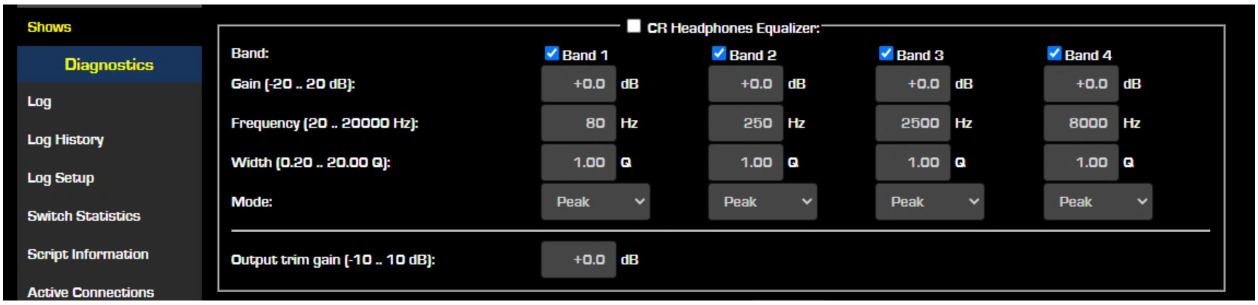
For each one of these sections, selectable with the corresponding buttons, it is possible to select different sources to listen, such as Program 1 to 4 bus, Record and Phone bus, Auxiliary 1 to 8 bus, and three EXTERNAL monitor inputs available on the Engine.

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 availble 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 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 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 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 to adjust the global parameters of the entire Automixer, such as Attack and Release times and the Pre/Post fader position of the automix gain control stage.

CALL CONTROL

For use in multiline workflows, such as those available with the Telos VX system, the CALL CONTROL tab will provide access to 8 phone lines and the ability to assign those to two different channel strips.

The view is organized with two columns for assignment 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.

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

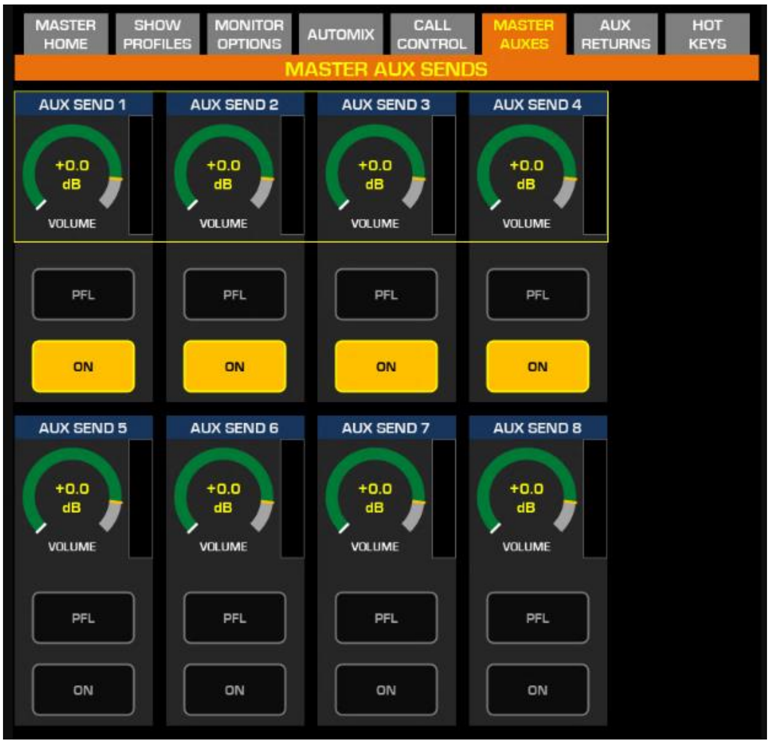
MASTER AUXES

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 AUX1-4 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** key 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** key.

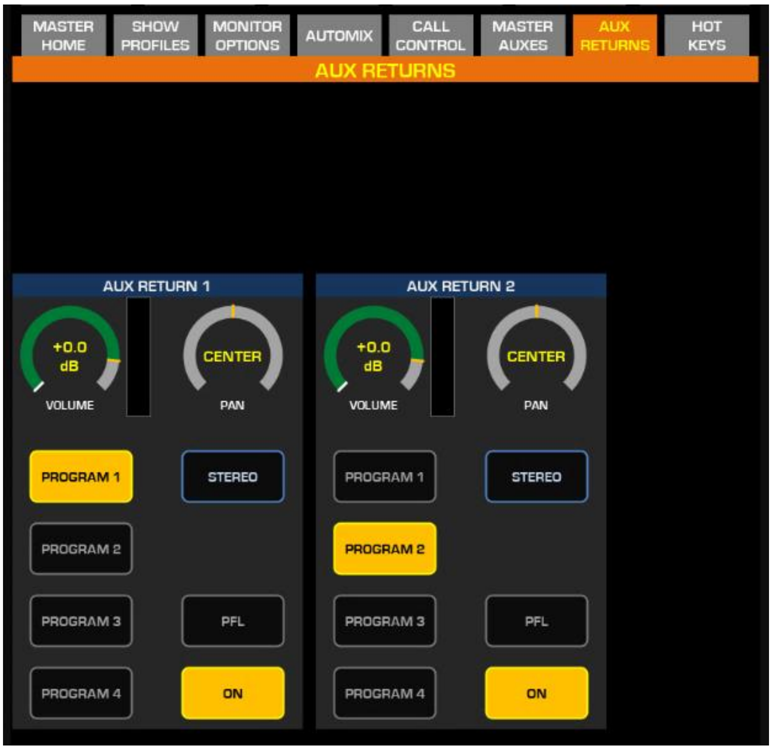


AUX RETURNS

The system includes two AUX return inputs.
These are basically two "short" input channels, (without DSP processing) typically used to bring 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.



HOT KEYS

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

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.

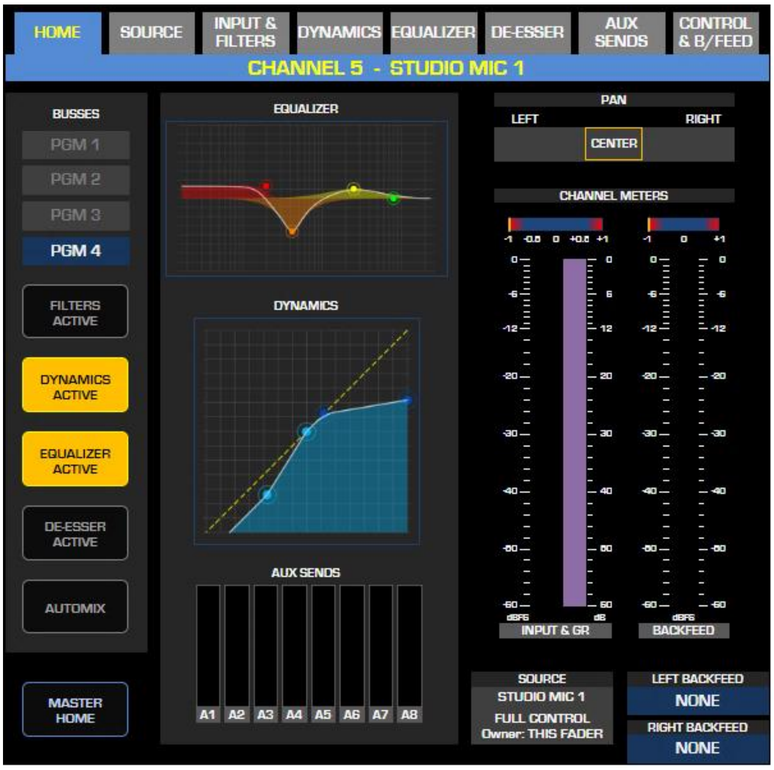
For detailed information about configuring the Hot Keys, please refer to Configuration Chapter 1 of this Manual: Hot Keys.

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 channel is selected from the Master module's Touchscreen UI.

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 mix assignments, EQ curve, Dynamics, Backfeed source, Panpot, and audio meters. There are buttons which allow to enable or disable the various audio processing stages and the automixer. The Channel meters provide at-a-glance monitoring of the level before and after processing.



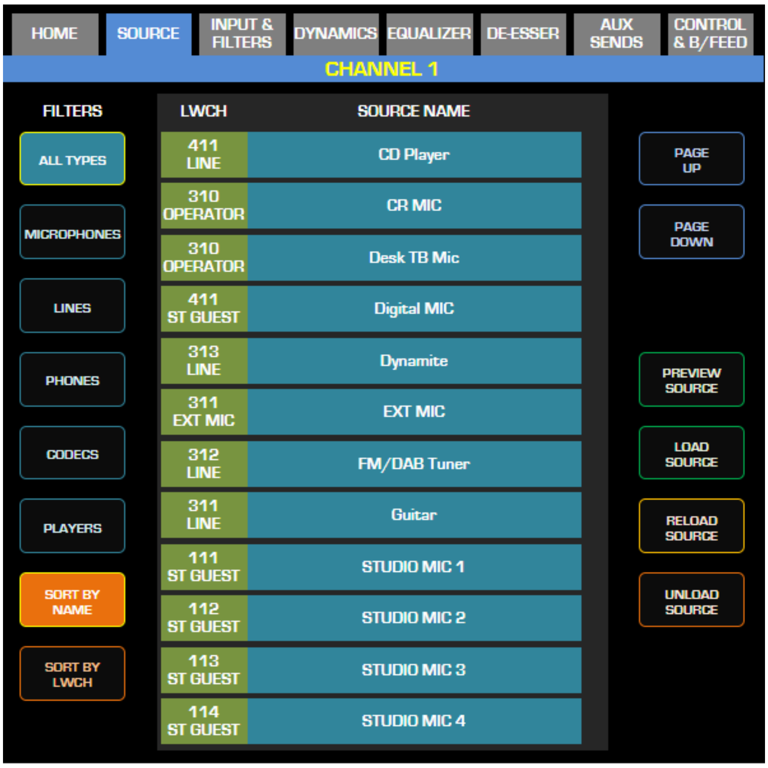
Note – The position of the FILTERS, DYNAMICS, EQUALIZER and DE-ESSER tabs in the channel UI, follows the chaining order of the DSP processing blocks within the channel.

SOURCE

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

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

The right hand column are 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 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 BACKFEED section provides control over the **Dim Gain** of the channel's backfeed. This determines by how much the main audio will be dimmed when the Talkback is engaged.

The OUTPUT section provides:

- **Output meter:** monitors the level sent to the mix bus (after processing, and before the 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.

DYNAMICS

New and improved algorithms in the Quasar Engine offer more accurate control over the audio. 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 its maximum level (>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, so it can be used to manually match the compressed signal level to the original input level. This is designed to finely adjust 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.

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.

EQUALIZER

The Quasar EQ was modeled after many hours of listening to the best analog EQs available on the market, including some very cool vintage gear.

The four 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.

DE-ESSER

New and improved De-Esser algorithm in the Quasar introduces more control over the challenges audio engineers face with sibilance. 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.

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

For detailed information about configuring the De-Esser, please refer to the next Chapter.

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.

CONTROL & B/FEED

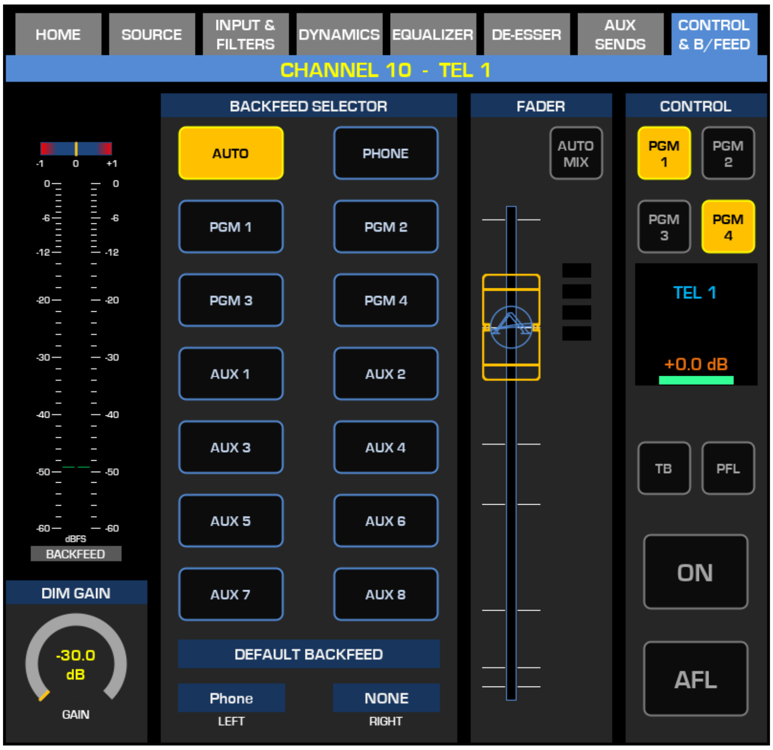
Backfeed is our definition for return streams. Some source profiles require a backfeed to be generated and this backfeed can be either a mix-minus bus or a headphone feed.
In either case, the backfeed will be a copy of one of various mixes of the console.

In the case of mix-minus, the backfeed would be the final mix minus the source (eg Program 1 minus the codec).

The 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 <u>Default</u> 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.
- **Headphone Control**: is displayed only in case the Source type is a Studio Microphone, and and the Backfeed is set as <u>Default</u> type. Provides level control for the HP Feed to the Guest Microphone (Mic Backfeed).

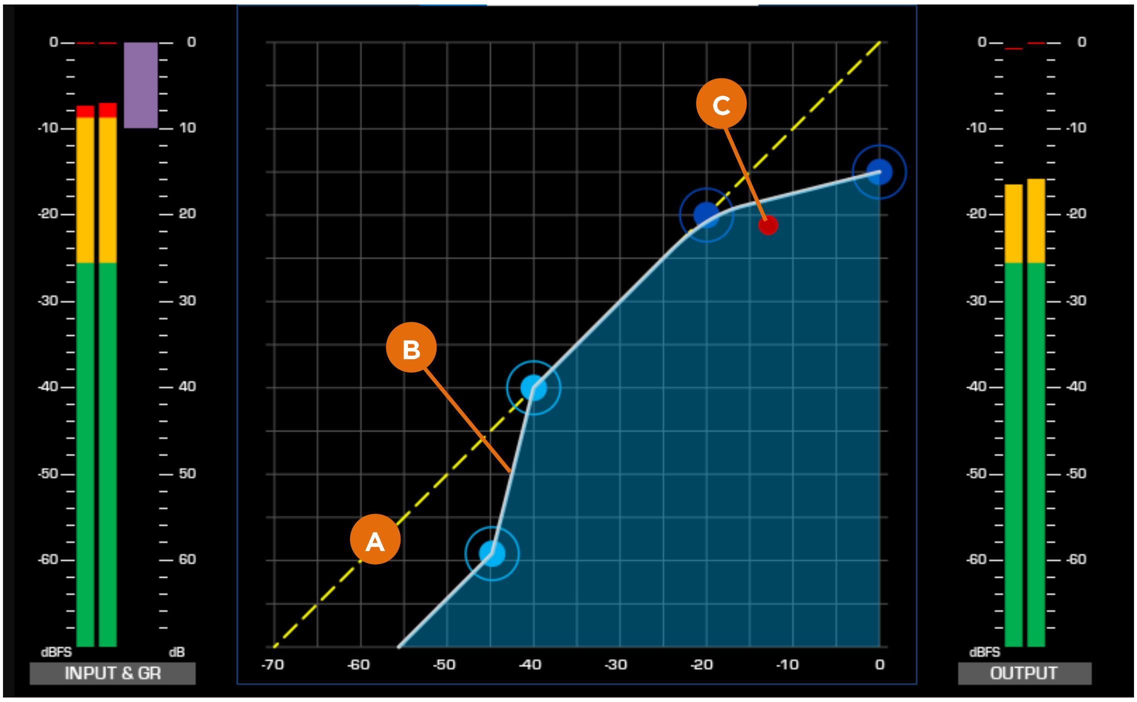


Working With The Dynamics

The Dynamics processor in the Quasar Engine channel is a very powerful tool, sophisticated yet easy 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 and 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 commpressor THRESHOLD and RATIO settings, while the light blue cricled dots do the same for the expander.

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 user, and help him in *real-time* when he's tweaking the parameters: the red GAIN DOT.

The GAIN DOT (C) continuosly 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 GAIN 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/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

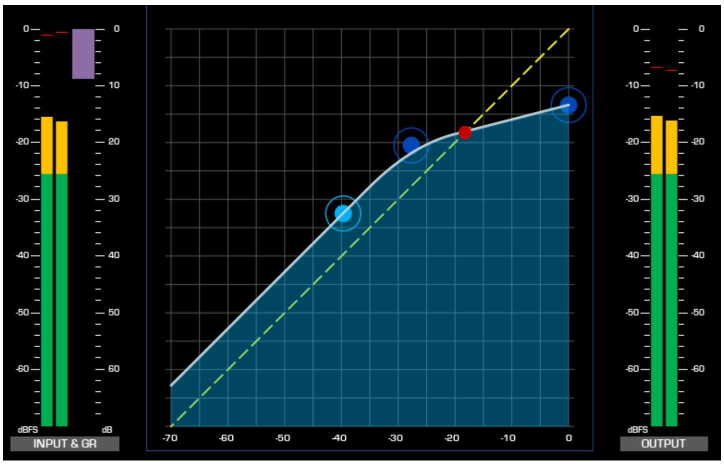
This function, that we designed specifically for broadcast, will let you adjust the output gain compensation *while you are on the air*, without altering the overall level of the audio sent to the program bus. Check the output meter to the right, while adjusting this Make-up Gain, and you will notice 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.



Limiter

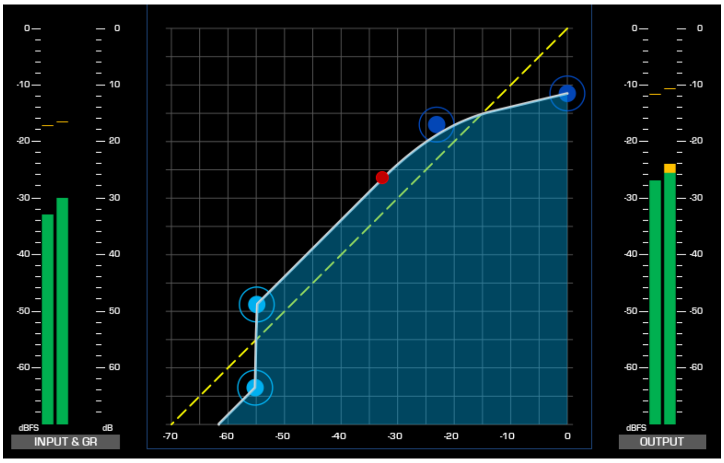
The compressor RATIO can be set from 1:1 to 50:1. Setting ratio to its maximum value turns the compressor into a limiter. In this case, the text "Limiter" appears and the Ratio is set to infinity:1. Meanwhile, the Detector, that is the internal part of the algorithm that controls the threshold and triggers the compression when the signal exceeds it, is switched from RMS to Peak. This makes the compressor a True Peak Limiter.

> **Tip -** If very fast attack times (below 1ms) are set with the Limiter, you will obtain the equivalent of a so-called "Brickwall Limiter".

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 conjuction with the Expander RATIO, it will let you seamlessly morph the expander into a noise gate.

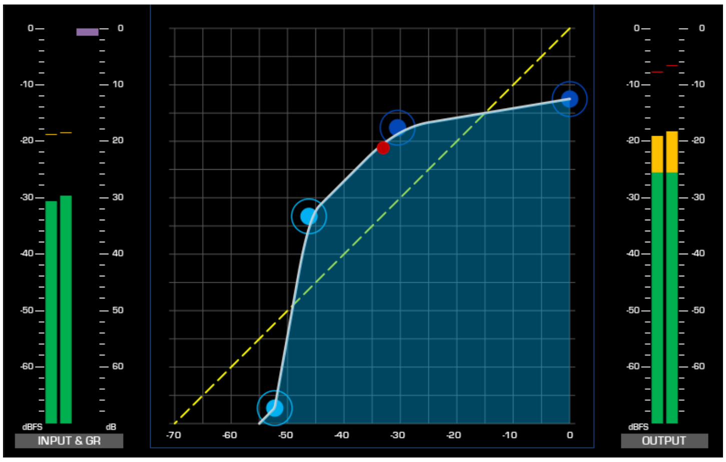
Auto Attack/Release

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.

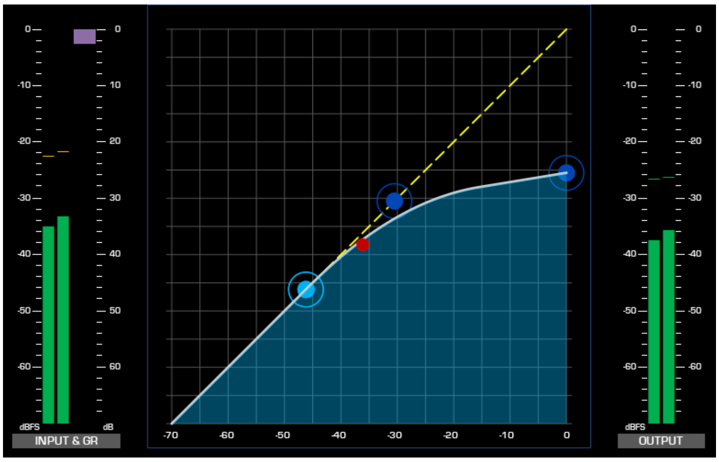


Applying compression to the Talent's microphone will help the voice sound closer and more detailed, letting you play with the proximity effect of the microphone to generate deep and warm voice tone, while controlling peaks, and make it stand out from your mix.

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 threat 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 were our Gain 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.

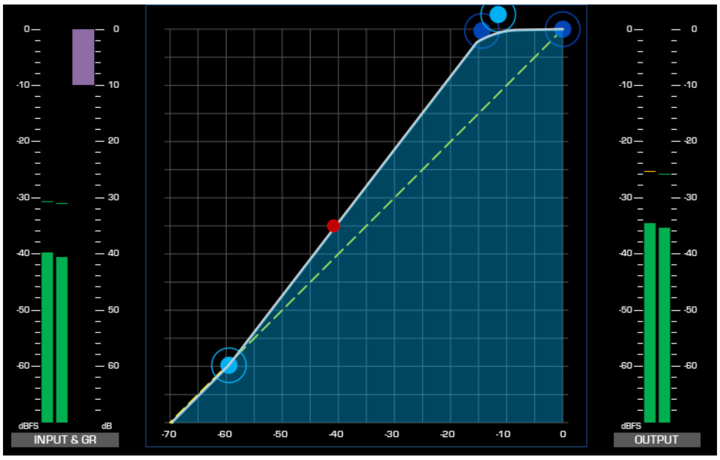


Variable Ratio Compressor

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 de-compress 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.

Working With The 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 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)

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 and enable the PFL function.
- 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.
- 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.
- Adjust the THRESHOLD by rotating the corresponding encoder backwards, from the 0.0dB position, until the gain reduction meter stwhile looking at the gain reduction meter in the top left area of the screen.
- 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.

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 boosts channels with higher signal levels relative to other paths in the group, while it attenuates those with lower signal levels. In reality, the Automixer only ever applies attenuation, and signals are never actually boosted.

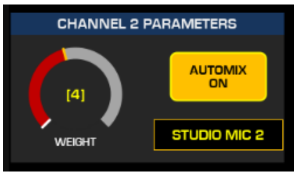
An exclusive gain sharing algorithm compares each input level with the sum of all inputs and adjust the gain of each mic so that the potential acoustic gain of the sum off all mics is constantly equal to that of one mic.

The attenuation is applied by an additional gain stage that can be placed before or after the Fader level control. The default settings is Post-Fader. In this case, the amount of attenuation remains proportional to the fader balance set by the operator.

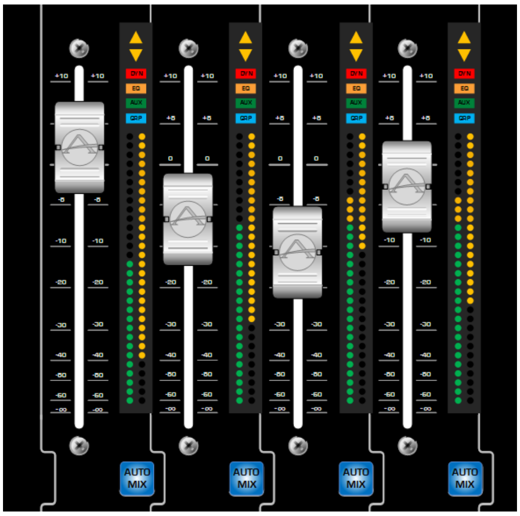
In broadcast, typical examples of usage scenarios for this feature, are:

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

In all of the above cases, the Automixer will produce a mix in which the total ambient/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.

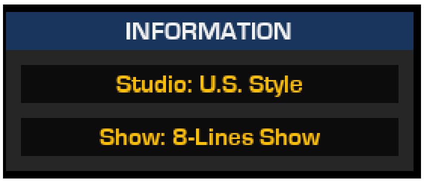
Quasar offers two Automixer groups, one for the Program outputs and one for the Record mix. The settings of these two groups are always linked, and not independent. The record automix is enabled if the fader channel is assigned to the PGM4/REC bus.

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.

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 placed 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.

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 (ooposite 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 # / Caller ID on the first text line (Caller ID always overrides the Line #)
- Line Status on the second text line

The Line is ready for incoming or outgoing calls

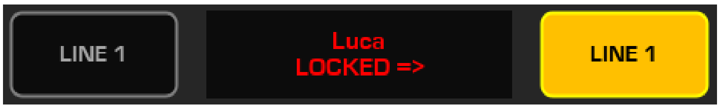
The Line is on Fader 1

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



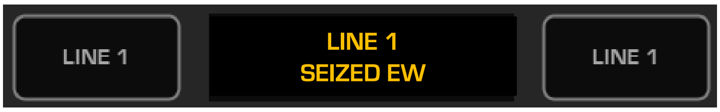
The Line is Locked on Fader 2

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 a 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 tranfer your call to, then you have the following two choices:

- Push the **GO** button and it will transfer the frist 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.

> **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.

Intentionally Left Blank

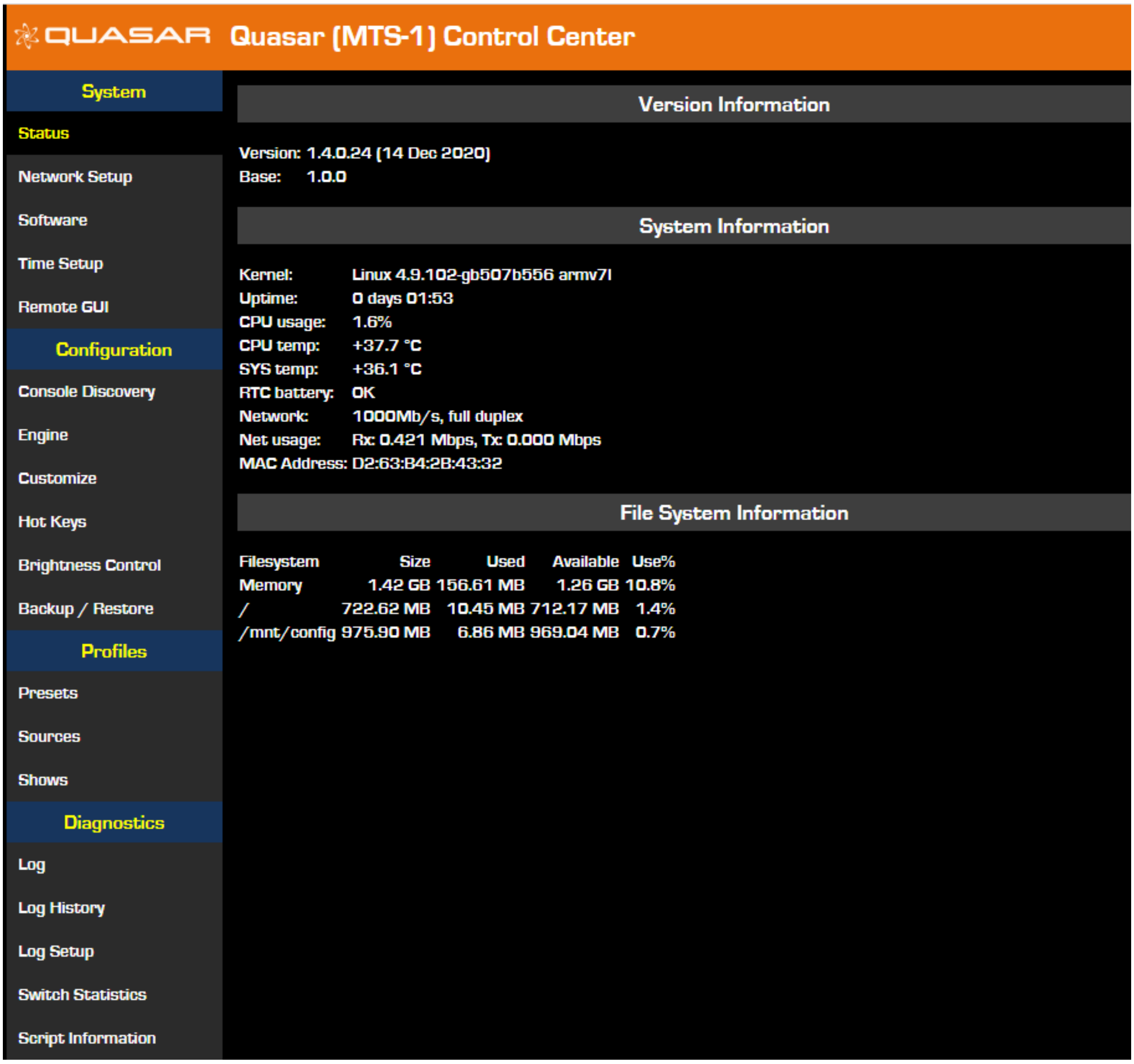
Configuration - 1

Web UI Reference: MTS-MON Module

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.

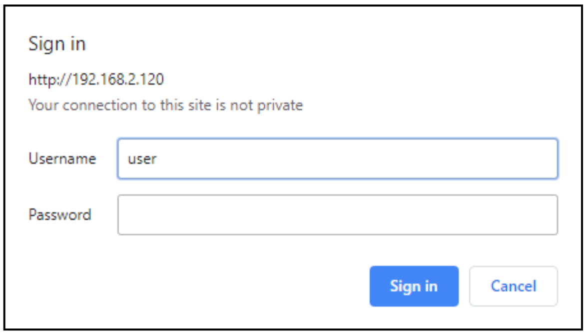
There is no need for the board operator to access the console Web UI during normal console operation. This is designed to let the Station engineer or system Administrator, entirely configure the console, without the need for a specialized Telos engineer. That's the beauty of our system!

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:



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.



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

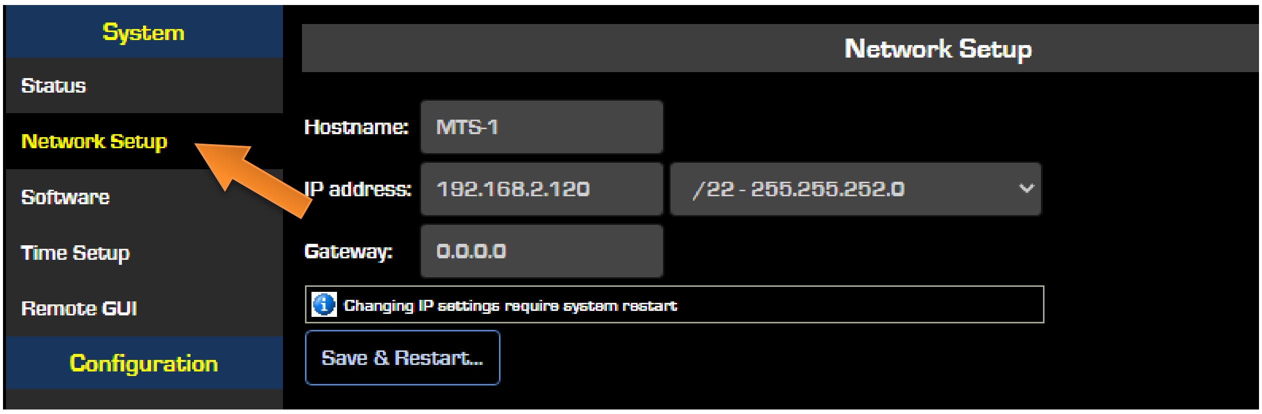
This landing page displays all the most important information about the installed Software Version, the System's internal hardware components, and its memory usage.

Network Setup

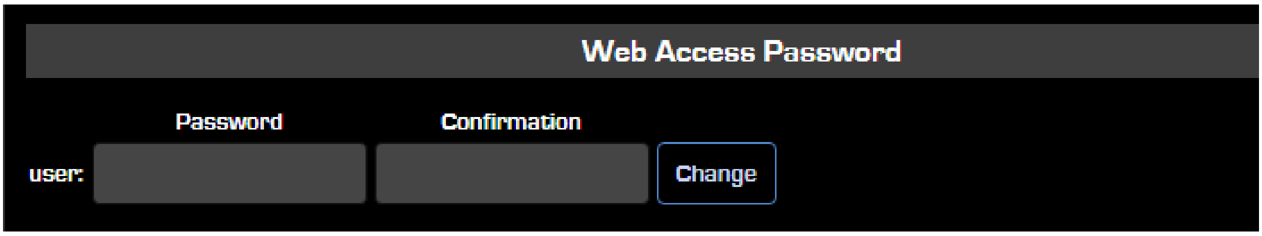
From this page you can set up or change your Quasar network configuration.
Here you can enter the Console Name (Hostname), the IP Address and Subnet information.

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:

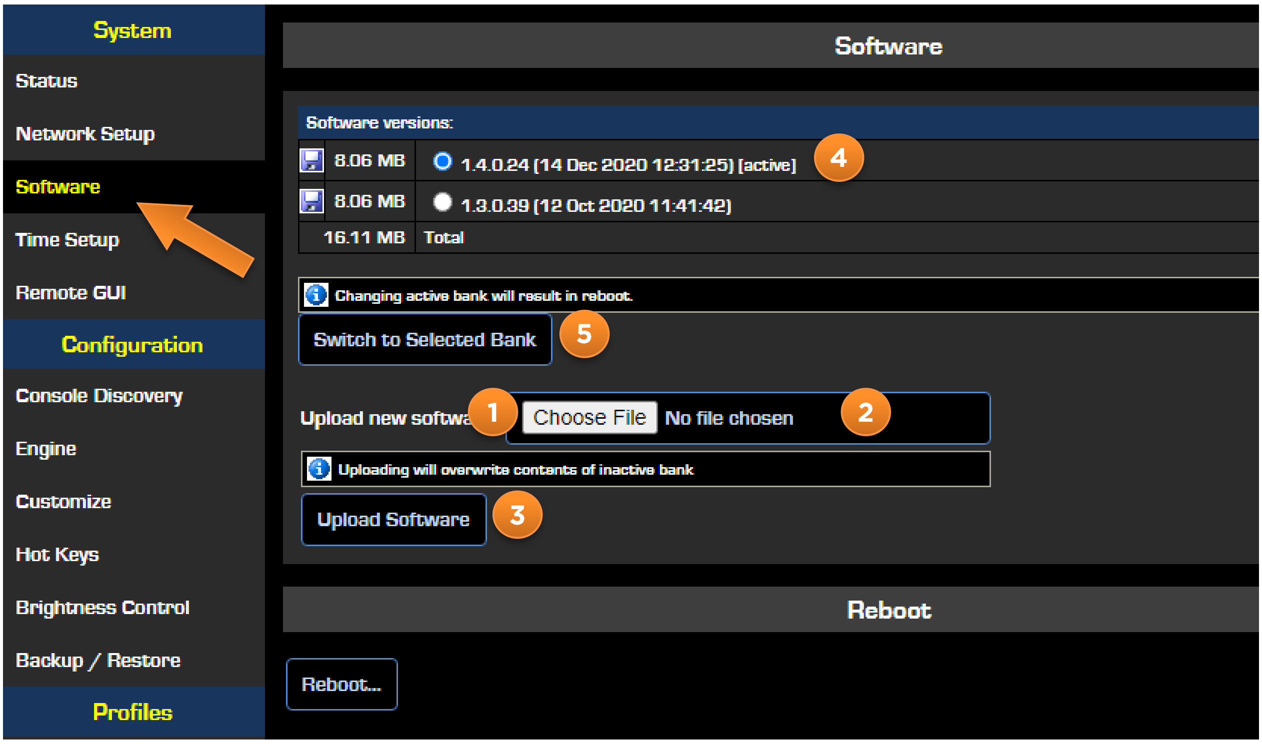


The Username is always “user” and cannot be changed.

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. 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 software bank, connections to all modules and Engine will be reset, and audio will be interrupted for a few seconds.

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 modules requires time to be set for this clock, and for its logging activity.
From this page you can set the sytem time from you PC, simply by cklicking 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.
Remember to press the **Apply** button before you exit the page.

Remote GUI

Selecting this link opens up an HTML5 representation of the Console Touchscreen.
The entire MTS UI, and all the controls accessible from the Touchscreen are duplicated here, allowing the System Administrator, or an operator, to access and operate the console entirely from remote. The controls in this page do not copy, or mirror, the console Touchscreen controls. They act independently, leaving the local operator freedom to access, for example, the EQ tab, while the remote operator is accessing the MASTER AUXES tab.

Normally, on a 24” touchscreen monitor, a 1:1 copy of the console’s 12” touchscreen can be displayed by setting the browser’s zoom at 75% magnification.

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

Note – With the Remote GUI, the console fader strips (or input channels) can be operated only one at a time, by accessing each of the channel’s screen and selecting the CONTROL & B/FEED tab.

Tip – It is possible to connect a Touchscreen PC, or a Tablet, to the Quasar Remote GUI to let a second operator access the console controls, in case a complex workflow requires dual operators.

At the time of writing this manual, when the HTML5 GUI is opened on a browser, it will load default gain values for all the rotary encoders (Gauges). These are then synchronized only at the moment the encoder on the console is touched, or the gauge on the Remote GUI is moved. Direct synchronization will be introduced with the next software versions.

A Note about Internet Browsers

Quasar’s Remote GUI was tested with Firefox, Chrome and MS Edge. Although we support all these browsers, we recommend using Chrome or Firefox.

This is because it seems that MS Edge has a feature that in fact works out not so well for us: When loading an HTML page, Edge loads the page, and then tries to load resources (styles, images, etc) in parallel threads opening several connections, in order to increase page loading speed. This feature was probably designed with Internet Web servers in mind.
Being a mixing console, Quasar’s Http server was designed with a different goal in mind, so we do not support this feature. This may result in a non-optimal navigation experience when using MS Edge with Quasar.

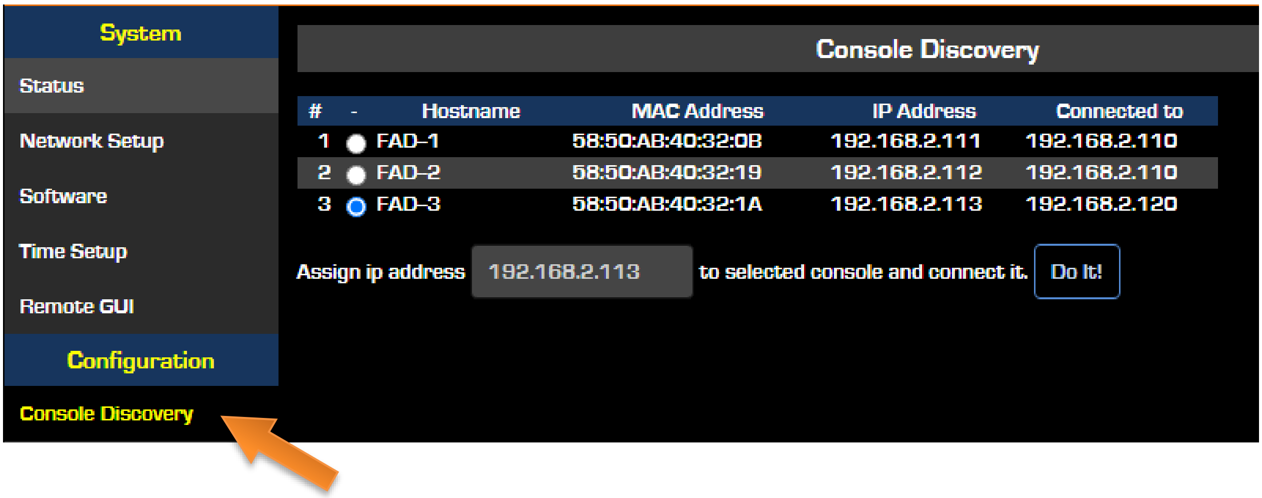
Configuration Menu

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

Console Discovery

This page displays all the information about the Fader Modules (or Accessory Modules, when they will be available) connected to the Master.

This page is normally used during factory configuration, but the user, or system administrator, might need to access it in case he needs to connect a new (additional or replacement) module to the console.

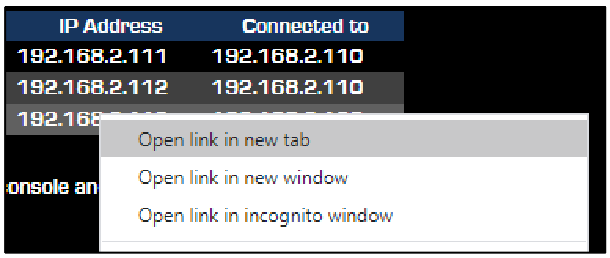


Selecting the Console Discovery link, activates the discovery process, during which the Master module will scan the network for available modules:

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:

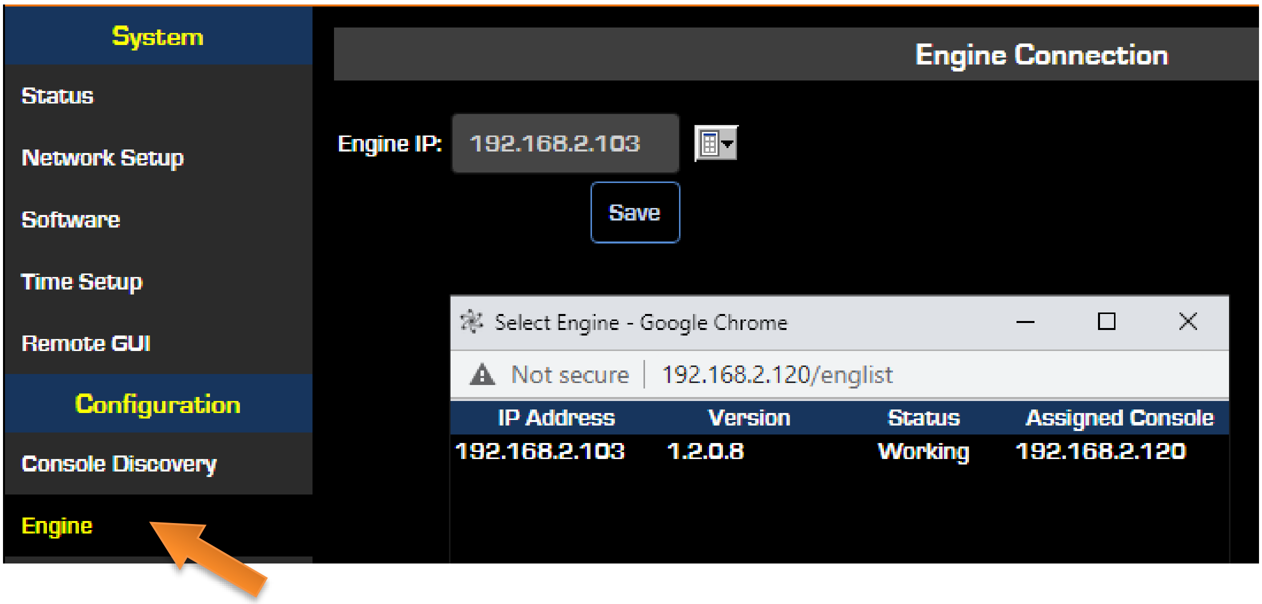
1. Select the module you want to assign, using the radio button in the left column
2. Enter the desired IP address in the field below
3. Click on the **Do It!** button. This will trigger a reboot on the selected module
4. Wait for a moment until the module restarts, then refresh the browser page to check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

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" .



