

Axia® iQx

AES67 Console with Integrated Engine



Installation & User's Guide

Version 1.0 February, 2019
1490-00203-003 --- USER MANUAL: AXIA IQX



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User Warnings and Cautions

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 100-240VAC. 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: HAZARDOUS VOLTAGES

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 on the rear 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.

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Patent information can be found at www.TelosAlliance.com/legal

Updates

Axia iQx features and operations are determined largely by software. The Telos Alliance strives to provide the most stable and feature-rich software available. We encourage you to check for software updates from time to time by visiting our website or by contacting us directly.

Feedback

We welcome feedback on any aspect of our products 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 or suggestions.

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You may reach our Telos Support Team in emergencies by calling +1 216-622-0247. For billing questions or other non-emergency technical questions, call +1 216-241-7225 between 9:00 AM to 5:00 PM USA Eastern Time, Monday through Friday.

By Email.

Non-emergency technical support is available at Support@TelosAlliance.com.

By Web

The Axia Web site has a variety of information that may be useful for product selection and support. The URL is <https://www.telosalliance.com/Axia>.

SERVICE

You must contact Telos Alliance before returning any equipment for factory service. We will need your unit's serial number, located on the back of the unit. We 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 Telos Alliance Technical Support at +1-216-622-0247. All other customers should contact local representative to make arrangements for service.

Warranty



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Register your product today to get the full benefits of our warranty, support, and product updates. telosalliance.com/product-registration/

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Table of Contents

	Warnings and Cautions	ii
	We support you.	iv
	Warranty.	iv
	<hr/> <i>Quick Start Guide</i>	<hr/> 1
1	<i>Introduction to iQx</i>	7
	iQx Surface	7
2	<i>Surface Operation</i>	9
	Channel Strip.	9
	A - ON/OFF Buttons	9
	B - Soft Button	9
	C - Channel Display	10
	D - Preview Button	10
	E - Fader	10
	F - Mix (Bus) Assignment Buttons	10
	G - Option Knob	11
	Monitor Section.	12
	A - Meter Toggle Button	12
	B - Clock/Timer Buttons	12
	C - Record Button	12
	D - Show Profile Button	12
	E - Studio Knob	12
	F - Studio Display	12
	G - Talk to Studio Button	13
	H - Talk to Backfeeds Button.	13
	I - Control Room Display	13
	J - Control Room Monitor Select Buttons	13
	K - Control Room Knob	13
	L - Operator Headphone Knob	13
	M - Preview Knob	13
	N - Preview in Headphone Button	13

	Remote	14
	Overbridge	14
	Audio Meters (A)(B)	14
	Clocks (C)	15
	Telco Option	15
	Fixed (“Euro”) Mode	15
	Selectable (Multi-Line) Mode	16
	User Button Option	17
3	<i>Audio Configuration</i>	19
	Source Profiles	19
	Source Profile Options	20
	Show Profiles	25
	Editing Show profiles	26
	Capture a Profile.	29
	Changes	29
	Audio Outputs	30
4	<i>GPIO</i>	33
	Line Profile GPIO Table	35
	Computer Player Profile GPIO Table	36
	Phone Profile GPIO Table	37
	Codec Profile GPIO Table	38
	Operator Microphone Profile GPIO Table	39
	CR Producer Microphone Profile GPIO Table.	40
	CR Guest Microphone Profile GPIO Table	41
	Studio Guest Microphone Profile GPIO Table	42
	External Microphone Profile GPIO Table	43
	Studio Feed Profile GPIO Table	44
	Control Room Profile GPIO Table.	45
	Studio Profile GPIO Table	46
	Film Cap buttons Profile GPIO Table	47
	Record (Enable) Mode Profile GPIO Table	48
	External Timer Control Profile GPIO Table.	49
	V-Mixer Profile GPIO Table	50
	Show Profile Profile GPIO Table	51

5	<i>System Configuration</i>	53
	Console Config	53
	V-Mixer and V-Mode	54
	Brightness Control	54
	Synchronization	54
	Backup & Restore	55
	General System Setup	55
	Code Updates.	55
A	<i>Appendix: Mechanical</i>	57
B	<i>Appendix: Tear Sheets – Specific Use Case Configurations</i>	59
	AES67 input	59
	AES67 output.	59
	On Air Light	60
	Phone setup to VX.	61
	Split recording of telephone calls	62
	<i>Warranty</i>	65

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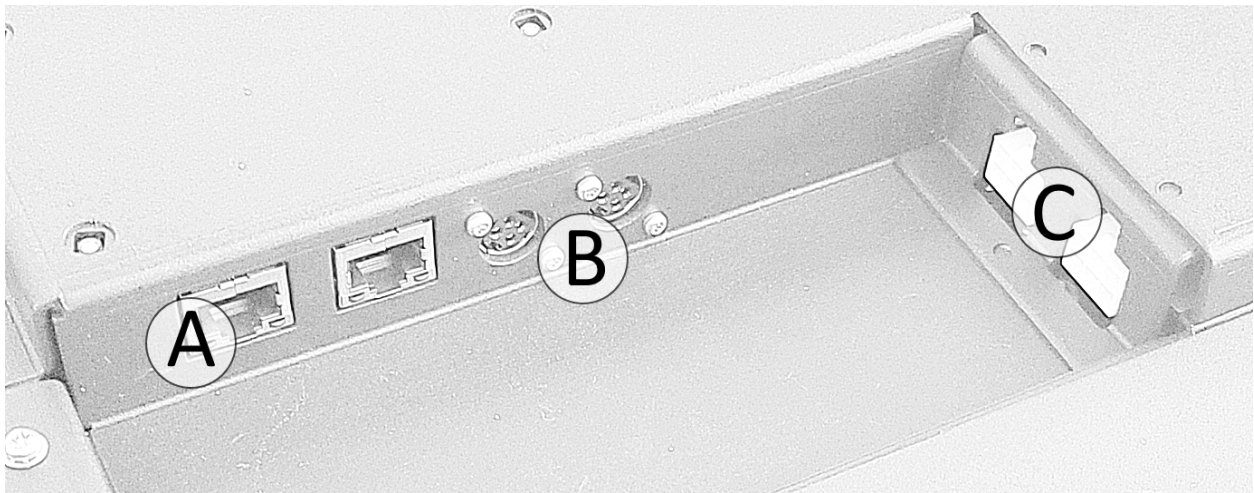
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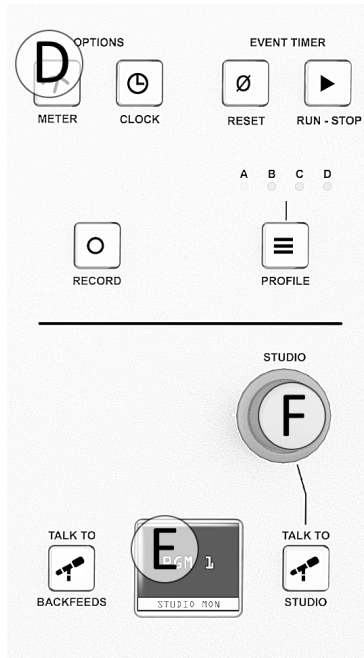
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Quick Start Guide

On the underside of the iQx console, find the connections for network, power, and expansion modules. Connect the uplink port (A) to a 1 gigabit ethernet link on your switch. Connect power (B) to either power ports. Connect expansion modules (optional) to the expansion module ports (C) . Flip the console right side up and connect the provided power supply to the AC main power.



Wait for the bootup (about 45 seconds) .



Press and hold the METER button (D) until the Studio display (E) changes to a menu.

Rotate the Studio knob (F) to highlight NETWORK, press the knob in to enter.

Rotate the knob to highlight IP ADDRESS, press the knob to enter.

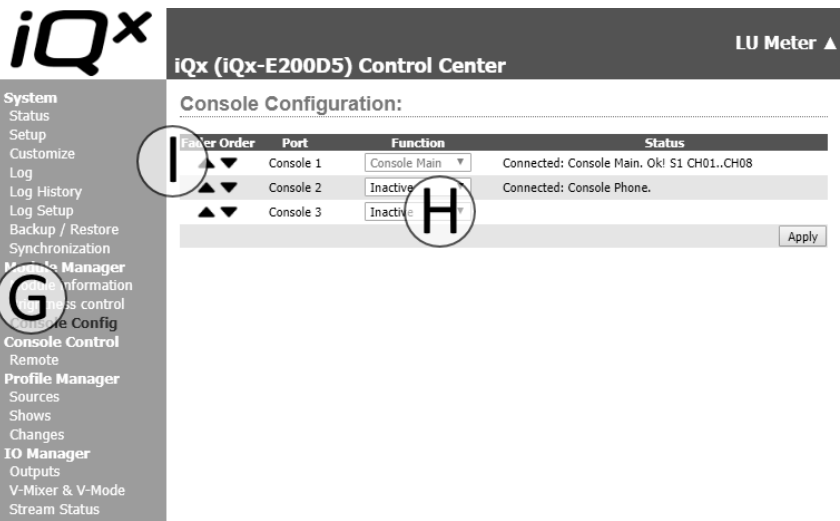
The display will show the 4 octets of an IP address. Rotate the knob to highlight an octet, press to enter edit mode, rotate the knob to increment or decrement the value, press the knob to end edit mode. Do these steps to edit the IP address of the device.

When the IP address is complete, rotate the knob to the top of the menu, two dots are shown, and press the knob to exit IP ADDRESS.

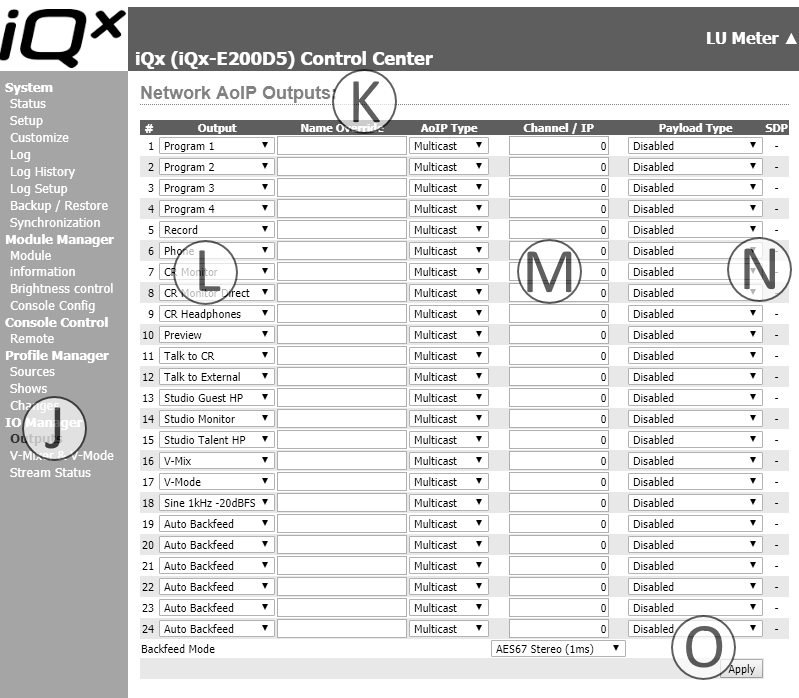


Repeat the procedures for netmask and gateway based on the needs of your network. After all NETWORK settings have been edited, select the SAVE menu item. The screen will return to its default function and the surface is ready for system configuration.

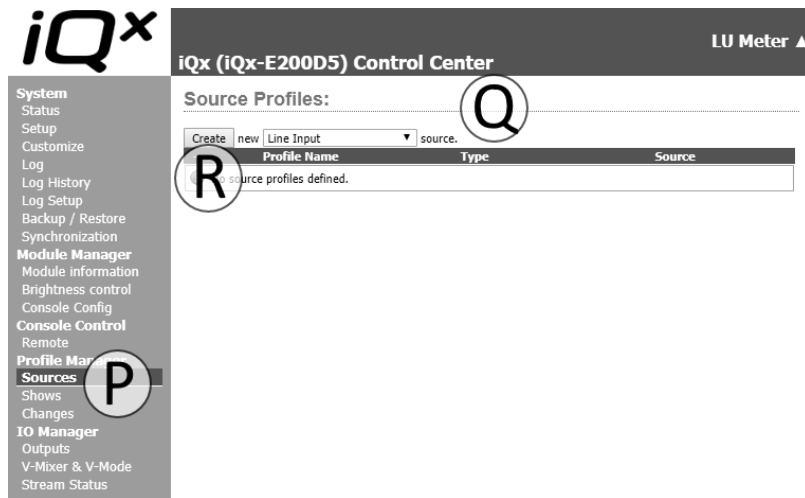
Using a PC connected to your studio network, launch a web browser and enter the assigned IP address into the browser's address bar. When prompted for authentication enter user name "user", password is left blank.



If installing expansion modules (optional) , select the Console Config link (G) . Console 2 and 3 ports are inactive by default, use the drop down (H) to select the correct expansion module that is connected. Use the Fader Order (I) arrows to organize the fader numbers to the physical placement of the expansion modules. The status column will indicate the fader numbers assigned to the module. Complete the console configuration by pressing the Apply button to the right.



In the iQx Control Center navigation in the left column under IO Manager, click “Outputs” (J) to view the Outputs screen (K) . Configure the outputs that are needed for the studio. As an example, select CR Monitor from the Output dropdown (L) , enter in a Livewire channel number 2907 into the Channel/IP field (M) , and select AES67 Stereo (1ms) in the Payload Type (N) . When the needed outputs are designated, select the Apply button to the bottom right (O) .



In the iQx Control Center navigation in the left column under Profile Manager click “Sources” (P) to view the Sources screen (Q).

Press the Create button (R), while noting the type of source is the default *Line Input* (review Chapter 3 for details on the options here). This will open the view to the Source Profile screen (S). Enter the word *TONE* into the Source name field (T). Select the *Sine 1kHz Tone -20dBFS* from the Source Input (U). Complete the profile generation by pressing the OK button (V). The screen will return to the Source screen with the *TONE* profile now listed. Press the create button again (R) and in the Source name field (T) enter *AoIP Src*. Press the browse button next to the Primary source field (W) and select a Livewire stream from the network (alternatively enter in the multicast value of an AES67 stream on the network).

