QUASAR

User Manual

Includes Quasar Engine User Guide

AXIA
THE TELOS ALLIANCE

By L.L.R. - Manual Rev 1 (1.0)
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Notices and Cautions

CAUTION:

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 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 Manual

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WARRANTY

This product is covered by a five year limited warranty, the full text of which is included in this manual.

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.

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. All other customers should contact local representative to make arrangements for service.

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 E-MAIL:

- 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

Did you know that all Telos Alliance products come with a 5-Year Warranty? Take a moment to activate your coverage online at http://telosalliance.com/product-registration/ .

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Creating the Most Exciting and Engaging Audio Experiences Imaginable

Congratulations on your new Telos Alliance product!

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

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

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Quick Start

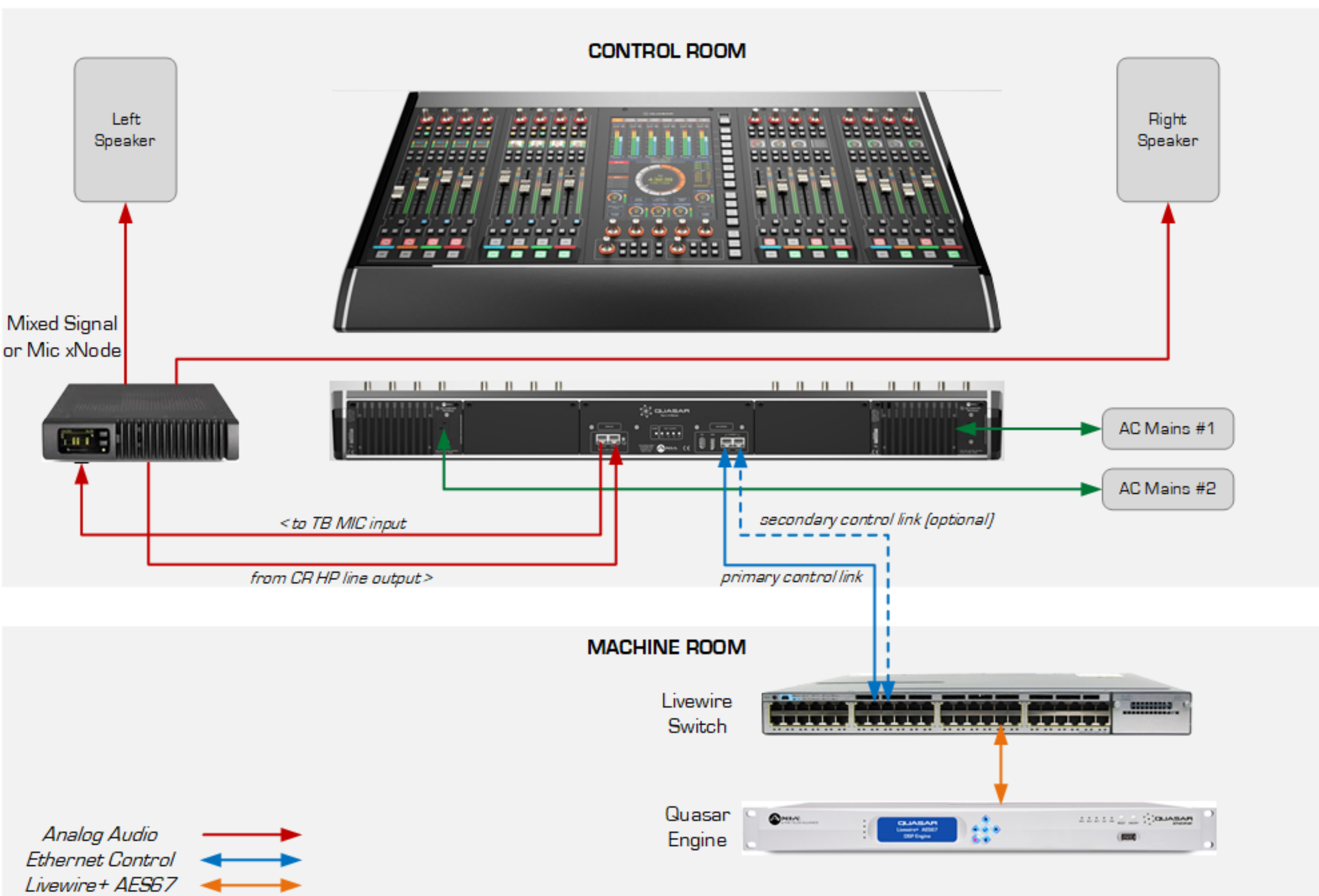
Setting up the Surface with the Engine

Setup of a networked audio mixing console, though we have made it as easy as possible, is challenging to condense into a few pages. The following material is intended to help the busy engineer get up and running in a few minutes.

This Quickstart Guide assumes a few things:

- That the reader has some knowledge of network basics and terminology,
- That the reader is familiar with other Axia Livewire products,
- And that the reader has a network switch correctly configured for Livewire

We will take you through the basic steps to configure your new Quasar and have you up and running in no time. But before we get started, let's make sure you are familiar with the Quasar System anatomy.



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

There are a few additional items required for a successful installation. Please check the list below.

Included items:

- Quasar Control Surface
- Quasar Mains Cable(s)
- CAT5 Ethernet Cable

Items you need to supply:
- A Quasar Engine mixing engine
- Switch configured for Axia network
- PC with access to your Axia network

Then, let's check your installation type and make sure you have what you need.

Installation type: You could be installing a Table-Top or Flush-Mount console, with single or split frame. Only the Quasar Engine can be connected to the Quasar Surface. This consists in an industrial rackmount server platform, with redundant PSU and fans, so it will have to go in a machine room. 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:

Network Switch – Configuration for Livewire
Quasar Surface - Physical connections
Quasar Surface - Network configuration
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Checking Connection to the Quasar Engine

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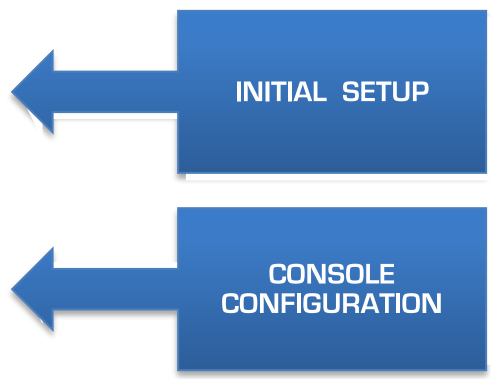
Input Source Creation and Configuration
Assign Sources to Input Channels
Program assignment & monitoring



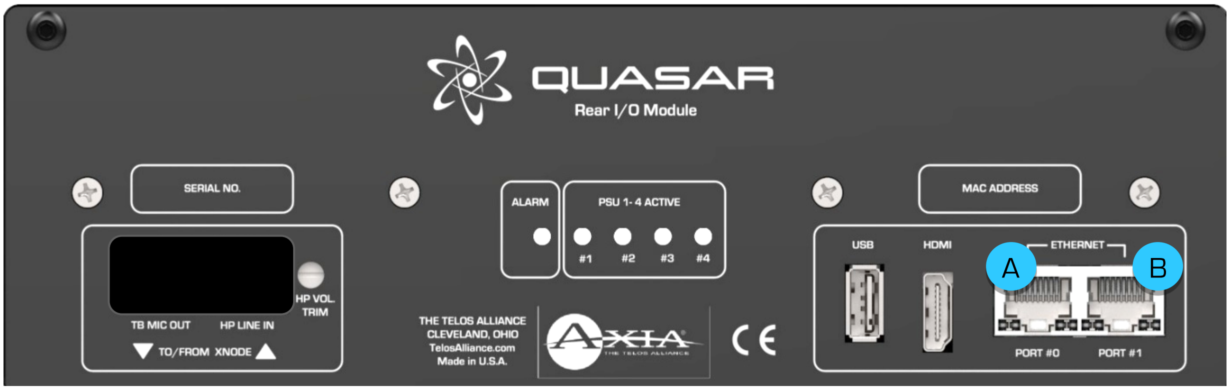
Network Switch – Configuration for Livewire

Please do not assume that any network switch will work out of the box, for handling the intense traffic generated by AoIP audio streams. Multicast traffic generated by AoIP streams can "flood" the switch if not properly managed and shaped via a correct configuration of the switch ports.
A list of approved Ethernet switches is published on our website, at this link: https://www.telosalliance.com/Axia/What-Ethernet-Switches-has-Axia-Approved. In the *Configuration* section of this webpage, you will find guidelines for specific switch models.

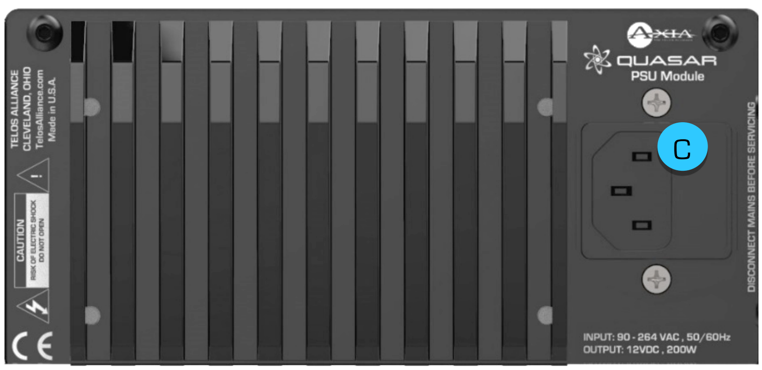
An Axia Applications Engineer will be happy to assist you in determining the correct switch for your application – please contact us for assistance.

Quasar Surface - Physical connections

1. **Connect** one Ethernet cable to the primary network port labeled "PORT #0"(A) on the Quasar Console Rear I/O board, **connect** the other end to a configured Gigabit Ethernet switch.



2. **Optionally,** it is also possible to use the secondary Ethernet port, labeled “PORT #1” (B) to create a redundant connection link. This will require a specific configuration of the switch ports. Please refer to the following chapters for details about this option.
3. **Connect** an Ethernet cable to the Livewire port (F) on the Quasar Engine, and **connect** the other end to a configured Gigabit Ethernet switch.



4. **Connect** the Quasar surface Power Supply Unit (or Units, if you have more than one installed in your frame) to AC Mains (C) using the Axia Power Supply power cable (or cables) provided. If you have more than one power supply installed and choice of standard and uninterrupted mains in your studio, it is a good idea to connect each PSU to a different type of Mains supply.
5. **Make Sure** your AC Mains are properly grounded! The Quasar (like every other professional device) is grounded through the AC Mains cables, and does not require a separate chassis ground.
6. Continue to Network Configuration with your Axia Quasar surface modules.

Quasar Surface - Network configuration

Once the console is started up, the Quasar Master Touchscreen Module (MTS Module) will present you with its Home page, were a round clock is displayed. Above the clock you will notice a warning message: “No Connection to Engine”. This is normal.

Now you need to access the System Setup page: push and hold for 5 seconds the Monitor Options key (labeled “MON OPT” and located between the two Control Room volume pots). Here you will need to assign an IP address to your console, as well as the Network Mask, and bind the surface to the engine.

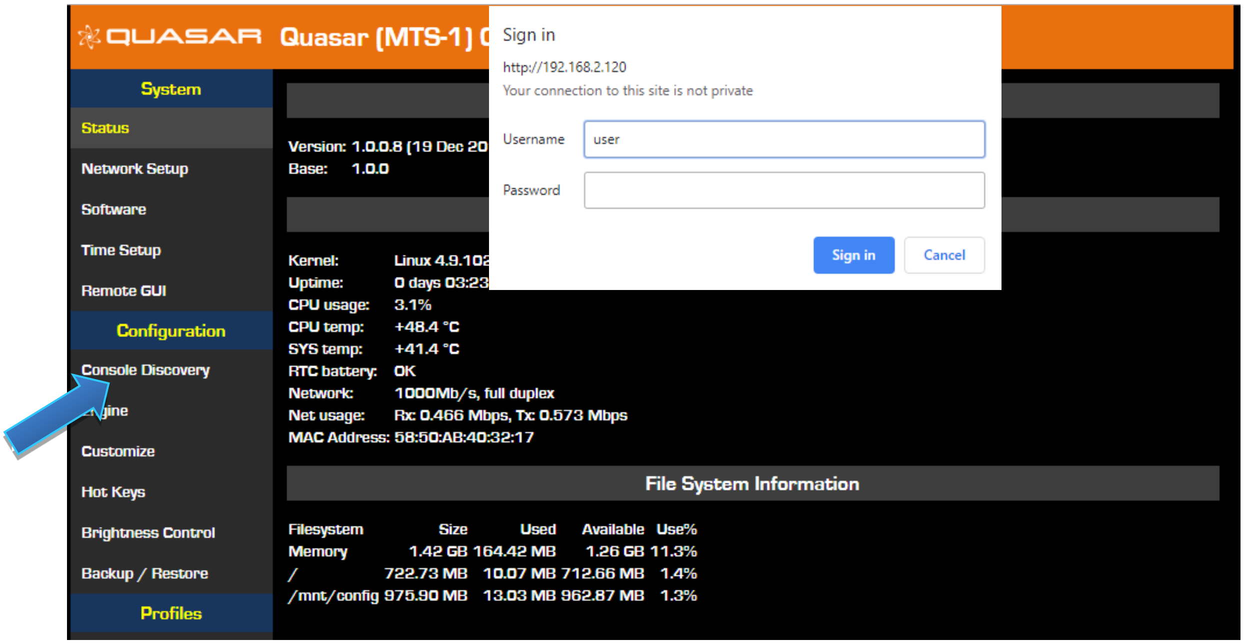
At the bottom of this page you can associate your Quasar Engine to the surface, by entering the Engine’s IP address. Addresses can be entered directly from the touchscreen, or by using the rotary encoders below the display. A configured gateway is optional, and not required for normal console operation in a closed network.

Press the “Save and Reboot” button when done. Once the Master Module IP address is set, you can now access the console Web UI to configure each fader module’s IP address.

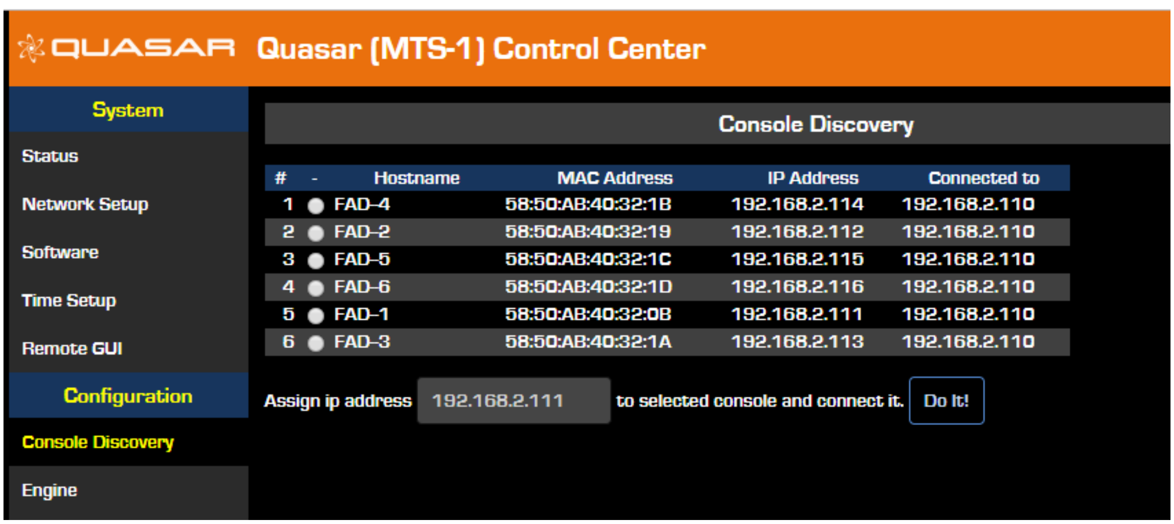
Quasar Surface – Modules’ discovery & configuration

Using a PC connected to your studio network, **launch** a web browser and **enter** the IP address you previously assigned to the MTS into the browser’s address bar.

In the left column, click on *Console Discovery* menu. When prompted for authentication **enter** user name “user”, and leave the password field blank.



Here you will find all Fader modules connected to the Master, listed by their MAC and IP addresses, which will be pre-assigned at the factory. If the pre-assigned addresses fits your networking scheme, then you can skip to the next step.



In case you need to change these addresses according to your network scheme, you can assign a new IP address to each module, by selecting it with the radio button, **entering** the desired address in the box at the bottom and **pushing** the “Do It!” button.

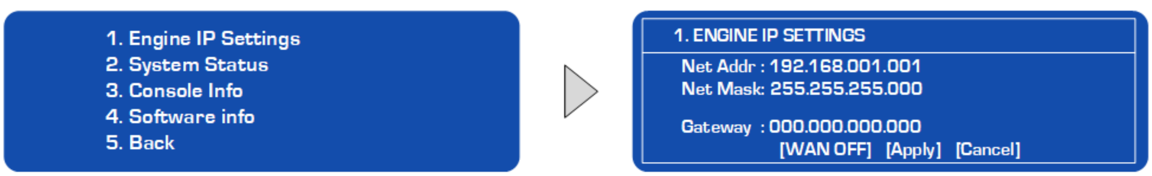
<u>Tip:</u> We recommend choosing for the Master module an IP address with the fourth octed ending with a 0 (zero), and an address with the fourth octet ending with 1,2,3,….etc for the first, second, third, Fader module in your surface and so on. This will make easier to remember each module’s IP address for fast UI access.

Once all addresses have been set, please check access to each fader module by entering its IP

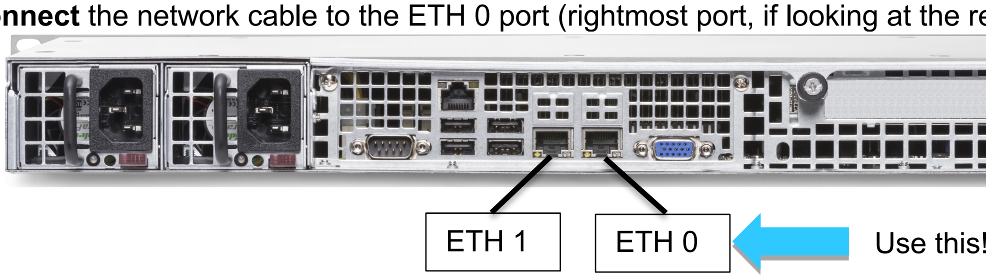
address in your browser and by navigating each module's Web UI. You can also click the IP address directly from the list you see in the Console Discovery.
The default access credentials will be the same used for the MTS Module: user – *no password.*

Quasar Engine – Installation & configuration

1. **Install** your Quasar Engine platform in a suitable environment, like an air conditioned machine room. Please make sure there is one empty position in your rack above the engine, and one below, in order to let the air flow inside from the front panel. Please refer to the user manual for details about the engine installation.
2. **Connect** 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. An IP address is needed.
3. **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.



4. **Push** the UP/DOWN arrows buttons to select *Net Address* field, and **Push** the **V** button to enter.
5. **Use** the LEFT/RIGHT arrow buttons to select each digit, the UP/DOWN arrows buttons to increment or decrement the value, and the **V** button to enter.
6. **Move** the cursor to the righ and select the ↵ symbol. **Push** the **V** button to confirm the setting.
7. Repeat for Netmask as needed.
8. Once on the Engine IP settings are entered, move cursor down to *[Apply]* and **Push** the **V** button to select.
9. 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.
10. **Connect** the network cable to the ETH 0 port (rightmost port, if looking at the rear panel).



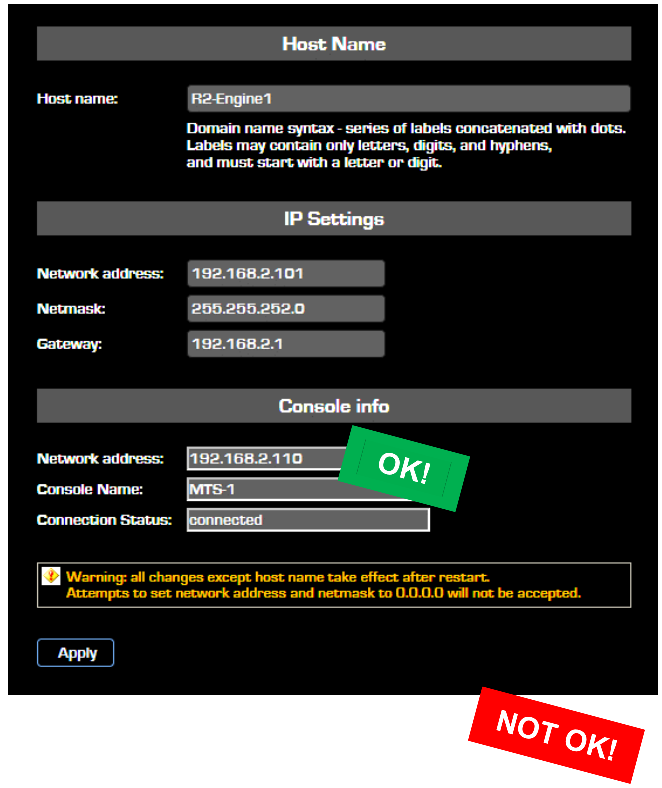
Checking connection to the Quasar Engine

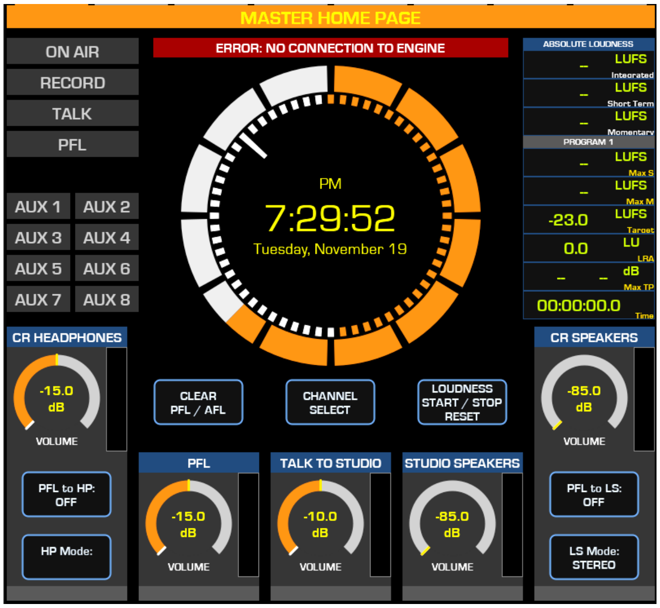
Using a PC connected to your studio network, **launch** a web browser and **enter** the IP address assigned to the Engine into the browser's address bar. When prompted for authentication **enter** user name "user", and leave the password field blank.

Let's double-check the connection to the Surface from the Quasar Engine Web UI, by going to the *Network* menu.

In *Console info*, make sure that the Network Address field shows the IP address of you Quasar Master Touchscreen module, abind the Connection Status is "connected".

Now please double-check that the warning message above the clock on the MTS Home page is disappeared.





The Quasar Surface is now recognized by the network and linked to the Quasar Engine. You're ready for the next step!

Now you can proceed with configuring your Console.

Quasar Engine – Audio outputs configuration

1. Using a PC connected to your studio network, go to the Engine Web UI home page
2. In the left column, **select** *Program and Monitor outputs*.
3. **Enter** the planned channel numbers for Quasar Engine (Livewire Sources) outputs.
4. **Enable** the streams you need to be active on the Engine
5. **Click** the "Apply" button.

Tip: We recommend choosing a Channel ID which has the first 2 or 3 digits corresponding to the last

Host name: R2-Engine1

Main, Auxiliary and Monitor Outputs

	Channel [1..32766]:	Mode:	Delay[0..400ms]:	Status:	Audio:
Program 1	1011	Live Stereo	0	OK	
Program 2	1012	Live Stereo	0	OK	
Program 3	1013	Live Stereo	0	OK	
Program 4	1014	Live Stereo	0	OK	
Program 4 Record	1015	Live Stereo	0	OK	
Aux Send 1	1021	Live Stereo	0	OK	
Aux Send 2	1022	Live Stereo	0	OK	
Aux Send 3	1023	Live Stereo	0	OK	
Aux Send 4	1024	Live Stereo	0	OK	
Aux Send 5	1025	Live Stereo	0	OK	
Aux Send 6	1026	Live Stereo	0	OK	
Aux Send 7	1027	Live Stereo	0	OK	
Aux Send 8	1028	Live Stereo	0	OK	
CR Monitor Direc	1040	Live Stereo		OK	
CR Monitor	1041	Live Stereo		OK	
CR Headphones	1042	Live Stereo		OK	
Preview	1043	Live Stereo		OK	
Talk to CR	1044	Live Stereo		OK	
Guest Headphones	1045	Live Stereo		OK	
Studio Monitor	1046	Live Stereo		OK	
Talent Hdphones	1047	Live Stereo		OK	
Talk to Ex[illegible]al	1048	Live Stereo		OK	

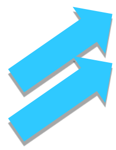
Apply

octet of the Engine IP address.

In this example, the engine has IP address 192.168.2.101

This will make easier to spot the right Engine channels when browsing a large network.

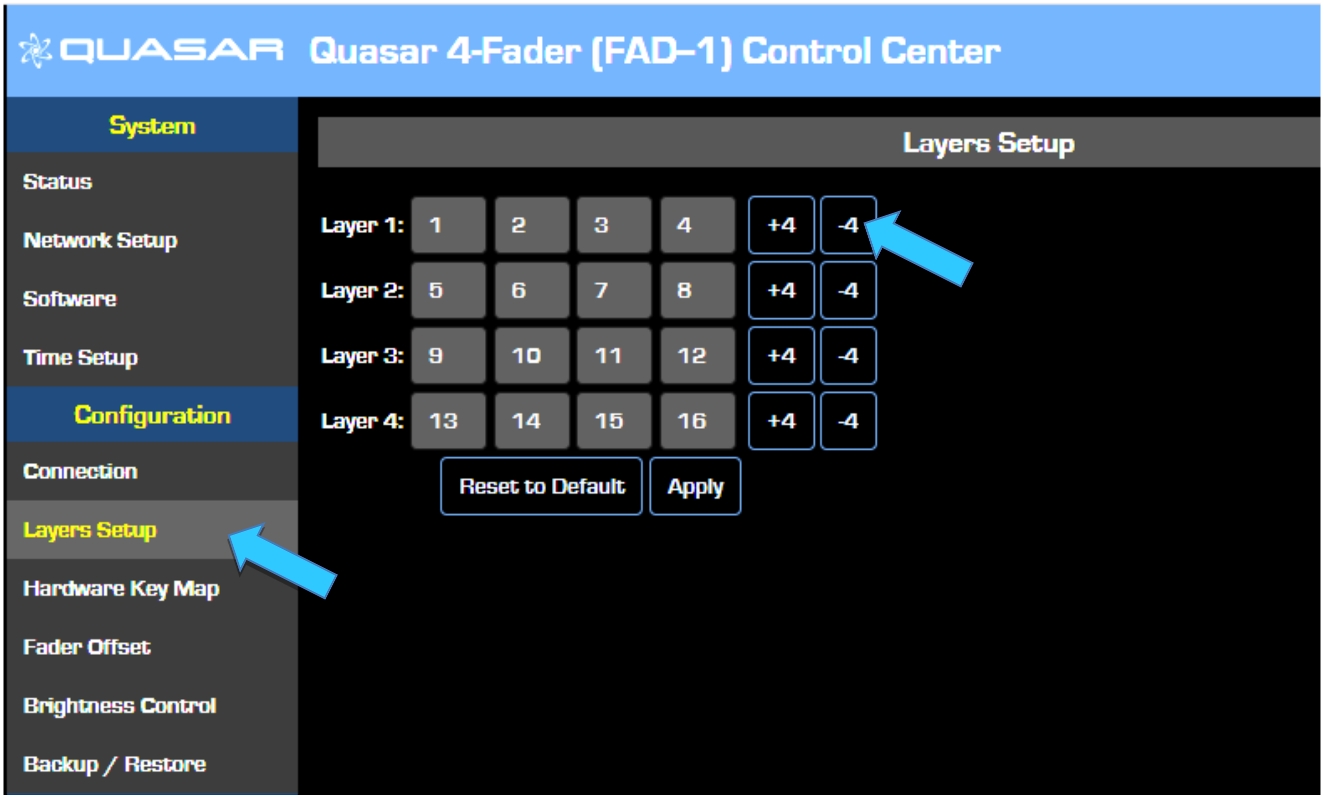
Don’t forget to assign the CR Monitor and CR Headphones streams to the Destinations (outputs) of the xNode you connected to your speakers and HP Amp. Or you will hear no Sound!



Surface Layer Configuration

Layers are useful when you need to access a number of DSP input channels on your engine which is larger than the number of faders (or phisical channel strips) available on your Quasar surface.
Since any fader on any module could access any input channel on the Quasar Engine, you will need to assign four input channel to each of the 4 layers available, on every Fader Module.

1. Using a PC connected to your studio network, **launch** a web browser and **enter** the IP address assigned to the first (leftmost) fader module into the browser’s address bar. When prompted for authentication **enter** user name “user”, and leave the password field blank.
2. In the left column, in the Module Manager section of the menu, **select** *Layer Setup*
3. Enter the number of the Engine input channel you want to assign to each physical fader, for each of the four Layers.

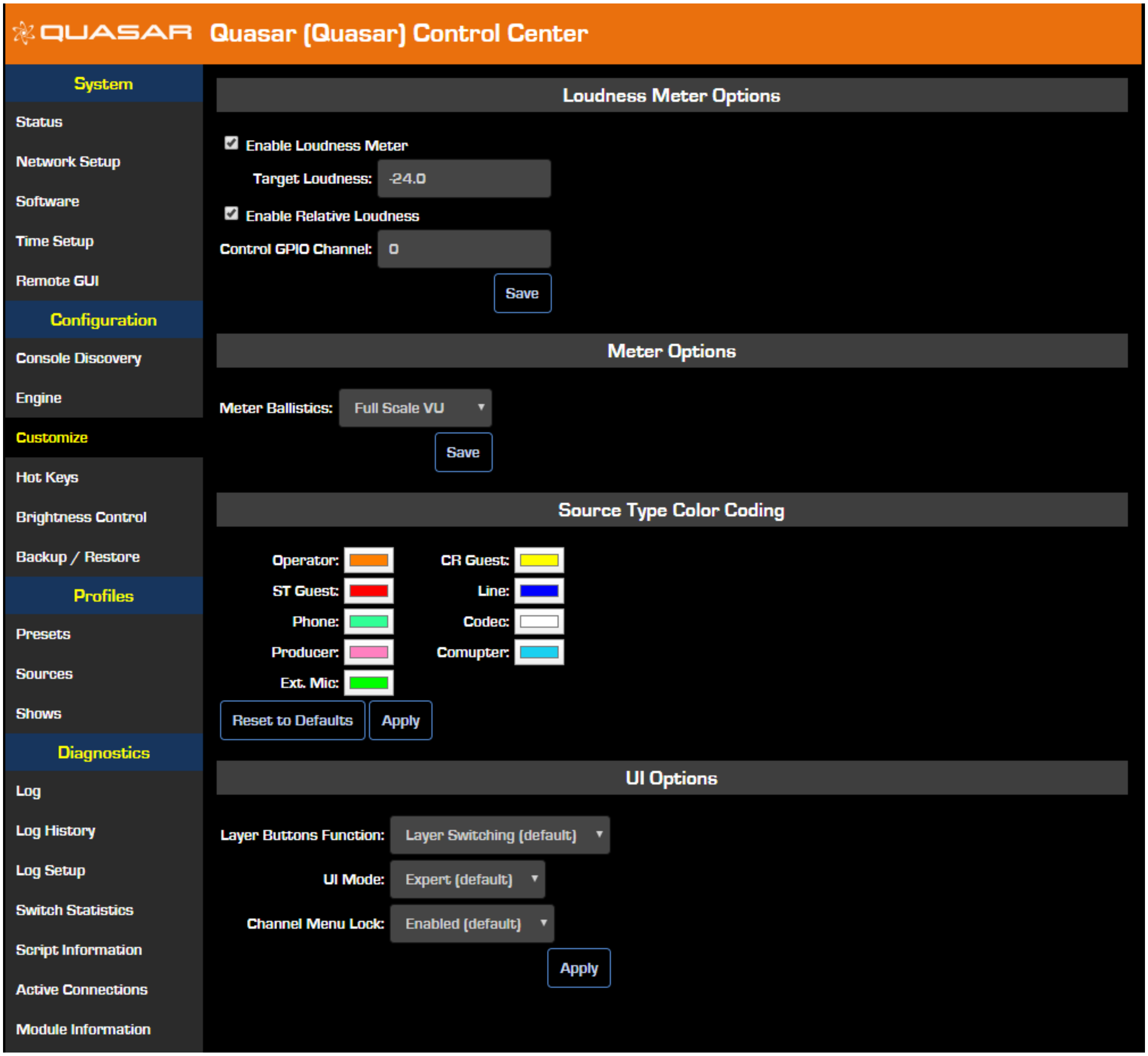


The table shown in the picture above shows which of the 64 input channels available in the Quasar Engine, will be loaded on the four channel strips each time the LAYER 1, 2, 3, 4 buttons will be pushed on the Master Module.

