

IP Intercom

Installation & User's Guide



Manual v1.3.8 - October 2017

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Notices and Cautions

CAUTION:

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

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



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



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

CAUTION: DOUBLE POLE/NEUTRAL FUSING

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

WARNING:

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

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

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

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

WARNUNG:

Die Installations-und Serviceanleitung in diesem Handbuch ist für die Benutzung durch qualifiziertes Fachpersonal. Um Stromschläge zu vermeiden führen Sie keine andere Wartung durch als in dieser Betriebsanleitung aufgeführt, es sei denn Sie sind dafür qualifiziert. Überlassen Sie alle Reparaturarbeiten qualifiziertem Fachpersonal.

Dieses Gerät hat eine automatische Bereichseinstellung der Netzspannung.
Stellen sie sicher, dass die verwendete Netzspannung im Bereich von 100-240V liegt.
Das Symbol ~, falls verwendet, bezeichnet eine Wechselstromversorgung.



Dieses Symbol, wo immer es auftaucht, macht Sie auf nicht isolierte, gefährliche elektrische Spannung (ausreichend um einen Stromschlag hervorzurufen) innerhalb des Gehäuses aufmerksam. Spannungen.



Dieses Symbol, wo immer es auftaucht, weist Sie auf wichtige Bedienungs-und Wartungsanleitung hin. Lesen Sie die Bedienungsanleitung.

ACHTUNG: ZWEIPOLIGE ABSICHERUNG / NULLEITER ABSICHERUNG

Das Netzteil des Gerätes hat eine interne Sicherung eingebaut. Auch wenn die Sicherung durchgebrannt ist, können auf einigen primären Bauteilen noch gefährliche Spannungen vorhanden sein. Wenn ein Austausch der Sicherung erforderlich ist, ersetzen Sie die Sicherung nur mit gleicher Art und Wert für den kontinuierlichen Schutz gegen Feuer.

WARNUNG:

Das Gerätenetzkabel ist die Haupttrennvorrichtung. Die Steckdose sollte sich in der Nähe des Gerätes befinden und leicht zugänglich sein. Das Gerät sollte nicht so angeordnet sein, dass der Zugang zum Netzkabel beeinträchtigt ist. Wird das Gerät in ein Rack eingebaut, sollte eine leicht zugängliche Sicherheitstrennvorrichtung in den Rack-Aufbau mit einbezogen werden.

Um die Gefahr von Stromschlägen zu verringern, darf dieses Produkt nicht Regen oder Feuchtigkeit ausgesetzt werden. Dieses Gerät ist nur für die Benützung im Innenbereich. Dieses Gerät erfordert freie Luftzirkulation für eine ausreichende Kühlung. Blockieren sie nicht die Lüftungsschlitze auf der Geräteoberseite und den Seiten des Gerätes. Unzureichende Belüftung kann das Gerät beschädigen oder Brandgefahr verursachen. Platzieren Sie das Gerät nicht auf einem Teppich, Poster oder andere Materialien welche die Lüftungsöffnungen beeinträchtigen könnten.

Wird das Gerät anders als in der, vom Hersteller angegebenen Weise verwendet, kann der, durch das Gerät gegebene Schutz beeinträchtigt werden.

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.

IP Intercom Manual

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WARRANTY



Telos Alliance Radio Division has an industry-leading five-year warranty. As a company, we are committed to our products and the pursuit of providing better quality to our customers.

Find the Warranty online at www.telosalliance.com/warranty.

UPDATES

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

FEEDBACK

We welcome feedback on any aspect of Fusion, or this manual. In the past, many good ideas from users have made their way into software revisions or new products. Please contact us with your comments.

SERVICE

You must contact Axia before returning any equipment for factory service. We will need your unit's serial number, located on the back of the unit. Axia will issue a return authorization number, which must be written on the exterior of your shipping container. Please do not include cables or accessories unless specifically requested by the Technical Support Engineer. Be sure to adequately insure your shipment for its replacement value. Packages without proper authorization may be refused. US customers, please contact Axia Technical Support at +1-216-622-0247. All other customers should contact local representative to make arrangements for service.

We Support You...

BY PHONE / FAX:

- You may reach our 24/7 Support team anytime around the clock by calling +1-216-622-0247.
- For billing questions or other non-emergency technical questions, call +1-216-241-7225 between 9:30 am to 6:00 PM, USA Eastern time, Monday through Friday.
- Our Fax number is +1-216-241-4103.

BY E-MAIL:

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

VIA WORLD WIDE WEB:

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

REGISTER YOUR PRODUCT

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

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War of the Waves

Dear Valued Customer,

It's with great pride and a tip of the hat to an incredible team that I congratulate you on your new Telos Alliance product. Everything we do here at the radio division of the Telos Alliance is with one end goal in mind: To help broadcasters declare victory in extremely competitive environments. By purchasing this product from us, in essence, you have declared war on your competition.

After all, the majority of Telos Alliance employees were broadcasters themselves once, and the products we've developed over the years have been designed as solutions to specific issues faced on the front lines of our industry. We're right there in the trenches with you and have the weapons you need in your arsenal.

Telos Systems is a catalyst to out-of-this-world sound, with the most powerful and popular broadcast telephone systems in the industry; IP/ISDN codecs and transceivers; plus processing/encoding for streaming audio. We built an industry on the back of these amazing telephony systems, and they are still going strong.

While we at the Telos Alliance never forget our roots, we are also blazing trails in terms of new technologies like stream-encoding and AoIP, so that all types of broadcasters can excel in this ever-evolving digital world. Omnia Audio not only lets you stand out on the dial with your unique signature sound via legendary audio processors, audio codecs, and microphone processing, it lets you give your listeners a better streaming experience across devices with innovative stream encoding/processing software and hardware.

Axia Audio is a driving force behind the AES67 AoIP standard, and its networked AoIP radio consoles, audio interfaces, networked intercom, and software products continue to move AoIP adoption forward and help broadcasters streamline operations with cohesive, smart, and feature-rich AoIP ecosystems.

Last, but certainly not least, 25-Seven has traditionally been known for its audio delays, but its Voltair watermark monitor/processor has made a name for itself more recently as the disruptive product that helped broadcasters take back their ratings and harness the true power of their listening audiences.

You work so hard on your programming day-in and day-out, it deserves technology that will optimize sound and performance at every point in the airchain and online. Armed with Telos products, you have what you need to set your competition squarely in your crosshairs.

With that, I'll leave you to prep your armaments. I hope that you will enjoy your Telos Alliance products for many years to come!

Sincerely,
Frank Foti
CEO, The Telos Alliance



Chapter One:

Introducing Axia

IP-Intercom

IP Intercom Overview

The family of Axia IP-Intercom products includes several models of rack-mounted panels, desktop units, console-mounted modules, and software panels.

Rackmount Stations:

- IC.20 – 20-station panel with OLED displays
- IC.1 – 10-station panel with user-labeled buttons

Desktop Stations:

- IC.20D – 20-station desktop panel with OLED displays
- IC.1D – 20-station desktop panel with user-labeled buttons

Axia Console Stations:

- 20-Station OLED – Element CanBUS module with OLED display that takes two fader positions
- 10-Station OLED – Element CanBUS module with OLED display that takes a single fader position
- 10-Station Film-Cap – Element CanBUS module with user-labeled buttons that take a single fader position
- 20-Station OLED - Fusion module that takes two fader position
- 10-Station Film-Cap - Fusion module that takes one fader position

Software Stations:

- SoftCom – 20-station windows application that uses the Mic input and Speaker output of the PC.

Axia's IP-Intercom family may operate as a standalone intercom system. The IP-Intercom system also integrates with select Axia consoles and routing functions of any existing Axia Livewire network. Axia xNodes can be added to a standalone system to add I/O for integration with other systems.

Axia IP-Intercom equipment introduces AEC (Advanced Echo Cancellation) technology licenses from Germany's Fraunhofer Institute. AEC is a revolutionary echo-cancellation technology which enables full duplex operation without the use of headphones. AES allows simultaneous Talk and Listen without feedback or comb-filtering.

IP-Intercom Stations: Front Panel

The IP Intercom family shares common control interfaces for an easy familiarity to the end user. The requirements will differ between devices so some options are available on some devices while not on others.

Front Panel Connectors (Rackmount and Desktop)

Microphone Jack

A 1/4 inch Tip-Ring-Sleeve receptacle is provided on rackmount and desktop units. The TRS gooseneck microphone is common amongst other brands as well. The Telex MCP-90-8 is a nice popular gooseneck microphone.

Headset Connector

The 4-pin XLR-M panel connector found on rackmount and desktop models is used to connect a standard intercom headset that incorporates both a microphone and a headphone. One example is from Production Intercom, model SMH210. Numerous other manufactures of headsets use the following 4-pin wiring convention:

- Pin 1: Mic common
- Pin 2: Mic hot
- Pin 3: Headphone common
- Pin 4: Headphone hot (mono)

Intercom Station Controls

Each intercom station will have a number of keys and the ability to talk or listen with the assigned channels. Intercom units with OLED displays will use the display to show key assignments and have a button for Talk and another button for Listen. The SoftCom mimics the OLED display functionality. Other models without OLED displays will have a single Programmable button that is configured with a Talk, Listen, or Talk+Listen function. The button caps are manually labeled by either using your choice of labeler or using the “Print labels” function from the intercom configuration interface to print labels on your local PC.

Each hardware model includes a rotary volume control to adjust the audio output of the received audio. Next to the volume knob are two buttons for mute microphone and mute speaker for privacy control.