Source Profile:

Source type:

Source name:

Source input:

Primary source:

Signal mode:

Signal phase:

Signal mode for Record bus:

Record Insert mode:

Input trim gain [-25 .. 25 dB]:

Panorama position [-24 .. 24]:

Audio delay (AES67 Link Offset) [0 .. 250 ms]:

Line Input

Preview

Stereo

Normal

Stereo

Post-Fader

+0.0 dB

0

0 ms

Equalizer:

Band:

Frequency:

Gain [-25 .. 15 dB]:

Mode:

Low

25 Hz

+0.0 dB

Auto

Mid

1000 Hz

+0.0 dB

Peak

High

16000 Hz

+0.0 dB

Peak

Source availability:

☒ Channel 1

☒ Channel 2

☒ Channel 3

☒ Channel 4

☒ Channel 5

☒ Channel 6

☒ Channel 7

☒ Channel 8

☒ All Channels

☒ External

☒ V-Mixer

Fader mode:

Preview mode:

Preview switching:

Auto-start timer:

Logic port:

Normal

Pre-Fader (CUE)

☐ Channel ON turns Preview OFF

☐ Preview ON turns Channel OFF

☐ Enabled

Exclusive mode

☐ GPIO ready enabled

Live Controls:

SOURCE

GAIN

MODE

PANORAMA

EQUALIZER

☐ Disabled

☐ Disabled

☐ Disabled

☐ Disabled

☐ Disabled

☒ Enabled

☒ Enabled

☒ Enabled

☒ Enabled

☒ Enabled

☐ Primary

☐ Primary

☐ Primary

☐ Primary

☐ Primary

Apply

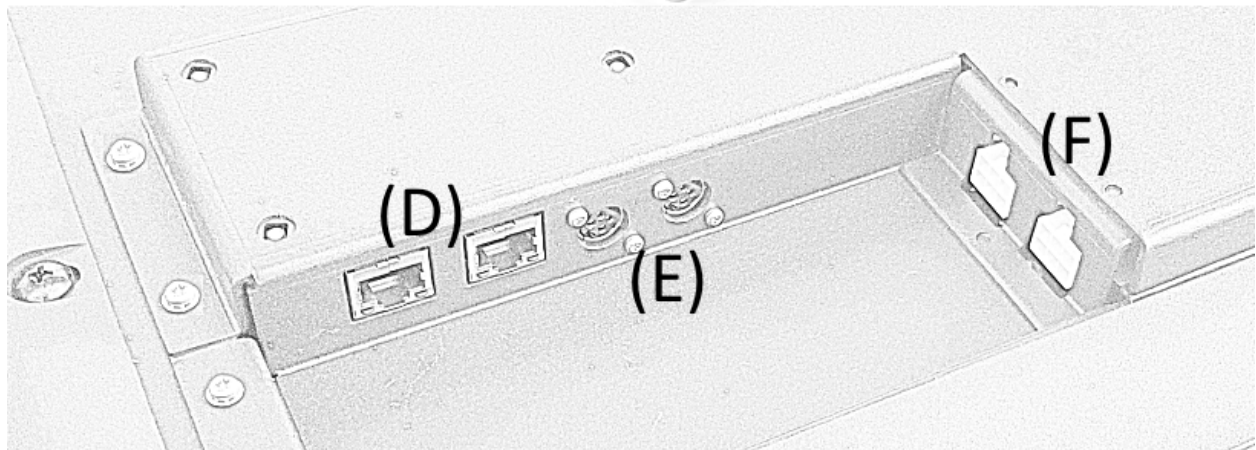
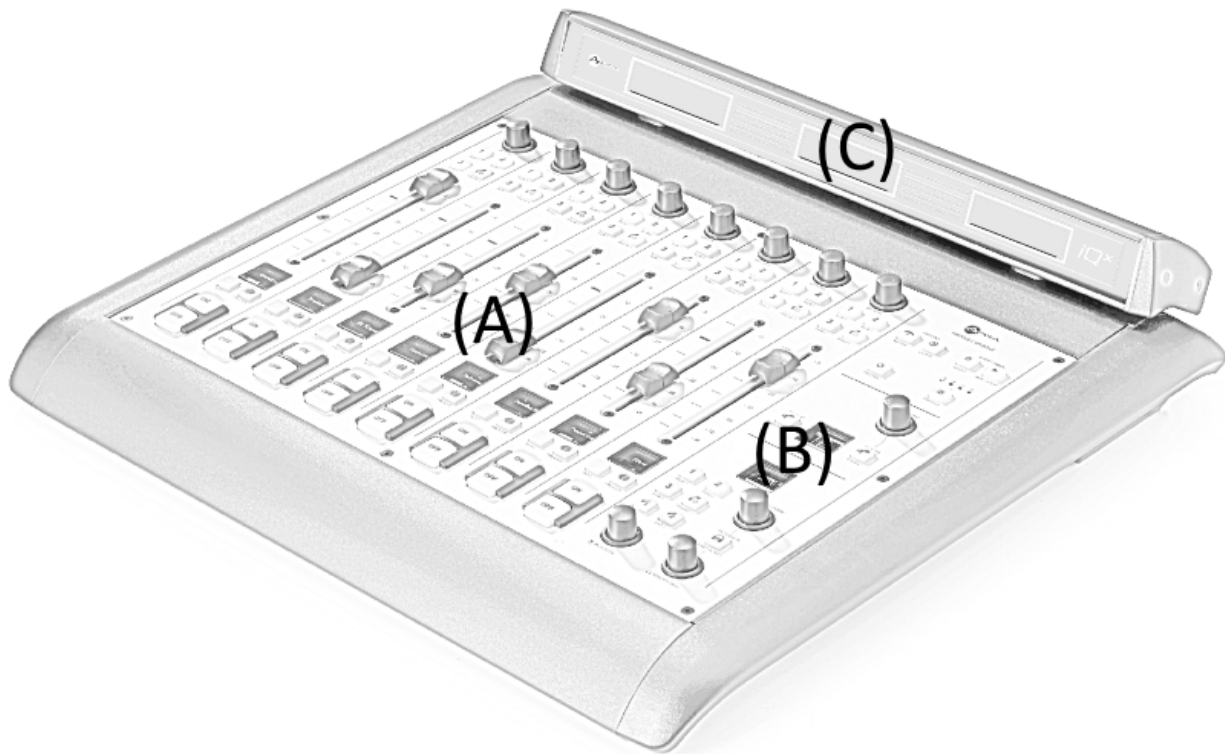
OK

Cancel

With some profiles created, load them to a channel on the console by pressing the knob at the top of any channel, in the channel display, highlight the source by rotating the knob, select the source by pressing the knob. Make sure you have the channel assigned to Program 1 by pressing the PGM1 button at the top and also that the monitor section also has PGM1 selected. You should be able to hear tone from the speakers that are configured to receive CR Monitor. Configuring the speaker or a node feeding an amp is not part of this Quick Start guide.

1 Introduction to iQx

Thank you for choosing Axia® as your choice in AoIP studios. We know there are many options available to you and believe you will be satisfied with your purchase thanks to the ease of setup, technical support, and reliability of the product. This chapter will introduce you to the iQx system.



iQx Surface

The main surface is the heart of the iQx system. In addition to the integrated DSP engine, it includes:

A - 8 console channels (covered in chapter 2)

B - Monitor control section (covered in chapter 2)

C - Meter overbridge (covered in chapter 2)

D - Dual network ports: These allow for a connection to the network (the port furthest from power) and for connection to a local IO product (the port nearest to the power).

E - Dual power inputs: Allow for redundant power supply for those concerned about having a single power supply powering the system. The two inputs are diode steered.

F - Two expansion ports: For use with the expansion modules ("sidecars") from the iQ product line, available in 8-fader, 6-fader with Telco, and 6-fader with User buttons modules. The optional expansion modules allow for expanding the fader count up to 24 channels (two 8-fader sidecars) and/or adding user-friendly Telco control or User buttons.

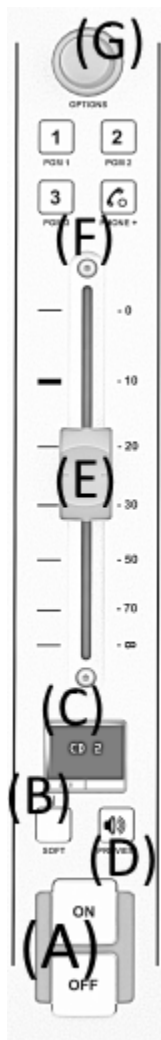
The main module and all three sidecars are designed for tabletop placement (no cutouts in the table are needed other than for wire access). When adding sidecars to the system, you can keep them as unique chassis (as they are shipped) or join the chassis together to make a unified console (refer to Appendix A).

All of the iQx settings are set from a web interface (browser-accessed UI). At first you will set an IP address to the unit by interfacing with the hardware on the surface. Once the IP address is set, all configuration is done from a browser on your computer. Chapter 3 will cover basic configuration items and chapter 5 will cover the more obscure items.

The iQx system was designed with latest AoIP standards in mind. AES67 is the target audio transport and with that, the iQx system requires a PTP grandmaster on the network. Native Livewire® functionality of advertisement, GPIO, and control are still supported. Other discovery and control protocols are planned for in the product.

2 Surface Operation

This chapter will cover the physical layout and user operation of the console surface. (For configuration, please refer to later chapters.)



Channel Strip

The main surface and each expansion module (sidecar) have a set amount of channel strips, either 8 or 6.

A - ON/OFF Buttons

At the bottom of the channel are the ON and OFF buttons. Pressing a channel's ON button will switch the channel to feed the designated mixes that are assigned (F). Pressing the OFF button is the opposite. ON and OFF state changes will trigger logic conditions and impact Livewire GPIO. Refer to chapter 4 for more details on GPIO behavior. The ON/OFF buttons will illuminate to indicate the state of the channel. The buttons may also blink to indicate a logic condition as follows:

1. **ON button illuminates while OFF blinks in 1-second intervals:** This indicates the channel has been instructed to change its source profile, but is prevented from doing so because the channel is in the ON state. Turn off the channel to clear the issue.
2. **OFF button illuminated while ON blinks in 1 second intervals:** This indicates the channel is in *Fader Start with Arm* mode (see E- Fader section below) and is armed.

B - Soft Button

The Soft button above the ON/OFF buttons is a context-sensitive button. Based on the profile loaded to the channel, the button is given a function. The display (C) will label the function. The possible functions are as follows:

1. **Next command (Computer Player profile):** Triggers GPO, which can be used to start the next event in an automation system if your automation system supports this function.
2. **Talk command (Phone, Codec, or Studio profiles):** Initiates talkback to the mix-minus from the operator microphone or alternate source as defined in the *Show* profile.
3. **Mute command (Operator profile):** Mutes the source (cough button). Note that when the channel is ON and the ON button is pressed, the same *Mute* function is enabled and the Soft key is illuminated to confirm the mute function is engaged.

C - Channel Display

Each channel has a display, which provides labeling, audio confidence metering, and user feedback information. A *Source* profile (chapter 3) defines the label with a 10-character limit that will be shown for the end user. This label appears at the center of the display when the profile is assigned to the channel. Additionally, an audio confidence meter for the source will appear at the bottom right of the display. If the profile defines a backfeed (mix-minus feed), there will be a confidence audio meter to the top right representing the return audio (what is called “backfeed”). Under the label may be other text based on conditions of the source. For example, if the source is owned by another console, the “*Used EW*” (used elsewhere) message will appear. If the source supports and is using Extra Label feature available in some Axia products, this text will appear under the label.

D - Preview Button

Next to the Soft button is the Preview button, which is used to route the audio to the Preview mix for OFF AIR monitoring of the source. Some know this as the “CUE” function. By default the Preview button behaves in “interlock” mode, which means pressing a second Preview button will disengage the first. You can monitor multiple channels at the same time by pressing all buttons at the same time or holding down an active Preview channel while pressing additional Preview buttons. To remove individual channels from Preview, press any active Preview button followed by pressing the channels you wish to be removed. To end the preview of multiple channels or a single channel, press the preview button that is illuminated.

E - Fader

The fader controls the audio level feeding the mixes that the channel is assigned to. The top most position of the fader is +10dB (by default, but can be changed). A thick mark near the top represents the unity gain position (+0dB). The behavior of the gain due to the fader position can be adjusted by the *Fader Offset* setting under *Customize / Fader Options* in the web UI.

The fader position may also impact the ON/OFF state of the channel if the profile is defined with a *Fader Start* or *Fader Start with Arm*. This setting is found in each *Source* profile in the *Fader Mode* option. Fader start will turn the channel to the ON state when the fader is moved away from negative infinity. When using the Arm option, the channel ON button must be pressed to arm the channel followed by sliding the fader away from the bottom negative infinity position. Pressing the OFF button at any position of the fader will change the channel state to OFF.

F - Mix (Bus) Assignment Buttons

Near the top of each channel are 4 buttons that are labeled 1, 2, 3, and a Phone handset icon. These buttons assign the channel to feed the associate mixes, after the Fader gain and ON/OFF switch. The first three buttons (1,2, and 3) are assigned to the Program 1, 2, and 3 mixes. The 4th button (with the phone handset) has association with three different mixes, Program 4, Program 4 Record, and Phone. These three mixes are tied to the same button and behave in the follow manner:

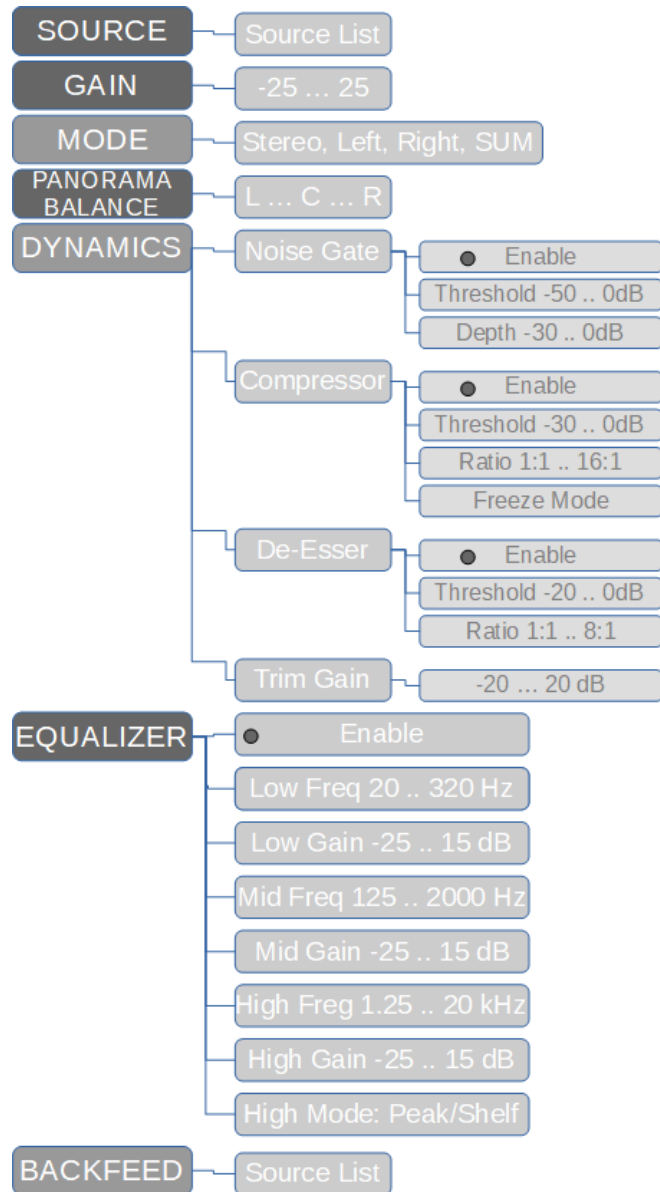
Program 4: Same as the other program mixes in that the channel feeds the mix post ON state and post Fader position. So the channel needs to be ON and the fader needs to be above negative infinity to feed into the mix.