All fader modules default to the channel assignement displayed above. In order to quickly increment or decrement channels, in banks of four, just click on the “+4” or “-4” buttons

<u>In case you don’t need to use Layers</u>, just make sure that they are disabled.

You can disable Layers by navigating the Master Module Web UI and selecting the *Customize* menu. This will switch off the Layer buttons on the Master module.

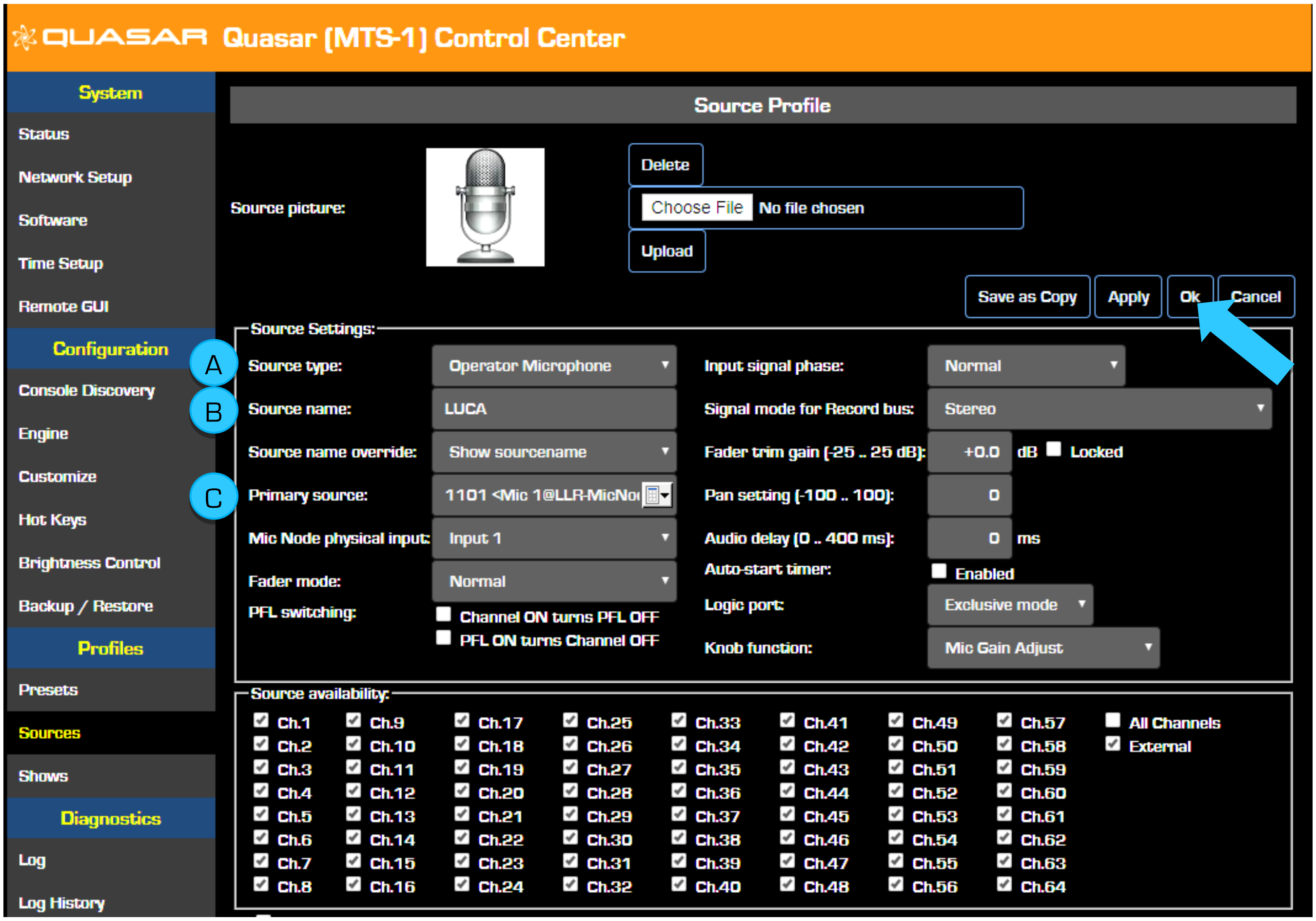


Input Source creation and configuration

You'll use your Web browser to create your first audio source in just a few fast steps:

1. Browse the Quasar MTS module Web UI. In the left column under *Profiles* **select** *Sources*
2. **Click** the button labeled "Create New Source Profile"
3. On the Source Profile screen, **select** *Source Type* from the drop down list (A).
4. **Enter** the name of the source in the *Source Name* field (B).
5. **Click** the browse button to the right of the *Primary Source* field (C) and **select** the desired source from the list.

6. **Click** *Ok*.
7. **Repeat** this operation to create more sources. These will then appear in the Channel Input > Sources menu of your Quasar.



<u>For more detailed information on how to create and configure all your Network sources, please refer to chapter….</u>

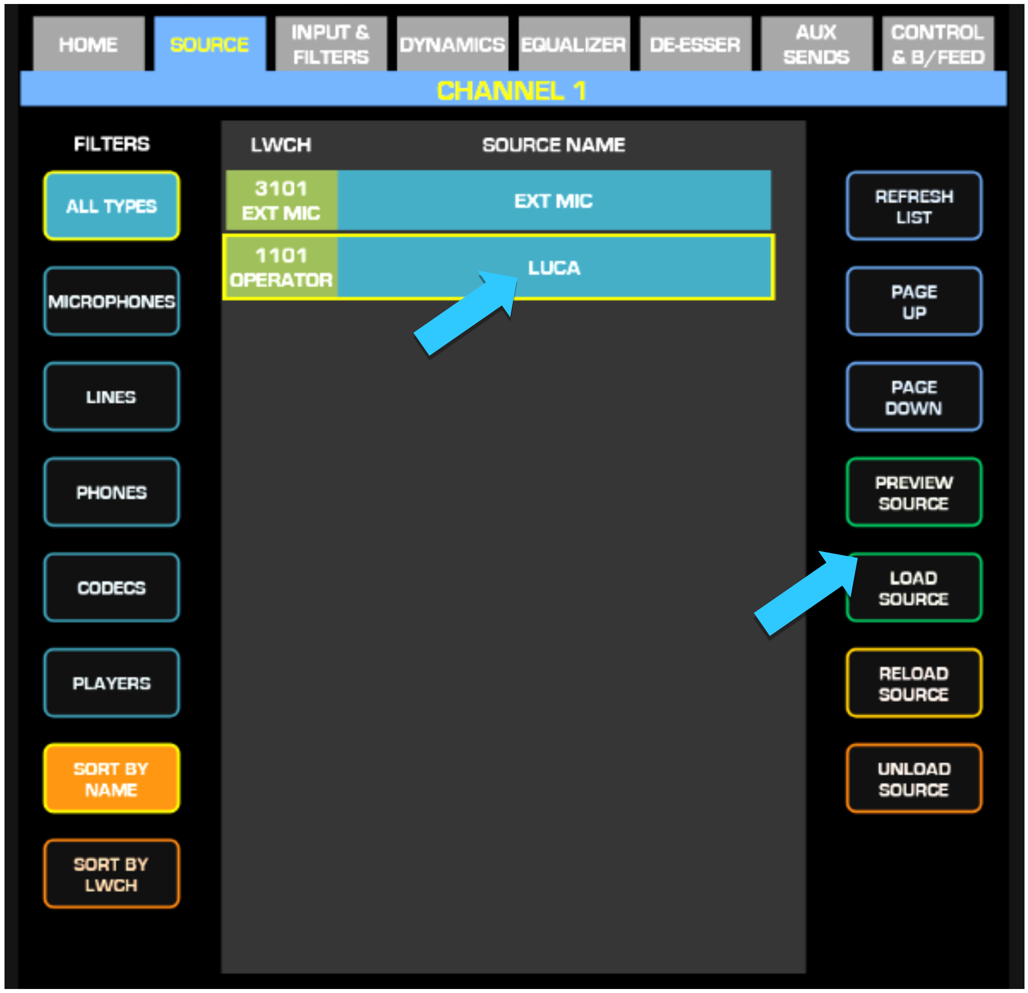
Assign Sources to Input Channels

Once all the required Sources are created, you need to assign them to the input channels of your Quasar Engine, and save this configuration in what we call “Show Profile”.
You can do so in two ways:

1. Assigning Sources directly **from the console**, and capturing your configuration into a Show
2. Creating a Show Profile **from the Web UI** and assigning sources to your channels from within this page

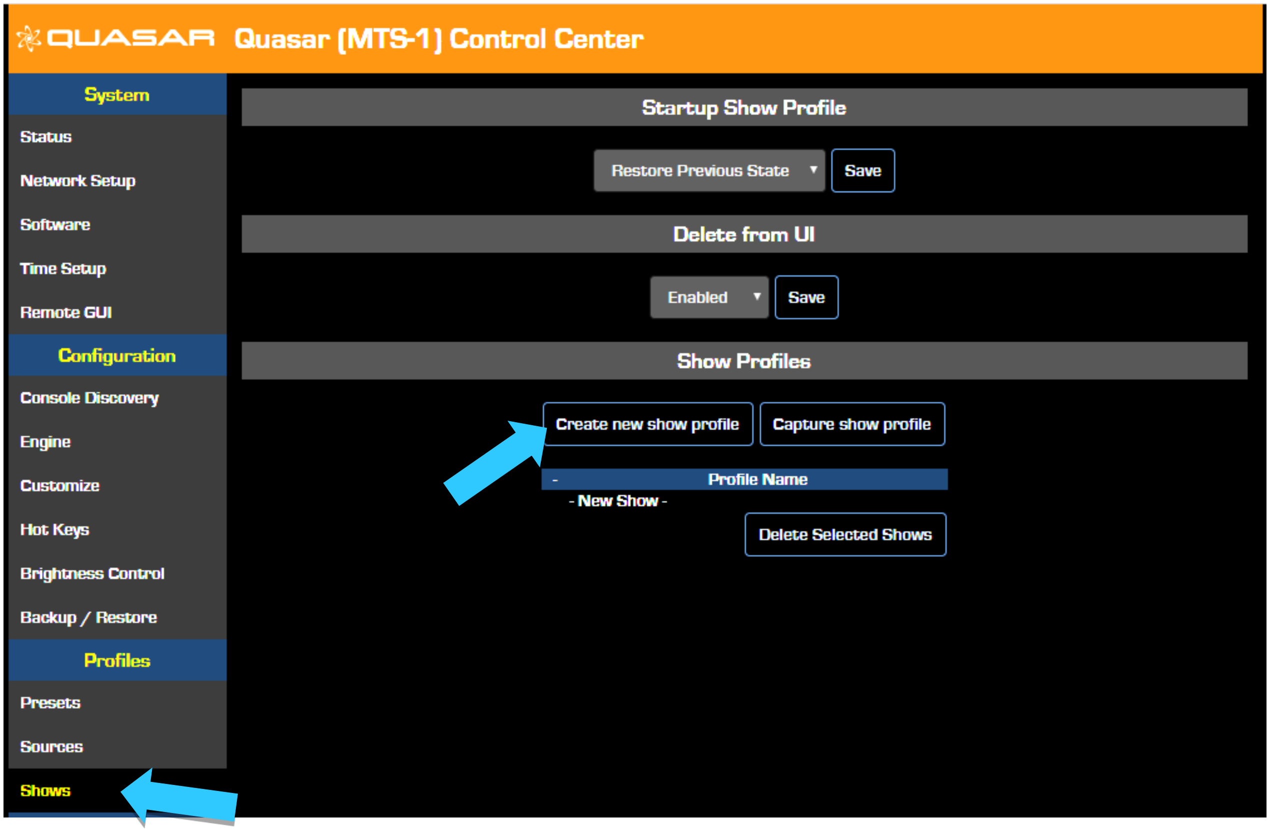
From The Console:
Push the top encoder of the channel strip you want to load with a source, and select the *Source* tab in the Master Touchscreen module.

Here you will be presented with a list of all the sources that have been configured on your console, and are active onto the network, and you will be able to select one by scrolling the touchscreen and pushing the *Load Source* button on the right.

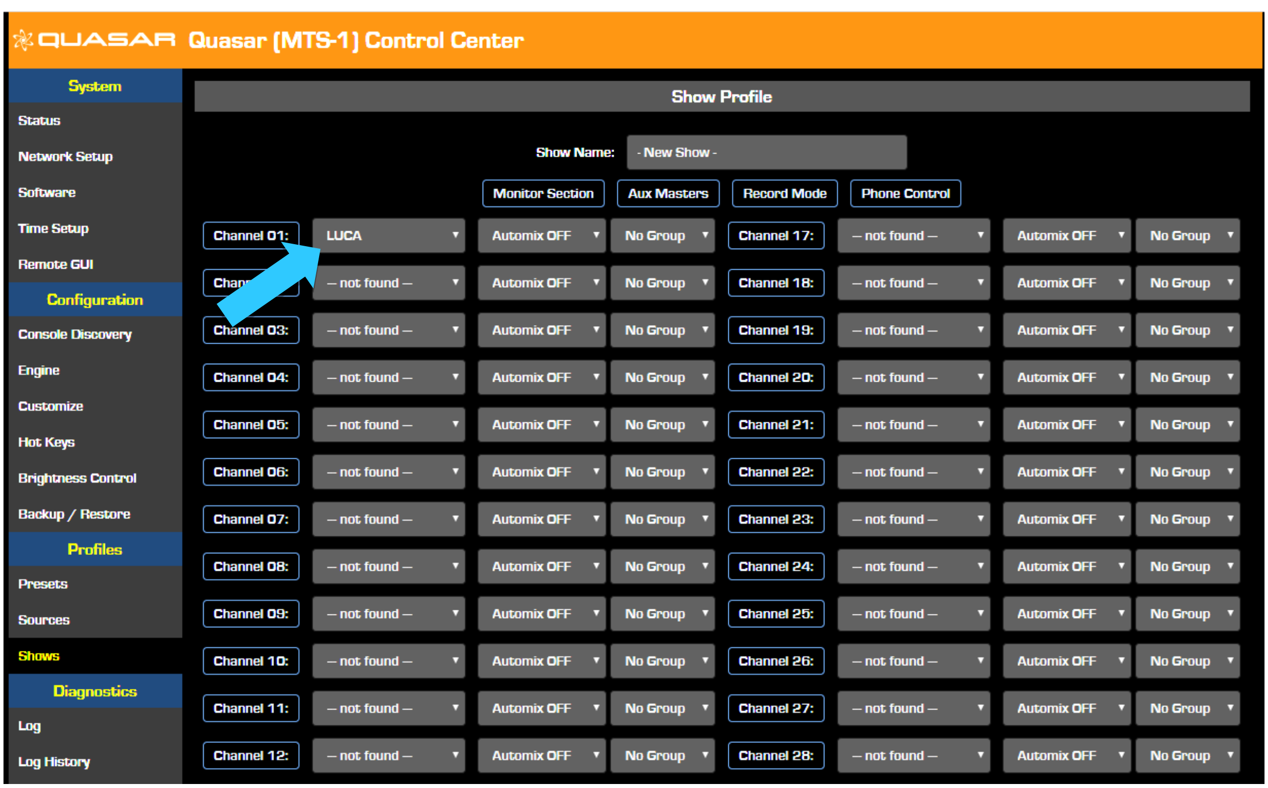


Note: inactive sources – those generated by devices which are disconnected from the network, or switched off – will not be detected by the Quasar and therefore will not show up on the above list.

From the Web UI:
From your PC, navigate the “Shows” main page, and create a New Show Profile



After clicking on “Create new show profile” button, a dialog will appear, to let you type in the new profile name and confirm. Then the following page will appear, and here you will be able to assign sources to each channel.



Program assignment & monitoring

Press the Program 1 key on the fader strip; it will illuminate to show you’ve assigned that fader to PGM-1. Press the ON key at the bottom of the fader strip and move the fader up.

Congratulations: you’ve got audio on your main Program bus!
The meters on your display should be active, as shown below.

To hear the audio, make sure you have selected Program 1 as source to your CR Headphones or Speakers, using the controls on your Monitor module, and that the volume is at an appropriate level (the level meters onscreen will show the relative volume you’ve set). Also, make sure the xNode feeding your speaker has been configured to the CR Monitor source channel number.

Your setup is complete!

That’s it! You’re ready to Rock with Quasar!!!

Now, sit back and enjoy some music. Try exploring the intuitive UI of the Quasar channel strip…. And have fun!

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Chapter 1
Quasar Components

The Quasar is more than just a mixing console — it is a complete studio system, with various components that serve different functions. Its main components are:

- The Quasar modular Control Surface
- The Quasar Engine DSP platform

This chapter gives an overview of the different components to help familiarize you. The following chapters will build on this familiarity as we dive deeper into Quasar's features and capabilities.

Quasar Control Surface

The Quasar Control Surface is a highly modular desk, consisting generally of a Frame populated with modules. Its main components are:

- The Quasar surface Frame, which includes:
    - one Rear I/O module,
    - a variable number of Rear PSU modules
    - any Rear Blank Plates

- one Master TouchScreen & Monitor module
- a variable number of Fader Modules
- any optional modules like:
    - the MIC in / HP out Module
    - the single channel blank strips
    - the 4-Channel Blank Module

The Quasar Control Surface offers a reduced footprint and a completely flat outline, since the traditional meter bridge has been eliminated and the displays integrated directly into the surface.

Quasar does not require an external display to operate; however, a display could be connected to show a duplicate of the Touchscreen UI, in landscape Full HD format.

Quasar surface Frame

Quasar Frames are available in both Table-Top and Flush-Mount versions and can be converted from one type to the other. The Quasar surface frame comes available in 7 different lenghts:

2.5 Units - 3.5 Units - 4.5 Units - 5.5 Units - 6.5 Units - 7.5 Units - 8.5 Units.

Each "Unit" corresponds to the width of 4 fader strips. One Fader module corresponds to 1 Unit, and one Master module corresponds to 1.5 Unit. So for example, an 8.5U frame will accommodate upt to 7 Fader modules (28 faders) plus the Master.

The Master TouchScreen & Monitor module (MTS-MON module)

The Quasar Master TouchScreen and Monitor module (we will refer to it as "MTS module" in this manual) is the brain and heart of your Quasar System. It is the host of the studio/console logic, and provides the control and feedback of various studio functions through the touchscreen and hardware. Some hardware buttons/encoders are soft controlled (their function changes based what you are doing) and some have capacitive touch functionality. One MTS-MON module is always required to operate a console.

It features a 12.1" TFT IPS Touschreen display, 7x Touch-Sensitive high-resolution optical encoders, up to 4x Banks of 8 Touch-Sensitive RGB User Buttons, 4x Layer buttons, 7x High Resolution Touch-Sensitive optical encoders.

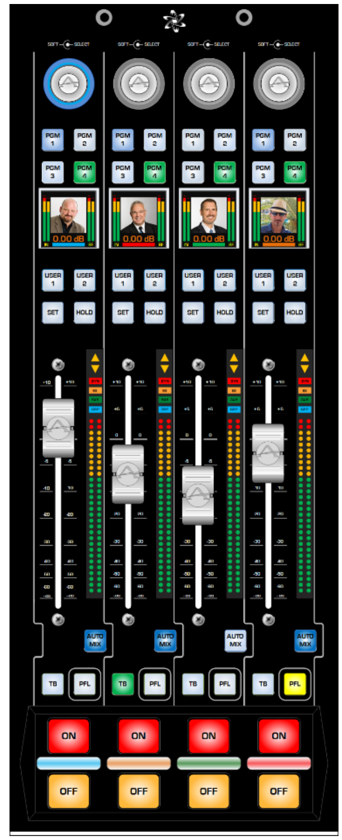
Internally, the MTS module has dual Gigabit Ethernet, HDMI, and USB connectivity, and dual 12VDC PSU inputs. These provide connectivity to the rear I/O module, and the fader modules. Depending on the type of fader module, a USB or Ethernet control loop will be used.

The HDMI connection permits the option to connect an external PC display which will mirror the display of the master module, in a landscape format. This is useful for emergency (in case the touchscreen gets damaged) …or just to show off a little bit with your Quasar!

NOTE: The USB ports on the front panel of the MTS module and at the back of the console are for future use.

The XR-4FAD module (extended access, 4-Fader module)

The XR-4FAD is the "full optional" fader module in the Quasar range.



This module provides all the standard controls needed for studio operation along with user definable buttons per fader.
Each hw button is RGB (Red, Green Blue) , and is programmable by the user via the Module's Web UI.

Both the top encoders and the motorized fader knobs are touch sensitive, and can interact with the operator.
Each fader-channel has LED meters for source level indication or Automix gain adjust information. Above each meter are LED status indicators for the channel.

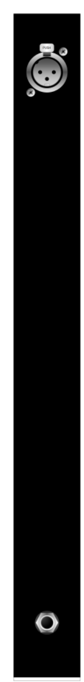
Each fader channel also includes a color display for channel/source information. Between the two large buttons at the bottom of the module is an RGB color bar which can be used for fader identification.

The Master module with the fader modules provide the functionality expected in a studio. With this comes some standard functions that automatically happen based on what is known (or assumed) by the console, such as:

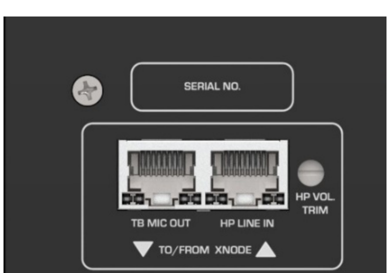
- Muting
- ON/OFF commands
- GPIO control
- Mix conditions
- Source ownership (which console has control of an active source)
- Phone control

…and many others.

Internally, the XR Fader module has dual Gigabit Ethernet , USB connectivity, and dual 12VDC PSU inputs. These provide connectivity to the rear I/O module, and the MTS module. Depending on the type of fader module, a USB or Ethernet control loop will be used.



The MIC in / TB out Module



The MIC in / TB out module is an optional accessory designed to provide connectivity for one Microphone (input) and one Headphone (output) right at the surface, so that a gooseneck microphone can be used for Talkback, for example. In case the surface is ordered with this module, the Rear I/O module will come equipped with the additional HP Amplifier board, which has audio I/O connections to/from one of our xNodes.

Note: Connectors A and B are not Ethernet sockets… They are Analog audio I/Os!

This module simply extends the audio I/O of the node up to the surface, adding powerful and high-quality amplification for your headphones. Connect the TB MIC OUT (A) socket to the input of your xNode , and the HP LINE IN (B) socket to one of the outputs of your xNode using standard RJ-45 cables. Use the HP Volume Trimpot (C) to adjust the internal Amplifier's gain up to the desired level.

Accessory Modules

Future Modules

The Quasar surface is primarily made up of the master module and the fader modules. Being a modular design and networked, we have only just begun the fun.

Future module will come out which can add function and options for our customers More optional module will be released in the future.

Quasar Engine

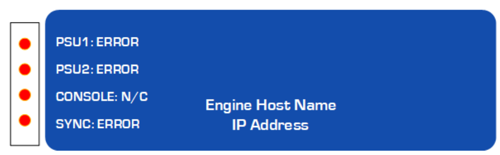
The Quasar Engine serves as the DSP platform of the studio. Audio from the AoIP network enters the mixing engine and is manipulated per the user's actions on the Quasar surface. Processed audio becomes streams (sources) from the Engine into the network.

The front-panel of the QuasarEngine has a display and navigation buttons. The display normally provides basic system information, along with an “OK” status indicator. If the “OK”indicator is missing from the screen, it’s because something needs attention — in its place will be an error message describing any items the unit’s self-diagnostics consider out-of-the-ordinary.

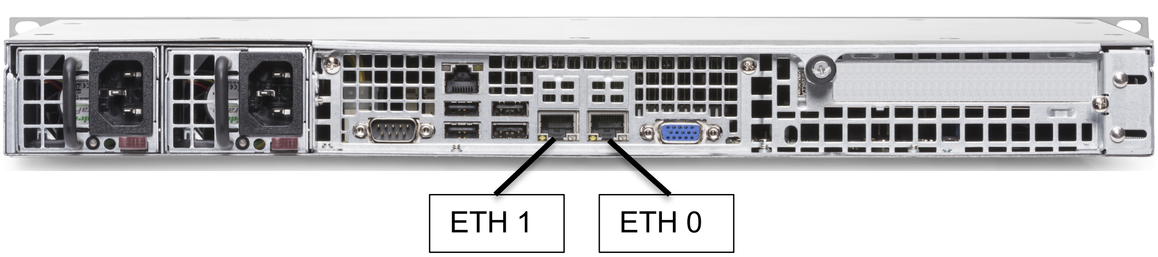
Four LED indicators located to the left of the display can turn green or red to provide quick visual feedback about the status of the following parameters:

- PSU1 status
- PSU2 status
- Connection to the surface
- Audio Sync

The display also provides access to the Quasar Engine’s menu options. From the OK status view, tapping the buttons will navigate through the menu system.



The **OK** indicator shows that the DSP engine is functioning and is receiving clock from the network. If it is not lit, either there is no network sync available to the DSP engine, or the main CPU has failed.
The rear of the QuasarEngine has two power supplies, each with standard IEC connectors. Physically there are two network interfaces: ETH 0 is the port that you want to use, when connecting to your core switch. The second interface is for future functionality.



What’s Next

Follow on as we dive into deeper configuration territory in the following chapters.

Chapter 2

Configuration Basics

Working With Profiles

A standard console configuration step is defining the sources that will be used on the console.

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 sources configured, we can go one additional step and define channel layout and monitor settings.

The Axia terminology for these source settings is “Profiles”. There are two types of profiles:

1. **Source Profiles**, which define the different audio sources and how they function within the studio system, and

2. **Show Profiles**, which define what sources are placed on which console input channels, and how the console’s Monitor section is configured.

This chapter reviews some specifics of Profiles and in turn is the basic knowledge needed to configure the Quasar surface. A description on each specific option is available in the Appendixes.

Source Profiles

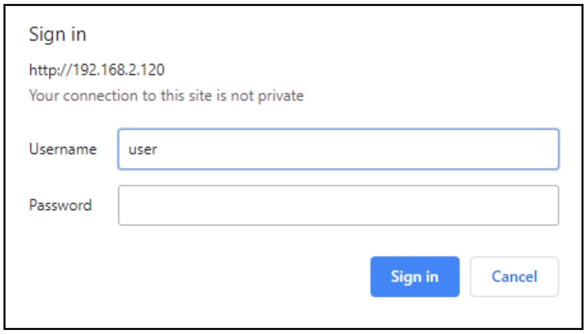
The fundamental difference between an Axia console (not just Quasar, but all models!) and its competitors, is that we created a set of logic attributes that can be associated to any audio source, which describes how that source behaves when it is loaded onto the console. This “logic layer” can be manipulated 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.

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:

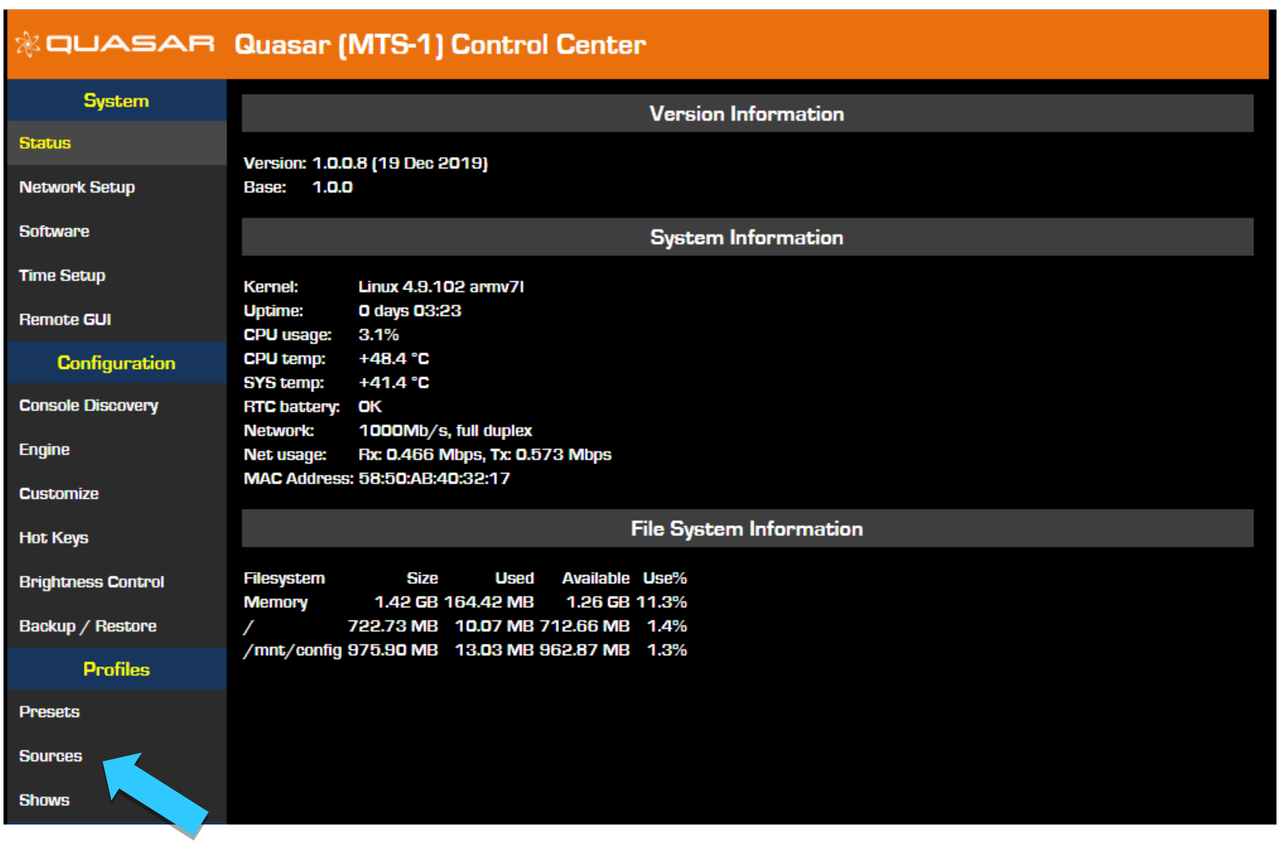
- **Operator Microphone** (Operator source type) – the Board Op’s mic
- **Guest Microphone** (CR Guest source type) – an additional microphone in the same studio
- **CD Player** (Line source type) – any basic audio source
- **Caller** (Phone source type) – a source that would require a mix-minus return
- **Codec** with specialized return feed needs (Codec source type with custom backfeed)

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.