Engine

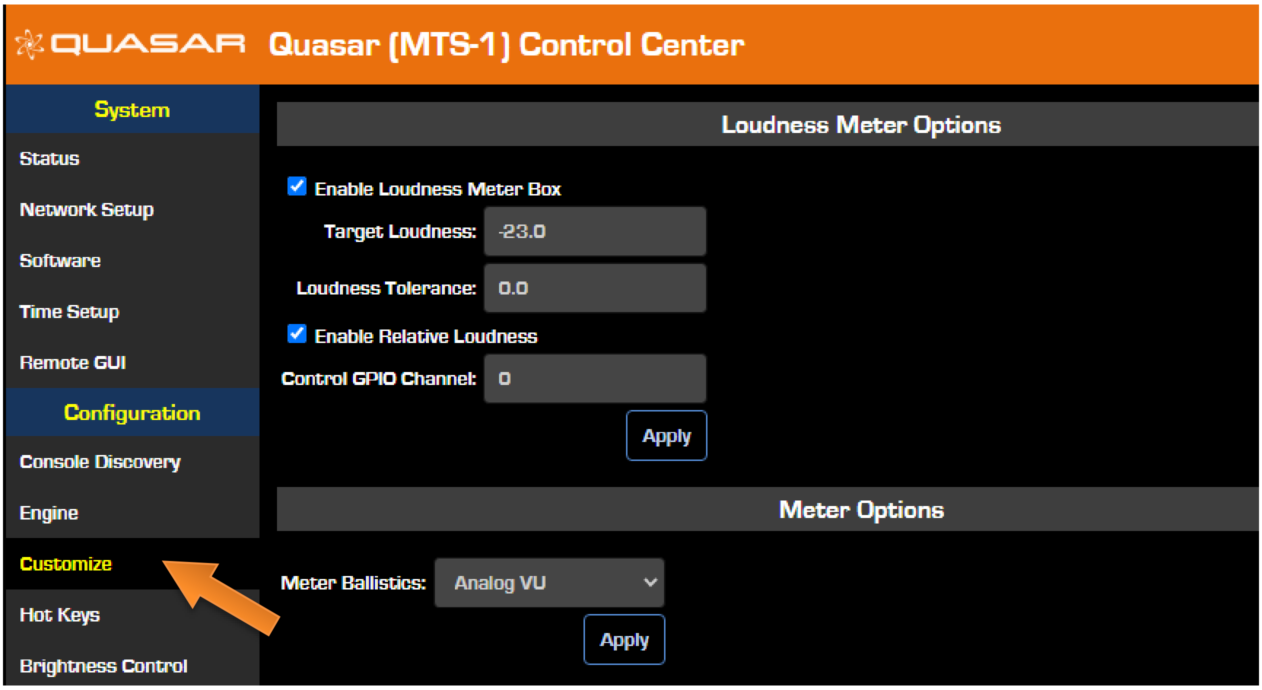
This page displays information about the Quasar Engine connected to your surface:



This page is normally used during factory configuration, but the user, or system administrator, might need to access it in case he connects a new (or different) Engine to the console.
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, Operating Status and Console Connection information.

Customize

Navigate to the **Customize** menu of your Quasar Master Module:



This page offers the following configuration areas:

- Loudness Meter Options
- Meter Options
- Source Type Color Coding
- 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.

Selecting the **Enable Loudness Meter Box** option and pressing the **Apply** button will turn on the loudness Metering box on the MTS Home UI page.

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 the options below the text fields. Depending on region, these values can vary.
For example EBU R128 calls for -23 LUFS and the American Calm Act recommends -24 LKFS (LUFS = LKFS).

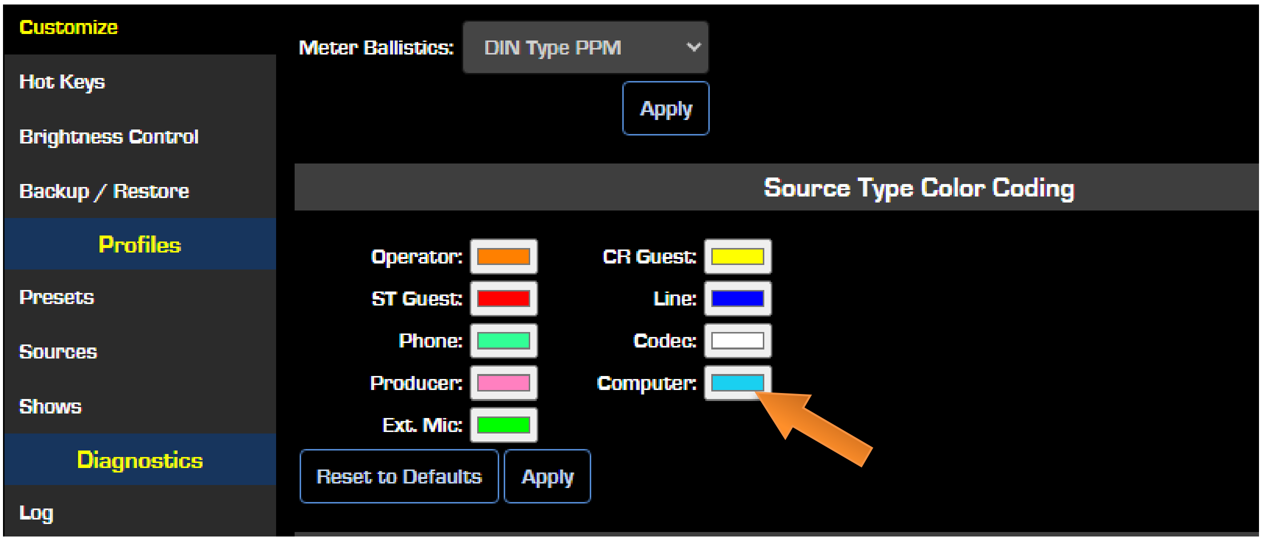
Meter Options

This menu lets you select the Ballistics options for the six Main Meters displayed in the Master Home screen. These options are covered in detail later on, in the Configuration section of this manual. For information about setting these options, please proceed to Chapter 6: Audio Metering Options.

Source Type Color Coding

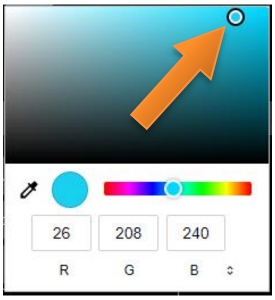
This menu lets the user define colors for each type of Source Profiles, so that when a Source is loaded onto a channel, and appears on the fader strip, the color-coding indicators will let the operator easily spot all the sources that belong the same type, like Microphones, Codecs, Phones, and so on.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

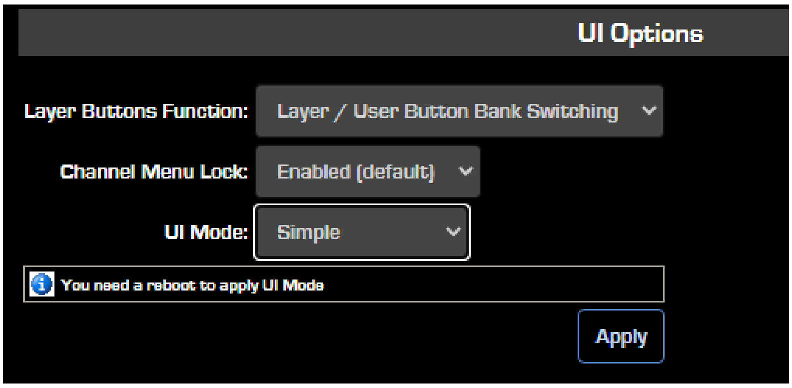


Selecting one of the color boxes, will display a pop-up window that lets you pick a custom RGB color and assign it to the selected Source Type.

You can directly enter an RGB code, or pick a color by dragging the small circled dot.



UI Options



This section of the Customize menu lets you define some important options for working with your Quasar.

Layer Buttons Function

This drop-down menu offers the following three options:

- **Disabled:** Completely disables the four LAYER-BANK keys on the surface. The surface will operate with a single, fixed layer (LAYER 1), in this case. The User Bank B,C,D selection is disabled too. Bank A remains active.
- **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 hold for one second.

Layers and User Keys operation is described in the Configuration Chapter 4 of this Manual.

Channel Menu Lock

This 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 sdisabled.

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. You can do so from the **Software** page.

UI Mode

This option 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.

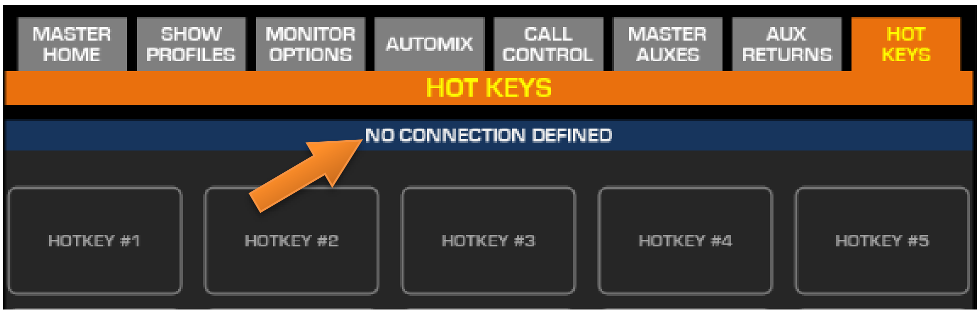
After the Simple UI Mode is enabled, you will need to reboot the MTS Module to apply the new UI. You can do so from the **Software** page.

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

Note - At the moment of writing this manual, 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.

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



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.

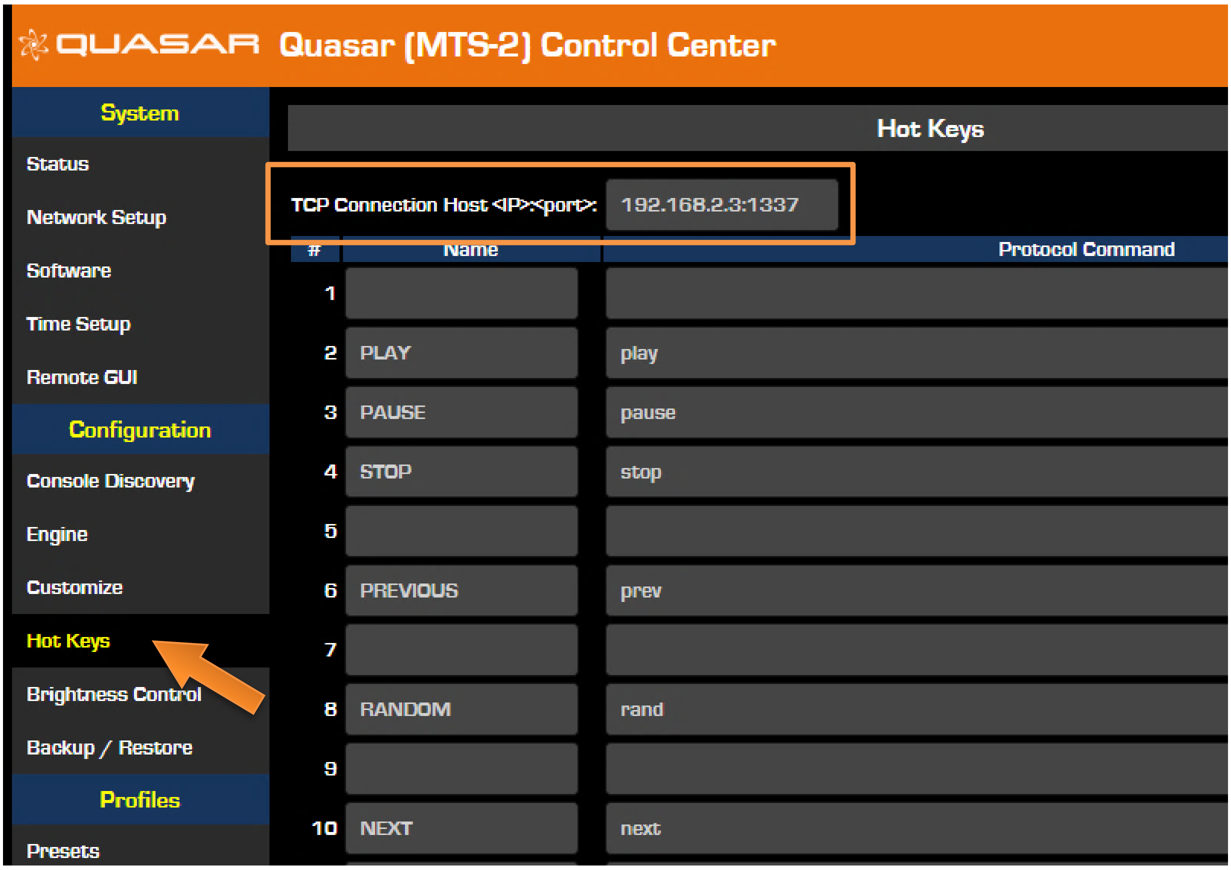
In the **Name** field, type the name you want to be displayed by each of the Hot Keys.

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

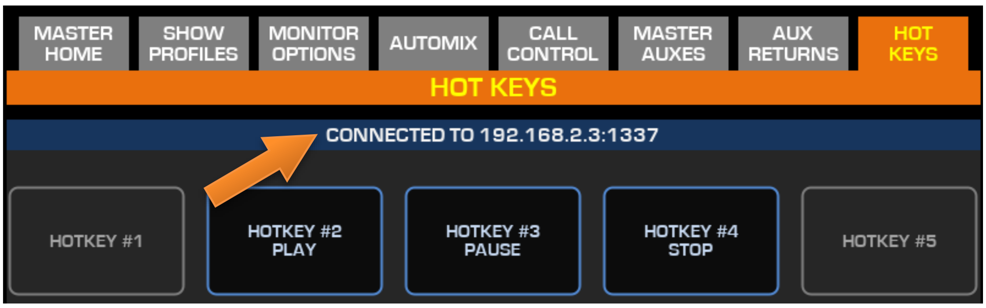
In the **TCP Connection Host** field, enter the IP address of the device you want to control (like a Cart Player or a Jingle Machine) , followed by ":" and its TCP port number.

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 appropriate syntax. Please refer to the device's User Manual to obtain this information.



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

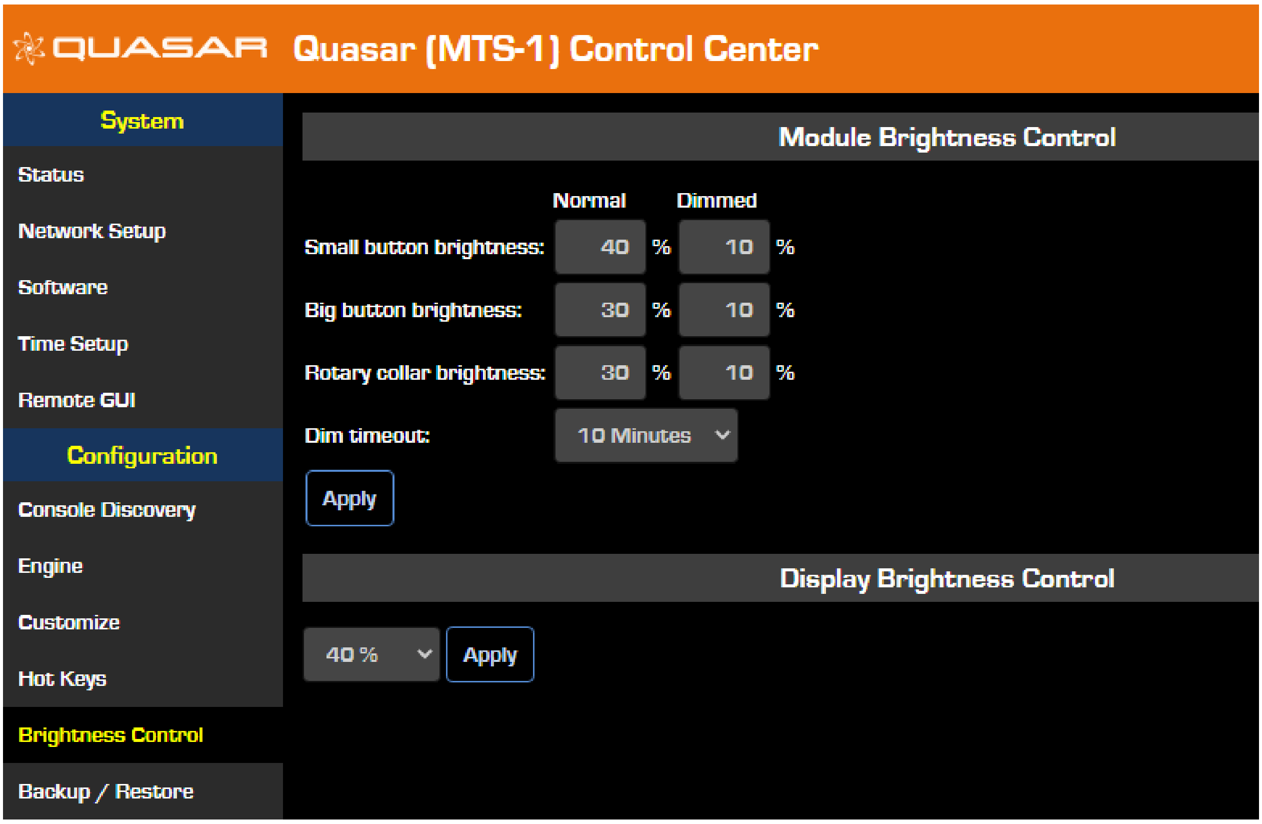


Brightness Control

This page is for setting the brightness of each group of hardware controls found in the Master Module. Two settings are available for controlling group 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 **Dim Timeout** is the time it takes to transition from the Normal to the Dimmed state,



> **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.

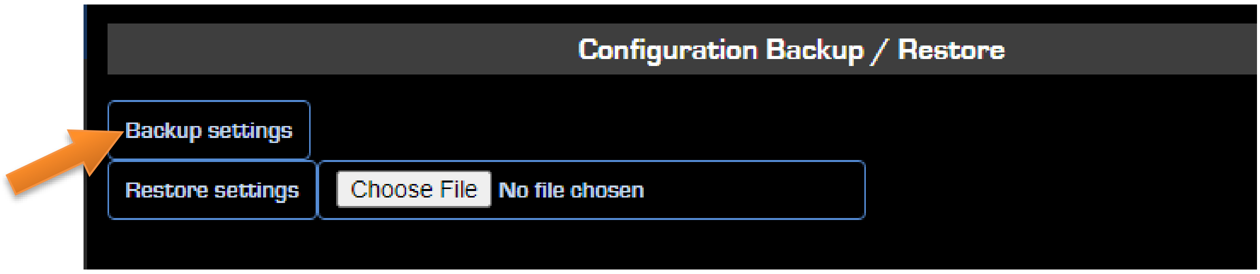
The **Display Brightness Control** allows setting of the Touchscreen brightness.

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.

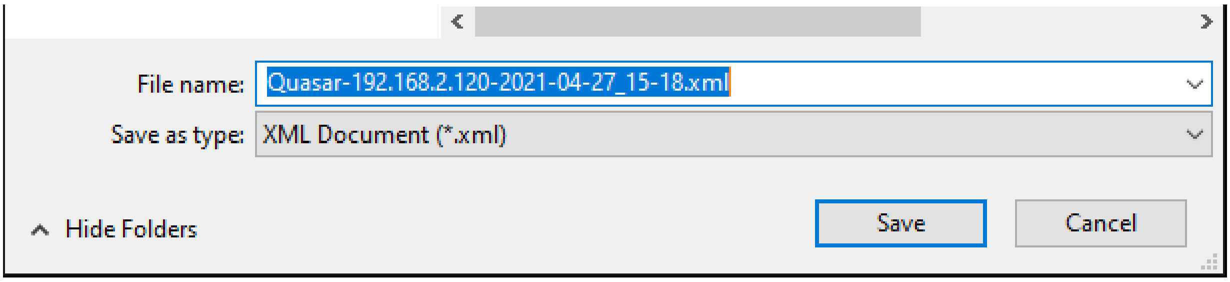
In order to select a different value, pick one of the preset levels from the drop-down menu and press **Apply** . The new brightness level will be maintained until the next reboot of the module (or console).

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 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.



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 Fusion, in the fact that its size is not fader-driven, but channel-driven. So it all 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 viceversa. 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.

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.

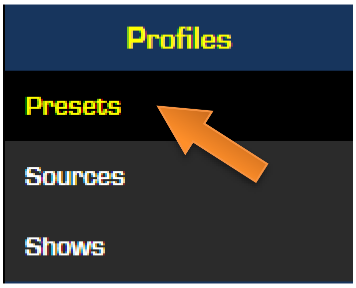
This section offers separate menus for:

- Presets
- Source Profiles
- Show Profiles

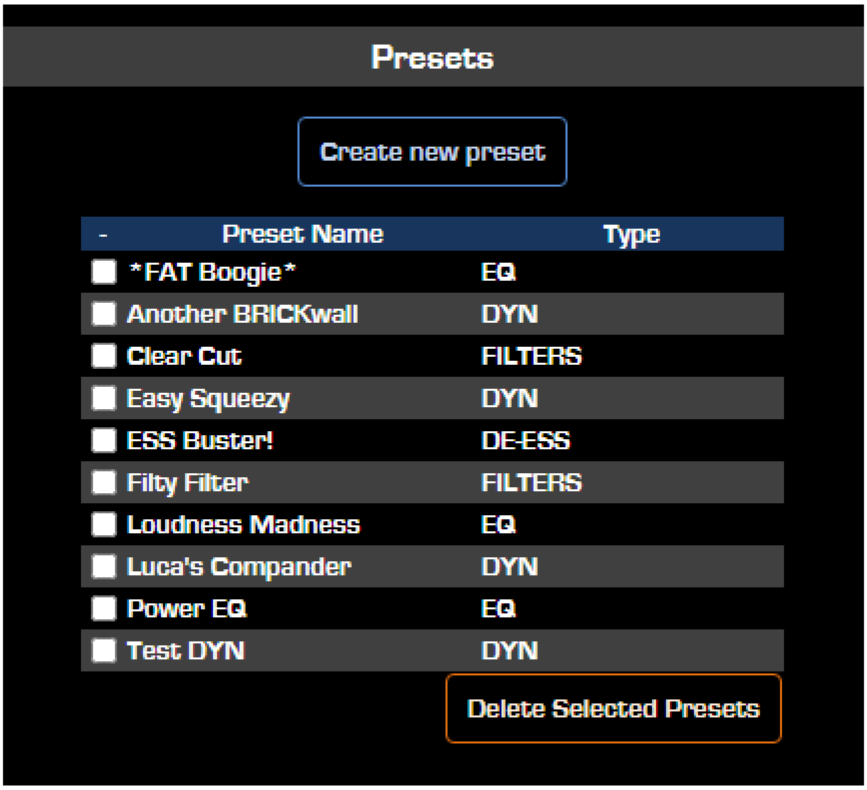
It is very important to master this part of the Quasar configuration, so we dedicated the next Configuration Chapter of this manual entirely to Source and Show Profiles, providing a detailed explanation of how these are created and managed.

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 favourite 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.

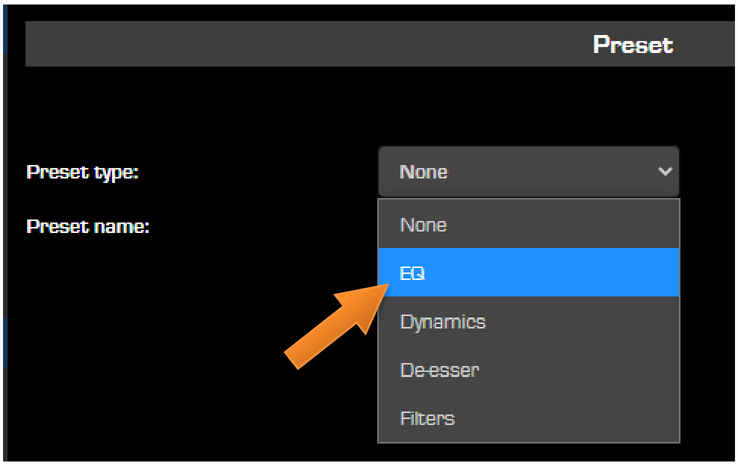


Presets also let you organize and move settings between Shows without having to manually copy each setting from the Show, or from a Source.

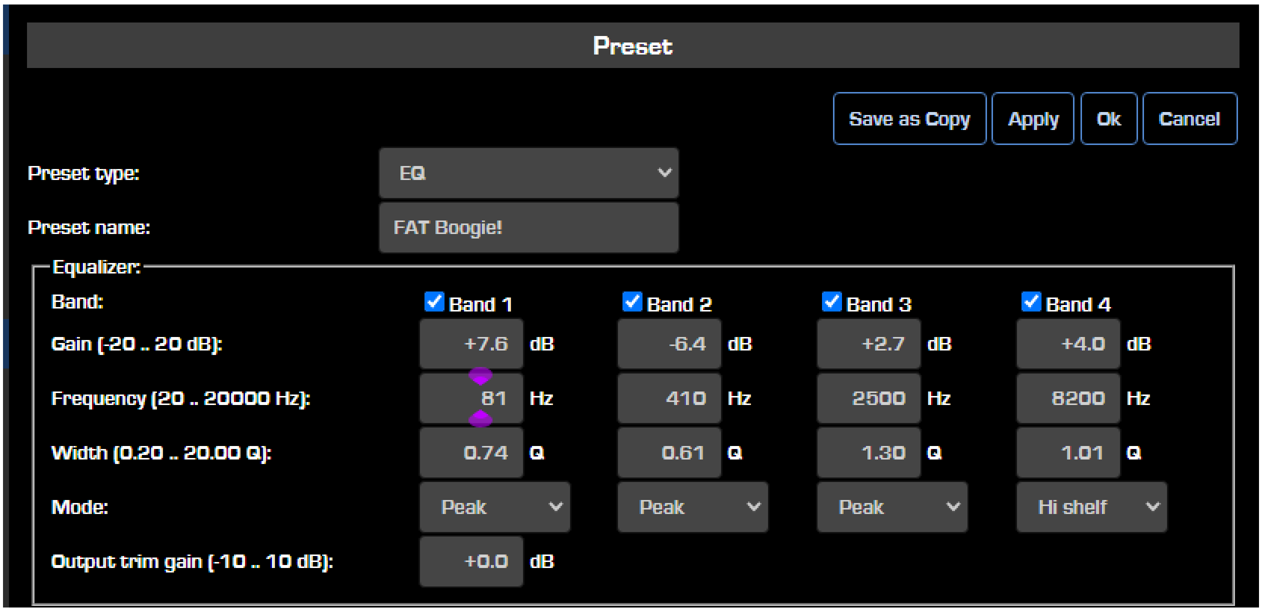


Creating Presets from the Web UI

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

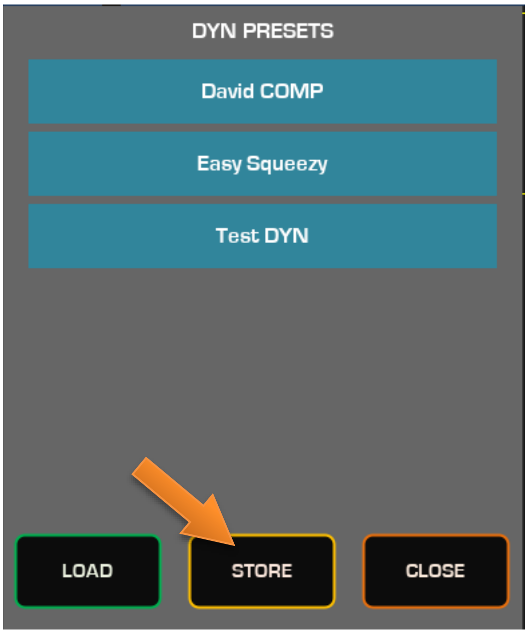
- Create your favourite channel processing (EQ, DYN, etc.) settings

--- *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 favourite EQ settings (for example) from one Show Profile to another, you can do so in two different ways:

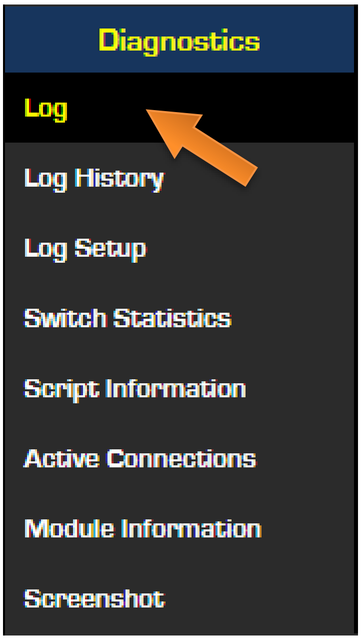
1. Load the Show with the EQ settings you want to copy, save them as a Preset, then Load the target Show, load the desired Preset so that its parameters are transferred into the Show's EQ settings, Then, update the Show in order to permanently store those settings in the Show configuration.
2. Or, you 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 – At the time of writing this Manual we don't have a function that copies a preset to a Source. But you can do so manually from the web UI.

Diagnostics Menu

Access to the 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.

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 file is created and saved. Each file has an upper size limit of 1MB. You can download individual files, by clicking on the floppy icon, while you can delete individual files by selecting them with the checkbox and pressing the **Delete Selected Files** button. The files are sorted top to bottom, from newest to oldest.

Log Setup

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).

Switch Statistics

This page is for monitoring the activity of the Ethernet switch built-in the Master Module:

- 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.
- Collisions should never happen. They indicate that the Network port is not working in Full Duplex mode.
- CRC Errors normally indicate bad Ethernet cable, or bad Ethernet connection.
- Rx and Tx Drop Packets normally indicate bad buffer operation on the transmitting or receiving ends of the connection.

Active Connections

Here a list of all modules connected to the Master is available. Mostly used for checking modules' IDs and if a module is properly connected or alive.

Module Information

The physical state of all Hardware controls is displayed in this page. This is useful debug tool for telling whether or not a physical button, or a touch key is working properly or not. Touch an encoder, or press a button, the refresh the web page to check if its state has changed.

Screenshot

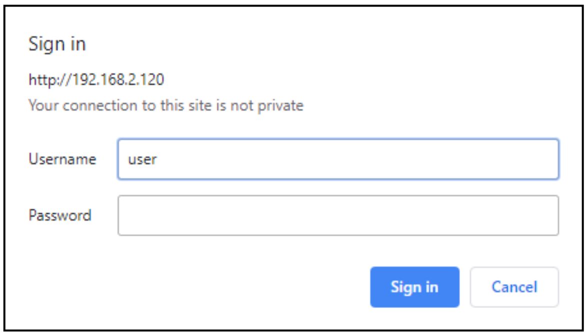
This useful feature lets you retrieve a snapshot of the graphics displayed by the Quasar MTS Touchscreen. It is very useful in case a remote operator needs to quickly monitor the console state, as if he was in front of the console. Here are some examples:

- Meter activity will show if the console is passing audio to its mix busses
- Loudness Meters will indicate if the audio from the console is too loud or soft
- The banner above the clock in the MTS Home page will show the currently loaded Show Profile, or a warning message in case connection to the Engine is lost
- The ON AIR and RECORD indicators in the MTS Home page will give a feedback on the operation status.
- By refreshing the screenshot multiple times, it could be possible to understand if the console is left unattended, or if it is being operated.

Web UI Reference: XR-4FAD Module

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.

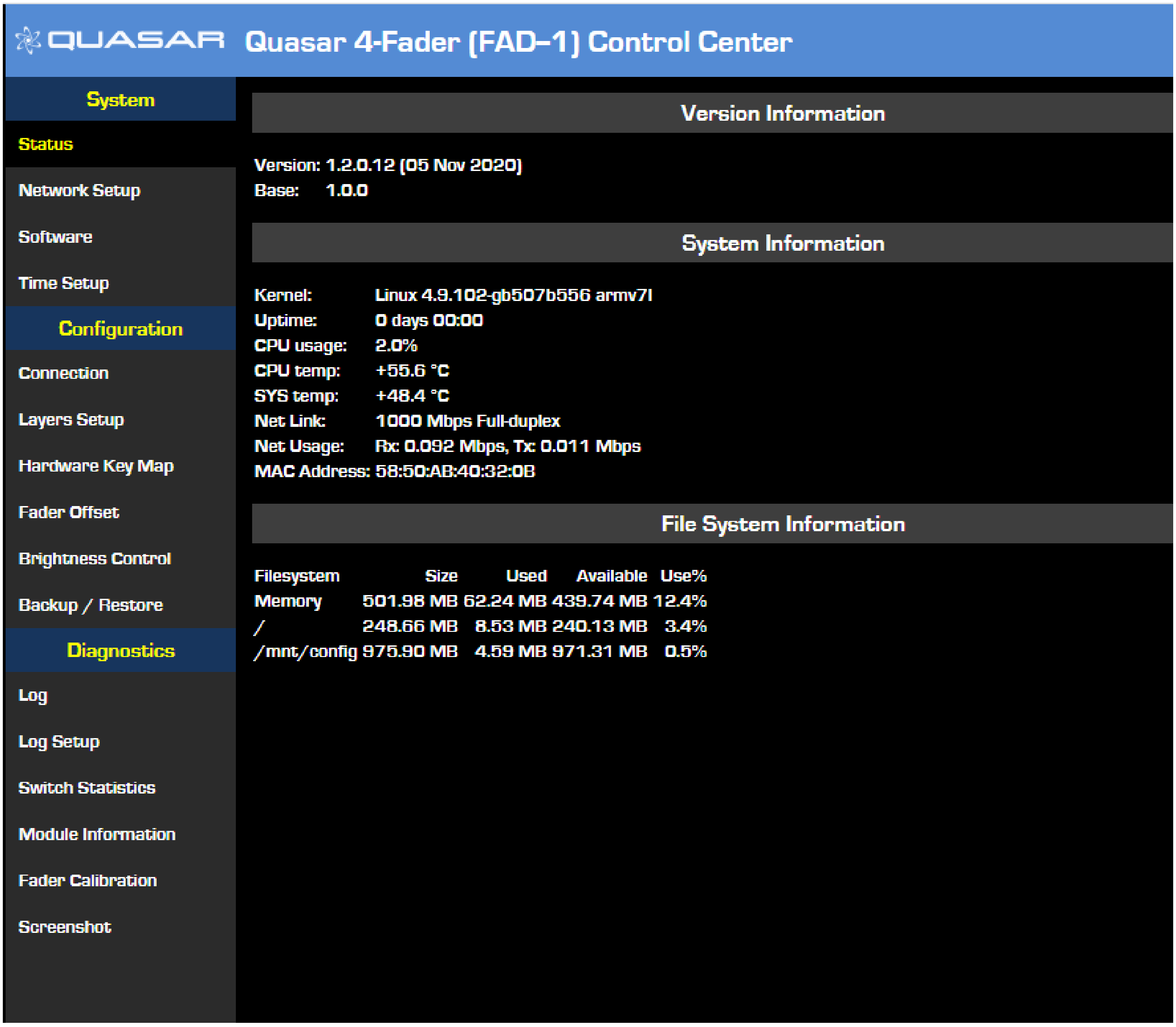
Connect a PC to the console network, and open up a browser. Enter the IP address of one of the XR-4FAD Modules.



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:



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 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.

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. 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.

Click the **Switch to Selected Bank** button.

Time Setup

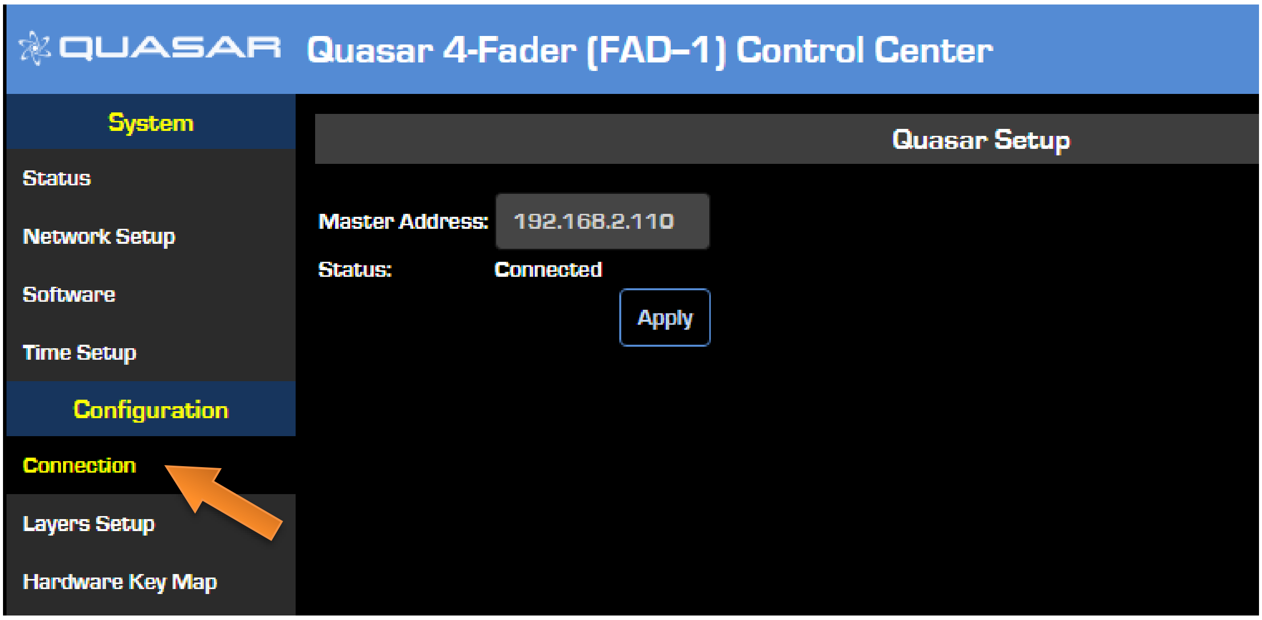
The Quasar XR-4FAD Module requires time to be set for its logging activity.
From this page you can set the sytem time from you PC, simply by cklicking on the **Set Time from PC** button. The clock can be set to display 24 hour time (0:00 to 23:59) or 12 hour time (AM/ PM). Timezone and Day Light Saving (DST) options are also available.
It is also possible to enter the IP address of an NTP server, to automatically update the clock.
Remember to press the **Apply** button before you exit the page.

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 (identified as "Console") which the Fader Module is connected to, and its connection Status.



It is possible to change the connected Master from both the Fader module end, using this menu, or from the Master module end, from the **Console Discovery** menu found in the MTS-MON web UI.

> **Note -** It is also possible to change this parameter using an external command from Pathfinder

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.

Fader Offset

Some users may want to change their "nominal" fader position. This is a global setting (applies equally to all 4 faders) that permits you to do this.

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.

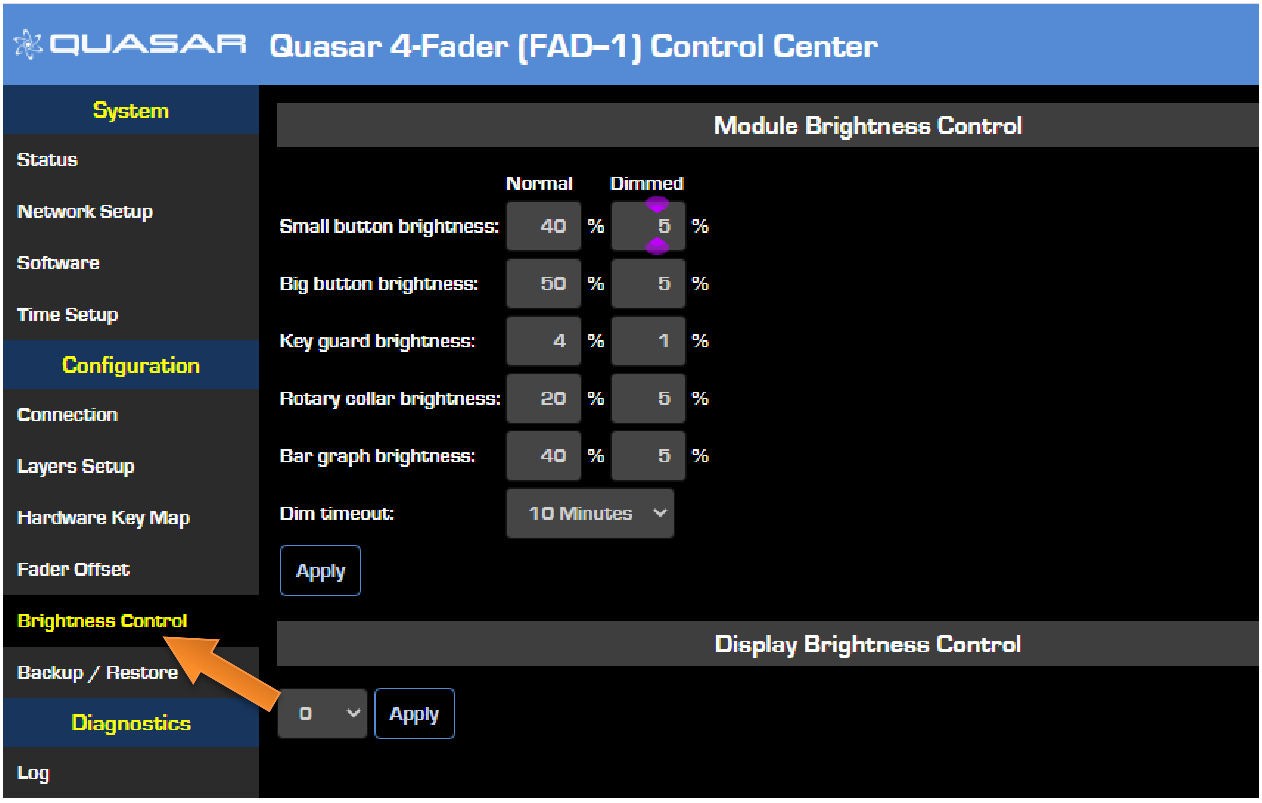
Example 2: a setting of +6 will increase the "sensitivity" of your faders by 6 dB. The faders in this example would have to be run 6 dB lower in order for the PGM audio output to be the same as the -0dB default setting.

> **Note -** Your meters will always display the actual audio levels, no matter what the offset is.

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 group the **Normal** and **Dimmed** states of each group. Enter the desired value, directly in each of the field and click **Apply** .

The **Dim Timeout** is the time it takes to transition from the Normal to the Dimmed state.
The **Display Brightness Control** allows setting of the four Channel Displays' brightness.



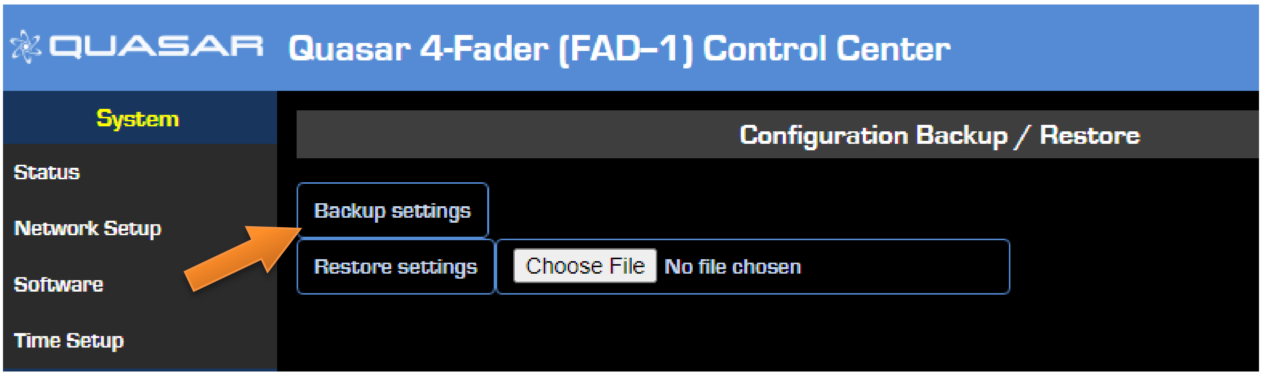
The default value is 80%. If the room ambient lighting condition permit, we recommend leaving this parameter set at the default value. This will help extending the lifetime of the display's backlight unit.

In order to select a different value, pick one of the preset levels from the drop-down menu and press **Apply** . The new brightness level will be maintained until the next reboot of the module (or console).

> **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.



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.

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

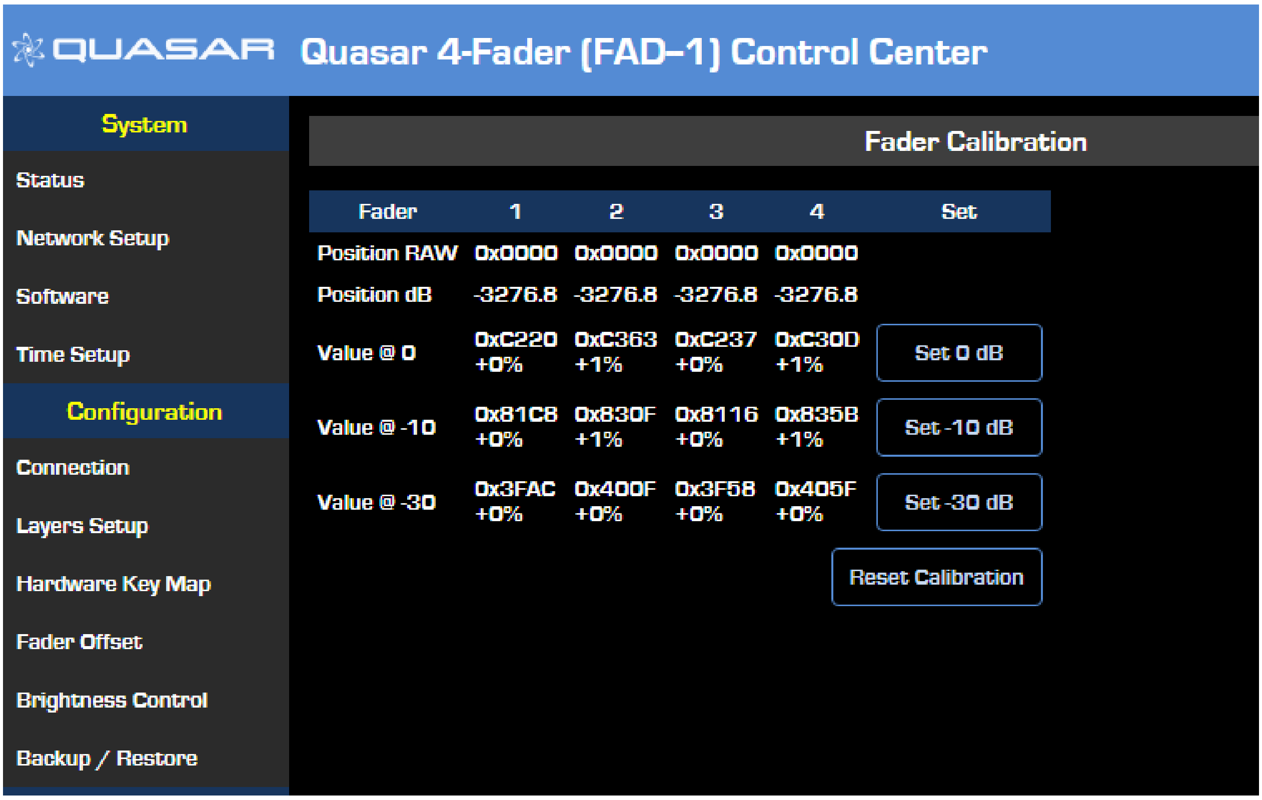
Access to the Fader Diagnostics Menu is never required during Console Operation, or System Configuration. Generally this section is used to retrieve specific information about the System when asked by Telos Support, such as Logs, Network Ports activity, Connected Modules and Active Firmware version.