Program 4 Record: A specialized mix intended for OFF AIR recording and level control. The channel feeds the mix pre ON state and post fader position. So the channel feeds Program 4 Record based on the fader position and without regard to the ON/OFF state of the channel.

Phone: A specialized mix used in mix-minus feeds. The channel feeds the Phone mix without regard to ON state nor fader position. Basically, if the 4th button is engaged, the source is feeding the Phone mix. The Phone mix is used in backfeed selection. This topic is covered in more detail in the source profile configuration in chapter 3.

Pressing a button in the Mix section will engage that mix, illuminating the button. Pressing the button again will extinguish the button, indicating a no assignment to the mix.

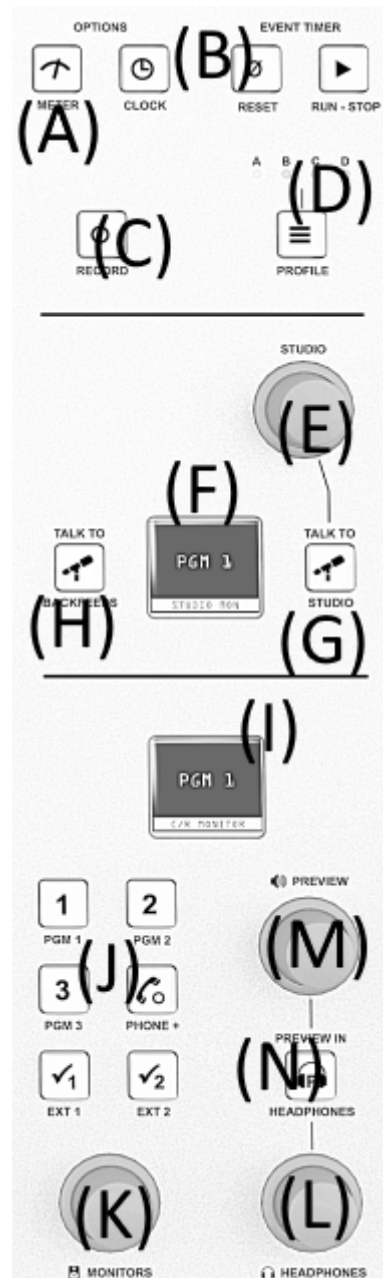
G - Option Knob



The option knob can be rotated or pressed to trigger system actions. By default, pressing the knob will trigger the display (C) to show a menu, which can be further navigated by rotating the knob and pressing to select. The options available in the menu will depend on the profile. The chart shown here shows the menu tree. Not all branches are universally available. Source branch and Gain are both universally available, but the Mode branch is only available to source types that support stereo. The Backfeed branch is only available to sources that produce mix-minus. The source profile can define some of the menu items to be filtered, not populating to the end user and thus limiting their control.

There is an option to define quick access control to some of the controls available through the menu tree (see above). Instead of navigating through menus, the knob can be assigned a quick access ("Live Controls"). For example, you can adjust the Gain option by rotating the knob without pressing it first. This allows for quick access to items that are often adjusted and avoiding delays due to menu selection.

Monitor Section



The Monitor section gives control to the monitors located in the control room and in an external talk studio, and provides control to an event timer. You'll also find the console's meter presentation and clock presentation are accessible here. This is also where you will find Show profiles, a layout configuration for the surface, which the operator can change by accessing the monitor section.

A - Meter Toggle Button

Pressing this button will toggle the last meter on the overbridge between various sources. The sources that are toggled depend on the amount of meters presented (refer to "Customize" topic in chapter 5). For example, if two meters are shown, the second meter will toggle between Pgm2, Pgm3, Pgm4, and Monitor. If four meters are shown, the last meter (number 4) will toggle between Pgm4 and Monitor.

B - Clock/ Timer Buttons

The *CLOCK* button will toggle the mode that the timer operates. The options are Manual or Auto. Manual mode requires the use of the Event Timer buttons to control the timer located on the overbridge. In Auto mode, source profiles with the Auto-Start timer option enabled will start the timer. Once started, the *RUN-STOP* button will illuminate and can be pressed again to stop the timer. The *RESET* button will set the timer back to zero.

C - Record Button

The *RECORD* button will trigger the record mode as it is defined in the Show Profile. Refer to Show Profile configuration in chapter 3 for more information on Record mode.

D - Show Profile Button

Pressing the *PROFILE* button will change display (F) to show the names of four show profiles. A fifth option is also present, which loads nothing, "NONE". This is a clear option that can be filtered from the Show Profile configuration page, and thus not available to end users. Selecting one of the show profiles will load the console layout as defined in the profile. Chapter 3 covers the topic of Show configuration. The LEDs at the top will also illuminate to indicate which of the four profiles is loaded.

E - Studio Knob

The *STUDIO* knob rotation will control the volume (audio level) to the Studio Monitor feed. When rotating the knob, display (F) will show a horizontal bar that increases/decreases to show the relative level. Pressing the knob will load sources to display (F), followed by rotation of the knob to highlight which audio source feeds the Studio Monitors. Once the desired source is highlighted, press the knob once more to select.

F - Studio Display

The display is used to provide information on which source is feeding the Studio Monitors, the status of the monitors (for example "[MUTED]") and is the display used for Show Profile selection.

G - Talk to Studio Button

The button below the Studio knob is used to talk into the Monitor Feed for cueing purposes. Pressing *TALK TO STUDIO* will take the audio of the Operator profile and feed it directly to the Studio Monitor feed. By Default the source defined as the Operator will be used for this talkback and this source would need to be loaded to a fader. The Show Profile provides the option to define an “External Source,” which avoids the requirement to load the Operator microphone to a channel.

H- Talk to Backfeeds Button

Similar to *TALK TO STUDIO* button, *TALK TO BACKFEEDS* is a global talkback button that will feed the operator microphone audio to all backfeeds that are generated by the console.

I - Control Room Display

The Control Room display indicates conditions on three different audio feeds related to the Control Room. These three feeds are the Control Room Monitor Feed, the Preview Feed, and the Operator Headphone Feed. By default the display shows the source selected to the monitors. By pressing the Preview knob (M) or the Headphone knob (L), the display will temporarily indicate which source is feeding those destinations. By rotating any of the knobs, (K) (L) (M), the display will show a horizontal bar indicating the volume. The display will indicate [MUTED] or [DIMMED] states.

J - Control Room Monitor Select Buttons

These six buttons are used to select what is monitored in the control room. The first four buttons are designated to the four Program mixes. The last two buttons are labeled “EXT n”—external—and are assigned to any source. A typical use case is to monitor the tuner set to your frequency and thus monitor what is broadcast on the air via the ON AIR Monitor. The operator can assign a source to a button by pressing the button for a few seconds until display (F) changes to show a list of sources. In this condition, use knob (E) to highlight the desired new source followed by pressing the knob to select. To exit the mode, press the *EXT* button again. Pressing the *EXT* button will route that audio to the control room monitors and operator headphones as well as list the name on display (I).

K - Control Room Knob

The control room knob is the volume for the control room monitor speakers. First-time operators should be instructed to use this knob to adjust the volume and not the channel faders. The channel faders are for adjusting the audio level in relation to what is shown on the meter bridge.

L - Operator Headphone Knob

The headphone knob is for adjusting the volume feeding the operator headphone. The knob is not intended to adjust the volume of any other headphone. The headphone feed would come from an xNode or other AoIP I/O device. The iQx console provides no analog audio interfaces.

M - Preview Knob

The preview knob will adjust the volume of the Preview mix. As items are selected for preview, the operator may need to adjust the volume to better hear the audio. By default the main monitors will dim when any channel is engaged into Preview in order to assist the operator.

N - Preview in Headphone Button

This button will route the Preview mix into the operator headphone feed. The result will be a mix of Preview and the selected (J) source. The selected source will be dimmed.

Remote

System

Status

Setup

Customize

Log

Log History

Log Setup

Backup / Restore

Synchronization

Module Manager

Module information

Brightness control

Console Config

Console Control

Remote

Profile Manager

Sources

Shows

Changes

IO Manager

Outputs

V-Mixer & V-Mode

Stream Status

Remote Control:

Show profile: B:SHOW B ▼ Load

Record mode: [enabled] ☐ Active Apply

Fader Channels							
#	Source	Status	Fader Gain	ON	PGMx	Pview	Apply
1	Live_Op ▼	Full Control	-2.4 dB	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Apply
2	PHONE1 ▼	Full Control	-5.4 dB	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Apply
3	PHONE2 ▼	Full Control	-4.3 dB	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	Apply
4	2018 ▼	Full Control	-2.9 dB	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	Apply
5	Multi 1 ▼	Full Control	-1.1 dB	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Apply
6	vmix ▼	Full Control	-1.0 dB	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Apply
7	TONE ▼	Full Control	-8.9 dB	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Apply
8	1402 PC ▼	Full Control	-0.7 dB	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Apply

Apply All Channels

Monitors				
#	Source	Status	Volume	Apply
CR Monitor	Program 1 ▼		-26.0 dB	Apply
CR Headphones	-		-30.0 dB	Apply
Preview	-		-30.0 dB	Apply
Studio Monitor	--- no source --- ▼		-30.0 dB	Apply

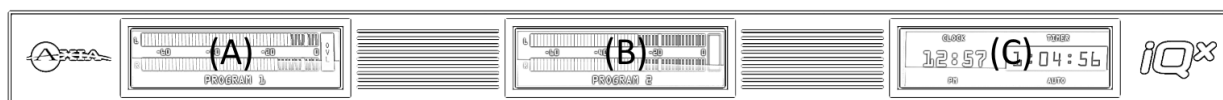
Apply All

Many of the important operations that are achieved from the surface can also be controlled remotely from the web browser. A simple static display with controls is offered from the *Remote* link that allows for simple control.

From the page, the Show Profile, Record mode, Source Profiles, Fader Gain, Channel States can be changed, and you can control the monitor section.

Overbridge

The overbridge is composed of three displays. Two displays are dedicated to displaying audio meters while the third display is a clock and count-up timer.



Audio Meters (A)(B)

You can access the web UI configuration through the *Customize* link. There you'll find the *Meters* section, giving the mode and count options as explained below.

You can configure the two displays for audio meters to be in one of three count modes, "2 meters," "3 meters," or "4 meters."

In 2 Meters Display, the audio meters will span the full width of the displays. The *METER* button on the monitor section would be used to toggle the (B) display to different sources. The (A) display would be fixed to Program 1.

In 3 Meters Display, the (B) display would be split in two with two meters shown. Again the *METER* button would toggle the last meter while the second meter would be fixed to Program 2.

In 4 Meters Display, both displays would be split showing Program 1, 2, 3 and the fourth meter being toggled between Program 4 and the Control Room monitor selection.

The meters have three ballistic options (modes), Full Scale VU, EBU Digital Peak, and PPM. The Full Scale VU is an RMS presentation with an added Peak indicator. The other two are peak presentations based on those standards. The mode is changed from the web UI and takes effect immediately.

Clocks (C)

The third display is split in half to show the time and a count up timer. The clock can be configured to show the day's date, seconds, or military time (24 hours). Note that AES67 sync comes from PTP and a grandmaster sync source would include real time. Because of this, the clock would default to the time presented by the PTP source. If the PTP source has an arbitrary clock (not recommended), you will need to set up an NTP source or set the time and date after a bootup.

Configuration options of the clock are found in the *Customize* link in the web interface. Configuration options are to show 12 or 24-hour time, seconds, date, timezone settings, daylight saving (DST) settings, and NTP source server.

The timer has two modes—Manual or Automatic. In automatic mode, designated source profiles would cause the timer to start or reset (if already started). In manual mode, the timer would only start by use of the Event Timer buttons on the monitor section. In either mode the Event Time buttons would Start/Stop and reset the timer.

Telco Option

The iQx has native support for Telos phone interface products. The Telos products supporting the VX protocol (iQ6 or any VX product), can be interfaced with by the console. There are two modes of operation, fixed or selectable. Configuration is covered in chapter 3.

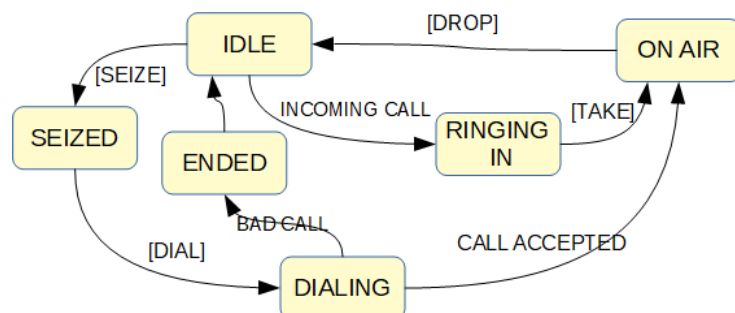
Fixed (“Euro”) Mode

Fixed operation dedicates a single phone line to channel (source profile). If the optional iQx 6-Fader with Telco sidecar is added to the system, the operator could make and receive phone calls. If no Telco sidecar is added to the system, the operator would only be able to receive phone calls.

When a correctly working phone profile is loaded to the channel, the display will indicate the status of the phone line. The various states are *IDLE* (ready and waiting), *RINGING IN* (incoming call), *ON AIR* (active call).