Having said that, connect a computer to the Quasar and access its configuration Web UI, by entering the MTS Module IP address into your Web browser, and follow along. 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.
To get started, select the **Sources** link, as shown here:

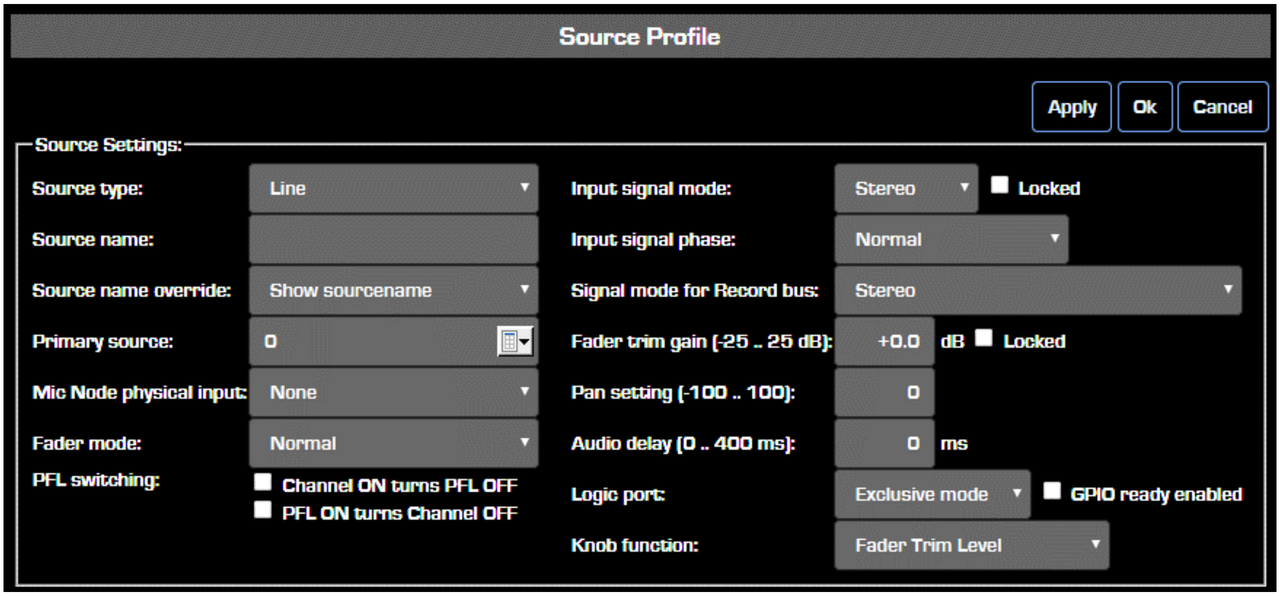


Once you've done this, you'll see any sources that have already been configured, as well as to "Create new source profile" by clicking on the button at the top of the page.

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**. You'll see, in part, the following :



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.

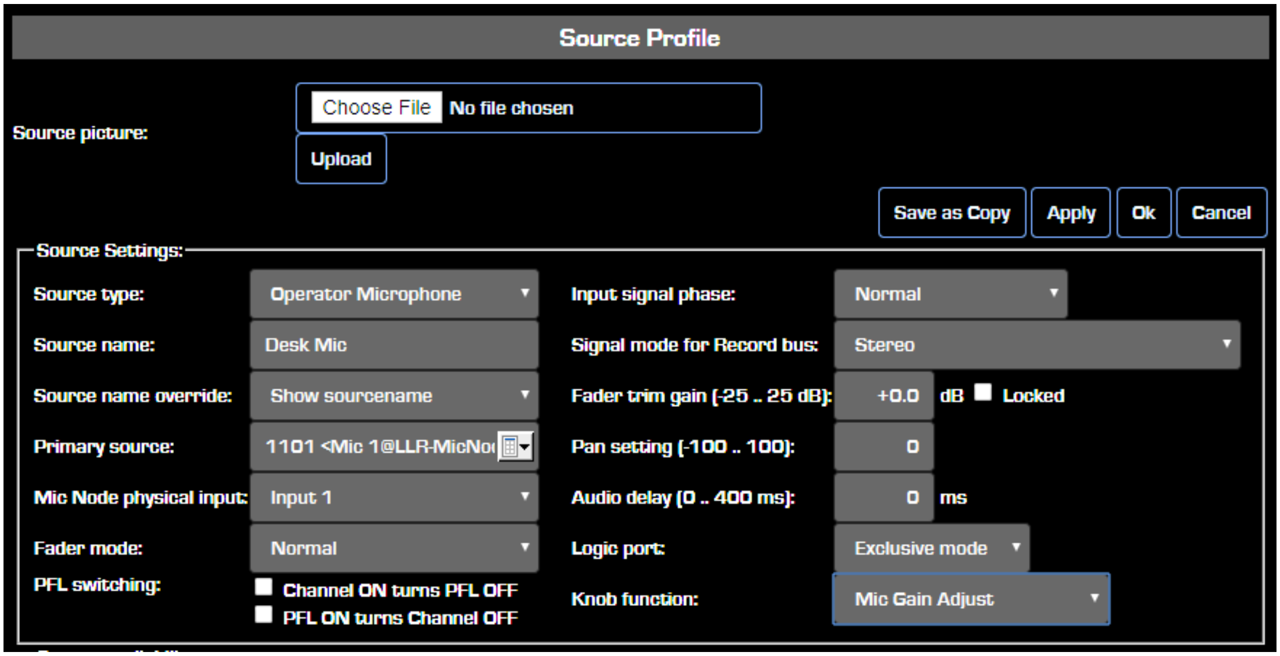
In the **Primary source** field, enter the board mic's Channel Number. (Each audio source in the network has its own unique Channel ID number.) If you know the number, just type it in; if not, use the Browse button to the right of the field to select the source from your network.

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.

In case you want to be able to control the Microphone Gain and Phantom Power directly from the console (these are settings normally belonging to the xNodes' Web UI) you can do so by entering the number of the **Physical Input** your microphone is connected to.

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.



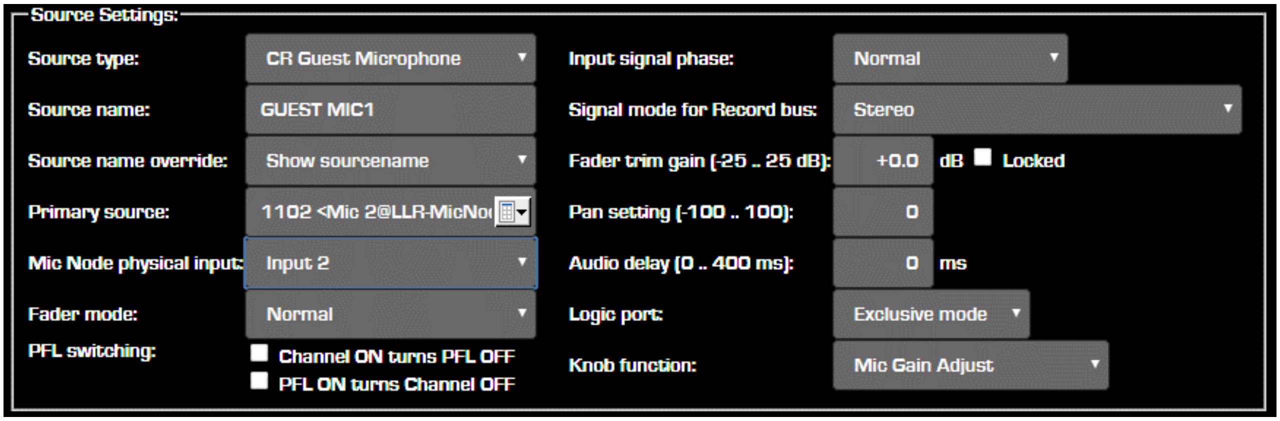
In the Source Availability box, you have the option to control what faders and Monitor channels the source may be assigned to. Uncheck the **External** — it's not very likely you'd assign the Operator's mic directly to the monitors!

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.

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 Node physical input** which allows for input gain and Phantom Power control directly from the surface.
You might also want to select “**PreAmp Gain Adjust**” 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: Knob Function

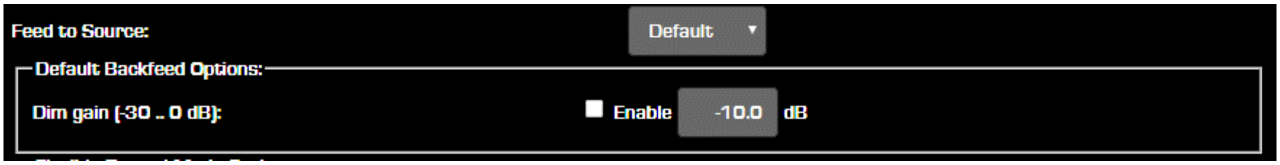
Pressing the **Channel Options** encoder at the top of any fader strip opens the Channel Option screen on Quasar master module.

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** dropdown box.

- If **Fader Trim Level** is selected, the control does not affect the Input gain of the source, but simply adjusts the range of the fader itself. This is useful when one of a group of similar audio sources is higher or lower than its siblings, and the operator wants to maintain a similar physical fader position.
- If **Mic Gain Adjust** is selected , the board op will be able to use the knob to quickly boost or cut the level of the Analog Mic Preamp stage in the xNode to compensate for audio that’s too “hot” or too low.
- If **Line Input Gain Adjust** is selected , the board op will be able to use the knob to quickly boost or cut the level of the source at its Line Input stage to compensate for audio that’s too “hot” or too low.
- **Automix Weight Control** is for adjusting the automix weight priority for the selected channel strip. The Automix can then be Enabled or Disabled using the dedicated button right below the fader

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 Talkback key on a fader strip, the board-op's mic is routed, pre-fader, to the "backfeed" of that channel — an IFB function, if you will. 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, Preview, or Cue) mix.

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).

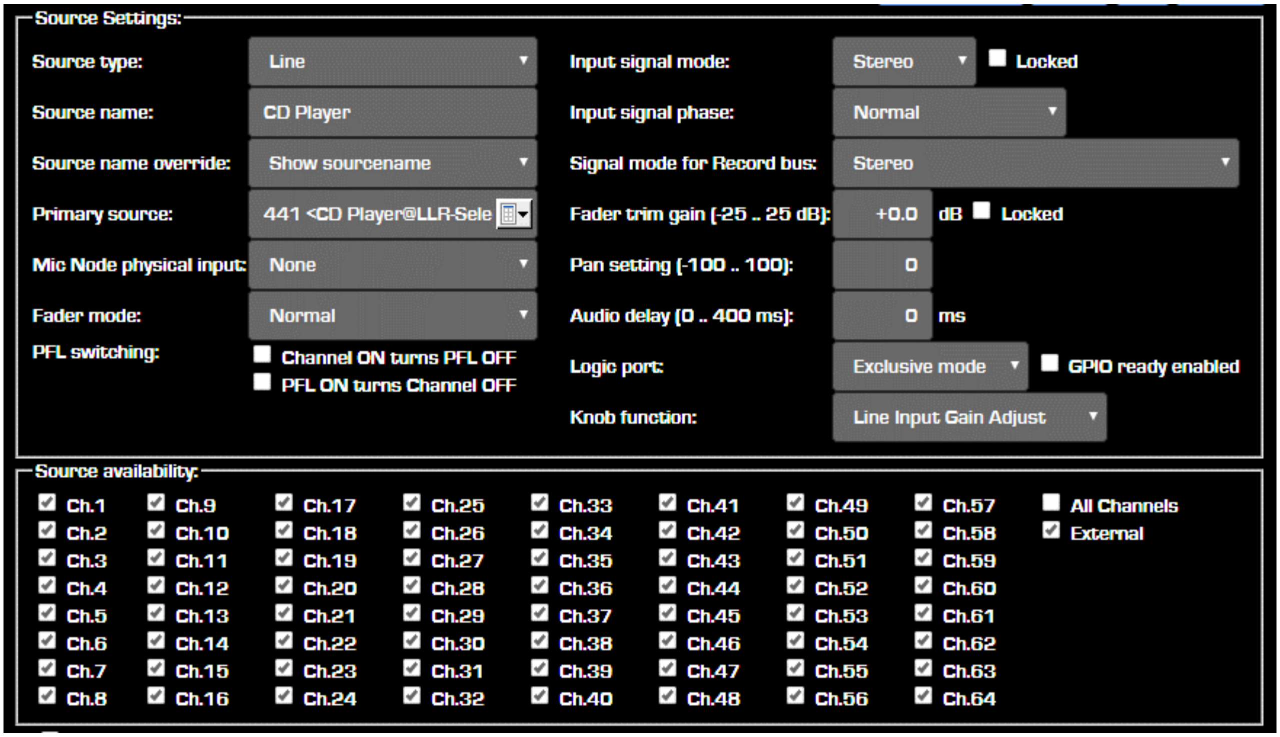
Line Source is perfect for creating Source Profiles for devices such as CD players, DAT decks, Satellite receivers, PCs, etc.

Setting up this Source Profile is just as easy as the previous two profiles.

From the **Source Type** dropdown, select "Line". Enter a name for the device, such as "CD Player" or “CD 1”.

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.



Note the example we show 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* twice which will unselect all 64 channels with the second click.

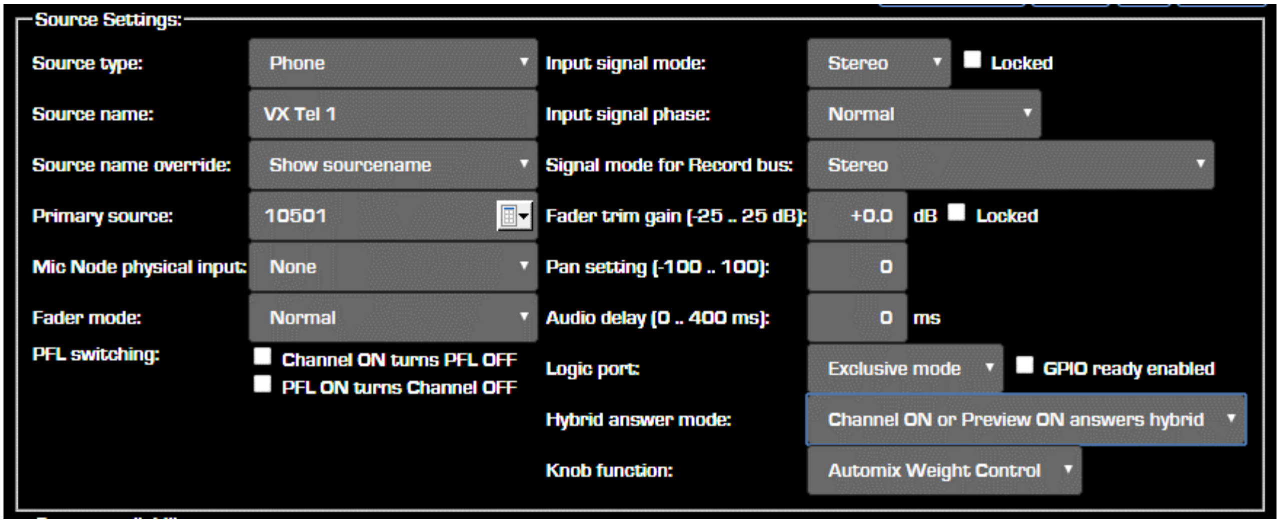
Another check box that’s useful for Line source types is the one marked **GPIO ready enabled**. This controls the OFF lamp on the fader strip that the source is assigned to. Some operation practices require an indication of source readiness; when this box is checked, the fader’s OFF lamp will only illuminate 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** dropdown, and some phone-specific options will appear.

In the **Source Name** field, type the name you’d like to be displayed on the console surface.

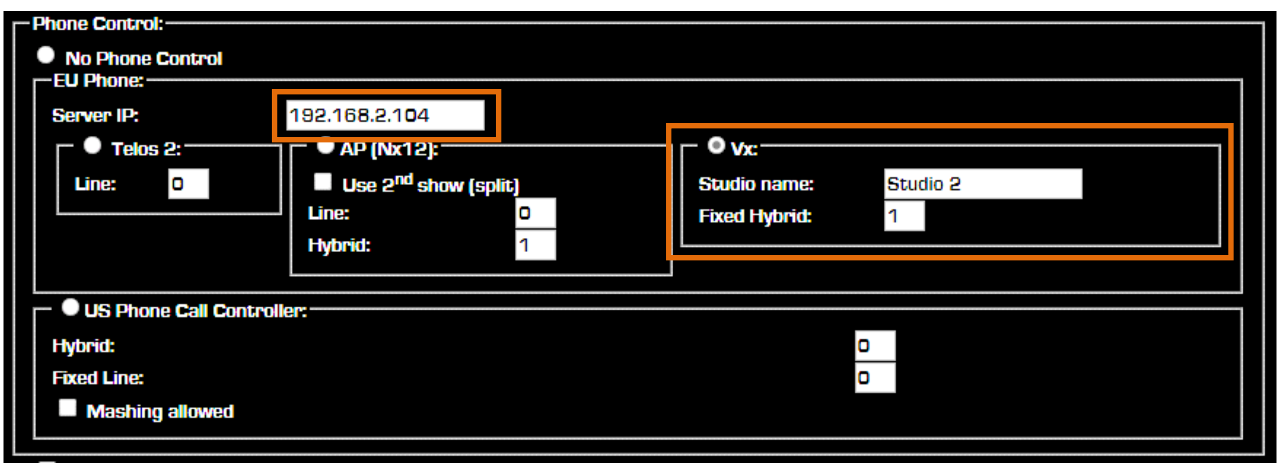
Now, to define the Phone controls. Select from one of the following options that best suits your studio’s phone gear:

- **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 Quasar’s GPIO capabilities.

  To do so, scroll to the **Hybrid Answer Mode** dropdown box in the Source Settings box. Select either “Channel ON answer hybrid” or “Channel ON or Preview ON answers hybrid”.

  In the first case, when the Phone source is assigned to a console fader strip, turning that channel ON picks up the phone. In the second case, your operator may either turn the channel ON *or* use the channel’s PFL (preview, cue) 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.

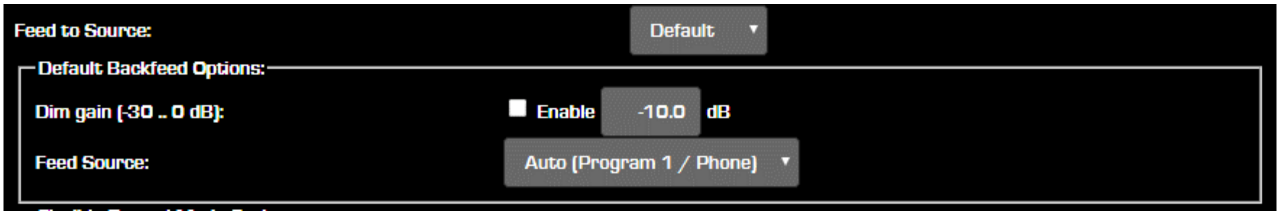


- In the example shown above, we're mapping a hybrid from a VX Broadcast VoIP system, using the sections highlighted in orange:
    - Type in the IP address of the VX Server into the **Server IP** box.
    - Select the **VX** radio button.
    - Enter the **Studio name** configured in your VX system into the **Studio Name** box.
    - Enter the number of the VX hybrid from the specified VX Studio into the **Fixed Hybrid box.**
    - Note: If the Server IP has authentication requirements, the authentication syntax in the Server IP field is *username:password@IP ADDRESS* . 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 soft interface in the master module. Enter the **Hybrid** number to tell the fader which hybrid the Source Profile controls.

- To access the Call Controller, move to the console and select the **CALL CONTROL** tab in the Master module's touchscreen. The screen will change to provide an 8 line controller.

The final step is to define how you want to handle mix-minus. Quasar (and all Axia consoles) automatically generates mix-minus (N-1, "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** dropdown.



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 engaging any PGM 4 button 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.

Advanced Stuff: A Codec Source With Custom Backfeed

If you've followed along and 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, of course). If the codec is bi-directional, 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 or named Backfeed.
However, when the Talkback key (TB) on the channel strip the codec is assigned to is pressed by the operator, the board op's mic audio (pre-fader) is routed to only the left channel; the right channel remains unchanged.

By now, you'll have noticed that, when you begin to create a Source Profile from scratch, all profile types have either "Disabled" or "Default" displayed in the **Feed to Source** option field.

But your operation might have needs which require a more specialized Backfeed than the Default behavior. And so we provide the “Custom” option, which allows high-level control of mix-minus behavior based on channel state logic. Here’s the section of the Source Profile which pertains to this Custom Feed to Source functionality:

As you can see in the next 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 make the automatic backfeed switch happen only when this condition is met.

NOTE: The grey outlined headres (like "While Channel is OFF", for example) always refer to the settings below the headr and not above. This is a standard convention adopted in all UI pages.

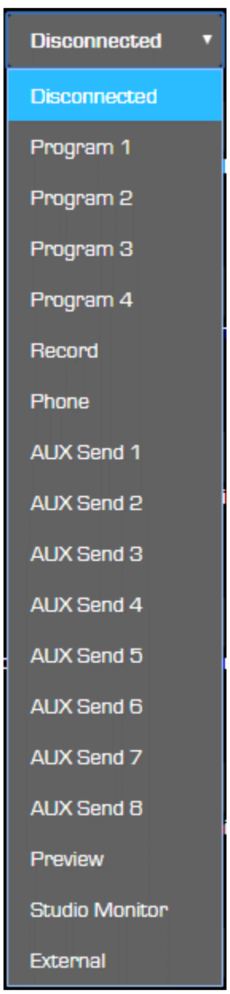
External Feed Src allows you to pick any audio channel in your Livewire network to send to your source's backfeed.

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 Standard Mode:
  - **While Channel is OFF**
  - **While Channel is OFF, But in Preview**
  - **While Channel is ON**
- In Record Mode:
  - **Record Mode: While Channel is OFF**
  - **Record Mode: While Channel is OFF, But in Preview**
  - **Record Mode: While Channel is ON**

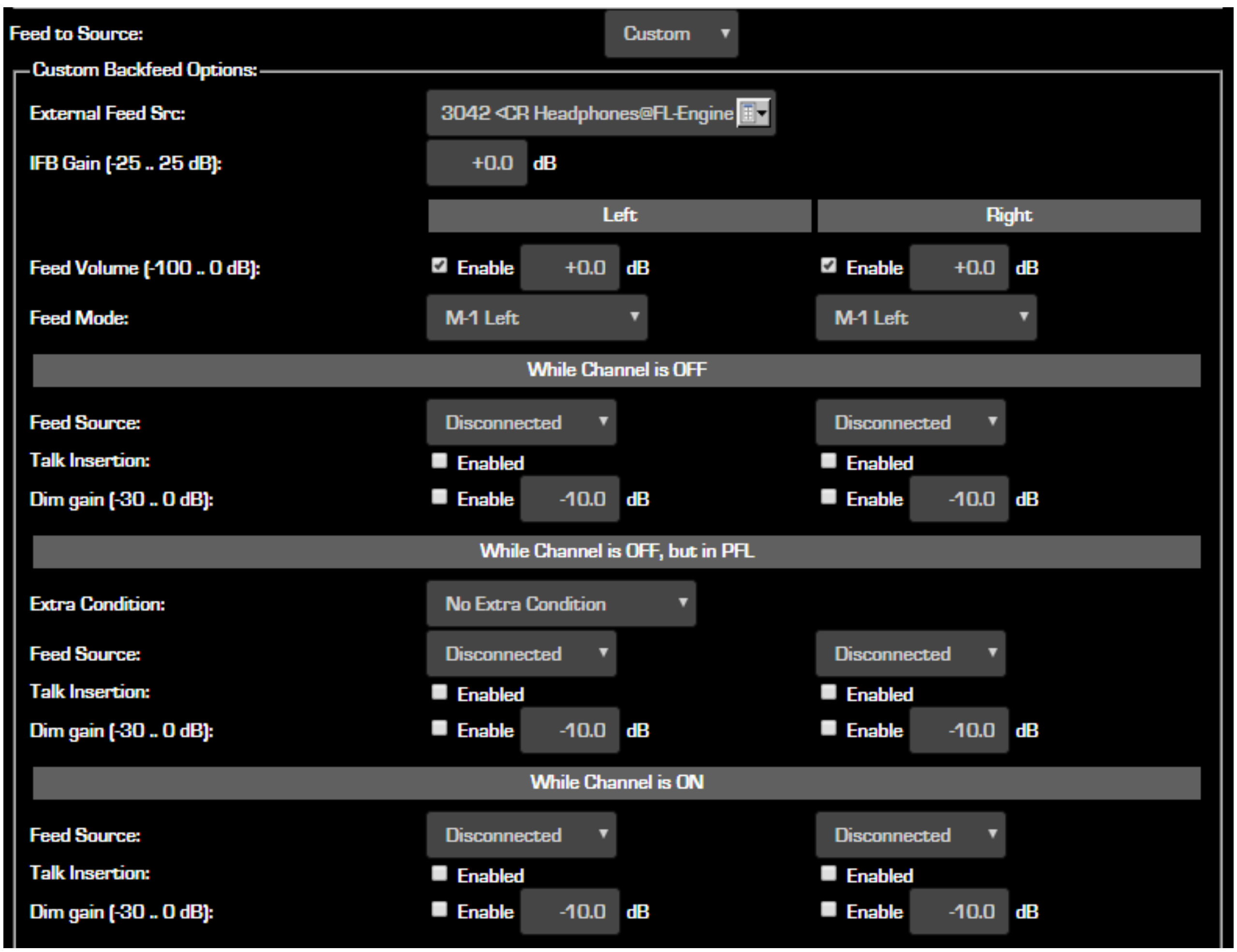
Using the **Feed Source** dropdown boxes, you can pick the Backfeed source that will be fed to the Codec's IFB channel in each of the six possible channel states noted above. Your choices are:

- **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 PGM4. The Record mix is postfader 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.
- **Preview.** Sends the audio from any sources assigned to the Preview (cue, PFL) mix.

- **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.** Box at the top of the Custom Backfeed Options box.

Note: 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. We’ll cover use of this function in later chapters.

So, say you want to construct 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 Preview, and a PGM-1 mix-minus when the channel is ON, you’d set it up like so:



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.

Show Profiles are, essentially, snapshots of an entire console. A Show Profile keeps track of :

- what sources are loaded to each of your console's channels
- what settings you have on your Monitor section
- what settings you have on your Auxiliary Master section
- what monitor settings you have in your Record Mode
- what control settings you have for your Telos Phone system

> NOTE: With the words "console channels" we always indicate the *Quasar Engine's DSP input channels,* and not the physical channel strips on your mixing surface.

Using Show Profiles, each user can have the board set just the way they like it — sources placed where they're most useful, monitors set to the appropriate feed, headphones conforming to personal preference.

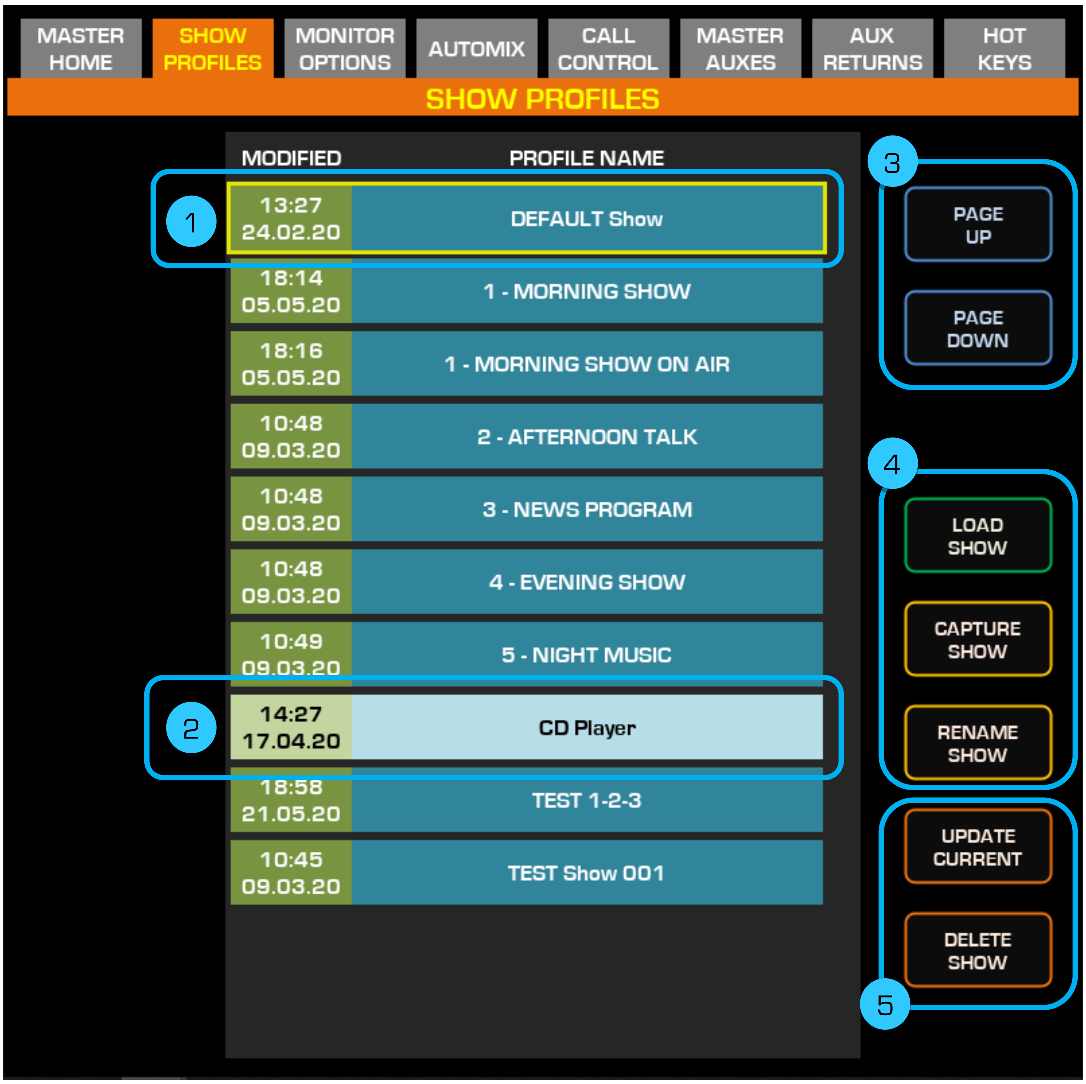
You can also use Show Profiles to define different types of broadcasts – one for the morning show, one for talk segments, one for musical guest interviews, one for unattended operation – that instantly recall your saved configuration when loaded.

Up to 9999 Show Profiles can be saved in a console.

How to Create A Show Profile

The easiest way to create a Show Profile is to set up your Quasar's console for a show, then save a Show Profile for it by taking a "snapshot." We call this process **Capture** in the console UI.

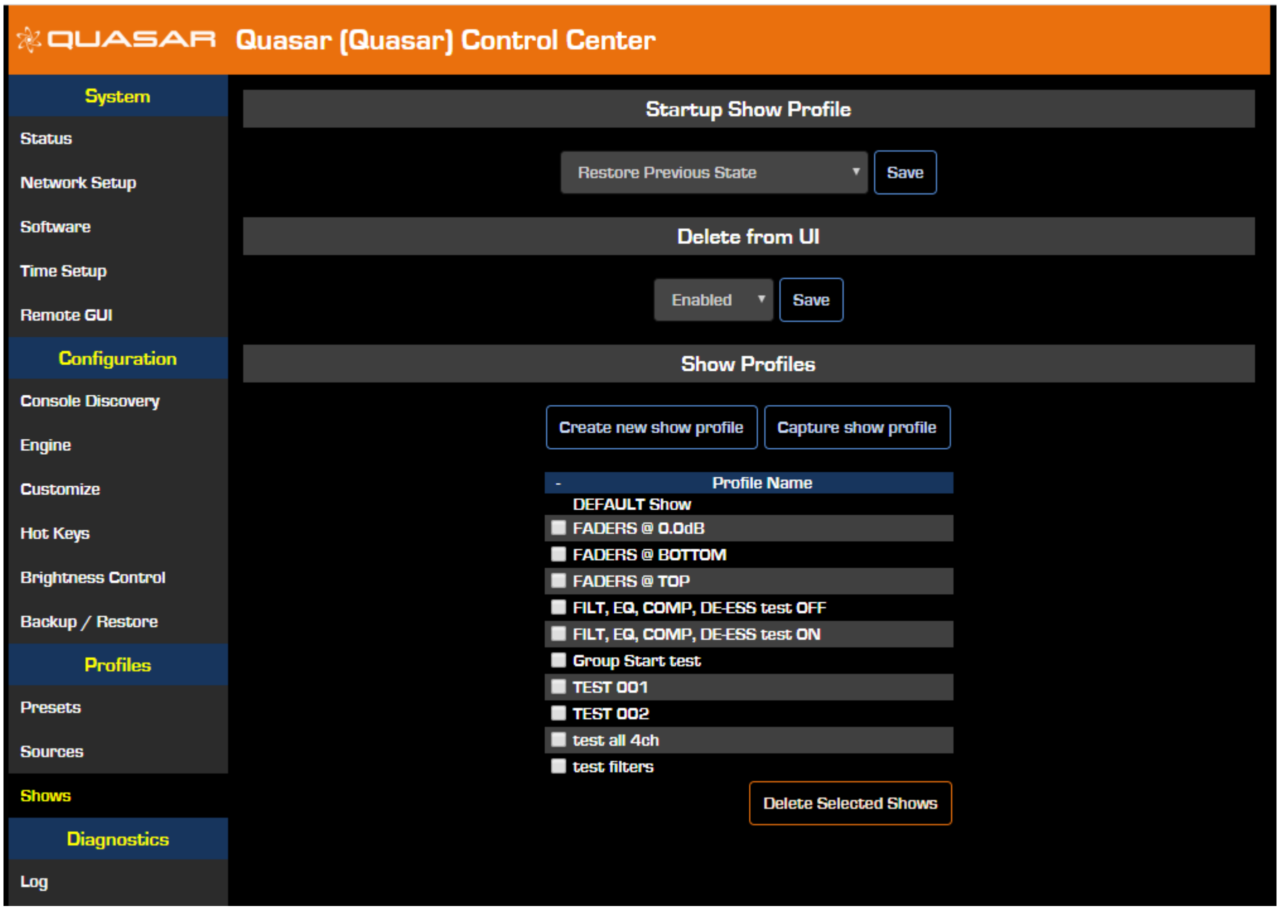
1. Selected Show : relates to LOAD SHOW, RENAME SHOW, DELETE SHOW buttons
2. Currently Loaded Show : relates to UPDATE CURRENT button
3. Paging Buttons: useful alternative to touch scroll, when lots of Show Profiles are listed
4. Non-Destructive buttons: used for every-day Show Profile management
5. Destructive buttons: used for expert Show Profile management. Can be disbled in the Web UI



- Get started by assigning a source to each fader strip using the Options knob at the top of each fader 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, hit the **Profile** button on your Master Module and press the **Capture Show** button. An on-screen keyboard will appear. Type a name for your new Profile, press OK , and the game is done!