Select models include the CALLSTACK control which is composed of an OLED display and button(s) for Talk/Listen. The CALLSTACK allows for additional intercom stations to interface the station. The CALLSTACK displays the external station name and the buttons provide the option to continue communication. This is used when all channels are assigned to other stations.

The OLED display in the CALLSTACK is also used with the Select knob, Group button, and Assign button for extra functionality.

A small select of models also include a keypad. The 12-button telephone-style dialing keypad is scheduled to be used for future features.

Livewire Status Indicators (Rackmount and Desktop)

Rackmount and desktop units include four LEDs for status indications. Net indicator is used to show active network connection and will be illuminated under normal use. If no network connection is present, the Net LED will flash. Livewire indicator will illuminate once multicast data is transmitted and received. Sync or Master indicator will be illuminated to indicate the device’s state within the Livewire network. Only a single device would be the master while all other devices are synchronized to the master. The intercom unit will synchronize with the master clock of a Livewire network.

IP-Intercom Stations: Rear Panel

Rackmount and desktop intercom models have Livewire network connection ports which are for connections to a correctly configured managed network. A recessed ID button can be found next to the Livewire connector which is used for IP address setup and panel testing. The underside of Element intercom modules have an RJ45 connection which in this model is used to connect to the Element CanBUS.

The Rackmount models include audio input and output connections at the rear. These connections are a 3-pin XLR-F for microphone, 3-pin XLR-F or 8-pin RJ45 modular jack for analog line level, and a 3-pin XLR-M or 8-pin RJ45 modular jack for analog line level out. The RJ-45 line connection is common in other Axia products for its small form factor. The pin out of the RJ-45 is located in the following table.

PIN	DESCRIPTION
1	Left Channel +
2	Left Channel -
3	Right Channel +
4	not used
5	not used
6	Right Channel -
7	not used
8	not used

The XLR connectors use the standard pin out convention of

Pin 1: Ground

Pin 2: Analog +

Pin 3: Analog –

All inputs are summed together as a single station input.

The rackmount models include a 15-pin control connector, GPIO – General Purpose Input Output, for logic signaling. The operation of this port is discussed in Chapter 2 and the setup of GPIO control is discussed in Chapter 3.

Rackmount and desktop models have an IEC connector for the internal power supply. The AC input accepts voltages from 90-240 VAC at 47-63 Hz.

IP-Intercom Stations: Software

The rackmount and desktop models run the same software and are upgradeable through the HTML interface. The software supports a small “web server” which is used as the interface to configure each station. Two banks of storage are available for software allowing for the upload of a new version while the unit is in use. Selecting the new software to run requires a reboot.

The Element based modules run Intercom software on the DSP engine associated with the Element. The DSP engine is the host to the software and provides an HTML interface for configuration of the Element intercom station. Uploading new intercom software to the DSP engine replaces the running intercom software and does not require any reboot.

Chapter Two:

The Nickel Tour Setup

This chapter is not intended to replace the content of the full manual. The goal here is to expose some of the critical items and get the hurried engineer some level of functionality. Please refer to the operations chapter for an understanding of some of the functions. The setup and configuration chapter goes into full details on the options. Also, there are case examples that can be reviewed to understand the functionality and perhaps give solutions to some questions.

The manual assumes that the customer understands the need for stations to be connected to a correctly configured network per recommendations to support Livewire. If the network switch has not been configured, please refer to switch configuration documentation available on AxiaAudio.com or request support from support@TelosAlliance.com.

Defining each station

The Axia Livewire Intercom is not based on a central device. Each station is a unique device and uses mDNS for discovery of other devices. Therefore a critical step to functionality is correctly defining each station. Once defined, stations can discover each other.

Rackmount and Desktop stations

First step is to define each station with a unique IP address. Download and run bootps.exe which is available from AxiaAudio.com support downloads. Connect the PC to the same network that intercom stations have been connected to. With the application running press the recessed ID button located at the back panel.

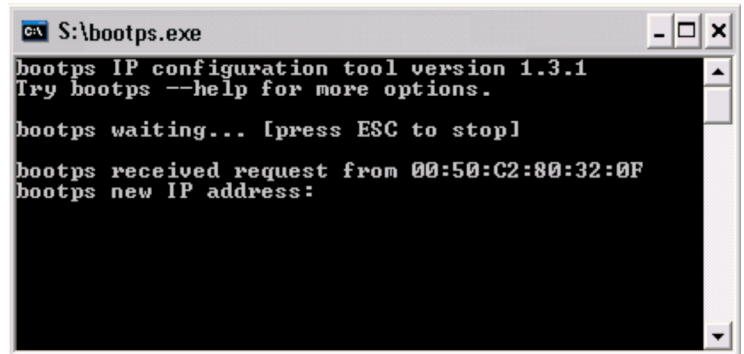


Figure 2-1: Bootps.exe window with address request

When the request is observed, enter in the unique IP address for the device.

With a web browser, enter the address just given to the station to access the device's configuration page. Default authentication is

Username: user

Password: (blank)

Select the *Configure Intercom* link and enter in a 10 character name that defines the station; Prod-01, Ctrl-A, Bob, NewsGuy.

Enter the same value in the Alternate label or another name of equal value; Production, KXYZ, GenManager, Ron B.

Set the Intercom talk to *Live Stereo* and provide a unique Livewire channel value (a number from 1-32767). A suggestion is to use the last octet of the IP address as the preface to the Livewire channel, as in 192.168.100.41 has a channel of 4101.

Press the Apply button.

Axia Console Intercom Module

The console module connects to the controller bus within the surface frame. This module is a controller to the intercom station which is resident in the DSP Engine. To configure this intercom station you must access it through the web interface of the DSP Engine. Select the *Intercom* link from the many links available with the DSP Engine.

Define a unique Intercom Label for the station and an alternate label.

Define two unique Livewire channels. The Preview mix channel number will be used as a console source profile.

Configure a console source profile with the name ExtPreview with Preview mix channel number used as the Primary source. Make sure to select this source profile as the External Preview source in each show profile. If you are not familiar with how to create profiles, please refer to the manual for the console.

If having problems accessing this html interface, make sure that the module is connected to the surface. If the module is not active, the Intercom station in the DSP engine will shutdown.

Softcom

The installation of Axia Livewire Softcom requires also the installation of Apple Bonjour to be installed. The install package should include an installer for bonjour for a 32-bit or 64-bit systems.

Once the application is started, select the Preferences item under the Livewire menu.

In the new window, select the network interface connected to the Livewire network.

Enter in a unique intercom label (default name takes the host name of the PC), and alternate label.

Define the unique Livewire channel source. Once these items are set, the Softcom is ready to participate with other intercom stations.

Defining Key assignments

Intercom stations have keys which are assigned to other stations or external sources. If all stations have been defined as explained above, then assigning keys is the next step. The topic of external sources is covered later in the manual.

Rackmount and desktop

Access the device through the web browser and select the *Key Assignment* link.

For each key, select a station from the drop down list. The drop down list is generated by the discovery process, so all stations that are configured and connected to the network should appear. If this is not the case, insure you

have followed all the items from earlier in this chapter.

Axia console intercom module

Go to the *Intercom* link of the DSP engine.

Under the Intercom Key Assignment heading are the channels available on the module. Using the drop down list, assign a source to the channel. The drop down is created through the discovery process. Don't forget to press the Apply button at the bottom of that channel group.

Softcom

Sofcom has no web interface but uses a front panel interface similar to the IC.20 hardware model. Assign sources (stations) to a channel by selecting a source from the drop down located at the top right. With a source selected, press the Assign button.

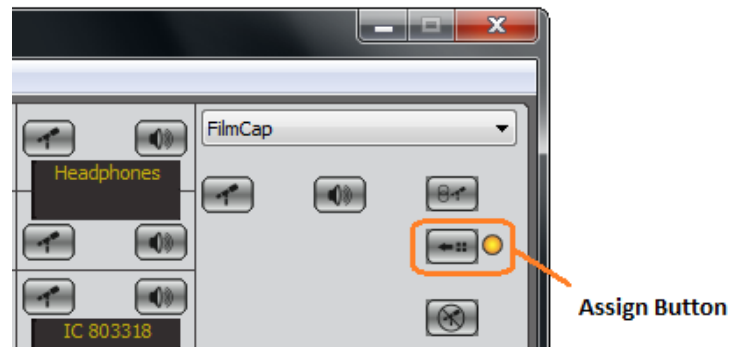


Figure 2-2: Assign Button

A faux LED next to the button begins to blink, this signifies assign mode. Press a button associated with a key to assign the source to the key and end assign mode.

Start talking

With microphones connected, press a channel and speak to the other station. That's it! To get more details on using external sources, interface with other devices, set hierarchy, or more information on the various options; please review the remainder of the manual.

Chapter Three:

Operation

Each intercom station is a standalone device without dependence on a central master unit. Each key can be used for talking to another location, listening to a source, or establishing a bi-directional communication. Some models have more features than others. Some models provide more flexible allowance of channels. This chapter should help you understand how the intercom is intended to operate. The next chapter will guide you in configuration of the intercom station.

Key Controls

The buttons are used to trigger Talk, Listen, or a Talk/Listen state with sources assigned to the keys. The buttons can be used in a “tap-and-latch” mode or a “push-to-talk” (PTT) mode. “Tap-and-latch” feature works by tapping the button to latch the function ON until the button is tapped again, releasing the latch. This mode is useful in allowing the operator’s hands to be free during the duration of a conversation. The mode can be disabled if it is desired to only use the “push-to-talk” mode. The PTT mode requires the operator to maintain the button press for the duration of a Talk or Listen.