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, 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.



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.

Configuration - 2

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.

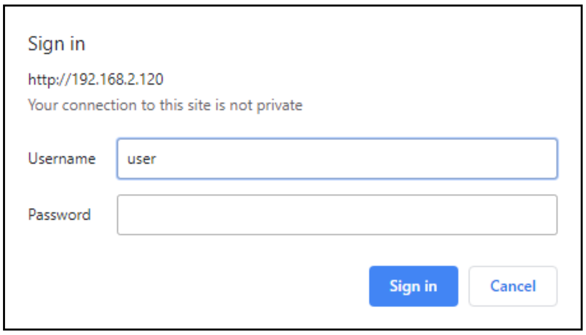
The fundamental difference between an Axia console (not just Quasar, but all our models) and its competitors, is that we created a *set of logic attributes* that can be associated to any audio source, which describes how that source behaves when it is loaded onto the console. This “custom logic layer” can be manipulated directly by the end user via the console built-in Web UI (no configuration softwares 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.

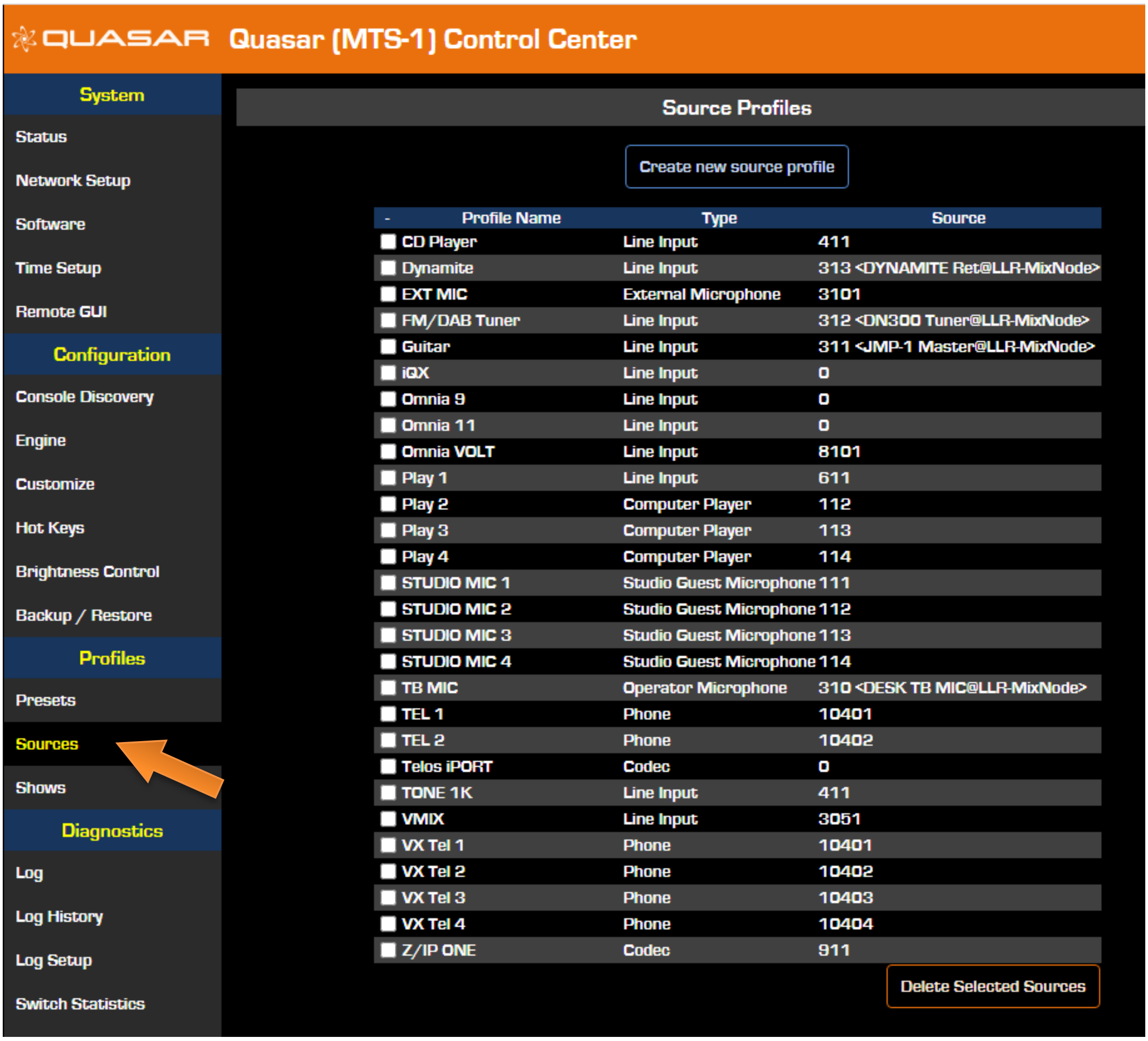
This is accomplished through the use of Source Profiles. Source Profiles allow you to specify which network audio sources are to be used on each console in your facility. They also include Input processing, Panpot, EQ and Dynamics settings. Even mix-minus source selections, can all be set in the Source 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.



select the **Sources** link, as shown here:



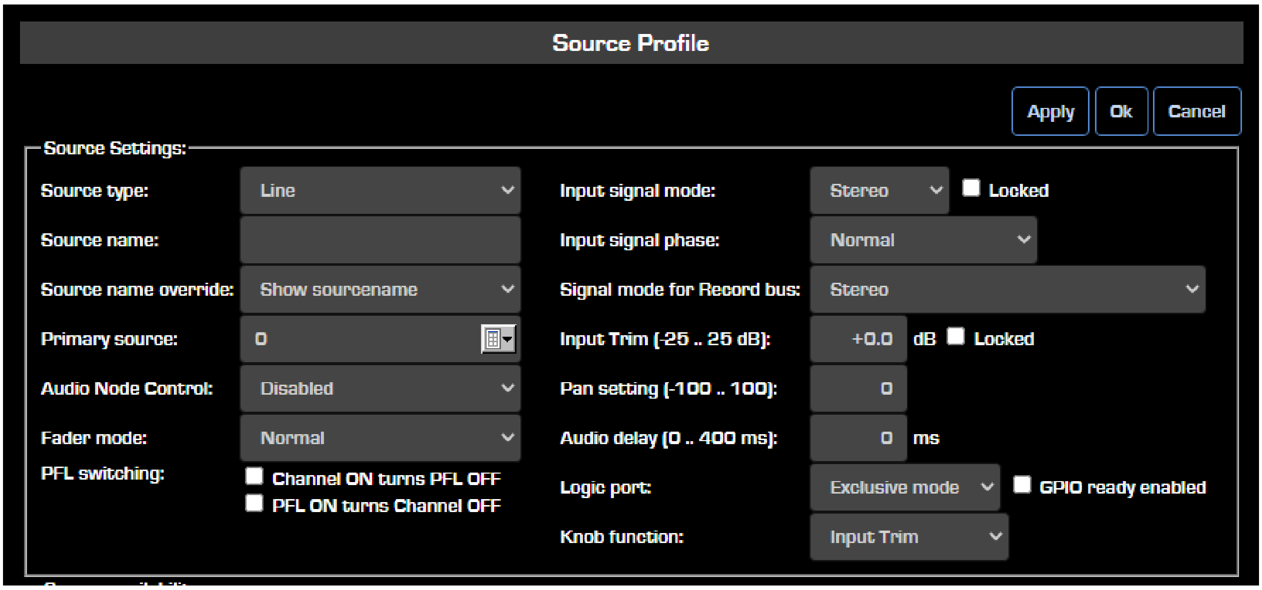
In this screen, you'll be able to see any sources that have already been configured, as well as to create a new Source by clicking on the **Create new source profile** button at the top of the page.

In order to Delete Source Profiles, select any of the checkboxes to the left, and press the **Delete Selected Sources** button.

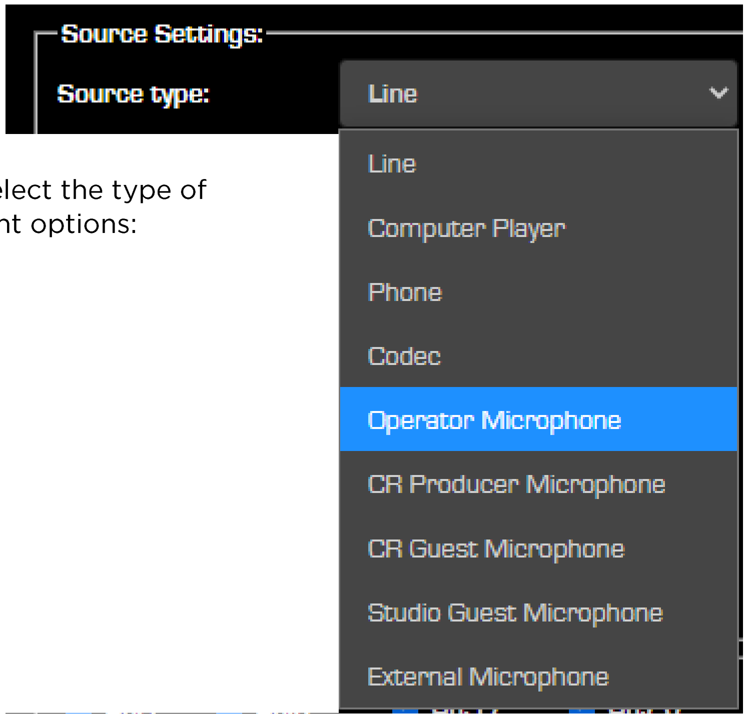
Basic Source Settings

Selecting **Create new source profile** displays a configuration page for your new Source. The layout of this page has been designed to be compact, for minimizing scrolling during the navigation, and with the settings grouped in different sections, sorted from top to bottom from the most to the least frequently used.

All the basic Source settings (those that define its Logic) are in the top section:



Note - While the options visible in the left part of this screen 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 becaming available.



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 below shows which options are available to each of the 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		•	•	•	•	•	•		•	

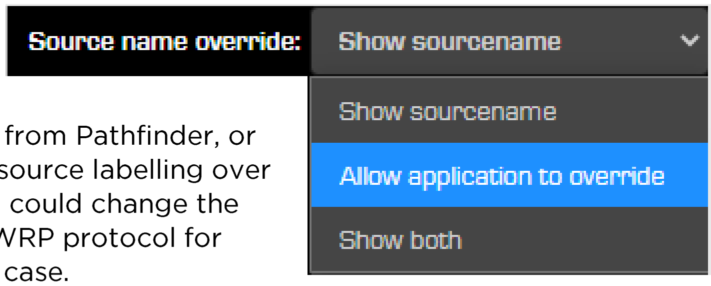
Source Name

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 Field to the Source Profile.
- Selecting **Allow Application to override** will override the Source name and display the LABEL information sent by the the external system.

- Selecting **Show Both** will display both the Sourcename and the Label on two lines of the Quasar channel displays.

Primary Source

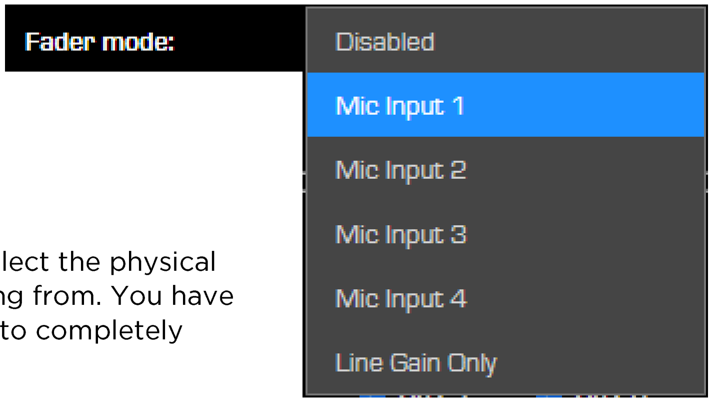
Select the appropriate **Primary Source** using the adjacent drop-down Source Selector box to pick the desired Livewire channel from your Livewire Network.

Alternatively, you can type the source channel ID or directly its multicast address.
You can type "0" if you want to clear the stream for this Source.

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 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 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.

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 casn 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 GPIO contact to Fader Start, please refer to Configuration Chapter 7 of this manual: GPIO Computer Playback Device Logic.

PFL 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.

- Both options un-selected**:** The **PFL/AFL** bus assignment will be retained even after the fader is turned ON.
- **CHANNEL ON turns PFL OFF:** If the PFL function is active on this Source, turning its channel ON will clear its PFL assignment.
- **PFL ON turns Channel OFF:** Does just what it says - useful for Phone channels.

Input Signal Mode

Defines the mono/stereo setting for the Source. Three 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.
- **Locked** if enabled, will prevent the user from changing the source's signal mode (as set above) from the Console surface.

Note – The Input Mode drop-down menu is not available when a Microphone source type is selected

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.

When the **Locked** checkbox is enabled, will prevnt operators from making changes to the above trim gain setting, from the Console surface.

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.

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.

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.

GPIO ready enabled

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:** Element 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.

Knob Function

This drop-down menu turns the Option Knob (located at the top of each channel strip) into a soft encoder. It allows you 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:

- **Input Trim**
- **Mic Gain**
- **Line Gain**
- **Automix Weight**
- **Channel Pan**

Source Availability

These checkboxes determine which Input Channels (not fader strips) will be allowed to load this source. To prevent a source from appearing on these inputs' selection lists, uncheck the desired box.

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" box.

Tip : A quick way of removing checkmarks from all boxes is to double-click the **All Channels** checkbox.

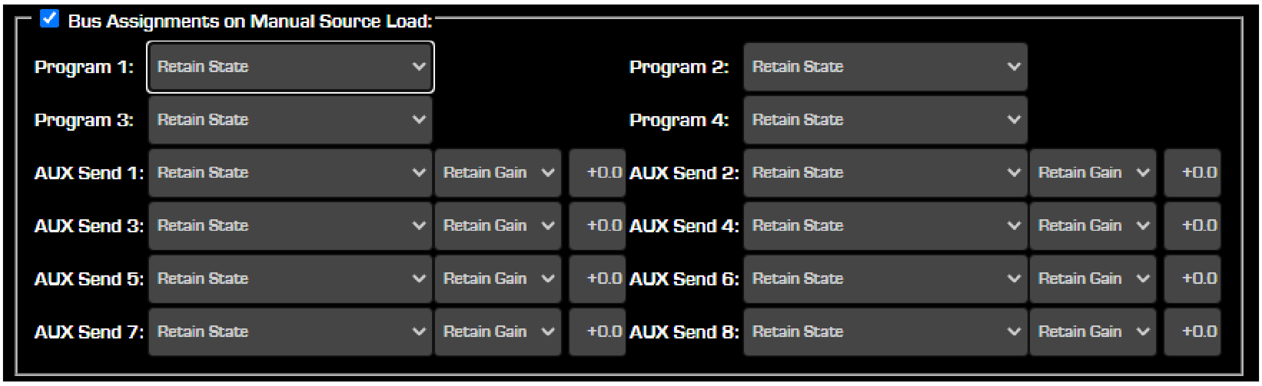
Generally speaking, normal sources such as computers and CD players will have fader assign capability only while sources such as off-air receivers, or on-air processors, will have monitor only capability.

Bus Assignment On Manual Source Load

This section covers a very important aspect: it defines the default assignement status for the four Program and eight Aux Busses, when the Source is loaded manually, to a channel.

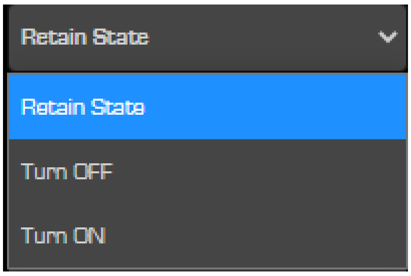
In fact, the behavior of the Source when this is loaded through a Show profile, is defined in the Show Profile settings page.

Select the checkbox if you want the assignments (and level, where applicable) to be different from defaults "Retain" settings.



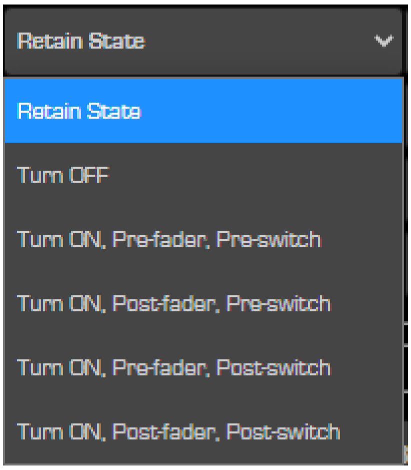
The options available for the **Program 1 – 4** Busses are:

Retain State , Turn OFF , Turn ON



- 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.

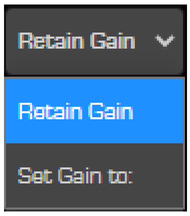
The options available for the **Auxiliary 1 – 8** Busses are:



- 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 <u>before</u> 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.

Filters, Dynamics, & Equalizer

These sections let you predefine the default Filters, Dynamics, and 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”.

> **Note –** Dynamics and De-Esser settings cannot be pre-defined in the Source Profile, for Line and Computer Player source types. You can still store these settings for those type of sources in the Show profile Channels, if you desire.

That’s it. These were the most important functions that need to be configured when creating a new Source. 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 Configuration Examples

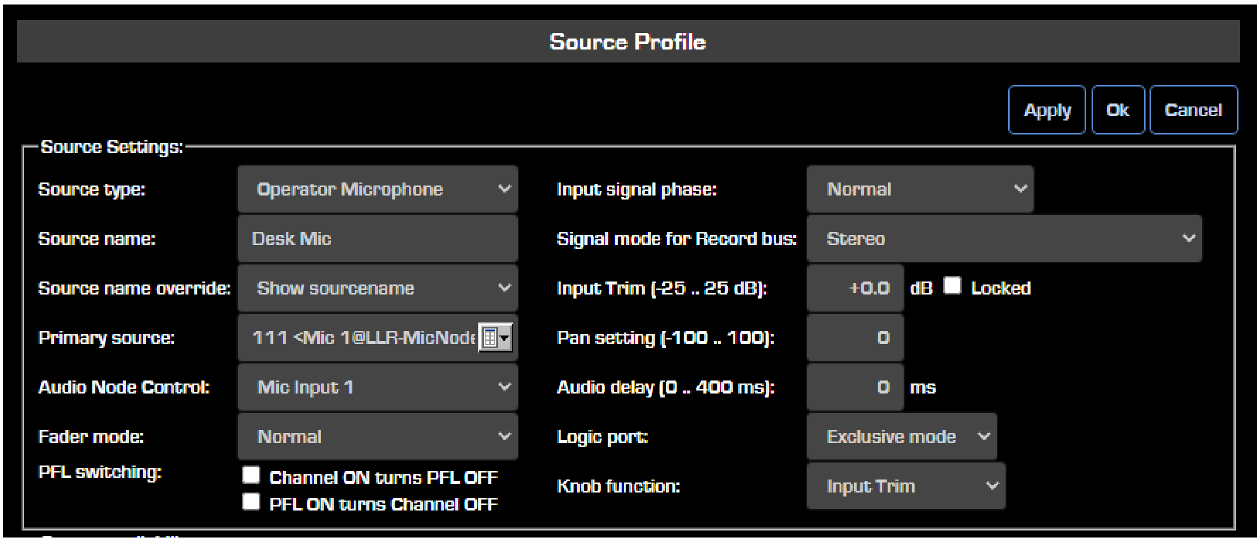
Following are some examples to show how different source types can be created and configured. A good way to understand Source Profile configuration options is to jump in and build a few common sources that almost any studio would need, such as:

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

Although the type and number of source profiles that need to be built at any facility differs from the next, these five samples represent the basic types of sources found in most studios; with this foundation (followed by a review of the full Source Profile options found in the Appendix), you should be able to build profiles to satisfy all of your needs.

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:



Now enter the following settings:

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.

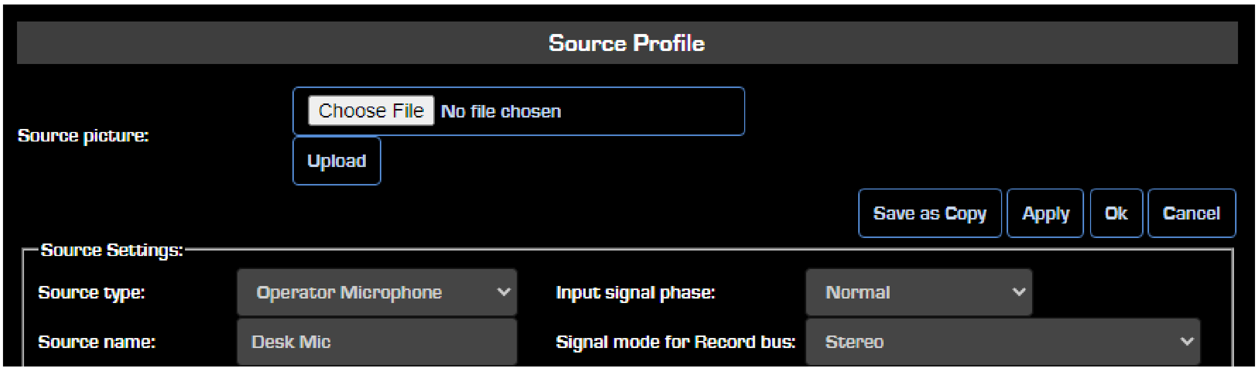
In the **Primary source** field, enter the board mic's Channel Number. (Each audio source in the network has its own unique Channel ID number.) If you know the number, just type it in; if not, use the browse button to the right of the field to select the source from your network.

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.

Click **Apply** **.** The page will refresh and display a new field called **Source picture.** This will let you upload a picture that will be associated to the source, and show up on the console channel strip displays when the source is loaded. The picture must be .PNG or .JPG type, max.192x192pixels, and max. 8kB 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.

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** — it's not wise to assign the Operator's mic directly to the monitors!

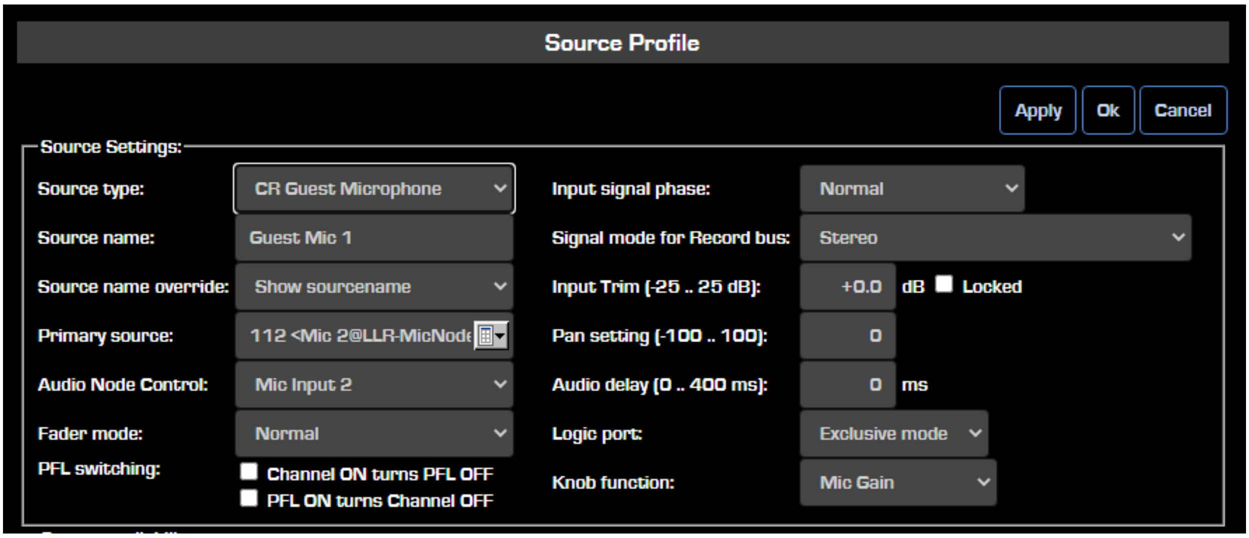
Press the Ok button at the bottom of the page and your new Mic source is ready to use – you've just created the Operator Mic Source Profile.

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.

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 last section.



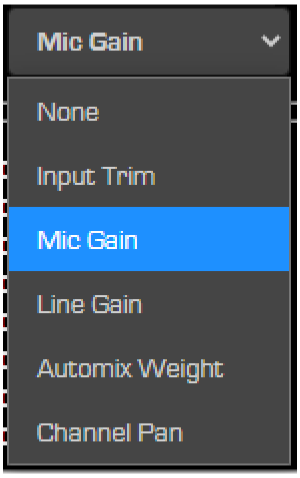
From the **Source type** drop-down, select **CR Guest Microphone** and type in a friendly name of up to 10 characters in **Source name**.

In the **Primary Source** 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.

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. 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.

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 opens the Channel Option screen on Quasar 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:

Input Trim adjusts the 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, and compensate for audio that's too "hot" or too low.

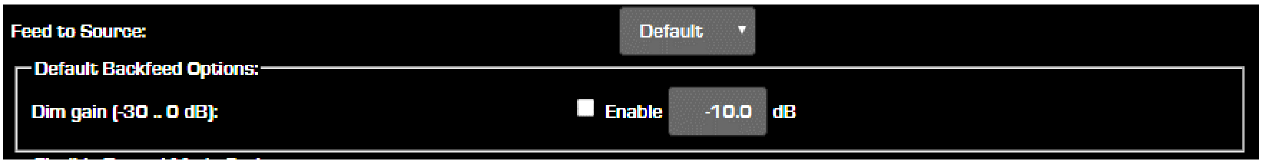
Line Gain allows the board operator to use the knob to quickly boost or cut the level of the xNode's Line Input stage, and compensate for audio that's too "hot" or too low.

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 Backfeed's 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.

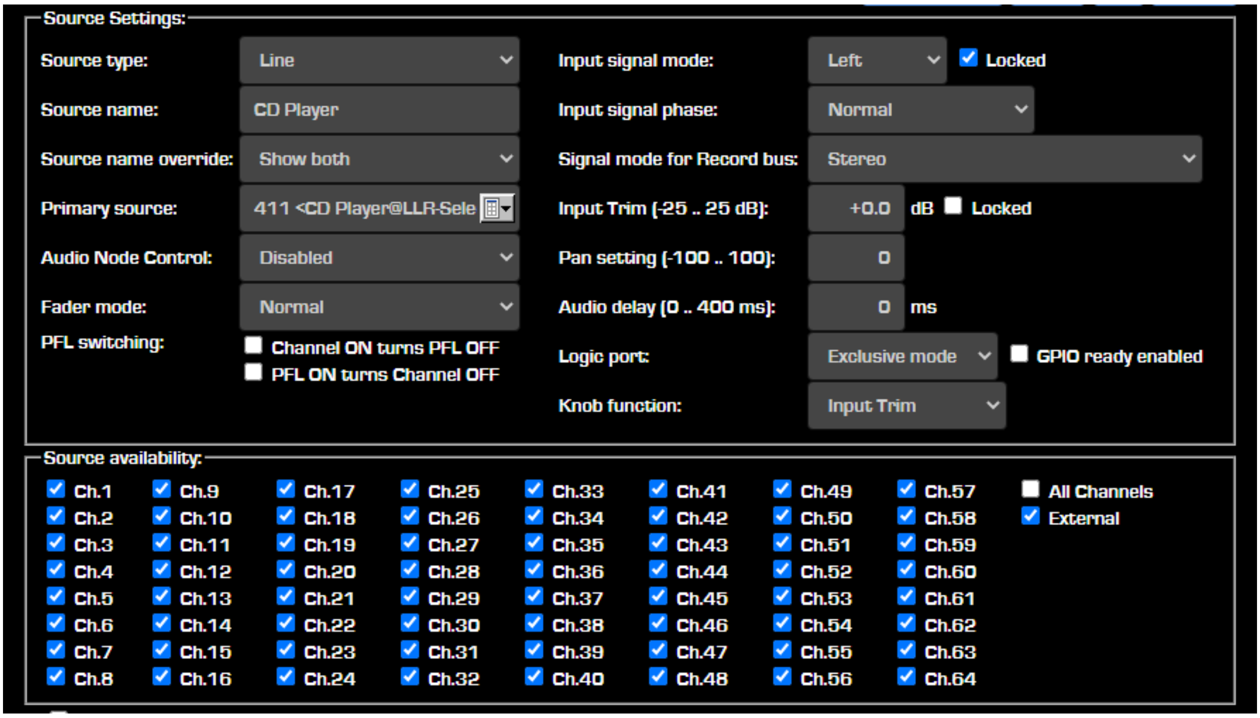
The **External** microphone source type is used for any microphone that will benefit from a headphone feed, and located in a space that does not require monitor speaker muting.

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. Setting up this Source Profile is just as easy as the previous two profiles. From the **Source Type** drop-down, select **Line**, then enter a name for the device.

In the **Primary Source** 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.

Now click the Ok button at the bottom of the screen; your CD player is configured and ready to be assigned to a fader.



The example shown above has the "External" box checked in the **Source Availability** section. Doing this allows the board op to assign the CD player to the monitors for direct auditioning.

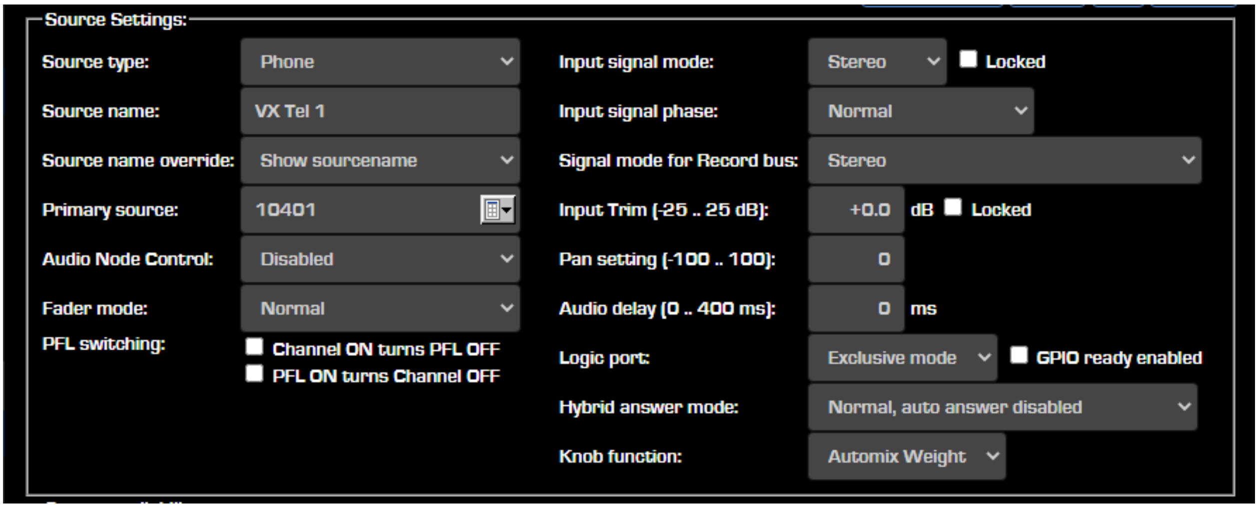
A more practical application for this option is when creating a Source Profile for an Air Monitor, so that talent can directly monitor the over-the-air broadcast signal. To do this, you'd create a **Line** source type for the air monitor receiver, and *uncheck* all the Channel availability boxes *except* 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)

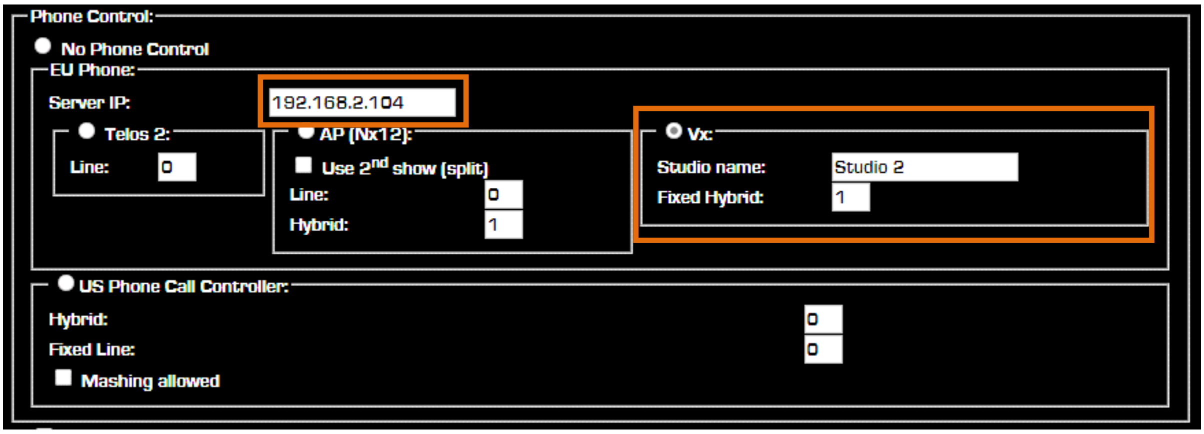
Putting phones on-air is one of the basic operations of the modern studio. Quasar's **Phone** Source Type helps ease the task of handling outboard phone hybrids.



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:

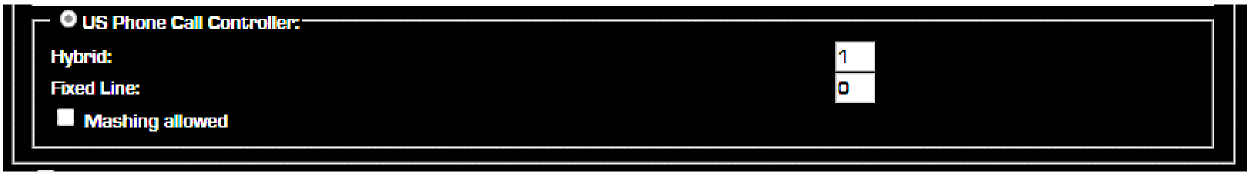
- Type in the IP address of the VX Server into the **Server IP** field
- Select the **VX** radio button.

- 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*.

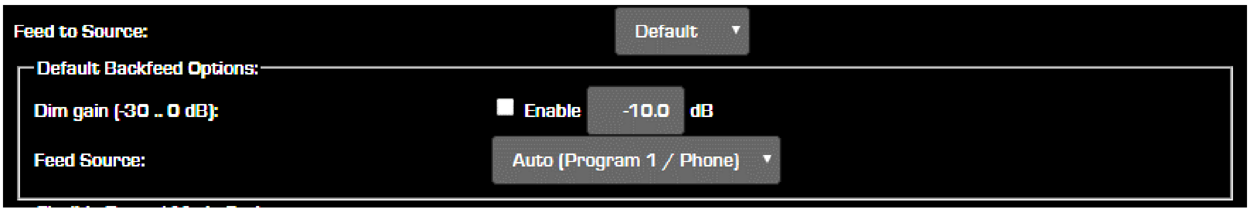
- **US Phone** ties the Source Profile to the Call Controller available in the Master module.

  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 pecifically 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. To configure this, you'll scroll to the **Default Backfeed Options** box and select the desired audio mix from the **Feed to 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 strips **PGM 4** key; the PHONE mix is then created pre-fader and pre-ON/OFF, so that pressing any **PGM 4** key will send the audio of that channel to the phone mix at unity gain.

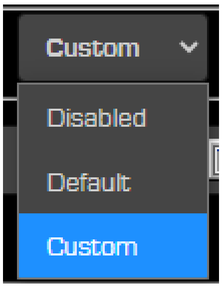
We suggest leaving the selection at its default, **Auto**, unless special circumstances dictate otherwise. No matter what you choose here, the board operator can quickly override it and select a different function, using the controls made available a the master modules touch display.
When your setup 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: "Working with 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.

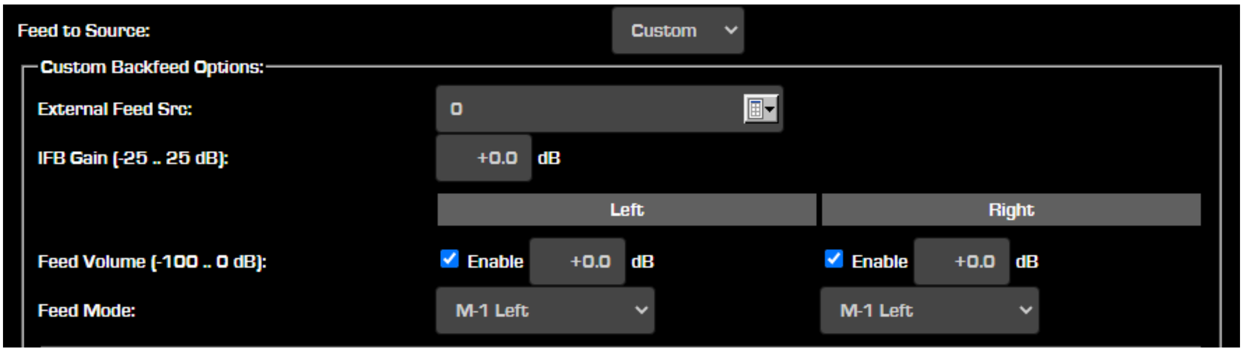


When you create a Source Profile from scratch, all Profile types have the **Feed to Source** option either "Disabled" or "Default". But your workflow might require a more specialized Backfeed than the Default behavior. That's what the "Custom" option is for: it allows high-level control of mix-minus behavior based on channel state logic.

As you can see from the previous page's screenshot, 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.

Below is the section of the Source Profile where Custom Feed to Source can be configured:



External Feed Src: allows you to pick any audio channel in your Livewire network to send to your source's backfeed.

IFB Gain: allows you to tailor the volume of the IFB audio with a cut or boost of up to 25 dB.

Feed Volume: 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 Mono Sum/Left/Right:** sends a mix-minus of the stereo Program output, minus the source, as either a summed Mono signal, or as the Left or Right channel of the stereo Program output, minus the source.
- **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.

  Notice that 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 include:

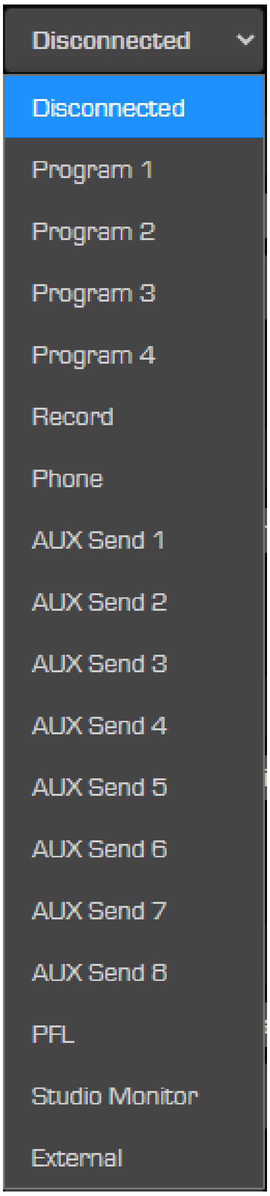
In normal operation Mode:

- 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, prefader and pre-on/off to allow speaker-phone style operation thru the Operator's mic. Selecting Phone feeds 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 Custom Backfeed Options box.

Note – The Record Mode is a special "macro" mode that helps talent record audio for later use with a single press of the Quasar's console's "Record" key, located on the Monitor Module. A detailed description of how to use this function in the Configuration Chapter – 5 of this Manual: Standard vs. Flexible Record Mode.

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, snapshots 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 and 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 <u>not</u> 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.

How To Create a Show Profile

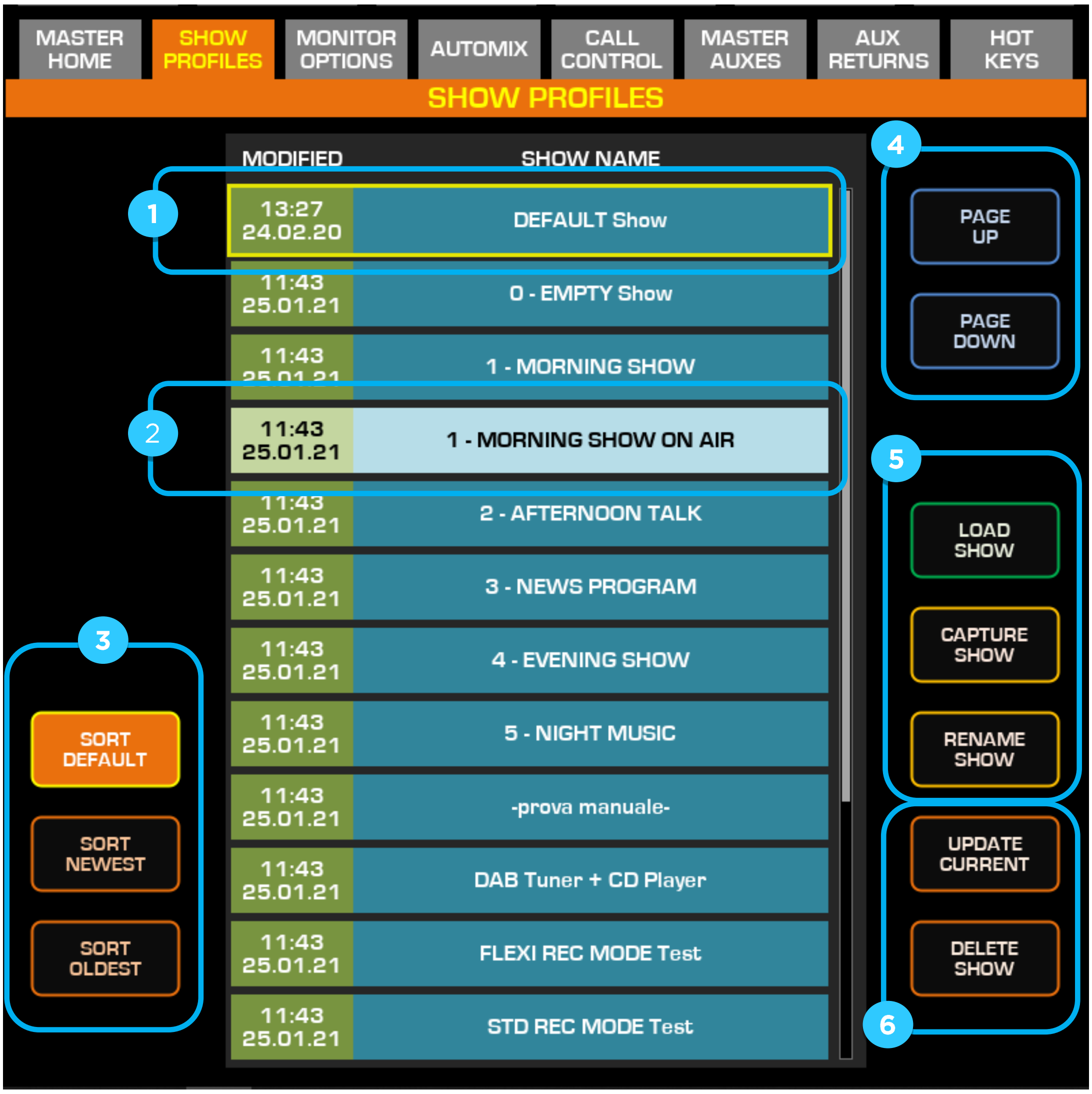
The easiest way to create a Show Profile is to set up the console the way you like it using the controls on the board itself – then use the **CAPTURE SHOW** button found in the SHOW PROFILES tab of your Quasar Touchscreen.
This will take a snapshot of the entire console – fader assignments, EQ, Dynamics, 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.

- Get started by assigning 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 your Monitor and Headphone audio choices 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 ouchscreen) and push the CAPTURE SHOW button: an on-screen keyboard will appear: type a name for your new Profile, and press OK .