Alternatively, you could connect a computer to your Quasar's IP address into your Web browser, and select the **Shows** link, as shown here:



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 **Profile** key at the console's Master Monitor module.

The other link shown here, **Create new show profile**, allows you to construct an entire Show Profile completely from scratch using your computer's on-screen controls. This is the "expert" way of making a show profile. We'll cover these options in detail later on.
You can save up to 99(?) different Show Profiles to make console reconfiguration fast and easy.

> Don't worry – 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. mixing surface.

Show Profiles Options

Show Profiles are snapshots that allow you to set up a console the way your talent wants it, then save and recall those settings with one button press. Up to 99 Show Profiles can be saved per console.

The easiest way to create a Show Profile is to set up your console the way you like it using the controls on the board itself – then use the **Capture Show Profile** link found on the **Shows** screen of your Fusion control center web page. This will take a snapshot of the entire console – fader assignments, source EQ, mic dynmics, Monitor
assignments, even headphone EQ settings – and save them to a new Show Profile that you can use as-is, or edit later.

There are also some options, such as Record Mode setup, that are *not* able to be set from the Fusion surface itself, and must be set up by editing the Show Profile itself. In the next few sections, we'll give you a line-item reference listing of the options found on each Show Profile screen.

Channel Screen Options

There's a Channel Screen for each of your installed faders. Following is a listing of options presented on each screen.

NOTE: choosing The "Retain Source Setting" option for any item leaves that item unchanged when loading the Show Profile.

General Controls Section

Source ID

- Dropdown box selects fader's assigned source from saved Source

Feed To Source Mode

If the loaded source has a mix-minus, chooses the source for mix-minus creation.

- **Auto** lets the console choose a mix-minus source automatically.
- **Phone, PROGRAM** and **AUX SEND** options let you manually choose the bus to use as the basis of the mix-minus.

Auto-Start Timer

- **Enable:** Console interval timer starts when fader ON key is pressed.
- **Disable:** Timer is unaffected by fader ON key.

Signal Mode

- **Stereo:** sets 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.

Signal Mode Locked

- **Unlocked**: allows board op to change the Signal Mode from the surface.
- **Locked**: prevents board op from making changes.

Fader Trim Gain

- Select the Use: radio button to specify a fader trim cut or boost between -25 and +25 dB.

Fader Trim Lock

- **Unlocked:** allows board op to trim the fader using Channel controls on the console.
- **Locked**: prevents board op from making changes.

Panorama Position

- Select the **Use:** radio button to Pan the input signal left or right of center.

Phase

- **Normal:** No phase change to input signal.
- **Invert Left:** Reverses the phase of the left input channel only.
- **Invert Right:** Reverses the phase of the right input channel only.
- **Invert Left and Right:** Reverses the phase of both input channels.

EQ Active

- **Bypass:** Loads the input source with no EQ adjustment.
- **Active:** Loads the source with EQ active, and applies EQ settings specified in following sections.

EQ High Mode

- **Shelf:** Selects high-shelf style of EQ application.
- **Peak:** EQ is applied to a selected frequency, "notch filter" style.

EQ High/Mid/Low Frequency

- Select the **Use:** radio button to set the center frequency of the selected band.

EQ High/Mid/Low Gain

- Select the **Use:** radio button to specify an EQ cut or boost between -25 and +15 dB to the selected frequency.

Assign To…

- **On:** Assigns fader strip to the specified Program or Aux Send bus upon Show Profile load.

- **Off:** Removes fader strip from the specified Program or Aux Send bus upon Show Profile load.

NOTE: Channels may be simultaneously assigned to any combination of Program 1 – 4 and Aux Send 1 – 4 mixing buses.

Aux Send A-B-C-D Pre/Post Fader

- **Pre-Fader:** Sends the input source to the selected Aux Send bus *before* fader gain adjustment.

- **Post-Fader:** Sends the input source to the selected Aux Send bus *after* fader gain adjustment.

Aux Send A-B-C-D Pre/Post ON

- **Pre-ON:** Sends the input source to the selected Aux Send bus *before* the channel's ON/OFF switch.

- **Post-ON:** Sends the input source to the selected Aux Send bus *after* the channel's ON/OFF switch.

Aux Send A-B-C-D Gain

- Select the **Use:** radio button to specify any additional gain or cut to be applied to the assigned source before sending to the selected Aux Send bus.

Noise Gate Status

- **Bypass:** Turns off the noise gate feeding the Vocal Compressor section.

- **Active:** Turns on the Compressor noise gate.

De-Esser Status

- **Bypass:** Turns off the Compressor De-Essing function.

- **Active:** Turns on De-Essing function.

Noise Gate Threshold

- Select the **Use:** radio button to specify the level, up to -50 dB, at which the Noise Gate activates to attenu- ate input signals that fall below that level.

Noise Gate Depth

- Select the **Use:** radio button to specify the amount, up to -30 dB, of attenuation to apply to the input signal when the specified Threshold is met.

Compressor Threshold

- Select the **Use:** radio button to specify the ceiling, up to -30 dB, at which the Compressor activates to attenuate input signals above that level.

Compressor Ratio

- Select the **Use:** radio button to specify the aggressiveness of the compressor, up to 16:1.

De-Esser Threshold

- Select the **Use:** radio button to specify the ceiling, up to -20 dB, at which the De-Esser activates to attenuate sibilance.

De-Esser Ratio

- Select the **Use:** radio button to specify the aggressiveness of the De-Esser, up to 8:1.

Compressor Mode

- **No Freeze:** The compressor is "free" to operate no matter the input level.
- **Freeze:** Prevents the compressor from "sucking up" room noise during brief pauses in audio input.

Backward Feed Dim Gain

- Select the **Use:** radio button to specify the amount of attenuation, up to -30 dB, will be applied to program audio being fed back to this source (if it has an associated Backfeed, such as an IFB or Talkback channel).

Channel On/Off Status

- **Safe ON:** If the fader is ON, and the Show Profile specifies a new source be loaded to this fader, the new source will be queued until the fader is turned off.
  If the new source is the same as the old source, the fader is immediately turned ON when the Show Profile loads.
- **Force OFF:** Immediately turns the fader OFF when the Show Profile loads.
- **Force ON:** Immediately turns the fader ON when the Show Profile loads. If a new source is to be loaded to the fader, it is changed immediately, regardless of whether audio from a previous source is passing through the fader.

Control Lock Map

- **PGM1 – PGM4, Options, Preview, On/Off, Fader, Talkback, HP Source:** Place a check mark in any of these boxes to keep the board op from changing the Show Profile's pre-selected options for this fader.

Group Start

Fusion contains a Group Start feature that enables the user to turn several faders ON by pressing the ON key of the Master fader; useful in roundtable discussions or multi-talent bullpens.

This function can be controlled in this screen, or by setting the option in each individual Channel Options screen.

- **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.

- **Independent:** Normal ON/OFF operation.

Fader Position

- Select the **Use:** radio button to specify the level at which the fader should be set when the Show Profile is loaded. You may specify any level between -73 dB and +10 dB.

  This setting operates in conjunction with the **Channel On/Off Status** described above. Using these two settings, you can set a channel to turn ON, and its fader to assume a preset output value, when the Show Profile is loaded. This is useful when creating a Show Profile for use with automation systems; combine it with the **Control Lock Map** controls to set up a Show Profile for unattended, automated operation with controls that cannot be inadvertently changed by careless operators.

Individual Headphones Section

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.

Current Source

- If a Microphone source is loaded to this channel, choose the audio source to be fed to its Individual Head- phone Feed (Backfeed), if desired. Choices include all Program and Aux Send buses, Monitor bus, or a direct feed from any individual source.

Source For Preset 1 / Preset 2

- Select the audio source assigned to the Preset keys on Axia Talent Headphone accessory panels. Choices include all Program and Aux Send buses, Monitor bus, or a direct feed from any individual source.

Headphones Master Gain

- Select the **Use:** radio button to specify the level at which Individual Headphone channel should be fed. You may specify any level between -85 dB and 0 dB.

Talkback Start Volume Low Limit

- Select the **Use:** radio button to specify the level at which Talkback will be fed to the Individual Headphone channel. You may specify any level between -85 dB and 0 dB.

Record Mode Section

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 Record Mode is entered and exited.

Program 1 – 4

- **Assign, While In Record Mode:** Assigns this channel to the selected Program bus when Record Mode is active.

- **Remove, While In Record Mode:** Removes this channel from the selected Program bus until Record Mode is exited.

- **No Change, While In Record Mode:** This channel's bus assignments do not change when Record Mode is active.

ON/OFF

- **Disable, While In Record Mode:** Prevents the operator from changing the channel ON/OFF state when Record Mode is active.

- **No Change, While In Record Mode:** Channel ON and OFF keys function normally during Record Mode.

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

Auxiliary Send & Return Screen Options

This screen allows you to pre-select settings pertaining to the 4 Aux Send and 2 Aux Return mixing buses.

Aux Send A-B-C-D Master Gain

- Select the **Use:** radio button to specify the master gain level at which the selected Aux Send bus will be set. This is the gain applied to the Aux Send bus after any sources have been assigned to it. You may specify any level between -60 dB and +10 dB.

Aux Send A-B-C-D On/Off Status

- **Off:** Turns off selected Aux Send bus upon Show Profile load.
- **On:** Turns on selected Aux Send bus upon Show Profile load.

Aux Return A-B Gain

- Select the **Use:** radio button to specify the gain level at which the selected Aux Return bus will be set. You may specify any level between -60 dB and +10 dB.

Aux Return A-B On/Off Status

- **Off:** Turns off selected Aux Return bus upon Show Profile load.
- **On:** Turns on selected Aux Return bus upon Show Profile load.

Aux Return A-B Signal Mode

- **Stereo:** sets bus signal mode to Stereo.
- **Left** / **Right:** takes the chosen side of the bus' stereo signal and sends it to both sides of the stereo bus output channel.
- **Sum:** Creates a sum of both stereo channels and sends the sum to both sides of the stereo bus output chan- nel.

Aux Return A-B Panorama Position

- Select the **Use:** radio button to Pan the output signal of the bus to left or right of center.

Aux Return A-B Assign To PGM1-2-3-4

- **On:** Assigns the output of the specified Aux Return bus to the selected Program bus(es) upon Show Profile load.
- **Off:** Removes the output of the specified Aux Return bus to the selected Program bus(es) upon Show Pro- file load.

Aux Return A-B Source ID

- Dropdown box selects Aux Return's assigned source from saved Source Profiles. Note that Program and other Aux buses are not available in this dropdown; choices are limited to external devices only.

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

Monitor Section Screen Options

Options in this section permit you to define Control Room and Studio Monitors & Headphone selections, and determine how on-screen Meters and Timers will behave when this Show Profile is loaded.

General Monitor Settings Section

Timer Mode (Onscreen "Count-Up" Timer)

- **Manual**: Gives board operator manual control of the event timer using the Timer keys found on the con- sole's Monitor Module.

NOTE: This setting interoperates with the Auto-Start Timer option. The Auto-Start Timer option can be set in two ways:

• Check the Auto-Start Timer box in a Source Profile to enable the function whenever a source is loaded to any fader.

• Use the Auto-Start Timer selection found in the Show Profiles Channel Options screen to override the Source Profile setting for a particular Show.

- **Auto-Reset:** Timer resets to zero and begins counting when a fader is turned ON.
- **Auto-Add:** Timer begins counting when a fader is turned ON and stops counting when that fader is turned OFF. In this mode, the timer will not reset to zero when it is restarted.

Show Tenths On

- **No Timers:** Tenths-of-a-second are hidden on both Count Up and Countdown timers.
- **Down Timer:** Tenths are only shown on the Countdown timer.
- **Up Timer:** Tenths are only shown on the Count Up timer.
- **Both Timers:** Tenths are shown on both timers.

Preview Interlock Mode

- **Disabled:** Board op can "button mash" multiple Preview buttons to preview multiple sources.
- **Enabled:** Button-mashing is disabled; selecting a Preview button removes any other source from the Pre- view bus.

Switched Meter Source Select

- **Program-4:** Selects Program 4 mixing bus to be metered on the #4 on-screen meter display.
- **CR Monitor/Preview:** Switches the #4 meter to the Control Room Monitor channel. When a source is placed in Preview, meter switches to display loudness of the

source in Preview.

Program 3 Meter Input

- Dropdown box allows you to select from **Program 3**, **Record bus, Phone bus** or the sources assigned to the Monitor Module's **External 1** and **External 2** choices to be displayed on the #3 meter.

Preview Speaker Master Gain

- Select the **Use:** radio button to specify the level at which the Preview bus will be set when the Show Profile is loaded. You may specify any level between -85 dB and 0 dB.

Preview Speaker Muted State Gain

- Select the Use: radio button to specify the level at which the Preview bus will be heard when it is muted. You may specify any level between -85 dB and 0 dB.

NOTE: Normally, turning a Control Room microphone channel ON mutes the Preview speakers entirely. This setting allows you to let the Preview bus be heard in the Control Room at a reduced level, even while CR Mics are active.

Source ID For External Preview

- Dropdown box allows any Source not assigned to a fader to be heard in the Preview channel. Note that this must be enabled by a pin on the CR Monitor GPIO.

  Example: you have an intercom system you wish to feed into the console's Preview channel. To do this, use the Dropdown box to select the intercom's audio source, then take the GPO from the intercom and use it to gate open the External Preview input, which would be fed by the intercom audio.

Talkback Gain

- Select the **Use:** radio button to trim the level of the Talkback channel. You may specify any level of cut or boost between -30 dB and +10 dB.

Feed To Source Sum Gain

- Select the **Use:** radio button to specify a gain compensation for mono-sum of Talkback audio sources. You may specify between -6 dB and -3 dB of attenuation.

User Buttons GPIO Channel

- If you have a Quasar Expert Monitor Module and wish to control an external device using its 4 User keys, enter the GPIO Channel Number of the device here.

GPIO Channel For Up Timer Control

- Allows an external device to trigger the Up Timer. Enter the GPIO Channel Number for the desired device.

<u>GPIO Channel For Down Timer Control</u>

- Allows an external device to trigger the Down Timer. Enter the GPIO Channel Number for the desired device.

Additional Meters Section

<u>Extra Meter 1-2-3-4 Input</u>

- This section allows to you specify meter sources for the “extra” meters that can be displayed in the center section of Quasar’s on-screen display when using an Expert Monitor Module. The selections for each of the meters are extensive and include all **Program, Aux, External** and **Monitor** busses, the **Phone** and **Record** buses, all **Fader Channel sources and backfeeds,** all **VMix inputs**, **Direct, Sub** and **Main** outputs, as well as all **VMode inputs and outputs.**

- The extra meters are displayed when the board operator presses the **Meter Options** key and presses the #6 Function knob to select the **More Meters** option.

Sources For External 1 & 2 Section

<u>Source ID For External Input 1 – 2</u>

- Use the Dropdown boxes to select sources to be loaded to the **External 1** and **External 2** Monitor Selection keys when the Show Profile is loaded.

- Board operators can override these selections by pressing and holding the External keys for 5 seconds, then choosing from the onscreen list of sources.

Control Room Monitor Options Section

<u>Source</u>

- Use the radio buttons to choose the source that will be selected to feed the Control Room Monitor speakers upon Show Profile load.

<u>CR Monitor Master Gain</u>

- Select the **Use:** radio button to specify the level of the CR Monitor speakers upon Show Profile load. You may specify a value between -85 dB and 0 dB.

<u>Signal Mode: CR Monitor</u>

- **Stereo:** sets 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 the sum to both speakers.

CR Monitor Dim Gain

- Select the **Use:** radio button to specify the amount of attenuation, up to -30 dB, to be applied to the CR Monitor channel when Talkback or Preview are in use.

CR Monitor Muted State Gain

- Select the **Use:** radio button to specify the amount of attenuation, up to -85 dB, to be applied to the CR Monitor channel when muted.

NOTE: Normally, turning a Control Room microphone channel ON mutes the CR Monitor speakers entirely. This setting allows you to let the monitors to be heard in the Control Room at a reduced level, even while CR Mics are active.

GPIO Channel For CR Monitor

- Enter the GPIO channel number used to trigger On Air lamps or other GPIO functions. This option should typically be programmed with the same logic channel number for all Show Profiles.

Control Room Headphones Options Section

Source

- Use the radio buttons to choose the source that will be selected to feed the Control Room headphones upon Show Profile load.

CR Headphones Master Gain

- Select the **Use:** radio button to specify the level of the CR Headphones upon Show Profile load. You may specify a value between -85 dB and 0 dB.

Signal Mode: CR Headphones

- **Stereo:** sets Headphone channel to Stereo.

- **Left / Right:** takes the chosen side of the Headphone channel and sends it to both speakers.

- **Sum:** Creates a sum of both sides of the Headphone channel and sends the sum to both speakers.

CR Headphones Independent

- **Follow Monitors:** CR Headphone source mirrors CR Monitor source selection.
- **Use Headphones Source Select:** CR Headphones and CR Monitors are selected independently.

Preview-In-Headphones Mode

- **Off:** Sources assigned to Preview bus are not heard in CR Headphones.
- **Stereo:** Sources assigned to Preview bus are heard in both sides of the CR Headphones.
- **Split:** Sources assigned to Preview bus are summed to mono and heard only in right side of the CR Head- phones. Regular audio is heard in left side.

CR Headphones EQ Active

- **Bypass:** No headphone EQ.
- **Active:** Headphones are EQd using the following settings.

CR Headphones EQ High Mode

- **Shelf:** Selects high-shelf style of EQ application.
- **Peak:** EQ is applied to a selected frequency, “notch filter” style.

CR Headphones EQ High/Mid/Low Frequency

- Select the **Use:** radio button to set the center frequency of the selected band.

CR Headphones EQ High/Mid/Low Gain

- Select the **Use:** radio button to specify an EQ cut or boost between -25 and +15 dB to the selected frequency.

Studio Monitor Options Section

Source

- Use the radio buttons to choose the source that will be selected to feed the Studio Monitor speakers upon Show Profile load.

Source ID For External Input

- Dropdown box chooses what source will be auditioned when External is chosen from the Studio monitor assignment list.

Studio Monitor Master Gain

- Select the **Use:** radio button to specify the level of the Studio Monitor speakers upon

Show Profile load. You may specify a value between -85 dB and 0 dB.

Studio Monitor Dim Gain

- Select the **Use:** radio button to specify the amount of attenuation, up to -30 dB, to be applied to the Studio Monitor channel when Talkback or Preview are in use.

Studio Monitor Muted State Gain

- Select the **Use:** radio button to specify the amount of attenuation, up to -85 dB, to be applied to the Studio Monitor channel when muted.

NOTE: Normally, turning a Studio microphone channel ON mutes the Studio Monitor speakers entirely. This setting allows you to let the monitors to be heard in the Studio at a reduced level, even while Studio Mics are active.

GPIO Channel For Studio Monitor

- Enter the GPIO channel number used to trigger On Air lamps or other GPIO functions. This option should typically be programmed with the same logic channel number for all Show Profiles.

Master Module Control Lock Map Section

- This functions like a "master lock" for console functions – a superset of the Control Lock Map found in the Show Profiles Channel Options pages. You can essentially prevent any setting on the board from being changed if you so desire.

  Place a check mark in any of these boxes to keep the board op from changing the Show Profile's pre-select- ed options.

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

Record Mode Screen Options

Record Mode is like a "macro" that helps talent quickly prepare to record phone bits, interviews or other program segments for later airing. Any source assigned to the Program-4 bus automatically feeds the Record and Phone buses as well.

Sources assigned to Program-4/Record will follow the Record options in their Source Profiles; the options here primarily affect the behavior of Monitors when Record Mode is invoked.

Record Mode Configuration Section

Record Mode Activation

- **Disabled:** Disables Record Mode entirely for this Show Profile.

- **Enabled:** Allows activation of basic 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.

GPIO Channel For Recorder Control

- 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.

Flexible Record Mode Options Section

When "Flexible" is chosen as the Record Mode Activation option, the following options are active.

NOTE: The options below allow you to customize the Monitor, Headphone and Meter settings that are auto- matically selected when you *ENTER* Record Mode.

CR Monitor Source

- **Retain:** Does not change the Monitor feed; keeps the Monitor selection the board operator was using prior to entering Record Mode.

- **Program 1 – 4, Record, Phone, Auxiliary A – D, External 1 – 2:** Changes the Monitor feed to the selected bus or Monitor channel when Record Mode is engaged.

- **Recall:** Re-loads the Monitor selection that was manually selected by the board operator *the last time* Record Mode was active.

Studio Monitor Source

- Same options as described in "CR Monitor Source" above, but for the Studio Monitor feed.

CR Headphone Source

- Same options as described in "CR Monitor Source" above, but for the Control Room Headphone feed.

4th Meter Source

- **Retain:** Does not change the #4 Meter on the Quasar display; keeps the Meter selection the board operator was using prior to entering Record Mode.

- **Program 4, Record, Phone, Auxiliary A – D, External 1 – 3, Monitor:** Changes the #4 Meter to the selected bus, external source, or the audio of the CR Monitor when Record Mode is engaged.

- **Recall:** Re-loads the #4 Meter selection that was manually selected by the board operator *the last time* Record Mode was active.

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

Group Start Screen Options

Group Start

Quasar contains a Group Start feature that enables the user to turn several faders ON by pressing the ON key of the Master fader; useful in roundtable discussions or multi-talent bullpens.

This function can be controlled in this screen, or by setting the option in each individual Channel Options screen.

- **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.

- **Independent:** Normal ON/OFF operation.

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

Phone Screen Options

The Phone screen is used when setting up a Telos talkshow system for use with your console. Please refer to Chapter 5, "Working With Phone Hybrids", for details. If no Call Controller module is installed, this screen will be empty.

Working With Layers

Quasar is the first Axia console that introduces 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.

The Layer Concept explained

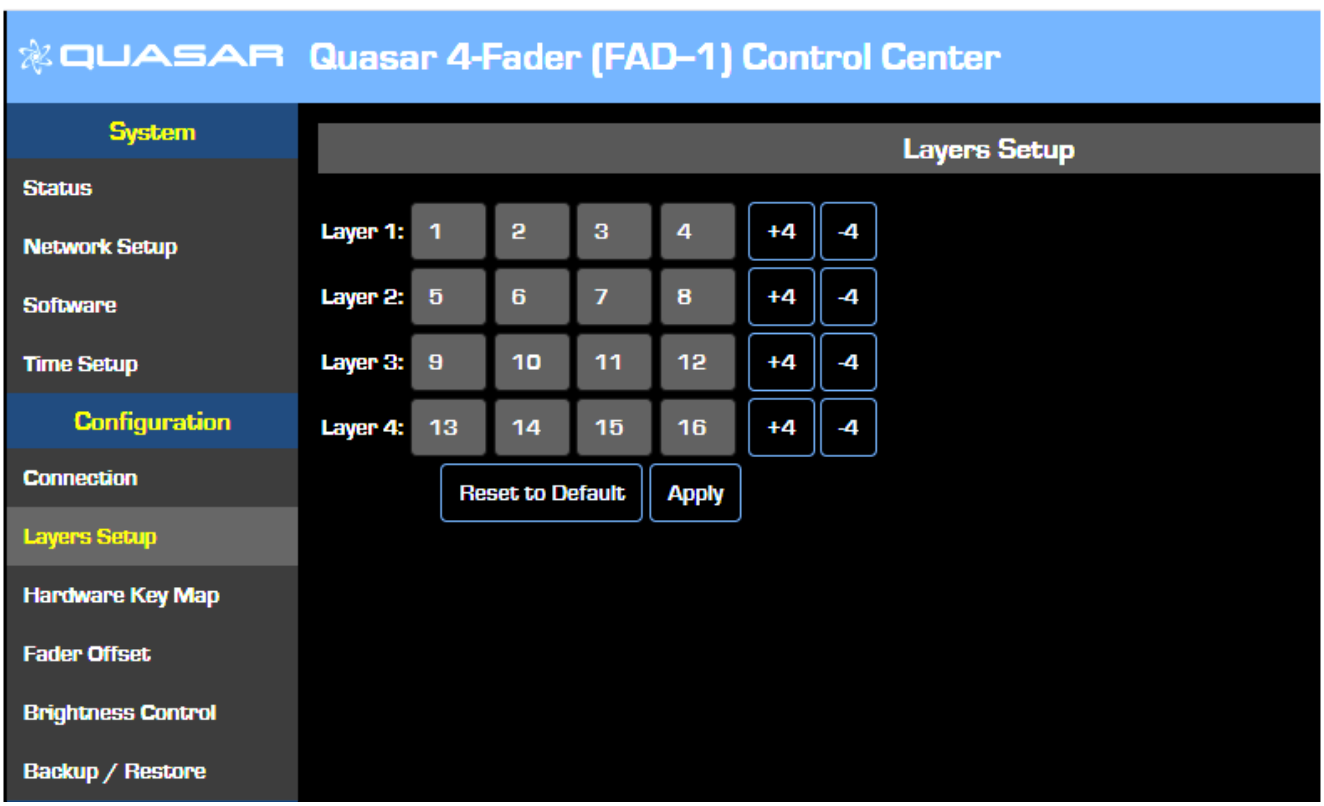
The Quasar Surface can be ordered with a variable number of faders strips, while the Quasar Engine has a fixed number of DSP Input channels. In fact, a Quasar Engine (at the time of writing this manual) always comes with 64 channels.

Since there is no longer a 1:1 correspondence between fader strips and input channels, a way to assign 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 the surface in any order, and multiple views of the same channels, if desired. Up to four Layers can be configured on a surface.

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, 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 buttons on the Master Touchscreen module.

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

Layer 1:	1	2	3	4
Layer 2:	17	18	19	20
Layer 3:	33	34	35	36
Layer 4:	49	50	51	52

Fader Module # 2

Layer 1:	5	6	7	8
Layer 2:	21	22	23	24
Layer 3:	37	38	39	40
Layer 4:	53	54	55	56

Fader Module # 3

Layer 1:	9	10	11	12
Layer 2:	25	26	27	28
Layer 3:	41	42	43	44
Layer 4:	57	58	59	60

Fader Module # 4

Layer 1:	13	14	15	16
Layer 2:	29	30	31	32
Layer 3:	45	46	47	48
Layer 4:	61	62	63	64

Layer 1:	1	2	3	4	+4	-4
Layer 2:	17	18	19	20	+4	-4
Layer 3:	33	34	35	36	+4	-4
Layer 4:	49	50	51	52	+4	-4

Reset to Default | Apply

TIP:
This work can be done manually, by typing the Input Channel number in each field, or by using the +4 or -4 buttons
available the right edge of each layer. These will increment or decrement the channel assignments four at a time.

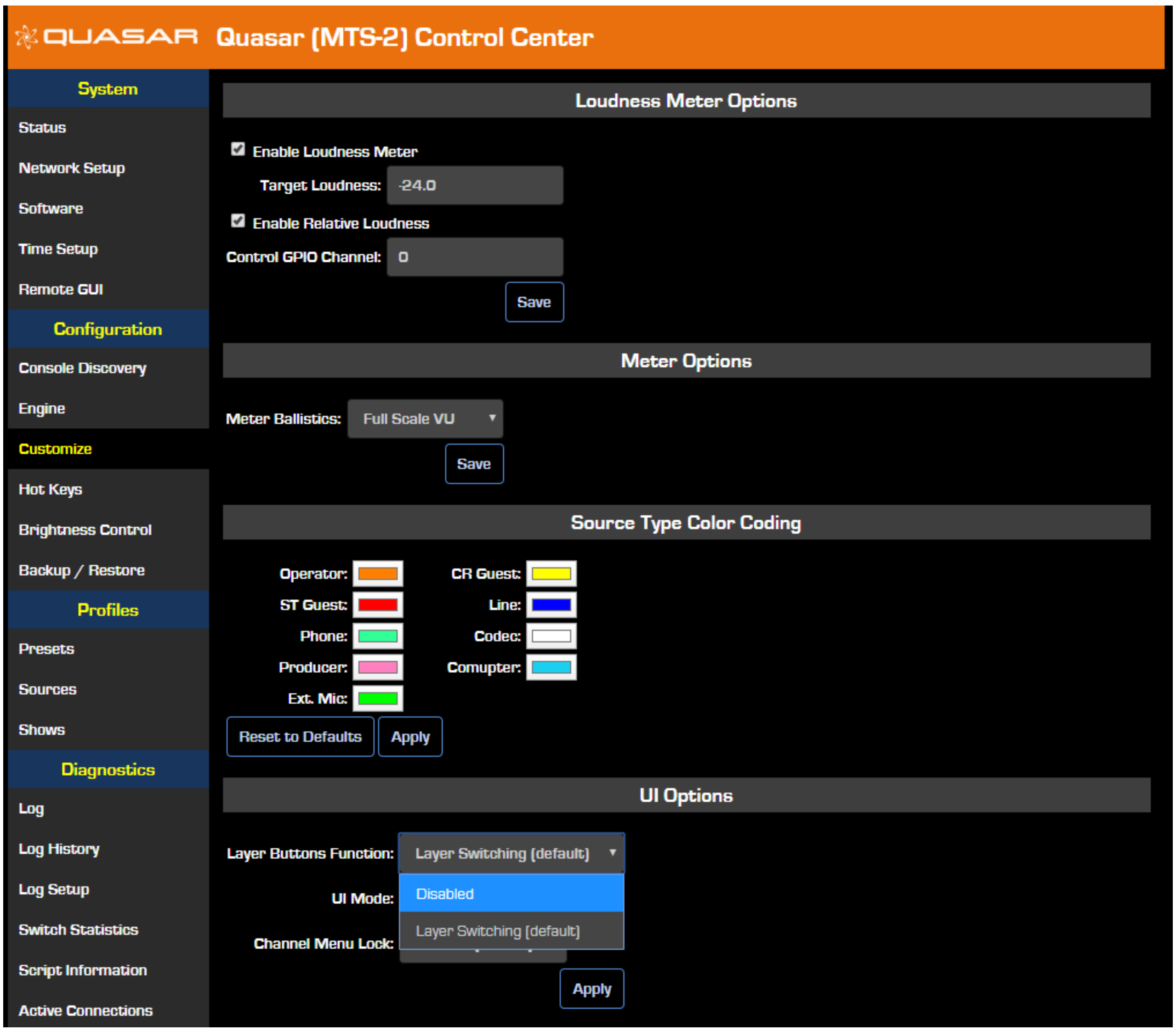
You can also go back to Defaults hitting the Reset button.

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 and therefore not stored in Show profiles .

Enabling or Disabling Layers

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:

Using a PC connected to your network, launch a web browser, and enter the IP address assigned to the 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.**



Other Configuration Options

Beyond Source and Show profiles, there are some global items that allow customization of the console to your needs. In the column of links on the left hand side is the **Customize** option. From this link you can configure a few global options.