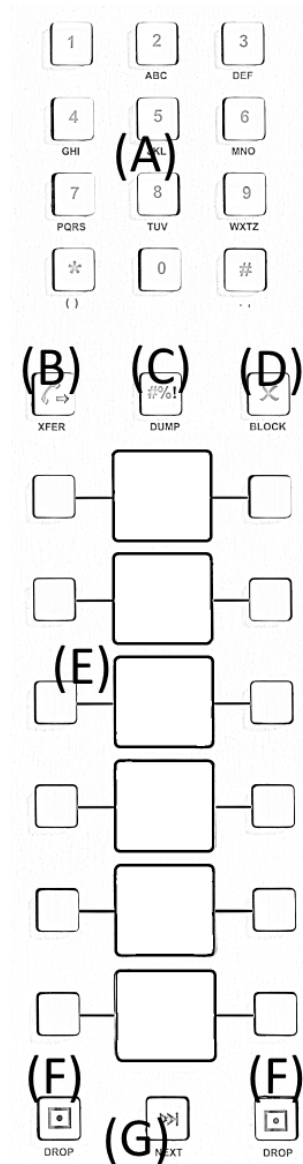
When the phone extension receives a phone call, the Soft button of the channel will blink with the label above the button indicating *TAKE*. Pressing the button will pick up the call. At this point, the label above the Soft button changes to *DROP* indicating the new function for the button. Pressing the button again will drop the call.

When the line is in *IDLE* mode, the Soft button will be labeled *SEIZE* and can be pressed to pick up the line. At this point the status changes from *IDLE* to *SEIZED* and the Soft button indicates a function of *DROP*. Without a Telco expansion sidecar, there isn't much to be done but drop the seized line. If the expansion sidecar is installed, the operator can dial the number to call on the keypad followed by pressing the # key (or the Soft key, which has again changed function to *DIAL*) to initiate the call. The status of the line, indicated on the display of the channel, will show *DIALING*. When the call is taken at the far end, the status will change to *ON AIR*, otherwise the status will indicate *ENDED* if the call could not be made.



Phone Line States - Fixed Mode

Selectable (Multi-Line) Mode



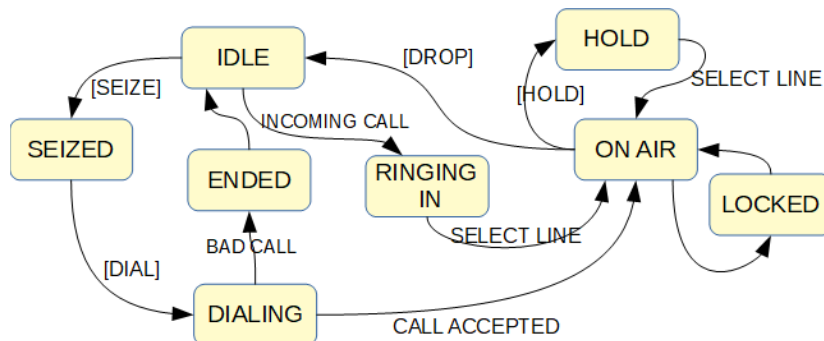
With the Telco expansion sidecar installed, the operator has access to a 6 line selection panel for controlling two phone channels.

The key pad (A) should be familiar to anyone who has used a phone. The XFER (transfer) button (B) can function depending on the version of Telos VX and the PBX system.

To transfer a call, press the *XFER* button (B) followed by the Line button (section E, which includes 12 buttons) to be transferred. Dial the new number on the keypad (A) followed by the # key. When the call is made at the far end, the line will be dropped locally and the display will indicate *ENDED*.

The Dump button (C) is a GPIO command to the Telos VX system. The Telos VX will provide configuration for what GPIO action to take place when this button is pushed. The Block button (D) is used to keep all the lines busy. This is commonly used prior to a call-in condition so that no lines are active with a call before the host instructs listeners to call in (“Caller 10 wins the prize!”). The line selection (E) is made up of 12 buttons and 6 displays. The 12 buttons make two columns, the left column for a first phone interface and the right column for a second phone interface. The 6 displays present the state of the 6 possible phone lines. There are two *DROP* buttons (F) for dropping calls, one for each column (phone interface). Any phone line not in a *LOCKED* state will be dropped. The *NEXT* button (G) is used when multiple calls are on hold and the operator wishes to take the first call in line.

When receiving a call, the display (E) will indicate *RINGING IN* and the buttons on either side of the display will blink. Selecting the left button will take the call on the first phone interface. It is a best practice to have the profile for the first interface on the left of the telco control panel. The display will indicate the *ON AIR* state and the Soft button on the channel will change to a *HOLD* function. The operator now has the option to place the call on hold (pressing the Soft button of the channel) or pressing the left *DROP* button (F) to end the call. If the operator would like to protect the call from being accidentally dropped, pressing the line select button (E) a second time will change the state of the line from *ON AIR* to *LOCKED*. In the locked state, the Soft button changes from *HOLD* to *TALKBACK*, permitting off-air communication to the other end of the line (useful for queuing). Pressing the associated *DROP* button will have no effect on the line. To undo the locked condition, press the line button again.



User Button Option

The iQx 6-Fader with USER buttons sidecar is an optional expansion that provides customizable buttons for the studio. Refer to the configuration in chapter 5 for setup. As for usage, it really depends on how it is set up. There is no predefined button function. If the button is pushed, it will trigger an action at a GPIO port, in channel logic, or within Axia Pathfinder Routing Control. The button will illuminate when the other end triggers the appropriate command. The two sets of five buttons make up a total of ten buttons. Each set of five is setup as a group. So the left set can be configured to interface with a GPIO port and the set on the right can be configured differently. The most useful option is to have the two sets of buttons interface with Pathfinder Core PRO (default). From here, all the tools available through Pathfinder Core PRO can be utilized.

3 Audio Configuration

In the iQx console, we use profiles to categorize audio that is already in the network and to define the behavior of the console. The iQx uses *Source Profiles* to define the audio inputs to the console. The use of *Show Profiles* is to define the layout of the console and in some cases how the console is to be used. As for returning audio into the network, there is a very flexible and easy-to-understand setup. This chapter will cover these three items, Source Profiles, Show Profiles, and Audio Output.

It should be noted that you access the main configuration of the iQx with a browser application (same used to surf the internet) and typing the IP of the iQx in the address (URL) field. When an authentication window appears, type in *user* in the username field and leave the password field blank. This is the default password to the iQx.

OK

Cancel

Source Profiles

System

Status

Setup

Customize

Log

Log History

Log Setup

Backup / Restore

Synchronization

Module Manager

Module information

Brightness control

Console Config

Console Control

Remote

Profile Manager

Sources

Shows

Changes

IO Manager

Outputs

V-Mixer & V-Mode

Stream Status

Source Profiles:

Create

new

Line Input

source.

No source

Line Input

Computer Player

Phone

Codec

Operator Microphone

CR Producer Microphone

CR Guest Microphone

Studio Guest Microphone

External Microphone

Studio Feed

The Source Profile configuration is accessed through a link on the left side column, *Sources*. Pressing this link will change the view to the right of the menu. A fresh iQx will have no profiles and the view would be rather bland. The first step is to select the appropriate source type from the drop down, followed by the *Create* button.

The options to select are:

- ◆ **Line Input** – Used for any general audio source. A GPIO logic port can be used to provide machine start/stop pulses if desired. Refer to GPIO tables (chapter 4) for assigned logic.
- ◆ **Computer Player** – Similar to Line Input but has different logic control that is common with PC automation systems. Pressing the ON button while source is already in the ON state would trigger a NEXT event in the automation. Refer to GPIO section in chapter 4 for more specifics.
- ◆ **Phone** – Defines this source as a hybrid or broadcast phone system input. A summed mono mix-minus is automatically created for the source and controls for Telos products will be available.
- ◆ **Codec** – This profile will generate dual mono mix-minus for the source; one PA feed (right) and one talent feed with talkback (left).
- ◆ **Operator Microphone** – The board operator's mic. It is the source mic for iQ TALK TO functions. Its logic mutes the CR monitors and Preview when ON.
- ◆ **CR Producer Microphone** – Used for in-studio producer's mic positions. It has associated GPIO logic that can operate TALK TO functions from a remote producer's panel. It also mutes CR monitors and Preview when ON.
- ◆ **CR Guest Microphone** – Used for any other guest mic in the control room. Its associated logic automatically mutes the CR monitors and Preview when ON.
- ◆ **Studio Guest Microphone** – Used for any mic located in a separate studio. Its logic mutes the Studio monitors when ON.
- ◆ **External Microphone** – Used for any mic located outside of a CR or studio. It functions like a Guest microphone but does not include any muting logic.
- ◆ **Studio Feed** – Defines a source that receives a backfeed and generates an IFB backfeed in return. This is intended for an external studio source that you wish to generate a talkback channel for.

After pressing the *Create* button, the display will change to show all the options associated with the source type that was selected. Some options are common to all source types, while other options are exclusive to other source types.

Source Profile Options

Source Name

Each Source needs a name for display on the console's channel displays. Names can be up to 10 characters long and may include spaces or underscore characters. When you create a source profile, these options will appear. Once saved, you can return to these options by selecting the source from the list view of profiles. An example of how to do this is covered in the Quick-Start Guide at the beginning of this manual

Source Input

This setting defines the AoIP source or internal DSP source. Which option you select dictates the input into the *Primary Source* field. The options are:

- ◆ **Livewire** – the original AoIP.
- ◆ **Livewire Backfeed** – the backfeed streams produced by consoles can be ingested if your application requires it.
- ◆ **Livewire Surround** – the original Livewire Surround stream had a stereo package in the last two channels, these two channels are ingested with this input type.
- ◆ **AES67 SIP** – The unicast options within the AES67 standard.
- ◆ **AES67 Multicast** – Thought to be the most typical AES67 stream type.
- ◆ **AES67 Multicast 16-bit** – 16-bit switch to help differentiate from those streams that match in bit utilization from 24-bit.
- ◆ **Sine 1kHz -20dBFS** (internal DSP source) – A tone for audio testing.
- ◆ **V-Mixer** (internal DSP source) – An internal mixer for special use cases.

Primary Source

This field is where specific AoIP information is applied. The information can be manually entered or in some cases the use of the browse button can be used to show a list of advertised sources.

- ◆ **Livewire** – a Livewire channel number (1-32767) is expected. The Browse button will show any sources discovered through the Livewire discovery channel.
- ◆ **Livewire Backfeed** – a Livewire channel number (1-32767) is expected. The Browse button will show any sources discovered through the Livewire discovery channel that happen to be Backfeed streams.
- ◆ **Livewire Surround** – a Livewire channel number (1-32767) is expected. The Browse button will show any sources discovered through the Livewire discovery channel that happen to be Livewire surround type streams.
- ◆ **AES67 SIP** – a SIP URI scheme is expected (sip:user@host) . The browse button will identify any that are discovered.
- ◆ **AES67 Multicast** – a multicast address is expected.
- ◆ **AES67 Multicast 16-bit** – a multicast address is expected.

Signal Mode

Determines whether source will be treated as mono or stereo.

- ◆ **Stereo** feeds incoming L/R signal to left and right channels of assigned mixes.
- ◆ **Left** feeds incoming left channel to both channels of assigned mixes.
- ◆ **Right** feeds incoming right channel to both channels of assigned mixes.
- ◆ **Sum L+R** creates mono mix of incoming stereo source and feeds it to both channels of assigned mixes.

Signal Phase

Determines whether source will maintain normal or inverted phase relation.

Signal Mode for Record Bus

The Record mix (bus) is a special variation of PGM 4 that is post-fader but pre ON/OFF (default) . This field defines how the source will feed the mix. The options are:

- ◆ **Stereo** – delivers this source in stereo to the Record mix (default) .
- ◆ **Sum to Left** – sums L+R of this source and delivers the sum to the Left channel of the Record mix.
- ◆ **Sum to Right** – sums L+R of this source and delivers the sum to the Right channel of the Record mix.

The last two options give the ability to create a split Record, which is commonly used for phone recorders. The Operator's mic can be recorded on one side of a stereo channel while all other sources are recorded on the other side.

Record Insert Mode

Is an option to feed the Record mix pre-fader. By default the Record mix is post-fader and may not be desired in some applications, thus this option.

Input Trim Gain

This setting allows you to specify an amount of gain from -25 to +25 dB. Fader Trim Gain is applied in the signal path just before the fader.

Panorama Position

Lets you preset pan settings for the source. Pan is variable in 49 steps, center being 0, far left -24, and far right +24. This setting can be adjusted on the fly by the operator if permitted by show profile settings.

Audio Delay

This setting applies a delay in the audio for use in video voice sync or additionally in AES67 link offset adjustment.

Microphone Processing (Microphone Types)

Dynamic audio processing is available to Microphone source types.

Microphone Processing:

☐ **Noise Gate:**
Threshold [-50 .. 0 dB]: dB
Depth [-30 .. 0 dB]: dB

☐ **Compressor:**
Threshold [-30 .. 0 dB]: dB
Ratio (1:1 .. 16:1): :1
☐ Freeze Mode

☐ **De-Esser:**
Threshold [-20 .. 0 dB]: dB
Ratio [1:1 .. 8:1]: :1

Post-Processing Trim Gain [-20 .. 20 dB]: dB

The check box to the left of each component, *Gate*, *Compressor*, and *De-Esser*, enables the component. The options below are specific to the component.

- ◆ **Noise Gate**
 - ◇ **Threshold**, down to -50 dB, at which the Noise Gate activates to attenuate the input signal that fall below this value.
 - ◇ **Depth**, up to -30 dB, of attenuation to apply to the input signal when the specified Threshold is met.

- ◆ **Compressor**
 - ◇ **Threshold**, down to -30 dB, at which the Compressor activates to attenuate input signals above this value.
 - ◇ **Ratio**, the aggressiveness of the compressor, up to 16:1. A zero change ratio would be 1 to 1 (1:1). Most aggressive of 16 to 1 (16:1) would be 16 dB of input change to result in 1 dB of output change.
- ◆ **De-Esser**
 - ◇ **Threshold**, up to -20 dB, at which the De-Esser activates to attenuate sibilance.
 - ◇ **Ratio**, aggressiveness of the De-Esser, up to 8:1.
- ◆ **Post-Processing Trim Gain**, a gain stage after the processing.

Equalizer

Three band quasi-parametric equalizer for audio tonal adjustment.

☐ **Equalizer:**

Band:	Low [20 .. 320 Hz]	Mid [125 .. 2000 Hz]	High [1250 .. 20000 Hz]
Frequency:	25 Hz	1000 Hz	16000 Hz
Gain [-25 .. 15 dB]:	+0.0 dB	+0.0 dB	+0.0 dB
Mode:	Auto	Peak	Peak ▼

The checkbox to the top left enables the Equalizer. Set the frequency of the three bands and the gain value for each band.

The low band will switch between peak or shelf based on being a gain or attenuation on the low band. The High band offers the choice of being Peak or shelf.

Source Availability

The check boxes determine which faders (or other inputs) you'll allow the source to be assigned to. For example, if your source is a Control Room mic, and you only want it to appear on the left-most fader, you would check only the box marked Channel 1. For sources only for monitoring, uncheck all Channels to prevent the source to go into final mix (potential audio loop). The All Channel check box is a quick toggle option for selecting all or none.