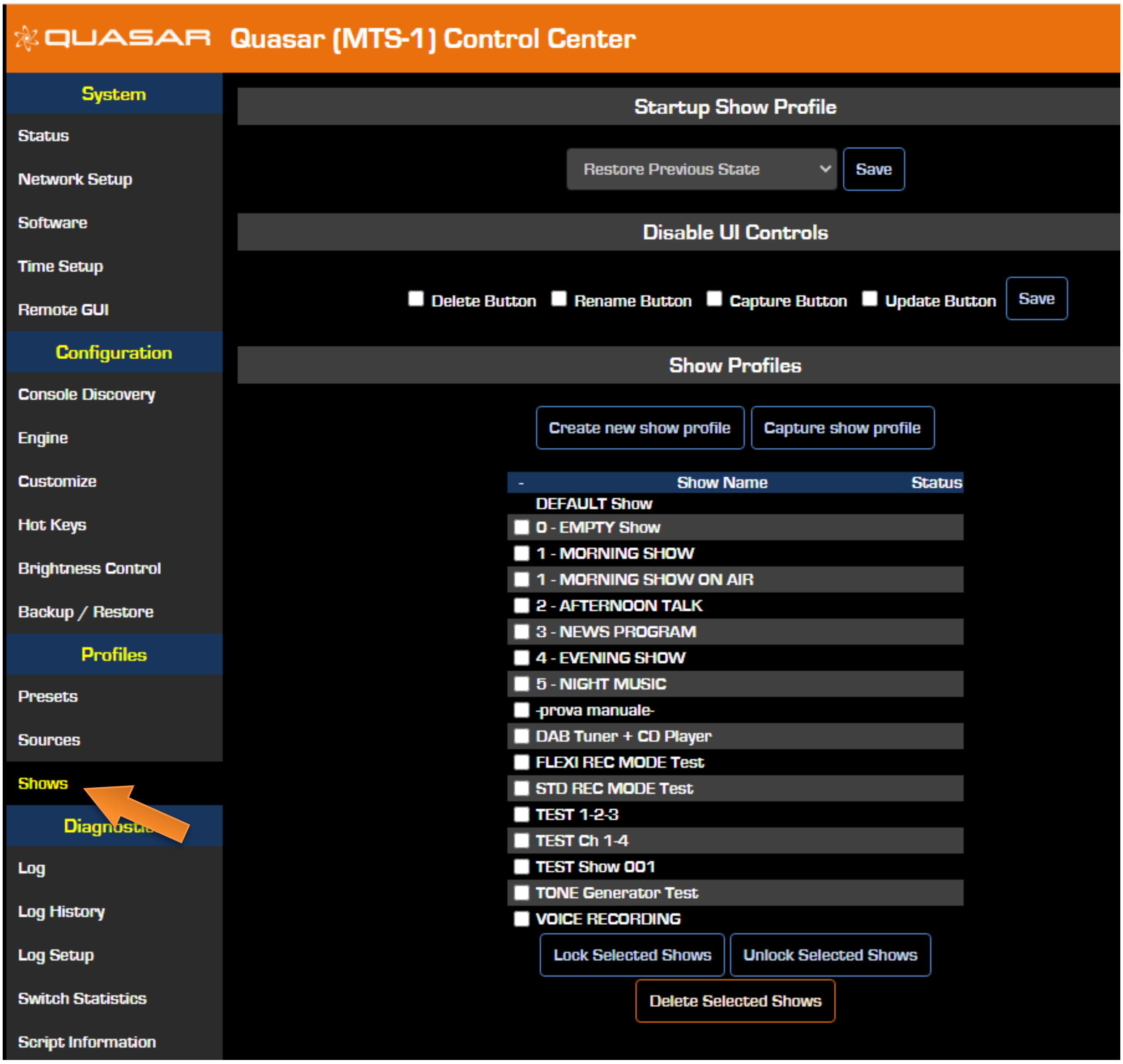
The picture below shows the Show Profile page on the Touchscreen UI. From a logical standpoint this is divided into the following areas:

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 (Deafault) or saving date/time.
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 in the Web UI.

Alternatively, you could enter your Quasar's IP address into a Web browser, and select the **Shows** link, as shown here. The Show Profiles menu page will be displayed:



Simply click on the **Capture show profile** button, and you'll be prompted to name your new Show Profile. Type in a name, click **Ok**, and you've got a new Show Profile that can be loaded by pressing the **PROFILES** key at the console's Master Monitor module.

The **Create new show profile** button allows you to configure an entire Show Profile completely from scratch 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.

In the **Disable UI Controls** section, you can enable or disable individual UI buttons. Disabled buttons will disappear from the console touchscreen, once the screen is refreshed.

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 explaned in detail later on in this chapter.

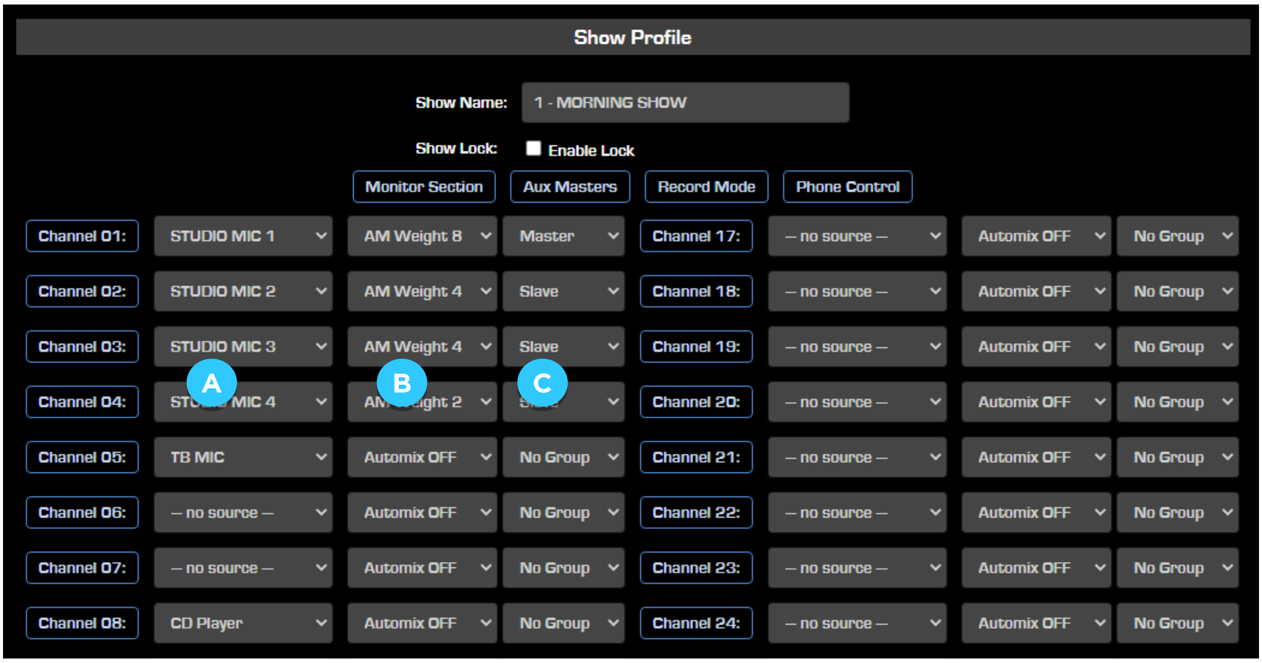
Remember to save the Show after you've made changes in any of these menus.

> **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 ad Weight parameter (B)
- set the **Group** activation and Master/Slave status (C)



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.

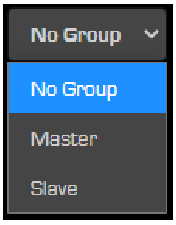
Automixer

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 this settings duplicated in eachof the Channel Options pages.

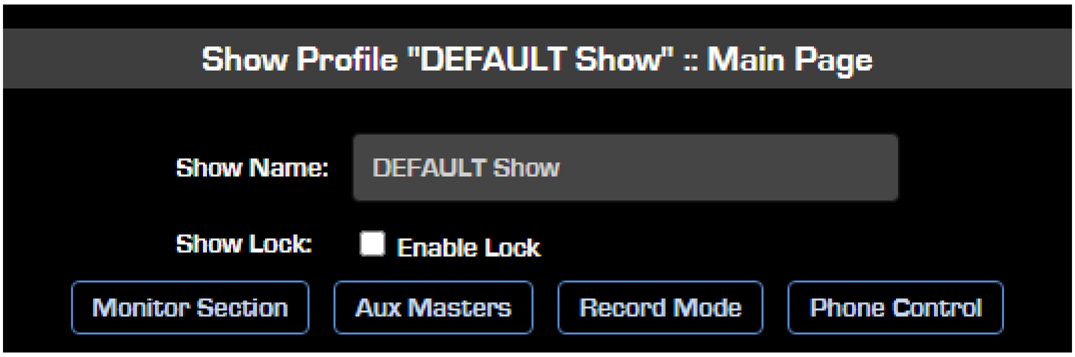


- **Master:** Designates this fader channel as a Group Start Master.
  Pressing its **ON** or **OFF** keys will turn Slave faders on and off as well.
- **Slave:** Designates this fader channel as a Slave. It will mirror the ON/OFF state of the Group Start Master fader.
- **No Group:** Normal ON/OFF operation.

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 four pages that can be accessed via the top menu in the Show Profile Main screen: **Monitor Section**, **Aux Masters**, **Record Mode**, and **Phone Control**.



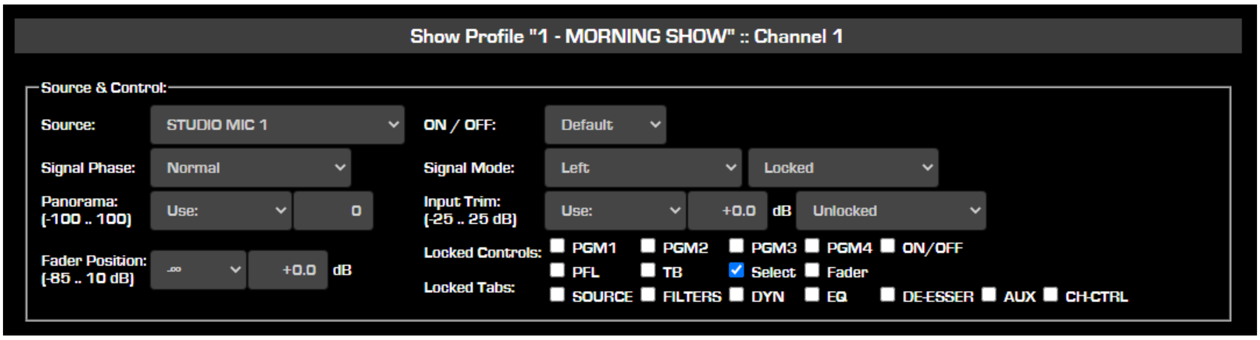
Channel Configuration Page

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

Signal 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 -85.0dB 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.

Signal 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 **Signal 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 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 (press the Option Knob) , and Fader, by selecting any of the checkboxes.

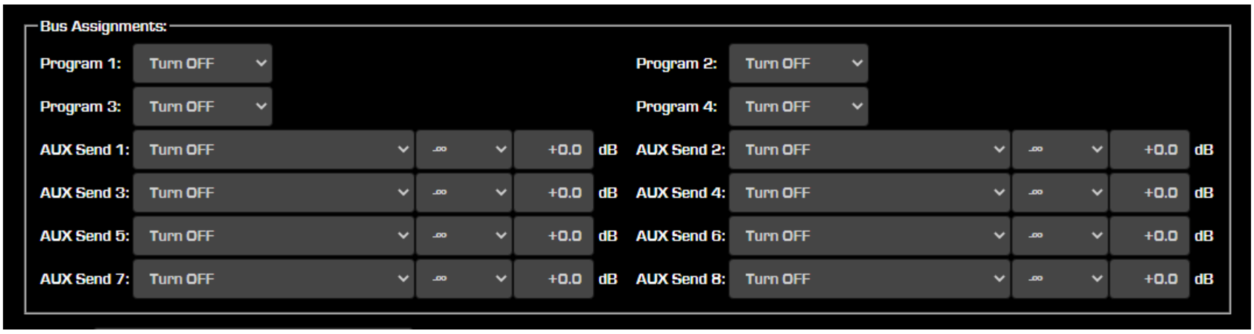
The above Locks will prevent the operator from changing the status of the selected parameters, from the console's Channel Strips. Only the Fader Module hardware controls (pusbuttons, faders, knobs) will be affected.

> **Note -** It will still be possible to control those functions from the Master Module, via its Touchscreen UI. For example, it will still be possible to control PGM bus activation by selecting 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



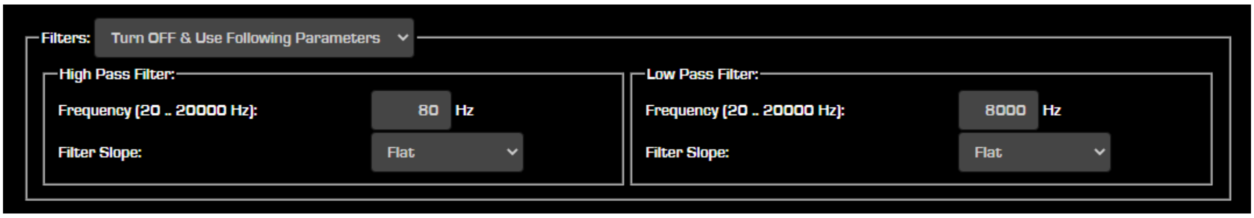
In this section, it is possible to pre-define the activation status of each of the four Program Busses, and each of the eight AUX busses, upon Show Profile load.

- 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



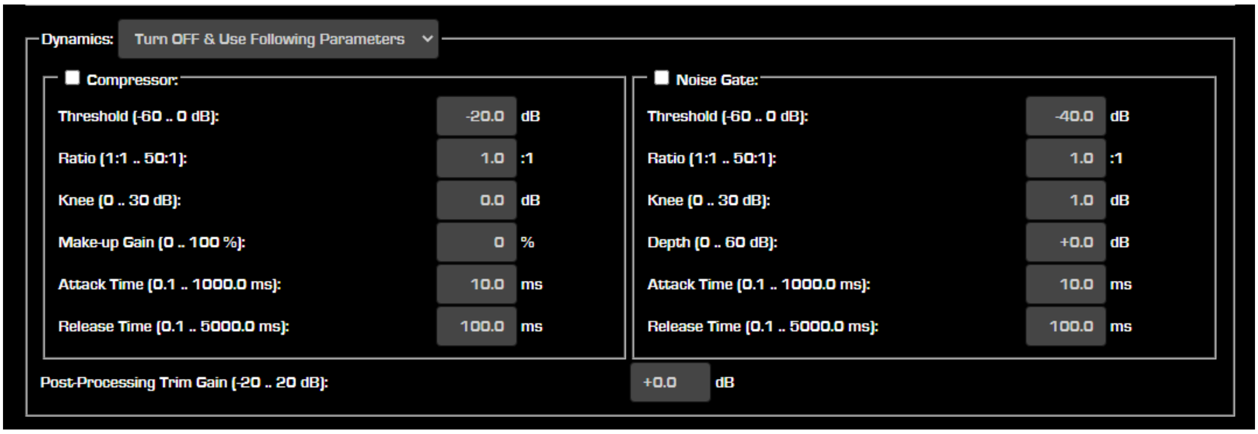
In this section, it is possible to pre-define the activation status and parameter settings of the channel's High-Pass and Low-Pass filters, upon Show Profile load. Three options are available:

- 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

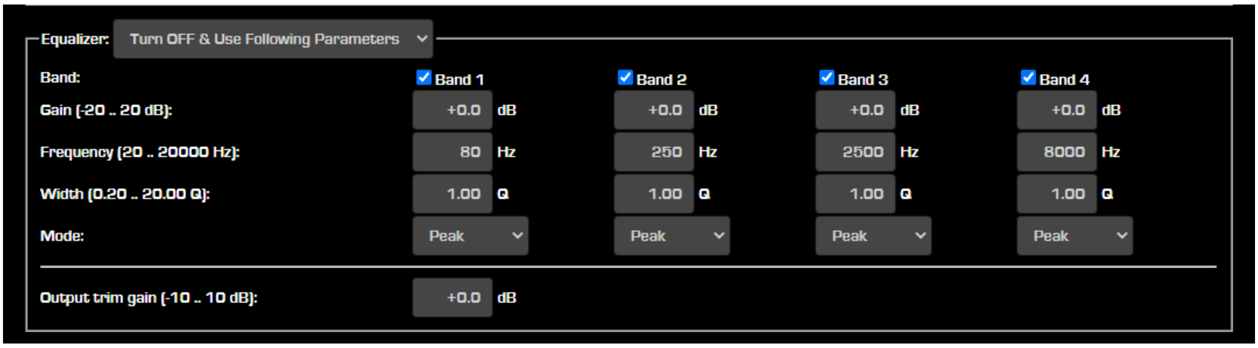
In this section, it is possible to pre-define the activation status and parameter settings of the channel's Compressor/Limiter and Expander/Noise Gate, upon Show Profile load.

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.

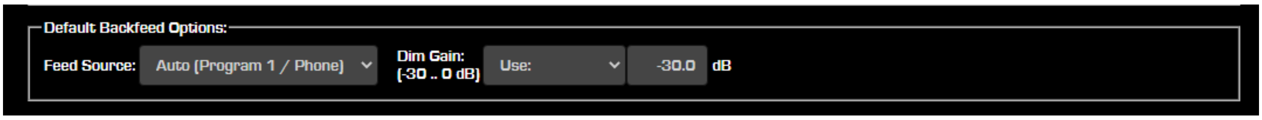
De-Esser

In this section, it is possible to pre-define the activation status and parameter settings of the channel's De-Esser, 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.

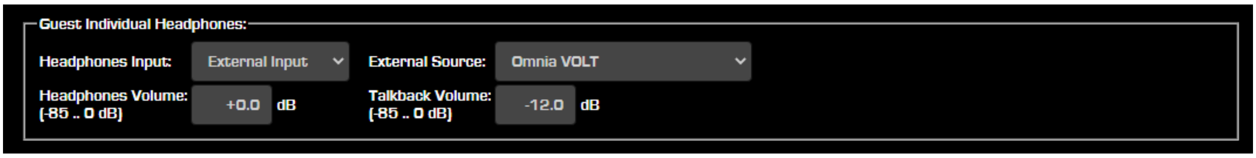
Any Quasar input defined as a Microphone source can have a dedicated headphone feed, to facilitate individual Talkback (IFB) to and from the board operator or other talent.
The options below affect this dedicated headphone feed, if enabled.

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

If in the previous menu "External Input" is selected as source for the Headphone Input, from this menu you can choose any of the Sources configured in the console, and feed it to the Guest Headphones.



Headphones Volume

It is posssible 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.

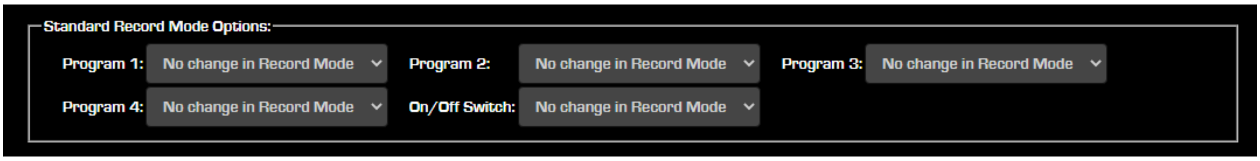
Talkback Volume (Low Limit)

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:** Prevents the operator from changing the channel ON/OFF state when Record Mode is active.

When done editing the Channel options, be sure to click the **Save Show** button.
You will be returned to the Show Profile options screen.

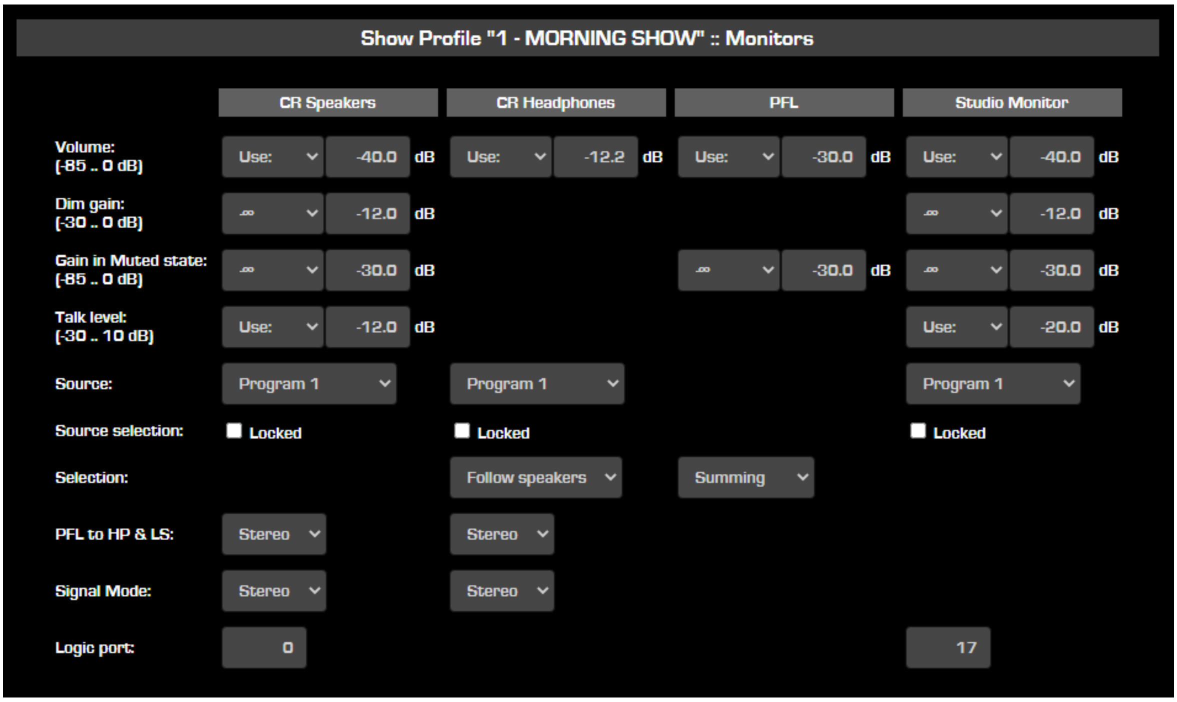
Monitor Section Configuration Page

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 by a fixed dB value. Enter the desired value in the next field. You may specify any level between -85.0dB and 0.0dB.

Dim Gain

- Select **Retain:** to keep the current Dim Gain level unchanged, when the Show is loaded.
  •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.

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 by 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 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. You may specify any level of cut or boost between -30.0dB and +10.0dB.

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 setting on the board from being changed if you want.
If desired, these options can be flagged to Lock the monitor Source selection, preventing the operator from changing the Show Profile's pre-selected options.

Selection (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.

PFL to HP & LS

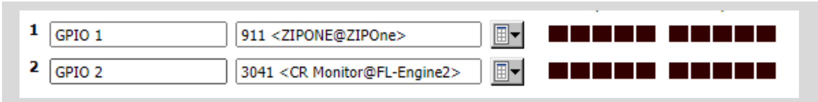
- Select **Retain:** from the drop-down menu to keep the current PFL monitoring status, when the Show is loaded.
- Select **Off:** to disable the PFL to Loudspeakers and PFL to Headphones selection
- Select **Split:** to retain the current selection on the left channel, and send a mono sum of the PFL bus to the right channel of the selected monitor section.

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

- Enter the GPIO channel number used to trigger On Air lamps or other GPIO functions, from the CR Speakers or Studio Monitor sections.



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. For simplicity, it is good practice to use the CR monitor LW id#, so that the IDs can be easily identified, and possible conflicts are avoided.

> **Tip -** This port can be used in conjunction with the EXT PFL Source selection to trigger the injection of an external signal into the PFL bus. 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 adjusts 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

EQ 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.

Feed to Source

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.

Automixer

This section allows to predetermine the Global Parameters for the entire Automixer. The individual Channel Parameters, (such as Activation and Weight) can be pre-set in the Show Profiles Main Page.

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.

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 (summing it, not switching it) into the PFL channel.

> **Note -** this must be enabled by a pin on the CR Monitor GPIO. Please see Appendix B for obtaining the wiring diagram pinout.

Example #1: you have an intercom system you wish to feed into the console's PFL bus. To do this, use the drop-down to select the intercom's audio source, then take the GPO from the intercom 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. To do this, use the drop-down to select the Playout Computer's PFL output channel, then take the GPO from the PC 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.

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, without the need for assigning the Operator's Microphone to one of the console channels.

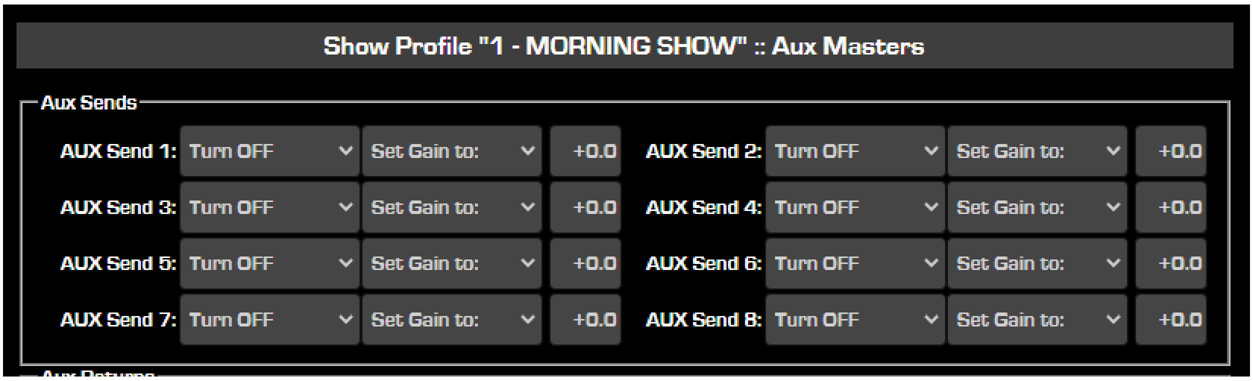
When done editing the Monitor Section options, be sure to click the **Save Show** button. You'll be returned to the Show Profile options screen.

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" channels, without any DSP processing feature.

Aux Master Sends 1-8



On/Off Status

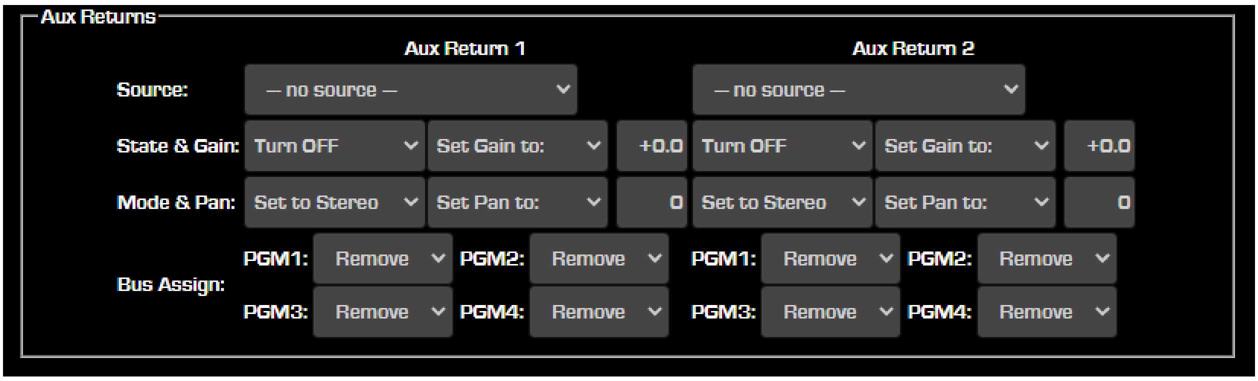
- Select **Retain State:** if you want the On/Off state to remain unchanged when the Show profile is loaded.
- Select **Turn OFF:** to switch off the selected Aux Send bus upon Show Profile load.
- Select **Turn ON:** to activate the selected Aux Send bus upon Show Profile load.

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.

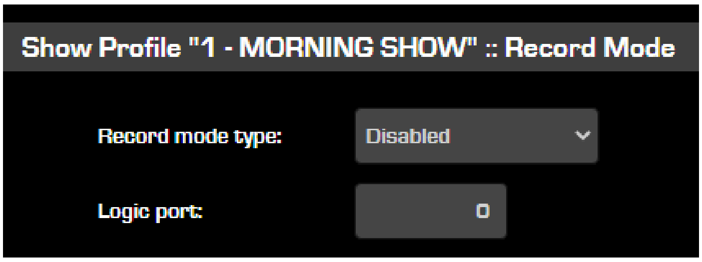
Tip - When done editing the Aux Send & Return options, be sure to click the Save Show button. You'll be returned to the Show Profile options screen.

Record Mode Configuration Page

From this section of the Show Profiles Web UI it is possible to set the desired type of Record Mode.

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

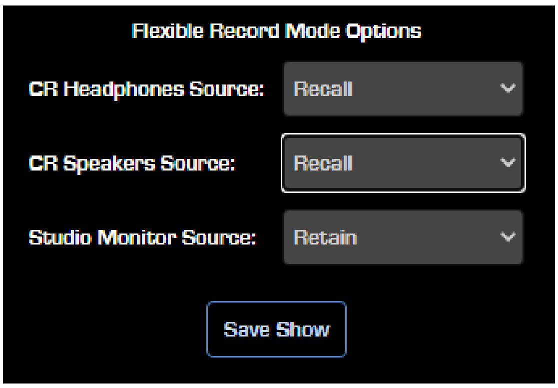
In this field you can enter the GPIO channel number used to trigger your dedicated recording device. This option should typically be programmed with the same logic channel number for all Show Profiles.

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.

Flexible Record Mode Options

When “Flexible” is chosen as the Record Mode Activation option, these options become active, allowing you to customize the Monitor and Headphone settings that are automatically selected when Record Mode is entered. The initial (original) Show Profile settings will be loaded upon exiting the Record Mode.

These options 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 Profile (see Source Profiles section of this chapter).



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

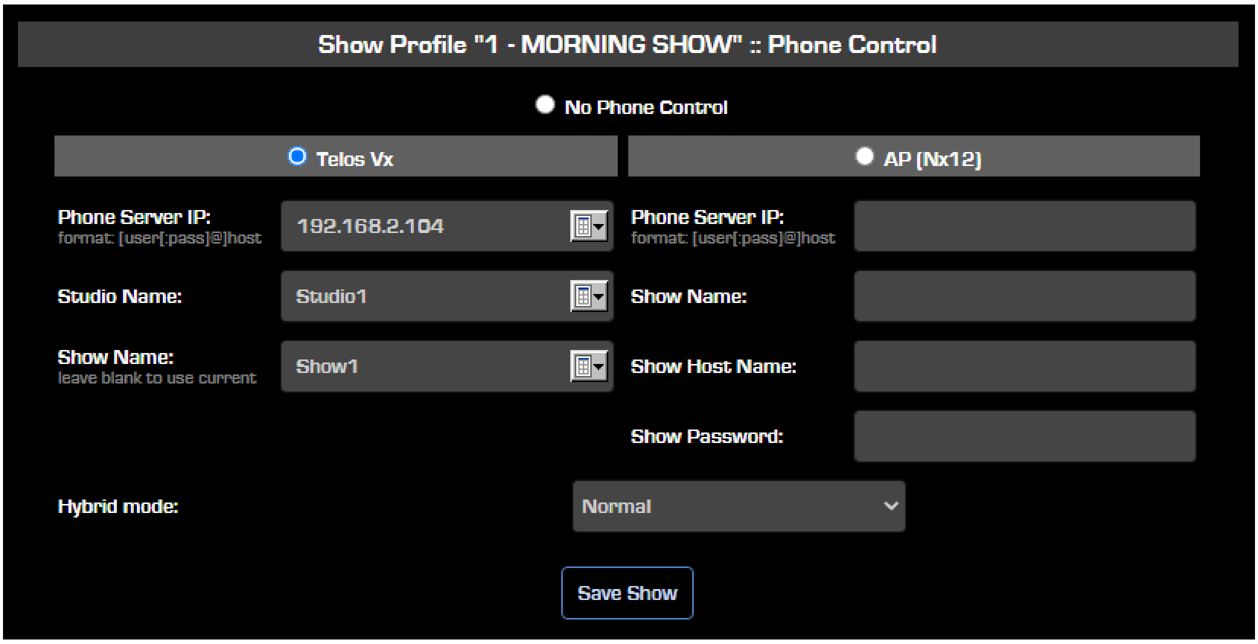
The same options as described in the previous menu are available for the Control Room Speakers section.

Studio Monitor Source

The same options as described in the CR HP menu are available for the Studio Monitor section.

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) . 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 reverses the order.

Please refer to the next Chapter, "Working With Phones", for more details.

Configuration - 3

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 your Quasar console 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 selction 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.

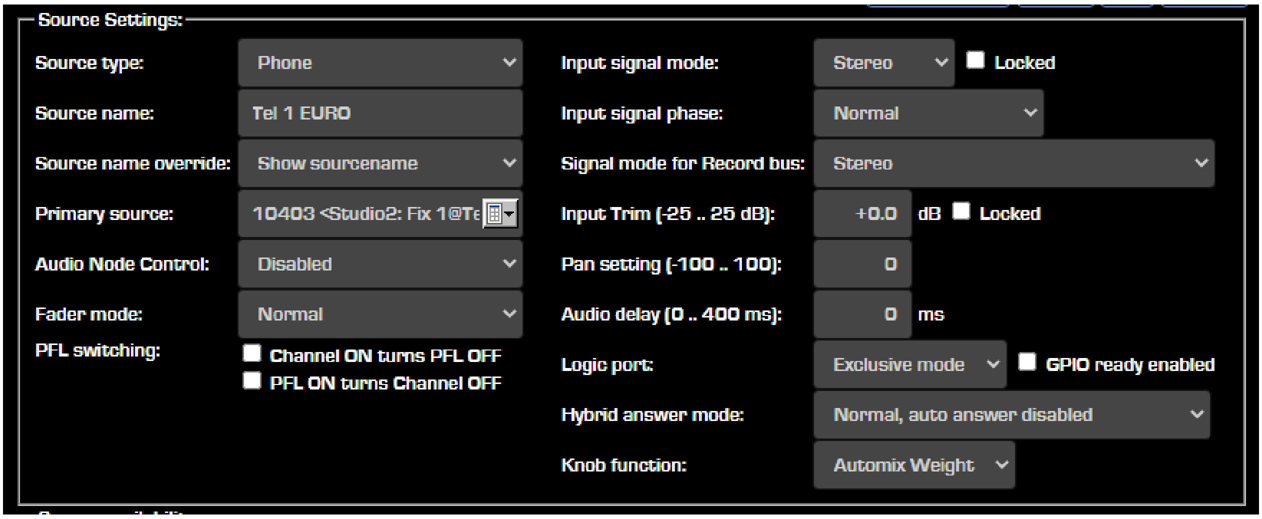
No Phone Control is used for controlling Telos Hx1 and Hx2 products , or third-party hybrids which do not support Telos control protocols. In this second case, basic GPIO closures are used to "take" and "drop" lines.

EU Phone Operation Setup

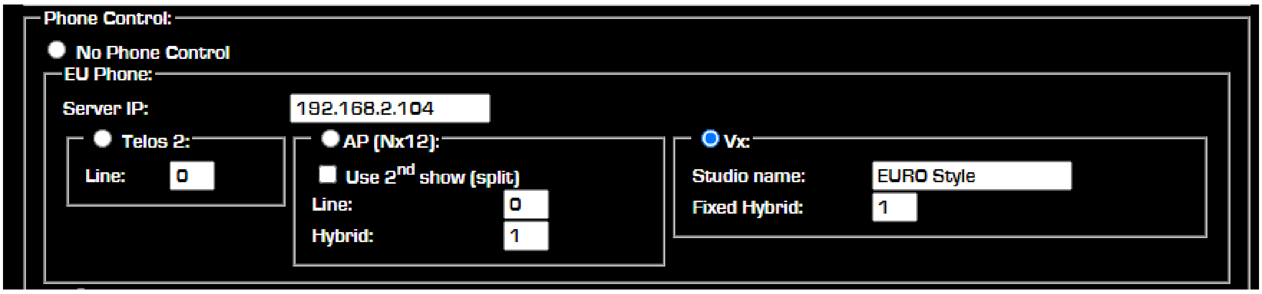
In this case you only need to create a Source Profile for each of your fixed line hybrids. In legacy Telos Phone systems, you may have 1, 2, or even 4 hybrids; with VX systems, the number may be many more.

Source Profile Settings for EU Phone Operation

1. Select the **Phone Source** type.
2. Select the appropriate **Primary 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. 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 2nd 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.

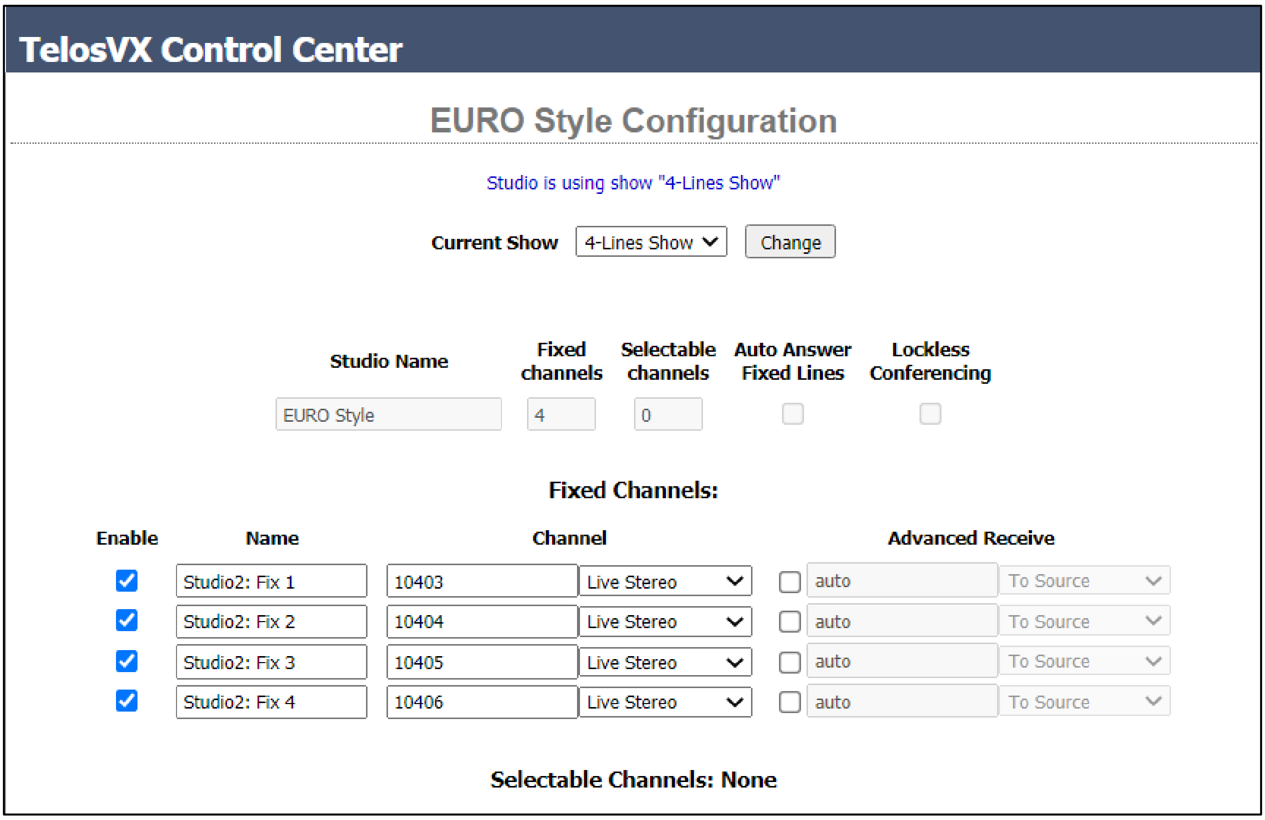
Telos VX settings for EU Phone operation

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:

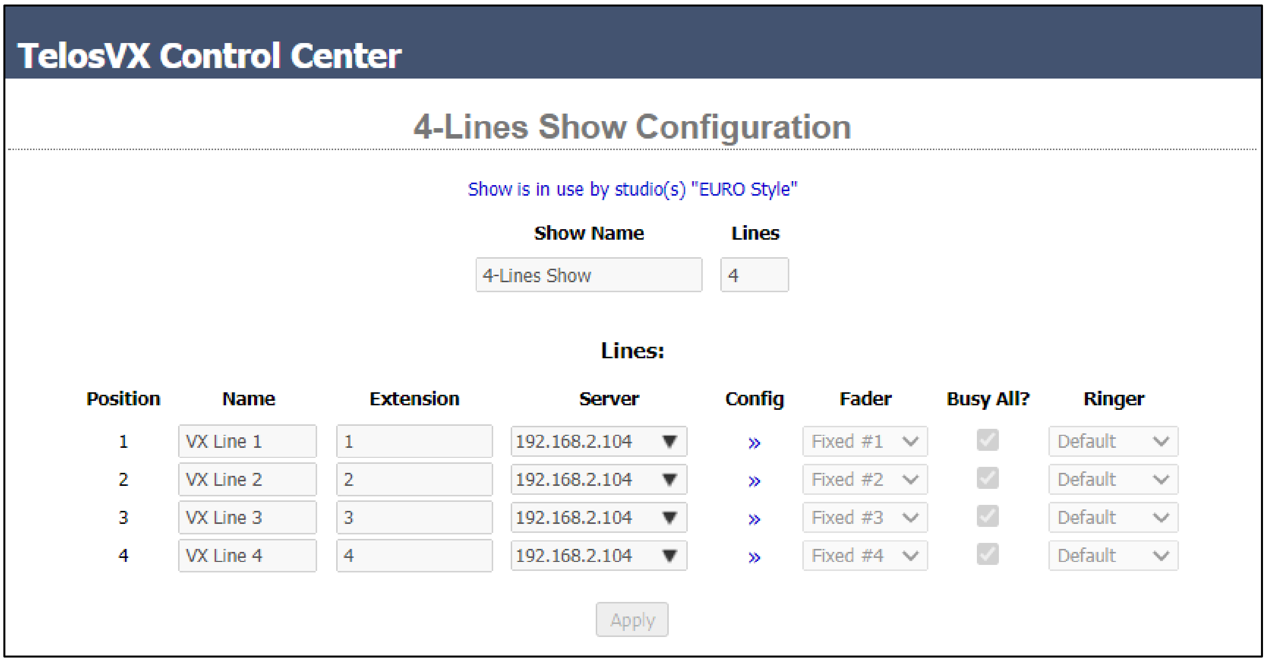
In VX, you will need to:

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.

Below is 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:



Following, is how the VX Show associated to this Studio could look like:



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

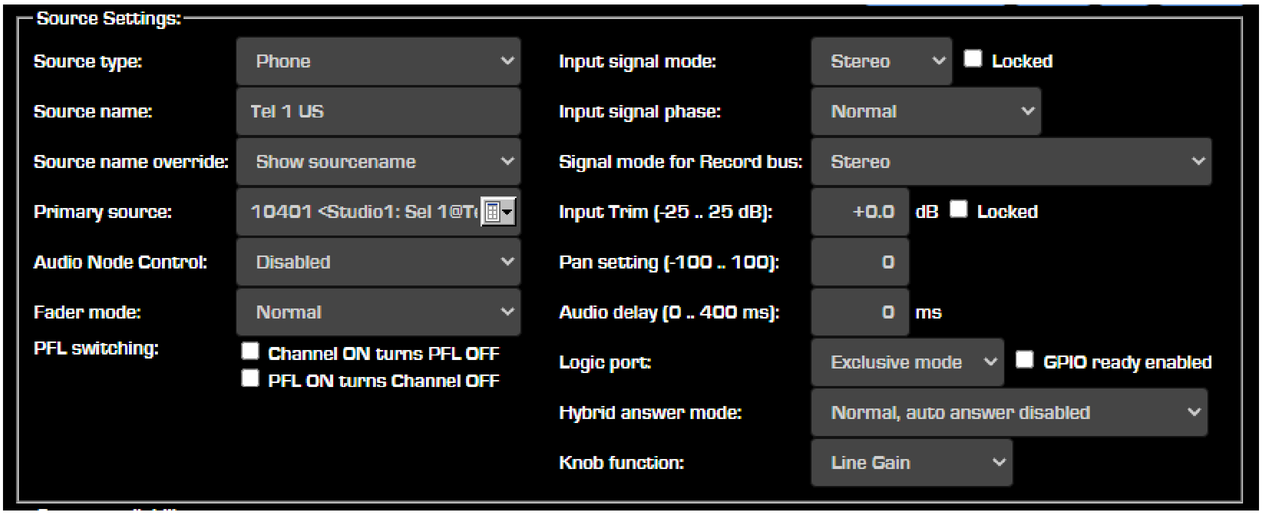
Using the US Phone method of operation requires the use of the Quasar Call Controller: this soft module lets you control up to 8 incoming lines from 2 faders, in a hghly flexible way. With this method, you can use Quasar's Show Profiles feature to instantly recall show setups that choose between different 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.

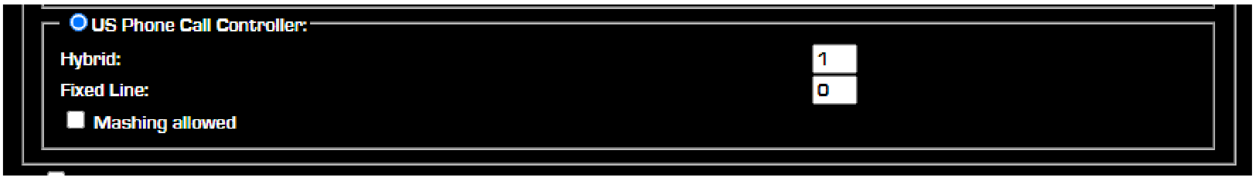
Source Profile Settings for US Phone Operation – VX & Nx Systems

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. Select the **Phone Source** type.
2. Select the appropriate **Primary 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.