Clock:

The Quasar Master Module display home view has time information and this can be configured in various ways. By pressing the **Set time from PC** button, the system time will adjust to the time which is on the PC accessing the page. The digital 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 provided so that time can be shown for your region. If you have an NTP server on the network, you can define its IP address for the Quasar time synced to the server.

What's Next

With configuration basics covered, the next chapter moves on to Console Operations – a tutorial on the controls the board operator will use to run the console.

Chapter 3

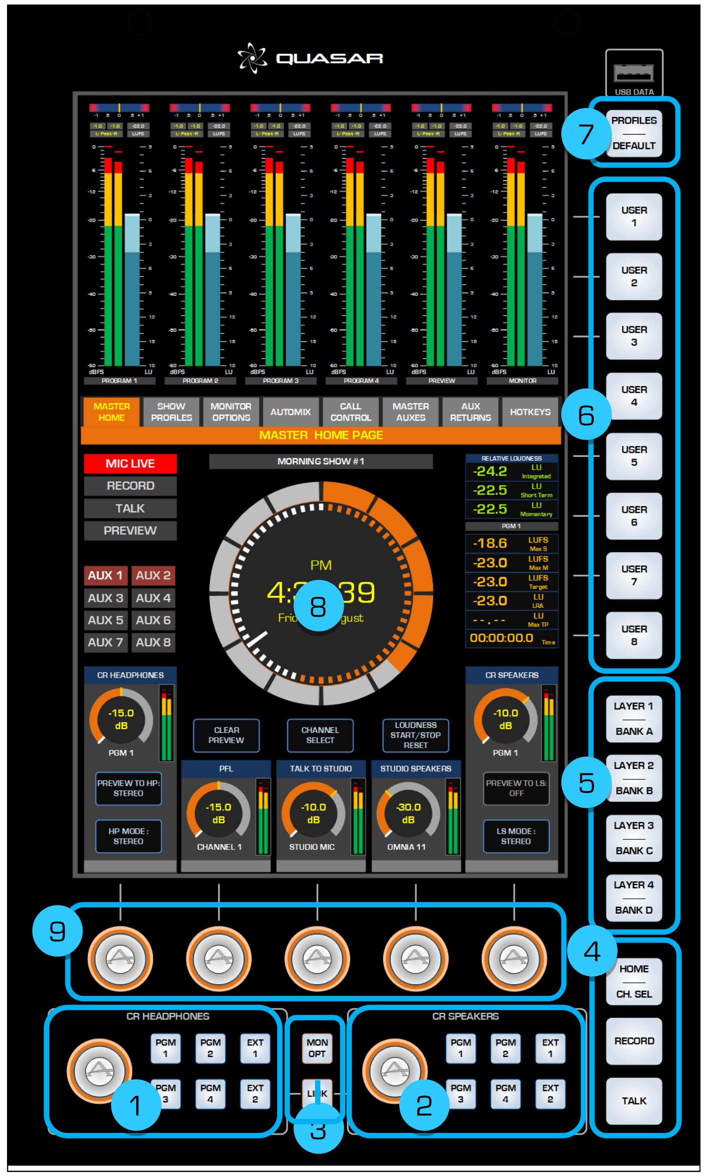
Operation

Quasar was designed with control in mind. Each facilities needs are unique and the ability to customize the workflow is important. Additionally different people have different workflows and we've made it possible to access some functions in multiple ways. This way, the board operator 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 modules to help familiarize you. Then, we'll have a look at some of the more specialized options.

The MTS-MON Master Module

Also referred to as "MTS Module" (or "Master Module") is the center point for all control in the Quasar system. A combination of fixed buttons, soft hardware (hardware that changes function based on the user operation), and touch display controls.



1. **CR Headphones section.** There are 6 buttons and a touch sense rotary encoder that are fixed to the operator's headphone control. The 6 buttons are used to select the more common mixes that need to be monitored. The rotary encoder, when touched will highlight Headphone information on the display (8) and a rotation of the encoder will adjust the level to the operator's headphones.

2. **CR Speakers section.** The same as above (CR HP) where six buttons and a touch sense rotary encoder are fixed to control the main monitor speakers in the control room. Between both sections is a ***LINK*** button which causes the two sections to copy each other. Selecting a mix in the Speaker section will also select that same mix in the Headphone section.

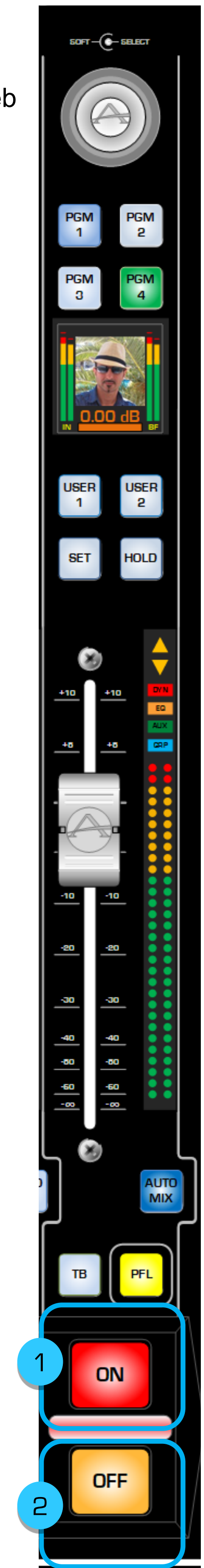
3. **MON OPT and LINK.** The MON OPT button enables the operator to have more control options over the monitoring. Pressing this button changes the display (8) and provides additional sources to be monitored, control to dim values, dedicated talkback functions and activation of CR Headphone 4-Band EQ. The LINK button links the input selection of the two CR monitor sections, by making the HP follow the SPK selection.

4. **Fixed Function buttons.** Quick access buttons for operators to navigate their workflow. The ***HOME/CH.SEL*** returns the display to the HOME screen (Starting point of navigation) when pressed momentarily. A press-and-HOLD of this button will change the display so that any channel can be selected for modification. Channel modification options are covered later. ***RECORD*** button will enter into (or exit) the record mode. Record mode offers customization

needed for recording workflows. Currently the ***TALK*** button will trigger the Talk to Studio function.

5. **Layer / User buttons.** Quasar offers 4 surface layers. These buttons enable the operator to quickly select the layers. Layers can be configured by accessing the Web UI built in each fader module. If layers are not needed, the system could be configured so that each layer is a duplicate of the previous and these buttons can be used as four Bank (A-B-C-D) buttons for the eight user programmable buttons located above. With the use of Pathfinder, the buttons can achieve any custom needs of the facility.

   Alternatively, the Layer buttons can be completely disabled in the MTS Module Web UI, (by browsing the Web UI *Customize* page).



6. **Touch sense User buttons.** Eight user buttons with capacitive touch sense are available for custom functionality. With the use of Pathfinder, these buttons can be made to initiate changes with the surface or some other property within the facility. The touch sense function will drive a "*pop-up*" on the display which acts as a dynamic labeling option for the buttons. This will the button label when the finger touches the button, <u>before it is pressed</u>. Pathfinder can change the text in the pop-up and change the color of the button (RGB color illumination).

7. **Profile button.** A fixed button which shows the defined show profiles onto the display for user selection. Currently, a press-and-HOLD of the button will load the *last loaded profile* (back to its default settings).

8. **Display.** 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 will cover this item in more detail.

9. **Five touch sense rotary encoders.** The five rotary encoders are soft driven, meaning their functionality changes based on actions by the operator. The display will 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 display is not viewing the current functionality of the rotary encoder.

The XR-4FAD Fader Module

The XR 4-Fader module is a full control module that is connected through a network connection.

The XR 4-Fader module is composed of the following items:

1. **Channel encoder**. Can respond to touch, press, and rotate. 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. A short press can trigger the Channel select which drives the master section into Channel options (blue tabs).
2. **PGM 1-4 buttons**. Main program mix assignments are the default configuration for these 4 buttons which assign the channel to feed the designated mix bus.
3. **TFT Color display**. Provides feedback to the user on the state of the channel, which profile is loaded, gain values, confidence metering, or an image can be shown as configured as part of the source profile.
4. **USER buttons**. Two of these buttons are generally configured as User buttons, while the other two are generally configured for integrated Phone workflows and assigned by default to the following functions:

- SET – used to select the channel or seize the line
- HOLD – place a call on hold.

5. **Channel Fader**. This high-quality spill-proof motorized fader controls the gain of the input channel.
6. **LED feedback**. The arrow indicators show fader position in relation to system level. When the fader is touched by a finger, both arrows light up to indicate that the touch sensor is active. After movement, this double indication will turn off after 2 seconds. The four rectangular LEDs provide status indication for the channel's Dynamics processing (DYN), Equalization (EQ), AUX send assignment (AUX), and Group assignment (GRP).
7. **Channel Bargraph**. This LED bar normally works as stereo level meter for the channel input (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.
8. **Dedicated buttons** for Talkback, PFL, and Automix functions.

- Talkback (TB) permits the operator to communicate through a backfeed to the source if a backfeed is enabled. A tap will latch the talkback condition where a press and 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.
- Automix is used to engage the channel into Automixer.

9. **ON/OFF buttons** for the channel.

NOTE: For those seeking an advanced customization of their surface, we remind that ALL buttons on every channel strip can be configured by the user with a predefined function as an alternative to the default one, or work as custom buttons driven by an external control system such as Pathfinder.

Quasar Touchscreen User Interface

The Quasar UI workflow is based on the following concept:

- 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

The display is divided into three areas:

1. **Meters.** The top area is the meter area and this area maintains a view of the final audio mix meters. Some channel option functions will replace the audio meters with views in support of the active function. For example, when adjusting the EQ of a channel, this area will have a view of the frequency vs amplitude so the operator can visualize the changes being made.

2. **Menu tabs.** The tabs area is for navigating the various UI pages. The tabs have two colors to help the see at a glance where they are in the navigation:

   Orange is the ***Home*** color: all tabs in this menu provide access to controls typically found in any console's Master Module. These controls define functions belonging to the entire surface.

   Blue is the ***Channel*** color: 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.

   Based on which tab is selected, that tab will be illuminated in its color while others within the class are grey. The bottom area will display the items related to selected tab.

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.

Quasar Master menu (Orange Tabs)

Master Home Tab

It's the starting page of the Quasar Touchscreen User Interface. The Master Home page 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.

- **PFL to HP:** routes the PFL (Pre-Fader Listen) bus to the Control Room Headphones out.

- **HP Mode:** Selects the listen mode for the CR Headphones (Stereo, Left, Right, Mono Sum).
- **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.
- **Channel Select:** Brings an option to select any channel from the touchscreen, as an alternative to pushing the encoder located at the top of each fader strip.
- **Loudness Start / Stop / Reset:** Controls the Loudness metering counters.
- **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 Tab

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 and press the **LOAD SHOW** button to the right.

Other buttons to the right let the userperform the following operations:

- **REFRESH LIST:** refreshes the list of the available Show Profiles, in case this has been edited from the web UI
- **PAGE UP:** pages up the list
- **PAGE DOWN:** pages down the list
- **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.
- **DELETE :** removes the profile from the library of show profiles.

Note: 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.

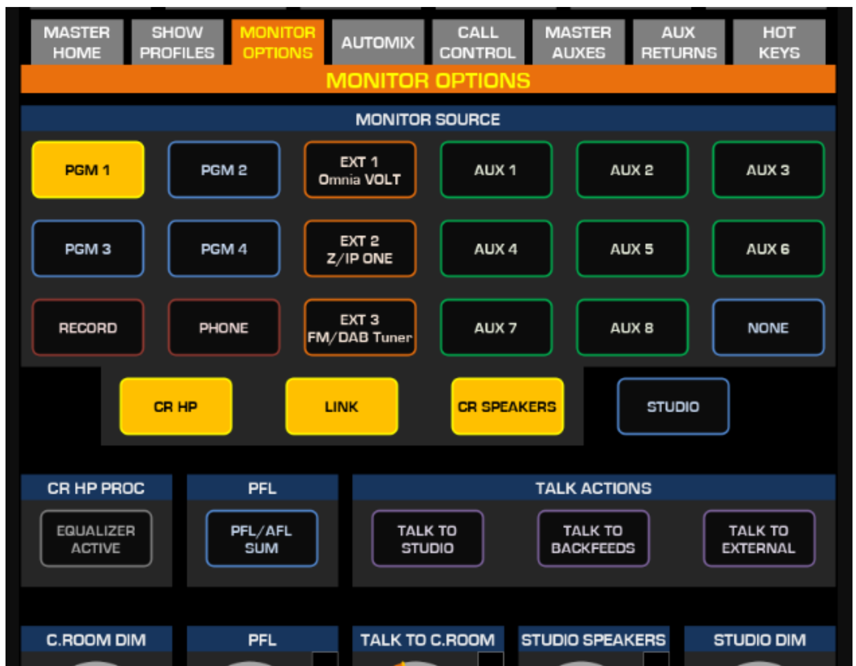
Monitor Options Tab

This view provides more detailed control over the monitors. As detailed earlier, there are some dedicated hardware buttons for monitor control which provide the most common sources. If the operation 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. Additionally, touching the display and sliding the finger to the left or right will also adjust the level.

In the Monitor Options page it is possible to set the source for the three Monitor Sections:

- Control Room Headphones (CR HP)
- Control Room Speakers (CR SPK)
- Studio Monitors



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. In order to assign a source to the external inputs, press and 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.

Automix Tab

The view is divided in three main sections:

at the top, the CHANNEL SELECTOR section allows to toggle selection of the channels in the system and to see which ones are enabled into automix.

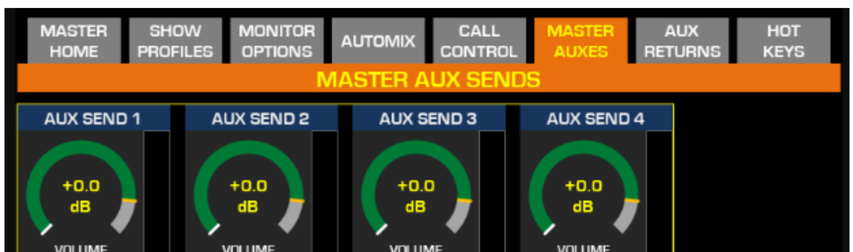
At the bottom left, the CHANNEL PARAMETERS section allows to change the enable state or adjust the priority of the channel within the automix.

At the bottom right, the GLOBAL PARAMETERS section 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.

Call Control (Orange tab)

For use in multiline workflows, such as is available with the Telos VX system, the Call Control view will provide access to 8 phone lines and the ability to assign those to two different channels. The view is organized with two columns for assignment to two different channels. In conjunction with the SET/HOLD buttons on the channel, the operator can take callers and make calls through this interface. For more on phone workflows, please refer to Appendix-YYY.

Master Auxes Tab



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

Aux Returns Tab

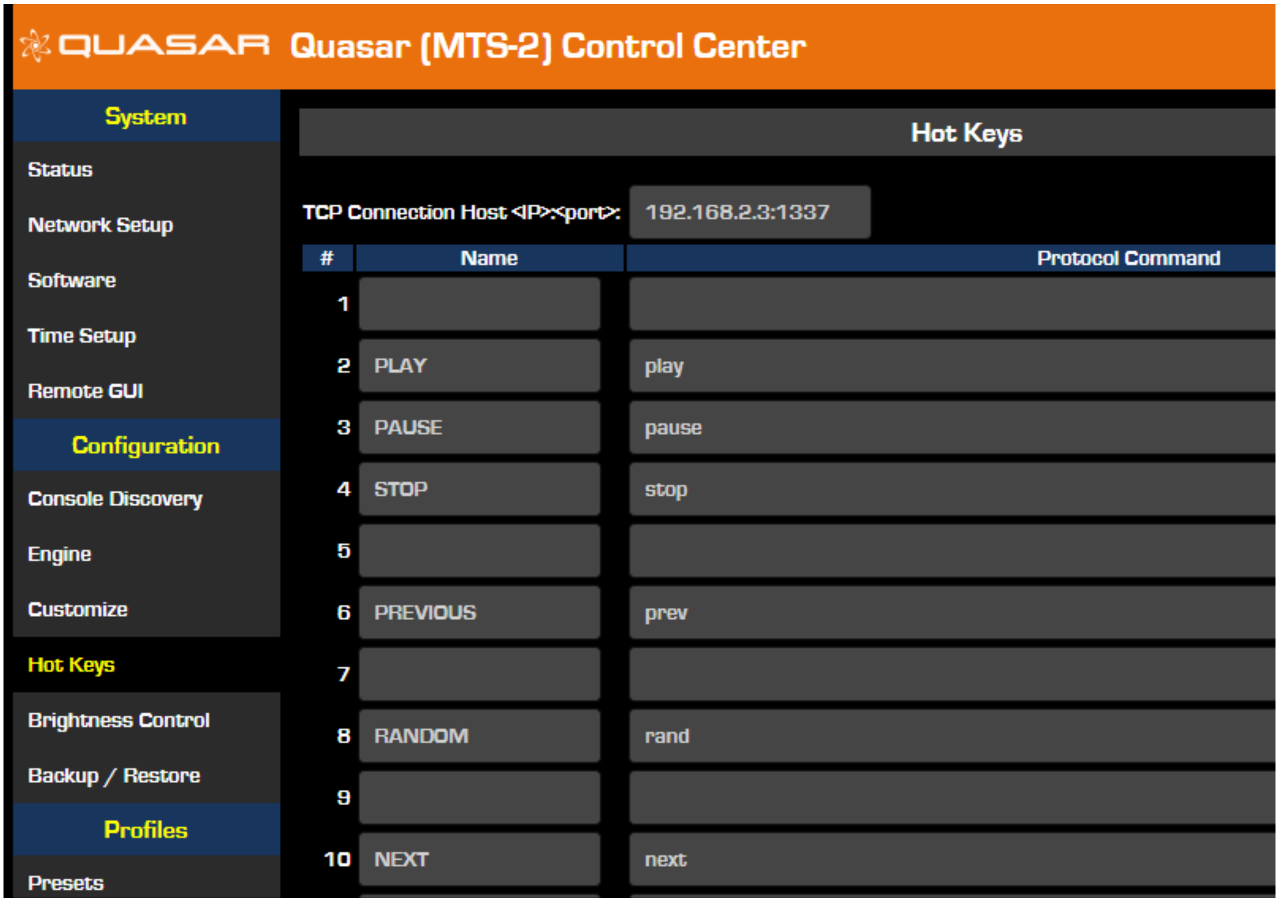
The system includes two AUX returns. These are basically two “short” input channels, (without DSP processing) typically used to bring any external FX processor return back into the mix.

This view will provide a 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 Tab

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

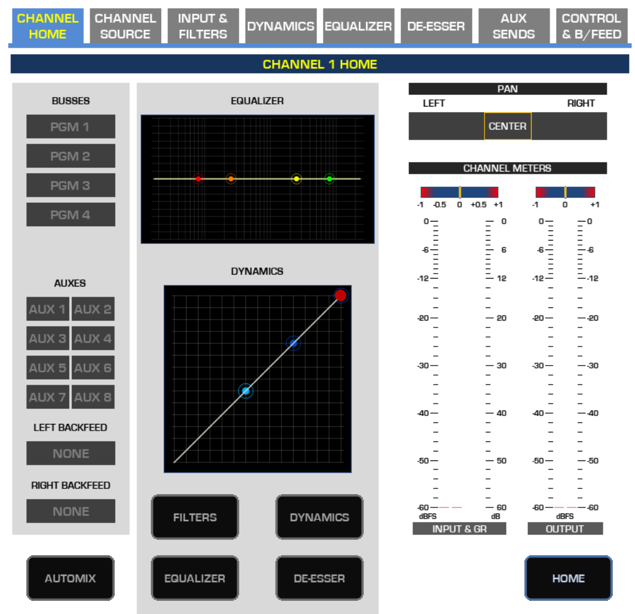

QUASAR Quasar (MTS-2) Control Center
System
Status
Network Setup
Software
Time Setup
Remote GUI
Configuration
Console Discovery
Engine
Customize
Hot Keys
Brightness Control
Backup / Restore
Profiles
Presets
Hot Keys
TCP Connection Host <IP>:<port>: 192.168.2.3:1337
Name Protocol Command
1
2 PLAY play
3 PAUSE pause
4 STOP stop
5
6 PREVIOUS prev
7
8 RANDOM rand
9
10 NEXT next

Quasar Channel menu (Blue Tabs)

The channel controls, or blue tabs, are made available when the option knob is engaged on a surface channel or one of the channel select buttons are pressed on the master module.

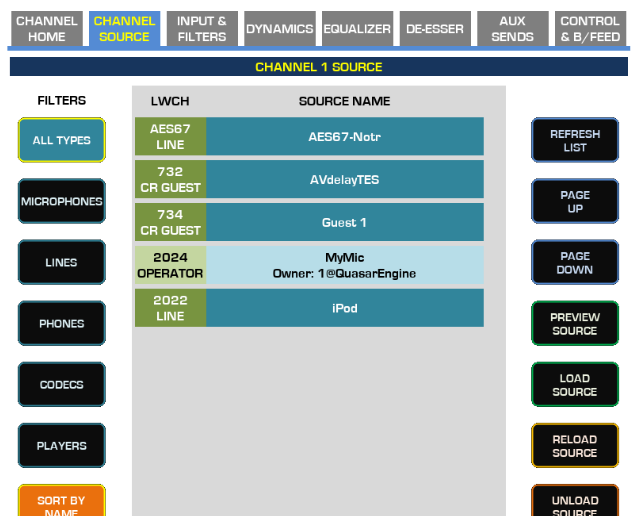
Channel Home (Blue tab)

This is the default screen anytime entering into channel control. The screen provides status of the channel, such as mix assignments, EQ curve, dynamic, Backfeed source, pan, and audio meters (pre/post audio meters). There are buttons which allow to enable/disable audio processing or automix.



Channel Source (Blue tab)

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 by type: MICROPHONES, PHONE, CODECS, PLAYERS, LINES. The list can be organized by name or be Livewire channel (LWCH).
The right hand column are additional buttons to navigate the list, preview the source, of loading options.



Input & Filters (Blue tab)

This view provides various controls to the input as well as provide feedback information.

- Input audio meter as the source enters into the DSP.
- Input gain control
- Mode (Stereo, Left, Right, Sum)
- Node control (applicable if profiles defines the node device)
  - Mic Gain
  - Phantom power

- Trim gain

- Pan & Phase
- Filters (High/Low)
- Backfeed audio confidence meter
- Dim gain control (when Talkback is engaged, the main audio will be dimmed this value)

Dynamics (Blue tab)

New and improved algorithms behind the Quasar offer more control over the audio. The compressor/limiter section provides

- Threshold level (xxdB to 0dB) to where the limiter begins to function.
- Ratio (xx:1 – 1.0:1) setting for how aggressive the processing would be.
- Knee control for the finer details on the transition.
- Auto Makeup is a special type of Makeup Gain: It 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 control is designed to let the user operate fine adjustments of the Compressor Threshold while on the air, because the output level is always aligned with the input
- Attack time control for how quickly the processor is to respond.
- Release time for how slowly to release the processing.
- Auto Attack/ Release will provide optimized Attack and Release time settings, for either music or vocal programs. The Speech preset is optimized to react quickly to voice transients, while the Music preset will provide natural compression for music programs.

The Expander/Noise Gate section provides

- Threshold level (xxdB to 0dB) to where the processing begins to function.
- Ratio (xx:1 – 1.0:1) setting for how aggressive the processing would be.
- Knee control for the finer details on the transition.
- Depth (-xxdB – 0dB) to the limiting
- Attack time control for how quickly the processor is to respond.
- Release time for how slowly to release the processing.

The options section provides the ability to copy the settings present and to paste those same settings to other channels.

Equalizer (Blue tab)

New and improved algorithm behind the Quasar introduces a four band EQ with more control. Each band provides

- Gain (range) : Gain adjustment for the selected band

- Frequency (range) : full range adjustment for the band's center.
- Width (range) : adjustment for the size (Q factor) of the band
- Peak/shelf button : determines the type of filter applied for that band
- Active button: for easy control and bypass of each individual band.
-

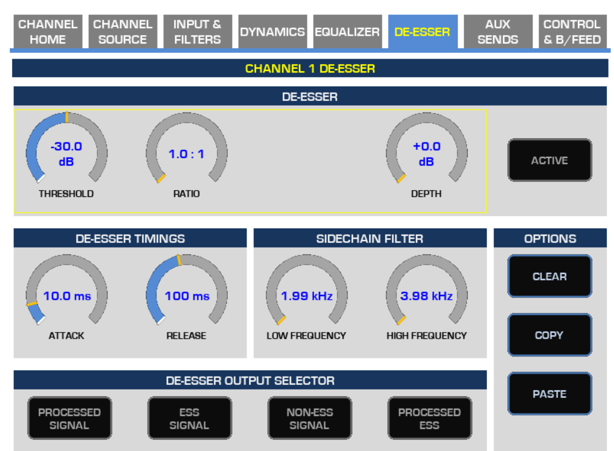
The Four bands are completely overlapping.

The options section provides the ability to copy the settings present and to paste those same settings to other channels.
To do so, press the COPY after you created your settings, select a surface different channel (by pressing its top encoder or from the MTS Channel Select menu) and press the PASTE button.

A layout button allows for grouping the control differently for alternate workflows. The controls are tied to the encoders below the screen and the layout 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.
The EQ button is the enable/disable button for the entire EQ processing chain.

De-Esser (Blue tab)

New and improved algorithm behind the Quasar introduces more control over the challenges audio engineers face with sibilance. Controls provided are



- Threshold for what level the processor is triggered
- Ratio for how aggressive the processor should be
- Depth for the amount attenuation the processor should take
- Attack and release times for the rate the processor goes in and out of processing.
- Sidechain filter controls to define the band that triggers the processor.
- Output selector buttons allow for controlling the output of the processor. This can be used to PFL the audio for what would be filtered.
- Options to copy the settings and apply to other channels.

Aux Sends (Blue tab)

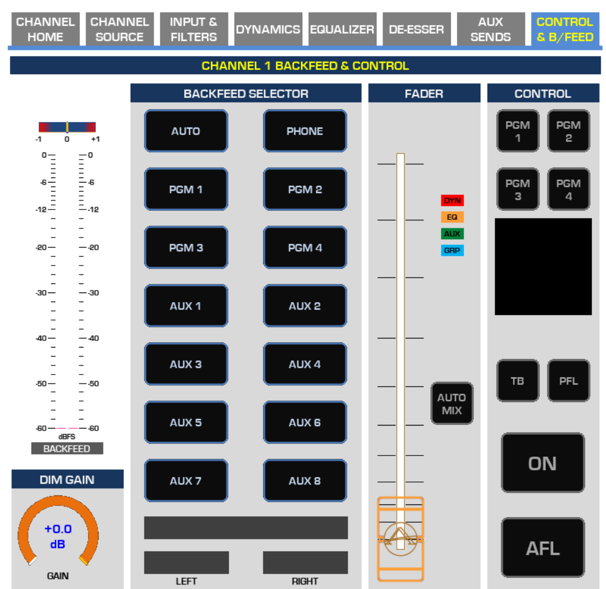
Eight AUX send mixes are available with Quasar. The controls provided are

- Gain (range) level of the send. 0dB means no additional gain or attenuation.
- ON/OFF button for enabling the send
- Post/Pre Fader button to define the signal flow of the send
- Post/Pre ON state button to define the signal flow of the send

Control & B/Feed (Blue tab)

Backfeed is the title given to return streams. Some source profiles define a backfeed to be generated and the backfeed is either a mix-minus 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). This tab provides status and controls which are

- Backfeed audio meter. As a confidence meter to verify that audio is returning to the codec or the guest
- Dim gain adjust for situations that a talkback is engaged and the main content should be dimmed to facility the needed communication
- Selector shows all the mixes available. Note there is an *AUTO* option which is a state sensitive case where when the channel is OFF, the backfeed is the PHONE mix and when the channel is ON, the backfeed is the PGM 1 mix. Phone mix is a Pre-fader and Pre-ON mix.
- Channel controls such as Fader, main mix assignments, Talkback, PFL, ON/OFF state, and AFL



What's Next

In the next chapter, we'll learn about ….

Chapter 4
Working With 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 console for use with Telos talkshow systems. But even if 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.

For details on how to use the Quasar Call Controller module to control multi-line phone systems directly from your console, please refer to the chapter entitled <u>Operations</u>, earlier in this manual.

Phone Setup Choices

There are three methods of setting up phone control with the Quasar console. These methods are referred to as:

- **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 is also known as "hybrid per fader" mode; that is, there is no switching between multiple lines on a single hybrid. Each line is presented on its own dedicated console fader. If four phone lines are desired for use, each line's hybrid is presented on a separate fader. No on-console line controller module is used.

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 Phone Controller module is used for this mode of operation.

Most Telos multi-line call systems support both of these two methods.

No Phone Control is a way of controlling third-party hybrids which do not work with Telos control protocols; instead, basic GPIO closures are used to "take" and "drop" lines.

Setting Up for EU Phone Operation

First, create a Source Profile for your fixed hybrid (as described in the section in Chapter 2 of this manual entitled "Working With Profiles"). Then:

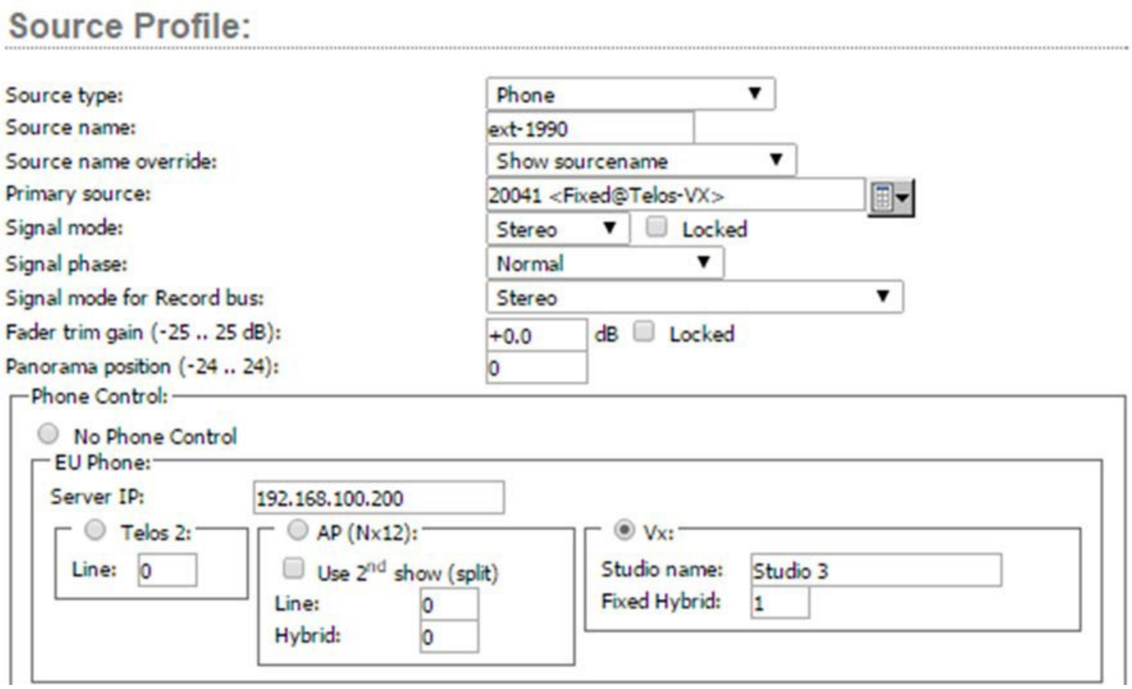
1. Select the **Phone Source** type.
2. Select the appropriate **Primary Source** using the adjacent dropdown Source Selector box to pick the de- sired Livewire channel from your Telos phone system.
3. In the **Phone Control** portion of the Source Profile, there will be an **EU Phone** section as shown in the im- age below.
   - » Enter the IP Address of the Telos phone system in the **Server IP** box.
   - » 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.

à If you're using a Telos TWO, enter the number of the line to be used in the **Line** box.

à If using an Nx system, enter the Line and Hybrid numbers in their respective boxes. Select the **Use 2nd Show** check box if split shows are enabled on your phone system.

à If using a VX system, enter the **Studio Name** as configured in your VX Engine, and the number of the **Fixed Hybrid**.



When finished, remember to save your Source Profile.

Setting Up for US Phone Operation

Using the US Phone method of operation requires the use of the Quasar Call Controller module. 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.