Fader Mode

Defines fader start actions and start logic:

- ◆ **Normal** conforms to a common method of requiring users to press buttons to define the on and off state of the channel.
- ◆ **Fader Start** follows an alternate standard of activating the channel and associated machine logic when the fader is raised from -∞.
- ◆ **Fader Start with Arm** works just like the Normal setting when the fader is not set to minimum position. When the fader is set to the minimum position, it can be in an Armed state. This armed state is entered by pressing the ON Button when the fader is down. The armed state means that the channel will go to the ON state when the fader is moved up. When in Arm state, the channel OFF lamp will be illuminated. The channel ON lamp will flash in "wink" mode—a long ontime and short off-time.

For all fader modes, pressing the OFF button at any time will place the channel in the OFF state.

Preview Mode

Permits choosing whether Preview acts in CUE (pre-fader) or SOLO (post-fader) mode.

Preview Switching

Check all options that apply to the preferred workflow.

- ◆ Channel ON turns Preview OFF
- ◆ Preview ON turns Channel OFF

Auto-Start Timer

When selected, the count-up timer will start when the channel is turned ON and the counter is set to *Auto* (Show Profile setting).

Logic Port

Option to disable GPIO machine logic to source device or enable it in two modes. Exclusive mode permits GPIO control associated with only a single fader. Shared mode allows more than one user to send ON/OFF/START signals to the source via GPIO. Some profile types support an optional “GPIO Ready mode” and there would be a check box to enable this option. This allows devices such as CD players or tape machines to signal their READY state to the operator through the OFF button LED. If a device is cueing, its OFF lamp will not be lit. Linking the device’s OFF lamp to the Ready command means that a machine must be cued and ready before the OFF lamp illuminates. If enabled, the OFF lamp illuminates when the fader is OFF and the Ready command is active on the GPIO. If disabled, the OFF lamp illuminates normally whenever the fader is turned OFF. Refer to GPIO section for more specifics on GPIO for each source type.

GPIO Hybrid Control (Phone Type)

Define the method of control to the phone interface.

- ◆ Soft-Key Only (default) uses only the software button (*soft* button) next to the *Preview* button to control the state of the interface (on hook / off hook).
- ◆ Channel ON Answers Hybrid - GPIO control of a phone interface would be triggered when the ON state is changed.
- ◆ Channel ON or Preview Answers hybrid - GPIO control of a phone interface would be changed based on ON state or Preview state of the channel.

Refer to GPIO section for more specifics on GPIO call outs for source profiles.

Feed to Source (Phone, Codec, and Studio Feed Type)

The backfeed (mix-minus) for a source is automatically generated, but this field defines the content of said backfeed. As an example, a phone mix-minus would be the content of Program 1 minus the phone, so this field would be set to Program 1 and the workflow would be achieved. The options available are:

- ◆ Disabled - no backfeed.
- ◆ Auto (Pgm-1/Phone) feeds the Program 1 mix (minus the source) when this source’s fader is ON, and feeds the “Phone” mix (minus the source) when the fader is OFF. Remember, the Phone mix is a special variant of Program 4. The Phone mix is pre-fader and pre-on/off to allow speaker-phone style operation through the Operator’s mic.
- ◆ Program 1 always feeds the Program 1 mix, minus the source, regardless of the on/off state of the channel.
- ◆ Program 2 feeds the Program 2 mix, minus the source.
- ◆ Program 3 feeds the Program 3 mix, minus the source.
- ◆ Program 4 feeds the Program 4 mix, minus the source.
- ◆ Phone feeds the Phone mix, minus the source. Phone is a pre-fader and pre-on/off variation of Program 4
- ◆ Talk Only is a “silent backfeed” that operates when the Talk To function is engaged, creating an IFB to the source.

Feed to Source Sim Gain (Phone, Codec, Microphone, and Studio Feed Type)

Talkback is sent over dimmed backfeed audio (default of -10dB) . If you want a hard interruption (Talkback interrupts Program audio) , set the value to -inf. Adjustment to the attenuation amount is in the range is 0dB to -30dB.

Hybrid for Telos Phone (Phone Type)

This field helps define the Telos VX configuration to interface with. The default of *None* indicates no interfacing (assume GPIO control) . If interfacing with a Telos VX system, the options to choose are *Fixed* or *Selectable* followed by the number of index value as defined in the VX Studio. Connection to the VX will be defined in the Show Profile section.

Live Controls

The group of radio buttons permits “live” functions for the Options control on each of the console’s fader strips. Available options will vary depending on the source type. One setting may be defined as the Primary or “live” control, which means that the Options control will vary that setting as soon as you adjust the knob. For example, if PANORAMA has been defined as the Primary function, simply rotating the Options control will pan the source. The other functions may be enabled or disabled as desired. Those that are Enabled may be chosen by the operator “on-the-fly” by depressing the Options control. When the control is pressed, the display will list the options that you’ve checked as Enabled in this section of the configuration. Pressing the Options control selects the highlighted option and turning the Options control will adjust the parameter for that option. Depending on the enabled options, the operator will be able to adjust Source, Gain, Mode, Pan, Dynamics, Equalizer, and Backfeed.

Show Profiles

Show profiles define how the console is laid out channel by channel and some behavior items, which will be covered in this section. Access through a link on the left side column, *Shows*. Pressing this link will change the view to the right of the menu. A fresh iQx will show four shows that are not defined.

System

Status

Setup

Customize

Log

Log History

Log Setup

Backup / Restore

Synchronization

Module Manager

Module information

Brightness control

Console Config

Console Control

Remote

Profile Manager

Sources

Shows

Changes

Show Profiles:

Logic Port for Show Switching:

☐ Hide [NONE] in show list

Profile Name	Actions		
A:Show	<input type="button" value="Edit"/>	<input type="button" value="Capture"/>	<input type="button" value="Reset"/>
B:Show	<input type="button" value="Edit"/>	<input type="button" value="Capture"/>	<input type="button" value="Reset"/>
C:Show	<input type="button" value="Edit"/>	<input type="button" value="Capture"/>	<input type="button" value="Reset"/>
D:Show	<input type="button" value="Edit"/>	<input type="button" value="Capture"/>	<input type="button" value="Reset"/>

The root page has a few options. The ability to change shows can be done through a GPIO port. The top item allows for defining a GPIO channel, which would also be applied to a GPIO port of an xNode. Doing so will allow the the GPI triggers to change the show and likewise create GPO contacts to identify which show is active. GPIO section covers the pin specific information.

Next item is an option to hide the NONE option from the user at the console. The NONE option is a blank show.

The four profiles can be edited, reset, or “Capture” the current console layout and save it as the show profile. Editing the profile will open a new view with the parameters for the show.

Editing Show profiles

Show Profile:

Show name:

Fader Channels				
#	Source	ON	PGMx	Group Start
1	--- no source --- ▼	<input type="checkbox"/>	<input checked="" type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>	Independent ▼
2	--- no source --- ▼	<input type="checkbox"/>	<input checked="" type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>	Independent ▼
3	--- no source --- ▼	<input type="checkbox"/>	<input checked="" type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>	Independent ▼
4	--- no source --- ▼	<input type="checkbox"/>	<input checked="" type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>	Independent ▼
5	--- no source --- ▼	<input type="checkbox"/>	<input checked="" type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>	Independent ▼
6	--- no source --- ▼	<input type="checkbox"/>	<input checked="" type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>	Independent ▼
7	--- no source --- ▼	<input type="checkbox"/>	<input checked="" type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>	Independent ▼
8	--- no source --- ▼	<input type="checkbox"/>	<input checked="" type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>	Independent ▼

Show Name

A seven character friendly name for the operators to ID their desired profile.

Fader Channels

A table that represents the amount of fader channels enabled in the system. Each row represents a channel and the options are to select a *source* profile (A) that loads when the show is called. Use the drop-down to select from the source profiles that have been enabled for the channel.

Fader Channels				
#	(A) Source	(B) ON	PGMx	Group Start (C)
1	--- no source --- ▼	<input type="checkbox"/>	<input checked="" type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>	Independent ▼

In the center (B) the row is divided into two rows. The top row is for enabling the option at show profile load, while the bottom is for locking the state so the operator is not able to make changes to the item. The control items here are the ON state of the channel and the four PGM mix buttons. If the first PGM was checked at the top and bottom, at show load the Program 1 mix would be engaged and the operator would not be able to disable the state of the Program mix.

The Group Start (C) section links channel ON states. Define as many *Master* and *Slave* channels. ON state changes on a *Master* will reflect in all other *Master* or *Slave* channels.

Phone Connection

Phone Connection

Phone Server IP:
format: [user[:pass]@]host

Studio Name:

Show Name:
leave blank to use current

Interfacing with a VX system is configured in the Phone Connection section. Having each show profile with the option to connect to either a different server or different studio adds flexibility.

The Phone Server IP field is for the IP address of the Telos VX product. The Studio Name field is a *Studio Configuration* defined in the VX. The Show Name is optional, if declared, the VX Show will be made the active VX show. Some prefer to leave this field empty and allow for VX show calls from another means.

The Phone Connection settings need to be in place in order to use Fixed or Selectable Phone types.

Monitor Section

	CR Monitor	CR Headphones	Preview	Studio Monitor
Volume (-85 .. 0 dB):	<input checked="" type="checkbox"/> Set <input type="text" value="-30.0"/> dB	<input checked="" type="checkbox"/> Set <input type="text" value="-30.0"/> dB	<input checked="" type="checkbox"/> Set <input type="text" value="-30.0"/> dB	<input checked="" type="checkbox"/> Set <input type="text" value="-30.0"/> dB
Volume adjust:	<input checked="" type="checkbox"/> Allowed	<input checked="" type="checkbox"/> Allowed	<input checked="" type="checkbox"/> Allowed	<input checked="" type="checkbox"/> Allowed
Dim gain: (-inf, -30 .. 0 dB)	<input type="text" value="-12.0"/> dB	<input type="text" value="-12.0"/> dB		<input type="text" value="-12.0"/> dB
Gain in Muted state: (-inf, -85 .. 0 dB)	<input type="text" value="-inf"/> dB		<input type="text" value="-inf"/> dB	<input type="text" value="-inf"/> dB
Source:	<div>--- no source --- ▼</div>			<div>--- no source --- ▼</div>
Source selection:	<input checked="" type="checkbox"/> Allowed			<input checked="" type="checkbox"/> Allowed
Preview in:	<input type="checkbox"/> Enabled	<input type="checkbox"/> Enabled		
Preview mode:		<div>Normal ▼</div>		
Logic port:	<input type="text" value="0"/>			<input type="text" value="0"/>

The monitor section is divided into four columns for each of the monitoring points;

- ◆ CR Monitor - the control room is where the surface resides
- ◆ CR Headphone - the operator of the surface
- ◆ Preview - the Preview (or cue) output
- ◆ Studio Monitor - the external studio to the surface, commonly used as the talk studio.

There are nine rows of items that can be set;

- ◆ Volume - set the volume to a defined value on show load or do not set (keep existing level)
- ◆ Volume Adjust - allow the operator control of the volume (effectively a lock option)
- ◆ Dim Gain - define the attenuation during a dim event. -inf is silence of the main content where 0 is no attenuation.
- ◆ Gain in Muted State - to defeat the mute state common in radio, for example in TV applications.
- ◆ Source - define the default source to load on show load.
- ◆ Source sSelection - allow the operator control of source selection.

- ◆ **Preview In** - define if the Preview mix is mixed into other monitor points.
- ◆ **Preview Mode** - define if Preview feeds the operator headphone in stereo or mono mix to one channel.
- ◆ **Logic Port** - define the Livewire channel number for the monitor logic. This value would be applied to a GPIO port. Refer to the GPIO section for more specifics on GPIO logic of the monitors. This item is commonly used for ON AIR lights.

External Inputs

Most of the monitor section is geared around local DSP-produced sources. In some cases there is a need to utilize a source that is in the AoIP network. This section allows for defining those sources to be called for local use. Note, that a source profile is needed to be defined to show up in these drop downs. As noted in Source Profile configuration, the *Source Availability* is used to help manage if sources are to be used for mix channels, monitor, or both.

- ◆ **External 1 (and 2)** - the monitor section has two EXT buttons as preset buttons to route an external source to the monitors. Defining a profile from the drop-down sets the source to the EXT button. Typically used for ON AIR tuners to monitor what is being transmitted on air.
- ◆ **External Talk** - used to defeat the default nature of TALKBACK audio originating from operator microphone channel. The default assumes that the operator microphone is assigned to a channel on the console. In some cases, the operator would never contribute audio to a production mix and thus using a channel is costly. This option allows for defining the operator microphone (or any other source) to be the source when a TALKBACK function is called.
- ◆ **External Preview** - defines the source that would be applied to the Preview mix if an EXT PREVIEW logic state was engaged. This can be used for Intercom (or other) audio to enter into the monitor section of the console. Triggering EXT PREVIEW is covered in the GPIO section.

External Inputs			
	External 1	External 2	External Talk
Source:	--- no source --- ▼	--- no source --- ▼	--- no source --- ▼
Record Mode			
Record mode:	Disabled ▼		
Logic port:	0		
User Interface			
Timer mode:	Manual ▼		
Preview interlock mode:	<input checked="" type="checkbox"/> Enabled		
Switch-able meter source:	Follow Monitors ▼		
Logic port for timer:	0		
Talk to CR level (-30 .. 10 dB):	-12.0 dB		
Control lock map (selected actions are allowed):			
<div> <input checked="" type="checkbox"/> Meter source <input checked="" type="checkbox"/> Talk to backfeeds </div> <div> <input checked="" type="checkbox"/> Timer mode <input checked="" type="checkbox"/> Talk to studio </div> <div> <input checked="" type="checkbox"/> Timer controls <input checked="" type="checkbox"/> Preview in HP </div> <div> <input checked="" type="checkbox"/> Record </div>			
<div> <div>Apply</div> <div>Ok</div> <div>Cancel</div> </div>			

Record Mode

The surface has a record button and the function of that button is defined here.

- ◆ **Disabled** - effectively the button does nothing and there is no Record mode
- ◆ **Enabled** - toggle the historical Axia record mode. When in record mode, the monitor section will change to the specialized Record mix, muting the CR monitors, and trigger GPIO. Refer to GPIO section for GPIO-specific information.
- ◆ **CR Ready** - use to indicate through GPIO of a ready condition. GPIO pin 5 will latch in this mode when in record mode.
- ◆ **CR Ready with Auto-Reset** - same as CR Ready but the latch will reset the moment a mute state is triggered.
- ◆ **GPIO Button** - treat the button as a GPIO button. GPIO pin 1 of a port will follow the button press. Inverse, GPIO pin 11 of the GPIO port will trigger the illumination of the *Record* button.