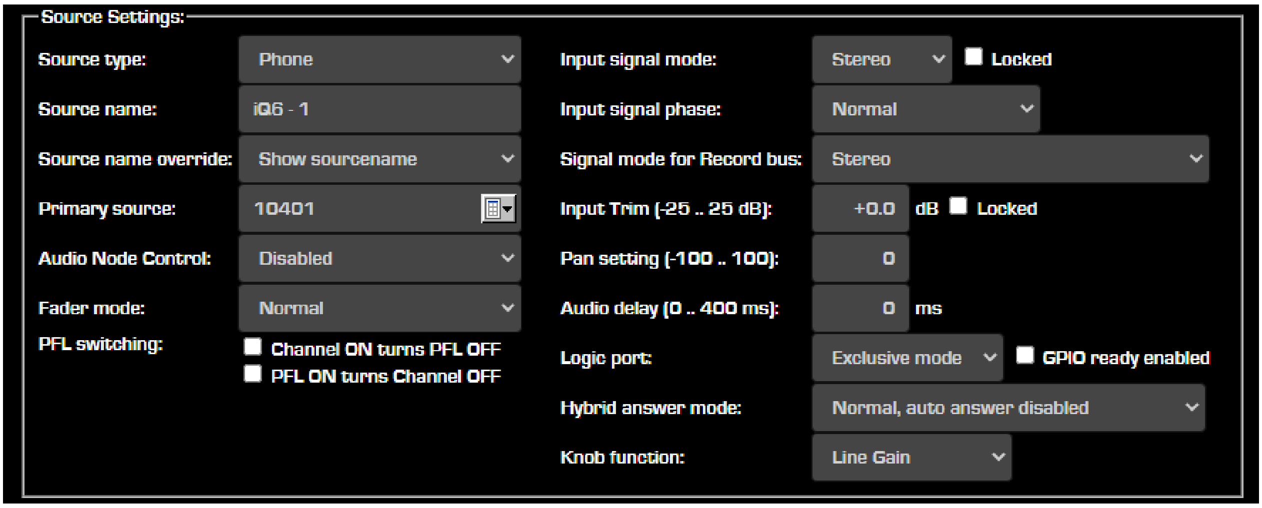
Note - If your Phone Source uses a Fixed Channel from the VX, you must set the **Fixed Line** parameter to any number different from 0 (it doesn't matter which number). But in this case you will not be able to control this line from a Call Controller, but directly from the console fader strip. You will 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.

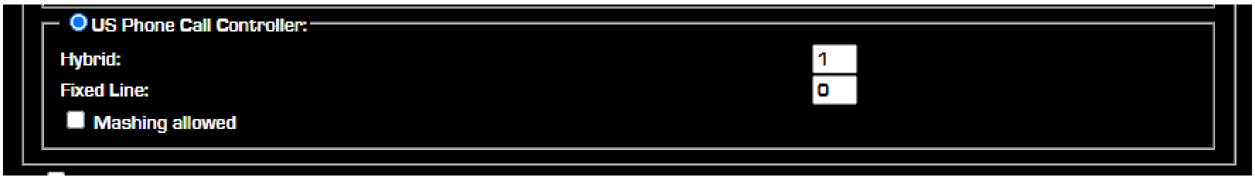
Source Profile Settings for US Phone Operation – Hx6 & iQ6 Systems

As with the other systems discussed here, you'll begin by creating a Source Profile for each of your Hx6 or iQ6 hybrids.

1. In the Source Settings, select the **Phone Source** type.
2. Select the appropriate **Primary 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).

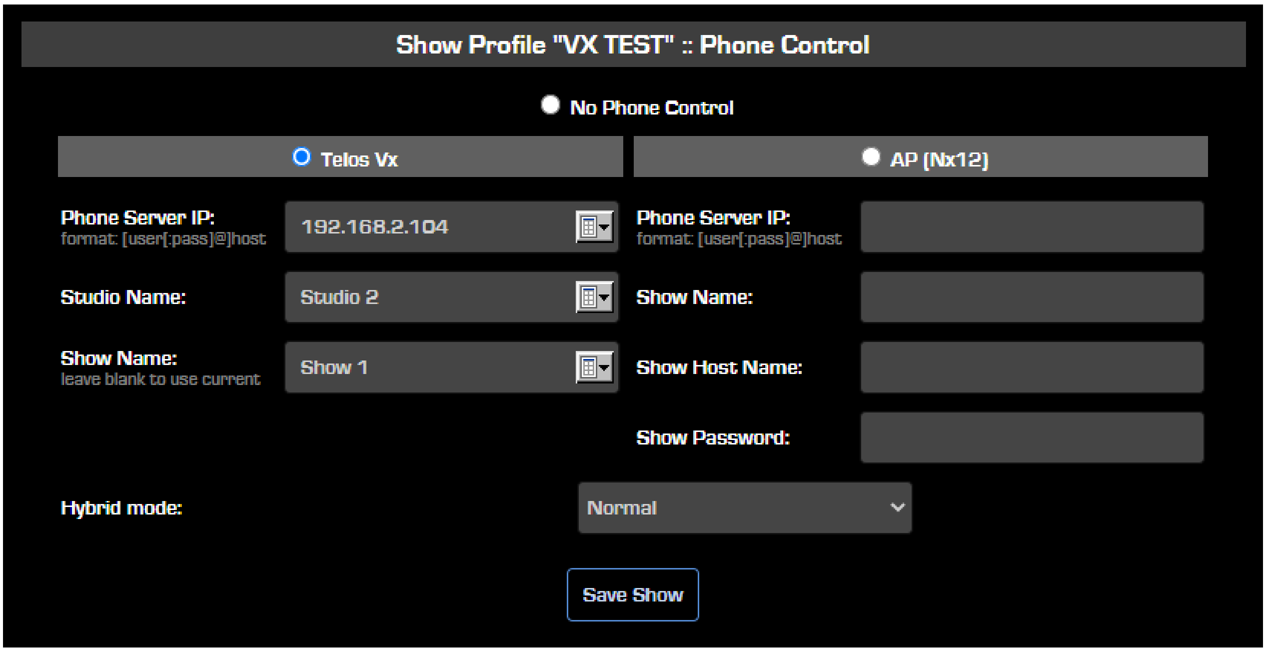
Show Profile Settings For US Phone Operation

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:

The **Phone Type / No Phone Control** 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, otherwise you would likely select the VX option.

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.

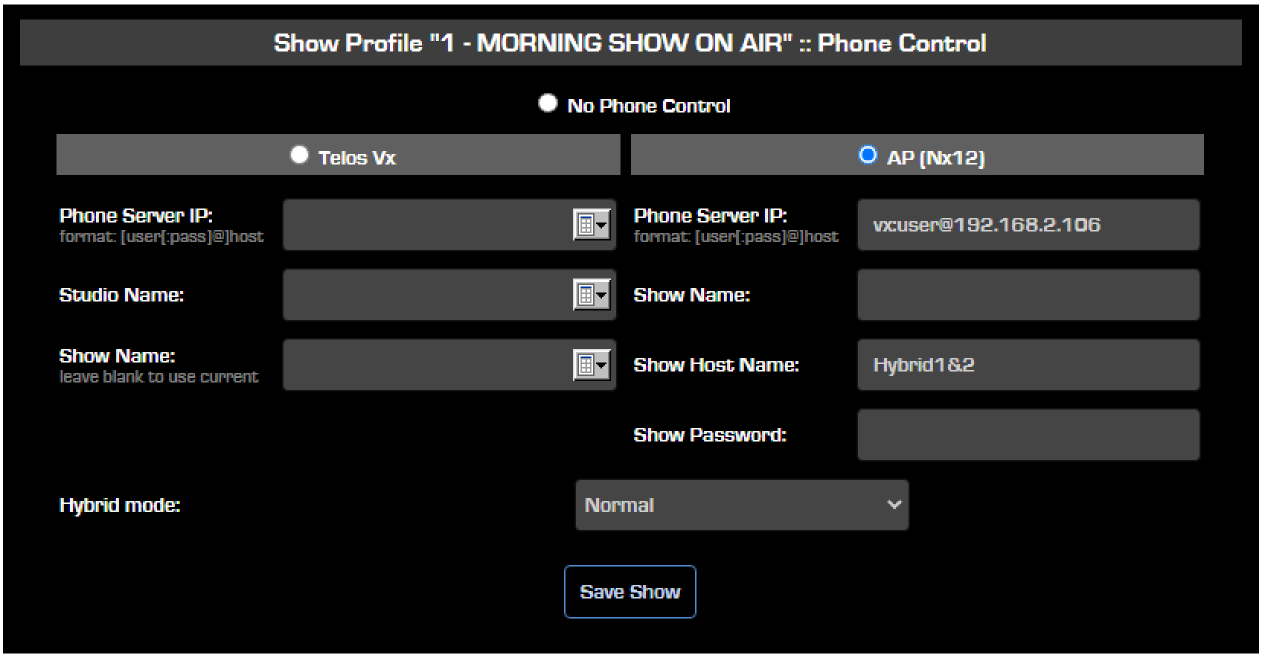
The **Show Name** field is used to identify a phone system configuration when using a Telos Series 2101 or VX system. These names are defined in your phone system's setup. Enter the configuration name you wish your Call Controller to log into.
With the VX system, this field can be left blank, allowing 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 **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 line 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.

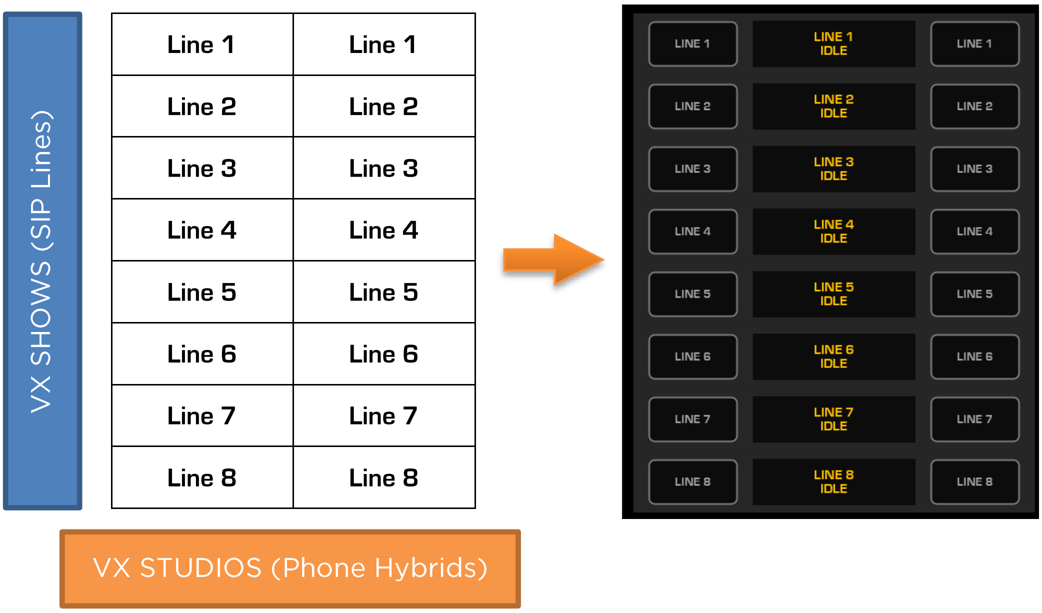
> **Tip -** When selecting a line directly using the Call Controller, the default behavior is for the Left column of buttons to assign the line to Hybrid 1, and the Right column to assign the line to Hybrid 2. But when using phone systems with more than two hybrids, you may assign lines to hybrids other 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.

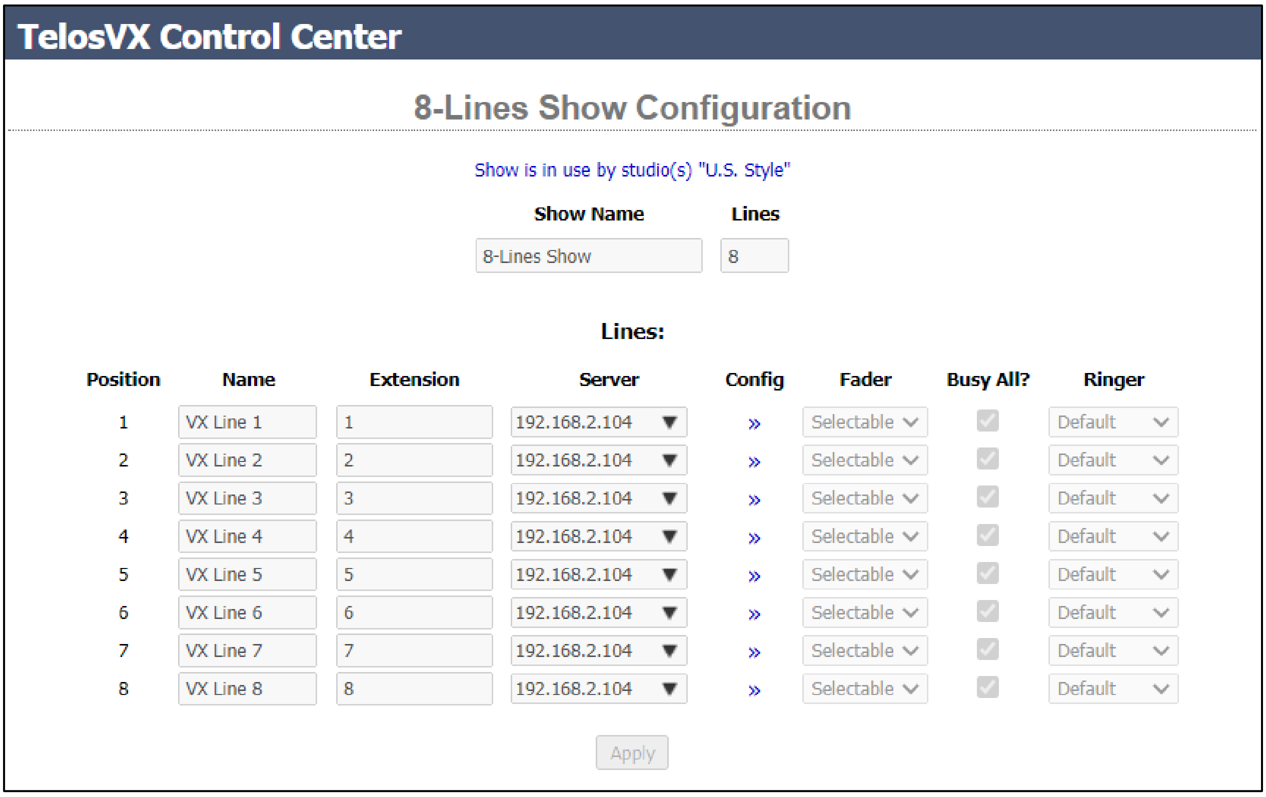
In VX, you will need to:

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"

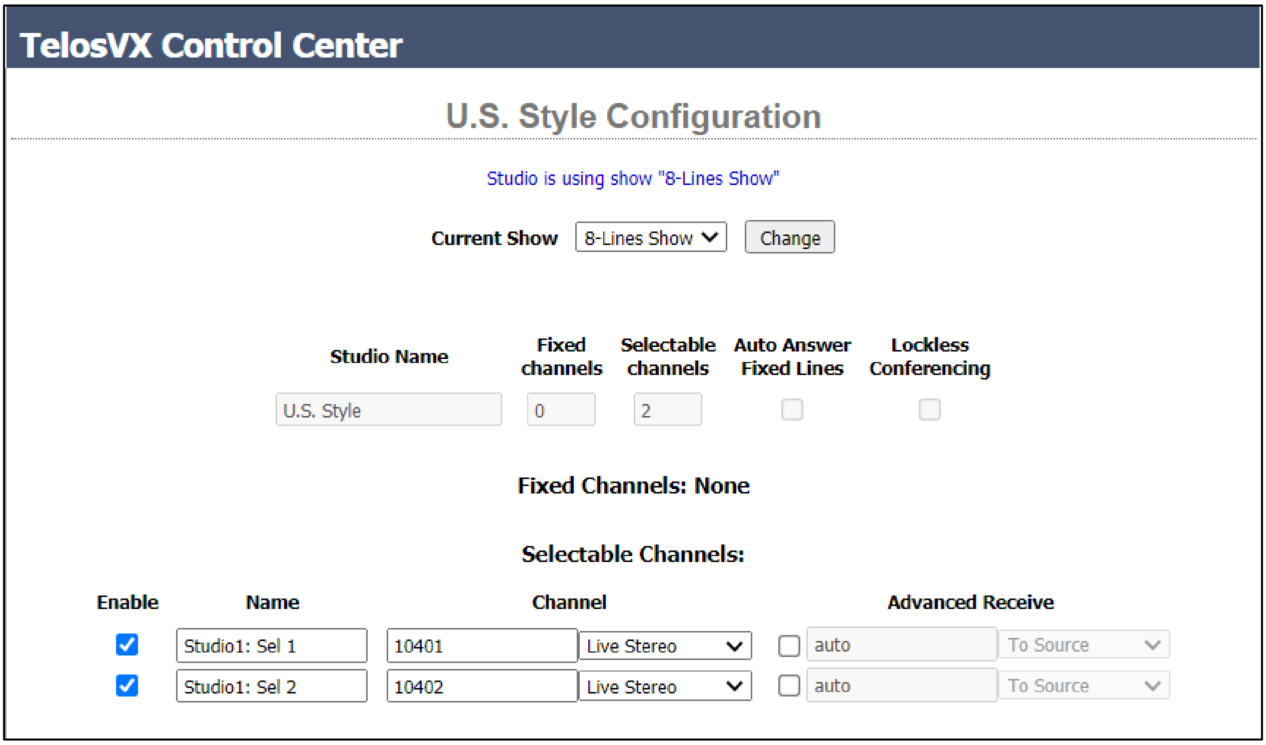
Show Name	Lines
8-Lines Show	8

Lines:

Position	Name	Extension	Server	Config	Fader	Busy All?	Ringer
1	VX Line 1	1	192.168.2.104	»	Selectable	✓	Default
2	VX Line 2	2	192.168.2.104	»	Selectable	✓	Default
3	VX Line 3	3	192.168.2.104	»	Selectable	✓	Default
4	VX Line 4	4	192.168.2.104	»	Selectable	✓	Default
5	VX Line 5	5	192.168.2.104	»	Selectable	✓	Default
6	VX Line 6	6	192.168.2.104	»	Selectable	✓	Default
7	VX Line 7	7	192.168.2.104	»	Selectable	✓	Default
8	VX Line 8	8	192.168.2.104	»	Selectable	✓	Default

Apply

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 8-Lines Show Change

Studio Name	Fixed channels	Selectable channels	Auto Answer Fixed Lines	Lockless Conferencing
U.S. Style	0	2	☐	☐

Fixed Channels: None

Selectable Channels:

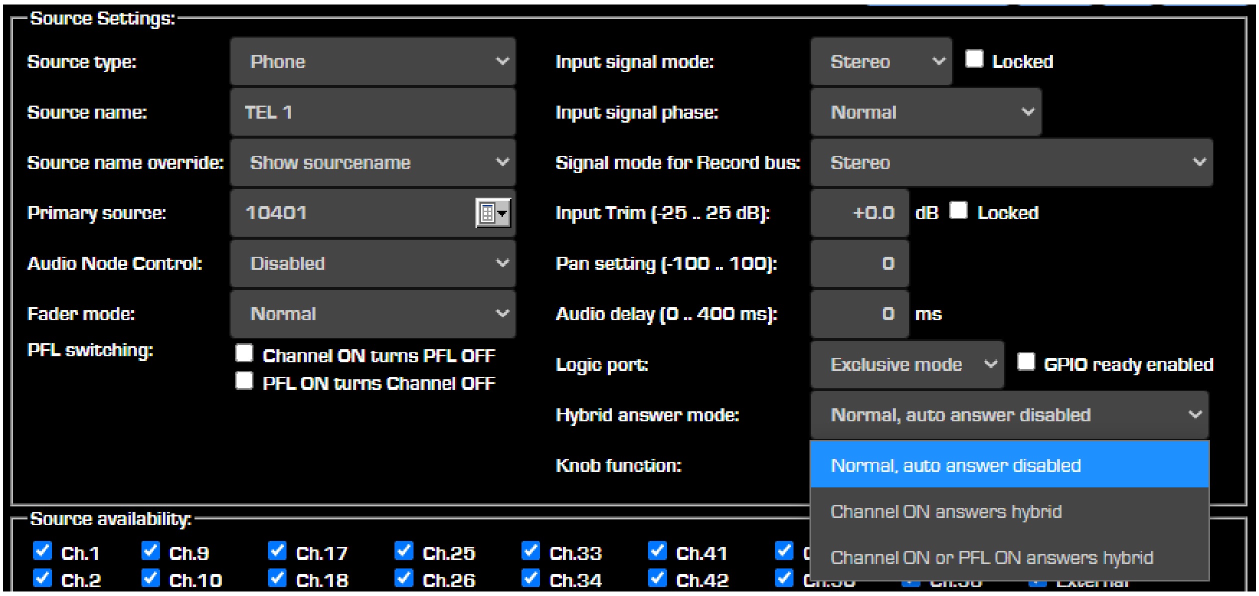
Enable	Name	Channel		Advanced Receive		
✓	Studio1: Sel 1	10401	Live Stereo	☐	auto	To Source
✓	Studio1: Sel 2	10402	Live Stereo	☐	auto	To Source

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 salient 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.

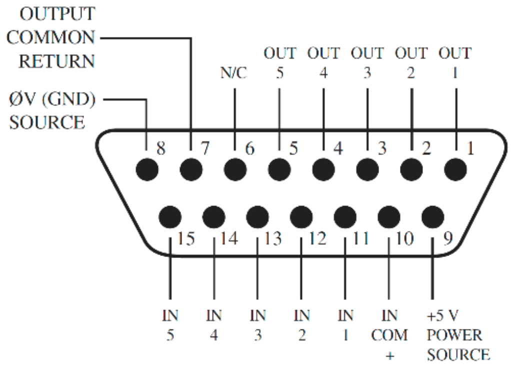


The default setting for Hybrid Answer mode is **Normal, auto answer diasbled**. If your desire is to achieve pulses on pins 4 and 5 of the GPIO logic port associated with your phone hybrid, then you must choose one of the two other options. These other options will create pulses when the channel changes 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.

Name	Pin	Type	Notes
INPUTS			
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 indi-cate source's readiness
OUTPUTS			
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 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.
POWER & COMMAND			
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 Layers

Quasar 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 4 user-definable combinations.

Layers are available only to Quasar consoles equipped with XR-4FAD Fader Modules.

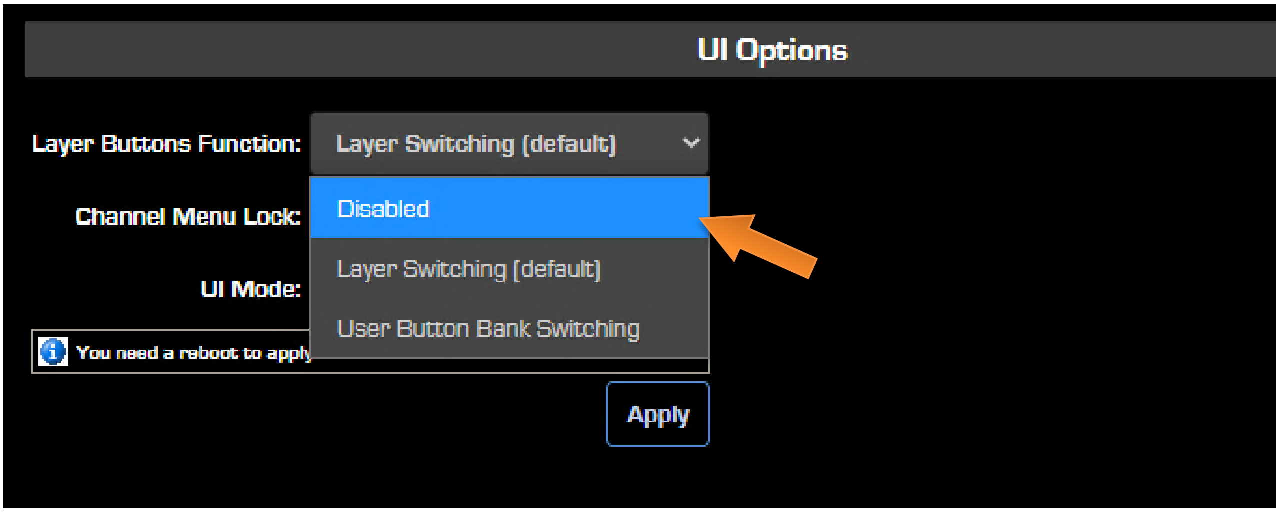
The Layer Concept Explained

The Quasar Surface can be ordered with a variable number of faders strips, while the Quasar Engine has a fixed number of DSP Input channels. In fact, a Quasar Engine (at the time of writing this manual) comes with up to 64 DSP channels.

Since there is no longer a 1:1 correspondence between fader strips and 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 four Layers can be configured on a surface.

Layers are enabled by default. If you use Layers, proceed to the next section **Configuring Layers.** If you do not use Layers, make sure they are disabled by following these steps:

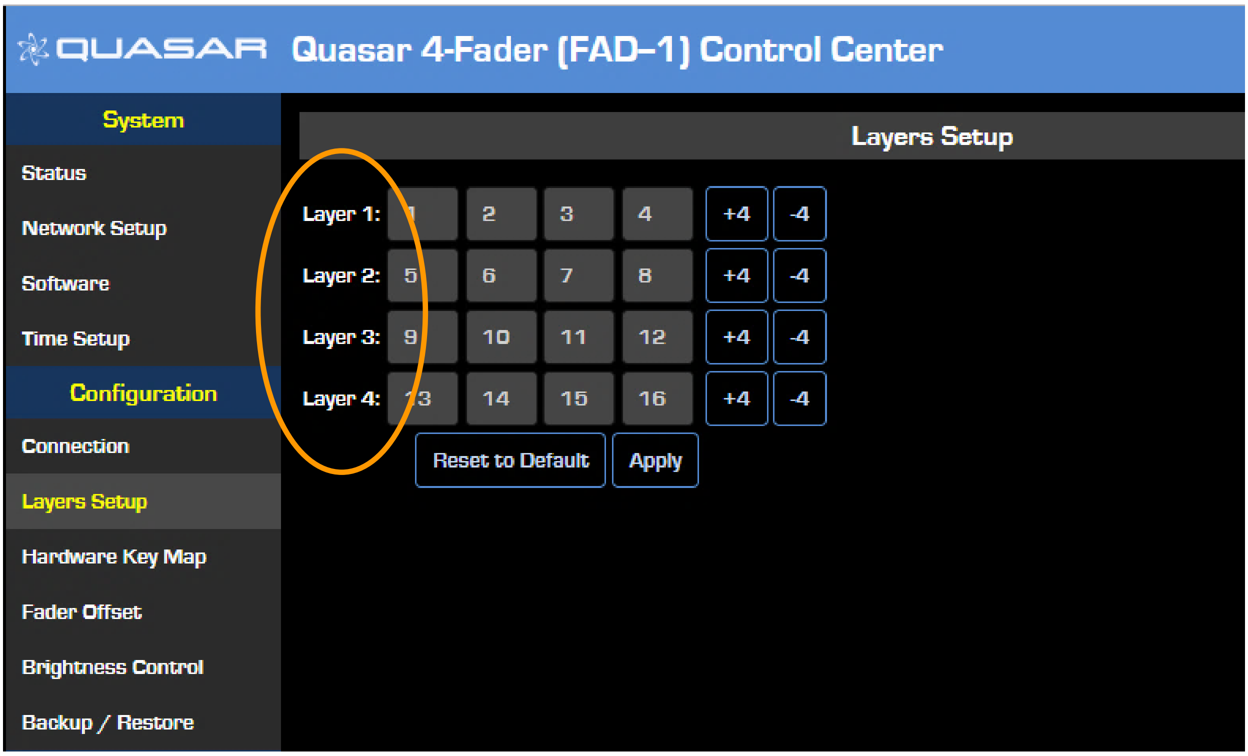
In order to disable Layers, connected a PC to your network, launch a web browser, and enter the IP address of your MTS module. Under the **Configuration** header, click on **Customize**.
In the **UI Options** section at the bottom of the page, select **Disabled** from the drop-down list, then click **Apply** .



Configuring Layers

Quasar ships 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 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.

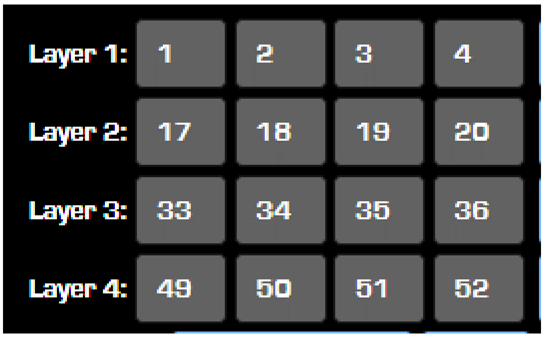
With Layers, you could also mirror two or more fader modules by setting both modules to control the same input channel, or channels, providing that they belong to the same logical surface (connected to the same Master). And it doesn't matter where they are physically installed - they can be both in the same surface frame or be distant from each other, as long as they are on the same LAN.

So, the fundamental difference between a Show Profile and a Layer is the following:

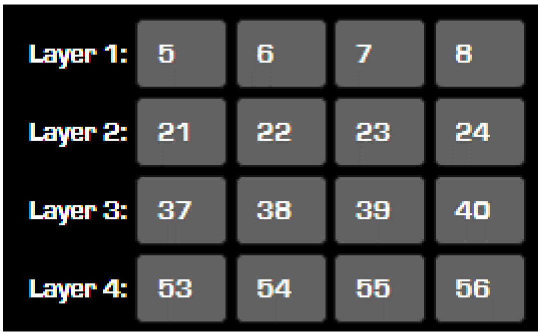
- **Shows** are used to map **input sources** from your network to the **input channels** of your Engine.
- **Layers** are used to map the **input channels** of the Engine to the physical **fader strips** of the Surface**.**

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:

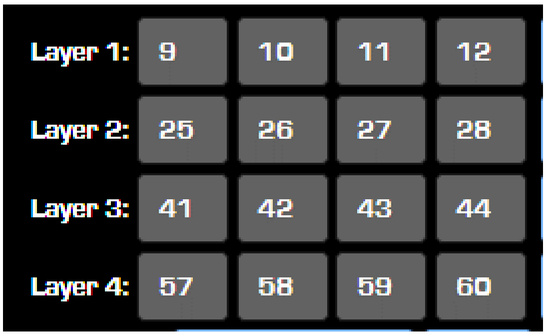
Fader Module # 1



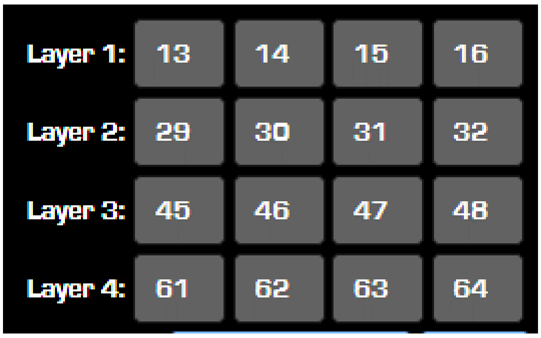
Fader Module # 2

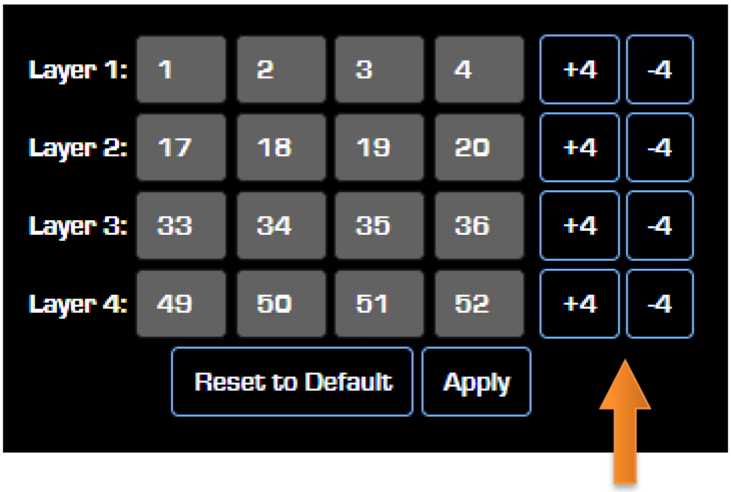


Fader Module # 3



Fader Module # 4





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 function and Pathfinder-defined function.

> **Note -** Pathfinder refers to them as "buttons" (being soft), so we could refer to hw keys as "buttons", only in this chapter.

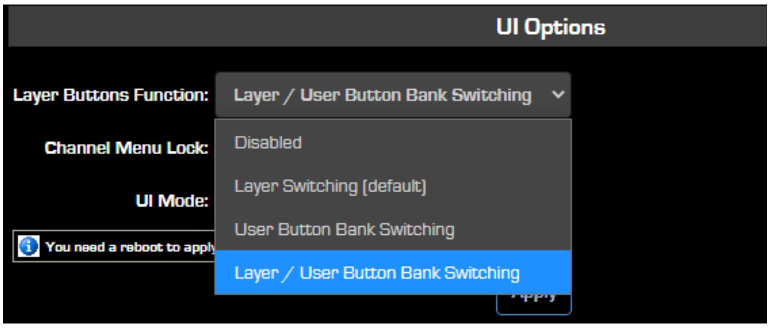
1. A Predefined function is a shortcut to a function which already exists in the console, but is simply not available directly from its hardware keys. The keys can be configured from each module's Web UI, and their function selected from a drop-down list.
2. A Custom PF function is a function defined with Pathfinder. When a User button is defined as Custom PF in the web UI, it becomes a PF button.

With the release of software version 1.2 and following, Quasar offers 4 Banks of eight User-Programmable keys in the MTS module, 4 Dedicated User-Programmable keys per channel strip, in the XR-4FAD module, and the possibility to remap any of the Fader module keys, for those users who need a high level of customization.

MTS-MON Module User 1-8 Buttons

By default, Quasar comes with the first bank (Bank A) of User 1-8 buttons enabled, as well as the Layer Switching function enabled.

The User buttons found in the Quasar MTS module belong to the second class, and are PF-defined only.



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**. You can choose two enable options from the **Layer Button Function** drop-down list at the bottom of the page:

- **User Button Bank Switching** will enable all the user Banks, and disable Layer 2,3,4 selection. 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

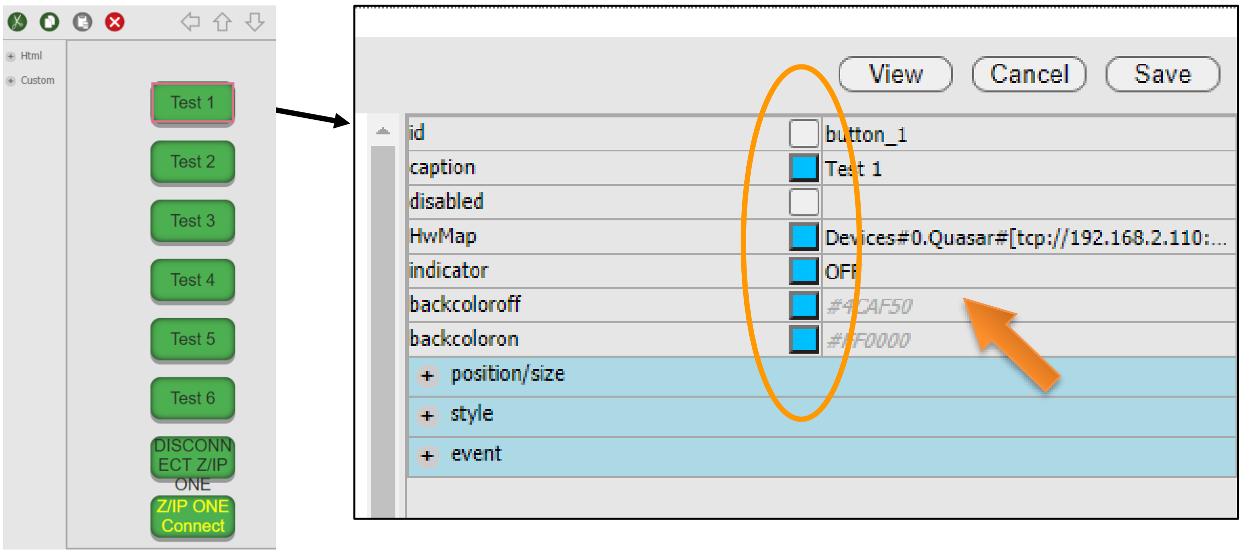
Click **Apply** .

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** line of the property list.



3. Pathfinder will detect 32 buttons for each Master Module detected, and display the following list of properties:

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

Select Hardware Button

Show 100 entries

Device Name	IP Address	Module	Button	
None		0	0	
Quasar	192.168.2.110	mon#0	1	UserPanels#0.UserPanel#MTS_Bank_A.Page#index.Control#button_1
Quasar	192.168.2.110	mon#0	2	UserPanels#0.UserPanel#MTS_Bank_A.Page#index.Control#button_2
Quasar	192.168.2.110	mon#0	3	UserPanels#0.UserPanel#MTS_Bank_A.Page#index.Control#button_3
Quasar	192.168.2.110	mon#0	4	UserPanels#0.UserPanel#MTS_Bank_A.Page#index.Control#button_4
Quasar	192.168.2.110	mon#0	5	UserPanels#0.UserPanel#MTS_Bank_A.Page#index.Control#button_5
Quasar	192.168.2.110	mon#0	6	UserPanels#0.UserPanel#MTS_Bank_A.Page#index.Control#button_6
Quasar	192.168.2.110	mon#0	7	UserPanels#0.UserPanel#MTS_Bank_A.Page#index.Control#button_7
Quasar	192.168.2.110	mon#0	8	UserPanels#0.UserPanel#MTS_Bank_A.Page#index.Control#button_8
Quasar	192.168.2.110	mon#0	9	UserPanels#0.UserPanel#MTS_Bank_B.Page#index.Control#button_1
Quasar	192.168.2.110	mon#0	10	UserPanels#0.UserPanel#MTS_Bank_B.Page#index.Control#button_2
Quasar	192.168.2.110	mon#0	11	UserPanels#0.UserPanel#MTS_Bank_B.Page#index.Control#button_3
Quasar	192.168.2.110	mon#0	12	UserPanels#0.UserPanel#MTS_Bank_B.Page#index.Control#button_4
Quasar	192.168.2.110	mon#0	13	UserPanels#0.UserPanel#MTS_Bank_B.Page#index.Control#button_5
Quasar	192.168.2.110	mon#0	14	UserPanels#0.UserPanel#MTS_Bank_B.Page#index.Control#button_6
Quasar	192.168.2.110	mon#0	15	UserPanels#0.UserPanel#MTS_Bank_B.Page#index.Control#button_7
Quasar	192.168.2.110	mon#0	16	UserPanels#0.UserPanel#MTS_Bank_B.Page#index.Control#button_8
Quasar	192.168.2.110	mon#0	17	UserPanels#0.UserPanel#MTS_Bank_C.Page#index.Control#button_1
Quasar	192.168.2.110	mon#0	18	UserPanels#0.UserPanel#MTS_Bank_C.Page#index.Control#button_2
Quasar	192.168.2.110	mon#0	19	UserPanels#0.UserPanel#MTS_Bank_C.Page#index.Control#button_3
Quasar	192.168.2.110	mon#0	20	UserPanels#0.UserPanel#MTS_Bank_C.Page#index.Control#button_4
Quasar	192.168.2.110	mon#0	21	UserPanels#0.UserPanel#MTS_Bank_C.Page#index.Control#button_5
Quasar	192.168.2.110	mon#0	22	UserPanels#0.UserPanel#MTS_Bank_C.Page#index.Control#button_6
Quasar	192.168.2.110	mon#0	23	UserPanels#0.UserPanel#MTS_Bank_C.Page#index.Control#button_7
Quasar	192.168.2.110	mon#0	24	UserPanels#0.UserPanel#MTS_Bank_C.Page#index.Control#button_8
Quasar	192.168.2.110	mon#0	25	UserPanels#0.UserPanel#Test_with_Dan.Page#index.Control#button_1
Quasar	192.168.2.110	mon#0	26	UserPanels#0.UserPanel#Test_with_Dan.Page#index.Control#button_2
Quasar	192.168.2.110	mon#0	27	UserPanels#0.UserPanel#Test_with_Dan.Page#index.Control#button_3
Quasar	192.168.2.110	mon#0	28	UserPanels#0.UserPanel#Test_with_Dan.Page#index.Control#button_4
Quasar	192.168.2.110	mon#0	29	UserPanels#0.UserPanel#Test_with_Dan.Page#index.Control#button_5
Quasar	192.168.2.110	mon#0	30	UserPanels#0.UserPanel#Test_with_Dan.Page#index.Control#button_6
Quasar	192.168.2.110	mon#0	31	UserPanels#0.UserPanel#Test_with_Dan.Page#index.Control#button_7
Quasar	192.168.2.110	mon#0	32	UserPanels#0.UserPanel#Test_with_Dan.Page#index.Control#button_8

Showing 1 to 33 of 33 entries   1 row selected

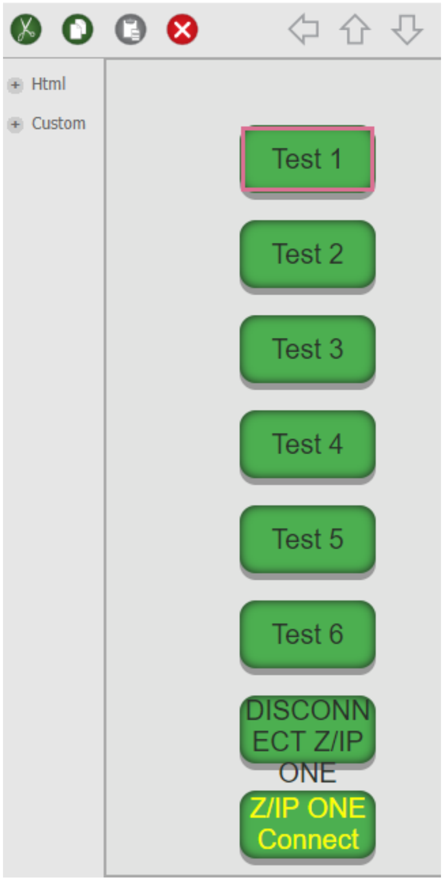
Select   Cancel

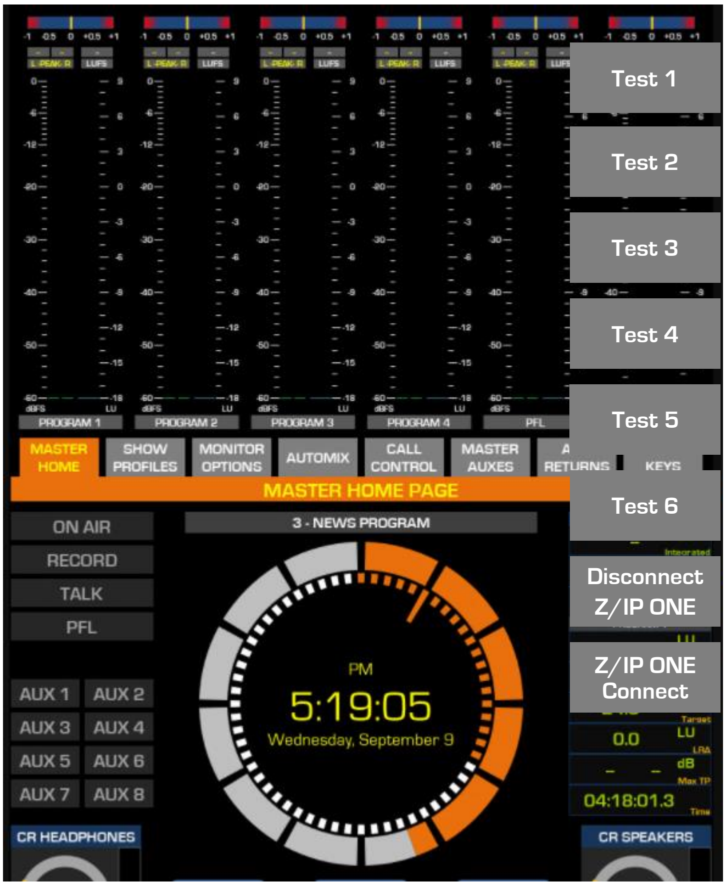
Lines with “FaCH xx” indicate Fader (User 1-4 keys).

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.