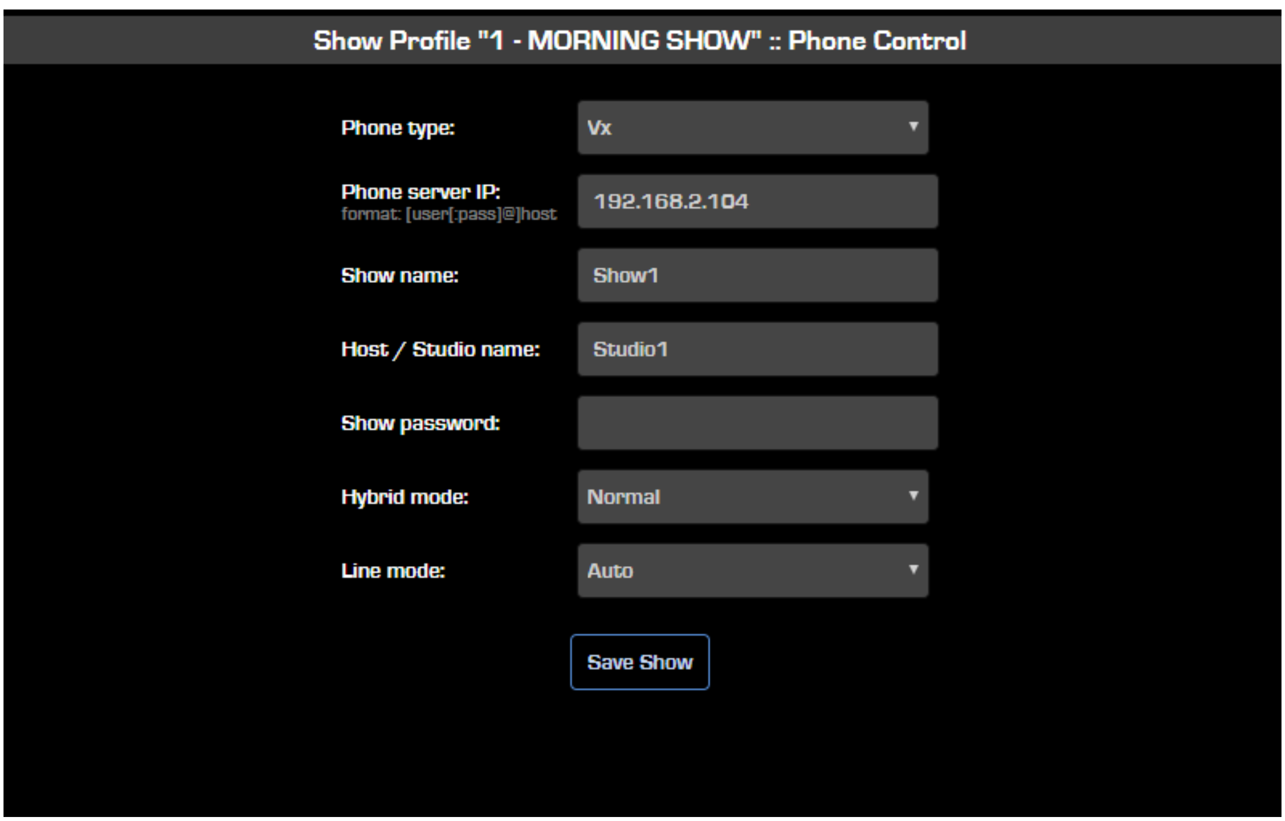
Using this mode requires two steps: defining a **Show Profile**, then defining **Source Profiles.**

Show Profile Settings For US Phone Operation

First, create a new Show Profile using the instructions found in the "Show Profiles" section of Chapter 2 of this manual.

Then, configure your new show profile. The Show Profile page of your Quasar's Web interface displays the name of the show, all the Channels (faders) of the surface, followed by several additional items, one of which is the **Phone** link. If more than one Call Controller module is installed in your console, multiple instances of this configuration option will be shown.

Select the **Phone** link to display the following screen:



The **Phone Type** is where you select the protocol used to communicate with the call 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.

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 the VX IP address here.

In cases where the non-standard credentials are used, you would add those here using the following syntax: *user:password@xxx.xxx.xxx.xxx*

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.

Note: with the VX system, this field can be left blank, permitting the changing of the show assigned with a VSet phone, or the VX Web UI interface, or a remote control connection.

If you set the show name here, upon connection the Quasar will select that show in the VX.

If you don't, the VX will keep its current show configuration regardless of the Show Profile loaded on the console.

The **Host / Studio Name field** is used differently with different systems.

- With **VX systems,** use this field to define the "Studio" name. This information allows 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.

- In legacy Telos phone systems, 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**)

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.

The Telos VX system does not use a password to protect shows, so this field will be completely ignored by a VX. You can leave it blank.

The **Reversed Hybrid** option swaps 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.

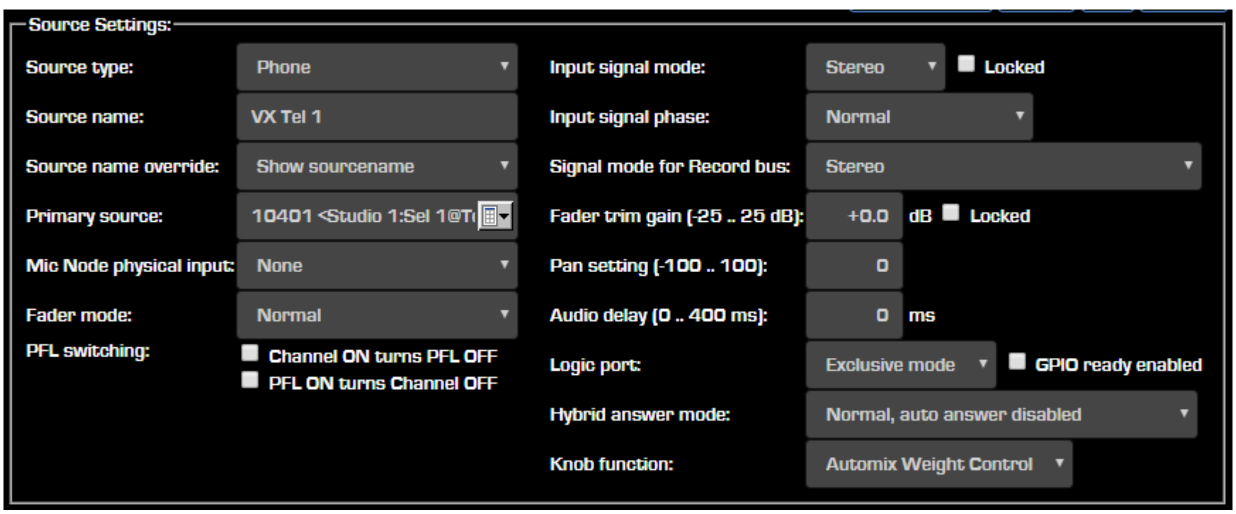
The **Mode** option is not used by Quasar. Please leave this set to Auto.

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

You'll need to create a Source Profile for your fixed hybrid (as described in the section in Chapter 2 of this manual entitled "Working With Profiles"). In legacy Telos Phone systems, you may have 1, 2, or even 4 hybrids; with VX systems, the number may be many more.

To begin:

- Select the **Phone Source** type.
- Select the appropriate **Primary Source** using the adjacent dropdown Source Selector box to pick the de- sired Livewire channel from your Telos phone system.



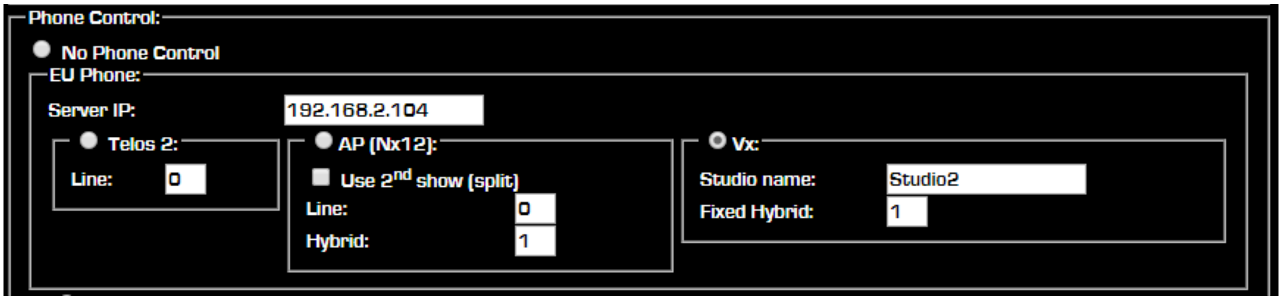
Now you have three main configuration options:

1. No Phone Control
2. EU Phone
3. US Phone Call Controller

No Phone Control is used to connect Quasar to a telephone Hybrid system which is controlled over GPIO only, and has no control over IP.

EU Phone is used in case you want European-Style control of a Telos Hybrid system which is controlled over IP.

Enter the IP Address of the Telos phone system in the Server IP box.



Select the Telos phone system you're interfacing with. You can 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** box.

- If using an Nx system, enter the Line and Hybrid numbers in their respective boxes. Select the **Use 2nd Show** check box if split shows are enabled on your phone system.

- If using a VX system, enter the **Studio Name** as configured in your VX Engine, and the number of the **Fixed Hybrid**.

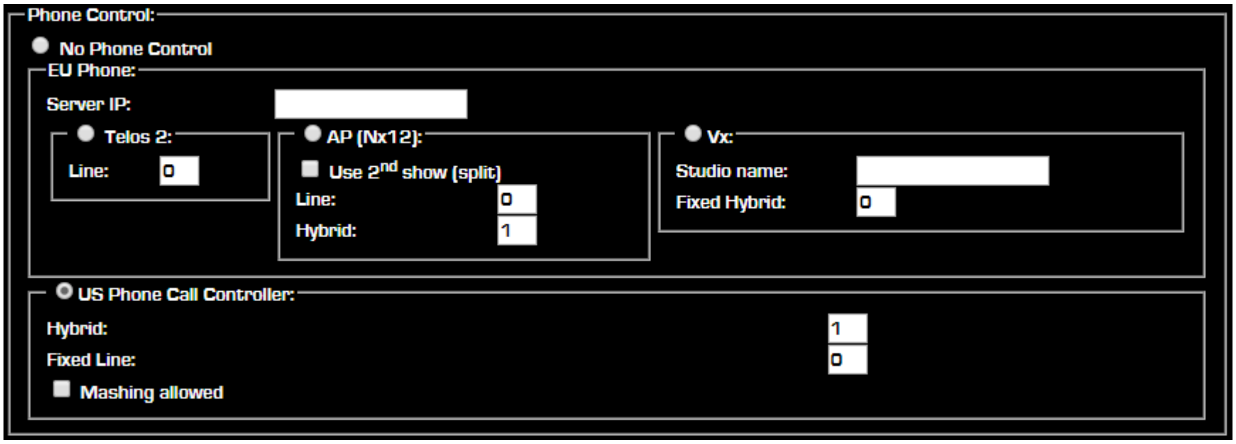
» Select the **Call Controller 1** radio button (if you have multiple Call Controllers installed, select the appropriate one).

» In the **Hybrid** box, enter the hybrid number you want this Source Profile to be associated with. This corresponds to the Channel ID number found in the Studio Information page of the VX Web UI.

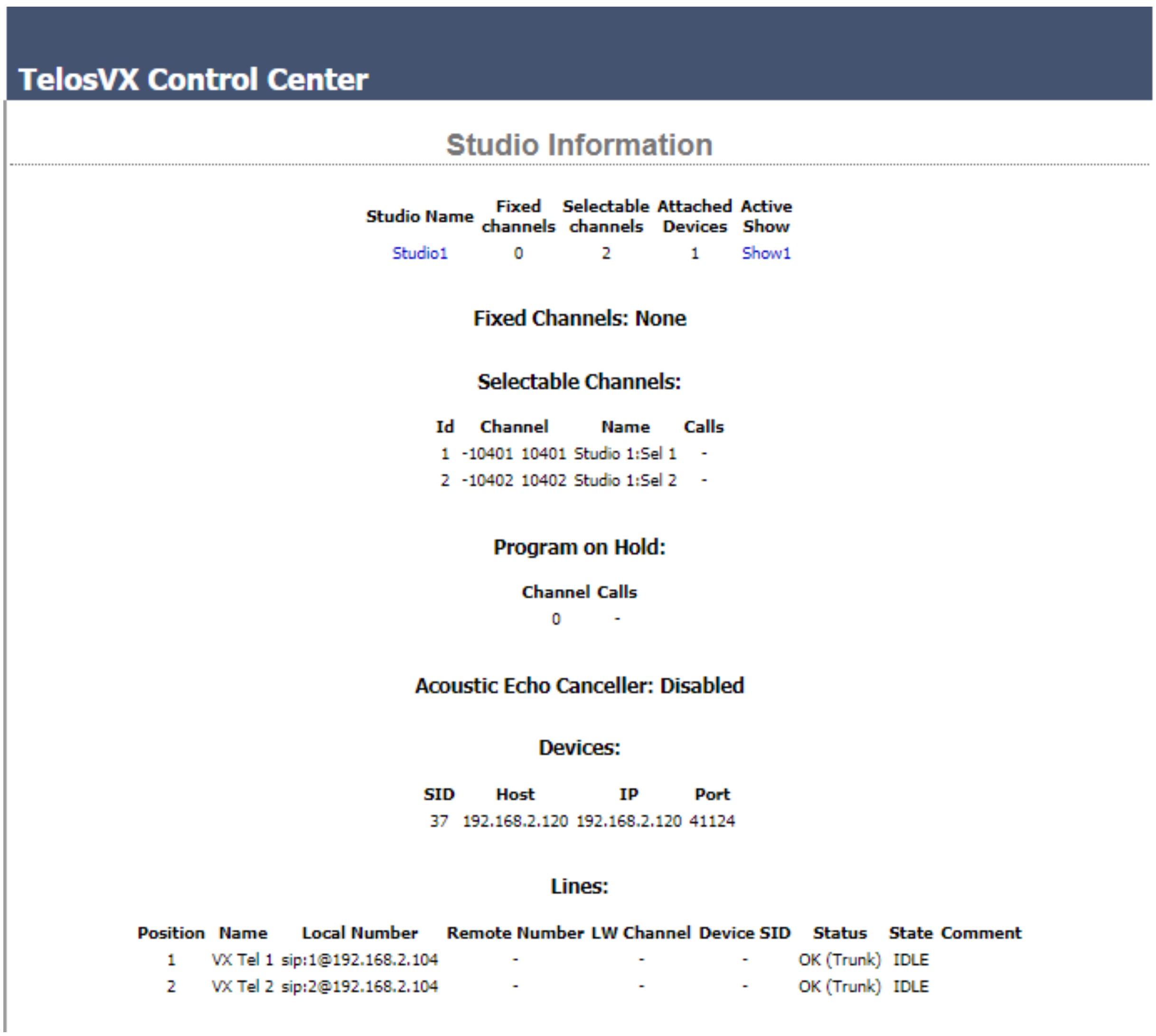
» The **Fixed Line** box is another method to do what the EU Phone method accomplishes; i.e., assign- ing a dedicated line to a dedicated fader. Most users with Call Controllers will not use this option. Leave this set to 0 if you are using the Call Controller.

» The **Mashing Allowed** checkbox permits a single hybrid to handle multiple callers at once by allow- ing the board operator to select multiple line buttons. Check (or un-check) this box as you desire.

US Phone Call Controller is used in case you want US-Style control of a Telos Hybrid system which is controlled over IP, and you're going to use the built-in soft Call Controller to do so.



On a correctly configured VX, the info page will look like this:


TelosVX Control Center

Studio Information

Studio Name	Fixed channels	Selectable channels	Attached Devices	Active Show
Studio1	0	2	1	Show1

Fixed Channels: None

Selectable Channels:

Id	Channel		Name	Calls
1	-10401	10401	Studio 1:Sel 1	-
2	-10402	10402	Studio 1:Sel 2	-

Program on Hold:

Channel	Calls
0	-

Acoustic Echo Canceller: Disabled

Devices:

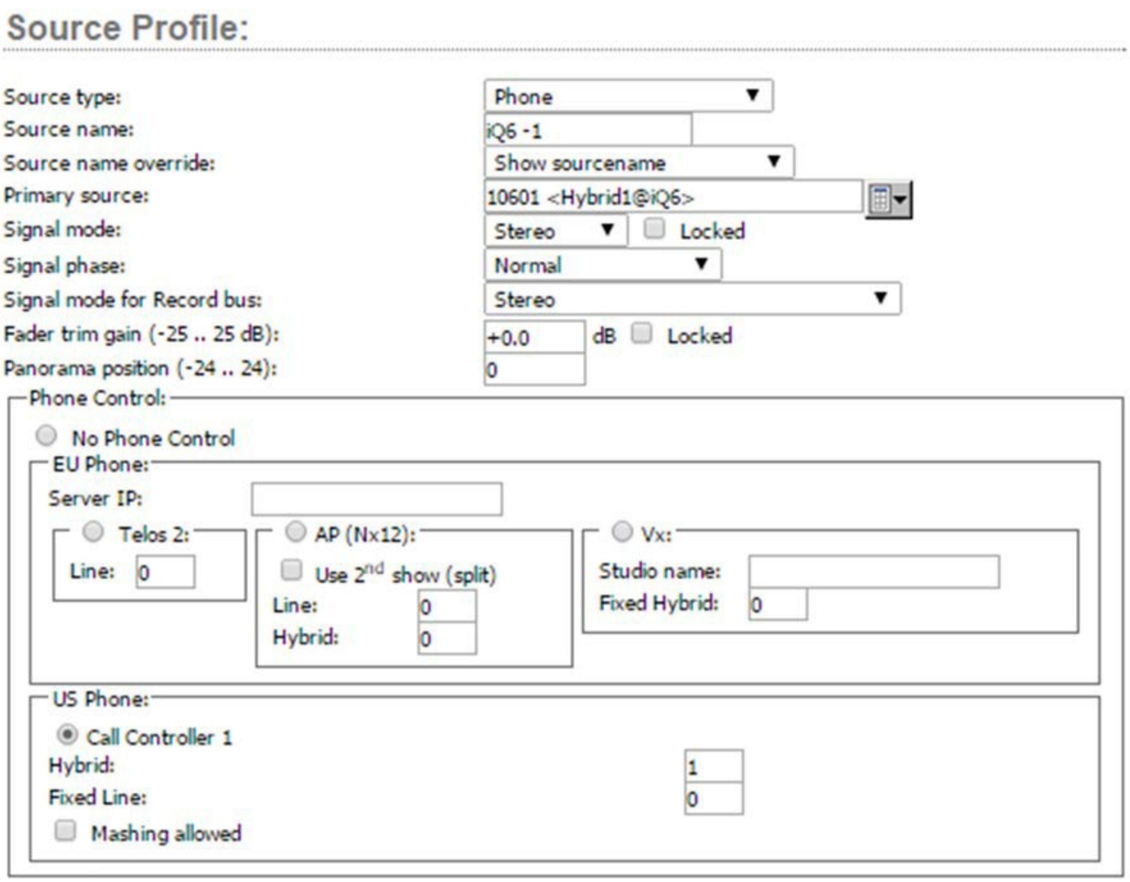
SID	Host	IP	Port
37	192.168.2.120	192.168.2.120	41124

Lines:

Position	Name	Local Number	Remote Number	LW Channel	Device SID	Status	State	Comment
1	VX Tel 1	sip:1@192.168.2.104	-	-	-	OK (Trunk)	IDLE	
2	VX Tel 2	sip:2@192.168.2.104	-	-	-	OK (Trunk)	IDLE	

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. Select the **Phone Source** type.

2. Select the appropriate **Primary Source** using the adjacent dropdown Source Selector box to pick the de- sired Livewire channel from your Telos phone system.

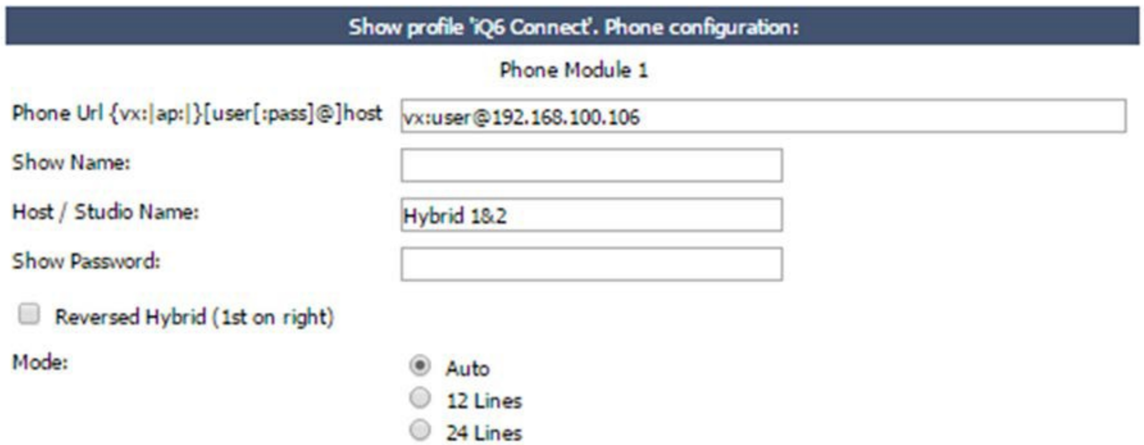
3. In the **Phone Control** portion of the Source Profile, you'll skip the EU Phone section and move to the **US**
Phone section.

   » Select the **Call Controller 1** radio button, then and enter the hybrid number in the **Hybrid** box. The iQ6 and Hx6 have two hybrids, so the option will be either 1 or 2.

   » Leave the **Fixed Line** entry at 0.

Repeat these steps to create a Source Profile for your phone system's second hybrid.

The Show Profile itself also needs configuration, to allow the console to log into the Telos Phone system as a client. To do this, click on the desired Show Profile in your Quasar's Web UI, and select the **Phone** link.



Enter the server information in the **Phone URL** field, so:
vx:username:password@xxx.xxx.xxx.xxx
where *xxx.xxx.xxx.xxx is the IP address of your phone system. (Hx6 and iQ6 systems use Telos VX control protocol, so the* **vx** *prefix in this example is correct.)*

A shortcut in Quasar assumes default values. If default values are used in your phone system, it would be possible to enter
vx:xxx.xxx.xxx.xxx
in this field, without explicitly entering the username and password. Check your Telos product to verify username and password. Default settings of the iQ6/Hx6 is a username of **user** with no password.

In the **Host/Studio Name box,** enter the value of Hybrid 1&2.

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.

Operating Options
When selecting a line directly using the Call Controller module, 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.

For instance, in the example illustrated in the screen capture above, the third Hybrid in an Nx system is configured for use with a console that has a single call controller installed. This Source Profile would be assigned to any of the standard fader strips on the console (not to the fader associated with the Call Controller).

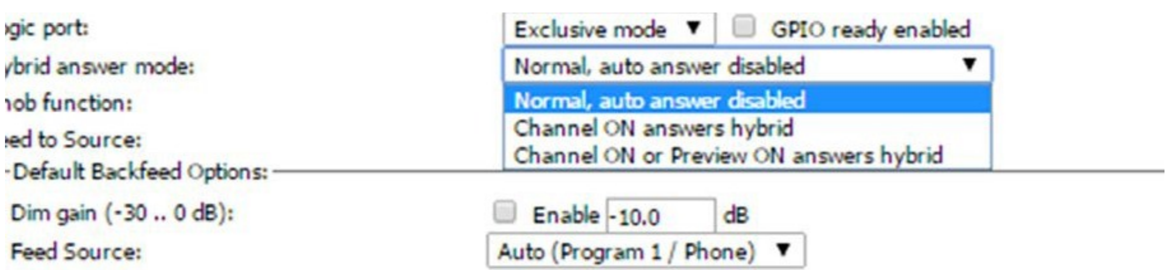
To select a line to use with this #3 hybrid, the board operator would press the **A key** on the fader strip. The words **Take Line** will then appear on the channel display; the operator can then select any line from the Call Controller and it will be assigned to this hybrid for use.

Setting up for GPIO control ("No Phone Control")

The **No Phone Control** option is the default selection when you create a new Phone Source Profile. 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.

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** dropdown box.



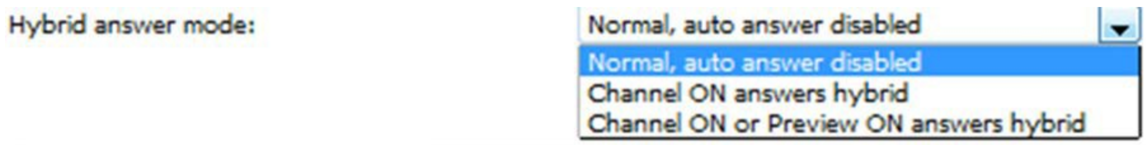
The default setting for Hybrid Answer mode is **Normal, Auto Answer Diasbled**. If your desire is to achieve pulses on pins 4 and 5 of the GPIO logic port associated with your phone hybrid, then you must choose one of the two other options. These other options will create pulses when the channel changes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

Wiring from a GPIO port that has been configured with this channel to the hybrid is the next step. Again, refer to Appendix B for the appropriate pinout charts, but here's a quick reference chart:

Telos Hx1 DE-9 Pin	Axia GPIO port DA-15 P
Pin-1	Pin-7
Pin-2	Pin-4
Pin-3	Pin-5

Additional Phone Type Source Profile Options

There are a few more phone-specific options that are available in Phone source profiles. Here's an overview:

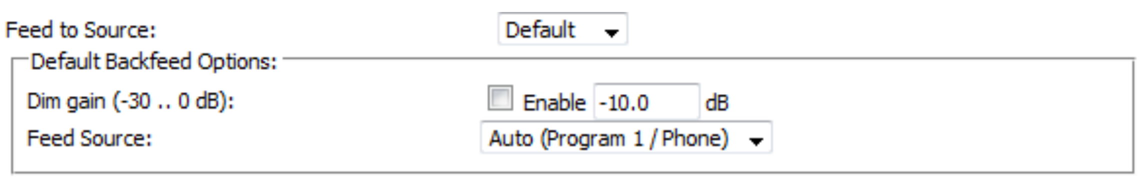


Hybrid Answer Mode. Your choices are:

» **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 Preview ON Answers Hybrid.** As above. Additionally, placing the fader channel in **Preview** answers the call; the board operator can then talk to the caller through the Operator Mic.



- **Feed To Source:** This setting controls the mix-minus that's automatically generated and fed back to the phone caller.

  » **Disabled** means no mix-minus will be generated for the caller, so they will not hear you talking to them!

  » **Default** (the option most often chosen) enables mix-minus for the caller, and enables a couple of additional options:

    à **Dim Gain** lets you trim the volume of the generated backfeed.

    à **Feed Source** defaults to **Auto**, which automatically generates a backfeed constructed of PGM-1 *minus* caller audio when the fader is On, and feeds the offline **Phone** mix to the caller when the fader is Off. This is the most commonly used mode, but you may also specify any of the other Program or Aux Send buses.

  » **Custom** enables a power-user's toolkit of mix-minus options to build custom backfeeds with select- able GPIO Monitor and Program Bus assignments based on channel On/Off/Preview state. (We'll cover details of these settings in the Advanced Configuration section of this manual.)

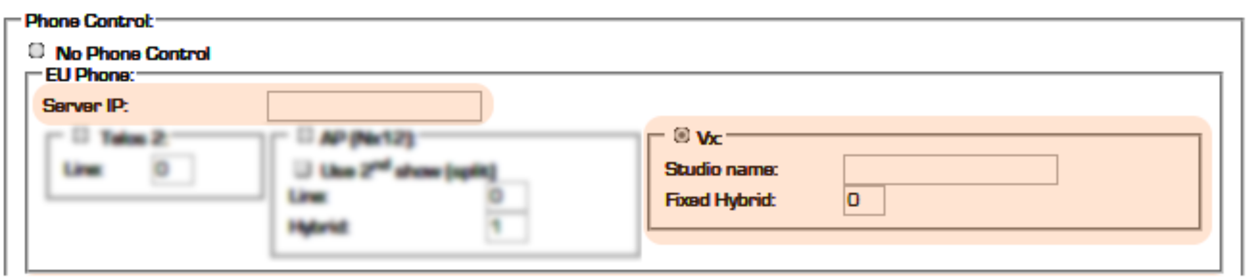
When you are done entering configurations, be sure to save your Show and Source profiles, and then load your Show Profile and newly created Phone sources to your console.

VX System integration (single line)

Single line is the method most used in European countries, wherein a single line is assigned to a single hybrid. This is also known as “hybrid per fader” mode; that is, there is no switching between multiple lines on a single hybrid. Each line is presented on its own dedicated console fader. If four phone lines are desired for use, each line’s hybrid is presented on a separate fader. No line controller interface is used.

Setting Up for Single Line Operation

1. Create a Source Profile for your fixed hybrid
2. Select **Phone** in the **Source type** field.
3. Select the appropriate **Primary Source** using the adjacent dropdown Source Selector box to pick the desired Livewire channel from your VX phone system.
4. In the **Phone Control** portion of the Source Profile, there will be an **EU Phone** section as shown in the image below.
   - » Enter the IP Address of the Telos phone system in the **Server IP** box.
   - » Select the Telos phone system you’re interfacing with. Choose the **VX** radio button and enter the appropriate settings for the device.
     - ☒ enter the **Studio Name** as configured in your VX Engine, and the number of the **Fixed Hybrid**.



Other phone integration (GPIO)

GPIO is a way of controlling third-party hybrids which do not work with Telos control protocols; instead, basic GPIO closures are used to “take” and “drop” lines.

VX System integration (multi line)

Multi line 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. With this method, you can use Show Profiles feature to instantly recall show setups that choose between different phone systems, or even different phone system configurations.

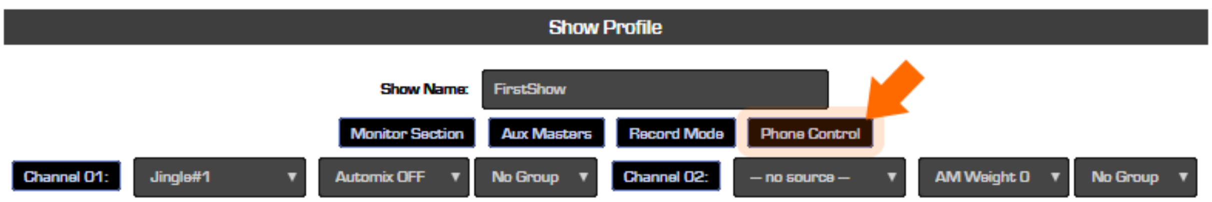
For example, one Show Profile could call up the configuration needed for a Talk format, while another might recall the configuration needed for music request lines, allowing a single studio to be the Control Room for any station in a cluster.

Using this mode requires two steps: defining a **Show Profile**, then defining **Source Profiles.**

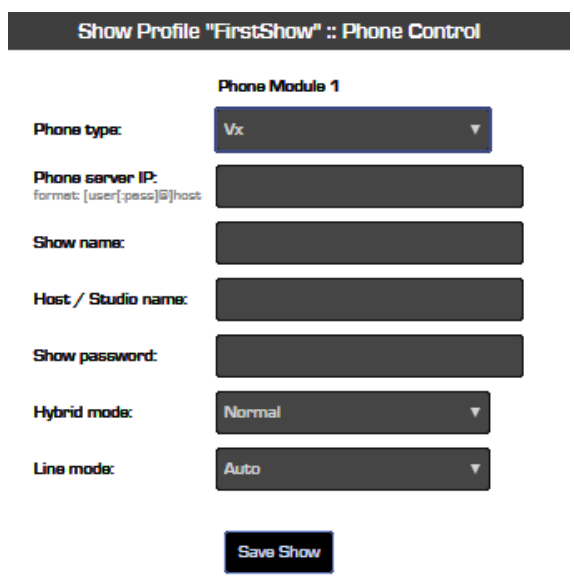
Show Profile Settings For Mulit Line Operation

Create a new Show Profile. If not familiar with how to do this, please review the instructions found in the “Show Profiles” section of Chapter 2 of the Quasar manual.

The Show Profile configure page displays the name of the show, all the Channels (faders) of the surface, above the channels are several additional items, one of which is the **Phone Control** link.



Select the **Phone Cotnrol** link to display the following screen:



The **Phone Type** is where you select the protocol used to communicate with the call 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.

The **Phone Server IP** is where you identify the IP address of the Telos product that is managing the calls. In cases where the non-default user name of password is to be used, you would add those in the text box. An example is MyUser:MyPass@192.168.2.66 .