Talk

The user will press either a dedicated Talk button or the programmable button to engage a talk session to the source assigned to the key. If the intercom station is configured with a lower trigger value than the station being called, the receiving station is required to acknowledge the call. Talk can be used to call another intercom station, interact with another device through GPIO, or introduce the stations audio through system route changes. Details on these can be found in the case examples later in this chapter.

Listen

The dedicated button on OLED models or the programmable button configured for “listen” only behave

similar to the Talk button, but for receiving audio from the assigned source. This function can be used to monitor external sources such as the Off Air feed, the Program 1 mix from a main control room, the audio feed from a TV set, or any other source within the network that needs to be heard.

Talk/Listen

A Talk/Listen state applies mostly to the programmable button as it is a single button assigned to a key. The Talk/Listen state is the bidirectional communication that takes advantage of the AEC (Advanced Echo Cancellation) technology available from the intercom station. The programmable button, by default, is set to Talk and Listen when pressed.

LED feedback

The buttons have associated LED that illuminate to indicate a mode is active. The Talk button on OLED models has a red LED that is illuminated when the microphone is active. In addition, if the station assigned to a key is “listening”, the red LED would be illuminated to indicate the condition. The dedicated Listen button has a yellow LED. If another station calls, the associated Listen LED (yellow) illuminates during the duration of the call. Once the call is complete, the LED will blink for the defined duration. The default time is 5 seconds and is adjusted per station.

The programmable buttons use different color LEDs to indicate the active state of the key.

Red indicates Talk

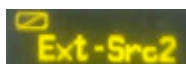
Green indicates Listen

Yellow indicates Talk/Listen

The LED indicators will also blink based on a particular trigger level relation between stations. If one station has a lower trigger value than another, the key for the assigned higher stations will blink. This blink indication demonstrates that the other station is required to acknowledge the call prior to the audio being sent. Once a call is accepted, the calling station is allotted a 5 minute window where any additional calls will be allowed without acknowledgement. This feature is known as the trigger level.

OLED Display

The OLED display will present the 10 character label or alternate label of the station or source assigned to the key. Rackmount and desktop models will present additional indicators on the display if the Talk button has been disabled, if the trigger value is higher than the station, or if the call request has been issued but a response is pending.



Talk Disabled



Call Pending

Other Controls

CALLSTACK

The CALLSTACK key is a flexible key used to allow additional station-to-station communication even though that a station is not assigned. Select models have a CALLSTACK. When one station calls another station and the receiving station does not have the calling station assigned to a channel, the call is processed through the CALLSTACK. If the trigger levels are set accordingly, the call will be heard by the receiving station and the CALLSTACK will show an indication of the calling station on the OLED. The button(s) of the CALLSTACK can then be used to respond back. If the receiving station has a higher trigger value, then the receiving station would need to press the button on the CALLSTACK to acknowledge the call.

NOTE: On models without a CALLSTACK, a call will still be received by the 2nd station by default, but the station will have no indication of who called. The receiving station will have no means to engage with the caller. The caller would need to establish a Talk/Listen state in order for two-way communication. Also be aware of trigger levels. If the receiving station has a higher trigger value and no CALLSTACK, then the call will not be processed because the receiver has no acknowledge button.

On select models, in addition to the CALLSTACK are a *Select* knob, *Assign* button, and *Group* button. The OLED of the CALLSTACK will display the last 4 processed calls. With the use of the *Select* knob, one of the 4 stations can be highlighted and the Talk button can be used to engage in communication. With a station highlighted, pressing the *Assign* button will cause the *Assign* LED to blink and allow for assigning the station to a key by pressing a button of the key. Note that any Talk or Listen settings associated with the key do not change through this process.

The *Group* button enables the talk state to multiple channels with a single button. Assigning channels to the group is done by pressing the *Assign* button so that the assign function is enabled (indicated by the assign LED blinking) and then pressing the *Group* button. The *Group* LED will blink to indicate the ability to define the group. Pressing keys assign the keys to the group. The LED of the key will illuminate to indicate assignment. Pressing the button again will un-assign. Pressing the *Group* button again terminates the assignment to the group.

Other than showing the last processed calls, the CALLSTACK OLED will also present some menu items. The menu items are as follows:

- Assign
- Shift (if additional shift pages have been defined)
- Status (for Rackmount/Desktop units)

The *Select* knob allows for highlighting a menu. While a menu is highlighted, the Assign icon is shown to the right of the menu title. Pressing the *Assign* button enters into that menu. Note this behavior as highlighting of a menu item changes the function of the *Assign* button.

Assign menu shows a listing of stations and defined external sources which can be defined to keys. Scroll through the list with the Select knob and pressing the Assign button to enter into assign mode (indicated by the Assign LED blinking). Pressing any key assigns the selection to the channel and exits assign mode. At the bottom of the list is a Back option which exits the menu.

Shift menu is shown if multiple Shift pages have been defined. Shift pages allow for different collection of key assignments. Entering into Shift menu shows the listing of Shifts, selecting the shift and pressing the Assign button will change the channel assignments. If the

new shift excludes the last 4 calls in memory, the CALL-STACK OLED will return to the top level where the previous calls are listed along with the menus.

Volume Knob

The Volume knob has three different functions; adjust overall volume, adjust per channel volume, and adjust headset volume.

Overall – rotating the volume knob any time other than by the methods defined below.

Per Channel – Holding down the Listen button while adjusting the volume knob will adjust the volume for that source loaded to the key. The setting is saved to the source in volatile memory.

Headset – Pressing the knob in while rotating will adjust the volume to the headset.

Mute Microphone (MUTE MIC)

The *Mute Mic* button will mute the audio inputs. When engaged the LED will be illuminated red. If any other station attempts to listen while the *Mute Mic* is enabled, the audio received will be muted. This help to ensure privacy to the operator of the intercom station. If the *Mute Mic* is engaged and the station attempts to talk to another station, the *Mute Mic* will be disengage.

Mute Speaker (MUTE SPKR)

The *Mute Spkr* button toggles the muting of the internal speaker. When muted, the red LED will illuminate. Mute control is also available through GPIO. With the Element module, the *Mute Spkr* button will mute the audio that is delivered to the External Preview circuit.

Keypad

Select models have a dial keypad for use with future versions of software.

External Sources

Intercom stations discover all other stations within the network. This provides an ease in configuration, but there are plenty of applications where a user needs to interact with a source that is not another intercom station. External source option allows for an intercom station to interact with additional devices. Creating an external source defines an additional option which can be

defined to a key. Using the talk and listen states operates the same for the operator, but behind the panel the actions are different. Configuration of external sources is covered in the next chapter. Case examples are provided at the end of this chapter.

GPIO

General Purpose Input/Output logic is provided in intercom stations to allow for external triggering and signaling. All stations have 8 virtual ports and a single main port. Rackmount stations include an additional port that is a physical GPIO port on the back panel. The 8 virtual ports are used for external source triggering and are defined as you configure an external source. The Main GPIO port has predefined functions as follows:

Main GPIO port

PIN	FUNCTION
GPO Pin1	Mute Speaker command
GPO Pin2	Busy (also mutes speaker)
GPO Pin3	
GPO Pin4	
GPO Pin5	
GPI Pin1	Listening Lamp
GPI Pin2	
GPI Pin3	Ring Lamp
GPI Pin4	Mic Mute Lamp
GPI Pin5	Talking Lamp

The physical port on the back of a rackmount station is intended to be given function by the use of GPIO port routing. GPIO port routing is a feature present in Axia products that links one port to another. By doing a port route, the GPI activity of one port result in GPO activity of the other:

GPI change of state in Port 1-> route (link)-> GPO change of state in Port 2

So by routing the Main GPIO port to a physical port, a “Ring Lamp” indication would be present on GPO pin 3 of the back panel GPIO. Likewise routing the physical port to the Main port, first GPI triggered on the physical

port results in the Main GPO Pin 1 would engage which is the “Mute Speaker” command. Through port routing, the physical port would have the following pin-out on the DA-15 connector:

Physical GPIO port (w/port routing)

PIN	FUNCTION	
Pin1	GPO 1	Listening Lamp
Pin2	GPO 2	
Pin3	GPO 3	Ring Lamp
Pin4	GPO 4	Mic Mute Lamp
Pin5	GPO 5	Talking Lamp
Pin6	not used	
Pin7	GND	GPO common
Pin8	common Out	0v Source (GND)
Pin9	+5 volts	+5v Source
Pin10	common In	GPI common
Pin11	GPI 1	Mute Speaker command
Pin12	GPI 2	Busy (also mutes speaker)
Pin13	GPI 3	
Pin14	GPI 4	
Pin15	GPI 5	

The desktop models do not have a GPIO port on the rear, but with the use of GPIO port routing it is possible to assign these functions to a GPIO port located anywhere within the network. This helps with keeping the countertop cleaner with fewer cables. Configuration of GPIO routing and more is covered in the next chapter.

Trigger Level

Each Intercom station is given a trigger level through configuration which defines its importance. The trigger value defines how station-to-station calling is managed. Basically, higher valued stations can talk to any other stations. Equal valued stations are allowed to talk freely. Lower valued stations can call and an acknowledgment by the receiving station is required prior to a talk session. In cases where an acknowledgment is given, a 5 minute window is opened between the two stations that allows the lower triggered station continuous option to engage a talk session without further acknowledgement from the higher valued station.