XR-4FAD Module User 1-4 Buttons

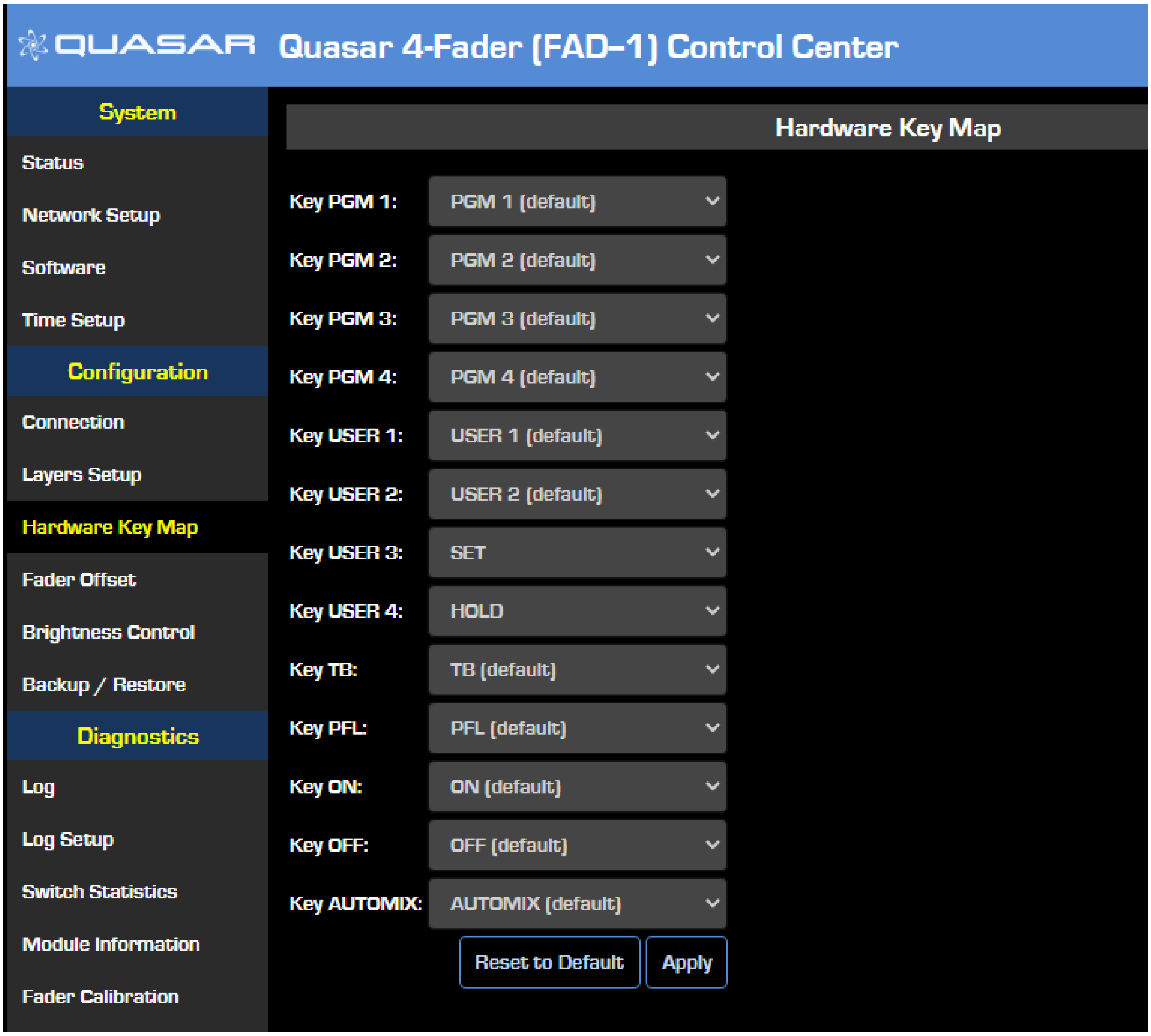
User Buttons 1-4 are available in the XR-4FAD Module, and located on each channel strip right below the channel display.

Remapping The XR-4FAD Module Keys

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** . Below is the list of all possible functionalities available for each key:

PGM 1	Activates the Program 1 bus
PGM 2	Activates the Program 1 bus
PGM 4	Activates the Program 1 bus
PGM 3	Activates the Program 1 bus
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
AFL	Activates the After-Fader Listen function
AUTOMIX	Activates the Talkback function
ON	Turns the Channel ON, only
OFF	Turns the Channel OFF, only
ON-OFF	Ch defaults to OFF state - push to turn ch ON
OFF-ON	Ch defaults to ON state - push to turn ch OFF, or CUT
SET	Selects the Phone channel for controlling a Hybrid/VoIP System
HOLD	Puts the Hybrid/VoIP System call on hold
EQ	Activates the Equalizer on the selected channel
DYN	Activates the Compressor and Expander on the selected channel
DE-ESSER	Activates the Equalizer on the selected channel
FILTER	Activates the Filters on the selected channel
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

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 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.

Programming Fader User Buttons With Pathfinder

In order to assign a function to these buttons from PF, we need to look for the "FaCH#xx" lines, (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

In fact, 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.

Configuration - 5

Practical Configuration Guides

This chapter contains single help guides of common setups. If we forgot one, 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 **Feed to Source** 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.

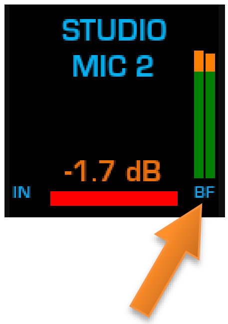
2. *Engineer + Studio Guest:* The Studio Guest is the program host, and uses the Studio Guest Microphone to talk to the Control Room. The console operator only uses the CR Microphone for the talkback, and never goes on air from it.

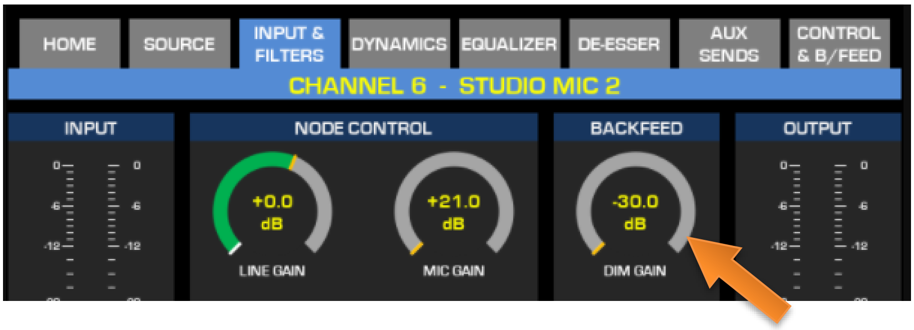
   In this case, having the CR Microphone loaded on one of the console channels is not required.

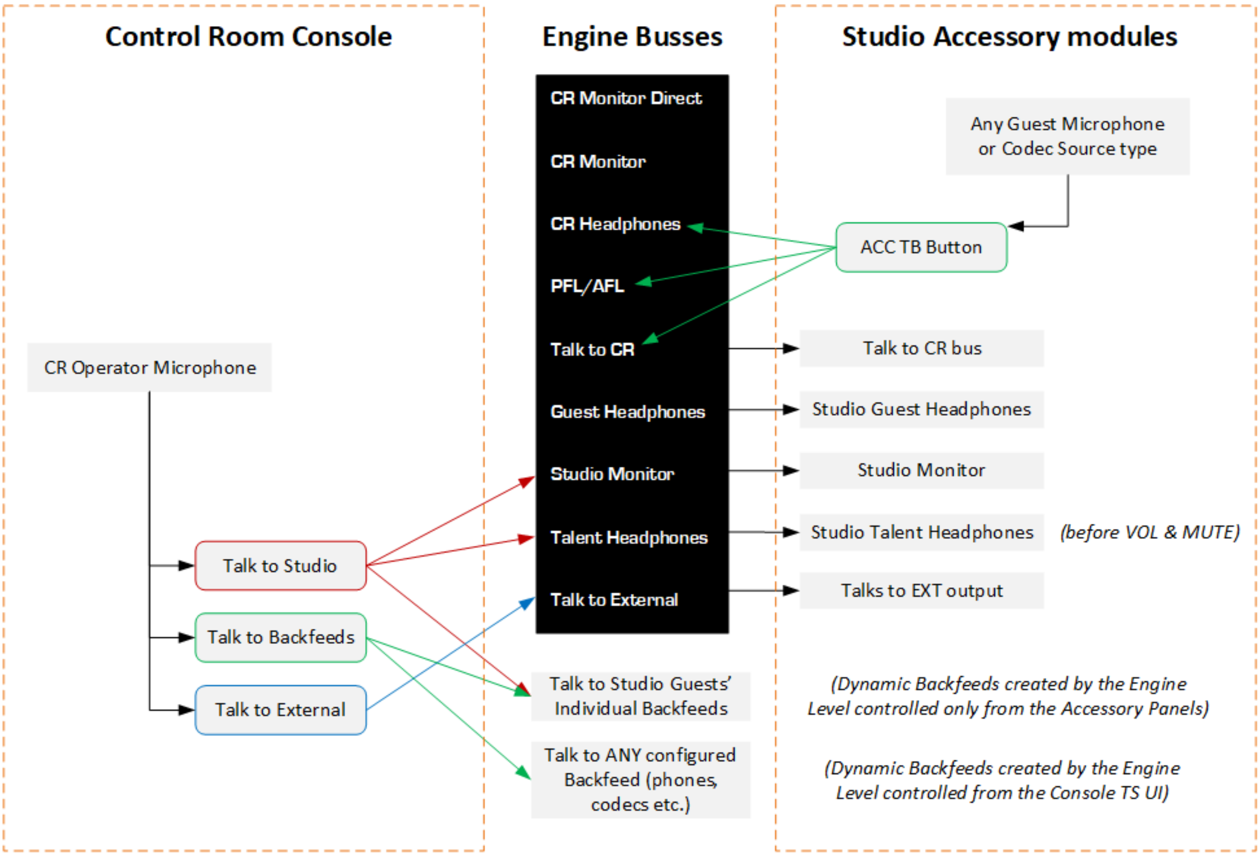
When the Operator is the Host

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' input menu. The channel BF meter will display the level drop when the TB is engaged.

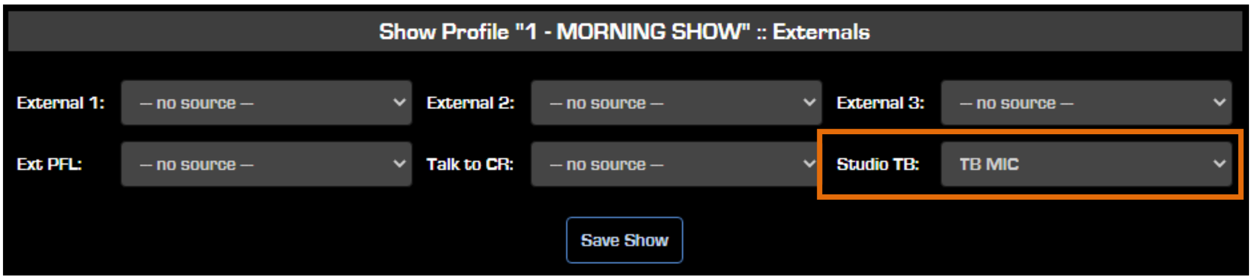






When the Studio Guest is the Host

Unlike the Fusion, Quasar <u>does not require</u> 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.



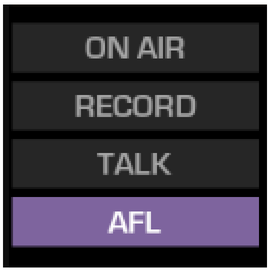
Only when the Talkback is properly configured in one of the two ways we just described, the TB circuit will be active. To talk to the Studio, push the hardware **TALK** key located in the bottom right corner of the Monitor module. To talk to one particular channel only, push the small **TB** key onto the fader strip. Both these actions will enable the Talkback function, turning the corresponding signal on in the MTS Home screen.

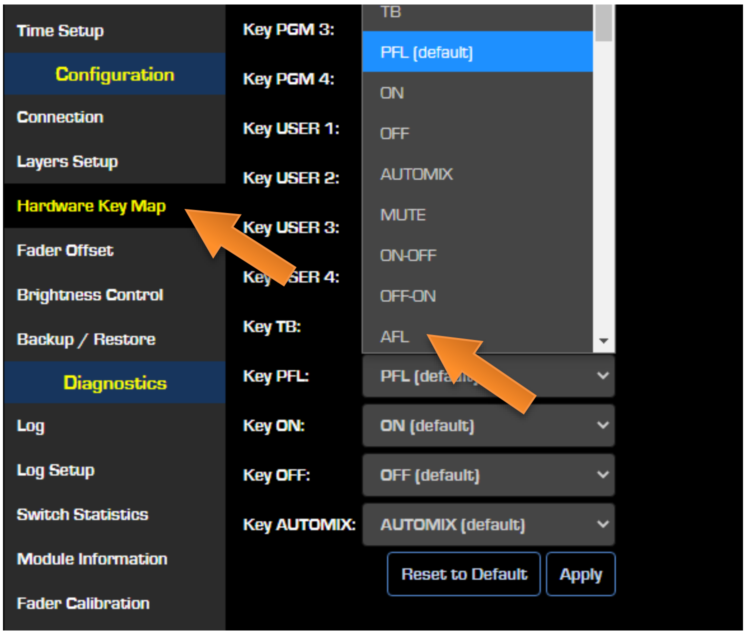
Setting Up The Monitors & PFL/AFL

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.





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 directly access the AFL function on every channel 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 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. In this case the level is controlled by the LS PFL control only.
2. **PFL to HP : STEREO.** The PFL bus is routed to the L & R HP outputs <u>before</u> the CR HEADPHONES volume. The PFL bus level is fixed, and the CR HP output level is controlled by the CR HEADPHONES volume.
3. **PFL to HP : SPLIT.** The PFL bus is routed to the Right HP output only, <u>before</u> the CR HP volume. The level of this output is controlled only by the CR HP volume. The previously selected monitor bus remains routed to the Left HP output only. The level of this output is controlled by the CR HP volume only.

Monitoring PFL in the CR Loudspeakers

PFL to LS Workflow

When the operator is monitoring 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 volume only.
2. **PFL to LS : STEREO.** The PFL bus is routed to the L & R LS outputs <u>after</u> the CR SPEAKERS volume, therefore the CR SPEAKERS volume control has no effect on the output level. The PFL bus level is controlled by the PFL volume, and the only way to control the LS output level is by using the PFL volume.
3. **PFL to LS : SPLIT.** The PFL bus is routed to the Right LS output only, <u>after</u> the CR SPEAKERS volume, therefore the CR SPEAKERS volume control effect only on the Right output level. The PFL volume controls the Right Speaker Output. The previously selected monitor bus is still routed to the Left Speaker output only. The level of this output is controlled by the CR SPEAKERS volume.

Triggering an ON AIR light

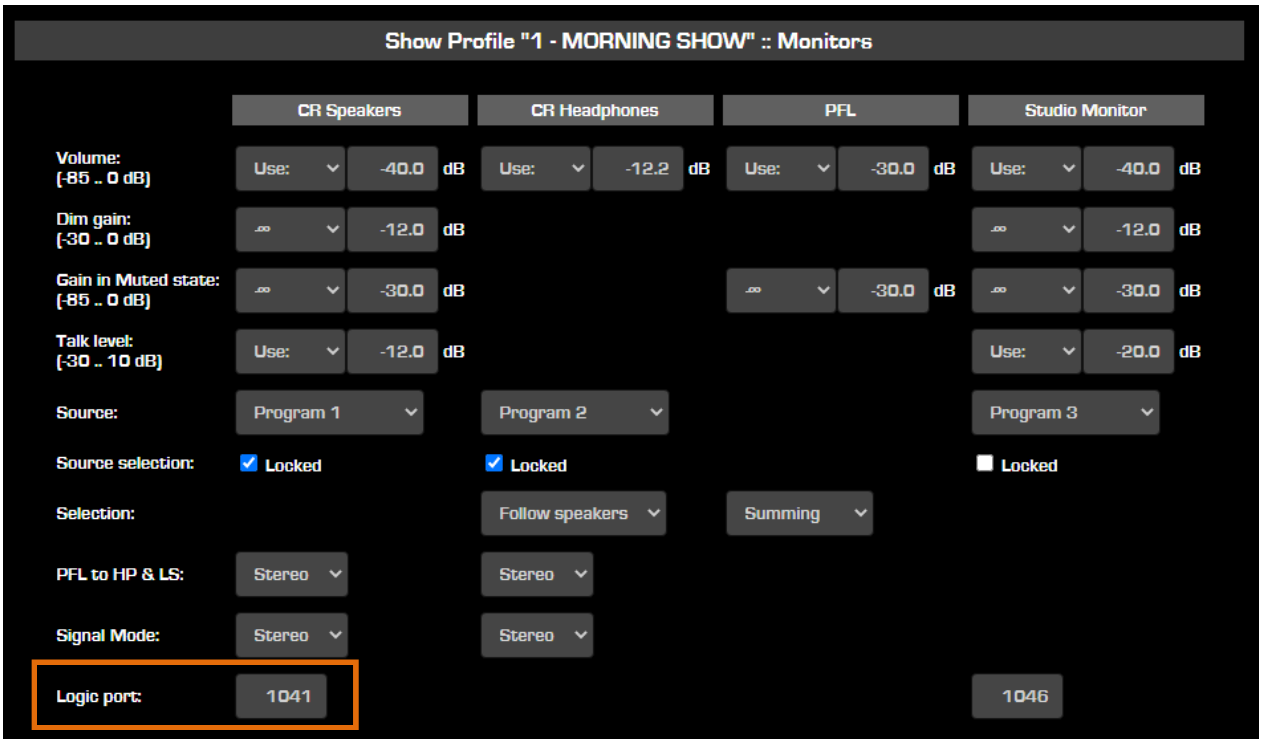
An Air light traditionally is illuminated when a microphone is ON. Additionally, the console has logic built in which will mute the monitor speakers if a localized microphone is ON. The Control Room (CR) speakers have GPIO logic built into the product. This logic is as follows.

GPIO Control Room Monitor Logic

Name	Pin	Type	Notes
INPUTS			
MUTE CR Command	11	Active Low Input	Mutes CR monitors and Pre- view 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
TALK TO EXT Command	14	Active Low Input	Turns on Talk to External Audio.
Not used.	15	Active Low Input	
OUTPUTS			
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

The first output is the ON AIR lamp and is illuminated anytime the CR monitors are mutes. To use this logic to trigger a light circuit is follows these steps:

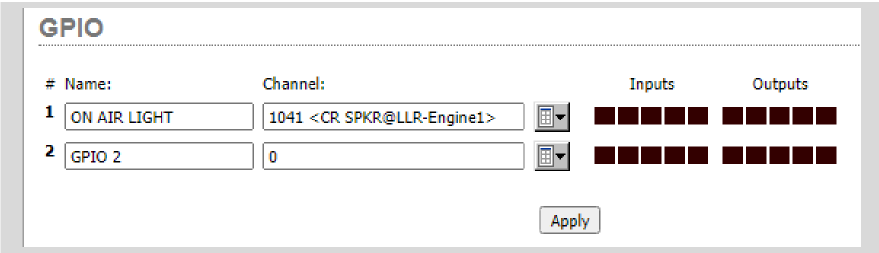
1. Navigate to the Monitor Section of your Show Profile
2. Locate the **Logic port** field for the CR Speakers:



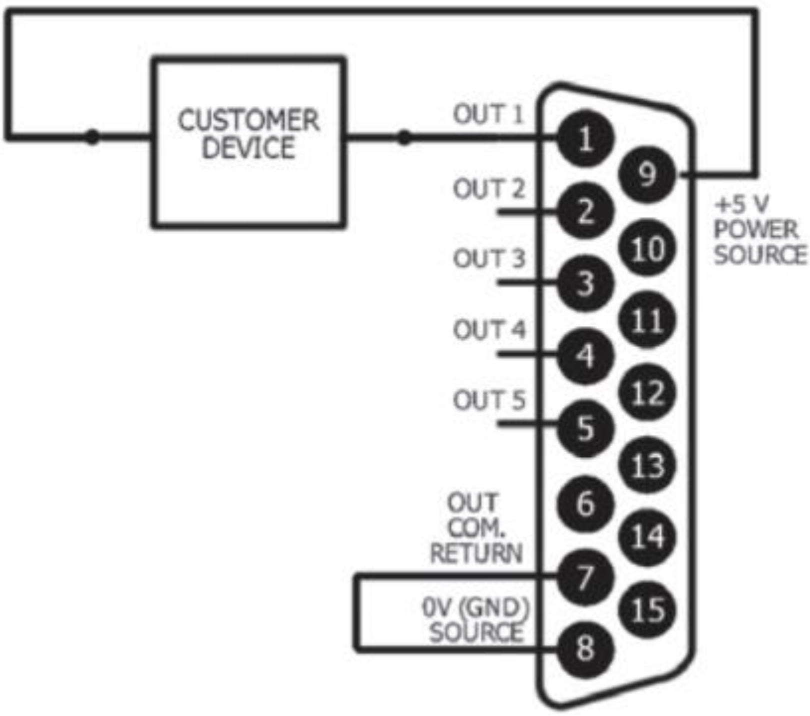
3. Enter a unique Livewire channel number. A common practice is to enter the LW ch# used for the CR Speaker audio.
4. Save the show profile and load the profile to the surface. When a microphone is turned on, the CR monitor speakers will mute and the logic state will be reported into the network.

The next step is to assign a GPIO port in the network to respond to this message.

5. Log into an xNode that has GPIO functionality.
6. Assign one of the ports to the same channel that was assigned earlier. In the example here, 1041 was used.



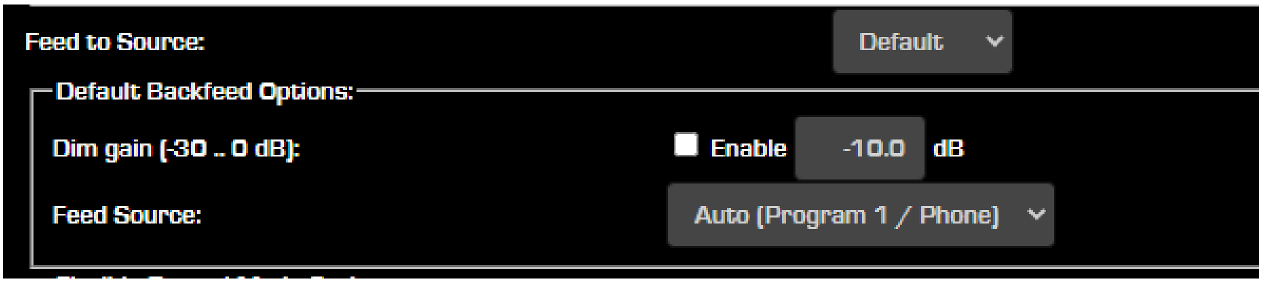
7. In the case of electrically isolated circuit, you would wire the GPIO port as shown in the pinout diagram below:



Setting Up Mix-Minuses

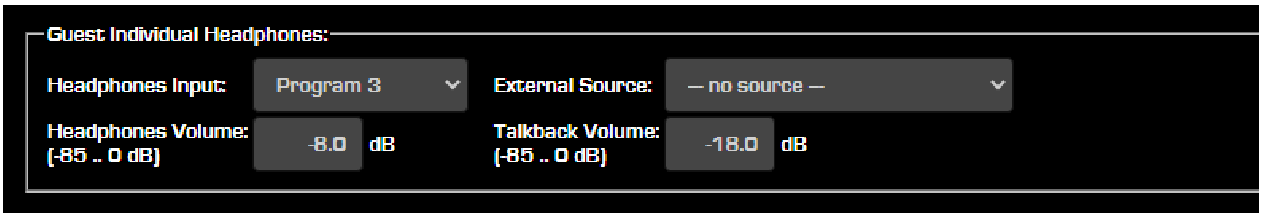
In Show profiles, this is where you set the Mix-Minus <u>for Codec and Phone type sources:</u>

There is an equivalent section in the Source Profile:



Again, this <u>will appear only in case of Codec and Phone type sources.</u>

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.

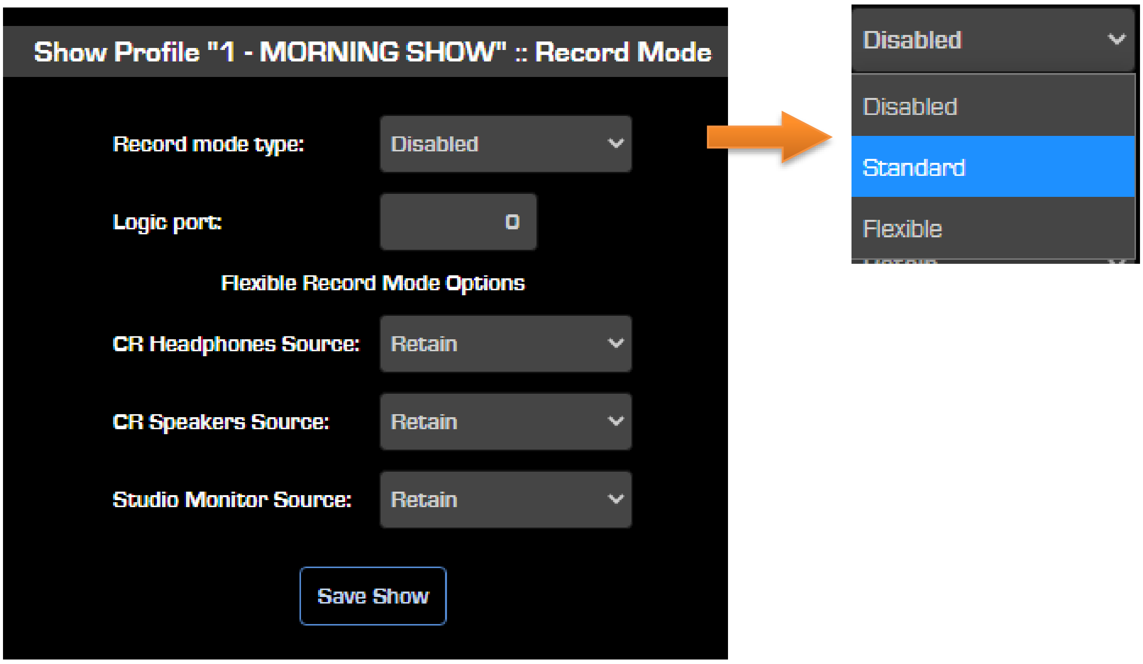
Standard Vs. Flexible Record Modes

Record Mode consists in a special "macro function" that helps 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** section of the Show Profile main page.

Please keep in mind that:

1. in case Standard Record Mode is selected, the logic is associated to the Show Channels: the options that define the channel behavior must be configured in each Channel of the Show Profile.
2. 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 Custom Backfeed options section.



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** is configured in the Show Profiles, on a global, per-Show basis.

To enter the Standard Record Mode:

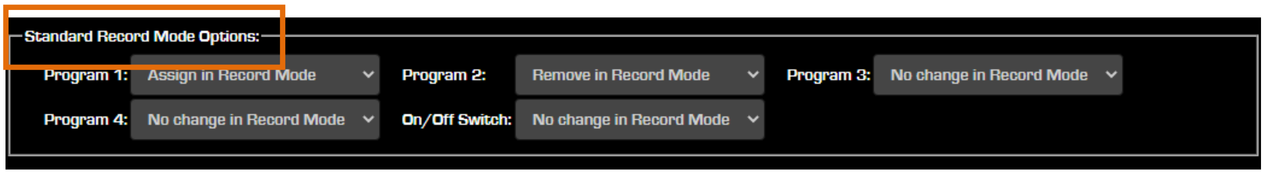
- Load the Show you have configured for this mode
- Press the hardware **RECORD** key in the bottom right corner of the Master Monitor module.

When this mode is active:

- The hardware **RECORD** key will turn Red.
- The “RECORD” 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.

The different color of the **PGM4** keys will also help reminding the operator that Record Mode is active, and he will need to exit this mode at some point.

Each of the Show Channels can be configured to recall different PGM bus assignments and control the Channel ON/OFF switches, by navigating to the Standard Record Mode Options section of the Channel, and selecting the appropriate drop-down menus.



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.

The Flexible Record Mode

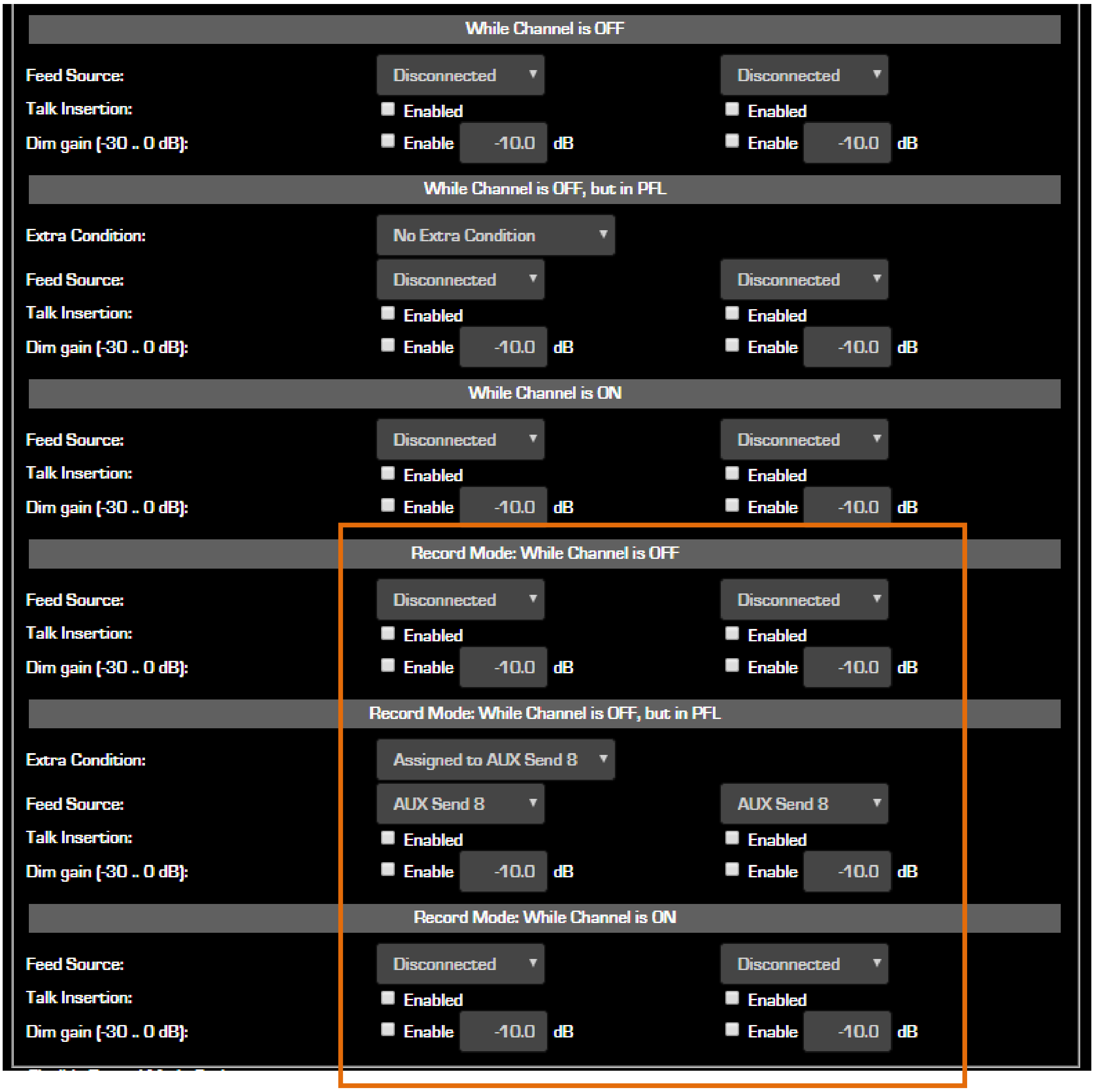
When the options available in the Standard Record Mode settings do not suit your workflow, you can use the Flexible Record Mode.
On top of defining the Show Profile Global settings, the **Flexible Record Mode** options must be configured in each of the Source Profiles assigned to the Channels of that particular Show.

When this mode is active:

- The hardware **RECORD** key will turn ORANGE.
- The “FLEX RECORD” indicator will turn on in the MTS Home page.
- MTS Home page will turn the “FLEX RECORD” indication on.
- The **PGM4** keys on each channel will not change color, to indicate that they have been activated by the record mode.

The options available to each Source let you independently assign any 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 Feed to Source mode 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 the Configuration Chapter 2 of this manual: Create a Codec Source with Custom Backfeed.

Configuration - 6

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
- Program 4 bus
- PFL / AFL bus
- CR Monitor bus

<u>The ballistics of these meters is programmable from the Web UI.</u>

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 Input Meter (right after source input, before Delay and Trim Gain)
- Backfeed Output Meter
- Dynamics Input, Gain Reduction, and Output meters
- Ess Signal Meter (monitors the De-Esser sidechain)
- Auxiliary 1-8 Sends Meters (confidence class)
- Channel Output Meter (after fader, before Automix Gain stage)

<u>These meters are not programmable.</u> 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)

- Backfeed Output Meter (confidence class)
- Fader Bargraph Meter (……..)
- Automixer Gain Reduction meter (……..)

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

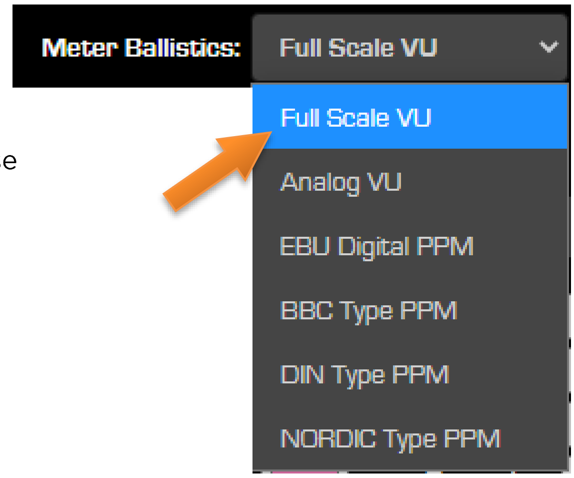
Starting from the leftmost meter, the first four are permanently assigned to each of the four Program busses, the fifth meter is assigned to the PFL/AFL bus, and the sixth one is assigned to the CR Monitor bus.

The Label below each of the meters, indicating the bus name can be of two different colors depending on the assignement 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.

Meter Ballistics



The **Meter Ballistics** option, available in the Web UI Configuration>Customize menu, lets the user choose between six possible Meter Types, each with its own Ballistics, for the Main Meters:

1. Full Scale VU
2. Analog VU
3. EBU Digital PPM
4. BBC Ttype PPM
5. DIN Type PPM
6. NORDIC Type PPM

Quasar Meter Types Specifications

Full Scale VU
Scale: -60 to 0 dBFS
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

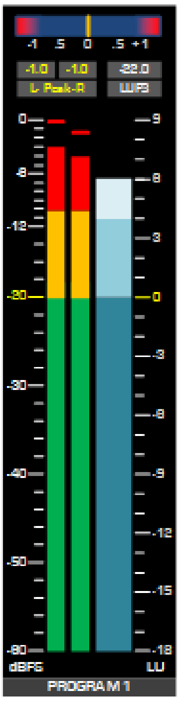
Analog VU
Scale: A VU scale, extending from –20 to +3, with 0 marked at about 71% max scale reading
Reference level mark: 0
Color transition: 0dBr/-20dBFS
Attack time: 300ms for 20dB
Release time: same as attack time
Calibration: 0VU = +4dBu

EBU Digital PPM
Scale: -60 to 0 dBFS
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

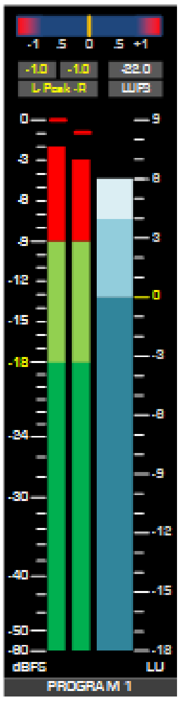
BBC Type PPM
Reference level mark: none (4 usually corresponds to -18dBFS and to 0dBu analog)
Color transition: 6.2 (-9dBFS)
Attack time: 80% in 10ms
Release time: 24dB in 2.8s (8.6dB/s)
Calibration: -18dBS = 0dBu
Normally, “4” corresponds to -18dBFS. But some installations prefer to have -12dBFS at this mark.

DIN Type PPM
Scale: -60 to +5 dB
0dB mark: -9dBfs
Reference level mark: -9dBr/-18dBFS
Color transition: 0dBr/-10dBFS
Attack time: 80% in 5ms
Release time: 20dB in 1.7s (12dB/s)
Node: -18dBS = 0dBu

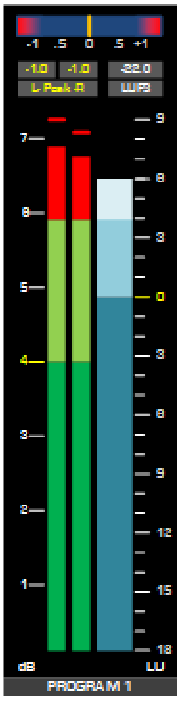
Nordic Type PPM
Reference level mark: “TEST” at 0dBr/-18dBFS
Color transition: +6dBr/-9dBFS
Attack time: 80% in 5ms
Release time: 20dB in 1.5s
Calibration: -18dBFS = 0dBu



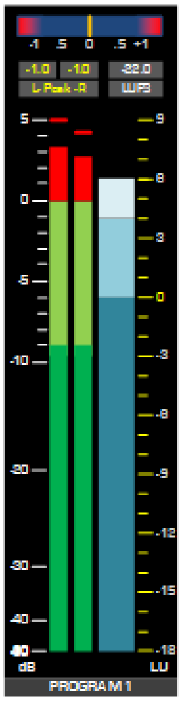
FULL SCALE VU



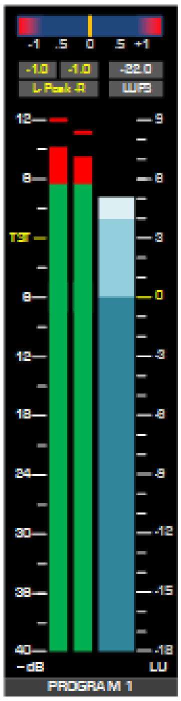
EBU DIGITAL



BBC-TYPE PPM



DIN-TYPE PPM



NORDIC N9

Programme Meter Type	Recommenda-tion	PML [a] 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

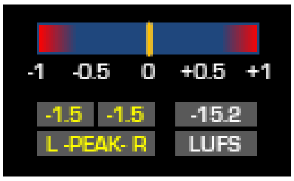
Above each main meter, there is a Max True Peak Meter, and Max LU Meter. The L-PEAK-R Indicator shows max true peak level, as described in ITU-R BS.1771 and uses oversampling to calculate the maximum peak levels. LUFS shows integrated loudness value since reset in LUFS units.

Peaks are measured by the single segments and text indicators with the label "Peak".
You must push the **LOUDNESS START/STOP/RESET** 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 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 300 ms 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 300 ms 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.

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, 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!
- 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 600 ohm 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, the solid bar is still measuring RMS levels. This bargraph is a Full Scale VU Meter. It is not a Peak Program Meter (PPM),

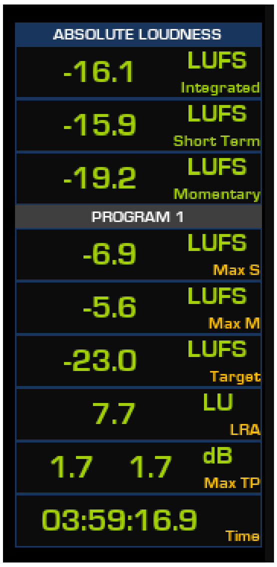
The Loudness Meters

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).

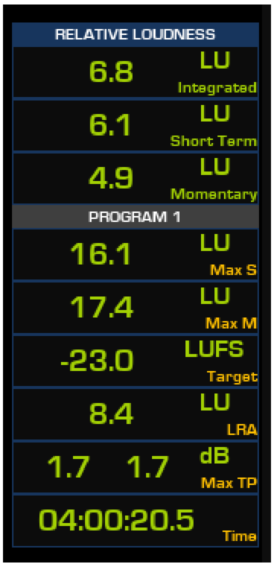
You can choose between Absolute and Relative modes, by selecting an option in the Web UI.



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:

Pin	Function
1	START
2	STOP
3	RESET

Configuration - 7

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 your 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

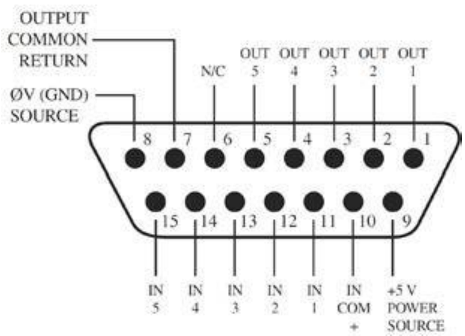
Axia Quasar consoles must be paired with xNode GPIO Nodes, in order to control, or be controlled via GPIO. Each GPIO port can be associated with a device in your studio, and provides five opto- isolated inputs and five opto- isolated outputs per device.

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.

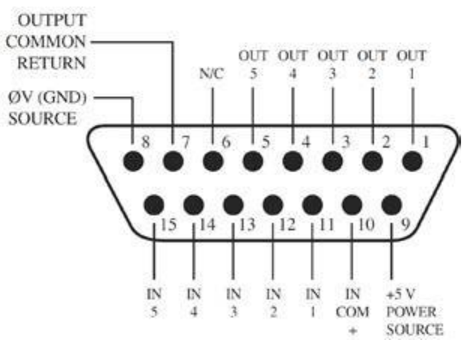


GPIO Operator's Microphone Logic

Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to Monitor 2) Command	13	Active Low Input	Activates the TALK TO MON2 function and routes mic audio to the Talkback bus.
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to PFL SOURCE) Command	15	Active Low Input	Activates the TALK key on every source currently in PFL and routes mic audio to the Talkback bus.
OUTPUTS			
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 Monitor 2) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK TO MON2 is active
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to PFL SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when TALK to PFL SOURCE is active.
POWER & COMMON			
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
INPUTS			
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
NOT CONNECTED	15		
OUTPUTS			
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
NOT CONNECTED	5		
POWER & COMMON			
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 volt- age; 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		

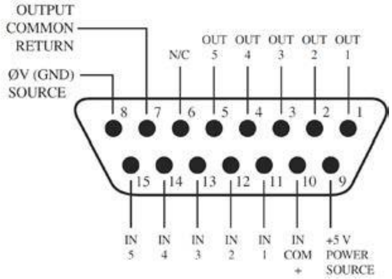


GPIO Producer's Microphone Logic

Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to MONITOR 2) Com- mand	13	Active Low Input	Activates the TALK to MON2 function and routes mic audio to the Talkback bus.
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to PFL SOURCE) Command	15	Active Low Input	Activates the TALK key on every source currently in PFL and routes mic audio to the Talkback bus.
OUTPUTS			
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 MONITOR 2) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK to MON2 is active.
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to PFL SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when TALK to PFL SOURCE is active.
POWER & COMMON			
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 volt- age; 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
INPUTS			
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 indi- cate source's readiness
OUTPUTS			
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
POWER & COMMON			
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 volt- age; 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		

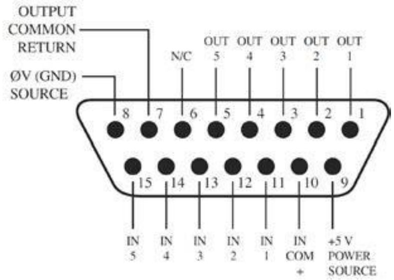


GPIO Codec Logic

Name	Pin	Type	Notes
INPUTS			
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.
OUTPUTS			
ON Lamp	1	Open Collector to Logic Com- mon 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.
POWER & COMMON			
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 volt- age; 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		

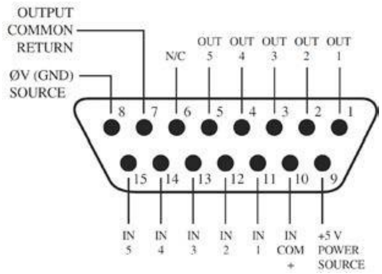
GPIO Telephone Hybrid Logic

Name	Pin	Type	Notes
INPUTS			
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 indi- cate source’s readiness
OUTPUTS			
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.
POWER & COMMAND			
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 volt- age; 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		



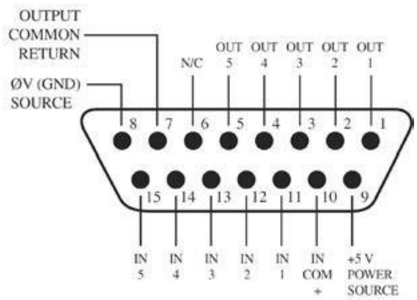
GPIO Control Room Monitor Logic

Name	Pin	Type	Notes
INPUTS			
MUTE CR 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 bus
TALK TO EXT Command	14	Active Low Input	Turns on Talk to External Audio.
Not used.	15	Active Low Input	
OUTPUTS			
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
POWER & COMMON			
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 volt- age; 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 Computer Playback Device Logic

Name	Pin	Type	Notes
INPUTS			
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 indi- cate source’s readiness
OUTPUTS			
NEXT Pulse	1	Open Collector to Logic Common Return	A 100 mS PULSE sent when ON key is depressed, ex- cept 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.
POWER & COMMON			
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 com- mon
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing volt- age; 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		



Comparative Table Of GPIO Commands

Source **Type**	**GPI / GPO**				
	11 / 1	**12 / 2**	**13 / 3**	**14 / 4**	**15 / 5**
Line	ON	OFF	PFL/AFL	Reset / Start*	Ready / Stop*
PC	ON / NEXT*	OFF	PFL/AFL	- / Start*	Ready / Stop*
Phone	ON	OFF	PFL/AFL	Reset / Start*	Ready / Stop*
Codec	ON	OFF	Talk to CR	MUTE	Talk to Source^
Operator	ON	OFF	Talk to Studio	MUTE	Talk to Preview
Producer	ON	OFF	Talk to Studio	MUTE	Talk to Preview
CR Guest	ON	OFF	Talk to CR	MUTE	Talk to Source^
Studio Guest	ON	OFF	Talk to CR	MUTE	Talk to Source^
External Mic	ON	OFF	Talk to CR	MUTE	Talk to Source^
Studio Feed	ON	OFF	Talk to CR	MUTE	Ready / Talk to Source^
CR Monitor	Mute CR / ON-AIR CR +	DIM CR	EXT PFL / PFL	Talk to External	- / Talk to CR
ST Monitor	Mute ST / ON-AIR ST	DIM ST			- / Talk to ST
Record	- / ON	- / OFF	-	- / Start*	- / Stop*
Rec (CR Ready Mode)	-	-	-	-	- / Ready CR
Circuit (CMsg)	**13 / 8**	**12 / 7**	**11 / 6**	**10 / 5**	**9 / 4**

* Sends 100ms pulse.

+ Quasar is set to ON AIR state only in case of open Microphones.

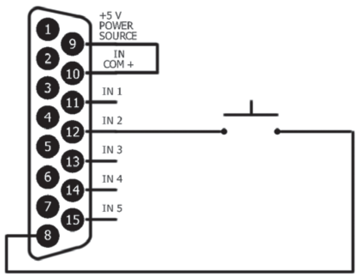
^ Requires active Backfeed.