The **Show Name field** is used to identify a phone system configuration when using a Telos VX system. These names are defined in your phone system’s setup. Enter the configuration name you wish to log into.

Note: 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.

The **Host / Studio Name field** is used differently with different systems.

- With **VX systems,** use this field to define the “Studio” name. This information allows the surface to interface with the appropriate configuration as defined in the VX.

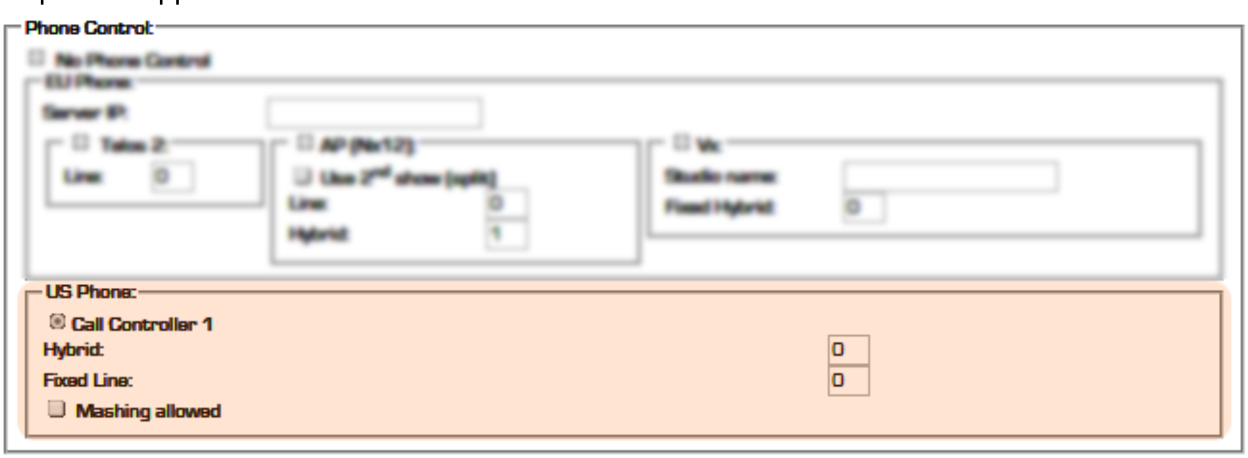
The **Show Password** field is used to permit access to the Call system that has password protection at show levels. If you have a password for the phone system Show you want to use, enter it here.

The **Hybrid Mode** option 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.

The **Line Mode** option defines the operation of the two Call Controller columns for line selection. The default is **8 lines,** which provides control for Hybrid 1 and Hybrid 2. The **16 lines** option is available for high call requirements and, and when setting each button as a different line.
Once you’ve entered the information required for your phone system, save your changes and proceed to Source Profile configuration.

Source Profile Settings for Multi Line Operation

1. Create a source profile
2. Select **Phone** in the **Source type** field.
3. Select the appropriate **Primary Source** using the adjacent dropdown Source Selector box to pick the desired Livewire channel from your VX phone system.
4. In the **Phone Control** portion of the Source Profile, you’ll skip the EU Phone section and move to the **US** Phone section.
   - » Select the **Call Controller 1** radio button
   - » In the **Hybrid** box, enter the hybrid number you want this Source Profile to be associated with. This will be a 1 or 2.
   - » The **Fixed Line** box is another method to do what the EU Phone method accomplishes; eg. assigning a dedicated line to a dedicated fader. Most will not use this option so this can be left as 0.
   - » 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.
5. Typically these steps are done twice for the two hybrid interfaces
6. You should consider editing the show profile and define which channel these two source profiles appear.



Chapter 5
Practical Help Guides

In many cases, our customers just want to know how to do one thing and why bother reading through an entire manual to figure that out, this chapter is for you. This chapter is filled with single help guides of common setups. If we forgot one, let us know!

Trigger an ON AIR Light

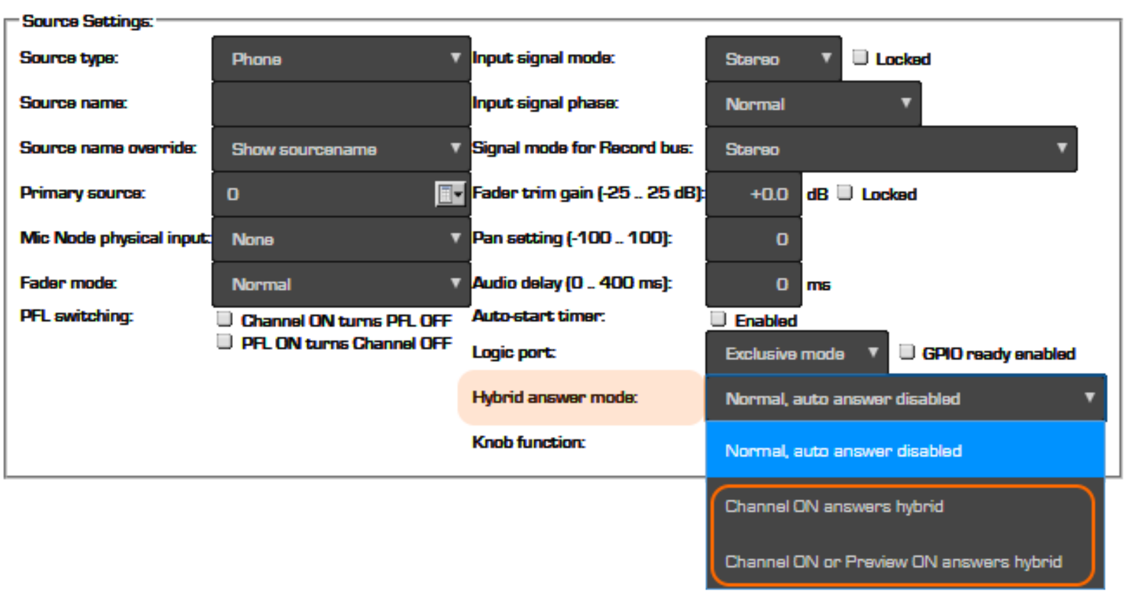
{page break here}

Setting up for GPIO control

The **GPIO** option is the default selection when you create a new Phone Source Profile. 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.

Please refer to the GPIO Telephone Hybrid Logic chart for the appropriate pinouts.

When configuring a Source Profile for this operational mode, the salient field is the **Hybrid answer mode** dropdown box.



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 Preview state.

- **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 Preview ON answers hybrid.** As above. Additionally, placing the fader channel in **Preview** 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 a Axia GPIO port. :

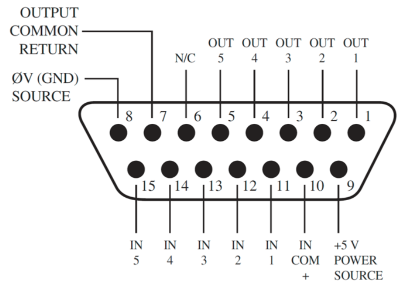
Telos Hx1 DE-9 Pin	Axia GPIO port DA-15
Pin-1	Pin-7
Pin-2	Pin-4
Pin-3	Pin-5

The following table applies to a GPIO port on an xNode or similar device.

GPIO Telephone Hybrid Logic

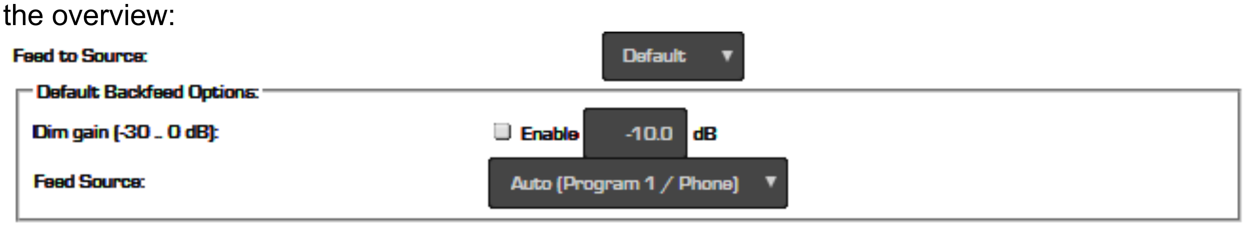
Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF
PREVIEW Command	13	Active Low Input	Turns preview ON
RESET Command	14	Active Low Input	Turns channel off while not sending a STOP pulse
READY Command	15	Active Low Input	Illuminates 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
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW 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		



Additional Phone Type Source Profile Options

There are a few more phone-specific options that are available in Phone source profiles. Here’s the overview:



- **Feed To Source:** This setting controls the mix-minus that’s automatically generated and fed back to the phone caller.

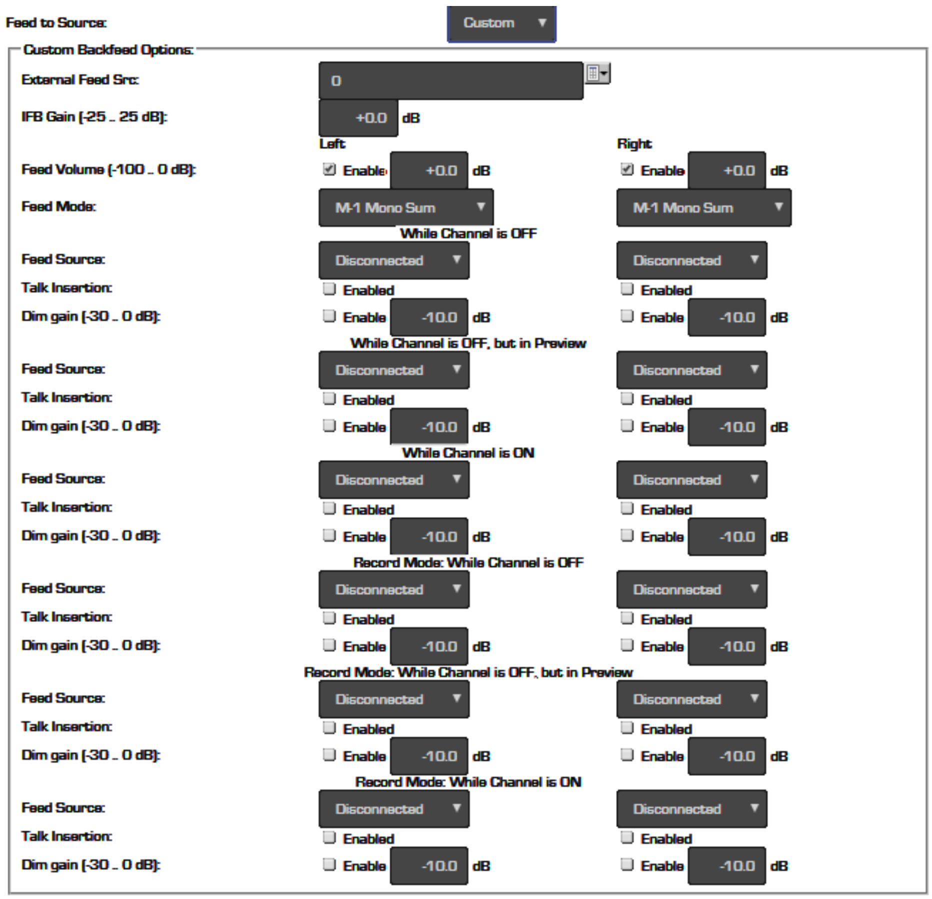
» **Disabled** means no mix-minus will be generated for the caller, so they will not hear you talking to them!

» **Default** (the option most often chosen) enables mix-minus for the caller, and enables a couple of additional options:

☒ **Dim Gain** lets you trim the volume of the generated backfeed.

☒ **Feed Source** defaults to **Auto**, which automatically generates a backfeed constructed of PGM-1 *minus* caller audio when the fader is On, and feeds the offline **Phone** mix to the caller when the fader is Off. This is the most commonly used mode, but you may also specify any of the other Program or Aux Send mixes.

» **Custom** enables a power-user’s toolkit of mix-minus options to build custom backfeeds with selectable Monitor and Program mix assignments based on channel On/Off/Preview state.


Feed to Source:
Custom
Custom Backfeed Options:
External Feed Src:
0
IFB Gain [-25 .. 25 dB]:
+0.0 dB
Left
Right
Feed Volume [-100 .. 0 dB]:
Enable +0.0 dB
Enable +0.0 dB
Feed Mode:
M-1 Mono Sum
M-1 Mono Sum
While Channel is OFF
Feed Source:
Disconnected
Disconnected
Talk Insertion:
Enabled
Enabled
Dim gain [-30 .. 0 dB]:
Enable -10.0 dB
Enable -10.0 dB
While Channel is OFF, but in Preview
Feed Source:
Disconnected
Disconnected
Talk Insertion:
Enabled
Enabled
Dim gain [-30 .. 0 dB]:
Enable -10.0 dB
Enable -10.0 dB
While Channel is ON
Feed Source:
Disconnected
Disconnected
Talk Insertion:
Enabled
Enabled
Dim gain [-30 .. 0 dB]:
Enable -10.0 dB
Enable -10.0 dB
Record Mode: While Channel is OFF
Feed Source:
Disconnected
Disconnected
Talk Insertion:
Enabled
Enabled
Dim gain [-30 .. 0 dB]:
Enable -10.0 dB
Enable -10.0 dB
Record Mode: While Channel is OFF, but in Preview
Feed Source:
Disconnected
Disconnected
Talk Insertion:
Enabled
Enabled
Dim gain [-30 .. 0 dB]:
Enable -10.0 dB
Enable -10.0 dB
Record Mode: While Channel is ON
Feed Source:
Disconnected
Disconnected
Talk Insertion:
Enabled
Enabled
Dim gain [-30 .. 0 dB]:
Enable -10.0 dB
Enable -10.0 dB

Trigger an ON AIR light

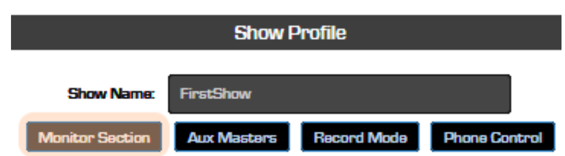
An Air light traditionally is illuminated when a microphone is ON. Additionally, the console has logic built in which will mute the monitor speakers if a localized microphone is ON. The Control Room (CR) speakers have GPIO logic built into the product. This logic is as follows.

GPIO Control Room Monitor Logic

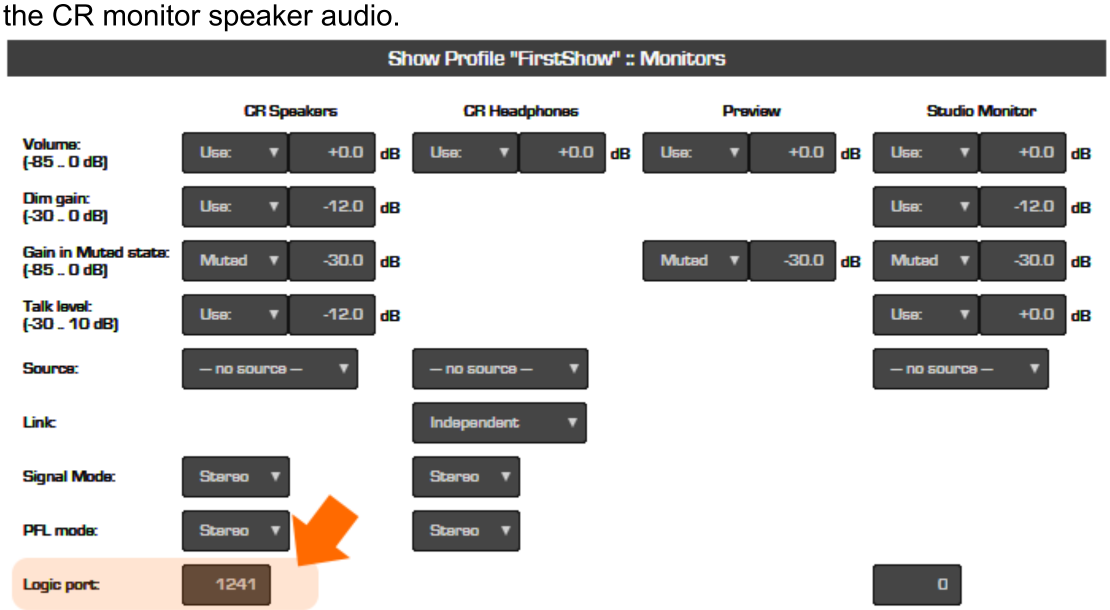
Name	Pin	Type	Notes
INPUTS			
MUTE CR Command	11	Active Low Input	Mutes CR monitors and Pre-view speakers
DIM CR Command	12	Active Low Input	Allows external dimming of CR monitor speakers.
Enable EXT PREVIEW Com-mand	13	Active Low Input	Feeds External Audio Input to PREVIEW
TALK TO EXT Command	14	Active 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
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW 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 as follows;

1. Within a show profile configuration editor, select Monitor Section.



2. Within the specific profile's Monitors section, locate the Logic port for the CR Speakers. Here you should enter a unique Livewire channel. A common practice is to enter the same number that is used to define the CR monitor speaker audio.

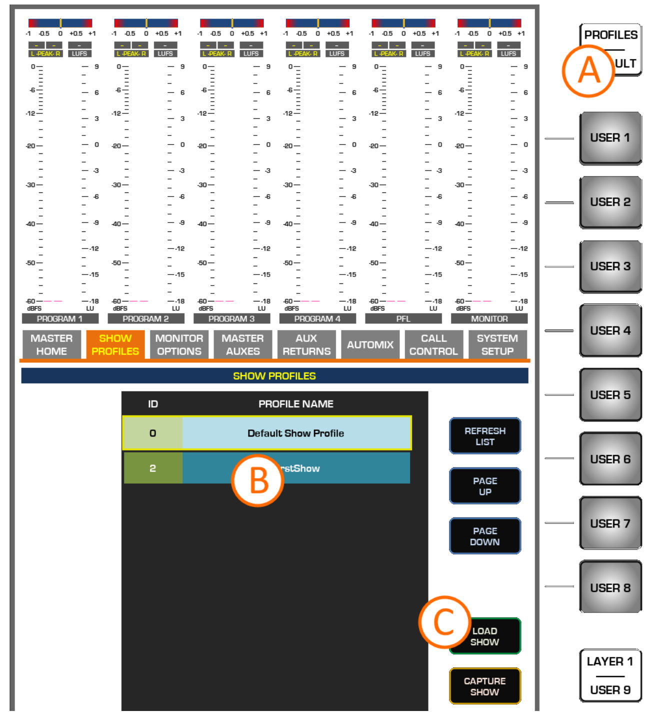


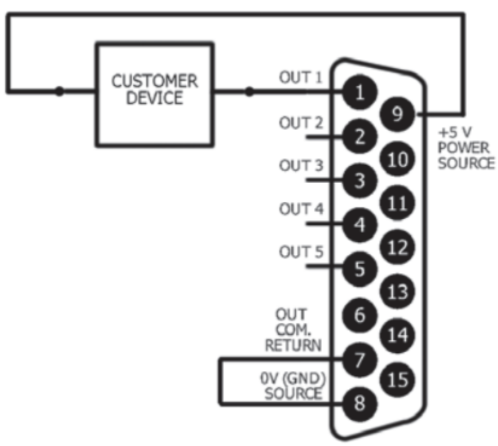
3. Once defined, save the show profile and load the profile to the surface. When a microphone is turned on, the CR monitor speakers will mute and the logic state will be reported into the network. The next step is to assign a GPIO port in the network to respond to this message.

4. Log into an xNode that has GPIO functionality.

5. Assign one of the ports to the same channel that was assigned earlier. In the example here, 1241 was used.

6. In the case of electrically isolated circuit, you would wire the GPIO port as follows





Chapter 4 (Fusion)

VMix and VMode

There's a lot of DSP processing horsepower in Axia mixing engines. Rather than let that go to waste, we've used it to power two tools that many broadcasters have found to be indispensible: VMix and VMode.

VMix (short for Virtual Mixer) provides up to 16 channels of "virtual mixing", which can be used to pre-mix up to 5 audio sources each for presentation on a single physical fader (or software fader, if Axia SoftSurface soft- console software is used). VMix works completely independently of the Fusion 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.

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.

Virtual Mixing with VMix

What's it all about?

In addition to the regular mixing capabilities of your Fusion console, there is an 80-input "virtual" mixer accessed using the console's HTTP interface (or with PathfinderPC routing tools).

This mixer consists of 80 stereo input channels, a direct output for each channel, 16 submixer outputs, and one master out. The 80 channels are divided equally among the 16 subgroups, providing 5 stereo channels feeding each subgroup mixer.

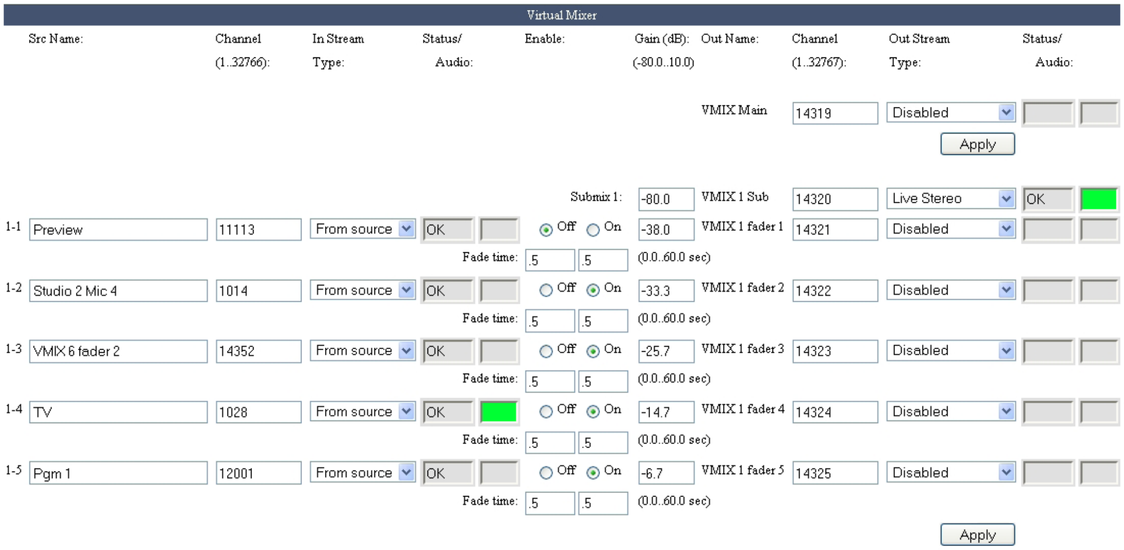
The various VMix outputs described above are sources that can feed your Livewire network and they can be manipulated in the same manner as any other audio source. A VMix source can be applied to a console fader, assigned to an audio node destination, or monitored by Pathfinder.

To understand this concept, think of VMix as a standalone piece of hardware. If you visualize wiring an external line mixer to your network, Livewire audio sources would be connected to the mixer inputs; your VMix outputs then become mixdowns that you can use anywhere else on the network, just like any other audio source.

To access VMix setup, open a browser on a computer connected to your Axia network and browse to the IP Address assigned to your PowerStation or StudioEngine or, if you have Axia iProbe, open the UI page by right- clicking on the device's icon in the left pane. Then, choose the **V-Mixer and V-Mode** item from the navigation bar.

The following image is an excerpt of the VMix/VMode setup screen. It shows the VMix Main and Sub-Mixer 1 controls. These sub-mix controls are duplicated for all 16 VMix submixers.

Note that adjustments made to VMix take effect as soon as you apply them — so changes saved "on the fly" will affect your output streams immediately.



VMix Main Controls

The **VMix Main** output provides a summed mix of all 16 submixes. 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. There are only a few settings for this control:

- **Out Name.** You can enter a friendly name for the VMix Main output, which will be displayed as a source name on your Axia network.
- **Out Stream Type.** Choose from **Live Stereo, Standard Stereo,** or leave it **Disabled** if no Main stream is desired.
- The **Status** window will normally display "OK" when the stream is enabled.
- The **Audio** box will be green when audio is present.

VMix Submixer Controls

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.

First, you'll want to enable the Submixer you're working with. At the top of the section, you'll see (in this case)
Submix 1 displayed on the screen. Next to that are controls for:

- **Gain.** Set this at whatever output level you want the submix output stream volume to be.
- **Channel.** All Livewire audio streams are assigned a Channel number; put a unique value in this box.
- **Out Stream Type.** Choose from **Live Stereo,** or **Standard Stereo. Disabled** turns off the submixer.Each **VMix Sub** submixer input includes an on/off setting, a gain setting, and automatic fade-up/fade-down time parameters. Using one is easy; let's walk through the

steps:

- In the **Src Name** box, enter the name of the source you'll be assigning to the input.

- In the **Channel** box, enter the unique Livewire channel number of your audio source.

- **In Stream Type** will normally be set to "From Source", meaning that the source itself (a mic, CD player, etc.) will be providing the audio. However, you can select "To Source" to use the source's automatically generated Backfeed (mix-minus audio) as in 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.

- Select the **Enable** box next to the input to turn the Submixer input on.

- Leave the **Fade Time** boxes set at their default.

  The **Fade Time** function won't be used in normal operation, but can be used to create cross fades between sources when Pathfinder is dynamically making changes to the VMix.

  If, say, 1.0 is entered in the first box, 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.

  The second box works the same way, but controls the ON to OFF fade time.

- In the **Gain** box, enter the setting, in dB, that you want for the Submixer input. Each input has its own individual Gain setting.

That's all you need to do to configure a VMix Submixer input stream. You can configure up to 5 input streams per VMix Submixer.

At the bottom of each of the submix sections is an **Apply** button. Any changes you make will be saved when this button is pressed. Be sure to save the changes for each submixer as they are configured.

VMIx Submixer Advanced Options

There are some additional options provided for advanced users.

In most cases only the Submix output itself needs a unique channel number, but if you so desire, each VMix input can also be sent back to the network as a unique source, post the ON/OFF and gain stage of the VMix. Some users find this useful for constructing "cascaded" mix channels to suit unique situations. To do so, enter values for these controls:

- **Out Name**: The name you give the post-on/off submix channel to send back to the Axia network.
- **Channel**: the channel number you assign to the post-on/off submix channel to send back to the Axia net- work.

- **Out Stream Type.** Choose from **Live Stereo, Standard Stereo** to enable the direct output stream for this VMix “fader”. **Disabled** means no stream will be generated.

Be sure to click **Apply** after you’ve finished making any options changes.

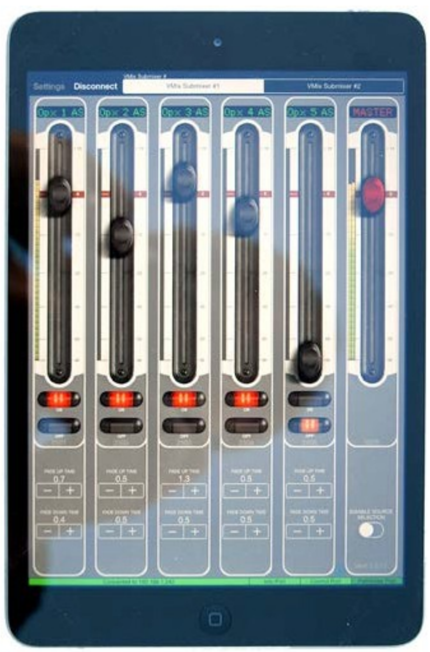
Some VMix Examples

Now that you know how to enable and set up VMix submixers, what might you do with it? Here are some examples.

A Mix of Sources to Monitor

Some facilities may need to monitor one or more sources in addition to program audio, like the “squawk channel” some satellite feed providers use to relay announcements.

If you wanted to monitor this “squawk” audio on your Preview speaker without taking up a fader assignment, you could create a mix of the “squawk” source and the Preview mix from the engine. The VMix Sub Channel would be the audio source that you would route to the Preview speaker.



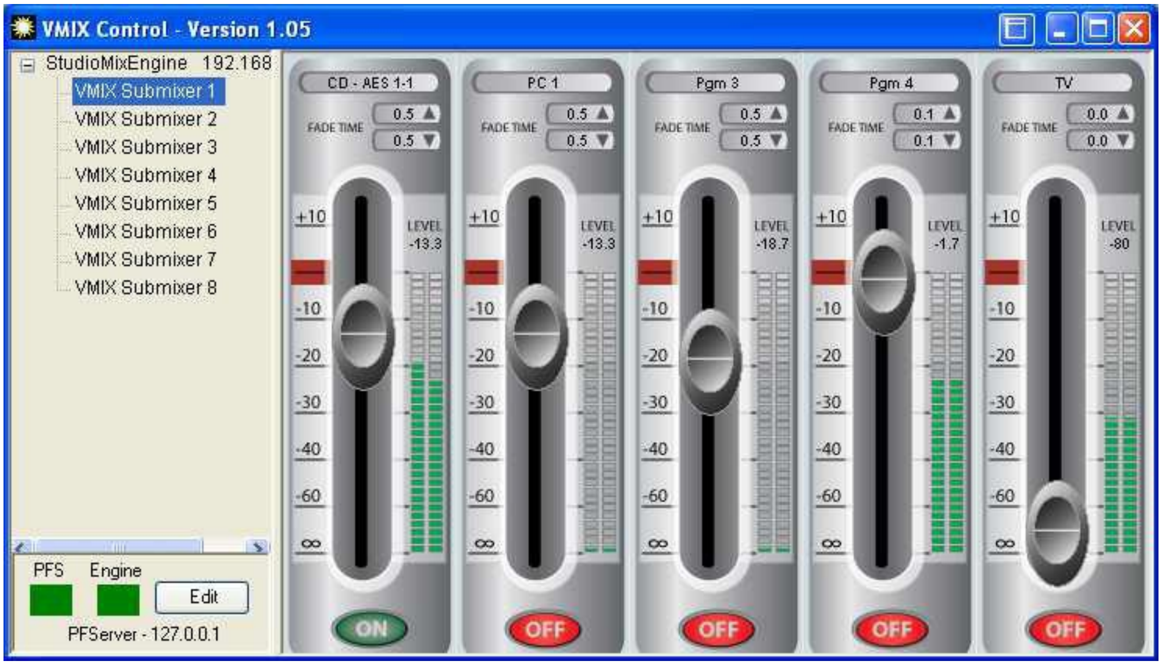
Virtual VMix Control

Using third-party products like VMix from BSI, or by taking direct control of a console’s mixing engine with Axia SmartSurface software, VMix channels can be assigned to a “virtual control surface”, giving studio talent or a producer direct control of VMix without the need for a physical console.

Combining VMix with Pathfinder Routing Control

Axia Pathfinder routing tools can be used to “control” VMix in two different modes.

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 and VMix could duplicate the function provided by other products that are controlling audio switching in many radio facilities. (Refer to documentation on Pathfinder for further information.)



Second, there's Pathfinder's VMix Control feature. This is a software fader control option that is provided with the PRO versions of Pathfinder. VMix Control brings the operation of VMIX out of the background and provides a graphical user interface with software faders, as shown here.

There are several other ways that Pathfinder can be used for background control of VMix. VMix functions can be used both as qualifiers and actions in routing salvos. This means that a designer can select GPIO triggers, time based events, user button pushes, 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 stacking events qualifiers.

Finally, Pathfinder's Software Authority protocol translator includes 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.

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 jock 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 and hold" function to allow the jock to monitor the sources momentarily. (This example is only possible with Pathfinder control of VMix.)

Manipulating Streams With VMode

Each of the 16 VMode instances supports audio streams with up to 8 channels at input and up to 8 channels at 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.

The VMode view provides status of the current configuration.

V-Mode

Input Selector	Src Name:	Address	Status/Audio	Audio Mode:	Out Name:	Address	Status/Audio	
None				Pass stereo	V-Mode 1	401		Configure...
None				Pass stereo	V-Mode 2	402		Configure...
None				Pass stereo	V-Mode 3	403		Configure...
None				Pass stereo	V-Mode 4	404		Configure...
None				Pass stereo	V-Mode 5	405		Configure...
None				Pass stereo	V-Mode 6	406		Configure...
None				Pass stereo	V-Mode 7	407		Configure...
None				Pass stereo	V-Mode 8	408		Configure...
None				Pass stereo	V-Mode 9	409		Configure...
None				Pass stereo	V-Mode 10	410		Configure...
None				Pass stereo	V-Mode 11	411		Configure...
None				Pass stereo	V-Mode 12	412		Configure...
None				Pass stereo	V-Mode 13	413		Configure...
None				Pass stereo	V-Mode 14	414		Configure...
None				Pass stereo	V-Mode 15	415		Configure...
None				Pass stereo	V-Mode 16	416		Configure...