When defining external sources which may interact with a station, a trigger value can be defined to the external source which permits the external source to interrupt

or require acknowledgment from the station.

Intercom stations can also be assigned a different value of trigger based on the GPIO logic of “BUSY”. This allows for creating equal trigger among all station unless a station is Busy which does not allow stations to interrupt.

Trigger value can also adjust volume presence in cases where multiple stations call a single station. Higher triggered stations will be given more presence as others are dimmed or muted.

Case Examples

In order to help understand the many ways that the stations can be operated, the following section describes some basic as well as some more imaginative means to operate the Axia Intercom station.

Icom Station to Icom Station

The basic operation of an Axia Intercom stations is fundamental and the use of the user interface so simple that it is almost silly to mention it. By configuring each station with a label name and selecting those names in the key assignment interface, each station is now able to talk and listen to each other. Don’t forget to include a microphone!

Icom Station call to other device

Other intercom device is used in a broad sense and the rules of engagement may vary. The assumption is that GPIO triggering is the key to enabling communication. By defining an External Source with GPIO for Talk and Listen defined, the Axia Intercom station user can press the Talk button and a GPO trigger will be activated, which will enable a PTT button on a radio transceiver, trigger a third party intercom’s listen function, or activate other logic circuits or Pathfinder Stack Events to trigger an event.

Likewise, external GPI trigger can enable an Axia Intercom station’s listen function which allows for the external device to send audio to the station.

Icom Station monitor (Listen) of external source

With the External Source option in each Axia Intercom station, a list of sources can be defined that are not devices one would “talk” to, but instead only wish to “listen” to. This is useful in cases where multiple audio feeds need to be monitored and a user interface is available where each source is listed and pressing of buttons can enable or disable a source to the speaker. With the use of shift pages, multiple listings of 20 sources per listing can be listed for review by a single station.

Icom Station call to codec

In most cases a codec would be receiving a dedicated source or a mix-minus from a mixing console. In applications where a producer monitors the audio from a codec and wishes to insert cues or other bits of valuable information, the Axia Intercom station’s External Source option allows to define an IFB route. This allows the user of the intercom station to press the Talk button and cause a route change where the intercom’s microphone is now feeding the codec. When the Talk button is released, the previous source is routed back to the codec.

Icom Station audio insert to Axia Element source backfeed

Axia Element surfaces create return audio to codecs, phones, and microphone positions. These return audio streams are called backfeeds. Starting with Element version 2.6-series (Powerstation 1.1.4-series), a new source type is created that permits the intercom to inject its microphone audio into the backfeed. This allows for seamless interaction of multiple producers, the control operator, and show hosts during show production.

Icom dedicated listen

Axia Intercom station’s have an option within External Source options that allows for defining an audio source to be enabled in Listen mode by default. This is useful in cases where an operator does not have access to an Axia Element control surface and needs some level of integration between the intercom system and the studio environment. By defining the normal headphone feed as an external source and enabling the Listen On option, the channel assigned to the source will always default on. This permits the use of the intercom to feed the operator’s headphones and that any talk sessions received will be heard.

Chapter Four:

Setup and Configuration

The different intercom stations reside in different environments and thus some configuration will differ based on those environments. For example, Element intercom modules are resident in the Element and DSP engine and take the properties from the host device. Similar, the softcom will take properties from the PC that the software is running. The rackmount and desktop units are independent devices that have a need for some configuration which is not needed for other types of stations. This chapter will provide setup information and in some cases the information will be specific to some stations while not others.

Assigning an IP Address

Rackmount and desktop stations require the assignment of an IP address. This is a simple first step to setting up an intercom station that only needs to happen once per station. Each station has an ID button at the back panel and pressing the button will issue a bootstrap (BOOTP) request and change the front panel interface of some models. The rackmount and desktop stations can be differentiated by those that have OLED displays or filmcap buttons. With OLED models, pressing the ID button will change the front panel display and provide values for each octet of the IP address per OLED screen as seen in figure 4-1.



Figure 4-1: IC.20 Front Panel IP Address Displays BootP Server

- Use the Mic and Spkr buttons above the display to adjust value.
- Press the Mic or Spkr button below the first display to select either IP address or subnet mask.

- Press the Mic or Spkr buttons below the 4th display to save or cancel changes.

For stations with filmcap buttons, pressing the ID button will not change the front panel interface.

All rackmount and desktop models will issue a bootp request when the ID button is pressed. With the use of a simple bootp server application or iProbe, the broadcast message sent by the station will be recorded and provide an option by the application to configure the IP address of the device.

Bootps.exe

Bootps.exe is a simple DOS application that is available from the Axia audio website support and software download section. This may be used to configure the IP address of your intercom stations.

- Run the bootps.exe application
- Press the ID button, you will get a response similar to:
bootps received request from 00:50:C2:80:32:0F
bootps new IP address: [0.0.0.0]
- Enter in the desired IP address for the device and press enter.

If you are having trouble, check firewall settings of the PC you are using. In some cases where there are multiple network cards installed, you may need to define the interface to be used:

```
>bootps.exe -i 192.168.2.89
```

You may also try to force the issue with the `-set` option:

```
>bootps.exe -i 192.168.2.89 -set 00:50:C2:80:32:0F/192.168.2.100
```

For more information on the bootps application, type `bootps.exe -help`.

iProbe

iProbe is an application that helps with the administration of the Axia devices and with it includes support for the bootstrap protocol. Within the application is a button with an image of an investigator. When bootstrap requests are received, a number will appear on the button. Pressing the button will open the bootstrap window. The window will permit the setting of the IP address and restoring of backup configurations.

Station Setup

Axia intercom stations use mDNS in order to avoid the need for a central controller. With the use of mDNS, each intercom station is able to learn of other stations but this requires that each station is configured with a host name and station label. Each station sends Livewire audio as part of the audio communication between stations, so proper setup of the audio is also required prior to successful operation of the intercoms.

Rackmount and Desktops setup

A factory fresh station will have a default host name based on the device's MAC address. The host name is used as part of the mDNS process to learn of IP addressing. Its best practice to change the host name once it has changed at all. This is accessible from the System page.

The main configuration options are located on the Intercom Configuration page.

Intercom Configuration

Intercom Label unique intercom name	<input type="text" value="RackRoom"/>	max 10 characters
Intercom Alternate Label also Livewire source name	<input type="text" value="TechRoom"/>	max 10 characters
Intercom talk Livewire channel	<input type="text" value="Live Stereo"/> - # <input type="text" value="4101"/>	1-32767
Listen mix preview/headphones (optional) Livewire channel	<input type="text" value="Disabled"/> - # <input type="text" value=""/>	1-32767
Call drop flash	<input type="text" value="5"/>	s
Trigger Level	<input type="text" value="4 - Default (Dim lower priority sources; Mute IFB)"/>	
Private Trigger Level On-Air or operator busy	<input type="text" value="4 - Default (Dim lower priority sources; Mute IFB)"/>	
Dim level	<input type="text" value="10"/>	[dB] (0 - no dimming, * - mute)
Tap to latch timeout	<input type="text" value="200"/>	[ms] (0 - Push-To-Talk/no-latch, default: 200)

Audio I/O Configuration

Connector	Enable/Gain [dB]	Phantom Power
Front Microphone	<input type="text" value="54.0"/> (+21 - +77, *-mute)	<input type="text" value="Enabled"/>
Headset	<input type="text" value="21.0"/> (+21 - +77, *-mute)	<input type="text" value="Disabled"/>
Rear Panel Microphone	<input type="text" value="21.0"/> (+21 - +77, *-mute)	<input type="text" value="Disabled"/>
Line In	<input type="text" value="Enabled"/>	

Figure 4-2

The **Intercom label** is used in mDNS and is used by each station to populate its own discovery table. For this reason it is important to set a label name and maintain it. Setting up a system of stations and then changing a station's label could create undesired results. Alternate labels are used to display on station OLEDs by default.

The **Intercom talk** Livewire channel is the audio that the station generates into the network. When the station's audio is to be received by another station, the audio channel defined in this field will be routed. The numerical value can be in the range of 1 – 32767 and must be a number unique per station. To enable the channel, select the Live or Standard Stereo.

Note: In Audio over IP (AoIP) there are different means of packaging the audio into packets. Livewire has traditionally provided two stream types known as Live and Standard. So what is the difference? Live streams are specifically designed to achieve low delay and in doing so carries 12 samples (250 microseconds of audio). This provides fairly low network bandwidth efficiency. Each packet takes 154 bytes on the wire but carries only 72 bytes of audio. The packet overhead puts the

Live streams at about 47% efficiency. Live stream packets are sent at 4000 packets/s rate. Each stream requires 4.9Mbits/s bandwidth.

Standard streams provide higher network bandwidth efficiency. Each packet of a Standard stream carries 240 samples (5 milliseconds of audio). This provides the highest network bandwidth efficiency. Each packet takes 1522 bytes on the wire, which is the maximum allowed Ethernet frame size. Each packet carries 1440 bytes of audio. The network efficiency of Standard Streams is 95%. Standard stream packets are sent at 200 packets/s rate. Each stream requires 2.5Mbits/s bandwidth.

Listen mix preview/headphone is an optional item allowing for the creation of a network audio stream which mimics the speaker of the intercom station for the exception that it does not mute. Volume control of this stream is by pressing the volume knob inward and rotating the knob.

Call drop flash is a value in seconds which is used to define the duration of time that a station's channel will blink to indicate who had just called. For example, if you received audio where the caller asked if you were present and the call ended prior to you looking at the station, the key associated with the caller will blink for 5 seconds (default call drop flash value). This allows you to know who called and you could respond back to them by interfacing with that key and let the caller know you are always working and not to bother you anymore.