About GPIO Connections

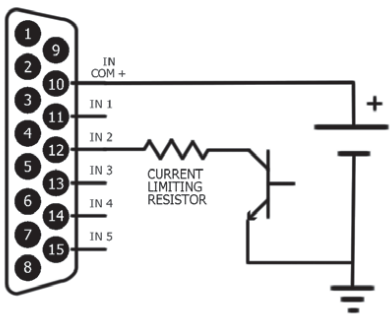
I NPUT	
VDC	External Series Resistor
5	0
6	0
12	680 @ 1/4 watt
24	1.8k @ 1/2 watt
48	3.9k @ 1 watt

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.

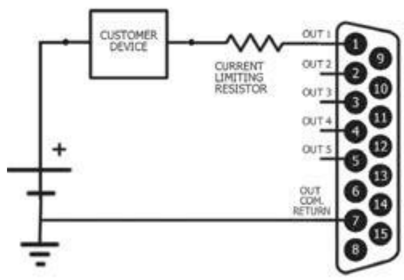
If the equipment being controlled is electrically isolated, than the use of the GPIO port's power supply is acceptable as shown here.

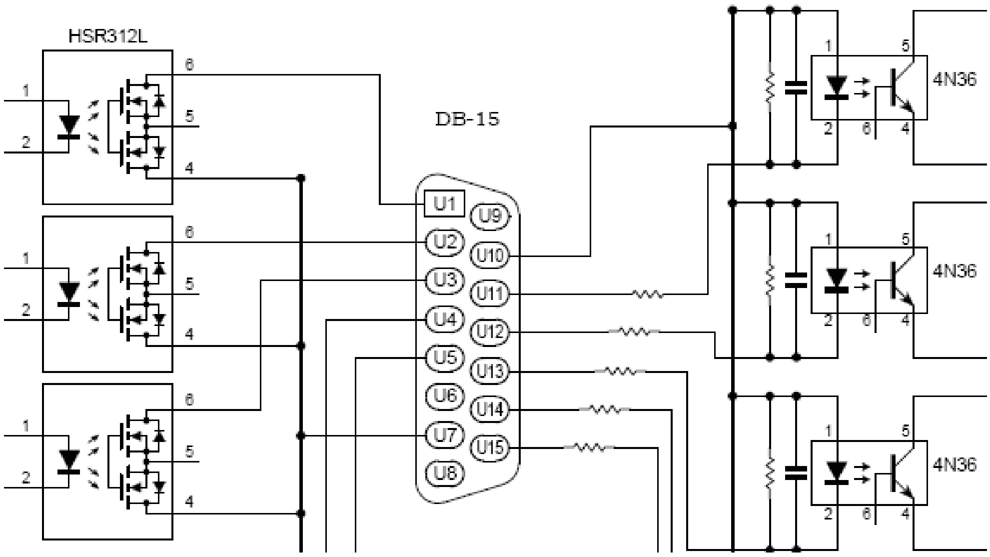


Take note to use current limiting resistors per the above chart if the voltage supplied is above 6vdc. The intention is to limit the current to 20mA for each GPI pin.



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.

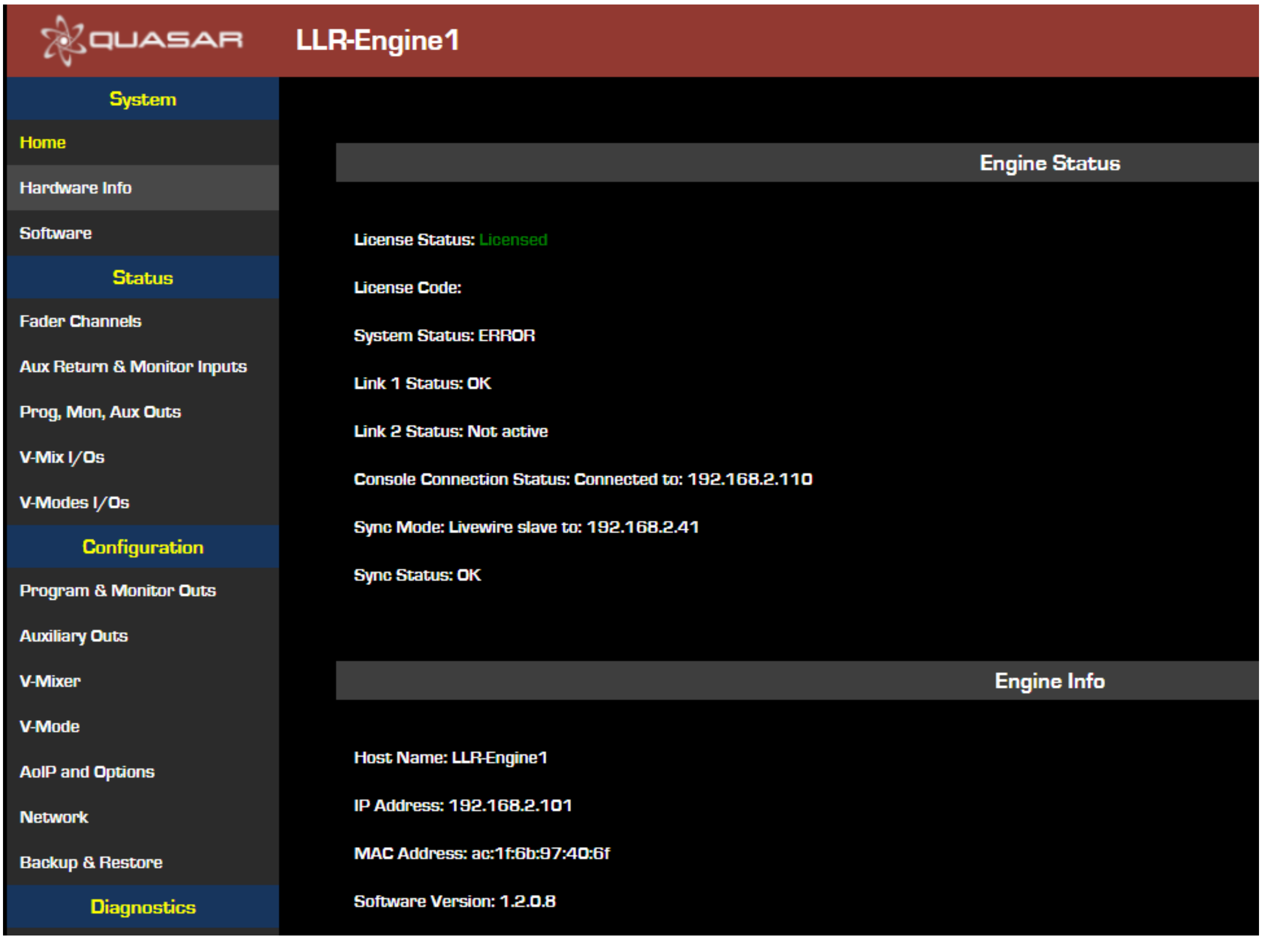
Note, all of the inputs and outputs on a specific GPIO port are “grouped together”. The 5 “Outputs” are on 5 separate output pins, however, they share the same “Common Return” connection on Pin #7. Similarly,
the 5 “Inputs” pins would be pulled to ground to activate them, and they share a common pin 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

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**, **Status**, **Configuration**, 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.

Home

This landing page displays all the most important information about:

- Engine Licensing status
- Audio Synchronization status
- Network address, connection status and link activity
- Running Software Version

License Status / License Code

With software v.1.2 , we added Licensing protection to the Quasar Engine. This license is pre-installed at the factory onto new engines, and it will be automatically installed during field update (from the previously released v1.0.0.60) if the Engine detects valid hardware.
You do not need to obtain a license from Telos, if you are updating your Engine with a web update downloaded from the Telos website.

System Status

If an ERROR message appears here, you need to check the **Hardware Info** page to see what's generating this error.

Network Link 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

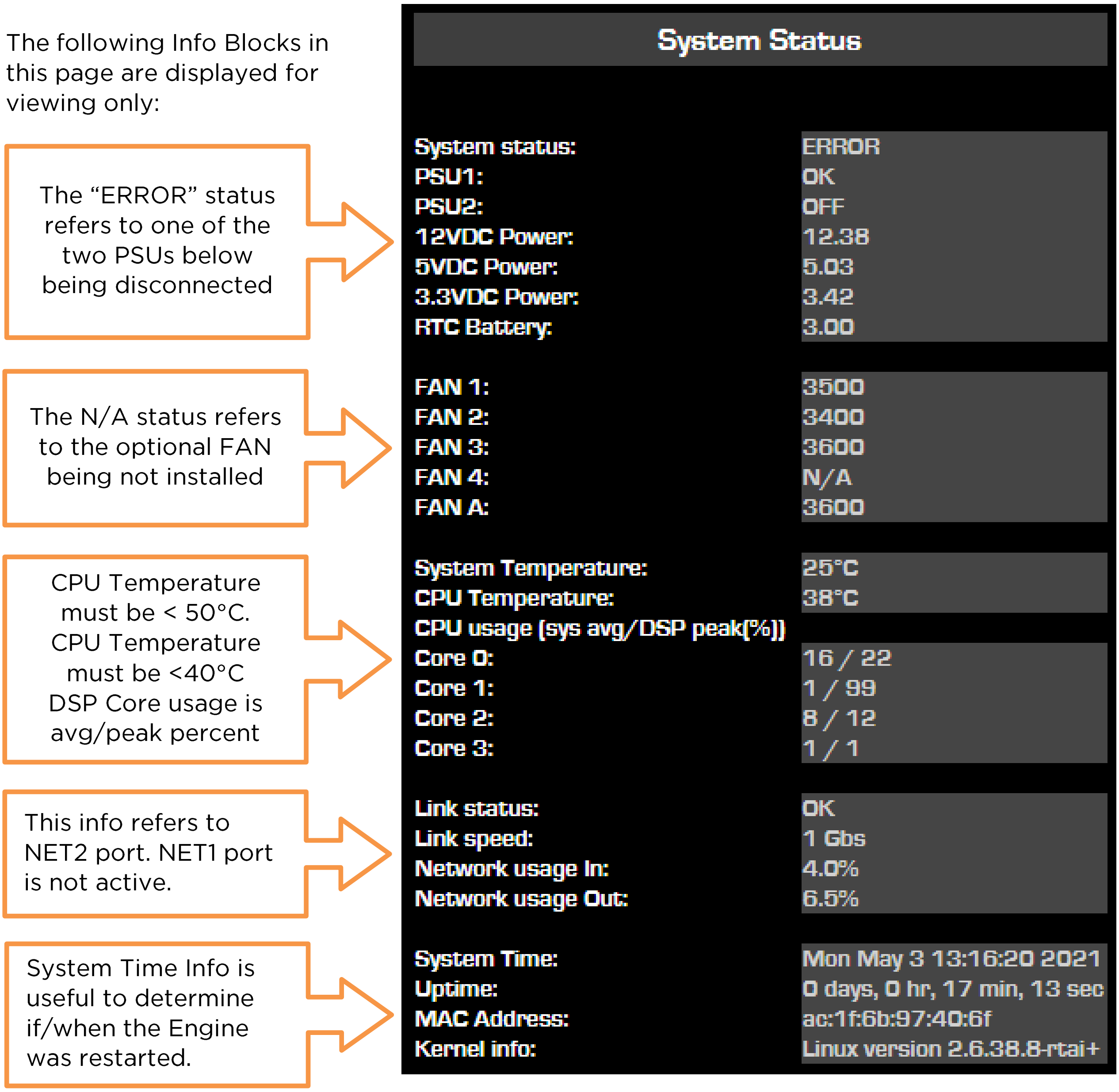
This line displays the IP address of the device (typically an Axia Audio Node) in the AoIP newtork, which is generating the Livewire clock. The Sync Mode must be "slave" and the status must be "OK" for the Engine to operate properly.

> **Note -** Both xNodes and legacy 8x8 Nodes (silver, 1RU model) are supported. GPIO nodes do not supply audio sync.

Important : The Quasar Engine has NO internal Audio Clock Generator, and requires an external Livewire Clock source to operate. Typically, this clock source will be an Audio Node in your network which is set as MASTER. At the time of writing this manual, the option to use PTP clock is not available.

Hardware Info

The following Info Blocks in this page are displayed for viewing only:

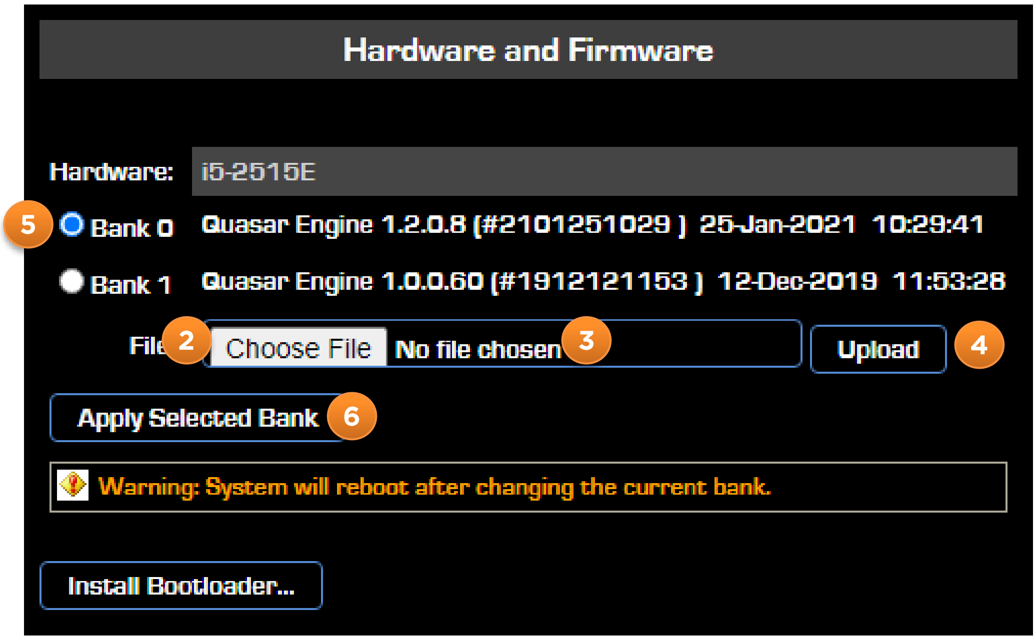


Software

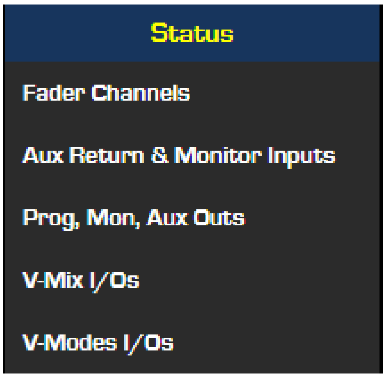
From this page it is possible to update the Quasar System software. In order to install a new software package, you must perform the following steps:

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.
4. 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
5. Select the new software bank

6. Click the **Apply Selected Bank** button.



Warning – The Engine will automatically reboot after selecting a new software bank, connections to the surface will be reset, and audio will be interrupted for more than one minute.



Status Menu

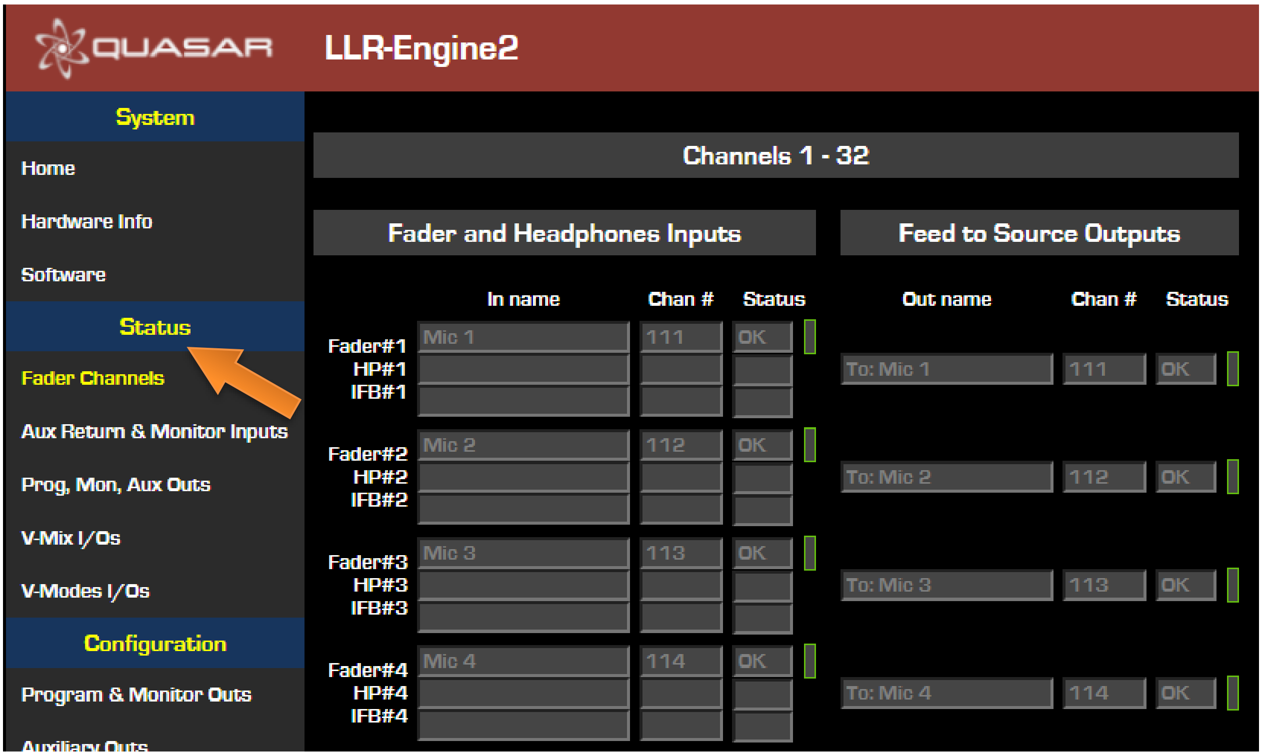
As the name of this section suggests, the Status Menu provides feedback about the status of all, Engine's Input (Fader) Channels, Output Busses, Monitor Inputs, VMIXes and VMODEs.

Note - This entire section, with its five pages, is for viewing only. No parameters can be configured in this section.

Fader Channels

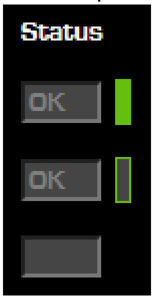
The status of all Fader Input Channels is reported here. The view is split across two blocks of 32 Input channels, each with an indication of the assigned Source Name, AoIP Channel (Livewire Stream) number, and Stream Status, along with audio presence indicators.
In case the channel has a Backfeed (audio return channel) or an IFB (Interruptible FoldBack channel), information about its Source Name, AoIP Channel (Livewire Stream) number, Stream

Status and audio presence will be displayed.



Audio Presence Indicators

During navigation of the Quasar Engine WebUI, you wil encounter these indicators in multiple places:

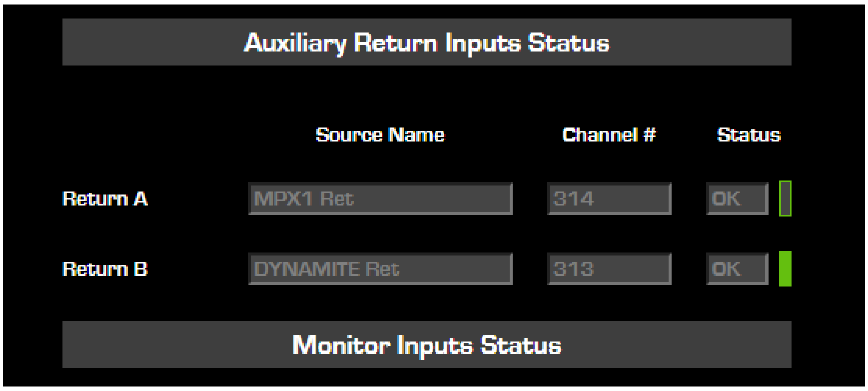


- A solid green bar indicates that the stream is active, and audio is present, on particular output.
- An empty bar with green outline indicates that the stream is active but no audio is present, on that output.
- No bar indicates the stream is not active, or not enabled.

“ERR” message in the text box indicates that the audio stream was not loaded successfully. “OK” indicates it was successfully loaded.

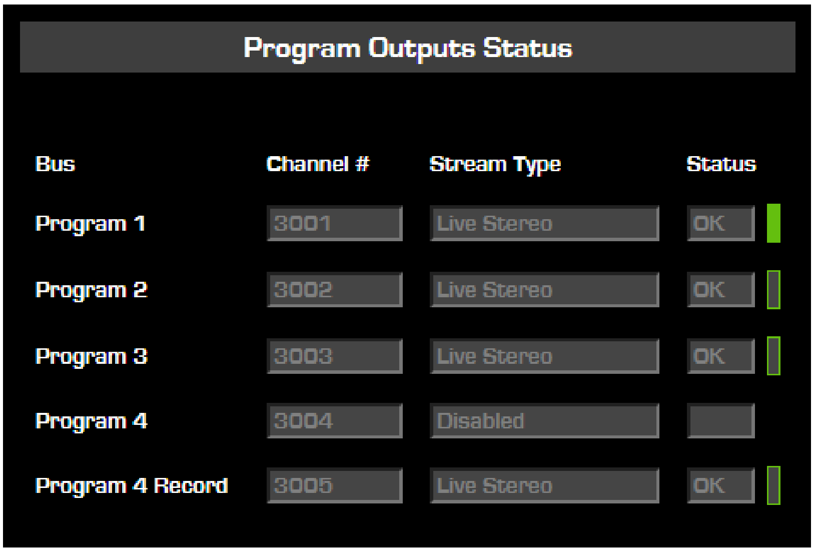
Aux Returns and Monitor Inputs

For each of the Engine’s Auxiliary Return Inputs, and Monitor Section Inputs, an indication of the assigned AoIP Channel (Livewire Stream) number, and Stream Type and Status, along with audio presence indicators, is provided.



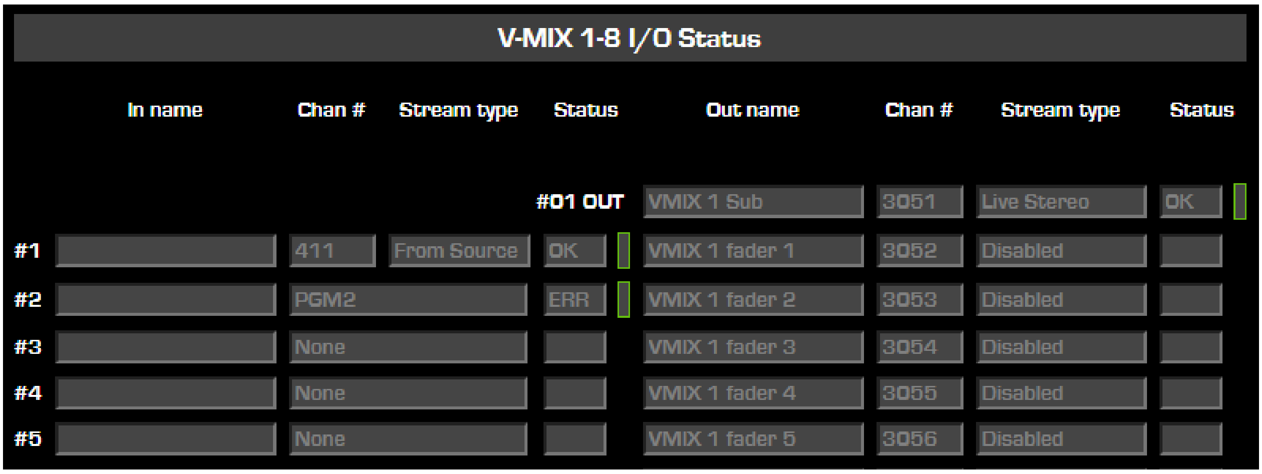
Prog, Mon, Aux Outs

For each of the Program, Monitor and Auxiliary busses outputs, an indication of the assigned AoIP Channel (Livewire Stream) number, and Stream Type and Status, along with audio presence indicators, is provided.

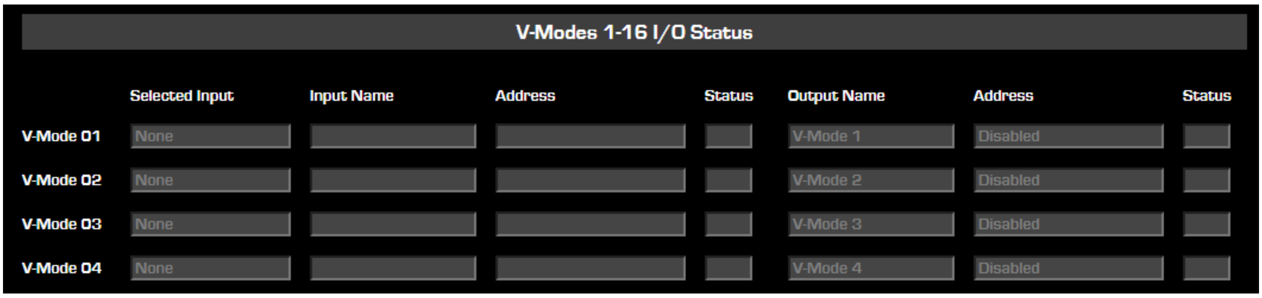


VMIX and VMODEs I/Os

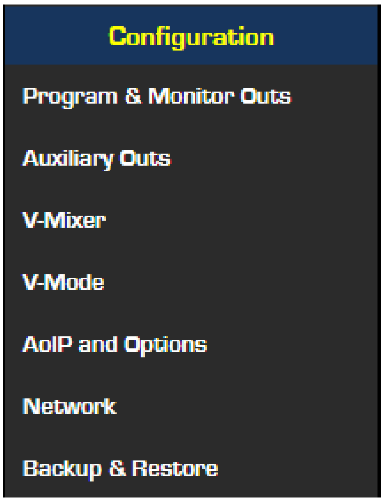
For each of the 16 VMIXERs available, the status of the five Input Channels, and Sub Outs is reported here. The view is split across two blocks of 8 VMIXERs, each with an indication of the assigned Source Name, AoIP Channel (Livewire Stream) number, and Stream Status, along with audio presence indicators.



The same applies to the VMODEs. Since any type of AoIP stream can be entered for mode conversion, here the full Stream Address is displayed, along with the other parameters' status.



Configuration Menu

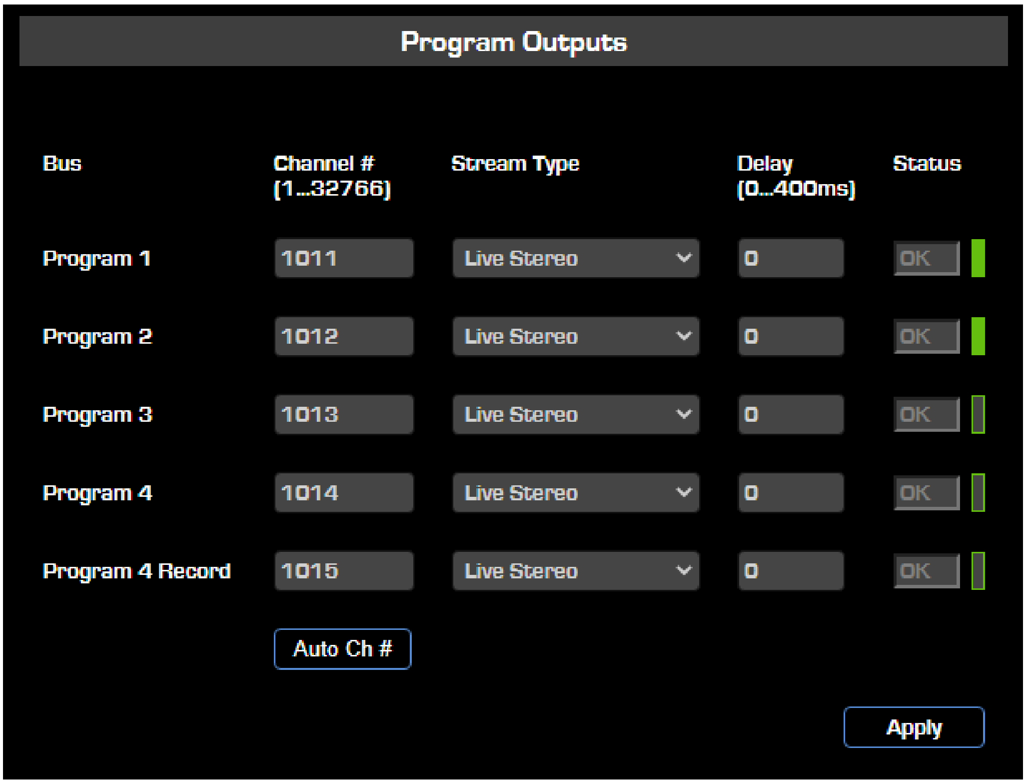


As the name of this section suggests, the Configuration Menu lets you enter all the parameters for configuring your Engine's Program, Monitor and Auxiliary output busses, Monitor Inputs, VMIXes and VMODEs.

In this paragraph only the Configuration basics is described. V-Mixer and V-Mode are described in the next two paragraphs of this Chapter.

Program & Monitor Outs

This is where you enable and assign unique channel numbers to all of your console's outputs. If any of your Quasar's outputs or monitor feeds are disabled in this menu, that channel will have no audio output.



Program Outputs

- **Program 1 - 4:** Your console's main Program bus outputs.
- **Program 4 Record:** Special output for recording devices which feeds the contents the Program 4 bus post-fader and pre-On/Off switch.

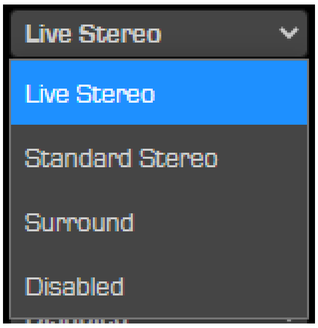
Monitor Outputs

- **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 output
- **Studio Monitor:** The source selected to feed the Studio Monitors is routed to this output
- **Talent Headphones:** The source selected to feed the Talent Headphones is routed to this output
- **Talk to External:** When any console **Talk** key is pressed, CR Mic audio is routed, pre-fader, to this channel.

Channel

The Quasar Engine is capable of generating up to 199 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.
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 **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.



Stream Type

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), back left, back right, stereo left, and stereo right.

- **Disabled:** Turns the stream off, making it invisible to the entire AoIP network.

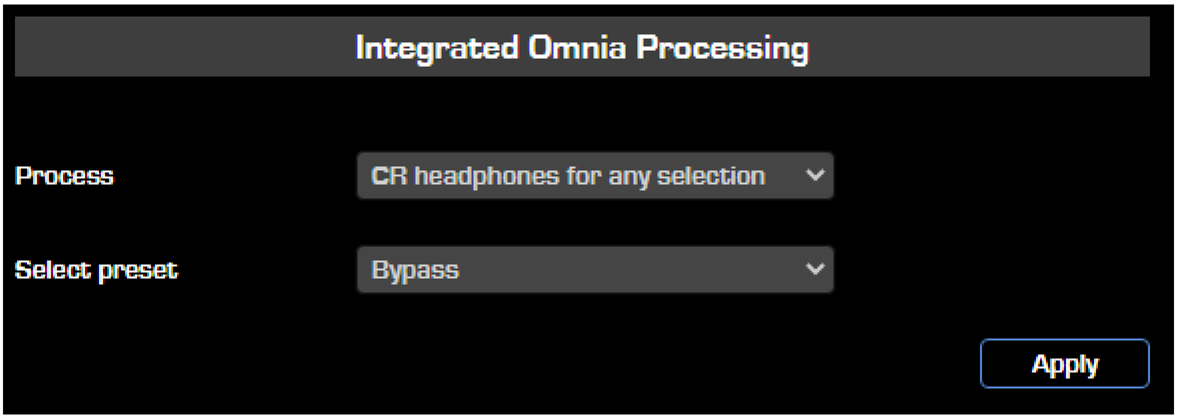
> **Note -** Typically, all the output Streams are all set to "Live Stereo", unless you have limited badwidth budget on your AoIP switch , and require keeping the traffic as low as possible.

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.

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.

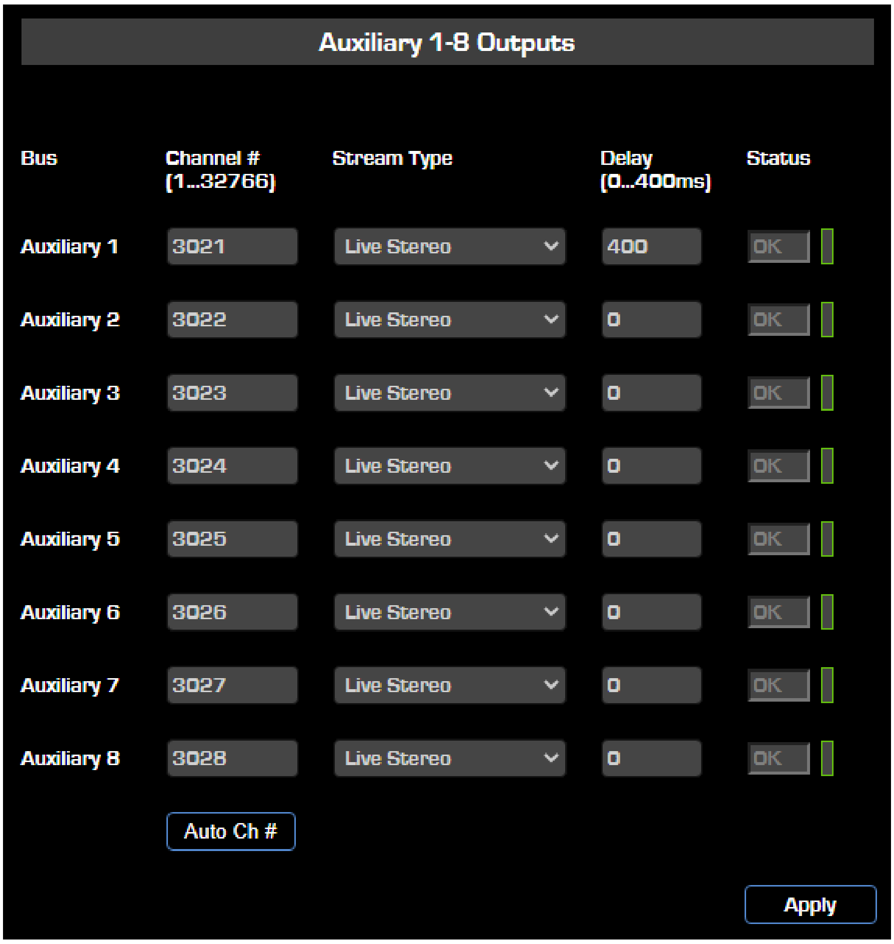
Choose one from the drop-down and click **Apply** .

> **Note –** You can also apply a 4-band EQ on the CR Headphones , by activating 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 Outputs

This is where you enable and assign unique channel numbers to 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 settings we described in the previous section, for the Channel #, Stream Type, and Delay, apply also to this section

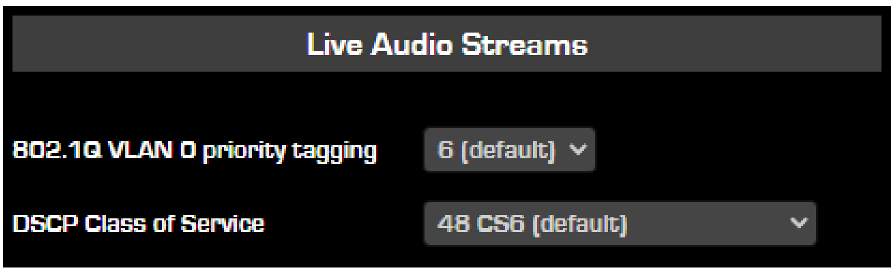


<u>The V-Mixer and V-Mode deserve their own Paragraph, so we will describe these in detail later on in this chapter.</u>

AoIP and Options

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.

Receive buffer size

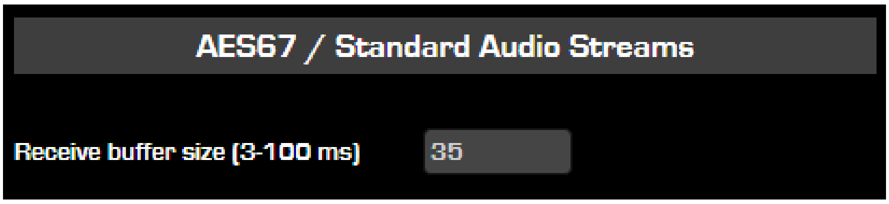
The Quasar Engine offers the possibility to tune the streams receive buffer size, except for the "Live Stereo" type of streams, in order to ensure optimal latency performance on every specific network, and maintain the lowest possible jitter. Users could experiment with the receive buffer setting to 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.

Note 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 next paragraph for more details about buffer underruns/overruns.



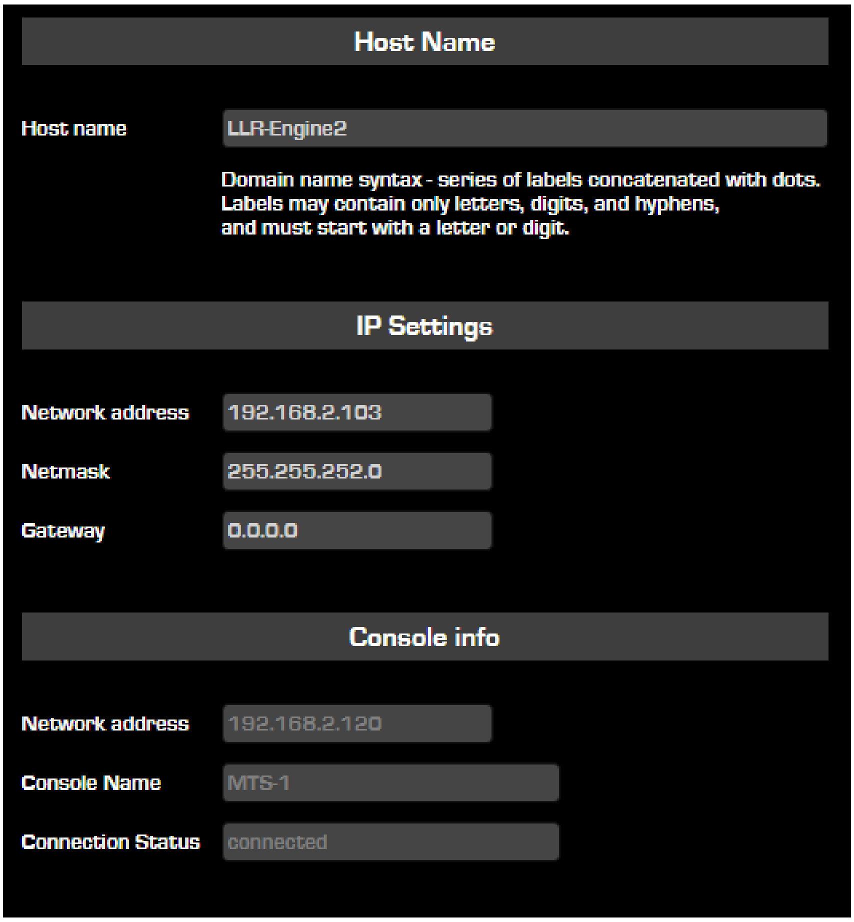
AES67 support

The Quasar Engine supports 1ms packt time streams for all its input channels. With VMode, it is possible to convert AES67 streams with Livewire streams.

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. Changes to this setting will be applied after Engine restart.

Network Setup

From this page you can set up or change your Quasar network configuration, as wellas 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 he 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 the these settings take effect only after restart.

Console Info

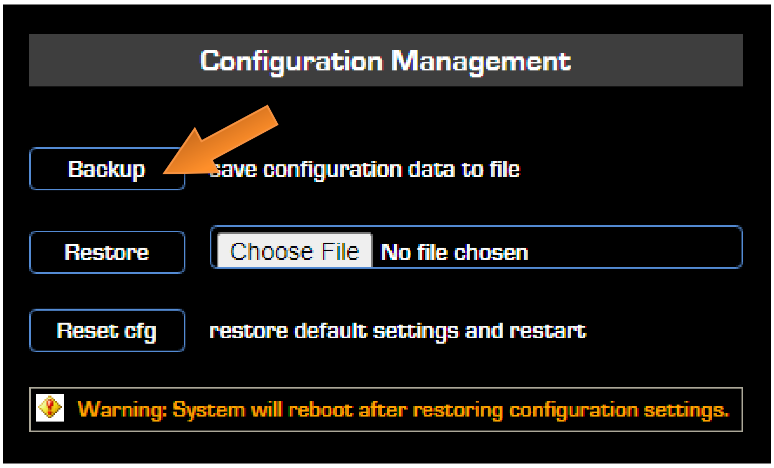
This section is only for viewing purposes.
It will show information about the connected console. In order to change the console connected to the Engine, you need to access the Quasar MTS Web UI , and select the **Engine** menu.

Backup / Restore

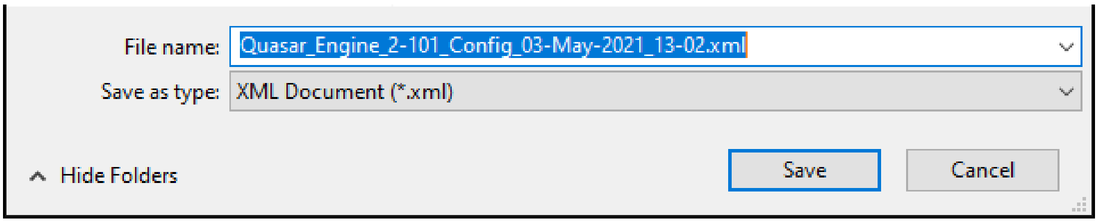
From this page, it is possible to download and save to your PC a full backup of the console configuration, in the form of an .xml file.

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.

The file name will be generated automatically , based on the Engine's IP address, Date and Time of the day.



Select Choose File , if you want to upload a previously created backup file from your archive. then press **Restore** .



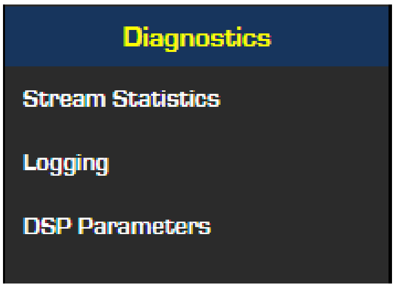
The configuration backup includes all parameters, except:

- License codes
- IP settings, conditionally by dialog option
- UI password

Warning - The System will reboot after restoring the configuration settings.

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 297 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:	Normal Packets:	Overruns:	Stream Resumed:	Time-stamp Sequence Defects:	DSP Underruns:
Input 1	Mic 1	111/0	86035472	2	1	0	0
Input 2	Mic 2	112/0	86035402	2	1	0	0
Input 3	Mic 3	113/0	86035363	2	1	0	0
Input 4	Mic 4	114/0	86035328	2	1	0	0
Input 5	DESK TB MIC	310/0	86051907	0	1	0	0
Input 9	JMP-1 Master	311/0	86051820	1	1	0	0

Individual Headphones Inputs

	Source Name:	Channel or Address:	Normal Packets:	Overruns:	Stream Resumed:	Time-stamp Sequence Defects:	DSP Underruns:

Interruptible Feedback Inputs

	Source Name:	Channel or Address:	Normal Packets:	Overruns:	Stream Resumed:	Time-stamp Sequence Defects:	DSP Underruns:

Return and Monitor Inputs

	Source Name:	Channel or Address:	Normal Packets:	Overruns:	Stream Resumed:	Time-stamp Sequence Defects:	DSP Underruns:
Studio Monitor	INTERNAL PGM1	312/0	86051920	1	1	0	0

For each Group of inputs, the following fields are displayed:

- Source Name
- Channel or Address
- Normal Packets
- Overruns
- Stream Resumed
- Time-Stamp Sequence Defects
- DSP Underruns

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 AoIP Stream number, of the connected source.

Normal Packets
This field shows the total number 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.

Overruns
The Engine checks the duration of all its internal processes execution, and the maximum number of clock cycles needed for executing a process can be constantly measured and monitored, to make sure it does not take longer time than it should. If more packets have arrived than the receive buffer can accommodate - a buffer overflow, or overrun, occurs.

As an example, when a buffer fill level is used to determine the time position, at the "micro level" an overrun means that the first packet received from this stream has arrived late relative to its normal position. When subsequent packets arrive normally, an excess amount is formed at the input, sooner or later causing an overrun. Once it happens, the stream is re-synchronized, and, unless there was another packet arriving too late, the position would be normalized and stay so. Globally, presence of overruns means high packet delay variation on the network. The latter is often being called "jitter".

This field shows the total number of Overruns that occurred to the input after the stream was connected. Ideally, the value in this field should be 0.

Stream Resumed
This field shows the total number of streams resumed after an interruption. An interruption means completely emptying the receive buffer, resulting in a stream re-synchronization. Practically this counter is close to an underrun counter, with the exception that this would not count a gap until the first packet after it. Ideally, the value in this field should be 1.

Time-Stamp Sequence Defects
This counter is increased in case a missing packet is detected, based on timestamp comparison between subsequent packets. Missing packets in absence of overruns and underruns may indicate some network interface hardware problems leading to loss of packets. Also, on a LAN with complex multi-layer topology, packets may get lost due to congestions in transit connections. However, the latter is likely to appear also as packet delay variation on terminal links. Ideally, the value in this field should be 0.