To change the configuration, select the far right “Configure…” button.

Input:

Input (Destination)	
Input selector:	None ▼
Source name:	
Address:	
Stream type:	L24 2ch (Livewire From source) ▼
Using the "Address" field:	
Blank	Disconnected.
Integer number	Channel number of Livewire multicast stream, range 1 to 32766. Standard RTP port number 5004 will be used.
<Multicast IP>	Address of AES67 multicast stream, range 239.0.0.0 to 239.255.255.255. Standard RTP port number 5004 will be used.
[0.0.0.0]:<port>	Unicast stream receive port number. A stream from any sender will be accepted.
<Unicast IP>:<port>	Unicast stream sender, and the receive port number. Only a stream from the indicated IP address will be accepted.

The **Input selector** is used to select any source native to the Engine or select an “EXTERNAL” source from the network.

Source name is a text field used to document the input.

Address is used to define the Livewire channel number of an “EXTERNAL” source or the address of an AES67 stream.

Stream type is used to define the source. With AES67, the stream could be a linear 24 bit up to 8 channels or linear 16 bit. The Livewire type sources are identified. Make sure you select the correct type.

Mode:

Audio Channel Routing	
Mode:	Pass stereo ▼

Manipulation within the matrix is referred to as routing and will be configured by a mode selection. 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

Mode option “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

Mode option “Pass stereo” passes only the stereo part of the matrix transparently, and blocks the multichannel part.

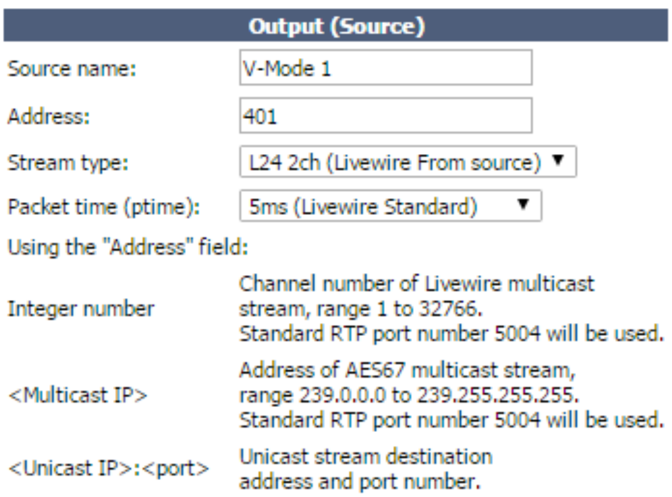
“Transpose ...” options move two channels between the stereo part of the matrix and channels 1 and 2 of the multi-channel part.

“Upmix ...” and “Downmix ...” options mix, copy, and/or apply gains to input channels, to obtain output channels. Generally, upmixing is a process that creates a bigger channel count at output from a smaller channel count at input (for example, multi-channel from stereo), and downmixing is the opposite (stereo from multi-channel).

“Split …” options create a mono sum from a stereo input, apply the specified attenuation, and send it to either Left or Right output channel as indicated in the name of the option, while sending no signal to the other.

“Combine ...” options are special in that they take input from two input streams (designated A and B) and output to one stream. The routing is indicated in the name of the option, *Combine A, B*, where the “side” identified with the *A* field results as the *Left* side of the output and *B* field in the *Right* side . For example “Combine Right,Left” takes the stereo *Right side* from the *A stream* input into the *Left side* of the output channel, and Stereo *Left side* from the *B stream* input to the *Right side* of output.

Output



:

The output of the matrix will result into a network source. You can define this source to be a Livewire source type or one of the many AES67 options.

Source name is used to identify the source in the network. Livewire sources are advertised and this name is used in the information propagation. AES67 does not define a required method for how sources are advertised. Once this is defined, this name may be utilized.

Address is used to define the Livewire channel number or an AES67 address.

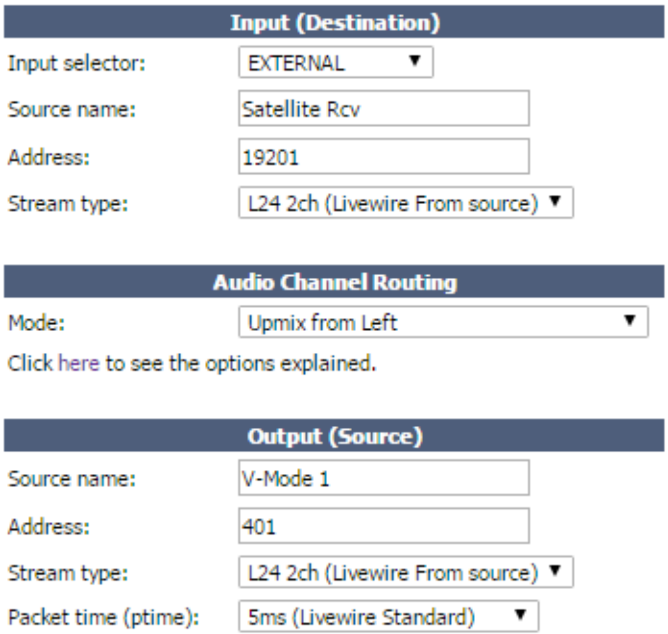
Stream type defines the type of stream to be generated from the matrix output. Similar to the input type field, there are many options available because of AES67 and the type used in Livewire are identified.

Packet time defines the latency and bandwidth consumption of the stream. Livewire has historically given this field a user friendly name such as Live or Standard stereo. The time value is the amount of audio encapsulated within the packet. The shorter time period permits the transmission of audio more frequently with the result of high band- width utilization and lower delay at receiver. The longer period results in better packet utilization and thus a lower stream bandwidth with a slower reception at receiver. The ptime used by Livewire devices are identified.

Some VMode examples

Create a Mono Stream From One Side of a Stereo Channel

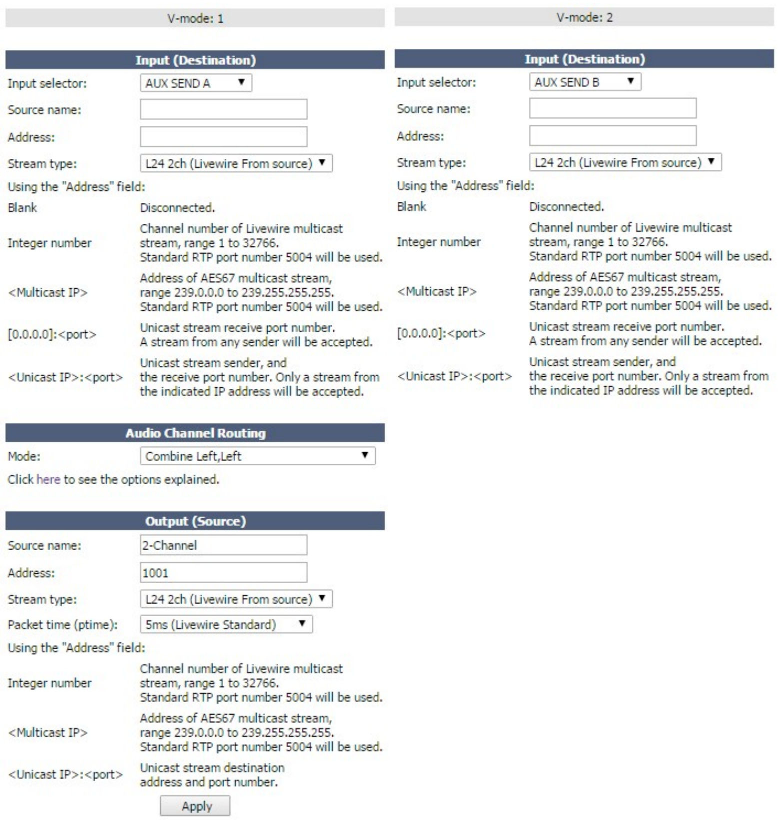
Sometimes, the program content that is fed on a satellite downlink is received on only the Left channel while another content or information on the Right channel that you don't want to air. Using VMode, you can split the Left and Right sides and create a new source using just the channel you want. Here's how:



- Select EXTERNAL
- Provide a useful name
- Define the source stream from the network you wish to input. The example assumes Livewire channel 19201.
- Define the type, the example assumes a Livewire stereo source.
- "Upmix from Left" mode
- Define the stream address or Livewire channel number.
- Define the type, again we assume Livewire 2-channel.
- Define the ptime, in the example we define a Livewire Standard Stereo packet.

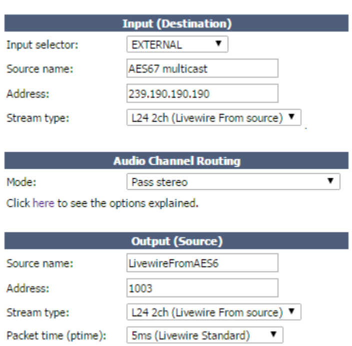
Create a Split Record Feed from Multiple Sources

You want to create a two channel recording with independant content and gain control for each channel different from Program 1 of the Quasar console. This single stream will be recorded on a stereo recorder. You can construct a custom VMode stream to satisfy this requirement, like so:



The resulting Livewire 2-channel output would have the Left side of AUX A on the Left and Left side of AUX B on the Right.

Create a stereo Livewire stream from a 2-channel AES67 stream.



What's Next

Telephones are an important part of many stations' on-air programming. Since Axia Audio is a part of Telos, Quasar features tight interoperation with Telos multi-line phone systems. In the next chapter, we'll discuss how to set up your console and phone system to work together seamlessly.

Appendix A (Fusion)

Advanced Configuration Reference

This appendix lists the advanced options available when setting up Show Profiles, and the “magic key” combinations available from the console surface.

Appendix B (Fusion)

Configuring GPIO

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 Appendix 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 PowerStation mix engines feature 4 built-in GPIO connections; Fusion consoles using StudioEngine mix engines must be paired with xNode GPIO Nodes. 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.

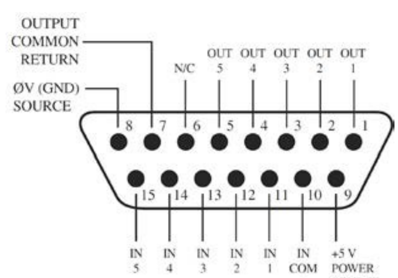
Axia GPIO ports can deliver unique command sets for the following types of devices:

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.

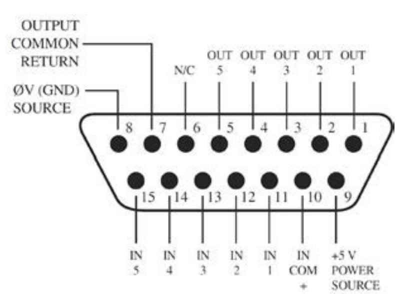
GPIO Operator's Microphone Logic

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 PREVIEWED SOURCE) Command	15	Active Low Input	Activates the TALK button on every source currently in preview 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 PREVIEWED SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when TALK to PREVIEWED 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 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.
Input Common	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



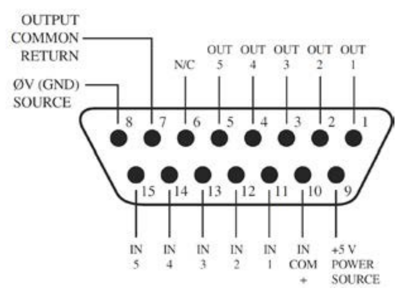
GPIO Control Room Guest Microphone Logic

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 PVW 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 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		



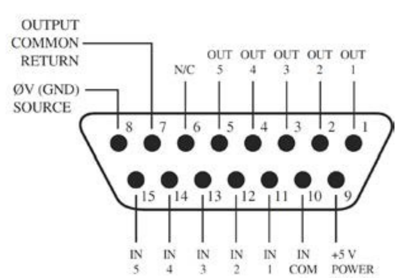
GPIO Producer's Microphone Logic

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) 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 PREVIEWED SOURCE) Command	15	Active Low Input	Activates the TALK button on every source currently in Preview 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 PREVIEWED SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when TALK to PREVIEWED 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.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



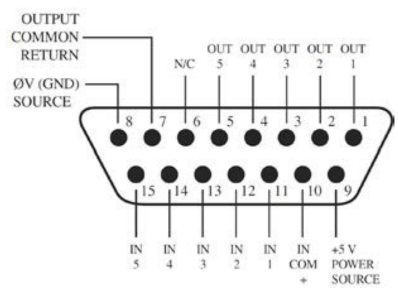
GPIO Line Input Logic

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
PREVIEW Command	13	Active Low Input	Turns preview 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
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW 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		



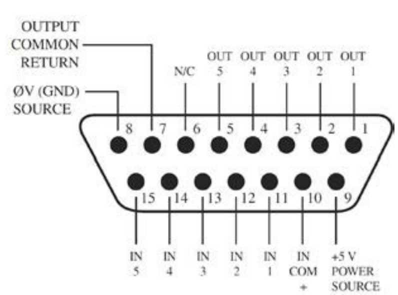
GPIO Codec Logic

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 PVW speakers
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to SOURCE) Command	15	Active Low Input	Allows an external button 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		



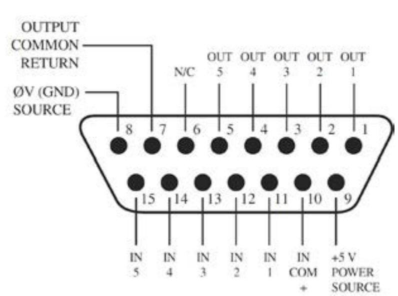
GPIO Telephone Hybrid Logic

Name	**Pin**	**Type**	**Notes**
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF
PREVIEW Command	13	Active Low Input	Turns preview ON
RESET Command	14	Active Low Input	Turns channel off while not sending a STOP pulse
READY Command	15	Active Low Input	Illuminates OFF lamp to indicate source's readiness
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
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW 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		



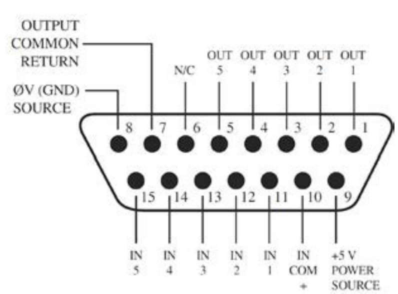
GPIO Control Room Monitor Logic

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 PREVIEW Com-mand	13	Active Low Input	Feeds External Audio Input to PREVIEW
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
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW 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

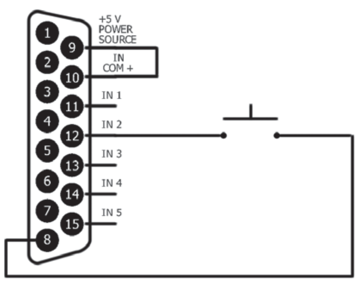
Name	**Pin**	**Type**	**Notes**
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF & sends 100 msec STOP pulse
PREVIEW Command	13	Active Low Input	Turns preview ON
Not Used	14	Active Low Input	
READY Command	15	Active Low Input	Illuminates OFF lamp to indicate source's readiness
OUTPUTS			
NEXT Pulse	1	Open Collector to Logic Common Return	A 100 mS PULSE sent when ON button is depressed, except when initially turned ON.
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF and READY is active
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW 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 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		



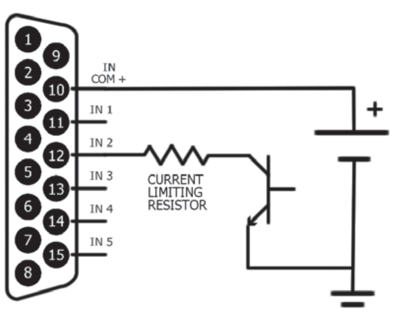
About GPIO Connections

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

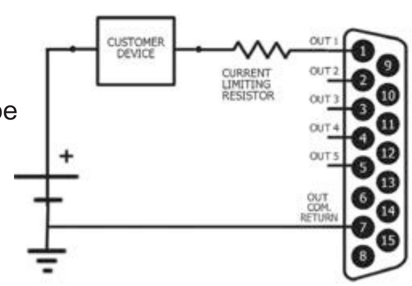
The maximum voltage allowed for an external power supply for logic control is 48 volts DC. The use of a current limiting resistor is required for some voltages.



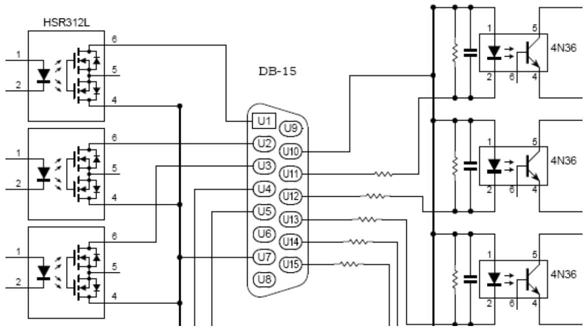
If the equipment being controlled is electrically isolated, than the use of the GPIO port’s power supply is acceptable as shown here.



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 buttons. 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.

Appendix C (Fusion)

Specifications

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,
- +/- 2 ppm
- 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,
- +18 dBu output
- 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

Audio Processing (Fusion)

Equalizer

- Frequency Bands: 20Hz to 320Hz, 125Hz to 2KHz, 1.25KHz to 20KHz.
- Cut/Boost range on each band: -25dB to +15dB.
- Q-factor: Automatic - bandwidth varies based on amount of cut or boost.

Compressor

- Threshold: -30dB to 0dB Ratio: 1:1 to 16:1
- Post-processor Trim Level: Adjustable from -20dB to +20dB

Expander/Noise Gate

- Threshold: -50dB to 0dB Ratio: -30dB to 0dB

De-esser

- Threshold: -20dB to 0dB Ratio: 1:1 to 8:1

Power Supply AC Input, StudioEngine

- Auto-sensing, field-replaceable modular supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, inter- nal fuse
- Power consumption: 100 Watts

Power Supply AC Input, Element Power Supply/GPIO

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 150 Watts

Power Supply AC Input, PowerStation Aux & Main

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 500 Watts

Operating Temperatures

- -10 degrees C to +40 degrees C, <90% humidity, no condensation

Appendix D (Fusion)

CE Declaration of Conformity

Declaration of Conformity

MANUFACTURER'S DECLARATION OF CONFORMITY

Number: TF4082

Declaration: This product is in conformity with Directive 2006/95/EC of the European Parliament and of the Council of 12 December 2006 on the harmonisation of the laws of Member States relating to electrical equipment designed for use within certain voltage limits (codified version).

This product is in conformity with Directive 2004/108/EC of the European Parliament and of the Council of 15 December 2004 on the approximation of the laws of the Member States relating to electromagnetic compatibility and repealing Directive 89/336/EEC Text with EEA relevance.

Manufacturer: Axia Audio, TLS Corp.
1241 Superior Avenue E,
Cleveland, Ohio 44114 USA

Product Identification:

Model: Fusion Console & PSU

Standards Used :

EN60950-1:2006 +A11:2009 +A1:2010	Information technology equipment – Safety Part 1: General requirements
EN 55103-1:2009	Electromagnetic compatibility - Product family standard for audio, video, audio-visual and entertainment lighting control apparatus for professional use -- Part 1: Emissions
EN 55103-2:2009	Electromagnetic compatibility - Product family standard for audio, video, audio-visual and entertainment lighting control apparatus for professional use -- Part 2: Immunity

Technical File: TF4082

Means of Conformity: Technical File

Signature of Manufacturer: SCOTT STIEFEL / [signature] / CHIEF OPER OFFICER
Name/Signature/Title

Place: TLS Corp.
1241 Superior Avenue E,
Cleveland, Ohio 44114 USA

Date: 2/23/2015

Appendix E (Fusion)

Warranty

Telos Alliance Limited Warranty

This Warranty covers “the Products,” which are defined as the various audio equipment, parts, software and accessories manufactured, sold and/or distributed by or on behalf of TLS Corp. and its affiliated companies, collectively doing business as The Telos Alliance (hereinafter “Telos”).

With the exception of software-only items, the Products are warranted to be free from defects in material and workmanship for a period of five (5) years from the date of receipt of such Product by the end-user (such date of receipt the “Receipt Date”). Software-only items are warranted to be free from defects in material and workmanship for a period of 90 days from the Receipt Date. Telos will repair or replace (in its discretion) defective Products returned to Telos within the warranty period, subject to the provisions and limitations set forth herein.

This warranty will be void if the Product: (i) has been subjected, directly or indirectly, to Acts of God, including (without limitation) lightning strikes or resultant power surges; (ii) has been improperly installed or misused, including (without limitation) the failure to use telephone and power line surge protection devices; (iii) has been damaged by accident or neglect. As with all sensitive electronic equipment, to help prevent damage and or loss
of data, we strongly recommend the use of an uninterruptible power supply (UPS) with all of our Products. Telos products are to be used with registered protective interface devices which satisfy regulatory requirements in their country of use.

This Warranty is void if the associated equipment was purchased or otherwise obtained through sales channels not authorized by Telos.

EXCEPT FOR THE ABOVE-STATED EXPRESS WARRANTY, TELOS MAKES NO WARRANTIES, EXPRESS OR IMPLIED (INCLUDING IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE).

In no event will Telos, its directors, officers, employees, agents, owners, consultants or advisors (its “Affiliates”), or authorized dealers or their respective Affiliates, be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from the use of any Product or the inability to use any Product either separately or in combination with other equipment or materials, or from any other cause.

In order to invoke this Warranty, the Product must be registered via Telos’ website (found at: http://telosalliance. com/legal/warranty) at time of receipt by end-user and notice of a warranty claim must be received by Telos within the above stated warranty period and warranty coverage must be authorized by Telos. Contact may be made via email: support@telosalliance.com or via telephone: (+1) 216-241-7225. If Telos authorizes the performance of warranty service, the defective Product must be delivered to: Telos, 1241 Superior Avenue, Cleveland, Ohio 44114 or other company repair center as may be specified by Telos at the time of claim.

Shipping Costs and Warranty Service:

If the date the customer's notice of warranty claim is received by Telos (such date the "Warranty Claim Notice Date") is within the first 90 days following the Receipt Date, Telos will pay the costs of shipping such warranted Product to and from the end user's location, and the cost of repair or replacement of such warranted Product.

If the Warranty Claim Notice Date occurs after the first 90 days following the Receipt Date and before the end of the second (2nd) year, the customer will pay the freight to return the warranted Product to Telos. Telos will then, at its sole discretion, repair or replace the warranted Product and return it to the end user at Telos' expense.

If the Warranty Claim Notice Date occurs between the end of the second (2nd) year following the Receipt Date and the completion of the fifth (5th) year, the customer will pay the costs of shipping such warranted Product to and from the end user's location. Telos will then, in its sole discretion, repair or replace the warranted Product at Telos' expense. Telos also reserves the right, if it is not economically justifiable to repair the warranted Product, to offer a replacement product of comparable performance and condition direct to the customer at a discounted price, accepting the failed warranted Product as a trade-in.

The end user will in all cases be responsible for all duties and taxes associated with the shipment, return and servicing of the warranted Product.

No distributor, dealer, or reseller of Telos products is authorized under any circumstances to extend, expand or otherwise modify in any way the warranty provided by Telos, and any attempt to do so is null and void and shall not be effective as against Telos or its Affiliates.

Out of warranty units returned to the factory for repair may be subject to a $500 evaluation fee, which fee must be prepaid prior to shipping the unit to Telos. If no repairs are required, the $500 fee will be retained by Telos as an evaluation charge. If repairs are required, the $500 fee will be applied to the total cost of the repair.

To activate your product warranty, visit http://www.telosalliance.com/product-registration .

Appendix F (Fusion)

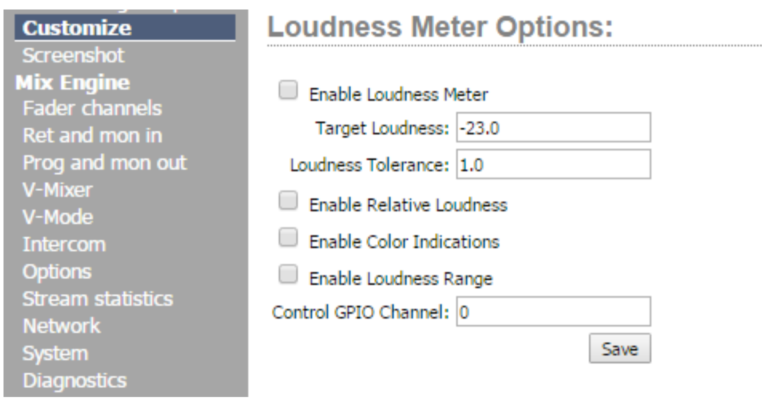
New Features

Version 3.1 adds new features to the Fusion console and those features are noted in this Appendix.

- Add support for loudness metering based on ITU.BS1770 and supporting EBU R128.
- Add support for "Live mode" control of Voco8.
- Add support for Source "Unload". Removes a source from console that has ownership of a source, permitting other console to take ownership.
- Add AES67 support in Studio Engine
- Support Accessory Panel Support and control of Custom Headphone backfeeds.
- Add Phase meter to display.

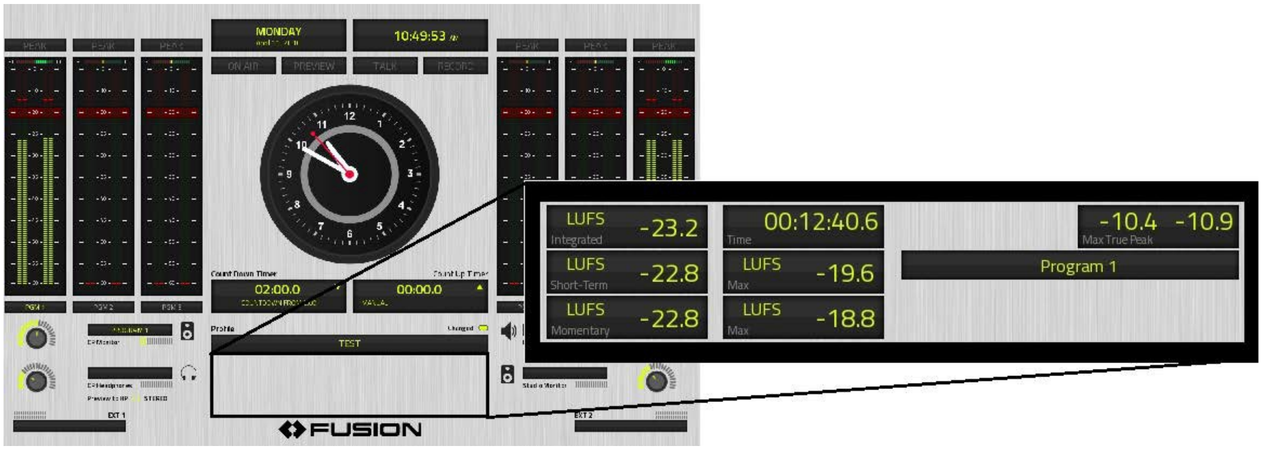
Loudness Metering

With version 3.1, there is an added sub section to the customize option for enabling Loudness metering. Loudness metering is based on ITU.BS1770. Some of the options provided are specific to EBU R128.



Enable Loudness Meter

Checking this option and pressing save will turn on loudness metering on the main display view.



The loudness meters present numerical values for the momentary (shortest time scale of 0.4s), short (intermediate time scale, 3s), and integrated (segment-wise time scale based on timer value) scales. The maximum value recorded for momentary and short scales within the segment time is also shown. To the right of the numerical meters is the max recorded true peak values. Below the true peak meter is the measured source. The options for measured source are the various mix busses of the surface (PGM1, PGM2, etc). To select another bus to view, rotate the first encoder on the overbridge of the master module.

Target Loudness, Loudness Tolerance

Two text fields are provided for entering your desired target loudness value and the acceptable +/- deviation from that 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).

Relative Loudness

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.

Color Indication

To have the meter change color to indicate cold, good, hot, select the *Color Indication* option and make sure the Tolerance is set appropriate. Green is an indication that the loudness is within the tolerance of the target. Red is an indication of above the tolerance value and blue is an indication of being below tolerance.

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

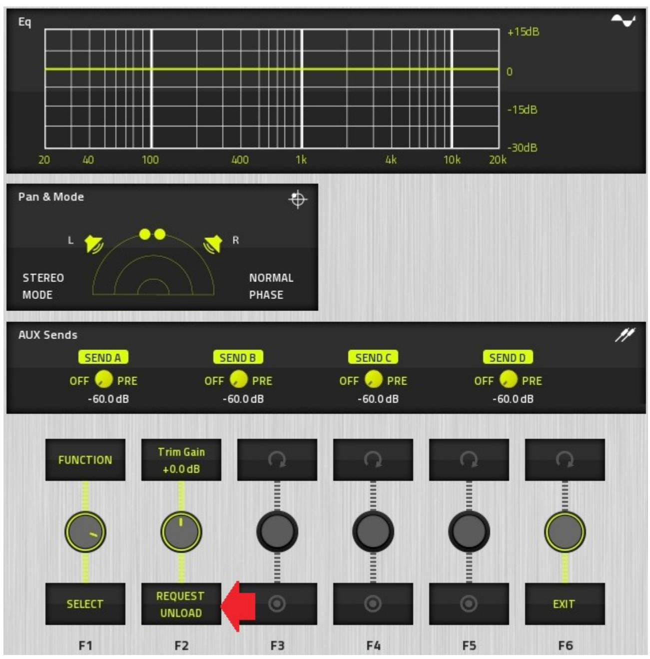
Other than enabling the loudness meter for your facility's needs, the program segment length is needed to be defined by the operator or other control point. When loudness metering is enabled, the timer control buttons are used for control of the integration timer and no longer for the count up timer. To control the up (and down) timer, press the timer option button to control the timers with the overbridge knobs. The control of integration time will be the START/STOP and RESET buttons. 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

Source“unload”

When loading shared resources, typically a facility codec, and said resource is in ownership by another studio, the console will load the source into *Listen Only* mode. This permits the console to have access of the audio from source, but is not able to control the source or provide backfeed audio. The console will provide indication of who is the owner of the source so that the source so action can take place if the newly requesting console can take ownership. In some cases, the last operator of that other studio may have left, requiring someone to walk but customers have told us they don’t want to walk down to the other studio. The *Request Unload* option now appears on general Channel Option screen when a source is loaded in *Listen Only* mode. Pressing the *Request Unload* function will open a new view requiring the operator to rotate the first knob and select the confirmation that the source is to be unloaded. By pressing the knob in, after selecting to unload, requires a three step process to unload the source from the other console. This is a security feature to avoid accidental operations. If the source is in the ON state on the active owner, the source will not unload as an extra protection and the requesting console will remain in *Listen Only* mode.



AES67 support

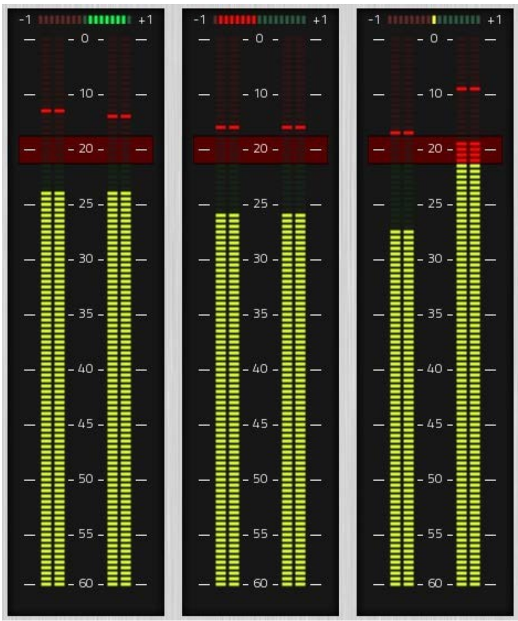
Version 3.1 introduces support for AES67 in the Studio Engine. This support is introduced in VMode and is further discussed in the VMODE section of Chapter 4 of the manual. With VMode, it is possible to convert AES67 streams with Livewire streams.

Accessory Panel Support

Version 3.1 also provides support of customized backfeed control with the Fusion Accessory panels. In addition, support for a Fader panel will be introduce. With the Fader panel, a positions can control the level of their own microphone away from the operator control or control a source other than the microphone, often a playout computer.

Phase meter

A phase meter is introduced at the top of each mix bus meter. A presentation of full green (+1) indicates the Left and Right channels are in phase together. A presentation of full red (-1) 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.



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