Trigger Level is how the hierarchy between stations is defined. The value of 1 is given the least significance where the top of the totem pole is at 8, most significant. The trigger settings define how a station receives the audio of a calling station. The different behaviors are the following:

- A station's key assigned to a higher triggered station will "wink" to acknowledge the difference in hierarchy.
- When a lower triggered station calls (Talk) a higher triggered station, the "wink" is changed to a "blink", indicating the need for a response from the receiving station.

- A lower triggered station calling (Talk) to a high triggered station requires an acknowledgment by the receiving (Listen) station. Once acknowledged, a 5 minute window is open to allow continual communication by the lower triggered station without acknowledgment.
- When higher triggered station talks to a lower triggered station, the same 5 minute window is open for the lower triggered station to respond.
- When two stations are talking to a single station, the lower triggered station will be dimmed or muted depending on the values

- » 8 - Highest (Mute lower priority sources)
- » 7 - High (Mute lower priority sources)
- » 6 - Above Normal - (Dim lower priority; Mute Low Priority and IFB)
- » 5 - Above Normal - (Dim lower priority; Mute Low Priority and IFB)
- » 4 - Default (Dim lower priority sources; Mute IFB)
- » 3 - Low Priority - (Dim lower priority sources; Mute IFB)
- » 2 - Low Priority - (Mute IFB)
- » 1 - IFB

The trigger value of a station can be changed based on a "busy" state being achieved and this is the purpose of the **Private Trigger Level**. A busy condition is signaled by the use of GPIO and is covered in GPIO discussions.

Dim level is a dB value for the amount of dimming applied by the station to any callers that are required to dim because of a trigger setting. The value can be from 0 (defeat dimming) to 100 (mute).

Tap to latch timeout is the time value in ms for the button to be pressed to enable the latch function. The intercom stations have a “tap to latch” function that allows the operator to tap the button and latch the talk or listen state. The feature was mainly introduced to allow for disabling of latching by setting the value to 0. This is a deprecated feature.

At the bottom of the page is the Audio i/o options. Presented are the different audio inputs with gain options, phantom power, or enabling. All inputs are summed together to create a single audio source that is used for the station.

Software setup

The installation of Axia Livewire Softcom to a machine requires also the installation of Apple Bonjour to be installed. The installer package should include an installer of Bonjour for a 32-bit or 64-bit systems.

Once the application is started, select the Preferences item under the Livewire menu. The window will provide the needed options for selection of the network interface connected to the Livewire network, an intercom label (default name takes the host name of the PC), an alternate label, and the Livewire channel source. Once these items are set, the Softcom is ready to participate with other intercom stations.

Axia console module setup

The console module connects to the controller bus within the surface frame. This module is a controller to the intercom station which is resident in the DSP Engine. When the controller module is online, the intercom station located in the DSP Engine will notify other stations through mDNS of its presence. Without the module online, the intercom station will shut down. The intercom station will take the host name from the DSP Engine. To configure this intercom station you must access it through the web interface of the DSP Engine. Select the *Intercom* link from the available links of the DSP Engine.

To define the station for other stations, define the unique **Intercom Label** and provide an **Alternate Label** which is displayed on OLED displays by default.

The screenshot shows the 'Intercom Configuration' web interface. The title bar indicates 'Intercom Configuration' and 'Intercom service version 1.3.5.9'. The form contains the following fields and values:

- Intercom Label** (unique intercom name): Engine21 (max: 10 characters)
- Intercom Alternate Label** (also Livewire source name): Studio2 (max: 10 characters)
- Livewire channel number**: 222 (range: 1-32767)
- Preview Mix channel number**: 220 (range: 1-32767)
- Call drop flash**: 5 s
- Trigger Level**: 4 - Default (Dim lower priority sources; Mute IFB)
- Private Trigger Level** (On-Air or operator busy): 8 - Highest (Mute lower priority sources)
- Dim level**: 80 [dB] (0 - no dimming, * - mute)
- Tap to latch timeout**: 200 [ms] (0 - Push-To-Talk/no-latch, default: 200)

An 'Apply' button is located at the bottom right of the form.

Figure 4-3

Define two unique **Livewire channels**. The first is the audio source of the intercom station. When this station sends audio to other stations, this is the audio channel received. The second channel, Preview mix channel, is used as a console source profile for insertion into External Preview function of the console. When other stations send audio to this station, the received audio is in the preview speaker. When you have the second channel defined, don't forget to generate a Source profile and define this source as the External Preview source in each of the Show profiles configured to the surface.

The **Call drop flash**, **Trigger levels**, **Dim level**, and **Latch timeout** settings are the same as described previously for the rackmount/desktop units.

Channel Setup

With all stations given initial setup properties, the next step is to assign the channels to other stations. For all stations this is similar in nature.

With Rackmount units, going to the *Key Assignment* link will present the page with all keys. Each key has a drop down which is populated with all stations that are discovered through the mDNS process. Selecting a station assigns that key to that station.

With Axia console modules, the key assignment options are presented on the single Intercom link in the DSP Engine. The drop down and other channel settings are the same as for Rackmount/Desktop units.

With Softcom, there is no user interface for config-

uration, only the screen display. There is a single drop down to the right of the window which presents all discovered stations and allows the user to use the *Assign* function to set channels to a station. This is done by selecting a source in the drop down, pressing the Assign button, then select the channel to assign to.

Some stations have two buttons per channel and other stations have a single button. The options available per each channel reflect this button count.



Figure 4-7: Channel options for Filmcap and OLED

The filmcap options define which function is associated with the button press. If both Talk and Listen (Lstn) are checked, pressing the button will engage bidirectional communication. If only the Talk is checked, the button press acts as a talk function and ability to hear the other end would require that the other end engages a talk function.

For OLED channels that have two buttons, the check boxes enable or disable the buttons.

The Latch option, when checked, enables support for the Tap-and-latch feature. Useful when setting up a key as a monitor channel to a source and tapping the key latches the source to the speaker.

the <new> option from the drop down. This will present a blank key assignment page and allow for the configuration of each key. At the bottom of the assignments is a “Save as” text box for naming the shift page. Pressing the Apply button will save the changes made. If the need is to create another shift page with only a few changes, make those changes, change the Save as field and check the Save as copy option. This will prevent the creating of a fresh un configured key assignments. By changing the Save as field and not checking the Save as copy will rename the shift in view.

When more than two shift pages are defined, the CALLSTACK will provide a Shift option that allows the user to view all shifts, select a shift, and assign the shift to the front panel.

External Source setup

Each station has an option to define sources other than intercom stations. External sources can be audio sources that are used for monitoring, other two way audio device that will use GPIO for signaling, or integration with Axia consoles return feeds. Defined external sources will be presented in the drop down list of the key assignment configuration.

Each external source defined will be a row and many options are available within the row, but not all fields are required. Presented below is the function of each option. Case examples on how to use these options is presented in the Case examples at the end of the chapter.

Label	Livewire Channel and Stream Type	Trig Level	GPIO Port	- Talk Pin	- Lstn Pin	Lstn On	T8	IFB Destination	-login	Delete
	From source	4				<input type="checkbox"/>	<input type="checkbox"/>			NEW

Figure 4-8

Select stations have support for “Shift pages”. A Shift page is the complete configuration of the keys. So one shift page may be the keys required to interface with all station associated with Studio A and another shift page can be defined for different grouping of stations associated with Studio B. To create a new shift page, select

Label

The label field is used to give the defined source a name that is used in the Key Assignment drop down listing and is shown in the OLED display to users. Provide a Label value that is suitable to define the source.

Livewire Channel and Stream Type

The channel value defines the source that is to be listened to or in one case defines the source that will be interacted with. The option for Stream Type defines if this is a source or a return feed from an Axia control surface. If these values are not known, the “Browse” button can be used to view active sources on the network and selected from a pop-up window.

Trigger (Trg) Level

The trigger level property is the same as that defined per station. Higher value levels are for greater importance. Trigger level between stations supports acknowledgement function where trigger level with external sources is only for dimming and muting conditions.

GPIO Port

External sources may have GPIO ports associated with their Talk and Listen states for signaling to external device. The options available in the drop down show the virtual GPIO ports available with the station. The virtual port is either routed to a physical port or used with control software (i.e. Pathfinder). More about GPIO is covered in the following section.

Talk / Listen Pin

The pin definition is used in conjunction with the defined GPIO port. The pin number defined the GPI and GPO, state and command, associated with a function. More on GPIO is the in the following section.

Listen ON (Lstn On)

The checkbox defines if the external source should be a default listen state. If enabled, the source would be in listen state upon the source loading to a key.

Talkback (TB)

The TB checkbox defines the external source as an Axia console backfeed enabled source. This requires that the Axia console support the Customizable Backfeed (available in Element version 2.6-series for example). This feature allows the intercom station to tap into the return audio going to a codec or a headphone position (tap into the backfeed associated with a source). Refer to case examples for a better picture of this feature.

IFB Destination

The text field known as the IFB destination is used to define an IP address and port of a device that supports Livewire routing protocol (most Livewire enabled devices). This allows the intercom station to issue a route change when Talk is engaged and send the intercom’s audio to the device. The format of the text is written in the following manner:

```
IP.ADDRESS/PORT
192.168.2.35/3
```

In the above example, an Axia analog xNode (ip address 192.168.2.35) has output 3 feeding into a stereo codec. Engaging the Talk mode on the intercom station will route the intercom’s mic to the output and when the mode is released, the previous source will be re-routed.