DSP Underruns

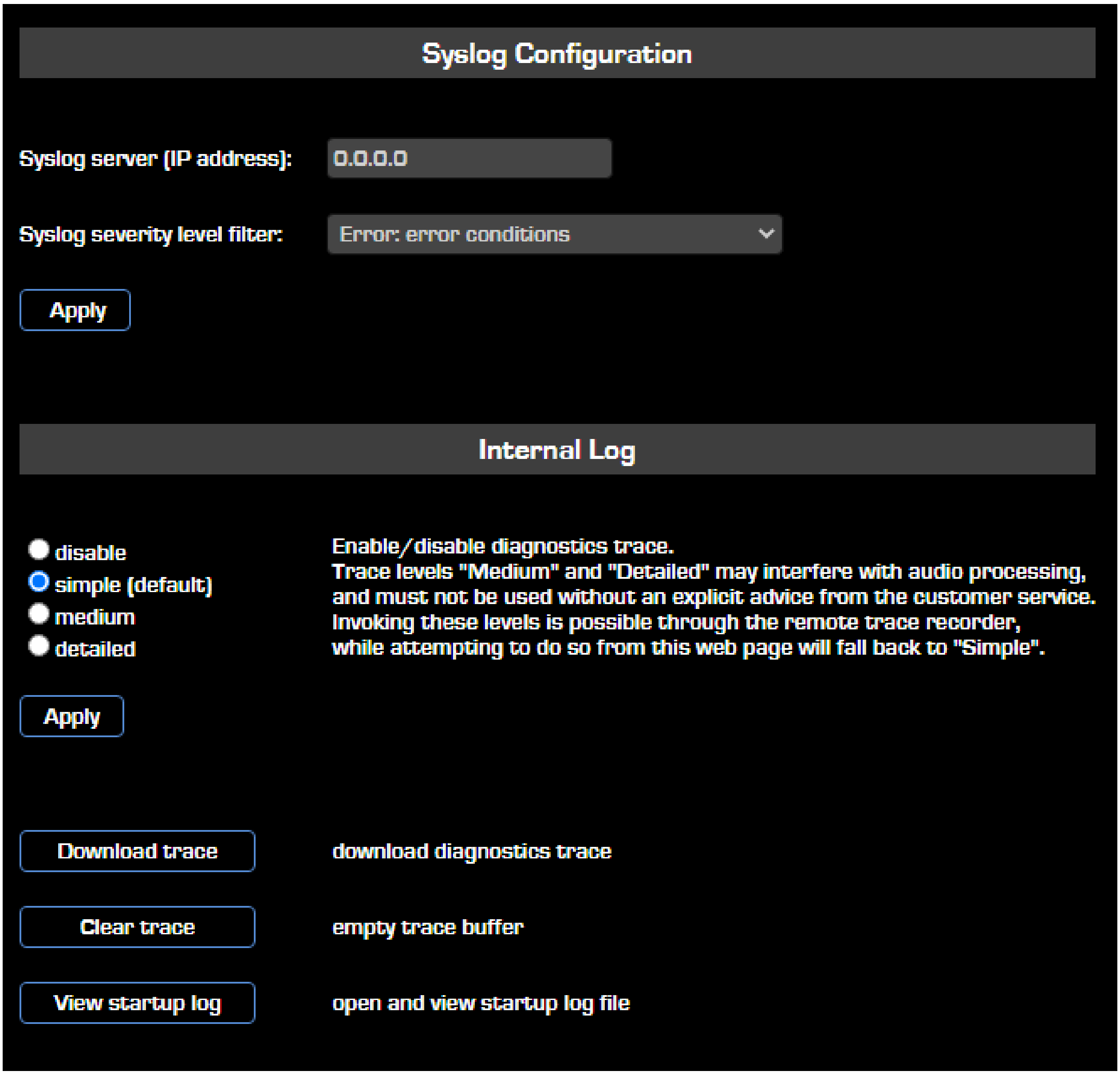
A DSP Underrun occurs when an input buffer is too short, and not capable to provide data unitl the next read cycle starts. This field counts the number of 12-sample blocks missed by the DSP input. This indicates the amount of audio data that was actually lost. A single missing "live" packet would count as 1. A single missing "standard" packet would count as 20.

Like Overruns, DSP underruns are a direct consequence of packet delay variation on the network, just measured in different units. When they occur, some audible clicks in the audio signal will be present, for example. Ideally, the value in this field should be 0.

Logging

Log Setup

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).



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 to reveal if a command sent from the control surface, is correctly acknowledged and executed by the Engine.

> **Note –** Every time a parameter is changed on the console, this page must be refreshed to display the change

Two views are available: Mixing Attributes and Metering Attributes.

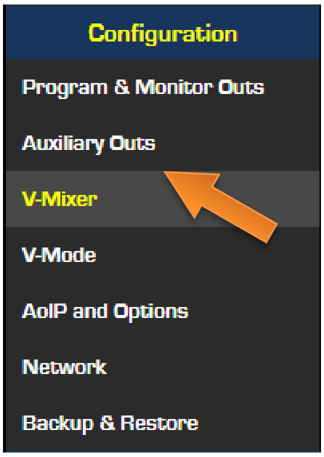
When you've done checking the DSP parameters, use the **Back to Main Page** button to go back to the main menu. Using the browser "Back" button will result in slower performance.

V-Mixer and V-Mode

In addition to the regular mixing capabilities of your Quasar console, 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.

V-Mixer

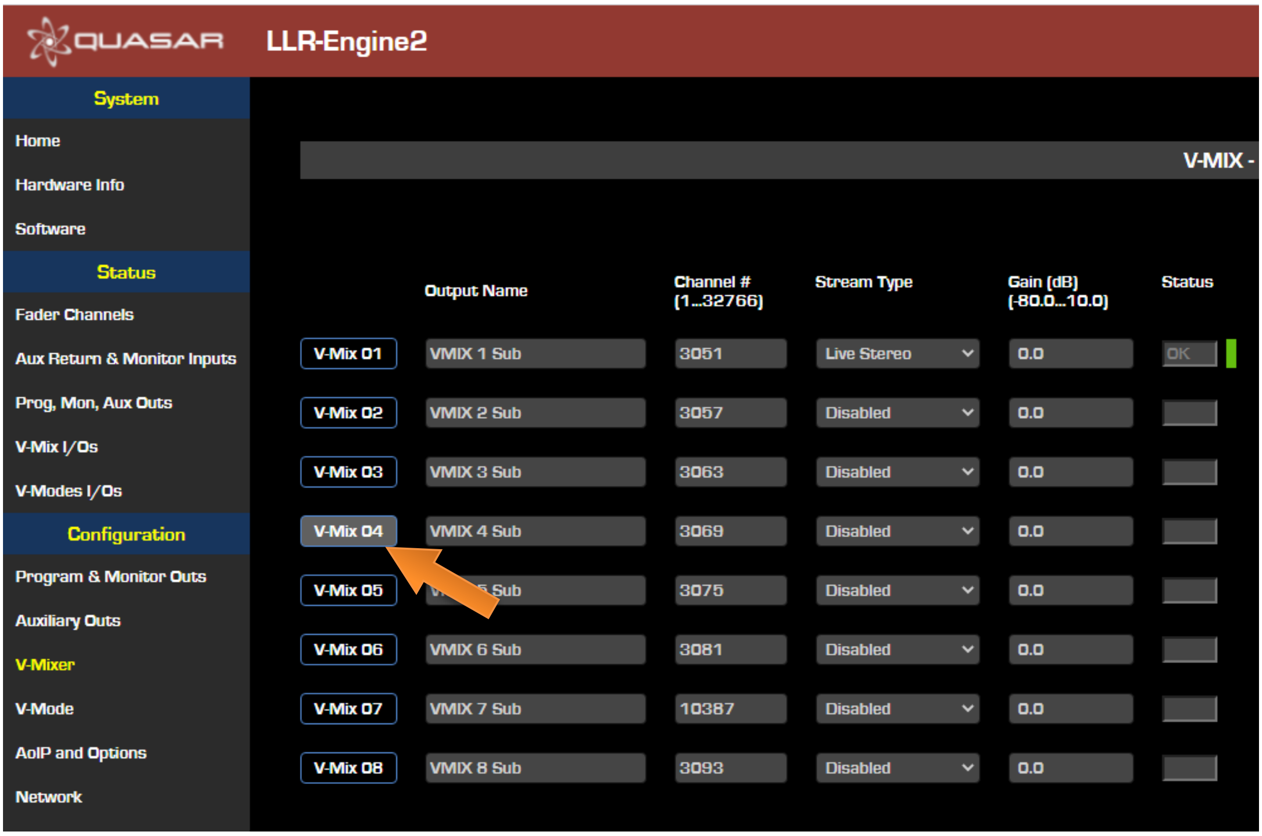
Each of the 16 VMix is a 5-Input “virtual mixer”, which can be used as a “subgroup mixer” to pre-mix up to 5 audio sources and generate a new submix source. The outputs of the 16 V-Mixers are directly available as Livewire Sources, and also summed into the Main VMix Output bus, with its own Main Output Source. This gives you the equivalent of an 80-input “virtual” mixer.



VMix works independently of the Quasar surface. 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. 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 available in the Quasar Engine.



For each V-Mixer, the following basic parameters are displayed:

Output Name
Here you can type in the name you want to assign to each V-Mixer

Channel
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 **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.

Stream Type
This drop-down lets you select the type of AoIP stream generated by the Engine.
The same options described earlier on for the Program busses, apply to this section, with the exception that Surrond type is not available here.

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.

Status
The same audio precence and stream status indicators as in other sections of the Engine are available for each VMix output.

- 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 the submixers 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. If **EXTERNAL** is selected, then 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 Ch#

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. Once a *valid* channel is selected, a small status/audio presence indicator will appear next to the button.

External Input Stream Type

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.

Gain

Here you can set the “fader gain level” for each of the Submixer input channels, by entering any value between -80.0dB and +10.0dB.

ON/OFF
These are channel ON and OFF buttons. The logic is intercanceling.

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 -∞ to the **Gain** value set in the next 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 time, and works in the same way.

Direct Out Name
The name you give the post-on/off submix channel to send back to the Axia network.

Direct Out Stream Channel
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 Stream Type
Here you can choose from **Live Stereo, Standard Stereo** to enable the direct output stream for this VMix channel. **Disabled** means no stream will be generated.

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 Submixes you intend to actively use; active submixes without any activity clutter up your network with empty streams.

Some VMix 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

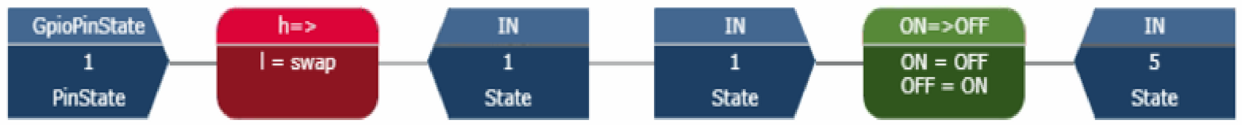
Another application is to premix the PFL channel of the Playout system directly with the Engine's PFL/AFL output. The VMix Sub Channel would then be the audio source that you would route to the PFL speaker.

Combining VMix with Pathfinder Core PRO Routing Control

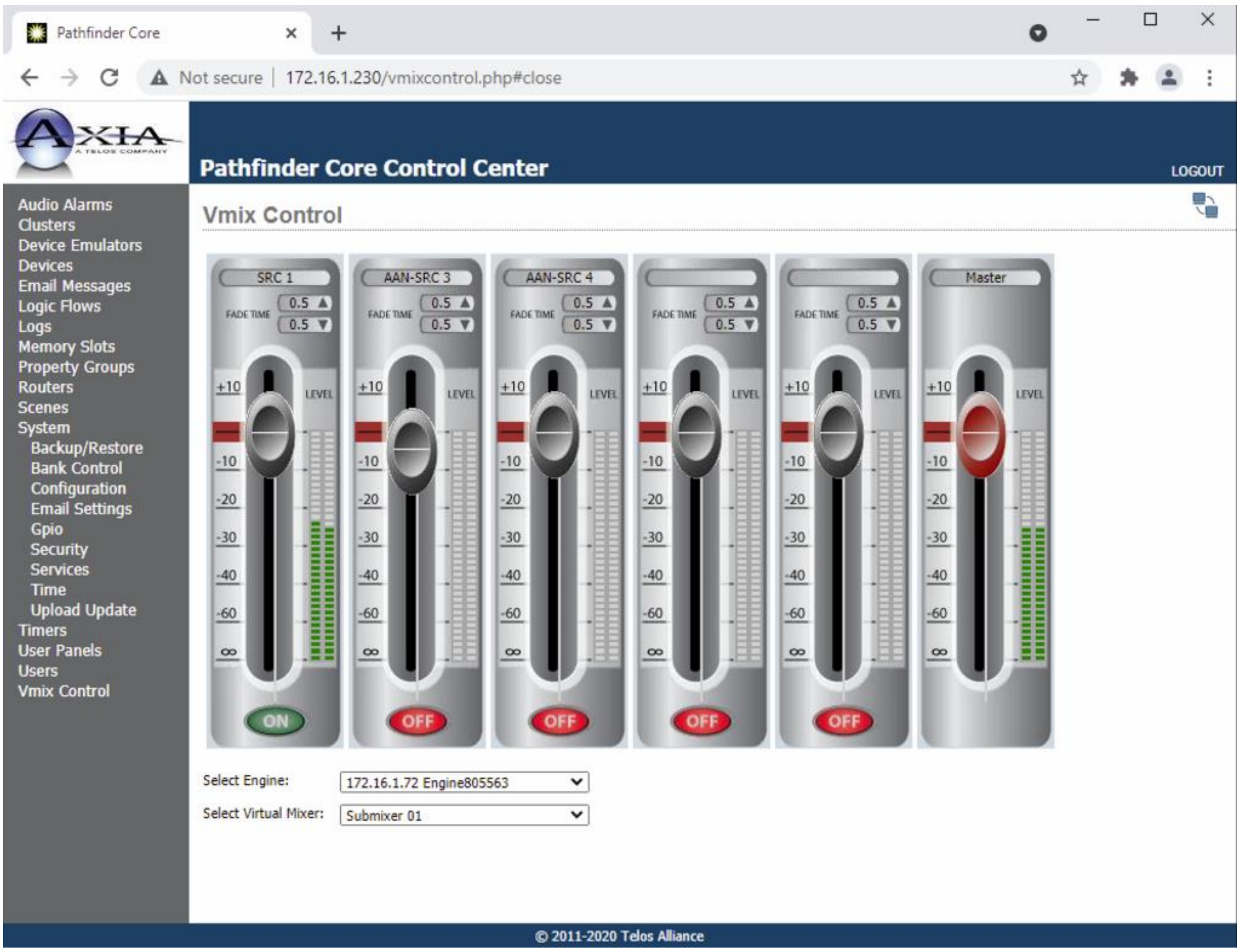
Axia Pathfinder Core PRO routing tools can be used to “control” VMix in several different ways.

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.)

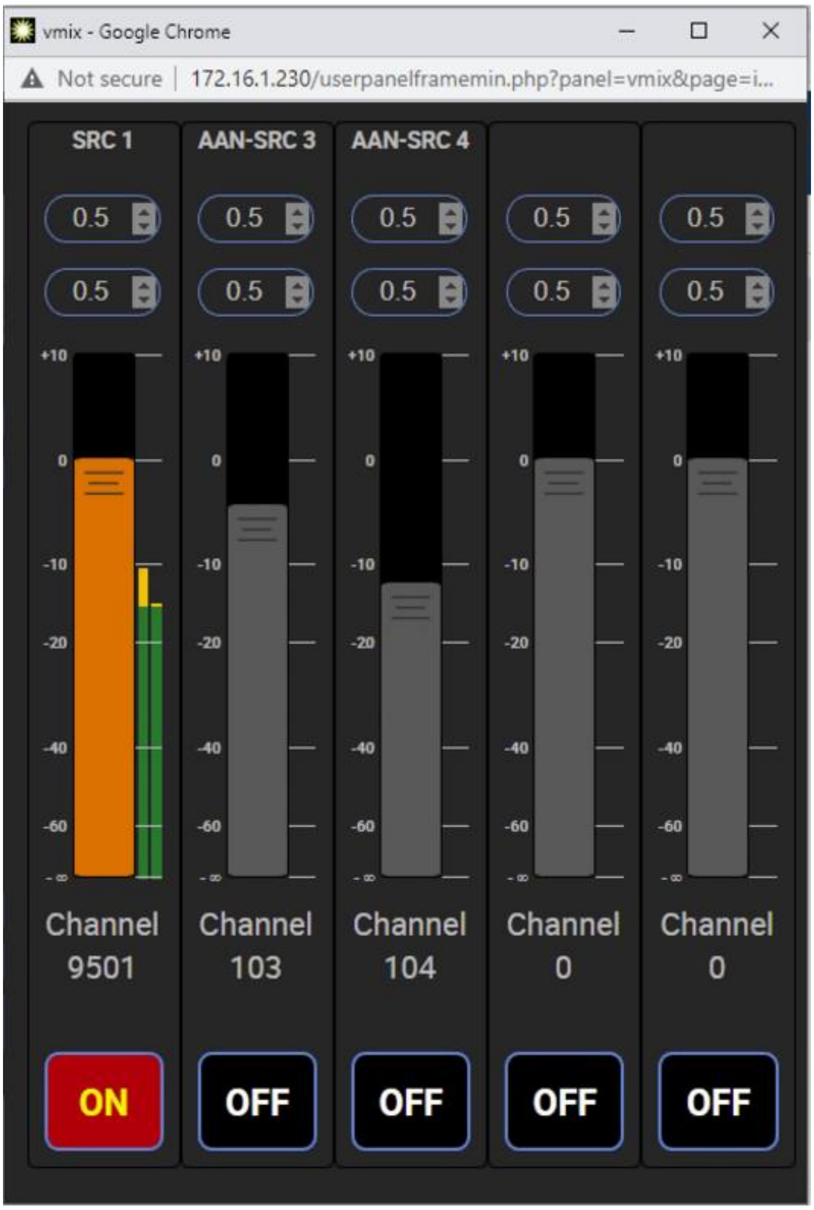


Second, Pathfinder Core PRO has a built-in VMIX controller web page specifically to provide a quick mixer user interface to the vmixers in your network. You can use parameters to specify which engines and vmixers a user may access.



Third, you can access vmixers 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 vmixers 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 VMIXer based on any of the logic flows.



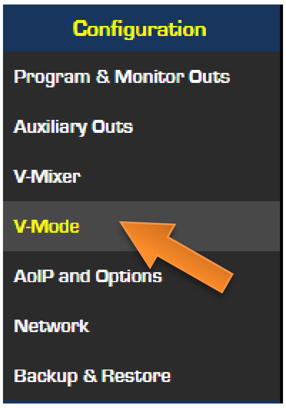
Finally, Pathfinder Core PRO's API and Generic emulators include commands to control any VMix fader that's active, so any machine that can send user defined serial or TCP commands can also control and read VMix functions through Pathfinder Core PRO.

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



VMode (short for Virtual Mode) 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 VMode instances in a Fusion Engine.

Each of the 16 VMode instances 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:

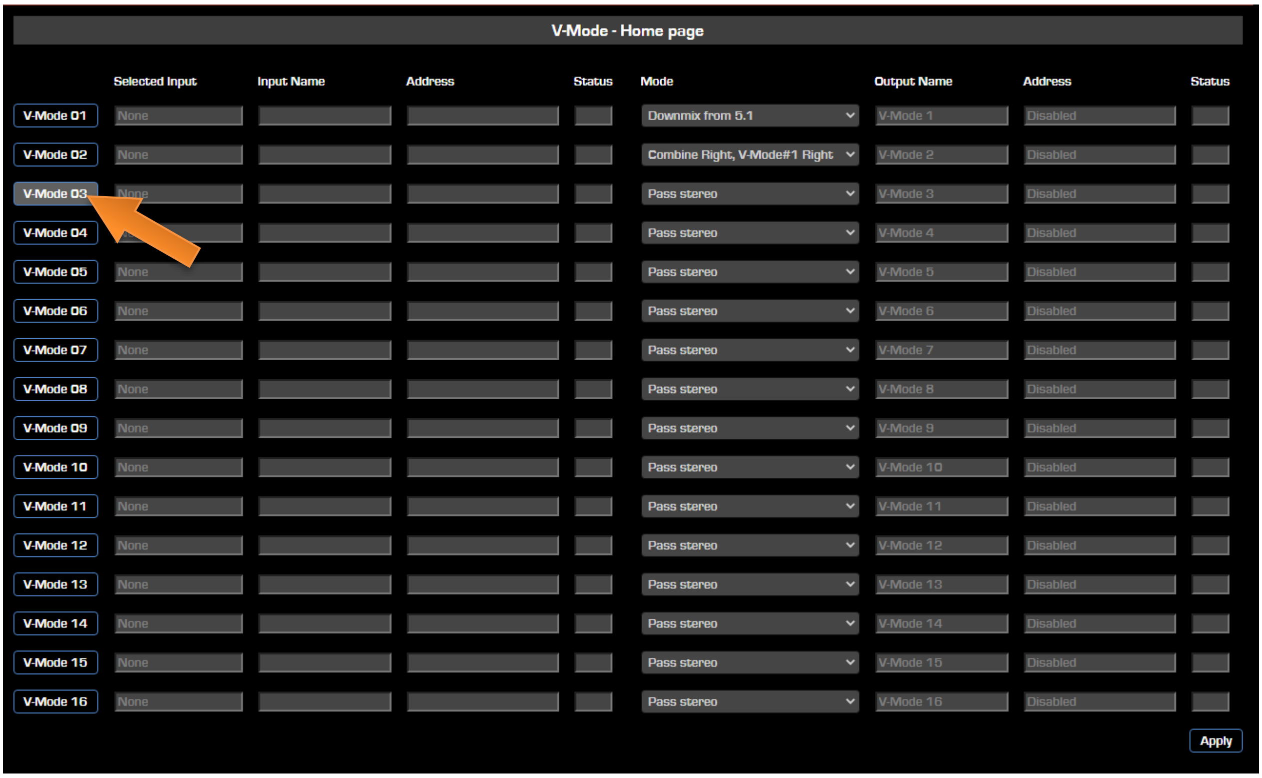
- Class "mono/stereo": streams with 1 or 2 channels
- Class "multi-channel": streams with 3 to 8 channels

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 VMode Main page offers an overview of all the 16 V-Modes available in the Quasar Engine. For each V-Mode, the following basic parameters are displayed:

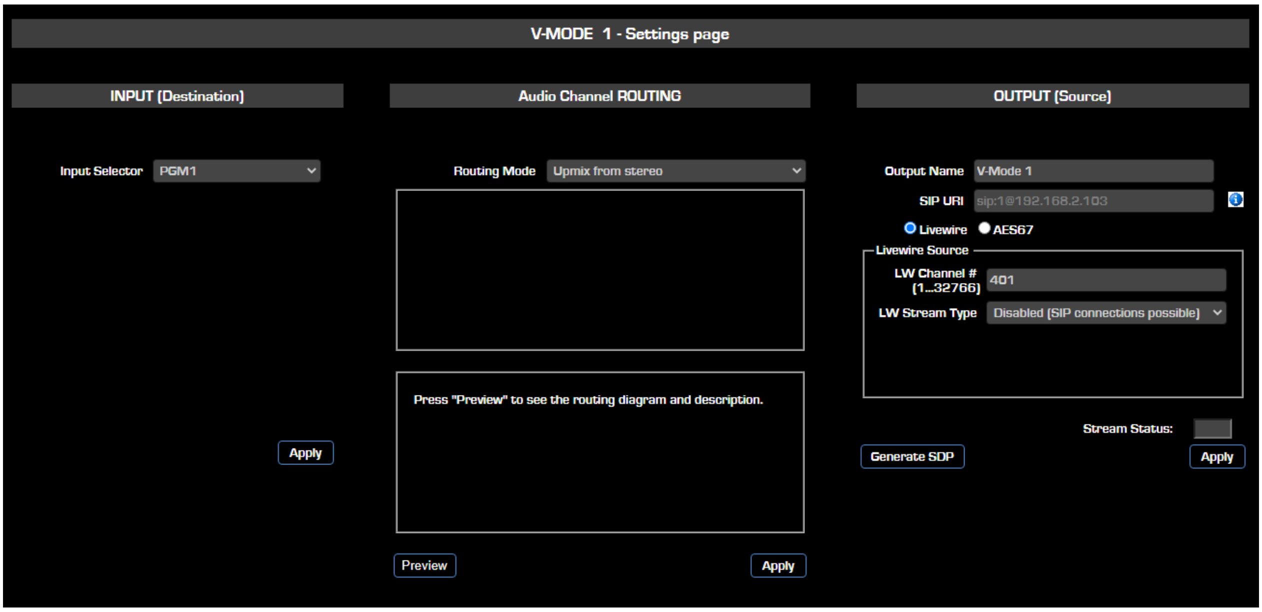
- Selected Input
- Input Name
- Input Address
- Input Status
- Mode
- Output Name
- Output Address
- Output Status



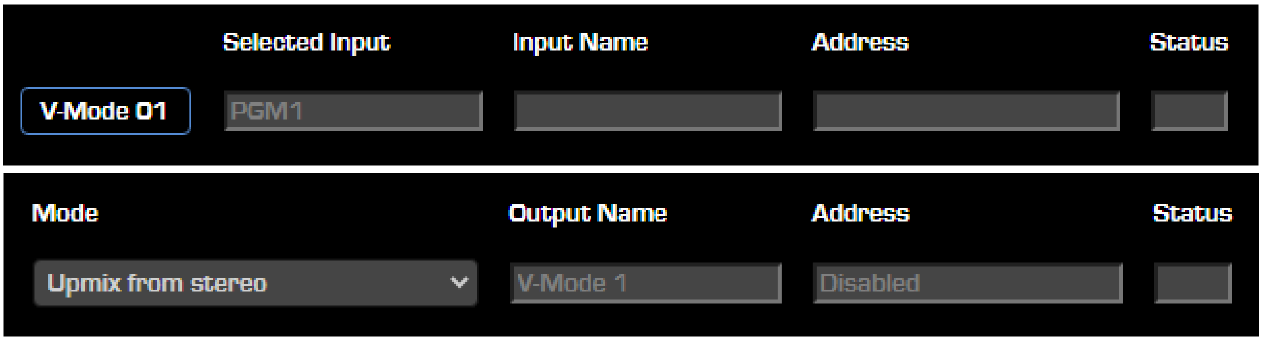
Only the **Mode** settings could be edited from this page: we duplicated here for the user's convenience. The other parameters fields in this page, which was designed with the goal of providing a view of all 16 V-Modes at a glance, are for view only.

V-Mode Settings page

In order to access and configure each V-Mode, press one of the selection buttons to the left of this screen: The V-Mode Settings page will be opened, where it is possible to configure each parameter of the 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:

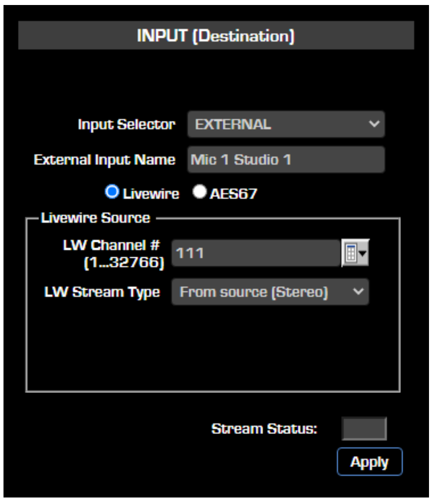
Input (Destination)

The **Input Selector** 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:

External Input name - A text field used to label the input.
External Input Type – Lets 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.



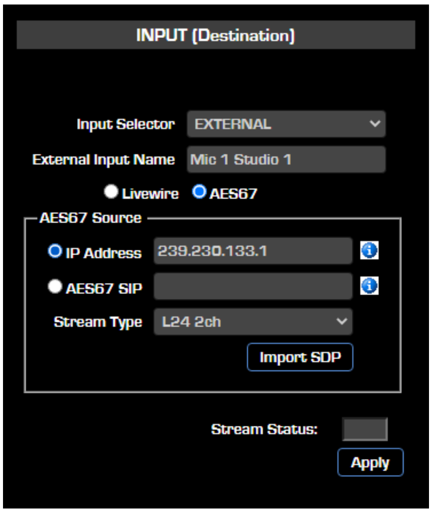
From the **LW Stream Type** drop-down, choose between:

- 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 reiceve port number. Only a stream from the indicated IP address will be accepted.



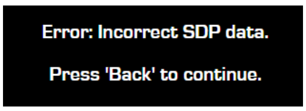
Or an **AES67 SIP** connection string can be entered, in order to initiate a SIP Unicast connection, in AES67 format. The maximum length of the string is 299 characters.

> **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.

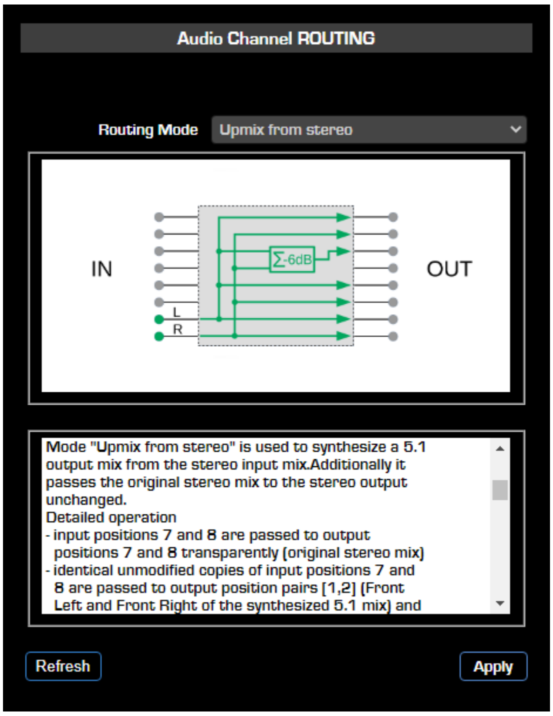
Alternatively, you can import an SDP (Session Description Protocol) string and apply it directly to the AES67 Source configuration, by clicking on the **Import SDP** button. The following page will appear:

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 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.



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 viceversa).

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

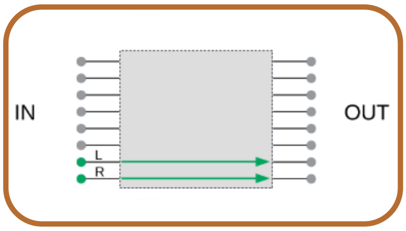
Furthermore:

11 Modes are provided, that alter the internal channel routing and gain level within the matrix,

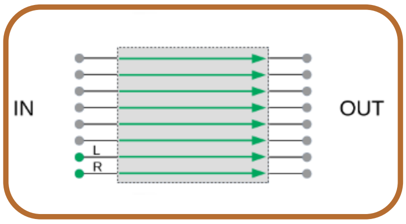
and 12 Modes are provided, that route signals between two adjacent V-Modes and alter the signal routing inside each matrix, to recombine the signals before they are sent to the outputs of the two V-Modes.

Following, is a detailed description of each Mode option:

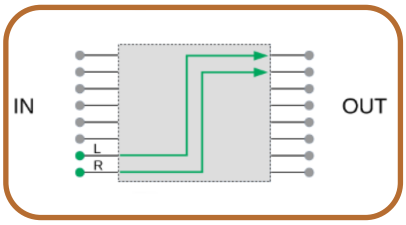
Pass stereo
transparently passes a mono or stereo input to the output.
Detailed operation:
- input 7 and 8 are passed to output 7 and 8 transparently
- silence is fed to output 1 to 6



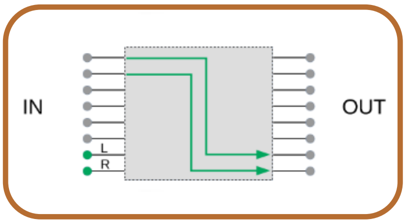
Pass multi-channel
transparently passes a multi-channel input to the output.
Detailed operation:
- input 1 to 8 are passed transparently through the matrix



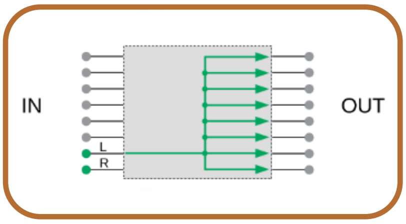
Transpose stereo to multi – channel [1, 2]
passes a mono or stereo input to the lowest channel numbers of a multi-channel output.
Detailed operation:
- input 7 and 8 are passed to output 1 and 2
- silence is fed to output 3 to 8



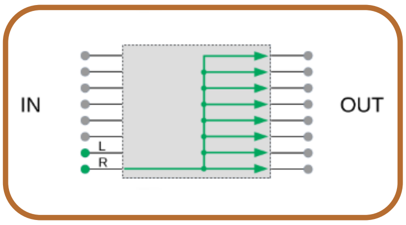
Transpose multi – channel [1, 2] to stereo
passes the lowest channel numbers of a multi-channel input to a mono or stereo output.
Detailed operation:
- input 1 and 2 are passed to output 7 and 8
- silence is fed to output 1 to 6



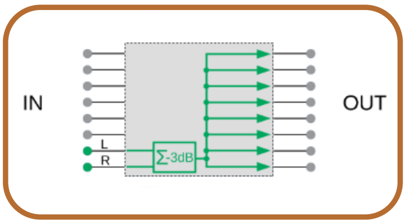
Upmix from Left
replicates a mono input or the Left channel of a stereo input to all 8 output channels.
Detailed operation:
- identical unmodified copies of input 7 are passed to output 1 to 8



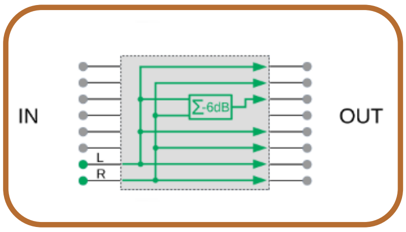
Upmix from Right
replicates the Right channel of a stereo input to all 8 output channels.
Detailed operation:
- identical unmodified copies of input 8 are passed to output 1 to 8



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

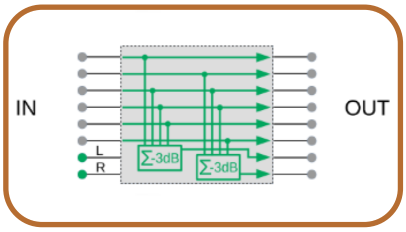


synthesizes a 5.1 output mix from the stereo input mix. Additionally it passes the original stereo mix to the stereo output unchanged.

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

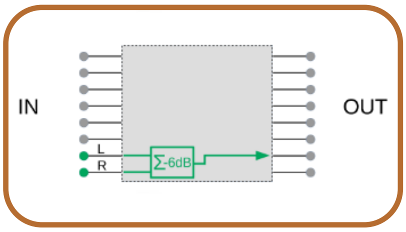


synthesizes a stereo output mix from the 5.1 input mix. Additionally it passes the original 5.1 mix to the 5.1 output unchanged.

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

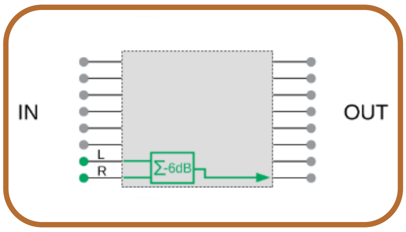


Sends a mono sum of the input stereo mix to the Left channel of the stereo output.

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

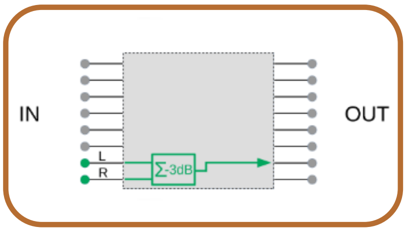


Sends a mono sum of the input stereo mix to the Right channel of the stereo output.

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

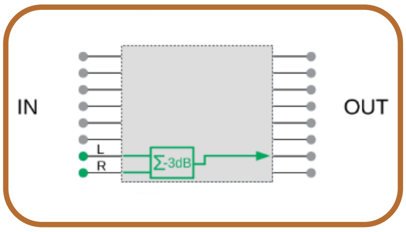


Sends a mono sum of the input stereo mix to the Left channel of the stereo output.

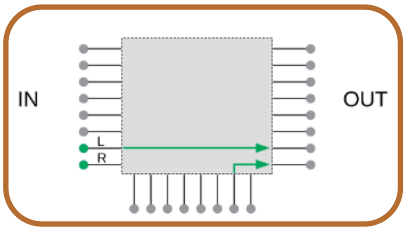
Detailed operation:

- 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

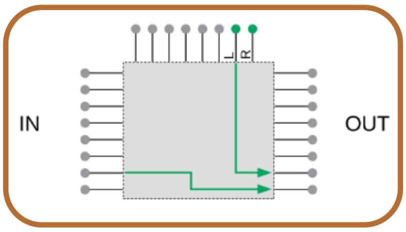
Split Right - 3dB
Sends a mono sum of the input stereo mix to the Right channel of the stereo output.
Detailed operation:
- the sum of input 7 and 8, attenuated by 3 dB, is passed to output 8
- silence is fed to output 1 to 7



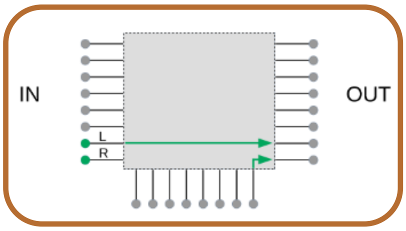
Combine Left, V-Mode#2 Left
Creates a single output from a pair of inputs.
This is an odd-number instance, and it is paired with the next even-number instance. The output signal is created from the Left channel of this instance's own stereo input shown on the left, and the Left channel of the stereo 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



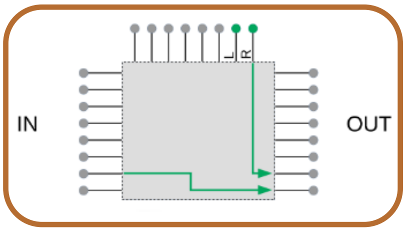
Combine Left, V-Mode#1 Left
Creates a single output from a pair of inputs.
This is an even-number instance, and it is paired with the previous odd-number instance. The output signal is created from the Left channel of this instance`s own stereo input shown on the left, and the Left channel of the stereo 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



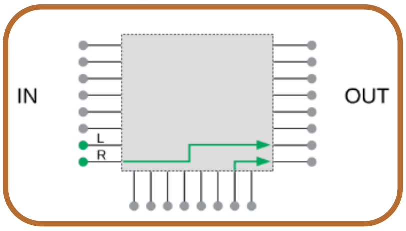
Combine Left, V-Mode#1 Right
Creates a single output from a pair of inputs.
This is an even-number instance, and it is paired with the previous odd-number instance. The output signal is created from the Left channel of this instance's own stereo input shown on the left, and the Right channel of the stereo 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.

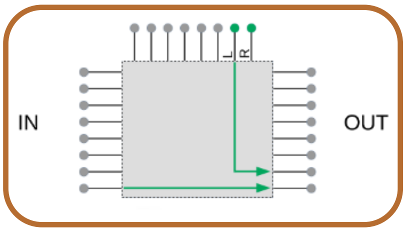


Detailed operation:

- 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.

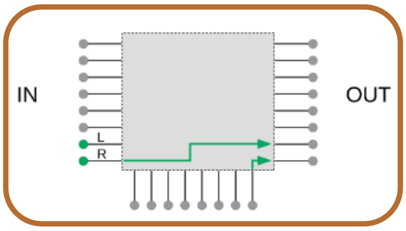


Detailed operation:

- 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.

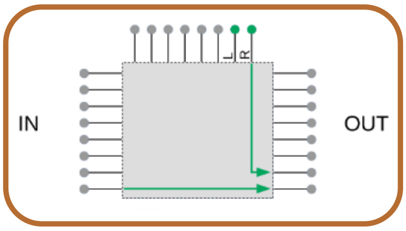


Detailed operation:

- 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

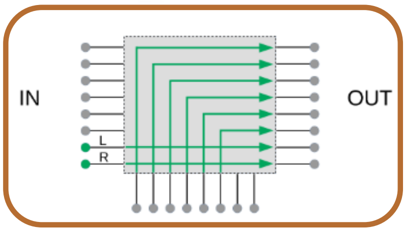
Creates a single output from a pair of inputs.
This is an even-number instance, and it is paired with the previous odd-number instance.
The output signal is created from the Right channel of this instance's own stereo input shown on the left, and the Right channel of the stereo input of the paired instance, shown above the matrix.



Detailed operation:

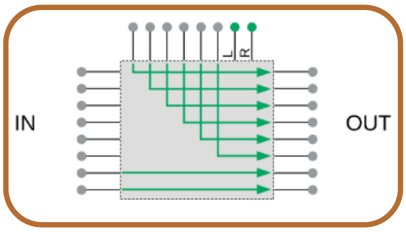
- input 8 from the paired instance is passed to output 7
- input 8 from this instance is passed to output 8
- silence is fed to output 1 to 6

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.



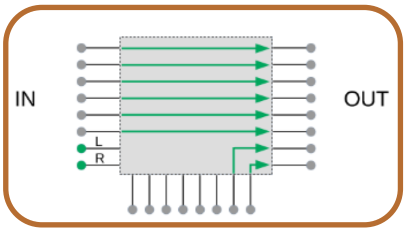
Detailed operation:
- input 7 and 8 from this instance are passed to output 7 and 8
- input 1 to 6 from the paired instance are passed to output 1 to 6

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.



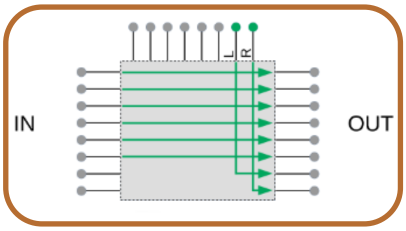
Detailed operation:
- input 7 and 8 from this instance are passed to output 7 and 8
- input to 6 from the paired instance are passed to output 1 to 6

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

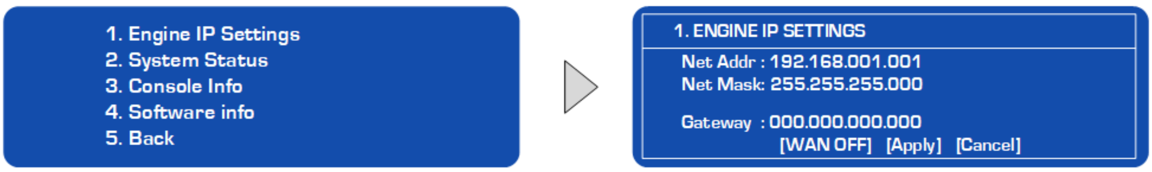
Intentionally Left Blank

Appendix A

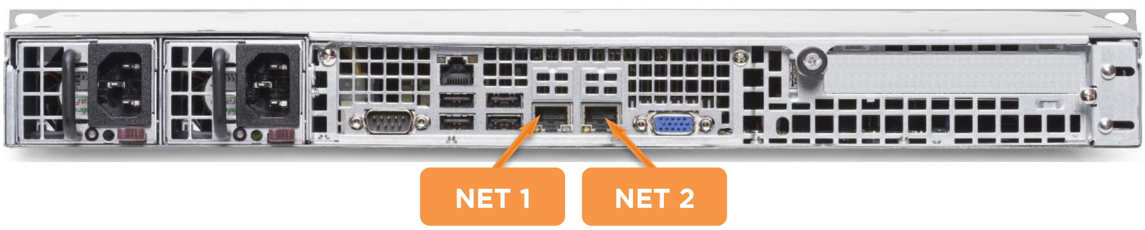
Previous Quasar Engine Platforms

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

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
Mechanical Data
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 -
Power Supply
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
Operating Temperatures
-10 degrees C to +40 degrees C, <90% humidity, no condensation
CE CONFORMANCE 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 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, -75 dBu to -20 dBu
Input Headroom: >20 dB above nominal input
Output Level: +4 dBu, nominal
Analog Line Inputs
Input Impedance: >40 k Ohms, balanced
Nominal Level Range: Selectable, +4 dBu 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: +4 dBu
Maximum Output Level: +24 dBu
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 0 dBFS,

105 dB “A” weighted to 0 dBFS

Analog Input to Digital Output: 105 dB referenced to 0 dBFS

Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB “A” weighted

Digital Input to Digital Output: 138 dB

Equivalent Input Noise

Microphone Preamp: -128 dBu, 150 ohm source, reference -50 dBu input level

Total Harmonic Distortion + Noise

Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input, +18dBu ouput

Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output

Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS

Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

Crosstalk Isolation, Stereo Separation and CMRR

Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kH

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

CE

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/ Model Number		
	Quasar MTS-MON Module	2001-00568
	Quasar XR-4FAD Module, Motorized	2001-00569
	Quasar Mic in/HP out Module	2001-00579
	Quasar table-top frame 2.5U – 8.5U	2001-00550, 2001-00551, 2001-00552, 2001-00553
		2001-00554, 2001-00555, 2001-00556

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 62368-1:2014/A11:2017 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: APRIL 7, 202

UK Conformity Assessed

UK
CA

DECLARATION OF CONFORMITY

MANUFACTURER:	Telos Alliance 1241 Superior Avenue Cleveland, Ohio 44114 USA	
PRODUCT / MODEL:	Quasar MTS-MON Module	2001-00568
	Quasar XR-4FAD Module, Motorized	2001-00569
	Quasar Mic in/HP out Module	2001-00579
	Quasar table-top frame 2.5U-8.5U	2001-00550, 2001-00551, 2001-00552, 2001-00553, 2001-00554, 2001-00555, 2001-00556
REPORTS:	F2P23592A-01E, F2P23529A-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 62368-1:2014/A11:2017

BS 63000:2018

This declaration of conformity is issued under the sole responsibility of the manufacturer.

Signature of manufacturer: ______________________ **DATE** APRIL ~~07~~ 07, 2021