The last three require a Logic port value to be applied (number between 1 and 32767).

User Interface

- ◆ **Timer Mode** defines if the count up timer on the overbridge is triggered manually or by those source profiles with timer control enabled.
- ◆ **Preview Interlock Mode** allows for the function of pressing a second Preview button that would release the earlier Preview button. If not enabled, each Preview button pressed would need to be pressed again to remove the channel from preview.
- ◆ **Switchable Meter Source** defines what source the last audio meter on the overbridge follows.
- ◆ **Logic Port for Timer** allows for remote manual control of the timer through GPIO. Refer to the GPIO section for specifics.
- ◆ **Talk to CR Level** defines the gain (attenuation or amplification) of the *Talk to Control Room* audio.
- ◆ **Control lock** allows control of the selected item. Any item not checked will not function for the operator.

Capture a Profile

Pressing the Capture button at the root Shows page will capture the current state of the console and load those values to the Show Edit page, allowing you to make changes before saving or simply saving the current configuration.

Changes

The page will show what is different between profile settings and active settings and give options in storing the volatile active settings to memory.

Audio Outputs

The outputs page is accessed via the *Outputs* link under IO Manager. The page provides 24 outputs that can each be defined to the facility needs. The default setting is for unused outputs to be reserved for *Auto Backfeed*. When a source is loaded that calls for a backfeed, one of the outputs from the pool of *Auto Backfeed* will be assigned (and configured) as the backfeed. The profile would need to be unloaded to reclaim the output. If no pool is available, the channel with the calling profile will show an error condition on the display. Other options for the *Output* column are the audio content generated by the DSP.

iQx (iQx-E200D5) Control Center LU Meter ▲

System
 Status
 Setup
 Customize
 Log
 Log History
 Log Setup
 Backup / Restore
 Synchronization
Module Manager
 Module
 information
 Brightness control
 Console Config
Console Control
 Remote
Profile Manager
 Sources
 Shows
 Changes
IO Manager
Outputs
 V-Mixer & V-Mode
 Stream Status

Network AoIP Outputs:

#	Output	Name Override	AoIP Type	Channel / IP	Payload Type	SDP
1	Program 1		Multicast	0	Disabled	-
2	Program 2		Multicast	0	Disabled	-
3	Program 3		Multicast	0	Disabled	-
4	Program 4		Multicast	0	Disabled	-
5	Record		Multicast	0	Disabled	-
6	Phone		Multicast	0	Disabled	-
7	CR Monitor		Multicast	0	Disabled	-
8	CR Monitor Direct		Multicast	0	Disabled	-
9	CR Headphones		Multicast	0	Disabled	-
10	Preview		Multicast	0	Disabled	-
11	Talk to CR		Multicast	0	Disabled	-
12	Talk to External		Multicast	0	Disabled	-
13	Studio Guest HP		Multicast	0	Disabled	-
14	Studio Monitor		Multicast	0	Disabled	-
15	Studio Talent HP		Multicast	0	Disabled	-
16	V-Mix		Multicast	0	Disabled	-
17	V-Mode		Multicast	0	Disabled	-
18	Sine 1kHz -20dBFS		Multicast	0	Disabled	-
19	Auto Backfeed		Multicast	0	Disabled	-
20	Auto Backfeed		Multicast	0	Disabled	-
21	Auto Backfeed		Multicast	0	Disabled	-
22	Auto Backfeed		Multicast	0	Disabled	-
23	Auto Backfeed		Multicast	0	Disabled	-
24	Auto Backfeed		Multicast	0	Disabled	-

Backfeed Mode: AES67 Stereo (1ms) ▼

Apply

- ◆ Program 1, Program 2, Program 3, and Program 4 are the four standard output mixes.
- ◆ Record is a special mix driven by the P4 button (phone handset icon) and is a mix which is Pre-ON and Post-Fader.
- ◆ Phone is another special mix driven by the P4 button and is a Pre-ON and Pre-Fader mix.
- ◆ CR Monitor is the control room monitor driven by the monitor section of the console.
- ◆ CR Monitor Direct is a pre-volume-knob source, which is not impacted by mute states and follows the CR Monitor selection.
- ◆ CR Headphone is the operator headphone feed and is driven by the console's Headphone section of the Monitor section.
- ◆ Preview is the preview mix (cue mix) driven by channels that are applied to *Preview*.
- ◆ Talk to CR is audio content of any profile that initiates a *talkback* to the Control Room. This is mixed in with the Preview mix for the operator to hear the audio, but if there is an application for only the *talkback* audio, this is the source to utilize.

- ◆ **Talk to External** is audio content from the designated Talk source when the CR Monitor logic of TALK TO EXTERNAL is triggered (refer to the GPIO section) . As mentioned in the Show Profile section above, the designated TALK source is either the Operator Microphone assigned to a channel or the show profile EXT TALK designation.
- ◆ **Studio Guest HP** is a feed that follows the Studio monitor assignment and is not muted. This source is designed for use with the headphones other than the operator.
- ◆ **Studio Monitor** is the feed also driven by the Studio monitor selection of the monitor section on the console. This source is muted due to mute states and receives *talkback* audio when the TALK TO STUDIO button is pressed.
- ◆ **Studio Talent HP** is similar to Studio Guest HP but the source receives *talkback* audio when the TALK TO STUDIO button is pressed.
- ◆ **V-Mix** is the output of the V-Mixer stage (refer to chapter 5)
- ◆ **V-Mode** is the output of the V-Mode stage (refer to chapter 5)
- ◆ **Sine 1kHz - 20dBfs** is a DSP-generated source of a 1kHz tone at -20dBFS.

Next to the output is a *Name Override* option that is useful if you have multiple consoles in the facility all reporting a Program 1 output. You can override the name to be something like “Studio A Prog1” for easier selection of the correct source.

The *AoIP Type* is the switch for selecting the two different AES67 forms for transmission—specifically *multicast* or *unicast*. When selecting *multicast* (default) , the next field is also required. When selecting *unicast*, a SIP URI will be provided that is given to the other end of the link. The intended configuration for unicast is that the receiving end issues a SIP request from the transmitter device.

The *Channel / IP* field is used to define the multicast value. If unicast is declared, this field will populate with a SIP URI for use in establishing a unicast transfer. In multicast mode, the field can be populated with a multicast address or a Livewire channel number (1 - 32767) . NOTE: When using a Livewire channel number, the device is advertised through the Livewire advertisement protocol. When defining a multicast address, the system determines this is an AES67 source and thus is not advertised. As the AoIP standards mature, we should expect to see a standard for these streams to be advertised through.

The *Payload Type* defines the type of packet. In technical terms, this defines the packet time, or ptime. Ptime is the amount of audio contained within the packet. AES67 packets are defined to be 1 msec. The other option that can be selected is the Livewire Standard Stereo packet (or ptime of 5 msec) , which is a high-efficiency amount (fill the packet to its max allowed size and result in lower bandwidth consumption) . Selecting the Livewire Standard Stereo may be required for interfacing with legacy Livewire equipment.

The last item is the *SDP* file. An SDP file is defined as a method for exchanging AES67 stream information between devices. Select the SDP link to download the file for use with other products or press the UP ARROW to view the file in a text box for copy-and-paste into other products.

At the bottom of the output page is a global drop-down for defining the backfeed payload type. Some versions of Telos VX, for example, will not accept a 1 msec ptime packet and thus this should be configured to *Standard Stereo* (5 msec) for compatibility.

At the very end is the Apply button; any changes you make require the press of the Apply button.

4 GPIO

General Purpose Input Output (GPIO) has been commonplace in the radio studio for years. Though AoIP has revolutionized how the studio gear interfaces with one another, logic states between equipment have still been widely dependant on this older method of wiring. Axia, from its beginning, has made the process of configuring GPIO as plug-and-play as possible. The strategy to achieve this is via defined behavior through profiles. An audio source is defined through the *Source Profile* (chapter 3) and this profile creates GPIO behavior. This behavior is not arbitrary but has been carefully defined with knowledge of radio workflows. Understanding each of the profiles will give the engineer the knowledge on how best to configure the studio console. Each profile defines how the 5 GPO and 5 GPI pins of the 15 pin GPIO ports behave. This chapter will provide tables for various profiles that we introduced earlier.

Specifically, the profiles are:

- ◆ **Line** - generic audio source
- ◆ **Computer Player** - logic commonly used with PC automation systems
- ◆ **Phone** - single-line telephone hybrid workflows commonly desire the line to be seized when a state is changed on the console, like the channel ON or channel Preview state
- ◆ **Codec** - links to remotes may require IFB communication and GPIO may be a means to trigger the appropriate states
- ◆ **Operator Microphone** - the operator of the console has a different function than other microphones
- ◆ **CR Producer Microphone** - a producer may not have access to the console, but may require the same controls as an operator
- ◆ **CR Guest Microphone** - microphone locations could always use ON/OFF/ COUGH (MUTE) /TALKBACK button interfaces at their fingertips
- ◆ **Studio Guest Microphone** - similar to CR Guest but are located in a different room (impact muting differently)
- ◆ **External Microphone** - is used for any mic located outside of a CR or studio. This profile defeats the muting logic
- ◆ **Studio Feed** - defines a source that receives a backfeed, and generates an IFB backfeed in return. This is intended for an external studio source that you wish to generate a talkback channel for
- ◆ **Control Room** - the control room has logic states, most critical, the ON AIR one
- ◆ **Studio** - like the control room, ON AIR and other logic states
- ◆ **Film Cap Buttons** - the sidecar with 10 buttons has GPIO functionality
- ◆ **Record Mode** - the various *Record* button functions available though show profile configuration
- ◆ **External Timer Control** - external control of the count up timer through GPIO
- ◆ **V-Mixer** - Control of the Vmixer ON states through GPIO
- ◆ **Show Profile** - control and status of show profiles through GPIO

The following tables show pin numbers as they would appear on an Axia xNode for install support. The Axia GPIO ports are 15 pin. Some of the pins are not reflected in the tables because the pins of power are common pins as shown below:

Pin 7: GPO common pin. Typically tied to GND by application

Pin 8: GPIO port GND

Pin 9: 5vDC supply

Pin 10: GPI common pin. Typically tied to 5vDC by application

Line Profile GPIO Table

Inputs (GPI)

ID	Name	Pin	Notes
GPI 1	ON Command	Pin 11	Momentary turns ON the console channel, triggers START pulse
GPI 2	OFF Command	Pin 12	Momentary turns OFF the console channel, triggers STOP pulse
GPI 3	PREVIEW Command	Pin 13	Active = PREVIEW ON Not Active = PREVIEW OFF
GPI 4	RESET Command	Pin 14	Momentary turns OFF the channel w/o generating a GPO STOP pulse
GPI 5	READY Command	Pin 15	Latch to indicate READY (OFF button illuminated) for profiles defined with the READY enable

Outputs (GPO)

ID	Name	Pin	Notes
GPO 1	ON lamp	Pin 1	Latching, during ON state
GPO 2	OFF lamp	Pin 2	Latching, during OFF state, and READY command if enabled
GPO 3	PREVIEW lamp	Pin 3	Latching, during PREVIEW state
GPO 4	START pulse	Pin 4	Momentary at transition to ON
GPO 5	STOP pulse	Pin 5	Momentary at transition to OFF

Computer Player Profile GPIO Table

Inputs (GPI)

ID	Name	Pin	Notes
GPI 1	ON Command	Pin 11	Momentary turns ON the console channel, triggers START pulse
GPI 2	OFF Command	Pin 12	Momentary turns OFF the console channel, triggers STOP pulse
GPI 3	PREVIEW Command	Pin 13	Active = PREVIEW ON Not Active = PREVIEW OFF
GPI 4	Not used	Pin 14	
GPI 5	READY Command	Pin 15	Latch to indicate READY (OFF button illuminated) for profiles defined with the READY enable

Outputs (GPO)

ID	Name	Pin	Notes
GPO 1	NEXT pulse	Pin 1	Momentary when ON button pressed and already in the ON state
GPO 2	OFF lamp	Pin 2	Latching, during OFF state, and READY command if enabled
GPO 3	PREVIEW lamp	Pin 3	Latching, during PREVIEW state
GPO 4	START pulse	Pin 4	Momentary at transition to ON
GPO 5	STOP pulse	Pin 5	Momentary at transition to OFF

Phone Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	ON Command	Pin 11	Momentary turns ON the console channel, triggers START pulse if ON is configured to answer Hybrid
GPI 2	OFF Command	Pin 12	Momentary turns OFF the console channel, triggers STOP pulse if ON is configured to answer Hybrid
GPI 3	PREVIEW Command	Pin 13	Active = PREVIEW ON Not Active = PREVIEW OFF
GPI 4	Not used	Pin 14	
GPI 5	READY Command	Pin 15	Latch to indicate READY (OFF button illuminated) for profiles defined with the READY enable. Note: could be use to provide ring indication.