-login

The login text field supports the IFB Destination in cases where the device has a user password defined. Entering in the password value allows the intercom station to successfully login to the device and make a route change.

Delete

This checkbox, when selected and pressing the Apply button will remove the defined external source. Any key assigned to the source will still be configured to the invalid source. You will be required, in addition, to edit any key assignments set to the external source which is deleted.

GPIO setup

General Purpose Input and Output (GPIO) has been the means of logic state exchange between devices for years. The Axia Intercoms provide several different GPIO ports to accomplish the application needs. All stations include 8 virtual ports which follow the standard Axia GPIO of 5 inputs and 5 outputs. The virtual ports are used with External Source configuration. All stations will have a main GPIO port which is configured with predefined station logic. Some stations include an additional GPIO port which is the hardware port on the back panel if available.

So the virtual ports have no assigned function unless specified through the external source configuration of

Talk and Listen states.

The main GPIO port has the following defined logic.

Main GPIO Port

PIN	FUNCTION
GPO Pin1	Mute Speaker command
GPO Pin2	Busy (also mutes speaker)
GPO Pin3	
GPO Pin4	
GPO Pin5	
GPI Pin1	Listening Lamp
GPI Pin2	
GPI Pin3	Ring Lamp
GPI Pin4	Mic Mute Lamp
GPI Pin5	Talking Lamp
Pin12	GPI 2
Pin13	GPI 3
Pin14	GPI 4
Pin15	GPI 5

The hardware port has no assigned functions. Its function will depend on configuration. Typically the hardware GPIO port would be mapped to the main GPIO port and this is done through GPIO routing.

GPIO Routing

Axia devices support a function known as GPIO routing. When a port is configured for this, the GPOs (outputs) of the configured port will mimic the behavior of the GPIs (inputs) from the designated port.

Just as we think of audio routing, the output of a device is the input of another device.

GPI pin of Port B goes low -> GPO pin of Port A goes low

To accomplish this, a port should be configured with the IP address and port value of a designated device. Examples of this are:

- 192.168.2.32/3 port 3 on device at IP value 192.168.2.32
- 192.168.2.50:4013/4 port 4 on device at IP 192.168.2.50 on TCP port 4013
- 127.0.0.1/2 same device (loop back address) on port 2

In order to assign a physical port the functions of the station's Main GPIO port, use port routing.

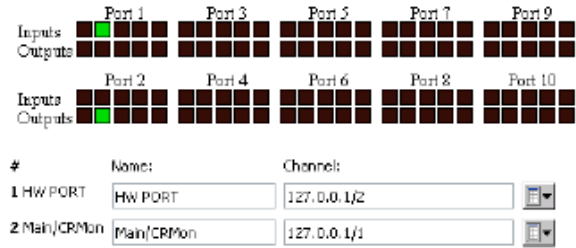


Figure 4-9

As shown in figure 4-9, the hardware port and the main port are mapped to each other, or referred to as GPIO routing. The inputs on the hardware port will be reflected as GPOs in the main port based on the line “127.0.0.1/1”. So the active input on pin 2 results in GPO pin2 being active which triggers the Busy function. In the reverse, when port 1, the hardware port, is configured with “127.0.0.1/2”, the GPI activity of the main port will equate to a GPO (output) on the physical port. Engaging the Mic Mute function will trigger a GPI on pin 4 of the main and a GPO pin 4 on hardware port.

In the case of desktop intercom stations, there is no hardware GPIO port. Using the GPIO port of equipment that is secured in an equipment rack and GPIO routing allows to bring the logic from the station to other equipment without needing to wire direct to the desktop station.

For a situation where an Axia console based intercom station has an external source configured to interface with a PTT (Push-to-talk) radio, the Axia GPIO xNode near the radio would have a port configured as 192.168.2.50:4013/2. Where the DSP engine associated with the surface is IP address 192.168.2.50. Port 4013 is the port where GPIO traffic is supported on DSP Engines (this is different than other devices that use port 93). The Aux 1 GPIO port used for the External source is known as value 2 (since value 1 is the main port).

Mute of Station speakers

A common need is the muting of the speaker associated with the station. This applies to rackmount and desktop models. Axia console intercom stations use the External Preview function, the preview mix is already muted when a local mic is active.

The Main GPIO port has defined two commands which will mute the speaker.

GPO Pin 1 – Mute Speaker

GPO Pin 2 – Busy (also mutes speaker)

Using a hardware GPIO port which has been configured with GPIO routing (previous section) and wiring a relay to GPI pin2 of the port will trigger the Busy function when the relay is active.

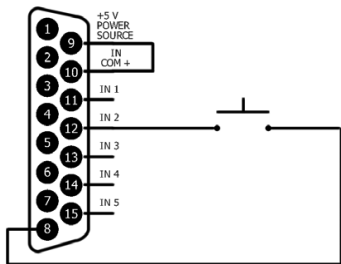


Figure 4-10

If the desire is to only mute the speaker and not trigger the Busy indication, then wiring to pin 1 as opposed to pin 2 would accomplish this.

If using an Axia console, the Control Room (CR) monitor logic has active logic when the monitors are muted. This logic can be tied into the intercom station if the main GPIO port is configured to the same LW Channel logic as defined in the show profile of the Axia console. Figure 4-12 shows the busy signal is active when the monitors are muted.



Figure 4-11

Indicators

Intercom stations provide indicators on the Main GPIO port which can be routed to a hardware GPIO port for triggering of external devices. Pathfinder can be used to monitor the Main GPIO port of these state changes and events can be trigger without the use of hardware GPIO ports.

The indicators, also referred to as lamps, are placed as GPI logic in the Main GPIO port so that GPIO routing will resolve as a GPO on hardware ports. Pathfinder is able to monitor both GPI and GPO state changes.

PIN	FUNCTION
GPI Pin1	Listening Lamp
GPI Pin2	
GPI Pin3	Ring Lamp
GPI Pin4	Mic Mute Lamp
GPI Pin5	Talking Lamp

Listening Lamp (GPI Pin 1) is engaged any time the station is listening to a source. This can either be in cases where the station is monitoring a source or when another station is talking to the station, as this is also a Listen state.

Ring Lamp (GPI Pin 3) is engaged when a lower triggered station calls. When the call is accepted, the ring lamp will release.

Mic Mute Lamp (GPI pin 4) is engaged when the Mic mute button on the front panel is pressed, engaging the Mic mute state. Pressing the button again releases the Lamp and the mute state.

Talk Lamp (GPI pin 5) is engaged when the station is talking to another station or also when another station is listening.

Intercom Station Web Pages

The following section covers the HTML user interfaces available. Some of the information may be a repeat of items mentioned earlier.

Home Page

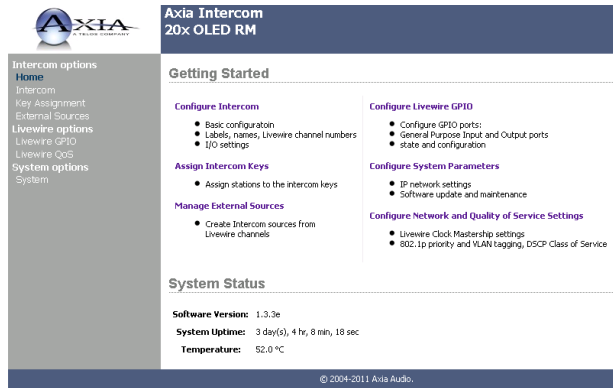


Figure 4-12

The home page provides links to other configuration pages as well as provide software version, uptime, and internal temperature. The left hand side of the page is consistent with other pages as a menu with all the needed links too. When accessing the device for the first time, an authentication window comes up where the default username is user and there is no password. The home page will be presented after entering in the username/ password.

Intercom Configuration Page

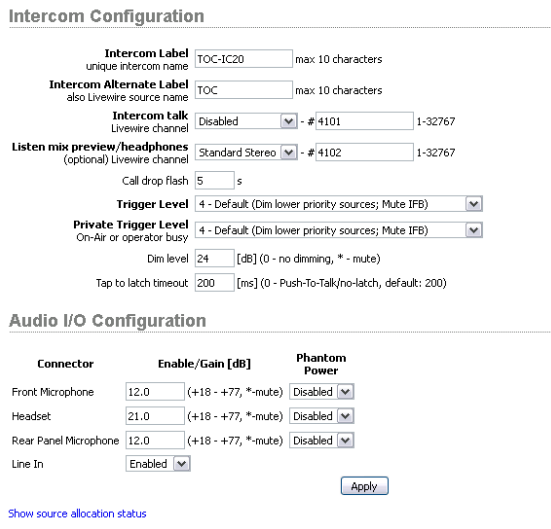


Figure 4-13

The configuration page defines how the station will behave and how it interacts with other stations.

External Sources Page

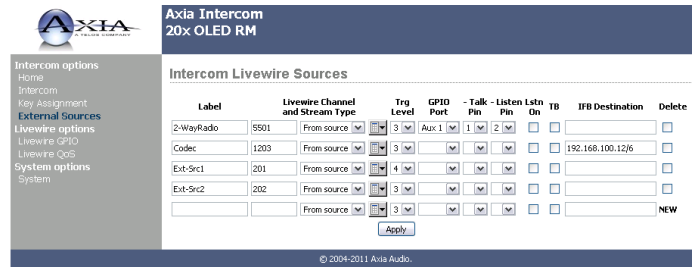


Figure 4-14

External sources can be used in different ways such as Axia console source insert, constant monitor source, 3rd party intercoms, IFB to a source (route change), a source for monitoring, push to talk radios, and likely more.

Key Assignment Page

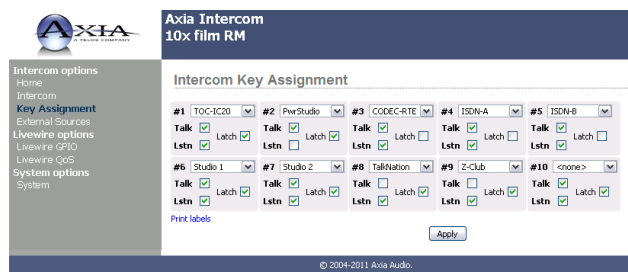


Figure 4-15

Key assignment page is where the each channel is defined with a source and if Talk, Listen, or latch is enabled. The options will vary based on device. An 10 channel intercom station with filmcap buttons is shown. A 20 station OLED would be slightly different.

Livewire GPIO Page

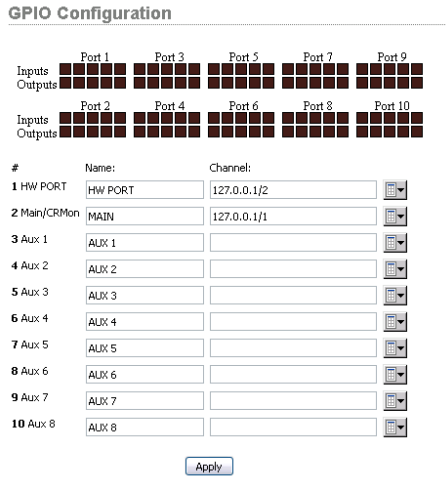


Figure 4-16

Intercom stations will have different amount of ports shown on the GPIO page. The Aux1-8 reflect virtual ports as mentioned in the GPIO section. The Main GPIO port is where the standard Intercom stations logic is located. The Hardware port (if available) represents a physical port and is assigned a function.

DSP Engine based web page

The Intercom stations associated with an Axia console have a single html interface resident in the DSP Engine of that console. The single page is a composition of the items above.

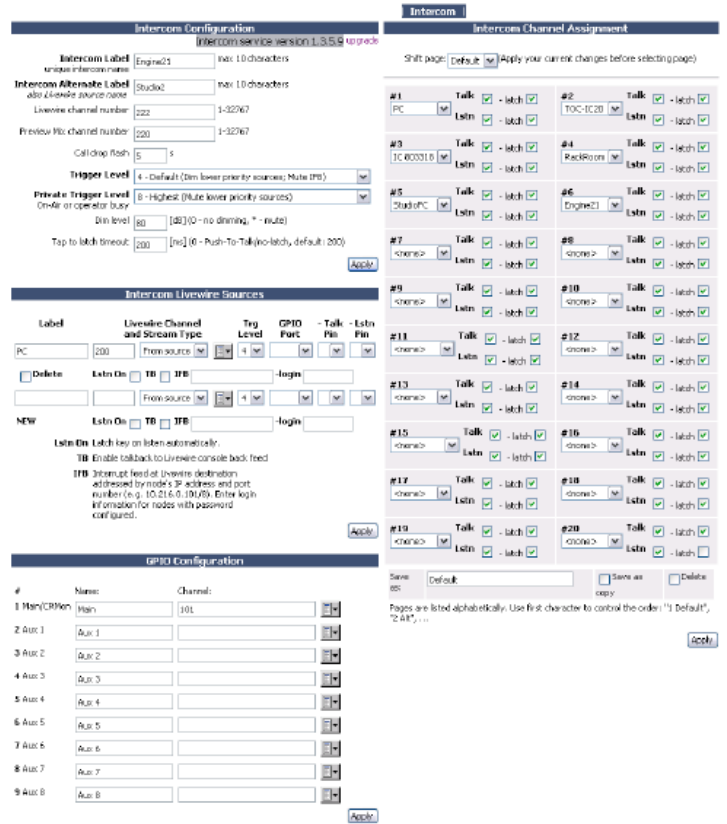


Figure 4-17

Network and Quality of Service Page (rackmount + Desktop)

Network and Quality of service page provides settings for Livewire synchronization. The clock mode should maintain the “Livewire IP low rate” as the other options are legacy options and are deprecated.

The clock master priority defines if a device should have a lower or high priority to being the source of clock. If all devices are set to an equal value they will automatically determine a device as master source. Typically the default setting is suitable.

Live Audio/ Clock Streams section define the priority tagging on Live Stereo audio. These settings should remain unchanged unless your network has been designed differently than suggested by Axia documentation.

Standard Audio streams section is suitable and doesn't need any changes unless there is a specific reason. The buffer size is used for any Standard Stereo streams received. The priority tagging is disabled.

System Configuration Page (rackmount + Desktop)

The system page provides the network options, password, and software control.

Host name is used to give the device a name.

Network address is typically set by other means, but if the need came to change the IP address it could be done so.

Netmask is standard IP setting. Default settings will work for most customers.

Gateway would be defined if the network had a router in place and interactivity was occurring between devices from different networks.

NTP server setting is used only in cases where the device has been configured to synchronize from an NTP source.

Syslog server settings is to define where and what degree of items are sent to a PC running syslog software.

There is no password by default. If there is a need to include a password for accessing the HTML interface, it would be entered into the two text fields. Setting it back to no password would require deleting the values in the two field and pressing Apply.

Firmware version provides two banks where code is stored. The active bank will be selected. To upload a new version, you must download the version from the internet onto a computer. From the computer you can use the browse button to pick the software package, once selected, pressing the Apply button will upload the new code into Bank 1. Once the code is uploaded, you can select Bank 1 and press apply. This will reboot the station into the new version. If running on software that is loaded to Bank 1, you must commit the software to Bank 0 by selecting the available checkbox and pressing apply. This will commit Bank 1 to Bank 0 and Bank 1 would be available for a new upload.

Factory Reset (rackmount + Desktop)

Rackmounts and Desktop models provide an option to reset back to factory default. This is done by removing power from the unit. Locate the ID button on the back panel. Press and hold the button while power is applied to the unit. Keep pressing the button for about 10 seconds. OLED models will provide a reset count down time on the display. Filmcap models will flash the buttons several times and then stop.

Display Tests (rackmount + Desktop)

The intercom stations have a test mode that tests the displays, buttons and provides an option to adjust the brightness. With power applied, press and hold the ID button for 10 seconds.

When in test mode, the four status LEDs (NET, SYNC, Livewire, Master) will scroll in sequence. OLED displays will show TEST. Pressing a button will illuminate the LED associated with the button. Pressing a button on filmcap models will engage the LED for the button. Each additional press will change the color of the light (Green, Red, Yellow, none). Rotating encoders on OLED models will display numerical values on the OLED. For filmcap models, the buttons should be illuminated and rotating the knob will change the brightness of the buttons.

Exit test mode is done so by pressing the ID button once on the rear panel.

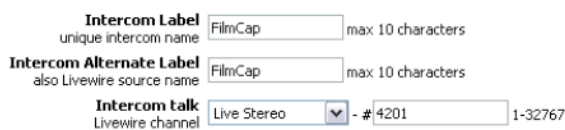
Console modules can be tested by accessing the underside of the module and rotating a 16 position switch to the F position.

Appendix A: Case examples

Icom Station to Icom Station (native Icom config)

Set up an intercom station with a label of FilmCap and a valid unique Livewire channel.

Intercom Configuration




Intercom Label: FilmCap (max 10 characters)
 Intercom Alternate Label: FilmCap (max 10 characters)
 Intercom talk Livewire channel: Live Stereo, #4201, 1-32767

Figure A-1

Setup the second station with the label of OLED and a valid unique Livewire channel.

Intercom Configuration



Intercom Label: OLED (max 10 characters)
 Intercom Alternate Label: OLED20 (max 10 characters)
 Intercom talk Livewire channel: Live Stereo, #4101, 1-32767

Figure A-2

Setup the key assignments on the two stations so that the first key is assigned to the other station.

Intercom Channel Assignment



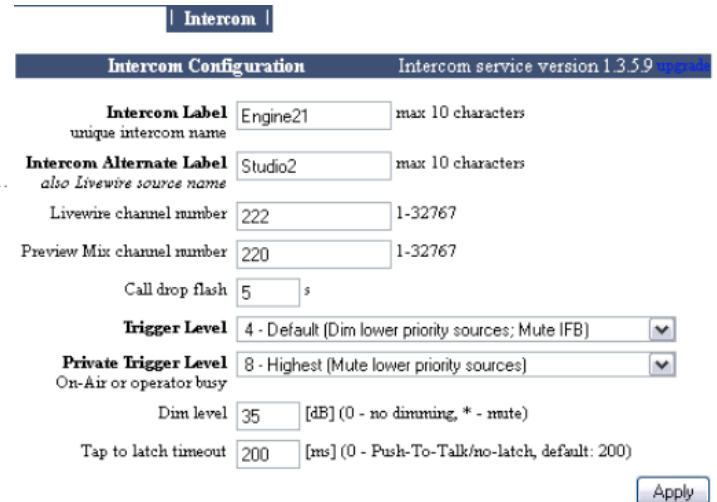
#	Source	Talk	Lstn
#1	<none>	<input type="checkbox"/>	<input type="checkbox"/>
#2	<none>	<input type="checkbox"/>	<input type="checkbox"/>
#3	<none>	<input type="checkbox"/>	<input type="checkbox"/>
#4	<none>	<input type="checkbox"/>	<input type="checkbox"/>
#5	<none>	<input type="checkbox"/>	<input type="checkbox"/>
#6	<none>	<input type="checkbox"/>	<input type="checkbox"/>
#7	<none>	<input type="checkbox"/>	<input type="checkbox"/>
#8	<none>	<input type="checkbox"/>	<input type="checkbox"/>
#9	<none>	<input type="checkbox"/>	<input type="checkbox"/>
#10	<none>	<input type="checkbox"/>	<input type="checkbox"/>

Figure A-3

Pressing the button (or Talk button) on the first key now engages a two way communication between the two stations.