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	ON lamp	Pin 1	Latching during ON state
GPO 2	OFF lamp	Pin 2	Latching, during OFF state, and READY command if enabled
GPO 3	PREVIEW lamp	Pin 3	Latching, during PREVIEW state
GPO 4	START pulse	Pin 4	Momentary based on Hybrid control configuration, to seize line
GPO 5	STOP pulse	Pin 5	Momentary based on Hybrid control configuration, to release line

Codec Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	ON Command	Pin 11	Momentary turns ON the console channel
GPI 2	OFF Command	Pin 12	Momentary turns OFF the console channel
GPI 3	TALK TO CR Command	Pin 13	Active = Routes channel audio to Preview and mutes channel
GPI 4	MUTE Command	Pin 14	Mutes console channel
GPI 5	TALKBACK Command	Pin 15	Active = Initiates Talkback to Source from operator mic or Ext Talk

GPO (Outputs)

INPUTS (GPI)D	Name	Pin	Notes
GPO 1	ON lamp	Pin 1	Latching during ON state
GPO 2	OFF lamp	Pin 2	Latching, during OFF state
GPO 3	TALK TO CR lamp	Pin 3	Latching during active TALK TO CR state
GPO 4	MUTE lamp	Pin 4	Latching during an active MUTE state
GPO 5	TALKBACK lamp	Pin 5	Latching during an active TALKBACK

Operator Microphone Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	ON Command	Pin 11	Momentary turns ON the console channel
GPI 2	OFF Command	Pin 12	Momentary turns OFF the console channel
GPI 3	TALK TO STUDIO Command	Pin 13	Active = TALK TO STUDIO Will MUTE channel
GPI 4	MUTE Command	Pin 14	Mutes console channel
GPI 5	TALK to source in Preview Command	Pin 15	Active = Initiates Talkback to channels that are in PREVIEW from operator mic Will MUTE channel

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	ON lamp	Pin 1	Latching during ON state unless TALK or MUTE active
GPO 2	OFF lamp	Pin 2	Latching, during OFF state
GPO 3	TALK TO STUDIO lamp	Pin 3	Latching during active TALK TO STUDIO state
GPO 4	MUTE lamp	Pin 4	Latching during an active MUTE state
GPO 5	TALK to source in Preview lamp	Pin 5	Latching during an active TALK

CR Producer Microphone Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	ON Command	Pin 11	Momentary turns ON the console channel
GPI 2	OFF Command	Pin 12	Momentary turns OFF the console channel
GPI 3	TALK TO STUDIO Command	Pin 13	Active = TALK TO STUDIO Will MUTE channel
GPI 4	MUTE Command	Pin 14	Mutes console channel
GPI 5	TALK to source in Preview Command	Pin 15	Active = Initiates Talkback to channels that are in PREVIEW from producer mic Will MUTE channel

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	ON lamp	Pin 1	Latching during ON state unless TALK or MUTE active
GPO 2	OFF lamp	Pin 2	Latching, during OFF state
GPO 3	TALK TO STUDIO lamp	Pin 3	Latching during active TALK TO STUDIO state
GPO 4	MUTE lamp	Pin 4	Latching during an active MUTE state
GPO 5	TALK to source in Preview lamp	Pin 5	Latching during an active TALK

CR Guest Microphone Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	ON Command	Pin 11	Momentary turns ON the console channel
GPI 2	OFF Command	Pin 12	Momentary turns OFF the console channel
GPI 3	TALK TO CR Command	Pin 13	Active = TALK TO CR (Preview channel) Will MUTE channel
GPI 4	MUTE Command	Pin 14	Mutes console channel
GPI 5	Not used	Pin 15	

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	ON lamp	Pin 1	Latching during ON state unless TALK or MUTE active
GPO 2	OFF lamp	Pin 2	Latching, during OFF state
GPO 3	TALK TO CR lamp	Pin 3	Latching during active TALK TO CR state
GPO 4	MUTE lamp	Pin 4	Latching during an active MUTE state
GPO 5	Not used	Pin 5	

Studio Guest Microphone Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	ON Command	Pin 11	Momentary turns ON the console channel
GPI 2	OFF Command	Pin 12	Momentary turns OFF the console channel
GPI 3	TALK TO CR Command	Pin 13	Active = TALK TO CR (Preview channel) Will MUTE channel
GPI 4	MUTE Command	Pin 14	Mutes console channel
GPI 5	Not used	Pin 15	

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	ON lamp	Pin 1	Latching during ON state unless TALK or MUTE active
GPO 2	OFF lamp	Pin 2	Latching, during OFF state
GPO 3	TALK TO CR lamp	Pin 3	Latching during active TALK TO CR state
GPO 4	MUTE lamp	Pin 4	Latching during an active MUTE state
GPO 5	Not used	Pin 5	

External Microphone Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	ON Command	Pin 11	Momentary turns ON the console channel
GPI 2	OFF Command	Pin 12	Momentary turns OFF the console channel
GPI 3	TALK TO CR Command	Pin 13	Active = TALK TO CR (Preview channel) Will MUTE channel
GPI 4	MUTE Command	Pin 14	Mutes console channel
GPI 5	Not used	Pin 15	

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	ON lamp	Pin 1	Latching during ON state unless TALK or MUTE active
GPO 2	OFF lamp	Pin 2	Latching, during OFF state
GPO 3	TALK TO CR lamp	Pin 3	Latching during active TALK TO CR state
GPO 4	MUTE lamp	Pin 4	Latching during an active MUTE state
GPO 5	Not used	Pin 5	

Studio Feed Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	ON Command	Pin 11	Momentary turns ON the console channel
GPI 2	OFF Command	Pin 12	Momentary turns OFF the console channel
GPI 3	TALK TO CR Command	Pin 13	Active = TALK TO CR (Preview channel) Will MUTE channel
GPI 4	MUTE Command	Pin 14	Mutes console channel
GPI 5	Not used	Pin 15	

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	ON lamp	Pin 1	Latching during ON state unless TALK or MUTE active
GPO 2	OFF lamp	Pin 2	Latching, during OFF state
GPO 3	TALK TO CR lamp	Pin 3	Latching during active TALK TO CR state
GPO 4	MUTE lamp	Pin 4	Latching during an active MUTE state
GPO 5	Not used	Pin 5	

Control Room Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	MUTE CR Command	Pin 11	Mutes Preview and Monitors while active
GPI 2	DIM Command	Pin 12	Dims Monitors while active
GPI 3	EXT PREVIEW Command	Pin 13	Routes External Preview defined in show to Preview
GPI 4	TALK TO EXT Command	Pin 14	Routes Operator mic or Ext Talk (Show profile defined) source to Talk to External output
GPI 5	Not used	Pin 15	

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	MUTE lamp ON AIR lamp	Pin 1	Latching during CR MUTE state
GPO 2	DIM lamp	Pin 2	Latching during DIM state
GPO 3	PREVIEW lamp	Pin 3	Latching during active PREVIEW
GPO 4	TALK TO EXT lamp	Pin 4	Latching during an active TALK TO EXT
GPO 5	TALK TO CR lamp	Pin 5	Latching during an active TALK TO CR

Studio Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	MUTE Studio Command	Pin 11	Mutes Monitors while active
GPI 2	DIM Command	Pin 12	Dims Monitors while active
GPI 3	RESET Timer Command	Pin 13	Starts or resets Timer on console
GPI 4	Not used	Pin 14	
GPI 5	Not used	Pin 15	

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	MUTE lamp ON AIR lamp	Pin 1	Latching during Studio MUTE state
GPO 2	DIM lamp	Pin 2	Latching during DIM state
GPO 3	START/RESET lamp	Pin 3	Momentary indication of Timer change state
GPO 4	Not used	Pin 4	
GPO 5	TALK TO STUDIO lamp	Pin 5	Latching during an active TALK TO STUDIO

Film Cap buttons Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	BUTTON 1 lamp (top button)	Pin 11	Illuminates the button while active
GPI 2	BUTTON 2 lamp	Pin 12	Illuminates the button while active
GPI 3	BUTTON 3 lamp	Pin 13	Illuminates the button while active
GPI 4	BUTTON 4 lamp	Pin 14	Illuminates the button while active
GPI 5	BUTTON 5 lamp (bottom lamp)	Pin 15	Illuminates the button while active

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	SWITCH 1 (top button)	Pin 1	Active when button pressed
GPO 2	SWITCH 2	Pin 2	Active when button pressed
GPO 3	SWITCH 3	Pin 3	Active when button pressed
GPO 4	SWITCH 4	Pin 4	Active when button pressed
GPO 5	SWITCH 5 (bottom button)	Pin 5	Active when button pressed

Note: When configuring the button to interface with a GPIO port, select the “*Livewire Mirror*” option followed by a Livewire number (1 -32767) in the GPIO Channel field. The corresponding GPIO port should have a matching Livewire number.

Record (Enable) Mode Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	Not used	Pin 11	
GPI 2	Not used	Pin 12	
GPI 3	Not used	Pin 13	
GPI 4	Not used	Pin 14	
GPI 5	Not used	Pin 15	

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	RECORD ACTIVE lamp	Pin 1	Latching while in Record mode
GPO 2	RECORD NOT ACTIVE lamp	Pin 2	Latching while Record mode off
GPO 3	Not used	Pin 3	
GPO 4	RECORD pulse	Pin 4	Momentary when entering Record mode
GPO 5	STOP pulse	Pin 5	Momentary when exiting Record mode

External Timer Control Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	START command	Pin 11	Momentary will start the timer
GPI 2	STOP command	Pin 12	Momentary will stop the timer
GPI 3	RESET command	Pin 13	Momentary will reset the timer
GPI 4	FREEZE command	Pin 14	Latching will freeze the timer view
GPI 5	Not used	Pin 15	

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	START lamp	Pin 1	Momentary at start of timer
GPO 2	STOP lamp	Pin 2	Momentary at stop of timer
GPO 3	RESET lamp	Pin 3	Momentary at reset of timer
GPO 4	FREEZE lamp	Pin 4	Active during freeze state
GPO 5	Not used	Pin 5	

V-Mixer Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	In 1 ON command	Pin 11	Normal: Latch to turn ON Toggle: Momentary for ON/OFF Radio: Last momentary is ON, all others OFF
GPI 2	In 2 ON command	Pin 12	
GPI 3	In 3 ON command	Pin 13	
GPI 4	In 4 ON command	Pin 14	
GPI 5	In 5 ON command	Pin 15	

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	In 1 ON lamp	Pin 1	Active when ON
GPO 2	In 2 ON lamp	Pin 2	Active when ON
GPO 3	In 3 ON lamp	Pin 3	Active when ON
GPO 4	In 4 ON lamp	Pin 4	Active when ON
GPO 5	In 5 ON lamp	Pin 5	Active when ON

Show Profile Profile GPIO Table

GPI (Inputs)

ID	Name	Pin	Notes
GPI 1	None command	Pin 11	Changes to the blank profile
GPI 2	Profile A command	Pin 12	Change to the A profile
GPI 3	Profile B command	Pin 13	Change to the B profile
GPI 4	Profile C command	Pin 14	Change to the C profile
GPI 5	Profile D command	Pin 15	Change to the D profile

GPO (Outputs)

ID	Name	Pin	Notes
GPO 1	Profile None lamp	Pin 1	Active when None
GPO 2	Profile A lamp	Pin 2	Active when A
GPO 3	Profile B lamp	Pin 3	Active when B
GPO 4	Profile C lamp	Pin 4	Active when C
GPO 5	Profile D lamp	Pin 5	Active when D

5 System Configuration

Console Config

Beyond the main 8-fader surface with a monitor section, the system can be expanded with sidecars connected to the underside of the main. There are three options of sidecars, 8-Fader expansion, 6-fader expansion with Phone control, and the 6-fader expansion with User buttons. The underside of the main provides two connection points. The system needs to be configured to use the ports, and this is done through the *Console Config* link.

Console Configuration:

Fader Order	Port	Function	Status
▲ ▼	Console 1	Console Main ▼	Connected: Console Main. Ok! S1 CH01..CH08
▲ ▼	Console 2	Console Phone ▼	Connected: Console Phone. Ok! S1 CH09..CH14
▲ ▼	Console 3	Console Ext 6 ▼	Connected: Console Ext 6. Ok! S1 CH15..CH20

Apply

User buttons:

ID	Role	GPIO Channel
3. Left	User (LWCP) ▼	0
3. Right	User (LWCP) ▼	0

Apply

What is presented will depend on what is connected. The Function column provides drop-downs to assign the port to an expansion option. Depending on which options are selected, additional options towards the bottom of the page will appear. For example the 6-Fader with User buttons will present *User buttons* configuration. The First column, *Fader Order*, offers arrows to move the ordering of the different surface modules. This is so the system numbers the faders in a logical order.

The *User buttons* options provide configuration of the two sets of 5 buttons. The options are *User (LWCP)*, *Livewire*, *Livewire Mirror*.

- ◆ User (LWCP) is the configuration when planning to use Pathfinder Core PRO to interface with the buttons for customization of workflows for the facility.
- ◆ Livewire with a Livewire number (1-32767) in the GPIO channel field is the configuration when using the buttons in “node” logic mode. This is to say the buttons play the role of a GPIO port and interface with console logic. The Livewire number would be shared with the Source number to be controlled.
- ◆ Livewire Mirror with a Livewire channel in the following GPIO Channel field is the configuration when using the buttons in “console” logic mode. A reciprocal mode allowing the buttons to interact with GPIO ports. A button-press would result in a GPO action and likewise a GPI event would result in illumination of the button. This also requires any GPIO ports in the network share the same Livewire number assignment.

V-Mixer and V-Mode

The V-Mixer and V-Mode are small utilities included that help with odd audio needs every studio runs into.

V-Mixer:

#	Control Type	Logic Port	Gain
V-Mix Master	Disabled ▼	0	-25.0 dB

#	Source	Status	On	Gain
In 1	--- no source --- ▼	Idle	<input type="checkbox"/>	+0.0 dB
In 2	--- no source --- ▼	Idle	<input type="checkbox"/>	+0.0 dB
In 3	--- no source --- ▼	Idle	<input type="checkbox"/>	+0.0 dB
In 4	--- no source --- ▼	Idle	<input type="checkbox"/>	+0.0 dB
In 5	--- no source --- ▼	Idle	<input type="checkbox"/>	+0.0 dB

V-Mode:

Input: ▼

Mode: ▼

The V-Mixer is a 5-input mixer with an ON switch and gain stage for each input. The V-Mixer output goes through a final gain stage. The V-Mixer inputs are defined through Source Profiles. The ON states of the inputs can be controlled through GPIO in three different modes, *Normal*, *Toggle*, or *Radio*. Assign the Logic Port a Livewire number and define the same value to a GPIO port. Chapter 4 covers the GPIO behavior.

- ◆ **Normal** - an active GPI will be an ON state
- ◆ **Toggle** - momentary GPI will toggle ON/OFF states
- ◆ **Radio** - the last GPI will be the active ON channel

V-Mode accepts one of the main mixes as input and can produce a V-Mode output that is a manipulation of the channels as defined in the Mode options.

Brightness Control

Settings that impact the illumination of different objects of the console. Normal operation and dim operation are two states. A dim state is entered after a timeout of no operation.

Synchronization

It should be noted that the iQx is an AES67 console and thus requires PTP for system time. The synchronization page is mostly a status page providing information on the PTP source. It is recommended to have a PTP grandmaster that supports real time as this time is ingested by the system and impacts log time stamping and clock presentation. If there are audio problems in the system, verify that this sync information indicates proper operation. Sync status would show a green SYNC. Otherwise there is a problem with the grandmaster or the network which the iQx is connected.

Backup & Restore

All the hard work in setting up profiles and the smaller details are saved in a configuration backup file which can be downloaded from the *Backup / Restore* link. From the same page, a backup file can be selected and uploaded into the system. A backup is a recommended step after first install and after any major changes to the configurations. A standard practice is to also grab a backup prior to any system update.

At the bottom of the options is also a Factory Reset button. This clears most things.