Addition of Axia Console module

Log into the Intercom page of the DSP Engine. Provide a Label and provide two unique Livewire channels.



Intercom Label: Engine21 (max 10 characters)
 Intercom Alternate Label: Studio2 (max 10 characters)
 Livewire channel number: 222 (1-32767)
 Preview Mix channel number: 220 (1-32767)
 Call drop flash: 5 s
 Trigger Level: 4 - Default (Dim lower priority sources; Mute IFB)
 Private Trigger Level: 8 - Highest (Mute lower priority sources)
 Dim level: 35 [dB] (0 - no dimming, * - mute)
 Tap to latch timeout: 200 [ms] (0 - Push-To-Talk/no-latch, default: 200)

Figure A-4

With the Preview Mix channel number defined, go to the Source profile configuration of the console and create a new Source profile. Type in the Preview Mix channel number in the primary source field, name the profile ExtPreview, and deselect all the source availability options. Save the changes. Edit a Show profile, select the “Monitor section” and in the EXTERNAL preview drop down, select the ExtPreview source. Reload the show profile. Return to the DSP Engine intercom page, edit the Intercom Key Assignment section and select the two previous stations to channels. Return to the two previous stations and add the Axia console intercom station you just setup. From the original intercom station, press the channel associated with this third station and the intercom station audio should appear in the Preview speaker. If the Operator source is loaded to a fader on the console, the third station is able to Talk back to the station that just called.

Icom Station to other devices

(External Sources)

In some applications the desire is to not interact with another station, but a device that is not an intercom station or maybe another type of communication device. External Source option provides some options to accomplish these tasks.

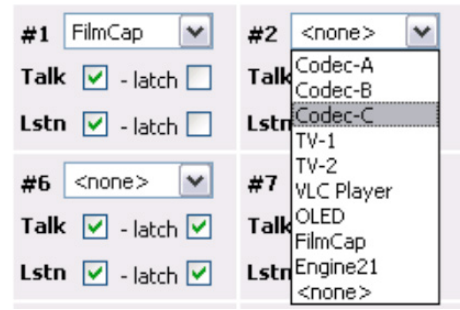


Figure A-6

Icom Station monitor (Listen) of external source

Intercom stations can act in the role of a monitoring station. The station can be configured to communicate with other stations and monitor particular sources. For example a news room would want to be able to talk to various studios, but also monitor the TV audio.

Go to the external sources page of the intercom station. Provide a label descriptive of the source. Enter in the Livewire channel number of the source, for example a source of an analog xNode.

Label	Livewire Channel and Stream Type	Trg Level	GPIO Port	- Talk Pin	- Lstn Pin	Lstn On	TB	IFB Destination
Codec-A	3201 From source	4						
Codec-B	3202 From source	4						
Codec-C	3203 From source	4						
TV-1	3501 From source	4						
TV-2	3502 From source	4						
VLC Player	200 From source	4						
	From source	4						

Figure A-5

If not familiar with creating sources from an Axia xNode, please refer to documentation for that product.

Once the external sources are defined, edit the Key Assignments. The drop down for each key should provide the External sources as options.

Icom Station to PTT Radio

Intercom stations support logic triggering when a channel is in Talk mode and Listen mode. Configure the external source with a Livewire channel, define a GPIO port out of the 8 virtual ports, and define a pin for Talk and a pin for Listen.

Label	Livewire Channel and Stream Type	Trg Level	GPIO Port	- Talk Pin	- Lstn Pin	Lstn On	TB
PTT Radio	5100 From source	4	Aux 1	1	2		

Figure A-7

Setting up an xNode's GPIO port with the intercom station is by configuring a GPIO route between two ports as seen in the following figure.



Figure A-8

The Talk or Listen mode occurring on the Intercom Station will cause the corresponding GPO pin on the xNode to go low. Likewise, if the GPI pin corresponding to either mode is triggered on the xNode, then the Intercom Station will engage into that mode. So in the case where Pin 1 is assigned as Talk, the PTT Radio source assigned to a channel, the channel's Talk button is pressed, the xNode's GPIO port pin 1 will go low.

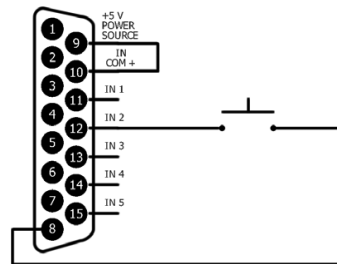


Figure A-11

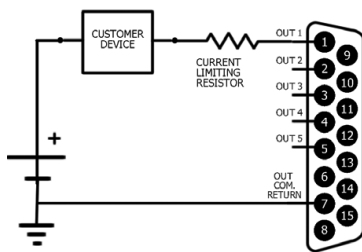


Figure A-9

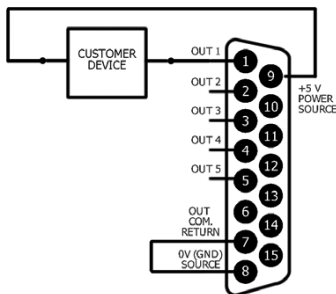


Figure A-10

Wiring up the GPIO node to engage the radio and having audio routed to the radio will allow the Axia Intercom station to key the radio and talk.

If the radio provides a carrier operated switch or similar, this can be wired to the xNode's GPIO port to trigger the Listen mode of the Intercom Station.

GPI on pin 12 (known as the second pin of 5 GPIs) would trigger a Listen mode if the Listen pin was set to 2.

Icom Station call to codec (IFB routing)

In some cases you may have multiple Intercom Stations that need the ability to talk to a single external source. The IFB Destination option assists with this issue as it allows an Intercom Station to make a route change to a destination when a Talk session is engaged. The codec needs to feed cues from a producer, the producer has an Intercom Station with a channel assigned to the external source of the codec.

The IFB Destination field is filled in with the IP address of the Livewire enabled codec or a Livewire I/O device. After the IP address should be the port number, for example if an analog xNode is feeding the stereo codec through the 4th port, the field will be filled as 192.168.100.98/4. In the case of a Zip/ONE, there is a single stereo Livewire port so the field is 192.168.100.101/1.

This will create a route change to the destination when the Talk mode is engaged. Note, this is a stereo (2 channel) route change.

Icom Station call to Codec/Headphone (console backfeed insert)

Intercom Stations have an option to tap into backfeeds that are created by some Axia consoles. With Element version 2.6-series, a new source type known as a Customizable Backfeed supports a Talk insertion option. The environment is the Element is used in a control room to a talk studio. Behind the console operator are producers who need to give cues to the hosts that are in the talk studio. The producers are each provided an Intercom Station.

Define an external source for each host position and select the TB check box. The channel number is the same number used to define the source within the Element.

Within the Element's *Customizable Backfeed* profile, define the host positions and enable the "Feed to Source" and enable the "Talk Insertion". When the source is loaded to a console fader, pressing the Talk button on the Intercom Station's channel assigned to the External source, the audio from the Intercom will be inserted into return feed for the host headphones.

Icom dedicated monitor

The Intercom Station can be used to drive the audio delivered to headphones. As an example, having the station feed audio to the operator behind an Axia iQ console. By setting up the CR Headphone source as an External source that is normally in Listen state, it allows for the operator to be more integrated with the intercom system. Allowing for other stations to call the operator and have the audio inserted into the headphone feed. To do this, setup an External Source of the CR Headphone audio. Select the Listen On (Lstn On) check box. Assign the External Source to a channel. This enables the listen state of the channel to be turned on automatically.

Appendix B: Troubleshooting

Syslog

Each intercom station should support a syslog function which allows for the defining of a syslog IP address. If defined, the station will report activity to the syslog within the network and provide information of what has occurred if users experience some trouble. With rackmount and desktop units, the syslog definition is located on the system page. With intercoms that are a part of a console, the syslog definition is on system page of the DSP Studio Engine. There are many syslog applications available to the public through the internet. If you are having trouble locating one, contact our support for guidance.

In addition to syslog activity, the rackmount and desktop units have hidden web pages with other technical details as follows:

Intercom stations discovery table

<http://IP.ADDRESS/pub/icd.txt>

Intercom stations static external sources table

<http://IP.ADDRESS/pub/ics.txt>

Livewire sources (from the advertisement)

<http://IP.ADDRESS/pub/lwsrscs.txt>

Most recent Syslog messages

<http://IP.ADDRESS/pub/log.txt>

Support

If you are experiencing troubles, make sure you collect as much data on the problem and to be precise with when the problem occurred. Just providing syslog data to support will not help point to the problem but are merely clues of what was happening over time. As with any murder mystery game, the more clues gathered make the attempt at finding the culprit (the reason for the problem) an easier task.

Warranty

Telos Alliance Limited Warranty



Telos Alliance Radio Division has an industry-leading five-year warranty. As a company, we are committed to our products and the pursuit of providing better quality to our customers.

Find the Warranty online at www.telosalliance.com/warranty.



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