General System Setup

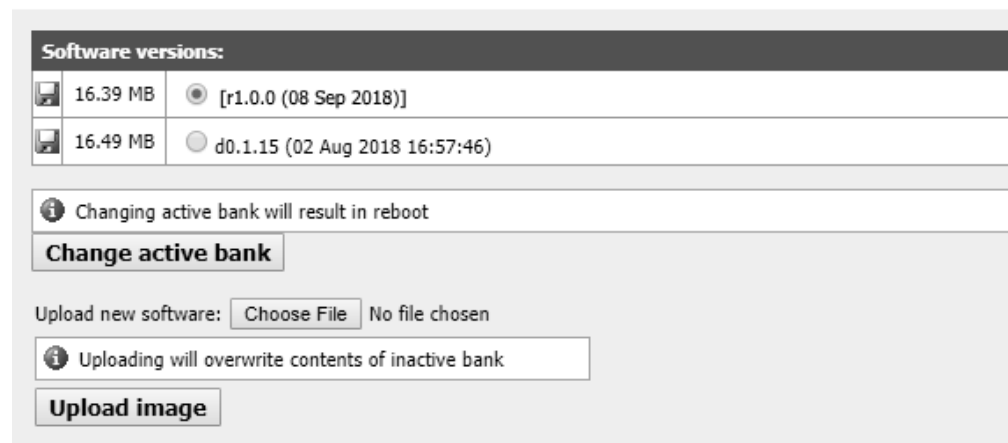
The *Setup* link provides Network configuration items which some of this should already be in place if you are accessing the setup page. Items that could use some work might be the *Hostname* (defaults is last three octets of MAC number) or clean up of the items that are setup from the front panel.

The setup page provides password control configuration. The default password is no password (blank) . If it is desired to change the password, it is entered here twice.

Code Updates

The system has two banks for software. These are found at the *Setup* link.

Software update:



The screenshot displays the 'Software versions' section of a system's update interface. It features a table with two rows of software versions. The first row shows a 16.39 MB file with a radio button selected next to '[r1.0.0 (08 Sep 2018)]'. The second row shows a 16.49 MB file with a radio button next to 'd0.1.15 (02 Aug 2018 16:57:46)'. Below the table, there is a warning message: 'Changing active bank will result in reboot'. A button labeled 'Change active bank' is positioned below the warning. Further down, there is a section for uploading new software, which includes a 'Choose File' button and the text 'No file chosen'. Below this, another warning message states: 'Uploading will overwrite contents of inactive bank'. At the bottom of this section is a button labeled 'Upload image'.

Software versions:	
16.39 MB	<input checked="" type="radio"/> [r1.0.0 (08 Sep 2018)]
16.49 MB	<input type="radio"/> d0.1.15 (02 Aug 2018 16:57:46)

Changing active bank will result in reboot

Change active bank

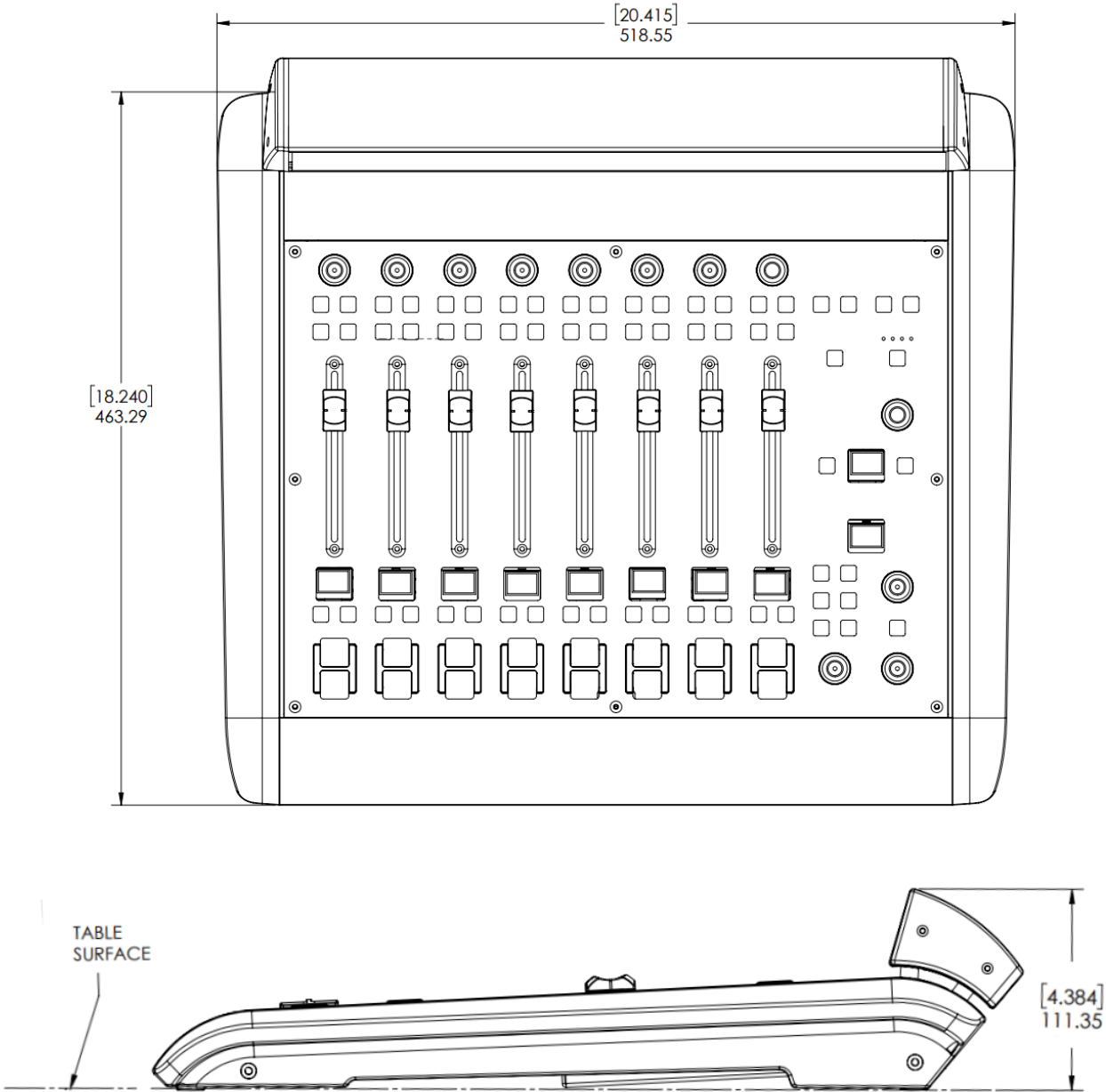
Upload new software: No file chosen

Uploading will overwrite contents of inactive bank

Upload image

To upload a new version to the system, first select the Choose File button to select the new package from the computer. Once selected, press the Upload Image button to load to the non active bank. When ready, select the other bank and press Change active bank. This will cause the system to boot into the other bank. All audio will stop for 45 seconds.

A Appendix: Mechanical



B Appendix: Tear Sheets - Specific Use Case Configurations

AES67 input

1. Define a source profile by selecting the *Sources* link followed by selecting the type of source and the create button. The type of source will only impact muting. The function of GPIO logic or mix minus generation are dependant on Livewire utilization.
2. Provide a useful *Source name* for the operator to identify the source .
3. For *Source input*, select the appropriate AES67 stream type.
 - a. AES67 SIP
 - b. AES67 Multicast
 - c. AES67 Multicast 16-bit
4. In the *Primary source* field, enter the appropriate SIP URI or multicast address for the stream.
 - a. A SIP URI would be in the form of 'sip:<user>@<domain>'. (ie. sip:SRC1@192.168.2.123)
 - b. A multicast value. (ie. 239.192.123.231)
5. If the stream is on a port other than 5004, after the address, enter a colon followed by the port value. (ie. 239.192.123.231:9999)
6. The remaining properties of the source profile are to the user needs. For more information on any of the items, refer to the beginning of Chapter 3 in the iQx manual.
7. The *Audio delay (AES67 Link Offset)* property should be increased in value if the source is jittery.
8. Once the Source Profile is completed, the source can be requested by console channel strips, through Show Profiles, or as inputs into the VMIX.

AES67 output

1. Define an output by selecting the *Outputs* link.
2. On the new page, use one of the available outputs and select the content of the output. The first column, labeled *Outputs*, has a drop down identifying different DSP outputs.
3. Select Multicast or SIP Unicast in the *AoIP* column
4. If Multicast is selected, you must define the multicast address in the *Channel/IP* column.
5. The *Payload Type* should be selected to AES67 Stereo (1ms) .
6. Once Apply button is pressed, for outputs that are multicast, an SDP file download will appear to the fire right. Next to the SDP download is an UP arrow which will show the SDP data in a pop up if you need to copy/paste the value into the receiver unit.

In cases where SIP unicast was selected, a SIP URI will appear in the *Channel/IP* column. This link should be copied and applied to the receiver unit in order for a unicast link to be negotiated.

On Air Light

On Air lights are a function of the Show profile. A show profile contains a Control Room (or Studio) monitor logic. Chapter 4 of the iQx manual has various tables describing GPIO behavior. The Control Room Profile (or Studio Profile) table from that chapter shows the GPIO behavior associated with the GPIO logic. The below information will define the steps to configure and associate a GPIO port to this logic.

1. Select the *Shows* link to open the view of the 4 profiles.
2. Select a profile to edit.
3. In the middle of the new page will be a *Monitor Section*.

Monitor Section				
	CR Monitor	CR Headphones	Preview	Studio Monitor
Volume (-85 .. 0 dB):	<input checked="" type="checkbox"/> Set <input type="text" value="-26.0"/> dB	<input checked="" type="checkbox"/> Set <input type="text" value="-30.0"/> dB	<input checked="" type="checkbox"/> Set <input type="text" value="-30.0"/> dB	<input checked="" type="checkbox"/> Set <input type="text" value="-30.0"/> dB
Volume adjust:	<input checked="" type="checkbox"/> Allowed	<input checked="" type="checkbox"/> Allowed	<input checked="" type="checkbox"/> Allowed	<input checked="" type="checkbox"/> Allowed
Dim gain: (-inf, -30 .. 0 dB)	<input type="text" value="-12.0"/> dB	<input type="text" value="-12.0"/> dB		<input type="text" value="-12.0"/> dB
Gain in Muted state: (-inf, -85 .. 0 dB)	<input type="text" value="-inf"/> dB		<input type="text" value="-inf"/> dB	<input type="text" value="-inf"/> dB
Source:	<input type="text" value="Program 1"/> ▼			<input type="text" value="--- no source ---"/> ▼
Source selection:	<input checked="" type="checkbox"/> Allowed			<input checked="" type="checkbox"/> Allowed
Preview in:	<input type="checkbox"/> Enabled	<input type="checkbox"/> Enabled		
Preview mode:		<input type="text" value="Normal"/> ▼		
Logic port:	<input type="text" value="6707"/>			<input type="text" value="6714"/>

4. The CR Monitor and Studio Monitor both have a Logic port declaration at the bottom of each. Defining a numerical value here will enable the GPIO behavior as defined in Chapter 4. In the image, it shows 6707 applied to Control Room Monitor. Apply a value that works for your facility and press the OK button.
5. Load the show to the console. If the show was already running, reload the show. Any changes made to a profile do not take place unless the profile is called.
6. Log into a device that provides GPIO ports. Navigate to the web page that permits assigning a GPIO port a value. In the case of an xNode, this would be the *Simple Setup* link. In the text box to assign the port a channel, type in the same value that was applied to the *Logic port* property mentioned above. If the web page provides pin status, you should be able to see the first GPO go active when the mute status of the monitor is active. Most Axia GPIO ports provide continuity from the pin to a common pin. The common suggestion is to tie the GPO common to the GND reference on the same connector. This will provide an active low contact to your logic circuit which will drive the ON AIR light. Refer to the documentation of the xNode or other device for current and voltage limits of the closure. In most cases the limits will be good for logic values and not for the power to drive the light. An external device will be needed to interface to the light.

Phone setup to VX

VX defines Studios and Shows within its own configuration. These VX properties are used in the communication between the client (iQx) and the server (VX Engine). The act of the iQx accessing as a client to VX is defined within the iQx Show profile.

1. Select the *Shows* link to open the view of the 4 profiles.
2. Select a profile to edit.
3. In the middle of the new page will be a *Phone Connections* section.

Phone Connection	
Phone Server IP: <small>format: [user[:pass]@]host</small>	192.168.101.200
Studio Name:	iQx
Show Name: <small>leave blank to use current</small>	

4. Enter the IP address of the VX Engine in the *Phone Server IP* field.
5. Enter the VX defined *Studio Name* from the VX configuration.
6. (optional) Enter the VX defined *Show Name* from the VX configuration. A common practice is to leave this blank.
7. Save the settings by pressing OK button. Reload the show for the new setting to take place. Now any Phone source types will access this VX Studio settings.
8. Define Phone Source type (Fixed line or Selectable). These depend on the settings that are defined within the VX Studio configuration.
 - a. Each fixed or selectable interface definition in the VX Engine has a numerical identifier, usually starts at 1 and increments from there. For Selectable lines to operate, you need the iQ 6-Fader Telco Expansion Frame (p/n 2001-00284).
 - i. Select Sources followed by selecting the Phone source type and pressing the Create button
 - ii. Towards the bottom of the new page is *Hybrid for Telos phone* property, select Fixed or Selectable for the needed configuration.
 - iii. Enter the numerical identifier in the following text field.
 - b. In the Source profile, for *Source input* property, select the Livewire option (default). Make sure to enter in the Livewire channel number for the designated Fixed line or Selectable line.
 - c. The mix minus will automatically be created and the content of the mixminus will be based on the Source profile's *Feed to source* property. Refer to chapter 3 of the iQx manual for information on the different options.
 - d. Make sure that there are enough Outputs to support the automatically generated mix-minus. Any none defined outputs are assigned as a pool of auto-generated backfeeds. When a phone source is loaded to a fader, a backfeed is generated. When the source is removed, the backfeed is broken down and the output is available again in the pool.

Split recording of telephone calls

A common practice is to record phone conversation in a two track recording, giving access to post edit the caller or host audio. Within a Source profile is the Signal mode for Record bus property. For each profile we define the routing to the record mix.

1. Select the Sources link.
2. Select the Source profile for the operator microphone. Edit the profile.
3. Find the Signal mode for Record bus property and select 'Sum to Left'.
4. Hit the OK button.
5. Select the Source profile for the phone interface. Edit the profile.
6. Again, find the Signal mode for Record bus property and select 'Sum to Right'.
7. Hit the OK button.
8. Make sure the Record output is one of the assigned outputs and that this network stream is routed to the recording device.
9. Assign the operator microphone and phone interface to fader channels on the console. If already loaded, make sure to reload them so the new setting in their profiles take place.
10. The operator and phone faders should both be applied to P4 (Program 4 assignment button, which has a Phone handset icon).
11. Keep the channels in the off position and move the faders to the unity gain position.
12. You should have recording levels from the operator microphone and the caller with gain adjustment by moving the faders.

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